Avaya Aura® Call Center Elite Feature Reference

Release 10.2.x
Issue 1
December 2023

© 2019-2023, Avaya LLC
All Rights Reserved.

Notice

While reasonable efforts have been made to ensure that the information in this document is complete and accurate at the time of printing, Avaya assumes no liability for any errors. Avaya reserves the right to make changes and corrections to the information in this document without the obligation to notify any person or organization of such changes.

Documentation disclaimer

“Documentation” means information published in varying media which may include product information, subscription or service descriptions, operating instructions and performance specifications that are generally made available to users of products. Documentation does not include marketing materials. Avaya shall not be responsible for any modifications, additions, or deletions to the original published version of Documentation unless such modifications, additions, or deletions were performed by or on the express behalf of Avaya. End user agrees to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation, to the extent made by End user.

Link disclaimer

Avaya is not responsible for the contents or reliability of any linked websites referenced within this site or Documentation provided by Avaya. Avaya is not responsible for the accuracy of any information, statement or content provided on these sites and does not necessarily endorse the products, services, or information described or offered within them. Avaya does not guarantee that these links will work all the time and has no control over the availability of the linked pages.

Warranty

Avaya provides a limited warranty on Avaya hardware and software. Please refer to your agreement with Avaya to establish the terms of the limited warranty. In addition, Avaya’s standard warranty language as well as information regarding support for this product while under warranty is available to Avaya customers and other parties through the Avaya Support website: https://support.avaya.com/helpcenter/getGenericDetails?detailId=C20091120112456651010 under the link “Warranty & Product Lifecycle” or such successor site as designated by Avaya. Please note that if the product(s) was purchased from an authorized Avaya channel partner outside of the United States and Canada, the warranty is provided by said Avaya Channel Partner and not by Avaya.

“**Hosted Service**” means an Avaya hosted service subscription that You acquire from either Avaya or an authorized Avaya Channel Partner (as applicable) and which is described further in Hosted SAS or other service description documentation regarding the applicable hosted service. If You purchase a Hosted Service subscription, the foregoing limited warranty may not apply but You may be entitled to support services in connection with the Hosted Service as described further in your service description documents for the applicable Hosted Service. Contact Avaya or Avaya Channel Partner (as applicable) for more information.

Hosted Service

THE FOLLOWING APPLIES ONLY IF YOU PURCHASE AN AVAYA HOSTED SERVICE SUBSCRIPTION FROM AVAYA OR AN AVAYA CHANNEL PARTNER (AS APPLICABLE), THE TERMS OF USE FOR HOSTED SERVICES ARE AVAILABLE ON THE AVAYA WEBSITE, HTTPS://SUPPORT.AVAYA.COM/LICENSEINFO UNDER THE LINK “Avaya Terms of Use for Hosted Services” OR SUCH SUCCESSOR SITE AS DESIGNATED BY AVAYA, AND ARE APPLICABLE TO ANYONE WHO ACCESSES OR USES THE HOSTED SERVICE. BY ACCESSING OR USING THE HOSTED SERVICE, OR AUTHORIZING OTHERS TO DO SO, YOU, ON BEHALF OF YOURSELF AND THE ENTITY FOR WHOM YOU ARE DOING SO (HEREINAFTER REFERRED TO INTERCHANGEABLY AS “YOU” AND “END USER”), AGREE TO THE TERMS OF USE. IF YOU ARE ACCEPTING THE TERMS OF USE ON BEHALF A COMPANY OR OTHER LEGAL ENTITY, YOU REPRESENT THAT YOU HAVE THE AUTHORITY TO BIND SUCH ENTITY TO THESE TERMS OF USE. IF YOU DO NOT HAVE SUCH AUTHORITY, OR IF YOU DO NOT WISH TO ACCEPT THESE TERMS OF USE, YOU MUST NOT ACCESS OR USE THE HOSTED SERVICE OR AUTHORIZE ANYONE TO ACCESS OR USE THE HOSTED SERVICE.

Licenses

The Global Software License Terms (“Software License Terms”) are available on the following website https://www.avaya.com/en/legal-license-terms/ or any successor site as designated by Avaya. These Software License Terms are applicable to anyone who installs, downloads, and/or uses Software and/or Documentation. By installing, downloading or using the Software, or authorizing others to do so, the end user agrees that the Software License Terms create a binding contract between them and Avaya. In case the end user is accepting these Software License Terms on behalf of a company or other legal entity, the end user represents that it has the authority to bind such entity to these Software License Terms.

Copyright

Except where expressly stated otherwise, no use should be made of materials on this site, the Documentation, Software, Hosted Service, or hardware provided by Avaya. All content on this site, the documentation, Hosted Service, and the product provided by Avaya including the selection, arrangement and design of the content is owned either by Avaya or its licensors and is protected by copyright and other intellectual property laws including the sui generis rights relating to the protection of databases. You may not modify, copy, reproduce, republish, upload, post, transmit or distribute in any way any content, in whole or in part, including any code and software unless expressly authorized by Avaya. Unauthorized reproduction, transmission, dissemination, storage, or use without the express written consent of Avaya can be a criminal, as well as a civil offense under the applicable law.

Virtualization

The following applies if the product is deployed on a virtual machine. Each product has its own ordering code and license types. Unless otherwise stated, each Instance of a product must be separately licensed and ordered. For example, if the end user customer or Avaya Channel Partner would like to install two Instances of the same type of products, then two products of that type must be ordered.

Third Party Components

The following applies only if the H.264 (AVC) codec is distributed with the product. THIS PRODUCT IS LICENSED UNDER THE AVC PATENT PORTFOLIO LICENSE FOR THE PERSONAL USE OF A CONSUMER OR OTHER USES IN WHICH IT DOES NOT RECEIVE REMUNERATION TO (i) ENCODE VIDEO IN COMPLIANCE WITH THE AVC STANDARD (“AVC VIDEO”) AND/OR (ii) DECODE AVC VIDEO THAT WAS ENCODED BY A CONSUMER ENGAGED IN A PERSONAL ACTIVITY AND/OR WAS OBTAINED FROM A VIDEO PROVIDER LICENSED TO PROVIDE AVC VIDEO. NO LICENSE IS GRANTED OR SHALL BE IMPLIED FOR ANY OTHER USE. ADDITIONAL INFORMATION MAY BE OBTAINED FROM MPEG LA, L.L.C. SEE HTTP://WWW.MPEGLA.COM.

Service Provider

WITH RESPECT TO CODECS, IF THE AVAYA CHANNEL PARTNER IS HOSTING ANY PRODUCTS THAT USE OR EMBED THE H.264 CODEC OR H.265 CODEC, THE AVAYA CHANNEL PARTNER ACKNOWLEDGES AND AGREES THE AVAYA CHANNEL PARTNER IS RESPONSIBLE FOR ANY AND ALL RELATED FEES AND/OR ROYALTIES. THE H.264 (AVC) CODEC IS LICENSED UNDER THE AVC PATENT PORTFOLIO LICENSE FOR THE PERSONAL USE OF A CONSUMER OR OTHER USES IN WHICH IT DOES NOT RECEIVE REMUNERATION TO: (i) ENCODE VIDEO IN COMPLIANCE WITH THE AVC STANDARD (“AVC VIDEO”) AND/OR (ii) DECODE AVC VIDEO THAT WAS ENCODED BY A CONSUMER ENGAGED IN A PERSONAL ACTIVITY AND/OR WAS OBTAINED FROM A VIDEO PROVIDER LICENSED TO PROVIDE AVC VIDEO. NO LICENSE IS GRANTED OR SHALL BE IMPLIED FOR ANY OTHER USE. ADDITIONAL INFORMATION FOR H.264 (AVC) AND H.265 (HEVC) CODECS MAY BE OBTAINED FROM MPEG LA, L.L.C. SEE HTTP://WWW.MPEGLA.COM.

Compliance with Laws

You acknowledge and agree that it is Your responsibility to comply with any applicable laws and regulations, including, but not limited to laws and regulations related to call recording, data privacy, intellectual property, trade secret, fraud, and music performance rights, in the country or territory where the Avaya product is used.

Preventing Toll Fraud

“Toll Fraud” is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there can be a risk of Toll Fraud associated with your system and that, if Toll Fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya Toll Fraud intervention

If You suspect that You are being victimized by Toll Fraud and You need technical assistance or support, please contact your Avaya Sales Representative.

Security Vulnerabilities

Information about Avaya’s security support policies can be found in the Security Policies and Support section of https://support.avaya.com/security.

Suspected Avaya product security vulnerabilities are handled per the Avaya Product Security Support Flow (https://support.avaya.com/css/P8/documents/100161515).

Downloading Documentation

For the most current versions of Documentation, see the Avaya Support website: https://support.avaya.com, or such successor site as designated by Avaya.

Contact Avaya Support

See the Avaya Support website: https://support.avaya.com for Product or Cloud Service notices and articles, or to report a problem with your Avaya Product or Cloud Service. For a list of support telephone numbers and contact addresses, go to the Avaya Support website: https://support.avaya.com (or such successor site as designated by Avaya), scroll to the bottom of the page, and select Contact Avaya Support.

Trademarks

The trademarks, logos and service marks (“Marks”) displayed in this site, the Documentation, Hosted Service(s), and product(s) provided by Avaya are the registered or unregistered Marks of Avaya, its affiliates, its licensors, its suppliers, or other third parties. Users are not permitted to use such Marks without prior written consent from Avaya or such third party which may own the Mark. Nothing contained in this site, the Documentation, Hosted Service(s) and product(s) should be construed as granting, by implication, estoppel, or otherwise, any license or right in and to the Marks without the express written permission of Avaya or the applicable third party.

Avaya is a registered trademark of Avaya LLC.

All non-Avaya trademarks are the property of their respective owners.

Avaya, the Avaya logo, Avaya one-X® Portal, Communication Manager, Application Enablement Services, Modular Messaging, and Conferencing are either registered trademarks or trademarks of Avaya Inc. in the United States of America and/or other jurisdictions.

All non-Avaya trademarks are the property of their respective owners.

Contents

Chapter 1: Introduction 13
Purpose 13
Chapter 2: ACD fundamentals 14
Communication servers 14
Trunks, trunk groups, and extensions 14
Communication server attendant 14
ACD 15
Automatic-in processing 15
DID processing 16
DID processing example 16
Split queue call processing 17
Priority and normal split queues 17
Dynamic queue slot allocation 17
Announcements for calls in a split queue 18
Answer supervision 19
Abandoned calls 19
Intraflow and interflow 20
How to administer ACD splits 20
Queue threshold assignment 21
Intraflow assignment type 21
How to administer Intraflow or Interflow 21
Night Service 22
Hunt Group Night Service 22
Trunk Group Night Service 22
System Night Service 22
Call distribution methods 23
Call distribution methods without EAS 23
Call distribution methods with EAS 25
How do agents handle calls 26
Chapter 3: ACD features 35
Abandoned Call Search 35
Abandoned Call Search considerations 35
ACD field options for agents 36
ACD field setting considerations 36
Add/Remove Skills 36
Add/Remove Skills considerations 37
Add/Remove Skills interactions 37
Agent/Caller Disconnect Tones 38
Agent/Caller Disconnect Tones scenarios 38

Country-specific disconnect tone definition ... 39
Agent/Caller Disconnect Tones interactions ... 40
Agent Call Handling ... 40
Agent work modes ... 41
Agent answering options ... 45
Agent capacity and related limits ... 45
Agent login and logout ... 47
Agent request for supervisor assistance with or without an active ACD call ... 47
Automatic Answer ... 48
Block hang-up by logged-in auto-answer agents ... 48
Caller information ... 49
Call Work Codes ... 52
Login to a skill ... 53
Logout ... 53
Manual Answer ... 54
Option to not apply ringback to callers ... 54
Ringer-off control of auto-answer for non-ACD calls ... 54
Stroke Counts ... 55
Agent Call Handling considerations ... 55
Agent Call Handling interactions ... 56
Agent Mobility integrates with Avaya Extension to Cellular ... 58
Operational differences between working on a deskphone and a mobile station ... 59
Features not currently supported with Agent Mobility ... 59
Auto-Available Split/Skill ... 61
Agent login with Auto-Available Split/Skill ... 61
Agent logout with Auto-Available Split/Skill ... 61
Auto-Available Split/Skill considerations ... 61
Auto-Available Split/Skill interactions ... 62
Automatic Call Distribution ... 62
ACD methods ... 64
Agent Availability Query dial tones ... 68
Basic Call Management System ... 68
BCMS interactions ... 69
Business Advocate ... 70
Call Prompting ... 70
Call Prompting command set ... 71
Touch-tone collection requirements ... 71
Call Prompting considerations ... 72
Call Prompting digit entry-collect digits command ... 73
Functions and examples ... 75
Dial-ahead digits-collect digits command ... 81
ASAI-requested digit collection ... 84
ASAI-provided dial-ahead digits - collect digits command ... 84

Call Prompting interactions 85
Expert Agent Selection 85
EAS terminology 86
EAS administration 87
EAS benefits 88
EAS-PHD - 120 skills/16 skill levels 89
Call Handling Preference 90
Direct Agent Calling 90
How to identify caller needs 96
Additional agent login ID capabilities 98
EAS functions and examples 100
EAS considerations 113
EAS interactions 114
Forced Agent Logout by Clock Time 123
Forced Logout Override 124
Forced Agent Logout from Clock Time example 124
Forced Agent Logout from Clock Time interactions 125
Forced Agent Logout/AUX Work by Location/Skill 127
Logging out all the agents in a location 128
Remotely forcing all agents in a location/skill to logout/AUX work mode 129
Remotely forcing all agents in a location/skill to logout/AUX work mode in one step 130
Forced Agent Logout from ACW 131
Forced Agent Logout from ACW interactions 131
Inbound Call Management 132
ICM applications 132
ICM considerations 136
ICM interactions 136
Information Forwarding 136
Information Forwarding benefits 137
Data handled by Information Forwarding 138
ASAI shared UUI data conversion 138
Information Forwarding detailed description 139
ISDN network requirements 143
Information Forwarding example 143
SIP network requirements 144
Information Forwarding considerations 144
Information Forwarding interactions 145
Information forwarding support for BSR and LAI 145
Advanced Information Forwarding for ISDN and SIP trunks 148
Interruptible Aux 153
Interruptible Aux thresholds 154
Interruptible Aux considerations 156
Intraflow and Interflow 157

How does Intraflow work ... 157
Interflow and Intraflow considerations ... 158
Interflow and Intraflow interactions ... 158
Location Preference Distribution ... 159
Location numbers ... 159
Call handling conditions ... 161
Local Preference Distribution interactions ... 162
Match Agent and Skill Partitions ... 164
Match Agent and Skill Partitions interactions ... 165
Maximum Agent Occupancy ... 166
MAO for manual-in agents ... 166
MAO for auto-in agents ... 167
Manual override of MAO AUX work mode ... 167
CMS database items for evaluating MAO ... 167
MAO interactions ... 167
Multiple Call Handling ... 168
MCH applications ... 168
MCH Settings ... 169
MCH considerations ... 171
MCH interactions ... 172
Proactive Contact outbound calling improved reporting ... 173
Prerequisites for using Proactive Contact outbound calling improved reporting ... 173
Switch-classified and non switch-classified calls ... 174
The Proactive Contact Outbound Calling Process ... 174
Proactive Contact improved reporting interactions ... 177
Queue Status Indications ... 177
Types of Queue Status Indications ... 177
Queue Status Indication interactions ... 178
Reason Codes ... 178
Types of reason codes ... 179
Reason code considerations ... 180
Reason code interactions ... 181
Redirection on IP Failure ... 182
How ROIF works ... 182
ROIF considerations ... 183
ROIF interactions ... 184
Redirection on No Answer ... 185
RONA detailed description ... 186
RONA application examples ... 186
RONA considerations ... 190
RONA interactions ... 191
Interactions with other ringing call timers ... 197
Redirection on OPTIM Failure ... 198

ROOF considerations ... 199
ROOF interactions ... 199
Remote Logout of Agent ... 200
Locally logging out an agent ... 200
How to administer Remote Logout of Agent using a VDN ... 200
Remotely logging out an agent using the assigned VDN ... 201
Remote Logout of Agent interactions ... 202
Reporting Adjuncts ... 202
Measured trunks and unmeasured facilities ... 203
Reporting adjuncts considerations ... 203
Service Level Maximizer ... 207
MAO ... 207
SLM requirements ... 207
SLM operations ... 207
SLM algorithms ... 212
SLM reporting ... 213
SLM interactions ... 215
Service Observing ... 216
No Talk FAC for Service Observing ... 217
Observing agent login IDs ... 217
Observing remotely by FAC ... 218
Observing VDNs ... 219
Service Observing with Exclusion ... 219
ASAI Single Step Conference calls with Exclusion ... 220
VDN Observing by Location ... 220
Service Observing indicators ... 222
Service Observing considerations ... 227
Service Observing interactions ... 230
Service Observing interactions for SIP phones ... 232
Service Observing with Multiple Observers ... 233
Service Observing with Multiple Observers example ... 233
Service Observing with Multiple Observers interactions ... 234
Universal Call ID ... 234
How does UCID work? ... 235
What creates UCIDs? ... 235
When are UCIDs created? ... 235
How are UCIDs transmitted? ... 235
UCID application requirements ... 236
UCID tracking scenarios ... 236
UCID considerations ... 242
UCID interactions ... 242
VDN of Origin Announcement ... 242
VOA considerations ... 244

VOA interactions ... 245
VDN Time Zone Offset ... 247
Reason to use VDN Time Zone Offset ... 247
Operation ... 247
Example of VDN Time Zone Offset ... 247
VDN Time Zone Offset interactions ... 248
VuStats ... 248
VuStats forms and fields ... 248
VuStats considerations ... 251
VuStats interactions ... 252
Zip Tone Burst for Callmaster® Endpoints ... 254
Chapter 4: Call Vectoring features ... 256
Adjunct Routing ... 256
Adjunct Routing considerations ... 257
Receiving and implementing an ASAI call route ... 258
Data sent with an ASAI call route request ... 259
Special vector processing considerations associated with adjunct routing ... 261
Advanced Vector Routing features ... 268
Advanced Vector Routing command set ... 268
When to use wait time predictions ... 269
Expected Wait Time (EWT) ... 270
Factors that increase EWT for a split priority level ... 274
Factors that decrease EWT for a split priority level ... 275
Rolling Average Speed of Answer (ASA) ... 275
VDN Calls ... 277
Attendant Vectoring ... 278
Attendant queue ... 279
Attendant VDNs ... 279
Console Parameters screen example ... 282
Hunt group queue for Attendant Vectoring ... 283
Night service ... 283
Redirecting calls to attendant VDNs ... 283
Restrictions for attendant and non attendant vectoring ... 284
TN assignments ... 284
Vector screen example ... 285
Best Service Routing ... 285
Benefits of BSR ... 286
BSR terminology ... 287
BSR requirements ... 289
BSR considerations ... 291
How BSR determines the best resource ... 291
Tips for writing BSR vectors ... 293
Alternate Selection on BSR Ties ... 294

Singlesite BSR ... 294
Multisite BSR ... 302
BSR local treatment for calls queued remotely ... 328
BSR interactions ... 334
Holiday Vectoring ... 337
Administering Holiday Vectoring ... 338
Holiday Vectoring considerations ... 338
Holiday Vectoring command set ... 339
Holiday table command syntax ... 340
Using the holiday table commands ... 341
Changing vector processing for holidays ... 342
Look-Ahead Interflow ... 344
LAI requirements ... 344
LAI considerations ... 345
Two switch configuration ... 349
LAI command set ... 349
How traditional LAI works ... 350
Enhanced LAI ... 353
LAI-initiated path replacement for calls in vector processing ... 359
DNIS and VDN Override in an LAI environment ... 360
LAI with network ADR ... 361
Multisite applications for Enhanced LAI ... 362
Meet-me Conference ... 363
Meet-me Conference command set ... 363
Activating Meet-me Conference ... 365
Creating a Meet-me Conference VDN ... 366
Creating a Meet-me Conference vector ... 367
Meet-me Conference capacity issues ... 368
Meet-me Conference call processing scenario ... 368
Meet-me Conference interactions ... 369
Network Call Redirection ... 371
NCR options supported by PSTNs ... 371
Protocols not supported by NCR ... 371
NCT-type feature operations ... 371
AT&T In-Band Transfer Connect service ... 376
NCR implementation methods ... 379
NCR considerations ... 385
NCR interactions ... 387
NCR support with SIP ... 388
SIP Network Call Redirection protocols ... 389
SIP limitations on call redirection ... 389
SIP NCR and Information Forwarding ... 389
NCR operations for SIP by Call Vectoring method ... 390

BSR queue-to best vector step activation of NCR 390
SIP NCR interactions 391
Percentage Allocation Routing 392
Considerations for implementing percentage allocation routing 392
Rules for percentage allocation routing 394
Percentage allocation routing example 394
Service Hours Table Routing 395
Service Hours Table Routing considerations 395
Goto processing for Service Hours Table Routing 395
Service Hours Table 395
Goto step command for Service Hours Table Routing 396
Goto vector command for Service Hours Table Routing 396
Service Hours Table Routing scenario 396
Variables in Vectors 398
VDN In a Coverage Path 399
VICP 399
VICP considerations 399
VICP interactions 399
VDN Override 403
VDN parameters associated with the active VDN 404
Application of VDN Override 404
VDN Override for ASAI messages 405
VDN Override interactions 406
VDN Return Destination 406
VDN variables 408
Reasons to use VDN variables 408
Voice Response Integration 409
VRI 409
VRI detailed description 409
VRI interactions 411
Call Vectoring considerations 416
Displaying VDN names for vector-initiated DACs 416
Call transfer to VDNs 419
Call Vectoring interactions 420
Call Vectoring and BCMS/CMS interactions 425
Chapter 5: Avaya 96X1 SIP agent deskphones 436
96X1 SIP agent deskphone feature support 436
96X1 SIP value-added capabilities 437
Chapter 6: Security issues 439
Remote access 439
Front-ending remote access 439
Replacing remote access 440
EAS 440

Limiting outside access using VDN COR restrictions..... 440
Meet-me Conference security issues..... 441
Vector-initiated service observing..... 441
Voice Response Integration (VRI)..... 442
Attendant Vectoring security issues..... 442
Chapter 7: Resources..... 443
Documentation..... 443
Finding documents on the Avaya Support website..... 444
Accessing the port matrix document..... 444
Avaya Documentation Center navigation..... 445
Training..... 446
Viewing Avaya Mentor videos..... 446
Support..... 447
Using the Avaya InSite Knowledge Base..... 447
Glossary..... 449

Chapter 1: Introduction

Purpose

This document contains Avaya Aura® Call Center Elite high-level feature descriptions for Automatic Call Distribution and Call Vectoring and provides details about feature characteristics, capabilities, capacities and interactions.

This document is intended for people who want a high-level understanding of Avaya Aura® Call Center Elite features and capabilities.

Chapter 2: ACD fundamentals

Communication servers

A communication server is an electronic device that processes and connects incoming, outgoing, and internal calls to the destination.

A Central Office (CO) is the communication server of a phone company in your local area.

Trunks, trunk groups, and extensions

Incoming calls to a company first pass through the CO. The CO sends calls to the communication server of the company. Trunks are phone lines that carry calls between two servers, between a CO and a server, or between a CO and a phone.

The CO receives the dialed digits from the caller, processes the digits, and seizes a trunk that is assigned those digits. From the seized trunk, the CO sends a continuing transmission to the destination phone or communication server. No other calls can be sent over that trunk until the current call disconnects.

As a trunk can carry only one call at a time, trunk groups are usually created and are assigned to the same digits. With a trunk group, the CO receives the digits of a dialed phone number and checks if any trunk assigned to that number is available. The CO then seizes an available trunk. You can make as many calls simultaneously over a trunk group with the available trunks. A trunk group, therefore, can carry multiple calls for the same phone number. When a trunk group carries incoming calls, that is, calls made outside the company communication server location to the communication server, the communication server connects the calls to their proper destinations within the company.

In addition to connecting incoming calls to the proper destinations, communication servers act like private COs for company employees. Employee phones connect to a communication server by phone lines called extensions. You can assign numbers to extensions and the numbers become the employee phone numbers for internal calls.

Communication server attendant

Incoming calls can also go to a communication server attendant who manually routes calls to their proper destinations using an attendant console. Normally an attendant serves as an internal operator who transfers calls to the proper extensions. A communication server often has more than one attendant, and all of the communication server's attendants answer calls directed to the attendant queue, which holds calls until an attendant is available. The attendant queue receives

internal calls from employee extensions, through Direct Inward Dialing (DID) processing and Automatic-in processing. If the attendant has an automatic-in number, it normally is the number published in a phone book, and the DID number is used by off-site employees who know only the extension number of the attendant.

With Centralized Attendant Service (CAS), you can consolidate attendants at one private network location. The attended location is called the CAS main and each unattended location is called a CAS branch. At branch locations, calls requiring attendant services route by way of Release Link Trunks to the main location.

ACD

Automatic Call Distribution (ACD) processes and distributes high-volume incoming, outgoing, and internal calls to groups of extensions called hunt groups or splits. Avaya Aura® Communication Manager sends information about ACD operations to Call Management System (CMS) or Avaya IQ, which stores and formats data and produces real-time, historical, and exception reports.

With ACD, system administrators can create an efficient call management environment by:

- Adding or removing splits or skills, announcements, and agents.
- Adding trunk groups.
- Routing calls to splits or skills.
- Assigning a split supervisor to each ACD split or skill.
- Determining the ACD measurement criteria.
- Using CMS for reports that indicate the operational efficiency of ACD.

An agent is a person or a voice response port that answers ACD calls. Companies that operate high-volume call centers, for example, a catalog sales, reservations, or customer service center, use ACD to process and distribute incoming calls to agents. The split supervisor can use Communication Manager and CMS features to monitor the split and agent performance and provide assistance.

Automatic-in processing

With automatic-in processing, the CO processes all the digits of an incoming call. The CO then seizes a trunk from the trunk group. As processing is complete, the call connects directly to a destination identified in the communication server software. The destination can be a phone, a queue in which callers wait to be answered in the order in which their call was received, or a special treatment like an announcement.

You can assign each automatic-in trunk group to an ACD split. Communication Manager directs all incoming calls from an automatic-in trunk group to the assigned split. ACD then distributes the calls to the agent extensions assigned to the split based on the assigned call distribution method.

DID processing

With Communication Manager, you can directly dial various extensions such as a Vector Directory Number (VDN), a hunt group, an agent, or a login ID. You can assign each extension to a split as a Direct Inward Dialing (DID) extension. With DID processing, incoming trunks do not connect the CO directly to the phone of the employee. Instead, the communication server pools the incoming trunks and the employee phones share the pool of trunks. Extension numbers serve as the final digits of employee phone numbers for incoming calls, that is, the CO assigns a 2-digit, 3-digit or 4-digit prefix to a trunk group. When a caller dials a 7-digit employee phone number, call processing is as follows:

1. The CO processes the prefix of the dialed number and seizes a trunk in the trunk group that is assigned the prefix.
2. The CO passes the remaining digits of the dialed number to Communication Manager.
3. Communication Manager recognizes the remaining digits as an employee extension number and sends the call to that extension.

Note:

Automatic-in trunk groups carry calls only to the split, whereas DID trunk groups carry calls to any extension identified in Communication Manager, not just to a split.

DID processing example

As an example of DID processing, say that employee A has an external phone number 538-1000 and an extension number 1000. Employee B has a phone number 538-9999 and an extension number 9999.

The steps for completing calls to employees A and B are as follows:

1. Employee A’s client dials 538-1000.
2. The CO serving employee A’s company identifies the digits 538 (the common prefix for all phone numbers to that company) and seizes Trunk 1 in the trunk group assigned the digits 538.
3. The CO passes the digits 1000 to the communication server at employee A’s company.
4. The communication server identifies the digits 1000 as employee A’s extension number and sends the call to employee A’s extension.
5. Employee A’s phone rings and employee A answers.
6. Meanwhile, employee B’s client dials 538-9999.
7. The CO identifies the digits 538 and seizes Trunk 2 in the trunk group assigned the digits 538.
8. The CO passes the digits 9999 to the communication server.
9. The communication server identifies the digits 9999 as employee B’s extension number and sends the call to employee B’s extension.

10. Employee B’s phone rings and employee B answers.

While employees A and B continue to talk, Trunks 1 and 2 in the 538 trunk group cannot accept any more calls, so another call beginning with the digits 538 seizes yet another trunk in the trunk group.

Split queue call processing

A split queue is a holding area for calls waiting to be answered, usually in the order in which the calls are received. When a call is put in a queue, the caller hears a delay announcement, music or silence, depending on the treatment assigned to the split. You can administer the call treatment for calls in the queue.

Calls enter the queue at the bottom and move up towards the top of the queue. After a call reaches the top of the queue, the call connects to the next available agent.

For Communication Manager with the Call Vectoring feature, call vectors specify all call treatment including routing, queuing, announcements, and music. When a call arrives at a split, ACD checks if an agent is available to handle the call. If an agent is unavailable or is busy, the call enters the split queue.

Calls queue only if all agents are unavailable. If the queue is full, the caller hears a busy tone or the call goes to coverage. If the split is vector controlled, this step fails. Furthermore, if no agents are logged in to the split or if all agents are in the AUX work mode, calls do not queue up.

Priority and normal split queues

Each split can have two queues: normal and priority. A split always has a normal queue but you can assign a priority queue to a split. ACD distributes all calls to the priority queue, if one exists. ACD distributes calls to the normal queue only after the priority queue is empty.

You can assign priority queuing in the Class of Restriction (COR) that is associated with the split extension number. You can also assign a split priority queuing on Intraflow, which means that calls to that split, if rerouted to another local split, enter the priority queue of the destination split.

Dynamic queue slot allocation

Communication Manager dynamically allocates queue slots to hunt groups or skills. Communication Manager extracts a queue slot resource from a common pool. When Communication Manager removes a call from a queue, the queue slot resource is released and returned to the common dynamic pool.

Dynamic queue slot allocation offers the following advantages:

- Reduced administration
- Increased capacities such as increased skill availability in a call center
- Elimination of lost or blocked calls when all queue slots are full

Note:

You can limit the actual number of calls that can be queued for a specific hunt group by using the *calls-queued* conditional in the **check split/skill if calls-queued** or **goto step/vector if calls-queued in split/skill** vector commands.

Administer the **Queue Limit** field to specify the maximum number of calls that are queued to a hunt group.

Announcements for calls in a split queue

When a call enters a split queue, the caller hears the ringing tone until Communication Manager connects the call to an agent or an announcement. The caller hears an announcement, music, or silence.

An announcement is a recorded message that provides information such as the destination the call has reached or the business hours of a company.

Communication Manager supports internal and external announcement devices. You can assign announcements and delay time to splits through the communication server administration.

Delay time is the amount of time a call waits in queue before receiving an announcement. If a call connects to an agent before the delay time expires, the caller does not hear the announcement. If a call connects to an agent while an announcement is playing, the announcement stops. After the first announcement, the caller hears music or silence until the second announcement or the call connects to an agent.

The announcement delay time can be from 0 to 99 seconds. A 0-second delay time causes a forced announcement, which means that callers always hear the entire first announcement, whether an agent is available, that is, idle or not. You can administer a second announcement for each time the first announcement delay time expires.

Rules for announcements

The system plays an announcement from beginning to end unless an agent becomes available. When an agent becomes available, the system interrupts the announcement and provides a ringback tone. Calls remain in the queue while the system plays the announcement. Any feedback that the system provides before an announcement continues until the announcement is played. For example, a wait with some music or ringback.

Without vectoring: If an announcement queue is full, the system continues to try to connect the call to the proper announcement until the call connects to an agent, to an announcement, or enters the announcement queue. The following rules apply to announcements without vectoring implemented:

- Calls that directly enter a split queue always receive a forced first announcement. The caller also hears first and second delay announcements, if administered, and the delay intervals are met.
- Calls that reach a split by way of Call Coverage from another split (Intraflow) or a station do not receive a forced or delay first announcement at the destination split. The caller hears a second delay announcement, if administered, and the delay interval is met.

- Calls that reach a split by way of Call Forwarding from another split (Interflow) or station do receive delay first and second announcements if administered and the delay intervals are met.

With vectoring: If the announcement queue is full, the call retries the announcement step for an indefinite period before any new vector steps are processed. If an **announcement** command follows a failed **adjunct routing** command, the announcement is interrupted. If the **adjunct routing** command succeeds, that is, the communication server receives a destination from the ASAI adjunct, the announcement terminates immediately. The **announcement** command step is skipped and vector processing continues at the next vector step, whenever any of the following conditions exist:

- Requested announcement is busied out, not available, or not administered
- Integrated board is not installed
- External aux trunk or analog equipment is not attached

Announcement queuing

The number of calls that can be queued to an announcement depends on the size of the communication server. The capacity tables in the *Avaya Aura® Communication Manager System Capacities Table* have details for each communication server model.

Internal announcements are delivered by a multiport or channel announcement board, and a call receives an announcement only when the call connects to one of the announcement ports or channels. Therefore, all calls wait in a single queue to gain access to a channel on the announcement board regardless of the split announcement. The same announcement can be delivered over multiple channels. Announcements are delivered on demand, so a call that connects to a channel receives an announcement immediately and does not have to wait for the announcement to finish and start again.

Answer supervision

Answer supervision is a signal that Communication Manager sends to the CO. The signal indicates to the CO that an incoming call is sent to an agent phone and that the CO must begin tracking toll charges, if applicable, for the call. Communication Manager sends answer supervision before a call connects to the agent phone, music, or announcement.

Abandoned calls

An abandoned call is a call that reaches a call center, but does not connect to an agent because the caller hangs up before the call connects. A caller can abandon a call while in queue or while the phone is ringing at an agent station. Abandoned calls represent lost sales or lost goodwill. Adequate split staffing and effective use of announcements reduces the number of

abandoned calls. You can staff the splits so that calls do not have to wait in queue for long and announcements persuade the caller to wait until an agent answers the call.

If Communication Manager sends an answer supervision signal before a caller abandons the call, a ghost call can occur. A ghost call is a call that an agent receives after the caller hangs up. Ghost calls occur because, after a caller disconnects, some COs wait for 2 seconds to 25 seconds before sending a disconnect signal to Communication Manager. Ghost calls waste agent time preventing other calls from connecting to an agent. To reduce the instances of ghost calls, you can assign the Abandoned Call Search feature to specific trunk groups.

With Abandoned Call Search, Communication Manager checks the incoming trunk before delivering an ACD call to an agent. If the trunk is on-hook at the CO, that is, the caller has abandoned the call, Communication Manager releases the trunk and does not deliver the call. If the call is still in progress on the trunk, Communication Manager delivers the call to an agent.

Intraflow and interflow

You can use Intraflow and Interflow to redirect ACD calls to another split, an internal destination, or a remote destination. Use Intraflow to redirect calls to an internal destination. Use Interflow to redirect calls to a destination outside the communication server.

You can administer three intraflow destinations or one interflow destination for a split. Intraflow uses the Call Coverage feature to redirect ACD calls to a coverage path that includes more than one of the following internal destinations or call treatments:

- An extension.
- A hunt group or an ACD split that includes a messaging system and message center splits. A hunt group is a group of extensions that receive distributed calls. A split is a hunt group that is measured by a reporting adjunct such as CMS.
- An attendant group.
- An announcement followed by a forced disconnection.

You can also administer call forwarding and ACD splits to intraflow calls unconditionally.

For external call redirection, you can administer a Centralized Attendant Service (CAS) attendant in addition to the destinations and call treatments that you use with Intraflow.

How to administer ACD splits

If you assign more than one intraflow destination to a split, Communication Manager attempts to redirect calls to each destination. If no destination accepts the call, Communication Manager leaves the call in the original split queue. If you assign an interflow destination, Communication Manager redirects the call only to the interflow destination. If the interflow destination does not accept the call, the caller hears a busy signal.

When you administer ACD splits to intraflow calls unconditionally, Communication Manager redirects all calls to the specified destination. Use unconditional intraflow to redirect calls when a split is not staffed.

You can also administer ACD splits to intraflow calls if one or all of the following criteria are met:

- Don’t Answer: When you administer call redirection to **Don’t Answer**, Communication Manager redirects the call if an agent fails to answer the call within the assigned interval. You can set a value from 1 to 99 ringing cycles.
- No Agents Staffed or All Agents in the AUX work mode: Communication Manager redirects calls if agents are unstaffed or if all agents are in the AUX work mode.

Queue threshold assignment

To increase the priority of an intraflow call, you can assign priority queuing on intraflow of the call to the destination split. You can also assign an inflow threshold to the queue of the intraflow destination. The inflow threshold is the length of time that the oldest call waits in queue. Once the queue reaches the inflow threshold, the queue does not accept any intraflow call and Communication Manager redirects the call to the next administered destination.

Intraflow assignment type

You can administer the following types of intraflow or interflow for a split:

- Don’t Answer Time Interval Intraflow with Call Coverage.
- Busy Intraflow with Call Coverage.
- Unconditional Intraflow with Call Forwarding-All.

When you administer intraflow of calls with Call Coverage, Call Management System reports inflowed and outflowed calls only if the call queues to the original split.

For example, a call that covers with the busy criterion is not recorded as inflowed or outflowed because the call did not queue to the original split. Calls that queue before covering with the Don’t Answer criterion are recorded as inflowed or outflowed calls.

How to administer Intraflow or Interflow

You can administer a split for Intraflow or Interflow, but not both. However, for a given split, you can use conditional and unconditional intraflow simultaneously. Conditional intraflow uses the Call Coverage feature and unconditional intraflow uses the Call Forwarding feature. Communication Manager uses the unconditional criterion before the conditional criterion to forward a call to the destination. When you administer unconditional and conditional intraflow for simultaneous use, Communication Manager applies the conditional intraflow criteria to the *forwarded-to* destination, and not to the original split.

You can administer unconditional and conditional intraflow criteria such that the Dialed Number Identification Service (DNIS) numbers appear on the agent phones. For these criteria, the DNIS number is a split extension with unassigned agent extensions. The intraflow destinations are splits with staffed agents. With such a configuration, CMS reports incoming calls for the DNIS number

that is redirected using unconditional intraflow to real splits as *outflowed*. CMS also reports the calls to the destination split as *ACD* and *inflowed* calls.

Administer console permissions and the Call Forwarding dial access code so that a split supervisor can use a station to activate unconditional intraflow or interflow by entering the dial access code, the split extension, and the destination number.

A split supervisor cannot establish conditional intraflow from a phone.

Note:

You cannot use CMS to set up or activate Intraflow or Interflow.

Night Service

You can use Night Service to redirect all calls to one of the following internal destinations:

- ACD split
- Extension
- Attendant group
- Announcement with forced disconnect

Night Service is available for a hunt group, a trunk group, or a system.

Hunt Group Night Service

Hunt Group Night Service redirects all calls arriving at a split to an internal destination. The Night Service destination for the split and the telephone button used to activate the feature are assigned through communication server administration.

Trunk Group Night Service

Trunk Group Night Service redirects all calls arriving over a split's assigned trunk groups to an internal destination. The Night Service destination for the trunk group and the telephone button used to activate the feature are assigned through communication server administration.

Trunk Group Night Service by itself does not guarantee that all calls to a split will be redirected. Calls from local extensions and DID calls will still connect to the split.

Trunk Group Night Service and Hunt Group Night Service can both be active at the same time. If the Trunk Group Night Service is active, the destination is used for calls that come in over the trunk group even if the calls go to a split that has a Hunt Group Night Service destination assigned.

System Night Service

System Night Service redirects all calls arriving over all trunk groups to the Night Service destination. System Night Service overrides any Hunt Group Night Service set up for an individual

split. If Trunk Group Night Service is active for a particular trunk group, System Night Service does not affect that trunk group. When any type of Night Service becomes effective, calls already in a split's queue are not redirected. Agents must continue to staff the split until the queue is empty.

Call distribution methods

The call distribution method, also known as hunting, that you assign to the split or skill determines the delivery of ACD calls to agents.

To establish queues for agents and calls, enable **ACD** and **Queue** on the Hunt Group screen.

Queues handle incoming calls based on the call distribution method, the availability of which depends on whether you enable Expert Agent Selection (EAS).

Call distribution methods without EAS

Without EAS, you can use the following call distribution methods:

- Direct Department Calling
- Uniform Call Distribution-Most Idle Agent (UCD-MIA)

Note:

In the following descriptions of ACD, Multiple Call Handling (MCH) is set to `n`. Agent availability is different for splits where MCH is set to `y`.

Direct department calling

Communication Manager searches for an available agent in the order of assignment of extensions to the split. With this call distribution method, you can use the most effective or experienced agents to handle more calls. Agents are ranked from the most to the least effective and are then assigned to the split in that order.

If you administer a split for Direct Department Calling (DDC), Communication Manager routes incoming calls to the first available agent extension in the administered sequence. If the agent is not available, the call routes to the next available agent.

UCD-MIA

If you use UCD-MIA, Communication Manager searches for an agent extension that has been idle for the longest time and delivers the call to the extension if the agent is available to receive an ACD call. The UCD-MIA method ensures a high degree of equity in agent workloads even when call-handling times per call vary.

Communication Manager determines which agent extension has been idle the longest by maintaining an ordered list of agents who are eligible to receive the next ACD call. Eligible agents enter the queue at the bottom and move towards the top of the queue. The agent who has been in queue for the longest time receives the next ACD call unless the agent is unavailable at the time

when Communication Manager delivers the call. If the agent at the top of the queue is unavailable, Communication Manager checks the availability of the next agent in queue until an available agent is found.

When an agent completes an ACD call, the agent is added to the bottom of the eligible-agent queue for the split associated with the call. The **MIA Across Splits/Skills** field is used to put an agent at the bottom of all split or skill queues that the agent is logged into when the agent completes an ACD call. Agents move toward the top of the eligible-agent queue as long as the agents remain staffed and available or on AUXIN or AUXOUT extension calls from the available state, or on an ACD call for another split, unless the MIA across splits or skills option is turned on. You can decide if the agents in the After Call Work (ACW) or AUX Work modes are in the eligible-agent queues.

An agent is marked as unavailable to take an ACD call if the agent is:

- In ACW or AUX.
- On an AUXIN or AUXOUT extension call from the available state.
- On an ACD call for another split or skill.

The agent remains in queue moving toward the top of the queue. Agents in multiple splits enter multiple eligible-agent queues. The agent progress in each queue is independent of any activity in other queues.

You can administer Communication Manager to maintain a separate queue for available agents in each split or skill, or you can create one combined queue for agents in all splits or skills.

If the **MIA Across Splits/Skills** field on the Feature-Related System Parameters screen is set to `n`, Communication Manager maintains the available agent queue for each split or skill. When an agent answers a call, Communication Manager removes the agent from the available agent queue for the split or skill at which the call arrived. If the field is set to `y`, the agent is removed from all split or skill queues that the agent is logged in to when the agent answers a call for any of the assigned splits or skills.

The agent is returned to the agent queues, based on how you administer the following:

- If forced Multiple Call Handling applies, the agent is placed in the queue when the call stops alerting.
- If the **ACW Agents Considered Idle** field on the Agent LoginID screen is set to `y` or `system` and the **ACW Agents Considered Idle** field on the Feature-Related System Parameters screen is set to `y`, the agent is queued when the call completes.
- If the **ACW Agents Considered Idle** field is set to `n`, the agent is queued when ACW completes.
- If the **AUX Agents Considered Idle (MIA)** field on the Agent LoginID screen is set to `y` or `system` and the **AUX Agents Considered Idle (MIA)** field on the Feature-Related System Parameters screen is set to `y`, the agent is queued when the agent logs on. Agent queuing also depends on the value set in the **ACW Agents Considered Idle** field, as mentioned in the previous list item, and also on when the agent enters the AUX work mode, if the agent was not previously in the queue.

Note:

If you use an Expert Agent Distribution (EAD) method, that is EAD-MIA or EAD-LOA, the agent is put back in the queue after completing an ACD call against the administered skill level. If you do not use the EAD method, the agent is put at the bottom of the queue after completing an ACD call or otherwise entering the eligible-agent queue.

Call distribution methods with EAS

With EAS, you can use the following call distribution methods:

- Uniform Call Distribution-Most Idle Agent (UCD-MIA)
- Expert Agent Distribution-Most Idle Agent (EAD-MIA)
- Uniform Call Distribution-Least Occupied Agent (UCD-LOA)
- Expert Agent Distribution-Least Occupied Agent (EAD-LOA)

The following table summarizes the different call distribution methods, which are further defined in the following sections.

Agent selection method	Communication Manager distribution
EAD-MIA	The highest skill level, downward from 1 to 16, and most idle agent.
UCD-MIA	The most idle agent without regard to skill level.
EAD-LOA	The highest skill level agent with the lowest occupancy.
UCD-LOA	The least occupied agent without regard to skill level.

UCD-MIA

UCD-MIA works in the same manner in the EAS environment as in a non EAS environment, except that the Communication Manager searches for the most idle agent with the required skill and does not use any per agent administered skill level preference.

UCD-MIA does not select an agent based on the skill level. Therefore, if an agent is the most idle agent with the required skill, even if the skill is assigned a secondary skill level for that agent, the call is delivered to the agent.

EAD-MIA

The EAD-MIA call distribution method selects the most idle agent with the highest skill level for the required skill to handle the call. This method of call distribution adds a layer of processing on top of the Most Idle Agent (MIA) distribution call processing.

EAD-MIA sorts the agents in the eligible-agent queue into multiple queues based on the skill level. Agents with the skill assigned at higher-priority levels appear in the eligible-agent queue ahead of agents with the skill assigned at lower-priority levels. The call is delivered to the most idle and most expert agent available.

When you use EAS Preference Handling Distribution (EAS-PHD), the agent can enter the MIA queue at one of the 16 levels. The lower the level, the higher the level of expertise. An agent with

skill level 1 is the most qualified to answer a call to that skill. Without EAS-PHD, agents enter the MIA queue as either level 1 (Primary) or level 2 (Secondary) agents.

When agents with a lower skill level become idle, the agents enter the MIA queue in front of agents with a higher skill level.

UCD-LOA

When the UCD-LOA call distribution method is in use, Communication Manager delivers the call to the least occupied agent without regard to the skill level.

Least Occupied Agent (LOA) is the agent who has spent the lowest percentage of time on ACD calls. The position of the agent in the queue of available agents is determined by this percentage. The agent occupancy, that is, the percentage of time on calls, is always calculated separately for each agent skill, so there is an available (idle) agent queue for each skill.

If the **AUX Agent Remains in LOA Queue** field on the Agent LoginID screen is set to `y` or `system` and the **AUX Agent Remains in LOA Queue** field on the Feature-Related System Parameters screen is set to `y`, the occupancy is calculated from the time the agent logged in. Otherwise, the occupancy is calculated from the time the agent last entered a working state.

EAD-LOA

When the EAD-LOA call distribution method is in use, Communication Manager delivers the call to the least occupied agent with the highest skill level.

The agent occupancy is calculated as described in the UCD-LOA section.

MIA across skills for EAS

You can use UCD and EAD with the **MIA Across Splits/Skills** field.

You can use the field to:

- Retain the agent position in other splits or skills MIA lists when handling an ACD or Direct Agent (DA) call. This is the default administrative setting.
- Remove the agent from all MIA lists when handling a call from a split or skill

Distribution is based on the total call activity and not on the activity in a single skill.

How do agents handle calls

An agent can receive split calls and personal calls. Calls distributed to an agent phone by Communication Manager are ACD calls.

Calls, such as internal and DID extension calls, that are dialed directly to an agent are called Extension-In (EXT-IN) calls. Outgoing calls that the agent makes are called Extension-Out (EXT-OUT) calls. EXT-IN and EXT-OUT calls are non ACD calls. You can administer an agent phone to receive EXT-IN calls or to make EXT-OUT calls.

Communication Manager distributes ACD calls only to available agent extensions. An agent must log in to become staffed and select a call answering mode, that is, automatic-in or manual-in.

Staffing non-EAS agent extensions

To staff a non-Expert Agent Selection (EAS) agent extension on Communication Manager, an agent must:

1. Type an access code to log in or press the **LOGIN** Abbreviated Dialing (AD) button on the deskphone.
2. Type a split number and a login ID.

You can use Communication Manager to administer the login access code, the **LOGIN** AD button, and the length of the login ID. You can also assign the split number to the **LOGIN** AD button or to another button on the deskphone.

Staffing multiple splits

An agent can log in from any extension assigned to a split. An agent can log into as many as four splits. To the communication server and CMS, each login counts toward the maximum number of agent members that can be measured. That is, if four agents are each logged into three splits, the agent member count is 12.

Agent login

Agent login lets ACD and CMS know if an extension is active and logged-in to the system.

An agent presses the login button and then follows the system login procedure to become staffed in the AUX work mode. However, this procedure varies based on the type of system.

The administered work mode is overridden in the following cases:

- When an agent logs in to Call Center Elite using ASAI or CTI and the agent has entered a work mode using ASAI or CTI.
- When an agent logs in to Call Center Elite using Avaya one-X® Agent and the agent has specified a work mode.

Agent logout

Agent logout lets ACD and CMS know that an extension is no longer active.

Agent request for supervisor assistance

An agent presses the **assist** button or dials the ASSIST FAC to bring the supervisor on line. Pressing the **assist** button automatically places the current call on hold.

Agent states and call-answering modes

Agent state is the current status of an agent. Work modes are the work functions that the agent is performing at a given time. When the agent is engaged in an ACD call, the agent is in the ACD state. After staffing an extension, the agent is in the AUX work mode, which is a non ACD work mode.

Auto-In versus Manual-In

In the Auxiliary (AUX) work mode, the agent is unavailable to receive ACD calls. To become available for ACD calls, the agent must press **manual-in** or **auto-in**.

Work mode	Description
Auto-In	The **auto-in** button indicates that the agent is available for an ACD call. When the call ends, the agent is immediately available for another ACD call based on the administered call distribution method. The agent does not have to press any button to receive another ACD call. This type of call answering increases the number of calls that the agents can answer in a given period and is most effective if the agents have little or no call-related work after finishing ACD calls. Communication Manager has a Timed After Call Work (TACW) feature for the auto-in mode. This option automatically puts the agent in ACW for a preset length of time at the end of the call. When the time is up, the agent automatically becomes available to receive another ACD call. Manual-in and auto-in dial access codes and phone buttons are assigned through the Communication Manager administration.
Manual-In	The **manual-in** button indicates that the agent extension is available for an ACD call. Communication Manager distributes a call based on the administered call distribution method. When the call ends, the agent automatically enters the ACW state. While in ACW, the agent is unavailable to receive ACD calls. When ACW ends, the agent presses **manual-in** to receive another ACD call. The manual-in mode is most effective if an agent must perform call-related tasks after finishing each ACD call. Manual-In dial access codes and phone buttons are assigned through Communication Manager administration.

Ringing versus zip tone for incoming calls

When a call arrives at an agent phone, the agent hears a ringing or zip tone based on the phone administration.

Administer a ringing tone when an agent answers calls using the handset. When a call connects to the agent phone, the phone rings and the agent picks up the handset to answer the call.

Administer zip tone when the agent uses a headset to answer calls. The agent can use the zip tone with a handset, but the agent must hold the handset and wait for the zip tone. When a call connects to an agent phone, the agent hears a single burst of zip tone for calls dialed directly to the split. The staffed agent does not have to push any button to greet the caller. This process is commonly known as Call Forcing.

Ringing or zip tone is established on each phone through Communication Manager administration. The ringing tone is called manual answer and the zip tone is called automatic answer.

Auxiliary and After Call Work

An agent presses the **aux-work** or the **acw** button to temporarily stop ACD calls from arriving at the agent phone.

Mode	Description
AUX work	The agent is involved in non ACD work, is on break, in a meeting, or at lunch. Call Management System recognizes the extension as staffed but does not want ACD to route calls there for an extended time. AUX-IN implies that the extension received an extension-in call while in the AUX work mode. AUX-OUT implies that the agent placed an outgoing call while in AUX work. The **aux-work** button temporarily stops ACD calls from arriving at the phone. The agent normally presses this button before doing non ACD-related work such as taking a break, or doing personal business. Instead of unstaffing the extension or logging off, an agent can press the button, which places the agent in the AUX work mode. To receive ACD calls, the agent presses the **manual-in** or **auto-in** button. The **aux-work** button or the dial access code, if no button is available, is assigned through Communication Manager administration. If an agent is normally logged into more than one split, an **aux-work** button for each split can be assigned. When the agent presses the **aux-work** button for a particular split, the agent does not receive calls from the split. However, the agent is still available for calls from the other splits the agent is logged into. If an agent logs in to more than one split or skill and receives an ACD call, the agent is unavailable for calls for other splits or skills. When the service level threshold for an interruptible hunt group exceeds, Communication Manager notifies agents with the interruptible skill who are in the AUX work mode with an interruptible reason code. The notification consists of a display message `You are needed`, flashing **auto-in** or **manual-in** buttons, and an audible tone. Agents who move to an Interruptible Aux mode after the threshold is exceeded are also notified. The duration of notification to *Auto-In-Interrupt* agents is administrable using the **Interruptible Aux Notification Timer (sec)** field on page 13 of Feature-Related System Parameters screen.
ACW	The agent is engaged in work associated with a call, but not on a call. ACW-IN implies that the station received a call while the agent was in the ACW mode. ACW-OUT implies that the agent made an outgoing call while in ACW. The **acw** button temporarily stops ACD calls from arriving at the station. An agent who is in auto-in mode presses the **acw** button during a call so that when the call is finished, the agent does not receive another ACD call and can, instead, do ACD call-related work such as filling out a form, completing data entry, or making an outgoing call. The lamp indicator next to the **acw** button lights when the agent is in ACW. When in the manual-in mode, an agent automatically enters ACW when the call ends. However, if the agent needs to get out of auto-in mode or the auxiliary work state to do additional call-related work, the agent can press the **acw** button or dial an appropriate access code. An agent can press the **manual-in** button or dial the appropriate access code while on an ACD call to automatically enter ACW when the call ends. If an agent is logged in to more than one split, pressing the **acw** button makes the agent unavailable for calls in all splits. Call Management System measures the agent state as **OTHER** state for all splits other than the split in which the agent is currently in ACW.

The following table lists additional agent states or work modes.

Agent state or work mode	Description
UNSTAFF	Unstaffed The agent is not logged in and is being tracked by Call Management System(CMS) or Avaya IQ.
DACD	The agent is on a direct agent ACD call.
DACW	The agent is in the ACW state for a direct agent ACD call.
OTHER	The agent is doing other work. If an agent is working in three splits or skills and receives a call from one, the ACD puts the agent in OTHER for the other two.
UNKNOWN	Call Management System does not recognize the current state. Unknown remains until the condition is cleared or the agent completes the current ACD call and any current ACW, or a current agent state message is sent to Call Management System from Communication Manager.
RING	The time a call rings at a phone after leaving the queue and before the agent answers.

Trunk states

Trunk State indicates the current status of a specific trunk.

Trunk state	Description
Idle	The trunk is waiting for a call.
Seized	The trunk is seized by an incoming or outgoing call.
Queued	An ACD caller has the trunk and is waiting for the agent to answer.
Conn	The agent and caller are connected in an ACD call.
Abandoned	The queued caller has just abandoned the call.
Fwrd	A queued call has been intraflowed outside the ACD or has been interflowed to another PBX or communication server.
Mbusy	Maintenance busy or out of service for maintenance purposes.
Hold	The agent has put the call on hold.

Station button assignments

Station button type	Description
ADD SKILL	Logged-in, that is, staffed Expert Agent Selection (EAS) agents or phone users with console permissions can press this button to add a skill. This button an Abbreviated Dialing (AD) button programmed with Add Skill Feature Access Code (FAC).
ALERT CHANGE	The lamp associated with this button flashes when another user changes an assigned skill or moves an agent from the current skill to a different skill. The lamp does not flash when agents change skills.

Table continues…

Station button type	Description
ASSIST	Agents press this button to request the split supervisor for assistance. When an agent presses this button, Communication Manager puts the current call on hold, dials the extension of the supervisor, and connects the agent to the supervisor.
AUDIO TROUBLE	Agents press this button for CMS reporting of a call with poor audio quality. Communication Manager sends a message to CMS that includes the extension number of the agent, the trunk, and the time of day when the incident occurred. CMS reports this information in the CMS Exception reports, which helps in troubleshooting trunk problems and extension problems. Use Stroke/Event Count button 0 for reporting audio difficulty.
CALL APPEARANCE	Agents use this button to make and receive calls. This button has two status lamps: red and green. The red lamp lights when an agent presses an appearance button to make or receive a call. The green lamp flashes to indicate an incoming call. Except with Multiple Call Handling (MCH), incoming ACD calls always arrive at the first call appearance. However, you can assign more than one appearance to the phones for more call handling capabilities. For example, an agent can use a second call appearance to transfer calls or make calls. On a double appearance phone, agents can use the second appearance to make calls.
CONFERENCE	Agents press this button to add another person to a two-person call. An agent with a multiappearance phone can add up to four people to a two-person call. For single appearance phones, the agents can add only one person. Single appearance phones do not have a **conference** button. Agents must use the **recall** button to conference a call. If an agent adds another agent to a conference call, the resulting conference call is not an ACD call for the added agent, but an Extension-In call.
CALL WORK CODE	Agents press this button and enter up to 16 digits to record the occurrence of a customer-defined event. CMS stores Call Work Codes (CWCs).
CALLER INFORMATION	With Call Prompting, agents can press this button to view the digits collected by the last **`collect digits`** vector command.
EMERGENCY	Agents press this button to report a malicious call to the controller. The controller then traces the call.

Table continues…

Station button type	Description
HOLD	Agents press this button to put a call on hold. ACD does not send any more calls to an agent who has a call on hold. For Communication Manager with MCH, an agent can put an ACD or non-ACD call on hold and press **auto-in** or **manual-in** to receive an ACD call. With MCH, Communication Manager can deliver multiple ACD calls to an agent in the auto-in or manual-in work mode, provided that there is an unrestricted line appearance is available on the phone. Single appearance phones do not have a **hold** button. Agents must use the **recall** button or the switch hook of the phone to put a call on hold. Agents cannot use a single appearance phone to handle multiple ACD calls.
LOGIN	Agents press this button to staff the extension and start CMS collection of agent data. This button is an AD button programmed with the login FAC.
LOGOUT	Agents press this button to unstaff the extension and end CMS collection of agent data. This button is an AD button programmed with the logout FAC.
RECALL/Flash switchhook	To put calls on hold, transfer calls, and create conference calls, agents using single appearance phones press the **recall** button or flash the switch hook if the phone does not have **recall** button.
RELEASE	Agents press this button to disconnect a call. Communication Manager logs an agent out of the system if the agent presses **drop**.
REMOVE SKILL	Logged-in, that is, staffed EAS agents or phone users with console permissions can press this button to remove a skill. This button is an AD button programmed with the remove skill FAC.
STROKE COUNT	Agents press this button to record call events. Stroke Count (SC) is also known as Event Count. Therefore, the terms can be interchanged. You can assign up to nine SC buttons. CMS records and reports SC information. SC button 0 is reserved for audio difficulty.
TRANSFER	Agents press this button to transfer calls to other agents or the split supervisor. This button is available only on multiappearance phones. Single appearance phone users must use the switch hook of the phone. Agents can also use this button to transfer calls to external destinations. External transfer must be assigned to a phone as a feature over and above the normal transfer feature. If an agent transfers a call to another agent, the call is not an ACD call for the agent receiving the call unless the transferring agent dialed a split extension, Vector Directory Number (VDN), or agent login ID, an EAS capability known as Direct Agent Calling. ACD treats the agent receiving the transfer to be on an Extension-In (EXT-IN) call. For the agent transferring a call, the call is counted as an Extension-Out (EXT-OUT) call.
VUSTATS	Agents with display phones press this button to view agent, split or skill, VDN, or trunk group data similar to that reported by CMS.

Table continues…

Station button type	Description
VDN-INFO	Agents with display phones press this button, Communication Manager sends the complete VDN name for the active call in the existing display format to the phones. After 10 seconds, the phones display gets restored to their original display. The button press is recognized only if the agent is in the normal display mode. If the agent is in any mode other than the normal display mode, such as the inspect mode, the button press is ignored.

Queue status lamps

The lamps associated with the queue status buttons provide the following information.

Lamp	Description
NQC	The NQC lamp, which is associated with the Number of Queued Calls (NQC) button, indicates that calls are in queue and that the number of calls in queue has met or exceeded the assigned queue threshold. If no calls are in the split queue, the status lamp associated with the button is dark. When more than one call is in queue, the lamp lights steadily. When the number of calls in queue reaches the assigned queue threshold, the lamp flashes on and off.
OQT	The OQT lamp, which is associated with the Oldest Queued Time (OQT) button, indicates that calls are in queue and that the oldest call in the queue has been waiting longer than the assigned wait time threshold of 0 to 999 seconds. If no calls are in the split queue, the status lamp is dark. When calls are in queue, the lamp lights steadily. When the assigned wait time threshold has been met or exceeded, the lamp flashes on and off.
Auxiliary queue status lamps	An auxiliary queue status lamp indicates that either the NQC or the OQT threshold is reached. The lamp lights when the assigned threshold is met or exceeded. Unlike the lamps on a phone, the auxiliary queue status lamp does not indicate when calls queue to the split.

Display buttons

The following phone button controls the information that appears on the display.

Button	Description
NORMAL	Agents press this button to view information on an active call appearance.

Split supervisor phone buttons

A split supervisor is normally assigned to each split. The capabilities that allow monitoring of agent performance, adding and removing agents, and performing other split-related activities must be assigned with separate Communication Manager administration procedures.

Supervisor phone buttons

The following phone features can be used for supervisory use:

Button	Description
NIGHT SERVICE	The split supervisor presses this button to send all calls to night service. The night service can be trunk group night service or split night service. There are separate buttons for each type of night service.
SERVICE OBSERVE	The supervisor presses the service observing button and enters an extension to observe. This can be a VDN extension, station extension, or a logical Agent ID extension. The Service Observe feature permits the supervisor to check the call-handling techniques of an agent. The phone can also be assigned the **serv-obsrv** or **sip-sobsrv** button so that the supervisor can listen to conversations of another agent. This capability is useful for agent training. Service Observing can be set up for listening only or for both listening and talking. The supervisor can press the **so-coach** button or the **Coach** soft key on SIPCC endpoints while observing a call to coach the agent by whispering advice to the agent without the customer hearing the conversation between the supervisor and the agent. Supervisors can coach only if the following conditions are fulfilled: • The supervisor is Service Observing an ACD agent in Listen-Only or Listen & Talk modes and not in the No-Talk mode. • The supervisor is Service Observing an active call, which is not on hold. • Supervisors must always be observing a call before trying to coach a call. For conference calls, prior to Release 8.1.1, coaching on a conference call was blocked. With Release 8.1.1, if the **Coach on SSC** option is enabled, supervisors can coach agents using the Service Observing Coach functionality even while the call is being recorded by recording applications that use the invisible ASAI SSC option to record calls. Prior to this option being available, supervisors could not coach agents on any call that was being recorded. If the **Coach on SSC** option is disabled, agent coaching continues to be blocked on any other conference call. Note that if the call is an invisible ASAI SSC call, the phone displays for the other parties on the call do not say “Conference”.
VU STATS	Split supervisors and agents with display telephones press this button to view agent, split or skill, VDN, or trunk group data similar to that reported by CMS.

Chapter 3: ACD features

Abandoned Call Search

Communication Manager uses Abandon Call Search to identify abandoned calls.

Note:

Use Abandoned Call Search only for old COs that do not provide timely disconnect supervision. Most COs provide timely disconnect supervision and do not require this feature.

Before an incoming ACD call rings a hunt group member or agent station, Communication Manager checks if the calling party has abandoned the call to ensure that an abandoned call is not delivered to the agent.

If the calling party abandons the call, Communication Manager determines if the calling party is still connected to the ground-start trunk at the CO by opening the tip-ring loop at the CO end of the trunk for 150 to 200 milliseconds (ms). If the calling party is still connected, the CO does not respond. If the calling party is not connected, the CO sends a disconnect signal to Communication Manager within 800 ms. Communication Manager interprets the signal as an abandoned call and releases the trunk.

Abandoned Call Search considerations

- Abandoned Call Search works with ground-start analog trunks that do not provide disconnect supervision and react to a 500-ms break.
- Some older COs can take up to two minutes to notify Communication Manager about a call disconnect. Communication Manager must, therefore, determine within one second if the call has been abandoned, before delivering the call to a hunt group. Call-handling statistics that CMS generates are more accurate because it is clear when a call is abandoned.
- Abandoned Call Search adds an overhead of up to one second to each call delivered to an agent.

ACD field options for agents

For each agent, you can administer the following ACD field options on the Agent LoginID or the Feature-Related System Parameters screen:

- **ACW Agent Considered Idle**
- **AUX Agent Remains in LOA Queue**
- **AUX Agent Considered Idle (MIA)**
- **Aux Work Reason Code Type**
- **Forced Agent Logout Time**
- **Logout Reason Code Type**
- **MIA Across Skills**

ACD field setting considerations

- The ACD field settings on the Agent Login ID screen take precedence over the system-wide settings on the Feature-Related System Parameters screen.
- If any field value is set to `system` on the Agent Login ID screen, the system-wide setting is applied.
- The `system` field value is the default setting for both new and upgraded systems.
- The settings on the Feature-Related System Parameters or the Agent Login ID screen override the default setting and apply to both ACD and DA calls.
- The agent must log out and log in again for the changes to take effect.

Add/Remove Skills

Expert Agent Selection (EAS) agents or supervisors can use Add/Remove Skills to add a new skill or remove an assigned skill.

A skill is a numeric identifier in Communication Manager that refers to agent ability. For example, you can assign a Spanish-speaking skill with an identifier number 20 to an agent who speaks Spanish. The agent adds skill 20 to the set of working and already administered skills. If a customer needs a Spanish-speaking agent, Communication Manager routes the call to the agent. Skills are typically expressed as 4–digit values.

Agents or supervisors use Feature Access Codes (FACs) to add or remove skills. When supervisors change skills, agents receive notifications through an administered Alert Agent/ Change (**alrt-agchg**) lamp on the phone.

Agents and supervisors can use the following indicators to determine whether to add or remove skills:

- Queue-Status Indications
- Basic Call Management System Reporting Desktop (BCMSRD) VuStats
- BCMS or Call Management System (CMS) information

For addition of skills, agents and supervisors must type the skill level or the reserve level if Business Advocate is active.

Add/Remove Skills considerations

An agent or a supervisor cannot add or remove a skill if:

- A direct agent call is in a queue for the skill.
- A move is pending from Call Management System (CMS) or Communication Manager.
- The agent does not log in to the system.
- The agent is on an ACD call or in the After Call Work (ACW) mode for the skill.
- The agent login ID does not have a Class of Restriction (COR) number that permits change of skills.
- The skill number is invalid. For example, the skill number is Auto-Available Split or Skill (AAS) and the agent login ID is non-AAS.
- The supervisor does not have a COR number that permits change of skills and a Class of Service (COS) number with console permissions.
- The Tenant Number (TN) of the skill does not match the TN of the agent login ID.

Add/Remove Skills interactions

Interaction	Description
Auto-Available Skills (AAS)	To grant permission to an agent to add a skill that is administered as Auto-Available, use the **AAS** field on the Agent Login ID screen.
Basic Call Management System (BCMS)	BCMS tracks new skills as soon as an agent adds the skills. When an agent removes a skill, the real-time agent information specific to the skill is removed from the real-time reports, but the skill still appears on the historical reports.
Expert Agent Selection-Preference Handling Distribution (EAS-PHD)	Use the **EAS-PHD** field so that agents cannot remove the assigned direct agent skill. In an EAS environment, agents must have at least one assigned skill during a login session. With EAS-PHD, agents can add up to 120 skills. Without EAS-PHD, agents can add up to four skills.
VuStats	BCMS interactions are applicable to VuStats as BCMS gathers the VuStats display information.

Agent/Caller Disconnect Tones

The call center policy requires an agent or a call center resource to stay on a call until the caller disconnects or Communication Manager redirects the call to an automated entity such as an after-call survey. You can administer **Agent/Caller Disconnect Tones** on the Feature-Related System Parameters screen to play one of the two disconnect tones. With the tone, the supervisor can identify if the agent disconnected from an active incoming ACD or Direct Agent (DA) call first. A supervisor can use the tones to alert an agent if the caller disconnects or to monitor the call as a service observer during agent training.

The remaining parties hear the following disconnect tones on an incoming, internal, or external ACD or DA call:

- Caller: Three beeps when the caller ends the call.
- Agent: Two beeps when the agent or other call center party ends the call.

The remaining party that hear the disconnect tone include agents, Auto Available Split/Skill (AAS) agents, service observers such as call recorders, and any additional conferees other than the caller depending on who disconnects first. If the caller is the only remaining party on the call, the caller hears no tone.

Communication Manager plays the caller disconnect tone if the caller disconnects first while an agent, a station, or other active party is still on the call. The remaining parties on the call hear the tone. Service observers on the call hear the caller disconnect tone along with the active party.

Communication Manager plays the agent disconnect tone if the caller is the last active party on the call and a minimum of one observer, for example service observer or call recorder, is still on the call. The observers hear the tone but the caller hears no tone.

Note:

If an incoming trunk is measured, CMS and IQ offer reports that indicate which party hung up first. If the incoming trunk is not measured, CMS and IQ cannot determine which party hung up first. This is true regardless of whether these tones are used. Alternatively, you can compare call recording with reports to track discrepancy. To guarantee consistency, always measure the trunks.

Agent/Caller Disconnect Tones scenarios

The following table covers the operation of relevant scenarios for the disconnect tones.

Party leaving call	Remaining parties	Communication Manager action
Agent	Caller	Plays no tone and ends the connection
	Caller and more than one observer	Plays the agent disconnect tone for the observers and ends the connection
	Caller and a transferred-to party	Plays no tone, but the call remains active

Table continues…

Party leaving call	Remaining parties	Communication Manager action
	Caller and more than one conferee	
Call center party or trunk connection	The caller is last party after a conference or transfer	Plays no tone and ends the connection
	The caller is last party after a conference or transfer and there are observers	Plays the agent disconnect tone for the observers and ends the connection
Caller	Agent	Plays the caller disconnect tone for the observers and ends the connection
	Agent and more than one observer	
	Agent and more than one conferee	Plays the caller disconnect tone for the agent and the conferees, but the call remains active
	• Transferred to single party connection • Transferred to single party connection with observers	Plays the caller disconnect tone and ends the connection
	• Conference connection • Conference connection with observers	Plays the caller disconnect tone, but the call remains active
Remaining parties: Up to two observers can monitor an agent or station. Once the observed agent or station disconnects, Communication Manager removes the observers from the call connection. More than one observer can monitor the call using VDN Observing. VDN observers remain on the call until Communication Manager drops the connection.		
Action: If VDN Return Destination (VRD) applies to the call, Communication Manager does not drop the connection after the agent or call center party releases the call.		

Country-specific disconnect tone definition

Tone	Country	Definition
Agent Disconnect	US	• 440 hz at -17 db for 200 msec • Silence for 200 msec • Repeat once
	Australia	• 525 hz at -11 db for 100 msec • Silence for 100 msec • Repeat once
	Italy	• 425 hz at -4 db for 100 msec • Silence for 100 msec • Repeat once

Table continues…

Tone	Country	Definition
	Hong Kong	• 440 hz at -13 db for 200 msec • Silence for 200 msec • Repeat once
	Belgium	425 hz at -11 db for 200 msec
	France	440 hz at -11 db for 300 msec
Caller Disconnect	All countries regardless of the administered country	• 404 hz at -16 db for 100 msec • Silence for 50 msec • Repeat twice

Agent/Caller Disconnect Tones interactions

- Agent/Caller Disconnect Tones apply after the VDN Return Destination (VRD) occurs.
- Agent/Caller Disconnect Tones apply to No Hold Conference or ASAI Single Step Conference (SSC).
- Any tones applied to the connection, such as Service Observing Warning Tone or Conference Drop Tone, applies and takes precedence if there is a conflict in applying one tone versus a disconnect tone.
- Agent/Caller Disconnect Tones does not apply to an outgoing call, OCM-ACD or otherwise, placed by or for the agent.
- The application of Agent/Caller Disconnect Tones does not apply to shuffled calls. Calls preserving failover do not hear the tones.
- The caller disconnect tone applies when the caller disconnect is received while the caller is on hold.
- When the agent and caller drop the call, the tone is played for the first party. However, the drop of the second party can immediately cancel or truncate the playing of the disconnect tone, resulting in a three burst tone sound similar to single or double burst tone.

Agent Call Handling

With Agent Call Handling, you can administer all functions that ACD agents use to answer and process ACD calls.

You can define the following agent capabilities:

- Agent login and logout
- Agent answering options: Automatic zip tone answer or manual answer
- ACD work modes: Auxiliary Work (AUX), Auto-In, Manual-In, or After Call Work (ACW)

- Timed ACW (TACW)
- Stroke/Event Counts
- Call Work Codes (CWCs)
- Forced entry of Stroke/Event Counts or Call Work Codes (CWCs)
- Agent request for supervisor assistance
- Block accidental hang-up by logged-in auto-answer agent
- Option to not apply ringback to the caller for auto-answer calls
- ACD call disconnect using the **release** button

Note:

All the agent capabilities mentioned in the list are also supported through the CallVisor ASAI.

Agent work modes

An agent can be in one of following work modes:

- After Call Work (ACW)
- Auto-In
- Auxiliary (AUX) work
- Manual-In

An agent can change the work mode anytime. To enter a work mode, an agent presses a work button or types a Feature Access Code (FAC). You can determine how an agent enters a work mode.

If an agent has no active calls or held calls, the work mode button lights steadily and Communication Manager notifies CMS or BCMS about the change in agent work mode. If the agent has active calls or held calls, the lamp flashes until all calls are dropped. The new work mode button lights steadily and Communication Manager notifies CMS or BCMS about the change in agent work mode.

Communication Manager cancels requests from an agent to change the work mode and plays an intercept tone if the agent:

- Tries to enter a work mode for an invalid split or skill.
- Tries to enter a work mode for a split or skill of which the agent is not a member.
- Types an invalid FAC.

ACW mode

An agent must enter the After Call Work (ACW) mode to perform ACD-related activities, such as filling a form related to the ACD call. The agent is unavailable for ACD calls to all splits or skills when the agent is in the ACW mode. You can use Communication Manager to decide whether the agent remains in the Most Idle Agent (MIA) queue while in the ACW mode.

When an agent is in the Manual-In work mode and the agent presses **release** to disconnect an ACD call, the agent enters the ACW mode. Although the agent is unavailable for ACD calls, the

agent can receive non-ACD calls. Communication Manager notifies CMS, Avaya IQ or Basic Call Management System (BCMS) when an agent, in the ACW mode, receives a non-ACD call or makes a call.

Timed After Call Work

With Timed ACW administered either on the active VDN for the call (takes precedence) or the hunt group (skill) that the call queues to, an auto-in agent is immediately placed in the ACW state for the specified length of time after completing the currently-active ACD or Direct Agent Call (DAC) call. When the Timed ACW Interval expires, the agent automatically returns to the available auto-in work mode. If the agent manually activates ACW while not on a call, the agent is placed in ACW (not timed) state regardless of whether the agent is in auto-in or manual-in mode.

Timed ACW applies to an agent when handling an ACD/DAC call that has the feature enabled for the active VDN for the call. The active VDN for the call is based on the VDN Override rules. When the call is placed in queue for a split/skill which also has a Timed ACW interval assigned on the hunt group screen, the interval assigned to the active VDN is applied when the agent is put into ACW.

When the Timed ACW Interval is administered, if the caller drops while on hold or the agent transfers the call, an auto-in agent is immediately made available instead of being placed in timed ACW. For these cases, the Timed ACW After Xfer or Held Call Drops option can be used to place the agent in the Timed ACW mode (for the assigned interval) instead of available. For more information, see About Timed ACW after transfer or held call drops option.

Use Timed ACW to allow agents to rest between incoming ACD calls, or to pace agents when agents have to complete work from the previous call within an allotted time.

Timed ACW after transfer or held call drops option

You can place an auto-in agent handling an incoming ACD or DAC call into the Timed After Call Work (ACW) mode instead of immediately making the agent available if the caller drops from a held call or the agent transfers the call. You can enable this option for agents in a hunt group or for calls delivered from a VDN by setting the **After Xfer or Held Call Drops** field to `y` on the same line as the **Timed ACW Interval (sec)** field when **Timed ACW** is set to `y` (this field is set to a non-zero value) either on the Hunt Group or the VDN screen.

Communication Manager puts an agent in the Timed ACW mode if the agent places an ACD or DAC call on hold, or holds a conference that includes an ACD or DAC call, and the caller disconnects the call. Timed ACW is specified in the associated **Timed ACW Interval (sec)** field assigned on the VDN or Hunt Group screen if you enable the **After Xfer or Held Call Drops** field. As with the basic Timed ACW feature, the setting on the active VDN for the call takes precedence over the setting on the hunt group. This feature also places an auto-in agent into the Timed ACW mode after the agent transfers an incoming ACD or DAC call.

The following table explains the call treatment based on the Timed ACW after transfer or held call drops feature for an ACD or DAC call.

Agent work state	Scenario	Agent work state if the field is set to `n`	Agent work state if the field is set to `y`
Auto-In	Agent or caller drops the call.	Timed ACW	Timed ACW
	Caller on hold drops the call.	Available	Timed ACW
	Agent transfers a call and drops the call.	Available	Timed ACW
	Agent initiates a consultative transfer and the caller on hold drops the call.	Available	Timed ACW
	Caller drops while the call is ringing or the initial VDN of Origin Announcement (VOA) is playing.	Available	Available
	Call drops after the agent selects the Pending ACW mode.	ACW	ACW
	In a Multiple Call Handling (MCH) condition, an ACD or DAC call is on hold, another ACD call is active and a Timed ACW interval setting applies based on the second call (whether Timed ACW applies in this scenario is based on the settings for the last active ACD or DAC call), the held ACD or DAC call with MCH drops and the second active ACD call drops and Timed ACW applies.	Available	Timed ACW (matches Manual-In operation)
	In an MCH condition, an ACD or DAC call is on hold, a station call is active and Timed ACW applies, the held ACD or DAC call with MCH drops and then an active station call drops.	Available	Timed ACW (matches Manual-In operation)

TACW After Held Call Drops/Call Transferred Option Interactions

1. If the agent places an ACD/DAC call on hold while initiating a consultative conference or transfer and the caller disconnects, TACW is pending while the agent is active making a call or talking to another party, then is activated after the active call drops.
2. When application of Timed ACW is based on the VDN for the call and VDN Override is set, the setting of the After Xfer or Held Call Drops option is taken from the active VDN for the call on call delivery along with the timed interval from the associated **Timed ACW Interval (sec)** field.
3. Timed ACW as well as the option can apply to VDN calls to Auto Available Split/Skill (AAS) stations or AAS stations in a hunt group with a Timed ACW interval set on the hunt group screen. The timed interval provides a period when a call will not be delivered to the AAS station port and the station port will then be automatically made available after the timed period expires.

Canceling Timed ACW

Timed ACW is canceled under the following conditions:

Agent activates Auto-In or Manual-In mode

When an agent activates Auto-In or Manual-In mode during Timed ACW, the agent becomes available and timed ACW is canceled. An agent can change to Manual-In mode before or during a call. The system cancels Timed ACW and applies ACW (not timed) mode when the call is released. The agent remains in ACW until the agent requests another mode. When the agent releases an ACD call, the ACW lamp (if provided) lights. At the end of the administered Timed ACW interval, the ACW lamp goes dark and the Auto-In lamp lights.

Agent manually activates ACW

Timed ACW is canceled when an agent presses the ACW button or dials the ACW FAC.

Agent activates Auxiliary Work mode

If an agent activates Auxiliary Work mode during Timed ACW, the agent is placed in that mode and Timed ACW is canceled.

Ringing or held ACD call is dropped by the caller

If the Timed ACW after held call drops option is not enabled and a ringing or held ACD call to an auto-in agent is dropped by the caller, the agent is immediately made available in the auto-in mode. However if the Timed ACW after held call drops option is enabled and the caller disconnects while on hold, Timed ACW applies. The agent will still be made available if a ringing call is disconnected. For more information, see About Timed ACW after transfer or held call drops option.

Note:

To prevent agents from canceling Timed ACW by pressing the Manual-In or ACW buttons, do not assign these buttons to the agents' phones. Timed ACW cannot be assigned to AAS, adjunct-controlled, messaging system, Remote AUDIX, or Message Center splits/skills. In addition, VDN-Timed ACW does not apply to calls routed to a converse split or skill by way of the **converse-on** vector command. Timed ACW assigned to a converse hunt group applies.

BCMS and CMS track Timed ACW as standard ACW work states. Time spent in Timed ACW is not specifically identified.

Auto-In mode

In the Auto-In work mode, an agent automatically becomes available to receive new ACD calls after the agent disconnects from the current ACD call.

AUX work mode

An agent must enter the Auxiliary (AUX) work mode when taking a temporary break. This makes the agent unavailable for ACD calls and removes the agent from the Most-Idle-Agent (MIA) queue. CMS and BCMS can continue to track the staffed agent.

In a non-EAS environment, when an agent is in the AUX work mode for a particular split, the agent is available for ACD calls to other splits that the agent is logged into, depending on the state of the agent in the splits. Even in the AUX work mode, the agent is available for non ACD calls. CMS or

BCMS is notified whenever an agent in the AUX work mode receives an incoming non ACD call or makes an outgoing call. When an agent logs in to a split, the agent automatically enters the AUX work mode for the split.

Note:

Agents in vector-controlled splits or skills can go into the AUX work mode even if the agents are the last agent and calls are queued to the split or skill.

Although an agent in the AUX work mode is unavailable to receive ACD calls, such an agent can be available to receive ACD calls if the agent:

- Has moved to AUX with an interruptible reason code.
- Is logged in to more than one skill administered as interruptible.
- Is logged in to more than one skill that has exceeded an administered service level threshold.

The agent is either automatically forced to an available state or requested to become available, depending on how you administer the skills. After changing state, agents are available to receive calls from any skill assigned to the agents, not just the one that caused the interruption from AUX. Applicable agent selection criteria are used once the agent is available.

Manual-In mode

In Manual-In mode, the agent automatically enters ACW mode for the split or skill upon disconnecting from an ACD call and is not available for any ACD calls. To become available for ACD calls, the agent must manually reenter either auto-in mode or manual-in mode.

Agent answering options

An agent can answer ACD calls by using a headset, handset, or speakerphone. You can determine which calls to agents will be automatic or manual answer calls.

Agent capacity and related limits

Agent sizing adds an overriding capacity limit to the number of logged-in ACD agents. You can use agent sizing to limit the number of logged-in ACD agents to a number less than or equal to the maximum supported by the system configuration.

The logged-in ACD agent limit applies to ACD agents in non Expert Agent Selection (EAS) ACD splits or in EAS skills. Auto-Available Split/Skill (AAS) agent ports are counted when first assigned, while the non-AAS agents are counted when the agents actually log in. Each logged-in agent is counted as a single agent independent of the number of splits or skills logged in to for the logged-in ACD agents limit. AAS and non-AAS agents are counted towards the limit whether the agents are Basic Call Management System (BCMS)/Call Management System measured or not.

Administrators must consider EC500–mapped mobile agents while deciding about agent sizing. Mobile agents might take calls beyond specific work hours since they are mobile and off-premise. In this case, the on-premise users cannot login to the station extension being used by mobile agents. Sharing of station extensions between multiple agents would lead to situations where the same ACD call simultaneously rings the cellular phones of multiple agents. Therefore, a station

extension cannot be shared between multiple agents and additional station extension licenses are required.

Avaya personnel administer the agent sizing license limit. The **Logged-In ACD Agents** and the **Logged-In Advocate Agents** fields on the System-Parameters Customer-Options screen are set by the loaded license file. The maximum number of logged-in ACD and Business Advocate agents is set to match the configuration that you purchase.

For more information, see the *Using Avaya Business Advocate* document.

The Logged-in Advocate Agent Count feature

The Logged-in Advocate Agent Count feature counts the number of Avaya Business Advocate agents who are logged in at the call center. The feature bases the count on whether or not a logged-in agent has any Avaya Business Advocate features, except Predicted Wait Time, assigned or associated with the agent. With this feature, Advocate-counted agents are still counted as ACD agents.

Agent sizing when agents work in shifts

For agent sizing, if you have agents working in shifts, you must purchase for the agent capacity to allow for a smooth shift change. If agents on a subsequent shift are logging in before agents in the previous shift have logged out, agents login can be denied because too many agents are currently logged in. Additionally, you must also pay attention to the non-ACD or non-agent (AAS/VRU) use of hunt group resources. Call center managers must be aware of the logged-in ACD agent and other related limits when adding agents to handle a traffic peak or when planning a special campaign. Some resource utilization is displayed dynamically on the Display Capacities screen.

Limit considerations

In addition to the logged-in ACD agents limit, the number of agents is dependent on the upper limits that the system platform supports. Following are the other limit considerations:

- Maximum hunt group members
  - Non-ACD members include hunt groups with or without queues, message center service groups, messaging-system groups, and remote messaging-system groups. Each line or port in a group is counted once when assigned.
  - ACD members, also called agent-split pairs or agent-skill pairs with EAS. For agents in multiple splits or skills, each combination is counted as a member. For example, an EAS agent logged into four skills or a non-EAS agent assigned to four splits counts as four members. Non-EAS ACD members are counted when assigned. Note that many splits can be assigned to an agent than can be logged in but each agent-split pair is still counted towards the limit. EAS ACD members are counted when the agents log in.
  - Avaya Business Advocate Agents - Each logged-in Avaya Business Advocate agent is counted as both an ACD member and as an Avaya Business Advocate agent.
- Hunt group members per group: Count of non-ACD or ACD members within a split or skill. Counting is similar to that in maximum hunt group members.

- Additional traditional ACD, non-EAS agents limits:
  - Maximum logged-in agents system limit.
  - Maximum splits an agent can log in to.
- Additional EAS limits:
  - ACD members administered: Limits skill assignments to agents. Each AAS port is counted as one skill pair.
  - Agent login IDs administered: Limits number of AAS ports and EAS agents that can be preassigned.
  - Agent login IDs logged-in system limit: Upper limit on the number of EAS agents and AAS ports that can be logged-in simultaneously.
  - Skills per agent: The maximum number of skills a particular agent can be assigned.
- CMS logged-in ACD members, that is, agent-split or skill pairs limit is assigned. Both an Avaya setup and a customer-administered limit are assigned in CMS. The limits are related to the CMS memory or hardware configuration equipped and are passed over the link to the Communication Manager to reduce the externally measured logged-in ACD member component of the hunt group member limit to that supported by CMS.
- BCMS internally measured ACD agents system limit. Non-EAS ACD agents counted when assigned while EAS agents are counted when logged in.

When the maximum number of ACD agents are logged in or any of the other above limits are reached, an agent who attempts to log in hears a reorder tone or is otherwise denied log in. Also with EAS, an agent logging in does not have all the assigned skills logged in if the ACD member limit is reached.

The administrator of a non-EAS system can be blocked from adding agents to splits using the Hunt Group screen.

The administrator of an EAS system can be blocked from assigning additional login IDs or skills to an agent using the Login ID screen if the relevant system limits are reached.

Agent login and logout

Agents must log in to the system to receive Automatic Call Distribution (ACD) calls. Communication Manager treats logged-in agents as staffed agents.

If you use Basic Call Management System (BCMS) or Call Management System (CMS) to track skills, agents must type a login ID.

Agent request for supervisor assistance with or without an active ACD call

To request for supervisor assistance, an agent presses the **assist** button or puts the call on hold and dials the assist FAC and the split or skill number. The agent must be logged in to the split or skill. Assist generates 3-burst ringing at the supervisor station. If a split or skill supervisor is not assigned, the agent hears an intercept tone.

Attendants must press the **start** button before pressing the **assist** button. The attendants can then transfer the call. The action causes the call to ring as a priority call at the supervisor set.

When the agent presses the **assist** button:

1. Communication Manager places the ACD call on hold and connects the agent to the split or skill supervisor. If the agent is not active on an ACD call, Communication Manager connects the agent to the supervisor.
2. Communication Manager notifies CMS or BCMS about the agent request. The supervisor phone display indicates that the call is a request for assistance. The call to the supervisor rings as a priority call with 3–burst ringing.
3. The caller hears nothing or music on hold.
4. After speaking with the supervisor, the agent can drop the assist call and return to the ACD call, set up a conference call with the supervisor and the calling party, or transfer the call to the supervisor.

When the agent puts the call on hold and dials the assist FAC and the split or skill number, Communication Manager handles the request as if the agent pressed the **assist** button, except that the assist call does not follow the coverage path administered for the supervisor phone.

Automatic Answer

An agent assigned to Automatic Answer hears a zip tone and connects directly to incoming calls.

Note:

You can administer Automatic Answer to apply only to ACD calls or to all calls terminating at the agent phone. If all calls are Automatic Answer and the agent receives direct-extension calls, the agent must always activate Call Forwarding or Send All Calls when leaving temporarily or for an extended period to ensure that calls do not terminate to an unstaffed station.

Block hang-up by logged-in auto-answer agents

You can set the **Block Hang-up by Logged-In Auto Answer Agents** field to `y` on page 12 of the Feature-Related System Parameters screen. The field setting to `y` prevents an agent in the auto-answer mode from accidently logging out or dropping an active call. The default field setting is `n`.

The enhancement is available for agents using H.323 IP and DCP phones, but does not apply and is not required for agents using BRI, analog, CallMaster®, or SIP agent deskphones.

With the default field setting, the following events occur if an auto-answer agent goes on-hook when on an ACD call:

- Communication Manager drops the active call and logs the agent out.
- The reporting adjunct fails to record the call and to track the logged-in agent.

With the **Block Hang-up by Logged-In Auto Answer Agents** field set to y, an agent cannot use the speakerphone, headset, or handset buttons to drop calls. The agent can perform the following functions to drop calls:

- Press **release** to disconnect the call and stay logged-in.
- Press **drop** to release the last connected party.
- Select another call appearance and enter the logout Feature Access Code (FAC). The action drops calls if the **auto-hold** field on page 6 of the Feature-Related System Parameters screen is set to n.
- Unplug the phone if you are using an H.323 phone.

If the auto-answer agent disconnects the handset or headset, the appearance is *on-hook* with no voice path. The caller from the previous call does not hear anything and eventually ends the call. If the **Block Hang-up by Logged-In Auto Answer Agents** field is set to y and the appearance is *on-hook*, Communication Manager maintains the agent login status, agent tracking continues, and the agent is available for another call. If a new call rings at the station, the agent must go off-hook, that is, connect the handset or headset to answer the call. If the agent does not hang-up the appearance before completion of the next call that rang-in, the next call is delivered auto-answer resuming the normal auto-answer operation.

With RONA administered for the skill, Communication Manager redirects the call if the agent fails to answer the call within the administered number of rings.

Note:

The **Block Hang-up by Logged-In Auto Answer Agents** field is applicable if you set **EAS** to y.

Caller information

Most call centers require agents to view call-related data when agents are in the After Call Work (ACW) mode. Agents must be able to view and use data without copying data during the call.

The caller information (callr-info) feature meets the U.S. government privacy requirements as specified in Health Insurance Portability and Accountability Act (HIPAA). HIPAA specifies that medical records must be inaccessible to unauthorized personnel.

When Communication Manager delivers a call with the associated caller information to the agent phone, the **callr-info** status lamp on the phone flashes till the agent presses **callr-info**. If the agent presses **callr-info**, Communication Manager sends the call-related information to the phone display. The status lamp then becomes steady.

The **callr-info** status lamp flashes on all phones regardless of the phone type, that is, the lamp flashes on one-line and two-line display phones.

For one-line display phones, you must select **y** in the **Enhanced Callr-Info display for 1-line display phones** field on the Station screen to use the **Clear callr-info** field options that are available on the Feature-Related System Parameters screen.

If you select **leave-ACW** or **next-call** in the **Clear callr-info** field, and if the agent releases the call, Communication Manager does not clear the call-related information regardless of the information on the phone display. For instance, if an agent, while on a call, does not press **callr-**

info to view the associated information, Communication Manager replaces the existing information on the phone display with callr-info when the agent releases the call.

When an agent using multiline display phone releases a call, Communication Manager moves callr-info to the top line of the display instead of the line appearance display.

Important:

After the agent releases the call, if the agent presses **vu-display** or **cwc**, or if the agent makes a call, Communication Manager clears the call-related information. The agent can no longer view the information even if the agent presses **callr-info**.

You can administer the **Callr-info Display Timer (sec)** field on the Feature-Related System Parameters screen to clear the call-related information on softphones, H.323 phones, and one-line display phones for which the **Enhanced Callr-Info display for 1-line display phones** field on the Station screen is administered as **n**.

If you administer the **Enhanced Callr-Info display for 1-line display phones** field as **y**, Communication Manager ignores the timer and displays the caller information till the agent presses **callr-info** again. Communication Manager then clears the caller information and blinks the status lamp. The agent can toggle **callr-info** to view the call-related information till the call is active.

Enhanced callr-info interactions

Interaction	Description
96X1 H.323 phones	The enhanced callr-info feature is not applicable to the 96X1 H.323 phones as Communication Manager automatically displays the caller information on the agent phone to support the Agent Greetings feature. These phones use the administered **Callr-info Display Timer (sec)** field to determine the display duration of the associated caller information.
96X1 SIP phones	The enhanced callr-info feature is not applicable to the 96X1 SIP agent phones as these phones support the display of callr-info between calls.
After Call Work (ACW) and Timed ACW	When you administer **Clear callr-info** on the Feature-Related System Parameters screen to `leave-ACW`, Communication Manager clears the caller information after the agent work state changes from ACW to Available or AUX or if the agent logs out of the system.

Table continues…

Interaction	Description
Call Work Code (CWC)	During a call, an agent can press **callr-info** to view the associated caller information. If an agent presses **cwc** to activate a CWC session, Communication Manager displaces the caller information, but the agent can press **callr-info** again. However, if the agent presses **cwc** while in ACW, Communication Manager clears the associated caller information and the agent can no longer view the caller information. If Communication Manager does not activate the CWC session when the agent presses **cwc**, the associated caller information remains on the phone display.
Hold, Conference, and Transfer	If the agent puts an active call on hold or adds a conference party, the lamp associated with **callr-info** does not flash. When the call is active again, the lamp flashes to indicate the presence of the associated caller information. The agent can then press **callr-info** to view the information. If an agent transfers the call, Communication Manager clears the caller information on the agent phone display and turns off the lamp.
IP Agent or One-X Agent	Softphones, such as IP Agent or One-X Agent, use their internal application interface to determine the display duration of the associated caller information after the agent presses **callr-info**. The enhanced callr-info feature, therefore, is not applicable to softphones.
Logout	If the agent logs out or leaves the ACW mode, Communication Manager clears the associated caller information. The agent can no longer view the information by pressing **callr-info**.
Multiple Call Handling (MCH)	The interaction with MCH is similar to the interaction when an agent puts a call on hold.
Non-ACD calls	If an agent releases an ACD call and makes or receives a non-ACD call, Communication Manager clears the associated caller information. The agent can no longer view the information by pressing **callr-info**.
User-to-User Information (UUI)	If an agent presses **callr-info** when the phone displays UUI, the information that is associated with the call replaces UUI. To view UUI, the agent must press **uui-info** while the agent is still on the call.
VuStats	If an agent presses **callr-info** when a VuStats session is active, Communication Manager sends the associated caller information to the phone display and ends the VuStats session. The agent can press **vu-display** anytime during the call to activate VuStats. This displaces the caller information. While the call is still active, the agent can press **callr-info** to view the caller information. If the agent ends the call and presses **vu-display**, the agent can no longer view the caller information.

Call Work Codes

With Call Work Codes (CWCs), agents with multi-appearance phones, such as Callmaster™, can type up to 16 digits to record customer-related information. Up to 100 agents can use CWCs simultaneously.

Note:

Third-party call control support APIs do not require Feature button configuration on the agents phone.

To use CWCs, an agent must be:

- On an ACD call.
- A manual-in agent in the After Call Work (ACW) mode.
- An auto-in agent in the Pending for ACW or Timed ACW mode.

The sequence of events is as follows:

1. The agent presses **work-code**.
2. Communication Manager lights the CWC lamp steadily and displays a C: prompt on the phone display. The agent waits for a ready indication or the caller hears the touchtone digits.
3. The agent types up to 16 digits on the dial pad. The agent can press an asterisk (*****) sign to delete the digits.
4. The agent presses the pound (**#**) sign to record the information.
5. Communication Manager sends the CWC to Call Management System (CMS), turns off the lamp indicator, and returns the phone display to normal.
6. If the agent presses a button or releases the call during digit collection, Communication Manager cancels the CWC entry, turns off the lamp indicator, and clears the phone display.

Forced entry of stroke counts and call work codes

You can administer a split or skill so that agents enter Stroke Counts (SCs) or Call Work Codes (CWCs) before agents become available to receive other calls.

Note:

Agents must have multiappearance deskphones or attendant consoles to enter SCs or CWCs.

To enter an SC or a CWC, an agent must be:

- On an ACD call.
- In the After Call Work (ACW) mode after the agent, if in the manual-in mode, releases the call.
- In the Timed ACW (TACW) mode after the agent, if in the auto-in mode, presses **release**.

After the agent releases a call, the agent automatically enters the ACW mode for manual-in or the TACW mode for auto-in. The agent must enter an SC or a CWC before the agent returns to the manual-in mode or available mode with TACW.

Communication Manager flutters the lamp or plays an intercept tone if the agent does not enter an SC or CWC, but does one of the following:

- For manual-in, presses **manual-in** to become available when in the ACW mode.
- For auto-in, presses **auto-in** to become available when in the TACW mode.
- Uses Feature Access Codes (FACs) to become available.

After the agent enters an SC or a CWC and presses **manual-in**, **auto-in**, or types an FAC, the agent returns to the manual-in mode or the available mode.

You can assign a forced entry of SCs or CWCs to any of the agent splits or skills. If the agent work mode changes to Auxiliary (AUX) work mode for any split or skill, Communication Manager removes the forced entry of SCs or CWCs for all other splits or skills.

Login to a skill

To log in, an agent goes off-hook and dials the system assigned login FAC, followed by the assigned login ID and password, if required. If login is successful, the agent automatically enters the Auxiliary Work mode for all the skills assigned to the agent and the assigned skills are displayed on the station set. The **aux-work** button lamp on the station set lights steadily and the agent hears the confirmation tone.

Canceled logins

Login is canceled and the agent receives an intercept tone if any of the following occur during login:

- The agent dials an invalid login FAC.
- With non-EAS the agent:
  - Dials an invalid split number.
  - Dials the number of an unassigned split.
  - Dials the number of a split the agent is logged in to.
  - Is logged in to the maximum number of splits (4).
  - Dials an invalid or unassigned BCMS/VuStats login ID.
- With EAS the agent dials an invalid agent login ID or password.

Login is canceled and the agent receives a reorder tone if the maximum number of agents are already logged in.

An EAS agent can be denied login to some of the assigned skills if the maximum number of agent-skill pairs is reached. The display screen indicates an “*” for each skill not logged in.

Logout

An agent must log out when the agent leaves for an extended period or is unavailable to receive ACD calls. If Call Management System or BCMS measures the split or skill and an agent logs

out, Call Management System or BCMS receive a message to not measure the split or skill as the agent has logged out. In a non-EAS environment, if an agent is logged in to multiple splits, the agent must log out of each split.

When temporarily unavailable for calls, an agent must use the AUX work mode, instead of logging out. Call Management System or BCMS can continue to track the auxiliary work time of the agent.

To log out of a split, an agent goes off-hook and dials the logout FAC followed by the split number. To log out of a skill, the agent dials the logout FAC and is automatically logged out of all the assigned skills. If the logout is successful, the agent hears confirmation tone and work mode button lamps darken. The logout is canceled and the agent receives an intercept if any of the following occur during logout:

- The agent dials an invalid logout FAC or split number.
- The agent dials a split number for a split that the agent is not logged in to.

If an agent uses a handset in the Automatic Answer mode, the agent can log out by hanging up or turning off the headset, which does not mean pressing the **release** button on a Callmaster® phone. This does not apply to quick-disconnect. If the agent pulls the handset to log out, the agent is automatically logged out of all splits that the agent has logged in to.

Manual Answer

An agent assigned to Manual Answer hears ringing, and then goes off-hook to answer the incoming call.

Option to not apply ringback to callers

You can set the **Apply Ringback for Auto Answer Calls** field on the VDN screen to `n` to prevent returning short burst ringback to callers. Setting the field to `n` is useful when auto-answer delivery to the agent is delayed by timing for ROIF.

Ringer-off control of auto-answer for non-ACD calls

Agents can use the **ringer-off** button to prevent non-ACD auto-answer calls from ringing.

Previously, agents besides hearing an answer tone in their headsets also heard an audible single ring for non-ACD auto-answer calls on their station set with any of the following settings:

- The **Auto-Answer** field on the Agent LoginID screen is set to `all`
- The **Auto-Answer** field on the Agent LoginID screen is set to `station` and the **Auto-Answer** field on the Station screen is set to `all`

Some call centers prefer quiet environments and do not want to hear the audible ring for every non-ACD call.

Administration: Set the **Allow Ringer-off with Auto-Answer** field on the Feature-Related System Parameters screen to `y`.

Stroke Counts

With Stroke Counts (SCs), agents can press a **stroke-cnt** feature button to record the number of occurrences of a customer-related event.

You must have ACD splits or skills to use SCs with Call Management System (CMS).

You can assign 10 SC button types with SC zero (0) for tracking audio difficulty or poor quality of transmission. Agents can record up to nine events for each call. For troubleshooting purpose, CMS records the equipment location of the trunk that the agent used when the agent pressed **stroke-cnt** for audio difficulty.

If an agent is on an ACD call or in the After Call Work (ACW) mode for an ACD call and the agent presses **stroke-cnt**, Communication Manager lights the feature lamp steadily for two seconds and sends the SC to CMS. If an agent is not on an ACD call or in the ACW mode, Communication Manager causes the feature lamp on the phone to flutter but sends no message to CMS.

Note:

Third-party call control support APIs do not require Feature button configuration on the agents phone.

Agent Call Handling considerations

Agents assigned to hunt group and ACD calls

- Do not use agents for hunt group calls and ACD calls simultaneously. Otherwise, all calls from one split or skill, either ACD or hunt group, are answered first.
- The oldest-call-waiting termination works only for agents who receive ACD calls.

Agents logged in to multiple splits or skills

If an agent is logged in to multiple splits or skills, the agent is unavailable for calls to one split or skill because of activity at another split or skill. For example, if an agent enters the ACW work mode for one split or skill, the agent becomes unavailable for calls to other splits or skills.

An agent cannot log in to a split or skill while a call is on hold at the extension.

Agents with Automatic Answer

Agents who use Automatic Answer must use a headset. The agent hears a zip tone through the headset and automatically connects to a call. If either the incoming trunk group or the agent extension is data-restricted, the agent does not hear a zip tone. Therefore, do not assign data-restriction to the extension of a agent who uses a headset .

Do not use Automatic Answer with a handset or speakerphone. The handset or speakerphone must be off-hook, that is, handset lifted or speakerphone turned on, all the time for the agent to hear zip tone.

If automatic answer is assigned for all calls, when a non-ACD call arrives, non-ACD Auto-Answer agents hear Incoming call ID tone, but no ring.

Callmaster® phones

Calls for Callmaster® digital phones and attendant stations are indicated with double tones. The tones that are doubled are zip tones for auto-answer ACD agent calls and Incoming Call Identification (ICI) for end of VDN of Origin Announcements (VOA) and all other auto-answer calls. The user hears part of the first tone and all of the second tone.

Non vector-controlled splits or skills

For non vector-controlled splits or skills, the last available agent in a split or skill cannot enter the AUX work mode if calls remain in the queue. However, the agent can log out.

When the last available agent tries to enter the AUX work mode, the following occurs:

- The **aux-work** button flashes indicating the change is pending.
- New calls on the ACD split or skill either receive a busy tone or redirect to coverage. Calls in the queue continue to route to the last available agent until the queue is empty.
- At the last available phone or console, the **aux-work** button lamp flashes until the queue is empty. The phone then enters the AUX work mode and the associated lamp lights steadily.

Release button

Agents using Automatic Answer are logged out of all splits or skills when the agents disconnect from an ACD call by hanging up or by using the **drop** button. Therefore, agents must always use the **release** button to force the release of a connection.

Agent Call Handling interactions

Interaction	Description
Abbreviated Dialing (AD)	Assign AD buttons to make agent login easier. You can program an AD button to dial an access code, split number, or agent login ID. You can use the Autodial feature buttons to assign login and logout feature buttons.
Auto Answer with Interruptible Aux	Even if an agent skill is administered to be forced interruptible, that is, Reserve Level (RL) is set to `a` or `m` on the Agent LoginID screen, the agent administered as `Auto Answer` is treated as requested interruptible, that is, RL set to `n`, not forced. Otherwise, a forced interruptible agent administered as `Auto Answer` is made available using the Interruptible Aux feature even if the agent is not physically present or ignores the call. In such a situation, a call is delivered and automatically answered by the endpoint but no one answers the caller.

Table continues…

Interaction	Description
Bridging	ACD split or skill calls are not bridged. If an agent bridges on to a call, the call is a non-ACD EXT-IN call. The agent is not available for an ACD call unless the agent is a member of a many-forced, one-forced, or one-per-skill MCH split or skill. The agent can put the call on hold and become available to receive ACD calls even in non-MCH splits or skills if only bridged appearances are active.
Call Coverage	If an ACD call routes to an agent as a result of covering to a VDN, where the VDN is the last coverage point in the coverage path, Timed ACW applies as administered for the VDN, split, or skill.
Call Forwarding	If an ACD call routes to an agent after being call forwarded to a VDN, Timed ACW applies as administered for the VDN, split, or skill.
Call Management System (CMS)	Timed ACW is reported on CMS reports in the same way as any other ACW. CMS gives exception notification only on ACW intervals that are longer than the defined threshold.
Call Pickup	When an ACD agent answers a call with Call Pickup, the call is treated as an incoming non-ACD call. The agent can put the call on hold and become available for additional calls.
CallVisor ASAI Adjunct	If a split or skill hunt group has CallVisor ASAI as the controlling adjunct, you cannot administer Timed ACW for the split or skill. Additionally, if an ACD call is routed to an agent in an adjunct-controlled split or skill, the agent is not placed in Timed ACW when the call ends.
Call Work Codes	The CWC 100-agent limit is shared with reason codes. Therefore, no more than 100 agents can simultaneously enter either a CWC or reason code.
Conference	If an agent receives an ACD call through a VDN and conferences in other agents, the agents added to the call use the Timed ACW interval associated with the number dialed to conference the other agents. An ACD agent on conference with more than three parties can cause inaccurate CMS measurements.
Expert Agent Selection (EAS)	When EAS is active, all ACD hunt groups are assigned as vector-controlled skills. Agents log in using Logical Agent IDs. Skills can be preassigned to login IDs, however, assignment on the Agent Login ID screen does not actually assign a non-AAS login ID to the skills until the ID is logged in. When the login ID is logged in, each skill is counted as a hunt group member towards the system hunt group member limit, the per-group member limit, and each agent is counted as a logged-in ACD agent.
Interruptible Aux Notification Indicators	Interruptible Aux notification indicators on an endpoint, that is, display message, flashing lamp and ring-ping alert tone, persist across call-preserving system resets, levels 1 and 2. The notification indicators keeps agents aware of critical service level conditions even when certain serious system events occur.

Table continues…

Interaction	Description
Manual Answer with Interruptible Aux	If a forced interruptible agent answer condition is administered as `Manual Answer` and is moved to an available mode, when a call is delivered to the endpoint the agent must answer the call. RONA applies to the calls delivered to the endpoint that are not answered within the set number of rings.
Multiple Call Handling (MCH)	If MCH calls are on hold at a telephone and the agent completes a call that normally is followed by Timed ACW, the agent is not placed in ACW. If no MCH calls are on hold, but one is alerting at the station when the Timed ACW call completes, the agent is placed in ACW. MCH affects when agents can enter different work modes and when calls are delivered to agents in manual-in or auto-in work modes.
Transfer	If an agent receives an ACD call through a VDN and transfers the call to another agent, the second agent uses the Timed ACW interval assigned to the number that was dialed to transfer the call. For an EAS agent, this is the Timed ACW interval associated with direct agent skill. For an agent receiving a call transferred to a second VDN, this is the VDN Timed ACW interval of the second VDN. The agent who originally transferred the call uses the ACW associated with the VDN, split, or skill that first received the call.
VDN Override	If a VDN has VDN Override set to `n` and the vector routes a call to a second VDN, the Timed ACW interval of the first VDN is used for Timed ACW. If VDN Override is set to `y`, the Timed ACW interval of the second VDN is used. If no interval is set for the second VDN, no Timed ACW is associated with the call.
Voice Response Integration (VRI)	If an ACD call routes on a **converse** vector command, any VDN Timed ACW associated with the call is ignored for agents in the converse split or skill. However, if the converse split or skill has an administered Timed ACW interval, the answering agent associated with the split or skill is placed in Timed ACW when **converse** vector command processing completes.

Agent Mobility integrates with Avaya Extension to Cellular

Agent Mobility integrates with Avaya Extension to Cellular (EC500) enabling Expert Agent Selection (EAS) agents to function outside the corporate network. Administrators can configure EC500 mapped mobile agents to log in and work when they are working remotely. With the EC500 feature, users can use a single number to make inbound and outbound calls.

For information about configuring Agent Mobility, see *Administering Avaya Aura® Call Center Elite*.

Using Feature Name Extensions (FNEs), mobile agents can log in, log out, change work-modes, and query their work-mode. Agents can also use FNEs to perform additional functions, such as

Idle Appearance Select, Conference Complete, Conference on Answer, Transfer Complete, and Transfer on Hang-Up.

Note:

You must have sufficient licensing of Off-PBX Telephones - type EC500 available.

Important:

- Do not configure EC500 mobile agents with local SIP extensions.
- Do not use mobile agents in an outbound contact center configuration or for SIPCC configured agents.

Related links

Operational differences between working on a deskphone and a mobile station on page 59
Features not currently supported with Agent Mobility on page 59

Operational differences between working on a deskphone and a mobile station

Agents working on a mobile station might notice the following operational differences from working on a deskphone:

- Communication Manager cannot detect calls outside the Communication Manager domain. Because of this, mobile agents can receive an ACD call while they are on a personal call on their mobile phone.
- Communication Manager cannot detect the state of a phone outside the Communication Manager domain. This might result in inaccurate login duration if the mobile phone goes out of the coverage area or loses battery power.

  For example, a mobile phone goes out of the service area or loses battery power at 10:30 a.m., and does not resume until 11:00 a.m. During that half an hour, 10:30 a.m. to 11:00 a.m., Call Management System (CMS) will report that the agent as available. If the agent was on a deskphone, Communication Manager would detect the loss of the station, log the agent out, and report it accordingly to CMS.
- A Public Switched Telephone Network (PSTN) does not support functionality such as displays and auto-answer. Because of this, mobile agents do not see vu-stats displays, callr-info displays, or call-prompting digits, and their mobile station does not auto answer.

Related links

Agent Mobility integrates with Avaya Extension to Cellular on page 58

Features not currently supported with Agent Mobility

Agent mobility currently does not support the following functionality or features:

- Station extension being shared by mobile agents: Sharing can cause errors and confusion when multiple mobile phones are associated with the same extension.

For example, two agents, Tom and Rick, share the same station extension x5001. Tom's agent ID is x4001, his mobile number is 303 538 1234, and he works in the morning shift. Rick's agent ID is x4005, his mobile number is 303 538 6789, and he works in the evening shift. Before enabling agent mobility for these users, Avaya recommends that the administrator allocates an additional station extension. If mobile agents share the deskphone extension and Rick logs in from his mobile station, both Tom and Rick might receive the same ACD call on their individual mobile phones. So Tom might receive Rick's ACD calls, and Tom is prevented from logging in with his agent ID while Rick is logged in.

In a contact center environment, the administrator must ensure that internal users do not use the station extension assigned to a mobile agent. Administrators can protect the extension using a station password.

- Service provider voice mail: In an exemplary scenario, Communication Manager extends an inbound contact center call to a mobile agent. If the cellular service provider has configured a voice mail for the mobile agent number, the call might go to the mobile voice mail system if:
  - The mobile phone is out of the coverage area.
  - The mobile phone is switched off, or the mobile phone's battery is discharged.
  - The mobile phone user rejects the call.

  To avoid such situations, Avaya recommends that you turn off the voice mail on the mobile agent's mobile phones. In addition, administer all skills or VDNs or both used with mobile agents with Redirect On No Answer (RONA).

  ROIF and ROOF apply to the deskphone and not to the mobile phones.

  The **Cellular Voice Mail Detection** field on the off-pbx-telephone configuration-set is used in Unified Communications (UC) deployments for cellular voice mail detection. However, using this field is not advised because it might not work with all service providers globally. Also, the feature might not work when the call goes through a PSTN that is not directly connected to Communication Manager.
- Abandoned Call Search: Communication Manager cannot detect the ACD call extended to a mobile agent using EC500 mapping in the Abandoned Call Search algorithm.
- Supervisor Assist and Stroke Count: Mobile Agents cannot access the Supervisor Assist or Stroke Count features from mobile phones. However, non-mobile users can use the supervisor assist functionality to assist the mobile agent.
- Call Work Codes: Mobile agents cannot set call work codes.
- Multiple Call Handling: This feature is not supported.
- Auto-Available Skills: This feature is not supported.
- Integration with Avaya one-X® Agent: This feature is not supported.
- Other: Applications that use ASAI/AES to generate, answer, transfer, conference, hold, or release calls perform those operations on the deskphone where the agent is logged in, instead of the mobile phone.

Related links

Agent Mobility integrates with Avaya Extension to Cellular on page 58

Auto-Available Split/Skill

If you set **Auto-Available Split/Skill (AAS)** to y, members of an ACD split or skill are in the auto-in work mode continuously. An agent in the auto-in work mode is immediately available for another ACD call after disconnecting the previous ACD call.

Use AAS to bring ACD agents back in the auto-in work mode after the system restarts. You can use this feature for splits or skills containing only non human members, for example, recorders or Voice Response Units (VRUs). To allow an IVR to receive back-to-back calls, you can assign a Timed-ACW delay of 1 to 2 seconds to AAS.

Agent login with Auto-Available Split/Skill

For splits or skills with Auto-Available Split/Skill (AAS), agents are automatically logged-in under the following circumstances:

- Call Management System (CMS) completes an agent move request to AAS.
- A maintenance-busied-out port, which is defined as an agent in AAS, is released.
- Communication Manager restarts and requires agents to login again.
- The **AAS** field on the Hunt Group screen is administered and the agent login ID is administered in AAS.

Communication Manager immediately moves an agent that is logged in to AAS to the Auto-in work mode and denies subsequent requests to change the work mode.

Agent logout with Auto-Available Split/Skill

For splits or skills with Auto-Available Split/Skill (AAS), agents are automatically logged out under the following circumstances:

- CMS completes an agent move request out of AAS.
- The auto-available agent port is unavailable due to maintenance.
- A split or skill is administered as AAS.
- An agent from an existing AAS split or skill is removed.
- Redirection on No Answer (RONA) redirects a call that the agent has not answered after an administered number of rings.

Auto-Available Split/Skill considerations

- Use Auto-Available Split/Skill (AAS) for non BX.25 and non ASAI PBX adjuncts, such as an IVR system VIS, that require extra help in getting PBX ports back online after a system restart. AUDIX uses BX.25 messages to automatically activate the ACD agent ports after a system restart and is, therefore, incompatible with AAS.
- Use AAS for non human agents and do not administer an auto-answer phone as a member of an AAS.

- Do not use AAS for any agent port hardware that can change the work mode state since Communication Manager denies a request to move to any state other than auto-in. However, administration of such a phone is not blocked.

Auto-Available Split/Skill interactions

Interaction	Description
Auto-Answer	If you administer a non-analog phone as auto-answer and an agent is logged in to a split or skill, the system logs the agent out of the system when the phone goes on-hook. If you administer an analog phone as auto-answer and an agent is logged in to a split or skill, the agent must enter a logout Feature Access Code (FAC) to log out of the system. As Communication Manager denies a logout FAC, do not administer an auto-answer phone as a member of Auto-Available Split/Skill (AAS). To log an agent out of the system: • Busy out the extension. • Remove the agent ID from the split or skill when the agent is on an ACD call. If an AAS agent with an auto-answer phone goes off-hook, Communication Manager logs the agent in another AAS of which the agent is a member. To log out of an AAS split or skill, an agent goes off-hook to be placed in the Auxiliary (AUX) work mode. The agent then presses **release** on non-analog phones or **disconnect** on analog phones. As Communication Manager does not place agents in the Auto-in work mode immediately, agents can place emergency or personal calls instead of responding to ACD calls in the queue.
Call Management System (CMS)	For each agent, AAS notifies CMS about agent login, logout, or change to the Auto-in work mode. In a non-EAS environment, CMS identifies and AAS agent with a login ID equivalent to the administered extension. In an EAS environment, the AAS login ID and port are assigned on the Agent Login ID screen. Use the CMS Move Agent request to move an agent from one AAS split or skill to another while the agent is logged in to the system.

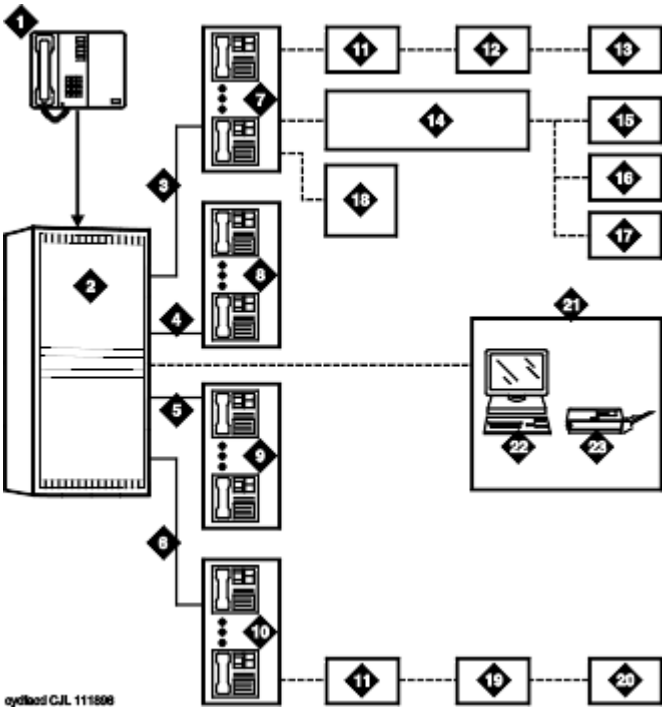
Automatic Call Distribution

With Automatic Call Distribution (ACD), Communication Manager connects incoming calls to specific splits or skills. An ACD split or skill is a hunt group that receives a high volume of similar calls. ACD hunt groups with basic ACD, but non-EAS, are called splits. ACD hunt groups with basic ACD and EAS are called skills. Communication Manager distributes calls to a specific split or skill or to hunt group members assigned to the split or skill. Calls queue to the split or skill until an agent is available.

An ACD agent can be a physical phone extension, an individual attendant extension, or an agent login ID in an EAS environment. An agent can log in to multiple splits or skills. However, in a non-EAS environment, agents can log in to only one split if that split is administered for Multiple Call Handling (MCH).

You can assign a supervisor to each split or skill. The split or skill supervisor listens to agent calls, monitors the split or skill queue status, and assists agents on ACD calls. Although supervisors can assist agents on ACD calls, supervisors do not receive ACD calls unless supervisors are members of the split or skill.

If you have CMS or Basic Call Management System (BCMS), you can measure and create reports on the status of ACD agents, splits or skills, and trunks.



1. Incoming calls
2. ACD switch
3. Trunk group 1
4. Trunk group 2
5. Trunk group 3
6. Trunk group 4
7. Split 1 Business Travel with 10 agents
8. Split 2 Personal Travel with 8 agents
9. Split 3 Group Travel with 5 agents
10. Split 4 General Information with 15 agents

11. Queues
12. Announcement 1
13. Announcement 2
14. Intraflow (Call Coverage)
15. Split 2 Personal Travel (3rd choice)
16. Split 3 Group Travel (2nd choice)
17. Split 4 General Information (1st choice)
18. Supervisor with Service Observing
19. Announcement
20. Disconnect
21. CMS
22. Terminal
23. Printer

ACD methods

The following table summarizes the different call distribution methods.

Agent selection method	call selection	Feature availability
Direct Department Calling	The first available agent found in the hunt sequence.	Non-EAS
UCD-MIA	The most idle agent, without regard to skill level.	Non-EAS, EAS, Avaya Business Advocate
EAD-MIA	The highest skill level, most idle agent.	EAS, Avaya Business Advocate
UCD-LOA	The least occupied agent, without regard to skill level.	EAS, Avaya Business Advocate
EAD-LOA	The highest skill level agent with the lowest occupancy.	EAS, Avaya Business Advocate

Basic ACD split queuing and announcements

Communication Manager automatically allocates queue slots so callers to wait for an agent to become available, that is, idle. The next available agent is automatically connected to the call in the queue.

For non-vector-controlled splits, calls do not queue in the following cases:

- No agents are logged in.
- All logged-in agents are in the Auxiliary (AUX) work mode.

The caller hears a busy signal unless a call comes in using the automatic-in Central Office (CO) facility. In this case, the caller hears a ringback from the CO and Communication Manager continuously attempts to place the call in the queue.

You can assign two announcements to each split and administer a second announcement to repeat. When an incoming call is directed to an ACD split, the call is either directed to an agent or is automatically connected to the first announcement. For information about how Call Forwarding and Call Coverage affect announcements, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

First announcement

After a call enters a queue, the caller hears ringing and the first announcement delay interval begins. If an agent becomes available during the first announcement delay interval, the call is connected to the agent. Otherwise, the interval expires and the system tries to connect the incoming call to the first announcement, with one of the following results:

- If the first announcement is available, the caller hears ringing, then the first announcement.
- If the announcement is busy and has no queue, the caller hears ringing and the first announcement delay interval is reset. The system tries to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, then:
  - If the queue is full, the caller hears ringing and the first announcement delay interval is reset. The system tries to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the caller hears ringing, then the first announcement. The system then tries to connect the call to an agent.
- If the announcement is not busy, but is still unavailable, the second-announcement delay interval begins and the system attempts to connect the call to the second announcement.

If there is no first or second announcement, the call remains in queue until answered or removed from the queue.

Forced first announcement

The first announcement delay interval defines how long a call remains in queue before the call is connected to the first announcement. If the delay interval is 0 seconds, the incoming call is automatically connected to the first announcement. This is a forced first announcement. The call is not routed to an agent until after the caller hears the first announcement.

With a forced first announcement, the following occurs:

- If a first announcement is available, the caller hears ringing and the first announcement. The system attempts to connect the call to an agent.
- If the announcement is busy and has no queue, the system waits 10 seconds and attempts to access the announcement.
- If the announcement is busy and has a queue, then:
  - If the queue is full, the system waits 10 seconds and attempts to access the announcement.

  - If the queue is not full, the call enters the announcement queue. The caller hears ringing and the first announcement. The system attempts to connect the call to an agent.
- If the announcement is not busy but is still unavailable, for example, it is deleted, then the system attempts to connect the call to an agent.

After a forced first announcement, the caller always hears ringback or music-on-hold, if administered until the call is answered or is connected to a second delay announcement. After a first or second delay announcement, the caller hears music-on-hold, if administered.

Second announcement

After the first announcement, the second-announcement delay interval begins and the caller hears ringing (if there is no forced first announcement), or music, if provided. If an agent becomes available during the interval, the call is connected. Otherwise, the interval expires and the system tries to connect the incoming call to the second announcement, resulting in one of the following:

- If the second announcement is available, the caller hears ringing or music, then the second announcement.
- If the announcement is busy and has no queue, the caller hears ringing and the second-announcement delay interval is reset. The system tries to access the announcement again when the interval expires.
- If the announcement is busy and has a queue, then:
  - If the queue is full, the caller hears ringing (only if the first announcement has not been heard) and the second-announcement delay interval is reset. The system tries to access the announcement again when the interval expires.
  - If the queue is not full, the call enters the announcement queue and the caller hears ringing (only if the first announcement has not been heard), then the second announcement. The system then connects the call to an agent.
- If the announcement is not busy but is still unavailable, the call remains in queue until answered or removed from the queue.

After the second announcement, the caller hears music, if provided, or silence and then:

- If you administered the split or skill to repeat the second announcement, the system tries to connect the call to the second announcement after the delay expires.
- If you administered the split or skill not to repeat the second announcement, the call remains in the queue until answered or removed from the queue.

Forced disconnect

You can connect an incoming call directly to an announcement and then disconnect the call after the announcement has completed in one of two ways:

- Administer an announcement extension as the incoming destination. The caller is directed to the announcement and is disconnected, without being queued for a split.
- Administer an announcement extension as a point in a split coverage path. Calls that have been in the queue for a long time are forced to go directly to the announcement and are disconnected.

Announcement rules

The following rules govern the announcements that a caller hears:

- Calls that reach a split directly always hear a forced first announcement, if assigned, regardless of subsequent call coverage, call forwarding, night service, or busy signal processing. If the calls queue long enough, the first and second announcements play.
- Calls that reach a split using call coverage receive a second announcement only, if administered. The assumption is that a caller has likely heard a first announcement at the original split or station before being redirected.
- Calls that reach a split using call forwarding receive first and second announcements at the destination split, if administered. The calls can receive a forced first announcement at the original split, if administered, but not at the split the calls are forwarded to.

Entering the queue

When a forced first announcement is not assigned, the system tries to connect an incoming call to an available agent. If an agent is available, the call is connected to the agent. If all agents are active (either on an ACD call or in ACW mode), the call enters the split or skill queue.

When you have administered Intraflow and Interflow with Call Coverage and Call Forwarding All Calls, the caller hears a busy tone or the call is redirected in any of these cases:

- No agents are logged in
- All logged-in agents are in AUX work mode, and the incoming facility is a digit-oriented facility (digits are sent to the communication server as in DID, incoming wink, or immediate tie trunks)

Note:

Central office trunk (non-DID) calls receive ringback from the CO, so the PBX cannot give these callers a busy signal. The system tries to put such calls into queue until successful or until the call is abandoned.

Priority queuing

Priority queuing allows priority calls to be queued ahead of calls with normal priority. You can implement priority queuing in two ways:

- Assign Priority Queuing to the COR of the calling party.
- Assign Priority on Intraflow to an ACD split. This allows calls from the split, when intraflowed into another split, to be queued ahead of non priority calls.

Queue status indications

You can assign queue status indications on agent or supervisor phones or consoles for ACD calls in queue. For more information, see Queue Status Indications.

Agent Availability Query dial tones

Communication Manager enables Mobile Agents to dial the **Agent Availability Query Access Code** Feature Name Extensions (FNEs) and understand their current work status. Depending on the agent state, Communication Manager plays a tone as described in the following table:

Agent state	Communication Manager action
• The incoming cell phone number has Avaya Extension to Cellular (EC500) mapping with the principal extension. • Agent is logged in and is associated with principal extension. • Agent is available to receive ACD calls.	Plays the confirmation tone.
• The incoming cell phone number has EC500 mapping with the principal extension. • Agent is logged in and is associated with principal extension. • Agent is not available to receive ACD calls as the agent is in AUX work mode.	Plays the busy tone.
• The incoming cell phone number has EC500 mapping with the principal extension. • Agent is logged in and is associated with principal extension. • Agent is not available to receive ACD calls as the agent is in After Call work (ACW) mode.	Plays the fast busy tone.
• The incoming cell phone number has EC500 mapping with the principal extension. • Agent is not logged in to the principal extension.	Plays the denial tone.

Basic Call Management System

With BCMS reports, you can manage agents, ACD splits or skills, VDNs, and trunk groups. You can view BCMS reports on a terminal or print a paper copy.

Real-time reports	Historical reports
• Split/Skill Status • System Status • VDN Status	• Agent • Agent Summary • Split/Skill • Split/Skill Summary • Trunk Group • Trunk Group Summary • VDN • VDN Summary

BCMS interactions

Call redirection and conference calls

For information on how BCMS records redirects and conferences calls, see the *Communication Manager Call Center Software - Basic Call Management System (BCMS) Operations* document.

Move agents from CMS

If agents are moved from one split or skill to another split or skill using CMS Supervisor, measurements are stopped for the agents from split or skill and started for the agent to split or skill.

If an attempt is made to move an agent from a non BCMS-measured split or skill to a measured BCMS split or skill using CMS Supervisor, and the move exceeds the maximum number of measured agents, Communication Manager rejects the move. Otherwise, internal BCMS measurements are started for the agent. If the an agent is moved from a split or skill that is measured by BCMS to a split or skill that is not measured by BCMS using CMS Supervisor, internal measurements for the agent stop.

Night Service

When night service is activated for a split or skill, new calls go to the alternate destination. BCMS does not record the calls as OUTFLOW. If the destination is a measured split or skill, BCMS treats the calls as new incoming calls, that is, BCMS does not record the calls as INFLOW.

System Measurements

The system can simultaneously produce BCMS reports, adjunct CMS reports, and Communication Manager traffic measurements.

Although some of the CMS and BCMS report information is similar, BCMS measurements are not determined in the same way as trunk group and hunt group measurements are reported in CMS. Therefore, representation of data in the two report types is not identical.

Business Advocate

Business Advocate is a Call Center Elite feature that uses a patented routing algorithm to:

- Manage agents and call volumes.
- Meet service levels.
- Predict call wait time.
- Reduce agent burnout.

Business Advocate automates the activation of reserve agents to prevent overflow of calls in a queue.

Dynamic Advocate, which is a Business Advocate feature, automatically adjusts the overload threshold based on the service level requirements.

Business Advocate leverages the following features to balance business needs such as service levels, caller segmentation, and multiskilled agent management:

- Percent Allocation for call selection and Percent Allocation Distribution (PAD) for agent selection
- Predicted Wait Time (PWT), which is applicable during call surplus conditions, as a systemwide call selection measurement
- Service Level Supervisor (SLS) with Call Selection Override and Reserve Agent
- Service Objective (SO) by Skill or Vector Directory Number (VDN)

For more information, see *Using Avaya Business Advocate* on the Avaya Support website at http://support.avaya.com.

Call Prompting

Use Call Prompting for flexible call handling based on the information collected from a calling party. The information is in the form of dialed digits that originate from an internal or external touchtone telephone or from an internal rotary telephone that is on the same Communication Manager as the vector. Call Prompting facilitates temporary transfer of call management control to the caller.

When you enable Call Prompting and Call Vectoring, Communication Manager collects Caller Entered Digits (CED) and Customer Database Provided Digits (CDPD) from the network. Communication Manager receives Call Information Forwarding (CINFO) digits in the ISDN message of an incoming call when the AT&T Network Intelligent Call Processing (ICP) service is in use. Communication Manager then collects and forwards the digits to another Communication Manager using **`interflow`** commands.

With Voice Response Integration (VRI), you can use the **`converse-on split`** command to access a Voice Response Unit (VRU) script. The script returns the digits to Communication Manager. You can also use the digits for call management.

You can use Call Prompting in various applications to handle calls with more flexibility. Call Prompting uses specialized vector commands to process incoming calls based on information collected from the caller or from an ISDN-PRI message.

The following list provides a brief description of some Call Prompting applications:

- Automated Attendant: Allows the caller to enter the extension of the party the caller wants to reach, and routes the call to the desired extension.
- Data In/Voice Answer (DIVA) Capability: Allows the caller to hear an announcement based on the digits the caller enters, or to be directed to a hunt group or another system extension.
- Data Collection: Allows the caller to enter data that can be used by a host or adjunct to assist in call handling. The data, for example, can be the caller's account number.
- CINFO Routing: Allows call routing based on the digits the network supplies in an ISDN-PRI message.
- Message Collection: Allows the caller to leave a message or to wait in the queue for an agent.

Call Prompting command set

The following table show the commands that are used for Call Prompting.

Command category	Action	Command
Information collection	Collect information from the calling party, from the public network in an ISDN SETUP message, from a Voice Response Unit (VRU), or from CallVisor Adjunct Switch Application Interface (ASAI).	**collect digits**
Treatment	Play an announcement. Delay with audible feedback of silence, ringback, system music, or an alternate audio/music source.	**announcement** **wait-time**
Routing	Leave a message. Route the call to a number that is programmed in the vector. Route the call to digits that are supplied by the calling party.	**messaging split** **route-to number** **route-to digits**
Branching/ Programming	Go to a vector step. Go to another vector. Stop vector processing.	**goto step** **goto vector** **stop**

Touch-tone collection requirements

Ensure that Communication Manager has a collection resource to accept the touch-tone digits entered by a caller. The resource used for collecting and interpreting touch-tone digits is a unit of hardware called a Touch-Tone Receiver (TTR). The TTRs are provided on the call classifier and tone detector circuit packs, one of which is required for Call Prompting.

The number of TTRs that are required is configured according to the following sources:

- Customer input to the Avaya Account Team.
- Account team input to the configurator tool.

For existing systems that are adding a Call Prompting application, the appropriate number of TTRs can be based on two the following factors:

- Account team input to the configurator tool.
- Application review by the Avaya Design Center.

The process of collecting CINFO digits does not require TTRs.

Outside callers must have a touch-tone telephone to enter the digits that are requested by the **collect digits** command. For callers who are using rotary dialing, the Call Prompting time out takes effect, the **collect digits** command times out, and vector processing continues at the next step. As a precaution, always provide a default treatment such as a **route-to attendant** command, or a **queue-to split** command, in the vector script unless the script is created exclusively for users of touch-tone telephones.

Note:

You can administer any number of seconds from 2 to 10 for the **Call Prompting Interdigit Timeout** field on the Feature-Related System Parameters screen.

Call Prompting considerations

Consider the following points when working with Call Prompting:

- To enter the digits requested using a **collect digits** command, outside callers must have a touch-tone telephone. For such callers using rotary dialing, a 10 second inter-digit time out takes effect, and the **collect digits** command is omitted. As a precaution, a default treatment, for example, **route-to attendant** command, **queue-to split** command, must always be provided in the vector script unless the script is created exclusively for users of touch-tone telephones.
- If a caller does not enter the full number of digits specified in a **collect digits** step, an administered time out occurs. Thereafter, vector processing continues with subsequent vector steps and an attempt is made to process the call using the digits that have been collected. If the digits entered do not represent a valid destination, and if Automated Attendant is being implemented using a **route-to digits** command, the **route-to digits** command fails, and vector processing continues at the next step, which must be a default treatment.
- In case a **route-to attendant** command fails, provide a disconnect announcement.
- Before some vector steps, it is a good practice to insert a brief wait step so that vector processing can handle any possible delays that might cause the vector to fail and the call to not be routed. Follow this best practice when programming a vector:
  - Start all vectors with a “wait-time 0 seconds” step. Adding this step helps ensure you will have a touch-tone receiver (TTR) for vector processing.

- When using an “adjunct routing” step, it is also advise to put a “wait-time 0 seconds” step before the “adjunct routing” step to provide a pause before routing the call to the adjunct system.
- Any time digits are collected from a caller using a “collect digits” step, the step should be preceded by a “wait-time 0 seconds” step, hearing either ringback or music. Again, this ensures you will have a TTR to collect those digits.

- From time to time, all the touch-tone receivers can be in use. As a result, do not start the main vector with a **collect digits** command, since the caller on a DID or tie trunk in this case receives no audible feedback if the caller has to wait for a receiver to become available. Accordingly, include some treatment, for example, a **wait-time 0 seconds hearing ringback** step, before the initial **collect digits** step. The **wait-time** step is not necessary if the **collect** step is collecting ced or cdpd digits.

Call Prompting digit entry-collect digits command

Collect digits command

The **collect digits** command collects the touchtone digits entered by a Call Prompting user. The command allows the system to collect up to 24 digits from a touchtone phone. The first 16 digits are collected immediately, while the remaining digits are stored as dial-ahead digits.

Call Prompting offers some flexibility when entering digits. The caller can:

- Delete the incorrect digit string.
- Enter a variable-length digit string.
- Enter the dial-ahead digits.

Removing incorrect digit strings

About this task

You can include an announcement in the call treatment requesting the caller to enter digits. As an option, the announcement can instruct the caller to enter an asterisk (*) if the caller enters incorrect data.

When the caller enters an asterisk, the following happens:

Procedure

1. Digits collected for the current **collect digits** command are deleted.

   Note:

   Any dial-ahead digits entered, which do not exceed the maximum digit count of 24, are deleted. (Dial-ahead digits are explained later.)

2. Digit collection is restarted.
3. The announcement is not replayed.

Result

Once the caller enters an asterisk, the caller can re-enter digits for processing.

Variable-length digit strings

When administering the **collect digits** command, specify the maximum number of digits to be requested from the caller. Sometimes, you can prompt the caller to enter fewer digits than the maximum specified. In fact, the number of digits that the caller enters can vary for several variations of one **collect digits** command. Each such grouping of digits is called a variable-length digit string.

By using an end-of-dialing indicator in the form of the pound (#) sign, the Call Prompting permits variable-length digit strings does the following:

- Tells the system that the caller has finished entering digits.
- Causes the next vector step to be processed immediately.

When the caller is permitted to enter a variable-length digit string, the announcement part of the **collect digits** command specifies the largest possible number of digits that can be entered. Accordingly, each **collect digits** command must be administered to collect no more than the intended maximum number of digits. The caller can enter a pound sign part of a variable digit string entry either:

- At the end of each variable digit string that is entered. In this case, the pound sign must be included in the count of the number of maximum digits that can be entered.
- At the end of each such string that, not counting the pound sign, contains fewer characters than the maximum number of allowable digits. In this case, the pound sign must not be included in the count of the number of maximum digits that can be entered.

If the caller enters more digits than the maximum number specified, the additional digits are saved as dial-ahead digits for subsequent **collect digits** commands. If the vector or vectors chained to it do not contain another **collect digits** command, the extra digits are discarded.

If the caller enters fewer digits than the maximum number specified and does not complete the entry with the pound sign, a Call Prompting timeout occurs. The timeout ends the command and any digits collected before the timeout are available for subsequent vector processing. The Call Prompting timeout period is set to 10 seconds by default. But can be changed to a value between 2 to 10 seconds using the **Prompting Timeout** field on the Feature-Related System Parameters screen.

⚠ Caution:

Avaya recommends that you do not set the timeout to less than 4 seconds except in special cases. If the timeout is set to less than 4 seconds, the short timeout can cause the caller to miss entering the next digit in a sequence. The caller can miss entering the next digit in a sequence if the callers are unaware that the digits must be entered quickly. The setting of this timer is systemwide and affects digit entry for ALL collect digits steps in all vectors.

For more information, see “Collect digits command” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Using an application involving the entering of variable-length digit strings the user can dial either the number for the attendant or an extension to reach the wanted destination. If the maximum number of digits that can be entered is administered to be 3 and the user wishes to reach the attendant, the user must dial 0#. However, if the user chooses to dial a 3-digit extension, the user must dial, for example, 748 and not 748#. As the maximum number of digits that can be dialed in this case is three, dialing 748# causes the pound (#) sign to be saved as a dial-ahead digit. On the other hand, if the caller dials 748# and if the maximum number of digits that can be entered is four, the pound (#) sign is not saved as a dial-ahead digit as the sign is the last of four digits.

Dial-ahead digits

When digit collection for the current **collect digits** command is completed, vector processing continues at the next vector step. However, the switch continues to collect any digits that the caller subsequently dials until the TTR disconnects. For more information, see collecting digits on the switch in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document. The dialed-ahead digits are saved for processing by subsequent collect digits commands. Dial-ahead digits are explained fully in dial-ahead digits - collect digits command.

Functions and examples

Call Prompting uses some of the functions found in basic call vectoring. It also provides some additional functions that involve digit processing.

Treating digits as a destination

With Call Prompting, you can route calls according to the digits collected from the caller. Once the digits are collected by the **collect digits** command, the **route-to digits** command attempts to route the call to the destination that the digits represent. The command always routes the call to the destination indicated by the digits processed by the most recent **collect digits** command.

The digits can represent any of the following destinations:

- Internal (local) extension, for example split or hunt group, station, and announcement
- VDN extension
- Attendant
- Remote access extension
- External number such as a Trunk Access Code (TAC) or an Automatic Alternate Route/ Automatic Route Selection (AAR/ARS) FAC followed by a public network number, for example 7-digit Electronic Tandem Network (ETN), 10-digit DDD.

The following example shows how a call is routed by digits that are collected from a caller.

Using Call Prompting to route by collected digits

```
1. wait-time 0 seconds hearing ringback
2. collect 5 digits after announcement 300 [You have reached Redux Electric in
Glenrock. Please dial a 5-digit extension or wait for the attendant.]
3. route-to digits with coverage y
4. route-to number 0 with cov n if unconditionally
5. stop
```

In this vector, the caller is prompted to enter the destination extension of the party that the caller wants to reach (step 2). The extension in this vector can contain up to 5 digits. The vector collects the digits and routes to the destination by the **route-to digits** command in step 3.

If the **route-to digits** command fails because the caller fails to enter any digits, or because the extension number entered is invalid, the **route-to number** command in step 4 routes the call to the attendant, which is the default routing option. However, as long as the destination is a valid extension, the **route-to digits** command succeeds, coverage applies, and vector processing terminates. If the destination is busy, vector processing terminates because coverage call processing takes effect.

Note:

Occasionally, all of the system's TTRs can be in use. As a result, when you are collecting digits from a caller, do not start your main vector with a **collect digits** command, since the caller in this case receives no audible feedback if the caller has to wait for a TTR to become available. Include some treatment, for example wait-time 0 seconds hearing ringback, before the initial collect digits step.

In addition, if calls are to be transferred to this vector, add a wait-time step of sufficient length before the collect step to allow the transferring party time to complete the transfer.

Using digits to collect branching information

Call Prompting allows you to direct a call to another step or vector based on the digits that are entered by the caller. This branching is accomplished with a goto step. For example, in the following vector example, digits are used to route calls to different vectors based on an assigned customer number.

Using Call Prompting to branch by collected digits

```
1. wait-time 0 seconds hearing ringback
2. collect 5 digits after announcement 200 [
Please enter your customer number.
]
3. goto vector 8 if digits = 10+
4. goto vector 9 if digits = 11+
5. goto vector 10 if digits = 12+
6. route-to number 0 with cov n if unconditionally
7. stop
```

The wildcard + indicates that the two digits can be followed by zero or any number of additional digits. Callers with a number that begins with the digits 10 are routed to vector 8, callers with a number that begins with the digits 11 are routed to vector 9, and callers with a number that begins with the digits 12 are routed to vector 10.

Vector Routing Tables

You can test digits against entries in a Vector Routing Table. Vector Routing Tables contain lists of numbers used to test a **goto...if digits** command. Digits that are collected with the **collect digits** step can be tested to see if the digits are either in or not-in the specified table. Entries in the tables can include either the "+" or "?" wildcard.

- The "+" represents a group of digits and can only be used as the first or last character of the string.

- The “?” represents a single digit. Any number of “?” can be used at any position in the digit string.

Tables are entered on the Vector Routing Table screen. For information on Vector Routing Tables, see the *Administering Avaya Aura® Call Center Elite* document.

Use the following Call Vector example to test against the numbers provided in the Vector Routing Table.

Testing for digits in Vector Routing Table

```
1. wait-time 0 seconds hearing ringback
2. collect 7 digits after announcement 200 [“Please enter your account number.”]
3. goto vector 8 if digits in table 10
4. queue-to split 5 pri l
5. wait-time 10 seconds hearing ringback
6. announcement 2771
7. wait-time 10 seconds hearing music
8. goto step 6 if unconditionally
```

If the caller enters an account number listed in the Vector Routing Table, the call is routed to vector 8. If the caller enters an account number that matches the wildcard entry, for example 1345987, the call is routed to vector 8.

If the caller enters an account number that is not listed in the Vector Routing Table, or if the caller does not enter an account number, the call is queued to split 5.

Suppose that, instead of containing a list of premier accounts, the Vector Routing Table contains a list of accounts with a poor payment record. The following example shows a vector that only queues calls with account numbers that are not in the table. Calls in the table route to the collection department.

Testing for digits not in Vector Routing Table

```
 1. wait-time 0 seconds hearing ringback
 2. collect 7 digits after announcement 200 [“Please enter your account number.”]
 3. goto step 11 if digits = none
 4. goto step 6 if digits not-in table 10
 5. route-to number 83456 with cov y if unconditionally      [collections]
 6. queue-to split 5 pri l
 7. wait-time 10 seconds hearing ringback
 8. announcement 2771
 9. wait-time 10 seconds hearing music
10. goto step 8 if unconditionally
11. route-to number 0 with cov n if unconditionally
12. stop
```

If no digits are collected, the call is routed to the operator.

Note:

Entries in Vector Routing Tables can be tested against the telephone number of the caller Automatic Number Identification (ANI). For more information, see “ANI /II-digits routing and Caller Information Forwarding (CINFO)” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Using digits to select options

Call Prompting makes it possible to provide a menu of options that callers can use to satisfy their information needs. Callers select the desired option by entering the appropriate requested digit.

Once the digit is entered, a conditional branch to the appropriate treatment is made. The treatment is usually provided by the **route-to number** command.

The following example shows how digits are used to select options.

Using Call Prompting to select options

```
1. wait-time 0 seconds hearing ringback
2. collect 1 digits after announcement 3531 [Thank you for calling Bug Out
Exterminators. If you wish to learn about the services we provide, please dial 1. If
you would like to set up an appointment for one of our representatives to visit your
home or place of business, please dial 2.]
3. route-to number 4101 with cov y if digit = 1
4. route-to number 4102 with cov y if digit = 2
5. route-to number 0 with cov n if unconditionally
6. disconnect after announcement none
```

In step 2 of this vector, the caller is asked to enter either 1 or 2, depending on the service used by the caller. If one of the digits is entered, an appropriate step (3 through 4) routes the call to the relevant extension, that is, either 4101 or 4102. If one of the digits is not entered, the call is routed to the attendant (step 5).

Displaying digits on an agent phone

If the agent has a two-line display phone and the display is in `normal` or `inspect` mode, the collected digits are automatically displayed on the second line. As a default, the digits remain on the line until the digits are overwritten, even after an agent releases the call. Administrators can decide when to clear the caller information on the agent phone display. For other types of phones, the agent must press the **callr-info** button to view the collected digits.

You can administer **callr-info** to reduce the agent time on the searching information that is related to the caller. For example, you can administer **callr-info** to collect information such as a customer account number, thereby eliminating the need for the agent to ask for the information.

Callr-info displays the caller information in the following format:

x = Info: 1234567890

where:

- x is a call appearance letter, for example a, b, and c.
- 1234567890 represents the digits that are collected from the caller.

The digits entered by the caller are collected by the most recent **collect digits** command. Any digits dialed ahead and not explicitly requested by the most recently executed **collect digits** command are not displayed.

If the agent presses **callr-info** when the call rings at the agent phone or when the station is active on a call appearance, the following events occur:

- The 10-second timer for display interval is set.
- The status lamp associated with the button is lit.
- The display is updated. Specifically, the incoming call identification, calling party ICI, is replaced with the collected digits in the format described earlier in this section. Only the digits collected for the last **collect digits** command are displayed.

If all the conditions to use the button, except for the collection of digits, are set and the agent presses the button, the status lamp associated with the button flashes denial.

More than one event can occur during a successful execution after the agent presses the button. The events include the following:

- The 10-second timer times out.
- The incoming call arrives at any call appearance.
- An active call changes status, for example another caller is added to the conference.

If any of the events occur, the following takes place:

- The status lamp associated with the button is turned off.
- The display is updated as previously described.

Note:

The agent can press **callr-info** again to view the digits again, provided the agent is active on the call or the call is still ringing. The agent can also flip between the collected digits and the ICI by alternately pressing **callr-info** and **normal**.

Clearing caller information from the station display

Administrators can decide when an agent's station display is cleared of caller information. Options include:

- Clearing the existing call information when the next call is received
- Clearing the existing call information when the call is released - whether the agent enters After Call Work (ACW) or not
- Clearing the existing call information when the agent leaves ACW mode or if the agent does not enter ACW, when the call is released

Passing digits to an adjunct

Call Prompting allows for the passing of information in the form of collected digits to an adjunct for further processing. Digits are passed to the adjunct by the ASAI Adjunct Routing capability.

An adjunct is any processor that is connected to a switch by the ASAI link. The adjunct makes a routing decision using the **`adjunct routing link`** command according to caller information and/or agent availability, and it returns the routing response to the switch. For example, the adjunct can indicate that the call be routed to a specific ACD agent. This is known as Direct Agent Calling (DAC).

A maximum of 16 Call Prompting digits from the last **`collect digits`** command can be passed to the adjunct by the **`adjunct routing link`** command.

The following example, shows how Call Prompting digits are passed to an adjunct.

Using Call Prompting to pass digits to an adjunct

```
1. wait-time 0 seconds hearing ringback
2. collect 10 digits after announcement 300 [Please enter your 10-digit account number]
3. adjunct routing link 15
4. wait-time 10 seconds hearing music
```

```
5. route-to number 52000 with cov y if unconditionally
6. stop
```

In step 2 of this vector, the caller is asked to enter a 10-digit account number. Once the account number is entered, the adjunct receives this information from the **adjunct routing link** command in step 3. This command then makes the appropriate routing decision if it is able to do so. If the command succeeds within the specified wait time, the command routes the call to the appropriate destination, and the call leaves vector processing. If the command fails, vector processing proceeds to the next step.

In addition to the adjunct routing capability, collected digits also can be passed by way of ASAI to an adjunct by prompting for the digits in one vector and then routing the call to a VDN that is monitored by an Event Notification (VDN) association. The collected digits (up to 16) are sent to the adjunct in a Call Offered to Domain Event Report. For detailed information, see *Communication Manager CallVisor ASAI Technical Reference*.

Note:

Adjunct Routing is fully discussed in Adjunct (ASAI) Routing.

Creating service observing vectors

Service observing vectors can be constructed to allow users to observe calls from a remote location or local station. When combined with Call Prompting, Service Observing vectors can route calls to a:

- Remote access service observing vector
- User-entered Feature Access Code (FAC) and extension
- Pre-programmed FAC and extension

Remote access Service Observing vector example

The following vector example connects a user to Remote Access. Once connected, the user can dial either a listen-only or listen/talk Service Observing FAC followed by the extension number to be observed. Although it is not required, Call Prompting increases security by providing passcode protection with remote service observing.

Remote access Service Observing vector

```
1. wait-time 0 secs hearing ringback
2. collect 5 digits after announcement 2300 [
Please enter your 5-digit      security
  code.
]
3. goto step 5 if digits = 12345 (security code)
4. disconnect after announcement 2000
5. route-to number 5000 with cov n if unconditionally
6. stop
```

User-entered FAC and extension example

The following vector example connects a user directly to the Service Observing FAC and extension based on the digits that are collected by Call Prompting.

Service Observing vector with user-entered FAC and extension

```
1. wait-time 0 secs hearing ringback
```

```
2. collect 5 digits after announcement 2300 [
Please enter your 5-digit      security
  code.
]
3. goto step 5 if digits = 12345 [security code]
4. disconnect after announcement 2000
5. wait-time 0 seconds hearing ringback
6. collect 6 digits after announcement 3245 [
Please enter the number 11 for
      listen-only observing or the number 12 for listen/talk observing      followed by
 the number of the extension you would like to observe.
]
7. route-to digits with coverage n
8. stop
```

Pre-programmed FAC and extension example

The following example shows a vector that connects a user to a pre-programmed FAC and extension using Call Prompting to allow the observer to select the extension that the observer wants to observe. In this example, the observer is Service Observing a VDN.

Service Observing vector with programmed FAC and extension

```
1. wait-time 0 secs hearing ringback
2. collect 5 digits after announcement 2300 [Please enter your 5-digit security code.]
3. goto step 5 if digits = 12345 [security code]
4. disconnect after announcement 2000
5. wait-time 0 seconds hearing ringback
6. collect 1 digits after announcement 2310 [Enter 1 to observe sales, 2 to observe
billing.]
7. route-to number 113001 with cov n if digit = 1 [11 = listen-only observe, 3001 =
Sales VDN]
8. route-to number 113002 with cov n if digit = 2 [11 = listen-only observe, 3002 =
Billing VDN]
9. goto step 6 if unconditionally
```

Dial-ahead digits-collect digits command

Dial-ahead digits

With Dial-ahead digits, callers can skip announcements. Only subsequent **collect digits** commands can use the digits. Other vector commands that operate on digits, for example, **route-to digits** and **goto...if digits**, do not use the digits until Communication Manager collects the digits.

Communication Manager does not forward the digits with interflowed calls and does not display the digits to an agent who presses **callr-info** until Communication Manager completes the **collect digits** command operation.

Collection of dial-ahead digits continues until one of the following occurs:

- Vector processing stops or is terminated.
- The sum of the digits collected for the current **collect digits** command plus the dial-ahead digits exceeds the storage limit of 24. Communication Manager discards additional digits until subsequent **collect digits** commands create more storage.

Note:

Asterisk (*) and pound sign (#) digits are part of the 24 digit limit, as do any dial-ahead digits that callers enter after the asterisk (*) or pound sign (#) digit.

- The TTR required by the caller to collect digits is disconnected, which can happen when one of the following conditions is true:
  - A successful or unsuccessful **route-to number** step is encountered during vector processing, except where the number routed to is a VDN extension.
  - A successful or unsuccessful **route-to digits** step is encountered during vector processing, except where the number routed to is a VDN extension.
  - A successful or unsuccessful **adjunct routing link** step is encountered during vector processing.
  - A successful or unsuccessful **converse-on** step is encountered during vector processing.
  - A Call Prompting time out occurs, during which time the caller does not dial additional digits, asterisks (*) sign, or pound sign (#).
  - Vector processing stops or is terminated.
  - A successful or unsuccessful collect ced step or cdpd step is encountered.

  When the TTR is disconnected due to a **route-to number**, **route-to digits**, **converse-on**, **adjunct routing link** , or **collect ced/cdpd** step, Communication Manager discards all dial-ahead digits. Therefore, following a failed **route-to**, **converse**, or **adjunct routing link** step, a subsequent **collect digits** step always requires callers to enter digits.

Dial-ahead digit vector examples

The vectors in the following examples illustrate a situation where a caller can enter dial-ahead digits. In this case, the caller must have a touch-tone telephone. Program an alternative handling sequence in case the caller has a rotary telephone or the caller does not dial a touch-tone digit before the time out period.

Step 2 of vector 30 offers the caller two options, each of which provides different information. The caller is prompted to enter either 1 or 2, depending on the information the caller wants to hear. Once the caller enters a digit, the digit is collected by the **collect digits** command. Thereafter, an attempt is made by the **route-to number** command to route the call to the appropriate vector (step 3 or 4). If the caller enters a digit other than 1 or 2, the appropriate announcement is provided (step 5), and the digit entry cycle is repeated (step 6).

If the caller enters 1, vector 31 is accessed.

Using dial-ahead digits to bypass announcements, example 1

```
VDN (extension=1030   name=''Coastal''   vector=30)
Vector 30:
1. wait-time 0 seconds hearing ringback
2. collect 1 digits after announcement 3000 [Thank you for calling Coastal League
Baseball Hotline. You must have a touch-tone telephone to use this service. If you wish
to hear the scores of yesterday's games, please press 1. If you wish to hear today's
schedule of games, please press 2.]
```

```
3. route-to number 1031 with cov y if digit = 1
4. route to number 1032 with cov y if digit = 2
5. announcement 301 [Entry not understood. Please try again.]
6. goto step 2 if unconditionally
```

In step 1 of vector 31, the caller is offered three options that supplement the original option provided in vector 30. The caller is prompted to enter either 3, 4, or 5, depending on what information the caller wants to hear. If the caller enters an incorrect digit, the customary digit correction routine is implemented (steps 5 and 6). Once an appropriate digit is entered, the call is routed, in this example by a **goto step** command (step 2, 3, or 4), to the appropriate announcement (step 7 or step 9).

In step 10 of vector 31, the caller is prompted with the choice of returning to the main menu provided in vector 30 or of terminating the call. If the caller selects the former option (by entering 9), the call is routed to vector 30 and the entire process is repeated.

Using dial-ahead digits to bypass announcements, example 2

```
VDN (extension=1031  name=''Scores''   vector=31)
Vector 31:
 1. collect 1 digits after announcement 4000 [If you wish to hear scores of games in
both divisions, please press 3. If you wish to hear scores for Northern Division games
only, please press 4. If you wish to hear scores for Southern Division games only,
please press 5.]
 2. goto step 7 if digits = 3
 3. goto step 7 if digits = 4
 4. goto step 9 if digits = 5
 5. announcement 301 [Entry not understood. Please try again.]
 6. goto step 1 if unconditionally
 7. announcement 4002 [Northern Division scores]
 8. goto step 10 if digits = 4
 9. announcement 4003 [Southern Division scores]
10. collect 1 digits after announcement 4004 [If you wish to return to the main menu,
please press 9. Otherwise, press 0.]
11. route-to number 1030 with cov n if digit = 9
12. goto step 15 if digit = 0
13. announcement 301 [Entry not understood. Please try again.]
14. goto step 10 if unconditionally
15. disconnect after announcement none
```

Vector 32 is similar in design to vector 31. The major difference is the information provided and the requested digit entries.

In this example, the caller has to go through two sets of options to get the desired information. Each option set is introduced by an announcement. However, because of the dial-ahead digit capability, the caller can bypass the announcements if the caller wishes to. The caller can enter 1 and 5 within a matter of seconds to hear yesterday's Southern Division scores.

The caller can enter digits while the call is being queued for an announcement or while the announcement is playing. If digits are entered during an announcement, the announcement is disconnected. If digits are entered while a call is queued for an announcement, the call is removed from the announcement queue.

Dial-ahead digits, example 2

```
VDN (extension=1032   name=Schedule   vector=32)
Vector 32
 1. collect 1 digits after announcement 5000 [If you wish to hear today's schedule of
games in both divisions, press 6.
    To hear today's schedule of games in the Northern Division only, press 7. To hear
today's schedule of games in the
    Southern Division only, press 8.]
```

```
 2. goto step 7 if digits = 6
 3. goto step 7 if digits = 7
 4. goto step 9 if digits = 8
 5. announcement 301 [Entry not understood. Please try again.]
 6. goto step 1 if unconditionally
 7. announcement 5002 [Northern Division schedule]
 8. goto step 10 if digits = 7
 9. announcement 5003 [Southern Division schedule]
10. collect 1 digits after announcement 4004 [To return to the main menu, press 9.
Otherwise, press 0.]
11. route-to number 1030 with cov n if digit = 9
12. goto step 15 if digits = 0
13. announcement 301 [Entry not understood. Please try again.]
14.goto step 10 if unconditionally
15.disconnect after announcement none
```

ASAI-requested digit collection

With ASAI-requested Digit Collection, an adjunct can request that a DTMF tone detector be connected for the purpose of detecting user-entered digits. The collected digits are passed to ASAI monitoring or controlling adjuncts for action. Communication Manager handles the digits as if the digits were dial-ahead digits. The caller can request for Sequence Dialing after the call has been routed to the final destination and has resulted in an unanswered call, that is busy, no answer.

The digits are not necessarily collected while the call is in vector processing. The digits can be sent to an ASAI adjunct, used by Call Prompting features, or both.

You must enable **ASAI Adjunct Routing** and **Call Prompting** for the feature to work.

ASAI-provided dial-ahead digits - collect digits command

The ASAI-provided digits feature allows an adjunct to include digits in a Route Select capability. The digits are treated as dial-ahead digits for the call. Dial-ahead digits are stored in a dial-ahead digit buffer and can be collected one at a time or in groups using the **collect digits** command. Although the adjunct can send more than 24 digits in a Route Select, only the first 24 digits (or 24-x, where x is the number of digits collected by vector processing prior to executing the **adjunct routing link** vector command) are retained as dial-ahead digits. An application can use this capability to specify the digits that Communication Manager must pass to the VRU as part of the **converse-on** vector step.

Note:

The maximum number of dial-ahead digits that can be stored in the buffer is dependent on the number of digits that were already collected for the call by a previous **collect digits** vector command. If x digits were collected by vector processing prior to executing an **adjunct routing link** vector command, the x digits collected reduces the maximum number of digits that can be stored as dial-ahead digits as a result of a Route Select. The rest are discarded.

Call Prompting interactions

The following interactions apply specifically to Call Prompting. For general Call Vectoring interactions that affect Call Prompting applications, see “Call Vectoring”.

Authorization Codes

If authorization codes are enabled, and a **route-to** command in a prompting vector accesses AAR or ARS, if the VDN’s FRL does not have the permission to use the chosen routing preference, then the system does not prompt for an authorization code and the **route-to** command fails.

CallVisor ASAI

ASAI-provided digits can be collected by the Call Vectoring feature using the **collect** vector command as dial-ahead digits. CINFO is passed to CallVisor ASAI.

Hold

With the exception of CINFO, if a call is put on hold during the processing of a **collect** command, the command restarts, beginning with the announcement prompt, when the call is taken off hold. All dialed-ahead digits are lost. Similarly, if a call to a vector is put on hold, vector processing is suspended when a **collect** command is encountered. When the call becomes active, the **collect** command resumes.

Inbound Call Management (ICM)

You can use Call Prompting to collect information that can be used by an adjunct to handle a call.

Transfer

If a call to a VDN is transferred during a **collect** command, the **collect** command restarts when the transfer is complete, and all dialed-ahead digits are lost. Similarly, if a call to a vector is transferred, vector processing is suspended when a **collect** command is encountered. When the transfer is complete, the **collect** command resumes. This is not true when a **collect** command collects CINFO digits. In this case vector processing is not suspended. Attendant extended calls do suspend vector processing in the same way as transferred calls.

Expert Agent Selection

You can use Expert Agent Selection (EAS) to match caller needs and agent skills. Matching the call to relevant agents reduces call transfer and call-holding time and increases customer satisfaction.

- EAS associates the phone with an agent only when the agent is logged in. While the agent is logged in, calls to the agent login ID are directed to the agent.
- With Direct Agent Calling (DAC), calls to a particular agent are treated as ACD calls.

You can use EAS only if you enable **ACD** and **Call Vectoring**. All of the existing ACD features and Call Vectoring capabilities can be used within EAS applications.

As with Call Vectoring calls, EAS calls are directed to VDNs, which in turn point to vectors. However, unlike Basic Call Vectoring, skills can be assigned in EAS to VDNs, or can be associated with vector steps to represent caller needs. As for Call Vectoring calls, EAS calls are queued to ACD hunt groups. However, with EAS enabled, ACD hunt groups are called “skill hunt groups”, not splits.

Skill hunt groups deliver calls to EAS agents. Agent skills are administered on the Agent Login ID screen.

Note:

These are the same login IDs that are used by CMS and BCMS.

Logical Agent implies that telephones are no longer preassigned to hunt groups. When the agent logs, the telephone becomes associated with all of the skill hunt groups that are assigned to that agent login ID.

With **EAS** enabled, ACD calls can also be directed to a particular agent, and not to the skill hunt group. The direct agent call is treated as an ACD call, but waits in queue for a specific agent to become available. Direct agent calls have a higher priority than skill hunt group calls.

EAS terminology

Term	Description
Agent skill	The type of call that a particular agent can handle. With EAS, you can assign up to four skills to each agent, with a primary (level 1) or secondary (level 2) skill level. With the following releases of Communication Manager for EAS-PHD: • Prior to 2.0, an agent can be assigned as many as 20 skills • With releases higher than 2.0, an agent can be assigned up to 120 skills
Caller needs	The reasons why customers call your call center. Caller needs are determined by the VDN number that the caller dialed, by Call Prompting, or by Automatic Number Identification (ANI) database lookup. You define caller requirements in the vector in order to route calls to an ACD agent with particular skills to match the needs of the caller. The caller needs, which translate to skills, become active for an ACD call whenever a queue to the main skill or check backup skill vector command is executed and the threshold condition is met.
Skill	A specific caller or business need of your call center. You define your skills based on the needs of your customers and your call center. You specify skills by skill numbers, which are assigned to agents and are referenced in vectors to match caller needs with an agent who is skilled to handle those needs. When configuring your call center for skills, a particular skill number always has the same meaning, whether it is an agent skill, VDN skill, or skill hunt group.
Skill hunt group	Calls are routed to specific skill hunt groups that are usually based on caller needs. Agents are not assigned to a skill group. Agents are assigned specific skills that become active when the agents log in.

Table continues…

Term	Description
Skill level	An administrator can assign a skill level for each agent skill. With EAS-PHD, skill levels can range from 1 to 16, with 1 being the highest skill level (also known as the highest-priority skill). Without EAS-PHD, skill levels can be defined as primary (level 1) or secondary (level 2), with the primary being the highest-priority skill. When calls are queued for more than one of the agent's skills and the agent's call-handling preference is by skill level, the agent receives the oldest call waiting for the agent's highest level skill. If an agent's call-handling preference is by greatest need, then the agent receives the highest-priority, oldest call waiting for any of that agent's skills, regardless of skill level.
Top agent	Agents in a given skill who have that skill assigned as their top skill, which is the highest-level skill of all the skills assigned to an agent. If an agent has more than one highest-level skill, the top skill is the first highest-level skill that an administrator assigns to the agent.
Top skill	For EAS-PHD, top skill is an agent's first-administered and highest-priority or highest-level skill. For EAS, top skill is an agent's first-administered primary skill or first-administered secondary skill if the agent has no primary skill. With call handling preference by skill level, an agent is most likely to receive a call for the top skill.
VDN skill preference	Up to three skills can be assigned to a VDN. Calls use VDN skills for routing based on the preferences that you specify in the vector. VDN skill preferences are referred to in the vector as 1st, 2nd, and 3rd.

EAS administration

Expert Agent Selection (EAS) uses vectors to route calls to agents.

To administer EAS, you must:

- Assign skills to VDNs on the Vector Directory Number screen.
- Create vectors to route calls to skills.
- Assign skills with priority levels on the Agent Login ID screen.

EAS administration	Description
VDN	You can administer up to three VDN skill preferences on the Vector Directory Number screen. The skill fields on this screen indicate the skills that are required to handle calls to this VDN. All the VDN skills on this screen are optional. For example, only the first and third or only the second and third VDN skills can be assigned. Vector steps can then refer back to these fields to route calls. For example, **`queue-to skill 1st`** routes calls the skill administered as first on this screen.
Vector	When a call routes to a VDN, the VDN directs the call to a vector that is specified on the Vector Directory Number screen. The vector then queues the call to the skill specified in a vector step. You can write vectors that route calls to specific skill numbers or to the skill preferences administered on the Vector Directory Number screen.

Table continues…

EAS administration	Description
Agent	Assign skills to each agent and a skill level to each skill. When a vector routes incoming calls to a skill, Communication Manager delivers the call to an available agent with the assigned skill. If agents are unavailable, Communication Manager keeps the call in the queue until an agent with the required skill is available to answer the call.
Agent selection	The administered agent selection method and call handling preference determine which agent receives an incoming call.
Agent selection method	EAS can use Uniform Call Distribution (UCD) or Expert Agent Distribution (EAD) to select agents for calls. Both methods use the Most-Idle Agent (MIA) or Least Occupied Agent (LOA) algorithm to select agents.

Note:

You cannot enter group information on the Station screen with the **EAS** field enabled when a work mode button is selected. The **assist** and **queue status** buttons require entry of group information.

EAS benefits

Because you can match caller needs to an agent who has the appropriate skills to handle the call, your call center can achieve the following:

- Maximum profitability.
- Greater customer satisfaction because the caller reaches, on the first call, an agent with the necessary skills to handle the call.
- Greater responsiveness to customer needs because you can base call distribution on either skill level or greatest need.
- Improved agent performance and satisfaction because agents have the required skills.
- Improved agent performance because supervisors have the option to have agents handle calls based on either skill level or greatest need. For agents, it offers an opportunity to learn new skills.
- Ability to track the number of calls that are handled by particular skills from the VDN perspective. You can see whether vectors are performing as expected.

Skill-based call distribution

With EAS, call distribution is based on agent skills. Caller needs are determined by the VDN called or by voice prompting. An agent, who has a minimum of one skill that a caller requires, is selected to handle the call. Assign skills and skill levels to agents to determine which types of calls go to which agents and to determine the order in which agents serve waiting calls.

Interruptible Aux

With Interruptible Aux, you can make Expert Agent Selection (EAS) agents in the Auxiliary (AUX) work mode available to receive incoming calls. However, you must assign interruptible reason codes to the agent login IDs.

Greatest need call distribution

With EAS, you can set call distribution based on greatest need instead of skill level. You can distribute the highest-priority, oldest call waiting to an agent with an appropriate skill, even if that skill is not the agent's highest-priority skill.

Percent allocation call distribution

With percent allocation, you can assign a percentage of an agent time to each of the assigned skills to comprise a total of 100 percent of the agent staffed time. Percent allocation then selects the call that is the best match for an agent's administered skill percentages.

Percent allocation is available with Avaya Business Advocate. For more information, see *Using Avaya Business Advocate*.

Percentage allocation routing for EAS

With Percentage Allocation Routing, you can use VDNs to distribute calls to local or remote destinations. You can administer the distribution of work by routing incoming calls based on the percentage that you specify for each destination.

ACD queuing and vector commands

ACD queuing and the vector commands, such as **`queue to skill`** and **`check skill`**, are used to route a call to an agent with the appropriate skill to handle the call.

Route calls by skill level

Under agent surplus conditions, you can use the feature to request selection of an agent with a particular skill level. Using vector processing, you can apply skill level as well as skill (and other factors) in agent selection. The **`check skill`** vector command is used to route calls to an agent with a particular skill. For more information on check skill vector command, see the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document. For more information on how to route calls by skill level, see the *Administering Avaya Aura® Call Center Elite* document.

EAS-PHD - 120 skills/16 skill levels

With EAS-PHD, you can assign up to 120 skills to an agent ID. For each skill, assign one of the 16 skill levels, with 1 being the highest skill level and 16 being the lowest skill level.

If calls are waiting for some of the agent skills and the agent call-handling preference is `skill level`, the agent receives the call that requires the highest-priority skill. For an agent, the first-administered, highest-priority skill is known as the top skill of the agent. The top skill represents the skill for which the agent is most likely to receive a call.

If an agent call-handling preference is `greatest need`, the top skill is not useful, because the agent receives the highest-priority, oldest call waiting that requires any of the agent skills, regardless of skill level.

Call Handling Preference

Administer Call Handling Preference on the Agent LoginID screen to route calls based on the greatest need or on an agent skill level. The following table summarizes call routing based on greatest need or agent skill level administration with either UCD or EAD distribution.

Condition	EAD or UCD with skill level	EAD or UCD with greatest need
Agents are available. When a new call arrives, the call is delivered to:	• EAD: Most-idle agent with the highest skill level for the skill required to provide service to the caller. • UCD: Most-idle agent with the skill required to provide service to the caller.	• EAD: Most-idle agent with the highest skill level for the call's skill. • UCD: Most-idle agent with the call's skill.
Agents are not available and calls are in queue. When an agent becomes available, the agent receives:	• EAD: Highest priority oldest call waiting for the highest skill level of the agent. • UCD: Highest priority oldest call waiting for the highest skill level of the agent.	• EAD : Highest priority oldest call waiting for any skill. • UCD: Highest priority oldest call waiting for any skill.

Direct Agent Calling

Calls to an agent login ID are treated as direct agent calls if the caller and the agent have the DAC Class of Restriction (COR). Direct agent calls can be originated by stations or trunks. If the caller or agent does not have the COR, the call is treated as a normal non ACD call.

DACs are treated as ACD calls and receive a zip tone answer, queue as other ACD calls do, allow the agent to enter after call work following the call, and are measured by BCMS and CMS.

Any of the agent skills can be the direct agent skill. If you set `greatest need` as the Call Handling Preference, the agent always receives direct agent calls before skill calls. This is because DACs have a higher priority than skill calls. However, if you set `skill level` as the Call Handling Preference, the agent receives DACs first only if the direct agent skill has the highest skill level of an agent. Otherwise calls from a skill with a higher level are distributed before DACs. If the direct agent skill and another skill are the same skill level, the agent receives direct agent calls before the other skill calls because DACs have a higher priority.

A **`route-to vector`** command with an EAS login ID as the destination is treated as a DAC if the VDN and agent have the COR and the **Direct Agent** field is set to `y`.

Note:

DAC requires CallVisor Adjunct-Switch Application Interface (ASAI) or EAS. You must administer the originating and called party Class of Restrictions (CORs) to allow Direct Agent Dialing.

With DAC, callers can:

- Contact a specific agent instead of a skill hunt group

- Choose to wait in queue for an agent if the agent is busy
- Use the agent loginID for callback and transfer
- Hear a system-wide DAC delay announcement while holding
- Follow the coverage path of an agent, if the agent does not answer the call immediately

How does DAC work

- Callers can dial the agent login ID as part of a DID or from auto attendant as an extension number.
- Direct agent calls have a special ringing sound regardless of the agent work state. The current work mode button on the agent phone flashes.
- If the agent is on a call, the agent can use multiple call handling to decide whether to put the call on hold in order to take the direct agent call.
- If the agent is available, the call is delivered according to the answering and alerting options.
- If the agent is not available or if multiple call handling is not used, call coverage or RONA routes the call to backup.
- While on direct agent calls, agents are unavailable for subsequent ACD calls. If the agent logs off by unplugging the headset, the agent can still answer a direct agent call in the queue by logging back in to become available. Agents who have direct agent calls waiting are not allowed to log off using a FAC. If the agent is in the manual-in mode or pushes the **acw** button while on a direct agent call, the agent enters the ACW mode.

Generally, direct agent calls are queued and served in first-in, first-out order before other calls, including priority calls. However, if you administer a skill level for Call Handling Preference, you must assign the highest priority to direct agent calls. Otherwise, calls with a higher skill level are distributed before direct agent calls.

You can use Multiple Call Handling (MCH) to allow agents to answer a direct agent call with another ACD call active.

Direct agent calls follow the receiving agent coverage and call forwarding paths, if the features are administered. Once a call goes to coverage or is forwarded, the call is no longer treated as a direct agent call and CMS is informed that the call has been forwarded.

DAC considerations

Maximum number of agents

If an agent is assigned to more than one split or skill, each assignment applies to the maximum number of agents. When computing the number of agents measured by BCMS, count one agent as one agent regardless of the number of logged-in splits or skills. For CMS sizing, count one agent for each agent in each split or skill measured by CMS, for example, count one agent logged in to three splits or skills as three agents.

Using the Number of Agents System Capacity screen, you can view the *Used*, *Available*, and *System Limit* counts.

MIA across splits or skills

You use the MIA Across Splits/Skills feature to equally distribute calls to agents with multiple splits or skills. When agents handle a call for one split or skill, the agent moves to the bottom of the idle agent list. With MIA Across Splits/Skills, agents cannot receive calls from all of the splits or skills. If, for example, split 20 has a very short average agent idle time and split 22 has a very long average agent idle time, agents with both of the skills cannot be the most-idle for skill 22 because the agents are continuously taking calls for split 20.

Announcements

Announcements can be analog, aux trunk, DS1, or integrated. Integrated announcements use TN2501AP, co-resident announcement board, or Avaya Aura® Media Server and queuing is based on whether one of the playback channels is available. When a channel becomes available, any announcement on the board can be accessed, including the announcement already being played. A caller can be in queue for an announcement because a channel is not available, even though that announcement is not being used.

Queues for analog and aux trunk announcements are on a per-announcement basis. You can also install multiple Integrated Announcement boards to use more announcements.

If you use a delay announcement, answer supervision is sent to the CO when the caller is connected to the announcement. Call charge, if applicable, begins when answer supervision is returned.

Storage and retrieval of messages

Leave Word Calling messages can be stored for an ACD split or skill and retrieved by a split or skill member, a covering user of the split or skill, or a system-wide message retriever. The message retriever must have a phone display and authorization. You can also assign a remote Automatic Message Waiting lamp to an agent phone to indicate when a message has been stored for the split or skill.

Class of Restriction

You must assign a Class of Restriction (COR) to each ACD split or skill and individual agent. You can use Miscellaneous Restrictions to prohibit selected users from accessing certain splits or skills. You can use Miscellaneous Restrictions or restrictions assigned through the COR to prevent agents from being accessed individually. Unless you administer such restrictions, each agent can be accessed individually as well as through the split or skill.

An agent with origination and termination restriction can receive ACD calls and use the assist function. A phone in a COR with termination restriction can receive ACD calls.

If you are using Service Observing, administer a COR for observers and agents being observed.

Trunk groups and ACD splits

- If you assign an ACD split extension as the incoming destination of a trunk group and the split extension is later changed, you must also change the incoming destination of the trunk group to a valid extension.

- Calls incoming on a non-DID trunk group can route to an ACD split instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the ACD split or a VDN.
- For MEGACOM 800 Service with DNIS over a wink/wink-tie trunk, if all agents are logged out or in the AUX work mode, incoming MEGACOM calls receive a busy signal if no coverage path is provided (unlike other automatic-in trunk groups, which receive ringback from the central office).
- CO communication servers usually drop calls that remain unanswered after two to three minutes. Therefore, if an incoming CO call queues to a split without hearing an announcement or music, and the caller hears CO ringback for two to three minutes, the CO drops the call.

Agent considerations

- Do not use agents for hunt group calls and ACD split or skill calls simultaneously. Otherwise, all calls from one split or skill are answered first. For example, if ACD calls are answered first, none of the hunt-group calls are answered until all of the ACD calls are answered.
- Agents with multi-appearance phones can receive only one ACD call at a time unless MCH is active. Without MCH, a phone is available for an ACD call only if all call appearances are idle. The agent can, however, receive non ACD calls while active on an ACD call.

Vector-controlled splits or skills

- You can enhance ACD by using Call Prompting, Call Vectoring and EAS. Vector-controlled splits or skills must not be called directly using the split or skill extension (instead of using a VDN mapped to a vector that terminates the call to a vector controlled split or skill). However, if split or skill extensions are called, the calls do not receive any announcements, are not forwarded or redirected to coverage, and do no intraflow or interflow to another hunt group.
- The oldest-call-waiting termination, which is available with Call Vectoring, is supported for agents who are servicing ACD calls only.

Changing hunt groups from ACD to non ACD

Before you change a hunt group from ACD to non ACD, all agents in that hunt group must be logged out. When you change a hunt group from ACD to non ACD, the system places all agents in that hunt group in busy state. If any phones in the hunt group have an **aux-work** button, the button lamp lights. To become available for calls, the agent presses the **aux-work** button or dials the Hunt Group Busy Deactivation FAC followed by the hunt group number.

DAC interactions

Interaction	Description
Attendant Call Waiting	An attendant can originate or extend a call to an ACD split. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the split queue.
Attendant Intrusion	Attendant Intrusion does not work with ACD split extensions because an ACD extension has many agent extensions. Determining which agent extension to intrude is difficult.

Table continues…

Interaction	Description
Automatic Callback	Automatic Callback calls cannot be activated for an ACD split or skill.
Call Coverage	• Calls can redirect to or from an ACD split or skill. A vector-controlled split or skill cannot be assigned a coverage path. • If the queue is not full, a call enters the queue when a minimum of one agent is on an ACD call or in the ACW mode. Queued calls remain in queue until the **Coverage Don't Answer Interval** elapses before redirecting to coverage. If a split or skill agent is available, the call is directed to the agent. • Calls that redirect on the *Don't Answer Coverage* criterion are reported to BCMS/CMS as intraflowed calls. • If a call is queued for an ACD split or skill and redirects using Call Coverage directly to an announcement, the call is dropped after the announcement. • Calls to a split or skill that are directed to an agent do not follow the call coverage path of the agent. If an agent activates **Send All Calls**, the distribution of ACD calls is not affected. An ACD split or skill call directed to an agent station follows the split or skill call coverage path, once the **Don't Answer interval** is met. • For a call to an ACD split or skill to be redirected to call coverage on the *Busy Coverage* criterion, one of the following conditions must exist: - All agents in the split or skill are active on a minimum of one call appearance and the queue, if there is one, is full. - No agents are logged in. - All agents are in the AUX work mode.
Call Forwarding All Calls	**Call Forwarding All Calls** activated for an individual extension does not affect the ACD functions of the extension. When activated for the split or skill extension, calls directed to the split or skill are forwarded from the split or skill. Calls receive no announcements associated with that split or skill, other than a forced first announcement, if administered. Communication Manager informs BCMS/CMS that calls are queued on the split or skill. Communication Manager informs CMS when the call is removed from the queue and forwarded. Calls can be forwarded to an off-premises destination to activate Intraflow and Interflow.
Data Call Setup	Phone or data terminal dialing can be used on calls to or from a member of an ACD split or skill.
Data Restriction	If the trunk group used for an ACD call has data restriction activated, agents with **Automatic Answer** set to y do not hear the usual zip tone.

Table continues…

Interaction	Description
DCS	CMS cannot measure ACD splits or skills on a Distributed Communications System (DCS) network as if the splits or skills are on one Communication Manager. Agents for a split or skill must be all on the same Communication Manager. If a call to an ACD split or skill is forwarded to a split or skill at another DCS node, the caller does not hear the forced first announcement at the second split or skill. If an ACD split or skill is in night service, with a split or skill at second DCS node as the night service destination, a call to the first split or skill is connected to the second split or skill's first forced announcement.
Dial Intercom	An agent with origination and termination restriction can receive ACD calls and make or receive dial intercom calls.
Forced Agent Logout from ACW mode	After an agent handles a DAC, the Forced Agent Logout from ACW feature applies when the agent enters the ACW state after the DAC is released.
Hold	If an agent puts an ACD call on hold, information is reported to the CMS using Personal Call Tracking. CMS records the time the agent actually talks on the call.
Individual Attendant Access	Individual attendant extensions can be assigned to ACD splits. Unlike phone users, individual attendants can answer ACD calls as long as there is an idle call appearance and no other ACD call is on the console.
Internal Automatic Answer (IAA)	Internal calls directed to an ACD split or skill are eligible for IAA. You cannot administer IAA and ACD Automatic Answer simultaneously on the same station.
Intraflow and Interflow	Intraflow and Interflow, when used with Call Forwarding All Calls or Call Coverage, allows splits or skills to be redirected to other destinations in and outside the system.
Multi appearance Preselection and Preference	All assigned call appearances must be idle before an ACD call is directed to a phone.
Location Preference Distribution	DAC calls take precedence over Location Preference Distribution.
Night Service - Hunt Group	When **Night Service** is activated for a split or skill and the night service destination is a hunt group, a caller hears the first forced announcement at the original split or skill. The call is redirected to the night service destination hunt group.
Terminating Extension Group (TEG)	A TEG cannot be a member of an ACD split or skill. • Transfer: Calls cannot be transferred to a busy split or skill. The transfer fails and the agent transferring the call is reconnected to the call. If an agent presses the **transfer** button, dials the hunt group extension number and disconnects while the split or skill is busy, the call is disconnected. • Phone Display: For calls dialed directly to an ACD split or skill extension, the identity of both the calling party and ACD split or skill are shown on the phone display.

How to identify caller needs

Caller needs for a particular call can be identified by any of the following methods:

- Interpreting information that is passed from the network in the screen of DNIS digits or ISDN messages.
- Processing Call Prompting digits, digits entered at a Voice Response Unit (VRU), or CINFO digits that are forwarded by the network.
- Using ASAI or a VRU such as Avaya Interactive Response in a host database lookup.

To show how a call center manager matches caller needs and agent skills, let us say the call center receives inbound calls from automobile club members who speak Spanish or English. The callers, in this case, require assistance in planning a vacation route or have trouble with their car and are calling for assistance. The following table provides example associations between caller needs and agent capabilities.

Example of caller need-to-agent skill matching	
Caller requirement	**Capability required**
Tourist information	Knowledge of the region
To speak Spanish	Bilingual
Emergency assistance	Handle stressful callers
Tow truck	Access to dispatch systems

The following list looks at the strategy of the call center manager in matching the caller needs to the capabilities of the agent:

- Tourist information or knowledge of the region

  Travelers require information while traveling or about a future trip. All assigned agents can provide this information.

- To speak Spanish or bilingual

  Separate numbers are published and used as part of Spanish membership information, or Call Prompting is used after a general number is dialed.

- Emergency assistance/handle stressful callers

  Separate emergency road service numbers are published and used, or Call Prompting is used after a general number is dialed. For example, a number is provided for towing.

Note that the call center chose to implement Call Prompting to identify Spanish-speaking callers and callers who require emergency assistance. This allows for quicker and more specialized treatment and therefore better satisfies the caller needs.

In addition, some callers prefer to speak to the same agent the caller spoke to on a previous call. To accommodate this request, a call center manager can implement Direct Inward Dialing (DID) at the call center. Also, Direct Agent Calling (DAC) can be used to direct a call to a specific agent.

The following sections explain further how caller needs are identified.

DNIS/ISDN called party

A set of DNIS digits can be interpreted as a VDN. The following table presents four services and their corresponding telephone number including DNIS digits that can be provided to the caller.

Service	Telephone number	Corresponding DNIS
Emergency road service (English)	800-765-1111	6001
Emergency road service (Spanish)	800-765-2222	6002
Route planning (English)	800-765-3333	6003
Route planning (Spanish)	800-765-4444	6004
General (Call Prompting)	800-765-5555	6005

Note:

DNIS digits must be extensions that are reflected in the dial plan.

Call Prompting, VRU, and CINFO digits

The Call Prompting, VRU, and CINFO digits are entered by the caller in response to any recorded question about a caller's needs, or in the case of CINFO ced or cdpd digits are provided by the call center host computer. For example, a hot line for a product can request that a product code be entered, or a travel service can request a 2-digit state code to indicate the state to which the caller wants to travel. The following table provides a prompt that encourages the caller to enter the appropriate Call Prompting digit for the required service from the automobile club.

"For emergency road service, dial 1."
"Para asistencia con su automovil, marque el dos."
"For travel route directions, dial 3."
"Para informacion sobre rutas, marque el cuatro."

Host database lookup

A host database lookup uses DNIS and ANI to determine the following:

- Skills
- Agent

For example, the database can show that the caller speaks Spanish and communicates with Agent 1367. To access the host information, use either Adjunct Switch Application Interface (ASAI) or a VRU in conjunction with a **`converse-on skill`** step.

Additional agent login ID capabilities

Capability	Description
Auto-Answer	When you set **EAS** to y, you can assign auto-answer settings to agent IDs on the Agent LoginID screen. The auto-answer settings apply to the station where the agent logs in to become staffed. If the auto-answer settings for the station and the agent ID are different, the agent settings override the station settings.
Calls	To call an EAS agent, the caller dials the login ID extension. The call is extended to the physical extension where the agent with the login ID is logged in. Calls to the login ID reach the agent irrespective of the phone the agent is currently using. For example, when agents use multiple phones the agents can be reached by the login IDs irrespective of their current location.
Name	Calls to the login ID display the name associated with the login ID and not the name associated with the phone. This is also true for calls made from a phone with an agent logged in.
Coverage	When an agent logs out, or when calls go to coverage because the agent is busy, or does not answer, calls to the login ID go to the coverage path associated with the agent and not the phone. When an agent logs out, calls go to the agent's busy coverage destination.
Restrictions	Calls to the login ID or from the agent use the restrictions associated with the agent, and not the phone. Phones are fully functional when an agent is not logged in. The restrictions, coverage, and name revert to the phone administration when the agent logs out.

Other agent login capabilities

In addition to skill assignments, the following capabilities are associated with agent login IDs.

Call routing

A call to the login ID reaches the agent independent of the telephone that the agent is currently using. In other words, such a call is sent to the telephone at which the agent is currently logged in.

If the proper Class of Restrictions (COR) is set, callers can initiate a direct agent call either by dialing the login ID extension directly or by calling a VDN that points to a vector that contains first a prompt for the login ID and then a **route-to digits** command. This allows external callbacks by way of Direct Inward Dialing (DID) or an 800 number. Both the receiving agent's login ID COR and the originator's, that is, the caller's COR must have Direct Agent Calling (DAC) set to y. The caller's COR is for the following:

- Telephone extension (for internal calls or transfers)
- Trunk group (for DID calls)
- VDN (for prompted calls)

If the call covers or is forwarded, the COR of the originator (or VDN) and the final agent is used. All feature functionality for ACD calls, except Queue Status indications, is available for direct agent calls.

Internal and external users can originate direct agent calls by dialing the agent's login ID. Also, DAC can be used to transfer ACD calls from one agent to another agent.

If an agent who is receiving the direct agent call is staffed but unavailable, the call waits in front of the skill calls in the skill that is administered as the agent's direct agent skill until either the call is answered or a coverage timeout occurs. Also, the caller hears an optional direct-agent announcement that is followed by music or silence. There is one direct agent announcement per system. The agent, on the other hand, receives a ring-ping, and the current work mode button flashes. If the agent is available, the call is delivered to the agent according to the answering and ringing options. Calls are answered and handled in the same manner as ACD calls. For more information, see "Feature-Related System-Parameters screen" in *Administering Avaya Aura® Communication Manager*.

Login ID name on the telephone display

A call to a logged-in EAS login ID by default displays the name associated with the login ID and not the name that is associated with the telephone. This is also true on the receiving party's display for a call that is made from a telephone with an agent logged in. However, the user can display the name of the physical telephone where the EAS agent is logged in. The user must be active on a call with the agent, and must have a telephone with an alphanumeric display and an inspect button. When the inspect button is pressed during a call to or from the EAS agent, the physical telephone name of the agent is displayed.

Coverage path

Call coverage can occur whether or not the agent is logged in. If the agent is not logged in, the busy criteria is met and the call follows the points on the coverage path. If the agent is logged in but fails to answer, the don't answer criteria is met and the call follows the points on the coverage path. A call to the login ID goes to the coverage path that is assigned to the login ID rather than to the coverage path that is assigned to the telephone extension.

Agent restrictions

A call to the login ID or from the agent uses the restrictions that are associated with the agent and not the telephone.

Telephones are fully functional if an agent is not logged in. The restrictions, coverage, and name revert to the telephone administration when the agent logs out.

If a number of agents are sharing one telephone due to job sharing or shifts, a unique login ID extension is assigned to each agent. Therefore, when an agent is logged out, any calls to the agent login ID are sent to the coverage path of the agent. As a result, login IDs can be used to reach people independent of the location. Such people include those who use more than one phone because the people have more than one office or in the case of security guards, for example, who sit at more than one desk.

Because AAS or messaging-system ports are not mobile, the ports are administered to agent login IDs. When the **AAS** or **AUDIX** field is set to `y`, a field that requests the port number is brought up and the **password** field disappears.

EAS functions and examples

Skills administration

A skill is an attribute that is:

- Administered as a skill hunt group
- Administered to VDNs
- Assigned to agents

A skill hunt group is administered for each skill. A skill hunt group is a set of agents trained to meet particular customer needs.

Generally, if the ability “Spanish-speaking” is assigned to skill 127, for example, it follows that agent skill 127 and VDN skill 127 both signify “Spanish-speaking”. However, note that the agent skill can be assigned a skill term that is broader than that for the corresponding VDN skill. For example, agent skill 127 can be labeled “bilingual” for agents that can handle calls in English and Spanish.

Skills for an application are shown in the following table, which presents a very abbreviated example of such a skill distribution for an automobile club.

Supergroup-99	
Emergency road service-bilingual-22	Route planning-bilingual-44
Emergency road service-English-11	Route planning-English-33

In the table, five skills are defined. Each skill indicates knowledge or an ability on the part of the agent or a need for knowledge on the part of the caller. More than one of the skill can be attributed to the agent according to the agent’s expertise with the corresponding highway services and the language-speaking ability.

The table is arranged in such a manner that the agents at the top level have the broadest knowledge, that is, the agents can handle emergency road service and route planning calls and can speak Spanish. The top level skill group is called Supergroup and the group contains agents who, as a group, can take any type of call regarding the automobile club. Accordingly, this skill group serves as a backup skill group. As you descend through the table, each sublevel corresponds to a group of agents who have more specific skills and can therefore take more specialized calls.

Calls can be distributed to the most-idle agent by using either the Uniform Call Distribution (UCD) option or the Expert Agent Distribution (EAD) option. UCD distributes calls from the skill hunt group to the most-idle agent who has this skill assigned at any priority level. This scenario provides a more even distribution to calls and therefore keeps agents equally busy. EAD distributes calls from the skill hunt group to agents to an available agent who has the highest skill level. Skills that are assigned to an agent at higher skill levels indicate a higher level of expertise or preference by the agent than any lower skill level skills that are assigned to that agent. EAD distribution provides the caller with the best or most expert agent match.

Agents are usually given a preference for higher skill level calls. However, the system can be administered to give agents a preference for the greatest need call. The greatest need call is the highest priority oldest call waiting for any of the agent's skills.

Multiple Call Handling on Request and Forced Multiple Call Handling make it possible for an agent to receive additional ACD calls either after putting a call on hold, or when active on another ACD call. Forced Multiple Call Handling can be used to give priority to an ACD call over an in-progress non-ACD call, or to give priority to a call from one skill over an in-progress call from a different skill.

To administer skills, set the **Skill**, **ACD**, and **Vector** fields to `y`. Instructions for completing the Hunt Group screen are included in *Administering Avaya Aura® Communication Manager*.

VDN skills

Expert Agent Selection (EAS) enhances Call Vectoring and Automatic Call Distribution (ACD) by distributing incoming calls based on:

- Specific skills that are assigned to a Vector Directory Number (VDN) or used in a vector.
- Skills assigned to an agent.

For example, a caller dials a particular number VDN that uses a vector to queue the call to an agent with a skill that matches the VDN skill.

You can assign up to three different skills to a VDN. The first skill assigned to a VDN can be the skill required to meet the needs of the caller who called the VDN. The second and third skills assigned to the VDN can represent backup skills that can also meet the needs of the caller.

Skills assigned to a VDN are commonly called *VDN Skill Preferences*. VDN skill preferences are labeled 1st, 2nd, or 3rd.

Note:

While skills can be optionally assigned to VDNs, the vector controls when and to what VDN skill the call queues.

The following table shows how skill preferences can be assigned to five VDNs that are used for the automobile club. For each VDN, the corresponding call type and the number of the vector to which the VDN points are indicated.

Call type	Skill preferences				
	VDN	1st	2nd	3rd	Vector
General Number	6005				1
Emergency Road Service (English)	6001	11	22	99	3
Emergency Road Service (Spanish)	6002	22		99	2
Route Planning (English)	6003	33	44	99	3
Route Planning (Spanish)	6004	44	99		2

In the table, note that two VDNs point to vector 3, two VDNs point to vector 2, and one VDN points to vector 1. Note also that a 1st and 3rd VDN skill preference, but no 2nd VDN skill preference,

are assigned to VDN 2222. This implies that the call to the VDN, if not already answered, waits longer before queuing to the backup skill, Supergroup-99 in our example, provided the vector is designed to execute accordingly.

The following table shows the skill preferences assigned for one specific VDN (6003) that is used for the automobile club.

Preference	Number	Description
1st	33	Directed to an agent who is trained in Route Planning and speaks English.
2nd	44	Directed to an agent who is trained in Route Planning and is bilingual.
3rd	99	Directed to an agent who can field all calls

In the table, the first VDN skill preference corresponds to a knowledge area that is a subset of the knowledge area represented by the second and the third preference. Similarly, the second VDN skill preference corresponds to a knowledge area that is a subset of the knowledge area represented by the third preference. Such an approach is commonly used to assign VDN skill preferences. The result of the approach is that the longer a call waits, the larger the pool of agents that the ACD checks for handling the call.

Recall that the vector numbers for each VDN associated with the automobile club are listed in the example of VDN skill preferences assignments. VDN 6003 points to vector 3. As such, the skill requirements associated with the VDN are forwarded to the vector. This process is shown in the following figure.

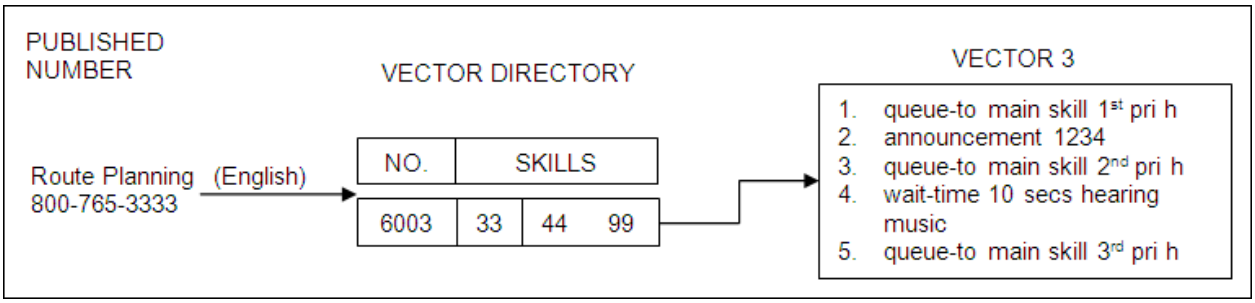


Figure 1: Example of VDN skill implementation

In the example, the English-speaking caller requires information on route planning and dials the appropriate number (800-765-3333). Network 800 features direct the call to VDN 6003, the call enters Communication Manager and is directed to VDN 6003, which points to the appropriate vector. As shown in skill preferences assignments for VDN 6003, VDN skill preferences 33, 44, and 99 are administered as the 1st, 2nd, and 3rd skill preferences, respectively, for VDN 6003.

Vector processing of this application is described in *delivering the call to the skill queue*.

Vector Directory Number screen

The Vector Directory Number screen shown in the following example is used to administer VDN skills.

Vector Directory Number (VDN) screen, page 1

```
change vdn xxxxx                                                page 1 of 2
                    VECTOR DIRECTORY NUMBER

                             Extension: 2001
                                  Name: vdn 2001
                         Vector Number: 1

                     Attendant Vectoring? n
                      Allow VDN Override? n
                                    COR: 1
                                     TN: 1
                               Measured: internal
       Acceptable Service Level (sec): 20
              Service Objective (sec):

        VDN of Origin Annc. Extension:
                               1st Skill:
                               2nd Skill:
                               3rd Skill:
```

Vector Directory Number (VDN) screen, page 2

```
change vdn xxxxx                                                page 2 of 2
                     VECTOR DIRECTORY NUMBER

                               Audix Name:
                  Messaging Server Name:
                     Return Destination:
                  VDN Timed ACW Interval:
                        BSR Application:
          BSR Available Agent Strategy: 1st-found
              Observe on Agent Answer?: n
```

Note:

Skills can be optionally assigned to VDNs, however, the vector controls when and to what VDN skill the call queues.

Call Vector screen

Completion of the Call Vector screen is required for using vectors with EAS. The screen contains three pages. However, if the vector contains less than 11 instructions, complete the first page only, as shown in the following example.

Call Vector screen (Page 1 of 3)

```
change vector 20                                          Page 1 of 3

                            CALL VECTOR
   Number: 20                 Name:_____________________________
Multimedia? n     Attendant Vectoring? n              Lock? y
     Basic? y  EAS? y   G3V4 Enhanced? n    ANI/II-Digits? n     ASAI Routing? n
 Prompting? n  LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? y         Holidays? y
 01 ________________
 02 ________________
 03 ________________
 04 ________________
 05 ________________
 06 ________________
 07 ________________
 08 ________________
 09 ________________
```

```
10 _________________
11 _________________
```

For more information, see the *Administering Avaya Aura® Communication Manager* and *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* documents.

Agent skills

Agent skills represent and define the ability of an agent to handle calls that require the skills. You can assign skill numbers to agents based on characteristics such as training or knowledge, access to systems or information, language ability, and interpersonal traits. Examples of agent skills include the following: speaks Spanish, knows about widget X, can handle complaint calls, has access to a database.

You can assign up to 120 skills with EAS-PHD or four skills without EAS-PHD. You can designate a skill level between 1 and 6 (EAS-PHD) or 1 and 2 (EAS), with 1 being the highest skill level, which is the highest-priority skill.

If an agent has multiple skills, you can create a single skill group for each set of skills. Administer the agent skills on the Agent Login ID screen.

Create a separate skill hunt group for direct agent calls. Direct agent calls are queued to the skill that is administered as the direct agent skill on the Agent LoginID screen. If an agent is not able to log in to the direct agent skill, direct agent calls are queued to the first-administered highest-level skill.

The following table shows the assignment of agent skills.

Example of agent skill assignments				
Agent	**Skills assigned**			
Jan O’Hara	22 (L1)	44 (L2)		
Sam Lopez	99 (L1)			
Sue Carlson	22 (L1)	11 (L1)	44 (L2)	33 (L2)
Mark Davis	44 (L1)			
Amy Brown	44 (L1)	22 (L2)		

Without EAS-PHD, you can assign a maximum of four agent skills to any one agent with one of the two preference levels. With EAS-PHD, you can assign up to 120 skills to each agent with one of 16 preference levels. The skill assignments table shows that four agent skills (22, 11, 44, 33) are assigned to Sue Carlson. The assignments indicate that Sue is bilingual and can service callers who require emergency road service or information on route planning. Only one agent skill, 99-Supergroup, is assigned to Sam Lopez. This means that Sam is serving as a backup.

An L1 or L2 next to the skill number indicates whether the agent skill is assigned as a level 1 or level 2 skill. For example, Jan O’Hara has *Emergency Road Service-Bilingual* as a level one skill and *Route Planning-Bilingual* as a level two skill. When Jan O’Hara is available for an ACD call, provided **Call Handling Preference** is set to `skill-level`, the ACD software first looks for English-speaking callers who are requesting information on the emergency road service. Only if there are no callers requesting emergency road service does the ACD software look for English-speaking callers who are requesting information on route planning. If **Call Handling**

Preference is set to `greatest-need`, Jan O'Hara receives the highest priority, oldest call waiting for either *Emergency Road Service* or *Route-Planning-Bilingual* each time Jan is available.

For any given application, EAS puts no restrictions on which agent skills can be assigned to an agent.

Note:

Agent skills are administered on the Agent Login ID screen.

Preference Handling Distribution

Preference Handling Distribution enables an agent to take calls based on either skill level or greatest need.

If an agent's call-handling preference is by skill level, the agent receives the call that requires the skill for which the agent's skill level is highest.

If an agent's call-handling preference is by greatest need, the agent receives the highest-priority, oldest call waiting that requires any of the agent's skills.

It is recommended that in any skill, all agents have the same call handling preference. This ensures the most consistent distribution of calls by either greatest need or skill level.

Preference Handling Distribution Examples

The following table is an example of how calls queue with Preference Handling Distribution.

Agent is assigned skills and skill levels...	These calls are in queue...
Skill 11, skill level 1	Waiting 15 seconds, priority medium.
Skill 21, skill level 8	Waiting 30 seconds, priority low.
Skill 31, skill level 16	Waiting 45 seconds, priority medium.

Logical Agent capability

With EAS, Communication Manager routes calls to agents based on the agent login ID, not the extension number of the station. The login ID must be consistent with the Communication Manager dial plan. When an agent uses a phone to log in to the system, the login ID overrides the extension ID as far as ACD tracking and characteristics, such as name and Class of Restriction (COR), are concerned.

When a specific login ID is called, Communication Manager routes the call to the phone that the agent is currently logged in to. With Logical Agent, agents can be called regardless of the phone that the agent uses. Calls to agent login IDs can be delivered as DAC calls with the correct COR set for both the originating and the receiving login ID.

Agents are not assigned to skill hunt groups with Logical Agent. Instead, an agent has specific skills that are assigned to login ID. When an agent logs in, the agent is associated with the assigned skill hunt groups and tracking begins for the assigned skills.

Call Management System (CMS) automatically measures a logical agent administered with a minimum of one measured skill when the agent logs in.

Logical Agent uses a single set of work-mode buttons for all skills, that is, an agent is available or in the AUX work mode for all skills at the same time. An agent cannot be available in some skills and in the AUX work mode in others.

Button assignments do not follow the agent login ID because the options are associated with the physical extension and not the agent login ID.

Note:

Converting to EAS requires a change to the CMS login ID if the current ID is not a valid extension number or cannot be made available in the Communication Manager dial plan. Agent login IDs are assigned names from the Dictionary-Login Identification window through CMS Supervisor. Login IDs must be different from the phone extensions.

Delivering the call to the skill queue example

This example shows how a call is delivered to a skill hunt group queue by vector processing.

The skills that are assigned to a VDN define the requirements in the vector for routing calls to an ACD agent with a particular set of skills. These skills become active for an ACD call whenever a **queue-to skill** command is executed. The skills also become active whenever a **check skill** command is executed and the threshold condition is met. Once a skill is active for an ACD caller, the call cannot be delivered to an available ACD agent unless the agent also has one of the active VDN skills.

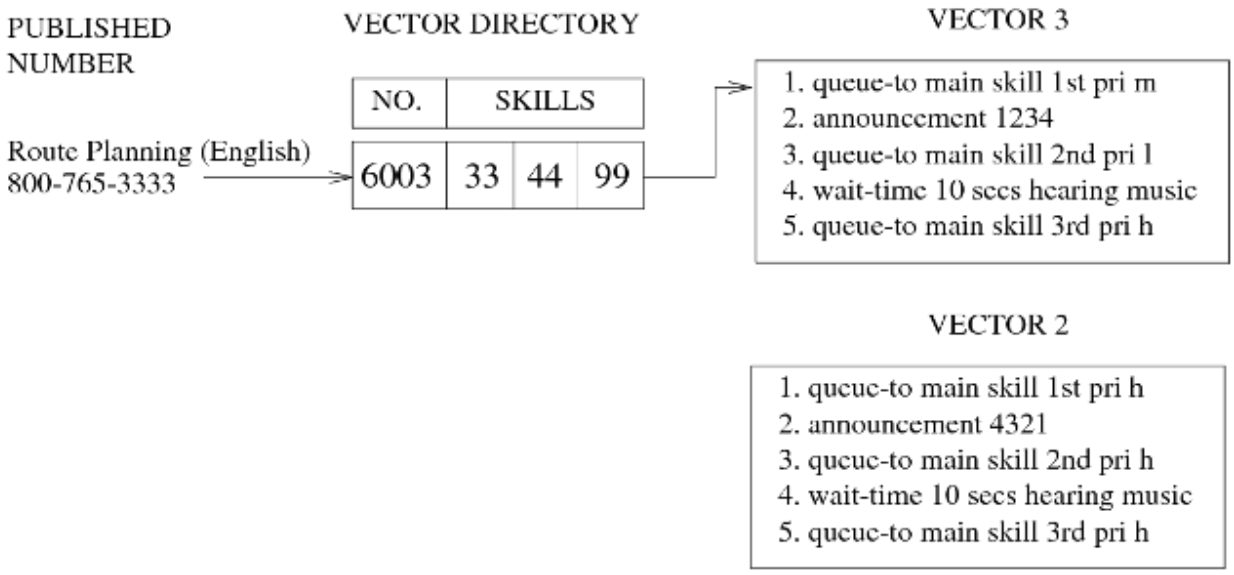


Figure 2: Process for delivery of a call to a skill queue

In the example, an English-speaking caller requires information on route planning and dials the appropriate number (800-765-3333). The call enters Communication Manager, which directs the call to VDN 6003 that is pointing to vector 3. Once vector processing starts, the **queue-to skill** command in step 1 queues the call to the skill hunt group that corresponds to the 1st VDN skill (33-Route Planning-English). If an agent with skill 33 is available, the agent answers the call. If the agent is not available, the call is eventually queued to the skill hunt group that corresponds

to the 2nd VDN skill (44-Route Planning-Bilingual) by the **queue-to skill** command in step 3. This time, if an agent with skill 44 is available, the agent answers the call. If the call is still not answered, the call is eventually queued to the skill hunt group that corresponds to the 3rd VDN skill (99-Supergroup) by the **queue-to skill** command in step 5.

In the process, vector 2 is executed if a Spanish-speaking caller calls. The announcement provided in vector 2 is in Spanish, the announcement in vector 3, which is executed in our example, is in English.

Note also that each of the **queue-to skill** commands in vector 2 queues the call at a high priority, whereas only one of the **queue-to skill** commands in vector 3 queues the call at this high a priority level. The strategy presented here is valuable when there is a limited number of bilingual agents because the bilingual such agents will be available more quickly to service callers who speak only Spanish.

VDN skills can also be used in **check skill, messaging skill,** and **converse-on skill** commands. Within any of the commands, a specific skill number can be used instead of a VDN skill preference, provided that the relevant skill hunt group is correctly administered. For example, step 5 reads queue-to skill 99 pri h.

Example using Call Prompting

The procedure described in the previous section can be enhanced by using Call Prompting. For example, the user can dial a general telephone number where the VDN points to a Call Prompting vector.

Staying with the automobile club example, recall that in examples of services and corresponding DNIS digits, 800-765-5555 is a general telephone number for the service. Recall also that in VDN skill preferences assignments example, 6005 is the VDN for the 800 number and VDN 6005 points to vector 1.

The following vector shows how vector 1 appears.

Call Prompting vector for the automobile club

```
1.  wait-time 0 seconds hearing ringback
2.  collect 1 digits after announcement 5678 ["For emergency road service, dial 1. Para
asistencia con su automovil,
    marque el dos. For travel route directions, dial 3. Para informacion sobre rutas,
marque el cuatro."]
3.  route-to number 6001 with cov n if digit = 1 [English Emergency Road Service VDN]
4.  route-to number 6002 with cov n if digit = 2 [Bilingual Emergency Road Service VDN])
5.  route-to number 6003 with cov n if digit = 3 [English Route Planning VDN]
6.  route-to number 6004 with cov n if digit = 4 [Bilingual Route Planning VDN]
7.  route-to number 6002 with cov n if unconditionally [Bilingual Emergency Road
Service VDN]
```

Once the caller dials 800-765-5555, the call enters Communication Manager and is directed to VDN 6005, which points to the Call Prompting vector. Vector processing begins. Step 1 provides ringback if the caller has to queue for the announcement in step 2. The **collect digits** command in step 2 first provides an announcement that requests the caller to dial 1, 2, 3, or 4, based on the caller need and language. If the caller dials a digit other than one of the four specified, each of the **route-to...if digits** commands in steps 3 through 6 fails and control passes to the **route-to...if unconditionally** command in step 7, which unconditionally

routes the call to VDN 6002. This VDN is assigned the “bilingual emergency road service” skill and points to vector 2, which is provided in the previous section.

If the caller dials 4, step 3 through 5 fail because the required digit, 1, 2, or 3, respectively, was not dialed. Thereafter, control passes to step 6, where the **`route to...if digit`** command finds a digit match and consequently routes the call to VDN 6004. This VDN is assigned the “bilingual route planning” skill and also points to vector 2, which is provided in the previous section.

Note:

VDN Override applies to the skills assigned to the VDN.

Super agent pool

EAS allows a skill hunt group to function as a super agent pool. A super agent pool is a backup group of more than one agent that is able to handle many if not all types of calls coming into the application. In the automobile club examples, Skill Hunt Group 99 (Supergroup) serves as a super agent pool. You can also recall that 99 appears as both a VDN skill and an Agent skill. However, a super agent pool can be assigned a skill hunt group number that is not assigned to a VDN skill. This can and must be done whenever the application requires four levels within the skill table distribution, as shown in the following table.

Supergroup-99			
Emergency road service- bilingual-88		**Route planning-bilingual-77**	
English-66	Spanish-55	English-44	Spanish-33
Bostonian-11	Castilian-13	Bostonian-15	Castilian-17
New Yorker-12	South American-14	New Yorker-16	South American-18

Besides a new skill numbering scheme, our modified skill table has four levels instead of the three levels that are provided in example of a skill table for an automobile club. Except for the skill numbering scheme, the top two levels (Supergroup-99 and Emergency Road Service-Bilingual-88/ Route Planning-Bilingual-77) remain unchanged. However, note that the next level is reorganized into segments to indicate the ability to speak English or Spanish. Finally, note that a new level is added to denote particular types of accents or pronunciation in English and Spanish.

The following table shows how some of the skills in modified skill table for the automobile club are administered to one relevant VDN (VDN 1616).

VDN 1616 - Skill preferences		
1st:	16	Knows about Route Planning, speaks English, has New York accent
2nd:	44	Knows about Route Planning, speaks English
3rd:	77	Knows about Route Planning, is bilingual

Modified vector to accommodate a super agent pool

```
1. queue-to skill 1st pri m
2. announcement 4555
3. queue-to skill 2nd pri l
4. wait-time 10 seconds hearing music
5. check skill 3rd pri l if calls-queued < 3
```

```
6. announcement 4666
7. check skill 99 pri l if available-agents > 0
```

For example, an English-speaking caller needs information on route planning and wants to speak to an agent with a New York accent. In this case, the caller dials the appropriate number, for example, 800-765-1616. Accordingly, the call enters Communication Manager and is directed to VDN 1616, which points to the vector in the previous screen. Once vector processing starts, the **queue-to skill** command in step 1 queues the call to the skill group that corresponds to the 1st VDN skill (New Yorker-16). If an agent with skill 16 is available, this agent answers the call. If such an agent is not available, the call is eventually queued to the skill group that corresponds to the 2nd VDN skill (English-44) by the **queue to main skill** command in step 3. This time, if an agent with skill 44 is available, this agent answers the call. If the call is still not answered, the **check skill** command in step 5 attempts to queue the call according to the parameter indicated (if calls-queued < 3) to the skill group that corresponds to the 3rd VDN skill (Route Planning-Bilingual-77). If the call is queued, and if an agent with skill 77 is available, this agent answers the call. If the call is not queued, or if it is queued and an agent with skill 77 is not available, the **check skill** command in step 7 is executed.

Before discussing the execution of step 7, note that a specific skill hunt group number (99) and not a VDN skill Preference designation (1st, 2nd, or 3rd) is included within the **check skill** command. Since the skill table for the application involves four levels of skills, and since there can be no more than three VDN skills, the specific skill group number (99) for the super agent pool must be included within the queuing command to allow caller access to the pool. Whereas a VDN skill is always represented in a vector by the term 1st, 2nd, or 3rd, a super agent pool is always represented by a whole number according to the parameters of the relevant switch. For the queuing commands, see Call Vectoring commands.

Returning to the vector execution, the **check skill** command in step 7 attempts to queue the call according to the parameter that is indicated (if available-agents > 0) to the super agent pool (Supergroup-99). If the call is queued, and if an agent in the super agent pool is available, this agent answers the call.

Note:

If the call has already queued to all three VDN skill hunt group preferences, it does not queue to the specific skill hunt group. This reflects the restriction that a call can only queue to a maximum of three splits or skills. The best approach is to test the splits or skills first to determine where to queue the call.

Routing calls to an agent

With you set **EAS** to `y`, an agent is associated at login with more than one skill hunt group. A single set of work mode buttons applies to all the skills assigned to a logged-in agent. For example, if the agent selects AUX work, the agent is in the AUX work mode for all the skills associated with the agent. Therefore, logged-in agents require only a single set of work mode buttons for all relevant skill hunt groups.

Calls can be routed to the agent from a skill hunt group by dialing an agent login ID or by dialing an agent phone extension directly.

Delivery from a skill hunt group

An incoming call is matched to an agent who has one of the three VDN skills required to handle the call. This match is done by ACD queuing and the **queue-to skill, check skill,**

`messaging skill,` or **`converse-on skill`** commands in the vector. If more than one agent is available for a call, the call is delivered according to whether EAD or UCD is administered for the skill hunt group.

For any one login session, an agent can have a maximum of four skills, or a maximum of twenty skills with EAS-PHD. Each agent skill is administered with a skill level.

Remember that when the **Call Handling Preference** is administered as `greatest need`, the agent receives the highest priority oldest call waiting for any of the agent's skills. If the **Call Handling Preference** is `skill-level`, the ACD software distributes the call that is waiting for the agent's highest skill-level skills when the agent becomes available. If no calls are waiting for the highest skills, the queued calls for the next highest skills are distributed to the agent. The following scenario describes call distribution when the **Call Handling Preference** is `skill level`.

Once an agent becomes available, the agent receives a waiting call in the following order:

1. Oldest DAC waiting for the agent if the direct agent skill is administered at the highest skill level of the agent.
2. Oldest call waiting at the highest priority for the highest skill-level skill.
3. Oldest call waiting at the next highest skill-level skill.

For example, Jill is the only agent with skills 22 (L1), 13 (L1), 23 (L1) and 47 (L2). While Jill is in the AUX work mode, five calls are queued, as shown in the following table, which also shows the skill level and priority level associated with each call.

Example of skill call queue sequence			
Call	**Time in queue**	**Skill number**	**Priority level**
A	8:00	13	Medium
B	8:01	47	Top
C	8:02	23	Direct Agent
D	8:03	22	Top
E	8:04	22	Medium

Given this scenario, the next table indicates and explains the order in which Jill handles the five calls.

Example of skill call distribution for a single agent	
Call handled	**Reason**
C	Only direct agent call queued at highest level skill.

Table continues…

Example of skill call distribution for a single agent	
Call handled	**Reason**
D	Oldest call waiting at the highest priority for highest skill-level skills (Call B has the same priority level (Top), but it is assigned a lower skill level (47). Also, Call E has the same skill (22), but it has a lower priority level (Medium) and has not been waiting as long as Call D).
A	Oldest call waiting at the highest priority level for highest skill-level skills (Call E also has a primary skill (22) and the same priority level as Call A, but Call A has been waiting four minutes longer than Call E).
E	Only remaining call with the highest skill level (22) (Call B has a lower skill level (47)).
B	Last remaining call, and the only one that has the lower skill level (47).

If no calls are waiting when an agent becomes available, the agent is placed into the agent queue according to the call distribution method that is in effect. For UCD, the agent is placed at the bottom of the most-idle agent queue. For EAD, the agent is placed at the bottom of the agents with the same skill level.

The following table shows a call scenario that is valid for either UCD or EAD.

Example of UCD/EAD call scenario		
Time	**Event**	**Skills**
9:00	Jill logs in	22(L1), 13(L1), 47(L2)
9:01	Jill available	22(L1), 13(L1), 47(L2)
9:02	Jack logs in	22(L1), 47(L1)
9:03	Jack available	22(L1), 47(L1)
9:04	Call A arrives	47
9:05	Call A drops	47
9:06	Call B arrives	13
9:07	Call B drops	13
9:08	Call C arrives	22

Given the scenario, the following table shows how Calls A, B, and C are distributed by UCD and EAD.

Example of call distribution by UCD and EAD			
Time	**UCD or EAD**	**Result**	**Reason**
9:04	UCD	Jill receives Call A.	Jill is the most idle agent for skill 47.
	EAD	Jack receives Call A.	Jack is the more expert agent because he has skill 47 as a level 1 skill whereas Jill has skill 47 as a level 2 skill.
9:06	UCD	Jill receives Call B.	Jill is the only agent who is logged in to skill 13.

Table continues…

Example of call distribution by UCD and EAD			
Time	**UCD or EAD**	**Result**	**Reason**
	EAD	Jill receives Call B.	Jill is the only agent with skill 13.
9:08	UCD	Jill receives Call C.	Jill is the most idle agent for skill 22. She receives Call C even if she handled Call A.
	EAD	Jill receives Call C.	Both Jill and Jack have skill 22 as a level 1 skill, but Jill has been logged in 2 minutes longer than Jack; that is, she is the most idle agent.

ACD login ID dialing

The ACD login IDs used in EAS are extension numbers that are included in a station numbering plan but not administered as stations. The IDs are administered on the Agent Login ID screen as shown in the following example. If **EAS-PHD** is set to `n`, you can administer only four skills.

Agent Login ID screen

```
add agent-loginID 9011                                        Page  1 of  1
                            AGENT LOGINID

                Login ID: 9011_                                  AAS? _
                    Name: ____________________________         AUDIX? _
                      TN: 1__                        LWC Reception: spe
                     COR: 1                 AUDIX Name for Messaging: _______
           Coverage Path: ____    Messaging Server Name for Messaging: _______
           Security Code: ____                LoginID for ISDN Display? n
      Direct Agent Skill: ___                                Password: _________
Call Handling Preference: skill-level          Password (enter again): _________
       Service Objective? _                                Auto Answer: ______

   SN  RL SL  PA       SN  RL SL  PA       SN  RL SL  PA       SN  RL SL  PA
1: __  _  __  ___   6: __  _  __  ___  11: __  _  __  ___  16: __  _  __  ___
2: __  _  __  ___   7: __  _  __  ___  12: __  _  __  ___  17: __  _  __  ___
3: __  _  __  ___   8: __  _  __  ___  13: __  _  __  ___  18: __  _  __  ___
4: __  _  __  ___   9: __  _  __  ___  14: __  _  __  ___  19: __  _  __  ___
5. __  _  __  ___  10: __  _  __  ___  15: __  _  __  ___  20: __  _  __  ___

      WARNING:  Agent must log in again before skill changes take effect
```

With EAS, an agent login ID is associated with a specific phone only when the agent actually logs in at that phone. When the agent logs off, the association of the agent login ID with a specific phone is removed. If an agent does not answer a call or if the agent is logged out, the call goes to the busy points on the coverage path.

When the agent logs in, the phone display indicates the skill assignments of the agent.

The agent logs in by:

- Going off-hook or selecting a line appearance.
- Entering the login FAC or selecting the login AD button after the dial tone.
- Entering a 1-digit to 16-digit login ID after the dial tone.

  Note:

  If someone is already logged in at that phone, the agent hears an intercept tone.
- Entering the 0-digit to 9-digit password after the dial tone. This is optional.

Note:

If the agent uses a DCP phone, such as a CallMaster®, the password digits are not shown unless an abbreviated dial button is used. BRI phones show the password digits.

Once the login is accepted, confirmation tone is given. The skills that are assigned are displayed for five seconds on the phone display. If more skills are assigned than can be displayed, a plus (+) sign appears at the end of the display. If a skill is administered, but the agent was not logged in to the skill, the skill number is displayed with an asterisk (*). The previous login sequence allows an ACD call to be directed to a specific agent and to have that call tracked and treated as an ACD call.

When an EAS agent logs in to a station with the station administered for audible message waiting, the agent receives an Audible Message Waiting tone only when calls are waiting for the agent login ID extension. When the agent logs out, Audible Message Waiting tone then applies again to messages that are waiting for the physical extension. This field has no impact on whether an agent hears the EAS Login-ID Message Waiting tone during the login process.

The message waiting lamp by default tracks the status of messages that are waiting for the logged in EAS agent Login ID rather than messages for the physical telephone. The operation of the Message Waiting Lamp can be changed so that it tracks the status of messages that are waiting for the physical phone where the agent is logged in. For more information, see “Feature-Related System-Parameters screen” in *Administering Avaya Aura® Communication Manager*.

EAS considerations

Station User records cannot be shared between TTI ports and EAS login ID extensions. This causes a reduction in the number of possible EAS login ID extensions allowed by the system depending on the number of administered TTI ports. For example, if 2,000 TTI ports are administered, the maximum number of allowable EAS login IDs is reduced by 2,000.

EAS agent login IDs are also tracked for personal calls. CMS uses the first skill an EAS agent is logged into to track personal calls. If the first logged-in skill is unmeasured, CMS credits the agent login ID with the personal call, but no skill hunt group is credited with the personal call.

The system can have either splits or skill hunt groups but not both simultaneously. Non ACD hunt groups can exist with either splits or skills. Skill hunt groups are required when using EAS.

When you implement the EAS feature, read the following EAS considerations:

- With EAS, skill hunt groups replace splits. You cannot administer both skills and splits on the same Communication Manager. All ACD hunt groups must be administered as either splits or skills. If you enable **EAS**, all ACD hunt groups are skill hunt groups.
- With EAS, all skill hunt groups except for messaging-system hunt groups must be vector-controlled.
- With EAS, non-ACD hunt groups are allowed, but cannot be vector-controlled.
- 2

  Agent login IDs are extensions in the dial plan and decrease the total number of stations that can be administered.
- With EAS, agents have a different login procedure and a single set of work mode buttons, regardless of the number of skills that are assigned to the agents.

- Skill hunt groups can distribute a call to the most-idle agent (UCD) or to the most-idle agent with the highest skill level for that skill (EAD). In either of these cases, the call can route to the most-idle agent for the specified skill, or to the most idle agent in all of the skills. Direct Department Call (DDC) distribution is not allowed for skill hunt groups.
- With either UCD or EAD distribution, the system can be administered to deliver calls based on greatest need or skill level. This is the Call Handling Preference that is administered on the Agent LoginID screen. When calls are in the queue, greatest need delivers the highest priority oldest call waiting for any of the agent's skills. With skill level administration, the system delivers the highest priority oldest call waiting for the agent's highest level skill with calls in the queue.
- The EAS-PHD customer option adds additional capabilities to the basic EAS capabilities.
  - It increases the number of skills an agent can log in to from 4 to 20
  - It increases the number of agent skill priority levels from 2 to 16

For information about converting a basic call center to EAS, see the *Planning for an Avaya Aura® Call Center Elite Implementation* document.

EAS interactions

Unless otherwise specified, the feature interactions for skill hunt groups are the same as for vector-controlled splits.

Interaction	Description
Abbreviated Dialing	Abbreviated Dialing (AD) is used to log in or log out EAS agents. You can administer AD lists or buttons only for stations.
Add/Remove Skills	In an EAS environment, agents can use an FAC to add and remove skills during a login session. Other phone users with console permissions can add or remove an agent skill on behalf of the agent. Note that the ability to add or remove skills depends on whether a user has a COR that allows addition and removal of skills.
Administration Without Hardware (AWOH)	EAS login IDs are extensions without hardware. Login ID extensions require space in the dial plan.
Agent Work Mode States	With EAS, agents can be in a single work mode at a given time for all the assigned skills.
Assist	You can use the Assist feature with a skill hunt group. For example, where there is one supervisor per skill hunt group. When an agent presses the **assist** button, a call is placed to the supervisor associated with the skill.
AUDIX™	Calls to the EAS agent login ID can cover to AUDIX™.

Table continues…

Interaction	Description
AAS	If you administer a skill hunt group as AAS, you must also administer the EAS login IDs that are assigned to the skill as auto-available. When Communication Manager re-initializes, the login IDs are automatically logged in with the auto-in work mode. If any Communication Manager feature causes a change to the work mode to anything other to auto-in, the change attempt is denied. AAS is not intended for human agents.
Automatic Answering with Zip Tone	Automatic Answer can be administered for an agent login ID and a physical extension. When an agent logs in to the system, the settings of the agent login ID for auto answer override the settings of the physical extension, except when the **Auto Answer** field value is `station`.
Automatic Callback	You cannot activate Automatic Callback to an EAS agent login ID. You can activate Automatic Callback to the phone where the agent is logged in.
Call Forwarding	Skill hunt groups, which are vector-controlled, cannot be call forwarded. EAS agent login IDs cannot be forwarded, but the physical extension where the EAS agent is logged in can be forwarded.
Call Park	Calls cannot be parked on the skill hunt group extension.
Call Pickup	Skill hunt group extensions and EAS login ID extensions cannot be members of a call pickup group.
Class of Restriction (COR)	Skill hunt groups do have a COR, which is used if the skill hunt group extension is called directly. The COR for an EAS agent login ID overrides the COR of the physical extension where an EAS agent logs in.
Class of Service (COS)	EAS agents do not have a COS associated with their login ID. Therefore, the COS of the phone is not affected when an EAS agent logs in.
Directed Call Pickup	An EAS agent can use the Directed Call Pickup feature to pick up a call or have the calls picked up by another agent. The COR of the agent overrides the COR of the station where the agent is logged in. If both the station COR and the logged-in agent COR allow the call to be picked up using Directed Call Pickup, the agent picking up the call can use either the station extension or the agent login ID.
Displays - Phone	When an EAS agent logs in, the display shows the login ID and agent name, as administered on the Agent Login ID screen. Calls that the agent originates show the agent login ID and agent name at the receiving phone display. However, other users can view the name of the physical extension where the EAS agent is logged in. To do this, the user must be active on a call with the agent and must have a phone with an alphanumeric display and an **inspect** button. When the **inspect** button is pressed during a call to or from the EAS agent, the physical extension name of the agent is displayed.

Table continues…

Interaction	Description
Leave Word Calling (LWC)	Calls to the physical extension show the physical extension number and name on the originator's display. When an EAS agent is logged in to a station, the agent can only retrieve LWC messages left for that agent login ID. To retrieve LWC messages left for that station, the agent must log out. When an EAS agent is logged in to a station, the message lamp defaults to tracking the status of LWC messages waiting for the station. However, you can assign the message lamp to track the status of LWC messages waiting for the agent login ID.
Look Ahead Interflow	VDN skills are not sent to another ACD or PBX when a call interflows using LAI. If skills have the same meaning on both ACDs, an LAI command to a VDN with the same skills assigned provides a mapping of the skills.
Message Waiting Lamp	The Message Waiting Lamp tracks the status of messages waiting for the logged in EAS agent login ID rather than messages for the physical extension. The operation of the Message Waiting Lamp can be changed to track the status of messages waiting for the physical extension where the agent is logged in. Setting the **Message Lamp Ext** field on the station to an extension other than the station default overrides the agent login ID system option for the station. The lamp does not track the logged-in agent.
Queue Status Indications	Physical extensions can be administered with Queue Status Indicator buttons and lamps for skill hunt groups. Queue Status Indicators can be administered for all skills required by agents using that physical extension, provided buttons are available.
Service Observing	Service Observing is activated in the EAS environment by dialing either the physical extension of the phone where an EAS agent is logged in or the login ID of the agent.
Tenant Partitioning and Agent Skills	Tenant Partitioning is designed to support multiple customers using the same Communication Manager. Tenant Partitioning separates entities reducing the interactions between entities in different partitions. Assign the same partition number to agents, groups, and entities to prevent calls from blocking and to prevent any unexpected interactions that result from mixing tenant partitions. When Tenant Partitioning is active and used for restriction of service, assign the same partition number to: • ACD agents. • Hunt groups, splits or skills. • Other entities involved with ACD agents and hunt groups such as VDNs and announcements. **Note:** An agent skill set must contain skills belonging only to the *same* tenant partition. Not doing so can result in unintended behavior.
VuStats	VuStats displays can show an agent skill assignment and some measurements by skill.

EAS feature interactions

Unless otherwise specified, the feature interactions for skill hunt groups are the same as that for vector-controlled splits.

Interaction	Description
Abbreviated Dialing (AD)	Use Abbreviated Dialing (AD) to log in or log out EAS agents. You can administer AD buttons only for stations.
Administration Without Hardware (AWOH)	Although EAS login IDs are extensions without hardware, the IDs are not a part of AWOH.
Agents in multiple splits feature	With EAS, the agents in the multiple splits feature is called *Agents in Multiple Skills*. With the feature, an EAS agent can log in to multiple skills.
Agent work modes	With EAS enabled, an agent can be in a single work mode for all skills at any one time. For example, an agent cannot be in the AUX work mode in one skill hunt group and also available in another skill hunt group. Also, if the ACW mode button is selected, the agent is placed in ACW for the first skill that is administered and logged in to.
Assist	Use the **assist** button for skill hunt groups, that is, when there is one supervisor per skill hunt group. You can administer a telephone with more than one **assist** buttons for each skill. You can also administer an **assist** button with no associated skill. In this case, the supervisor for the skill that the agent is currently active on is called. If the agent is not active on any skill, the supervisor for the agent's first skill is called. The **assist** button selected is tracked as an assist for the current call, regardless of the skill assigned to the button. The administered association of an **assist** button with a particular skill and assigned supervisor is not affected when an EAS agent logs in to the station.
Audible message waiting	If messages are waiting for an EAS agent login-ID extension, an agent hears a special 5-burst EAS Login-ID Message Waiting tone, instead of a confirmation tone, after successfully logging in. This does not require Audible Message Waiting to be assigned to the telephone or the system. If Audible Message Waiting is enabled and assigned to an agent's telephone and messages are waiting for the agent login ID extension, the agent hears the Audible Message Waiting tone whenever the agent goes off-hook, or selects a line appearance and hears dial tone. Messages that are waiting for the physical extension do not cause an Audible Message Waiting tone when an EAS agent is logged in.
Auto-Available Skills (AAS)	If a skill hunt group is administered as an AAS, the EAS login IDs assigned to the skill must also be administered as Auto-Available. When the switch re-initializes, the login IDs are automatically logged in with the auto-in work mode. If any switch features attempt to change the work mode to anything except auto-in, this attempt is denied. Agents cannot have both Auto-Available and Non-Auto-Available Skills. This feature is not intended for human agents.

Table continues…

Interaction	Description
Automatic answering with zip tone	Automatic Answer can be administered for an agent login ID and a physical extension. When an agent logs in to the system, the settings of the agent login ID for auto answer override the settings of the physical extension, except when the **Auto Answer** field value is `station`.
Basic Call Management System (BCMS)	The BCMS user interface remains the same when EAS is enabled. The only change is that the labeling of the headings is changed from split to skill. When EAS is enabled, BCMS agent reports are based on the agent login IDs. BCMS tracks DAC calls as skill calls. DACs affect ACD talk time, ACW time, and ASA. When DACs are waiting, BCMS displays an asterisk (*) immediately after the CALLS WAITING column.
Best Service Routing (BSR)	EAS VDN skills, 1st, 2nd, and 3rd, can be used in the **`consider`** split or skill commands. EAS skills levels are used for the EAD-MIA and EAD-LOA BSR Available Agent Strategies.
Bridging	ACD calls do not alert on bridged appearances. However, bridged users can activate features on behalf of agents. Features that can be activated include log in, log out, change work modes, and assist.
Call Coverage	Call Coverage can occur whether or not an agent is logged in. If the agent is not logged in, the busy criteria is met and the call follows the points on the coverage path. If the agent is logged in but fails to answer, the don't answer criteria is met and the call follows the points on the coverage path. A call to the login ID goes to the coverage path that is assigned to the login ID rather than to the coverage path that is assigned to the telephone extension.
Call Detail Recording (CDR)	For skill calls, the **called party** field can optionally be the agent login ID.
Call Forwarding	Since skill hunt groups are vector-controlled, skill hunt groups cannot be call forwarded. EAS agent login IDs cannot be forwarded, but the physical extension where the EAS agent is logged in can be forwarded. If another station with console permissions tries to forward an EAS login ID, an intercept tone is given.
Call Park	To retrieve a parked call by an FAC, the agent dials the Answer-Back FAC and the extension where the call is parked. If the person who is unparking the call dials the Answer-Back FAC and the physical extension of the station where the call is parked, the person is connected to the parked call. In some cases, the person who is unparking the call can also be able to dial the Answer-Back FAC and the logical agent extension of the agent who parked the call. This operation is possible if the COR of both the agent parking the call and the telephone or agent who is unparking the call have a COR with the **DAC** field set to `y`. If the telephone that is unparking the call is not a logged-in agent, the telephone must have a COR with **DAC** set to `y`. If the station that is unparking the call is a logged in agent, then the COR of the logical agent extension must have **DAC** set to `y`.
Call Pickup	Skill hunt group extensions and EAS login ID extensions cannot be members of a call pickup group.

Table continues…

Interaction	Description
Class of Restriction (COR)	Skill hunt groups do have a COR. The COR is used if the skill hunt group extension is called directly. The COR for an EAS agent login ID overrides the physical extension's COR of the telephone that an agent logged in to.
Class of Service (COS)	EAS agents do not have a COS associated with their login ID. Instead, the COS is associated with the physical extension. Therefore, the COS of the telephone is not affected when an EAS agent logs in to that telephone.
Dial Plan	Agent login IDs are part of the dial plan reducing the total number of stations.
Direct Agent Calling (DAC)	If a called EAS agent login ID and the call originator, that is extension, trunk, or VDN, both have a COR that allows DAC calls, the call to the login ID is treated as a DAC call. A call to the telephone extension where an EAS agent is logged in, or a call to an EAS agent login ID where either the originator's or the login ID's COR does not allow direct agent calls, is treated as a personal or non ACD call.
Leave Word Calling (LWC)	When an EAS agent is logged into a station, the agent can only retrieve LWC messages left for the agent login ID. To retrieve LWC messages left for the station, the agent must log out. When an EAS agent is logged into a station, the message lamp defaults to tracking the status of LWC messages waiting for the station. However, you can assign the message lamp to track the status of LWC messages waiting for the agent login ID.
Look-Ahead Interflow (LAI)	Skills are not sent to another system when a call interflows using LAI. If skills have the same meaning on both ACDs, a LAI command to a VDN with the same skills assigned can provide a mapping of the skills.
Multiple Split Queuing	When EAS is enabled, the Multiple Split Queuing feature is called *Multiple Skill Queuing*, which has the same functionality. With multiple split or skill queuing, a call can queue to a maximum of three splits or skills.
Outbound Call Management (OCM)/ Expert Agent Selection (EAS)	If EAS is enabled, the Outbound Call Management (OCM)/EAS feature is required for a CallVisor Adjunct Switch Application Interface (ASAI) adjunct application to launch predictive OCM calls. Predictive Calling is an OCM feature often used in applications, such as sales or cold calling, where it does not matter which agent is accessed by a caller. While OCM predictive calling is an OCM application, the EAS environment provides a number of desirable features for inbound call handling. With OCM/EAS, you can enable both types of call handling. From a technical standpoint, if EAS is enabled, the feature is required for the following reasons: • All skill hunt groups are vector controlled. However, to launch a predictive OCM call in a traditional ACD environment, the ACD split cannot be vector-controlled. • The traditional ACD environment and EAS cannot be enabled at the same time. OCM/EAS extends the ASAI features to include launching predictive OCM calls from a VDN extension. Previously, ASAI hosts launched predictive calls only from ACD split extensions. A limited number of Call Vectoring commands are supported in the VDNs used to launch or process OCM predictive calls.

EAS and adjunct interactions

ASAI interactions with EAS

ASAI support for EAS is organized in the following categories:

- Call control
- Feature requests
- Value queries
- Event notification
- Adjunct-controlled skills

This section provides a high-level overview of the behavior of ASAI in the EAS environment.

Call control

Call control capabilities work the same way in the EAS environment as in the traditional ACD environment except for the following:

- User-classified third-party make calls, that is, calls classified by the originator, originate from an EAS login ID and terminate to a login ID. User-classified calls that terminate to a login ID are given the same direct agent treatment that is provided for such calls that are dialed from a station extension.
- Switch-classified third-party make calls, which are classified by a call classifier board and delivered to the originating hunt group, originate from or terminate to EAS login IDs.
- Direct agent third-party make calls, which are ACD calls terminated to a selected member of an ACD skill group, can be requested by including a direct agent option, an agent physical extension and a skill group extension (compatibility mode), or by requesting a user-classified third-party make call with a login ID destination. The primary differences between the two methods of requesting direct agent calls are that the compatibility mode allows the adjunct to specify the skill hunt group to which a given direct agent call is queued and that the non compatibility mode allows the adjunct to direct the call to a login ID, regardless of which station an agent is logged in to. Direct agent third-party make calls do not always originate from an EAS login ID.
- Supervisor assist third-party make calls, which are supervisor assist calls that are originated by a selected member of an ACD split, originate from an EAS login ID and terminate to an EAS login ID. Unlike dialed direct agent calls, supervisor assist calls terminated to a login ID behave as though the calls have been previously directed to the physical extension of the requested agent. For example, the calls do not cover if the requested agent is not logged in and if the originator display shows the physical extension and not the login ID of the agent.
- Extension (Domain) control cannot be requested for an EAS login ID, but can be requested on behalf of a Logical Agent physical extension. Auto-dial calls, which are calls that are initiated by an extension-controlled station, terminate to an EAS login ID, in which case the call is given direct agent treatment.
- Adjunct-routing calls, which are vector calls routed by an ASAI adjunct by the **`adjunct routing link`** Call Vectoring command, are similar to third party make calls. Such calls include a direct agent option, an ACD agent physical extension, and a skill extension. These

calls are given compatibility mode direct agent treatment and terminate to an EAS login ID, in which case the calls are treated as dialed direct agent calls.

- If you enable **EAS**, ASAI launches the OCM switch-classified or predictive calls from a VDN extension by the OCM/EAS feature. To launch a predictive call in a traditional ACD environment, an adjunct OCM application sends an ASAI request to Communication Manager with an ACD split number as the originating number. The application also sends flags that identify the call as a switch-classified call. In the traditional ACD environment, the ACD split cannot be vector-controlled.

Feature requests

EAS supports agent login, logout, and change in work mode requests. Agent login requests must contain an EAS agent login ID and an optional password delimited by the pound (#) sign in the login request user code IE. Agent logout requests and work mode change requests can contain the physical extension or login ID of the agent.

Call Forwarding and Send all Calls (SAC) feature requests are denied for EAS login IDs but can be requested for EAS physical extensions where an EAS agent is logged in.

Multiple monitors

Multiple Monitors provides the option for up to three ASAI applications to monitor the same ACD split or VDN domain.

This is useful in environments where OCM is primary and can also be used to add an OCM application to launch calls at off-peak times without disrupting the primary application. Multiple Monitors can also be used to monitor an ACD split over 2 links in call environments where ASAI link failure recovery is important.

Value queries

Value queries functions identically in the EAS and traditional environments, except that the Extension Type/Class Information Query returns a new indication that a requested extension is an EAS login ID along with an indication of whether the login ID is currently logged in and where, in other words, at which physical extension.

Event notification

As all skill hunt groups are vector-controlled, you cannot request for an event notification on the basis of a skill hunt group extension. However, you can request for an event notification on the basis of a controlling VDN extension. Most event reports that involve EAS agents contain the physical extension of the agent instead of the agent login ID.

Adjunct-controlled skills

Agents with adjunct-controlled skills are treated as adjunct-controlled agents. Adjunct-controlled agents exhibit the same behavior as agents within adjunct-controlled splits in the traditional ACD environment. The following list provides more details:

- Stations are locked for all logged-in adjunct-controlled agents. The only action an agent can take from the station is to go on hook (or unplug the headset) from an auto-answer station, which causes the agent to be logged out.

- Stations are unlocked whenever the controlling adjunct's ASAI link stops functioning. Stations are locked again when the adjunct's link is reestablished.
- The adjunct controls all skill and agent activities such as login, logout, and change work-mode (with the exception of agent logout using the telephone hook).
- Only adjunct-controlled calls can terminate to the extension of an adjunct-controlled agent.
- Only adjunct-controlled calls can terminate to an adjunct-controlled skill hunt group extension.
- Adjunct-controlled EAS Agents can be administered with only one skill. Accordingly, EAS agents cannot mix adjunct-controlled and non-adjunct-controlled skills.

Messaging system

Calls to the EAS agent login ID can cover to the messaging system. Each agent must enter the agent login ID when calling the messaging system to receive messages.

Messaging system agents are assigned to EAS agent extensions. The login IDs are used for reporting adjunct tracking if the associated messaging system skill hunt group is externally measured. The **aut-msg-wt** button or message waiting light can be used to indicate that the login ID has a message.

An agent cannot have both messaging system and non-messaging system skills.

CMS

Note:

CMS reports show only the first 15 skills that an agent is logged into.

The following items apply to Avaya CMS Agent Tables:

- Separate direct agent database items starting with DA_ are tracked.
- Standard reports combine statistics for direct agent calls and skill calls. However, reports can be customized to separate these statistical groupings.

The following is true for the CMS Skill Tables:

- Skill queues can be monitored for direct agent calls on the Queue/Agent Summary report.
- Direct agent calls are not tracked.
- Agent time while on a direct agent call is tracked as other time.
- Non-ACD calls while in direct agent ACW are tracked.

The following item is true for the CMS VDN/Vector Tables:

- Direct agent calls and skill calls are combined as ACD calls.

Speech-processing adjuncts

Speech-processing adjuncts that have a line interface to the Communication Manager can initiate direct agent calls by dialing the agent login ID.

When administering a split or skill, you can type `list members hunt-group` to verify that all agents are logged out and to identify any agents who are logged in. You can list all logged in agents for a split or skill, or limit the list to a range of login IDs or physical extensions:

- `list members hunt-group <hunt group nnn>`
- `list members hunt-group <loginid nnnn to-loginid nnnn>`
- `list members hunt-group <ext nnnn to-ext nnnn>`

For example, to list the agents logged in to skill 37:

1. Enter `list members hunt-group 37` and press `Enter`.

   The List Hunt Group Members report screen appears.

```
list members hunt-group 37

                           HUNT GROUP MEMBERS

Group Number: 37     Group Name: Platinum Card    Group Extension: 3002
  Group Type: ucd-mia       ACD? y     Skill? y           Members: 4

    Phys  Phys            Login Login          Agt     Per       Wrk
    Ext   Name            Ext   Name           Prf Lvl All  SO DF Tim Occ
 1: 1002  1002-Al MacInni 2902  Agent 2902     grt 04      y      10  33
 2: 1022  1022-Kelly Chas 2901  Agent 2901     lvl 14      n      15  55
 3: 1001  1001-Chris Pron 2904  Agent 2904     pal R2         n    0   0
 4: 1021  1021-Maria Esta 2903  Agent 2903     pal 08   30    y   18  45
```

   Note:

   This screen shows a system using EAS and Avaya Business Advocate. For systems without these features, the related columns are blank.

You can also use this command to list the agents *administered* in non-ACD hunt groups. However, since non-ACD hunt groups do not use agent login IDs, the report does not identify agents who are currently active.

Forced Agent Logout by Clock Time

With Forced Agent Logout from Clock Time, administrators can:

- Set a specific time when Communication Manager automatically logs out EAS agents.
- Set a logout reason code for agents on a systemwide basis.
- Administer Communication Manager so that agents can override the forced logout when agents press a forced logout override button.

If the agent is still on an ACD call when the forced agent logout time is reached, the agent is put into the pending logout mode. In pending logout mode, the forced logout override button flashes, and the agent hears a repeating tone. The forced logout occurs when the call is disconnected.

Forced Logout Override

Agents can perform any of the following actions to cancel Forced Logout:

- Press **logout-ovr** when on an ACD call.
- Press **logout-ovr** anytime during the shift. The forced logout time resets for the next day.
- Log in again. The forced logout time resets for the next day.

Note:

The agent can toggle the forced logout override button to remove the override.

The forced logout override button flutters in one of the following conditions:

- The administrator does not configure the logout time on the Agent LoginID screen, but the agent presses the forced logout override button.
- The forced logout time elapses, and the agent presses the button.

Forced Agent Logout from Clock Time example

The ABC call center has groups of agents in several different time zones. The shift log out times vary based on the local time for each of the groups of agents. Agents sometimes leave their position without logging out at their designated time.

The ABC call center wants to use the Forced Agent Logout by Clock Time to set a TOD for automatically logging out such agents. Forced Agent Logout by Clock Time will be set up for some of the agents who have in the past forgotten to logout.

In this example, the main system clock is in the central time zone. The following table shows the assignments required for the Forced Agent Logout by Clock Time feature.

Agent	Login ID	Local logout time	Forced Agent Logout After field setting on the Agent Login ID screen	Physical station	Location for physical station (used for time zone offset and DST rule)
1	1001	8:10 am	08:15	3001	2
2	1002	5:30 pm	17:30	3002	2
3	1003	12:15 pm	12:15	3003	1
4	1004	4:50 am	05:00	3004	7
5	1005	5:45 pm	17:45	3005	7

Set the Feature Related System Parameters field **Clock Time Forced Logout Reason Code** to 2. This sets logout reason code 2 to be used for this type of logout.

To help you understand how the local time will be determined, the following table shows an example of the locations screen setup for this configuration. Normally the locations screen setup will be part of the original system and station set configuration, and will not need to be configured as part of the Forced Agent Logout by Clock Time feature.

Location	Name	Time zone offset	DST rule
1	Main (in Central Time)	+ 00:00	1
2	Mountain time location (Phoenix)	- 01:00	0
...			
7	Eastern time location	+ 01:00	3

Note:

The time zone offset is defined relative to the main location.

Forced Agent Logout from Clock Time interactions

Only the features that are impacted by the Forced Agent Logout from Clock Time feature are described in this section.

Interaction	Description
Automatic agent logout and login	If Forced Agent Logout by Clock Time and automatic logout and login of agents are pending, Forced Agent Logout by Clock Time takes precedence over automatic logout and login. However, if an agent presses **logout-ovr** to cancel the Forced Agent Logout by Clock Time, automatic logout and login is applicable.
Call Hold	If the agent has ACD or DAC calls on hold when the forced logout time is reached, the agent is put into a pending logout mode. If the agent has non-ACD calls on hold, the forced logout occurs at the assigned time.
Call Work Codes or Stroke Counts	If a forced logout time is reached when the agent is in the process of entering a Call Work Code (CWC) or Stroke Count and the digits message is not sent to reporting adjunct, the message is aborted and the CWC session is closed. Even if the setting of the CWC is forced, the agent is allowed to be logged out during ACW and this log out takes precedence over the CWC entry. An agent on an ACD call remains connected while in pending logout mode and can complete entering and sending the CWC or stroke count.
Caller Info Display	When a forced logout occurs, the Callr-Info from collected digits that is displayed on the station is cleared.
Conference	A forced logout occurs when an agent is in a conference call unless it is an ACD call that came directly to the agent, or if the agent has an ACD call on hold. If either of the conditions exist, the agent is put into pending logout mode while remaining connected to the conference.
DAC	Forced Agent Logout by Clock Time is applicable to an agent handling a Direct Agent Call (DAC) after the DAC is released.
Forced Logout Override	Forced Logout Override is initiated when an agent presses the forced logout override button, causing the lamp to light. If the button is on, that is, lamp is lit, the system does not log the agent out when the forced logout time arrives.

Table continues…

Interaction	Description
Improved Integration with Proactive Contact	Do not use the Call Center 4.0 Forced Agent Logout by Clock Timer feature with the Call Center 4.0 Improved Integration with Proactive Contact Outbound Calling capability. The Proactive Contact system places a PC agent in the AUX work mode when the agent is making an outbound PC-imitated call. If the administered time for Forced Agent Logout by Clock Time is reached, the PC agent is logged out immediately.
Manual-In Mode, Pending ACW, and Timed ACW	A pending forced agent logout takes precedence. When the ACD call is released, the agent is logged out and not put into the ACW state.
Multiple Call Handling	Agents not on an ACD call or with ACD calls on hold are logged out when the administered time for forced logout occurs. If the agent has ACD calls on hold, the agent is put into pending logout mode. The calls remain connected and the agent is not logged out and reported to reporting adjunct until all of the ACD calls are disconnected.
Multiple Locations Feature	The forced logout is specified in the local time for the agent station. The local time for the agent is determined using the assigned multiple locations feature location number for time zone offset and Daylight Saving Time (DST) rule. The time zone offset and DST rule assigned to the location number for the agent is applied to the main switch clock time, which also has an assigned DST rule, to determine the current time local to the agent. If the multiple locations feature is not active, the default location 1 is used. In this case, the time zone and DST rule assigned to the main location is used to determine local time.
Non-ACD call connection	A forced logout occurs at the assigned time when an agent is connected to any incoming or outgoing destination that is not an established ACD, ACDO, or DAC call. The reporting adjunct sees the agent as logged out and does not record any subsequent actions from the station. No other events are logged even if the agent is in the middle of dialing, being called by an ACD or non-ACD call, on a trunk, hearing an announcement, in vector processing, or queued to an ACD or non-ACD hunt group.
Pending Logout Mode	If the agent is still on a call when the forced agent logout time is reached, the agent is put into pending logout mode. The forced logout override button flashes and the agent, a service observer or anyone else connected to the agent in a conference hears a repeating tone. The caller cannot hear this tone. This tone has precedence over any other tones including: Service Observing tones, call-waiting tones, music on hold, or conference tones. When the call is released or disconnects, the forced logout occurs.
Reason Codes	A specific logout reason code as defined on the Feature-Related System Parameters screen is sent to the reporting adjuncts and BCMS/VuStats as the reason for the forced logout.
Supervisor Assist	A forced logout does not occur if a supervisor is logged in as an agent and assisting an agent with an ACD call that the agent received. The logout does not occur because the forced logout is not directed at the supervisor’s login.

Table continues…

Interaction	Description
Transfer	If an agent is in the process of transferring an ACD call when a forced logout time is reached, the agent is put in pending logout mode. The forced logout occurs when the transfer is completed. If the agent is transferring a non-ACD call when a forced logout time is reached and there are no ACD calls on hold, the forced logout occurs at the assigned time.
VuStats	Active VuStat sessions reflect the system-assigned forced agent logout reason code when displaying the statistics for that reason code.

Forced Agent Logout/AUX Work by Location/Skill

Use Forced Agent Logout/AUX Work by Location/Skill to force all agents in a location or skill to log out of the system or enter the AUX work mode.

Note:

The feature does not apply to Auto-Available Splits/Skills (AAS).

Forcing agents to log out of the system or enter the AUX work mode is a two-step process that a supervisor can perform directly from a phone by entering a Feature Access Code (FAC) or a Vector Directory Number (VDN):

1. A supervisor, using an administered station, enters the administered FAC to perform one of the following tasks:

Feature Access Code (FAC)	Action
Forced Agent Logout by Location	Logs out all the staffed agents in a particular location.
Forced Agent AUX Work by Location	Puts all the agents in a location in the AUX work mode.
Forced Agent Logout by Skill	Logs out all the agents who are logged in to a particular skill.
Forced Agent AUX Work by Skill	Puts all the agents in a skill in the AUX work mode.

   When the supervisor enters the FAC for Forced Agent AUX Work by Location, the system validates the location number. The location digit string is not validated against the list of administered location numbers.
2. After entering one of the FACs on a phone, the supervisor hears another dial tone. The supervisor then enters the location or skill number of the agents to log out or to put the agent in the AUX work mode. For example, if the FAC for Forced Agent Logout by Location is *46, and the location is 9, the supervisor dials *46, waits for a second dial tone, then dials 9#. The supervisor hears a confirmation tone.

To access an FAC that performs Force Agent Logout/AUX Work by Location/Skill:

1. Create a vector that contains a **`route-to number`** vector step. The vector step references the relevant FAC. The vector can also contain, for example, vector steps to

prompt the supervisor for a password, and steps to prompt for and collect the location or skill number.

2. Create a VDN that references the administered vector.
3. The supervisor calls the VDN.

When the forced logout operation occurs, if an agent is on an ACD or DAC call, or has ACD/DAC calls on hold, the agent hears the forced logout tone and is put in the Pending Logout mode until the agent releases all calls. However, if the agent is on a non ACD call or has only non ACD calls on hold, forced logout occurs immediately, whereas a forced AUX work mode change remains pending until the agent releases all the non ACD calls. The agent stays connected to the non ACD call after being logged out.

For the forced logout or AUX work features, an agent is treated to be in a particular skill if logged in to the skill at the time the feature request is made. Pending logout applies only to those agents.

Note:

To assign the feature to multiple agents, assign a common skill number to the agents. Use the common skill number to force logout or change to the AUX work mode.

The supervisor can view entries of successful Forced Logout/AUX Work mode by Location/Skill events on the Events Report screen. The category for Forced Logout/AUX Work mode events is feat_event, Event Data 1 is the location or skill number.

```
display events

                            EVENTS REPORT

Event Event                        Event     Event      First        Last     Evnt
Type  Description                  Data 1    Data 2     Occur        Occur     Cnt

3800  Frcd lgout by loc invoked 1          0         12/03/16:42 12/03/16:42  1
3801  Frcd lgout by skl invoked 501        0         12/03/16:42 12/03/16:42  1
3802  Frcd aux by loc invoked   8          0         12/03/16:42 12/03/16:42  1
3803  Frcd aux by skl invoked   42         0         12/03/15:47 12/03/15:47  1
```

Related links

Location numbers on page 159

Logging out all the agents in a location

Before you begin

The Class of Restriction (COR) for the station which the supervisor uses to force an action must have the permission to do so, as indicated by the **Can Force a Work State Change** field.

Each agent whose work state is being changed must have the COR **Work State Can Be Forced** field set to `y`.

Forced Logout by Location applies only to the server on which you administered the feature. If the system is fragmented into multiple servers [Main/Local Survivable Processor (LSP) / Enterprise Survivable Server (ESS)], you must administer the feature separately on each individual server.

About this task

By completing the following steps, a supervisor can log out all the agents in the location 3.

Procedure

1. A supervisor, using an administered workstation or a **route-to number** vector step, dials the FAC for forced logout of agents in a location: `*46`.
2. The supervisor dials the location code with # (this terminates the digit string) suffix: `3#`.

Result

All the agents logged into location 3 are logged out.

Remotely forcing all agents in a location/skill to logout/AUX work mode

Before you begin

1. You must set up the Feature Access Codes (FAC). For information, see the *Administering Avaya Aura® Call Center Elite* document.
2. Assign a common skill or location number to all the agents to be forced to log out of the system or to move to the AUX work mode.
3. The Class of Restriction (COR) for the station which you are using to force an action must have the permission to do so, as indicated by the **Can Force a Work State Change** field.

About this task

Using the following steps, a supervisor can remotely force all agents in a location or skill to logout, or enter the AUX work mode:

Procedure

1. Create a vector that contains a **route-to number** vector step that routes to the relevant FAC. The vector can also contain, for example, vector steps to prompt the supervisor for a password and for collection of the location or skill number.

   For example, if the FAC is *22, the vector step is: `route-to number *22`
2. Create a VDN that references the administered vector.
3. Call the VDN.

   You will hear a second dial tone as a prompt to enter the location or skill.
4. Dial the appropriate location or skill number of the agents followed by the pound (`#`) sign.

Example

You can see entries of successful Forced Logout/AUX work mode by Location/Skill events on the Events Report screen. The category for Forced Logout/AUX work mode events is feat_event, Event Data 1 is the location or skill number.

```
display events

                              EVENTS REPORT

Event Event                        Event     Event      First       Last     Evnt
Type  Description                  Data 1    Data 2     Occur       Occur     Cnt
```

```
3800  Frcd lgout by loc invoked 1          0         12/03/16:42 12/03/16:42  1
3801  Frcd lgout by skl invoked 501        0         12/03/16:42 12/03/16:42  1
3802  Frcd aux by loc invoked   8          0         12/03/16:42 12/03/16:42  1
3803  Frcd aux by skl invoked   42         0         12/03/15:47 12/03/15:47  1
```

Note:

When forced logout or forced AUX work occurs, if an agent is on an ACD or DAC call or has ACD/DAC calls on hold, the agent is put in pending logout/AUX work mode until the calls are released. However, if the agent is on a non ACD call or has only non ACD calls on hold, forced logout occurs immediately while the agent remains connected to the non ACD call, whereas a forced AUX work mode change remains pending until any non ACD calls are released.

For forced logout and forced AUX work, an agent is treated to be in a particular skill if the agent is logged in to the skill at the time the feature request is made. Pending logout also applies only to those agents.

Remotely forcing all agents in a location/skill to logout/AUX work mode in one step

Before you begin

1. You must set up the FAC. For more information, see the *Administering Avaya Aura® Call Center Elite* document.
2. Assign a common skill or location number to all the agents to be forced to logout or to be put in the AUX work mode.
3. The COR for the station which you are using to force an action must have the permission to do so, as indicated by the **Can Force a Work State Change** field.

About this task

Using the following steps, a supervisor can remotely force all agents in a location/skill to logout or in the AUX work mode:

Procedure

1. Create a vector that contains a **`route-to number`** vector step that routes to the appropriate FAC. The vector can also contain, for example, vector steps to prompt the supervisor for a password as a security precaution and steps to prompt for and collect the location or skill number.

   For example, if the FAC is *22 and the location or skill is 1, the vector step is: `route-to number *221#`
2. Create a VDN that references the administered vector.
3. Call the VDN.

   All the agents logged in to skill or location 1 are logged out.

Forced Agent Logout from ACW

Forced Agent Logout from After Call Work (ACW) automatically logs out an EAS agent who spends more than the predetermined time in the ACW mode. You can specify the time out period on a per-system basis and can also choose to override this value on a per-agent basis. The time out is reported with an assignable reason code set on a per-system basis. The per-agent time out setting takes precedence over the per-system setting.

Note:

For auto-in agents, Timed ACW takes precedence over Forced Agent Logout from ACW.

Use Forced Agent Logout from ACW when you:

- Require that agents do not remain in ACW longer than a set time limit in order to monitor agents who exceed the time limit.
- Logout agents who walk away from their position while in the ACW mode.

Forced Agent Logout from ACW interactions

Only the features that are impacted by the Forced Agent Logout from ACW feature are described in this section.

Interaction	Description
Call Work Codes and Stroke Count	If the agent is in the process of entering a Call Work Code (CWC) or Stroke Count and the Forced Agent Logout from ACW timer expires before the Digits message is sent to CMS, the following actions occur: • The software aborts sending the message. • The CWC session is closed as the agent is being logged out. Even if the setting for CWC is forced, logging out of the agent is allowed and takes precedence over the CWC entry.
Direct Agent Calls	After an agent handles a Direct Agent Call (DAC), the Forced Agent Logout from ACW feature applies when the agent enters the ACW state after the DAC is released.
Multiple Call Handling	An agent in ACW is logged out because the Forced Agent Logout from ACW timer has expired, even if the agent has ACD calls on hold.
Timed ACW	Timed ACW immediately switches an auto-in agent into the ACW mode for a specific length of time after the agent disconnects from a call. • If the agent disconnects from a call while in auto-in mode, the Timed ACW settings apply and the agent is not logged out based on the Forced Agent Logout from ACW settings. • If the agent, after disconnecting from a call, uses the ACW button to enter ACW, or enters ACW while in Manual-In mode, the Forced Agent Logout from ACW feature settings apply.

Inbound Call Management

With Inbound Call Management (ICM), you can integrate Communication Manager features with host-application processing and routing, and automate the delivery of caller information on the display screen of the telephone. You can create a sophisticated system to handle inbound calls for applications such as telemarketing and claims processing.

To implement ICM, you must integrate Communication Manager features, such as ACD, Expert Agent Selection (EAS), Call Vectoring, Direct Agent Call (DAC), and Call Prompting, with an application on a host processor.

The host application or adjunct can be a CallVisor or a personal computer, an IVR system, or a telephony services server serving a local area network, or a vendor application using the CallVisor ASAI. A CallVisor ASAI link between Communication Manager and adjunct allows the adjunct to control incoming call processing and routing.

You can automate and associate an ACD agent telephone display with new and transferred calls, and assist calls to a supervisor. You can display incoming call information such as Calling Party Number (CPN), Billing Number (BN), and Dialed Number Identification Service (DNIS). You can also set up the adjunct to retrieve caller information from a database and display the information on an agent screen, based on the service dialed.

ICM applications

- Communication Manager passes the Calling Party/Billing Number (CPN/BN) information and routes the call to an adjunct application for screen pop and supervisory transfers with screen duplication.
- Communication Manager sends to the adjunct application both caller and prompter information on all incoming calls to a particular number. According to caller information in a database, the application directs Communication Manager to route the call. For example, the call can be routed to a preferred agent, to best customer treatment, or to accounts receivable.
- Communication Manager uses Call Prompting to retrieve a customer account number and passes the information to the adjunct for call routing or screen pop.
- Communication Manager connects the caller to a VRU, along with caller CPN/BN and DNIS information. The caller interacts with the VRU to direct how the call is handled. Communication Manager verifies the identity of the caller and provide access to database information such as claims status or account balance.
- With Direct Agent Call (DAC), an adjunct application can transfer a call to a specific ACD agent and have the call treated as an ACD call and tracked on CMS.
- An adjunct application can attach information used by another application to an ICM call using User-to-User Information fields. The adjunct transfers the call, along with the application-specific information, over Primary Rate Interface (PRI) trunk to a CallVisor ASAI application at another Communication Manager. For example, an application at one Communication Manager can determine the account or claim number of the caller and pass the information to a special list on another Communication Manager, where an application transfers the call.

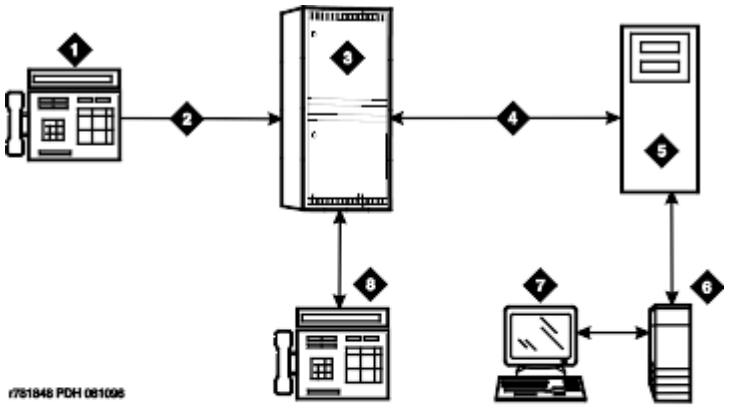
Agent data screen delivery applications

You can use an Interactive Voice Response (IVR) to deliver appropriate display data about callers to agents. You can pass information such as CPN/BN, DNIS, LAI information, digits collected from Call Prompting, and which agent is selected to an IVR system. The IVR delivers the appropriate data screen to the agent who takes the voice call. The IVR system can transfer or duplicate data screens for transferred or conferenced calls.

A simplified configuration for the use of an IVR for agent data screen delivery applications is shown in the following figure.

Note:

An IVR VIS is used as an example - other adjunct processors have similar capabilities but must be verified for a particular application. If the host supports ASAI, the IVR system is not needed.



1. Telephone
2. ISDN-PRI
3. Avaya switch
4. ASAI
5. IVR
6. Host
7. Agent data terminal
8. Agent telephone

General processing for this type of application occurs as follows.

1. An IVR system or host requests notification for events such as call offered, call ended, call connected, call dropped, call transfer, and alerting.

2. The communication server notifies the IVR system with event reports when the call arrives, when the agent answers, and when the call drops.
3. The IVR system sends information to the host application so the host application can send a data screen to the data terminal of the agent.

The IVR system can determine when a call drops before being answered and can track abandoned calls or use CPN/BN information for callbacks.

Integration with speech processing adjuncts

You can use ICM for integration with VRUs. The advantages of using ICM with the CallVisor ASAI in addition to the tip or ring interfaces are as follows:

- Data-screen integration is provided on transferred calls.
- Answer notification is provided on internal calls. The CallVisor ASAI capabilities let you know what happens with the call.
- ISDN network information, such as CPN/BN and DNIS, is delivered to agents. Call prompting for this information is not necessary.

A simplified configuration of this application is shown in the following figure.

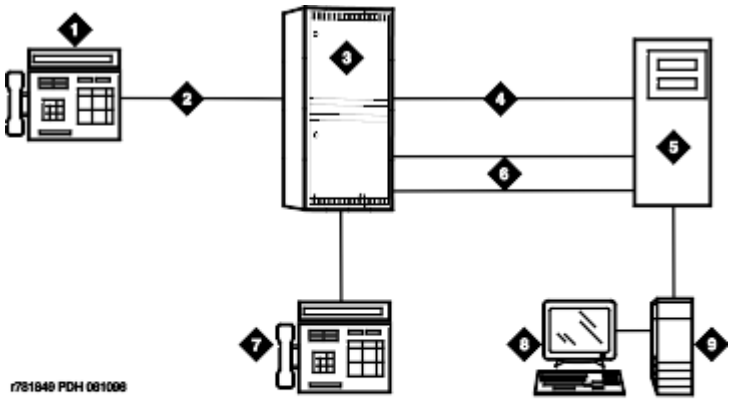


Figure 3: Simplified ICM configuration for speech processor integration

1. Phone
2. ISDN-PRI
3. Avaya switch
4. ASAI
5. Speech processor
6. Tip/ring lines

7. Agent phone
8. Agent data terminal
9. Host

General processing for this type of application occurs as follows:

1. Communication Manager uses the CallVisor ASAI link to pass incoming call information to the IVR.
2. The split or skill distributes the call to an available voice line.
3. After collecting the digits using a DTMF keypad, the IVR transfers the call to a split or skill or to a specific agent using CallVisor ASAI.
4. If the call is transferred to an agent, Communication Manager uses the CallVisor ASAI link to pass an event report on which agent receives the call.
5. The IVR forwards the agent ID to the host application which delivers a data screen to the agent.
6. Agents can view the collected digits on the data terminals. Except for the dialed number, information from an IVR cannot be carried with the call and displayed on a phone. For example, digits collected in an IVR adjunct cannot be passed to Communication Manager for display.
7. If the collected digits is an extension where the call is being routed, the routing digits are passed to Communication Manager as the destination in the CallVisor ASAI third-party make-call request. The IVR uses the request to set up various types of calls.

Host or adjunct call routing

The host or an IVR system adjunct uses incoming call information to route the call to a split or skill, vector, particular agent, or location off the communication server. The IVR system can also direct the system to handle the call as a priority call. Routing can be based on the caller's area code or country code, digits collected using Call Prompting, dialed number or service, agent availability, or information in a customer database.

To implement this type of call routing, make sure that calls come into a vector that contains an **`adjunct routing`** vector command. This command causes the communication server to initiate the route CallVisor ASAI capability. Vector processing occurs while the caller waits. A default split or skill or answering position can also be specified in the vector, in case the IVR system does not respond in the administered amount of time (determined by the announcement/wait steps). Announcement and wait steps are needed to give the host time to respond.

Note:

If the Display VDN for Route-to DAC option is enabled, and an **`adjunct`** vector step results in a direct agent call to an EAS agent, the VDN name is provided in the same manner as when a **`route-to digits`** or **`route-to number`** vector command is used.

For adjunct routing, if the call queues to a split or skill or leaves vector processing, a route-end request is sent to an IVR system.

ICM considerations

- ICM traffic.
- Rated Communication Manager capacity.
- CallVisor ASAI interface traffic.
- Rated capacity of the adjunct application processor.

Note:

Avaya Technical Design Center can provide planning assistance.

- CallVisor ASAI and BX.25 CPN/BN-ANI are not supported simultaneously.
- Direct agent calls are allowed only if the caller and the receiving agent have a Class of Restriction (COR) that allows Direct Agent Calling (DAC).
- Direct agent calls cannot go through vectors.
- Direct agent calls cannot be made over a DCS link. If the receiving agent is not an internal extension, the call is denied.

ICM interactions

Call Prompting: Digits collected by Call Prompting are passed with current call information to an IVR system adjunct.

Direct Agent Calling: DAC allows an adjunct to direct a call to a particular ACD agent and have the call treated as an ACD call. Calls that enter the communication server as ACD calls and are routed to a particular agent using adjunct routing, or are transferred using a third-party make-call request, are treated as ACD calls for the duration of the call. See Automatic Call Distribution for more information on direct agent calls.

Priority Calling: CallVisor ASAI allows both Priority Calling and DAC for the same call.

Information Forwarding

Information Forwarding sends call related information, such as Universal Call ID (UCID), ASAI data, collected digits, active VDN name, wait time, Automatic Number Identification (ANI), Calling Line Identification II (Information Indicator) Digits (CLID), and Customer Information Forwarding (CINFO), with calls over public and private networks using Integrated Services Digital Networks (ISDN) or Session Initiation Protocol (SIP) trunks. You can configure private networks for QSIG or non-QSIG protocols. You can also use the call data derived from Information Forwarding to enhance call processing, customer service, and data collection.

Whenever Communication Manager interflows a call over ISDN trunks such as PRI or BRI or SIP trunks by means of a **`route-to`** (with Look-Ahead Interflow active), **`queue-to best`**, or **`check`**

best command, the following information is sent with the call using user-to-user information transport and can be used by adjuncts or displayed at the receiving communication server:

- ASAI user information
- the name of the active VDN (LAI DNIS)
- other LAI information (a time stamp showing when the call entered the current queue, the call's priority level in its current queue, and the type of interflow)
- any collected digits (this does not include dial-ahead digits). These digits are available for processing at remote vectors and/or displaying to the agent.
- the number of seconds that the call has already spent in vector processing (called *in-VDN time*)
- Universal Call ID (UCID)

Note:

Sending of information depends on priority settings and activated features. Also the communication server version must be V6 or later.

You can set up Asynchronous Transfer Mode (ATM) and IP trunking to emulate ISDN PRI. For more information, see *Administering Network Connectivity on Avaya Aura® Communication Manager* and *ATM Installation, Upgrades and Administration using Communication Manager*.

Information Forwarding benefits

The following table lists Information Forwarding benefits:

Function	Benefit
Improved agent efficiency and service to call	Forwarding of original caller service requirements and entered prompted digits speeds service to the caller and saves the agent time.
Improved network-wide call tracking	Forwarding of UCID, In-VDN-Time and collected digits allows tracking as a single call and provides a network-wide view for call statistics.
Improved CTI integration	Forwarding of UCID, In-VDN-Time, and collected digits provides screen pop and database access applications across sites.
Forwarding of original call service requirements (VDN Name or DNIS)	Faster and more efficient agent handling, better service to the caller, and improved CTI integration
Transport of UCID	Improved call tracking as a single call and CTI integration
Collected Digits Transport	Better service to the caller because the caller doesn't have to repeat input of information, more information for the agent, better and faster call handling, improved call tracking because the collected digits are included with the call record, and improved CTI integration
Forwarding of In-VDN Time	Improved call tracking as a single call and end-to-end time-before-answer statistics

Table continues…

Function	Benefit
Support of ASAI user Information Forwarding	CTI integration

Data handled by Information Forwarding

The following table lists the call related information that you can send through Information Forwarding based on the trunk type.

Incoming call-related information	Supported trunks
Automatic Number Identification (ANI) and Calling Line Identification (CLID)	ISDN SIP
Information Indicator Digits (II-Digits)	ISDN
Caller Information Forwarding (CINFO)	ISDN
Adjunct Switch Application Interface (ASAI) provided user information	ISDN SIP (ASAI provided user information is the only information provided if the SIP trunk group is assigned as *service provider*.)
Look-Ahead Interflow (LAI) information, such as the in-queue timestamp, VDN name, and network provided caller information, including priority level and type of interflow.	ISDN
Universal Call ID (UCID): UCID provides a unique identifier for each call that is used to track the call. For more information, see *Avaya Aura® Communication Manager Feature Description and Implementation*.	ISDN SIP
Interflowed Collected Digits and in-VDN time data.	ISDN SIP

ASAI shared UUI data conversion

The outgoing trunk interface treatment controls whether the ASAI data format is shared or non-shared:

- If the outgoing trunk interface is non-shared, ASAI UUI data stored in shared format is converted to the non-shared (service provider) format.
- If the outgoing trunk interface is shared, ASAI UUI data stored in shared format is sent in shared format.

Information Forwarding detailed description

In the past, for ISDN trunks, look-ahead interflow transported the LAI Information Element (IE) in codeset 6 or 7, which functioned over non-QSIG private networks, but only over certain public networks.

Now, call centers can transport application information (including the LAI information) over many more public ISDN networks using User to User Signaling (UUS) Supplementary Services that incorporate user-to-user information (codeset 0 UUI). Information passes over QSIG private networks using manufacturer specific information (MSI - codeset 0 Facility IE) in various messages.

User-to-user information can also be transported over SIP trunks. You can use this feature to:

- Enable multiple applications on the communication server to share the contents of the UUI IE or MSI
- Allow backwards compatibility with software prior to the DEFINITY R6.3

Support of call center features

Information transport supports these call center features:

- Enhanced Look-Ahead Interflow: Routes calls from busy call centers to centers that are less busy.

  Note:

  Look-Ahead Interflow information can be forwarded using information transport or the traditional codeset 6/7 LAI IE.
- Best Service Routing: Routes calls to the best available agents irrespective of the agent location.
- Universal Call ID: Provides a means to collect and trace call data from multiple call centers.

Determining user information needs

User information rules

The network byte limit on user information contents, that is, the user data part of the UUI, must be large enough to carry the data needed for the customer application.

To forward information over a network that does not support a minimum of 82 bytes of user data, you must determine the space required for the application and adjust priorities accordingly.

The network byte limit on the user data part of the UUI user information must carry the data required for the customer application.

The UUI IE for ISDN uses three bytes for the IE header information and allows from 32 bytes to 128 bytes for the user data portion. For example, if the network can transport 32 bytes of user data, the UUI IE header length is 35 bytes.

The UUI for SIP includes the same data as the ISDN UUI IE except for the first two bytes in the UUI IE header information. For example, if the network can transport 32 bytes of user data, the UUI header length is 33 bytes.

The user information capacity need is determined by adding the space required for each data item to be transported based on the following rules.

Rule	Description
Minimum and maximum byte lengths	Communication Manager supports a maximum of 128 bytes of user data for UUI. Some network providers can impose limitations on the supported user data.
User data length	Each shared data item requires two bytes for the application header. The application header is the application identifier and the application data length. The application data length depends on the configuration of the customer application, except for UCID, In-VDN time, and Other LAI. The applications have a fixed byte length.
Byte length overruns	If the administered overrun for the data length is exceeded, the lowest priority items are not included until the remaining data fits. If a specific data item at a higher priority exceeds the administered Maximum UUI size setting, that item is not sent, leaving room for other lower priority items.
Priority settings	If the **data item priority** is left `blank` on the Shared UUI Feature Priorities page of the Trunk Group screen, the data item is not sent and no space is allocated for the data item.
QSIG considerations	QSIG signaling and networks do not have user information size limits. QSIG signaling and networks support sending of MSI for user data items at the maximums. You do not have to determine space allocation and administration of priorities for QSIG networks.

Table continues…

Rule	Description
ASAI data length considerations	If the network supports 128 bytes and less than 78 bytes of ASAI user data is required, you do not have to determine space allocation or administer priorities. If the ASAI user data is greater than 78 bytes, you can have up to 96 bytes of ASAI user data, that is, 98 bytes with the application header. When setting priorities and determining how much ASAI user data to support for the application, pay attention to the need for other interflow shared data transport. If the network supports all 128 bytes and all interflow data at their maximums is transported (48 bytes), the maximum length for ASAI user data is 80 bytes (78 bytes plus the application header). If all 96 bytes of ASAI user data is required (plus 2 bytes for the application header), only 30 bytes is available for other interflow data. With the service provider trunk group for UUI transport (non-shared UUI), the UUI transports only ASAI UUI. The first byte of the UUI contains the UUI codepoint, which is 0x7E, followed by the byte that is set to the length of the following bytes. The next byte is the UUI protocol discriminator which, is unusually set to 0x00, that means the following data (ASAI UUI data) is in a user specified format. Another possible setting for the protocol discriminator is 0x04, meaning that the following data is in the IA5 (ASCII) character format. With service provider non-shared UUI format the protocol discriminator setting is retained by Communication Manager and included in the ASAI UUI IE sent to a CTI adjunct. The shared format for the trunk group UUI transport is a multi-data element format that contains the UCID, ASAI user data, collected digits, and the VDN name associated with the call. Each data element is designated by a unique op code defined by the Avaya shared UUI specification. The ASAI user portion of the shared UUI consists of the op code (0xC8) and the data length byte followed by the user data. The ASAI UUI portion of the shared UUI does not contain a protocol discriminator field that is included with the non-shared UUI. When an ASAI UUI IE is created for passing to an adjunct, Communication Manager inserts a 0x00 for the protocol discriminator. **Note:** In most cases the application must not rely on the setting of the protocol discriminator since the discriminator is not meaningful except to account for the discriminator as being part of the format. If the Call Vectoring set command initially creates the ASAI UUI data, the protocol discriminator is always set to 0x04 in both the service provider and shared cases by Communication Manager, otherwise the protocol discriminator setting is not changed. In the case of shared the UUI IE protocol discriminator is set to 0x04 regardless of what else is carried by the shared UUI. The set command defines the data bytes following the protocol discriminator which is limited to the decimal digits (0 - 9) subset of the ASCII (IA5) character set.

Table continues…

Rule	Description
	The SIP UUI format is basically the same as the ISDN format but without the first two bytes (the 0x7E op code and length byte). The SIP UUI is in an ASCII string of hex characters (two for each byte) starting with the UUI protocol discriminator byte (00 or 04).

Bytes length ranges for UUI user data

The following table specifies the minimum and maximum byte lengths used to send user data over call center networks.

Type of user data	Total user data bytes (with 2-byte header)	Description
ASAI	2 to 98 or 0 (calculated by 1 byte per byte of ASAI user information)	Required for certain CTI applications when the CTI application sends user information and the amount of space is determined by the application. For example, 34 bytes is required if the application sends 32 bytes of data. Sending more than 78 bytes of ASAI data (80 bytes with the application header) reduces capacity for other interflow data.
UCID	10 or 0	A unique tag that identifies the call that this message is being sent for and the other information included in the UUI applies to. The priority for this element can be set between 1 to 7 (1 is the highest) or a blank. The default for this element is priority 2. If blank, this field's information is not forwarded. Used by BSR to track calls across multiple sites. Trunk group setting and/or system feature settings control transport of UCID data, even when the priority is set to 1. When the data item is not included, it does not take up any space.
In-VDN Time	4	Used by BSR to determine time before answer and call tracking across sites. This data type can be eliminated when short waiting times are anticipated. If the priority field is not blank, it is always included.
VDN Name	2 to 17 (calculated by 1 byte per character in name) maximum of 15	Used by BSR, but can be eliminated if receiving sites use dedicated VDNs that display equivalent information to the answering agent. An interflowed call that is received without the originating VDN name uses the incoming VDN name. If the priority field is not blank, the 2-byte header is always included.
Collected Digits	4 to 11 or 0 (calculated by 1 byte per 2 digits plus 1) maximum of 16 digits	Requires a whole byte for an odd number of digits. For example, 1 digit requires 2 bytes (1 plus 1), 7 digits need 5 bytes (4 plus 1), and 16 digits need 9 bytes (8 plus 1).

Table continues…

Type of user data	Total user data bytes (with 2-byte header)	Description
Other LAI Info	6	Required for existing CTI applications that use any of the following obtained from the from the LAI IE: • in-queue time stamp • queue priority • interflow type
Held Call UCID	10 or 0	The unique tag for the last call that was put on hold by the ACD agent placing this call to another system. This UCID can be used to identify the original or parent call that can eventually be conferenced or transferred to that other system. The UCID included in the element with default priority 2 is the tag for this new call placed by the agent while the original call is on hold. The priority for this element can be set between 1 to 7 (1 is the highest) or a blank. The default for this element is priority 7. If blank, this field information is not forwarded.

ISDN network requirements

Your network must meet the following requirements to support Information Forwarding:

- Both the private and public networks must support end-to-end transport of codeset 0 user data either as UUI IE or QSIG Manufacturer Specific Information (MSI) in the SETUP and DISCONNECT ISDN messages. Private networks can be configured for either non-QSIG transport by way of a codeset 0 UUI IE or QSIG transport by way of MSI packaged in a codeset 0 Facility IE. Public networks do not currently support QSIG and user data can only be transported by way of the UUI IE when supported by the network. Future public network offerings can support QSIG by way of a VPN.
- Communication Manager must support the ISDN country protocol.

  Important:

  If testing has not been done to verify operation over the public networks involved with the preferred specific configuration, use private ISDN trunking between the nodes until successful testing is complete.
- The network byte limit for the user data portion of user information contents must carry the data needed for the customer application.

Some public network providers require service activation or fees for user information transport.

Information Forwarding example

In the example, let the public network supports only 32 bytes of user information. The application requires 13 bytes of ASAI user information (15 bytes of user data), UCID (10 bytes of user data), and 8 collected digits (7 bytes of user data - 4 plus 1 plus 2 for the header). The application does not require Other LAI Information. Also, call time at the sending Communication Manager is brief

because calls are not queued before interflow takes place and tracking as a single call is not required.

By dedicating appropriately named VDNs at the receiving Communication Manager, the public network can support the application. Because the needed data items require the entire 32 bytes of user data, the priority fields for the **In-VDN Time**, **VDN Name**, and **Other LAI Information** must be set to blank.

SIP network requirements

Your network must meet the following requirements to support Information Forwarding:

- The public network must support end-to-end transport of SIP user data in a user-to-user information (UUI) header in the INVITE messages.
- The Communication Manager must support the SIP protocol and the UUI header.
- The network byte limit for the user data portion of user information contents must be large enough to carry the data that is needed for the customer application.

Information Forwarding considerations

Procedure

1. Enhanced information forwarding has been tested with several major carriers.

   To find out if the capabilities work with your carrier, check with your account team for the latest information. If testing is not done to verify operation over the public networks involved with the preferred specific configuration, use private trunking between the nodes until successful testing is completed.

2. Any communication server that acts as tandem node must have priorities assigned to the Shared UUI features for non-QSIG trunk groups.

   Even if this communication server does not create anything, the priorities must be set correctly to pass the information along.

3. The **Send codeset 6/7 LAI** trunk group field operates independently of the **UUI IE Treatment** trunk group field.

   However, if you turn both the options on, you will send the same information twice and possibly exceed the maximum ISDN message size. The communication server provides a warning message when both options are administered. There are two ways to correct when the user data exceeds the maximum message size, either:

   - Leave the **VDN Name** and **Other LAI Information** fields `blank` on the Shared UUI Feature Priorities screen or
   - Disable the **Send codeset 6/7 LAI** field.

4. For non-QSIG or QSIG trunk groups to the communication server that require information forwarding, the **UUI IE Treatment** must be `shared` and the **Send Codeset 6/7 LAI IE** must be `n`.

5. Information transported using the Shared UUI does not work with non-Avaya switches unless the information adheres to the proprietary encoding.

Information Forwarding interactions

Best Service Routing: Best Service Routing-related data is sent in addition to the associated ASAI user data and UCID.

Intraflow and Interflow: Intraflow and Interflow allow you to redirect ACD calls from one split or skill to another split or skill when the splits/skills are not vector-controlled. Intraflow redirects calls to other splits/skills within the system using Call Coverage or Call Forwarding All Calls. Interflow redirects calls to an external split or skill or location using Call Forwarding All Calls.

Information forwarding support for BSR and LAI

When a call is interflowed to another Communication Manager by BSR or Look-Ahead Interflow, the following data types are supported for Information Forwarding:

- Collected Digits - Any digits that are collected for the call are passed with the interflowed call, and automatically collected when the call enters vector processing at the receiving Communication Manager.
- Elapsed in-VDN time - The elapsed time that the call has already spent at the sending Communication Manager is passed with the interflowed call and automatically sent to the Avaya Call Management System (CMS) when the call enters vector processing at the receiving Communication Manager.
- UCID - Universal Call ID

Forwarding collected digits with interflowed call

The following list describes how forwarded collected digits are handled in interflowed calls:

- The last set of up to 16 collected digits, not including the dial-ahead digits, are forwarded with a call interflowed over ISDN or SIP facilities.
- When processing for the call at the remote location reaches the VDN, the forwarded digits are inserted in the collected digits buffer. Therefore, a TTR is not needed. The objective is to immediately provide the collected digits to the CMS in a DIGITS message and to ASAI by way of the VDN event report in the same manner as incoming ANI.
- The collected digits are available for further routing by steps in the assigned and subsequent vectors, and eventual display to the answering agent.
- All interactions with the collected digits are the same as digits that are collected using a collect step. For example, a subsequent collect step will clear the digits.
- If the call is further interflowed or tandemed over ISDN or SIP facilities, the collected digits are tandemed with the call. If more digits are collected at the tandem Communication Manager, the latest collected digits are tandemed.

Forwarding accumulated in-VDN time

The following list describes how forwarded in-VDN time data is handles in interflowed calls:

- When a call is interflowed, the in-VDN time in seconds, from 0 to 9999, is included. The in-VDN time is the elapsed time starting from the VDN that was originally called until when the Information Forwarding message is created.
- If the call was interflowed to the local system and in-VDN time was received for the call, the previous in-VDN time is added to the local in-VDN time.
- If the accumulated time exceeds the largest value that can be transported, the maximum value is sent.
- The accumulated in-VDN time that is received on an incoming interflowed call is forwarded to the CMS in the DNEVENT message when the call starts VDN/vector processing at the remote location.
- In-VDN time does not pass to the Basic Call Management System (BCMS) for reporting by BCMS.

Transport over ISDN

Use of codeset 0 supports information transport over ISDN PRI or BRI facilities (QSIG or non-QSIG) as well as supporting operation over public networks. The following list describes information transport over ISDN trunks:

- When a call is LAI or BSR interflowed, the following information is forwarded with the call over public or private ISDN networks using QSIG or non-QSIG protocols:
  - LAI information.

    Note:

    The forwarded LAI information is the same as that sent in the LAI IE: VDN name (also called LAI DNIS), put in queue time-stamp, priority level and type of interflow.
  - Collected digits.
  - in-VDN time data in the ISDN SETUP message.
- Other call related information, including calling party number (ANI), calling party name, II-digits and CINFO digits, that is tandemed with the interflowed call in the SETUP message is forwarded in the normal manner.

  Note:

  II-digits and CINFO are forwarded as codeset 6 IEs which can be a problem in some networks.
- At the remote end, the transported data is separated into its component parts for storage with the call, call vectoring, call processing and display, further interflow or tandeming, and forwarding to adjuncts. For example, the LAI info is treated as though it was received as an incoming codeset 6 LAI IE including forwarding over ASAI as a code set 6 LAI IE in event reports.

- When a status poll call is placed to the remote location, the Communication Manager only forwards the UCID and caller information that was received from the original call.
- In response to a status poll, the Communication Manager forwards the reply-best status data in the ISDN DISCONNECT message over public or private ISDN PRI/BRI networks. In this case, the DISCONNECT message has a cause value of 31 *Normal-Unspecified* for wider international interoperability.
- Multisite routing related data is in addition to the associated ASAI user data, which was previously sent in a non shared UUI IE, and the UCID data.

Transport over SIP

SIP supports information transport with the UUI header. The service provider must support SIP information transport.

The following list describes information transport over SIP trunks:

- When a call is LAI or BSR interflowed, UUI header information is forwarded with the call over SIP networks using non-QSIG protocols.
- Other call related information, including calling party number and calling party name is forwarded in the normal manner.
- At the remote end, the transported data is separated into component parts for storage with the call, call vectoring, call processing and display, further interflow or tandeming, and forwarding to adjuncts.
- Multisite routing related data is in addition to the associated ASAI user data, which was previously sent in a non-shared UUI, and the UCID data.

LAI backward compatibility issues

The following list summarizes LAI backward compatibility issues:

- An ISDN trunk group option is provided in the SETUP message for LAI interflowed calls to specify whether to include an LAI IE (codeset 6 or 7). When this option is set to `y` (default), an LAI interflow (using the existing or enhanced LAI vector command) will include a codeset 6/7 LAI IE to provide inter-operability in a mixed Communication Manager environment with pre-Definity R6 systems. The option must be set to `n` if the network does not support codeset 6/7 or this IE is not required.

  Important:

  Codeset 0 information transport by way of shared UUI is required for BSR polling calls.
- Administer the ISDN Trunk Group option: Send Codeset 6/7 LAI IE. This option is valid even if LAI at the remote site is not active for tandem situations. Use of this option for LAI does not depend on the setting of the Vectoring Best Service Routing customer option.
- If the ISDN trunk group option is set to send the LAI IE, this IE is sent in addition to the Information Forwarding by way of codeset 0 shared UUI transport when a call is LAI interflowed over a trunk in this trunk group. With shared UUI, you can set the LAI data to be excluded in the UUI IE.

- Administer the Shared UUI priorities for ISDN trunks. This is important when the network byte limit on the user data part of the UUI user information contents is not large enough to carry the data that is needed for the customer application. Note that Shared UUI priorities do not apply to QSIG. To determine customer application data sizes, see Determining user information needs.

Advanced Information Forwarding for ISDN and SIP trunks

This section explains ISDN (BRI or PRI) and SIP trunk group setting interactions with Information Forwarding, UCID, and Multisite Routing.

Advanced information forwarding

User information included in the SETUP message for an outgoing call at the sending Communication Manager server or in the DISCONNECT message sent back for an incoming call at the receiving Communication Manager server is based on the trunk group settings at the sending or receiving sites.

Shared user information forwarding supports various data items such as the following:

- UCID, collected digits, and In-VDN-Time
- LAI information such as VDN-Name and Other-LAI
- ASAI provided user data

Shared forwarding over non-QSIG ISDN trunks packs the data items in a codeset 0 UUI IE, called *shared UUI*. Where each item consists of a two-byte header, that is, application ID and data length. Shared forwarding over QSIG trunks transports the data items as MSI in codeset 0 Facility IEs.

BSR and shared data forwarding, that is, UCID and other new data items, requires QSIG or the shared **UUI Treatment** setting with non-QSIG trunk groups on both the sending and receiving trunks at Communication Manager. Shared settings are also required on tandem trunk connections through the switch that routes the calls. LAI info, UCID, collected digits, In-VDN-Time and ASAI provided user data can be forwarded with a call in the SETUP message, LAI or BSR interflowed call, a tandemed call, for UCID with any outgoing call and for ASAI user data any adjunct routed outgoing call. Only BSR reply-best data is returned with a BSR poll call and only ASAI user data is returned for a non-poll call in a DISCONNECT message. Both types of data are not included in the same DISCONNECT message. Shared UUI Priority settings do not affect what is put in the DISCONNECT message or data forwarded over QSIG trunks.

The protocols, QSIG or non-QSIG, are set on page 1 of the ISDN Trunk Group screen using the **Supplementary Service Protocol** field. The QSIG type as defined for shared MSI is protocol type b, another protocol type d, ECMA QSIG is treated as non-QSIG for information forwarding. The **Send Codeset 6/7 LAI** field on page 2 indicates whether or not to include an LAI IE in the SETUP message. The codeset used (6 or 7) is determined by the **Codeset to Send TCM, Look-ahead** field on page 1. The **Send UCID** field on page 2 indicates whether or not the UCID data item must be included as user information with calls routed over this trunk group. The **Send Codeset 6/7 LAI IE** field is ignored for BSR polls over the trunk group, that is, an LAI IE is included with BSR calls.

Non-QSIG protocol

UUI Treatment set to `service-provider` includes any application provided UUI in a codeset 0 UUI IE on a non-shared basis. That is, the data portion of the UUI IE only includes user info in the SETUP or DISCONNECT messages as provided by an application such as ASAI without the shared App-ID and length header fields. User data from only one application can be included in non-shared UUI. This setting is used for non-QSIG trunk groups when service-provider functionality is wanted (for example, where shared forwarding of the new data items is not required or for trunk groups to other vendor switches or network services that need user information from the trunk group in a non-shared UUI IE such as provided by ASAI). Incoming calls received with shared user information (shared UUI IE) that are routed outgoing over a non-QSIG service-provider trunk group will forward only ASAI provided user data in a non-shared UUI IE.

UUI Treatment set to `shared` allows all applications to include data items in the UUI IE on a shared forwarding basis. The Shared UUI Feature Priorities page settings along with the **Max. Size of UUI Contents** field on page 2 and the features configured for the system determines what actually is included in the UUI IE. This is the normal setting for non-QSIG trunk groups that route calls to the switch over private or public networks when information forwarding is required and must be used for BSR.

QSIG trunk group

UUI Treatment set to `service-provider` forwarded ASAI provided user data in a non-shared codeset 0 UUI IE and all other user data in codeset 0 Facility IEs as MSI. In this case the **Max. Size of UUI Contents** field is not relevant and the Shared UUI Feature Priorities page does not show nor apply. This setting is used for QSIG trunk groups to pre-R6.3 DEFINITY switches for compatibility with existing ASAI applications or when service-provider functionality is wanted (e.g., where shared forwarding of the new data items is not required or for trunk groups to other vendor switches that need user information from the trunk group in a non-shared UUI IE such as provided by ASAI). Incoming calls received with shared data (shared UUI IE) routed out over a QSIG service-provider trunk group, will separate any ASAI provided user data included in the shared UUI IE and forward it in a non-shared UUI IE.

UUI Treatment set to `shared` will forward all user information including ASAI provided user data in codeset 0 Facility IEs as MSI in the SETUP or DISCONNECT message. The UUI IE is never included over a shared QSIG trunk group. In this case, the **Max. Size of UUI Contents** field and the Shared UUI Feature Priorities page do not apply. This is the normal setting for QSIG trunk groups to the switch when information forwarding is required and must be used for BSR.

Send Codeset 6/7 LAI IE field interactions

The **Send Codeset 6/7 LAI IE** field is independent of the **Supplementary Service Protocol** and **UUI Treatment** settings to allow additional flexibility. Communication Manager can have a mix of trunk groups set with non-QSIG or QSIG protocol and with `service-provider` or `shared` settings. Calls interflowed over the shared non-QSIG trunk groups contain the data items to be forwarded with the call in the UUI IE while calls interflowed over the non-QSIG service-provider trunk groups do not contain the data items, except for ASAI which can always be sent in UUI.

Calls interflowed over the QSIG trunk groups always have MSI user information, except for ASAI where the transport method depends on the UUI treatment setting.

When a call is LAI interflowed over a non-QSIG service-provider trunk group, the **Send Codeset 6/7 LAI IE** field being active results in just the LAI IE being forwarded with the call in a SETUP message. When interflowed over a non-QSIG shared trunk group, setting the **Send Codeset 6/7 LAI IE** field to `y` includes a codeset 6/7 LAI IE in the SETUP message in addition to the same LAI information included as shared data in the UUI IE. You can set the LAI information fields to `blank` on the Priorities screen to exclude the data items from the UUI IE. When interflowed over a QSIG service-provider or shared trunk group with the **Send Codeset 6/7 LAI IE** field active, the LAI information is included as both MSI and in the LAI IE. However, in this case there is no mechanism to eliminate the duplication of data if the codeset 6/7 LAI IE is required.

The combinations can be used when calls are LAI interflowed to Communication Manager previous to the switch with existing ASAI applications using ASAI provided UUI that cannot be using the LAI IE. Note that codeset 6/7 IEs are not defined for QSIG and other vendor switch treatment of calls with a LAI IE is undefined can be ignored, blocked, or misinterpreted.

When the trunk group is set to `non-QSIG` and `shared` or to `QSIG` with `service-provider` or `shared`, do not set the **Send Codeset 6/7 LAI IE** field to `y` due to the overhead of sending duplicate information. In some cases, the configuration can exceed the SETUP message or user information byte count limits for the network and result in the user information being dropped. Also, transport can cost more in networks which charge for user transport by quantity of bytes transported. An administration warning message displays when the combination is set for the trunk group. In fact the combination is not recommended except in very limited cases where a mix of early and later switches can be reached over the same trunk group, using a public or switched private network, using LAI, and where BSR or UCID is not active or is being used and the data that must be forwarded with the call can be limited to that supported by the network.

The **Send Codeset 6/7 LAI IE** field must not be set to `y` with trunk groups where calls will be interflowed over public networks or virtual private networks that do not support codeset 6/7 transport. In such cases, the codeset 6/7 IE is not forwarded or the calls cannot be routed by the network. This can happen in some international situations, notably over networks in Germany.

UUI treatment	Send Codeset 6/7 LAI IE	Supplementary services protocol	
		Non-QSIG (other than b)	QSIG (SS b) MSI is sent in codeset 0 Facility IEs
service-provider	`n`	ASAI provided user info in codeset 0 UUI IE.	ASAI provided user info in a codeset 0 UUI IE and all other user info in codeset 0 MSI.

Table continues…

UUI treatment	Send Codeset 6/7 LAI IE	Supplementary services protocol	
		Non-QSIG (other than b)	QSIG (SS b) MSI is sent in codeset 0 Facility IEs
	y	ASAI provided user info in codeset 0 UUI IE & a codeset 6/7 LAI IE.	ASAI provided user info in codeset 0 UUI IE, all other user info in codeset 0 MSI and a codeset 6/7 LAI IE. With this combination, the LAI information (LAI-Name and Other-LAI) is sent both as MSI (in a Facility IE) and in the LAI IE. Note that LAI IE and shared MSI operation with other vendor switches is undefined.
shared	n	All user info in a shared codeset 0 UUI IE.	All user info in codeset 0 MSI.
	y	All user info in a shared codeset 0 UUI IE & a codeset 6/7 LAI IE. With this combination, the LAI information (VDN-Name and Other-LAI) is sent in both the UUI IE and in the LAI IE (setting the UUI Priorities for these items to blank can eliminate the duplication).	All user info as codeset 0 MSI and a codeset 6/7 LAI IE.

Situation	Trunk group field settings		
	UUI treatment		Send Codeset 6/7 LAI IE
	Non-QSIG	QSIG	
Trunk groups over which information forwarding is not required for LAI, BSR or UCID transport.	service-provider	service-provider	n
Non-LAI interflow or tandem calls to service providers or other vendor switches that do not recognize shared UUI.	service-provider	service-provider	n
LAI to pre-R6.3 switches over networks that block codeset 6/7 IE calls.	service-provider	service-provider	n

Table continues…

Situation	Trunk group field settings		
	UUI treatment		Send Codeset 6/7 LAI IE
	Non-QSIG	QSIG	
LAI to pre-R6.3 switches over networks that allow codeset 6/7 (traditional LAI) with or without ASAI applications that use UUI and/or LAI Info	service-provider	service-provider (With this combination, the LAI information is sent both as MSI (in Facility IEs) and in the LAI IE.)	y
LAI over public/virtual private network to mixed R6.3 and earlier switches, where the Avaya switches have shared information forwarding. The pre-R6.3 switches can use LAI Info in an ASAI application, but must not use UUI.	shared (With this combination, the LAI information (LAI Name and Other LAI) is sent in both the UUI IE and in the LAI IE.)	shared	y
LAI over public or virtual private network to mixed R6.3 and prior switches. The R6.3 and prior switches can use LAI info or UUI in an ASAI application.	service-provider (The LAI IE and ASAI non-shared UUI is supported, but BSR, UCID and other new data items are not.)	service-provider	y

Table continues…

Situation	Trunk group field settings		
	UUI treatment		Send Codeset 6/7 LAI IE
	Non-QSIG	QSIG	
BSR, LAI to all R6.3 or later switches All switches interflowed to must be R6.3 or later with shared incoming and outgoing trunk group settings. Tandeming or interflowing through R6.3 or later switches requires shared settings. Switches tandemed through can be older than R6.3 or other vendor switches that pass codeset 0 UUI or MSI transparently. This is the only combination that supports BSR and new data items information forwarding. In this scenario, do not set the **Send Codeset 6/7 LAI IE** field to `y` in order to save SETUP message space and to ensure operation over networks that do not allow codeset 6/7 IEs.	shared	shared	n

Interruptible Aux

With Interruptible Aux, you can make Expert Agent Selection (EAS) agents in the Auxiliary (AUX) work mode available to receive incoming calls. However, you must assign interruptible reason codes to the agent login IDs.

You can use Interruptible Aux to administer:

- The threshold fields on the Hunt Group screen as **calls-warning-threshold**, **service-level-target**, or **time-warning-threshold**.

  The **Interruptible** field on the Reason Code Names screen as **y**. This field option signifies that the corresponding reason code is interruptible.

- The **Reserve Level (RL)** field on the Agent Login ID screen as **a**, **m**, or **n**, where **a** is auto-in-interrupt, **m** is manual-in-interrupt, and **n** is notify-interrupt.

When an interruptible split or skill exceeds the administered threshold, Communication Manager notifies the agents with the interruptible skill. The notification consists of a display message, such as the following: `You are needed`. Communication Manager flashes the **auto-in** or **manual-in** buttons on the agent phone and plays an audible, full ring tone.

You can use **auto-in-interrupt** to force an agent to be automatically available and **manual-in-interrupt** to force an agent in After Call Work (ACW) to be available for calls. You can use **notify-interrupt** to request an agent in the AUX work to be available for calls.

You can administer the duration of the notification in the **Interruptible Aux Notification Timer (sec)** field on the Feature-Related System-Parameters screen. Notifications continue till a further event, such as an agent becoming available or logging off from the system, takes place or the interruptible skill no longer exceeds the administered threshold.

Communication Manager makes the forced interruptible agents in the AUX work mode automatically available after the notification timer expires, except if connected to or being alerted by a non-ACD call or if the agent logs in to the system as an auto-answer agent. Communication Manager treats a forced interruptible agent administered with auto-answer as *requested interruptible*, not forced, even if the **Reserve Level (RL)** field option is a forced interruptible type. This action prevents the situation where a call is delivered automatically and the agent is not present at the workstation. Therefore, forced interruption is applicable only to agents without the auto-answer administration.

The following table describes the reserve level treatment for agents. In the following example, the Interruptible Aux threshold is administered as **service-level-target**. Communication Manager changes the agent work mode to AUX work as follows.

Agent	Reason code	Reserve level	Agent action when skill exceeds the administered threshold	Agent action (if applicable)
5002	9 (not interruptible)	Auto-in	No notification.	—
5004	8 (interruptible)	Notify	Notified and moved from AUX work to auto-in.	—
5005			Notification received to be available for calls.	Presses **auto-in** to be available for calls to all the assigned skills.
5007				The agent is away from the phone and cannot press **auto-in** so the agent remains in AUX work.
5008		Manual-in	Work mode change from AUX work to manual-in.	—

The forced interruptible option is applicable to manual-in agents. With manual answer, you can administer Redirection on No Answer (RONA) to redirect a call if an agent does not receive the call. The forced interruptible option is inapplicable to an auto-in agent because the agent cannot respond to calls without a headset. You cannot apply RONA to auto-answer delivered calls.

Interrupted agents are available for all the assigned skills. This functionality is unlike Service Level Maximizer (SLM) auto-reserve agents, which limits agent availability to the skills that do not meet the service-level targets.

Interruptible Aux thresholds

Activation thresholds for Interruptible Aux:

Use Interruptible Aux to notify and make agents in the Auxiliary (AUX) work available to receive calls.

Threshold type	Activation criteria	Description
Calls warning threshold	More than X calls are in a hunt group queue	Communication Manager activates Interruptible Aux if the number of calls in a queue for a hunt group exceeds the threshold. For example, if you select **calls-warning-threshold** and administer the threshold at 20 calls, Communication Manager notifies interruptible agents in the AUX work mode if the number of calls in the queue is more than 20.
Service level target	Less than X % of calls answered in Y seconds	Communication Manager activates Interruptible Aux if the percentage of calls that agents answer within the specified number of seconds is less than the administered level. For example, you decide that agents must answer 90 % of the calls within 10 seconds, but agents do not meet the administered target. Communication Manager notifies interruptible agents in the Auxiliary (AUX) work mode to ensure that agents answer 90 % of the calls within 10 seconds.
Time warning threshold	The oldest call is in the queue for more than Y seconds	Communication Manager activates Interruptible Aux if the oldest call is in a queue for more than the administered number of seconds. For example, if you select **time-warning-threshold** and administer the wait time for the oldest call as 60 seconds, Communication Manager notifies interruptible agents in the Auxiliary (AUX) work mode if the oldest call is in the queue for more than 60 seconds.

Deactivation thresholds for Interruptible Aux:

If you administer an activation threshold, you must administer a deactivation threshold to turn off agent notifications.

Threshold type	Deactivation criteria	Description
Calls warning threshold	Less than X calls in a hunt group queue	Communication Manager turns off agent notifications when the number of calls in a queue is less than the administered threshold.
Service level target	More than X % of calls answered in Y seconds	Communication Manager turns off agent notifications when the percentage of calls that agents answer is more than the administered levels.
Time warning threshold	The oldest call is in the queue for less than Y seconds	Communication Manager turns off agent notifications when the oldest call is in the queue for less than the administered level.

You must use deactivation thresholds to prevent situations where Communication Manager triggers and turns off Interruptible Aux frequently if the Calls Warning, Service Level, or Time Warning threshold values fluctuate around the thresholds. The values of the activation and deactivation thresholds must differ by one unit at the minimum.

Administer the deactivation threshold to a value less than the activation threshold. For example, if the activation threshold for a Calls Warning threshold is 100, you can administer the deactivation threshold at 90. Interruptible Aux notifications begin when the number of calls in a queue reaches 100 and continue till the number of calls in the queue drops to 90.

Interruptible Aux considerations

- You can use Computer Telephony Integration (CTI)-based applications that use Adjunct Switch Application Interface (ASAI) to:
  - Change an agent work mode to Interruptible Aux.
  - Receive events when agents are available to receive calls.
- You can interrupt agents in the Auxiliary (AUX) work mode so that the agents are available to receive calls for all assigned skills.
- You cannot administer the default reason codes as interruptible. The defaults codes are IP Failure Aux Work, Maximum Agent Occupancy Aux Work, Redirection on No Answer Aux Work, and Redirection on OPTIM Failure Aux Work.
- You cannot apply the most idle agent algorithm when the only available agents are in interruptible mode. Here, the call goes to the first agent on the list.
- When more than 1000 forced interruptible agents are in the AUX work mode with an interruptible reason code, simultaneously moving all the agents to an available mode can adversely affect the performance of Communication Manager. To mitigate performance loss, only five agents are notified every half a second until all the required agents are notified. Similarly, agents are forced to an available mode at the same rate: Five agents every half a second.

Note:

It is still recommended that CTI clients do not send bulk agent change commands for Interruptible AUX.

Intraflow and Interflow

Use Call Coverage with Intraflow for conditional redirection of ACD calls from one split to another split based on the coverage path redirection criteria. For example, you can define a coverage path for a split to automatically redirect incoming ACD calls to another split when a phone is busy or an agent fails to answer the call. You can redirect calls to less busy splits for efficient call handling. Use Call Forwarding with Intraflow to unconditionally forward calls.

With Interflow, you can redirect ACD calls from a split on one Communication Manager to a split on another Communication Manager or an external location. Use Call Forwarding with Interflow to unconditionally forward calls to an off-premise location.

Important:

You cannot use Call Coverage with Interflow.

For information on the Call Coverage redirection criteria, see “Call Coverage” in the *Avaya Aura*® *Communication Manager Feature Description and Implementation* document.

How does Intraflow work

You can assign an inflow threshold for each split or skill receiving Intraflow and Interflow calls. The threshold prevents a split or skill from receiving new ACD calls if the oldest call has been in the queue longer than the threshold. If Communication Manager forwards an ACD call using Call Coverage, but cannot forward the call further to another split or skill or a coverage point, the call remains in queue at the original split or skill.

For a split or skill with administered Intraflow and Call Coverage, you can also assign priority on Intraflow. When an ACD call intraflowing from a split or skill with priority on Intraflow to a covering split or skill enters the queue, Communication Manager places the call ahead of non priority calls but behind other priority calls already in the queue.

For Intraflow using Call Coverage, Communication Manager does not connect the calls to the first delay announcement at the covering split or skill. For Intraflow using Call Forwarding, Communication Manager connects the calls to the first delay announcement at the forwarded-to split or skill. The calls, however, do not receive a forced first announcement.

The following scenario explains how Intraflow works:

- A call is intraflowed from split 1 to split 2 using Call Coverage.
- Split 1 is assigned priority on intraflow.
- Split 2 has a queue with three priority calls and four non priority calls.

- Split 2 has an inflow threshold of 90 seconds and the oldest call in queue at split 2 has been in queue for 60 seconds.
- Split 2 has been assigned a second delay announcement and has a second delay announcement interval of 45 seconds.
- Music-on-Hold is provided.

When the call is intraflowed from split 1 to split 2, the call is placed in the split 2 queue as the fourth priority call, ahead of the four non priority calls. The call stays in the queue for 45 seconds and is still not answered. The call is connected to the second delay announcement for split 2. After the announcement, the caller hears music until an agent answers the call.

You can assign a Coverage ICI button to an agent phone. The agents use the button to identify a call intraflowed from another split or skill. When an agent receives the call, the button lamp lights.

Interflow and Intraflow considerations

The same coverage path can be used for as many splits or skills as desired. You must administer redirection criteria for a split or skill coverage path so that calls are redirected under Busy or Don’t Answer conditions. Do not use All or Send All Calls as redirection criteria.

Interflow and Intraflow interactions

Interaction	Description
Call Coverage	All splits or skill with the same coverage path are automatically assigned the same Don’t Answer Interval. The default Don’t Answer Interval is 2. If Intraflow using Coverage is active, the Coverage Don’t Answer Interval associated with Call Coverage begins when a call enters the split or skill queue. If the Coverage Don’t Answer interval expires before either of the two delay-announcement intervals expires, a call is redirected to coverage. If either of the delay-announcement intervals expires before the Coverage Don’t Answer interval, the call is connected to a delay announcement, if available. If no coverage point is available to handle a call, a call remains in queue and can be connected to a delay announcement.
Temporary Bridged Appearance	If an ACD call is routed to an agent but is intraflowed to another split or skill before being answered, the Temporary Bridged Appearance at the agent telephone or console is no longer maintained.
Look-Ahead Interflow	Use Look-Ahead Interflow (LAI) to balance the load of ACD calls across multiple locations.

Location Preference Distribution

Use Location Preference Distribution to route incoming ACD calls to agents in the same location as the incoming trunk on which the arriving call originated. With Location Preference Distribution, you do not have to use trunks between locations. You can reduce the trunking costs and save trunks for additional incoming calls. If an incoming caller cannot be matched to an agent in the same location, the system routes the calls to agents at different locations. In this case, the administered distribution algorithms determine routing without regard to location.

When there is more than one choice for call delivery, Local Preference Distribution matches the trunk and the agent location numbers. The Multiple Locations feature defines the location number. Delivery preference is given to the agent whose location number matches the incoming trunk location number. Location Preference Distribution takes precedence over most other caller-agent selection features except for direct agent and reserve agent calls.

Location numbers

When an agent logs in to the system, Communication Manager assigns a location number to the agent login ID. Local Preference Distribution uses location numbers to match the incoming trunk and the agent.

Multiple Locations

With Multiple Locations, you can define location numbers for use with Location Preference Distribution. The Multiple Locations feature was developed to display the local time on phones that are located in a different time zone from the connecting communication server. You can use the feature with Location Preference Distribution to match the location of agents and trunks.

When an agent logs in to the system, Communication Manager sends the location number that is associated with the agent login ID to Call Management System (CMS) for tracking and reporting purpose. Communication Manager also sends the location number that is associated with the trunk from where agents receive calls.

Related links

VDN Observing by Location on page 220

Changes to the location number

During call center activity, location numbers can be changed through administration or the Multiple Locations feature can be disabled. If this happens, the Location Preference Distribution feature uses the agent ID obtained when the agent logged in and the call ID when the call was originally received. During such a change, the Location Preference Distribution matching is not appropriate until all agents log out and log back in again. The software does not check for the changes.

How trunks, stations, and agent endpoints obtain location numbers

Trunks, stations, or agent endpoints obtain location numbers as follows:

- Non-IP phones and trunks inherit the location number from their connected hardware. For example, a non-IP phone inherits its location number from a cabinet, remote office, or media gateway.
- IP phones indirectly obtain their location numbers when the location numbers are administered on the Network Region screen. This screen applies location numbers to all phones in the same network region.

  If the **location** field is left blank on the Network Region screen, the IP phone assigns its location from the CLAN board located on the cabinet or gateway where the phone is registered.
- H.323 IP and SIP phones receive the location defined on the **Location** field of the Station screen. This assignment overrides other location assignments.
- IP and SIP trunks obtain their location from the cabinet containing the CLAN or NIC that the trunk is signaling through.

If none of the above applies, location 1 is used as the default.

How to set up a location number

Important:

For information on how to administer locations for each station, see “Administer locations per station” in the *Avaya Aura® Communication Manager Feature Description and Implementation* document.

Establishing location numbers

About this task

Use the Locations screen to establish location numbers. This defines the characteristics of the location that can include:

Procedure

1. Time zone offset between local standard time and the remote server location
2. Daylight saving rules used by any Expansion Port Networks (EPNs) located in different time zones
3. Number plan area codes
4. An ARS prefix that is required for 10-digit calls.

   The ARS prefix defines calls that are routed to the relevant location, such as E911 local call routing

Assigning location numbers to EPN cabinets or to the Media Gateway

About this task

Perform one of the following tasks:

Procedure

1. Use the Cabinet Description screen to assign location numbers to the appropriate EPN cabinets.

   Use the **`change cabinet xx`** command.

2. Use the Media Gateway screen to assign location numbers to the Media Gateway.

Result

You can assign the same location number to more than one cabinet or gateway that is located in the same time zone. Note that you can assign all Avaya DEFINITY and Media Server configurations, except the S8100 Media Server configuration, to locations other than 1. The DEFINITY Server CSI and SI configurations default to location 1. Digital and analog station sets get their defined location number based on the port location of the cabinet or gateway. The circuit switch trunks also obtain their gateway number in the same manner.

Assigning the location by IP network region

About this task

Administer the location by IP network region on the IP-Network-Region screen.

This sets the following conditions:

Procedure

1. The correct date and time information and trunk routing based on the IP network region.
2. The correct date and time worldwide displayed for IP phones registered with a server, but located at a remote site or a site with a S8300 Media Server with a G700 or equivalent gateway.

   The IP phone can be administered in a different network region from other Communication Manager endpoints and in the same location as the S8300 Media Server or remote office users. The IP endpoint users can move from location to location and always have correct display information. Remote users are identified in a network region and location that routes the user calls to correct 911 services or notifies the users through announcements that the users are in a different 911 jurisdiction than where the users are registered.

Assigning the location on the Station screen

About this task

For information on how to administer locations for each station, see “Administer locations per station” in the *Avaya Aura® Communication Manager Feature Description and Implementation* document.

Call handling conditions

You can use Location Preference Distribution to administer how the system handles agent surplus conditions and call surplus conditions.

Agent surplus conditions

An agent surplus condition occurs when available agents are waiting for incoming ACD calls. The Location Preference Distribution algorithm routes new incoming ACD calls to an idle agent located within the same location number as the trunk or station of the calling party. If there is no match for an idle agent, the incoming ACD call is routed to the agent at the top of the free agent list of the skill based on the administered selection criteria.

Call surplus conditions

A call surplus condition occurs when there are ACD calls in queue waiting for an available agent. The Location Preference Distribution algorithm routes the next best queued call to a multi-skilled EAS agent that has the same location number as the call. The next best queued call is determined by the appropriate Avaya Business Advocate or non-Advocate algorithm. If there is no match between the queued ACD call and the skills administered for the agent, the normal best queued ACD call selection is made by the appropriate Avaya Business Advocate or non-Advocate algorithm. The selection is made without any location number preference of the queued ACD call or agent based on the administered selection criteria.

Location Preference Distribution selects calls only from the top of the queue for each skill. Location Preference Distribution does not try to match the agent's location from the skill queue. For example, when an agent with five skills becomes available and has calls in queue for three of his skills, Location Preference Distribution looks at the call at the top of each queue. If one or more calls match the location of the agent, Location Preference Distribution uses the administered selection criteria to pick a call for the agent from the same location.

Local Preference Distribution interactions

Feature	Description
Best Service Routing (BSR)	Local Preference Distribution is used to select an available agent within the call center during the **`consider`** and **`queue-to best`** step operations. Local Preference Distribution is not used across system sites. In this case, there is no multisite network region.
Business Advocate	Business Advocate provides call handling preferences based on: • A service objective that is administered on the Hunt Group screen. • Agent percentage allocation that is administered on the Agent Login ID screen on a per skill basis. Local Preference Distribution takes precedence over Business Advocate call handling preferences.

Table continues…

Feature	Description
Call Admission Control (CAC)	Location Preference Distribution does not interact directly with the CAC feature. However, when Location Preference Distribution selects a trunk-agent combination at the same location, not as much overall bandwidth is needed between locations. Location Preference Distribution cannot circumvent an internode bandwidth blockage between two Wide Area Network (WAN) switch or gateway sites when any of the following conditions exist: • All agents at an incoming trunk switch or gateway location are busy. • The WAN bandwidth has reached capacity between the incoming trunk location and a remote location. • An agent is unavailable at the remote location. In order to bypass the blocked WAN call path, BSR, or any other multisite feature, Location Preference Distribution routes an incoming ACD trunk call to an available agent at the remote location. The call is routed over the Public Switched Telephone Network (PSTN), Integrated Services Digital Network (ISDN) tie trunk, and other types of networks.
Conference and Transfer	When an incoming trunk call is transferred to an ACD hunt group and the agent is available when the transfer occurs, the location number is for the agent that transferred the call. If the transferred call queues and the transferring agent drops before an agent is available, the location number is for the incoming trunk.
Direct Agent Calling (DAC)	Direct Agent calls take precedence over Location Preference Distribution.
Dynamic Advocate	Dynamic Advocate provides call handling preferences based on: • A Percentage Allocation Distribution (PAD) group type preference assigned on the Hunt Group screen • A Percentage Allocation assignment for the skill assigned on the Agent Login ID screen. • The Service Objective (SO) field on the Hunt Group screen overrides the service objective assigned for Service Level Supervisor (SLS) on the Hunt Group screen. Local Preference Distribution takes precedence over Dynamic Advocate call handling preferences.
Expert Agent Selection (EAS)	EAS must be set to `y` before administering Local Preference Distribution.
Inter-Gateway Alternate Routing (IGAR)	IGAR provides the ability to alternately use the PSTN to carry the bearer portion of a call when the IP-WAN is incapable of carrying the bearer location. Local Preference Distribution does not interact directly with IGAR. However, when Location Preference Distribution selects a trunk-agent combination in the same location, the need for IGAR between locations is reduced.
Path replacement	When an incoming trunk call receives path replacement before the call is delivered to an agent, the original incoming trunk retains the location number for the call.

Table continues…

Feature	Description
Reserve agents	You can assign reserve agents using any of the following features: • Service Level Maximizer (SLM) • Service Level Supervisor (SLS) • Percent Allocation In most cases, the selection of an agent or a call based on Location Preference Distribution takes precedence over SLM, SLS, or Percent Allocation. Nevertheless, SLM, SLS, and Percent Allocation take precedence when the system chooses a reserve agent for the following reasons: • The skill is above the Estimated Wait Time (EWT) threshold with SLS. • The service level is below the threshold with SLM or Percent Allocation. **Note:** If more than one reserve agent is eligible for the call, Location Preference Distribution is used to choose the agent.
Separation of Bearer and Signaling (SBS)	The location number of an incoming SBS call is obtained from the bearer trunk assignment.
Service Level Maximizer (SLM)	See *auto reserve agents*.
Service Level Supervisor (SLS)	See *auto reserve agents*.

Related links

Auto reserve agents on page 210

Match Agent and Skill Partitions

Businesses that provide hosted contact center services can increase their cost advantage by sharing resources with multiple tenants. For example, service providers can share the maximum skill capacity of 8000 skills on a single Communication Manager instance with multiple tenants. However, sharing of resources presents problems related to voice and data security. For example, delivery of calls to unintended tenants is a breach in security.

With the Match Agent and Skill Partitions feature, administrators can ensure that calls for one tenant do not flow to another tenant. Communication Manager prevents the assignment of skills with Tenant Numbers (TNs) for which agents must not receive calls.

To apply administrative restrictions, this feature provides the following optional fields on the Agent LoginID screen:

- **Check skill TN to match LoginID TN**: This field is applicable when Tenant Partitioning is active for the system.
- **Include Tenant Calling Permissions**: This field is applicable when the **Check skill TN to match LoginID TN** field is **y**.

When you administer the **Check skill TN to match LoginID TN** field, Communication Manager checks all skills that you assign to an agent login ID. If the TN of a skill and the login ID do not match, Communication Manager displays an error message and prevents incorrect skill assignments to the login ID.

This administrative restriction is also applicable when:

- Supervisors use CMS Supervisor to change agent skills.
- You use the **`duplicate agent-loginid`** command to create multiple copies of an Agent Login ID screen.

Note:

When agents or supervisors use the Add Skill Feature Access Code (FAC) feature to change agent skills, Communication Manager performs TN assignment validation checks. These checks are independent of the field settings in the **Check skill TN to match LoginID TN** and **Include Tenant Calling Permissions** fields. These optional fields do not change the current operation that Communication Manager performs for the Add/Remove Skills via FAC feature.

You can grant tenant calling permissions on the Tenant screen and administer the **Include Tenant Calling Permissions** field. Agents can then receive calls for more than one TN.

If you change a skill TN and Tenant Partitioning is active for the system, Communication Manager displays a warning message. The warning message indicates that changing the TN might cause a mismatch of agent-skill partitions. Furthermore, if you change the TN of a login ID, Communication Manager rechecks all skills that you assigned to the login ID. If Communication Manager finds a mismatch, Communication Manager prevents change in the TN of the login ID.

For more information, see *Administering Avaya Aura® Call Center Elite*.

Match Agent and Skill Partitions interactions

Interaction	Description
Automatic logout and login of agents	If the TN of a skill and a login ID do not match, Communication Manager will return the mismatch error and will not perform the automatic logout and login of the agent.
Call Vectoring route-to step	If you use Tenant Partitioning, Communication Manager checks the TN of the Vector Directory Number (VDN) during call processing. Communication Manager then determines whether to deliver calls to the route-to destination. Communication Manager does not route calls to the destination in the following scenario: • The TN matches, but the Facility Restriction Level (FRL) of the Class of Restriction (COR) assigned to the VDN has a value that is lower than the value of the destination FRL, Instead, Communication Manager logs a denial event and vector processing moves to the next vector step.

Table continues…

Interaction	Description
Direct Agent Calls (DACs)	Tenant calling permissions on the Tenant screen determine the delivery of DACs to agents. As DAC is a skill that you assign to agent login IDs, Communication Manager checks the TN of the DAC skill and the login ID before call delivery. Therefore, ensure that the DAC skill has the same TN as the agent login ID.

Maximum Agent Occupancy

With Maximum Agent Occupancy (MAO), you can determine the total percentage of time that agents must spend on calls. Use the MAO feature to provide short breaks from work for agents with high occupancy rates. For example, agents with highest skill levels or single-skill Service Level Maximizer (SLM) agents.

When the percentage of work time for an agent exceeds the administered MAO threshold, Communication Manager:

1. Places the agent in the Auxiliary (AUX) work mode.
2. Flashes the **aux-work** button on the agent phone.
3. Sends the MAO AUX reason code to the reporting adjuncts

If the agent phone has more than one **aux-work** button, Communication Manager flashes the MAO-related button. If no MAO-related button is on the phone, Communication Manager flashes an **aux-work** button that has no assigned reason code.

When the percentage of work time for the agent is less than the MAO threshold, Communication Manager changes the agent work mode from AUX work to Available.

You can use MAO with SLM or Business Advocate, but you must use MAO with Expert Agent Selection (EAS).

MAO for manual-in agents

When the percentage of work time of a manual-in agent exceeds the MAO threshold, Communication Manager changes the agent work mode to After Call Work (ACW) when the agent releases the call. When the agent presses **manual-in** or types a Feature Access Code (FAC) to change the work mode to Available, Communication Manager changes the agent work mode to AUX work and sends the MAO reason code to the reporting adjuncts.

When the percentage of work time is less than the MAO threshold, Communication Manager changes the agent work state to manual-in so that the agent can receive calls.

MAO for auto-in agents

When the percentage of work time for an auto-in agent exceeds the MAO threshold, Communication Manager changes the agent work mode to AUX work and sends the MAO reason code to the reporting adjuncts.

Manual override of MAO AUX work mode

To override the work mode change due to MAO, an agent can:

- Press **auto-in** or **manual-in** twice.
- Enter the Feature Access Code (FAC) code twice.

CMS database items for evaluating MAO

Call Management System (CMS) includes database items that you can use to ensure that MAO functions properly.

In CMS Supervisor R12 or later, the historical Agent Summary report includes the following fields for evaluation of MAO:

- % Agent Occup w/ACW
- % Agent Occup w/o ACW

MAO interactions

Interaction	Description
Auto-Available Skills (AAS)	MAO is inapplicable to AAS ports as these ports are non-human. However, if you do not assign messaging systems and Voice Response Units (VRUs) as AAS, you can change the work mode to AUX work.
After Call Work (ACW)	When an agent is available or leaves ACW for a direct agent call, Communication Manager changes the agent work mode to AUX work if the agent occupancy exceeds the MAO threshold.
Auxiliary (AUX) work	If an agent manually enters the AUX work mode with the MAO reason code, Communication Manager does not treat the agent as an agent with pending availability. The agent must manually change the work mode.

Table continues…

Interaction	Description
Call Work Code (CWC) or Stroke Count (SC)	If the agent occupancy is more than the MAO threshold and you administer forced entry of CWCs and SCs, Communication Manager prompts the agent to type a CWC or SC. When the agent types the CWC or SC, Communication Manager changes the agent work mode to AUX work.
Hunt group type	If you use the Most Idle Agent (MIA) and Least Occupied Agent (LOA) agent selection methods for hunt groups, Communication Manager places agents who are in MAO-related AUX work in the last position of an idle queue when the agent becomes available to receive calls. Maximum Occupancy uses Least Occupied Agent (LOA) calculations to determine agent occupancy. Agent occupancy includes agent time with ACD calls ringing, calls active, calls on hold, and logged-in ACW if the system option or agent login ID has ACW as agent work time.
Multiple Call Handling (MCH)	An agent does not receive new MCH calls if the occupancy is more than the administered MAO threshold. An agent availability is pending when the agent releases all active ACD calls.
Reserve agents	If the occupancy of a reserve agent exceeds the MAO threshold, the agent availability is pending until the agent occupancy is less than the MAO threshold.
Timed ACW (TACW)	Communication Manager changes the agent work mode to AUX work after the agent time in TACW is complete. When the agent occupancy is less than the MAO threshold, Communication Manager makes the agent automatically available to receive calls.

Multiple Call Handling

MCH applications

With Multiple Call Handling (MCH), agents can receive additional calls without dropping the active call. Examples of applications include:

- An agent puts a call on hold to retrieve caller information.
- ACD calls are more important to business than non-ACD calls. Use MCH to interrupt agents who are on non-ACD calls.

- If calls from one skill such as sales are more important than calls from another skill such as service, use MCH to interrupt agents to receive calls from the more important skill.

Note:

MCH is not applicable to EC500 mapped mobile agents, since mobile agents can handle only one call at a time. This applies to both ACD and non-ACD calls.

You can use MCH in an EAS or non-EAS environment.

- With EAS, you can administer any combination of MCH and non-MCH skills for an agent. If an EAS agent is a member of both MCH and non-MCH skills, the agent can handle multiple ACD or DAC calls in the MCH skills.
- Without EAS, agents log in to only one split if the split is an MCH split. An agent logged in to a non-MCH split cannot log into an MCH split.

MCH Settings

On request

In on-request splits or skills, the following is true:

- If an agent enters the auto-in or manual-in work mode, but there are no calls in the queue, the agent is placed at the bottom of the MIA queue or at the bottom of the assigned skill level in the EAD queue, or is made available in the DDC queue.
- Agents must select the auto-in or manual-in work mode for each new ACD call the agents take while a call is on hold.
- The agent can take additional ACD calls as long as there is an available line appearance.

Use on-request MCH in conjunction with a feature, such as VuStats, which agents can use to see when the queue is getting full and take additional calls.

One forced

An agent who is idle or active on a non-ACD call is automatically interrupted with an ACD call from this split or skill when no other ACD call for any of the agent's splits or skills are alerting, active, or held. In addition, the following must also be true:

- The agent is in manual-in or auto-in work mode.
- The agent is the most idle or next available.
- An unrestricted line appearance is available.
- AUX work or Move from CMS are not pending.

As long as an ACD call is active or held, the agent does not automatically receive an additional call from the one forced split or skill. An agent in a one forced split or skill in auto-in or manual-in work mode is unavailable for that split or skill from the time that an ACD call rings until all ACD calls are abandoned, redirected, or dropped. However, the agent can request another ACD call from a one forced split or skill by placing the active call on hold and selecting manual-in or auto-in work mode.

If an agent with multiple skills is active on an ACD call for a group with one forced MCH, the agent can be forced to take an ACD call for one of other skills of the agent, depending on the MCH settings of the skill.

Because one forced MCH forces an ACD call to alert an agent who is not on an ACD call, use one forced when you want ACD calls to take precedence over other calls.

One per skill

You must have EAS to use one per skill MCH. An agent with no ACD calls for the skill is automatically interrupted with a single ACD call from this skill under the same conditions listed for one forced.

If a one per skill call is active or held, the agent does not automatically receive additional calls from the skill. However, the agent can request another ACD call from a one per skill in the usual way.

If an agent with multiple skills is active on an ACD call for a one per skill group, the agent can be forced to receive an ACD call for one of the other skills if those skills are many forced or one per skill MCH.

Use one per skill MCH when calls from one skill are of a higher priority than other ACD calls.

Many forced

Agents are automatically interrupted with an ACD call under the same conditions listed for one-forced. As soon as an agent answers an alerting ACD call, the agent immediately becomes available to receive another ACD call from a many-forced split or skill.

Agents in many-forced groups in auto-in or manual-in work mode are unavailable only when an ACD call is ringing.

Use many-forced MCH when agents must answer important or urgent calls, even when agents must put equally important calls on hold. You can also use many-forced MCH to force direct agent calls to an agent.

MCH example

In this example, an agent is logged into four skills, each with a different MCH option. The following table shows how calls are delivered when an unrestricted-line appearance is available and the agent is in auto-in or manual-in work mode (AUX work mode is not pending).

	Calls delivered?			
Condition	**Skill 1 (MCH=one-request)**	**Skill 2 (MCH=one-forced)**	**Skill 3 (MCH=one-per-skill)**	**Skill 4 (MCH=many-forced)**
No calls on set	yes	yes	yes	yes
One active extn call	no	yes	yes	yes
Skill 1 call active	no	yes	yes	yes

Table continues…

Calls delivered?				
Condition	**Skill 1 (MCH=one-request)**	**Skill 2 (MCH=one-forced)**	**Skill 3 (MCH=one-per-skill)**	**Skill 4 (MCH=many-forced)**
Skill 2 or 4 call active	no	no	yes	yes
Skill 3 call active	no	no	no	yes
Extn call held, no other action	no	yes	yes	yes
Skill 1, 2, or 4 call held, no other action	no	no	yes	yes
Skill 3 call held, no other action	no	no	no	yes
Extn call held, then AI/MI selected	yes	yes	yes	yes
Skill 1,2,3, or 4 call held, then AI/MI selected	yes	yes	yes	yes

Agents and supervisors in on-request MCH splits or skills can use Queue Status, VuStats, and BCMS/CMS reports to determine if a waiting call must be answered immediately.

MCH considerations

- Agents can receive multiple calls only when in the auto-in or manual-in work mode. All Forced MCH calls are terminated at the agent station, not with zip tone. Requested MCH calls are delivered with a ring or zip tone.
- Agents can toggle between the auto-in and manual-in work mode.
- If an agent selects the After Call Work (ACW) or the Auxiliary (AUX) work mode with calls on hold, the work mode is pending until all calls complete or until an manual-in call completes. New ACD calls are not delivered when the AUX work mode is *pending*. When an ACD or Direct Agent (DA) call with pending ACW completes, the agent enters ACW. When an agent is active on a non-ACD call with ACW pending, the agent can receive the Forced MCH calls.
- If an agent is either in the auto-in work mode and active on an ACD or direct agent call, or in auto-in or manual-in work mode and active on a non-ACD call, and a manual-in ACD or DA call abandons from hold, the agent is pending for the ACW work mode and the after-call button lamp flashes.
- If an agent reconnects to an ACD or direct agent call on hold, the work mode changes to the call work mode, that is, auto-in or manual-in.
- Do not use Forced MCH with Direct Department Calling (DDC) distribution because the first agent continues to receive calls until all line appearances are busy.

MCH interactions

Interaction	Description
Automatic Hold	To answer a ringing ACD call, an agent in a many-forced, one-forced, or one-per-skill split or split or skill pushes the line-appearance button. If automatic hold is administered, the active call is automatically placed on hold. Otherwise, the agent must first push hold.
Call Work Codes (CWCs) and Stroke Counts (SCc)	Agents who handle multiple ACD calls simultaneously with MCH can enter CWCs and Stroke Counts. When an agent does so with multiple calls on the station, the code/count is associated with the last call the agent handled. If an agent enters a code/count during an active call with calls on hold, the code/count is associated with the active call. If an agent with on-request MCH is active on a call that requires forced entry of CWC or stroke counts and places the call on hold without entering a code/count, the agent cannot request another call. If agents with many-forced MCH are in a split or skill with forced entry of CWC or stroke counts, the agent are forced to handle an ACD call even if the agents have not entered a code or a count.
Direct Agent Calling (DAC)	Agents can handle multiple direct agent calls if their direct agent skills have MCH. The queue-status indicator is not lit when a direct agent call queues to a split or skill. Agents are notified that calls are waiting with a ring ping and a flashing current-work-mode lamp.
Forced Agent Logout from ACW	An agent in ACW is logged out because the Forced Agent Logout from ACW timer has expired, even if the agent has ACD calls on hold.
Move Agent While Staffed	An agent with a move pending can place a call on hold and request another ACD call. All calls and ACW must complete before the pending move occurs.
Non ACD calls	If an agent activates the Auto-In or Manual-In work mode with calls on hold, the agent can answer or originate a non ACD call. With on-request MCH, the agent is temporarily unavailable for ACD or direct agent calls. With forced MCH, a call can be delivered. If an agent in ACW reconnects to an AUXIN/AUXOUT call, the agent remains in ACW.
Queuing	When an agent is available, the agent is placed at the end of the queue for Uniform Call Distribution (UCD) hunt groups or at the bottom of the skill type for Expert Agent Distribution (EAD) hunt groups, or is made available for Direct Department Calling (DDC) hunt groups. When the agent becomes the most available according to group type (UCD, EAD, or DDC), the agent receives a queued ACD or direct agent call. If the last agent on a forced MCH split or skill is pending for AUX work mode in a non vector-controlled split, the agent must empty the queue before going to AUX work mode. The agent continues receiving ACD calls until the queue is emptied.

Table continues…

Interaction	Description
Redirection on No Answer (RONA)	If an agent has a call active or on hold and the RONA timer expires for another ringing ACD call, RONA redirects the alerting call back to the split or skill or administered VDN. The agent is not taken out of service when the call redirects, but is placed at the bottom of the Most Idle Agent (MIA) or Expert Agent Distribution (EAD) queue.
Restricted line appearance	If you administer last-available line appearance as Restricted Last Appearance for an agent phone, the agent does not receive additional ACD calls because the appearance is reserved for making conference or transfer calls.

Proactive Contact outbound calling improved reporting

Avaya Call Center customers using a Proactive Contact integration can track and analyze the time agents in the AUX work mode spend on outbound calls using the same real-time and historical CMS reports used by customers with other Outbound Call Management (OCM) applications. This feature offers the following benefits:

- Delivers outbound Proactive Contact calls to agents that are in the AUX work mode as though the calls were ACD calls associated with a skill specified for reporting.
- Includes calls in the Least Occupied Agent (LOA) occupancy calculations since Communication Manager treats the calls as ACD calls.
- Tracks switch-classified and non switch-classified outbound calls on CMS as ACD-OUT calls instead of as AUX -IN calls.
- Allows Avaya IQ to track calls using the Proactive Contact event stream instead of the Management Information System (MIS) link Switch Protocol Interpreter (SPI) events off the connected Communication Manager.

Prerequisites for using Proactive Contact outbound calling improved reporting

You can use the improved integration with Proactive Contact outbound calling only if all of the following Call Center conditions are true:

- The **Expert Agent Selection (EAS)** field is set to `y` on the System-Parameters Customer-Options screen.
- The **ASAI Link Plus Capabilities** field or the **Computer Telephony Adjunct Links** field is set to `y`. Both fields are located on the System Parameters Customer-Options screen.
- The **Call Center Release** field is set to 4.0 or later on the System Parameter Customer-Options screen.

Switch-classified and non switch-classified calls

Switch-classified outbound calls are outbound calls placed by the Proactive Contact dialer and connected to the agents after the Communication Manager Call Classifier determines that the call has been answered.

Non switch-classified outbound calls are outbound calls that are automatically launched by Communication Manager and connected to an available agent during call setup. This configuration is also referred to as agent-classified calling.

The Proactive Contact Outbound Calling Process

Acquiring agents for outbound calling

About this task

The process described in this section applies to both switch-classified and non switch-classified calls. Proactive Contact selects agents specified for inbound calls and uses the agents for outbound calls as follows:

Procedure

1. Agents are assigned both inbound skills and a skill defined for outbound calling.
2. Agents log in to both Communication Manager and Proactive Contact and take inbound calls in the auto-in or manual-in mode.
3. Agents select an outbound campaign using the Proactive Contact terminal.
4. Proactive Contact acquires agents who have selected an outbound campaign when Proactive Contact determines that current staffing is more than adequate for handling inbound calls.

   The details are as follows:

   a. Proactive Contact obtains an available agent by placing a call to the outbound skill using an ASAI Third-Party Make Call operation with a phantom number as the originator.

      The call is made to a VDN whose vector has a queue-to outbound skill step. This setup is used to acquire agents for outbound calling.

   b. The **queue-to** step selects an available agent and Proactive Contact then changes the work state of the acquired agent to the AUX work mode using the ASAI Change Agent Work Modes request feature and then drops the connection.

   c. Proactive Contact then uses the Third-Party Make Call operation to send the call to an announcement extension using the acquired agent as the originator.

      The agent hears a recording that says, `You are acquired for outbound calling`. The connection then drops.

5. Proactive Contact launches an outbound call.

Launching switch-classified outbound calls

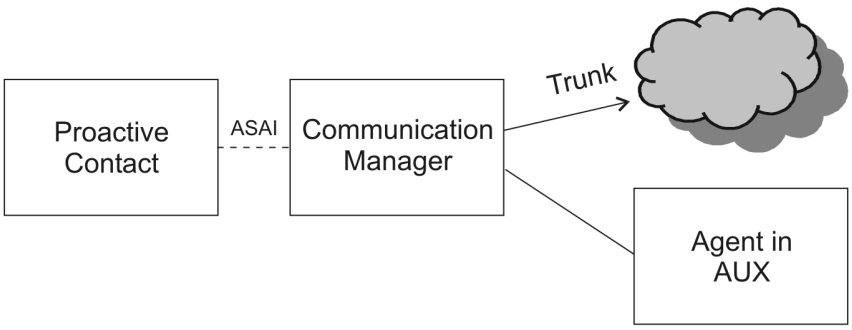
Procedure

1. Proactive Contact launches a switch-classified outbound predictive call request through an ASAI Third-Party Make Call operation using a VDN as the originator.
2. When the call is classified as answered, Communication Manager connects the call to the originating VDN.

   The adjunct routing step in the assigned vector requests a route.
3. Proactive Contact does one of the following steps:
   - If there is an acquired agent in AUX work mode that is available for that campaign, Proactive Contact immediately has Communication Manager connect the call to that agent using a route-select message.
   - If there is no available agent, Proactive Contact waits to send the route-select message until an agent is available. The vector provides a message to the called party, waits until the called party drops, or provides other programmed treatment.
4. The call delivery is reported to the reporting adjuncts.
   - CMS receives events for an ACD-OUT OCM (ACDO) call that is associated with the assigned reporting skill instead of an AUX-IN call.
   - Avaya IQ (if connected) receives a message to ignore these events.

Sample of Proactive Contact launching an outbound switch-classified call to an agent in AUX work mode

The following example shows how Proactive Contact launches an outbound switch-classified call to an agent in AUX work mode.



ASAI Third-Party Make Call operation for VDNx → Communication Manager places outbound call and classifies it as answered → Call is connected to VDNx
Vector z is processed
Route request sent to Proactive Contact
Route-select message sent to route the call to the available agent

```
VDNx - vector z
1. adjunct routing link 1
2. wait-time 2 secs hearing silence
3. announcment 42015
4. wait-time 10 secs hearing silence
5. goto step 3 unconditionally
6. disconnect after announcement 42020
```

Launching non switch-classified outbound calls

1. Proactive Contact launches an outbound call using an ASAI third-party make call operation to an available acquired agent in AUX work. The priority call parameter for the third-party make call operation must be set to `y`.
2. Communication Manager connects the call to the agent at the same time the outbound connections are being established.
3. The call delivery is reported to the reporting adjuncts.
   - CMS receives an ACD-OUT OCM (ACDO) call that is associated with the assigned reporting skill instead of an AUX-IN call based on the setting of the Third-Party Make Call priority flag. The call is reported to CMS only if Proactive Contact non switch-classified calls are administered. For more information, see “Administering PC non switch-classified calls for improved reporting” in the *Administering Avaya Aura® Call Center Elite* document.
   - Avaya IQ receives a message to ignore these events.

Proactive Contact improved reporting interactions

Interaction	Description
BCMS	You can use BCMS to track Communication Manager switch-classified and non switch-classified calls. BCMS reports for Proactive Contact (PC) include ACD-OUT triggered and ACD time tracking. There is no ring-time associated with PC calls since PC agents are logged in to auto-answer stations.
Forced Agent Logout by Clock Time	Do not use the Call Center 4.0 introduced Forced Agent Logout by Clock Timer feature with the Call Center 4.0 introduced Improved Integration with PC Outbound Calling capability. The PC system places a PC agent in the AUX work mode when the agent is making an outbound PC-imitated call. If the administered time for Forced Agent Logout by Clock Time is reached, the PC agent is logged out immediately.

Queue Status Indications

With Queue Status Indications, you can assign queue status indicators for ACD calls based on the number of split or skill calls queued and the time in a queue. You can assign the indications to lamps on agent, supervisor, or attendant phones or on consoles to help users monitor queue activity.

You can also define auxiliary queue warning lamps to track the queue status. On phones and consoles with displays, you can view the number of calls queued and time in queue of the oldest call in the split or skill.

You can use Queue Status Indications to provide status information for attendant groups or other hunt group types such as DDC and UCD. With attendant groups, button such as AQT and AQC, are different from those for split or skill queues, the display shows OPERATOR instead of the split or skill name or extension and all status information applies to the attendant group queue.

Types of Queue Status Indications

There are two types of Queue Status Indications:

- Number of queued calls (NQC) - The system reports the total number of calls, excluding direct agent calls, in queue at a hunt group.
- Oldest queued time (OQT) - The system reports the time in queue of the oldest call in a split or skill queue.

You can also use auxiliary queue warning lamps to provide both types of indications. Install the lamps at any location convenient to agents and supervisors.

If a queue status threshold is reached, the lamp next to the associated button flashes. If calls are queued but the threshold is not reached, the lamp lights steadily. If no calls are queued, the lamp goes dark.

If the OQT or NQC button on a phone or console with display is pressed, the following information is briefly displayed:

- Split or skill name, or extension, if name is not assigned.
- Oldest queued time.
- Number of queued calls.

Queue Status Indication interactions

Interaction	Description
Attendant and Phone Display Timers	The timer and the queue status information can be displayed at the same time. On one-line display phones, the timer is displayed in the last eight display positions and the number of queued calls is not displayed. On two-line display phones, the timer is displayed on the first line and the queue status information is displayed on the second line.
CMS	When you use CMS to move an agent from one split or skill to another, all buttons associated with the first split or skill, including NQC and OQT buttons, become associated with the second split or skill. With EAS skills, the move of an agent just adds or removes skills the agent is logged in to, but the skill assigned to the Queue Status Indicators do not change.

Reason Codes

With reason codes, staffed agents can enter a numeric 1–digit or 2–digit code that describes the reason for entering the AUX work mode or for logging out of the system. Reason codes give call center managers information about how agents spend time.

Use the data to develop precise staffing forecasting models or with schedule-adherence packages to ensure that the agents are performing scheduled activities on time.

You can administer the codes so that entry of the code is forced or optional. You can assign up to 100 AUX reason codes, including a default code of zero (0) and codes from 01 to 99.

You can use VuStats to view the reason code name or number. Use VuStats and Call Management System (CMS) or Avaya IQ to gather historical and real-time reason-code statistics.

Note:

You must set **EAS** to `y` to use reason codes.

Types of reason codes

Reason code type	Description
Forced reason code	If you have administered forced reason codes, agents cannot enter the AUX work mode or log out until agents enter a code. Agents can enter codes 1-99, but not the default code 0. If an agent enters an invalid code or fails to enter a code within the 10-second time out interval, the state change is denied. The agent remains in the current work mode. If the agent enters a Feature Access Code (FAC), the agent hears an intercept tone. If the agent pressed the **aux-work** button, the AUX lamp flutters and becomes dark or lights steadily if the agent was already in the AUX work mode with a different reason code.
Requested reason code	If you have administered requested reason codes, agents need not enter a code to enter AUX work mode or log out. Agents can enter the codes 0-99. If an agent enters an invalid code or fails to enter a code within the time out interval, the agent enters AUX work mode or logs out with default code 0.
Entering AUX reason code	An agent can enter an AUX reason code in one of three ways: • Pressing an **aux-work** button with an assigned code. • Pressing an **aux-work** button with no assigned code and responding to the prompt for a reason code. • Entering an **aux-work** FAC and responding to the dial tone prompt for a reason code. If no calls ring or are active, or held at the phone, agents enter AUX work mode immediately and the AUX lamp lights steadily. Otherwise, the AUX lamp blinks until the agent completes all calls. If a button for AUX work is associated with the reason code that the agent entered, the button lamp lights. If no **aux-work** button is administered, the system lights the first AUX button lamp with no administered reason code. You can assign an **aux-work** button without a reason code. With this button agents with a limited number of buttons can use all 100 reason codes.
Logging out	To log out with a reason code, the agent enters the logout FAC, hears a second dial tone, and enters a reason code. The agent hears confirmation tone and is logged out of the system.

Table continues…

Reason code type	Description
Default code	The default code zero (0) is used when the system moves an agent work mode to the AUX work mode or logs the agent out. For example: • When an agent logs in and is put in to the AUX work mode. • When an agent in the available state receives a non ACD call. • When the agent skill assignments are changed while an agent is staffed, the system automatically logs the agent out and back in. • When an agent forces a logout without entering a code. For example, by pulling the headset. • When an agent who is requested to enter a reason code fails to enter a valid code within the 10-second time out period. • When an agent with requested reason codes enters the pound (#) sign or the asterisk (*) sign.
Reason codes not interruptible	The following reason codes cannot be made interruptible: • IP failure AUX work • Redirection on No Answer (RONA) AUX work • Redirection on Off-PBX Telephone Integration and Mobility (OPTIM) Failure (ROIF) AUX work • Maximum Agent Occupancy (MAO) AUX work • Default reason code on the Reason Code Names screen

Reason code considerations

- For forced logout reason codes, if an agent in the auto-in or manual-in work mode enters the logout Feature Access Code (FAC) but does not enter a reason code, the agent is returned to the available state. ACD calls are delivered even if the agent has left the phone. To prevent call delivery when an agent is not at the station, ensure that agents enter the Auxiliary (AUX) or After Call Work (ACW) work mode before logging out.
- When an agent work mode changes to the AUX work mode and the AUX work reason code type is set to `none`, the agent work mode changes to the AUX work mode with the default reason code even if you have administered a different reason code for the AUX button. You can, thus, complete button administration before activating the feature.
- Do not administer AUX buttons without a reason code for hybrid station sets.
- When an agent in the AUX work mode is active on a non ACD call, the agent cannot immediately change the reason code. A work mode change is pending until the call drops.
- There is a limit to the number of agents who can simultaneously enter a reason code or a Call Work Code (CWC).
- With two-digit reason codes, Communication Manager notifies CMS or Avaya IQ about a change in the work state when agents enter:
  - A two-digit reason code.

- A one-digit reason code followed by the pound (#) sign or wait for 10 seconds after entering a one-digit reason code. If agents press any button during the wait time of 10 seconds, Communication Manager does not notify CMS or Avaya IQ. Furthermore, Communication Manager might not change the agent work mode to AUX work.

  Avaya recommends that agents press the pound (#) sign as the second digit for one-digit reason codes.

Reason code interactions

Reason Code	Description
Abbreviated Dialing (AD)	You can administer Feature Access Codes (FACs) for the AUX work mode or logout with or without an associated reason code on automatic-dial buttons or in abbreviated-dial lists. At the reason code prompt, when an agent selects an abbreviated-dial or automatic-dial button, the first digit of the button is taken as the reason code.
Agents in multiple skills	When an agent who is assigned to multiple skills enters the AUX work mode with a reason code, the agent enters the AUX work mode for all the skills with the same reason code.
Adjunct Switch Application Interface (ASAI)	With ASAI, a host can log an agent out and place an agent in the AUX work mode with a reason code. The host can query the current work mode of the agent and receive the reason code that is associated with the AUX work mode.
Auto-Available Split/ Skill (AAS)	The system logs AAS agents out for Redirection on No Answer with the default reason code.
Basic Call Management System (BCMS)	Statistics about the Aux work mode by reason code are not available in BCMS reports.
Call Management System (CMS)	CMS tracks agent time in the AUX work mode by reason code and displays the reason codes for agents currently in that mode. When an agent is moved from CMS while the agent is staffed, the system logs the agent out using the default code, and then logs the agent back in again. If an agent is in the AUX work mode when moved, the agent is returned to the AUX work mode with the same reason code after the move is completed.
Direct Agent Calling (DAC)	When a DAC is queued for an agent in AUX work mode with a reason code, the appropriate AUX button lamp flutters to alert the agent to the queued call. If there is no AUX button lamp, agents receive an audible alert, that is, ring-ping or a call-waiting tone. If there is an AUX button with no assigned reason code administered, the AUX button lamp flutters.
Redirection on No Answer (RONA)	When a call is redirected on no answer, an agent is placed into AUX work mode or is logged out with the RONA AUX work reason code, if the agent is in an Auto-Available Skill (AAS).
Redirection on IP Failure (ROIF), Redirection on OPTIM Failure (ROOF), and SIP Agent Reachability	The ROIF, ROOF, and SIP Agent Reachability features use the respective AUX work reason code.

Table continues…

Reason Code	Description
Forced Agent Logout for Unreachable Reason Code	You can define the reason code number in the range 0 to 9, which specifies that agents are logged out because of unreachability. The default value is 0.

Redirection on IP Failure

Redirection on IP Failure (ROIF) applies to:

- Agent login IDs with IP endpoints using H.323 with auto-answer only.
- Agent login IDs using H.323 CTI soft clients with either auto-answer or manual-answer.

Calls can sometimes be lost when Communication Manager delivers the calls to auto-answer agents that use IP phones. With ROIF, Communication Manager redirects the calls back in the queue or to the specified Vector Directory Number (VDN) when Communication Manager cannot connect calls to the stations due to loss of IP connectivity.

Note:

For auto-in agents, Timed ACW takes precedence over Forced Agent Logout from ACW.

How ROIF works

- When the **Switch Hook Query Response Timeout** field is set to y, Communication Manager checks for IP connectivity before delivering a call to the auto-answer agent using an IP phone.
- When you set the **Switch Hook Query Response Timeout** value, the degree of network congestion or delays determines the speed of the switch hook query response. The range value for the switch hook timer is 500-5000 ms. IP hardphones that are located geographically close to Communication Manager should respond to the timer within the 500-750 ms range. IP Agent endpoints on a PC will take longer to respond, requiring that the timer be set to 2000 ms or greater.
- Communication Manager invokes redirection upon detection of IP connectivity loss. The switch hook query is not acknowledged within the administered time out period. Communication Manager moves the call back in the queue or forwards the call to the VDN specified on the Hunt Group screen.
- Communication Manager puts a non Auto-Available Split/Skill (AAS) agent into the AUX work mode, redirects the call to the split or skill queue or the specified VDN if Communication Manager detects IP connectivity failure before delivering the call to the agent. If **Reason Codes** is set to y, CMS reports the change to the AUX work mode.
- Communication Manager logs out AAS agents instead of putting the agents in the AUX work mode.

You can retain the active VDN context when Communication Manager redirects a call due to IP connectivity failure. When you set the **Retain Active VDN Context** field as `y`, Communication Manager retains and uses the VDN context from the original active VDN.

If you set **Retain Active VDN Context** as `n` and ROIF occurs, Communication Manager uses the context of the applicable redirect-to VDN that is assigned to **Redirect on IP/OPTIM Failure to VDN** on the Hunt Group screen.

The VDN context includes the following information:

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement (VOA) Extension
- VDN Skills (1st, 2nd, 3rd)
- VDN Return Destination (VRD)

  Note:

  The VRD is set before being RONA/ROIF/ROOF redirected, and does not change by subsequent routing. The staffed agent receiving the redirect call views 'CR' at the right end of the display indicating a RONA/ROIF/ROOF redirected call.
- VDN Timed After Call Work (ACW) interval
- Best Service Routing (BSR) application
- BSR available strategy
- BSR tie strategy
- Display VDN for route-to Direct Agent Calling (DAC)
- Trunk Adjunct Switch Application Interface (ASAI) messages
- BSR local treatment
- VDN variables
- VDN time zone offset

If you retain the active VDN context, you can set up a generic VDN-vector that caters to calls redirected from multiple VDNs.

If Communication Manager redirects the call to a VDN when routing directly to a hunt group rather than through a VDN, the redirect to VDN is the active VDN regardless of the setting of the **Retain Active VDN Context** field.

ROIF considerations

Reason codes for ROIF

You can administer the IP failure AUX work reason code as a non-zero number not currently in use so the system can distinguish between AUX work changes that are caused by miscellaneous changes and those caused by a loss of IP connectivity.

Using the auto-in or manual-in button to test IP connectivity

If an agent has not received a call within the usual time frame, the agent must press the **auto-in** or **manual-in** button to ensure IP connectivity. The lamp update for the **aux-work** button is not always received by the IP station due to the loss of IP connectivity.

Call redirection alert button

A **noans-alrt** button can be assigned to non-SIP agent or supervisor phones to indicate that a call has been redirected.

ROIF interactions

The ROIF call and agent interactions are the same as for RONA, with the following additions:

- ROIF is applied system-wide. The default for the system is *not active*.
- ACD calls delivered from the split or skill queue and Direct Agent Calling (DAC) work the same as RONA. ROIF first attempts to redirect DAC to the coverage path. If the call cannot go to coverage, the call is redirected to the VDN that you administer in the **Redirect on IP/OPTIM Failure to VDN** field on the Hunt Group screen. Ensure that the direct agent skill group has a redirect VDN for occurrences of ROIF. If **Redirect on IP/OPTIM Failure to VDN** is not assigned, the call is re-queued to the same skill at a high priority. If there are no queue slots available, the caller hears a busy signal. If all fails, the caller receives a ringback until the system detects a caller disconnect. This also applies to priority DACs when the **Redirect on IP/OPTIM Failure to VDN** is not specified.
- The agent is unaware that the line is in AUX work during an IP connectivity failure. If the system restores connectivity during the TCP retry period, the station lamp indicates that the line is in the AUX work mode.
- The only indication that CMS receives after a ROIF has occurred is a state change and the resultant flow-out, flow-in, and DFWD-unknown indications for the call. Unlike RONA, the action is specifically not identified, other than by the reason code.
- As with RONA, the caller will hear a ringback during the ROIF timer period, and redirected calls receive a ringback when re-queued. Calls redirected to a VDN will hear the feedback determined by the assigned vector. If the call cannot be re-queued because no queue slots are available, the caller will hear a busy signal until the caller disconnects the call. In this case, a DFWD-unknown message is sent to CMS to decrease the tracking of call ringing.
- As with RONA, if the **Retain Active VDN Context** field value is `y`, the calls are redirected to the specified VDN with the previous active VDN context information.
- ROIF does not provide a lamp indication to the call center supervisor as is done for RONA.
- ROIF applies to AAS agents, Voice Response Units (VRUs), or Interactive Voice Response (IVR) ports if the ports are connected through IP and auto-answer is active. AAS lines are logged out if an IP failure is detected during call delivery.
- A manual answer station is always on-hook and, until answered, the call is delivered by ringing the station. With IP connectivity, the system invokes existing IP failure mechanisms

when the phone is not ringing so that the call is not lost and delivered to the next available agent.

Redirection on No Answer

Redirection on No Answer (RONA) redirects a ringing ACD skill or direct agent call after an administered number of rings. RONA prevents an unanswered call from ringing indefinitely especially for IVRs/VRUs where more than one port fails. The call can redirect either to a skill or to a VDN for alternative call handling. Direct agent calls route to the coverage path of an agent or to a VDN if no coverage path is administered.

You must enable **ACD** to use RONA. Administer RONA for each ACD hunt group. You can use RONA in Auto-Available Splits/Skills (AAS), or in splits/skills with agents operating in the auto-in or manual-in work mode. You can administer RONA for vector-controlled or non vector-controlled splits/skills. RONA only applies to manual answer station operation where the station or port is rung waiting for answer. RONA does not work with auto-answer configurations.

Do not administer RONA for splits or skills controlled by adjuncts or AUDIX, or for auto-answer agents assigned splits or skills, since calls must ring at a telephone to be redirected.

You can specify whether to retain the active VDN context when RONA redirects a call to an alternate VDN defined as the redirect VDN due to an agent that has not answered. When you administer the **Retain Active VDN Context** field as `y`, the VDN context from the previous active VDN is retained and used after the call is redirected to the specified redirect VDN. If you administer the **Retain Active VDN Context** field as `n` and RONA occurs, the system uses the context of the applicable redirected to VDN. The active VDN for a call is based on VDN Override rules, normally being the first VDN called unless overridden by a routed to VDN. For more information, see “VDN Override” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

The VDN context includes the following information:

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement (VOA) Extension
- VDN Skills (1st, 2nd, 3rd)
- VDN Return Destination (VRD)

  Note:

  The VRD is set before being RONA/ROIF/ROOF redirected, and does not change by subsequent routing. The staffed agent receiving the redirect call views 'CR' at the right end of the display indicating a RONA/ROIF/ROOF redirected call.
- VDN Timed After Call Work (ACW) interval

- Best Service Routing (BSR) application
- BSR available strategy
- BSR tie strategy
- Display VDN for route-to Direct Agent Calling (DAC)
- Trunk Adjunct Switch Application Interface (ASAI) messages
- BSR local treatment
- VDN variables
- VDN time zone offset

If you choose to retain the active VDN context, you can set up a generic VDN-vector combination that caters to calls redirected from multiple VDNs with specialized treatment based on the context parameters of the previous active VDN.

If the system redirects the call to a VDN when routed directly to a hunt group rather than through a VDN, the redirect to VDN is the active VDN irrespective of the setting of the **Retain Active VDN Context** field.

RONA detailed description

When RONA is invoked for a call, the system:

- Places an agent in the Aux work mode and the agent is unavailable to receive calls from other splits or skills. In an AAS, the agent is logged out.
- Redirects split or skill calls back to the split or skill or administered VDN. Redirected calls are requeued at the highest priority so that the calls are distributed before any other split or skill calls.
- Sends a message to CMS. When a RONA time out occurs, the **noans-alrt** lamp for the split or skill lights steadily. The supervisor presses the **noans-alrt** button to view the login ID or the extension and name of the last agent timed out with RONA.
- Records the redirection in BCMS or CMS.

RONA application examples

VRU applications

Typically, RONA is used with IVR/VRU applications in AAS configurations. RONA detects VRU failures and provides alternate operation. For example, an adjunct port failure is not detected by ACD call processing. RONA detects the failure, takes the port out of service, and provides notification of the failure.

Use Call Vectoring for flexible call handling in case of a VRU failure. Assign RONA to a converse split or skill connected to the IVR system or to equivalent VRU ports. Whenever RONA times out on a ringing call delivered using the **converse-on** command to an AAS VRU port, the port is logged out and the call is redirected back to the converse split or skill.

With a complete VRU failure, all VRU ports are eventually logged out and vector processing for the **converse-on** command bypasses that step for new calls.

The following vector example shows how to provide automatic backup for a complete VRU failure.

Example vector - Providing automatic backup for a complete VRU failure

```
                                CALL VECTOR

01 wait-time 0 secs hearing ringback

02 converse-on split... (VRU returns the digit "1" as a return code followed by
additional digits for the application)

03 collect 1 digits after announcement none

04 goto step 6 if digits = "1"

05 goto vector xxx (for backup when the VRU fails)

06 collect 2 digits after announcement none

07 ...
```

In the example vector shown above, the application works as expected as long as the VRU returns the digit string, which includes a return code of 1. In this case, the condition in Step 4 is satisfied and the program branches to Step 6, which provides normal application processing.

On the other hand, if all VRU ports in an AAS split or skill are logged out by a RONA time out, the **converse-on** command step (Step 2) is skipped, and no digits are collected by Step 3 (after the 10-second time out). The condition in Step 4 is not satisfied and vector processing proceeds to Step 5, which branches to vector xxx to connect the call to an agent.

Other applications

You can use RONA for applications that involve human agents with manual answering and other adjunct applications such as Home Agent. For example, a call is not answered because an agent left without entering the AUX work mode or logging out. You can use RONA to make the non answering agent unavailable and redirect calls to another agent or to the RONA VDN.

RONA routing sequences

The following tables describe how RONA redirects split or skill calls and direct agent calls.

RONA routing sequence for DAC	
Redirection destination	**Explanation**
Coverage Path	DACs redirect to a coverage path, if one exists. Priority calls do not route to coverage.
RONA VDN	Calls redirect to the VDN specified when RONA occurs. If no coverage path exists when a VDN is administered for RONA, direct agent calls redirect to the VDN. If the **Retain Active VDN Context** field is y, the system retains the VDN context from the previous active VDN after the call redirects. Otherwise, the system uses the context of the redirect to VDN.

Table continues…

RONA routing sequence for DAC	
Redirection destination	**Explanation**
VDN Return Destination	For external calls, if neither a coverage path nor a RONA VDN are administered, DACs redirect to the VDN return destination extension.
None	Calls continue ringing.

RONA routing sequence for split or skill calls	
Redirection destination	**Explanation**
RONA VDN	If a RONA VDN is administered, calls redirect to the VDN.If the **Retain Active VDN Context** field is y, the system retains the VDN context from the previous active VDN after the call redirects. Otherwise, the system uses the context of the RONA VDN.
Requeue to split or skill	If a RONA VDN is not administered, the call redirects back to the split or skill at a priority that is above the highest priority.
Coverage Path	In non vector-controlled splits, if calls cannot requeue to the split, the calls redirect to the split coverage path if one is administered.
VDN Return Destination	For external calls, if a coverage path or a RONA VDN is not administered and calls cannot requeue, the calls redirect to the VDN return destination extension.

Using BCMS/CMS reports with RONA

You can use BCMS and CMS reports to determine which agents had RONA time out and how were the calls redirected. The CMS Exception report lists agents who were timed out and made unavailable.

You can determine which agents were in the AUX work mode or logged-out with AAS.

With R3 CMS, you can use the real-time Split Status report to see which agents are in the AUX work mode, but you need a custom report to see logged-out agents.

With BCMS, use SAT to create a list of unstaffed agents for the split to see which agents are logged out (for AAS applications). With EAS, list agent-loginid specifying unstaffed and AAS set to yes.

With BCMS, agent mode changes to the AUX work mode appears in the BCMS Split (AGENT) Status report. In an AAS split, agents log out, so the agents do not appear in the Split Status report. When the call is requeued, the System Status report shows only the AVG ANSW SPEED time and AVG ABAND TIME time for the requeued call. The Historical Split and system reports show both a FLOWOUT (primary split) and FLOWIN (redirected split) for requeued calls, while the VDN report shows only a FLOWOUT.

Direct agent calls are recorded as ACD split or skill calls but the flowout is recorded only if an agent’s coverage path requeues the call to a split or skill.

Since BCMS does not report exceptions, RONA events are not reported. If you have BCMS, use the RONA split or skill lamp indicator for RONA event indication.

Generic VDNs for redirected calls handling

For the RONA, ROIF, and ROOF calls, you can set up a generic VDN and assigned vector that:

1. Utilizes the VDN/call parameters carried over by the VDN Override function to specialize the treatment and to further route the call
2. Facilitates specialized treatment for calls redirected from several different applications, based on their respective original active VDNs

Setting up a generic VDN to handle redirected calls

About this task

To use the VDN context or call parameters carried by the VDN Override function:

Procedure

1. Administer the following fields for the hunt group:
   a. Administer **Redirect On No Answer to VDN** or **Redirect on IP/OPTIM Failure to VDN** fields as `2110200`, which is the generic VDN number for call redirection.
   b. Administer the **Retain Active VDN Context** field as `y`.
2. Set up a generic vector for RONA, ROIF, and ROOF call handling.

```
                                          CALL VECTOR
1 announcement V1 (Using a VDN variable for the application to give a
personalized announcement)
2 queue-to skill 1st pri t (puts the call in a queue personalized for the
application)
3 queue-to skill 2nd pri t (adds another queue specific for the application)
4 announcement V2 (a further personalized wait announcement)
```

In the example, vector step 1 plays a personalized announcement identified by the extension stored in the active VDN's variable V1, that the caller gets if the original call delivery fails. The announcement is specified by a VDN variable (V1) assigned to the active VDN instead of the actual extension number so redirected calls from other VDNs can hear the announcement defined by the V1 assigned to the VDN the call was redirected from. Step 2 queues the call to the skill assigned to the 1st skill parameter assigned to the active VDN with top priority. Step 3 queues the call to the 2nd skill assigned to the active VDN Step 4 is for giving a personalized announcement for the waiting caller utilizing the 2nd VDN Variable (V2) assigned to the active VDN.

For a call routed to the generic vector for the redirected calls, the answering agent gets the following display: `303-555-0003 to Billing CR`

Where:

- 303-555-0003 = caller number.
- Billing = The name of active VDN the call is redirected from.
- CR = Indication that the call is redirected and not a directly routed call.

Returning AAS agents to service

When RONA redirects a call that was directed to an AAS, the agent is logged out. To return an AAS agent to service, readminister the agent as a member of the AAS split or skill to be logged in again in one of the following ways:

- For ACD splits, remove the agent from the split and then resubmit the split Hunt Group screen with the agent added to it. Alternatively, administer the agent in a different location in the split members list on the Hunt Group screen. Use the **`list unstaffed-agents`** command to get a list of all AAS agents that have been logged out, not just AAS agents that were logged out because of a RONA timeout.
- For EAS skills, readminister the Agent LoginID screen so that the AAS agent is automatically logged in. To determine which EAS agents are logged out, use the **`list agent-loginid`** command.
- For ACD splits and for EAS skills, you can busy-out the AAS agent station with the **`G3-MT busyout station`** command and release it with the **`release station`** command. Releasing the AAS agent station automatically logs the agent in. If all AAS agent ports on the circuit pack had a RONA timeout, busy-out and release the entire circuit pack.
- Use CMS Move Agents to move up to 32 agents at a time into a dedicated unused split or skill and then move the agents back into the AAS split or skill. You can set this up using the timetable on a manual-scheduled basis to activate when the VRU has been restored to service after a failure.
- Use ASAI to log the logged-out agents back in using ASAI login request messages.

RONA considerations

Set the **Redirection on No Answer Aux Work Reason Code** field on page 14 of the Feature-Related System Parameter screen to a non zero number not currently being used so the system can distinguish between the AUX work changes caused by miscellaneous changes and those caused by no answer.

- RONA can time out when an agent is at the station if the agent does not answer the call within the administered number of rings or has selected another work mode while a call is ringing. RONA handles the call as usual, making the agent unavailable. With ACD splits, the aux-work lamp when lit indicates that the agent is unavailable. The agents can press the **auto-in** or **manual-in** button to become available.
- Specify a coverage path or VDN for redirection for non vector-controlled splits or for logical agent IDs with EAS direct agent calls to ensure that calls are always redirected.
- A **noans-alrt** button can be assigned to non-SIP agent or supervisor phones to indicate that a call has been redirected. When a supervisor presses the **noans-alrt** button, the phone display shows the login ID or extension and the name of the last agent timed out with RONA.

RONA interactions

Interaction	Description
Auto-Available Split/ Skill (AAS)	Use AAS with RONA for Voice Response Unit (VRU) ACD non-ASAI adjunct-controlled split or skill applications. Assign AAS only to ACD hunt groups. When all lines in a vector-controlled AAS split or skill are logged out, the split or skill is treated as unavailable and vector processing skips the step in the vector for new calls. If RONA occurs on the last VRU port in an AAS split, the call is not requeued to the converse split but is processed by the next vector step. Any calls queued to a split or skill that has been taken out of service can be left at the split or skill. When the system reinitializes, all busied-out ports are automatically logged back into the AAS splits. New calls cause a RONA time out if the adjunct or agent still does not answer after the system reinitializes.
Abandoned Call Search	Abandoned Call Search, if defined for a trunk, is reapplied to call on that trunk that RONA requeued whenever the calls are routed to another agent.
Agents in multiple splits	When a RONA time out occurs, an agent is placed in AUX work mode with notification to Call Management System (CMS) for all splits that the agent is logged into. The agent is responsible for becoming available in each split. In AAS, agents are logged out of all splits that agents are logged into. You must log agents back into the AAS splits.
Agent logout	An agent can log out from a multifunction set while an ACD call subject to RONA is ringing the set. However, if the agent logs out before RONA times out, RONA timing is canceled, and RONA redirection and notification occur immediately.
Agent work modes	If an agent presses the **acw** button with an ACD call ringing, the change request is pending. If the agent has a pending change to ACW before a RONA time out occurs on a ringing ACD call, RONA timing continues. At time out, the call is redirected, CMS is notified, and the agent is placed in AUX work which overrides the pending ACW request. If an agent presses the **aux-work** button with an ACD call ringing, the work mode change request is pending. With ACD splits or skills, as the RONA time out changes the state to AUX work, there is no conflict with the pending AUX work change request. With AAS splits or skills, an agent-initiated work mode change to AUX work is denied as per existing operation.

Table continues…

<table>
<tr><th>Interaction</th><th>Description</th></tr>
<tr><td>Adjunct Switch Application Interface (ASAI)</td><td>
RONA applies to vector-processed calls that are routed by an adjunct to a split or an agent as a direct agent call.

You can assign RONA to ASAI adjunct-monitored splits and adjunct-monitored calls. An event report is not sent to the ASAI adjunct when a RONA time out puts an agent in the AUX work mode.

The adjunct makes an agent query as part of the value query capability group to determine the agent work state. Once the call is requeued to the split, the adjunct receives a call-queued event report if event reporting is active for the domain (VDN or non vector-controlled split or skill).

An adjunct-monitored split or skill can be assigned as an auto-available split or skill. The logout event for an AAS split or skill is sent to the adjunct when RONA time out logs an agent out.

You cannot assign RONA to an adjunct-controlled split or skill. An adjunct-controlled split or skill cannot be an AAS.

ASAI Interactive Voice Response (IVR) or VRU applications are configured with non vector-controlled splits or skills using manual-answer operation on analog lines to the IVR ports. The ASAI link provides event notification for the ACD split or skill for enhanced services. In addition, you can log in and log out the ports as required. AAS splits or skills are not used for this application because the ASAI link controls the login or logout.

You can assign RONA to the splits or skills to detect failure conditions in the same manner as non-ASAI VRU applications. RONA does not notify the IVR system of AUX work mode changes. An ASAI IVR system cannot query to determine the states of its ports. You must restore ports manually after a failure using the IVR system management screens. Complete failure is automatically restored when the IVR system reinitializes.

The following table describes ASAI events that the communication server sends the adjunct for various stages of the RONA call. Also included are the ASAI associations for which the events are provided. For the split or skill to have Notification association active, the split or skill must not be vector-controlled or adjunct-controlled.

When a call is redirected using ASAI Redirect Call, the RONA timer is canceled.
<table>
<tr><th>Stage of call</th><th>ASAI event</th><th>ASAI associations</th></tr>
<tr><td>1. RONA time out</td><td>Logout for AAS</td><td>Domain (agent) control</td></tr>
<tr><td rowspan="2">2. Call redirected to split</td><td>Call redirected</td><td>Domain (station) control for agent extension call is leaving</td></tr>
<tr><td>Call queued (only if the call queues)</td><td>Domain (station) control for new agent and for internal originator call control, notification</td></tr>
<tr><td>3. Call redirected to VDN processing</td><td>Call redirected</td><td>Domain (station) control (for agent extension that call is leaving)</td></tr>
</table>
</td></tr>
</table>

Table continues…

<table>
<tr><th>Interaction</th><th>Description</th></tr>
<tr><td></td><td>
<table>
<tr><th>Stage of call</th><th>ASAI event</th><th>ASAI associations</th></tr>
<tr><td></td><td>Call redirected (only if call is redirected to a VDN with Notification active)</td><td>Notification</td></tr>
<tr><td></td><td>Call offered to Domain</td><td>Notification (VDN)</td></tr>
<tr><td>4. Call delivered to agent</td><td>Alerting</td><td>Domain (station) control,
(for new agent & for internal originator) call control, notification</td></tr>
<tr><td>5. Call routed to split’s coverage path</td><td>Call redirected followed by existing operation of ASAI Events</td><td>Domain (station) control (for agent ext that call is leaving)</td></tr>
<tr><td>6. Infinite feedback to caller</td><td>Call redirected</td><td>Domain (station) control (for agent ext that call is leaving)</td></tr>
<tr><td>7. Continue vector processing</td><td>Call redirected followed by existing ASAI events</td><td>Domain (station) control (for agent ext that call is leaving redirecting to VDN)</td></tr>
<tr><td>8. Call routed to direct agent’s coverage path</td><td>Call redirected followed by existing operation of ASAI events</td><td>Domain (station) control (for agent ext that call is leaving)</td></tr>
</table>
</td></tr>
<tr><td>Attendant return call</td><td>If an attendant extends a call to an ACD split or VDN for which the return call timer is not activated, the call does not interact with RONA. The Attendant Return Call Timer is not set if an attendant extends the call to another attendant.</td></tr>
<tr><td>AUDIX Transfer</td><td>RONA applies to a call transferred by AUDIX to an ACD split. A redirected call to AUDIX does not go to split or agent coverage after it is transferred out of AUDIX. If RONA times out on this type of call, the call cannot be redirected.</td></tr>
<tr><td>Automatic answering</td><td>If an agent with automatic answering receives a call with zip tone instead of ringing, RONA timing is canceled.</td></tr>
</table>

Table continues…

Interaction	Description
Call Coverage	Direct agent calls are redirected to the agent's coverage path if a path is administered. A temporary bridged call appearance is not maintained for a call directed to an ACD hunt group or VDN, or for a direct agent call. When a call is redirected to a split or skill, the Coverage Subsequent Redirection/ CFWD No Answer timer is started on the call. Covered calls go to the next point in the split or skill coverage path. If no other point is available to accept the call, the call remains queued or continues to ring the current coverage point. When RONA times out at the coverage point, the following occurs: • RONA does not reset the Subsequent Redirection/CFWD No Answer timer. The timer that expires first controls the call. • If the coverage point for a covered call is a direct agent logical agent ID whose skill has RONA, and if RONA times out first, the call is sent to the next point in the skill coverage path, not to the agent's coverage path. The Subsequent Redirection/CFWD No Answer timer is reset when the call is redirected to the next coverage point. • If RONA was applied to an ACD call that was a previously redirected coverage call (that is, the RONA split was a point in the coverage path), RONA is used to requeue the call as specified for a non covered call. However, the call is not designed to go to split coverage or forwarding. The Subsequent Redirection/ CFWD No Answer timer is reset if RONA requeues the call to the RONA split. Both the RONA timer and Subsequent Redirection/CFWD No Answer timer are reapplied. • If RONA applies to an ACD call that was a previously-redirected coverage call (for example, the RONA split was the second point in the coverage path), the call is redirected to the next coverage point in the principal's coverage path if the call cannot be requeued to the RONA split. The Subsequent Redirection/CFWD No Answer timer is reset. • If no other point in the coverage path exists or other points are unavailable, the split-covered call that cannot be requeued or the direct-agent-covered call receives call-cannot-be-redirected handling.
Call Detail Recording (CDR)	When an agent is assigned to be recorded on the CDR record as the called number, the RONA redirected-to answering destination is recorded as the final called number. You can administer CDR to record the VDN, the hunt group, or the answering agent as the called number.
Call Forwarding All	If an adjunct direct agent call is made to an agent's extension that has Call Forwarding All assigned and it is redirected by RONA, the call follows the agent's coverage path. A call forwarded using Call Forwarding to a split or logical agent ID with RONA is sent to the principal's coverage path instead of going to the split's coverage path (if the call cannot be requeued) or to the agent's coverage path (for a direct agent call) on RONA redirection.

Table continues…

Interaction	Description
Call Pickup	A member of an agent's pickup group can pick up an ACD call that is being timed for RONA. RONA is cancelled.
Call Vectoring	RONA applies to vector-controlled ACD splits when calls are queued using the **queue-to split**, or **converse-on split**, or **check split** commands. Also, RONA applies to non vector-controlled and vector-controlled ACD splits when calls are routed to the split using a **route-to** or a **messaging split** command. Basic Call Vectoring handles an AAS with all agents logged out as unavailable and skips the relevant step. With an **adjunct routing** or **route-to with coverage** step that routes to a vector-controlled split with all agents logged out, the call is given a busy tone just as when the call cannot queue to a non vector controlled split according to the existing operation. Vector events are generated for a RONA timeout when **converse-on** processes a call or results in a RONA redirection failure, and when a vector step is skipped because all AAS agents are logged out. Do not assign vector-controlled splits coverage, forwarding, or night service, because Call Vectoring provides these functions. These functions do not apply to RONA-redirected calls involving vector-controlled splits.
Calling/Called Number Display	A call to a split or skill that RONA redirects is similar to a direct call to the split or skill. If the call goes to coverage, the destination display looks like it does for a normal covered call. An internal or DCS caller to an ACD hunt group or VDN sees displayed the hunt-group or VDN name and extension. This display remains when the call rings an agent. A direct agent call (with EAS) initiated at a phone displays the agent name and logical ID when the call rings the agent station. If the ACD split call or direct agent call goes to coverage, the name remains, but the extension or logical ID portion changes to *cover*. This also happens when RONA redirects a call.
Delay announcements	Delay announcements assigned to non-vector-controlled splits are applied to requeued RONA calls as usual for redirected calls.
Direct Agent Calling	RONA applies to direct agent calls from splits with RONA assigned. RONA timing applies when a direct agent call (from an adjunct or phone) is delivered to and rings an agent with manual answering. Agents are placed in the AUX work mode or logged out even if the agents are some of the last agents in the split and ACD split calls are queued. Direct agent calls that are queued for an agent remain queued and are not delivered because the agent is unavailable. Don't-answer (DA) coverage continues for the queued calls. If an agent with a coverage path is made unavailable by a RONA time-out on a non-covered direct agent call, the call follows the agent's coverage path. With EAS, the agent's logical extension coverage path for direct agent calls is used. If the agent has no coverage path or if the path is unavailable, the call cannot be redirected and the caller hears previously-provided feedback. If a direct agent call comes from a split that has forwarding or night service, the call is forwarded, precluding RONA timing. If the agent has forwarding or Send-All-Calls, the direct agent call is forwarded (ACD calls only) or goes to coverage, precluding RONA timing.

Table continues…

Interaction	Description
Direct Department Calling	RONA applies to DDC-type hunt-group ACD calls.
Home Agent	RONA applies to Home Agent lines that terminate on the IVR Home Agent system as a means to detect port failures. Home Agent lines use Manual Answer and are not present in AAS. Once RONA notification is made, you can correct the failure and restore service manually on the IVR system.
Inbound Call Management (ICM)	RONA applies to ICM-managed calls that ring an agent in an ACD split with RONA assigned.
Message Center/ Server Service	You can assign RONA to Message Center/Server ACD splits.
Multiple Call Handling (MCH)	If an MCH agent has a call active or on hold and the Redirection on No Answer timer expires for another ringing ACD call, the ringing call is redirected to the split or skill or administered VDN. When the call redirects, the agent is not made unavailable, but is placed in the queue of available agents.
Music-on-Hold access - Music on Transferred trunk call	Trunk callers who are transferred to another destination continue to hear administered music (or silence), not ringback, while the call rings. This applies while the transferred call queues to a split. If the trunk call (an ACD call or direct agent call) is transferred to a split with RONA, timeout applies to the call, but the caller continues to hear the previous feedback instead of ringback.
Night Service	When Night Service is activated, calls (including RONA calls) for the hunt group redirect to the night station extension. If the night service split has RONA assigned, RONA timing is reapplied to the redirected call.
Queue status indications	Calls that RONA requeues are counted in the queued calls total. When a RONA call is queued, the call's call-wait time is reset, so RONA does not affect the oldest call waiting (OCW) time.
Queuing	When redirected to a split, RONA timed-out ACD calls in a non vector-controlled split are queued at the highest priority. These calls are distributed before any other calls, except direct agent calls.
Stations	RONA applies to ACD split or direct agent ACD calls that ring at multifunction or hybrid stations with Manual Answering in an ACD hunt group. RONA applies to Off-Premises Station (OPS) lines in an ACD split.
Voice Response Integration (VRI)	You can assign RONA to converse splits. RONA timing applies to calls that a **converse-on** command queues and delivers. RONA timing is canceled if a call is delivered to an agent in another split to whom the system previously tried to queue a call. RONA interacts with a converse split that is an AAS like any other AAS. If RONA must redirect a call to an agent port in a converse split and the queue is full or all AAS agents are logged out, the call is processed by the next vector step while the caller continues to hear the previous vector feedback.

Interactions with other ringing call timers

Several features time the ringing when an ACD call is delivered to an agent. You can use the RONA timer in conjunction with other timers.

Note:

The timer that expires first applies to the call. RONA is canceled if any of the other timers expires first, except in the case of coverage timers.

When a coverage timer expires, RONA timing is canceled only when the call goes to coverage. If RONA times out first, the other timers continue timing or are stopped and can later be reset. RONA interactions with other timers are summarized in the following table.

Timer	Description	RONA timer Interaction	
		RONA timeout	Restarted after redirection
Split DA	Split Call Coverage Don't Answer (non vector-controlled)	Stopped	If requeued or delivered to another agent
Covering DA	Covering Point Direct Agent (DA) - Subsequent RONA	Stopped	If redirects to covering point
Agent DA	Agent DA Coverage	Stopped	If covers to direct agent with coverage
NATO	DID/CO Trk No Answer Time out	Continues	NA
WAST	Wait Answer Supervision Timer	Continues	If ringing destination or RONA redirection fails

If you want RONA notification and redirection, set the number of rings or equivalent time for a RONA time out to a period shorter than other time out periods. DA timers start when a call is placed in queue and continue when the call rings the station. Since RONA starts only when the call is ringing, the RONA interval is usually set to two or three rings, while the DA interval is set to more than 10 rings.

Since queue time is variable, assign a coverage time out period that is greater than the longest expected queue time plus three or four rings, the time the call rings the agent.

The NATO timer starts when the call seizes the incoming trunk. The timer can be the timing before the call is queued by vector processing. Therefore, set the NATO timer to a period greater than the longest expected time before the call rings the agent, including the time before and after being queued, and add three or four rings.

The WAST timer starts when the call rings the agent. Set the RONA timer to a slightly shorter interval, fewer than 10 rings, than the WAST 50-second interval.

Redirection on OPTIM Failure

You can use the Redirection on Off-PBX Telephone Integration and Mobility (OPTIM) Failure (ROOF) feature only for ACD agents who use SIP hardphones or softphones administered as Off-PBX Stations (OPS).

As with ROIF, ROOF provides redirection of calls back into the call queue or to the specified VDN when calls to ACD stations cannot be connected due to loss of IP connectivity and ROOF applies to agents using auto answer or manual answer. ROOF can also retain the active VDN context while redirecting the calls to the specified VDN.

Before delivering a call to an OPTIM station, ROOF checks for an IP connectivity. If the IP connectivity is not acknowledged, ROOF invokes RONA. Communication Manager places the agent in the AUX work mode and puts the ACD call back in the queue or forwards the call to a RONA-specified VDN. ROOF prevents a call loss during the period when Communication Manager does not detect IP connectivity failure.

You can specify whether to retain the active VDN context while redirecting the calls for a VDN. When you administer the **Retain Active VDN Context** field as `y`, the VDN context from the previous VDN is retained and used while the call is redirected to the specified VDN. If you administer the **Retain Active VDN Context** field as `n` and redirection occurs, the system uses the context of the applicable redirected to VDN.

The VDN context includes the following information:

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement (VOA) Extension
- VDN Skills (1st, 2nd, 3rd)
- VDN Return Destination (VRD)

  Note:

  The VRD is set before being RONA/ROIF/ROOF redirected, and does not change by subsequent routing. The staffed agent receiving the redirect call views 'CR' at the right end of the display indicating a RONA/ROIF/ROOF redirected call.
- VDN Timed After Call Work (ACW) interval
- Best Service Routing (BSR) application
- BSR available strategy
- BSR tie strategy
- Display VDN for route-to Direct Agent Calling (DAC)
- Trunk Adjunct Switch Application Interface (ASAI) messages
- BSR local treatment
- VDN variables

- VDN time zone offset

If you choose to retain the active VDN Context, you can set up a generic VDN that caters to calls redirected from multiple VDNs.

If the call is redirected to a VDN when routed directly to a hunt group rather than through a VDN, the redirect to VDN is the active VDN regardless of the setting of the **Retain Active VDN Context** field.

Note:

ROOF applies to the SIP phones. ROIF applies to H.323 IP phones, for example, the 4622SW IP phone.

ROOF considerations

Set the **Redirection on OPTIM Failure and Unreachable Aux Work Reason Code** field on page 14 of the Feature-Related System Parameters screen to a non zero number that is not currently used. As a result Communication Manager can distinguish between AUX work changes that are caused by miscellaneous changes and those caused by OPTIM failure.

ROOF interactions

ROOF and RONA interactions are similar, but ROOF has more interactions such as:

- ROOF is applied system-wide, where the default for the system is *active*.
- Communication Manager delivers ACD calls from the skill queue and Direct Agent Calling (DAC) interactions similar to RONA. With ROOF, Communication Manager attempts to redirect DAC to the coverage path. If the call does not go to coverage, Communication Manager redirects the call to the VDN that you administer in the **Redirect on IP/OPTIM Failure to VDN** field on the Hunt Group screen. Ensure that the direct agent skill group has a redirect VDN for occurrences of ROOF. Without a redirect VDN, Communication Manager puts the call in the same skill queue at high priority. If queue slots are unavailable, the caller hears a busy signal.

  If all fails, Communication Manager provides a ringback until the caller disconnects the call. This action also applies to priority DACs if no VDN is available for call redirection.
- Calls are redirected through ROOF when repeated attempts to deliver a skill call to the same SIP agent fail with the same SIP error code. This call redirection through ROOF sets the agent's work mode to AUX work.

  For SIP 500 or 380 failures, Communication Manager moves the agent to AUX work after three ROOFs of the same call on the agent. Communication Manager moves the agent to AUX work after one SIP 480 failure.
- For a DAC call, Communication Manager forces the agent into AUX work on the first SIP 380, 500, or 480 failure.
- During ROOF, the agent work mode is AUX work. If network connectivity restores during the TCP retry period, the station lamp indicates that the line is in the AUX work mode.

- The only indication that CMS receives after ROOF occurs is a work state change. Consequently, CMS will receive the resultant flow-out, flow-in, and DFWD-unknown indications for the call.
- A caller hears a ringback during the timer period and when Communication Manager puts calls in the same queue with high priority. Calls that the Communication Manager redirects to a VDN receive feedback from the assigned vector. If Communication Manager cannot put a call back in the queue because queue slots are unavailable, the caller hears a busy signal until the caller disconnects the call. Here, Communication Manager sends a DFWD-unknown message to CMS to stop call tracking.
- When ROOF occurs, Communication Manager does not provide a lamp indication to the supervisor.

Remote Logout of Agent

With Remote Logout of Agent, you can log out idle ACD or EAS agents without being physically present at their station. You can also log out agents from a remote location.

If an agent walks away from the station without logging out, ACD calls sent to the station are not serviced. Consequently, CMS, IQ reports or both reports will be inaccurate since the agent remains logged in even though no ACD calls are being handled at the station. Without Remote Logout of Agent, supervisors or other agents have to walk over to the station, and enter the logout Facility Access Code to change the work mode of the agent or to log out the agent.

Remote Logout of Agent is similar to the Add/Remove Skills feature.

Locally logging out an agent

Procedure

1. Use a local station assigned with the COS and COR to log out an agent locally within the communication server.
2. Enter the FAC established to activate the feature followed by the agent login ID or physical station extension.

   Important:

   - You must be in the same Tenant Partition as the agent.
   - The physical extension is used only in non-EAS systems.

How to administer Remote Logout of Agent using a VDN

Many call centers are geographically dispersed, but the reporting and tracking of agent activity takes place from one main location. Agents can log in to the system remotely and take calls using the Home Agent capability.

With the Remote Logout of Agent feature, a vector is administered with a **route-to number** step that contains the FAC for remote logout. The FAC can be followed by the agent loginID or the supervisor can dial the loginID after the VDN with an appropriately programmed vector. Note that Service Observing and remote logout are the only features that can use a VDN vector in this manner.

If the user is remote and calling into the communication server through a trunk, the user can reach the activation VDN to logout the agent. In this case, the entire FAC-with-EXT is required in the vector. Or, the vector can prompt the user to enter the extension and then route-to digits. An activation vector can also prompt the user for a password for additional security. Note that a remote trunk user does not always hear confirmation tone, varying with trunk type and trunk administration, and the user continues to hear feedback until the user hangs up.

To set up this capability, the communication server administrator must create an activation VDN and set the incoming destination of a CO trunk or DID dialed number to be the activation VDN. The VDN to which the call terminates must be assigned a COR and a TN that include the appropriate settings for use of the Remote Logout of Agent feature. COS assigned with console permissions is not required. In this example, the activation VDN includes the following vector.

```
01 wait time 0 seconds hearing ringing
02 collect 5 digits announcement 3501 (“enter password”)
03 goto step 5 if digits = 39744
04 disconnect after announcement 3502 (“bye”)
05 collect 1 digits announcement 3503 (“enter 1 to logout agent 89923, 2 to logout
agent  89924...”)
Note: The names of the agents can also be requested in the appropriate switch-setup.
06 route-to number *6389923 with cov n if digit = 1
07 route-to number *6389924 with cov n if digit = 2)
08 goto step 5 if unconditionally
```

Note:

In this example, *63 is the FAC assigned for Remote Logout of Agent. This example is one of many ways in which the vector can be written to activate the VDN.

Remotely logging out an agent using the assigned VDN

About this task

To log out an agent from a remote location using the assigned VDN:

Procedure

1. Dial in to the communication server from an outside line that reaches the activation VDN.
2. Enter the password as programmed in the vector.

   See Step 2 in the vector example described in How to administer Remote Logout of Agent using a VDN.
3. Enter the physical or logical agent extension or the digit corresponding to the desired agent you want logged out.
4. Enter `1`.

   The login ID associated with that prompt is the login ID or name of agent A.

Note:

Logout fails if a remote logout is attempted for an agent who is on an ACD call, has an ACD call on hold, or who is not logged in.

Remote Logout of Agent interactions

Interaction	Description
Auto-Available Split/Skill (AAS)	If an agent login ID is assigned to AAS, then the Remote Logout of Agent feature cannot be used to log the agent out. RONA can be used to automatically logout a port that is not answering calls.
AUDIX	If an agent is a member of an AUDIX hunt group and has no other splits/skills assigned to the agent login ID, then the Remote Logout of Agent feature will not successfully log out the agent, even though the user attempting the logout hears a confirmation tone.
Non ACD hunt groups	If an agent is a member of ACD splits/skills and is using a physical extension that is a member of a non ACD hunt group, then use of the Remote Logout of Agent feature will log the agent out of the splits/skills but allow the agent to continue receiving non ACD calls.
Non EAS agent operation	A non EAS agent is logged out of all splits even while active on an ACD call. This call is not dropped, but all Call Center reporting of the call is stopped.
Timed ACW	If an agent answers an ACD call for a hunt group with Timed After Call Work administered and then hangs up the call, the Remote Logout of Agent feature can be used to log out the agent during the ACW time.
Service Observing	An agent can be logged out using the Remote Logout of Agent feature while being service observed.

Reporting Adjuncts

Reporting adjuncts are applications for organizations that use Communication Manager to process large volumes of calls using ACD and the Avaya Proactive Contact (PC) outbound dialer or Proactive Outreach Manager (POM). Reporting adjuncts include Avaya Call Management System or Avaya IQ.

Reporting adjuncts support solutions for:

- Routing and agent selection tracking
- Multisite contact centers
- Remote agents
- Reporting
- Interfaces to other systems

- Workforce management
- Desktop applications
- System recovery
- Quality monitoring

Measured trunks and unmeasured facilities

CMS requires data allocation for unmeasured facilities trunk data. Unmeasured facilities are not actual trunks but are tracking records.

Unmeasured facilities on CMS are required for:

- Internal calls to a measured split or an agent
- Internal calls to VDNs
- Calls made by agents to internal destinations or on an unmeasured facility group
- Transfers and conferences until the transfer or conference is complete.

Reporting adjuncts considerations

- Display of IP or SIP trunk locations.
- Reporting adjunct representation of IP or SIP trunk member port-IDs.
- Display of physical (non-IP) member port-IDs.

IP trunk groups and SIP trunk groups

IP or SIP trunk members are defined using a virtual trunk equipment location number that is sent from the communication server to the reporting adjunct. Two types of messages can send information about single trunk group members and about status changes (from measured to unmeasured, or from unmeasured to measured using the reporting adjunct) of trunk group members. For H.323 trunks the equipment location can be just a port number.

When IP or SIP trunk groups are administered on the communication server, the gateway location of the associated C-LAN board is determined for each trunk group member. When message information is sent to the reporting adjunct, the “T” virtual port number is mapped into a 9-digit number with leading zeros. For example, an IP trunk member with a port-id of T00001 is displayed on CMS as 000000001, and an IP trunk member with port-id of T00400 will be displayed on the reporting adjunct as 000000400.

Display of IP or SIP trunk locations

IP or SIP trunk members are defined using a virtual trunk equipment location number that is sent from the communication server to the reporting adjunct for use in trunk reports. The virtual port numbers are defined in Communication Manager during administration using the letter “T” followed by 5 digits with leading 0’s. The first virtual port is defined by T00001 and ranges to the maximum port location of T24000.

When trunk information is sent to the reporting adjunct, the "T" virtual port number is mapped into a 9-digit number with leading zeros using the bbbXssccc format (the same as the G650 ports but

with the carrier always set to 0 to indicate that the equipment location is a virtual port and an extra 0 [“b”] in the “cabinet” field). To display IP trunk member port-IDs to the reporting adjunct as virtual trunk equipment location numbers, a 9-digit number starting with leading zeros is used.

For example, an IP trunk member with a port-id of T00001 is sent to CMS/IQ as “000 0 00 001” (cabinet, carrier, slot, circuit), and is displayed on CMS/IQ as 000000001. An IP/SIP trunk member with port-id of T00400 is sent to CMS/IQ as “000 0 00 400” and displayed on the CMS reporting adjunct as “000000400.” Due to the conversion process used by the software, the numbering above Txxx500 for each 1,000 port IDs are mapped to numbering 000 to 499 with an increment to the next 1000 in the series (a modulo 500 function). For example, T00500 is sent as “000 0 01 000” and the range of T00501 to T00999 is displayed on CMS/IQ as 000001001 to 000001499.

In SPI language 24, trunk locations are sent to reporting in two different ways. One way is described in the following table. The other way is sending a virtual trunk string that populates the total of 16-bit cab, car, and slot fields with a binary number that is the numerical value of the Txxxxx virtual H.323 or SIP trunk string identified by Communication Manager. The numerical value ranges from 00001 to 24000.

Use the following table to correlate IP trunk member port IDs on the communication server and the reporting adjunct.

Communication server representation	Reporting adjunct representation
T00001 through T00499	000 0 00 001 through 000 0 00 499
T00500 through T00999	000 0 01 000 through 000 0 01 499
T01000 through T01499	000 0 02 000 through 000 0 02 499
T01500 through T01999	000 0 03 000 through 000 0 03 499
T02000 through T02499	000 0 04 000 through 000 0 04 499
T02500 through T02999	000 0 05 000 through 000 0 20 499
T03000 through T03499	000 0 06 000 through 000 0 06 499
T03500 through T03999	000 0 07 000 through 000 0 07 499
T04000 through T04499	000 0 08 000 through 000 0 08 499
T04500 through T04999	000 0 09 000 through 000 0 09 499
T05000 through T05499	000 0 10 000 through 000 0 10 499
T05500 through T05999	000 0 11 000 through 000 0 11 499
T06000 through T06499	000 0 12 000 through 000 0 12 499
T06500 through T06999	000 0 13 000 through 000 0 13 499
T07000 through T07499	000 0 14 000 through 000 0 14 499
T07500 through T07999	000 0 15 000 through 000 0 15 499
T08000 through T08499	000 0 16 000 through 000 0 16 499
T08500 through T08999	000 0 17 000 through 000 0 17 499
T09000 through T09499	000 0 18 000 through 000 0 18 499

Table continues…

Communication server representation	Reporting adjunct representation
T09500 through T09999	000 0 19 000 through 000 0 19 499
T10000 through T10499	000 0 20 000 through 000 0 20 499
T10500 through T10999	000 0 21 000 through 000 0 21 499
T11000 through T11499	000 0 22 000 through 000 0 22 499
T11500 through T11999	000 0 23 000 through 000 0 23 499
T12000 through T12499	001 0 00 000 through 001 0 00 499
T12500 through T12999	001 0 01 000 through 001 0 01 499
T13000 through T13499	001 0 02 000 through 001 0 02 499
T13500 through T13999	001 0 03 000 through 001 0 03 499
T14000 through T14499	001 0 04 000 through 001 0 04 499
T14500 through T14999	001 0 05 000 through 001 0 05 499
T15000 through T15499	001 0 06 000 through 001 0 06 499
T15500 through T15999	001 0 07 000 through 001 0 07 499
T16000 through T16499	001 0 08 000 through 001 0 08 499
T16500 through T16999	001 0 09 000 through 001 0 09 499
T17000 through T17499	001 0 10 000 through 001 0 10 499
T17500 through T17999	001 0 11 000 through 001 0 11 499
T18000 through T18499	001 0 12 000 through 001 0 02 499
T18500 through T18999	001 0 13 000 through 001 0 13 499
T19000 through T19499	001 0 14 000 through 001 0 14 499
T19500 through T19999	001 0 15 000 through 001 0 15 499
T20000 through T20499	001 0 16 000 through 001 0 16 499
T20500 through T20999	001 0 17 000 through 001 0 17 499
T21000 through T21499	001 0 18 000 through 001 0 18 499
T21500 through T21999	001 0 19 000 through 001 0 19 499
T22000 through T22499	001 0 20 000 through 001 0 20 499
T22500 through T22999	001 0 21 000 through 001 0 21 499
T23000 through T23499	001 0 22 000 through 001 0 22 499
T23500 through T23999	001 0 23 000 through 001 0 23 499
T24000	002 0 00 000

Display of physical port locations

Physical port network trunk locations that terminate on the G650 gateway are represented by a cabinet, carrier, slot and circuit format. For example a location can be 1, A, 1, 22. This equipment location will show as 01A01022 on CMS.

Port IDs for physical circuit-switched trunks that terminate on an H.248 Media Gateway such as the G700 or G450, are displayed in a different format than that used for normal port network

equipment locations such as the G650. This difference is also reflected in the way those port IDs are listed in the reporting adjunct.

The following table compares the standard trunk equipment location format for port network port IDs to the H.248 Media Gateways format.

Regular port network trunk equipment location format like the G650	Media Gateway-terminated trunk equipment location format
bbXssccc where: bb = cabinet (1-64) X = carrier (A-E) SS = slot (1-25) CCC = circuit (1-256)	gggVscc where: ggg = gateway (replaces cabinet) (1-250) V = indicates H.248 media gateway (replaces carrier) s = slot (1-4) cc = circuit (1-32)

In reports, media gateway-terminated trunk equipment locations are displayed in a slightly different format than that used for port network gateways.

Depending on the reporting adjunct version, the trunk equipment location is shown in a fixed 8-character or 9-character format according to the following rules:

Gateway number: The gateway number will show as either 2-digits (the leading digit shown in the communication server display is dropped by CMS) or 3 digits. This variation in the gateway number format is the result of the following factors:

- The numerical designation assigned to a gateway can be any number from 1 to 250, but earlier Avaya communication server releases only supported up to 99.
- R3V11 CMS and CMS Supervisor do not provide a third-digit space for the gateway numbers.

Therefore, for R3V11 installations, when the gateway numbers that are greater than 99, the leading (hundred) digit of the gateway number is shown as the leading (tens) digit in the slot number and the following rules apply to display of slot numbers for H.248 Media Gateway-terminated trunk equipment on R3V11 versions of CMS and CMS Supervisor:

- For gateways 1-99, slot numbers range from 01 to 04.
- For gateways 100-199, slot numbers range from 11 to 14.
- For gateways 200-250, slot numbers range from 21 to 24.

Carrier number: The carrier number shows as the letter *V* followed by two digits (01-04) for the slot number.

Circuit number: The circuit number shows as 3 digits (001-032).

The following table shows how H.248 Media Gateway-terminated trunk equipment location formats are listed on Avaya communication servers and reporting adjuncts.

Avaya communication server	On CMS (pre-R3V11ag) and CMS Supervisor (pre-R3V11FJ.04)	On CMS (R3V11ag or later) and CMS Supervisor (R3V11FJ.04 or later)	On CMS (R3V12 or later), CMS Supervisor (R3V12 or later), and Avaya IQ
Example 1: gateway=12, slot number=2, circuit number=16			
012V216	12702016	12V02016	012V02016
Example 2: gateway=130, slot number=2, circuit number=16			
130V216	30702016	30V12016	130V02016
Example 3: gateway=240, slot number=2, circuit number=16			
240V216	40722016	40V22016	240V02016

Service Level Maximizer

Use Service Level Maximizer (SLM) to ensure that a defined service level of *X* % of calls are answered in *Y* seconds. When SLM is set to `y`, Communication Manager verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met.

MAO

When using SLM, you can also use an optional feature called Maximum Agent Occupancy (MAO) to set thresholds on the amount of time an agent spends on a call. MAO is used to prevent agent burnout.

SLM requirements

SLM works on all platforms and operating systems that are supported by Communication Managerand requires the Call Center Elite package.

You cannot use SLM and Advocate on the same system, because SLM and Avaya Business Advocate cannot be simultaneously enabled on the System-Parameters Customer-Options screen. Avaya Business Advocate provides a more flexible and functional form of achieving service level targets.

SLM operations

SLM agent selection

Agent selection methods are used when a call is queued to a skill and more than two agents are available to take the call. This is known as an agent surplus condition.

The algorithm for agent selection:

- Selects the agent with only one skill first.

- For agents with more than one skill, the agent skills that are *not* needed for the incoming call are reviewed and the agent with skills that are less likely to be needed for future calls is selected. This is another way of saying that the algorithm selects the agent with the lowest opportunity cost.
- In a tie, the most idle agent is selected.

The agent selection algorithm excludes the agent skill level and occupancy.

SLM call selection

Call selection applies when an agent becomes available and there are calls waiting in queue in more than two of the agent skills. This is known as a call surplus condition.

Call selection compares all the skills assigned to an agent that have a call in queue and picks the one with the lowest ART value. If any ART values are negative, the call in the skill with the highest negative ART value is chosen.

Note:

The SLM call selection method is applied to agents who have a minimum of one skill administered as `slm`.

The call selection algorithm excludes the agent skill level and call priorities.

SLM target service levels and agent opportunity costs

The SLM agent selection method is based on user-defined target service levels for SLM-administered skills and the concept of agent opportunity costs.

Target service level: You define specific target service level goals for each SLM skill based on the following format:

`SLM target service level = x percent calls answered in y seconds`

For purposes of SLM reporting, estimates of service level compliance for a skill are expressed as the Actual service level Relative to the Target service level (ART). At any point in time, an SLM skill can be below, equal to, or above the specified target service level. For example, if a skill has a target service level of 80 percent of all calls answered within 20 seconds and the current service level is 75 percent of all calls to the skill answered within 20 seconds, then the current ART value is -5 percent. Alternately, if the current service level indicates that 90 percent of all calls are being answered within 20 seconds, then the current ART value is +10 percent.

For information on how to administer service target levels for a skill, see “Service Level Maximizer” in the *Administering Avaya Aura® Call Center Elite* document.

Opportunity costs: SLM compares actual call service levels to target service levels for each SLM skill, so that when an incoming call arrives at a skill, service level data can be used as the basis to develop agent *opportunity cost* estimates. The opportunity cost for an agent at a given point in time is represented as a weighted estimate that checks the status of the agents skills relative to the target service levels of each skill.

The process that SLM uses to derive agent opportunity cost estimates can be summarized as follows:

- An incoming call arrives for an SLM skill and agents that are both assigned to that skill and currently available are identified.
- All skills to which the available agents are assigned are also identified. For each of the assigned skills (excluding the skill associated with the incoming call), a current service level estimate is calculated and compared to the target service level.

  Note:

  The opportunity cost for single skill agents is always equal to zero, since the agents can always be selected for an incoming call in the assigned skill with no impact on the service level status of any other skills.
- Based on the current overall service level for the skills of each available agent, SLM derives a weighted estimate that identifies which of the available agents is currently the least needed for their other assigned skills, where the *need* of a skill is (approximately) defined as the difference between the current service level and the target service level. This agent has the lowest overall opportunity cost.

Because of the way that SLM estimates agent opportunity costs in the agent selection process, available agents whose skills are currently closest to matching their specified target service levels are selected first, while agents whose skills are furthest from matching their specified target service level are selected last. This strategy maximizes the possibility that an agent will be available when a call arrives at a skill whose target service level is at risk.

For example, agents A and B, are both assigned to skill 4 as well as two other skills. When an incoming call arrives at skill 4 and both agents are available, SLM compares the current service level to the target service level for each of the skills to which the agents are assigned. The agent who currently has the lowest opportunity cost is identified and selected to receive the incoming call in skill 4.

The following table shows how the agent with the lowest opportunity costs is selected in two different call service level scenarios:

Note:

To simplify this example, the service level states for each skill are represented as ART values. The actual agent selection algorithms used by SLM are complex and do not rely directly on ART data.

Skill assignments	SLM skill 1	SLM skill 2	SLM skill 3	Skill 4 (incoming call)
Agent A	X	X		X
Agent B		X	X	X

Table continues…

Skill assignments	SLM skill 1	SLM skill 2	SLM skill 3	Skill 4 (incoming call)
	For Skill 1, if...	For Skill 2, if...	For Skill 3, if...	Then... Agent with lowest opportunity cost for incoming call is:
Scenario 1	ART (ART = Actual service level relative to Target service level, where the service level is defined as x% calls answered in y seconds.) = -5%	ART = +2%	ART = +2%	Agent B
Scenario 2	ART = -1%	ART = +5%	ART= -6%	Agent A

In scenario 1 in this table, Agent B has the lowest opportunity cost compared to Agent A because the skills other than skill 4 assigned to Agent B (skills 2 and 3) are both above target service level. At the same time, of Agent A's skills (skill 1 and skill 2), skill 1 is below target. Agent A is selected for skill 1. Therefore, of Agents A and B, it is better to select Agent B for the incoming call to handle skill 4.

In scenario 2, of Agent B's other skills (2 and 3) skill 2 is above target level but skill 3 is below target by 6%. At the same time, of Agent A's other skills (1 and 2), skill 1 is only below target by 1%. Therefore, in this scenario, Agent A has the lowest opportunity cost compared to Agent B, since Agent B has a skill in worse shape than Agent A.

SLM benefits

Because SLM is able to differentiate skills in terms of their current call service demands, it provides the following advantages over other agent selection methods:

- Since agent resource needs for each skill are assessed in real-time, you can use SLM to allocate agent resources to those skills that have the greatest call service demand in a dynamic manner, thereby reducing overall call response times.
- Potential problems associated with staffing exceptions, or fluctuating, intra-day call service demands are also reduced.
- SLM is especially useful for call center operations that are bound by contract or other legal obligation to meet specific service level requirements.

Auto reserve agents

Auto reserve is a feature that you can use to ensure that the service level is met in critical skills. When a critical skill does not meet the administered service level, auto reserve puts agents in standby for their other skills ensuring that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of the assigned skills are checked to see if any auto reserve skills are not meeting the target service level. If so, the agent is made available only in the those skills.

For information on administration of the auto reserve option, see “Service Level Maximizer” in the *Administering Avaya Aura® Call Center Elite* document.

Auto reserve agents

Auto Reserve Agents is a feature that you can use to ensure that critical skills meet the target service levels. When a critical skill does not meet the target service level, Communication Manager uses this feature to put agents in standby for the other skills. Communication Manager ensures that the next call for the critical skill has an available agent.

When an agent becomes available, Communication Manager checks all the assigned skills to see if an auto reserve skill does not meet the target service level. If so, Communication Manager makes the agent available for calls to the critical skill.

How auto reserve works

With SLM, you can specify auto reserve agents for a skill to ensure that the desired service level is met in critical skills. When an agent becomes available, the agent can be reserved for SLM skills that have a weighted service level below their assigned targets. When the agent is reserved in more than one of the assigned skills, the agent is made available to receive calls only from the assigned skills.

With SLM, an agent becomes reserved for an SLM skill, which has a Group Type “slm”, when the agent becomes available. At that time, the SLM software checks all the skills assigned to the agent to determine if any have a weighted service level below the target service level. Before the agent is automatically reserved for more than one of the skills:

- The skills must have a maximum auto reserve setting greater than zero, as set on the Hunt Group screen for the skill.
- The limit of reserved agents has not been exceeded for the skill

In other words, the agent is only available in that skill. The agent is made unavailable in the above-target skills, reserving each agent for the neediest skills.

Considerations for allocating auto reserve agents

Since auto reserve agents are unavailable in other skills, use auto reserve agents only in skills for which achievement of service level targets is critical. The addition of even a single auto reserve agent can have a significant impact on the service level. Therefore, initially set the number of auto reserve agents on the Hunt Group screen to 0 or 1, observe the impact on the service level, and gradually increase the number of auto reserve agents by increments of one until you achieve the service level target.

Rules for auto reserve agents

For agents assigned to any skills that use auto reserve, the following rules apply when an agent becomes available:

- If any of the auto reserve-enabled skills to which an agent is assigned are currently below their specified target service level, the agent is available only in those skills.
- The designation of auto reserve agents for a skill is continuously assessed as agents become available. If the maximum number of auto reserve agents has already been reached,

a single-skill agent who becomes available replaces the multi-skilled agent who has the highest opportunity cost.

- If more than one of the agent auto reserve-enabled skills are currently below the specified target service level, a *multi-skill* agent is put into the auto reserve state if one of the following conditions are met:
  - The maximum number of auto reserve agents for the skill is not yet filled.
  - The maximum number of auto reserve agents for the skill *is* filled, but the opportunity cost for an idle, multi-skilled agent is lower than the opportunity cost of a multi-skilled agent who is currently in the auto reserve state. In this case, the agent with the highest opportunity cost is released from the auto reserve state.

Release of auto reserve agents

To prevent an agent from being auto-reserved for a long time, the auto reserve agent is replaced as soon as another agent becomes available. However, the following conditions must apply:

- The maximum number of auto reserve agents for the SLM skill are in use.
- A single-skilled agent for that skill has become available or a multi-skilled agent with a higher occupancy than the auto reserve agent has become available.

In addition to being replaced, auto reserve agents can be released when the agents change the work mode or when a skill moves from *below target* to *above target*.

Agent selection rules in mixed skill environments

SLM skills can be co-resident on the system with skills that use other agent selection methods such as LOA. However, situations can arise in which a skill is not administered as an SLM skill, but includes agents that are also assigned to more than one SLM skill. In such a mixed skill environment, the following rules apply:

- If a non-SLM administered skill does not include any agents who are also assigned to SLM skills, agent selection is based on the agent selection method administered for that skill.
- If a non-SLM administered skill includes more than one agent who is also assigned to SLM skills, a current service level value of 100 percent is applied to the non-SLM skill for purposes of SLM service level and agent opportunity cost calculations.

Important:

In a mixed skill environment, the service level for non-SLM hunt groups must be administered to reflect the importance of the hunt group. For example, if you can keep inbound callers in queue for a longer amount of time, you can set the service level to be 75 percent (of calls answered) in 180 seconds. In other cases, when an extended wait time is not expected, but target service level compliance is not critical, you can set the service level to be 45 percent in 15 seconds.

SLM algorithms

You can choose an alternative algorithm for selecting agents and delivering calls to maximize service level targets. The original Weighted Service Level (WSL) algorithm used for maintaining the service level targets is changed to an Actual Service Level (ASL) algorithm that works better

with low staff or low traffic level conditions. The ASL algorithm also handles the higher staff and traffic conditions. ASL is determined as a percentage on a hunt group basis using the number of *accepted* calls in the current interval divided by the total calls in the current interval. A call is counted as *accepted* if it is answered within the target service level time period. You can still select the WSL algorithm on a system basis whenever required. The WSL algorithm is based on a weighting calculation that uses the difference between the target time and the estimated wait time.

Criteria for choosing algorithms

The ASL algorithm is an improved algorithm for maintaining service level targets. Use ASL for most situations, in addition to low staff levels or low traffic level conditions unless actual experience indicates that WSL provides better performance for your installation.

Administering the ASL algorithm

Before you begin

Ensure that the field option in the **Service Level Maximizer** field on the System Parameters Customer-Options screen is **y**.

Procedure

1. Administer the **Service Level Maximizer Algorithm** field on the Feature-Related System Parameters screen as **actual**.
2. Administer the **SLM Count Abandoned Calls** field on the Hunt Group screen to determine whether to include abandoned calls in the ASL algorithm calculations for SLM.
3. Administer the time interval in the **Service Level Interval** field on the Hunt Group screen.

SLM reporting

This section contains an overview of the CMS Supervisor report features with which you can evaluate various aspects of SLM performance.

For detailed information on:

- Database items related to SLM or MAO, see *Avaya Call Management System Database Items and Calculations*.
- ART reports, see *Avaya CMS Supervisor Reports*.

Evaluating target service level compliance

CMS includes database items that you can use in CMS Supervisor to evaluate how well your target service levels are met by SLM operations.

Note:

The service level used by the Communication Manager to route calls is based on a prediction of a call being answered in the target service level. The service level calculated by CMS is the actual service level being achieved.

ART reports

Supervisor provides several types of Actual Relative to Target (ART) reports that compares actual service levels to target service levels and expresses the difference on a percent basis in a graphical format.

Note:

If your service level targets are based on contractual agreements, verify that your assessment of service level performance is based on a time frame (days, weeks, months) that is appropriate for the terms of your contract.

A percent value that exceeds zero means that actual service levels exceed the target, while percent values less than zero mean that the service level is not being achieved. When actual and target service levels correspond closely, the percent difference between the two data sets that are displayed in ART reports will tend to be close to zero, which is an indication that staffing levels are consistent with call service goals.

Service level calculations

Service level calculations can also be used to evaluate service level compliance. In R12 new database items have been added to track the number of calls answered (TARGETACDCALLS), abandoned (TARGETABNS) and outflowed (TARGETOUTFLOWS) within the service level administered on the Communication Manager.

CMS uses the target service level that is administered on the Communication Manager to generate these items. The advantage to using these items is that if the target service level is changed, CMS receives the new service level value and automatically adjusts how these items are computed. These items can be included in custom reports.

Note:

The existing CMS service level calculation can be used only if the acceptable service level on CMS Split/Skill Call Profile matches the Target Service Level administered on the Communication Manager. If the target service level is modified on the Communication Manager, the CMS service level must be manually modified to match that value.

CMS report for skills in the auto reserve state

CMS Supervisor includes a *% Skills Available* column in the historical Agent Summary report. The value in the column is 100% when an agent spends no time in the auto reserve state. Percentages less than 100 indicate that the agents have spent time in the auto reserve state.

Reasons for low agent occupancy

SLM can impact the agent occupancy in the following ways:

- Reserving an agent as an auto reserve agent can impact the occupancy of the agent because the agent is made unavailable in some of the assigned skills. This might cause the agent to miss ACD calls to those skills. For example, an agent might miss a call that is queued to skill 1 and be made available for ACD calls that arrive at skill 2.

  The *% Skills Available* column in the historical Agent Summary report indicates if an agent is an auto reserve agent. Supervisors can view the column to determine the percentage of staffed skills that the agent was available in. The percentage is 100 for an agent who is not

an auto reserve agent. Otherwise, the agent is an auto reserve agent for some the assigned skills.

- Agent selection can impact the occupancy of an agent. In an agent surplus condition, Communication Manager delivers ACD calls to a single-skilled agent or an agent whose other assigned skills exceed the required service level targets. For example, skill 1 is assigned to agent A and agent B, and the other assigned skills for agent A are at or below target while the other assigned skills of agent B are above target. If an ACD call queues to skill 1, Communication Manager sends the ACD call to agent B. The delivery of ACD calls to agent B continues till the other assigned skills of agent A are above the target or agent A is the only available agent in skill 1.

  To determine which skills do not meet the service level targets, run the Actual Relative to Target Service Level report.

Note:

- Agents with the fewest skills typically have the highest occupancy.
- The *% Skills Available* column in the historical Agent Summary report does not display a lower percentage if Communication Manager is not delivering ACD calls to an agent whose other assigned skills are at or below target.

SLM interactions

Interaction	Description
Avaya Business Advocate	Do not enable both SLM and Avaya Business Advocate on the System-Parameters Customer-Options screen.
BCMS Reporting Desktop VuStats	If BCMS Reporting Desktop VuStats is used to display acceptable service level report data, the displayed value is identical to the seconds value that is set in the **Target Service Level (% in sec)** field on the Hunt Group screen. For more information on administering SLM skills, see “SLM administration” in the *Administering Avaya Aura® Call Center Elite* document.
Best Service Routing	With BSR, the best resource choice among the local skills and best skills of the remote sites is based on the lowest adjusted EWT or assigned available agent strategy rule. This rule does not check service level targets that are assigned to individual skills. However, when an SLM skill is selected as the best resource, the available agent selection is based on the specified service level target for the skill. Therefore, service level objectives are maintained within the local or remote skills, but not across sites.
Direct Agent Calls	For agents assigned to SLM skills and eligible to receive direct agent calls, direct agent calls have priority over ACD calls.
Least Occupied Agent	SLM does not use LOA as an agent selection method.

Table continues…

Interaction	Description
Location Preference Distribution	You can assign reserve agents using SLM. In most cases, the selection of an agent or a call based on Location Preference Distribution takes precedence over SLM. However, SLM takes precedence when a reserve agent is needed because the service level is below the threshold. **Note:** If more than one reserve agent is eligible for the call, Location Preference Distribution is used to choose the agent.
Non-SLM Skills	Agents that have a minimum of one assigned SLM skill have the administered Call Handling Preference (CHP) ignored and are treated as if the CPH is set to `slm`. The non-SLM skills are treated as if the skills are always at service level when it comes to agent and call selection.
Greatest Need	Greatest Need is not used when SLM is enabled, since call selection is driven by the target call service levels that are administered for each SLM skill.
RONA	Redirected calls are part of the service level calculations of any SLM skill to which redirected calls are sent.

Service Observing

With Service Observing, supervisors can monitor calls to extensions, attendants, Expert Agent Selection (EAS) agents, or Vector Directory Numbers (VDNs).

Supervisors or another observer can monitor calls in one of the following modes:

- Listen Only
- Listen/Talk
- No Talk
- Next Call Listen Only
- Coach

Note:

For coaching, the so-coach button must be administered.

In case of H.323 or DCP phones, an observer presses **serv-obsrv** to begin an observing session. Communication Manager puts the observer in the listen-only mode. While observing a call, the observer can press **serv-obsrv** to toggle between listen-only and listen/talk. The **serv-obsrv** button lamp on the observer deskphone indicates the observing mode. To stop the observing session, the observer can select another call appearance, put the receiver on-hook, or press **release** while on a call.

In case of SIP phones, an observer presses the **Service Observing** or **SO** feature key to begin an observing session. To stop the observing session, the observer can press the **Service Observing** or **SO** feature key again or can select another call appearance, put the receiver on-hook, or

press **release** while on a call. Administrators use the **Listen-Only?** and **Coach?** subfields of the **Service Observing** or **SO** feature key to control whether the observer can change to listen-talk or to coach modes. While observing a call, based on the administration done on the phone, the observer can use the **Listen**, **Talk**, and **Coach** softkeys of the **Service Observing** or **SO** feature key to toggle between the listen, talk, and coach modes. The **Service Observing** or **SO** feature key is administered as **sip-sobsrv**.

Note:

Only the 9608SIPCC, 9611SIPCC, 9621SIPCC, and the 9641SIPCC station types support the **Service Observing** or **SO** feature key.

You can add a warning tone to inform agents and callers when an observer is monitoring the call. Agents and callers hear a 440 Hertz (Hz) tone for 2 or 3 seconds before an observer connects to the call. Agents and callers then hear a half-second burst of the tone every 12 seconds during the observing session.

Important:

Service Observing is subject to federal, state, or local laws and rules, or requires the consent of one or both the calling parties. Familiarize yourself and comply with all applicable laws, rules and regulations before using the feature.

No Talk FAC for Service Observing

Note:

SIP phones do not support service observing using a FAC. SIP phones use the **Service Observing** or **SO** feature key to activate the No Talk feature.

An observer or a **route-to number** vector command can activate Service Observing in a No Talk mode. Communication Manager does not reserve a time slot for the talk path as the observer uses the No Talk mode when the observer wants to only listen to calls. This observing mode reduces unnecessary time slot usage. You can use the time slots for other purpose such as simultaneous call recording sessions.

If you use the No Talk mode, Communication Manager does not allow the observer to select the Listen/Talk mode because Communication Manager reserves no time slot for the talk path.

For call recording applications, use the No Talk mode for more than one simultaneous recording session.

To use the No Talk mode, you must assign a dial code in the **Service Observing No Talk Access Code** field on the Feature Access Code (FAC) screen. Observers must type the No Talk Feature Access Code (FAC) to activate an observing session for call recording.

Observing agent login IDs

With Expert Agent Selection (EAS), an observer can monitor all incoming Automatic Call Distribution (ACD), personal, and direct agent calls to agent login IDs.

Without Multiple Observers, only one observer can monitor an extension at a time. The observer cannot monitor an agent login ID at a physical phone that is already being observed. Similarly, an observer cannot monitor a deskphone extension that is being observed as the extension of an agent login ID.

Observing remotely by FAC

Observers can use the following Feature Access Codes (FACs) to monitor calls:

- Service Observing by Location Listen Only Access Code
- Service Observing by Location Listen/Talk Access Code
- Service Observing Listen Only Access Code
- Service Observing Listen/Talk Access Code
- Service Observing Next Call Listen Only Access Code
- Service Observing No Talk Access Code

Note:

Use the No Talk FAC if you do not want to reserve a time slot for the talk path.

Observers can use FACs for different observing modes, such as Listen Only and Listen/Talk, but observers cannot toggle between observing modes. Observers can monitor physical extensions, agent login IDs, and Vector Directory Numbers (VDNs).

Observers can begin remote observing through Remote Access or Call Vectoring.

- *Remote Access*: To gain access to Communication Manager, an observer uses a trunk group dedicated for Remote Access or a Direct Inward Dialing (DID) to the Remote Access extension. Remote observing works with all types of DID trunks, including ISDN-PRI and tie trunks, and DCS over analog, T1, or PRI.
- *Call Vectoring*: To gain access to Communication Manager, an observer types a VDN extension or Central Office (CO) trunk that has a VDN extension as the incoming destination.

  You can use **route-to** commands to create a Service Observing vector that allows a VDN call to gain access to an extension or a dial tone. At the dial tone, the observer can type the extension number. You can combine Call Prompting and Call Vectoring to provide security and to restrict observation.

  The following vector is an example of a Service Observing vector:

```
1.wait-time 0 seconds hearing ringing
2.collect 5 digits announcement 2300 ("Enter your security code")
3.goto step 5 if digits = 12345
4.disconnect after announcement 2000 ("You are not authorized to observe")
5.collect 1 digits announcement 2310  ("Enter 1 to observe Sales, 2 to observe
Billing")
6.route-to number 113001 with cov n if digit = 1 (11=listen-only observe,
3001=Sales VDN)
7.route-to number 113002 with cov n if digit = 2 (11=listen-only observe,
3002=Billing VDN)
8.goto step 5 if unconditionally
```

Note:

SIP phones do not support service observing using a FAC. SIP phones use **Service Observing** or **SO** feature key to activate all types of service observing modes.

Observing VDNs

To monitor a Vector Directory Number (VDN), an observer types a VDN extension and bridges onto calls, one call at a time, that have started vector processing for that VDN. The observer hears all the tones, call prompting, caller dialing, announcements, music, and speech that the agent and caller hear. The observer can press **voa-repeat** to listen to the VDN of Origin Announcement (VOA) if the observer has a Class of Restriction (COR) number that allows VOA.

Service Observing of VDNs is enhanced to start observing a call to the VDN when Communication Manager delivers the call to the agent or station. When this VDN option is active, VDN Service Observing activation still associates the observer with calls to the VDN, but the observer does not hear a call during vector processing. After initial activation, the first call to be observed passes through vector processing before the system begins an observing session. When the observing connection is completed for the first call, that is, the call is released, the observer is bridged onto the subsequent call to the VDN when the call is answered by an observable agent or station. The observer saves time because after VDN observing is active, the observer does not have to wait for each subsequent call to go through vector processing and for an agent to answer the call.

To begin VDN observing when Communication Manager delivers calls to an agent or a station, select **y** in the **Observe on Agent Answer** on the Vector Directory Number screen.

The observer views the name of the VDN, agent, or trunk as each is accessed by the VDN. During vector processing, the observer views the VDN name, but when the call connects to an agent, the observer views the agent name.

When the observer connects to a call in vector processing, Communication Manager maintains the connection until the call is disconnected or the observer hangs up, even if the call is routed or transferred externally.

If the observer does not disconnect after one observed call is disconnected, the observer is connected to another call on the same VDN. Observing is in the Listen Only mode as long as the call is in vector processing. When the call is out of vector processing, the observer can change from Listen Only to Listen/Talk, provided the observer has Listen/Talk observing permissions.

Service Observing with Exclusion

You can administer the **Allowed with Exclusion: Service Observing** field on the Feature-Related System Parameters screen to use Service Observing to listen to or record calls to a station or an agent login ID that has the Exclusion feature active.

If you administer this field as **y**, Communication Manager allows service observing of a station with Exclusion active by Class of Service (COS) or manual activation of the **exclusion** button.

If you administer this field as **n**, Communication Manager denies service observing of a station with Exclusion active. If an agent who is being observed presses **exclusion**, Communication Manager drops all bridged parties including the observer. **n** is the default option.

ASAI Single Step Conference calls with Exclusion

You can administer the **Allowed with Exclusion: SSC** field on the Feature-Related System Parameters screen to allow ASAI Single Step Conference (SSC) parties to bridge on to an existing call that has the Exclusion feature active.

If you administer this field as **l**, Communication Manager allows invisible SSC parties a connection towards a station with Exclusion active, either by Class of Service (COS) or by manual activation of Exclusion.

If you administer this field as **b**, Communication Manager allows both visible and invisible SSC parties a connection towards a station with Exclusion active, either by Class of Service (COS) or by manual activation of Exclusion.

If you administer this field as **n**, Communication Manager denies a SSC connection towards a station with Exclusion active. If an agent who is being observed presses **exclusion**, Communication Manager drops all bridged parties including the observer. **n** is the default option.

VDN Observing by Location

Note:

SIP phones do not support service observing using a FAC. SIP phones use **Service Observing** or **SO** feature key to activate the VDN Observing by Location feature. The system presents users with prompts to gather the necessary information.

Observers can use VDN Observing by Location to:

- Begin an observing association with a VDN by typing a Feature Access Code (FAC). Communication Manager establishes the observing connection when an agent in the required location answers the call.
- Indicate the location of the agents that observers want to observe by typing a location ID number. Communication Manager connects the observers to calls delivered to agents with the assigned location ID number.

  Note:

  Use the pound (#) sign after the location ID number to indicate end of digit entry as multiple options are present. For example, you can enter 1, 2, 3, or 4 digits of a location ID number.

  For information about location ID numbers, see *Multiple Locations*.

You can use the following access codes for VDN Observing by Location:

- Service Observing by Location Listen Only Access Code
- Service Observing by Location Listen/Talk Access Code

Processing of the VDN Observing by Location FACs is similar to that of the VDN Observing by FACs, except that you can enter the location ID number. For the VDN Observing by Location FACs, Communication Manager provides a third dial tone for entry of the location number.

For example, when an internal station user dials a VDN Observing by Location FAC, Communication Manager provides a second dial tone prompting the user to type a valid local VDN extension number. After the user types the extension number, Communication Manager provides another dial tone to accept the entry of 1, 2, 3, or 4 digits, ranging from 1 to 2000, of a location ID number. Communication Manager then checks if the number entered is a valid Multiple Locations number.

Note:

Use the pound (#) sign after the location ID number to indicate end of digit entry as multiple options are present. For example, you can enter 1, 2, 3, or 4 digits of a location ID number.

You can also activate VDN Observing by Location by the Remote Access feature or the **route-to** vector command.

Remote access feature activation

To activate VDN Observing by Location remotely, see Observing remotely by FAC on page 218. Communication Manager provides the first dial tone for the VDN Observing by Location FAC entry, the second dial tone for a valid VDN extension entry, and a third dial tone for a location ID number.

Route-to vector command

Assign at least one of the VDN Observing by Location FAC digit strings as the **route-to number** destination, or collected/assigned in the digits buffer for a **route-to digits** vector command. Communication Manager then provides a dial tone for entry of a VDN extension and a second dial tone for entry of a location ID number.

You can also enter the VDN Observing by Location FAC followed by the VDN extension. For example, route-to number *23345600 or route-to digits with DIGITS set equal to *23345600. In this case, the caller hears the dial tone and enters the location ID number.

If you activate VDN Observing by Location using the **route-to vector** command, the system plays a confirmation tone indicating that the observation of a VDN is successful. For an incorrect entry, the system plays a denial tone.

You can use a new special character sequence, *~L* for location, as a separator between the VDN extension number and the location ID number for a complete observe by location activation with the **route-to number** vector command.

For example, with an FAC of *23, a VDN 345600, and a location ID number 125, the destination parameter for the **route-to number** command is *23345600~L125#.

Note:

Do not use the special character sequence with the **route-to digits** command as digits are numeric or must start with an asterisk (*) sign for the command.

After successful activation, Communication Manager puts the observer in the wait-state for the VDN until a call, originally directed to the VDN, is delivered to an agent with the assigned location ID number. The observer can be in the wait-state for a long time if all agents in the desired location are busy or have skill level priorities. The observer can hang up to stop observing calls.

You can have only one VDN observer on a call. Observers of all calls to the VDN connect to a call at the beginning of vector processing and therefore have precedence over the VDN *Observe by*

Location observers of the same VDN. The *Observe by Location* observers can observe only those calls that are not already being observed when delivered to an agent in the desired location.

To increase the chances for observing calls by location, do not use observing calls to the VDN simultaneously as observing by location.

Non-by location observing of VDNs with the **Observe on Agent Answer** field administered as **y** still results in selecting a call from the start of vector processing and interacts with VDN Observe by Location. This interaction is in the same manner as observers of all calls although the observer hears only when Communication Manager delivers the call.

Related links

Multiple Locations on page 159

Service Observing indicators

The following tables apply primarily to non-SIP-based Service Observing endpoints. Users using SIP phones for observing generally hear an audible beep followed by a display message indicating the denial reason.

General indications to observer

The following table shows general Service Observing indicators that observers receive.

Condition	Button lamp	Tone
Not active	Dark	None
Denied activation	Broken flutter	Intercept/busy/reorder
Activated	Steady/Winking	Confirmation tone followed by silence or connection to call.
Observing (listen only)	Steady	Hear call
Observing (listen/talk)	Winking	Hear/talk on call
In wait state	Flash	None
Denied observing	Flash (wait state)	Silence/ineligible tone followed by silence

Tables showing what observers receive when using Service Observing

The following tables include the indications that observers receive when observers activate and use Service Observing. In the tables:

- Wait state means that the observer has activated Service Observing but there are no calls or a call cannot be observed. A call appearance is not reserved. The observer must have an idle call appearance available to be used by Service Observing when an observable call comes in.
- Ineligible tone is heard when an observed call becomes ineligible for observation. This tone is the “hold confirmation tone” - a rapid series of 5 short 440-Hz beeps. The observer does not hear this tone if the agent receiving the ineligible call hears zip tone.

Feedback to observers when activation denied			
Condition	**State**	**Lamp**	**Tone**
No such extension	denied	broken flutter	intercept
Extension not observable	denied	broken flutter	intercept
Not allowed COR (Extension COR cannot be observed or COR for observer calling permission does not allow observing the COR of extension to be observed.)	denied	broken flutter	intercept
Extension has Data Restriction	denied	broken flutter	intercept
Extension has Exclusion Active	denied	broken flutter	busy
Extension has Data Privacy Active on call	denied	broken flutter	busy
Extension already observed	denied	broken flutter	busy
Extension is an observer	denied	broken flutter	busy
Extension being busy-verified	denied	broken flutter	reorder
Extension has a 6-party conference	denied	broken flutter	reorder
COR doesn't allow SO activation	denied	broken flutter	intercept
Observe VDN not optioned	denied	broken flutter	intercept
Logical ID not logged In	denied	broken flutter	busy
Activation to logical with physical observed	denied	broken flutter	busy
Activation to physical with logical ID observed	denied	broken flutter	busy
Maximum VDNs being observed	denied	broken flutter	reorder

Feedback to observer when activation allowed - at time of activation			
Condition	**State**	**Lamp**	**Tone**
Active-eligible call	observing	steady or blinking	confirmation tone followed by connection to call
No active call	wait state	flash	confirmation tone followed by silence
Call ineligible	wait state	flash	confirmation tone followed by silence
Call has No Observe COR	wait state	flash	confirmation tone followed by silence
VDN call already being observed	wait state	flash	silence

Feedback to observer when activation allowed - after observe activated			
Condition	**State**	**Lamp**	**Tone**
Inactive or ineligible call	wait state	flash	silence
Call in 6-party conference	wait state	flash	silence
Call already being observed	wait state	flash	silence

Table continues…

Feedback to observer when activation allowed - after observe activated			
Condition	**State**	**Lamp**	**Tone**
Call is being busy-verified	wait state	flash	silence
Call has Data Privacy active	wait state	flash	silence
Call has Data Restriction	wait state	flash	silence
Call has Exclusion Active	wait state	flash	silence
Active-eligible call (in listen-only mode)	SO listen	steady	hear call
Active-eligible call (in listen/talk mode)	SO listen/talk	winking	hear/talk on call
Press button while observing in listen-only mode	SO listen/talk	winking	hear/talk on call
Observer presses Release	not observing	dark	none
Call has No Observe COR	wait state	flash	silence
VDN call already being observed	wait state	flash	silence
No active eligible call	wait state	flash	silence
Eligible VDN call	observing	steady/ winking	hear call
Eligible VDN call (in vector processing)	SO listen	steady	hear call
Eligible VDN call (out of vector processing in listen-only)	SO listen	steady	hear call
Eligible VDN call (out of vector processing in listen/ talk)	SO listen/talk	winking	hear/talk on call
Press button while observing in vector processing	SO listen	steady	no change to mode
Press button while not in vector and in listen-only	SO listen/talk	winking	hear/talk on call
Call being observed becomes ineligible	wait state	flash	ineligible tone followed by silence
Active call disconnects	wait state	flash	silence
Logical agent logs out	denied	broken flutter	busy, then silence
Observer (without button) hangs up	deactivates observing	n/a	n/a

The Service Observing button and IP phones		
Action by observer	**What happens**	**Condition**
No action.	The **serv obsrv** button is not highlighted.	The phone is idle.
The observer presses the Service Observing button.	The **serv obsrv** button is highlighted.	The system is waiting for dialing to begin.

Table continues…

The Service Observing button and IP phones		
Action by observer	**What happens**	**Condition**
The observer dials a station extension, Agent ID extension, or a VDN extension.	The **serv obsrv** button remains highlighted and displays two down arrows.	The observer has successfully established a session and is waiting for a call.
No action.	The **serv obsrv** button remains highlighted. The call display reflects the call being observed. Two arrows are displayed if the observer is in the Listen/Talk mode. No arrows indicate the Listen Only mode. Initially, Service Observing starts out in the Listen Only mode.	An observable call is offered to the observer.
The observer presses **serv obsrv** again.	The Service Observing mode of operation is toggled between Listen Only and Listen/Talk and vice versa. The mode is indicated by the appearance or lack of two down arrows on the **serv obsrv** button.	The observer has actively changed the Service Observing mode.

While observing, the observer can press only the following buttons:

- Call Appearance
- Service Observing
- Position Busy
- Auto-ckt Assure
- Release. This ends Service Observing
- Bridged Appearance
- Auxiliary Work
- Queue Status (NQC, OQT, AQC, and AQT)
- System Night Service
- Hold (ignored)

General security

Use the following Class of Restrictions (COR) to prevent unauthorized observing.

- For an observer, administer **Can Be A Service Observer** on the COR screen.
- For an agent to be observed, administer **Can Be Service Observed** on the COR screen.
- For an observer, grant permissions in **Service Observing Permissions COR table** on the COR screen.

VDN call security

Use the following CORs for VDN call observing.

- For a VDN extension to be observed, administer **Can Be Service Observed** on the COR screen.
- For a VDN destination, administer **Can Be Service Observed** on the COR screen.
- For an observer, enter the VDN extensions to be observed in **Service Observing Permissions COR table** on the COR screen.

Vector-initiated security

You can use the following guidelines for vector-initiated Service Observing (SO):

- Create a vector for SO.
- Grant calling permissions to an observer who monitors the Vector Directory Number (VDN) locally.
- Use Call Prompting commands in the SO vectors to provide passcode protection and to limit access to destinations or vector-verified, caller-entered digits.
- Use **`route-to`** commands to observe a VDN extension, but ensure that the extension has a COR number with SO permissions.
- Use time of day (tod) or day of week (dow) checks in the SO vectors.

In vector-initiated SO, the COR number that you assign to the following determines if the system permits SO:

- Agent login IDs. If you do not allow SO of a login ID, Communication Manager denies SO regardless of the COR permissions that you assign to a VDN or a caller.
- Internal caller extensions.
- VDN.

When a call routes through multiple VDNs, Communication Manager uses the COR number of the last VDN for calling or observing permissions regardless of VDN Override Rules.

If you administer a warning tone, the caller and the observer can hear the tone when Communication Manager connects the call to the answering or routed-to destination. You can use an announcement at the beginning of vector processing to tell the caller that the system observes all calls for quality and training purpose. The caller and the observer can hear the periodic tone during the call even if Communication Manager transfers the call to another communication server.

Remote access security

- Use barrier codes and authorization codes to limit the use of Remote Access to authorized users.

  For information about codes and other remote access security measures, see *Avaya Aura*® *Communication Manager Feature Description and Implementation*.

- Use different authorization codes for Service Observing (SO) permissions.
- Use Facility Restriction Levels (FRLs) and restrictions, such as the authorization code Class of Restriction (COR), to restrict Remote Access service observer access to other destinations such as stations or trunks.
- Use Call Prompting for additional security.

Assign the VDN, barrier code, and authorization code calling and Service Observing permissions and administer the **Can Be A Service Observer** field on the Class of Restriction screen.

Service Observing considerations

Observability

Although an agent can be a member of multiple splits/skills an agent can be observed by only one observer at a time. If two agents with different supervisors are observed and one agent calls the other, the originator's supervisor observes the call, and the other supervisor is placed in the wait state.

An attendant can be observed but cannot be an observer.

Coaching behavior

When coaching is activated for a an agent or station:

- The agent hears the coaching.
- A second observer hears the coaching.
- No other parties on the call hear the coaching.

When coaching is activated for a VDN:

- All agents on the call hear the coaching.
- All Service Observing agents on the call hear the coaching.
- No other parties on the call hear the coaching.

Ineligibility

A call to an agent or Vector Directory Number (VDN) extension is ineligible for observing when the call:

- Is already being observed. For more information, see *Service Observing with Multiple Observers*.
- Is being busy-verified.
- Has Data Privacy active.
- Has Data Restriction active, that is, the call is conferenced with an extension that has Data Restriction active or is a VDN call that reached an extension that has Data Restriction active.

- Has Privacy - Manual Exclusion active, that is, the call is conferenced with an extension that has Privacy - Manual Exclusion active or is a VDN call that reached an extension that has Privacy - Manual Exclusion active.

  Note:

  If you use Service Observing with Exclusion, observers can monitor calls when manual exclusion is active.

- Is in a conference where adding the observer results in more than six parties.
- Is a VDN-observed call that reaches an unobservable extension or VDN.

Trunk calls

If an agent being observed makes an trunk-call, observation starts after the agent finishes dialing. For Central Office (CO) trunks, dialing is treated as complete when answer supervision is returned or when answer supervision timeout occurs.

Multiple VDN observers

Multiple observers can observe a single VDN simultaneously, but only one VDN observer is observing a given call to the VDN. There is no limit to the number of observers observing a single VDN as long as the total number of observers actively observing VDNs does not exceed 999.

When you set **Allow Two Observers in the Same Call** to y, the VDN observed call connects to:

- An agent that is being observed by an observer.
- A station that is being observed by an observer.
- A conference that includes another observer.

Conferenced calls

An observer cannot initiate a conference while observing.

If an observed agent conferences a call and the number of parties is less than six, the observer is placed in the wait state until the call is connected. Then the observer observes the conference. In addition, the observer is bridged onto any call on which the agent becomes active before the conference is complete. When the conference is complete, the observer is again bridged onto that call.

If an observed agent conferences a call and the number of parties (including the observer) is six, the conference is denied.

A call to an observed VDN cannot be monitored if the observer, caller, and other parties bridged onto the call constitutes more than six parties.

If a conference is being observed because an observed agent entered the conference, when the agent hangs up, the conference is no longer observed. If a conference is being observed because an observed VDN call entered the conference, observing continues until the call is routed to an unobservable destination.

Conference members are observed during a conference regardless of their COR setting.

If a VDN call being observed is conferenced to an agent call being observed, the VDN observer continues to observe and the agent observer goes into wait state. If two observers (of either VDN or agent calls) are conferenced to a call, the first observer conferenced-in continues to observe and the second observer goes into the wait state. VDN or agent call observers hear the ineligible tone before going into wait state.

The same rules apply when multiple observers monitor transferred calls.

Supervisors must always be observing a call before trying to coach a call. For conference calls, prior to Release 8.1.1, coaching on a conference call was blocked. With Release 8.1.1, if the **Coach on SSC** option is enabled, supervisors can coach agents using the Service Observing Coach functionality even while the call is being recorded by recording applications that use the invisible ASAI SSC option to record calls. Prior to this option being available, supervisors could not coach agents on any call that was being recorded. If the **Coach on SSC** option is disabled, agent coaching continues to be blocked on any other conference call. Note that if the call is an invisible ASAI SSC call, the phone displays for the other parties on the call do not say “Conference”.

Transferred calls

Observers cannot initiate a transfer while observing.

When a service-observed agent starts a call transfer by pushing the Transfer station-button, all parties connected to the call are placed on hold, and the service observer is placed in the service observing wait state hearing silence. The service observer continues to hear silence while the agent dials a second party and hears ringback. When the second party answers the agent's call, the service observer is reconnected to the transferring agent. When the Transfer station-button is pushed again to complete the call transfer, all held parties are reconnected to the call, the transferring agent is disconnected from the call, and the agent's service observer is placed in the service-observing wait state hearing silence.

A VDN observer continues to monitor the transferred call until it is transferred or routed to a unobservable destination.

Service Observing support with users on AES CTI clients

This feature allows users on AES CTI clients to access the Service Observing feature. When the AES CTI client activates Service Observing, only specific supported devices (listed below) will function properly and recognize Service Observing activation. Only the listed devices will receive notification that Service Observing is activated by the AES CTI client. This supported SIP device must have the “sip-sobsrv” button administered.

The SIP devices that support this feature include:

- J169CC
- J179CC
- J189CC
- AES CTI devices

Note:

When using the “sip-sobsrv” feature button, coaching is only available from these SIP devices when the user is logged on as an agent.

Service Observing interactions

Interaction	Description
Adjunct Switch Application Interface (ASAI)	A call to an observed Vector Directory Number (VDN) continues to be observed after being routed to an adjunct. A call can be routed to a Service Observing (SO) Feature Access Code (FAC) by the **adjunct routing** command in the same way that the call is routed with the **route-to** command.
Assist	A VDN observer continues to observe a call during an assist operation. The observer observes the caller on hold and the conference, when the agent conferences the assist call with the VDN call.
Avaya Call Recording (Witness)	The listen-only/no-talk mode for Service Observing reduces the time slot usage required by the recording device to connect to monitored stations and ACD agents, allowing greater call recording capacity. Since the Service Observing No-Talk FAC used by the recording device to activate a recording session does not provide a talk-path, a warning tone cannot be inserted by the recording device when the recording session begins. To notify the parties in a call that the call can be recorded, design the vector to execute an **announcement** vector step before queuing or routing the call that provides an audible warning message. You can also administer Communication Manager to provide the warning tones by enabling the **Service Observing Warning Tones** on the Feature-Related System-Parameter screen.
Basic Call Management System (BCMS)	BCMS does not report on Service Observing. BCMS reports show normal measured-call and agent activity related to Service Observing calls. When a non-EAS agent is observed, the BCMS Report By Login ID shows the physical extension along with the login ID.
Bridged appearances	If an observer monitors agent extension 3082, the observer is bridged onto calls only to 3082. If the agent with extension 3082 has a bridged appearance for extension 3282, calls to extension 3282 are not observed. Although extensions 3082 and 3282 have a call appearance on the same telephone, the observer cannot observe both extensions at the same time.
Busy-verification	An observer cannot observe an agent call that is bridged onto by busy-verification. Also, an agent’s call that is being bridged onto by an observer cannot be busy-verified.
Call Coverage/Call Pickup	An observer cannot observe a call answered by a covering agent or member of a pickup group until the called agent bridges onto the call. The observer continues observing a call to an observed VDN call if the call is routed to a destination that forwards the call, using Call Coverage, Call Forwarding, or Call Pickup.
Call Park	An observer cannot park a call while observing the call. An observer observing a VDN continues observing after a call is parked.

Table continues…

Interaction	Description
Call Waiting	A call cannot wait on a single-line phone that is being observed.
Call Work Codes/ Integrated Directory	The observer does not hear agent dialing with these features because the digits are passed to the communication server in S-channel messages.
CMS	When an observer is bridged onto a VDN call, CMS is notified.
Conference and Transfer	A VDN observer who is bridged on a call follows the call on a conference and/or transfer operation.
Converse Command	Converse-split extension ports can be observed as physical extensions. A call to an observed VDN continues to be observed if the call is answered by a VRU through the converse command.
Converse-on Vector Command	Calls connected by the converse-on command are not observed by the VDN observer when the Observe on Agent Answer option is set to y. If the call is subsequently answered at an agent station or other destination using the route-to command, the VDN observer is bridged on the call.
DCS	To observe stations on another node (a DCS station extension), you must set up remote-access service observing. A DCS station can only observe another node using remote service observing. Service observing displays are not supported across DCS.
Dialed Number Identification Service (DNIS)	Observing by VDN provides monitoring by DNIS since the VDNs represent the DNIS of the service dialed.
Direct Agent Calling (DAC)	A direct agent call to a logical-agent ID is monitored by observing the Logical Agent not by monitoring the physical extension.
Hold	Observers cannot place calls on hold while observing. If an observed agent places a call on hold, the observer is put in wait state. A VDN observer continues to monitor the caller placed on hold.
Leave Word Calling	Parties on an observed call cannot use LWC.
Look Ahead Interflow	If an observed VDN call routes to another location using Look Ahead Interflow, the call continues to be observed. The observer hears a warning tone, if administered at the sending communication server, when the call arrives at the receiving communication server. The observer continues to hear the periodic tone while observing the VDN call.
Manual Answer	VDN observers are bridged on to the call when the agent answers the call that has been ringing the ACD agent extension with the Observe on Agent Answer set to y.
Move Agent/Change Skills	Moves or changes of physical of logical agents being observed occur according to the move or change rules. Observing continues.
Multiple Call Handling	While an agent extension or logical ID is observed, only the active call is monitored. If all calls are put on hold, the observer hears silence.
Music-on-Delay/ Music-on-Hold	If an observer is in listen/talk mode, neither caller nor observer hears music-on-hold. If an observer is in listen-only mode, the caller hears music-on-hold, but the observer does not. A VDN observer hears music provided to the caller.
Night Service	A VDN observer continues to observe when a call routes to night service.

Table continues…

Interaction	Description
Recorded Announcement	A VDN observer continues to monitor a call connected to an announcement. A Verify Announcement call placed by an observed physical or logical agent can also be observed.
Redirection on No Answer	A VDN observer continues observing a call after it is redirected or rings in limbo.
Route-to Number Vector Command	Calls connected by the **route-to number** command are observed by the VDN observer after “answer” is received when the **Observe on Agent Answer** field is set to y. This includes routing to internal destinations, such as stations, hunt groups, ACD splits or skills, the attendant, or to external destinations.
Trunks without disconnect supervision	Service observing cannot be activated over no-disconnect-supervision trunks. The caller hears denial indication.
VDN of Origin Announcement (VOA)	VDN observers with the **Observe on Agent Answer** field set to y are not bridged on the call until after the VOA is given to the agent. Therefore, the observer does not hear VOAs.
VDN Return Destination	You can create a prompting VDN with a return destination assigned so that, if you activate observing and it fails or the denial indication times out, the prompting VDN allows you to retry activation. This is true only if the denial and disconnection occur after the call leaves vector processing. If a vector step fails, the system proceeds to the next vector step. Disconnect or busy commands cause calls to be dropped and do not trigger return destination. When return destination is triggered, the call is monitored through each return destination operation until the caller disconnects. The observer bridged on the call follows the call when the VDN Return Destination feature, active on the VDN, redirects the call back through vector processing after the agent releases the call.
Telephone displays	The display for local observers matches exactly what is displayed on the observed physical or logical agent’s telephone display. For example: a=“3035001234 to Sales SO” While observing a VDN, an observer sees displayed the name of the VDN being observed while in vector processing. After the call leaves vector processing, the name of the agent or trunk group that the call is connected to is displayed.
VuStats	Non remote observers using 2-line displays can activate VuStats for an agent. An observer must activate VuStats before using Service Observing. The agent’s statistics appear on the second line of the observer’s display.
Zip tone	VDN observers do not hear the zip tone that the answering agent hears.

Service Observing interactions for SIP phones

Service Observing from a SIP endpoint works as follows:

- A service observer does not have to be logged-in as an agent to enable Service Observing on an object (VDN, agent, or station).
- If logged-in as an agent, a service observer must be in an AUX work state to enable Service Observing on an object (VDN, agent, or station).

Interaction	Description
Basic	The basic mode starts in the listen-only mode and can be changed to the listen-talk or coach mode, based on the configuration done on the phone.
No-talk	The no-talk mode starts in listen-only mode. This mode cannot be changed to the listen-talk or coach modes.
Next-Call	The next-call mode starts observing association when the next call arrives. This mode starts in the listen-only mode and can be changed to the listen-talk or coach mode, based on the configuration done on the phone. **Note:** Warning tones do not play when observing using the Next-Call mode.
VDN-By-Loc	For this mode, you must enter the VDN extension and location number. The VDN-By-Loc mode starts in the listen-only mode and can be changed to the listen-talk or coach mode, based on the configuration done on the phone. The VDN-By-Loc mode combines Listen Only and Listen/Talk versions of non-SIP FAC activated VDN SO.

Service Observing with Multiple Observers

Service Observing with Multiple Observers with **Allow Two Observers in the Same Call** set to y allows:

- Up to two observers to monitor the same agent Login ID or station extension using the Service Observing station button or using any of the following FACs: Service Observing Listen-Only, Service Observing Listen-Talk, Service Observing No-Talk
- Two separate calls, each with an associated service observer, to be conferenced with both service observers included in the conference call. However, if both observers are VDN observers, one VDN observer is dropped.
- Customers who use call recording products, such as the Avaya Witness Call Recording or NICE, to connect a voice-storage server to a station or Login ID extension to record agent-to-customer transactions acting as an observer.
- Customers who use call recording products to allow an observer to monitor a station or Login ID extension and record the transaction at the same time.

Note:

Service Observing with Multiple Observers does not allow multiple observers of a VDN to observe the same call with the Service Observing by VDN feature.

Service Observing with Multiple Observers example

A typical user of the Service Observing with Multiple Observers feature is a call center in a financial institution like a bank where all the calls must be recorded for security and quality control. The call center can use a call recording facility, such as Avaya Witness Call Recording, to record

the calls to each agent. Additionally a call center manager or supervisor can simultaneously use temporary service observing of an agent to monitor how the agent handles calls.

The benefit provided by Service Observing with Multiple Observers is exemplified when a call being recorded and observed simultaneously is conferenced with another call that is also being recorded. This feature allows the recording of both calls to continue.

Without Service Observing with Multiple Observers, one of the recording associations is temporarily removed during the duration of the conference because only one service observer, the recorder in this case, can be observing the same call.

Service Observing with Multiple Observers interactions

Interaction	Description
Agent Assist	If the supervisor an agent calls for help via the Assist button is currently observing the agent, when the supervisor picks up the call appearance to talk to the agent, the observing call appearance is dropped. Once the assist is over, the supervisor is no longer observing the agent.
Conference Tone	A second service observer added to a call does not have any effect on the current Conference Tone operation. If the second service observer drops or is dropped from the call, the tone is continued only if there are more than two parties left on the call.
Malicious Call Trace	If an incoming PRI trunk call with the MCT feature activated has been routed to an agent being monitored or on a conference call, the two observers are included in the MCT reporting for this call.
No-Hold Conference	No-hold call conference works the same way with two observers as it does when one observer monitors a call.
QSIG Path Replacement	If the QSIG Path Replacement operation takes place when multiple observers are either monitoring the call-legs at the originating Avaya communication server or are merged together when the path replacement takes place, the total number of observers left on the call will not exceed two observers.
Service Observing of VDNs	The Multiple Observers with Service Observing feature allows only one VDN observer on a call in vector processing. If a service-observing route-to number < FAC> vector step associated with the VDN is currently being observed by a VDN observer, a second observer can be added to the call only if the second observer is a station observer or a login ID observer.

Universal Call ID

Universal Call ID (UCID) is a unique tag that Communication Manager assigns to a call. UCID is an 8-byte data element that displays as a 20-character number. Communication Manager that is configured to create UCIDs assigns one UCID to every call, not just to ACD calls.

With UCID, you can track calls across multiple Communication Manager and Voice Response Units (VRUs). Call centers use UCID to track call history since UCID can uniquely identify every

call in a network of any size. For example, you can combine data from many locations and print reports to track a call throughout the life of the call.

In simple call scenarios, the tag stays with that call within a network that is based on communication servers connected by ISDN or SIP trunks. In complex call scenarios, the tag often merges with other tags.

Note:

The UCID data element is universal because it does not just identify a call on one particular communication server; a UCID uniquely identifies a call across a network of communication servers.

How does UCID work?

For every new call that comes into or is originated by the communication server or an IVR product, the product creates a UCID. Depending on the call scenario, the UCID will either remain unique to that call or merge with other UCIDs.

What creates UCIDs?

Both the communication server and the IVR system can create UCIDs once the capability has been enabled. In other words, neither product automatically creates UCIDs until the feature is enabled.

When are UCIDs created?

Once the communication server or the IVR system is administered to create UCIDs, these products assign a UCID to each call. For incoming calls over ISDN or SIP trunks, the communication server determines whether or not the call already has a UCID. If so, the communication server preserves the existing UCID and does not create a new one. If the call does not have a UCID, the communication server creates one when call processing begins.For incoming calls over trunks other than ISDN or SIP, the communication server does not create a UCID for the call because these trunks do not support the transmission of UCID.

For outgoing calls, the communication server creates a UCID when the caller goes off-hook.

How are UCIDs transmitted?

How communication servers transmit UCIDs depends on the sending and receiving equipment. The following table summarizes UCID transmission features.

Sender	Receiver	Connection	UCID contained in
Switch	Switch	ISDN (BRI or PRI) trunks using QSIG service protocol.	Codeset 0 Facility IE as manufacturer specific information (MSI) IE.

Table continues…

Sender	Receiver	Connection	UCID contained in
		ISDN (BRI or PRI) trunks using Shared UUI service protocol.	codeset 0 shared user-to-user information (UUI) IE.
		SIP trunks using Shared UUI service protocol.	User-to-User header of an INVITE message.
Switch	IVR	ASAI	Various ASAI messages.
IVR	Switch	ISDN-PRI	Codeset 0 shared UUI IE.
Switch	CMS	X.25	SETUP5 CMS message.
Switch	CTI adjunct	ASAI	Various ASAI messages.

UCID application requirements

Before you administer UCID, note that UCIDs are successfully transmitted only when all Communication Manager network components meet the requirements, software and connections, specified in the table.

Some requirements vary, based on the use of UCID. The following table lists requirements for different applications of UCID.

Operation	Requirement
UCID sent between two Communication Manager	ISDN (BRI/PRI) trunks with Shared User-to-User Information (UUI) or QSIG service protocol, or SIP trunks with Shared UUI
UCID sent from Communication Manager to the IVR system	ASAI link to the IVR system
Communication Manager receives UCID from the IVR system	ISDN-PRI connection (with shared UUI) between Communication Manager and the IVR system
Communication Manager sends UCIDs to CMS	Ethernet connection from Communication Manager to CMS
Communication Manager sends UCIDs to a CTI Application	ASAI link to adjunct

To maximize the benefits of UCID, use the updated version of Avaya Communication Manager.

In the case a Communication Manager network component cannot support UCIDs, administer the component, that is, trunk group, ASAI connection, or CMS software, to deny sending and receiving of UCID. For example, if an Avaya Communication Manager is connected to a non-Avaya switch, administer the connecting trunk to not send UCID over that trunk for outgoing calls.

UCID tracking scenarios

The way a network maintains and tracks a UCID depends on the call path. The following sections describe several call scenarios that illustrate UCID tracking throughout the life cycle of a call:

- Station-to-station calls
- Incoming Trunk Calls

- Outgoing Trunk Calls
- Simple Transfer or Conference
- Complex Transfer and Conference

Station-to-station calls

This scenario describes what happens when Phone I calls Phone II (both phones are on the same communication server).

The communication server creates a new UCID (such as UCID a) for any call originated by an internal station user.

Incoming trunk calls

UCID is assigned to an incoming call.

The communication server does one of the following:

- Receives UCID x information from an incoming call over an ISDN trunk.
- Creates UCID y for incoming calls that do not already have a UCID.

There is one CMS call history record for each incoming call.

Outgoing trunk calls

UCID is associated with the outgoing trunk call from Phone I.

The communication server creates a UCID (such as UCID x) for an outgoing trunk call and then sends it over an outgoing shared UUI or QSIG ISDN trunk.

The communication server creates a UCID (such as UCID x) for an outgoing trunk call even if the trunk does not support the transmission of a UCID.

Simple transfer or conference

This scenario describes a simple transfer or conference call scenario.

When an incoming trunk or station call is received by the station user at Phone I and transferred to or conferenced with another station user or outside party:

1. The communication server creates a UCID for the incoming call if it needs one.
2. The communication server creates a new UCID for the temporary conference or transfer portion of the call.
3. The communication server merges the temporary portion of the call with the original call when the conference or transfer is completed within the communication server. This is when the overriding UCID (such as UCID a), becomes the UCID for all parties within the communication server.

Note:

If the outgoing trunk does not support the sending of UCIDs, then the UCID of the outgoing call at the receiving communication server will be null.

If the call is transferred to another communication server, only the UCID for the transfer (UCID b) gets passed on. This is because the communication server cannot merge UCIDs if the call is not completed within the communication server.

Note:

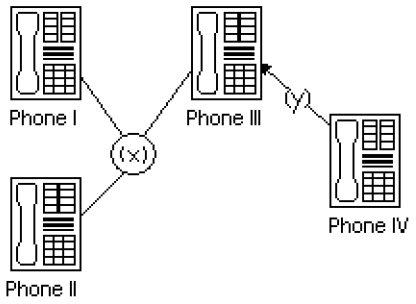
If, during the conference or transfer, the incoming call drops before the operation is complete, the two UCIDs will not appear to be associated because no merge of the two parts of the call was done.

Complex conference

The following complex call scenario illustrates when a station user adds an incoming call to an existing conference.

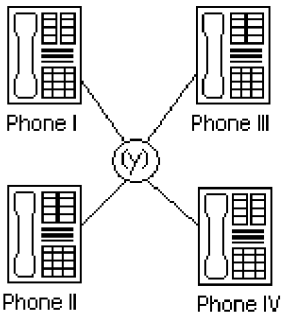
In this scenario,

1. Phones I, II, and III are in the same conference call with UCID “x”.
2. The person at Phone III receives an incoming call from Phone IV (this call has UCID “y” associated with it).



3. The person at Phone III puts the conference call on hold and answers the incoming call from Phone IV.
4. The person at Phone III decides to add Phone IV into the conference call.
5. The person at Phone III
   a. presses the Conference button
   b. presses the call appearance button to return to the conference call
   c. presses the Conference button again.

   This brings the conference call into the call between Phones III and IV.
6. UCID “y” overrides UCID “x” because the communication server views Phone IV as the primary party in the conference initiated by Step 5.



7. The UCIDs associated with each segment of the complex conference are sent to CMS if the parties in the call are measured (for this example, if the parties are ACD agents in a measured split or skill).

Configuration - communication server before the IVR system

The following scenarios describe what happens to UCID information when a call comes in to the switch before it goes to the IVR system. In this configuration, the IVR system serves as a Voice-Response Unit (VRU) that controls the routing of incoming ACD calls.

Note:

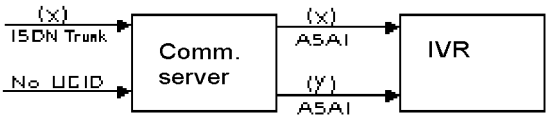
This configuration is more common than a call coming in to the IVR system before reaching the communication server.

This section describes two scenarios:

- Simple call tracking
- The IVR system transfers a call

Simple call tracking

The following call scenario describes when a call comes in to the communication server before the IVR system.



When the communication server is before the IVR system:

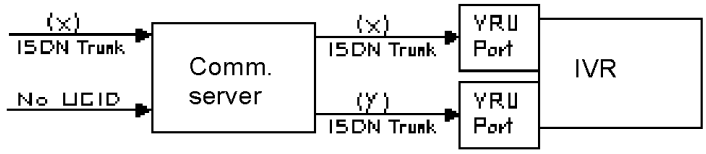
1. The communication server receives an incoming call over an ISDN trunk.
2. The communication server does one of two things:
   - If the incoming call has a UCID (such as UCID x), then the communication server passes it along.
   - If the incoming call does not have an associated UCID, the communication server creates a new one (such as UCID y).

3. The communication server passes the UCID to an Interactive Voice Response (IVR) voice system through an ASAI connection (using the activation of split or skill or VDN event notification by the IVR system.
4. UCID information is sent to the CMS if trunk, VDN(s), and/or split or skill(s) involved in the call are measured.

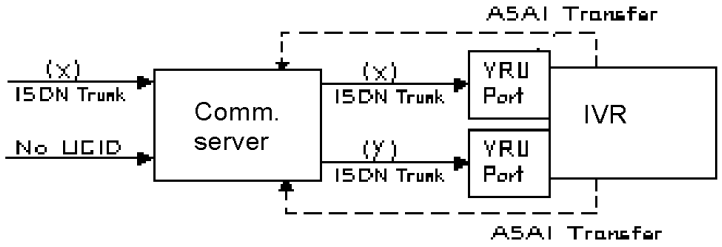
An IVR system transfers a call

The following call scenario involves an IVR system behind the communication server configuration when the IVR system initiates a call transfer after the call is answered by a port on the IVR system that serves as an ACD agent.

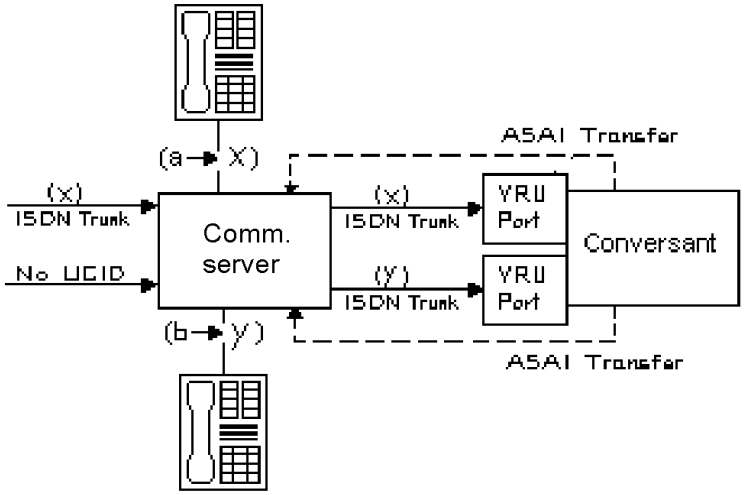
1. Call is directed to the IVR system VRU port (typically by call vectoring) with UCID information (UCID x or UCID y).



2. The IVR system determines the call's destination and transfers the call (using an ASAI third-party transfer operation).



3. The communication server temporarily creates a new UCID (such as UCID a or UCID b) for the transfer portion of the call (the original UCID is quickly merged into the call).



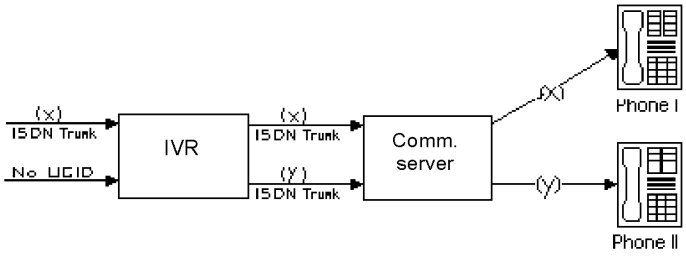
4. The UCIDs of the transfer segment and merged call are returned to the IVR system in ASAI acknowledgment messages.
5. The communication server sends UCID information to CMS if trunk, VDN(s), and/or split or skill(s) involved in the call are measured.

Configuration - IVR before the communication server

This scenario illustrates a system configuration where a call comes in to (Interactive Voice Response (IVR) before reaching the communication server. In this configuration, IVR provides voice response services or call screening so that the number of incoming calls to the communication server is reduced.

Note:

This configuration is less common than the communication server before the IVR configuration.



When IVR is before the communication server:

1. IVR receives an incoming call with UCID x.

   or

   IVR creates a new UCID y and associates it with the incoming call (if the call has no UCID already associated with it).

Note:

For IVR to recognize an incoming UCID (such as UCID x) from an ISDN trunk, special IVR scripting is required. When IVR receives a call from the public network, it automatically creates a new UCID because it cannot recognize whether or not the call already has a UCID.

2. IVR sends UCID to the communication server over an ISDN-PRI trunk.
3. The communication server receives UCID and reuses it for the incoming call.
4. The communication server reports UCID to the CMS if the trunks, VDNs, or splits/skills associated with the call are measured.

UCID considerations

UCID is tested with several major carriers. To find out if the UCID capabilities work with your carrier, check with your account team for the latest information. If testing is not done to verify operation over the public networks involved with the preferred specific configuration, use private trunking between the nodes until successful testing is complete.

UCID interactions

Interaction	Description
Distributed Communications System (DCS)	If you plan to use DCS in a network of Communication Manager where UCIDs are tracked, you must configure DCS with ISDN or SIP trunks having the `shared UUI` service protocol. Otherwise, calls handled through one of the DCS features, such as DCS Coverage, do not retain the UCID initially assigned to the call.
Remote Messaging System	For a remote messaging system over DCS, configure the DCS trunks used to accomplish the remote messaging system operation as described in Distributed Communications System to retain the UCID associated with a call.
Tandem Calls	When a tandem call is made through Communication Manager, you can choose to pass or block the UCID information through the tandem Communication Manager. To pass a UCID through a tandem Communication Manager, configure both the incoming and outgoing trunks at the tandem Communication Manager to handle UCIDs. For proper private and public network information forwarding administration, see “Information Forwarding”.

VDN of Origin Announcement

VDN of Origin Announcement (VOA) provides agents with a short message, no longer than 1.2 seconds in duration, about the city of origin of the caller or the requested service based on the VDN used to process the call. Use VOA messages to help agents to respond appropriately

to callers. For example, if you have two toll-free numbers, one for placing orders and one for technical support, you can administer two unique VDNs to route calls to the same set of agents. When an incoming call is routed to a VDN with a VOA assigned (for example, new order or tech help), the VDN routes the call to a vector, which can place the call in an agent queue. The agent hears the VOA message when answering a call, and can respond appropriately to the caller's request.

The agent cannot hear the caller while the VOA message is playing. The caller is not connected to the agent until after the message completes and cannot hear the message or the agent during the message. The caller hears ringback while the agent is listening to the VOA.

Agents logged in at multiline telephones see the call-appearance button for an incoming call flash until after the VOA completes. An agent can press the flashing call-appearance button to stop the VOA.

To repeat the VOA, an agent presses the VOA Repeat button. The VOA Repeat button lamp lights during the VOA. The VOA Repeat button lamp remains lit if the repeat request is queued. If an agent presses the VOA Repeat button while the lamp is lit, the VOA is stopped. If an agent presses the VOA Repeat button but there is no VOA or the system cannot play the VOA within three seconds, the lamp flutters.

Note:

VOA-repeat functionality is not supported on 96x1 SIP phone types. You must not select the voa-repeat button while configuring 96x1 SIP extensions.

You assign VOAs for each VDN. However, the VOA applies to a COR, so you must administer a COR for agents who will receive VOAs.

You can set up VOAs in four ways:

- Agents can hear a unique announcement based on the dialed number identification service (DNIS) received from the service office or carrier communication server. Assign each DNIS as the VDN of a vector. Set up the VOA to announce the services associated with the DNIS.

  Note:

  The announcement associated with the current VDN only plays if the VDN Override for the previous VDN is set to y. If VDN Override for the previous VDN is set to n, the VOA associated with that VDN plays.

- Use vector steps, an integrated prompting, or converse-on step to route calls to a VDN. Set up the VOA to announce the service the caller requested or to announce a condition that caused the call to route-to the VDN.
- You can route calls to a voice response system, directly or through a vector. Use voice prompting to direct the caller to enter a touchtone response, and route the call to a specific VDN based on the caller's response. Set up the VOA to indicate the service the caller selected.
- If agents require a caller's city of origin, assign the trunk group to a particular VDN. Set up the VOA to provide the location of the origin of the trunk group. Subsequent VDNs can be used to handle the call, or multiple VDNs can be assigned to a single vector.

Note:

VDN Override applies to VOA in the same way that VDN Override applies to display information. If a VDN with a VOA has VDN Override enabled, the system overrides the original VOA with VOAs in subsequent VDNs to which the call is routed.

VOA considerations

- Keep messages brief, no more than 1.2 seconds in length, as callers are kept waiting while a VOA plays. Agents must use a speakerphone or headset, so agents do not miss the VOA. If agents cannot use a speakerphone or headset, administer phones with a **VOA Repeat** button.
- If you have multiple announcement boards, you must place shorter VOAs on one board and longer recorded announcements on the other to prevent delay in the delivery of VOAs. If you have only one announcement board, place VOAs on the integrated board and install an auxiliary announcement device for longer announcements.
- Agents must be on the same communication server as the VOA.
- A VOA can be assigned to multiple VDNs, but a VDN can have only one VOA.
- If you use integrated announcements, the system maintains a separate logical queue for VOAs. If the VOA cannot be delivered to the agent within 1 second because of traffic or inoperative equipment, the system does not provide the announcement. VOAs are higher priority than other announcements. A burst of VOAs can delay other announcements.
- Auxiliary announcements are connected for a duration of 1 to 2 seconds on a barge-in basis, immediately after the agent answers, or is assigned the call for auto-answer, and the incoming call is extended to the agent. Integrated and non-barge-in auxiliary announcements are connected for the duration of the announcement. The communication server does not ensure that the integrated announcement is shorter than the allowed playback time.
- VOA supports Auxiliary Trunks (aux-trunk) with barge-in, queue, or without queue. For aux-trunk with or without queue, when the trunk is idle, a VDN call seizes the trunk to start the VOA and the system plays the entire announcement, not just 1 to 2 seconds. However, if the announcement is busy and if aux-trunk has barge-in, the call does not queue but bridges onto the announcement for 1 to 2 seconds. When the VOA completes, the trunk is released along with the listeners, and the next call requiring the VOA starts the process over again. For this reason, your aux-trunk announcements must consist of one short announcement that repeats during the full announcement time. For example, you can record `New Order` as many times as possible, so that when a call bridges to the announcement, the agent hears `New Order` no matter where the agent bridges into the announcement.
- If you use aux-trunk or integrated announcement without queue and a port is busy when a VDN call comes in, the system does not play an announcement. If you use aux-trunk or integrated announcement with queue, the system plays the current announcement for an agent and connects the next agent in the queue.

VOA interactions

Interaction	Description
Agent Call Handling - Automatic Answer	• ACD agents in the Automatic Answer mode hear a zip tone before VOA. You can administer a zip tone after VOA completes to alert agents that an announcement is complete and a call is connected. • Non ACD agents can receive VOA if a call is routed to the agents by vector processing. When non ACD agents in the Automatic Answer mode receive calls, the agents hear a call ID tone before VOA. Agents hear a second zip tone after VOA indicating call connection.
Agent Call Handling - Manual Answer	When non ACD agents in the Manual Answer mode receive calls, the agents hear ringing, answer the call, and then hear VOA.
ASAI Adjunct Routing	If a vector step includes Adjunct Routing, VOA plays for the agent to whom the call is routed.
Auto-Available Split/Skill (AAS)	AAS is intended to be used for splits or skills containing only non human adjuncts, such as a voice messaging system or an IVR system. However, VOAs can be directed to AAS.
Call Forwarding	VOAs apply to forwarded calls, including those forwarded to a hunt group. The answering station must be on the same Communication Manager. If VOA is forwarded, the message is played only if the destination extension is administered with a COR that allows VOA.
Call Pickup	With Call Pickup, an agent can pick up a ringing call on another extension. If the pick up extension has COR permissions for VOA, the agent can receive VOA.
Conference	If an agent receives a call and then conferences in additional stations, any station on the connection can use **VOA Repeat** button to replay VOA. Only the person using the button can hear VOA unless the call is being service observed.
Converse-on split or skill	A converse-on split or skill is one used in a **converse-on** vector step. When a **converse-on** vector step is executed, VOA is not applied. After returning to the vector, the call can be routed to a station or VDN where the answering agent receives VOA as if the **converse-on** step had not been processed.
Coverage	VOA applies to coverage paths.
Data Restriction	Data Restriction prevents tones from being applied to line or trunk circuits during a data call. VOAs do not play for data-restricted calls.
Direct Agent Calling	Direct Agent Calling (DAC) allows a vector to route a call to particular ACD agent and have the call treated as an ACD call. VOA applies only to DAC if the calls reach an agent through vector processing. DACs from a phone administered on Communication Manager are not vector-processed. Hence, VOA is not played.

Table continues…

Interaction	Description
Enhanced Automatic Wake-up	If you use enhancements to Automatic Wake-up with integrated announcements, there can be some contention for integrated announcement ports. VOAs have priority over Automatic Wake-Up announcements.
Expert Agent Selection (EAS)	When you use EAS, the logical agent COR definition determines the assignment of VOAs for each extension. EAS uses the COR of the logical agent instead of the COR for the phone that the agent uses.
Hold	Agents cannot use the **VOA Repeat** button if their calls are all on hold. The button applies only to active calls.
Home Agent	You can assign an initial VOA to a home-agent port on Communication Manager. However, home agents cannot use a **VOA Repeat** button because home agents need a Dial Access Code (DAC) to reach features and VOA replay does not use a DAC.
Hunt Groups	VOAs apply to calls that are routed to a hunt group. The COR for the extension of the answering station determines if the station can receive VOA.
Look-Ahead Interflow (LAI)	VOAs apply only to the Communication Manager server where the VDN is defined. If a call interflows to another Communication Manager, VOA is lost. You can administer the interflow to another Communication Manager access a VDN with the same VOA message as on the original Communication Manager.
Redirection on No Answer (RONA)	If a call requeues to a split or skill because the RONA timer expired, VOA applies to the call when an agent answers the call.
Service Observing	The system handles Service Observing calls as conference connections. If the observer presses the **VOA Repeat** button only the observer hears the announcement. However, if another party on the call presses the **VOA Repeat** button, the user and the observer hear VOA.
Supervisor Assist	If an agent requests for supervisor assistance and conferences the supervisor into a call, the agent or the supervisor can use the **VOA Repeat** button to replay VOA, but only the person who presses the button hears VOA.
Transfer	If an agent receives a VDN call and transfers the call, the answering party can use the **VOA Repeat** button to replay the message.
VDN Override	The VDN extension that is used for VOA follows VDN Override Rules. Communication Manager plays VOA for an agent who answers a call at the extension that is assigned to the active VDN.
VOA distribution	If you use long VOAs or multiple VOAs, there can be a delay between the zip tone and the announcement. Communication Manager provides multiple announcement circuit packs to help prevent announcement delays.

VDN Time Zone Offset

Reason to use VDN Time Zone Offset

If you have identical opening and closing times in different locations, you can use a single vector to handle the opening and closing time checks using VDN Time-Zone Offset in a manner similar to skill preferences. This simplifies programming and allows sharing of vectors. For example, you can use 9 to 5 as the time in all vectors without converting to the local time reflected on each Communication Manager clock.

Operation

Call Vectoring TOD goto step vector conditionals are calculated based on the main server system clock local time. The main server system clock uses the local server rules for the date, day, year, time-zone, and DST. The default setting for DST is for the main location (location 1) with the Multiple Locations feature active.

Using VDN Time-Zone Offset, you can modify the time used for the Time of Day (TOD) conditional calculation based on the active VDN for the call. You can base the TOD values on the local time relative to the VDN where the calls are directed. In addition, if you apply the offset on a VDN basis, you can apply common call flows using the same vector for calls to different VDNs whose application requires the TOD conditional calculations based on different time zones.

Example of VDN Time Zone Offset

In this example, a call center company has locations in London, New York, and Denver. The server is located in London and the gateways are located in London, New York, and Denver. All of the locations share the same opening and closing times. Opening time is 9:00 a.m. and closing time is 5:00 p.m. Calls routed to each of the locations are given a separate VDN, each dedicated to routing calls to that location. The company wants to program one vector to handle each of the locations, including the opening and closing time checks. The company can do this by using the VDN Time Zone Offset feature along with skill preferences. The company uses EAS and the system Communication Manager clock is set to GMT.

Assuming Daylight Saving Time is not active, the tod conditional check done in step 2 for calls to VDN1 is based on the server local time in London England, that is, the Greenwich Mean Time (GMT). For calls to VDN2, the time used is the server local time GMT-5 hours or Eastern Standard Time. For calls to VDN3 the time used is GMT-7 hours or Mountain Standard Time.

The VDN assignments are described in the following table.

VDN	Location	Extension number	Time zone offset	Skill preference
VDN1	London	10001	+00:00	lst = 51
VDN2	New York	10002	-5:00	lst = 60
VDN3	Denver	10003	-7:00	lst = 75

Each of the following VDNs are assigned to vector 201:

1. wait-time 0 secs hearing ringback
2. goto step 7 if time-of-day is all 17:00 to all 09:00
3. queue-to skill 1st pri 1
4. announcement 30002 [“All our agents are busy. Please wait.”]
5. wait-time 60 secs hearing music
6. goto step 4 unconditionally
7. disconnect after announcement 30003 [“Our hours are between 9 a.m. and 5 p.m. Please call back.”]

VDN Time Zone Offset interactions

VDN Time Zone Offset interacts with other features as follows:

- VDN Time Zone Offset does not apply to time calculations associated with the *doy*, *dow* and *tod* Variables in Vectors variable types. The variable types use the server local time.
- The VDN Time Zone Offset used is the one assigned to the active VDN for the call. The active VDN follows the VDN Override rules.
- The LSP and ESS servers must be synchronized with the main server system time. This ensures that if a switch over occurs to the survivability server, VDN offset is applied consistently when the vector TOD conditional steps are being processed by the survivability server.

VuStats

With VuStats, agents, supervisors, call center managers, and other users can press a button and view the statistics for agents, splits or skills, VDNs, and trunk groups.

The statistics reflect information collected during the current BCMS interval, since the agent logged in, since the day began, or historical data accumulated over an administered number of intervals. The information is limited to 40 characters displayed at a time. VuStats can display information on demand or update it periodically.

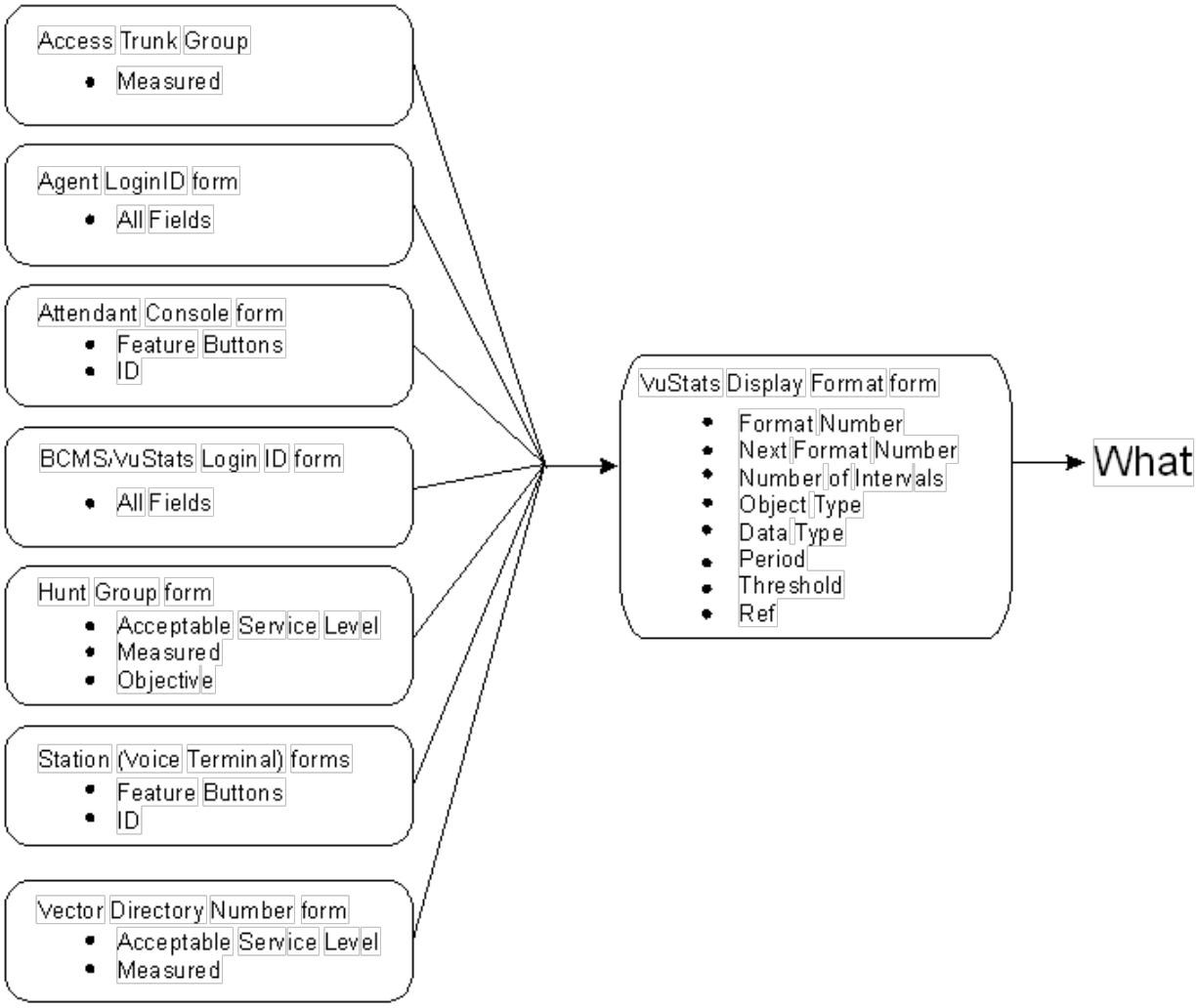
With VuStats, anyone with a phone that has a digital display can view the BCMS statistics, which are otherwise available only on BCMS reports or management terminals. Agents can monitor their performance and supervisors can use the statistics to manage splits or skills.

Although VuStats can run with BCMS or CMS, neither is required.

VuStats forms and fields

The following forms and fields determine information that VuStats displays.

Forms that determine what information appears on the VuStats display



Data type

Data type defines what data is displayed for an object type. For example, for an agent object type, VuStats displays the total number of calls that an agent has answered since login, the average time that the agent has spent on ACD calls, the number of agents available to receive calls for a split or skill, and the percent of calls within the acceptable service levels.

For split or skill object types, VuStats displays split or skill description and performance information, such as Average Speed of Answer (ASA), number of calls in a queue, and agent work states. VuStats also displays the acceptable service levels or percent of calls answered within the acceptable service levels for a split or skill.

For more information, see *Administering Avaya Aura® Call Center Elite*.

Period

VuStats can show statistics that have accumulated for the day or for an administered number of intervals. For example, if you administer VuStats to display the number of ACD calls for the past four completed intervals, VuStats displays the number of ACD calls received in the past two hours (half-hour intervals) or four hours (one hour intervals) plus those completed during the current interval using historical data can affect processor occupancy, depending upon the number of active users, the update rates, and the number of historical data types.

With agent or agent-extension object types, shift data is available for the number of ACD calls answered, the average ACD talk time, and AUX work mode time by reason code for an agent. You can clear shift data at midnight or the next time an agent logs in.

Threshold

Many data types can be administered with a threshold comparator and value. When the condition defined by the threshold is true, and the data type is shown on the display, the VuStats button lamp flashes. For example, suppose a format is created in which the oldest call waiting data type is administered with a threshold of >= (greater than or equal to) five minutes. Whenever that VuStats format is displayed, if the oldest call in queue has been waiting for five minutes or longer, the VuStats lamp flashes on the phone. Each time the display updates, the threshold is checked for each data type being displayed.

Format description

Use Format Description to create labels on the display to identify data. For example, in the example figure CallMaster with VuStats display, *AUX*= identifies the data type *split-agents-in-aux-all* that is, the number of agents currently in the AUX work mode for a specified split or skill. Text appears on the display exactly as you enter in the field. Text is optional.

Due to the 40-character limit, use abbreviations when possible. For example, use `S=` to indicate split number.

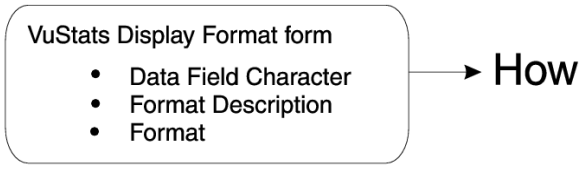
Display linking

Link display formats to increase the amount of information users can view. For example, link a display of information for an agent's first split or skill to a display of information for the agent's second split or skill. Or, link a display of information about the work states of all agents on a split or skill linked to another display of information about calls waiting, number of calls abandoned, or oldest call waiting for the split or skill.

If you use display linking, assign a Next button on agent phones.

How the information looks

The following fields on the VuStats Display screen determine how information looks on the VuStats display.



VuStats statistics appear on the second line of 2-line DCP telephone displays or on the first line of 1-line DCP telephones and all BRI telephones. On telephones with 2 x 24 displays, the display automatically wraps to the second line of the display. When VuStats is activated, it overwrites and cancels any display feature on the second line of a 2-line display and on the first line of a 1-line display.

You define the following format information on the VuStats Display Format screen:

- Labels for data types and the amount of space reserved for data
- Order in which data types appear on the display
- Format for time-related data types
- Display links

When the information updates

The following forms and fields determine when VuStats displays update.

Screen	Field
Feature-Related Systems-Parameters	**BCMS/VuStats Measurement Interval**
VuStats Display Format	• **Update Interval** • **On Change** • **Display Interval** • **On Change** • **Display Interval**

Most display features that use the second line of a 2-line display or the first line of a 1-line display overwrite and cancel VuStats. Reason codes and Call Work codes only suspend VuStats; when the prompt is removed, the VuStats display reappears.

Agents press the normal (exit) button to clear the VuStats display.

Administer VuStats to display information until agents press the normal button or another operation overwrites the VuStats display, or administer VuStats to display for an interval of 5, 10, 15, or 30 seconds.

You can also administer VuStats to update displayed statistics every 10, 20, 30, 60 or 120 seconds or every time an agent changes work mode or a BCMS Measurement Interval is completed, or not update at all.

VuStats considerations

Some VuStats data is accumulated for a login session of an agent. The shift data clears either at midnight or the next time the agent logs in depending on how the system is administered. If the

data clears at login and agents log out to go to lunch, the system clears the accumulated data when the agents log back in after lunch.

To accumulate the statistics for a day, agents and supervisors must keep a running total of all the login sessions. You can use historical data instead and require agents to enter the AUX work mode when agents are temporarily unavailable, or administer the system to clear shift data at midnight.

VuStats interactions

Interaction	Description
Basic Call Management System (BCMS)	VuStats displays data collected by BCMS.
Call Prompting	When an agent presses **vu-display**, the VuStats display overwrites the Call Prompting display.
Call Work Codes (CWC)	The CWC display prompts suspend VuStats. The system displays VuStats data after completing the CWC prompt. If an agent presses **vu-display** during the CWC prompt, the system cancels the CWC entry and displays VuStats data.
Change skills	The system cancels VuStats data when an agent changes skills.
Call Management System (CMS)	Moving an agent from one split or skill to another does not affect the ID assigned to the **vu-display** button. If an agent is moved from one split or skill to another, the system does not associate VuStats buttons from the previous split or skill to the new split or skill. Therefore, if you must frequently move agents between splits or skills, do not associate the VuStats buttons with a specific split or skill. Instead, associate the VuStats button with the agent format (without an ID) on each agent phone and use a split or skill reference to view the split or skill.
Expert Agent Selection-Preference Handling Distribution (EAS-PHD)	VuStats provides statistical data for up to 20 skills for each agent. However, agent statistics by skill, that is, agent or agent-extension object types, are available only for the current interval or for the shift-acd-calls and shift-average-acd-talk-time data types.
Integrated Directory	If an agent activates Integrated Directory, VuStats is automatically canceled. The Integrated Directory display overwrites the VuStats display. Agents can press **vu-display** to view VuStats data.

Table continues…

Interaction	Description
Queue Status (QStats)	The **q-calls** and **q-time** buttons cancel VuStats. Agents can press **vu-display** to view VuStats data.
Reason Codes	You can use VuStats data types to record real-time and historical AUX work mode time by reason code or AUX work mode time summed for each reason code. The Reason Codes display prompt suspends VuStats. The system displays VuStats data after completing the Reason Code prompt.
Service Observing	On phones with a single-line display, the Service Observing button display automatically cancels VuStats. Agent can press **vu-display** to view VuStats data.

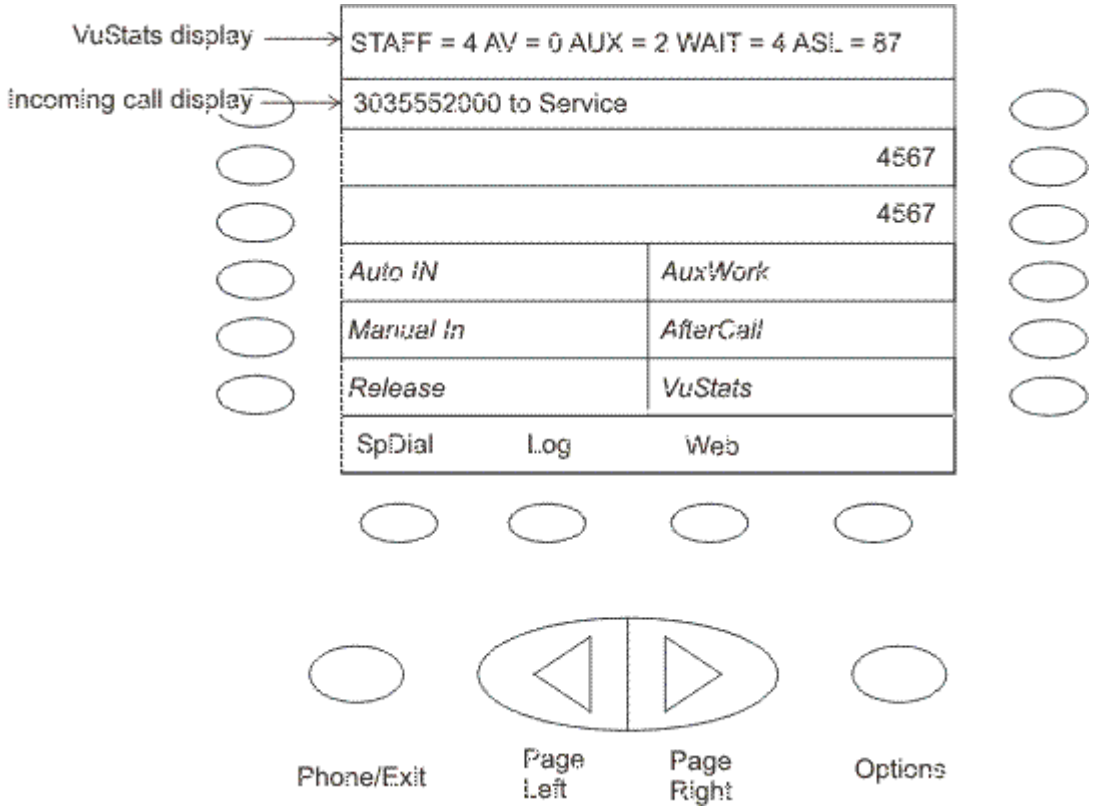
VuStats display interactions and modifications for IP telephones

If you are using Communication Manager and a 4610, 4620, 4621, 4622, 9620, 9630, or 9650 type IP telephone:

- Incoming calls are displayed on the incoming call display line (second from top), and the top information display line continues to display the VuStats after a slight flicker.
- For outgoing calls, the top VuStats display line is suspended when the agent originates an outgoing call. The digits dialed are echoed on the associated incoming call display line. When the call is being made, the VuStats appear again on the top line after a slight flicker.

Other single-display-line sets including the 2400 series function in the similar way like the 4xxx and 9xxx phones.

The VuStats and incoming call information displays on these types of telephones are depicted in the following diagram of a 4622 telephone display.



With certain VuStats formats, the VuStats display line cannot be viewed during agent login or logout, during any other Feature Access Code operation, or during an off- and on-hook sequence (such as a misdialed number). In these cases, the VuStats display line is restored on the next successful received or placed call or when the agent presses the **VuStats** button. As is normal with single-display-line sets, deactivate VuStats to see the Caller-Info (collected digits) display.

Zip Tone Burst for Callmaster® Endpoints

With Callmaster® phones, you can use a single burst of zip tone for calls to ACD agents.

Setting **Zip Tone Burst for Callmaster Endpoints** to `single` eliminates the second burst of zip tone and reduces the time taken by the agent to begin a conversation with the caller. The default entry is `double`, which keeps the current operation of double burst zip tones.

This field applies to:

- ACD calls with the station or agent ID auto-answer option set to `acd`.
- Incoming Call Identification (ICI) tones for auto-answer non ACD calls with the field setting as `all`.

Note:

Use the single burst zip tone only when the agent clearly hears a single burst of tone to recognize an incoming call.

Chapter 4: Call Vectoring features

Call Vectoring involves processing of incoming and internal calls as per a programmed set of commands. The commands, called vector commands, determine the type of processing for calls. For example, vector commands can direct calls to on-premise or off-premise destinations, hunt groups, splits or skills, or to specific call treatments such as an announcement, forced disconnect, forced busy, or delay. Vectors can queue or route calls based on different conditions. For instance, you can route important calls to the most skilled agents.

Although Call Vectoring is primarily used to handle the call activity of ACD splits or skills, you can use Call Vectoring to serve several purposes.

For more information on Call Vectoring applications, see “Call Vectoring application examples” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Adjunct Routing

Adjunct Routing provides a means for an Adjunct Switch Application Interface (ASAI) processor to specify the destination of an arriving call when the processor encounters an **`adjunct routing link`** vector command during vector processing. An adjunct is a processor connected to Communication Manager that uses the ASAI protocol. The adjunct makes routing decisions based on caller information or agent availability, and returns a call route response to Communication Manager.

Communication Manager provides information in an ASAI route request message. Adjunct applications use the message to access a database and determine a route for the call. In a typical application, an ASAI adjunct uses the dialed number, Calling Party Number/Billing Number (CPN/BN), or the digits collected by Call Prompting to access caller information and to determine an appropriate call route.

Adjunct routing can be used in conjunction with the Call Prompting and Look-Ahead Interflow (LAI). When combined with one of those features, the following rules apply:

- When combined with Call Prompting, adjunct routing can pass up to 16 digits that are collected from the last relevant **`collect digits`** vector command.
- When combined with LAI, adjunct routing can pass the LAI information element or other call center-related data (with enhanced Information Forwarding) that was passed from the originating Communication Manager in the message or associated with the call from the local Communication Manager.

Adjunct Routing considerations

Consider the following points when working with Adjunct Routing:

- Before vector steps, try to insert a brief wait step for vector processing to handle any possible delays that cause the vector to fail and the call not to be routed. Follow this best practice when programming a vector:
  - Start all vectors with a “wait-time 0 seconds” step. Adding this step helps ensure you will have a touch-tone receiver (TTR) for vector processing.
  - When using an “adjunct routing” step, try to put a “wait-time 0 seconds” step before the “adjunct routing” step to provide a pause before routing the call to the adjunct system.
  - Any time digits are collected from a caller using a “collect digits” step, the step should be preceded by a “wait-time 0 seconds” step, hearing either ringback or music. This ensures you have a TTR to collect those digits.
- An adjunct specified in an **`adjunct routing link`** command can route a call to an internal number, an external number, a split, a VDN, an announcement extension, or to a particular agent. An adjunct can also provide priority ringing, priority queuing, and specify that a route-to an agent be done as a direct agent call.
- You can also include more than two consecutive adjunct routing link steps in a vector. This approach offers the following advantages:
  - Redundancy for ASAI link or application failure.
  - Simultaneous processing of multiple route requests, which distributes incoming call load more efficiently and results in faster call processing times.
- Vector processing continues to occur while an ASAI route request is being processed. Consequently, the first step to follow more than one adjunct routing link steps must be an announcement or a wait time step that adheres to the following rules:
  - If an **`announcement`** step follows immediately after an **`adjunct routing link`** step, the announcement must not contain any information essential to the caller, such as further instructions, since the announcement immediately terminates when the switch receives a destination from the ASAI adjunct.
  - If a **`wait-time`** step follows immediately after an **`adjunct routing link`** step, the step must specify ringback or music, but not silence, as the feedback option. If the caller hears silence, the caller might abandon the call.

  Important:

  If an ASAI link or application specified in the **`adjunct routing link`** step is out of service, the step is skipped. If the next step is not a **`wait-time`**, **`announcement`**, or **`adjunct routing link`** step, as much as six minutes can elapse before the switch determines that the adjunct application is out of service.
- The field **IC Adjunct Routing** should be disabled on the CTI Link screen unless the CTI application requires it enabled. Then, program the associated vectors as instructed for that application.

- When IC Adjunct Routing is disabled, only the first **announcement** or **wait-time** step encountered is skipped after a sequence of adjunct route steps fails. The following paragraphs apply when IC Adjunct Routing is disabled.
- The second step after the **adjunct routing link** step must be implemented as a default treatment in case the host application or ASAI link is down. Speed of execution for the default treatment step, for example, **route-to number** *0* if unconditionally, is controlled by the following factors:
  - If the ASAI link is down and if the first non-adjunct routing link step is a wait-time or an announcement treatment, the treatment step is skipped and the default step that follows the skipped, treatment executes immediately.
  - If the host application is not down, the default step executes if the adjunct does not provide a route within the time defined by the first non-adjunct step. For example, if the first non-adjunct step is an announcement, the default step executes after the time defined by the length of the announcement is exceeded.
- When a vector contains an **adjunct routing link** command and an ASAI link or application fails, special rules apply to vector processing operations that result. Consider the special processing operations while designing adjunct routing vectors.
- Vector processing continues while an ASAI call route request is processed at an adjunct. Succeeding vector steps can terminate an ASAI call route request if the steps execute before a call route can be provided by the adjunct. Alternately, the adjunct can reject the call route request, and subsequent vector processing proceeds normally.
- The **wait-time hearing i-silent** command allows the adjunct to decide whether to accept an incoming ISDN-PRI call. When this step is encountered after an adjunct routing link step, the switch does not return an ISDN PROGRESS message to the originating switch. The **wait-time hearing i-silent** command is important for the Network ISDN and LAI features.

Receiving and implementing an ASAI call route

Communication Manager that receives an adjunct-supplied call route performs various checks to validate the call route implementation. When the adjunct-supplied route is validated, the resulting operations are similar to those in effect for a **route-to** *xxxxx* **with coverage = y** command. The caller hears normal call progress tones and feedback, and if the call routes to an extension with no available call appearances and no coverage path, the caller hears a busy signal.

Other features in effect at the adjunct-supplied destination, such as Send-All-Calls or Call Forwarding, interact with the routed call.

Look-Ahead Interflow operations are not applied when calls are routed over trunks. Instead, ASAI-routed calls are directed to their adjunct-supplied destination without waiting for call acceptance.

Validation for an adjunct-supplied call route

When the switch receives adjunct-supplied call route instructions, the switch validates the route according to the following process:

1. The switch verifies that the COR rules specified for the target VDN permit the call to be terminated at the adjunct-supplied destination.
2. The switch validates the following information:
   - Destination number
   - ACD split
   - TAC/AAR/ARS access code
   - Dial plan compatibility
   - Other options specified by the adjunct
3. If the ASAI adjunct specifies the direct agent call option, the destination number (agent) must be logged into the adjunct-specified ACD split.
4. If the destination for the call is external, the switch verifies that a trunk is available for the call.

Switch response to validated adjunct-supplied call routes

If the switch validates an adjunct-supplied call route, the following operations occur:

1. Vector processing in the VDN that contains the initiating **`adjunct routing link`** command terminates immediately.
2. The switch signals the ASAI adjunct that the route is accepted.
3. The switch routes the call to the destination specified by the ASAI adjunct.

Switch response to invalid adjunct-supplied call routes

If any of requirements for call route validation listed in Validation requirements for an adjunct-supplied call route are not met, items the following operations occur:

1. The switch discards the route.
2. The switch signals the ASAI adjunct that the route is invalid.
3. Vector processing of any other default treatment steps in the VDN that contains the initiating adjunct routing link proceeds.

Data sent with an ASAI call route request

When a call encounters an **`adjunct routing link`** command and if the call is not queued to a split, the switch sends an ASAI message that requests a call route over the specified adjunct link. The following list identifies the contents of the message, along with a comment or a brief explanation for each item:

Calling number information

The calling party number or billing number (CPN/BN) that is provided by ISDN-PRI or R2-MFC signaling facilities. If the call originates from a local switch extension, this extension is the calling number.

Originating line information (II-digits)

A two-digit code that is provided by ISDN-PRI facilities that indicates the type of originating line. This information is not provided by SIP facilities.

Called number

The originally called extension if a call is forwarded to a VDN, or the first VDN through which the call was routed if the call was not forwarded to the VDN.

If the VDN Override for the Trunk ASAI Messages feature is in effect for an incoming call, the active VDN extension (instead of the Called Number received in the SETUP or INVITE message) is sent in the Called Number for the Call Offered, Alerting, Queued, Connect, and Adjunct Route-Request ASAI Event Reports.

Routing VDN

The last VDN that routed the call to the vector that contains the **`adjunct routing link`** command.

Call identifier

An ASAI identifier that permits the ASAI adjunct to track multiple calls by either Event Notification or 3rd Party Call Control. For more information on ASAI, see *Communication Manager CallVisor ASAI Technical Reference*.

Enhanced Information Forwarding (related data) and Look-Ahead Interflow information (if any)

Includes the original VDN display information, the priority level of the call at the originating switch, and the time that the call entered vector processing.

Digits collected by Call Prompting or Caller Information Forwarding (CINFO) (if any, maximum of 16 digits)

Digits that are collected by the most recent **`collect digits`** command. For more information, see “Call Prompting”, “ANI /II-digits routing” and “Caller Information Forwarding (CINFO)” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

User-to-User Information (UUI)

User-provided data that is associated with the call. If provided by ASAI, this data was provided in a 3rd-Party-Make-Call, Auto-Dial, or Route-Select message. If provided over an ISDN or SIP trunk, the data was in the SETUP or INVITE message that delivered the call to this switch. Calls that contain UUI specifically used by ASAI allow ASAI UUI to be propagated to the new call during a manual transfer or conference operation. ASAI UUI is propagated to a new call during its establishment when the agent presses the transfer/conference button the first time. If the call is transferred to a remote switch, the ASAI UUI from the first call is copied into the SETUP or INVITE message sent for the second call, in which case, the alerting event message sent to an ASAI application contains the ASAI information.

Special vector processing considerations associated with adjunct routing

When you design call vectors that include more than one **adjunct routing link** command, you must be aware of a number of special operational features.

Effects of ASAI link/application failure on vector processing

An ASAI link failure can change the manner in which subsequent announcement or wait-time treatment steps are processed.

In the following simplified vector example, the step that follows immediately after an **adjunct routing link** command is a **wait-time** command. If the adjunct routing link step fails at either the ASAI link or adjunct application, the wait-time step is skipped.

The second step after the adjunct routing link step is often implemented as a default treatment. In the example shown above, the default treatment in step 3 is a route to an attendant. If the switch recognizes that the ASAI link or adjunct application is out of service, this step executes immediately. Otherwise, the step executes only if the application does not respond with a route within 60 seconds (the wait-time assigned in the example).

Simplified example of vector processing in an ASAI link/application failure condition

```
1. adjunct routing link 11  [link/application is down]
2. wait-time 60 seconds hearing ringback  [step is skipped]
3. route-to number 0 with cov n if unconditionally [step is executed]
4. disconnect after announcement 2000
```

Vector processing with goto steps in an ASAI link or application failure condition

Processing rules for a vector that includes more than one **adjunct routing link** command and has an ASAI link or application failure condition in effect summarized as follows:

- An announcement or wait time treatment is skipped when one of the following conditions is true:
  - The treatment step follows immediately after a failed **adjunct routing link** command.
  - The treatment step is the first non-goto step that follows a **goto** step that succeeds. In this context, a **goto** step is treated as a success when the specified goto condition is true and the call branches from the **goto** step to the treatment step.
  - The treatment step is the first non-goto step that follows a failed **goto** step. In this context, a **goto** step is treated as a failure when the specified **goto** condition is true, the call fails to branch and control proceeds to the treatment, which is the next step listed in the vector sequence.

Note:

The treatment step is skipped even when a failed **goto** step that precedes the treatment is, in turn, preceded by more than one successful **goto** steps.

The rules listed for vector processing under ASAI link or application failure conditions are illustrated in the following examples.

Example 1 - Vector processing with goto steps in an ASAI link or application failure condition

```
VDN (extension=1040  name=''Ad Route''  vector=40)
Vector 40
 1. adjunct routing link 10  [link/application is down]
 2. wait-time 10 seconds hearing ringback [step is skipped]
 3. adjunct routing link 20 [link/application is down]
 4. goto step 7 if available-agents in split 20 < 1 [step executes and condition is
false]
 5. wait-time 10 seconds hearing ringback [step is skipped]
 6. goto vector 50 @step 1 if unconditionally  [step executes, go to vector 50]
 7. goto step 10 if calls-queued in split 20 pri l > 50
 8. announcement 4001
 9. goto vector 50 @step 1 if unconditionally
10. route-to number 6000 with cov n if unconditionally
   VDN (extension=6000  name=''Message''  vector=60)
```

Based on the scenario presented in the example shown above, the following vector processing events occur:

Step 1 fails: In the example, the adjunct link or application is out of service. The **adjunct routing link** command in step 1 fails.

Step 2 is skipped: Because the **wait-time** command in step 2 immediately follows an **adjunct routing link** command whose adjunct link is out of service, the **wait-time** step is skipped.

Step 3 fails: For purposes of this example, step 3 contains another **adjunct routing link** command whose adjunct link is out of service. The step fails and control is passed to the **goto step** command in step 4.

Step 4 executes: A **goto** step that immediately follows a failed **adjunct routing link** command is always executed. In this example, the command fails to branch because there is a minimum of one available agent in split 20.

Step 5 is skipped: The **wait-time** step that follows the unsuccessful **goto** step (step 4) is skipped, because in an ASAI link failure condition, the first non-goto step to be processed after the first successful first **goto** step is always skipped if the step is either an **announcement** or a **wait-time** step. Control is passed to the **goto vector** command in step 6.

Step 6 executes: Step 6 routes the call to vector 50 (not shown), which is designed to queue the call and provide standard call treatment.

In the next example, the **goto** step succeeds when the specified condition is true, that is, no agents are available in split 20, and control is passed to step 7, where another **goto** step determines whether there are more than 50 calls in split 20. If the condition is true, step 7 succeeds and control is sent to step 10, where the **route-to number** command sends the call to vector 60.

The example processing events are described as follows:

Example 2 - Vector processing with goto steps in an ASAI link or application failure condition

```
VDN (extension=1040  name=''Ad Route''  vector=40)
Vector 40
 1. adjunct routing link 10 [link/application is down]
```

```
 2. wait-time 10 seconds hearing ringback [step is skipped]
 3. adjunct routing link 20 [link/application is down]
 4. goto step 7 if available-agents in split 20 < 1 [step executes and condition is
true]
 5. wait-time 10 seconds hearing ringback
 6. goto vector 50 if unconditionally
 7. goto step 10 if calls-queued in split 20 pri l > 50 [step executes and condition is
true]
 8. announcement 4001
 9. goto vector 50 if unconditionally
10. route-to number 6000 with cov n if unconditionally [step executes unconditionally]

VDN (extension=6000  name=''Message''  vector=60)
Vector 60

1. announcement 4000 ["We're sorry. We are still unable to connect you to an agent. If
you would like to leave a
   message, please do so after the tone."]
2. wait-time 6 seconds hearing silence
3. messaging split 18 for extension 1500
4. announcement 4010 ["We're sorry. We are unable to connect you to our voice mail. If
you would like to try to
   leave a message again, please do so after the tone. Otherwise, call back weekdays
between  8:00 A.M. and 5:00 P.M."]
5. goto step 2 if unconditionally
```

Based on the scenario presented in the example, the following vector processing events occur:

Step 1 fails: For purposes of this example, the adjunct link or application is out of service. The **adjunct routing link** command in step 1 fails.

Step 2 is skipped: Because the **wait-time** command in step 2 immediately follows an **adjunct routing link** command whose adjunct link is out of service, the **wait-time** step is skipped.

Step 3 fails: For purposes of this example, step 3 contains another **adjunct routing link** command whose adjunct link or application is also out of service. The step fails and control is passed to the **goto step** command in step 4.

Step 4 executes: A **goto** step that follows a failed **adjunct routing link** command is always executed. In this example, the command succeeds and branches to step 7, because no agents are available in split 20.

Step 7 executes: Again, a **goto** step that follows a failed **adjunct routing link** command is always executed. In this example, the command branches unconditionally to vector 60

Step 10 executes: In this example, step 10 (route-to number) is the first non-goto step immediately preceded by more than one **goto** step in an ASAI link fail condition. The step executes, because the step is not an **announcement** or **wait time** command.

Vector 60: Step 1 executes: The first step in this vector is an **announcement** command. In this example, this is the first step in the processing sequence to be either an **announcement** or **wait-time** step. However, this step is not skipped, since the step is not the first non-go to step in the processing sequence. Instead, step 10 in vector 40 (a route-to number step) is the first non-goto step.

Simultaneous processing of vector steps and ASAI call route requests

When Communication Manager sends a route request to an ASAI adjunct, vector processing continues for any vector steps that follow the **adjunct routing link** command. Therefore,

non-adjunct routing link step that follows immediately after an adjunct routing link step or multiple adjunct routing link steps in uninterrupted succession can determine:

- The maximum length of time that Communication Manager waits for a call route reply from the ASAI adjunct.
- In some cases, whether or not the ASAI call route request is allowed to finish processing.

If the next step is not a **wait-time**, **announcement**, or another **adjunct routing link** command, as much as six minutes can elapse before Communication Manager determines that the adjunct application is out of service. For this reason, design vectors so that the next step to follow an **adjunct routing link** command is either a **wait-time**, or **announcement** command.

Vector steps that terminate an ASAI call route request

If an adjunct routing link step is followed by a wait-time or announcement treatment and the treatment completes before an ASAI call route request is returned by the adjunct, call processing continues for any vector steps that follow the treatment. In this case, certain vector commands terminate the ASAI call route request after being executed. Vector commands that terminate an active ASAI call route request include:

- **busy**
- **check split**
- **converse-on split**
- **queue-to split**
- **collect digits**
- **disconnect**
- **messaging split**
- **route-to**

If Communication Manager receives a valid ASAI call route message before one of the listed vector commands can execute, Communication Manager routes the call to the destination specified by the adjunct route. Otherwise, the ASAI route request is terminated.

Note:

The adjunct can also reject a call request by negatively acknowledging the route request sent by Communication Manager. When Communication Manager receives a route request rejection message from the adjunct, Communication Manager terminates any announcement or wait-time step being executed. Call processing then continues with the next vector step.

Adjunct routing-initiated path replacement

You can use QSIG or DCS with Reroute using ISDN SSE for path replacement for calls in a queue. For calls that are waiting in a queue or in vector processing, even if the call is not connected to an answering user, Communication Manager can attempt a path replacement to find the optimum path for the call.

When adjunct routing is used with a call, path replacement can be initiated when the following criteria are true:

- The inbound call is over a QSIG trunk or DCS SSE trunk.
- A route select response is received from the CTI application after the **adjunct route** vector command has been executed.
- The routing destination that is contained in the route select ASAI message is to an outbound QSIG trunk or out bound DCS SSE trunk.

When all three criteria are met, the trunk is seized and used for the call.

Example vector for adjunct routing-implemented path replacement

The following Call Vector example shows how a vector for adjunct routing can be written to trigger path-replacement at the terminating switch.

Note:

In order for a path-replacement to be attempted, the incoming and outgoing trunks that are used for the call must be administered with the **Supplementary Service Protocol** field set to `b`.

Adjunct routing-initiated path-replacement vector

```
1. announcement 5996
2. adjunct routing link 11
3. wait 20 seconds hearing ringback
4. announcement 3111
```

At the terminating (receiving) switch, the vector that is executed by the incoming call must be programmed with an **announcement**, **wait hearing music**, or **wait hearing ringback** vector command. The use of one of these commands is what makes it possible for path-replacement to take place while the call is in vector processing.

Phantom calls

A phantom call originates from a nonphysical device by way of an ASAI application and can be placed anywhere. In general, phantom calls

- Use less resources.
- Are treated as voice calls.

How do phantom calls work?

First, an application requests a phantom call by sending an ASAI third_party_make_call or auto_dial capability message to Communication Manager.

If the specific extension of a station Administration Without Hardware (AWOH) is specified as the originator, Communication Manager places the call from that extension if the extension is available.

You can specify a hunt group extension with members that are AWOH extensions as the originator.

How are phantom calls used?

Applications use phantom calls to originate a call without using a physical device. For example, applications require to:

Action	Description
Reserve a queue slot	Many call centers handle incoming requests as voice, video, data, voice messages, faxes, and e-mail. Agents handle the mix of requests. However, a single queue must manage and distribute the work load for the agents. For each non-voice request, the application can place a phantom call into the queue. When the phantom call reaches the head of the queue, the call is delivered to the agent. The agent is then given the corresponding work item on the desktop, for example, the fax.
Conference control	Multiple parties, both internal and external, can be conferenced into a call. The initial call is placed as a phantom call. When answered, the call is placed on hold by the application and another phantom call is made. The two calls are then conferenced together. This process is repeated until all parties are added to the call.
Help with trunk-to-trunk transfers	Working with the Single Step Conference (SSC) feature, applications can use the phantom call feature to help with trunk-to-trunk transfers, that is, transferring a trunk-to-trunk call to another trunk.
Alerts (wake-up, maintenance, and security)	Applications can use phantom calls to alert users of various conditions such as wake-up, maintenance, or security.

How do phantom calls affect Call Vectoring?

Because phantom calls can be directed anywhere, you must properly configure the application and Communication Manager to ensure that the vector commands that are executed for the calls make sense. For more information, see *Communication Manager CallVisor ASAI Technical Reference*.

Communication Manager does not block phantom calls from executing any vector commands because phantom calls follow the same vector processing as regular voice calls. However, do not allow phantom calls to enter certain vector steps such as the following.

Steps	Description
Announcements	Because there is nobody listening to an announcement that is made to a phantom call, there is no sense in playing one.
Collect steps	In a phantom call, the **collect** step fails because the step cannot connect a tone receiver to a station Administration Without Hardware (AWOH). The step times out because there is nobody to enter the expected digits. The **busy** step provides a busy signal to the caller. In a phantom call, the **busy** step disconnects the call because Communication Manager clears a phantom call when the call cannot terminate at a specific local destination.

Phantom call administration

There are no administration screens that are specific to phantom calls, but the following criteria must be met in order for the feature to work:

- Some stations AWOH must be administered.
- If a hunt group is specified as originator, a non-ACD hunt group with AWOH members must also be administered.
- It is recommended that meaningful names are assigned for the stations AWOH that are used by phantom calls if the calling party name will appear on the agent's or Service Observer's display.

Single-step conference

With Single-Step Conference (SSC), an application can:

- Add a device into an existing call, for example, to play announcements or make voice recordings.
- Facilitate application-initiated transfers and conferences.

Stations that are AWOH can use SSC. The party can be added to a call in the listen only mode with no visibility or in the listen/talk mode with visibility.

SSC is only available through an ASAI link.

How does SSC work with Call Vectoring?

The call to which an extension is to be single-step conferenced is not allowed to be in vector processing unless the visibility option with the single-step conference request indicates no visibility.

To be transferred to a VDN, a conference call must not have more than two parties.

Note:

Invisible (listen-only) single-step-conference parties are not counted in the two-party limit for a conference call transfer to a VDN.

Multiple outstanding route requests

This feature allows multiple ASAI route requests for the same call to be simultaneously active. The route requests can be over the same or over different ASAI links.

Route requests are all made from the same vector. The requests must be specified without intermediate **wait-time**, **announcement**, **goto**, or **stop** steps. If the **adjunct routing link** commands are not specified back-to-back, standard adjunct routing functionality applies and previous outstanding route requests are cancelled when an **adjunct routing link** vector step is executed.

The first route select response that is received by Communication Manager is used as the route for the call and all other outstanding route requests for the call are canceled.

With multiple outstanding route requests, multiple adjuncts can process the route call request without waiting for the first route attempt to fail. An application can make use of this feature to distribute the incoming call load evenly across adjuncts based on the adjunct's current CPU load.

Note:

Each link has a unique extension number, even in a configuration where there can be multiple links to the same adjunct.

Multiple call route request example

The following example shows a typical vector where multiple adjunct route requests to multiple links are active at the same time. The first adjunct to route the call is the active adjunct and specifies which VDN the call must be routed to at that point.

Sample adjunct routing link vector with redundancy

```
1. wait-time 0 seconds hearing ringback
2. adjunct routing link 11
3. adjunct routing link 12
4. adjunct routing link 13
5. wait-time 6 seconds hearing ringback
6. route-to number 1847 with cov n if unconditionally
```

Advanced Vector Routing features

You can use the following advanced vector routing features to enhance the conditional routing capabilities of basic call vectoring, and to improve the operational efficiency of a call center.

Rolling Average Speed of Answer (ASA)

You can use Rolling ASA Routing to make routing decisions based on the current average time a skill or VDN takes to answer a call. This ensures that the vectors route calls to the skill or to the VDN with the shortest average time.

Expected Wait Time (EWT)

You can use EWT Routing to make routing decisions based on the wait time for a call or split, in queue. EWT can be passed to an Interactive Voice Response (IVR) or a Voice Response Unit (VRU) to notify the caller about the expected time in queue.

VDN Calls

You can use VDN Calls Routing to make routing decisions based on the number of incoming trunk calls currently active in a VDN. With the VDN calls conditional, you can use a vector to limit the number of simultaneous calls to a particular VDN. For example, if you enter into a contract with a service agency to handle 100 simultaneous calls for a client, you can route calls in excess of 100 to a **busy** step.

Advanced Vector Routing command set

The commands used in Advanced Vector Routing are listed in the following table.

Command category	Action taken	Command
Routing		
	Queue the call to a backup ACD split.	**check split**
Branching/programming		
	Go to a vector step. Go to another vector.	**goto step** **goto vector**

When to use wait time predictions

A number of factors can affect the accuracy of wait time predictions. Wait time predictions are best suited for medium-volume or high-volume call scenarios. The potential accuracy of a wait time predictor increases as the rate of removal from queue increases.

Under all conditions, EWT is the most accurate wait time predictor, but EWT is most accurate when the rate of removal from queue at a given split priority level is a minimum of one call every 30 seconds.

Predictions can be made for a split with multiple priority levels as long as the majority of calls are delivered to lower priority levels. If the majority of calls are queued at the higher-priority levels, any predictions made for the lower-priority levels not be accurate.

The following circumstances can limit the accuracy of the wait time predictions.

Factors limiting wait time prediction accuracy	Description
System restart or new split administration	The EWT algorithm uses a combination of historical and real-time information to make predictions. When no historical information exists, such as when a new split is added or a reset system 3 or 4 is completed, there is the potential for inaccuracies. To prevent inaccurate predictions when there is no historical information, administer the **Expected Call Handling Time** field on the Hunt Group screen. The value in this field is then used in place of the missing historical data. If the value of this field does not accurately reflect the call handling times of the split, EWT predictions can be inaccurate until some call history is generated. The algorithm normally requires about 30 queued calls to be answered from a split priority level before it reaches its maximum accuracy. You can change the value in the **Expected Call Handling Time** field by executing a **change hunt group** command. Changing the value does not disrupt EWT predictions by overwriting EWT history. The value is stored and used the next time a reset system 3 or 4 is executed.

Table continues…

Factors limiting wait time prediction accuracy	Description
Low call volume applications	Split priority levels where the rate of removal from the queue is very low can only be predicted with limited accuracy.
Sites with frequent staffing changes	Although EWT immediately adjusts for all types of staffing changes, since predictions can already be made for calls that are waiting in queue, the past predictions were based on staffing information which is now out of date. Therefore, the EWT in scenarios where large staffing changes are continually happening can only be predicted with limited accuracy.
Staffed agents who rarely answer calls to a split	The EWT algorithm takes account of agents in multiple splits in its calculation. However, suppose there are many agents who are assigned to a split but spend most of their time answering calls in their other splits. If a large number of these agents are moved to or from the split, the EWT for this split be temporarily inaccurate until it adjusts to those changes.
Applications with widely varying call handling times	If the majority of calls to a split are handled within a narrow range of times, the accuracy of any predictor will be much greater than that for a split where call handling times are widely different.

Expected Wait Time (EWT)

With EWT routing, you can make routing decisions based on the time that a caller can expect to wait in queue.

How EWT is calculated

Depending on how the EWT condition is used in a vector step, the predicted wait time calculation is derived by the following rules:

- If the call is currently queued to a split, the EWT is based on the actual current position of the call in the queue at a particular priority level and the rate of service of calls from the queue at that priority level.
- If the call is not yet queued to a split, the EWT is based on the assumption that the call is placed at the end of the queue.

EWT also adjusts for many other factors such as multiple split queuing, call handling times, and the impact of direct agent calls on the wait time of other calls to the split. The algorithm adjusts EWT immediately for changes in staffing, such as agents logging in or taking breaks in the AUX work mode.

The EWT can also be passed to a VRU so that a caller can be notified of the expected time in queue. The expected-wait condition can be used with either the **goto** or **check** commands.

Call Vectoring offers several conditionals to estimate predicted wait time on a queue, including EWT, rolling ASA and Oldest Call Waiting (OCW), but EWT uses the most accurate method of prediction. EWT checks the real-time and historical information, such as priority level, position in queue, and number of working agents.

EWT is responsive to changing call center conditions. For example, EWT adjusts instantly to any staffing changes in the split, or if agents moves in or out of auxiliary work mode, the wait time predictions immediately adjust.

EWT does not include the time in a call vector before the call enters a queue. It also does not include the time that the call rings at a telephone after it is removed from the queue.

EWT for a split

The EWT for a split is the time that an incoming call is expected to remain in queue if the call is queued to the split at the specified priority level. EWT is used to determine if a call must be queued to the split.

The following vector example shows how to use EWT to determine if a call must be queued to a split.

```
1. goto step 3 if expected-wait for split 1 pri l < 600
2. busy
3. queue-to split 1 pri l
4. announcement 3001
5. wait-time 998 secs hearing music
```

In the example shown above, the following wait time conditions are possible:

- If there are agents available, EWT is zero.
- EWT is infinite if:
  - There are no logged-in agents.
  - All logged-in agents are in the AUX work mode.
  - The split queue is full.
  - There is no split queue and all agents are busy.
  - The split queue is locked. This occurs when the last working agent in a non-vector-controlled split attempts to enter the AUX work mode.

EWT for a call

EWT for a call is the remaining time that a caller can expect to wait before the call is serviced from the queue. If the call is queued to multiple splits, the remaining queue time for each split is calculated, and the shortest calculation is used as the EWT.

A call must be queued to a minimum of one split to have an expected wait time. If the call is not queued, or if the call is queued to splits that are not staffed, the EWT value is infinite.

The following vector example shows how to use EWT to determine call treatment:

```
1. queue-to split 1 pri m
2. check split 2 pri m if expected-wait < 30
3. goto step 5 if expected-wait for call < 9999
4. busy
5. announcement 3001
6. wait-time 998 secs hearing music
```

Passing EWT to a VRU

The Expected Wait Time (EWT) for a call can be passed to a Voice Response Unit (VRU) to inform callers about the expected time in queue. EWT is passed to the VRU with the **converse-on** command as wait data. The value that is outpulsed to the VRU is the expected wait time of the call in seconds. The VRU then converts the seconds to a spoken message. The expected wait is calculated after the VRU port answers the call, so queuing to a converse split does not adversely impact the EWT value that is passed to the VRU.

No zero padding is added to the wait time that is passed to the VRU. If the EWT for the call is 128 seconds, the digits 1, 2, and 8 are outpulsed. If the EWT is 5 seconds, the digit 5 is outpulsed.

The wait time passed to the VRU is the most accurate prediction possible. On an average, 50 percent of the time the actual wait time is shorter and 50 percent of the time it is longer. Configure the VRU applications to make an upward adjustment of the prediction so that the majority of callers receive a predicted wait time that is either equal to or greater than the actual wait time.

The VRU can also announce EWT at set intervals while the call is in queue, but use this strategy with caution. Circumstances such as a reduction in the number of agents or a sudden influx of higher priority calls can cause the EWT of the caller to increase from one announcement to the next.

If the call is not queued or is queued only to splits that are unstaffed, or to splits where all agents are in the AUX work mode, the end-of-string character # is the only data item that is outpulsed to the VRU.

The following vector example illustrates routing based on the predicted split wait time and passing wait data to the VRU. The caller hears the wait time only if the caller is expected to wait for more than 60 seconds in queue. Callers who have to wait more than 10 minutes are requested to call back later.

```
1. goto step 3 if expected-wait for split 32 pri l < 600
2. disconnect after announcement 13976
3. queue-to split 32 pri l
4. wait-time 20 secs hearing ringback
5. goto step 7 if expected-wait for call < 40
6. converse-on split 80 pri l passing wait and none
7. announcement 11000
8. wait-time 60 secs hearing music
9. goto step 7 if unconditionally
```

Calls that have predicted wait times greater than 10 minutes fail step 1 and are disconnected after an announcement. If the expected wait time is less than 10 minutes step 1 routes the call to step 3 where it is queued to split 32 and waits 20 seconds hearing ringback. After 20 seconds if the expected wait time for the call is less than 40 seconds, step 5 routes the call to an announcement followed by a wait with music. If the expected wait time for the call is equal to or greater than 40 seconds, step 6 informs the caller of the amount of time the caller can expect to wait before the call is answered.

Notifying callers of wait time without a VRU

You can use EWT to notify callers of their expected wait time without a VRU. This can be done using recorded announcements and by associating each recorded announcement with a time band, as shown in the following example.

```
VECTOR 101
1. queue-to split 3 pri h
2. goto step 4 if expected-wait for call <= 600
3. busy
4. wait-time 12 seconds hearing ringback
5. announcement 3001 [Thank you for calling ABC Inc. All agents are busy, please wait
and we will get to your call as soon as possible]
6. goto vector 202 if unconditionally

_ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
VECTOR 202
1. goto step 13 if expected-wait for call > 280
2. goto step 11 if expected-wait for call > 165
3. goto step 9 if expected-wait for call > 110
4. goto step 7 if expected-wait for call > 55
5. announcement 3501 [Thank you for waiting. Your call will be answered within the next
minute.]
6. goto step 14 if unconditionally
7. announcement 3502 [Thank you for waiting. Your call will be answered within
approximately one to two minutes.]
8. goto step 14 if unconditionally
9.  announcement 3503 [Thank you for waiting. Your call will be answered within
approximately two to three minutes.]
10. goto step 14 if unconditionally
11. announcement 3504 [Thank you for waiting. Your call will be answered within
approximately three to five minutes.]
12. goto step 14 if unconditionally
13. announcement 3505 [We apologize for the delay. Due to heavy call volume, you may
have to wait longer than five minutes
    to speak to a representative. If possible, we suggest that you call between the
hours of 8am and 10am for the
    fastest service.]
14. wait-time 120 secs hearing music
15. goto step 1 if unconditionally
```

In step 1 of the example shown above, the call is queued to split 3 at high priority. If the call fails to get a queue slot in split 3, if split 3 has no working agents, or if the wait time in split 3 at high priority exceeds 10 minutes, step 2 fails and the caller receives a busy signal. If step 2 succeeds, the caller hears ringback and an announcement and is then sent to vector 202.

Steps 1 through 4 of vector 202 determine tests to determine a predicted time band interval for the remaining queuing time for the call. One of five recorded announcements is then played to provide the caller with an expected wait time.

You can program vectors so that few callers experience wait times that exceed the wait time of the announcement. In the example shown above, the EWT thresholds are set lower than the times that are quoted in the recorded announcements.

Using EWT to route to the best split

You can use EWT to change the normal queuing strategy for multiple splits to ensure that calls are answered in the shortest possible time. However, this strategy uses additional system resources and can make reading and analyzing split reports difficult. Alternately, you can use EWT to identify the best split for each call and not use multiple split queuing.

The following vector example shows a scenario that includes a main split (1) and a backup split (2). In this example, the preference is for an agent from the main split service the call, but a 30-second maximum wait time is a competing preference.

The strategy in this vector is to use the backup split only if the backup split can answer the call within 30 seconds and the main split cannot.

```
1. goto step 5 if expected-wait for split 1 pri m <= 30
2. goto step 5 if expected-wait for split 2 pri m > 30
3. check split 2 pri m if unconditionally
4. goto step 6 if unconditionally
5. queue-to split 1 pri m
6. wait-time 12 secs hearing ringback
7. announcement 3501
8. converse-on split 18 pri m passing wait and none
9. wait-time 120 secs hearing music
10. goto step 8 if unconditionally
```

In the example, step 1 branches to step 5 if the main split can answer the call within 30 seconds. If the main split cannot answer the call within 30 seconds, step 2 checks to see if the backup split can answer the call within 30 seconds. If the test fails, the call branches to step 5 and is queued to the main split. If possible, the call is queued to the backup split in step 3. At this point, the call is queued either to the main split or to the backup split, but not to both.

Steps 6 through 10 provide audible feedback to the caller while the call is in the queue. Note that in step 8, which is executed every 2 minutes, a VRU is used to provide the caller with the remaining wait time.

Factors that increase EWT for a split priority level

The most common causes for an increase in EWT for a split priority level are:

- The number of calls that are in queue increases.
- Agents log out.
- Agents go on break or are otherwise in the AUX work mode.
- Agents are moved to another split.
- Agents with multiple splits answer an increasing number of calls in other splits.

Following are the other conditions that can increase the EWT for a split priority level:

- The average talk time increases.
- The number of calls at a higher priority increases.
- The number of direct agent calls increases.
- The number of RONA calls increases.
- The number of abandoned calls decreases.
- The number of calls that are queued in this split but answered in another decreases.

Factors that decrease EWT for a split priority level

The most common causes for a decrease in EWT for a split priority level are:

- The number of calls in queue decreases.
- Agents log in and start answering calls.
- Agents return from break or otherwise are no longer in the AUX work mode.
- Agents are moved from another split.
- Agents with multiple splits answer fewer calls in other splits.

The following conditions can reduce the EWT for a split priority level:

- The average talk time decreases.
- The number of calls at higher priority decreases.
- The number of direct agent calls decreases.
- The number of RONA calls decreases.
- The number of abandoned calls increases.
- The number of calls queued in this split but answered in another increases.

Rolling Average Speed of Answer (ASA)

Rolling ASA Routing helps you to make routing decisions that are based on the current average time that it takes for a call to be answered in a split or VDN. In this way, a vector can route a call to the VDN or split where it is likely to be answered most quickly.

Rolling ASA versus interval ASA

The ASA calculation used for vector routing is called “Rolling ASA” to differentiate it from the “Interval ASA” that is recorded in Basic Call Management System (BCMS) and Avaya Call Management System (CMS) reports.

Rolling ASA is a running calculation that does not take into account the 15-minute, half-hour, or hour reporting intervals. Rolling ASA does not reflect interval boundaries.

The Interval ASA used for BCMS and CMS reports is calculated on reporting interval boundaries and clears to zero at the start of each reporting interval.

The Rolling ASA for a split or VDN is calculated based on the speed of answer for all calls recorded since system start up and is recalculated every time a call is answered. During each calculation, the speed of answer for the current call is given a weighted value that is greater than previous calls. Approximately 95 percent of the value of Rolling ASA is retrieved from the previous ten calls.

Note:

Calls that are not answered, such as calls that receive a forced busy, are not included in the Rolling ASA calculation.

The Rolling ASA is calculated for an entire split or VDN. The calculation does not include the priority levels of answered calls.

When to use rolling ASA

Use Rolling ASA to test whether vector processing queues the call to additional splits or skills when the main split or skill does not currently meet the targeted threshold.

Do not use Rolling ASA conditionals to prevent calls from queuing to the main split or skill, or being answered in the principal VDN. If no calls are being answered in the main split or skill or VDN, the value of rolling ASA does not change. This can result in all future calls being locked out of the main split or skill, or VDN unless there are other call vectors in the system directing calls.

Important:

To implement a call flow that tests whether or not to queue a call to a main split or skill, use the EWT feature.

Rolling ASA split calculation

The rolling ASA for a split is the average call answer time, as specified by the time interval that starts when call processing attempts termination to a split, and ends when the call is answered in that split. The measured interval includes both time in queue and ringing time at the agent station.

If the call is answered in another split or the call is abandoned by the caller, rolling ASA is not recorded for the call. If a call flows into a split from another split, the time queued and ring time for the previous split are not included. If a call is queued in multiple splits, only the rolling ASA for the split in which the call is answered is measured.

Rolling ASA VDN Calculation

The rolling ASA for a VDN is the average call answer time, as specified by the time interval that starts when call processing is initiated within the VDN until it is answered. The measured interval includes:

- Time elapsed in vector processing, including time in announcements.
- If the call is answered by an agent, time in queue and time ringing at the agent station.

Note:

If a call flows between VDNs, only the time elapsed in the answering VDN is used in the calculation.

Rules for specifying VDNs

Rolling ASA follows the rules used for other Advanced Vector Routing conditionals to specify a VDN in a goto step:

- A VDN number.
- The value designated as *latest*. The latest VDN is the VDN currently processing the call. The latest VDN is not affected by VDN override settings.
- The value *active*. The active VDN is the VDN of record, which is the called VDN as modified by override rules. For example, if a call routes from a VDN with override set to *yes* then the

new VDN is the active VDN. If a call routes from a VDN with override set to *no*, the previous VDN is the active VDN.

Combining VDN and ASA routing example

The following vector example combines VDN and split ASA routing.

```
1. queue-to split 10 pri h
2. goto step 6 if rolling-asa for split 10 <= 30
3. check split 11 pri h if rolling-asa <= 30
4. check split 12 pri h if rolling-asa <= 30
5. check split 13 pri h if rolling-asa <= 30
6. announcement 10000
7. wait-time 40 secs hearing music
8. goto step 3 if unconditionally
```

Step 1 queues the call to the main split. If the main split is currently answering calls within the target time of 30 seconds, step 2 bypasses all of the backup splits and goes directly to the announcement in step 6. The assumption is that the call will be handled by split 10 within the time constraints. However, if the call is not answered by the time that vector processing reaches step 8, the backup splits are checked.

If the rolling ASA for the main split is greater than 30 seconds, steps 3, 4, and 5 check the backup splits. The call is queued to any of these splits that have a rolling ASA of 30 seconds or less. If the call still is not answered by the time that vector processing reaches step 8, the backup splits are checked again.

VDN Calls

VDN Calls routing allows you use the counted-calls conditional to make routing decisions on the number of incoming trunk calls that are currently active in a VDN.

How VDN Call counts are calculated

The counted-calls conditional allows a vector to limit the number of simultaneous calls directed to a particular VDN. For example, if a service agency is contracted to handle 100 simultaneous calls for a client, calls in excess of that number can be routed to a busy step.

VDN Call counts follows the rules used for other Advanced Vector Routing conditionals to specify the VDN in a goto step:

- A VDN number.
- The value designated as latest. The latest VDN is the VDN currently processing the call. The latest VDN is not affected by VDN override settings.
- The value active. The active VDN is the VDN of record, which is the called VDN as modified by override rules. For example, if a call routes from a VDN with override set to **yes** then the new VDN is the active VDN. If a call routes from a VDN with override set to **no**, the previous VDN is the active VDN.

When Advanced Vector Routing is enabled, a count of active incoming trunk calls is kept for each VDN. The VDN counter increments each time that an incoming call is placed to the VDN and

decremented each time that an incoming call is released. A call is treated as active in a VDN from the time the call routes to the VDN until all parties on the call are dropped and the call is released.

Note:

The call is counted for the originally called VDN only. When a call is routed to another VDN, the call counter for the subsequent VDN does not increment, nor does the call counter for the original VDN decrement.

The VDN Call count includes the following types of calls:

- Incoming trunk calls routed directly to the VDN.
- Incoming trunk night service calls in which the VDN is the night service destination.
- Calls that cover or forward to the VDN if it is the first VDN routed to and the call is an incoming trunk call.
- Already counted calls that are conferenced with counted or not counted calls from the same VDN.

The VDN call count does not include:

- Internal calls to the VDN.
- Calls that are transferred to the VDN, unless they are a SIP transfer and the **Suppress Target VDN counted-calls?** field is set to `n`.
- Calls that are redirected to their VDN return destination.
- Conferenced calls that were previously counted on different VDNs.

Using the counted-calls conditional example

The following vector example shows how the counted-calls conditional can be used to route calls.

Using VDN call counting to route calls

```
1. goto step 3 if counted-calls to vdn 1234 <= 100
2. busy
3. queue-to split 60 pri l
4. wait-time 20 seconds hearing ringback
5. announcement 27000
6. wait-time 60 seconds hearing music
7. goto step 5 unconditionally
```

If more than 100 calls are active in VDN 1234, the caller hears a busy signal and vector processing is terminated. If 100 or fewer calls are active, the call queues to split 60.

Attendant Vectoring

With Attendant Vectoring, you can use a set of commands to write call vectors for calls to be routed in non-call center environments. When you set the **Attendant Vectoring** field to `y`, all attendant-seeking or dial 0 calls are processed using call vectors, and not by the normal attendant console call routing.

Attendant Vectoring is available in non distributed attendant environments and distributed attendant environments for IAS and QSIG CAS.

The main reason to use Attendant Vectoring is to allow flexible routing of attendant-seeking calls. If users are instructed to dial an attendant VDN, the call can be answered by an attendant, but the call can also be covered to the voice mailbox of a night station.

If you use Attendant Vectoring and Night Service to route calls to a voice mail system, you can also use Automatic Message Waiting (AMW) to notify after-hours personnel about messages in the night service station mailbox by assigning an AMW lamp on the backup telephones. When personnel see new messages, the personnel can check the messages and respond accordingly.

Attendant queue

If attendant vectoring results in putting a call in the attendant queue, it is placed in queue with the priority as administered on the console parameter screen. There are no changes made to the attendant priority queue for attendant vectoring. Even when partitioning is turned on and multiple attendant groups exist, all queues have the same priority assignments. Priority queue administration also applies for calls to an individual attendant, by way of the assigned extension.

Attendant VDNs

The fact that VDN extensions can be dialed directly or calls can be transferred to VDN extensions is unchanged for attendant VDNs.

Currently, VDN extensions can be assigned to the following night destinations:

Hunt group

An attendant vectoring VDN can be assigned as the night destination of a hunt group. Calls to the hunt group when it is in night service are redirected to the VDN and attendant vectoring applies. Hunt group night service does not apply if the hunt group is vector controlled. When **vector** on the Hunt Group screen is y, the **night service destination** field is removed from the screen. In order for a hunt group to be available in vectoring for the **queue-to hunt-group** command, the hunt group must be vector controlled. The hunt group in the **route-to** command can be in night service and the call can terminate to the indicated night service destination. If the hunt group is accessed using the **queue-to hunt-group** command no night service applies.

LDN and trunk

One or all trunk groups can be placed into night service and an attendant vectoring VDN can be assigned as the night service destination of the LDN or trunk group. If a night destination is assigned for LDN calls, it overrides the trunk night destination of the trunk group. Either of these destinations can be an attendant vectoring VDN. However, if Tenant Partitioning is administered and the trunk group night service destination is the attendant group, the call is redirected to the VDN associated with the tenant number of the trunk group. If, instead, the night service destination is explicitly assigned to a particular attendant vectoring VDN, it cannot be the resulted VDN had the night destination been the attendant group.

Trunk group incoming

The incoming destination can be an attendant vectoring VDN except for RLT trunk groups. As in trunk group night service, an assigned incoming destination to an attendant vector can result in

the call being sent to a different VDN than if the destination had been assigned to the attendant group.

Last coverage point in a coverage path

An attendant VDN can be assigned as a coverage point. If an attendant VDN is assigned as a coverage point, the VDN be the last point in the coverage path.

Abbreviated dialing lists

Attendant VDNs can be assigned to abbreviated dialing lists.

Emergency access redirection

An attendant VDN can be assigned to emergency access redirection. When the attendant's emergency queue overflows or when the attendant group is in night service, all emergency calls are redirected to this VDN. Careful thought must be given to routing the calls off-switch.

QSIG CAS number for attendant group calls

An attendant VDN can be assigned the QSIG CAS number which determines where attendant group calls at a QSIG branch are processed. This allows local vectoring at a branch prior to routing the calls to the main or elsewhere.

Auxiliary data for the following button assignments

In keeping with existing procedures, attendant VDNs is not denied as auxiliary button data for:

- Facility busy indication. Visual indication of busy or idle status for the associated extension.
- Manual message waiting indication. Lights a message waiting lamp on the station associated with the button.
- Manual signaling. Rings the station associated with the button.
- Remote message waiting indicator. Message waiting status lamp automatically lights when a LWC message is stored in the system for the associated extension.

Attendant Vectoring and attendant VDNs

When Attendant Vectoring is administered and if an attendant VDN is assigned, attendant group calls are intercepted and sent through vector processing. The attendant VDN can be assigned on the Console Parameters screen if Tenant Partitioning is turned off or on the Tenant screen if partitioning is turned on. If an attendant VDN is assigned, the call is redirected to the VDN for vector processing. If a VDN is not assigned, the call is directed to the attendant group. Attendant group calls can only be redirected to attendant VDNs.

Intercept attendant group calls

When calls are placed to the attendant group or become attendant group calls for the reasons listed below, a check is made for an assigned attendant VDN. If an attendant VDN is assigned and either the system is not in night service or the system is in night service and a night console is available, the call is redirected to the VDN for subsequent vector processing. Otherwise, the call is treated with typical attendant group procedures.

The following occurrences can cause a call to become an attendant group call:

- Listed Directory Number (LDN)

- Attendant group in coverage path
- Attendant control of trunk group access
- Calls forwarded to attendant group
- Controlled Restriction
- Dialed attendant access code
- DID/Tie/ISDN intercept treatment
- DID time-out due to Unanswered DID Call Timer expiration
- DID busy treatment
- Security Violation Notification (SVN)
- Multi frequency signaling with attendant group as terminating destination
- CDR buffer full with attendant group as Call Record Handling Option
- Trunk incoming destination is attendant group
- Trunk group night service destination is attendant group
- Hunt group night service destination is attendant group
- Automatic Circuit Assurance (ACA) referral
- VDN routes to the attendant access code.

Vector override always applies to attendant VDNs.

Allow override

VDN override always applies to attendant VDNs.

To provide the most flexibility possible, there are no restrictions placed on the vector that is assigned to a VDN. A non attendant vector can be assigned to an attendant VDN and an attendant vector can be assigned to a non attendant VDN. Obviously, doing so is not recommended. Assigning an attendant vector to a non attendant VDN severely restricts processing for basic call vectoring since only limited vectoring commands are available in attendant vectors. Assigning a non attendant vector to an attendant VDN also severely restricts attendant vectoring since the attendant-specific commands are not available in basic call vectoring. In addition, it removes basic call vectoring information from attendant VDNs. Also, there are no restrictions in vector chaining between attendant and non attendant vectors (for example, using the **goto vector** or **route-to number** commands).

Interflow between vectors

When calls interflow from one type of vector processing to another, the calls are removed from the queue and treated as new calls to vectoring, not continuations of vectoring.

Tenant Partitioning assignments apply to attendant and non attendant VDNs. Therefore, care must be taken that a VDN assignment on the Partitioning screen has a compatible TN number assigned to the VDN. For example, tenant partition 1 can be assigned a VDN which belongs to tenant partition 2 so long as partition 1's permissions allow access to partition 2. However, music source

selection is based on the tenant partition where the VDN is assigned rather than the partition to which the VDN belongs.

Music source

When music is to be provided for attendant vectored calls, the source that is assigned to the tenant partition of the attendant seeking call is used rather than the source that is assigned to the partition of the VDN.

Console Parameters screen example

When Attendant Vectoring is enabled, a field on the Console Parameters screen identifies the assigned Attendant Vectoring VDN. The following examples show the Console Parameters screens.

Console Parameters screen (Page 1)

```
change console-parameters                                    Page   1 of   4
                            CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                         COS: 1                                 COR: 1
       Calls in Queue Warning: 1                    Attendant Lockout? y
        Ext Alert Port (TAAS): 01A1216
                         CAS: none
                                                Night Service Act. Ext.: 195
                 IAS (Branch)? n                IAS Tie Trunk Group No.:
          IAS Att. access Code:                   Alternate FRL Station:
               Backup Alerting? y         DID-LDN Only to LDN Night Ext? n

Attendant Vectoring VDN: 2000
```

Console Parameters screen (Page 2)

```
change console-parameters                                    Page   2 of   4
                            CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 30           Return Call Timeout (sec): 30
  Time in Queue Warning (sec): 15

  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 10                     Alerting (sec): 10
                               Secondary Alert on Held Reminder Calls? y

ABBREVIATED DIALING
     List1:                   List2:                   List3: system

                          COMMON SHARED EXTENSIONS
            Starting Extension: 670              Count: 3
```

Console Parameters screen (Page 3)

```
change console-parameters                                    Page   3 of   4
                            CONSOLE PARAMETERS

QUEUE PRIORITIES

               Emergency Access: 1
                Assistance Call: 2
                        CO Call: 2
```

```
               DID to Attendant: 2
                        Tie Call: 2
             Redirected DID Call: 2
                 Redirected Call: 2
                     Return Call: 2
                     Serial Call: 2
    Individual Attendant Access: 2
                 Interpositional: 2
       VIP Wakeup Reminder Call: 2
             Miscellaneous Call: 2

       Call-Type Ordering Within Priority Levels? n
```

Console Parameters screen (Page 4)

```
change console-parameters                                    Page   4 of   4
                             CONSOLE PARAMETERS

ASSIGNED MEMBERS ( Installed attendant consoles )
               Type       Grp  TN            Type       Grp  TN
            1: principal   1   1           9:
            2:                            10:
            3:                            11:
            4:                            12:
            5:                            13:
            6:                            14:
            7:                            15:
            8:                            16:
```

Hunt group queue for Attendant Vectoring

If attendant vectoring results in putting a call in the hunt group queue, it is placed in the queue with the indicated priority. To use this command, the hunt group must be vector controlled.

Night service

There is no additional Night Service functionality provided for Attendant Vectoring. Night service routing can be provided using the existing night station service in conjunction with Attendant Vectoring. All existing night service rules remain in place, for example, night console service supersedes night station service, which supersedes TAAS. Attendant group calls are not redirected to Attendant Vectoring when the system is in night service unless a night console is available. Otherwise, the calls continue to be redirected to the applicable night service processing. To achieve Attendant Vectoring for calls when the system is in night service without a night console, the night station service extensions must be Attendant Vectoring VDN extensions.

Redirecting calls to attendant VDNs

The following types of calls cannot be redirected to the attendant VDN.

Type of calls	Description
Emergency Access	These calls are still sent directly to the attendant group. However, an attendant vectoring VDN can be assigned as the emergency access redirection extension.

Table continues…

Type of calls	Description
Attendant Return	These calls are still sent to the original attendant if the original attendant is available or will be placed into the attendant group queue if no attendants are available.
Serial	As with Attendant Return calls, Serial calls are still returned to the original attendant if the original attendant is available and are placed into the attendant queue if no attendants are available.
VIP Wakeup	These reminder calls are still sent directly to the attendant group.
Call Park Timeout	These calls result in a conference (caller, principal, and attendant) and Call Vectoring does not allow conferenced calls to be vectored.
Call Transfer Timeout	These calls are controlled by the attendant return call timer and are processed as though the calls are attendant extended calls, in other words, actual Attendant Return calls.

Restrictions for attendant and non attendant vectoring

No restrictions apply to attendant and non attendant vectoring. For example, an attendant VDN can point to a non attendant vector and vice versa. The same is true for vector commands. For example, an attendant VDN that points to an attendant vector can have a vector step that routes to another non attendant VDN. In this case, the call is removed from the queue and treated as though it just entered vector processing rather than as a continuation from one VDN to another. The reverse is also true if a non attendant VDN is routed to an attendant VDN.

TN assignments

Just as Tenant Number (TN) assignment determines the attendant group to which calls are terminated, the TN assignment also determines the VDN to which calls are redirected. If a VDN is administered, attendant group calls are redirected to the VDN rather than the attendant group. If a VDN is not assigned, calls terminate to the associated attendant group.

The selected TN for calls that are covered to an attendant group is the called user's TN, not the calling user's TN. When Tenant Partitioning is not administered, the system can have only one partition and attendant group. All attendant group calls are directed to attendant group 1. The screen to administer TN associations is not accessible, so system-wide console assignments apply. To follow the existing principals of this administration, the attendant vectoring VDN assignment appears on the Console Parameters screen when partitioning is turned off. When it is turned on, the field is removed from the console screen and the contents are automatically copied to TN 1.

Note:

Covered calls can be rerouted to an attendant group for a different tenant partition B when the call does not queue to the attendant group for tenant number A or for the out of hours case, and so on. The vector assigned to the attendant vectoring VDN for tenant A needs to include a failure branch to a "route-to ldn_number with cov y if unconditionally" step to route to the LDN extension of tenant B. The "with coverage" parameter of the route-to step must be set

to “cov y” and the original coverage path that covers to the tenant A attendant vectoring VDN must have the “Cvg Enabled for VDN Route-to party?” field set to “y”.

Vector screen example

The following example shows the Call Vector screen with the Attendant Vectoring field enabled.

Call Vector screen

```
change vector xxx                                        page 1 of 3
              CALL VECTOR

    Number: xxx                  Name: _______________________________
Multimedia? n
Attendant Vectoring? y
     Meet-me Conf? y           Lock? y
     Basic? n    EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? n    LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n

01 ____________
02 ____________
03 ____________
04 ____________
05 ____________
06 ____________
07 ____________
08 ____________
09 ____________
10 ____________
11 ____________
```

The **Attendant Vectoring** field appears only when Attendant Vectoring is enabled on the Customer Options screen. If either Basic Vectoring or Prompting are set to y, the Attendant Vectoring field defaults to n. If Basic Vectoring, Prompting, and Enhanced Conference are not enabled on the Customer Options screen, the Attendant Vectoring field defaults to y, and it cannot be changed to n. When the Attendant Vectoring field on the Call Vector screen is set to y, that vector is used as an attendant vector.

To associate VDNs and vectors for attendant vectoring, a field on the VDN and the call vectoring screens indicates attendant vectoring. When attendant vectoring is indicated for VDNs and vectors, all call center-associated fields (such as Skills and BSR) are not displayed.

Best Service Routing

Best Service Routing (BSR) is a Call Center Elite feature that provides singlesite and multisite load balancing and maximizes the agent staffing resources. BSR provides the best service to callers by comparing skills and routing calls to the best skill. To respond to changing conditions and operate efficiently, BSR monitors the status of the specified resources and adjusts call processing.

You can configure BSR for singlesite or multisite operations. Use singlesite BSR to compare local skills. Use multisite BSR to compare local and remote skills and to route calls to the skill that can provide the best service.

To use Best Service Routing (BSR) on a single Communication Manager, use special commands and command elements that are part of Call Vectoring. As a result, BSR for a single location can be easily added to existing vectors without modifying other parts of Communication Manager.

Multisite applications work similarly, but require additional administration. Since steps in a multisite BSR vector contact more than one remote locations, you must define the locations, administer Communication Manager to contact each location, and set up VDNs and vectors to handle communication between the sending Communication Manager and each remote Communication Manager.

Use three VDN or vector pairs in every multisite BSR application. The primary VDN or vector pair, on the sending Communication Manager, contacts the specified remote Communication Manager, collects information, compares the information, and delivers or queues the call to the resource that is most likely to provide the best service. Two VDN or vector pairs are required on each remote Communication Manager. A status poll VDN or vector pair provides information on the best resource in response to inquiries from BSR applications on other Communication Manager. Finally, you require an interflow VDN or vector pair to receive and process the calls that interflow from BSR applications on other Communication Manager.

Benefits of BSR

Singlesite and multisite BSR compare resources to find the resource that can provide the best service to a caller. With multisite BSR, you can integrate a network of call centers for better load balancing and optimal agent utilization.

Note:

If a call center network is overloaded and a significant number of calls are being blocked or abandoned, the use of BSR does not result in shorter wait times. Rather than reducing wait times, any productivity gains allows more calls to gain access to the network.

Benefit	Feature
Increased revenue	• Better agent utilization, allowing more calls to be handled with a given staff level. • Lower abandonment rates by balancing the load between resources, BSR reduces extremes in wait times across local resources or across an entire network. • In call centers with EAS, the ability to deliver calls to the best qualified or highest revenue generating agents.
Lower costs	• Better agent utilization. • Shorter trunk holding times. • Reductions of ineffective interflows. • Operation over ISDN-BRI trunks and public networks.

Table continues…

Benefit	Feature
Improved customer satisfaction	• Interflowing calls from centers with a surplus of calls to centers with a surplus of agents. You can achieve uniform service levels across your network. This means that all callers for a given application experience approximately equivalent waiting times. • Shorter wait times. • In call centers with EAS, the ability to deliver calls to the best qualified or highest revenue generating agents. • Robust information forwarding capabilities. Multisite BSR can forward original service requirements and any caller-entered digits with each call and can use both QSIG and non-QSIG information transport methods over private or public networks.
Increased performance and more efficient trunk usage	• Less messaging and processing required per call than in traditional LAI scenarios. • Eliminates phantom calls to remote agents. • Intelligent interflows that only route calls to centers with available agents.
BSR easy configuration	• Simple vector commands. You do not have to learn complex programming languages or design comparison steps. All that you have to do is list the local and remote resources to be considered for calls and instruct Communication Manager to queue or deliver the call to the best resource on the list.
Improved agent productivity	• Increased efficiency. Improve service without adding or reducing staff while maintaining the current level of service. Network-wide load balancing means that agents at one location are less likely to sit idle while calls wait in queue at another location. • No call delivery delays. In contrast to approaches that queue calls at all remote centers simultaneously, with BSR there is no delay in delivering a call when an agent becomes available.
Increased operating flexibility, easier staffing and scheduling	• Larger pool of agents available to take calls in a split or skill. Through its network-wide call distribution and information forwarding, BSR effectively converts distributed locations into a virtual call center. Staffing problems therefore, do not have to be solved on a center-by-center basis. BSR can automatically react to staff shortages at one center by routing more calls to other locations. • Automatic management of sudden and unexpected increases in call volume. Large increases in call volume for a single split or skill can be distributed across other splits or skills. Spikes in call volume at a single call center can be distributed across all call centers, provided trunk capacity is available between servers.
Improved service levels	• Lower Average Speed Answer (ASA).

BSR terminology

Term	Description
adjusted EWT	Expected Wait Time (EWT) plus a user adjustment set by a **consider** command.

Table continues…

Term	Description
agent selection method	The method that Communication Manager uses to select an agent in a hunt group when more than one agent is available to receive the next call. Possible methods are: • UCD-MIA • UCD-LOA • EAD-MIA • EAD-LOA The agent selection method is a property of hunt groups and is set in the **Group-Type** field on the Hunt Group screen. **Note:** To use an EAD available agent strategy, you must enable **Expert Agent Selection (EAS)**.
Alternate Selection on BSR Ties	BSR compares splits or skills using a series of **consider** steps and selects the one that provides the best service to a call. When that comparison results in a tie, the Alternate Selection on BSR Ties determines how BSR chooses which agent, skill, or location to select.
application	A general term for a system in any call center that handles calls of a particular type. In relation to BSR, any specific implementation of multisite BSR.
application plan	Used only in multisite applications, the application plan identifies the remote switches that can be compared in consider series. The plan also specifies the information that is used to contact each switch and to interflow calls to it.
best	Includes the following conditions • No agents available - When no agents are available in any of the specified splits or skills, the best resource is the one with the lowest adjusted EWT. • Agent available in one resource - When an agent is available in one and only one of the splits or skills that are specified in a consider series, that agent is the best and the call is delivered to that agent. If the **BSR Available Agent Strategy** field is set to `1st-found`, BSR ignores all subsequent steps in the consider series. If any other available agent strategy is used, all remaining resources are still checked before the call is delivered. • Agents available in more than two resources - When agents are available in more than two splits or skills, the best agent is the one that best meets the criteria that are specified in the **BSR Available Agent Strategy** field. For example, if the available agent strategy is UCD-MIA, the best agent out of those available is the agent with the longest idle time.
Best Service Routing (BSR)	A feature that is based on Call Vectoring and routes ACD calls to the resource that is best able to service each call. BSR can be used on a single switch, or to integrate resources across a network of switches.

Table continues…

Term	Description
BSR available agent strategy	A field that appears on the VDN screen when either version of BSR is enabled. The entry in this field is a property of the VDN and its assigned vector. Possible entries are: • 1st-found • UCD-MIA • UCD-LOA • EAD-MIA • EAD-LOA When the VDN is the active VDN for a call, as determined by VDN Override, this field determines how BSR commands in the vector identify the best split or skill when several have available agents.
consider series	**Consider** commands are written in a sets. This set of **consider** commands is called a consider series. A consider series in a status poll vector can have just one **consider** step.
consider sequence	A consider sequence is a consider series plus a **queue-to best**, **check-best**, or **reply-best** step.
Expected Wait Time (EWT)	Expected Wait Time is an estimate of how long a call in the queue has to wait before being connected to an agent.
Intelligent polling	An automatic feature of BSR that significantly reduces the number of status polls that are executed. When a remote location cannot be the best resource at a given moment in time, the intelligent polling feature temporarily suppresses polls to that location.
interflow	The process of routing an incoming call to an external switch without answering the call at the origin switch.
poll suppression	A component of BSR intelligent polling that eliminates wasteful polling of remote locations which have returned poor adjusted EWTs.
resources	An agent, split, skill, or location
status poll	A call that is placed by a **consider location** vector command to obtain status data from a remote location in a multisite BSR application.

BSR requirements

For singlesite BSR applications, Communication Manager must meet the requirements indicated in the following pages. The requirements for ISDN trunks and Look-Ahead Interflow (LAI) do not apply to singlesite BSR applications.

To use multisite BSR applications, all servers involved and the network connecting the servers must meet all the requirements that are described in this section.

Caution:

To ensure that the network meets the requirements for BSR support, contact your Account Executive about the BSR network certification.

Server requirements

Communication Manager must meet the requirements shown in the following table to support BSR.

Screen name	Page number	Field name	Field setting
System- Parameters Customer- Options	4	**ISDN-BRI Trunks** Multisite BSR operates over both BRI and PRI trunks. ISDN connectivity is necessary only to use multisite BSR, in which case set one or both the fields to `y`.	`y`
		ISDN-PRI You can set up ATM trunking, which is on page 3, and IP trunk, which is on page 4, to emulate ISDN PRI.	`y`
	6	**Vectoring (G3V4 Advanced Routing)**	`y`
		Vectoring (Best Service Routing)	`y`
		Lookahead Interflow (LAI) (To use multisite BSR)	`y`
Feature-Related System Parameters	12	**Adjunct CMS Release**	`R3V6` or later `blank`

Tip:

If you use BSR and then turn BSR off, you cannot set **Vectoring (Best Service Routing)** to `n` until you remove all BSR commands from the vectors. If you are using multisite BSR with LAI and want to turn LAI off, you cannot set **Lookahead Interflow (LAI)** to `n` until you remove all **`consider location`**, **`reply-best`**, and **`interflow-qpos`** commands from the vectors.

Network requirements

To support multisite BSR, networks must meet both the criteria for LAI call control operation over switched networks and the following criteria:

- The network must support end-to-end transport of codeset 0 user data, either as a User-to-User Information Element (UUI IE) or by a QSIG Manufacturer Specific Information (MSI IE), in the ISDN SETUP and DISCONNECT messages.
- With BSR poll calls, information is forwarded in the DISCONNECT message. In this case, the network must support forwarding of UUI in the first call clearing message, while the call is still in the call proceeding state, prior to the active state.
- Private networks can be configured for either QSIG using MSI packaged in codeset 0 Facility IEs or non-QSIG using a codeset 0 UUI IE transport. Currently, public networks cannot be configured for QSIG and user data can only be transported by the UUI IE when supported by the network. Future public network can be configured for QSIG, possibly by a Virtual Private Network (VPN).

- The switch must support the ISDN country protocol.
- The network byte limit for the user data portion of the user information contents must be large to carry the data needed for the customer application.

Note:

Some public network providers can require service activation, fees for user information transport, or both.

BSR, LAI, enhanced information forwarding, and UCID have been tested with several major carriers. To find if the capabilities work with your carrier, check with your account team for the most current information.

If testing has not been done to verify operation over the public networks that are involved with the preferred specific configuration, use private ISDN trunking between the nodes until successful testing is complete.

BSR considerations

- If more than one of the considered resources have an available agent, the resources with EWT are ignored. This means that there is an agent surplus.
- If the available agent strategy assigned to the active VDN is 1st-found, the adjust-by parameter is ignored and the first consider with an available agent is used for the queue-to best.
- If the available agent strategy is UCD-MIA, EAD-MIA, UCD-LOA, or EAD-LOA and there is more than one consider step with an available agent, the adjust-by parameter is applied as part of the algorithm to select the best of the possible choices.

Special BSR local treatment considerations

Trunk group status

Communication Manager displays the calls that queue remotely but receive local treatment as *active* trunk members if you execute the **`status trunk-group`** command on the interflowed trunk group. Even though the H.323 (IP) trunk member is *active*, no bandwidth is used because no voice packets are transmitted for local treatment.

Path replacement

Both ends of the connection must be answered for path replacement to work. When you set **BSR local treatment** to `y`, the local VDN answers, but the remote VDN where the call queues does not answer. Therefore, path replacement does not occur when local treatment VDNs queue a call remotely.

How BSR determines the best resource

BSR determines the best resource to service a call by examining one or all of the following variables:

- The Expected Wait Time (EWT) of the resource
- Any user adjustments

- The availability of agents
- The selection strategy for the active Vector Directory Number (VDN)

Note:

The BSR available agent strategy that applies to a given call is the strategy that is assigned to the active VDN for that call, as determined by VDN override.

Call surplus situation

Every BSR application compares a set of predetermined resources, splits or skills, and selects the best resource to service the arriving ACD call.

In a call surplus situation when no agents are available, the best resource is always the skill with the lowest EWT. For the purpose of calculations, the best resource in a call surplus situation, with BSR, you can adjust the EWT figure for any split or skill. The actual EWT for calls in queue is not changed. Only the figure used in the calculations performed by the BSR feature is changed. You do not have to enter adjustments, but the ability to adjust the EWT for splits or skills allows you to program preferences in vectors. Because of agent expertise, for example, or the availability or cost of tie trunks, you can choose that some resources do not service a call unless doing so significantly decreases the time in queue for the call.

You can make adjustments to agent availability using the **`consider`** step.

Agent surplus situation

In an agent surplus situation when more than one agent is available to take incoming calls, BSR delivers a new call according to the **BSR Available Agent Strategy** field specified on the VDN screen. The best resource is the split or skill that meets one of the five criteria that you have defined by the strategy as the VDN BSR Available Agent Strategy. BSR can use any of the five strategies shown in the following table to select an agent when agents are available.

Field setting	Call delivery
`1st-found`	The first available agent. BSR does not check any other resources as soon as BSR finds an available agent.
`ucd-mia`	The resource with a staffed agent who has been idle for the longest time. BSR compares all the splits or skills specified in the vector before delivering the call.
`ead-mia`	The resource with a staffed agent who has the highest (most primary) skill level that is relevant to the call and who has been idle the longest. BSR compares all the splits or skills specified in the vector before delivering the call.
`ucd-loa`	The resource with the least-occupied agent. BSR compares all the splits or skills specified in the vector before delivering the call.
`ead-loa`	The resource with an agent who has the highest (most primary) skill level that is relevant to the call and who is the least occupied. BSR compares all the splits or skills specified in the vector before delivering the call.

When agents are available in more than one of the specified resources, BSR does not check resources whether local or remote that return an Expected Wait Time value as is the case in a call queue or call surplus situation, when selecting the best place to send the call.

Note:

With the exception of the *1st-found* field setting, all other field settings must match the agent selection method used in the splits or skills checked by a BSR application.

Agent selection adjustments

An option has been provided to have the BSR adjust-by value apply in the agent surplus (agents available) situation. This adjustment provides the ability to use the **consider** step adjustment value to prioritize (handicap) agent resources when agents are available.

When the adjustment is used, the **consider** step uses the following syntax:

```
consider split/location adjust-by x
```

The server applies the agent adjustment in the same manner as the calls in queue/call surplus (lowest EWT) situation.

To select an adjustment, think in terms of reducing the importance of a resource/site and in relative percentage — the higher the adjustment, the less desirable it is to pick that agent/site. So, if **x** = 30, then the agent/site is 30% less desirable.

The available agent adjustment applies to the UCD-MIA, UCD-LOA, EAD-MIA, and EAD-LOA call distribution methods. For the most idle agent distribution methods, the adjust-by lowers the idle time value returned by the agent/site. For the least occupied agent distribution methods, the adjust-by raises the returned occupancy level of the agent/site. In either case, with EAD, the MIA or LOA is used as a tie breaker if more than one site has an agent available with the same highest skill level.

The same adjust-by value in the **consider** step applies to both agent surplus and call surplus situations.

Tips for writing BSR vectors

Before you write vectors using BSR, see the sample vectors provided and familiarize yourself with the new commands and command elements. Sample vectors are provided in singlesite BSR and multisite BSR. The new commands and command elements are explained in the Call Vectoring commands.

As you write BSR vectors, follow the guidelines:

- Arrange the **consider** steps in the order of preference.

  The **consider** step that tests the main or preferred, resource must be the first in the series. The second **consider** step must test the resource, that is, your second preference for handling the given call type. To prevent unnecessary interflows, use **consider** steps for local resources before steps for remote resources. This arrangement also provides a local best as a backup in case the interflow fails.

  Arranging **consider** steps in the order of preference for all BSR vectors. This arrangement is especially important when the active VDN for the call is using the 1st-found agent strategy

since the server delivers the call to the first available agent found, arranging **consider** steps in the order of preference ensures that calls are delivered to the best available resources.

- Do not put any commands between the steps of a **consider** series that can cause a delay. **Goto** commands are ok.
- Do not put a **consider** series in vector loops.
- Confirm that calls queue successfully.

  Since EWT is infinite for a call that is queued, a step that checks EWT after a queue attempt is a good confirmation method. After a **queue-to best** step, for example a command such as **goto step X if expected-wait for call < 9999** must be included.
- Do not use the wait-improved conditional in a vector before you have queued the call once.

  The wait-improved conditional compares the EWT of the call in the current queue to the best resource found by a **consider** series. If a call is not queued and a vector step such as **check best** if wait-improved > 30 is executed, the server interprets the current EWT of the call as infinite and the **check best** step always routes the call to the best resource. In other words, in this situation the **check best** step functions like an unconditional **goto** or **route-to** command.

Alternate Selection on BSR Ties

With Best Service Routing (BSR), Communication Manager uses a series of **consider** steps to compare splits or skills and to select the split or skill that can provide the best service to a call. When that comparison results in a tie, that is, the results are equal in value, Communication Manager uses Alternate Selection on BSR Ties to select an agent, skill, or location.

Communication Manager uses Alternate Selection on BSR Ties to select between the following:

- Skills or locations with the same Expected Wait Time (EWT).
- Available agents that are weighted with the same criteria, that is, the most idle agent or the least occupied agent, in a **consider** series designed to search for the best skill or best location.

Administer **BSR Tie Strategy** for the system or for each Vector Directory Number (VDN). Communication Manager uses each **consider skill/location** step to compare the current best choice with the previous **consider** step to the value obtained from the current **consider** step.

If **BSR Tie Strategy** on the active VDN for the call is **1st-found**, Communication Manager stops processing **consider** steps when Communication Manager finds an available agent at a **consider** step. Alternate Selection on BSR Ties is inapplicable in such cases.

Singlesite BSR

Singlesite BSR is a simple, logical extension of Call Vectoring. Like any other vector, vectors with BSR commands are assigned to more than one VDN. Using new vector commands and command elements, you tell the Communication Manager to compare specific splits or skills for

each call that is processed in that vector. Throughout the comparison, Communication Manager can remember which resource is the best based on how you define best. BSR vectors can deliver a call to the first available agent found, or check all the specified resources and deliver the call to the best split or skill. If no agents are available in any split or skill, Communication Manager queues the call to the split or skill with the shortest adjusted EWT.

Planning for singlesite BSR

About this task

To work efficiently, record goals, VDN extensions, vector numbers, and other information on paper before you begin your administration session. To do this, complete the following:

Procedure

1. Select the group of callers for which singlesite BSR is to be used and identify the VDNs and vectors to support the group.
2. Define your business goals.

   For example, your business goals in using BSR can be to achieve a faster Average Speed of Answer (ASA) for a certain call type, or perhaps better service by routing calls to the most qualified agents.

   Different VDNs or vectors have different goals.
3. Decide which agent selection strategy to assign to each VDN in order to achieve the goals relevant to the VDN.
4. Decide if the VDNs must follow VDN Override Rules.

Singlesite BSR command set

The following table shows the screens, the vectors, the vector commands and command elements used in singlesite BSR. The following table shows the vector commands and command elements used in singlesite BSR applications.

Commands and command elements		Purpose
Screens	Vector Directory Number	To link a VDN to a BSR vector. To set the agent selection strategy that is used for all calls to that VDN.
	Call Vector	To confirm that BSR is administered. To write vectors that use BSR commands.

Table continues…

Commands and command elements		Purpose
Commands	**consider split/skill**	To retrieve the EWT or agent data required to identify the best local resource. One **consider** step must be written for each split or skill that is to be checked. Since the **consider** command is designed to compare more than two resources, **consider** commands are written in a series of more than two commands with the sequence terminating in a **queue-to best** vector step. This set of **consider** commands and a **queue-to best** step is called a consider sequence.
	queue-to	With the *best* keyword to queue calls to the best resource identified by the consider sequence.
	check	With the *best* keyword to queue calls to the best resource identified by the consider sequence if the resource meets certain conditions.
Key word	**best**	Use the *best* keyword in **queue-to**, **check**, and **goto** commands that refer to the resource identified as *best* by a series of **consider** steps
Conditional	**wait-improved**	To prevent calls from being queued to an additional split or skill when the reduction in EWT is not enough to be useful. Wait improved means that the EWT of a call must be improved by a specific amount, specified in seconds, over the current EWT or Communication Manager does not queue the call to the additional split or skill.
User adjustment	**adjust-by**	To specify your preferences for the splits or skills to handle the calls for a particular application, reflecting factors such as agent expertise or reducing calls to a backup split or skill. When a vector checks a local resource you can make the selection of that split or skill less desirable. The higher the setting, the lesser the chance that the resource is selected over another with a lower setting. For example, set to 30 makes that choice 30 percent less desirable. With EWT returned, the setting increases the returned EWT for comparison with other returned EWTs. As a result, the split or skill is less likely to service the call unless its EWT is significantly less than that of any other available split or skill. Optionally, the adjust-by parameter applies in the available agent case. If you are using the UCD-MIA or EAD-MIA available agent strategy, the setting decreases the returned agent idle time, making the agent appear less idle. If you are using the UCD-LOA or EAD-LOA available agent strategy, the setting increases the returned agent occupancy, making the agent appear busier. In either case with EAD, the MIA or the LOA is used as a tie breaker if more than one site has an agent available with the same highest skill level.

Example of basic singlesite BSR

The central element of singlesite and multisite BSR is a VDN or a vector pair. The vector contains the commands that process the call, but the active VDN for the call contains information that is used by the vector steps.

For singlesite BSR, the active VDN for a call sets the available agent strategy that is used by the vector.

Singlesite BSR example VDN screen

```
change vdn xxxxx                                                 page 1 of 3
        VECTOR DIRECTORY NUMBER

                            Extension: 5000
                                 Name: Singlesite BSR
                       Vector Number: 234
                   Attendant Vectoring? n
                    Meet-me Conference? n
                    Allow VDN Override? n
                                  COR: 59
                                   TN: 1
                             Measured: internal
       Acceptable Service Level (sec): 20
              Service Objective (sec):
        VDN of Origin Annc. Extension:
                            1st Skill:
                            2nd Skill:
                            3rd Skill:
```

```
change vdn xxxxx                                                 page 2 of 3
        VECTOR DIRECTORY NUMBER

                           Audix Name:

                  Return Destination:
              VDN Timed ACW Interval:
                     BSR Application:31
         BSR Available Agent Strategy: 1st-found
```

In the example Vector Directory Number screen, the **BSR Available Agent Strategy** field is set to `1st-found`. If vector 234 uses BSR commands, when a **`consider`** step locates a resource with an available agent, all subsequent consider steps are skipped and the call is delivered to that resource. Resources that are specified in any subsequent **`consider`** commands are not checked. If no split has an available agent, the call is queued to the split with the lowest adjusted EWT.

If the **Allow VDN Override** is set to `n` and a second VDN and vector are used to process this call, the 1st-found strategy specified in VDN 5000 is still used.

In the preceding example, VDN 5000 is associated with vector 234, which is shown below. In this example, vector 234 compares two splits. No adjustment is assigned to either resource, indicating that both splits are equally suited to service calls since neither is preferred to the other. In reality, such a vector probably has additional steps after step 4, such as **`announcement`** or **`wait-time`** commands. The steps are omitted in this example for purposes of clarity.

Singlesite BSR example vector

```
1. wait time 0 secs hearing ringback
2. consider split 1 pri l adjust-by 0
3. consider split 2 pri l adjust-by 0
4. queue-to best
```

Note that the **consider** commands follow each other in unbroken sequence and that the **queue-to best** command immediately follows the last **consider** command. This structure is called a *consider series*. Write such series in uninterrupted order. You can use a few commands, such as the **goto** command, which cause little if any delay in the execution of the **consider** steps. In general, however, do not put other commands between **consider** steps, or between a **consider** step and a **queue-to best** step. Even if BSR still works in that situation, the performance can be impaired.

Consider commands collect and compare information. When a call is processed in the vector, the first consider step collects and temporarily saves the following information about split 1:

- The fact that split 1 is a local split.
- The queue priority that is specified in the consider step.
- The user adjustment that is specified in the consider step.
- The split's
  - Split number
  - Expected Wait Time (EWT)

If EWT=0, which indicates that more than one agent is available, the step also collects all the agent information required by the BSR available agent strategy. This includes:

- Agent Idle Time (AIT)
- Agent Occupancy (AOC)
- The skill level of the agent in the split or skill who receive the next call

In the example, the splits do not have an available agent when the consider series executes. If a split has an available agent, the call is delivered to that split by the **queue-to** best step. Since there are no available agents in either split, the complete set of saved data now defines the best resource. The second consider step collects and compares the same data to the current best data. For this example, if the EWT for split 1 is 40 seconds and the EWT for split 2 is 20 seconds, when the second consider step executes, the data replaces the best data from step 1 because the adjusted EWT is lower. The best data is essentially a placeholder. When a **queue-to** best step executes, the step reads the data saved as the best at the moment and queues the call to the split. In this case, the best data was collected from split 2, so the call is queued to split 2 at the specified priority.

What if there are available agents in both splits?

As *BSR Available Agent Strategy* in this example is 1st-found, the consider series skips the consider steps after step 2 and the queue-to best step delivers the call to split 1, which is the first split or skill with an available agent that is found by the vector.

In any BSR vector, the order of the consider steps must reflect your preferences for the resources. In the consider series, put the step that considers the most preferred split or skill first, followed by the other splits or skills.

What if there are several available agents in split 1? Which agent receives the call?

When more than one agent is available in a split, the BSR **consider** command collects agent data only for the agent who will receive the next call to that split. This agent is identified according to the agent selection method that is specified in the **Group-Type** field on the Hunt Group screen.

Note:

For greatest efficiency, the agent selection method used in the splits or skills checked by a BSR vector must match the BSR Available Agent Strategy that is assigned to the active VDN.

User adjustments in singlesite BSR

In singlesite and multisite BSR, with the *adjust-by* portion of the **consider** command, you can implement your preferences into the vectors.

You can assign a value of 0 to 100 in user adjustments. The server supplies the units of the value depending on the conditions whenever the **consider** step executes. For example, in the command **consider split 1 pri h adjust-by 20**, the server interprets *adjust-by 20* to mean *add 20 percent to the EWT, but add 20 seconds at the minimum*.

Note:

If you define the user adjustment as a number of seconds, BSR is not efficient when EWT was high. If you define the user adjustment as a percentage, BSR is not efficient when EWT was low. Such efficiencies, while always important, become critical in multisite BSR applications where issues of trunk cost and capacity are involved.

For EWT of 1 to 100 seconds, an adjustment of 20 adds 20 seconds. Over 100 seconds, the same adjustment adds 20 percent to the EWT for the split or skill that is specified in the **consider** step. The following table shows the results of user adjustment to a range of EWT.

EWT of resource (in seconds)	User adjustment	Adjustment applied by the server (in seconds)	Adjusted EWT used to select resource
10	20	20	30
60		20	80
120		24	144
300		60	360

Example of singlesite BSR with adjustments

The following example shows a more complex implementation of singlesite BSR. Four skills in an EAS environment are compared. The Expected Wait Time (EWT) for some skills is adjusted to reflect the administrator preferences.

Singlesite BSR example VDN screen

```
change vdn xxxxx                                          page 1 of 3
                    VECTOR DIRECTORY NUMBER

                           Extension: 5001
                                Name: Singlesite BSR
                        Vector Number: 11
                    Attendant Vectoring? n
                     Meet-me Conference? n
                     Allow VDN Override? n
                                  COR: 59
                                   TN: 1
                             Measured: internal
          Acceptable Service Level (sec): 20
```

```
              Service Objective (sec):
       VDN of Origin Annc. Extension: 501
                           1st Skill:
                           2nd Skill:
                           3rd Skill:
```

```
change vdn xxxxx                                              page 2 of 3
                  VECTOR DIRECTORY NUMBER

                            Audix Name:
                     Return Destination:
                VDN Timed ACW Interval:
                       BSR Application: 19
         BSR Available Agent Strategy: EAD-MIA
```

In the example, the **BSR Available Agent Strategy** field is set to `EAD-MIA`. If vector 11 uses BSR commands, calls are not automatically delivered to the first resource with an available agent that is found. All **`consider`** steps in vector 11 are executed and one of the following things happens:

Condition	Action
No skill has an available agent	The call queues to the skill with the lowest adjusted EWT.
One skill has an available agent	The call is delivered to the skill.
More than two skills have available agents	The call is delivered to the skill with the most expertise.
More than two skills have available agents with the same skill level	The call is delivered to the agents who has been idle the longest.

Also note that **Allow VDN Override** is set to `n`. If a second VDN and vector are used to process this call, the EAD-MIA strategy that is specified in VDN 5001 is used. If **Allow VDN Override** is set to `y` and vector 11 routes some calls to another VDN, the available agent strategy of the subsequent VDN governs the operation of **`consider`** steps in its vector.

The following example vector 11, which compares four skills.

Singlesite BSR example vector

```
1. wait-time  0 secs hearing ringback
2. consider skill  1 pri l adjust-by 0
3. consider skill  2 pri l adjust-by 30
4. consider skill 11 pri l adjust-by 30
5. consider skill 12 pri l adjust-by 30
6. queue-to best
7. wait-time 10 secs hearing ringback
8. announcement 1001
9. wait-time 30 secs hearing music
10. goto step 8 unconditionally
```

For this example, the EWT of the four skills are 95, 60, 180, and 50 seconds, respectively. Note that all **`consider`** steps except the first adjust the EWT returned by the specified skill. Skill 1 is the preferred skill to handle calls to VDN 5001, so the EWT is not adjusted. Skills 2, 11, and 12 can handle this call type, but the skills are not preferred. The adjustment of 30 means that, in call surplus situations, the skills do not handle calls to VDN 5001 unless the EWT is a minimum of 30 seconds better than the EWT in skill 1.

The following table shows the adjustments to be applied to each skill given the EWT and the user adjustment specified in the **`consider`** step. The last column shows the adjusted EWT the server uses to select a skill for the call.

User adjustments				
Skill number	**User adjustment in the consider step**	**Actual EWT (in seconds)**	**Adjustment applied by the server (in seconds)**	**Adjusted EWT used in BSR calculations (in seconds)**
1	0	95	0	95
2	30	60	30	90
11	30	180	54	234
12	30	50	30	80

Since the available agent strategy is not 1st-found, all four **consider** steps are executed each time that the vector processes a call. In this example, there are no available agents in any skill. In fact, EWT is high in the first three skills for the server to queue the call to skill 12.

When the **queue-to-best** step executes, the data in the best data placeholder is the data from skill 12 and so the call is queued to that skill. From this point on, if the call is not answered during the execution of step 7, a common vector loop regularly repeats an announcement for the caller while the caller waits in the queue.

User adjustments also apply to available agent situations, with a strategy other than first found, in a manner similar to EWT.

What if there is an available agent in one skill?

Since the “BSR Available Agent Strategy” in this example is EAD-MIA, the entire consider series will always be executed to check all of the skills for available agents. If only one skill has available agents, the call is delivered to that skill and user adjustments are not applied.

What if there are available agents in two skills?

Since the BSR Available Agent Strategy for VDN 5001 (the active VDN) is EAD-MIA, the call is delivered to the skill with the most expert agent. If there are available agents in both skills with the same skill level, their user adjusted idle times are compared and the call goes to the skill with the agent who has the longest adjusted idle time.

If a split/skill has more than one available agent, remember that it is the split/skill’s agent selection method that determines which agent’s data is used in BSR selection of the best resource.

What if no agents are staffed in a skill? Will the server recognize this?

Yes. Under any of the following conditions, the EWT returned from a split/skill is infinite:

- No agents logged in
- No queue slots available
- All agents in AUX work mode

The server logs a vector event and goes to the next vector step without changing the data in the best placeholder. A resource with an infinite EWT is never selected as the best resource.

Can VDN skills be used in consider steps?

Yes. For example, consider skill 1st [2nd, 3rd] pri m adjust-by 0 will collect data on the 1st [2nd, 3rd] skill, as defined for the active VDN.

Multisite BSR

Multisite BSR extends all of the capabilities of singlesite BSR across a network of Communication Manager. Multisite BSR compares local splits/skills and remote splits/skills, and routes calls to the resource that provides the best service.

Multisite BSR has special features that work to ensure efficient use of processor power and network resources in your BSR applications.

Multisite BSR command set

The following table shows the screens, vectors, special vector commands, and command elements required to administer multisite BSR applications.

Screen name	Description
Best Service Routing Application Plan	• To define the group of remote sites to be polled by a specific application. • To assign a unique name and number to each application. • To assign routing numbers for the status poll and interflow VDNs.
Vector Directory Number	• To link a VDN to a BSR application by the application number. • To link the VDN to a BSR vector. • To set the agent selection strategy to be used for all calls to that VDN.
Call Vector	• To confirm that BSR is administered and to program the vector steps for BSR.
ISDN Trunk	• To administer Communication Manager to forward user information by Shared UUI or QSIG MSI.
List Best Service Routing Applications	• To display a list of all the BSR applications by name and number.
System Capacity	• To monitor the number of BSR application-location pairs assigned in the system.
VDNs and Vectors	
Primary VDN, which is the active VDN for the call at the origin, as determined by VDN Override Rules	• To define the application plan and available agent strategy used by the vector assigned to this VDN.
Primary vector	• To control call processing at the original server and compare local and remote resources.
Status poll VDN or vector	• To respond to status poll calls from another server. The status poll vector checks a set of local splits or skills and returns data on the best resource to the original server.

Table continues…

Screen name	Description
Interflow VDN or vector	• To accept BSR calls from another server and to queue the calls to the best of the local resources.
Commands	
consider split/skill	• To obtain the EWT or agent data needed to identify the best local resource. One **consider** step must be written for each split or skill that is to be checked. Since the **consider** command is designed to compare more than two resources, **consider** commands are written in a series of more than two commands with the sequence terminating in a **queue-to best** vector step. This set of **consider** commands and a **queue-to best** step is called a consider sequence.
consider location	• To obtain the EWT or agent data required to identify the best resource at a remote server. One **consider** step must be written for each location that is to be checked. Routing information is obtained from the BSR Application plan for the active VDN.
reply-best	• To return data to another server in response to a status poll.
queue-to	• With the *best* keyword to queue calls to the best resource identified by the consider sequence.
check	• With the *best* keyword to queue calls to the best resource identified by the consider sequence if the resource meets certain conditions.
Key word	
best	• In **queue-to**, **check**, and **goto** commands that refer to the resource identified as *best* by a series of **consider** steps
Conditional	
wait-improved	• To prevent calls from being queued to an additional split or skill that can be local or remote when the reduction in EWT is not useful. Wait improved means that the EWT of a call must be improved by a specific amount, which is a figure that you specify in seconds, over the current EWT or the server does not queue the call to the additional split or skill.
User adjustment	

Table continues…

Screen name	Description
adjust-by	• To control long distance costs and limit trunk usage, reflecting factors such as availability of the trunks or agent expertise at remote locations. When a vector polls a local or remote resource, you can make the selection of that site less desirable. The higher the setting, the lesser the chance that the resource is selected over another with a lower setting. With EWT returned, the setting increases the returned EWT for comparison with other returned EWTs. Optionally, the *adjust-by* setting applies in the available agent case. If you are using the UCD-MIA or EAD-MIA available agent strategy, the setting decreases the returned agent idle time, making the agent appear less idle. If you are using the UCD-LOA or EAD-LOA available agent strategy, the setting increases the returned agent occupancy, making the agent appear more occupied. In either case with EAD, the MIA or the LOA is used as a tie breaker if more than one site has an agent available with the same highest skill level.

Multisite BSR applications

You can implement BSR at a single location by using the BSR commands in vectors. Using BSR across a network is more complex and requires additional administration.

A series of consider location steps in a multisite BSR vector contacts several remote locations. You must define the locations, determine how the server contacts each location, and set up VDNs and vectors to handle the communication between the origin server and the remote servers.

Note:

You can use any combination of split or skill numbers, VDN numbers, and vector numbers to support a single customer application or call type across a network. For clarity and simplicity, use the same BSR Application Plan number and the location numbers on all servers for a given application.

You must also set up the ISDN trunk groups, parameters for information forwarding (UUI Transport), and administer the numbering plans and the AAR/ARS tables.

Multisite BSR starts with the active VDN for a call, as determined by VDN Override rules. If you want any specific VDN or vector pair to interflow calls using multisite BSR, you must create a specific application for the VDN or vector. A multisite application must contain the elements shown in the following table.

BSR application elements	Purpose
The primary VDN	The primary VDN is the active VDN for a call at the origin server, as defined by the VDN Override rules. Therefore, the primary VDN in a BSR application does not have to be the VDN that originally received the incoming call. The primary VDN links its assigned vector to a BSR application plan and sets **BSR Available Agent Strategy**.

Table continues…

BSR application elements	Purpose
The primary vector that handles the incoming call on the origin server	The primary vector contacts the specified remote servers, collects information, compares the information, and delivers or queues the call to the resource that is likely to provide the best service.
An application plan	The application plan identifies the remote servers and specifies the information to be used to contact each server and to route calls.
Two VDN or vector pairs on each remote server: • Status poll VDN or vector • Interflow VDN or vector	Status poll VDN or vector The status poll vector compares splits at the location and replies to the origin server with information on the best of the splits. Each remote server in a given application must have a dedicated status poll VDN or vector.
	Interflow VDN or vector When a given remote server is the best available, the origin server interflows the call to the VDN or vector on the remote server. Each remote server in a given application must have a dedicated interflow VDN or vector. The steps in this vector deliver or queue the call to the best resource found by the status poll vector.

To create a multisite BSR application, first create an application plan on the origin server.

Note:

Remember that the terms local, origin, and remote are relative terms. In most networks that use multisite BSR, every server can interflow calls to other servers and receive interflowed calls from other servers. Therefore, every server in the network can have all the elements described in the table. For clarity in the following discussions, local or origin means a server that decides whether to interflow a call. Remote means a server that is polled by the origin server.

Application plans

The application plan identifies the remote servers and specifies the information that is used to contact each server and to route calls.

You can identify the plan for each application by the application number and a name. The plan specifies the remote servers to be polled by the application and identifies each with a number called the Location Number. The plan also specifies the numbers for the status poll and interflow VDNs for each remote server. Whatever you dial to reach the VDNs is what you must enter in the fields: full length numbers as well as AAR, ARS, UDP, or public network numbers work.

Create application plans on the Best Service Routing Application screen. A plan for an application with three remote servers looks like the following example.

Sample multisite BSR application plan

```
BEST SERVICE ROUTING APPLICATION PLAN
Number: 15    Name: Customer Service    Maximum Suppression Time: 60     Lock? y

Num  Location     Switch   Node          Status Poll VDN    Interflow VDN    Net Redir?
1    New Jersey   320      84015         84115                               n
2    Denver       18       913031234015                     913031234115     n
```

```
4    New York     12345    912121234015                    912121234115      n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
___ _____________      ____  ____  __________  __      _____________         n
```

The maximum number of application plans vary depending on your Communication Manager software release and platform. For more information, see *Avaya Aura® Communication Manager System Capacities Table*.

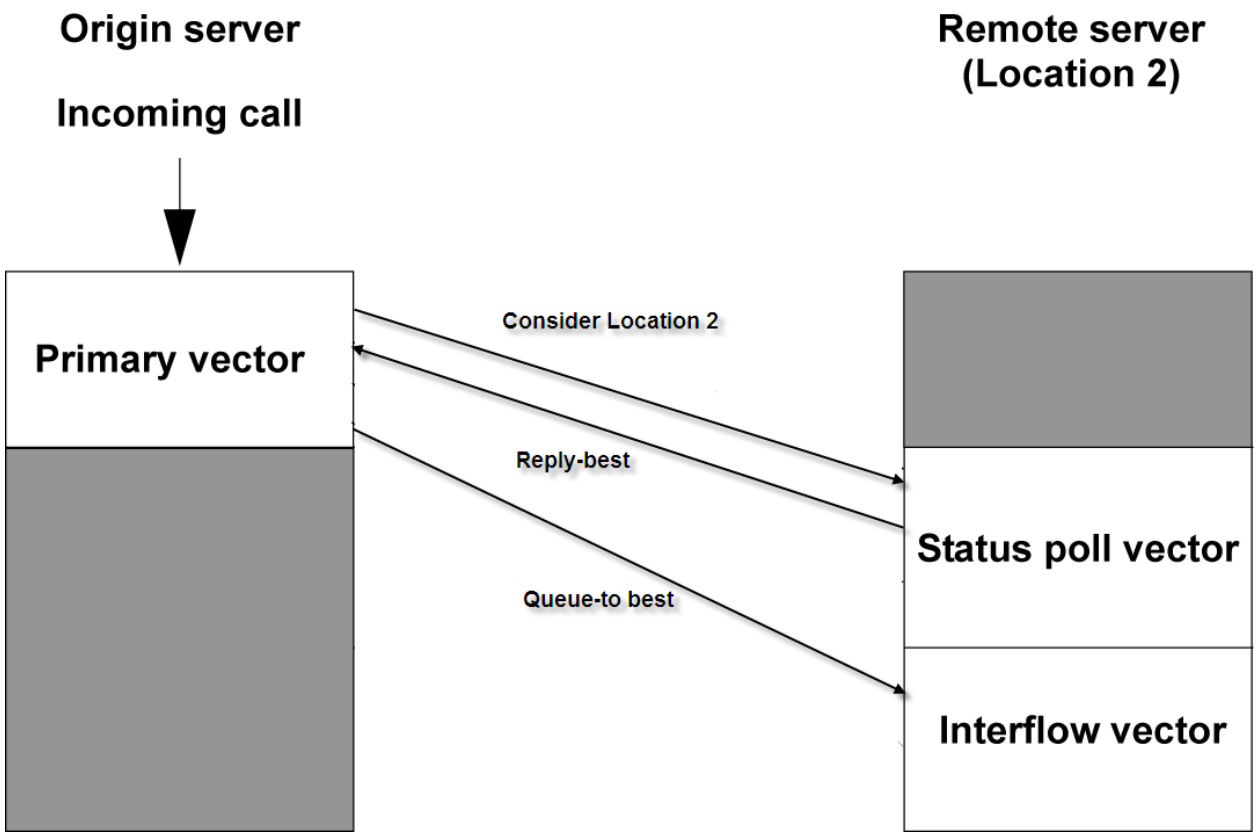
By entering the application number from this plan on a VDN screen, you can link a given VDN on your local server to this list of locations. This VDN becomes the primary VDN for the application. For example, if the primary vector contains instructions to check locations 1 and 2, the server places a status poll call to the status poll VDN at the New Jersey and Denver servers and compares the results. If location 2 is better than either location 1 or any splits that are checked on the originating server, the call is interflowed to the interflow VDN that is specified in the plan for location 2.

Simple multisite BSR

Multisite BSR compares local and remote splits or skills and queues calls to the resource that provides the best service. Remember that each BSR application has the following main parts:

- An application plan that identifies the remote servers for comparison.
- A set of the following VDN or vector pairs:
  - Primary: The primary vector, which is on the origin server, contacts the specified remote servers, collects information, compares the information, and routes the call to the server that is likely to provide the best service.
  - Status poll: The status poll vector, which is on the remote server, compares resources on the remote server and replies to the origin server with information on the best of the resource. Each remote server in a given application must have a dedicated status poll vector.
  - Interflow: When a given remote server is the best available, the origin server interflows the call to the vector on the remote server. Each remote server in a given application must have a dedicated interflow vector.

The general operational scheme for multisite BSR is shown in the following figure.



The following example shows the primary VDN using a multisite BSR application.

BSR example primary VDN

```
change vdn xxxxx                                                page 1 of 3
        VECTOR DIRECTORY NUMBER

                             Extension: 52222
                                  Name: Multisite BSR
                         Vector Number: 222
                     Attendant Vectoring? n
                      Meet-me Conference? n
                      Allow VDN Override? n
                                   COR: 59
                                    TN: 1
                              Measured: internal
       Acceptable Service Level (sec): 20
              Service Objective (sec):
        VDN of Origin Annc. Extension:
                             1st Skill:
                             2nd Skill:
                             3rd Skill:
```

```
change vdn xxxxx                                                page 2 of 3
VECTOR DIRECTORY NUMBER
                               Audix Name:
                     Return Destination:
                 VDN Timed ACW Interval:
                        BSR Application: 15
          BSR Available Agent Strategy: UCD-MIA
```

In the example shown above for VDN 52222, the entry in the **BSR Application** field links this VDN to BSR Application Plan 15. Also note the UCD-MIA entry in the **BSR Available Agent Strategy** field. If vector 222 uses BSR commands, calls are not automatically delivered to the first resource found with an available agent. All **consider** steps in vector 222 are executed, and one of the following happens:

If	Then
There is no available agent in the local or the remote splits	The call queues to the split with the lowest adjusted EWT.
Only one split has an available agent	The call is delivered to the split.
More than two splits have available agents	The call is delivered to the split with the most idle agent.

Note that the **Allow VDN Override** field is set to `n`. If a second VDN and vector are used to process the call, the UCD-MIA strategy and the application plan specified in VDN 52222 are used.

Application plan 15, which is shown in the sample multisite BSR application plan, identifies the remote server and provides the digit strings to dial into the VDNs for both the status poll vector and the interflow vector.

BSR primary vector

When a call arrives at the origin server, the primary vector processes the call. The vector begins the BSR process by checking the specified resources. The following example shows a primary vector used for that purpose.

BSR example of primary vector on origin Communication Manager

```
1. wait time 0 secs hearing ringback
2. consider split   1 pri m   adjust-by 0
3. consider location 2 adjust-by 30
4. queue-to-best
```

In this example, the **consider** commands in steps 2 and 3 collect information to compare local split 1 with more than one split at location 2. Location 2 is the Denver server identified on the BSR Application Plan screen. Step 4 queues the call to the best found split. As in singlesite BSR, with the adjust-by parameter of the **consider** command, you can set preferences for each resource, whether the resource is a remote location or a split/skill on the origin server. In multisite BSR, with the user adjustment, you can control the frequency of interflows by adjusting the EWT that is returned by a particular resource on a remote server. In this example, the Communication Manager administrator has chosen to adjust the EWT value for location 2 by 30.

BSR status poll vector

To collect information from the remote server, the **consider location 2 adjust-by 30** command in the primary vector places an ISDN call, known as a status poll, to the status poll vector on the server at location 2. The following example shows a status poll vector on the remote server used for the purpose.

BSR example of status poll vector on remote Communication Manager

```
1. consider split  2    pri m   adjust-by 0
2. consider split 11    pri m   adjust-by 0
3. reply-best
```

The status poll retrieves and returns information to the origin server. The call is not connected to the status poll VDN.

The vector compares splits 2 and 11, identifies the best split and sends the information back to server 1 with the **reply-best** command. Note that the **adjust-by** command can be used on the remote server to adjust the EWT returned by either of the splits. When EWT adjustments are applied at both the origin and remote servers, the two adjustments are added at the origin server.

The **consider** command is ISDN neutral and does not return answer supervision. The status poll call drops after vector processing executes the **reply-best** step, but the ISDN DISCONNECT message returned to server 1 contains the information from the best split considered at location 2. Once the remote server returns the necessary information, the **consider** series in the primary vector on server 1 can continue at the next vector step.

Caution:

Do not use the status poll vectors to poll other servers. Status poll vectors compare resources only on the server where the vector resides. Status poll vectors must always end with a **reply-best** command. Do not use a **busy** or **disconnect** command.

Note:

Multisite BSR includes mechanisms that automatically limit the number of status poll calls that can be placed over the network when such calls are unlikely to yield better service for the caller.

BSR interflow vector

In this example, no agents are available and split 11 (location 2) has the lowest adjusted EWT. The **queue-to best** command in the primary vector interflows the call to the interflow vector at location 2. The following example shows what the interflow vector looks like.

BSR example of interflow vector on remote Communication Manager

```
1. consider split   2   pri m   adjust-by 0
2. consider split 11     pri m   adjust-by 0
3. queue-to best
```

The interflow vector checks the status of both splits again to get current information and queues or delivers the call to the best split. Note that the consider sequences in the interflow vector and the status poll vector are identical aside from their last step. When a call is interflowed, the call is removed from any queues at the origin server and any audible feedback at the origin server is terminated.

Caution:

BSR will not operate correctly unless the consider series in the status poll vector and the interflow vector use the same splits or skills with the same queue priorities.

BSR call interflow with SIP

You can use SIP to interflow BSR calls, but not BSR polling. The polling must be done using either H.323 or ISDN trunks. Once polling determines the best site, the incoming call can be routed to another location by the queue-to-best step. The redirected call can be routed over a SIP trunk.

To use BSR local treatment for calls queued remotely, the queued call must be answered before expiration of the Session Establishment Timer for the SIP trunk signaling group. You must

increase the Session Establishment Timer to an appropriate value. For an approach to handling exceeding the timer, see the local treatment “Take back example” in the “Example vector for the local Communication Manager” topic.

Related links

Example vectors for the local Communication Manager on page 330

What happens to the call if the interflow attempt fails?

If the interflow attempt fails, for example, because there are no available trunks, the call is queued to the best local split. The call is not disconnected. The call is not dropped from vector processing on the origin server. For the call to be queued to a local split, however, that split must be checked at some previous point in the consider series. In writing primary vectors, always check local splits or skills before remote resources.

Adjusting the AIT or AOC returned by an available resource

Procedure

1. Enable the **Available Agent Adjustments for BSR** field on the Feature-Related System Parameters screen.
2. On the Vector screen, administer a **`consider split/skill`** or **`consider location`** vector command specifying both the split or skill, or location and the adjust-by parameter.

   You can use the adjust-by parameter to provide a percentage value during vector processing and can be:

   - A percentage (0 through 100)
   - A vector variable (A-Z, AA-ZZ)
   - A VDN variable (V1-V9)

Result

Once the vector command is executed, the adjustment factor has the following result when the remote site has an available agent:

- For the MIA strategies, the adjustment reduces the Agent Idle Time (AIT) received.
- For the LOA strategies, the adjustment increases the Agent Occupancy percentage (AOC) received.

Depending on the available agent strategy assigned to the VDN for the call, the adjusted AIT or the adjusted AOC is used for that local split or skill, or remote location when choosing between available agents over multiple locations.

Example: You have an agent whose current AIT is 40 percent. You can increase the agent idle time to 60 percent to prevent sending calls to the remote location. If the strategy is `ucd-loa`, use the following vector command:

```
consider location 4 adjust-by 50
```

The occupancy used for location 4 is increased by 50 percent of the actual occupancy. The occupancy originally sent was 40 percent. A 50 percent adjust-by results in addition of 40 to 50 percent of 40. Therefore, 40 + 20 = 60 percent.

BSR available agent strategies

In multisite BSR applications, the *1st-found* available agent strategy results in fewer interflows and minimizes the load on trunking. Communication Manager has less processing to perform for each call in BSR vectors as Communication Manager does not compare many resources to identify the best resource.

If the processing power and tie trunk capacity are issues in your multisite applications, use the 1st-found strategy.

The other strategies typically result in a much greater percentage of calls being interflowed, optimizing load balancing across locations.

More on status poll and interflow vectors

Read the following points before you write status poll and interflow vectors:

- Since status poll vectors do not return answer supervision, call charges are not normally incurred for the status poll portion of the call flow.
- When a **consider location** step performs a status poll, the command also checks for the availability of a B-channel. If no B-channel is available, the remote resource is not treated as the best since the call cannot be redirected to a remote resource.
- If only one split or skill on a remote server can service the call type handled in a BSR application, you do not have to write a consider series in the interflow vector. You can just queue the call to the appropriate resource.
- If status poll and interflow vectors check more than one split or skill, the VDNs for the vectors must be administered with the appropriate BSR available agent strategy.

User adjustments in multisite BSR

User adjustments are especially important in multisite applications, where unnecessary interflows are costly and use trunk capacity inefficiently.

User adjustments in multisite applications function in the same way as in singlesite BSR with one important difference: user adjustments is applied at the remote servers in an application as well as at the origin server. Since a status poll vector uses **consider** steps to evaluate resources on the server where the vector resides, with the *adjust-by* portion of each **consider** command, the administrator at each server can set preferences for the splits or skills at the server. In BSR applications, the status poll vector checks any such adjustment for a split or skill when selecting the best resource. The adjustment is then returned to the origin server along with the other data for that resource. When the server receives the adjustment from the remote server, the server adds the adjustment to any adjustment that was assigned to that location in the **consider** location step. In the following example, no agents become available during the time the vectors are processing the call.

The following example shows a primary vector that checks one remote location to which the vector assigns an adjustment of 30.

Vector with consider step for one location

```
1. wait time 0 secs hearing ringback
2. consider split     pri m  adjust-by  0
```

```
3. consider location 2 adjust-by 30
4. queue-to-best
```

The following example shows the status poll vector at location 2.

Status poll vector

```
1. consider split  2    pri m   adjust-by 0
2. consider split 11    pri m   adjust-by 20
3. reply-best
```

`Consider split/skill` commands in the status poll vectors work the same as in singlesite BSR vectors. User adjustments are applied to a single split or skill and not to the entire location. In this case, the two splits are assigned different adjustments. Say that split 11, despite having the larger adjustment, returns the lower adjusted EWT for a call. The **`reply-best`** command in step 3 returns the user adjustment of 20 to the primary vector on the origin server, along with the rest of the data for split 11.

In saving the data that is returned by location 2, the origin server adds the remote adjustment of 20 to the adjustment of 30 that is specified in step 3 of the primary vector. As a result, the call does not interflow to location 2 in this example unless the EWT for location 2 is more than 50 seconds better than the EWT in split 1 on the origin server.

Example of multisite BSR with limited trunking

Multisite BSR applications must balance improvements in wait times and agent utilization with the cost of interflows and the availability of inter-server trunking for status polls and interflows.

The following example shows an application used for balancing agent workload across the network while still limiting tie trunk usage.

BSR example of Application Plan

```
BEST SERVICE ROUTING APPLICATION PLAN

Number: 10   Name: International       Maximum Suppression Time: 60   Lock? y

Num Location Name    Switch Node           Status Poll    VDN   Interflow VDN  Net Redir

1 Kansas City   1111   919131234015        919131234115          n
2 New York      1112   912121234015        912121234115          n
3 Montreal      1113   915141234015        915141234115          n
3 London        1114   90114411234015      90114411234115        n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
___ _____________      ____  ____   __________  __      _____________          n
```

The following VDN example shows the VDN screen for VDN 51110, the VDN used in the BSR Application Plan example. In the example, the entry in the **BSR Application** field links the VDN to BSR Application Plan 10. Also note the `EAD-MIA` entry in the **BSR Available Agent Strategy** field.

If vector 100 uses BSR commands, calls are not automatically delivered to the first resource found with an available agent. In each consider sequence, when the **queue-to best** or **check best** step executes, one of the following happens:

Condition	Action
No skill has an available agent	The call queues to the skill with the lowest adjusted EWT.
One skill has an available agent	The call is delivered to that skill.
More than two skills have available agents	The call is delivered to the skill with the most expert agent, which is the agent with the lowest skill level.
More than two skills have available agents with the same skill level	The call is delivered to the skills that has the most idle agent.

Note that the **Allow VDN Override** field is set to n. If a second VDN and vector are used to process the call, the EAD-MIA strategy and the application plan specified for VDN 51110 is used.

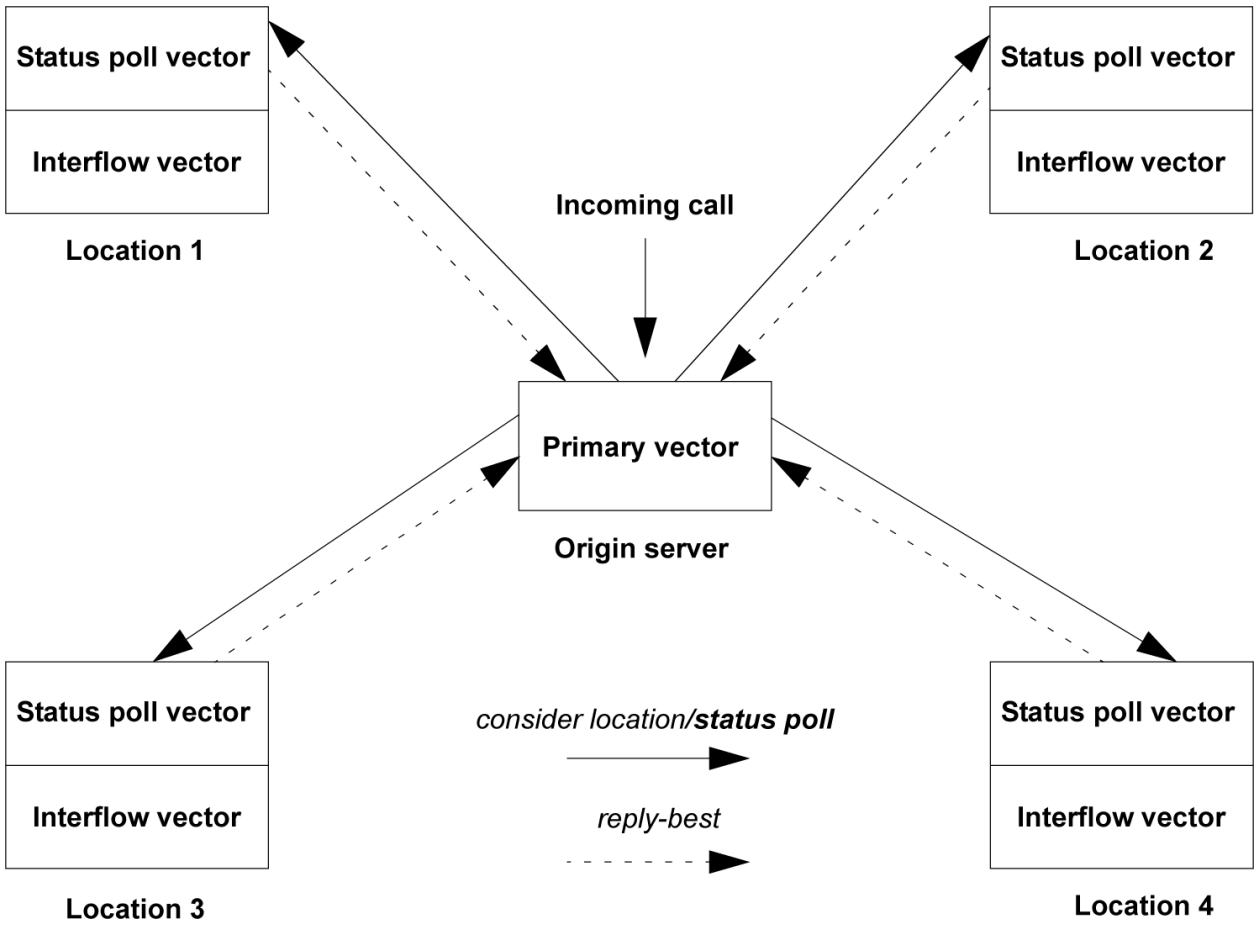
BSR example of primary VDN

```
change vdn xxxxx                                                 page 1 of 3
        VECTOR DIRECTORY NUMBER

                            Extension: 51110
                                 Name: Multisite BSR
                       Vector Number: 100
                   Attendant Vectoring? n
                    Meet-me Conference? n
                    Allow VDN Override? n
                                  COR: 59
                                   TN: 1
                             Measured: none
       Acceptable Service Level (sec): 20
              Service Objective (sec):
        VDN of Origin Annc. Extension: 1001
                            1st Skill:
                            2nd Skill:
                            3rd Skill:
```

```
change vdn xxxxx                                                 page 2 of 3
        VECTOR DIRECTORY NUMBER

                          Audix Name:
                 Messaging Server Name:
                    Return Destination:
                VDN Timed ACW Interval:
                       BSR Application: 15
         BSR Available Agent Strategy: UCD-MIA
             Observe on Agent Answer?: n
```

The overall application is represented in the following figure. Application plan 10 on the origin server identifies the remote servers and provides the digit strings to dial into the VDNs for both the status poll vector and the interflow vector on each server.

Each **consider location** command in the primary vector places a status poll call to its specified location. The status poll vector at that location executes a series of **consider skill** commands and returns data on the best resource to the origin server through a **reply-best** command.



The following example shows the primary vector for this application. The first **consider** series in the primary vector tests two local skills. If either skill has an available agent, step 4 jumps to step 9 and the call queues locally. No remote locations are polled. If no agents are available in either local skill, steps 5 to 8 test four remote locations. In general, do not use other commands between the **consider** steps. The use of the **goto** step is one of the few exceptions to the rule.

If the best remote location's adjusted EWT can reduce the call's current adjusted EWT, step 9 interflows the call to the location. In the vector, a local available agent is always favored over a remote available agent. The call is always directed to the most idle and skilled agent , irrespective of the location that services the call

Multisite BSR example

```
1. wait time 0 secs hearing ringback
2. consider skill 1 pri m   adjust-by  0
3. consider skill 2 pri m   adjust-by 20
4. goto step 9 if expected-wait for best = 0
5. consider location 1      adjust-by 30
6. consider location 2      adjust-by 30
7. consider location 3      adjust-by 50
8. consider location 4      adjust-by 50
9. queue-to best
10. announcement 1001
11. wait time 60 secs hearing music
12. goto step 10 if unconditionally
```

In the primary vector, note that user adjustments are entered for local skill 2 as well as for all the remote locations. The user adjustments indicate the administrator's preferences regarding both local and remote resources. If neither local resource has an available agent, the EWT is greater than 0.

Status poll vector in a multisite BSR application

Each receiving server in a multisite application must have a status poll vector. To collect information from the locations, each **consider location** command in the primary vector places a status poll to the status poll vector for the appropriate server.

The following example shows the status poll vector on the server at location 3.

BSR example of status poll vector at location 3

```
1. consider skill  2   pri m  adjust-by  0
2. consider skill 11   pri m  adjust-by 20
3. consider skill 21   pri m  adjust-by 30
4. reply-best
```

The status poll vector compares skills 2, 11, and 21, identifies the best skill, and sends information back to the origin server through the **reply-best** command. Note that user adjustments are applied to skills 11 and 21 to adjust the EWT. When EWT adjustments are applied at both the origin and remote servers, both the adjustments are added at the origin server.

If skill 11 has the best adjusted EWT at location 3, the skill data, including a user adjustment of 20, is returned to the origin server by the **reply-best** command.

Finding the best resource

Once the remote servers have returned the best data for each location, the second consider series in the primary vector can be completed. In this example, let's suppose that no agents are available at any remote location.

The following table shows how user adjustments at the origin and remote servers yield the adjusted EWT for each location.

Location	Actual EWT of remote best (sec)	User adjustment on origin server	User adjustment on remote server	Adjustment applied by origin server (sec)	Adjusted EWT used in BSR calculations (sec)
1	60	30	0	30	90
2	45	30	10	40	85
3	40	50	20	70	110
4	70	50	0	50	120

The second consider series identifies location 2 as the best remote location, with an adjusted EWT of 85, and the queue-to best step interflows this call to location 2.

Interflow vector in a multisite BSR application

The interflow vector on a remote server in a multisite application accepts the interflowed call from the origin server. The vector also executes the consider series as the status poll vector to identify the current best resource, in case conditions have changed since the status poll.

The following example shows the interflow vector on a remote server.

BSR example of interflow vector at location 2

```
1. consider skill  2     pri m   adjust-by 0
2. consider skill 11     pri m   adjust-by 20
3. consider skill 21     pri m   adjust-by 30
4. queue-to best
```

When a call is interflowed, the call is removed from any queues at the origin server and any audible feedback at the origin server is terminated.

⚠ Caution:

BSR does not operate correctly unless the consider series in the status poll vector and the interflow vector use the same splits/skills with the same queue priorities.

Example of multisite BSR with slow networks

Network response times are not an issue for most users. This example is intended for those users, if any, who experience such a problem. This example uses the same VDN, application plan, and a four-server network that is described in the example of multisite BSR with limited trunking. The vector in the example minimized interflows by using a **goto** step that skips the remote **consider** series if a local resource has an available agent. The design is especially useful if network response times are slow. Calls are always queued locally once before being queued at remote locations.

Both status polls and interflows are conditional. The call can wait in the queue for a local resource while BSR looks for a better split or skill at remote locations.

The example also shows the function of the **check best** command and the wait-improved conditional.

The following example illustrates the primary vector for the application, vector 100. The first consider series in the primary vector tests two local splits and queues the call to the best split. If the EWT for the best split is less than 30 seconds, step 5 jumps to the loop in step 11 and the second consider series is not executed. If the EWT for the best split is more than 30 seconds, steps 6 through 9 test 4 remote locations. If the best remote location reduces the EWT of a call by more than 30 seconds as compared to its EWT in the best local queue, step 10 interflows the call to the location.

⚠ Caution:

Queue calls once before using the wait-improved conditional in a vector step. If calls are not already queued when the step with the wait-improved conditional executes, the server reads the EWT of the call as infinite. This can result in a vector that interflows all calls, even if interflowing not the intended function.

Multisite BSR with EWT

```
1. wait time 0 secs hearing ringback
2. consider skill   1  pri m   adjust-by  0
3. consider skill   2  pri m   adjust-by 20
4. queue-to-best
5. goto step 11 if expected-wait for call <= 30
6. consider location 1         adjust-by 30
```

```
7. consider location 2         adjust-by 30
8. consider location 3         adjust-by 50
9. consider location 4         adjust-by 50
10. check best if wait-improved > 30
11. announcement 1001
12. wait time 60 secs hearing music
13. goto step 11 if unconditionally
```

A **consider** series can end with either a **queue-to best** or a **check best** step. The **check best** command lets you set conditions that must be met before a call is queued to the best resource. In this example, step 10 in the primary vector is **check best if wait-improved > 30**. In other words, step 10 interflows the call to the best location found by the **consider** series only if the EWT for the location is more than 30 seconds better than the EWT of the call in the local queue.

You can use up to 3 **consider** series in one vector. You can write more than 3 **consider** series in a vector, but there is no benefit in doing so. The server only allows you to queue a call simultaneously to 3 different local resources. Since each **consider** series ends by queuing a call, if no agents are available, using more than 3 series in a vector does not place the calls in additional local queues. If the call interflows to another Communication Manager, the call is removed from vector processing and any queues the call was in on the origin server.

You can combine singlesite and multisite **consider** series, as this example shows. Note that user adjustments are entered for local skill 2 as well as for locations 3 and 4. These indicate your preferences regarding both local and remote resources. In this example, say that step 2 queues the call to skill 1, which has an EWT of 65 seconds, before the second **consider** series is executed.

Status poll vector in a multisite BSR with slow networks

Each receiving server in a multisite application must have a status poll vector. To collect information from the locations, each **consider location** command in the primary vector places a status poll to the status poll vector for the appropriate server.

The following example shows the status poll vector on the server at location 3.

BSR example of status poll vector at location 3

```
1. consider skill  2  pri m  adjust-by  0
2. consider skill 11  pri m  adjust-by 20
3. consider skill 21  pri m  adjust-by 30
4. reply-best
```

The status poll vector compares skills 2, 11, and 21, identifies the best skill, and sends information back to the origin server through the **reply-best** command. Note that user adjustments are applied to skills 11 and 21 to adjust the EWT. When EWT adjustments are applied at both the origin and remote servers, both adjustments are added at the origin server.

If skill 11 has the best adjusted EWT at location 3, the skill data, including a user adjustment of 20, is returned to the origin server by the **reply-best** command.

The first **consider** series queues the call to local skill 1. If the second **consider** series identifies location 2 as the best remote resource, the **check** command in step 10 recalculates the current and unadjusted EWT of the call in skill 1. The **check** command compares the calculated value to the unadjusted EWT of location 2. If the call's actual (unadjusted) EWT can be improved by more than 30 seconds, the call is interflowed.

Note:

BSR uses adjusted EWT to determine the best resources in the **consider** series. Once the best resource is identified, subsequent **expected-wait** and **wait-improved** conditionals use the actual EWT values.

Interflow vector in a multisite BSR with slow networks

When a call is interflowed to any of the remote locations, the interflow vector on that server accepts the interflowed call from the origin server. It also executes the same consider series as the status poll vector to identify the current best resource, in case conditions have changed since the status poll. The following example shows such an interflow vector.

BSR example of interflow vector at location 2

```
1. consider skill 2   pri m  adjust-by  0
2. consider skill 11  pri m  adjust-by 20
3. consider skill 21  pri m  adjust-by 30
4. reply-best
```

Caution:

BSR will not operate correctly unless the consider series in the status poll vector and the interflow vector use the same splits/skills with the same queue priorities.

If the call is queued to a remote resource by step 10 in the primary vector, is the call removed from the local queue that it entered in step 4? When a call is interflowed, the call is removed from any queues at the origin server and any audible feedback at the origin server is terminated.

The second consider series can compare local and remote resources. If it does, and if step 10 queues the call to another local skill, will the call be removed from the local queue that it entered in step 4?

No. In general, the server can queue a call to as many as 3 local splits or skills simultaneously. BSR does not change this limit.

Example for handling excessive wait times

Step 4 of the vector directs calls to a **disconnect** step after an **announcement** step when wait time in the network exceeds five minutes.

The following example shows a simple primary vector.

Multisite BSR using disconnect for excessive wait times

```
1. wait 0
2. consider skill    1   pri m   adjust-by 0
3. consider location 2   pri m   adjust-by 30
4. goto step 6 if expected-wait for best ≤ 300
5. disconnect after announcement 3001
6. queue-to best
```

Advanced multisite routing

This section is intended for users whose call center networks meet the following criteria:

- More than five switches in the network.
- Combination of low-volume and high-volume locations.

Application architecture in multisite BSR

Multisite applications can be structured in a variety of ways. In general, however, most applications fit one of two models: distributed or centralized. When each switch in a network interflow calls to other switches and receive interflows, this is called a distributed system. A centralized system, by contrast, is one in which all calls are initially delivered to a single call center, the hub, and distributed from this site to queues at remote switches. A centralized system requires greater inter-switch trunking, since a greater percentage of calls must be redirected. However, if your organization has a significant investment in VRU and CTI technology at the hub, the configuration is appropriate.

Which architecture you choose for an application has direct implications for your choice of user adjustments and polling patterns.

User adjustment considerations

User adjustments in **`consider split/skill`** steps can be set at the user discretion. In distributed multisite applications, however, pay attention to adjustments due to costs and inter-switch trunk capacity. In centralized applications, all calls are redirected so you can use adjustments of zero (0). In distributed applications, a user adjustment of zero (0) for a **`consider location`** step is neither practical nor efficient.

In distributed applications, if the adjustments are small, the load balance across the network is less, but the percentage of calls redirected between switches is high increasing the demands on inter-switch trunking. Higher adjustments reduce interflows, but at the cost of allowing greater imbalance in the load between switches. It takes time and effort to find the best combination of user adjustments in a particular network, but the table contains recommended ranges for initial user adjustments under different conditions. Adjustments can vary between different call center applications so apply the guidelines for each of your applications separately.

Recommended adjustments	Criteria
10-15	• To balance wait times across the network. • Trunk facilities between Communication Manager are plentiful. • Each Communication Manager receives more than 1 call every 10-15 seconds, that is, more than 240-360 calls per hour, for the application.
20	• To balance wait times across the network. • Adequate trunk facilities are available to support the desired balance. • Each Communication Manager receives more than 1 call every 20 seconds, that is, more than 180 calls per hour, for the application.

Table continues…

Recommended adjustments	Criteria
More than 30	• Gains in agent efficiency are more important to you than balancing wait times across the network. • Trunk facilities are scarce. • Call interflow is costly. • Each Communication Manager receives no more than 1 call every 30 seconds, that is, less than 120 calls per hour, for the application.

In the first multisite application, you can begin with a remote adjustment of 30. You can reduce the adjustment later if inter-switch trunking is under-utilized. On the other hand, if trunk exhaustion is a common occurrence, user adjustments are probably set too low. When trunks are exhausted, no further load balancing takes place, deteriorating the overall balance.

Set high user adjustments so that calls are not interflowed to gain the equivalent of a fraction of a queue position. The following equation gives the minimum recommended user adjustment for each remote switch:

$$\frac{\text{AverageCallHandlingTime}}{\text{NumberOfFullTimeEquivalentAgents}} \leq \text{UserAdjustment}$$

Adjustments for remote locations is probably in the range of 10–30 in most distributed applications.

User adjustments and the balance in wait times

Changing conditions can produce significant variations between user adjustments and the balance in wait times across a network, but on average you can predict the balance in wait times for a given user adjustment.

Choose, for instance, a user adjustment of 20 for all remote resources in a network and poll all the remote sites. When waiting time is short, less than 100 seconds, the highest and lowest EWTs for the application on the network stays within a range of approximately 20 seconds. When waiting time is long, more than 100 seconds, the highest and lowest EWTs for the application stays within a range of approximately 20 percent.

Status polling

Status polls are the key element in multisite BSR applications. Status polls provide the communication link between Communication Manager that interflows a call and other Communication Manager that provides service to the call.

The vectors that you write in multisite applications must balance the costs of time and trunk usage with the benefit of better customer service. BSR is designed to help you achieve this balance, incorporating mechanisms to maximize improvements in customer service while minimizing interswitch communications with attendant delays and trunk usage. This section explains the mechanisms and the benefits.

How long do status polls take?

One **consider location** step polls one remote location. Does this mean that an optimal multisite BSR application polls every switch in a network? No.

Let us look at an example of a moderately large network, containing 16 switches. The primary vector on switch 1 can be written as shown in the following vector example. Polling response times are variable. If this is a slow response network and each status poll takes 1 second, the **consider** series in this vector adds as much as 15 seconds to a time of a call in vector processing. In fact, the vector in the example of what NOT to do.

Intelligent polling for multi switch networks

```
1. wait time 0 secs hearing ringback
2. consider skill     1 pri m  adjust-by  0
3. consider skill     2 pri m  adjust-by 20
4. goto step 20 if expected-wait for best = 0
5. consider location 1         adjust-by 30
6. consider location 2         adjust-by 30
7. consider location 3         adjust-by 30
8. consider location 4         adjust-by 30
9. consider location 5         adjust-by 30
10. consider location 6        adjust-by 30
11. consider location 7        adjust-by 30
12. consider location 8        adjust-by 30
13. consider location 9        adjust-by 30
14. consider location 10       adjust-by 30
15. consider location 11       adjust-by 30
16. consider location 12       adjust-by 30
17. consider location 13       adjust-by 30
18. consider location 14       adjust-by 30
19. consider location 15       adjust-by 30
20. queue-to best
21. announcement 1001
22. wait time 60 secs hearing music
23. goto step 21 if unconditionally
```

First, even in very large networks you can obtain nearly all of the possible benefits in agent utilization with very few polling connections. In a network of 16 switches, 99 percent of the total benefits possible with BSR can be obtained if each switch polls just 4 others.

Now our vector looks like the following. Is polling time now cut from 15 seconds to 4 seconds, proportional to the reduction in consider steps?

```
1. wait time 0 secs hearing ringback
2. consider skill     1 pri m   adjust-by  0
3. consider skill     2 pri m   adjust-by  0
4. goto step 9 if expected-wait for call = 0
5. consider location 5          adjust-by 30
6. consider location 10         adjust-by 30
7. consider location 13         adjust-by 30
8. consider location 15         adjust-by 30
9. queue-to best
10. announcement 1001
11. wait time 60 secs hearing music
12. goto step 10 if unconditionally
```

In fact, polling time in this vector can be around 0.4 seconds per call because of mechanisms in BSR that constantly react to network conditions and resource usage to minimize the number of status polls. The mechanisms, whose combined operation is called “Intelligent Polling”, also function to make each status poll as productive as possible.

Intelligent polling

A BSR application only polls the switches that are likely to provide the best service at any given time. If a remote switch is polled and returns an adjusted EWT greater than that of the current best resource, polling of the remote switch is suppressed for a period of time proportional to the difference between the two adjusted EWT values. In other words, polling of a given location is suppressed whenever the adjusted EWT returned by that location is subsequently replaced by a better adjusted EWT from another resource. The **consider** step for this location is skipped during the period and vector processing continues at the next step. When the suppression period is over, the **consider** step once again polls the location. If the location returns the best adjusted EWT, the next call processed by the vector also causes the location to be polled. If the location is not the best, polling is temporarily suppressed.

If no calls are in queue at the remote location an agent can become available at any moment, and BSR, therefore, does not suppress polling for longer than 5 seconds in such situations. BSR does not suppress polling of any remote location for more than 60 seconds, regardless of the differences between adjusted EWT returned by different switches.

Other conditions can also suppress status polls to a location:

- Resource exhaustion, that is, no trunks available, queue full.
- Administration errors, that is, badly written vectors, or no application plan.

This feature significantly reduces the average number of status polls placed per call. The greater the call volume, the greater the percentage reduction. Let's take another look at the vector in screen 2.

Let us say the network is operating in a balanced state. EWTs are 30 seconds at all locations and a call arrives every 3 seconds at each site. Adjusted EWTs are 30 seconds at the origin switch and 60 seconds for each remote switch. After each status poll under the conditions, polling is suppressed for 30 seconds. Each remote location is polled, therefore, by every 10th call. On average, this means that each call polls any one location 0.1 times. Since there are four **consider** steps, each call makes 0.4 polls. Remembering the 1-second polling response time given at the beginning of the example, the average time added to call processing for each call is 0.4 seconds.

The 1st-found available agent strategy, discussed in BSR can cut average polling times further. With the 1st-found strategy, BSR skips all subsequent **consider** steps in a series if a resource with an available agent is found and deliver the call to that resource.

Efficient polling patterns in large networks

Unless you have a small network, you cannot benefit by having every switch poll every other switch. This section explains how many remote locations each switch polls and provides guidelines for selecting which locations any given switch must poll.

How many switches must one switch poll?

You do not have to poll every Communication Manager in larger networks. Because of the intelligent polling capabilities of BSR, you can obtain 99 percent of the possible benefits in agent utilization with very few polling connections.

The following example is a laboratory network of 16 Communication Manager used for simulations of BSR multisite applications.

As shown in the following table, approximately 99 percent of the possible benefits are obtained when any one Communication Manager polled four others.

Number of remote sites polled by each switch	ASA across the network (seconds)	Approximate percentage of total benefits obtained (%)
0	192.8	0
1	26.2	89
2	10.6	95
3	7.6	98
4	6.5	99
15	4.7	100

For each Communication Manager to poll the other 11 Communication Manager in the network produces an additional percent gain in ASA and agent utilization, an improvement which is more than offset by the cost of additional messaging and trunking.

In most situations, you can get the optimal results with your multisite BSR applications if you follow the polling guidelines shown in the following table.

Number of switches in the network	Number of switches to poll
2-4	all the other switches
5-10	3 other switches
11-20	4 other switches
21-40	5 other switches
More than 41	6 other switches

Which remote switches must each switch poll?

In networks with less than five switches, each switch can productively poll all the other switches in the network. In larger networks, each switch does not poll every other switch. But which switches must each switch poll? The term *polling patterns* describes the relationships between switches in multisite BSR applications.

Do not use the following polling patterns:

- Mutual polling: As much as possible, 2 switches must not poll each other. This is difficult in small networks, but in large networks mutual polling can and must be minimized.
- Polling chains: For example, if switch A polls B and C, B polls C and D, this is a polling chain.

You can experiment with polling patterns appropriate for your network and applications, if you are not constrained by the physical structure of the network. The following table provides a template for creating polling patterns for applications of up to 12 switches. In the majority of situations, the patterns produce results that are close to optimal. To use the following table, first assign a number from 1 to x to each switch in your application. Next, find the column that matches the number

of switches in your application. As you read the column, see which switches each switch in the application must poll.

Switch	Must poll the specific switches shown in the column for your network size							
	5	6	7	8	9	10	11	12
1	2,4,5	2,4,5	2,4,6	2,4,7	2,4,6	2,4,7	2,4,8,10	2,4,8,9
2	3,5,1	3,5,6	3,5,7	3,5,8	3,5,7	3,5,8	3,5,9,11	3,5,9,10
3	4,1,2	4,6,1	4,6,1	4,6,1	4,6,8	4,6,9	4,6,10,1	4,6,10,11
4	5,2,3	5,1,2	5,7,2	5,7,2	5,7,9	5,7,10	5,7,11,2	5,7,11,12
5	1,3,4	6,2,3	6,1,3	6,8,3	6,8,1	6,8,1	6,8,1,3	6,8,12,1
6		1,3,4	7,2,4	7,1,4	7,9,2	7,9,2	7,9,2,4	7,9,1,2
7			1,3,5	8,2,5	8,1,3	8,10,3	8,10,3,5	8,10,2,3
8				1,3,6	9,2,4	9,1,4	9,11,4,6	9,11,3,4
9					1,3,5	10,2,5	10,1,5,7	10,12,4,5
10						1,3,6	11,2,6,8	11,1,5,6
11							1,3,7,9	12,2,6,7
12								1,3,7,8

In applications of more than 12 switches, the following table provides the formulae you must figure out the optimal polling pattern.

Number of switches in application	Switch to poll
13 or 16	i + 1, i + 3, i + 7, i +11
14 or 19	i + 1, i + 3, i + 7, i + 9
15	i + 1, i + 3, i + 7, i + 10
17 or 20	i + 1, i + 3, i + 7, i + 12
18	i + 1, i + 3, i + 7, i + 13
21-23	i + 1, i + 3, i + 7, i + 15, i + 17
24	i + 1, i + 3, i + 7, i + 15, i + 19
25	i + 1, i + 3, i + 7, i + 15, i + 20

To use one of the formulae, first assign a number from 1 to x to each switch in your application. Then, in the left-hand column of the table, find the number of switches in your application. The corresponding formula in the right-hand column is the one you must use.

In the formulae, i is the number of the switch for which you are calculating a polling pattern. To calculate the polling patterns in an application with 16 switches, the formula to use is: i + 1, i + 3, i + 7, i +11

As shown in the first row of the table. Here are the actual results of this formulae for the first 5 switches in this 16-switch application. The numbers wrap, that is, start over at 1, after you have

polled the last switch in the network: switch 5 polls switch 16 as its fourth poll and then the polling pattern for switch 6 has switch 1 in the fourth position.

Switch number	Switch to poll
1	2 ,4, 8, 12
2	3, 5, 9, 13
3	4, 6 ,10, 14
4	5, 7, 11, 15
5	6, 8, 12, 16
6	7, 9, 13, 1
7	8, 10, 14, 2

Minimizing variations in wait time

When a network contains large and very small resources, BSR and LAI can be effectively combined. This section presents two sample vectors. The first example illustrates a primary vector intended for the small resources in a network wherein a call in queue at one call center when an agent is available at another makes no sense. The design reduces wait time variation as well. The second example illustrates the best way to minimize wait times across a network with large resources.

LAI as a backup

If your principal concern is that a call should not wait in queue while an agent is available elsewhere, use BSR at all locations in the network. At smaller locations, write primary vectors that will perform LAI attempts to other preferably larger resources once the call has been queued.

```
1.  wait time 0  secs hearing ringback
2.  consider skill 1st pri m adjust-by  0
3.  consider location 12     adjust-by 30
4.  consider location 22     adjust-by 30
5.  goto step 7 if expected-wait for call < 600
6.  disconnect after announcement 3501 "Due to heavy call volume..."
7.  queue-to skill best
8.  announcement 3500 "Thanks for calling...."
9.  goto step 13 if expected-wait for call < 90
10. wait time 45 secs hearing music
11. announcement 3502 "Still busy..."
12. goto step 9 if unconditionally
13. route-to-number 913031234567 with cov n if interflow-qpos = 1
14. wait time 5 secs hearing music
15. goto step 13 if unconditionally
```

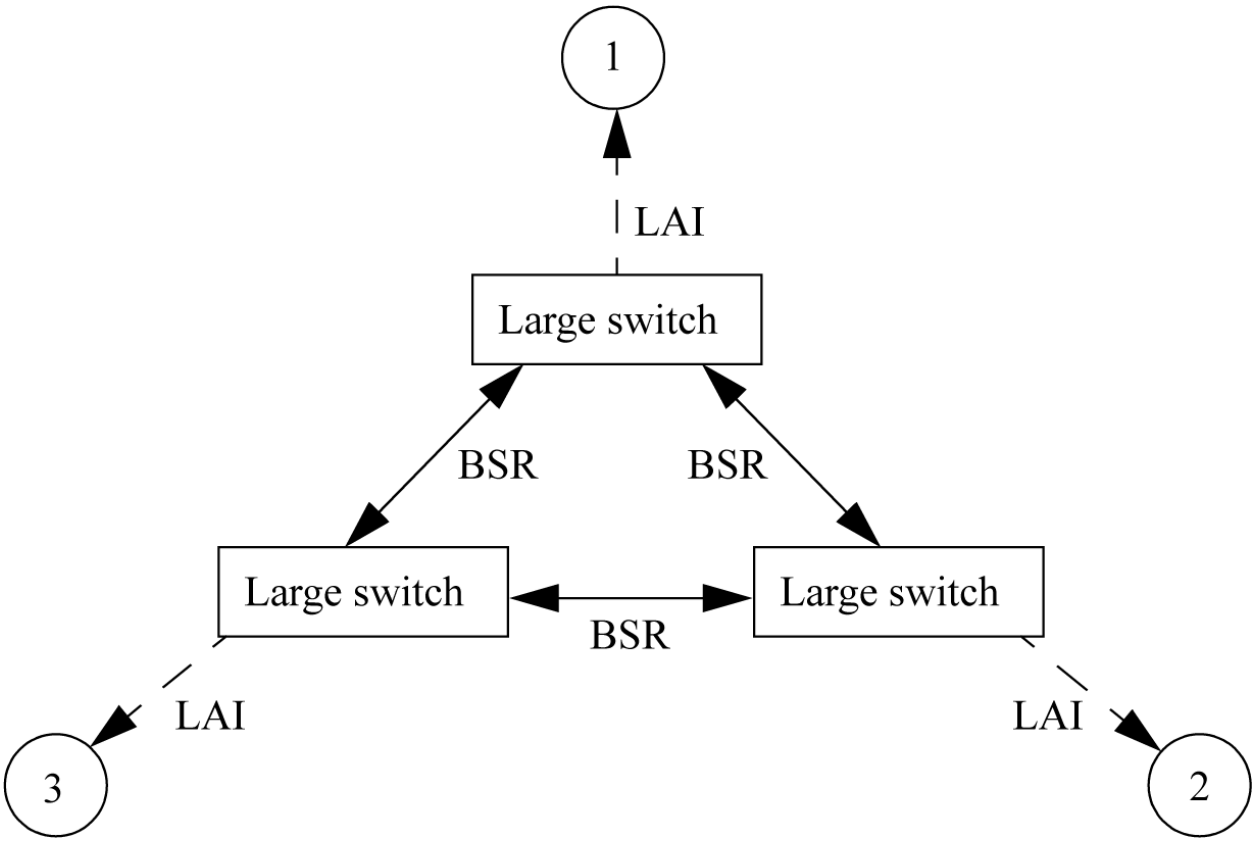
Steps 1 to 4 comprise a typical BSR vector. The origin Communication Manager checks a local resource and 2 remote resources. Before queuing or routing the call, however, the vector checks the expected wait time for the best resource. If this is more than 10 minutes, the caller receives a busy announcement. Otherwise, the queue-to best step sends the call to the best resource. Two vector loops follow: one 45-second loop with music and a delay announcement, and one 5-second loop that uses LAI. If the call is queued successfully in step 7 the first announcement loop (steps 9-12) executes until the call gets within a certain range of the head of the queue (at which point EWT is less than 90 seconds). At this time, step 9 sends the call to the second loop, where LAI attempts are placed every 5 seconds for the call at the head of the interflow eligible queue

(**interflow-qpos=1**). If an agent becomes available at the larger remote resource, any call at the head of the eligible queue at the smaller location is outflowed to the larger resource, normally within a period of 5 seconds.

Single-queue FIFO hybrid configuration

To minimize variations in wait time across a network, the best strategy is to let only the call centers with the larger resources receive calls. The following figure shows a network of three large and three small resources, that is, call centers with large skills and call centers very small skills in the same application.

The large locations use BSR and all poll each other, while each location with a small resource (numbered 1, 2, 3) is treated as a satellite of one of the larger locations and receives calls interflowed from the larger location. Mutual polling is not optimal in larger networks, but work well for switches in such a small network to poll each other. BSR is used to balance the load between the locations with the larger resources. Each large switch executes a rapid Look-Ahead Interflow (LAI) vector loop to one small switch to look for available agents. Since calls never queue at the small switches, the problem of highly variable wait times at the small resources is eliminated. This strategy will also give the best balance in wait times across resources.



The following vector example shows the primary vector that is used at the large locations with this strategy. This vector is almost identical to the vector shown in *using LAI as a backup*. The difference is at the application level. In contrast to the previous example:

- Only locations with the larger resources receive calls.

- The primary vector in this example resides on larger switches.

Steps 1 to 4 comprise a typical BSR vector. The origin Communication Manager checks a local resource and 2 remote resources. Before queuing or routing the call, however, the vector checks the expected wait time for the best resource. If this is more than 10 minutes, the caller receives a busy announcement. Otherwise, the queue-to best step sends the call to the best resource. Two vector loops follow: one 45-second loop with music and a delay announcement, and one 5-second loop that uses LAI. If the call is queued successfully in step 7, the first announcement loop (steps 9-12) executes until the call gets within a certain range of the head of the queue. At this time, step 9 sends the call to the second loop, where LAI attempts are placed every 5 seconds (only for the call at the head of the interflow eligible queue). If an agent becomes available at the smaller resource, any call at the head of the eligible queue at the larger location is outflowed to the smaller resource, normally within a period of 5 seconds.

Vector combining BSR and LAI

```
1.  wait time 0  secs hearing ringback
2.  consider skill 1st pri m adjust-by  0
3.  consider location 120    adjust-by 30
4.  consider location 220    adjust-by 30
5.  goto step 7 if expected-wait for best < 600
6.  disconnect after announcement 3501 "Due to heavy call volume..."
7.  queue-to skill best
8.  announcement 3500 "Thanks for calling...."
9.  goto step 13 if expected-wait for call < 90
10. wait time 45 secs hearing music
11. announcement 3502 "Still busy..."
12. goto step 9 if unconditionally
13. route-to-number 913031234567 with cov n if interflow-qpos = 1
14. wait time 5 secs hearing music
15. goto step 13 if unconditionally
```

Similar vector loops can be added to the interflow vectors at each of the large switches. In other words, each vector that processes calls at the larger locations can use rapid LAI loops to interflow calls to its satellite resource. This system maximizes agent utilization and the distribution of call load while evening out wait times across the network.

Low volume splits or skills

Very small resources, for example, 2-3 agents, have special needs. With BSR, you can maintain a balance of wait times across a network of call centers. However, for very small splits or skills, wait times for each call can vary significantly.

To see why this is the case, let us take an extreme example of a split with a single agent logged in with one call active and no call in the queue. Average call handling time is 3 minutes. Now, if a new call arrives in the queue, that call can be answered almost immediately or the call can be in the queue for more than 3 minutes. The variation in wait times is perhaps 5-180 seconds.

In general, the fewer agents logged in to a split or skill, the greater the variability in wait times because agents become available less often. BSR favors large resources, steering calls away from smaller resources when there are no available agents or wait times are not the best in the application. This tendency helps reduce the possibility that an individual caller can have a disproportionately long wait at a small resource.

If your network includes very small splits or skills, you have three options:

- If the operation is not badly affected by a small percentage of calls having variable wait times, use BSR across the network.
- If your principal concern is that a call does not wait in a queue while an agent is available elsewhere, use BSR, but write primary vectors at smaller locations to perform rapid look-ahead attempts to other resources once the call is queued. Rapid LAI vector loops use the **interflow-qpos** conditional, which is an enhancement to LAI.
- If you want agents to answer each call quickly, use the following configuration: Do not deliver or queue calls directly to the very small resources. Deliver or queue all incoming calls to larger resources and use BSR to balance the load across the larger locations. Ensure most of the larger locations perform rapid look-ahead attempts to more than one of the smaller resources. In this way, the members of the very small resource become an extension of the agent pool at one of the larger call centers.

In any network, do not have several large resources poll or make look-ahead attempts to a very small resource. As the status at the very small resource changes infrequently, frequent polls to that resource is a waste. A very small resource must receive look-ahead attempts or be polled only by other small resources or by one large resource.

BSR local treatment for calls queued remotely

As the **converse-on** vector step does not include any information on the queue position and the wait time for the call at the remote end, Voice Portal Call Back Assist does not work with BSR Local Treatment.

In a multisite BSR configuration, a call that arrives at a local Communication Manager server can be rerouted to a remote server located in a another part of the world. With BSR local treatment for calls queued remotely over IP (SIP or H.323) or ISDN trunks, you can provide local audio feedback to calls while calls wait in a queue on a remote server.

The feature provides the following benefits:

- Bandwidth savings for IP calls.
- No audio quality concerns that occur when music is sent over WAN that use low bit-rate codecs.
- Centralized management for announcements and other call treatment.

Local treatment operations

This section describes local treatment feature operations that occur when BSR redirects IP or ISDN calls to a remote queue.

Important:

Before following the local treatment operations, ensure that the required feature and vector administration steps are implemented on the local *and* remote Communication Manager servers.

The following steps describe the basic process for local treatment operations in a multisite BSR environment:

1. A call arrives at the local Communication Manager server and is processed by a VDN with BSR local treatment.
2. The local vector includes the **consider**, **queue-to best**, and **wait** hearing announcement steps that are required for BSR local treatment operations.
3. A skill on a remote server is identified as the best location and the local server attempts an interflow to the remote server. Vector processing is temporarily suspended on the local server while the interflow attempt is in progress.
4. If the interflow attempt succeeds, the remote server returns an ISDN_PROGRESS message with progress indicator of in-band information (8) or a SIP 182 QUEUED message, to indicate that the call is in a queue and local treatment operations can proceed.

   The remote server must meet the following requirements for the appropriate ISDN_PROGRESS message to be sent back to the local server:

   - The remote server is administered for BSR local treatment.
   - The call is directed to a VDN with BSR local treatment.
   - The vector associated with the VDN includes only those steps and commands that are required for successful local treatment operations.
5. The local server receives a SIP 182 QUEUED or an ISDN_PROGRESS message with progress indicator of in-band information (8), vector processing resumes with an appropriate treatment step and the caller receives feedback provided by the local server while the caller waits in the remote queue.

   Important:

   To ensure that the local treatment feature operates as designed, use the vector commands that are recommended for local treatment implementation. Although local treatment operations do not impose restrictions on the types of vector steps that are administered on the local server after call processing resumes, use of inappropriate vector steps can interfere with local treatment operations.
6. When an ACD agent on the remote server accepts the call, a 200 OK or an ISDN_ALERTING message is sent to the local server. Vector processing is discontinued on the local server and the remote server.

Local treatment system requirements

The BSR Local Treatment for Calls Queued Remotely Over IP or ISDN Trunks feature works on all platforms and operating systems that are supported by Avaya Communication Manager. To use the local treatment feature, the system license file must be configured to enable the following features:

- LAI
- BSR

- BSR Local Treatment for IP and ISDN

Implementation guidelines for local treatment vectors

Important:

Read the guidelines before you implement the local treatment feature.

Implementation of the local treatment feature requires use of specific vector steps to generate the correct ISDN or SIP messages between the local and remote Communication Manager.

For polling vectors: You must be careful to administer your local treatment polling vectors so that calls are not unintentionally dropped or phantom calls are generated. If the **queue-to best** step is followed by vector steps that include any commands other than **announcement**, **wait**, or **goto**, the trunk to the remote queue are dropped. For example, the addition of **consider** steps after a **queue-to best** command can cause intermittent call behavior. The addition of a **queue-to** step after a **queue-to best** step can cause phantom calls to be queued to the remote server.

Tip:

You can also exploit this functionality to allow the local server to “take back” calls that remain in queue on a remote server after a specified time limit is exceeded. For more information, see the “Take back example” in the example vector for local Communication Manager.

Interflow local treatment vectors on the remote Communication Manager: When the BSR Local Treatment feature is enabled, specific ISDN or SIP messages must be exchanged between the remote and local Communication Manager. If additional vector steps are included either before or after the **consider** steps and **queue-to best** in the interflow vector on the remote server, the following results occur:

- Either an ALERTING or PROGRESS message, with in-band information, is returned from the remote server to the local server.
- In response to the message, trunk bandwidth is immediately allocated and the call is removed from the local queue.
- Local treatment operations cease, trunk bearer resources are allocated for the call sooner than required and cost savings associated with the local treatment feature are not realized.

Example vectors for the local treatment feature

This section provides vector guidelines and examples that describe how to implement the local treatment feature. Vector administration typically requires polling vectors on both the local and remote Communication Manager and an interflow vector on the remote server. You must also administer the polling vector on the local server to provide an appropriate local call treatment.

Example vectors for the local Communication Manager

The following examples shows two different vector strategies to implement the local treatment feature on the local server. Vectors created for this purpose are the same as those used in all BSR polling operations, which include a **consider** series followed by a **queue-to best** step.

Important:

Administer the local treatment polling vectors so that calls are not dropped unintentionally.

After various skills and locations are polled and the call is placed in queue at the identified best location, the local server continues to maintain control of the call until the call is answered by an agent. While the call is in queue, the local server continues to provide additional vector steps to implement the local call treatment.

At a minimum, the local treatment vector must include **announcement** and **wait-time** steps to provide appropriate feedback to the caller. However, the local treatment vector can be designed to use either a continuous loop or take back strategy. The alternate local call treatment strategies are described in the following sections.

Continuous loop example

The following example shows a vector that provides a sequence of call treatment steps on the local server that proceed in a continuous loop until an agent answers the call at the remote location.

In the following vector example, step 6 places the call in queue at the identified best location. Step 7 provides an appropriate announcement and step 8 provides 10 seconds of music. Step 9 uses an unconditional **goto** step to loop call processing back to step 6, where the treatment process continues.

```
change vector 40                                                    Page 1 of 3
                                  CALL VECTOR

  Number: 40                    Name: Local BSR vector
                  Attendant Vectoring? n       Meet-me Conf? n           Lock? n
    Basic? y    EAS? y   G3V4 Enhanced? y     ANI/II-Digits? y   ASAI Routing? y
Prompting? y    LAI? y  G3V4 Adv Route? y      CINFO? n BSR? y       Holidays? y

01 announcement 3000
02 consider skill 4 pri m adjust-by 0
03 consider skill 6 pri m adjust-by 0
04 consider location 1 adjust-by 10
05 consider location 2 adjust-by 10
06 queue-to best
07 announcement 3001
08 wait-time 10 secs hearing music
09 goto step 7 if unconditionally
```

Take back example

The previous example set up the local treatment process as a continuous loop that repeats indefinitely while the call remains in queue at the identified best location. However, you can also design vectors that allow the local server to take back a call after the call remains in queue for a specified time.

In the following vector example, the **queue-to best** in step 6 is followed by a series of **announcement** and **wait-time** commands provided in steps 7 through 12. If the treatment steps complete and the call still remains in the remote queue, vector processing proceeds to step 13, which uses a **route-to** command that causes the call to the remote server to be dropped. The **route-to** step can be used to provide alternate services for the call.

Note:

When the call to the remote server is dropped, a type 305 vector event is logged.

```
change vector 40                                                    Page 1 of 3
                                  CALL VECTOR

Number: 40                    Name: Local BSR vector
                 Attendant Vectoring? n         Meet-me Conf? n           Lock? n
    Basic? y    EAS? y   G3V4 Enhanced? y      ANI/II-Digits? y   ASAI Routing? y
Prompting? y    LAI? y  G3V4 Adv Route? y       CINFO? n BSR? y       Holidays? y

01 announcement 3000
02 consider skill 4 pri m adjust-by 0
03 consider skill 6 pri m adjust-by 0
04 consider location 1 adjust-by 10
05 consider location 2 adjust-by 10
06 queue-to best
07 announcement 3001
08 wait-time 10 secs hearing music
09 announcement 3001
10 wait-time 10 secs hearing music
11 announcement 3001
12 wait-time 10 secs hearing music
13 route-to number 54010 if unconditionally
```

For another method to take back the call based on the time the call has been in the system, see “VDN type variable” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Related links

BSR call interflow with SIP on page 309

Example polling vector for the remote Communication Manager

The following example shows a call vector that polls skills on the remote server. This vector does not differ from other typical BSR polling vectors.

```
change vector 31                                                    Page 1 of 3
                                  CALL VECTOR

Number: 31                    Name: Remote BSR poll vector
                 Attendant Vectoring? n         Meet-me Conf? n           Lock? n
    Basic? y    EAS? y   G3V4 Enhanced? y      ANI/II-Digits? y   ASAI Routing? y
Prompting? y    LAI? y  G3V4 Adv Route? y       CINFO? n BSR? y       Holidays? y

01 consider     skill    3   pri m adjust-by 0
02 consider     skill    4   pri m adjust-by 0
03 reply-best
```

Example interflow local treatment vector for the remote Communication Manager

The following example shows a call vector that is used to interflow the call to the remote server while local treatment is provided for the call.

Important:

When the BSR Local Treatment feature is enabled, specific ISDN messages must be exchanged between the remote and local Communication Managers. If additional vector steps

are included either before or after the consider steps (if used) and queue-to best in the interflow vector on the remote server, the following results occur:

- Either an ALERTING or PROGRESS message (with in-band information) is returned from the remote server to the local server.
- In response to the message, trunk bandwidth is immediately allocated and the call is removed from the local queue.
- Local treatment operations cease, trunk bearer resources are allocated for the call sooner than required and cost savings associated with the local treatment feature are not realized.

```
change vector 32                                                    Page 1 of 3
                                    CALL VECTOR

Number: 32                    Name: Remote BSR interflow vector
                 Attendant Vectoring? n         Meet-me Conf? n           Lock? n
    Basic? y    EAS? y   G3V4 Enhanced? y      ANI/II-Digits? y   ASAI Routing? y
Prompting? y    LAI? y  G3V4 Adv Route? y       CINFO? n BSR? y       Holidays? y

01 consider     skill    3   pri m adjust-by 0
02 consider     skill    4   pri m adjust-by 0
03 queue-to     best
```

BSR-initiated path replacement for calls in vector processing

You can use QSIG or DCS with Reroute using ISDN SSE for path replacement for calls in a queue. For calls that are waiting in a queue or in vector processing, even if the call is not connected to an answering user, Communication Manager can attempt a path replacement to find the optimum path for the call.

The **queue-to best** command is used in BSR to initiate a QSIG path replacement for a call. The following scenario can take place:

If the terminating Communication Manager server receives a *Path Replacement Propose* operation for a call that is in a queue or in vector processing, Communication Manager immediately initiates path replacement using the path replacement extension if the **Path Replace While in Queue/Vectoring** field is y and the **Path Replacement Extension** field has a valid entry. The fields are available on the ISDN Parameters page of the Feature-Related System Parameters screen.

Example BSR vector written to trigger path-replacement

The following example shows how a BSR vector can be written to trigger path-replacement at the terminating Communication Manager.

Note:

In order for a path-replacement to be attempted, the incoming and outgoing trunks that are used for the call must be administered with the **Supplementary Service Protocol** field set to b.

BSR-initiated path-replacement vector

```
1. wait 0
2. consider skill 1
3. consider skill 5
```

```
4. consider location 10    adjust-by 10
5. consider location 24    adjust-by 20
6. queue-to best
```

At the receiving server, the vector that is executed by the incoming call must be programmed with an **announcement**, or **wait hearing music** vector command. The use of one of the commands is what makes it possible for path-replacement to take place while the call is in vector processing.

BSR interactions

Agent phone display

If Communication Manager forwards the collected digits with an interflowed call, the forwarded digits are displayed on the phones display of the agent who answers the call.

Look-Ahead Interflow (LAI)

Restrictions and interactions that apply to LAI also apply to BSR status poll and interflow calls.

Basic Call Management System (BCMS)

BCMS does not log LAI attempts. Therefore, BCMS does not log BSR status polls, which are treated as LAI attempts.

Call Vectoring

The following considerations apply to all vectors when you set **BSR** to y.

Call Vectoring considerations	
route-to VDN	If a call is routed to a new VDN, Communication Manager clears the best resource data defined by a series of consider steps in the previous VDN.
goto vector	If a **goto vector** command is executed, Communication Manager retains the best resource data defined by a series of consider steps in the original VDN and uses the data in the subsequent vector.
consider	• Do not use other commands within a series of **consider** steps, since the commands can delay the execution of the series. • Skills used in **consider** commands must be vector controlled.
converse	Communication Manager passes the collected digits that Communication Manager forwards with the call to the VRU using the digits data passing type.
best (keyword)	You can use the **best** keyword in the following commands, but only with the conditions listed: • **goto step** or **goto vector** commands using the expected-wait or wait-improved conditionals. • **check** commands using the unconditional, expected-wait, or wait-improved conditionals. You cannot use the **best** keyword as a replacement for a split or a skill in the following vector commands: • **converse-on split/skill** • **messaging split/skill**

Direct Department Calling (DDC)

BSR functions when the splits use DDC call distribution. Once Communication Manager determines the best resource, the actual call distribution follows the DDC setting of the split irrespective of the BSR available agent strategy. You cannot use DDC as a BSR available agent strategy.

Distributed networking using QSIG Manufacturers Specific Information (MSI)

BSR does not function with systems from other vendors, unless the vendor develops a corresponding capability that works with Communication Manager.

Expert Agent Selection (EAS)

EAS must use the EAD-MIA or EAD-LOA available agent strategy. You can use the EAS VDN skills, that is, first, second, or third in the **consider skill** commands.

Facility Restriction Levels (FRL)

The FRL applies to status poll and interflow calls in the same way FRL works with the **route-to number** command.

Integrated Services Digital Network (ISDN)

BSR and globally supported information transport are fully functional over ISDN-PRI or ISDN-BRI trunking facilities.

Note:

You can set up Asynchronous Transfer Mode (ATM) and IP trunking to emulate ISDN-PRI.

Location Preference Distribution

Use location preference distribution to select an available agent within the call center during the **consider** and **queue-to best** step operations. You cannot use local preference distribution across system sites. In this case, there is no multisite network region.

Look Ahead Routing (LAR) - BSR incompatibility

LAR and BSR are incompatible. If a trunk is not available at the site being polled, an alternative route using an Alternate Route Selection (ARS) pattern can be used to poll, provided there is a secondary route available that supports transporting shared UUI in the DISCONNECT message. This does not use LAR. If no route is available for polling when a **consider location** step is executed, BSR processing handles the situation and after a period of 30 seconds, subsequent calls attempt to poll the location again.

The use of alternative routes for polling only works if there are alternative routes for the interflow path, irrespective of whether LAR or BSR is in use.

Multiskill queuing

A call can be queued up to three times by **queue-to** or **check** commands in the same vector. One vector can therefore contain up to three series of **consider** steps. Each series must be followed by a **queue-to best** step. Each **consider** series selects the best remote resource from the options you specify and queues the call to the resource.

BSR can queue simultaneously only on the origin Communication Manager. BSR no longer controls a call once BSR queues the call at a remote resource.

Network access

BSR interflow operates over public, private, or virtual private ISDN-BRI and ISDN-PRI networks that meet the criteria explained in network requirements. BSR interflow can also operate over SIP trunks. BSR requires the network support transport of user-to-user data using MSI or UUI as a codeset 0 Information Element (IE), or as shared UUI over SIP trunks. The numbers administered on the BSR Application Plan screen are expected to access VDNs using ISDN, H.323, or SIP trunks, or using the Network Call Redirection (NCR) feature.

Administration or call processing does not prevent access to other types of trunks or to destinations that are not VDNs.

Operating Support System Interface (OSSI)

The new administration commands, conditionals, keywords and screens are available using OSSI.

Path replacement for QSIG/DCS ISDN calls

For calls waiting in queue or in vector processing, even if the call is not connected to an answering user, path replacement can be attempted to find an optimal path for the call. This results in more efficient use of the trunk facilities.

The QSIG ISDN or DCS ISDN trunk path-replacement operation can be triggered for ACD calls by the LAI **route-to number** vector step, BSR **queue-to best** vector step, and the adjunct routing vector steps.

QSIG

LAI, BSR, and information forwarding function over QSIG trunk facilities if the remote locations are Communication Manager.

Redirection on No Answer (RONA)

Calls redirected to a VDN by RONA can be subsequently processed by BSR or LAI applications. When RONA redirects a call to a VDN, any best resource data defined in a previous vector is cleared.

Service Level Maximizer (SLM)

The following interactions occur between BSR and SLM:

- The SLM algorithm applies only within a particular call center location, not across locations in a multisite configuration.
- Assignment of reserve agents applies only to skills within a local site.
- SLM always selects the agent for an SLM skill at the remote site. BSR uses SLM to determine the best available agent and when to route the call to that skill.
- The best skill selected at a particular site or across sites when due to multiple consider steps is based on an existing BSR operation. In other words, the shortest adjusted EWT or skill as defined by the available agent strategy.
- The selection of the agent and delivery of the call in the best-chosen skill is based on what is assigned to the skill.
- BSR does not override the skill distribution algorithms and pick a reserve agent, unless the skill distribution algorithm selects that agent due to the current conditions at that site.

Service Observing

You can observe a call in BSR or LAI processing as long as the call is still connected through the local communication server. All current restrictions on Service Observing still apply.

Transfer

If a call is transferred to a VDN, any best resource data defined in previous vector processing is cleared. Transferred calls do not forward any information such as previously collected digits and In-VDN time, that is forwarded with interflows.

Trunk Access Code (TAC)

Do not use routing numbers, that is, status poll or interflow, that utilize TACs, since the required in-band outpulsing slows the setup operation significantly.

VDN Override

VDN Override applies to the **BSR Application Number** and the **Available Agent Strategy** fields assigned on the VDN screen. VDN Override also applies to the VDN name forwarded using Information Forwarding. When a **consider** step is executed, the application number and available agent strategy assigned to the active VDN for the call is used.

VDN Return Destination (VRD)

Communication Manager clears the best resource data for a call when the call leaves vector processing and is not available if the call returns to vector processing.

BSR polling over IP without the B-channel interactions

- If you do not use a B-channel trunk facility, CMS records will poll attempts, but not actual trunk measurements.
- If you assign trunks, you can use the trunk group for H.323 IP voice calls, but you must install an IP media processor.

Holiday Vectoring

Holiday Vectoring simplifies vector writing for holidays. With Holiday Vectoring, you can route calls or provide special handling for date-related calls on a regular basis. You can administer as many as 999 different holiday tables and use the tables to make vectoring decisions. Each table can have up to 15 dates or date ranges. You can administer the Holiday Vectoring capabilities in advance to ensure seamless call routing during holidays when staffing reduces, or when call centers close.

When vector processing encounters a **goto** xxx if holiday in table # step, vector processing determines if the current date and time qualifies as a holiday in the specified table. The information is then used to decide whether the goto condition is true or false and in turn to decide whether to go to the given step or not. The date and time match is done at the time that the call is in vector processing. The date and time match is similar to the match done in the Time-of-Day (TOD) routing, which means that the system date and time is checked on the Processor Port Network (PPN), rather than the local port network time on the Expansion Port Network (EPN).

Holiday Vectoring is not limited to holiday use, but can also be applied to any date-related special processing. For example, vectors can be modified or created to perform special processing during a two-week television promotion or a semi-annual sale.

The Holiday Vectoring feature benefits administrators who have to administer as many as 30 bank holidays throughout the year. Holiday Vectoring streamlines vectoring tasks and ensures seamless operation during holidays or special events.

Without Holiday Vectoring, call center administrators have to write special vectors for each holiday or other special date related circumstance and ensure that the vectors are administered at appropriate times. In some cases, administrators are required to go to work on holidays just to administer vectors.

Administering Holiday Vectoring

About this task

To enable the **Holiday Vectoring** field you must first enable either **Vectoring (Basic)** or **Attendant Vectoring**.

You can use up to 999 different holiday tables if you enable the **Vectoring (3.0 Enhanced)** field. Otherwise, you can use up to 10 holiday tables.

On the Call Center pages of the System-Parameters Customer-Options screen, set the **Vectoring (Holidays)** field to `y`.

Holiday Vectoring considerations

- Administration of holiday tables is supported only on Communication Manager and cannot be changed using adjunct vectoring tools.
- Holiday vectoring is only available when you enable **Vectoring (Basic)** or **Attendant Vectoring**.
- There is no procedure to validate consistency among the 15 holidays in any table. If the same holiday is entered twice, the system stops checking after the first found entry.
- Date range overlapping is common for holidays that are in ranges of dates. When a call is in vector processing, holidays are checked from top to bottom on the table and the check stops if a match is found. Even though there can be multiple matching entries, the check stops at the first match.
- There is a validation that the day of the month that is entered is valid with the given month. Specifically, if the month is April, June, September, or November, then the date must be a number between 1 and 30. If the month is January, March, May, July, August, October, or December, then the date can be a number between 1 and 31. If the month is February, a the date can be any number between 1 and 29.

  Note:

  The year is not checked in holiday vector processing. This allows the same holidays to be used year-to-year when the holiday is on a fixed date. For holidays where the date changes from year-to-year, you must re-administer the holiday tables.

- When disabling the Holiday Vectoring feature, that is, changing the value of the **Vectoring (Holidays)** field from `y` to `n` on the System-Parameters Customer-Options screen, the vectors are checked for any **`goto if holiday`** steps. If any such steps are found, an error message is displayed and the change is not allowed. You must remove the vector steps first before you disable the Holiday Vectoring.
- You can use the **Use VDN Time Zone For Holiday Vectoring** field on the Vector Directory Number (VDN) screen that is active for the call so that vector processing uses the VDN time zone specified when applying holiday vectoring if the field is set to `y`, otherwise vector processing uses the system time.

Holiday Vectoring command set

Holiday Vectoring enables a set of commands that are used to create call vectors, which route calls on holidays or days when special processing is required. The following table shows the commands that are available for use in Holiday Vectoring.

Command category	Action taken	Command
Branching/programming	Go to a vector step	**`goto step`**
	Go to a vector	**`goto vector`**

Goto step command for Holiday Vectoring

Syntax and valid entries

Command name	Parameter	Variables	Condition	Variables	Description
`Goto`	step	<step #> 1-99	if holiday in table	<table #> 1-999, A-Z, AA-ZZ, V1-V9	This command directs the call to a specific vector step if the conditions of the call match a holiday that is in the specified holiday table.
			if holiday not in table		This command directs the call to a specific vector step if the conditions of the call do not match any of the holidays that are in the specified holiday table.

Goto vector command for holiday vectoring

Syntax and valid entries

Command name	Parameter	Variables	Condition	Variables	Description
Goto	vector	<vector #> 1-8000 @ step 1–99	if holiday in table	<table #> 1-999, A-Z, AA-ZZ, V1-V9	This command directs the call to a specific vector if the conditions of the call match a holiday that is in the specified holiday table.
			if holiday not in table		This command directs the call to a specific vector if the conditions of the call do not match any of the holidays that are in the specified holiday table.

Holiday table command syntax

Syntax 1

```
change holiday-table x
```

This command allows you to change the entries in a holiday table. To create a new holiday table, you must use the **change** command with a table number that is not in use. For example, change holiday-table 9, where table 9 has not been used to define holidays.

Syntax 2

```
display holiday-table x
```

This command allows you to view the entries in the specified holiday table.

Syntax 3

```
list holiday-table
```

This command lists all of the holiday tables in use.

Syntax 4

```
list usage holiday-table x
```

This command lists all vector steps that refer to the selected holiday table.

Using the holiday table commands

About this task

After enabling **Vectoring (Holidays)** on the Customer Options screen, enter:

```
change holiday-table 1
```

On the Holiday Table screen, which is shown in the following example, enter the holiday information.

Setting up a Holiday Table

```
change holiday-table 1                                    page 1 of 1
                                HOLIDAY TABLE
    Number: 1              Name: Bank Holidays
          START                   END
    Month Day Hour Min     Month Day Hour Min     Description
    12    24  00   00      12    31  23   59      Christmas
    01    01  00   00      01    02  06   00      New Year's Day
```

Note:

When using a range of dates, the end date must be greater than the start date. Ranges must be within one calendar year. There are two entries in the example, one for each calendar year.

The Holiday Table screen can be used to enter individual holidays or holiday ranges. The following rules apply for entering dates on this screen:

Procedure

1. If a day is entered, the corresponding month must be entered.
2. If a month is entered, the corresponding day must be entered.
3. If an hour is entered, the corresponding minute must be entered.
4. If a minute is entered, the corresponding hour must be entered.
5. If an hour and minute is entered, the corresponding month and day must be entered.
6. If a month and day is entered, the corresponding hour and minute is not required.
7. If an end month and day is entered, the corresponding start month and day must be entered.
8. If a start month and day is entered, the corresponding end month and day is not required.
9. To enter an individual holiday, enter a start month and day, but do not enter an end month and day.
10. To enter a holiday range, enter both a start month and day and an end month and day.
11. The start month, day, hour, and minute must be less than or equal to the end month, day, hour, minute.
12. The description field is alphanumeric.

Result

After creating a holiday table, use the **display holiday-table** command to view the entries. To list all of the holiday tables, use the **list holiday-table** command, as shown in the following example.

Listing Holiday Tables

```
list holiday-table
                                HOLIDAY TABLES

                            Table Number      Name
                                01            Business Holidays
                                02            Annual Promotion Dates
                                03            Summer Special
                                04
                                05
                                06
                                07
                                08
                                09
                                10
```

Changing vector processing for holidays

About this task

You must add or change vector processing for the administered holiday tables.

On the command line, enter `change vector` x (where x is the vector number). The Call Vector screen appears with editable fields. Customers can enter a new goto conditional for the holidays.

When you set the **Holiday Vectoring** field to `y`, a field on the Call Vector screen identifies if the vector on which you are currently working is a holiday vectoring vector, as shown in the following example.

```
change vector x                                                 page 1 of 3

                                   CALL VECTOR


    Number: xxx      Name: _____________________________
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y               3.0 Enhanced? y

01 ____________
02 ____________
03 ____________
04 ____________
05 ____________
06 ____________
07 ____________
08 ____________
09 ____________
10 ____________
11 ____________
```

The **Vectoring (Holiday)** field is a display-only field and appears only when you set the **Holiday Vectoring** field on the Customer Options screen to `y`. If either **Vectoring (Basic)** or **Attendant Vectoring** are set to `y`, the **Vectoring (Holiday)** field can be set to `y`.

The following examples use the **goto** command to route calls during holidays.

Example 1

```
 change vector 1                                                Page 1 of 3
                                  CALL VECTOR

    Number: 1        Name: In Germany
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? y

01 goto         vector 2  if holiday         in    table 1
02 route-to     number 123456789             with cov n if unconditionally
03
04
05
06
07
08
09
10
11
```

Example 2

```
change vector 3                                                 Page 1 of 3
                                  CALL VECTOR

    Number: 3        Name: In Ireland
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? y

01 goto         step   2  if holiday         in    table 2
02 route-to     number 45678          with cov n if unconditionally
03 stop
04 announcement 2721
05
06
07
08
09
10
11
```

After you have assigned holiday tables to several vectors, you can use the **list usage holiday-table** command to view which vectors and vector steps are using the selected holiday table.

```
list usage holiday-table x
                             LIST USAGE REPORT

     Used By
     Vector                Vector Number 1                      Step 1
     Vector                Vector Number 3                      Step 1
```

Look-Ahead Interflow

Look-Ahead Interflow (LAI) enhances Call Vectoring for call centers with multiple ACD locations. With LAI, call centers can improve their call-handling capability and agent productivity by intelligently routing calls among call centers to achieve an improved ACD load balance. This service is provided by ISDN D-channel messaging over QSIG or non-QSIG private networks, virtual private networks, or public networks. LAI is also supported on SIP trunks. The receiving Communication Manager can accept or deny interflowed calls sent by the sending Communication Manager.

LAI has the following basic attributes:

- Produces First in First Out (FIFO) or near-FIFO call processing
- Includes enhanced information forwarding, that is, codeset 0 user information transport

LAI requirements

The following items are criteria for basic LAI call control operation over a VPN or a public switched network:

- The sending and receiving call center locations must have ISDN PRI or BRI trunk facilities.

  Note:

  ATM trunking and IP trunking can be set up to emulate ISDN PRI, but first verify that the release of Communication Manager that you have still supports the use of ATM trunking.

- The switch must support the ISDN country protocol.
- LAI has been tested with several major carriers. To find out if the capabilities work with your carrier, check with your account team for the latest information. If testing is not done to verify operation over the public networks involved with the preferred specific configuration, use private ISDN trunking between the nodes until successful testing is complete.
- The ISDN SETUP and DISCONNECT messages are transported between sending and receiving locations, for example, Signaling System 7 (SS7) or equivalent public network connectivity.
- A receiving-end generated DISCONNECT message must transmit back to the sending switch call center without changing the cause value.

  Conversion of the DISCONNECT message to a progress message, with a **Progress Indicator Description** set to `1` and a cause value other than 127 included, is a valid reject message and compatible with LAI.

- Progress messages that are generated towards the sending end by intervening network switches must have the **Progress Indicator Description** set to `8` so that the switch does not treat the call accepted or rejected.
- ISDN codeset 0 user information transport supports LAI information forwarding. As an alternative, LAI can use dedicated VDNs at the receiving location to provide an equivalent display of the forwarding application identity and set trunk group options to not send either the codeset 6/7 LAI IE or codeset 0 information transport.

Note:

BSR cannot use the LAI alternatives. BSR must use ISDN codeset 0 user information transport.

LAI considerations

Consideration	Description
Carrier compatibility	LAI works with several major carriers. To find out if the LAI capabilities work with your carrier, check with your account team for current information. If testing has not been done to verify operation over the public networks involved with the preferred specific configuration, use private ISDN trunking between the nodes until successful testing is complete. ISDN routing with LAI enabled: All calls routed over ISDN facilities by a **route-to number** with cov n or **route-to digits** with cov n vector command on a communication server where LAI is enabled are treated as LAI call attempts. A vector can route a call over an ISDN facility to a destination that is not a VDN. The sending communication server processes this call as an LAI call even if the call is not an LAI call. ISDN processing at the receiving communication server causes the call to always be accepted. However, the DNIS and any other information in the LAI information forwarded with the call are ignored. Interim call handling before LAI is accepted by receiving communication server: Until the LAI attempt is accepted by the receiving communication server, the caller continues to hear any feedback applied by the sending communication server vector and remains in any split or skill queue.
Call handling with Route-to number or Route-to digits handling with coverage	**Route-to number** with cov y or **route-to digits** with cov y commands does not result in an LAI call attempt. The commands always complete the call. Do not use the command if the vector at the receiving communication server can deny the call, since the caller in this case hears a busy signal. Use this command with coverage y only when you want unconditional interflow with LAI active and the terminating communication server is set up accordingly.
Continuity during call transfer between communication servers	Audible feedback can be provided to the caller before interflow is attempted. Therefore, another audible feedback from the receiving communication server can confuse the caller. For example, a caller hearing ringback on the sending communication server can be confused if music is applied suddenly when the call interflows to the receiving communication server.

Table continues…

Consideration	Description
Backward compatibility of LAI applications	For backward compatibility of LAI applications between Avaya communication servers, leave the **Send Codeset 6/7 LAI IE** field on the Trunk Group screen to the default `y`. Existing LAI applications continue to operate as before, even after you upgrade. You can use enhanced LAI available in the communication server without any network or trunk administration changes, by adding the “interflow-qpos” conditional to original LAI vectors.
AAR/ARS	ISDN facilities used to provide LAI to a VDN on another communication server in a private network can use the AAR feature if private facilities are to be used for call routing.
Agent Telephone Display	If collected digits are forwarded with an interflowed call, the forwarded digits are displayed to the answering agent on the telephone display.
Attendant Control of Trunk Group Access	Calls do not route over a trunk with Attendant Control of Trunk Group Access set.
Authorization Codes	Authorization codes are not required for interflow routing. Assign a high enough FRL to the VDN so that the route desired for routing interflow calls can be used without requiring an authorization code entry. If a route choice is encountered that requires a higher FRL, the interflow is an invalid destination, rejected for LAI or not available for standard interflow, without the application of recall dial tone.
BCMS	Neither does BCMS log LAI attempts nor does BCMS report accumulated in-VDN time.

Table continues…

Consideration	Description
Call Detail Recording - Sending Server	No ineffective call attempt or outgoing call CDR records are generated for vector **route-to** commands that are unsuccessful including denied LAI attempts. If a local call to a VDN generates an LAI call attempt that is accepted and answer supervision is returned from the receiving communication server, one outgoing call CDR record is generated with the originating extension as the calling number. If an incoming call to a VDN generates an LAI call attempt that is accepted, and no answer supervision is returned from the receiving communication server, one incoming CDR record is generated. The VDN is the called number and the duration is from the time answer supervision was provided to the incoming trunk. If an incoming call to a VDN generates an LAI call attempt that is accepted and answer supervision is returned from the receiving communication server, two incoming CDR records are generated: • An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk. This is generated if the call is initially answered in the sending communication server before interflow takes place. • An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.
Call Detail Recording - Receiving Server	On the receiving communication server, an incoming LAI call is treated like any other incoming vector call. If answer supervision is returned by the vector and the call does not terminate to another destination, the VDN extension is recorded as the called number in the CDR record. If the call terminates to a hunt group, the VDN, hunt group, or agent extension is recorded as the called number. If the **Record VDN in Record** field on the Feature-Related System Parameters screen is y, the VDN extension overrides the **Call to Hunt Group - Record** field entry for vector calls.
Call Prompting	Digits collected at the sending Communication Manager, no matter how the digits are collected (caller-entered, ASAI provided, CINFO provided) are forwarded with interflowed calls and available at the remote Communication Manager using Information Forwarding. **Note:** Dial-ahead digits are not forwarded with the call. There is a maximum of 16 forwarded digits.
Centralized Attendant Service	A centralized attendant can be an LAI destination.

Table continues…

Consideration	Description
VDN Name Display	The VDN name, part of the LAI information forwarded with calls, can be up to 27 characters long. Any characters over this limit is dropped. On Communication Manager, a VDN name can be administered with as many as 27 characters.
Distributed Networking - Manufacturers Specific Information (MSI)	LAI, whether enhanced or no, does not function with systems from other vendors unless that vendor develops a corresponding capability that works with the Avaya communication server.
Facilities Restriction Level and Traveling Class Marks	The FRL for interflow over ARS/AAR route choices is assigned to the original VDN used for the incoming call.
Incoming Call Management	You can use the adjunct routing capabilities of vectoring at the sending communication server to determine if a call must be interflowed. You can use the adjunct routing at the receiving communication server to tandem the call to a far-end communication server. If the call terminates to a tandem trunk, two CDR records are generated: • An incoming record with the VDN as the called number and the duration as the time since answer supervision was provided to the incoming trunk. • An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number.
Network Access	LAI operates over public, private, or virtual private networks. ISDN-BRI and -PRI networks that meet minimum network requirements. The sending of an LAI codeset 6/7 information element is counted toward Message Associated User-to-User Information (MA-UUI) counts.
Path replacement for QSIG/DCS ISDN calls	For calls that are waiting in queue or in vector processing, even if the call is not connected to an answering user, path replacement using QSIG can be attempted to find a more optimal path for this call. This results in more efficient use of the trunk facilities. The QSIG ISDN or DCS ISDN trunk path replacement operation can be triggered for ACD calls by the LAI **`route-to number`** vector step, BSR **`queue-to best`** vector step, and the adjunct routing vector steps.
QSIG	LAI and information forwarding function over QSIG trunk facilities if the remote locations are Avaya communication servers. You can get LAI call control functionality with other vendors if an Avaya communication server is the starting point.
RONA	Calls redirected to a VDN by RONA can subsequently be processed and routed by LAI applications.
Service Observing	You can observe a call in LAI processing using VDN observing throughout the life of the call as long as the call is still connected through the local communication server. All current restrictions on Service Observing still apply. Incoming calls can be service observed at the remote communication server.

Table continues…

Consideration	Description
Trunk-to-Trunk Transfer	Interflowed calls can be transferred by a receiving communication server to another trunk connection.
VDN Override	The name of the active VDN for a call is displayed at the remote answering agent.

Two switch configuration

LAI is enabled through the use of call vectors and their associated commands. For a two-switch configuration, these vectors are included in both the sending switch, which processes vector outflow, and the receiving switch, which processes vector inflow.

LAI command set

The following table lists the call-acceptance vector commands that are used in LAI.

Command	Qualification
announcement	Announcement available Queued for announcement Retrying announcement
check split	Call terminates to agent Call queued to split
collect digits	Always, except for ced and cdpd digits, which are neutral
converse-on split	VRU answers the call Call queued to converse split
disconnect	With announcement and announcement available With announcement and queued for announcement With announcement and retrying announcement
messaging split	Command successful Call queued
queue-to split	Call terminates to agent Call queued to split
route-to	Terminates to valid local destination Successfully seizes a non-PRI trunk Results in a LAI call attempt, and the call is accepted by the far-end switch
wait-time	Always, except **wait-time hearing i-silent**, which is neutral

If the receiving Communication Manager is unable to accept the LAI call, one of the vector steps executes call denial.

Note:

Use the **busy** command instead of the **disconnect** command to allow for compatibility with similar network services such as Alternate Destination Redirection (ADR).

Vector command (for call denial)	Qualification
busy	Always
disconnect	Without announcement With announcement but announcement unavailable
reply-best	Always. Used with BSR

The vector commands in the following table are treated as neutral because the commands do not generate call acceptance or denial messages.

Neutral vector command	Qualification
adjunct routing link	Always
announcement	Announcement unavailable
check split	Call neither terminates nor queues
collect ced/cdpd digits	Always
consider	Always. Used with BSR
converse-on split	Call neither terminates nor queues
goto step	Always
goto vector	Always
messaging split	Command failure
queue-to split	Call neither terminates nor queues
route-to	Unsuccessful termination Trunk not seized LAI call denied by the far-end switch
stop	Always
wait-time hearing i-silent	Always **Note:** This command is used following an **adjunct routing link** command in applications where the adjunct decides whether to accept or reject the Look-Ahead calls.

How traditional LAI works

Use the traditional LAI feature when the preferred call flow performs LAI attempts before queuing the call.

LAI commands

LAI uses the commands included within the basic Call Vectoring and Call Prompting features:

- **route-to number with coverage n** or **route-to digits with coverage n** command on Communication Manager that has LAI enabled and that successfully seizes an ISDN trunk automatically results in a normal LAI call attempt being placed. The call attempt can be rejected or accepted by the remote end.
- **route-to number with coverage y** or **route-to digits with coverage y** command does not result in an LAI call attempt. The command always completes the call. Do not use the command when the vector at the receiving location ends up denying the call since the caller will hear a busy signal, or the call will disconnect. Use the command with coverage set to y only for those cases when an unconditional interflow is wanted (with LAI active) and the terminating Communication Manager is set up to handle this type of call.

When a LAI call attempt is made, Call Vectoring at the sending location checks a potential receiving location to determine whether to hold or send the call. While this is done, the call remains in queue at the sending location. As such, the call can still be connected to the sending-location agent if one becomes available before the receiving location accepts the call.

Call Vectoring at the receiving location determines whether to accept the call from the sending location or to instruct the sending location to keep the call. In the latter case, the sending location can keep the call, check other locations, or provide another call treatment. Conditions for sending, refusing, or receiving a LAI call attempt can include a combination of any of the following:

- Expected wait time for a split.
- Number of staffed or available agents for a skill.
- Number of calls in queue.
- Average speed of answer or the number of calls active in a VDN.
- Time of day and day of week.
- Any other legitimate conditional.

If the call is accepted by the receiving Communication Manager, the call is removed from any queues at the sending Communication Manager and call control is passed to the receiving Communication Manager. If the call is denied by the receiving Communication Manager, vector processing continues at the next step at the sending Communication Manager. Until the call is accepted by either Communication Manager, the caller continues to hear any tones applied by the sending Communication Manager. If the call is denied, the call vector can apply alternate treatment, such as placing another LAI call to an alternate backup Communication Manager.

Note:

The LAI operation is completely transparent to the caller. While a LAI call attempt is being made, the caller continues to hear any audible feedback that is provided by the sending Communication Manager vector. The caller also maintains position in any split queues until the call is accepted at the receiving Communication Manager.

LAI passes Call Prompting digits collected in the sending Communication Manager to the receiving Communication Manager by codeset 0 user information transport.

Example of traditional LAI

The vectors in the sending Communication Manager use the **goto** command to determine whether the call must be sent to the receiving Communication Manager. Recall that the **goto** command tests various outflow threshold conditions such as expected wait time. If the expressed condition is met, a branch is made to the appropriate **route-to** command. This command sends the call to the receiving Communication Manager, which, as already noted, can accept or deny the call.

The following example shows an outflow vector that can be included in a sending Communication Manager.

Using LAI with route-to commands to outflow calls

```
1. wait-time 0 secs hearing ringback
2. goto step 5 if expected-wait for split 3 pri m < 30
3. route-to number 5000 with cov n if unconditionally
4. route-to number 95016781234 with cov n if unconditionally
5. queue-to split 3 pri m
6. announcement 3001
7. wait-time 30 secs hearing music
8. goto step 6 if unconditionally
```

If split 3 has an expected wait time of less than 30 seconds (step 2), step 5 queues the call to the queue at a medium priority.

If the expected wait time is more than 30 seconds, LAI attempts are made in steps 3 and 4. If the call is accepted by one of the receiving Communication Manager call control passes to the receiving Communication Manager.

If the receiving Communication Manager deny the call, the call queues to split 3 and announcement 3001 plays. The caller hears music, interrupted by announcement 3001 every 30 seconds.

Receiving switch operation

When the receiving switch receives the LAI request, the call first routes to a VDN. The VDN then maps the call to the receiving switch's inflow vector, and vector processing begins, starting with inflow checking. Inflow checking is enabled by conditional **goto** commands in the inflow vector. The decision to accept or deny a call can be based on checks such as any of the following:

- Expected Wait Time
- Number of staffed agents
- Number of available agents
- Time-of-day or day of the week
- Number of calls in split queue
- Average Speed of Answer
- Active VDN calls
- Automatic Number Identification (ANI) and Calling Line Identification (CLID)
- II-Digits

- CINFO ced or cdpd digits
- Collected digits forwarded from the sending switch

Once inflow checking is complete, acceptance of the LAI call is accomplished by executing any of the vector commands listed in the call-acceptance vector commands.

Note:

For each command listed in the call-acceptance vector commands, neutral vector commands, and call-denial vector commands, only one of the corresponding qualifications must be true for the command to cause the desired result, which is call acceptance, call denial, or no effect on such acceptance or denial.

The following example shows an inflow vector that can be used by a receiving switch.

Using inflow checking for LAI requests

```
1. goto step 6 if expected-wait in split 1 pri h > 30
2. queue-to split 1 pri h
3. announcement 4000
4. wait-time 2 seconds hearing music
5. stop
6. busy
```

Step 1 of this inflow vector checks the inflow thresholds. The **goto step** command in step 1 checks the expected wait time in split 1. If the expected wait time is greater than 30 seconds, a branch is made to the **busy** command in step 6. If executed, the **busy** command denies the call, and the receiving switch returns a call denial message to the sending switch. The sending switch, in turn, drops the LAI call attempt and then continues vector processing at the next vector step.

If the expected wait time in split 1 is less than or equal to 30 seconds, the receiving switch returns a call acceptance message to the sending switch, and call control is passed to the receiving switch. Thereafter, the call is queued to split 1 in the receiving switch (step 2). Once queued, the caller receives the appropriate announcement in step 3 and is then provided with music until the call is answered by an agent or abandoned by the caller (steps 4 and 5). Remember that the **stop** command halts vector processing but does not drop the call.

If the sending switch does not receive a call acceptance or call denial message within 120 seconds after the LAI call request, the LAI attempt is dropped. The sending switch continues vector processing at the next step.

Enhanced LAI

Enhanced Look-Ahead Interflow (LAI) uses the same basic Call Vectoring commands as traditional LAI, but includes an **interflow-qpos** conditional. Use Enhanced LAI when the preferred call flow performs an LAI attempt after putting a call in a queue.

The **interflow-qpos** conditional:

- Produces First in First Out (FIFO) or near FIFO call processing.
- Uses less processing during LAI.

The simple way to achieve FIFO

You can use the *interflow-qpos* conditional in a **route-to** or **goto** command to achieve First In, First Out (FIFO) results.

For example, you can use the following **route-to** command with the conditional to achieve FIFO results:

route-to number 9581234 with cov n if interflow-qpos=1

If you have a lot of remote agents, set the **route-to** command as follows:

route-to number 9581234 with cov n if interflow-qpos<=2

The interflow-qpos conditional

The **interflow-qpos** conditional only applies interflow processes to a dynamic eligible queue and to calls that are queued locally before the **route-to** is attempted.

The eligible queue is that portion of the split/skill queue that:

- Includes only calls that are not expected to be answered locally during the interflow process at that moment relative to the call being processed
- Does not include direct agent calls because these calls are excluded from any interflow process.

The following is an example of the **interflow-qpos** conditional used in a **route-to** command:

```
route-to number _____ with cov _ if interflow-qpos CM x
```

where

- **CM** is the comparator. It is one of three symbols: =, <, <=
  - With **if interflow-qpos = x**, the call is interflowed if it is at the **x** position from the top of the eligible queue.
  - With **if interflow-qpos < x**, the call is interflowed if it is among the top **x-1** of the eligible queue.
  - With **if interflow-qpos <= x**, the call is interflowed if it is among the top **x** eligible calls.
- **x** indicates the call position in the eligible queue. Valid queue positions are 1 through 9. The top queue position is 1. The eligible queue is made up of calls from the first local split or skill that the call has been queued to due to previous steps in the vector.

Note:

Calls that are likely to be serviced locally before an LAI can be completed are not eligible for interflow since the calls are excluded from the eligible queue. Calls that are likely to be answered are identified based on conditions of the split/skill to which the call is queued and, under certain conditions, an administered minimum EWT threshold value.

The following is an example of the **interflow-qpos** conditional used in a **goto** command:

```
goto step/vector ____ if interflow-qpos CM x
```

where

- **CM** is the comparator. It is one of six symbols: =, <>, <, <=, >, >=
- **x** indicates the call's position in the eligible queue. Valid queue positions are 1 through 9. The top queue position is 1.

Calls that are likely to be serviced locally before an LAI can be completed are not eligible for interflow since the calls are excluded from the eligible queue.

When does a call not interflow?

A call does not interflow under the following circumstances:

- If the interflow-qpos conditional is not met.

  As with other conditionals, the route-to number... if interflow-qpos step or the goto step/vector branch is executed only if the conditional is met, otherwise vector processing goes to the next step.

- If the call is not in a split or skill queue, or not in the eligible portion of the queue when the conditional step is executed.

  If the call is not in queue when the "route-to number... if interflow-qpos" step is executed, a vector event is logged and vector processing continues at the next step.

  If the call is not in queue when a "goto... if interflow-qpos" step is executed, the queue position of the call is treated as infinite in determination of the conditional.

  Note:

  A vector event is not logged if the call is in queue, but is not in the eligible portion of the queue.

- Interflow failure or LAI rejection

  Interflow failure or LAI rejection will also go to the next step. Route-to operation and feature interactions will be the same as other configurations of the route to number command, for example, **route to number ___ with cov _ if digit CM x**.

The following table outlines what action is taken for different cases of interflow eligibility.

Case	Action at route-to step	Action at goto step
The call not eligible for interflow.	The call is never routed.	Treat as if the interflow queue position is infinite.
The call is not in any split queue.	The call is treated as if the interflow queue position is infinite.	Treat as if interflow queue position is infinite.
The call is eligible for interflow.	Act according to the conditional.	Act according to the conditional.

Setting the minimum EWT

About this task

The minimum expected wait time (EWT) threshold that is used to help determine which calls are more likely to be answered locally is administered on the Feature-Related System Parameters screen. Minimum EWT is used when the local agents, that is, in the first split/skill to which the call is queued, are handling a significant number of the calls. If these agents are not handling a significant number of calls, the call is eligible for LAI even if its EWT is lower than the threshold.

Note:

When enhanced LAI vectors or the look-ahead EWT threshold are administered inappropriately, remote agents can experience phantom calls or a delay between becoming available and receiving an ACD call.

Use the SAT terminal or terminal emulator to administer Communication Manager.

To set the minimum EWT threshold:

Procedure

1. In the command line, type `change system-parameters feature` and press `Enter`.

   The system displays the Feature-Related System Parameters screen.

2. Find the page of the Feature-Related System Parameters screen that has the **Interflow-Qpos EWT Threshold** field.

   If Look-Ahead Interflow is active, the **Interflow-Qpos EWT Threshold** field can be administrated.

3. In the **Interflow-Qpos EWT Threshold** field, enter the number of seconds, as a number from 0 to 9, that you want for the EWT threshold.

   The default of two seconds is recommended.

   Note:

   When the **look-ahead EWT threshold** field is set too low, remote agents can experience phantom calls.

4. Press `Enter` to save your changes.

Example of single-queue multisite operation

In the scenario, all new calls for a given customer application are routed by the public network to only one of the switches in the network, where the calls are put in the queue.

Local agents service the calls from the queue in the normal fashion. However, remote agents service calls by means of Enhanced Look-Ahead Interflow (LAI).

The Communication Manager server with the call queue does rapid enhanced look-ahead attempts to all other switches in the network that can service this call type, looking for an available agent.

Normally, the look-ahead attempts are placed only on behalf of the call that is at the head of the queue (**interflow-qpos = 1**). However, in scenarios where there are large numbers of agents at a remote Communication Manager, interflows on behalf of more than one call can be necessary to outflow a sufficient volume of calls to keep all agents busy (**interflow-qpos <= 2**).

Vector to back up split

```
1. announcement 3501
2. wait-time 0 secs hearing music
3. queue-to skill 1 pri m
4. route-to number 93031234567 with cov n if interflow-qpos = 1
5. route-to number 99089876543 with cov n if interflow-qpos = 1
6. wait-time 5 secs hearing music
7. goto step 4 if unconditionally
```

Interflow call attempts are placed on behalf of the call that is at the beginning of the queue every five seconds to the two other Communication Manager in the network.

If queuing times are very long, five minutes, for example, and the call is not near the beginning of the queue, going through the vector loop from step 4 to step 7 every 5 seconds is of no avail. The FIFO processing vector is more efficient.

Example of maintaining FIFO processing with LAI

One of the advantages of enhanced LAI is the ability to provide FIFO or near-FIFO call processing. The following example shows a vector that is used to achieve such call processing.

FIFO processing vector

```
1. announcement 3501
2. wait-time 0 secs hearing music
3.  queue-to skill 1 pri m
4. goto step 7 if interflow-qpos < 9
5. wait-time 30 secs hearing music
6. goto step 5 if interflow-qpos >= 9
7. route-to number 93031234567 with cov n if interflow-qpos = 1
8. route-to number 99089876543 with cov n if interflow-qpos = 1
9. wait-time 5 secs hearing music
10. goto step 7 if unconditionally
```

In this vector:

- The rapid look-ahead loop is only entered when the call reaches one of the top 8 positions in queue.
- The number of executed vector steps is reduced dramatically when call waiting times are long.

It is important to write vectors so that calls at the head of the queue have advanced to the rapid look-ahead loop by the time their turn to interflow has been reached. In the vector example shown above, if 8 calls can be serviced from queue in less than 30 seconds (which is the loop time on step 5), there can be a delay in outflowing calls to available agents at the remote sites.

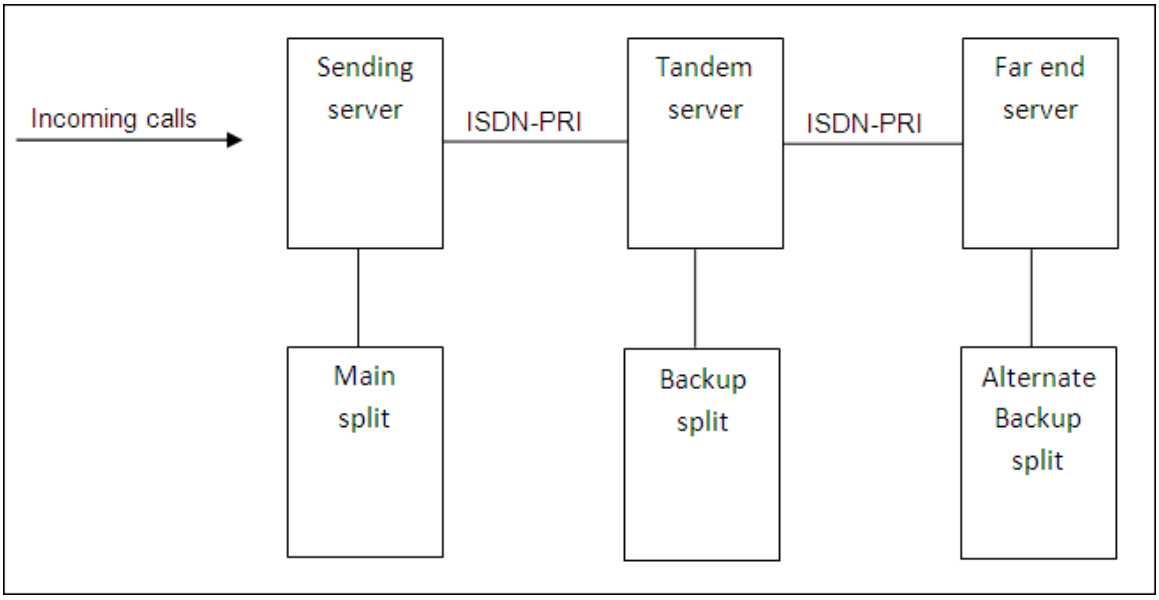
Single-queue FIFO considerations

- When there are available agents, calls are always delivered to available agents at the queuing Communication Manager before available agents at the remote Communication Manager.

- When there are calls in the queue and agents serve calls from multiple applications, the agents always service calls from the applications that are queued locally before calls from applications that are queued at another Communication Manager.
- Use backup VDNs and vectors ito provide continuous operation in the event of a failure at a queuing Communication Manager.
- EWT predictions cannot be made if the split or skill in which the calls are queued has no working agents.
- EWT predictions can be temporarily inaccurate if there are sudden, major changes in the number of working agents in the split or skill in which the calls are queued.

Example of LAI in a tandem switch configuration

To route calls over ISDN facilities, you can implement tandem LAI by using the **route-to** command that contain external destinations. This configuration is shown in the following figure.



Sending switch operation

The sending switch is unaware that its LAI call is being tandemed to an alternate switch. The operation of the sending switch in the tandem switch configuration is the same as that in the two-switch configuration.

Tandem switch operation

If the receiving Communication Manager executes a **route-to** command that routes the call over an ISDN facility before call acceptance, the **route-to** command is performed on a look-ahead basis in the same manner as a sending Communication Manager. If the call is accepted at a far-end Communication Manager, acceptance is passed to the sending Communication Manager, and call control is passed to the far-end Communication Manager, along with tandeming of the original calling party information and the original DNIS name. If the call is denied, the next step of the tandem switch vector is executed.

The following example shows a tandem switch vector.

Tandem switch vector example

```
1. goto step 6 if expected-wait in split 30 pri h > 30
2. queue-to split 30 pri h
3. announcement 200
4. wait-time 2 seconds hearing silence
5. stop
6. route-to number 4000 with cov n if unconditionally
7. busy
```

Step 1 of this vector checks the inflow threshold. If the inflow criteria are acceptable, the vector flow drops to step 2, where the **queue-to split** command provides acceptance to the sending Communication Manager. Thereafter, steps 3 through 5 provide a typical queuing-wait scheme.

If the inflow criteria are not acceptable, a branch is made to step 6. The **route-to** command in this step checks another Communication Manager that is enabled with LAI on a look-ahead basis. If this far-end Communication Manager rejects the call, a denial message is relayed back to the tandem Communication Manager, which then drops the LAI call attempt. On the other hand, if the far-end Communication Manager accepts the call, an acceptance message is relayed all the way back to the sending Communication Manager.

No ringback is provided in this tandem Communication Manager vector. This is necessary so that an acceptance message is not returned to the sending Communication Manager. This operation is appropriate for the caller because the sending Communication Manager has already returned an announcement before a LAI attempt is made to the receiving Communication Manager.

Be sure that the sending Communication Manager is not used as a backup location for the tandem Communication Manager or for any of the far-end Communication Manager. If the sending Communication Manager is administered in this manner, all trunk facilities can be tied up by a single call.

Far-end switch operation

The far-end switch is also unaware that tandeming has taken place. The far-end switch functions in the same manner as the receiving switch within the two-switch configuration.

LAI-initiated path replacement for calls in vector processing

Path replacement for calls in vector processing

You can use QSIG or DCS with Reroute using ISDN SSE for path replacement for calls in a queue. For calls that are waiting in a queue or in vector processing, even if the call is not connected to an answering user, Communication Manager can attempt a path replacement to find the optimum path for the call.

You can use the **route-to** command with LAI to initiate a QSIG path replacement for a call.

If the terminating Communication Manager server receives a *Path Replacement Propose* operation for a call that is in a queue or in vector processing, Communication Manager immediately initiates path replacement using the path replacement extension if the **Path Replace While in Queue/Vectoring** field is y and the **Path Replacement Extension** field has a valid entry. The fields are available on the ISDN Parameters page of the Feature-Related System Parameters screen.

Example LAI vector

The following example shows how an LAI vector can be written to trigger path-replacement at the terminating switch.

Note:

In order for a path-replacement to be attempted, the incoming and outgoing trunks that are used for the call must be administered with the **Supplementary Service Protocol** field set to `b`.

LAI-initiated path-replacement vector

```
1. wait 0 seconds hearing music
2. queue-to skill "n" if available-agents < 6
3. route-to number "ARS number for ISDN trunk" with cov n
4. wait 999 seconds hearing ringback
```

At the receiving Communication Manager, the vector that processes the incoming call must use an **`announcement`**, or **`wait hearing music`** vector command to enable path-replacement.

DNIS and VDN Override in an LAI environment

LAI handles Dialed Number Identification Service (DNIS) and VDN Override in various ways, depending on a number of different characteristics of the call. DNIS, as described in Call Vectoring fundamentals, allows any agent with a display-equipped telephone to receive visual displays that specify the name of the called VDN. VDN Override in its basic form allows the name of a subsequently routed to VDN to be displayed to the answering agent instead of the name of the originally called VDN.

The following sections discuss how LAI handles DNIS and VDN Override.

DNIS information displayed to answering agent

For LAI, the DNIS name, which is the called VDN name from the sending switch, is presented on the display for the answering agent on the receiving switch if all of the following are true:

- The LAI option is enabled.
- The call routes to a VDN.
- The DNIS name field is not blank.

The type of DNIS information that is displayed depends upon a number of different scenarios. This information is presented in the following table.

Scenario	Information displayed
Tandem LAI call	Look-Ahead Interflow DNIS information from the original LAI call.
No redirection at the sending switch	VDN name according to Override rules at the sending switch (active VDN).

Table continues…

Scenario	Information displayed
Redirection at the sending switch (VDN in coverage path)	Original VDN name, or If multiple VDNs are accessed, the name of the VDN that was last accessed by a **route-to** command.
Sending switch sends a blank DNIS Name field (that is, a name is not assigned to the sending switch called VDN) or the trunk group is administered to not send the LAI name.	Name associated with the receiving VDN. This name can be changed according to the rules of VDN Override at the receiving switch.

Note:

VDNs that map to vectors that place LAI calls must have their ISDN Calling Party Number (CPN) prefixes administered. If an ISDN CPN prefix is not administered, the assigned VDN name is not sent. Instead, a DNIS of all blank space characters is sent and displayed on the answering agent's terminal.

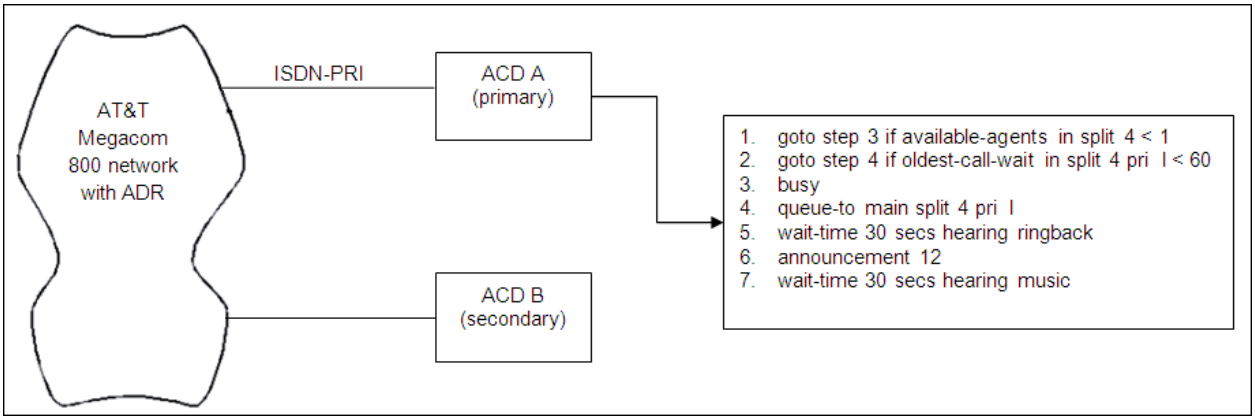
Originator display

For internal calls, the originator display contains the same information as for Basic Call Vectoring, but the originator can receive unwanted display updates during LAI call attempts. In this case, LAI calls must go out over trunk groups that have the **Outgoing Display** field set to `n` and internal callers who call the trunk group can view the dialed digits on their phone display.

LAI with network ADR

Call Vectoring and LAI are compatible with and supplement the network services Alternate Destination Redirection (ADR) rerouting feature or equivalent service from other network providers. ADR uses ISDN-PRI connectivity with the switch in the same manner as LAI to allow the receiving system to indicate whether a call is to be accepted or rejected. The same type of vector that is used as a receiving ACD for LAI is used at the ADR-receiving ACD. If the call is accepted, it is connected to the system. If the call is rejected, the network routing number is translated to another number that routes the call to the alternate location within dialing-plan constraints. ADR allows for only one alternate location. LAI can be used at the alternate location to test other locations for less-busy conditions.

The following figure shows the configuration for a multilocation application.



The network requires ISDN-PRI connectivity to primary location A. Connection to secondary location B does not have to be ISDN-PRI. ADR attempts to route the call to location A over the ISDN-PRI link using a routing number that selects a VDN that is assigned to the receiving vector shown.

When the routing attempt is made, Call Vectoring starts processing the vector. The example then proceeds at location A as follows:

1. Step 1 checks for staffing of the ACD split, and branches to step 3 if it is not staffed.
2. If the ACD split is staffed, step 2 checks the oldest call waiting time in the split, and branches to step 4 if it is less than 60 seconds.
3. If the ACD split is unstaffed or if the oldest call waiting time is more than 60 seconds, step 3 rejects the call and returns a busy indication to the network.
4. If the oldest call waiting time is less than 60 seconds, step 4 accepts the call and queues it. ADR then connects the call through to the receiving system.
5. Steps 5 through 7 provide ringback, announcement, and music to the caller.

If the vector at location A rejects the call by sending a busy indication back to the network over the ISDN-PRI link, ADR reroutes the call to location B which must accept the call. If location B is closed or too busy to take the call, location B can use Call Vectoring and LAI to check other locations. If other locations exist and can take the call, location B can forward the call. If other locations do not exist or cannot take the call, location B can use Call Vectoring to route the call to location A. If location A is not open, location B can use Call Vectoring to provide an announcement or a busy tone to the caller.

Multisite applications for Enhanced LAI

Enhanced LAI has the following applications in a multisite environment:

- You can implement a single queue First-In First-Out (FIFO) operation for any application. However, for maximum efficiency and flexibility, use BSR instead of LAI.
- You can use LAI with BSR in networks with extremely low call volumes.

Meet-me Conference

The Meet-me Conference feature allows you to set up a dial-in conference of up to six parties. The Meet-me Conference feature uses Call Vectoring to process the setup of the conference call.

Meet-me Conference can be optionally assigned to require an access code. If an access code is assigned, and if the vector is programmed to expect an access code, each user dialing in to the conference call must enter the correct access code to be added to the call.

The Meet-me Conference extension can be dialed by any internal or remote access users, and by external parties if the extension number is part of the customer's DID block.

Meet-me Conference command set

The following table lists the commands associated with Meet-me Conference.

Command category	Action taken	Command
Information collection	Collect information from the calling party.	**collect digits**
Treatment	Play an announcement.	**announcement**
	Play a busy tone and stop vector processing.	**busy**
	Disconnect the call.	**disconnect**
	Delay with audible feedback of silence, ringback, system music, or alternate audio or music source.	**wait-time**
Routing	Route to the appropriate meet-me conference and stop vector processing.	**route-to**
Branching or programming	Go to a vector step.	**goto step**
	Go to another vector.	**goto vector**
	Stop vector processing.	**stop**

Information collection commands

collect command

Syntax

```
collect 6 digits after announcement <extension>
```

When the **Meet-me Conf** field is set to `y`, the collect vector step has been modified to collect the next six digits and use those digits as the access code for a Meet-me Conference call. Though not required, the digits can be collected after a recorded announcement.

Treatment commands

announcement command usage for Meet-me Conference

Syntax

```
announcement <extension>
```

The usage for the **announcement** command is the same as in Basic Call Vectoring. For details on using this command, see the Basic Call Vectoring section.

busy command usage for Meet-me Conference

Syntax

```
busy
```

The usage for the **busy** command is the same as in Basic Call Vectoring. For details on using this command, see the Basic Call Vectoring section.

disconnect command usage for Meet-me Conference

Syntax

```
disconnect after announcement <extension>
```

The usage for the **disconnect** command is the same as in Basic Call Vectoring. For details on using this command, see the Basic Call Vectoring section.

wait-time command usage for Meet-me Conference

Syntax

```
wait-time <time> secs hearing <silence, ringback, music>
```

The usage for the **wait-time** command is the same as in Basic Call Vectoring.

Routing commands

The following section details the syntax that can be used for this command and any information that is specific to the Meet-me Conference feature.

route-to meetme command

Syntax

```
route-to meetme
```

The route-to vector step uses the condition meetme only for the Meet-me Conference feature. When successful, this condition adds the caller to the Meet-me Conference call and all parties on the call hear an entry tone to signify that another caller has joined the conference. This condition is valid when the caller has entered the correct access code and there are not already six parties on the call.

If the route to meetme step ever fails, vector processing stops and the caller hears busy tone.

Branching/programming commands for Meet-me Conference

Meet-me Conference uses several branching/ programming commands, including:

- goto step
- stop

The following sections detail the syntax that can be used for these commands and any information that is specific to their use in Attendant Vectoring.

goto step command for Meet-me Conference

Syntax 1

```
goto step <step #> if meet-me-idle
```

Syntax 2

```
goto step <step #> if meet-me-full
```

The goto step vector step has two conditions used for the Meet-me Conference feature:

- meet-me-idle
- meet-me-full

The meet-me-idle condition routes the first caller accessing a Meet-me Conference to the conference call. An announcement step informs the caller that the caller is the first party to access the call.

The meet-me-full condition is used when the Meet-me Conference already has the maximum of six parties on the call.

Syntax 3

```
goto step <step #> if digits = meet-me-access
```

The goto step vector step supports the option, meet-me access, for the digits condition to verify that the access code is valid. If the access code entered by the caller equals the access code administered for the VDN, vector processing continues.

stop command for Meet-me Conference

The use of the **stop** command is the same as in Basic Call Vectoring. For details on using this command, see the Basic Call Vectoring section.

Activating Meet-me Conference

About this task

Meet-me Conference is available for all Communication Manager releases that support Avaya Call Center R11 or later.

To enable the Meet-me Conference feature:

Procedure

1. Set the **G3 Version** field on the System-Parameters Customer-Options screen to V11 or later.
2. Set the **Enhanced Conferencing** field on the System-Parameters Customer-Options screen to `y`.

   Enhanced Conferencing has an Right-to-Use (RTU) cost and must be enabled through the License File process.

Creating a Meet-me Conference VDN

About this task

To create a Meet-me Conference VDN (using example VDN 36090):

Procedure

1. Enter the `add vdn 36090`. The following VDN screen appears:

```
add vdn 36090                                          Page   1 of   3   SPE A
                            VECTOR DIRECTORY NUMBER

                             Extension: 36090
                             Name: Enhanced Conf. Meet-me VDN
Vector Number: 90

                     Meet-me Conferencing? y
```

2. Enter a name, a vector number, and enter `y` in the **Meet-me Conferencing** field.
3. Press **NEXTPAGE** to display page 2.

   The system displays page 2 of the VDN screen:

```
add vdn 36090                                          Page   2 of   3   SPE A
                            VECTOR DIRECTORY NUMBER

                         MEET-ME CONFERENCE PARAMETERS
       Conference Access Code: 937821
        Conference Controller: 80378
              Conference Type: 6-party
              Route-to Number:
```

4. Enter a conference access code.

   If you do not want an access code, leave the field blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the “init” super user login.

   Security alert:

   Always assign an access code to a Meet-me Conference VDN.

5. Enter a conference controller extension.

   If an extension number is entered, a user at that extension can change the access code for the Meet-me Conference VDN using a feature access code. If this field is blank, only a station user that is assigned with console permissions can change the access code for the Meet-me Conference VDN using a feature access code. In addition, remote access users can change a Meet-me Conference access code using the feature access code.

6. Enter the conference type.

   This field can have the following values:

   - 6-party - Enter this value to administer a regular 6-party conference. This value is the default.
   - expanded - Enter this value to administer up to a 300-party conference.

7. If you set the **Conference Type** field to expanded, use the **Route-to Number** field to administer the ARS/AAR Feature Access Code, the routing digits, and the conference ID digits for the VDN.
8. Press **Enter** to submit the VDN.

Creating a Meet-me Conference vector

About this task

To create a Meet-me Conference vector (using example vector number 90):

Procedure

1. Enter:

   ```
   change vector 90
   ```

   The system displays the CALL VECTOR screen.

2. Enter **y** in the **Meet-me Conf** field.

   This designates the vector as a Meet-me Conference vector.

3. Create a vector as shown in the following example:

```
change vector 90                                        Page   1 of   3   SPE A
                                CALL VECTOR

    Number: 90               Name: Meet-me Vec
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? y           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n

01 collect      6    digits after announcement 12340
02 goto         step   6   if digits           =      meet-me-access
03 collect      6    digits after announcement 12341
04 goto         step   6   if digits           =      meet-me-access
05 disconnect   after announcement 12342
06 goto         step   11  if meet-me-idle
07 goto         step   14  if meet-me-full
08 announcement 12343
09 route-to     meetme
10 stop
11 announcement 12344
```

```
change vector 90                                        Page   2 of   3   SPE A
                                CALL VECTOR

12 route-to     meetme
13 stop
14 disconnect   after announcement 12345
15 stop
16
17
18
19
20
21
22
```

4. Press ENTER to submit the vector.

Meet-me Conference capacity issues

Meet-me Conference calls count towards the maximum number of 3-way and 6-way conference calls.

Users cannot add more parties to a conference call once the system maximum is reached.

For Category A, the number of Meet-me Conference VDNs is a subset of the total number of VDNs allowed in the system.

For Category B, the total number of VDNs and vectors is doubled from the normal limit if both Call Vectoring and Enhanced Conferencing are enabled. However, the maximum number of VDNs and vectors available for call center applications is unchanged.

Meet-me Conference call processing scenario

Joe Davis has a sales review scheduled with four associates located in different cities. He has reserved Meet-me Conference telephone number 865-253-6090. In switch administration, this number has been assigned to vector 90. See the following screen.

```
add vdn 36090                                          Page   1 of  3   SPE A
                          VECTOR DIRECTORY NUMBER

                           Extension: 36090
                                Name: Meet-me VDN                         Vector
Number: 90

                    Meet-me Conference? y
```

VDN 36090 is administered with an access code of 835944.

When each associate calls the Meet-me Conference telephone number, the following vector processing occurs:

```
change vector 90                                       Page   1 of   3   SPE A
                                CALL VECTOR

    Number: 90              Name: Meet-me Vec
                  Attendant Vectoring? n    Meet-me Conf? y          Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n

01 collect      6    digits after announcement 12340
02 goto         step   6   if digits           =      meet-me-access
03 collect      6    digits after announcement 12341
04 goto         step   6   if digits           =      meet-me-access
05 disconnect   after announcement 12342
06 goto         step   11  if meet-me-idle
07 goto         step   14  if meet-me-full
08 announcement 12343
09 route-to     meetme
```

```
10 stop
11 announcement 12344
```

```
change vector 90                                        Page   2 of   3   SPE A
                                 CALL VECTOR
12 route-to     meetme
13 stop
14 disconnect   after announcement 12345
15 stop
16
17
18
19
20
21
22
```

Each caller hears announcement 12340, which says something similar to “Welcome to the Meet-me Conferencing service. Enter your conference access code.” Each caller enters the access code 835944.

The **collect** vector step 1 collects the access code digits. If the access code is valid, vector processing continues with vector step 6. If the access code is invalid, vector processing continues with vector step 3, which plays announcement 12341. Announcement 12341 says something similar to “This access code is invalid. Please enter the access code again.” If the caller enters the wrong access code again, vector processing continues with vector step 5, which plays announcement 12342. Announcement 12342 says something similar to “This access code is invalid. Please contact the conference call coordinator to make sure you have the correct conference telephone number and access code. Goodbye.”

Vector step 6 is only valid for the first caller into the Meet-me Conference. The meet-me-idle condition routes the first caller to announcement 12344 (vector step 11). The recorded announcement says something similar to, “You are the first party to join the call.” The caller is then routed to the Meet-me Conference call by vector step 12 and vector processing stops.

Vector step 7 is used when the Meet-me Conference already has the maximum of six parties on the call. The meet-me-full condition disconnects the caller after playing announcement 12345 (vector step 14). The recorded announcement says something similar to, “This Meet-me Conference is filled to capacity. Please contact the conference call coordinator for assistance. Goodbye.”

If a caller enters the correct access code, is not the first caller, and the conference call is not full, vector processing continues with vector step 8, which plays announcement 12343. The announcement says something similar to “Your conference call is already in progress.” The caller is then routed to the Meet-me Conference call by vector step 9 and vector processing stops. As each caller enters the conference call, all parties on the call hear an entry tone.

When the conference call is over and callers drop out of the conference call, any remaining parties on the call hear an exit tone.

Meet-me Conference interactions

Both Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

If **Enhanced Conferencing** is enabled, but no other vectoring customer options are enabled, only Meet-me Conference vectors can be assigned.

A non Meet-me Conference vector cannot be assigned to a Meet-me Conference VDN and a Meet-me Conference vector cannot be assigned to a non Meet-me Conference VDN.

There will be no restrictions in vector chaining between Meet-me Conference and non Meet-me Conference vectors (for example, using the **goto vector** or **route-to number** commands. When calls interflow from one type of vector processing to another, the calls will be removed from any queue and treated as new calls to vectoring, not a continuation of vectoring.

Call Detail Recording

As parties join a Meet-me Conference, a call record is created if required by system administration. In the call record, the called party is the Meet-me Conference VDN number and the duration is the length of time that the party is included in the call. An individual record for each party is generated when the party drops from the call. One option that records all calls to Meet-me Conference VDNs is to activate the Intra-switch CDR feature and populate all the Meet-me Conference VDN numbers in the system.

If Intra-switch CDR is used with the Meet-me Conference VDNs, the condition code must be set to C for all call records as is done with traditional conference calls when Intra-switch CDR is active.

If Intra-switch CDR is not enabled for Meet-me Conference VDNs, the creation and contents of call records depends on the trunk group translations for external callers to the Meet-me Conference. For internal callers to the Meet-me Conference, no record is generated if the Intra-switch CDR feature is not active for either the Meet-me Conference VDN or the calling extension.

Changing vector types

To change a Meet-me Conference vector to a non Meet-me Conference vector and vice versa, you must first remove all vector steps. If either of the conditions exists, a warning message displays stating the VDNs currently assigned to the vector is not operating as expected. The next time you try to submit a change to the Meet-me Conference VDN, you have to assign the VDN to a Meet-me Conference vector.

Direct Inward Dialing

If the VDN extension is part of the customer DID block, external users can access the conference VDN. If the VDN extension is not part of the customer DID block, only internal callers on the customer network, including DCS or QSIG, or remote access callers can access the conference VDN.

Disabling Enhanced Conferencing

If Meet-me Conference VDNs are assigned when disabling the **Enhanced Conferencing** field, the change is not allowed and the message, “Must first remove all Meet-me Conf VDNs and vectors ” appears. You must remove the VDNs and vectors before you set the **Enhanced Conferencing** field to n.

Removing stations

A station that is administered as a controlling station for a Meet-me Conference VDN cannot be removed without first removing the assignment on the VDN. The following message displays:

```
Must first remove as conference controller on VDN form.
```

Network Call Redirection

Network Call Redirection (NCR) provides a call routing method between sites on a public network or a Virtual Private Network (VPN) reducing trunking costs. Cost reductions are particularly valuable in enterprises or multisite call center environments where trunk costs are high.

Public Switch Telephony Networks (PSTNs) or VPNs manage call redirection when an incoming call arrives at Communication Manager that has the NCR feature enabled. As a result, trunks that the server normally retains to accomplish a necessary trunk-to-trunk transfer are released after call redirection takes place.

Cost reductions associated with reduced trunk use can be significant, particularly when you implement Avaya virtual routing features such as Best Service Routing (BSR) with Expected Wait Time (EWT). You can achieve cost savings by using fewer trunks to handle the same number of incoming or outgoing calls after implementing NCR within the local Communication Manager.

NCR options supported by PSTNs

This section describes the various NCR redirection options that are supported by Public Switched Telephone Networks (PSTNs). All the NCR protocols described in this section support Information Forwarding using User-to-User (UUI) transport to the redirected-to location over the PSTN or VPN network.

Protocols not supported by NCR

The PSTN call-redirection protocols that are currently available but are *not* supported by the NCR feature include the following:

- AT&T 4ESS Out-of-Band Transfer and Connect
- Nortel Release Link Trunks (RLT)

NCT-type feature operations

A key advantage of Network Call Transfer (NCT)-type protocols invoked by Call Vectoring or by manual station call-transfer or call-conference/release operations is that the redirecting server retains control over the call and can continue to use a trunk-to-trunk connection if the PSTN switch does not accept the request to merge B-channels for both legs of the call. If the PSTN switch returns a `failure` message to the PSTN, the originating server maintains a trunk-to-trunk

connection for the call. For vector call processing, NCR call processing still treats the NCR attempt to be successful, but the following outcomes occur:

- A vector event is logged to indicate that the NCT operation as attempted with the PSTN failed.
- If CMS is used to track incoming calls to an externally measured VDN, the call is not counted as deflected.

For NCR invocation by Call Vectoring, the local Communication Manager sets up the second leg of the call, waits for the second site to be connected, and requests the PSTN or VPN switch to merge the first leg of the call with the second leg. If this request is accepted, the PSTN or VPN switch joins the original caller to the redirected-to endpoint, sends a `success` message back to the redirecting server and then drops both legs of the call at the redirecting server.

For NCR NCT or TBCT invocation by a station, ACD agent, VRU, or CTI-controlled call doing a call-transfer or call-conference/release operation, if the second leg of the call is set up over an outgoing trunk in the same signaling group as the incoming call, then call-redirection takes place when the call-transfer or call-conference/release occurs. For the NCR ISDN ETSI ECT protocol, the call redirection will take place when the outgoing trunk B-channel either has the same or a different D-channel than the incoming call.

Specific NCT-type protocols

Specific NCT-type protocols include the following protocols:

MCI Network Call Transfer over ISDN PRI

Network Call Redirection and PSTN switch operations associated with the MCI NCT protocol are consistent with those described in NCT-type feature operations on page 371.

MCI Network Call Redirection/Network Call Transfer is compliant with ANSI Explicit Network Call Transfer (ENCT) T1.643 (1995), the MCI Nortel variant of ANSI ECT (1995).

Note:

MCI NCT is offered in the United States by MCI for their Nortel DMS-250 and Alcatel DEX-600 PSTN switches.

Two B-Channel Transfer (TBCT) over ISDN PRI

Network Call Redirection and PSTN switch operations associated with the TBCT protocol are consistent with those described in NCT-type feature operations on page 371.

The Network Call Redirection/Telcordia Two B-Channel Transfer (TBCT) protocol is compliant with the Telcordia Two B-Channel Transfer and ANSI Explicit Call Transfer (1998) standards. For more information, see any of the following:

- Telcordia GR-2865-CORE
- ANSI T1.643 (1998)
- Lucent 99-5E-7268

Note:

TBCT is offered in the U.S. by AT&T for their DMS-100 PSTN switches configured with the NI2 network protocol. TBCT is offered in Canada by Bell/Canada for their DMS-100 PSTN switches and by AT&T/Canada, for their DMS-500 PSTN switches.

ETSI Explicit Call Transfer

Network Call Redirection and PSTN switch operations associated with the European Telecommunications Standard Institute (ETSI) Explicit Call Transfer (ECT) protocol are consistent with those described in NCT-type feature operations on page 371.

The Network Call Redirection/ETSI Explicit Call Transfer protocol is compliant with ETSI standard EN 300 369-1.

Note:

ETSI ECT is offered in Europe by France Telecom and other in-country PSTN service providers for their Ericsson AXE-10 PSTN switches. ETSI ECT is offered in the United Kingdom by the company initially known as MCI for their DMS-100 PSTN switches.

Selection of outbound call leg for NCT-type NCR protocols

For the MCI NCT and Telcordia TBCT NCR protocols, the PSTN switch requires that the outbound call leg of a redirected PRI call is in the same trunk group and has the same Direct Access Line (DAL) D-channel as the inbound call. For vector-initiated invocation of the NCR feature by either a BSR queue-to best or route-to number vector step, the Avaya Communication Manager enforces this requirement by automatically selecting an outbound B-channel that has the same signaling group as the incoming call's D-channel. This results in sending the NCR invocation request on the same D-channel used for the first call leg's associated signaling or for the same associated D-channel when the Non-Facility Associated Signaling (NFAS) D-channel backup configuration is used.

For the ETSI ECT NCR protocol, there is no restriction that the outbound PRI call leg must have the same Direct Access Line (DAL) D-channel used for the first call leg's associated signaling. If the PRI trunk group has more than one associated D-channel, NCR processing sets up the second call leg for call redirection using any B-channel in the trunk group independent of its associated D-channel.

NCD

The Network Call Deflection (NCD) operation by a PSTN switch can occur only if the incoming call to Communication Manager is not answered, that is, an ISDN CONNECT message is not sent to the PSTN switch from the incoming server.

NCD is compliant with ETSI Supplementary Services Network Call Deflection ETS 300 207-1 (partial call rerouting in the public network).

Important:

Some Call Vectoring commands cause CONNECT messages to be sent to the PSTN switch. If Call Vectoring methods are used to implement NCR and the PSTN switch supports the

NCD protocol, call vectors used to invoke NCR must not include any of the following vector commands:

- **announcement**
- **collect x digits**
- **converse-on split/skill**
- **wait hearing music**
- **wait hearing (announcement extension) then (continue, music, ringback or silence)**

When Communication Manager invokes NCD, the PSTN switch sets up the second leg of the call instead of the redirecting Communication Manager. There are two PSTN options for NCD specified by the ETSI standards: *retain call until alerting/connect* and *clear call upon invocation*. The *clear call upon invocation* option is commonly referred to as a *partial call reroute*.

When the *clear call on invocation* option is used, a successful NCR/NCD attempt is indicated when the PSTN or VPN switch has validated the NCR request and sends a *call reroute return* DISCONNECT message to the originating server. In this case, the server loses control of the call after it is transferred to the PSTN or VPN redirection endpoint, and no alternate transfer method is possible if the PSTN or VPN switch fails to transfer the call to the second location.

The *retain call until alerting/connect* option is not available because there are presently no known PTSN or VPN offers with this protocol. With this option, the PSTN or VPN switch sets up the second leg of the call, waits until an ALERTING message is received, and then sends a *call reroute return* FACILITY message followed by a DISCONNECT message to the originating server. In this case, if the second leg of the call fails, the server can redirect the call with a trunk-to-trunk connection so that the call is not lost.

NCD is offered in Europe by British Telecom for their Marconi/Plessey System X and Ericsson AXE10 PSTN switches and by Deutsche Telecom for their Siemens EWSD and Alcatel S12 PSTN switches. NCD is offered in Australia by Telstra for their Alcatel S12 PSTN switches.

Reserving trunk group B-channels for NCT-type redirection operations

Administer the Trunk groups used for NCR NCT-type operations as Call-By-Call (CBC) trunk types. Since CBC trunk groups can carry both incoming and outgoing call traffic, situations can occur in which transient levels of incoming traffic occupy all available B-channels. When no B-channels are available for outgoing calls, attempts to set up the outgoing leg for a redirected call fail and the call redirection fails.

Important:

When the NCR feature is used with high volumes of incoming calls, reserve a minimum number of trunk members for the outgoing leg of redirected calls by using CBC Trunk Group Allocation administration. However, the optimum number of trunk members to reserve depends on traffic patterns that are specific to each call center. Call traffic analysis determines if reservation of B-channels is necessary.

If a trunk group has multiple D-channel signaling groups, the CBC Trunk Group Allocation operation does not guarantee the reservation of outgoing trunks associated with a particular D-channel. Instead, the operation reserves outgoing trunks considering the entire trunk group. Therefore, when NCR is attempted for a trunk group having multiple D-channels, the CBC Trunk Group Allocation operation does not always prevent the blockage of the NCR second call leg setup due to unavailable outgoing trunk B-channels.

To reduce the blockage of NCR NCT-type operations due to no available outbound trunk B-channels:

About network facility types

Before you allocate a minimum number of trunk group members to the outgoing legs of NCR calls, you must administer more than one ISDN services or features. The Network Facilities screen includes two predefined features and ten predefined services. The predefined entries are associated with either Network Specific Facilities (NSF) Type 0 or Type 1. You can administer additional user-defined services or features on the Network Facilities screen. User-defined facilities can be Type 0, 1, 2, or 3. You must obtain support agreements with your PSTN service provider for Type 0 or Type 1 facilities.

Type 2 (incoming) and Type 3 (outgoing) facilities do not use NSF codings or require special support by the PSTN. The network facility types are offered because NSF information is not included with ISDN calls in some regions of the world.

Important:

If PSTN does not support NSF, you must specify a Type 3 facility when you reserve trunk members for NCR operations and enable the Usage Allocation Enhancements Optional feature before you can administer a Type 3 facility.

Example trunk allocation for PSTNs that supports NSF codings

The following example of the Network Facilities screen includes the basic default predefined services and features.

```
change isdn network-facilities                                  Page   1 of   2
                                NETWORK-FACILITIES
                         Facility                                Facility
        Name            Type  Coding           Name             Type  Coding
    sub-operator         0    00110       mega800               1    00010
    operator             0    00101       megacom               1    00011
    outwats-bnd          1    00001       inwats                1    00100
    sdn                  1    00001       wats-max-bnd          1    00101
    accunet              1    00110       lds                   1    00111
    i800                 1    01000       multiquest            1    10000

    ____________         _    _____
    ____________         _    _____
    ____________         _    _____
    ____________         _    _____
    ____________         _    _____
    ____________         _    _____
```

Once you specify the network facilities, you can allocate trunk members on the basis of specific facilities. The following example shows a CBC Trunk Group Allocation screen for a CBC trunk group for which a minimum of one B-channel is always available for the outgoing legs of

redirected calls when the mega800 service is used. The specific feature or service that you specify in this screen depends on the support provided by the PSTN.

```
change trunk-group 29
                                    CBC TRUNK GROUP ALLOCATION

Usage Allocation Plan 1        Usage Allocation Plan 2        Usage Allocation Plan 3

                 Min# Max#                      Min# Max#                      Min#
Max#
Service/Feature Chan Chan       Service/Feature Chan Chan       Service/Feature Chan
Chan
mega800           1   99
```

Example trunk allocation for PSTNs that do not support NSF codings

The following example of the Network Facilities screen includes the basic default predefined services and features and an additional user-defined, Type 3 (outgoing) feature (bsr-redirect).

```
change isdn network-facilities                                  Page   1 of   2
                            NETWORK-FACILITIES
                        Facility                              Facility
        Name           Type  Coding           Name           Type  Coding
    sub-operator        0    00110       mega800             1    00010
    operator            0    00101       megacom             1    00011
    outwats-bnd         1    00001       inwats              1    00100
    sdn                 1    00001       wats-max-bnd        1    00101
    accunet             1    00110       lds                 1    00111
    i800                1    01000       multiquest          1    10000
    bsr-redirect        3    _____
    ____________        _    _____
    ____________        _    _____
    ____________        _    _____
    ____________        _    _____
    ____________        _    _____
```

After administering the user-defined feature, you can specify a minimum number of reserved trunk channels to remain available for the outgoing legs of redirected calls when the feature is used.

```
change trunk-group 42
                                    CBC TRUNK GROUP ALLOCATION

Usage Allocation Plan 1        Usage Allocation Plan 2        Usage Allocation Plan 3

                 Min# Max#                      Min# Max#                      Min#
Max#
Service/Feature Chan Chan       Service/Feature Chan Chan       Service/Feature Chan
Chan
bsr-redirect      5   25
```

AT&T In-Band Transfer Connect service

This section describes PSTN redirection operations associated with the AT&T In-Band Transfer Connect℠ service. Details of the service are described in “AT&T Technical Reference 50075” on the AT&T Web site.

NCR provides Information Forwarding support for the AT&T In-Band Transfer Connect℠ network service ISDN D-channel data-forwarding capability. The Information Forwarding feature forwards the UUI that is associated with the call to the redirected-to location. When Call Vectoring and AT&T In-Band Transfer Connect℠ are used to transfer a call and NCR is enabled for the

system, the **disconnect** vector step causes UUI IE information to be inserted into the ISDN DISCONNECT message generated by a successful AT&T In-Band Transfer Connect operation.

Note:

For information on NCR administration and other administration measures that are required when the AT&T In-Band Transfer Connect℠ service is used, see *Administering Avaya Aura® Call Center Elite*.

AT&T In-Band Transfer Connect℠ operations can be initiated by Call Vectoring after performing the following administration steps:

1. Administering a **route-to number** vector step with an announcement extension, where the associated announcement is recorded with Dial Tone Multi-Frequency (DTMF) tones that include a *T followed by a PSTN endpoint number.
2. Administering a BSR location **VDN Interflow** field on the BSR Application Plan screen as an announcement extension, where the associated announcement is recorded with DTMF tones that include a *T followed by a PSTN endpoint number.
3. Administering a BSR location **VDN Interflow** field on the BSR Application Plan screen as a local switch VDN number associated with a vector that contains an announcement step, where the associated announcement is recorded with DTMF tones that include a *T followed by a PSTN endpoint number.

When the **route-to number** vector in action 1 is executed or when a **queue-to best** vector step is executed and the BSR location described in action 2 or action 3 is selected as the BSR best location for call interflow, the AT&T In-Band Transfer Connect℠ operation succeeds, but no UUI IE information is sent to the redirected-to PSTN endpoint by Communication Manager. However, for step 3, NCR can be administered for use with the AT&T In-Band Transfer Connect℠ service and a **disconnect hearing announcement none** vector added after the **announcement** step such that UUI information associated with the call is passed to the routed-to endpoint when the call redirection is completed. This UUI information can be used in agent screen pop-ups at the redirected-to PSTN endpoint where the call is interflowed.

BSR call flow resulting in AT&T In-Band Transfer Connect UUI IE

A typical BSR call flow that results in the UUI IE being inserted in the ISDN DISCONNECT message during a successful AT&T In-Band Transfer Connect operation is as follows:

1. A PRI call from the PSTN switch arrives at the local Avaya Communication Manager and is routed to a VDN that uses a vector to do subsequent BSR vector processing.
2. The BSR polling vector steps on the local server receive status information from various local skills and remote BSR locations, and identify a remote call center site as the BSR best location.
3. Call control passes to the interflow VDN selected as the BSR best location specified on the Best Service Routing Application Plan screen.

   For information on administering a BSR application plan, see “Call Vectoring methods used with AT&T In-Band Transfer Connect service” in the *Administering Avaya Aura® Call Center Elite* document.

Important:

The **Net Redir** field in the BSR application plan for the remote location must be set to `n`.

4. The vector associated with the interflow VDN for the BSR best location includes the following:
   - An **`announcement`** vector step that specifies an extension for which a special sequence of DTMF digits has been recorded. The recorded DTMF digits return in-band information about the redirected-to endpoint back to the PSTN. The DTMF digits provided in the announcement are entered from a Touch-tone keypad with the following format:

     *T + PSTN number

     T corresponds to the number 8 button on a DTMF keypad and the PSTN number represents the PSTN endpoint number where the call is redirected.

     Note:

     The phone equipment required to create the announcement is described in “About setting up DTMF announcements for the AT&T In-Band Transfer Connect service” in the *Administering Avaya Aura® Call Center Elite* document.
   - A **`wait-time hearing silence`** step providing a brief interval to allow sufficient time for the PSTN switch to process the DTMF digits.
   - A **`disconnect after announcement none`** vector step. This vector step sends an ISDN DISCONNECT message that includes a UUI IE. The UUI IE contains the Avaya Information Forwarding for the call that is sent to the PSTN switch.
5. The PSTN switch makes the connection to the specified redirected-to endpoint and releases the B-channel connection with Communication Manager.

Call Vectoring methods used with the AT&T In-Band Transfer Connect service

The BSR feature invokes the AT&T In-Band Transfer Connect℠ service. You can administer an interflow VDN in the BSR Application table to route to a near-end Communication Manager VDN. The VDN causes the execution of a *T announcement vector step rather than setting the same interflow VDN number to route directly to a VDN on a far-end Communication Manager.

You must administer the BSR location on the BSR Application Plan screen to enable a BSR polling vector to identify a BSR best location and route the incoming call to the location. The BSR Application Plan must meet the following requirements:

- The plan must include at least one interflow VDN that is associated with the vectors that include the vectors steps necessary for successful invocation of the AT&T In-Band Transfer Connect℠ service.
- The **Net Redir** field associated with a location where AT&T In-Band Transfer Connect℠ is used must be set to `n`.

Example: The following example shows how you can use BSR with the AT&T In-Band Transfer Connect℠ service to implement call redirection. In the example, local interflow and not remote interflow VDN numbers are assigned on the BSR Application Plan screen.

```
                  BEST SERVICE ROUTING APPLICATION

Number: 1    Name:        Maximum Suppression Time: 60      Lock? y

Num  Location Name    Switch Node     Status Poll VDN   Interflow VDN    Net Redir?
1    Omaha             320           95022011          4004                 n
2    Paris             320           95022111          4005                 n
3    Sydney            320           95032211          4006                 n
```

The example application plan lists VDN extension numbers that are local to Communication Manager. Each VDN is associated with vectors that are designed to invoke the AT&T In-Band Transfer Connect℠ service operations.

Each vector associated with the interflow VDNs listed in the application plan includes the following steps:

```
1. announcement 1234 [*8 and PSTN number for remote site]
2. wait-time 2 secs hearing silence
3. disconnect after announcement none
```

In the vector example, step 1 provides the extension for an announcement that plays the DTMF digits. Step 2 includes a **wait-time** step to give the PSTN switch time to process the in-band information sent by the announcement in the preceding step before the call is disconnected at Step 3. The **disconnect** command sends an ISDN DISCONNECT message that includes the Information Forwarding data for the call in a codeset 0 or 7 UUI IE element.

Important:

The type of Information Forwarding data sent to the PSTN depends on how the **UUI IE Treatment** field on the Trunk Features page of the Trunk Group screen is administered:

If the **UUI IE Treatment** field is set to `Service Provider`, the ASAI user data is forwarded to the PSTN in the ISDN DISCONNECT message. If the **UUI IE Treatment** field is set to `Shared`, the call center-related data described in NCR and Information Forwarding is forwarded to the PSTN in the ISDN DISCONNECT message.

You can use the call vector example to begin the AT&T In-Band Transfer Connect℠ service with non BSR vector programming. To execute the example vector in a non-BSR vector call flow, apply the guidelines and notes related to the *T announcement format.

NCR implementation methods

NCR operations by Call Vectoring method

Note:

The following description does not apply when the AT&T In-Band Transfer and Connect℠ service is used with NCR.

1. The PSTN switch sends an incoming ISDN call to Communication Manager, where the call enters vector processing.

2. One of the following occurs:
   - If the Communication Manager trunk group and PSTN or VPN switch are configured to use an NCT-type redirection protocol, the redirecting Communication Manager returns an ISDN CONNECT message to the PSTN switch. Any of the following vector commands can be used to return the message:
     - **announcement**
     - **collect** *x* digits
     - **converse-on split**
     - **wait** hearing *music*
     - **wait** hearing *(announcement extension)* then (“continue”, “music”, “ringback” or “silence”)

     Note:

     If the redirecting Communication Manager does not execute one of the listed vector steps, a CONNECT message is automatically returned to the PSTN switch.
   - If the server trunk group and PSTN or VPN switch are configured to use the NCD redirection protocol, a CONNECT message is not sent to the PSTN switch. Therefore, when the NCD protocol is applied, none of the listed vector commands are included in call vectors that implement NCR.
3. Call processing proceeds to either a **route-to number** ~*r* or to a BSR **queue-to best** vector step. Depending on which type of redirection is administered for the incoming trunk group, either NCT-type or NCD processes are initiated. In either case, a FACILITY message is sent to the public network over the D-channel associated with the incoming trunk to invoke redirection of the call.

   Note:

   The following items pertain to the PSTN or VPN endpoint number and to the receiving vector for the interflow location.
4. The PSTN or VPN switch indicates redirection success or failure consistent with the protocol-specific operations described in NCR options supported by PSTNs. An unsuccessful NCR attempt results in one of the following outcomes:
   - If an NCT-type protocol is used, the redirecting Communication Manager establishes a trunk-to-trunk connection.
   - If the NCD protocol is used and the Avaya DEFINITY version is earlier than load 37 of Release 10, vector processing continues to the next vector step that follows the **queue-to best** vector step without any best local BSR call treatment.
   - If the NCD protocol is used, the call can be redirected to the best location by means of a trunk-to-trunk connection. However, the ability of the originating server to establish such a trunk-to-trunk connection depends on the specific features of the NCD protocol in use.

Using BSR queue-to best vector step to activate NCR

NCR is especially useful for multisite call center operations in which the Best Service Routing (BSR) feature is activate as the number of PRI B-channels needed for call interflows is reduced. The **queue-to best** vector step can be used to interflow ISDN calls between Communication Managerover the PSTN. This method provides the best approach for balancing loads across a multisite environment and is more cost effective and accurate than pre-delivery routers.

NCR is activated by the **queue-to best** vector step when the BSR feature determines a BSR location is the best BSR location and that location is administered with the **Net Redir** option set to y on the BSR Application Table screen.

Note:

The administered **Interflow VDN** field on the Best Service Routing Application screen must be a PSTN or VPN endpoint number without a trunk/ARS/AAR access codes included. For some PSTN switch dialing plans, the long-distance access code. For example, 1 in the United States must be prefixed to the PSTN number for the call to be successfully routed by the PSTN switch.

As shown in the following example, the Best Service Routing Application Plan screen must include locations that have the **Net Redir** field set to y.

```
                  BEST SERVICE ROUTING APPLICATION

Number: 1    Name:        Maximum Suppression Time: 60      Lock? y
Num  Location Name   Switch Node    Status Poll VDN  Interflow VDN    Net Redir?
1    Omaha                          95552011       3035551211           y
2    Paris                          95552022       18005551234          y
3    Sydney                         95552033       18665553456          y
```

An appropriate vector is then used to identify a BSR best location and NCR is activated by the queue-to-best vector step.

```
wait 2 seconds hearing ringback
consider skill 1 pri l adjust-by 0
consider location 1 adjust-by 20
consider location 2 adjust-by 40
consider location 3 adjust-by 20
queue-to best
```

Using route-to number ~r vector step to activate NCR

This method can be used to invoke NCR when a **route-to number** vector step that specifies a number that begins with the ~r character. This method can be used to invoke NCR with or without the **LAI**option set to y or with Attendant Call Vectoring active.

Note that the administered route-to number vector step number field must be a PSTN or VPN endpoint number without a trunk/ARS/AAR access codes included. For some PSTN or VPN switch dialing plans, the long-distance access code (for example, a “1” in the United States) must be prefixed to the number for the call to be successfully routed by the PSTN or VPN switch.

Example route-to number ~r vectors: The following examples show vectors that include **route-to number** commands to activate NCR, either with or without use of the Attendant vectoring feature.

```
wait 0 seconds hearing ringback
goto step 4 if skill oldest-call < 30 secs
route-to number ~r13035403001
queue-to skill 35 priority m
...
```

```
goto step 6 if time-of-day is all 17:00 to 09:00
wait 0 seconds hearing ringback
queue-to attd-group
wait 999 secs hearing music
stop
route-to number ~r13035551002
```

Using vector/VDN variables with route-to number ~r to activate NCR

The **number** field of the **route-to number ~r** vector command can be administered with a global vector variable A-Z or AA-ZZ instead of a PSTN endpoint number after the leading *~r* characters. The **number** field of the **route-to number ~r** vector command can also be administered with a VDN variable V1-V9 instead of a PSTN endpoint number after the leading *~r* characters.

An example of using the **route-to number ~r** vector command with a vector variable in the **number** field is shown in the following example. For this example, it is required in the Variables for Vectors screen that the following administration is done:

- Vector variable A is defined as of type collect for digit-buffer and L for local
- Vector variable T as of type tod to contain the current system clock time-of-day value

```
1.  goto step 5 if T < 0700      [if time-of-day is less than 7:00 a.m., set up NCR
                                   call-redirection to out of hours PSTN endpoint]
2.  goto step 5 if T > 1800      [if time-of-day is after 6:00 p.m., set up NCR
                                  call-redirection to out of hours PSTN endpoint]
3. set A = none CATR 18005555555   [set digit-buffer to in-office
                                              hours PSTN endpoint number]
4. goto step 6 if unconditionally [jump to step 6 to do NCR call-redirection ]
5. set A = none CATR 18661111111 [set digit buffer to out-of-office hours
                                                 PSTN endpoint number]
6. route-to number ~rA [initiate NCR call-redirection operation]
```

NCR activation using ASAI Call Transfer and third-party Merge/Release operations

NCR NCT-type operations are activated by ASAI call processing when the Call Transfer or Third-Party Merge/Release operation is performed by a CTI application. This occurs in the following manner:

1. This is typically initiated by the CTI application user selecting an icon, menu item, or button to transfer an answered incoming ISDN call to another party over the PSTN.

Since the incoming ISDN call must be connected to a station user before the Call Transfer or Third-Party Merge/Release operation is requested, NCR can only initiate the call redirection if an NCT-type protocol is optioned on the trunk.

2. If a call arrives at an ASAI-monitored VDN, ASAI will send appropriate information in the ASAI disconnect event to notify the CTI application that the call has been redirected by NCR.

For the ASAI operations listed above to succeed, the following conditions must be in effect:

- The **Network Call Redirection** field is set on the System Parameters Customer Options screen.
- An NCR NCT-type protocol is administered for both the incoming and outgoing call ISDN trunk group.
- The PSTN number that the CTI application uses to redirect an incoming ISDN call to another PSTN endpoint must be added to the ARS digit analysis screen in such a way that for the NCR MCI NCT and TBCT protocols, the second leg of the call transfer uses the same trunk group with a trunk that has the same D-channel as the incoming call. For the NCR ETSI ECT protocol, the CTI-initiated second call leg can be over a different trunk group with a different signaling group than the incoming call leg.

Note:

NCR-related AAR/ARS routing table administration is required for station transfer or conferencing with MCI trunks.

Other things to know about using NCR with ASAI

Using ASAI data for call tracking

ASAI event reporting allows tracking of ISDN ACD calls that were redirected by NCR in a multi-server call center environment. These calls can be tracked by the UCID assigned to each call, or by the UUI information inserted by the application through either the Third Party Make Call or Adjunct Routing features.

ASAI drop event

Successful NCR call redirection causes an ASAI drop event to be sent to the CTI application with a CV_REDIR cause value of decimal (30) after the redirection is completed. Only one NCR drop event is received for a successful NCR operation when the NCT PSTN feature is used, even though two trunks are dropped by the PSTN.

ASAI third-party merge/call transfer

The CTI application requests a third-party merge/call transfer ASAI operation to transfer the call to the second Communication Manager. This is only used if Network Call Transfer is not available. Once the two calls merge, then ASAI sends a third-party acknowledgement, and when the call is completed, ASAI sends a drop event report, and the third-party call ends.

NCR activation using station call transfer or call conference/release operations

When an incoming ISDN call over a trunk with NCT-type PSTN service is answered at a station or voice response unit (VRU), the station user or VRU places the call on hold, and dials the number

for a PSTN of VPN endpoint where the outgoing trunk B-channel is determined by AAR or AAS routing. The station user initiates a call transfer using the Transfer feature button or a switch hook flash, or the VRU initiates a call transfer by using an analog or line-side E1/T1 switch-hook flash.

The switch automatically sends an “invoke NCT” FACILITY message when the transfer is completed if the following conditions are met:

- The **Network Call Redirection** field is set to `y` on page 3 of the System Parameters Customer Options screen.
- An NCT-type protocol is administered for both the incoming and outgoing call ISDN trunk group.
- The second leg call is eligible for redirection by means of an NCT-type protocol, which requires for the MCI NCT and TBCT protocols the second leg of the call is in the same trunk group and has the same signaling group as the incoming call. For the NCR ETSI ECT protocol, the second leg of the call can be over a different trunk group with a different signalling group than the incoming call leg.

If the station user or IVR initiates and completes a three-way conference instead of doing a call transfer operation as above, and releases or hangs up from the conference with the following condition also being met, the switch automatically sends an “invoke NCT” ISDN message to the PSTN or VPN switch if also the following condition is met:

- The number of parties in the conference including the conference originator must be no greater than three parties.

Note:

NCR-related AAR/ARS digit-analysis and routing table administration is required for correctly setting up the second call leg over NCT-type trunks associated with the station or IVR call transfer and call conference/release operations.

NCR activation using ASAI adjunct route operations

NCR can be invoked by specifying the activate NCR option for the ASAI route message in a route select ASAI message sent by a CTI application to the Avaya Communication Manager after an adjunct routing vector step is executed during call vector processing. This Communication Manager capability provides greater flexibility for CTI applications to directly route calls to PSTN or VPN endpoints without the need to specify a VDN extension in the route select ASAI message to route the call instead to a VDN and vector step that activates NCR via a route-to number ~r or queue-to-best vector step. The invocation of NCR by the adjunct routing vector step route select ASAI message for various NCT-type protocols follows the same rules as used for the route-to number ~r or queue-to-best vector step operations.

For more information, see the following ASAI documents:

- For information on the call options codepoint for NCR routing or the ASAI Call Route Selection message, see the *ASAI Protocol Reference* document.
- For information on feature interactions, see the *ASAI Technical Reference* document.

NCR considerations

Limitations of call redirection

Limitation	Description
NCR feature support	PSTN support for NCR varies with the geographical location and can be limited or absent in some areas. Consult your Avaya account team to determine the PSTN service provider availability of the NCR protocols in your area.
NCD redirection protocol support	At this time, no PSTNs offer the Network Call Deflection “retain call until alerting or connect” operation. Therefore, only the Network Call Deflection “clear call upon invocation” offer is available from PSTNs. Both methods are described in this document. Negotiate with your PSTN as the NCR feature works on either platform. NCR is limited by which PSTN platform is available to you.
Allowable number of redirection per call	There can be limits placed on the number of times a call can be redirected over the public network. The limits are imposed by the public network service provider. For example, in the United States, MCI currently allows only one redirection per call. In the United Kingdom, there is a limit of 20 call deflections per call. In addition, there can be additional charges associated with redirected calls.
User-to-User information forwarding support	Some public network service providers do not support forwarding of UUI, including ASAI user data, collected digits, VDN name, the VDN in-time, as reflected by the NETINTIME database items, and the UCID. In such situations, Information Forwarding is lost and the second leg of the redirected call looks like an entirely new call to the redirected-to server at the second location. One of the data items lost is the VDN name, which is rerouted to the originally called service DNIS information. Call forwarding indication can be achieved by using dedicated VDNs for call forwarding, but this strategy loses the benefits of Information Forwarding inherent with NCR and limits use of CTI applications. PSTN service providers typically charge by call or by a monthly rate for the redirect and UUI transport services. For more information on charges, contact your Avaya account team.

NCR operational considerations

Reserving outbound trunk B-channels to ensure NCR operations succeed

When the trunk group service type is set to `call-by-call`, the trunk group Usage Allocation capability can be used to reserve a minimum number of trunk channels for outgoing PRI B-channel calls within the same trunk group and same D-channel.

For more information, see “Reserving trunk group B-channels for NCT-type redirection operations” in the *Administering Avaya Aura® Call Center Elite* document.

Call Vectoring configuration required for successful MCI NCT operations

When NCR is used with the MCI NCT protocol, the VDN call vector the call is redirected to by a successful MCI NCT operation must immediately return an ISDN CONNECT message to the PSTN switch. To meet this requirement, either a **`wait`** *0 secs* hearing *music* or an **`announcement`** vector step must be the first step executed in the redirected-to call vector.

Configuring Ericsson AXE-10 for ETSI ECT operations

Before you begin

Verify that AXE-10 contains VN7 Translocal 4.2 or later software. This software is also called GOAS 2.1 in Ericsson.

Procedure

1. Configure AXE-10 for the `pure ETSI` level.
2. To subscribe to the AXE-10 `ETSI ECT` mode, configure all the PRI trunks used with Communication Manager NCR, or the ETSI ECT feature.
3. On the AXE-10 Trunk Configuration screen, enter the **ECT** category as `ON`.

Note:

Do not configure the AXE-10 PRI trunk to expect a “HOLD” ISDN message when the NCR ETSI ECT feature sends a message as a part of the ETSI ECT invocation sequence.

NCR and Information Forwarding

The Avaya Information Forwarding feature is supported with NCR when the PSTN supports ISDN UUI IE transport in conjunction with the specific network redirection protocol used by the switch.

UUI data included in Information Forwarding for ISDN calls

Information Forwarding forwards the following call center-related data (as User-to-User Information) with an ISDN call:

- Adjunct Switch Application Interface (ASAI) user data
- Universal Call ID (UCID)
- Collected digits
- In-VDN time
- VDN name

UUI data forwarding

When an NCT-type option is used for NCR, the UUI is forwarded by the Avaya Communication Manager in the ISDN SETUP message sent with the call to the second site.

When the NCD option is used for NCR, the UUI is included in the ISDN FACILITY invoke message sent from the Avaya Communication Manager to the PSTN. The PSTN then forwards the UUI to the second site.

When the AT&T In-Band Transfer and Connect service is used for NCR, the UUI is returned in an ISDN DISCONNECT message that includes the data in a codeset 0 or 7 UUI IE element.

PSTN terms used for UUI transport service

For NCT-type options and the NCD option, the PSTN service provider must configure the PRI trunks used with the NCR feature to transport the UUI data associated with the Information Forwarding feature.

The various PSTN terms used in different countries for UUI IE transport are listed in the following table.

Country	UUI transport term	Providers
Australia	UUS Service 1	Telstra
Canada	UUS Service 1	AT&T/Canada Bell Canada
France	Mini-Message	France Telecom
Germany	(included in basic ISDN package)	Deutsche Telecom
Singapore	Not supported	
UK	Not supported	
USA	N-Quest Type 1 Service MA UUI Type 1 Service	Verizon AT&T

NCR interactions

- Attendant Vectoring: Can use the **route-to number** vector step to route calls to attendants located at another Communication Manager node. The operation of the NCR feature using the NCT-type or NCD networks features to accomplish the call redirection is exactly the same as for redirecting ACD calls.
- Advice of Charge: No new capabilities are added for the NCR feature for the Advice of Charge PSTN feature. Use the Advice of Charge feature with the same trunk facilities used for the NCR feature.
- BCMS: No change is made to BCMS for support of NCR. Redirected calls are tracked as completed calls since the PSTN disconnects the incoming facility of the original call when the call is redirected to another site.
- Enhanced Information Forwarding: For the NCR feature, Enhanced Information Forwarding transports UUI for the incoming ISDN call to the PSTN endpoint that receives the redirected call. The use of the Enhanced Information Forwarding capability with NCR requires that the incoming call trunk group be assigned as `shared`, that is, the **UUI IE treatment** field is set to `shared`. However, if the trunk group is set up as `service provider`, only the ASAI user information or user information provided by the incoming ISDN call, is included in the UUI IE sent on a non-shared basis to the redirected-to PSTN endpoint. NCR supports Information Forwarding for AT&T In-Band Transfer and Connect service.
- Look-Ahead Interflow: NCR activation using the **route-to number** vector step does not require LAI to be active to provide multisite capabilities, which are required to check remote locations and to access the BSR Application Plan screen.
- Service Observing by VDN: If the Service Observing by VDN feature is used to service observe a VDN, where the NCR feature is used to redirect incoming ISDN calls, the service observer hears the same tones, music, or announcements heard by the incoming caller before the NCR feature reroutes the call to another PSTN endpoint. When the NCR operation is completed, the service observer is dropped as an observer of the incoming call and placed in the service observing queue associated with the VDN.

- Trunk-to-Trunk Transfer: If you enable the NCR feature and use the ASAI Third-Party make Call/Transfer operation to redirect an incoming ISDN to a PSTN endpoint, set the **Trunk-to-Trunk Transfer** field on the System-Related Customer-Options screen to `y` for the call redirection to succeed. If the route-to number or BSR queue-to best vector step uses the NCR feature to redirect an incoming ISDN call to a PSTN endpoint, the **Trunk-to-Trunk Transfer customer** option does not need to be set to `y`.

  For more information, see “Using the route-to command for NCR” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.
- VDN Return Destination: If you administer the VDN Return Destination (VRD) feature for the VDN associated with a vector that causes the NCR feature to be invoked, VRD is canceled when the call is redirected by NCR.
- CMS Database Items: The following Avaya CMS database items are affected by NCR:
  - DEFLECTCALLS: In the VDN CMS database tables, the DEFLECTCALLS item includes the number of calls that are redirected using NCR through the BSR feature by using the **`route-to number`** or **`queue-to best`** commands. Successful NCR attempts are pegged as DEFLECTCALLS.
  - INTERFLOWCALLS: In the VDN CMS database tables, the INTERFLOWCALLS item includes successful BSR interflows using NCR redirections.
  - LOOKATTEMPTS: In the VDN CMS database tables, the LOOKATTEMPTS item includes the number of times the LAI or BSR interflow was attempted for calls in the vector. Successful LAI or BSR attempts are also counted. NCR invoke attempts (NCD or NCT) are also reflected in LOOKFLOWCALLS.
  - LOOKFLOWCALLS: In the VDN CMS database tables, the LOOKFLOWCALLS item includes the number of INTERFLOWCALLS redirected by the LAI or BSR features. LOOKFLOWCALLS is a subset of INTERFLOWCALLS and includes LOOKATTEMPTS for the LAI or BSR interflows. With BSR interflow using trunk-to-trunk transfer or NCR, every LOOKATTEMPT is also be counted as a LOOKFLOWCALLS unless a failure occurs.

NCR support with SIP

Network Call Redirection (NCR) provides Communication Manager call routing method between sites on a public network or a Virtual Private Network (VPN) that can reduce trunking costs. The cost reductions are particularly valuable in enterprises or multisite call center environments where trunk costs are high.

When an incoming call arrives at a Communication Manager server that has the NCR feature activated, call redirection is managed by the SIP service provider or VPN switch instead of the local Avaya server. As a result, trunks that the server otherwise retains to accomplish a trunk-to-trunk transfer are released after the call redirection takes place.

The cost reductions associated with reduced trunk use can be significant particularly when Avaya virtual routing features such as BSR with Expected Wait Time (EWT), are implemented. The cost-savings are achieved by the Avaya customer requiring fewer trunks to handle the same number of

incoming or outgoing calls after the NCR feature is implemented within the local Communication Manager.

SIP Network Call Redirection protocols

NCR can occur over SIP trunks. The SIP REFER or SIP 302 Moved Temporarily messages contain all the information needed for Network Call Redirection. When the call is answered by an agent or call vectoring event, a SIP REFER message is sent. If a SIP call is not answered by an agent or a call vector process, a SIP 302 Moved Temporarily message is sent.

SIP limitations on call redirection

NCR feature support

SIP service provider support for NCR varies with the geographical location and can be limited or absent in some areas. Consult your Avaya account team to determine the SIP service provider availability of the NCR protocols in your area.

Allowable number of redirection per call

There can be limits placed on the number of times a call can be redirected over the public network. The limits are imposed by the public network service provider. There can be additional charges associated with redirected calls.

User-to-User information forwarding support

Some public network service providers do not support forwarding of UUI, including ASAI user data, collected digits, VDN name, the VDN in-time, as reflected by the NETINTIME database items, and the UCID. In such situations, Information Forwarding is lost and the second leg of the redirected call looks like an entirely new call to the redirected-to server at the second location.

One of the data items lost is the VDN name, which is rerouted to the originally called service DNIS information. The indication that the call has been forwarded can be achieved by using dedicated VDNs for call forwarding, but this strategy loses the benefits of Information Forwarding inherent with NCR and potentially limits use of the CTI applications.

SIP NCR and Information Forwarding

Information Forwarding is supported with NCR when the SIP service provider supports UUI transport in conjunction with the specific network redirection protocol used by the switch.

UUI data included in Information Forwarding for a SIP call

Information Forwarding forwards the following call center-related data (as User-to-User Information) with a SIP call:

- Adjunct Switch Application Interface (ASAI) user data
- Universal Call ID (UCID)
- Collected digits
- In-VDN time
- VDN name

UUI data forwarding with SIP

When NCR is used, the UUI is forwarded by the Avaya Communication Manager in the SIP REFER or 302 Moved Temporarily messages.

The UUI is forwarded by the Avaya Communication Manager in a SIP REFER message if the call has been answered. A call can be answered for example, if an agent answers the call, a vector plays an announcement or music, or the call is processed by a command that provides an answer such as collect digits. The REFER is sent back to the caller, causing the first call to be dropped after a second call is issued and established with the next location.

The UUI is forwarded by the Avaya Communication Manager in a 302 Temporarily Moved message if NCR is invoked before the call is answered. The first call is redirected to the next location.

NCR operations for SIP by Call Vectoring method

The processes by which NCR is implemented for SIP by a Call Vectoring method are summarized in the following steps:

1. The SIP service provider switch sends an incoming call to Communication Manager, where the call enters vector processing.
2. Call processing proceeds to either a **route-to number** ~*r* or to a BSR **queue-to best** vector step.

BSR queue-to best vector step activation of NCR

You can use NCR to reduce the number of trunks for call interflow. This method provides the best approach for balancing loads across a multisite environment and is more cost effective and accurate than pre-delivery routers. You can use the **queue-to best** vector step to interflow calls between Communication Manager over the SIP service provider network. You can use SIP to interflow BSR calls, but not BSR polling. You can use BSR polling through other methods such as Polling Over IP without B-Channel.

The **queue-to best** vector step activates NCR when the BSR feature determines that a particular BSR location is the best location and that location is administered with the **Net Redir** field set to y on the BSR Application Table screen. Note that the administered **Interflow VDN** field on the Best Service Routing Application screen must be a SIP service provider or VPN endpoint number without a trunk/ARS/AAR access codes included. For some SIP service provider switch dialing plans, you must prefix the long-distance access code, for example, 1 in the United States, to the SIP service provider number for the call to be successfully routed by the SIP service provider switch.

As shown in the following example, the Best Service Routing Application Plan screen must include locations that have the **Net Redir** field set to y.

```
                BEST SERVICE ROUTING APPLICATION

Number: 1    Name:        Maximum Suppression Time: 60      Lock? y
```

```
Num  Location Name    Switch Node     Status Poll VDN  Interflow VDN    Net Redir?
1    Omaha                            95552011       3035551211           y
2    Paris                            95552022       18005551234          y
3    Sydney                           95552033       18665553456          y
```

An appropriate vector is then used to identify a BSR best location and NCR is activated by the **queue-to-best** vector step.

```
wait 2 seconds hearing ringback
consider skill 1 pri l adjust-by 0
consider location 1 adjust-by 20
consider location 2 adjust-by 40
consider location 3 adjust-by 20
queue-to best
```

SIP NCR interactions

- Attendant Vectoring: Use the **route-to number** vector step to route calls to attendants located at another Communication Manager node. The operation of the NCR feature using the network call redirection features to accomplish the call redirection is exactly the same as for redirecting ACD calls.
- Basic Call Management System (BCMS): No change is made to BCMS for support of NCR. Redirected calls are tracked as completed calls since the SIP service provider disconnects the incoming facility of the original call when the call is redirected to another site.
- Enhanced Information Forwarding: For the NCR feature, Enhanced Information Forwarding transports User-to-User information (UUI) for the incoming call to the SIP service provider endpoint that receives the redirected call. The use of the Enhanced Information Forwarding capability with NCR (the recommended configuration) requires that the incoming call trunk group be assigned as shared, that is, the **UUI treatment** field is set to `shared`. However, if the trunk group is set up as service provider, only the ASAI user information (or user information provided by the incoming call) will be included in the UUI sent on a non-shared basis to the redirected-to SIP service provider endpoint.
- Look-Ahead Interflow (LAI): NCR activation using the route-to number vector step does not require LAI to be active to provide multisite capabilities, which are required to check remote locations and to access the BSR Application Plan screen.
- Service Observing by VDN: If the Service Observing by VDN feature is used to service observe a VDN, where the NCR feature is used to redirect incoming calls, the service-observer will hear the same tones, music, and/or announcements heard by the incoming caller before the NCR feature reroutes the call to another SIP service provider endpoint. When the NCR operation is completed, the service-observer will be dropped as an observer of the incoming call and placed in the service-observing queue associated with the VDN.
- Trunk-to-Trunk Transfer: If the NCR feature is optioned and the ASAI Third-Party make Call/transfer operation is used to redirect an incoming call to a SIP service provider endpoint, the **Trunk-to-Trunk Transfer** field on the System-Parameter Features screen must be enabled for the call redirection to succeed. If the route-to number or BSR queue-to best vector step uses the NCR feature to redirect an incoming call to a SIP service provider endpoint, the Trunk-to-Trunk Transfer customer option does not need to be set to `y`.

- VDN Return Destination (VRD): If VRD is administered for the VDN that is associated with a vector that causes the NCR feature to be invoked, VRD is canceled when the call is redirected by NCR.
- CMS database items: The following database items are affected by NCR:
  - DEFLECTCALLS: In the VDN CMS database tables, the DEFLECTCALLS item includes the number of calls that are redirected using NCR through the BSR feature by using the route-to number or queue-to best commands. Successful NCR attempts are pegged as DEFLECTCALLS.
  - INTERFLOWCALLS: In the VDN CMS database tables, the INTERFLOWCALLS item includes successful BSR interflows using NCR redirections.
  - LOOKATTEMPTS: In the VDN CMS database tables, the LOOKATTEMPTS item includes the number of times the LAI or BSR interflow was attempted for calls in the vector. Successful LAI or BSR attempts are also counted. NCR invoke attempts are also reflected in LOOKFLOWCALLS.
  - LOOKFLOWCALLS: In the VDN CMS database tables, the LOOKFLOWCALLS item includes the number of INTERFLOWCALLS that were redirected by the Look-Ahead Interflow or BSR features. LOOKFLOWCALLS is a subset of INTERFLOWCALLS and includes LOOKATTEMPTS for the Look-Ahead Interflow or BSR interflows. With BSR interflow using trunk-to-trunk transfer or NCR, every LOOKATTEMPT will also be counted as a LOOKFLOWCALLS unless a failure occurs.

Percentage Allocation Routing

Use Percentage Allocation Routing to distribute calls among a set of call centers or VDNs based on a specified percent allocation. Communication Manager can initially direct incoming calls that arrive at a particular VDN to a Policy Routing Table (PRT) for Percent Allocation instead of to a vector. The PRT distributes the calls to the administered Route-to VDNs based on the administered percent allocation targets.

This feature is useful for segmented call-handling, outsourcing, and optimizing call handling in a multiple-location enterprise. With Percentage Allocation Routing, you can allocate target percentages and, for example, do the following:

- Allocate certain call types among multiple answering groups with similar skills.
- Allocate maximum calls to a more economical calling group.
- Ensure that the organization meets the terms of a service level agreement.

Considerations for implementing percentage allocation routing

- Modification to the table while calls are being routed to the VDN(s) that the PRT is assigned to can result in calls being miss-routed, miss-appropriated or other indeterminate actions. To prevent such a situation from occurring, create a new PRT table with the required changes and replace the old version.

- The following are the valid entries in the **Period** field:
  - `100_count` (default): The call count, displayed in percentage, is reset after the total calls for the PRT reach 100 which is when the total calls match the target routing pattern percentages. This ensures that the routing points have equal distribution of calls all the time.
  - `max_count`: The call count is maintained until calls delivered to one of the VDNs exceed 65,400. At that point, calls are continued to be distributed over the VDNs, but the call counts are reset when the actual percentages equal the targets for all of the VDNs at the same time.
  - `Half-hour`: Resets the call count at the top of the hour and at the 30 minute point.
  - `Hour`: Resets the call count at the top of the hour.
  - `Daily`: Resets the call count at midnight, every night.
  - `Weekly`: Resets the call count at midnight on Saturday.
- The actual percentage and the call count are reset whenever the PRT screen is changed.
- Calls are routed to the VDN destination that is farthest from meeting its target allocation.
- Calls are routed based on the actual call count, not on the actual percentage. When there is no actual call count, at startup or after reset, the route point with the highest target is selected.
- Calls routed through a PRT are reported to BCMS, CMS, and Avaya IQ as though the calls were routed to a VDN assigned to vector 0. VDN reports can be created using the existing CMS custom reporting capabilities to show the percentages and the number of calls that have been distributed among the destination VDNs. Avaya IQ provides additional reporting including what VDN the call was routed to, the PRT table number and the type used, and the matched percentage.
- A PRT can be assigned to multiple VDNs. There is no restriction on the VDN that can be entered as a route point.
- There are limits for the number of PRT tables that can be defined and there is a system routing point (PRTs x VDN destinations) limit. Both the limits can be found in the Communication Manager System Capacity tables and displayed on the Display Capacity screen.
- In addition to add, change, display, list, and remove, the following commands also support PRT:
  - `list vdn`: Displays destination type, Vector or PRT.
  - `list usage policy-routing-table`: Lists the VDNs that use the specified PRT.
  - `list history`: Displays the history of add and change policy-routing-table commands.
  - `list trace vdn`: Displays calls to a PRT.
- When the destination is a PRT, the **Attendant Vectoring** and **Meet-Me Conferencing** fields do not appear.

- Routing through the PRT to the selected VDN is considered as a **route to** command operation and this triggers the *Allow VDN Override* rule that is assigned the previous VDN.
- For VDN domain monitoring with CTI/ASAI, a call to a VDN, that is, routed via a PRT. appears as if the call was routed to the destination VDN by a **route-to number** command in a vector assigned to the original VDN.
- To evenly distribute calls across three routing points, administer the routing points with a 33 percent target, add a fourth routing point that routes back to the PRT with a target of 1 percent. Set the **Period** field to `100_count` and calls are distributed evenly.

Rules for percentage allocation routing

- The target percentages must be in integers. You cannot use decimals or fractions.
- The target percentage of all the VDNs must add up to 100 percent before form submittal.

Percentage allocation routing example

Use the **Destination** and **Number** fields on the Vector Directory Number screen to specify the routing destination as either a vector or a PRT table. To implement percentage allocation routing, specify the routing destination as a PRT table (Destination: Policy Routing Table and Number: PRT table number).

Use the Policy Routing Table screen to specify various routing destinations and percentages allocated for each destination. The following example Policy Routing Table screen includes the Target and Actual traffic routed to each of the VDNs.

```
display policy-routing-table 1957

                              POLICY ROUTING TABLE
Number: 1957  Name: % distribution      Type: percentage   Period: max count

                                                    Target Actual Call
Index Route-to VDN  VDN NAME                         %      %      Counts
  1   2220071       Gizmo support                   25     27.2      3
  2   2221501       Ultra support                    5      9.0      1
  3   2220601       Customer Service South          35     27.2*     3
  4   2220511       Outsourcer Charlie              10      9.0      1
  5   2220501       Survey after service            10      9.0      1
  6   2220072       Outsourcer International        15     18.1      2
  7
  8
  9
 10
 11
 12
                                          Totals 100             11

Command:
```

Service Hours Table Routing

Service Hours Table Routing simplifies the vectors for handling calls based on office hours. Vectors use the service hours routing tables to determine how to handle calls that are received during working and non working hours. Customers can use this feature as an alternative to Time of Day (TOD) routing and can specify the working hours on a daily or hourly basis. You can administer as many as 999 different tables and use the tables to make vectoring decisions. One simple vector command can check if the call meets the administered service hours.

Without Service Hours Table Routing, customers have to add multiple TOD steps in the vectors to define the hours of operation for a specific business application.

Service Hours Table Routing considerations

- Service Hours Table Routing is not available when upgrading from a previous release.
- **Vectoring (Basic)** must be enabled.
- The Call Center Release field must be set to 4.0 or later.

Goto processing for Service Hours Table Routing

When vector processing encounters a goto if service-hours step, it determines if the current day of week and time is within the service hours listed in the corresponding table. This information is used to decide if the `goto` condition is true or false, and therefore, whether or not to go to the given step or vector. The day of week and time match is based on the system time on the Communication Manager that receives the call. The time used in the calculations is the time the call reaches the goto step.

Time adjustments on the Service Hours Table screen

The time used in the calculations can be adjusted using the **Use time adjustments from location** field on the Service Hours Table screen. This field indicates the location number on the Locations screen that specifies how the adjustments are performed.

You can make the following time adjustments using the **Use time adjustments from location** field:

- Adjust the daylight saving time from the system time
- Apply the time zone for a specific location
- Apply the daylight saving time for a specific location

If this field is blank, no adjustments are made.

Service Hours Table

Procedure

1. At the command prompt, type `change service-hours-table xxx`, where *xxx* is number of a service hours table. Press **Enter**.

2. Administer the following fields on the Service Hours Table screen:
   - **Description**
   - **Number**
   - **Start and End**
   - **Use time adjustments from location**
3. Press **Enter** to save the changes.

Goto step command for Service Hours Table Routing

Syntax 1

goto `step x if service-hours in table y`

The command directs the call to a specific vector step if the conditions of the call match the service hours specified in the Service Hours Table.

Syntax 2

goto `step x if service-hours not-in table y`

The command directs the call to a specific vector step if the conditions of the call do not match any of the service hours that are in the specified Service Hours Table.

Goto vector command for Service Hours Table Routing

Syntax 1

goto `vector x @step z if service-hours in table y`

The command directs a call to a specific vector if the conditions of the call match service hours that are in the specified Service Hours Table.

Syntax 2

goto `vector x @step z if service-hours not-in table y`

The command directs the call to a specific vector if the conditions of the call do not match any of the service hours that are in the specified Service Hours Table.

Service Hours Table Routing scenario

The following is a very basic scenario without time adjustments:

Basic Service Hours Table scenario

```
                                SERVICE HOURS TABLE

                                   Number: 99
                            Description: Call-ahead Reservations_____
                  Use time adjustments from location: 2__

     MON           TUE           WED           THU           FRI
  Start  End    Start  End    Start  End    Start  End    Start  End
  08:00 12:30   10:30 02:00   __:__ __:__   __:__ __:__   __:__ __:__
  13:00 16:30   15:00 20:30   __:__ __:__   __:__ __:__   __:__ __:__
  __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__
```

```
    __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__
    __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__   __:__ __:__

                          SAT          SUN
                       Start  End    Start  End
                       __:__ __:__   __:__ __:__
                       __:__ __:__   __:__ __:__
                       __:__ __:__   __:__ __:__
                       __:__ __:__   __:__ __:__
                       __:__ __:__   __:__ __:__
```

VECTOR 1:

goto vector 2 @step 1 if service-hours not-in table 99

<Service hours – Call-ahead Reservation processing>

VECTOR 2:

<After hours processing>

The following table shows how calls at different times will be processed.

Day of week or time of call	Which processing
Sunday or any time	After hours
Tuesday or 14:59	After hours
Tuesday or 15:00	Service-hours
Monday or 12:30	Service-hours
Monday or 12:31	After hours

The same scenario with time adjustments:

```
                                DAYLIGHT SAVING RULES
Rule          Change Day                  Month   Date    Time    Increment
 0: No Daylight Saving
 1: Start: first Sunday       on or after April     1    at 02:00   01:00
     Stop: first Sunday       on or after October   25   at 02:00
 2: Start: first Sunday       on or after April     1    at 02:00   02:00
     Stop: first Sunday       on or after October   25   at 02:00
```

```
LOCATIONS
ARS Prefix 1 Required For 10-Digit NANP Calls? y
 Loc.    Name          Timezone  Rule    NPA                              Proxy Sel.
 No.                   Offset                                             Rte. Pat.
 1:     Main          + 00:00     1
 2:     Branch        + 02:00     2
 3: :
 4: :
 5: :
 6: :
 7: :
 8: :
 9: :
10: :
11: :
12: :
13: :
14: :
```

The following table shows how calls at different times will be processed.

System time	Daylight saving	Adjustments	Adjusted time	Which processing
Tuesday or 07:30	yes	- subtract 1 hour (put system into standard time) - add 2 hours (location time zone) - add 2 hours (location DST)	Tuesday or 10:30	Service-hours
Tuesday or 11:01	yes	- subtract 1 hour (put system into standard time) - add 2 hours (location time zone) - add 2 hours (location DST)	Tuesday or 2:01	After hours
Monday or 06:00	no	- add 2 hours (location time zone)	Monday or 08:00	Service-hours
Monday or 10:31	no	- add 2 hours (location	Monday or 12:31	After hours

Variables in Vectors

With vectors variables, you can create variables to:

- Improve the general efficiency of vector administration
- Provide increased manager and application control over call treatments
- Create vectors that better serve the call center operations

Vector variables are defined in a central variable administration table, but the values assigned to some types of variables can also be quickly changed by means of special vectors, VDNs and FACs that you create for that purpose.

Different types of variables are available to meet different types of call processing needs. Depending on the variable type, variables can use either call-specific data, or fixed values that are identical for all calls. In either case, an administered variable can be reused in many vectors. You can view the vector variable usage by entering the `list measurements summary` maintenance command.

For more information on VIV capabilities, administration requirements, and vector examples, see the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

VDN In a Coverage Path

VICP

VDN in a Coverage Path (VICP) enhances call coverage and call vectoring. If you administer **Call Vectoring (Basic)** or **Call Prompting**, you can assign a VDN as the last point in a coverage path. A vector or Call Prompting processes calls that reach this coverage point.

VICP considerations

- A VDN is not allowed to be a member of a coverage answer group.
- A coverage answer group can only be a point in a coverage path. A vector cannot route a covered call to a coverage answer group.
- Removing a VDN from the system with the **`remove vdn`** <*extension*> command automatically removes the VDN from any coverage paths.
- By default, features such as Call Coverage, Call Forwarding, RONA, or Night Service cannot redirect calls that cover to a VDN. Therefore the **`route-to digits`** or **`route-to number`** command with **cov** set to `y` is treated the same as **cov** set to `n`.
- The **Cvg Enabled for VDN Route-to Party** field on the Coverage Path screen allows a second redirection (if needed) when a **`route-to`** command with cov = y vector step processes a previously covered call routed to a VDN. The default is `n` which retains the single redirection operation. If you set the field to `y` and the call has covered only once, the call redirects again as though the call was placed directly to the VDN. For example, the following second redirection can occur:
  1. the destination has a coverage path or another redirection feature.
  2. the calling user presses the **go-to-cover** button while the destination rings the principal phone on the second coverage path. The call stops ringing at the second principal phone and moves to the next coverage path.
  3. the second principal phone's **Hunt-to Station** path is checked for possible hunt before coverage or hunt after coverage redirection.
  4. send-all-calls at the destination station that go to coverage applies.

VICP interactions

Interaction	Description
AAR/ARS Partitioning	The Class of Restriction (COR) assigned to the VDN determines the Partition Group Number (PGN). The PGN in turn determines the AAR/ARS routing tables used by **`route-to`** commands.

Table continues…

Interaction	Description
ASAI	For direct calls to a VDN, the **adjunct routing** command operates like the **route to digits** command with coverage set to y. For calls that cover to a VDN, the **adjunct routing** command operates like the **route to digits** command with coverage set to n.
Attendant	Vectors connect calls, covering to a VDN, to an attendant queue or a hunt group. Internal calls that route to an attendant display the COR of the originating station if the attendant presses the **display COR** button. An attendant cannot establish a conference with a call covering to a VDN if the call is in vector processing. If a call placed to a local destination has covered to a VDN and the attendant attempts to add this call to a conference, conferencing is denied until the call has completed vector processing. If the attendant extends a call to a local destination that covers the call to a VDN, the attendant's return call timer cancels when vector processing begins and the **Return Call** button does not affect the call. If a call covers to a VDN and is then routed to an attendant, the attendant can transfer the call to another VDN.
AUDIX	Calls that cover to a VDN can be routed to an AUDIX by the **route-to** or **messaging** vector commands. Calls that cover to a VDN can be subsequently transferred to the AUDIX. Calls can also be transferred out of an AUDIX to a VDN.
Automatic Call Distribution (ACD)	A VDN can be the last point in an agent's coverage path for direct agent calls.
Call Coverage	A VDN cannot be a member of a coverage answer group. A vector cannot route a covered call to a coverage answer group. Coverage Callback and Leave Word Calling work normally when a vector delivers a call to a covering user.
Call Forwarding	Calls that have covered to a VDN cannot be redirected by Call Forwarding unless the **Cvg Enabled for VDN Route-To Party** field applies.
Call Park	A parked call does not cover to a VDN. When a call is parked at an extension with a VDN in its coverage path, the call continues to ring the extension. If the call parks to a hunt group extension and the call is in queue, the call remains in the queue until it is retrieved, answered by an agent, or abandoned by the caller. A vector event is generated for the calls when the administered coverage criteria are met. Once a call covers to a VDN, Call Park cannot be established until the call is delivered to an extension and vector processing ends.

Table continues…

Interaction	Description
Call Vectoring	The COR assigned to a VDN determines the PGN. The PGN in turn determines the AAR/ARS routing tables used by **route-to** commands. When a call covers to a VDN, VDN Override has no effect on the display shown on an answering display. This station shows the normal display for a covered call. Adjunct Routing: For direct calls to a VDN, the **adjunct routing** command operates like the **route to digits** command with coverage set to y. For calls that cover to a VDN, the **adjunct routing** command operates the same as a **route to digits** command with coverage set to n. Converse: Covered calls to a VDN work with the **converse** command. If a call in vector processing is connected to an agent in a converse split, the agent cannot activate Consult, Coverage Callback, or Coverage Leave Word Calling. Messaging: The **messaging** command handles covered calls differently depending on whether an extension is specified in the command. If the command **messaging split** *xxxx* extension none is used, the mailbox of the principal extension is used for the call. The number of the principal extension and the reason for redirection are passed to the messaging adjunct in the CONNECT message. When an extension is specified in the **messaging** command, no information on the principal extension is passed to the adjunct. Instead, the number of the extension specified in the command is passed to the adjunct in the CONNECT message along with the reason for redirection. The mailbox for the specified extension is used. Route-to: A call covering to a VDN can be routed to any valid destination by the **route-to** command. When the **route-to** command terminates a covered call locally, information identifying the principal and the reason for redirection are retained with the call. This information can be displayed on display phones or passed to an AUDIX or a Message Center system.
Class of Restriction	The COR assigned to the covering VDN governs the vector routing of the call.
Conference	Calls in an established conference do not cover to a VDN. Once a call covers to a VDN, a conference cannot be established until the call is delivered to an extension and vector processing ends.
Consult	The feature normally uses a Temporary Bridged Appearance on the principal's set. Call coverage to a VDN removes the Temporary Bridged Appearance from the principal's set, but the Consult feature still works.

Table continues…

Interaction	Description
Hunt Groups	A VDN can be the last point in a hunt group's coverage path. If the coverage vector for a split or hunt group routes calls to another using a **route-to** or **messaging** command, calls queue at the second resource with the queue priority assigned for the first split or hunt group. If a **queue-to**, **check**, or **converse** command is used, calls queue at the second split or hunt group with the priority specified in the command. If an inflow threshold has been assigned to a hunt group, the group does not allow new calls to queue when the oldest call in queue has exceeded the threshold. Therefore, covered calls are not connected to a hunt group when the group's inflow threshold has been exceeded. The interaction can also occur when a **messaging split**, or **route-to** command routes a covered call to a split that is not vector-controlled.
Look-Ahead Interflow	For calls that have covered to a VDN, LAI works like a **route-to digits/number** vector command with coverage = n. Any Dialed Number Identification Service (DNIS) digits sent with the interflowed call indicates the VDN to which the call covered, not any VDN the call encountered before it went to coverage.
Night Service	Calls that have covered to a VDN cannot be redirected by Night Service.
Personal CO lines (PCOL)	A VDN can be assigned as the last point in a PCOL coverage path.
Phone Display	Calls covering to a VDN and then directed to an agent in a split/hunt group by a **queue-to**, **check**, **converse**, or **route-to** command display the following information to the agent. `a=EXT 3174   to   EXT 3077     b` In this example, station A calls station B. Station B is busy, and the call covers to a VDN.
Redirection on No Answer (RONA)	RONA applies to calls that cover to a VDN. If the vector associated with the VDN queues the call to a resource (for example, a split or agent) that uses RONA, the call can be requeued for the same resource. The call cannot be redirected, however, since it has already covered to the VDN unless RONA occurs as a result of a **route-to** command with cov=y vector step and the **Cvg Enabled for VDN Route-To Party** field applies.
Terminating Extension Groups	A VDN can be assigned as the last point in the coverage path for a Terminating Extension Group (TEG), since it has already covered to the VDN.
Transfer	Calls can be transferred to extensions that cover to a VDN. Users who receive a covered call can transfer the call to a VDN. If a transfer attempt goes to coverage and covers to a VDN, the user at the answering station can complete the transfer by pushing the Transfer button or by flashing the switchhook on an analog station. Calls that cover to a VDN can be subsequently transferred to AUDIX. Calls can also be transferred out of AUDIX to a VDN.

VDN Override

The following table depicts how the *active* VDN extension is replaced when a call is routed through a series of VDNs by the **route-to number** or **route-to digits** vector steps.

The *active* VDN extension is determined by enabling the **Allow VDN Override** field for one of the previous VDNs to which the call was routed using a **route-to** command based on the following rules:

- If the **Allow VDN Override** field in the previous VDN is set to y, the extension of the current VDN overrides the active VDN extension.
- If the **Allow VDN Override** field in the previous VDN is set to n, the current active VDN extension remains the same.

The following example describes the VDN Override control of the active VDN extension for all calls routed to multiple VDNs by vector processing. VDN 1 is always the initial active VDN for the call.

Settings assigned for the Allow VDN Override field on the VDN screen								
VDN 1	y	n	n	n	y	y	y	n
VDN 2	y	y	n	n	n	n	y	y
VDN 3	y	y	y	n	y	n	n	n

Active VDN after the call is routed to the next VDN in the sequence								
After call is routed to VDN2	VDN2	VDN1	VDN1	VDN1	VDN2	VDN2	VDN2	VDN1
After call is routed to VDN3	VDN3	VDN3	VDN1	VDN1	VDN2	VDN2	VDN3	VDN3

Note:

With Expert Agent Selection (EAS) enabled for the system, if the **Allow VDN Override** field is set to y for the original VDN, the VDN skills (defined on page 1 of the Vector Directory Number screen) of the new VDN are used for vector commands where the skill group can be administered as 1st, 2nd, or 3rd. If the **Allow VDN Override** field is set to n on the original VDN, the VDN skills of the original VDN are used for such vector commands.

For Best Service Routing (BSR), if the **Allow VDN Override** field is set to y for the original VDN, the settings for the BSR application and **Available Agent Strategy** field (defined on page 2 of the Vector Directory Number screen) of the new VDN are used for BSR-related vector processing. If the **Allow VDN Override** field is set to n for the original VDN, the settings for the BSR application and **Available Agent Strategy** field settings of the original VDN are used for BSR-related vector processing.

VDN parameters associated with the active VDN

With VDN Override, you can use the information on the subsequent VDN to which a call is routed to be used instead of the information on the previously active VDN. The new VDN then becomes the *active* VDN for the call. The information that is associated with the *active* VDN and with the call as the call progresses through the vector steps includes the following:

- VDN Name
- Tenant Number (TN)
- VDN of Origin Announcement (VOA) Extension
- VDN Skills (1st, 2nd, 3rd)
- VDN Return Destination
- VDN Timed After Call Work (ACW) Interval
- Best Service Routing (BSR) Application
- BSR Available Strategy
- BSR Tie Strategy
- Display VDN for Route-to Direct Agent Call (DAC)
- Trunk Adjunct Switch Application Interface (ASAI) Messenger
- BSR Local Treatment
- VDN Variables
- VDN Time Zone Offset
- Apply Ringback for Auto-Answer Calls

Application of VDN Override

You can use VDN Override in conjunction with a vector that prompts the caller for a service type. For example, a call is placed to an automobile dealer with several departments, including Sales and Parts. If the caller wants to talk to someone in Sales, the VDN named Main directs the call to the main vector which redirects the call to the Sales vector. If VDN Override is assigned to the Main VDN, the agent phone displays the Sales VDN name when the call is connected to the agent.

Note:

When you enable **Variables in Vectors**, the VDN Override settings change the VDN extension number that is assigned to a *vdn* type vector variable. VDN Override is based on the *active* VDN for the call.

VDN Override for ASAI messages

You can use VDN Override when:

- A Computer Telephony Integration (CTI) application has set up an ASAI VDN or station event-notification association.
- The CTI application requires the called number ASAI message information to be the *active* VDN extension associated with the incoming trunk or internal call rather than the called number digits contained in the ISDN SETUP message for the incoming call.

VDN Override is useful for a CTI application that is monitoring a call where the *active* VDN extension is changed by the following vector scenario:

1. An incoming call is routed to a VDN. The vector prompts the caller to enter more than one digit.
2. The call is then routed to a subsequent VDN by a **route-to number** or **route-to digits** vector step.

ASAI messages

The VDN Override for Adjunct-Switch Application Interface (ASAI) Messages feature affects the following *called number* ASAI message information:

- Call Offered
- Alerting
- Queued
- Connect
- Adjunct Route-Request

Important:

Communication Manager activates this feature for an incoming ISDN trunk call when Communication Manager routes the call to a VDN that has the **VDN Override for ASAI Messages** field on the Vector Directory Number screen set to `ISDN Trunk`. To use the VDN Override for ASAI Messages feature for ISDN trunk calls and internal calls, set the field to `all`.

When Communication Manager activates this feature for a call, this feature remains in effect for the call regardless of the **VDN Override for ASAI Messages** field setting for subsequent VDNs to which the call is routed.

The **VDN Override for ASAI Messages** field setting affects the called number information in the following manner:

- If you set the field to `no`, the called number information is the called VDN extension in the called number Information Element (IE) sent in the incoming ISDN SETUP message or the local call number and does not change after routing to the called VDN and to subsequent VDNs.

- If you set the field to `ISDN Trunk`, when an incoming ISDN trunk call is routed to this VDN, the called number information sent in the ASAI event and Adjunct Route-Request ASAI message is the *active* VDN extension that becomes associated with the call based on the VDN Override rules. This field option does not apply to internal calls.
- If you set the field to `all`, the *active* VDN is used for the called number for all types of calls to the VDN including internal calls and incoming ISDN trunk calls.

VDN Override interactions

- If you set **VDN Override for ASAI Messages** to `ISDN Trunk` or `all`, Communication Manager does not keep the field setting when an ACD agent or a station user answers an incoming call. The Communication Manager call-transfer or station call-conference features transfer or conference the call with another agent or station user.
- If Communication Manager routes an incoming Central Office (CO) call to a VDN that has the **VDN Override for ASAI Messages** field set to `ISDN Trunk` or `all`, the field setting has no effect on the called number information for the ASAI messages.
- For calls that are routed through VDNs and are answered by Call-pickup button, then VDN override flag does not get applied.

VDN Return Destination

VDN Return Destination (VRD) allows calls to be placed back in vector processing after all parties, except the originator, drop the call. The originator can optionally be an internal call or connection as well as an external trunk. Internal calls include any type of calls that can be placed to, covered to, or transferred to a VDN that is not treated as an incoming trunk call.

The **Return Destination** field on page 2 of the Vector Directory Number screen allows you to enter a VDN extension as a return destination. The VDN which has this field administered is called the VDN with this feature active. The return destination VDN (the one specified in the field) is referred to as the *Return Destination*. Return destination is applied on the VDN that is treated as active for a call when the call left vector processing.

Every call that is processed through a VDN with this feature active is routed back to the assigned VDN when all parties on the call, except the originator, drop. For this feature, the originator is the party that originated the call at the time the call entered the VDN with this feature active.

Note:

VRD does not apply to DCS and attendant handled calls.

The VDN to which the call is placed to (when the originator is the only remaining party) is determined by the return destination. The VDN can be the original VDN or a different one.

The VRD status comes from the active VDN for the call when it finally reaches the destination. Once the call has been through vector processing, the VRD status cannot be changed by subsequent vector processing. The VRD status is that it either does not have a VRD that applies

(not the right origin or the active VDN assignment is blank) or that it has a VRD (an assigned VDN) that applies. If an internal call is released prior to the caller dropping, the assigned VRD extension and **Call Origin** value on the active Vector Directory Number screen determines VRD eligibility. An internal call is eligible for VRD if there is a VRD destination assigned and the **Call Origin** field on the Vector Directory Number screen is set to `internal` or `both`. VRD internal calls include:

- Station or Voice Response Unit (VRU)/Interactive Voice Response (IVR)/Voice Portal (VP) adjunct to a VDN
- An agent call to a VDN
- Agent/station/line port transfer of a previously ineligible call to a VDN
- Coverage or forwarding of a previously ineligible call to a VDN
- ASAI 3rd Party Makecalls to a VDN
- ASAI Merge connections, including Single Step Conference (SSC) to a VDN

Note:

An attendant call to a VDN or a call handled by an attendant is not eligible for VRD. If the call with a VRD reaches the PBX attendant, the VRD status will be canceled.

VDN Override applies to internal calls during the initial vector processing (not after leaving vector processing) if the call is routed to another VDN via a **`route-to`** command or adjunct route in the same manner as external calls. In this case the call origin criteria is also overridden so if for example an internal call to VDN A has **call origin** set to `internal` and override set to `y`, routing to VDN B, which has origin set to `external`, will remove the VRD. If VDN B does not have a VRD defined, the VRD will be also removed from the call since it overrides the assignment to VDN A. This feature keeps the call active after the original call has terminated. One typical use of VRD is for returning the caller, after being handled by an agent, to a VDN that is assigned to vector that routes to a VRU that surveys the caller. Another example of an application using VRD is to give the caller an opportunity to signal the need for sequence dialing (by entering a #). There are two ways this can happen:

1. When the destination drops on its own (after having answered), the call goes to the return destination which has a **`collect digits`** vector step. This step tries to collect the # sign entered by the caller.
2. When the call is not answered, the caller enters the # to request sequence calling (the ASAI-Requested Digit Collection feature collects the # sign). This # is reported to the adjunct. The adjunct requests the third_party_drop (or third_party_end_call) for the destination, and at that point the call goes to the return destination.

The VRD and ASAI-Requested Digit Collection features can be used independently, with the following rules:

1. If there is no ASAI request to collect digits, but a return destination is provided: when all parties, except the originator, drop, the switch routes the call with only one party active (the caller) to the return destination. At this point, the call enters vector processing for the VDN specified by the return destination.

The caller keeps returning to the same return destination indefinitely until either the caller hangs up or a busy or disconnect vector step is executed. Once a call leaves vector processing for the first time, the return destination is not changed.

2. If a request is made to collect digits but there is no return destination provided, the switch collects and passes the digits to an adjunct, which takes an action. However, if the adjunct drops one party on the call, the switch will drop the other party as well and clear the call (it cannot retain a call with only one party, if there is no return destination for further processing).

VDN variables

Many vectors use the same basic call flow, but are unique because each vector requires a different set of announcements, route-to destinations, holiday tables, vector routing table indexes, and conditional limits. With VDN variables, you can create a generic call flow vector. Use VDN variables to administer unique items on the VDN screen. VDN variables drastically reduce the number of vectors, ensure common flows and ease of administration during crisis times when you have to change the flows due to unforeseen events. Unforeseen events include problems with trunking, staffing, or messaging.

VDN variables enable VDNs to use a smaller set of vectors. You can:

- Assign up to nine variable fields, V1 through V9, on the VDN screen for each VDN.
- Use the VDN variables in all vector commands that support vector variables except as a for parameter with the **`collect-digits`** command.
- Use up to 16 digits to assign a number to the VDN variable and up to 15 characters to describe the VDN variable.
- Use VDN variables as indirect references to announcement extensions and other numerical values in vector commands.

Reasons to use VDN variables

You can create general-purpose vectors that support multiple applications with call-wait treatments that are tailored to the application. For example, you can create a single vector that can be used by multiple applications that are the same except for the announcement. Even when using only one vector, callers can still hear an announcement that is appropriate for their call. This can reduce the need for more vector capacity.

For more information about VDN variables, see the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Voice Response Integration

VRI

Voice Response Integration (VRI) integrates Call Vectoring with the capabilities of Voice Response Units (VRUs).

You can perform the following tasks:

- Pass data between the system and a VRU.
- Pool IVR ports for multiple applications.
- Run a VRU script while a call is in a queue, retaining the position of the call in the queue.
- Run a VRU script while retaining control of a call in vector processing.
- Tandem VRU data through a communication server to an ASAI host.
- Use a VRU as an external announcement device.
- Provide agent identifier in a system variable for vectoring use, when a customer call is redirected by the VDN Return Destination (VRD) feature into vector processing.

The **converse-on** command, which is part of Basic Call Vectoring, provides these capabilities. Use a converse-on step to integrate a VRU with ACD. With VRI, you can use the VRU capabilities while controlling a call in ACD.

Include VRUs with vector processing to take advantage of the following:

- Audiotex applications
- Caller information
- Increased recorded announcement capacity
- Interactive Voice Response (IVR) applications
- Local and host databases
- Speech recognition
- Text-to-speech capabilities
- Transaction processing applications

With VRI, contact centers can make productive use of call queuing time. For example, while a call is in a queue, the caller can listen to product information or can complete an IVR transaction. You can resolve caller queries, which reduces the time in a queue for other callers.

VRI detailed description

A call queued to a split or skill retains position in the queue while a VRU script is being run. When an agent becomes available, the line to the VRU is dropped and the caller connects to the agent.

The **converse-on** command delivers a call to a predetermined converse split or skill. A converse split or skill is administered and operates exactly like other splits or skills. Non converse splits or skills are accessed by the **queue-to** and **check** vector steps.

Members of a converse split or skill are the ports connected to the VRU. If all VRU ports are busy, a call queues to the converse split or skill with the administered priority. After the VRU answers the call, the **converse-on** command passes up to 2 data items to the VRU, depending on command parameters specified. You can pass data required by a VRU script or data that selects the VRU script to be run.

Whether or not you pass data, a caller is connected to the VRU, which runs the VRU script. Audible feedback provided by the vector is not heard and no further vector steps are run until the VRU script completes. The VRU returns data to the system and then drops the line to the system. Vector processing continues at the step following the **converse-on** command.

If the call was queued to a non converse split or skill before the **converse-on** command was run, the call retains the queue position. If an agent becomes available while the VRU script runs, the system drops the line to the VRU and connects the caller to the agent. The VRU detects the disconnect and terminates the VRU script.

With Call Prompting, you can collect and use digits that the VRU returns. The digits are handled as dial-ahead digits. Rules for collecting and processing VRU digits are the same as for Call Prompting.

You can use digits returned from the VRU in the following ways:

- To display for the answering agent, automatically for 2-line displays or with the callr-info button for other displays.
- As an extension in a **route-to digits** vector step. For example:

  ```
  converse-on split. . . . (VRU returns 4 digits)
  collect 4 digits after announcement none
  route-to digits coverage y
  ```

- For vector-conditional branching in an **if digits equals** vector step. For example:

  ```
  converse-on split . . . (VRU returns 1 digit)
  collect 1 digit after announcement none
  goto vector 101 if digits = 1
  goto vector 102 if digits = 2
  goto vector 103 if unconditionally
  ```

- Tandem to an ASAI host - collected digits are passed to ASAI hosts in call offered to domain event reports and in route request messages, therefore, caller digits or database information returned from the VRU can be sent tandem through the system to ASAI hosts. For example:

  ```
  converse-on split ... (VRU returns 9 digits)
  collect 9 digits after announcement none
  ```

```
adjunct route link Y
```

In this vector, the digits returned from the VRU are forwarded to the ASAI host in the adjunct routing route request message.

When you use a VRU application that returns data for a **collect-digits** step, the opportunity for toll fraud exists when the VRU application does not return any data. Following are some guidelines:

- If the collected digits are used to route calls internally, ensure that the COR for the VDN does not allow calls to route externally.
- If the collected digits are used to route calls externally, use a password to verify that the collected digits have been passed by the VRU application. For example, in the following vector, the VRU application returns a 3-digit password followed by the 8-digit external number. The vector routes calls without the correct password to a vector 23.

```
converse-on split 10 pri m passing none and none (VRU returns 11 digits)
collect 3 digits after announcement none
goto vector 23 if digits <> 234
collect 8 digits after announcement none
route-to digits with coverage n
```

The agent identifier is available in a system variable for vectoring use when a customer call is redirected by the VRD feature into vector processing. The agent identifier can be included in User-to-User Information (UUI) using existing vector commands before routing the call to an external IVR system. By adding the agent identifier in UUI, post call surveys can furnish reports details down to the agent level.

The *agent* type variable is read-only and cannot be set. The *agent* variable can be assigned only in the following conditions:

- By setting an *asaiuui* variable in a VRD vector
- As a **converse-on** operand [operand1 or *operand2*] in a VRD vector.

The following vector example illustrates using the *agent* variable, as a parameter in the **converse-on** vector command in a VRD vector.

```
01 converse-on skill 801 pri l passing LA and none    [LA - 'agent' variable]
```

The following vector example illustrates how to set an *ASAIUUI* variable to the last agent login ID in a VRD vector. By adding the agent identifier in UUI, post call surveys can furnish reports details down to the agent level. You can set an *asaiuui* variable to the last agent login ID by adding the *asaiuui* variable to the left of the = operand and the *agent* variable to the right of the = operand, with the SEL operator and the length specified as operand2.

```
01 set CH = LA  SEL  7     [CH - 'asaiuui' variable, LA - 'agent' variable]
```

VRI interactions

Converse splits interact like other vector-controlled splits unless noted here.

Interaction	Description
Adjunct Switch Application Interface (ASAI)	When a **converse-on** vector step places a call to an ASAI-monitored domain, ASAI event messages are sent over the ASAI link. When a **converse-on** step places an ASAI-monitored call, the ALERT message sent to the ASAI adjunct includes a cause IE, Coding Standard 3 value 23 (CS3/23), which informs the adjunct that the call has not been dequeued from any non converse splits. If a **converse-on** step is run while an adjunct routing request is outstanding, the request is canceled. ASAI cannot transfer or conference calls but can direct the system to perform the action.
Agents	You can use a **converse-on** step to deliver a call to a group of human agents. To agents, the call looks like an ACD call, except that agents cannot use certain features such as Transfer, Conference, and Supervisor Assist. The agent can return data to vector processing by pushing the **transfer** button or flash hook on analog and dialing the converse-on data return code and required digits.
Answer supervision	Answer supervision is returned only once during a call. If a call is answered because of a converse-on step, answer supervision is sent if not previously sent. If digits are passed to the Voice Response Unit (VRU), answer supervision is sent after the digits are sent.
AUDIX	If a **converse-on** step calls AUDIX, the call is handled as a direct call to AUDIX. The caller hears the AUDIX welcome message and can retrieve messages as usual. If a call is forwarded to a VDN and then delivered to an AUDIX hunt group by a **converse-on** step, the call to AUDIX is treated as a redirected call, and the caller can leave a message.
Auto-Available Split/Skill (AAS)	A **converse-on** vector step can place a call to an AAS. Use auto-available converse splits or skills for VRI except when ASAI controls the converse split or skill.
Automatic answering	When you administer ports on the Interactive Voice Response (IVR) system as agents of a converse split or skill, do not administer agents as automatic answer. The system-provided zip tone can interfere with the interaction between the IVR system and the calling party.

Table continues…

Interaction	Description
BCMS or CMS	BCMS tracks calls that a **converse-on** step places to a BCMS-measured hunt group. CMS tracks calls that a **converse-on** step places to a CMS-measured hunt group, split, or skill. The VDN tracks such calls as waiting in the vector. A call is treated as answered when answered by a non converse split or skill agent, not when answered by a converse split or skill agent. The converse split or skill tracks this as a separate answered call when the VRU answers. Though trunk and split or skill totals no longer match, VDN and trunk totals match.
Call Detail Recording (CDR)	The duration of a call to a VDN is recorded from when answer supervision is returned after a successful **converse-on** step. Unsuccessful **converse-on** steps do not generate ineffective call-attempt records. **Converse-on** steps cannot place calls. The steps simply direct a call to a hunt group.
Call Park	Calls that a **converse-on** step placed cannot be parked.
Call Pickup	Do not use Call Pickup with **converse-on** steps.
Class of Restriction (COR)	The system does not check CORs when a **converse-on** vector step routes a call to a split.
Conference	You cannot conference a call routed by a **converse-on** step.
Direct Department Calling (DDC)	You can administer a converse split or skill as a DDC split or skill.
Distributed Communications System (DCS)	If an incoming DCS call is placed to a vector with a **converse-on** step, the caller DCS extension is sent to the VRU.
Expert Agent Selection (EAS)	**Converse-on** steps can place calls to a skill hunt group.
Hold	An agent answering a converse call can put the call on hold, but the caller does not hear music on hold. If a call is queued to a backup split or skill before it was sent to the VRU and a non converse split or skill agent answers the call on hold, the agent who placed the call on hold is dropped, and the caller connects to the answering agent.
Hold - Automatic	Automatic hold applies to converse-on calls.

Hunt Groups

A **converse-on** step can deliver a call to a vector-controlled or AUDIX hunt group, ACD split, agent skill, or message center.

Interaction	Description
Integrated Services Digital Network (ISDN)	You can administer a **converse-on** step to send a Calling Party Number (CPN) / Billing Number (BN) to the IVR system using the caller keyword.
Intra-switch CDR	If a converse-on call is answered and either the caller or the VDN associated with the call is administered for intra-switch recording, timing for the call is started and the CDR record shows *calling party to VDN* as the originating and answering parties.

Table continues…

Interaction	Description
Line-side T1 connectivity	T1 connectivity between Communication Manager and the IVR system is supported for VRI. The DS1 board must be a TN767E or later, or TN464F or later. Administer all converse agents as DS1FD-type stations. Operation of the converse step using Line-side T1 is identical to that over a tip/ring line. In particular, delay-timing and outpulsing speed is the same as for analog lines. T1 connectivity to the IVR system is supported only in the United States and Canada.
Look-Ahead Interflow (LAI)	If an incoming call or a call routed by a converse-on vector step is answered by a VRU, or is queued to the converse split or skill while an LAI call attempt is outstanding, the attempt is accepted.
Message Center	**converse-on** steps can deliver calls to message hunt groups. Such calls are handled as direct calls to the message hunt group. If a call is forwarded to a VDN and a **converse-on** step delivers the call to a message split, the call is handled as a redirected call. A **converse-on** step can queue a call to three different skills and then to a converse skill group or split.
Music-on-Hold (MOH)	During the data return phase of a **converse-on** step, the caller is placed on hold, but does not hear music.
Non vector-controlled splits	A **converse-on** step cannot route a call to a non vector-controlled split.
Queuing	Converse-on calls queue when delivered to busy hunt groups. Call Vectoring audible feedback is not disconnected while a converse-on call is queued. If a **converse-on** step is run while a call is queued to a non-converse split or skill, the call remains in queue, even after being answered by the VRU. **Converse-on** steps can queue calls at one of four priority levels: low, medium, high or top. You administer the queue priority of a call on the **converse-on** step.
R2-MFC signaling	R2-MFC signaling trunks can send Automatic Number Identification (ANI) to VRUs using the *ani* data item on the **converse-on** step.

Table continues…

Interaction	Description
Recorded announcement	Use VRI to increase the system recorded announcement capacity by offloading some recorded announcements to a VRU. Using the **converse-on** step, redirect callers to a group of VRU ports by passing the number of the announcement to be played. The IVR system can play any announcement on any port. Although only one caller can be connected to each port, up to 48 callers can be connected simultaneously to the IVR system. The maximum number of callers that can be connected to a VRU simultaneously varies with each VRU.
Redirection on No Answer (RONA)	If a **converse-on** step calls a hunt group with *no answer time out* administered and the call rings an agent or port for longer than the time out interval, the call redirects and the agent or port is put in the AUX work mode or logged out if the agent is an AAS member. With RONA, the call is requeued to the split or skill. The call cannot requeue to the split or skill if it is an AAS with all agents logged out or if the queue is full. If the call cannot be requeued, the **converse-on** step fails, a vector event is logged, and processing restarts at the next vector step.
Service Observing	Calls delivered by a **converse-on** step can be observed. To prevent the observer from hearing tones associated with data being sent to the VRU, the observer is not connected to the call until after data is passed. If the VRU returns data, the observer is put in service-observing-pending mode and the caller is put on hold while the data is sent. When the converse-on session ends and the VRU drops the line, the observer remains in service-observing-pending mode and waits for the next call. In addition, the observer observing a VDN does not hear data being sent. After data is sent, the observer rejoins the call. Do not administer a service observing warning tone because the warning tone can interfere with the interaction between the IVR system and the caller.
System measurements	System measurements track converse-on calls to hunt groups.
Touch-Tone Dialing	A caller can use touch-tone dialing while digits are passed in a converse-on session. The data is not corrupted. The system does not collect the dialed numbers as dial-ahead digits. After the system sends digits to the IVR system, a caller can enter touch-tone digits at the IVR prompt. After the IVR system has returned data to the system and an additional collect <#> digits vector step is run, a caller can enter a touch-tone response to a system prompt.

Table continues…

Interaction	Description
Transfer	A call delivered by a **converse-on** step cannot be transferred. If an attempt to transfer a converse-on call is made, a vector event is logged, the line to the IVR system is dropped, and processing restarts at the next vector step. If a human agent tries to transfer a call, the transfer fails and the agent reconnects to the call.
Transfer out of AUDIX	If a **converse-on** step delivers a call to an AUDIX hunt group and the caller tries to transfer out of AUDIX, the transfer fails and processing continues at the next vector step.
Uniform Call Distribution (UCD)	You can administer a converse split or skill as a UCD split or skill.
VDN Display Override	If a call that accesses multiple VDNs encounters a **converse-on** step that passes VDN, normal display override rules determine which VDN number is sent to the VRU.
Vector-controlled splits or sSkills	**Converse-on** steps can deliver calls only to skills or vector-controlled splits.

Call Vectoring considerations

Displaying VDN names for vector-initiated DACs

The Display VDN for Route-to DAC feature improves the efficiency of call center agents who answer vector-initiated direct agent calls that originate from multiple Vector Directory Numbers (VDNs).

The type of information displayed at the agent station display with a vector-initiated direct agent call can be summarized as follows:

- When the Display VDN for Route-to DAC feature is not enabled, only the EAS LoginID name for the agent who receives the call is shown.
- When the Display VDN for Route-to DAC feature is enabled for such calls, the active VDN name associated with the call is shown.

Providing agents with the ability to see the VDN name associated with an incoming call improves agent efficiency and customer satisfaction. For example, if an agent receives incoming trunk calls for different products from three different VDNs, the VDN name displayed by the Display VDN for Route-to DAC feature allows the agent to answer the call as a sales representative of that product. This feature is especially useful when vector-initiated direct agent calls route incoming trunk callers to personalized agent providing services for new customers, special product offers, or premier levels of service.

Display VDN for Route-to DAC feature operation

The Display VDN for Route-to DAC feature is designed for call scenarios where a VDN-initiated call is routed to a vector where direct agent calls are originated by one of the following methods:

- A route-to number vector step with cov parameter set to y, where the **number** field is administered with a valid EAS loginID extension.
- A route-to digits vector step with coverage parameter set to `y`, where a **`collect digits`** vector step preceding this step is used to allow the caller to enter the digits for an EAS LoginID extension.
- An adjunct routing link vector step, where a direct agent call is originated by the Route Select digit information returned from a CTI application.

The Display VDN for Route-to DAC feature is activated for an incoming trunk call when the call is routed through a VDN that has the **Display VDN for DAC Calls?** field administered to `y`. When one of the above-listed vector steps routes such an incoming call as a direct agent call to an EAS loginID extension, the active VDN name is shown on the called agent station display instead of the called EAS agent's LoginID name. If this call is routed to another EAS agent in the initially-called EAS agent coverage path, the active VDN name will again be shown on the covered-to agent station display, instead of the initially-called EAS agent LoginID name.

Station display formats

If the Display VDN for Route-to DAC feature is activated for an incoming trunk call routed through a VDN to a vector that initiates a direct agent call to an EAS agent, the format of the called agent station display appears as one of the following:

```
<Incoming Trunk Name> to <VDN Name>
```

```
<Incoming caller ANI> to <VDN Name>
```

If the Display VDN for Route-To DAC feature is not activated for an incoming trunk call, the called agent station display appears as one of the following:

```
<Incoming Trunk Name> to <EAS loginID extension>
```

```
<Incoming caller ANI> to <EAS loginID extension>
```

Note:

If the EAS agent to which the call is routed by vector-initiated Direct Agent Calling (DAC) is not available, and the called EAS agent has a coverage path to other EAS agents, the Display VDN for Route-to DAC feature preserves the active VDN name and sends it to the agent station display for a covered-to EAS agent. If the call covers to a normal station extension in the called EAS agent coverage path, the Display VDN for Route-to DAC feature does not apply to the covered-to station display, and the EAS LoginID of the called EAS agent is displayed instead.

Methods for creating vectors that use the Display VDN for Route-to DAC feature

You can administer a vector in several different ways to utilize the Display VDN for Route-to DAC feature.

Note:

For any of the vector examples shown below, if an incoming trunk call is routed through a **VDN with the Display VDN for Route-to DAC?** field set to y, the direct agent call is activated with the VDN Display for Route-to DAC feature.

Using collect digits and route-to digits commands

The following vector example shows how to:

- Use a collect digits vector step to prompt a caller to enter digits for a valid EAS agent loginID extension
- Use a route-to digits vector step to route the call to an agent as a direct agent call:

```
wait-time 0 secs hearing ringback
collect 5 digits after announcement 3001
go to step 5 if digits < > 1????
route-to digits with coverage y
announcement 3002
goto step 2
```

Using route-to number commands

The following simple vector uses the route-to number vector step to originate a direct agent call to an EAS LoginID extension:

```
wait-time 0 secs hearing ringback
route-to number 85103 with cov y
```

Using adjunct routing link commands

You can also originate a direct agent call with a vector that includes an adjunct route vector step. When an incoming trunk call is routed through a VDN to a vector that includes an adjunct route vector step, vector processing treats this step like a **route-to number with cov** set to y vector step.

The following vector uses the adjunct route vector step to originate a direct agent call. In this example, the CTI application is designed to route the call as a direct agent call in a Route Select ASAI message.

```
1. wait 0 secs hearing ringback
2. adjunct route link 3
3. wait 30 secs hearing ringback
4. announcement 3501
5. disconnect
```

Interactions with other Communication Manager features

Interactions of the Display VDN for Route-to DAC feature with other Communication Manager features include the following:

Interaction	Description
Call Coverage	When the Display VDN for Route-to DAC feature is activated for a call, and a vector-initiated direct agent call is made to an EAS agent having a coverage path that has other agents as coverage points, the active VDN name associated with the call is displayed on a covered-to agent's station display instead of the originally-called EAS agent's LoginID extension.
Call Forwarding	Display VDN for Route-to DAC has no impact on the Call Forwarding feature.
Station Conference/ Transfer	When an EAS agent transfers or conferences a vector-initiated direct agent call that has the Display VDN for Route-to DAC feature activated to another agent or station user, the station display of the answering agent or station does not show the active VDN name that was previously displayed for the call. This is consistent with the existing station display treatment for transferred or conferenced calls that have a VDN name shown as the "to" party for a call.
VDN Override	Active VDN name station display rules for the VDN Override feature are applied to the Display VDN for Route-to DAC feature. For example, if an incoming trunk call is routed through a VDN where the VDN Override feature is enabled, and the call is routed to a second VDN by a route-to number vector step where the **Display VDN for Route-To DAC?** option is set to y, the station display for an EAS agent that receives a subsequent vector-initiated direct agent call shows the second VDN's name for the call instead of the called EAS agent's LoginID extension.
Redirect on No Answer (RONA)	The Display VDN for Route-to DAC feature is activated only for vector-initiated direct agent call to an EAS LoginID extension. When the RONA timer expires after the call is not answered, one of the following results occurs: • If subsequent vector processing again routes the call to an EAS LoginID extension by means of the Direct Agent Calling (DAC) feature, and the Display VDN for Route-to DAC feature is enabled, the active VDN name is shown on the covered-to agent station display. • If subsequent vector processing again routes the call to an EAS LoginID extension by means of the DAC feature, and the Display VDN for Route-to DAC feature is not enabled, then the EAS LoginID for the covered-to agent is shown on their station display.
Messaging systems for EAS Agents	The Display VDN for Route-To DAC feature has no interaction with messaging systems for a vector-initiated direct agent call that is routed to an EAS agent and subsequently covers to the agent's messaging-system mailbox.
Adjunct Routing	If a call is routed through a VDN having the Display VDN for Route-to DAC feature set to y, and an adjunct route vector step is executed that results in a direct agent call to an EAS agent, the active VDN name is displayed on the routed-to agent's station display instead of the called EAS agent's LoginID.

Call transfer to VDNs

Care needs to be taken when writing a vector to which callers will be transferred. This is especially true if the vector manipulates or tests data that is delivered with the incoming call, such as ANI, II-digits, or CINFO digits.

To understand why care is needed, it is necessary to understand how a transferred call is treated. There are three main steps in a call transfer.

1. The transferring party hits the transfer button. The caller is put on hold. A second call is created with the transferring party as the originator.
2. The transferring party dials the VDN extension. Vector processing starts. The transferring party, not the caller, hears the initial vector provided feedback, if any.
3. The transferring party hits the transfer button for the second time. The two calls merge. The transferring party is dropped from the call. The caller becomes the originator of the new call. The caller now begins to receive vector provided feedback.

Between transfer steps 2 and 3 there is always a small but finite amount of time during which it is the transferring party who is connected to the vector. Any testing of ANI, II-digits, or CINFO digits during this time window applies to the transferring party and not to the caller. For this reason, it is recommended that vectors not start with an ANI, II-digit, or collect cdpd/ced step. Insert a delay of sufficient length to allow the transferring party to complete the transfer.

A delay is not required before a collect x digits after announcement step because a collect announcement is restarted for the caller when the transfer is complete.

Call Vectoring interactions

Interaction	Description
AP Demand Print	A VDN cannot be used as an argument to the FAC for AP Demand Print.
Attendant Control of Trunk Group Access	If a **route-to** step in a vector dials a controlled trunk group, vector processing continues at the next step.
Attendant Recall	Attendant Recall to a VDN is blocked.
AUDIX Interface	A **route-to** step in a vector call the AUDIX extension. If a voice port can be seized to that adjunct, vector processing is terminated. The system sends a message to AUDIX requesting retrieval of messages for the originating extension (not the VDN). AUDIX also be accessed by the **queue-to split** and **check split** commands. Also, the messaging step use an AUDIX hunt group in its operation.
Authorization Codes	If authorization codes are enabled, and if a **route-to** command in a prompting vector accesses Automatic Alternate Routing or Automatic Route Selection and the VDN's Facility Restriction Level (FRL) does not have the permission to utilize the chosen routing preference, then no authorization code is prompted for and the **route-to** command fails.
Automatic Alternate Routing (AAR)/Automatic Route Selection (ARS)	Any **route-to** command in a vector can dial an AAR/ARS FAC followed by other digits. It cannot dial only the Facility Access Code.

Table continues…

Interaction	Description
Automatic Callback	Automatic Callback cannot be used for calls placed to a VDN.
Bridged Call Appearance	VDN extensions cannot be assigned to bridged appearance buttons. A **route-to** command to an extension with bridged appearances updates bridged appearance button lamps.
Busy Verification - Terminals, Trunks	Busy verification of VDNs is denied and intercept tone is returned.
Call Coverage	A VDN be administered as the last point in a coverage path.
Call Forwarding	Calls can be forwarded to a VDN. Calls placed by a **route-to** command to an extension that has call forwarding activated are forwarded. An attendant or phone with console permission cannot activate/deactivate call forwarding for a VDN. An attendant or phone with console permission cannot activate/deactivate call forwarding for a vector-controlled hunt group.
Call Detail Recording (CDR)	You can administer the Feature-Related System Parameters screen so that the VDN extension is used in place of the Hunt Group or Agent extension. This overrides the *Call to Hunt Group - Record* option of Call Detail Recording (CDR) for Call Vectoring calls. If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply. For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned. If answer supervision is returned by the vector, and the call never goes to another extension, then the VDN extension is recorded as the called number in the CDR record. If the call terminates to a hunt group, then the VDN, hunt group, or agent extension is recorded as the called number as per the administration described above. If the call terminates to a trunk, then the following two CDR records are generated: • An incoming record with the VDN as the called number and the duration from the time answer supervision was provided to the incoming trunk. • An outgoing record containing the incoming trunk information as the calling number and the dialed digits and the outgoing trunk information as the called number. Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number. No Ineffective Call Attempt records are generated for Call Vectoring **route-to** commands that are unsuccessful.
Call Detail Recording - Account Code Dialing	If a **route-to number** command in a vector specifies an CDR account code, vector processing continues at the next step.

Table continues…

Interaction	Description
Call Park	Calls cannot be parked on a VDN.
Call Waiting Termination	If an extension is busy and has call waiting termination administered, the **route-to with cov n** operation is treated as unsuccessful and vector processing continues at the next step. **Route-to with cov y** is successful (call will wait) and vector processing terminates.
Class of Restriction (COR)	Each VDN in the system has a Class of Restriction (COR) associated with it. This VDN COR is used to determine the calling permissions/restrictions, the AAR/ARS PGN, and the priority queuing associated with a vector.
Code Calling Access	A VDN cannot be used as the argument to the code calling access feature access code. If a **route-to number** command in a vector specifies the code calling feature access code, vector processing continues at the next step.
Conference	A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call.
Data Restriction	Music will play on calls from data restricted extensions when the call receives music as the result of a wait-time vector step.
Facilities Restriction Level	If a **route-to** command dials an external number using AAR/ARS, the FRL associated with the VDN COR is used to determine the accessibility of a routing preference in an AAR/ARS pattern.
Facility Busy Indication	The facility busy lamp indication for a VDN is always off. A facility busy button can be used to call a VDN.
Facility Test Calls	If a **route-to number** command in a vector specifies a Facility Test Call, vector processing continues at the next step.
Forced Entry of Account Codes	If a COR requiring entry of account codes is assigned to a VDN, the **route-to number** commands executed by the associated vector are unsuccessful and vector processing continues at the next step.
Individual Attendant Access	A call sent to an attendant by a **route-to number** command can wait in the attendant priority queue. The call is removed from vector processing.
Integrated Directory	VDN names and extensions are not available in the Integrated Directory feature.
Intercept Treatment	A VDN cannot be used for Intercept Treatment.
Inter-PBX Attendant Calls	A **route-to number** command in a vector can dial the Inter-PBX Attendant. If the call attempts to access a controlled trunk group, vector processing continues at the next step.
Intraflow and Interflow	The functionality of intraflow and interflow can be obtained using the **check** and **goto** Call Vectoring commands. Calls can intraflow from an ACD split or skill that is not vector-controlled into one that is vector-controlled.
Leave Word Calling (LWC)	Leave Word Calling (LWC) messages cannot be stored, canceled, or retrieved for a VDN.

Table continues…

Interaction	Description
Night Service	A VDN can be administered as a night service destination. **Route-to** commands that route to destinations with night service activated redirect to the night service destinations.
Priority Calling	A VDN cannot be used with the priority calling access code. Intercept tone is supplied to the user. If a **route-to number** in a vector specifies the priority calling access code, vector processing continues at the next step.
Property Management System Interface	VDNs cannot be used with the following features and functions: Message Waiting Notification, Check-In, Check-Out, Room Status, and Automatic Wakeup.
Recorded Announcement	The first announcement extension, second announcement extension, first announcement delay, second announcement delay, and recurring second announcement do not exist for a vector-controlled hunt group.
Redirection on No Answer	If an ACD split or skill or direct agent call is not answered after an administered number of rings, RONA can redirect that call to a VDN for alternate treatment.
Ringback Queuing	External call attempts made using **route-to** commands with coverage no are not queued using Ringback Queuing when all trunks are busy. External call attempts made using **route-to** commands with coverage yes are.

Table continues…

Interaction	Description
Route Calls to Agent by skill level	Using the skill level preference parameters, you can request a preference to route calls to an available (idle) agent who has a specified skill level or is in a skill level range. This option is specified by using the skill level preference parameters on the **check** Vectoring command. In an agent surplus condition (available-agents > 0), you can request that the call is routed to an available agent who has a specific skill level or a skill level within a specified range. If you select pref-level, the system displays only the Skill Level1 field, where you can enter a skill value of 1-16. If you select pref-range, the system displays two fields, Skill Level1 and Skill Level2. Using these two fields, you can enter a range of preference levels, such as 5 (Skill Level1) to 13 (Skill Level2). The values in both these fields need to be between 1 and 16. The number in Skill Level2 field needs to be equal to or greater than the number you enter in the Skill Level1 field. • `Skill`: Skill level preference options are only available for the check skill version of the command, not for check split or check best. • `if available-agents > 0 or greater`: This option ensures that the skill level preference is applied only in agent surplus conditions. • `all-levels`: The system ignores the skill level of the agent. This is the default value. • `pref-level`: The system displays the Skill Level1 field in which you can enter a skill level value for the agent from 1 to 16. Preference level of 1 is the best while 16 is the least. • `pref-range`: The system displays two fields, Skill Level1 and Skill Level2. Using these two fields, you can enter a range of preference levels, such as 5 (Skill Level1) to 13 (Skill Level2). The values in both these fields need to be between 1 and 16. The number in Skill Level2 field needs to be equal to or greater than the number you enter in the Skill Level1 field. Following is a sample vector command: `check skill 5 pri h if available-agents > 0 pref-range 1 to 3 queue-to skill 17 pri t` For example, in agent surplus conditions, this feature allows you to route high-value and critical calls to the best skilled agents for this skill and route the low valued calls to trainees or novice agents who have the assigned same skill. For more details on syntax and usage of check skill command and skill level preference parameters, see the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Table continues…

Interaction	Description
Send All Calls	If the destination of a **route-to with coverage no** command has the Send All Calls (SAC) feature active, calls are not redirected. If there is an idle appearance, the call terminates and vector processing stops. If not, vector processing continues at the next step. If the **send-calls** button is pressed after a vector call is terminated, button activation is denied.
Time of Day Routing	Since a **route-to number** command in a vector can specify the Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) access codes, the Time-of-Day (TOD) routing algorithm can be used to route the call.
Timed After Call Work (ACW) Associated field: **After Xfer or Held Call Drops**	A Timed After Call Work (TACW) interval can be assigned to a VDN. For incoming ACD or DAC calls, an Auto-In agent is placed into Timed ACW mode, instead of immediately making the agent available, if the caller drops a held call or the agent transfers the call. You can enable this feature for the agents in a hunt group or for calls delivered from a VDN when the Timed ACW Interval field is set to a non-0 value.
Timed Reminder	The attendant Timed Reminder is not available for calls placed, transferred, or extended to a VDN. Vectoring causes all other timers to be ignored.
Transfer	Calls can be transferred to a VDN.
Traveling Class Mark	A TCM is sent when a **route-to** command dials a seven-digit Electronic Tandem Network (ETN) or 10-digit Direct Distanced Dialed number using AAR/ARS. This TCM is the FRL associated with the VDN Class of Restriction (COR).
VDN in a Coverage Path (VICP)	A call covering to a VDN can be routed to any valid destination by the **route-to** command. You can enable the **Cvg Enabled for VDN Route-to Party** field on the Coverage Path screen to allow a second redirection when a **route-to** command with cov = y vector step processes a previously covered call routed to a VDN. The default is `n` which retains the single redirection operation. When the **route-to** command terminates a covered call locally, information about the principal and the reason for redirection is retained with the call. This information is displayed on the phone, or passed to an AUDIX or a Message Center system. The COR assigned to a VDN determines the PGN. The PGN in turn determines the AAR or ARS routing tables used by **route-to** commands. When a call covers to a VDN, VDN Override rules have no effect on the display shown on an answering display telephone. This station shows the normal display for a covered call.

Call Vectoring and BCMS/CMS interactions

Call Vectoring interacts with a management information system that helps monitor and report Call Vectoring activities. The management system is either an Avaya Call Management System (CMS) or an Avaya Basic Call Management System (BCMS).

CMS, which is installed on an adjunct processor, collects and processes ACD information to generate reports. BCMS, which is part of Communication Manager, collects ACD information and

generates a limited number of reports. CMS reporting and data storage capabilities are much more extensive than those of BCMS.

Call tracking

Tracking is the identification of call flows and other actions relevant to call handling. You can classify call flow into split flow, VDN flow, and vector flow.

BCMS and CMS track the following types of call flow:

- Abandons
- Answers
- Busies
- Dequeues
- Disconnects
- Inflows (flow ins)
- Outflows (flow outs)

A contact center supervisor can use VDN flow and vector flow to evaluate the effectiveness of vector programming at a site. The supervisor can use split flow to determine how splits in a site handle incoming calls.

Defining and interpreting call flows

The manner in which specific call flows are defined and interpreted depends upon the call flow class in question, the management system in effect, and the version of the switch being used. Management systems include CMS and BCMS.

The following sections define and interpret specific call flows according to these parameters.

Answered and abandons

The most important tracking items for most VDNs and vectors are the number of calls answered and the number of calls abandoned. The CMS provides VDN profiles that show when calls are answered and abandoned. Ten service level intervals are administered for these profiles. These intervals can have smaller time intervals around the time most calls are answered and when most calls abandon to get more detailed information.

This data can be used to determine what an acceptable service level is for most callers. The percentage answered within the administered acceptable service level is also shown on the Call Profile reports. For VDNs, the calculation is ACD calls answered and non ACD calls connected within the service level divided by calls offered to the VDN (including calls that inflow to the VDN).

For split/skill statistics, the calculation is ACD calls answered within the service level divided by calls queued to the split/skill (answered calls, abandoned calls, calls that flow out, calls that dequeue). In most cases the VDN percentage will be higher than the split percentage since calls dequeued from a split/skill are counted as answered, abandoned, or outflows for the VDN.

Changes made to a vector or to staffing will typically affect the VDN call profile. Even the wording of an announcement can affect the abandon profile. It is worthwhile to review the VDN’s call profile before and after any change to determine if the change had a positive impact.

Busies and disconnects

Busy calls and forced disconnects reported on the CMS indicate how many calls the VDN or vector turned away. If forced disconnect is used out of business hours, the item indicates how many customers expected the call center to be operating during a specific time interval. If busies are given when the queues are full or waiting times are long, the number of busies in an interval suggests a staffing change. If disconnect is used to deny an LAI attempt, a large number of denials indicate a busy time at multiple sites.

VDN inflows and outflows

CMS and BCMS standards for interpreting VDN inflows and outflows

Flow type	Management system	Description
VDN flow in	CMS	Calls that flow into the VDN using a **route-to VDN** command or by Redirection on No Answer to a VDN.
	BCMS	(Not tracked.)
VDN flow out	CMS	Calls that successfully flow out of a VDN to another VDN or to an external location using a **route-to** command.
	BCMS	Same as for CMS.

Vector inflows and outflows

CMS standards for vector inflows and outflows

Vector flow-in pertains to calls that flow into a vector from another vector using a route to or a goto vector command. Vector flow out pertains to calls that successfully flow out of a vector using a route to or a goto vector command.

CMS and BCMS standards for interpreting split flows

CMS and the BCMS are grouped together because both the systems interpret two split flow types identically. The flows include inflow and outflow. CMS interprets another split flow type, dequeue. BCMS does not interpret this split flow type because BCMS does not have a dequeue tracking item.

Primary split is defined as the first split in a VDN to which a call actually queues. Therefore, the split is not necessarily the first split referenced in the vector.

Another split becomes the primary split if either of the following events occurs:

- Call cannot queue to the originally-targeted split because the split has no queue slots available.
- Call leaves the VDN, for example, using a **route-to** VDN command, and is queued to another split as a result.

If the call leaves vector processing and does not queue to another split, for example, as a result of a **route-to extension** command, there is no new primary split.

The following table indicates how CMS and BCMS interpret split flows.

Flow type	Management system	Interpretation
Inflow	CMS	Calls that ring at an agent in a split other than the primary.
	BCMS	Same as for CMS.
Outflow	CMS	Calls that are dequeued from a primary split using a **route-to** or **messaging split** command, or by ringing at or being answered by an agent in another split to which the call is also queued.
	BCMS	Same as for CMS.
Dequeue	CMS	Calls that are dequeued from and not answered by any split other than the primary split in a VDN.
	BCMS	Not tracked.

When a call is not answered due to an outflow, abandon, busy, or disconnect, the disposition of the call is tracked for the primary split as long as the call is still queued when the call abandons, outflows, etc. However, if the call abandons or outflows from ringing, the disposition is recorded for the split. On the CMS, the other splits to which the call is queued track a dequeue when the call outflows, abandons, is given busy treatment, or is disconnected.

If the primary split in a VDN is unmeasured, an outflow, abandon, busy, or disconnect is not tracked for the call. Also, an answer is not tracked if the call is answered by an agent in the primary split.

Examples of split flow tracking

The following sections provide some examples of tracking in CMS and BCMS. Each section first presents a scenario of Call Vectoring events. The scenario is then followed by a table in which the tracking for the various splits involved is recorded. Following each tracking table, an explanation of the tracking procedure is provided.

The scenarios presented include the following:

- Call answered by a primary split.
- Call answered by a non primary split.
- Call abandoned from queue.
- Call answered by a primary split after a route to VDN.
- Call answered by a non primary split after a route to VDN.
- Call answered after a route to split.

Note:

Inflows, outflows, and dequeues are not tracked for splits administered by the **converse-on split** command. However, if a converse split and subsequently a non converse split answer the call, CMS or BCMS tracks an answer for each split. However, CMS or BCMS treats a call as answered only when a non converse split answers the call. Therefore, traffic measurements for converse splits must be used only to measure converse split traffic and not to calculate the total number of calls.

Call answered by a primary split

The following scenario involves a call answered by the primary split. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2 and 3.
2. Call is answered in split 1.

The following table shows the tracking table for this scenario.

Split tracking			
	1	2	3
CMS	answer	dequeue	dequeue
BCMS	answer		

Comments:

- CMS: Dequeue is tracked in split 2 and split 3 because the call is answered by the primary split, that is, split 1 and is, therefore, dequeued from splits 2 and 3 without being answered in the splits.
- BCMS: No dequeue tracking item is available.

Call answered by a non-primary split

The following scenario involves a call answered by a non primary split. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2 and 3.
2. Call is answered in split 2.

The following table shows the tracking table for call answered by non-primary split.

Split tracking			
	1	2	3
CMS	outflow	inflow answer	dequeue
BCMS	outflow	inflow answer	

Comments:

- CMS: Outflow is tracked in split 1 because the call is answered by an agent in another split to which the call is queued, that is, split 2. Although the call is removed from split 1 after the call is answered in split 2, dequeue is not tracked in split 1 because split 1 is the primary split. Inflow is tracked in split 2 because the call is answered in this split and the split is not the primary split. Dequeue is tracked in split 3 because the call is removed from the split without being answered there. When the call is removed from split 3, outflow is not tracked in split 3 because this split is not the primary split.
- BCMS: Outflow is tracked in split 1 because the call is answered by an agent in another split to which the call is queued, that is, split 2. Inflow is tracked in split 2 because the call is answered in this split and the split is not the primary split. When the call is removed from split 3, outflow is not tracked in split 3 because this split is not the primary split.

Call abandoned

The following scenario involves a call abandoned by the caller. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2, 2 and 3.
2. Call is abandoned.

The following table shows the tracking table for abandoned calls.

Split tracking			
	1	2	3
CMS	abandon	dequeue	dequeue
BCMS	abandon		

Comments:

- CMS: Abandon is tracked in split 1 because this split is the primary split. Dequeue is tracked in splits 2 and 3 because the call is dequeued from these splits without being answered in either split.
- BCMS: Abandon is tracked in split 1 because this split is the primary split. Tracking is not recorded in splits 2 and 3 because no dequeue tracking item is available.

Call answered by a primary split after a route to VDN

The following scenario involves a call answered by the primary split after a **route-to** VDN command is executed. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2 and 3.
2. Vector executes a **route-to VDN** step.
3. Call is then queued to splits 4, 5 and 6.
4. Call is answered in split 4.

The following table shows the tracking table for call answered by primary split after route to VDN.

Split tracking						
	1	2	3	4	5	6
CMS	outflow	dequeue	dequeue	answer	dequeue	dequeue
BCMS	outflow			answer		

Comments:

Split 1 is the original primary split, because this is the first split to which the call actually queues. However, split 4 becomes the new primary split because:

- Call leaves the original VDN upon execution of the **route-to VDN** step.
- Split 4 is the first split to which the call queues upon execution of this step.
- CMS: Outflow is tracked in split 1 because this split is the original primary split, and the call is dequeued from this split using a **route-to VDN** step. Dequeue is tracked in splits 2, 3, 5, and 6 because the call is dequeued from each of the splits without being answered in any one of the splits.

- BCMS: Outflow is tracked in split 1 because this split is the original primary split.

Call answered by the non-primary split after a route to VDN

The following scenario involves a call answered by the non primary split after a **route-to VDN** command is executed. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2 and 3.
2. Vector executes a **route-to VDN** step.
3. Call is then queued to splits 4, 5 and 6.
4. Call is answered in split 5.

The following table shows the tracking table for call answered by non-primary split after route to VDN.

Split tracking						
	1	**2**	**3**	**4**	**5**	**6**
CMS	outflow	dequeue	dequeue	outflow	inflow answer	dequeue
BCMS	outflow			outflow	inflow answer	

Comments:

- CMS: Outflow is tracked in split 1 because this split is the original primary split, and the call is dequeued from this split using a **route-to** VDN step. Dequeue is tracked in splits 2, 3, and 6 because the call is dequeued from each of the splits without being answered in any one of the splits. Outflow is tracked in split 4 because this split becomes the new primary split after the **route-to VDN** step is executed and the call is subsequently dequeued from this split by being answered in another split (split 5) to which the call is also queued. Finally, inflow is tracked in split 5 because the call is answered in this split, and the split is not the primary split.
- BCMS: Outflow is tracked in split 1 because this split is the original primary split. Outflow is tracked in split 4 because this split becomes the new primary split after the **route-to VDN** step is executed. Finally, inflow is tracked in split 5 because the call is answered in this split, and the split is not the primary split.

Call answered after a route to split

The following scenario involves a call answered after the call is routed to a split using a **route-to digits** or **messaging split** command. The scenario is as follows:

1. Call comes into a VDN where the vector queues the call to splits 1, 2 and 3.
2. Vector executes a **route-to digits** or **messaging split** step.
3. Call is queued to split 4 and answered by an agent in split 4.

The following table shows the tracking table for call answered after route to split.

Split tracking				
	1	**2**	**3**	**4**
CMS	outflow	dequeue	dequeue	answer
BCMS	outflow			answer

Comments:

- CMS: Outflow is tracked in split 1 because this split is the original primary split, the call is dequeued from this split using a **`route-to digits`** or **`messaging split`** step, and the call is answered in split 4, which becomes the new primary split. Dequeue is tracked in splits 2 and 3 because the call is dequeued from each of the splits without being answered in any one of the splits.
- BCMS: Outflow is tracked in split 1 because this split is the original primary split and the call is answered in split 4, which becomes the new primary split.

Evaluating split performance

By using the information presented to this point, along with the information from various reports (as discussed in the next section), the split supervisor can answer more than one question concerning split performance and then make adjustments, if necessary. Here are some of the questions the supervisor can answer:

1. How many ACD calls offered to my split were mine (that is, were offered to this split as the primary split)?

   Note:

   Split ACD calls include direct agent calls for BCMS, but not for CMS, which tracks direct agent calls separately.

2. How many of my ACD calls did my split not answer?
3. How many ACD calls that I didn't answer weren't mine?

The following sections present the answers to these questions from the perspective of the CMS and BCMS.

CMS

The following answers reflect the use of the CMS:

- The number of calls offered to my (primary) split that were mine can be determined by an examination of the CMS Split Summary Report. The algorithm is as follows: CALLSOFFERRED - INFLOWCALLS - DEQUECALLS (that is, the total number of calls offered minus the number of calls not mine that I answered minus the number of calls not mine that I didn't answer.)
- The number of my calls that my split didn't answer can be determined by an examination of the CMS VDN Report. The algorithm is as follows: ABNCALLS + BUSYCALLS + DISCCALLS + OUTFLOWCALLS (that is, the number of abandoned calls plus the number of busy calls plus the number of disconnected calls plus the number of calls outflowed from my split tagged as a primary split).
- The number of calls not mine that my split didn't answer is DEQUECALLS, which is indicated in the CMS Split Summary Report.

BCMS

The number of calls offered to my split that were mine can be determined by an examination of the BCMS Split Report. The algorithm is as follows: ACDCALLS + ABNCALLS + OUTFLOWCALLS - INFLOWCALLS (that is, the total number of calls answered plus the total number of calls abandoned from my split tagged as a primary split plus the number of calls that outflowed my split

tagged as a primary split minus the number of calls answered that were not directed to my split tagged as a primary split).

Using CMS and BCMS reports to evaluate Call Vectoring activity

There are a number of CMS and BCMS reports that allow you to evaluate Call Vectoring activity. Some of these facets include the call flows present within Call Vectoring as well as the speeds at which calls are answered. The sections that follow identify and discuss the CMS and BCMS reports that indicate this activity.

CMS reports

CMS has real-time, historical, and integrated reports. Most of the CMS historical reports are available in the following four formats: intra-hour, daily, weekday, and monthly. The following list identifies and describes several CMS reports that summarize Call Vectoring activity. For more information, see *Avaya CMS Supervisor Reports*.

Note:

The reports described in this section are generated in CMS R3 and newer releases of the CMS. Corresponding CMS R2 reports do not provide information that reflects capabilities that are new to the switch, for example, internal or external call tracking.

- Split Summary Report summarizes the call activity for an entire split. Among other information, the report provides the number of calls answered, the total number of flow ins (inflows), flow outs (outflows), dequeues, and abandoned calls.

  The report also indicates the average speed of answer (interval ASA) for calls. This refers to the sum of the queue time and ring time for a call within the answering split only. Finally, the report indicates the dequeued average queue time, which is the average time a call waits until it is answered by another split to which the call is also queued.

- VDN Report summarizes VDN activity for specific vectors. Among other information, the report provides calls answered, connected, abandoned, the number of VDN Flow Ins/Outs, calls forced busy, and calls forced disconnect. VDN Flow In pertains to calls that flow into a VDN from another VDN using a **`route-to`** command. VDN Flow Out pertains to calls that successfully flow out of VDN to another VDN or external location using a **`route-to`** command.

- Vector Report summarizes vector activities. Among other information, the report provides the number of calls offered, calls answered, calls abandoned, Vector Flow Ins/Outs, calls forced busy, and calls forced disconnect. Vector Flow In pertains to calls that flow into a vector from another vector using a **`route-to`** or **`goto vector`** command. Vector Flow Out pertains to calls that successfully flow out of a vector using a **`route-to`** or **`goto vector`** command.

BCMS reports

BCMS has a real-time split report, split historical reports, real-time VDN reports, and VDN historical reports. The following list identifies and describes several BCMS reports that summarize Call Vectoring activity. For more information on these and other related reports, see the *Avaya Communication Manager Call Center Software - Basic Call Management System (BCMS) Operations* document.

BCMS Split Report

Summarizes the call activity for an entire split. The information can be requested either daily or by the administered time period. Among other information, the report provides the total number of flow ins (inflows) and flow outs (outflows), the calls answered and calls abandoned. The report also provides the average speed of answer time for calls handled by the split during the indicated time period.

VDN Summary Report

Summarizes statistical information for all internally-measured VDNs. The information can be requested by the administered time interval or daily. The list bcms vdn report gives multiple time periods or days for a single VDN. The list bcms summary vdn report gives a one-line summary per vdn (with data from the specified times or days), but can give the data for numerous vdns.

The report also indicates the total number of flow outs, specifically, the number of calls that route to another VDN or to a destination external to the switch. However, calls that encounter a **goto vector** command are not shown as outflows. No further measurements are taken on the calls once the calls have outflowed. If an outflowed call later abandons, this is not indicated in the report.

Among other information, the VDN report provides a total for offered calls, answered calls, abandoned calls, and also one for calls that were either forced busy or forced disconnect.

VDN Real-Time Report

Provides statistical information including the number of calls currently waiting and the oldest call waiting. The VDN real-time report has the same characteristics as other real-time BCMS reports.

Using CMS in an EAS environment

The same tracking and database items used within a traditional Call Vectoring environment are used within an EAS environment but there are also new items that are specific to EAS. All existing custom reports work when you are upgrading to EAS.

Related topics explain how the following entities are tracked in an environment with EAS optioned.

Agents and skills

The fields under the Extn column in the CMS Real-Time Agent Report show the extension that the agent is logged into. The fields can be used to locate the agent or to service observe the agent.

With **EAS** enabled, the Skill Status Report replaces the Split Status Report. The report indicates the skills logged into and the skill level of each skill. If too many calls are waiting, or if calls are waiting too long (also shown on the Skill Status report), it is possible that not enough agents have the skill administered at a high enough skill level.

An agent can be denied login to some skills if the maximum agents or skill number is met or if the CMS limit on agent or skill pairs logged in has been reached.

CMS reports show only the first 15 skills that an agent is logged into.

DAC calls

Waiting direct agent calls are not included in the calls waiting and Oldest Call Waiting (OCW) report fields for skills because such calls are not skill calls. However, direct agent calls are included in the two report fields for VDNs.

The Queue/Agent Summary Real-Time Report lists separately the direct agent calls waiting in a skill queue. direct agent calls are queued to the skill that is administered as the direct agent skill. To manage the skill's queue slots effectively, it is recommended that a skill be dedicated for direct agent calls.

Since direct agent calls are not skill calls, the skill tables do not track direct agent calls; however, the tables do monitor skill queue slots. The agent's time is tracked as OTHER in the skill tables. In the agent tables, there are separate direct agent call items. The standard CMS agent reports add the direct agent calls and the skill ACD calls and report these calls as ACD calls. The VDN tables track direct agent calls as ACD calls.

Non-ACD calls

The first measured skill that an EAS agent is logged into is used by CMS to track non-ACD calls unless the agent has an ACD call on hold. If an ACD call is on hold, outgoing non-ACD calls are counted for the skill of the held ACD call.

VDN skill preferences

VDN skill preference data is collected to provide information on what groups of agents (skills) are handling calls and on how effectively each skill group handles a particular VDN.

Real-time and historical VDN Skill Preference reports can be used to compare the percentage of calls being answered by the 1st, 2nd, and 3rd VDN preferences against an objective. If too few calls are being answered by the 1st skill preference, the vector can be adjusted to allow more time for the 1st skill preference group to answer calls; another alternative is to train or hire more agents with the 1st skill preference.

You can use VDN skill preference data to compare the average talk time and average ACW time for agents in the 1st, 2nd, and 3rd skill groups. If these times vary too much across groups, more training is required for the backup groups, that is, the 2nd and 3rd skill groups.

VDN skill preference data is tracked according to the skill preferences (1st, 2nd, 3rd) assigned to the VDN. Whenever a vector step either references a 1st, 2nd, or 3rd skill or specifies a skill number that matches the 1st, 2nd, or 3rd skill administered, the new database items are tracked. For example, if VDN 1000 has Skills 21, 22, and 23 administered as the 1st, 2nd, and 3rd skills, respectively, and if the vector associated with VDN 1000 has a queue to main skill 22 step, tracking occurs for the 2nd VDN skill preference if the call is answered by an agent in Skill 22. Skill preference tracking also occurs for Skills 21 and 23. This allows users who prefer to specify the actual skill number in the vector to take advantage of the tracking for VDN skill preferences.

EAS administration from CMS

CMS can be used to administer vectors as well as skills for agents and VDNs. The ACD Administration: Change Agent Skills CMS screen is used to display and modify the skills and levels assigned to an agent, as well as the assigned direct agent skill and call handling preference.

The ACD Administration: Change VDN Skill Preferences screen is used to request a VDN's skill preferences and to modify the VDN's skills.

The CMS Vector Contents screen is used to create and modify vectors. CMS supports the Call Vectoring commands that queue calls to the 1st, 2nd, or 3rd VDN skill.

Chapter 5: Avaya 96X1 SIP agent deskphones

9608, 9611G, 9621G, and 9641G are part of the multiline 9600 Series IP Deskphones. The deskphones are signaling protocol independent with two telephony applications, Avaya Deskphone H.323 and Avaya Deskphone SIP, that support H.323 and SIP respectively.

9621G and 9641G are touch-based deskphones with a color display. 9611G and 9608 are button-based deskphones. 9611G has a color display while 9608 has a monochrome display. With the 9641G, 9608, and 9611G models, you can use a dual headset adapter so that two persons can listen to calls. You can also attach more than one Button Module (BM) to these deskphone models to extend call appearances, bridge appearances, or feature keys. For an agent, the deskphones offer convenient features and capabilities, including a phone screen to view and manage calls, and icons that indicate agent status, call state, feature status, queued calls, and missed calls.

Related links

96X1 SIP agent deskphone feature support on page 436
96X1 SIP value-added capabilities on page 437

96X1 SIP agent deskphone feature support

The 96X1 SIP agent deskphones supports the following call center features:

- Agent greeting
- Agent login and logout buttons instead of Feature Access Codes (FACs): The phone shifts the button for login to logout after the agent logs in to the system.
- Auto and manual answer.
- Automatic display of collected digits with an incoming call that follows a call transfer.
- Auxiliary (AUX) work and After Call Work (ACW) buttons.
- Call Work Codes (CWCs) button.
- Coaching: The **sip-sobsrv** button is used by supervisors to monitor and coach EAS agents when agents speak with callers. For coaching, supervisors need to press the **Coach** soft key on SIP CC endpoints.
- Correct messaging to the reporting adjuncts.
- Display of active Vector Directory Number (VDN).

- Display of Adjunct Switch Application Interface (ASAI) User-to-User (UUI) information.
- Entry of reason codes for change to the AUX work mode and for logout.
- Hold, mute, transfer, conference, message waiting, elapsed call timer, date and time display, exit, and a minimum of three call appearances.
- Insertion of Vector Directory Number (VDN) of Origin Announcement (VOA) after answer with manual-answer operation, or accompanying zip tone for autoanswer operation.
- Interruptible AUX work.
- Manual-in and Auto-in work modes.
- Message Waiting Indicator (MWI) tracking for Expert Agent Selection (EAS) agent login IDs.
- Queue Status button: The **q-calls** button displays the number of calls in a queue and the time the oldest call is in a queue.
- Release button.
- Service Observing: The **sip-sobsrv** button is used by supervisors to monitor and coach EAS agents when agents speak with callers. Observers can hear VOA only after they have joined the observed call.

  Service Observing from a SIP endpoint works as follows:

  - A service observer does not have to be logged-in as an agent to enable Service Observing on an object (VDN, agent, or station).
  - If logged-in as an agent, a service observer must be in an AUX work state to enable Service Observing on an object (VDN, agent, or station).
- Stroke/Event Count button.
- Third-party MWI button.
- Visible and audible confirmation of feature activation and status change to the agent.
- VuStats button.

Note:

VOA-repeat functionality is not supported on 96x1 SIP phone types. You must not select the voa-repeat button while configuring 96x1 SIP extensions.

Related links

Avaya 96X1 SIP agent deskphones on page 436

96X1 SIP value-added capabilities

The 96X1 SIP agent deskphones offer the following value-added capabilities:

- Agent log in and log out using a button on the deskphone.
- Consistent view of information on the Agent Information Line. Deskphone displays VuStats or Q-calls/Q-time information.

- Consistent view of information on the Agent Status Line. Deskphone displays collected digits without user action if information is available with a call.
- Contact lists, call logs, and menu of options and settings to customize agent deskphones.
- Deskphone screen for viewing and managing calls.
- Display of the list of assigned skill numbers with alerts when the skills change.
- Display of the number of direct agent calls waiting for an agent.
- Feature buttons to change agent work modes and work states.
- Handling of confirmation or denial tones when an agent logs in to the system.
- Improved user interaction with entry for features with specific displays and validations: The Communication Manager option settings are passed to the deskphone to customize the user interface.
- Softkeys to select an option or action displayed on the deskphone screen.
- Text display of denial and error conditions.
- Text or icon displays, in addition to lamp indications for the auto-in and manual-in agent work modes and work states, such as Avail, On-ACD, AUX, and ACW, as well as pending changes.

Related links

Avaya 96X1 SIP agent deskphones on page 436

Chapter 6: Security issues

Call Vectoring can be integrated into the security of your switch. For example, Call Vectoring and Call Prompting can be used to help prevent unauthorized users from gaining access to the switch using the Remote Access feature. This section explains how this is done.

Remote access

Abuse of remote access on the switch is one of the main methods by which unauthorized users obtain telephone services illegally. This section explains how a number of Call Vectoring features can be used to prevent unauthorized use of the remote access feature. No new development is required for any of these services.

Two methods are available:

- Front-ending remote access
- Replacing remote access

Front-ending remote access

This method gives authorized external callers a VDN extension to call instead of the remote access extension, which is kept private. The corresponding call vector can then implement a number of security checks before routing callers to the remote access extension. Routing can be done using a **`route-to number`** or **`route-to digits`** step.

The following advantages are possible using this method:

- Call Vectoring can introduce a delay before the dial-tone is provided to the caller. Immediate dial-tone is often one criterion searched for by a hacker's programs when the hacker is trying to break into a system.
- A recorded announcement declaring that the use of Communication Manager services by unauthorized callers is illegal and that the call is subject to monitoring or recording can be played for the caller.
- Call Prompting can be used to prompt for a password. In such a case, the call is routed only if there is a match on the password.
- Use of the remote access extension can be limited to certain times of the day or certain days of the week.

- Real-time and historical reports on the use of the remote access feature can be accessed from CMS or from BCMS.
- Different passwords can be used on different days of the week or at different times during the day.
- Many VDNs that call the remote access extension can be identified. Accordingly, individuals or groups can be given their own VDN with unique passwords, permissions and reports. Any abuse of the system or security leak can then be attributed to an individual or a group.
- The caller can be routed to a VRU using the **converse-on** step where more sophisticated security checking, such as speaker recognition, can take place.
- Anyone failing any of the security checks can be routed to a security VDN that routes the caller to security personnel with a display set or to a VRU. Such a call shows the attempted password on the display. If the call is passed to a VRU, the VDN, the ANI or the prompted digits can be captured. CMS and BCMS reports on this security violation VDN gives information on how often and when security violations occur.

Replacing remote access

For this method, the remote access extension is not used. More than one VDN is designed to access call vectors that can employ all of the security checks described in the previous section. The same reports and monitoring or recording capabilities described in the previous section can also be used. Instead of routing to the remote access extension, the vector collects digits from the caller and routes to the given destination if there is a match on the password.

Again, multiple VDNs can be created for individuals or groups with different security checks and different permissions or restrictions. Destination numbers provided by callers can be screened by the vectors and denied if the user does not have permission to access that destination. For example, an individual user can be restricted from placing calls to numbers beginning with area codes 303 and 908.

EAS

With EAS, agent stations can be locked when not staffed. This is accomplished by assigning the station a COR that does not allow outbound calls or restricts outbound calls from toll calls.

EAS agents have an optional password of up to nine digits to log in. This password is not displayed on DCP terminals when the agent is entering the password on the dial pad.

Limiting outside access using VDN COR restrictions

Routing calls through Communication Manager with Call Vectoring can raise some security issues. A VDN has a COR. Calls processed by the vector carry the permissions and restrictions associated with the COR of the VDN.

For example, if a vector in Communication Manager is written to collect digits and then to route to the digits dialed, the restrictions on what calls can be placed are determined by the COR of the latest VDN. Also, checks can be made on the digits that are dialed, using **goto if digits** vector commands. For example, **goto if digits = 123** to disallow routing to undesired destinations. The **collect digits** step can also be limited to collect only the number of digits required, for example, only collecting five digits for internal dialing.

An incoming caller can access Trunk Access Codes, some FACs, and most other sets of dialed digits. To deny incoming callers access to outgoing facility paths, configure the COR of the VDN to deny outgoing access. The configuration must include the following:

- Lowering the Facility Restriction Level (FRL) in the COR to the lowest acceptable value. FRL=0 provides the most restricted access to network routing preferences.
- Assigning a Calling Party Restriction of Toll or Outward denying Facility Test Call capability.
- Blocking access to specific CORs assigned to outgoing trunk groups using the Calling Permissions section of the Class Of Restriction screen.

Review the CORs assigned to the VDNs. If not restricted, assign restrictions on the VDN or use goto tests on the digits to prevent callers from exiting the system using the vector.

Meet-me Conference security issues

Meet-me Conference is a potential security problem. If Meet-me Conference VDNs are assigned without access codes, hackers can tie up Meet-me Conference facilities, keeping others from conducting legitimate business and can potentially access the switch and use the switch to make unauthorized calls. Therefore, all Meet-me Conference VDNs must have access codes known to the users only on a need-to-know basis. You must also change the access codes on a regular basis to reduce the risk of unauthorized access.

If a user tries to change the access code of a Meet-me Conference and is unsuccessful, or if a user tries to access a Meet-me Conference and uses an invalid access code, a meet-me event is logged. For more information, see “Tracking unexpected events” in the *Programming Call Vectoring Features in Avaya Aura® Call Center Elite* document.

Vector-initiated service observing

The following restrictions can be used with vector-initiated service observing to guard against unauthorized use.

- Call prompting commands can be used in service observing vectors to provide passcode protection, and to limit access to observing specific destinations or verified caller entered digits.

- Time of Day/Day of Week checks can be incorporated in service observing vectors.
- A vector can be created to be used exclusively for service observing.
- For a VDN to be observed as the result of a route-to command, the VDN must have a COR that allows it to be observed.
- The calling permissions of the COR assigned to the service observing VDN in conjunction with the can be observed settings of the COR assigned to the destination determine what agents, stations, or VDNS can be observed.

Voice Response Integration (VRI)

When a **converse** step is used to access a VRU application that returns data for a collect digits step, the opportunity for toll fraud exists when the VRU application fails to return any data. Ensure that one of the following is true to prevent toll fraud:

- If the collected digits are used to route calls internally, be certain that the COR for the VDN does not allow calls to route externally.
- If you have to use collected digits to route calls externally, use a password to verify that the collected digits have been passed by the VRU application. In the following vector example the VRU application returns a three-digit password followed by the eight-digit external number. The vector routes calls without the correct password to a different vector and routes calls with the correct password to the collected digits.

VRI security

```
converse-on split 10 pri m passing none and none
collect 3 digits after announcement none
goto vector 23 if digits <> 234
collect 8 digits after announcement none
route-to digits with coverage n
```

Attendant Vectoring security issues

Security Violation Notification (SVN) referral calls can be directed to an attendant group. These are priority calls and, as such, cannot terminate to a VDN. However, when the calls are sent to the attendant group, the calls are treated as ordinary calls. Priority does not apply to attendant group processing. So, the calls are treated as normal attendant group calls and are sent through vector processing.

Chapter 7: Resources

Documentation

Title	Use the document to:	Audience
Supporting		
Avaya Aura® Communication Manager System Capacities Table	Read about the system capacity and system scalability.	Implementation engineers, sales engineers, and solution architects
Programming Call Vectoring Features in Avaya Aura® Call Center Elite	Write and edit call vectors.	Implementation engineers and system administrators
Avaya Aura® Call Center Elite Call Vectoring Feature Description	Learn how the Call Vectoring feature work and get to now details about feature characteristics, capabilities, capacities, and interactions.	Implementation engineers and system administrators
Understanding		
Avaya Aura® Call Center Elite Feature Reference	Read about Automatic Call Distribution (ACD) and Call Vectoring features.	All users of Avaya Aura® Call Center Elite
Avaya Aura® Communication Manager Feature Description and Implementation	Read about Avaya Aura® Communication Manager features.	All users of Communication Manager
Using		
Using Avaya Business Advocate	Learn how to use Business Advocate for agent selection and call selection.	Contact center managers, system administrators, and supervisors
Using Avaya 96X1 SIP Agent Deskphones with Avaya Aura® Call Center Elite	Know the prerequisites for using Avaya 96X1 SIP agent deskphones with Call Center Elite.	Implementation engineers and system administrators
Administering		
Administering Avaya Aura® Call Center Elite	Administer all the Call Center Elite features.	System administrators
Troubleshooting		

Table continues…

Title	Use the document to:	Audience
Troubleshooting Avaya Aura® Call Center Elite	Know how to troubleshoot common problems and denial events related to Call Center Elite.	All users who perform troubleshooting tasks in Call Center Elite
Implementation		
Planning for an Avaya Aura® Call Center Elite Implementation	Know how to transition from a basic call center environment to an Expert Agent Selection (EAS) and a Call Vectoring environment.	All users who perform Call Center Elite site preparation and planning tasks, including implementation engineers and sales engineers.

Finding documents on the Avaya Support website

Procedure

1. Go to https://support.avaya.com.
2. At the top of the screen, click **Sign In**.
3. Type your **EMAIL ADDRESS** and click **Next**.
4. Enter your **PASSWORD** and click **Sign On**.
5. Click **Product Documents**.
6. Click **Search Product** and type the product name.
7. Select the **Select Content Type** from the drop-down list
8. In **Select Release**, select the appropriate release number.

   For example, for user guides, click **User Guides** in the **Content Type** filter. The list only displays the documents for the selected category.
9. Press **Enter**.

Accessing the port matrix document

Procedure

1. Go to https://support.avaya.com.
2. At the top of the screen, click **Sign In**.
3. Type your **EMAIL ADDRESS** and click **Next**.
4. Enter your **PASSWORD** and click **Sign On**.
5. Click **Product Documents**.

6. Click **Search Product** and type the product name.
7. Select the **Select Content Type** from the drop-down list
8. In **Choose Release**, select the required release number.
9. In the **Content Type** filter, select one or both the following categories:
   - **Application & Technical Notes**
   - **Design, Development & System Mgt**

   The list displays the product-specific Port Matrix document.
10. Press **Enter**.

Avaya Documentation Center navigation

For some programs, the latest customer documentation is now available on the Avaya Documentation Center website at https://documentation.avaya.com.

Important:

For documents that are not available on Avaya Documentation Center, click **More Sites** > **Support** on the top menu to open https://support.avaya.com.

Using the Avaya Documentation Center, you can:

- Search for keywords.

  To filter by product, click **Filters** and select a product.
- Search for documents.

  From **Products & Solutions**, select a solution category and product, and then select the appropriate document from the list.
- Sort documents on the search results page.
- Click **Languages** ( ) to change the display language and view localized documents.
- Publish a PDF of the current section in a document, the section and its subsections, or the entire document.
- Add content to your collection using **My Docs** ( ).

  Navigate to the **Manage Content** > **My Docs** menu, and do any of the following:
  - Create, rename, and delete a collection.
  - Add topics from various documents to a collection.
  - Save a PDF of the selected content in a collection and download it to your computer.
  - Share content in a collection with others through email.

- Receive collection that others have shared with you.

- Add yourself as a watcher using the **Watch** icon ( ).

  Navigate to the **Manage Content** > **Watchlist** menu, and do the following:

  - Enable **Include in email notification** to receive email alerts.
  - Unwatch selected content, all content in a document, or all content on the Watch list page.

  As a watcher, you are notified when content is updated or deleted from a document, or the document is removed from the website.
- Share a section on social media platforms, such as Facebook, LinkedIn, and Twitter.
- Send feedback on a section and rate the content.

Note:

Some functionality is only available when you log in to the website. The available functionality depends on your role.

Training

The following courses are available on www.avaya-learning.com. Enter the course code in the **Search** field, and click **Go** to search for the course.

Course code	Course title
ACIS-7391	
73600V	Implementing Avaya Aura® Call Center Elite 40 hours
7391X	Avaya Aura® Call Center Elite and Avaya Aura® Call Center Elite Multichannel Implementation Exam 1.50 hours
ACSS-7491	
74600V	Supporting Avaya Aura® Call Center Elite 16 hours
7491X	Avaya Aura® Call Center Elite and Avaya Aura® Call Center Elite Multichannel Support Exam 1.50 hours
2416W	Avaya Aura® Call Center Elite Fundamentals 0.5 hour for all audiences
2412W	Using Avaya Workspaces for Call Center Elite – Agents 0.5 hour for end-users
2414W	Using Avaya Workspaces for Call Center Elite – Supervisors 0.5 hour for end-users

Viewing Avaya Mentor videos

Avaya Mentor videos provide technical content on how to install, configure, and troubleshoot Avaya products.

About this task

Videos are available on the Avaya Support website, listed under the video document type, and on the Avaya-run channel on YouTube.

- To find videos on the Avaya Support website, go to https://support.avaya.com/ and do one of the following:
  - In **Search**, type `Avaya Mentor Videos`, click **Clear All** and select **Video** in the **Content Type**.
  - In **Search**, type the product name. On the Search Results page, click **Clear All** and select **Video** in the **Content Type**.

    The **Video** content type is displayed only when videos are available for that product.

  In the right pane, the page displays a list of available videos.
- To find the Avaya Mentor videos on YouTube, go to www.youtube.com/AvayaMentor and do one of the following:
  - Enter a keyword or keywords in the **Search Channel** to search for a specific product or topic.
  - Scroll down Playlists, and click a topic name to see the list of videos available. For example, Contact Centers.

Note:

Videos are not available for all products.

Support

Go to the Avaya Support website at https://support.avaya.com for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

Related links

Using the Avaya InSite Knowledge Base on page 447

Using the Avaya InSite Knowledge Base

The Avaya InSite Knowledge Base is a web-based search engine that provides:

- Up-to-date troubleshooting procedures and technical tips
- Information about service packs
- Access to customer and technical documentation
- Information about training and certification programs

- Links to other pertinent information

If you are an authorized Avaya Partner or a current Avaya customer with a support contract, you can access the Knowledge Base without extra cost. You must have a login account and a valid Sold-To number.

Use the Avaya InSite Knowledge Base for any potential solutions to problems.

1. Go to https://support.avaya.com.
2. At the top of the screen, click **Sign In**.
3. Type your **EMAIL ADDRESS** and click **Next**.
4. Enter your **PASSWORD** and click **Sign On**.

   The system displays the Avaya Support page.
5. Click **Support by Product** > **Product-specific Support**.
6. In **Enter Product Name**, enter the product, and press `Enter`.
7. Select the product from the list, and select a release.
8. Click the **Technical Solutions** tab to see articles.
9. Select **Related Information**.

Related links

Support on page 447

Glossary

AAR	When resources are unavailable, Communication Manager uses the Automatic Alternate Routing (AAR) feature to route calls to a different route than the first-choice route.
abandoned call	An inbound call in which the caller disconnects the call before an agent can answer the call.
access code	A 1-digit, 2-digit, or 3-digit dial code that activates or cancels a feature, or accesses an outgoing trunk.
access trunk	A trunk that connects a main communications system with a tandem communications system in an Electronic Tandem Network (ETN) on page 455. You can use an access trunk to connect a system or tandem to a serving office or service node. Also called an access tie trunk.
ACD	Automatic Call Distribution (ACD) is a telephony feature for processing and distributing inbound, outbound, and internal calls to groups of extensions.
ACW	An agent enters the After Call Work (ACW) mode to complete ACD call-related activities, such as filling forms or taking notes. An agent in the ACW mode is unavailable to receive ACD calls.
AD	Abbreviated Dialing (AD) makes agent login easier as agents can press the **AD** button to dial an access code, split number, or login ID.
adjunct	A processor that does tasks for another processor and is optional in the configuration of the other processor. See also application on page 450.
adjunct routing	A means of evaluating calls before the calls are processed by requesting information from an adjunct. Communication Manager requests instructions from an associated adjunct and makes a routing decision based on agent availability or caller information.
adjunct-controlled split	An ACD split that is administered to be controlled by another application. Agents logged in to such splits must do all telephony work, ACD login, ACD logout, and work mode changes through the adjunct, except for auto-available adjunct-controlled splits, wherein agents cannot log in, log out, or change the work modes.

adjunct-monitored call	An adjunct-controlled call, active-notification call, or call that provides event reporting over a domain-control association.
adjusted EWT	A Best Service Routing (BSR) term for Expected Wait Time (EWT) plus a user adjustment set by a **consider** command.
administration terminal	A terminal that is used to administer and maintain a system.
AES	Application Enablement Services (AES) is an Avaya product that provides a platform for the development of CTI-based applications for Communication Manager.
agent	A member of an ACD hunt group, ACD split, or skill. Depending on the ACD software, an agent can be a member of multiple splits/skills.
agent report	A report that provides historical traffic information for internally measured agents.
ANI	Automatic Number Identification (ANI) is a display of the calling number for agents to gain access to information about the caller.
appearance	A software process that is associated with an extension and whose purpose is to supervise a call. An extension can have multiple appearances. Also called call appearance, line appearance, and occurrence.
application	An adjunct that requests and receives ASAI services or capabilities. Applications can reside on an adjunct. However, Communication Manager cannot distinguish among several applications residing on the same adjunct. Hence, Communication Manager treats the adjunct and all resident applications as a single application. The terms application and adjunct are used interchangeably throughout the document.
application plan	A plan used in multisite Best Service Routing (BSR) applications. The application plan identifies remote Communication Manager servers for comparison in a consider series. The plan specifies information that is required to contact each Communication Manager server and to interflow calls to the selected Communication Manager server.
applications processor	A micro-computer based, program controlled computer providing application services for the switch. The processor is used with several user-controlled applications such as traffic analysis and electronic documentation.
ARS	Automatic Route Selection (ARS) is a feature that Communication Manager uses to automatically select the least cost route to send a toll call.

ASAI	Adjunct-Switch Application Interface (ASAI) is an Avaya protocol that applications use to gain access to the call-processing capabilities of Communication Manager.
association	A communication channel between adjunct and switch for messaging purposes. An active association is one that applies to an existing call on the switch or to an extension on the call.
attendant	A person at a console who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations.
attendant console	The workstation used by an attendant. The attendant console allows the attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some system operations. Also called console.
AUDIX™	Audio Information Exchange (AUDIX™) is an Avaya messaging system.
auto-in	A call-answering mode in which an agent automatically receives ACD calls without pressing any button to receive calls.
Automatic Callback	A feature that enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both originating and receiving parties when the receiving party becomes available.
Automatic Circuit Assurance (ACA)	A feature that tracks calls of unusual duration to facilitate troubleshooting. A high number of very short calls or a low number of very long calls signify a faulty trunk.
automatic trunk	A trunk that does not require addressing information because the destination is predetermined. A request for service on the trunk, called a seizure, is sufficient to route the call. The normal destination of an automatic trunk is the communications-system attendant group. Also called automatic incoming trunk and automatic tie trunk.
AUX work	Agents enter the Auxiliary (AUX) work mode for non-ACD activities, such as taking a break, going for lunch, or making an outgoing call. Agents in the AUX work mode are unavailable to receive ACD calls.
auxiliary trunk	A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.
available agent strategy	A strategy that determines how the Best Service Routing (BSR) commands in a vector identify the best skill when multiple skills have available, that is, idle agents.

Avaya Aura®	A converged communications platform unifying media, modes, network, devices, applications. Avaya Aura® is based on the SIP architecture with Session Manager at the core.
Avaya Aura® Media Server	Avaya Aura® Media Server (Avaya Aura® MS) is a software-based media platform. Communication Manager uses Avaya Aura® MS to provide IP audio, tone generation and detection, and announcement capabilities similar to legacy H.248 media gateways or port networks with media processors.
AWOH	Administration Without Hardware (AWOH) is a feature that allows administration of ports without associated terminals or other hardware.
barrier code	A security code used with remote access to prevent unauthorized access to the system.
BCMS	A software package residing on Communication Manager that monitors the operations of ACD systems. Basic Call Management System (BCMS) collects data related to the calls on Communication Manager and organizes data into reports that help supervisors manage ACD facilities and personnel.
best	The split/skill or location that can provide the best service to a caller as determined by BSR.
BHCC	Busy-Hour Call Completion (BHCC) is a measure of the number of calls that Communication Manager successfully completes during the peak hour of a network.
BRI	Basic Rate Interface (BRI) is an ISDN configuration that offers two bearer (B) channels for voice and data and one data channel for signaling.
bridge (bridging)	The appearance of a phone extension at other phones.
bridged appearance	A call appearance on a telephone that matches a call appearance on another telephone for the duration of a call.
BSR	A feature that provides singlesite and multisite load balancing and maximizes staffing resources. Communication Manager uses Best Service Routing (BSR) to compare skills and to route calls to the best skill.
Business Advocate	A Call Center Elite feature that establishes different service levels for different types of calls. For example, a company decides that a premium customer must receive service before the other types of customers.
call appearance	For an attendant console, the six buttons labeled **a** to **f** for making calls, receiving calls, or holding calls. For a deskphone, a button labeled with an extension for making calls, receiving calls, or holding calls.

cause value

A value that is returned in response to requests or in event reports when a denial or unexpected condition occurs.

CCS or hundred call seconds

A unit of call traffic. Call traffic for a facility is scanned every 100 seconds. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter C. The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated CCS. If a facility is busy for an entire hour, the facility is said to have been busy for 36 CCS.

channel

1. A circuit-switched call.
2. A communications path for transmitting voice and data.
3. In wideband, all of the time slots (contiguous or noncontiguous) necessary to support a call. Example: an H0-channel uses six 64-kbps time slots.
4. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

circuit

1. An arrangement of electrical elements through which electric current flows.
2. A channel or transmission path between points.

circuit pack

A card with microprocessors, transistors, and other electrical circuits. A circuit pack is installed in a switch carrier or bay. Also called a circuit board or circuit card.

CMS

A software program for reporting and managing agents, splits, trunks, trunk groups, vectors, and VDNs. With Call Management System (CMS), you can also administer some ACD features.

CO

Central Office (CO) is a switch that a local phone company owns to provide local phone service (dial-tone) and access to toll facilities for long-distance calling.

communications server

A software-controlled processor complex that interprets dialing pulses, tones, and keyboard characters and makes the proper connections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the connections. A communications system provides voice and data communications services, including access to public and private networks, for telephones and data terminals on a customer’s premises. Previously called a switch or a Private Branch eXchange (PBX).

confirmation tone

A telephone tone confirming that feature activation, deactivation, or cancellation has been accepted.

connectivity	A connection of disparate devices within a single system.
consider sequence	A consider series plus a **`queue-to best`**, **`check-best`**, or **`reply-best`** step is called a consider sequence.
consider series	A series of **`consider`** commands typically written in sets. A set of **`consider`** commands is called a consider series.
COR	Class of Restriction (COR) is a feature that allows classes of call-origination and call-termination restrictions for phones, phone groups, data modules, and trunk groups.
COS	Class of Service (COS) is a feature that uses a number to specify if phone users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features.
coverage answer group	A group of up to eight telephones that ring simultaneously when a call is redirected by Call Coverage. Any one of the group can answer the call.
coverage call	A call that is automatically redirected from the called party's extension to an alternate answering position when certain coverage criteria are met.
coverage path	An order in which calls are redirected to alternate answering positions.
coverage point	An extension or attendant group, VDN, or ACD split designated as an alternate answering position in a coverage path.
covering user	A person at a coverage point who answers a redirected call.
CWC	Call Work Codes (CWCs) are up to 16–digit sequences that agents type to record the occurrence of customer-defined events, such as account codes or social security numbers.
data link	A configuration of physical facilities enabling end terminals to communicate directly with each other.
data terminal	An input/output (I/O) device that has either switched or direct access to a host computer or to a processor interface.
dial-repeating tie trunk	A tie trunk that transmits called-party addressing information between two communications systems.
dial-repeating trunks	A PBX tie trunk that is capable of handling PBX station-signaling information without attendant assistance.
direct agent	A feature, accessed only through ASAI, that allows a call to be placed in a split queue but routed only to a specific agent in that split. The call receives normal ACD call treatment (for example, announcements) and is measured as an ACD call while ensuring that a particular agent answers.

Direct Inward Dialing (DID) trunk	An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.
DMCC	Device, Media, and Call Control (DMCC) is the new name for Communication Manager Application Programming Interface (API), that is, CMAPI.
domain	A group of VDNs, ACD splits, and stations.
Dynamic Host Configuration Protocol (DHCP)	A protocol that dynamically assigns IP addresses to devices when the devices connect to the network.
Dynamic Percentage Adjustment	A Business Advocate feature that automatically adjusts the agent target allocations to meet the administered service level targets.
Dynamic Queue Position	A Business Advocate feature that queues calls from multiple VDNs to a single skill, while maintaining the service objectives of each originating VDN. For instance, DQP positions a premium customer call with an assigned service objective of 10 seconds before a regular customer call with an assigned service objective of 25 seconds. Dynamic Queue Position (DQP) is also known Service Objective by VDN.
Dynamic Threshold Adjustment	A Business Advocate Service Level Supervisor (SLS) feature that meets the administered service levels by automatically adjusting the overload thresholds to engage reserve agents.
EAD-LOA	Expert Agent Distribution - Least Occupied Agent (EAD-LOA) is an agent selection method for call delivery. With EAD-LOA, calls are delivered to the available agent with the highest skill level and the lowest percentage of work time since login, when compared to other available agents with the same skill level.
EAD-MIA	Expert Agent Distribution - Most Idle Agent (EAD_MIA) is an agent selection method for call delivery. With EAD-MIA, calls are delivered to the available agent with the highest skill level who has been idle the longest since the last ACD call, when compared to other available agents with the same skill level.
ECC	External Call Controller (ECC) is an external Media Gateway Controller (MGC) that communicates with the G250 or G350 media gateways in a network.
Electronic Tandem Network (ETN)	A large private network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each telephone is assigned a unique extension.

Exclusion	A feature that allows multi-appearance telephone users to keep other users with the same extension from bridging onto an existing call.
Expansion Port Network (EPN)	A port network that is connected to the Time Division Multiplex (TDM) bus and packet bus of a processor port network. Control is achieved by indirect connection of the EPN to the processor port network using a port-network link.
Expected Wait Time (EWT)	A prediction of how long a call waits in queue before the call is answered.
extension-in (EXT-IN)	A work state agents go into when answering a non-ACD call. If the agent is in manual-in or auto-in and receives an EXT-IN call, the call is recorded by the reporting adjunct as an AUX-IN call.
extension-out (EXT-OUT)	A work state that agents go into when placing non-ACD calls.
external call	A connection between a communications system user and a party on the public network, or on another communications system in a private network.
facility	A telecommunications transmission pathway and the associated equipment.
glare	A simultaneous seizure of a 2-way trunk by two communications systems resulting in a standoff.
ground-start trunk	A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.
holding time	A total length of time in minutes and seconds that a facility is used during a call.
ICC	Internal Call Controller (ICC) is an internal MGC that communicates with the G250 or G350 media gateways in a network.
IMS	IP Multimedia Subsystem (IMS) is an architectural framework for delivering IP multimedia services.
in-use lamp	A red light on a multiappearance telephone that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.
intelligent polling	An automatic feature of Best Service Routing (BSR) that significantly reduces the number of status polls executed. When a remote location

	cannot be the best resource at a given moment in time, the intelligent polling feature temporarily suppresses polls to that location.
intercept tone	An tone that indicates a dialing error or denial of the service requested.
interflow	An ACD term that refers to the ability to establish a connection to a second ACD and overflow a call from one ACD to the other.
internal call	A connection between two users within a system.
internal measurement	A BCMS measurement that is made by the system.
intraflow	An ACD term that refers to the ability for calls to redirect to other splits on the same Communication Manager to backup the primary split.
ISDN	Integrated Services Digital Network (ISDN) is a communication standard for digital transmission of voice and data in a public switched telephone network.
ISDN Gateway (IG)	A feature allowing integration of the switch and a host-based telemarketing application using a link to a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based telemarketing application of call events.
ISDN trunk	A trunk administered for use with ISDN-PRI. Also called ISDN facility.
line	A transmission path between a communications system or Central Office (CO) switching system and a telephone.
line port	A piece of hardware that provides the access point to a communications system for each circuit associated with a telephone or data terminal.
link	A transmitter-receiver channel that connects two systems.
maintenance	Activities involved in keeping a telecommunications system in proper working condition: the detection and isolation of software and hardware faults, and automatic and manual recovery from these faults.
major alarm	An indication of a failure that has caused critical degradation of service and requires immediate attention. Major alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote maintenance facility, if applicable.
management terminal	A terminal that the system administrator uses to administer the switch. The administrator can also use the management terminal to gain access to the BCMS feature.

manual-in	A call-answering mode in which an agent must press **manual-in** to receive an ACD call.
MAO	Maximum Agent Occupancy (MAO) is a feature that Communication Manager uses to set thresholds on the amount of time that an agent spends on a call. The MAO threshold is a system-administered value that places an agent in the AUX work mode when the agent exceeds the MAO threshold for calls.
message center	An answering service that supplies agents and stores messages for later retrieval.
message-center agent	A member of a message-center hunt group who takes and retrieves messages for telephone users.
MGC	Media Gateway Controller (MGC) controls the phone services on a media gateway.
minor alarm	An indication of a failure that affects customer service. Minor alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported to a remote maintenance facility.
multiappearance telephone	A telephone equipped with several call-appearance buttons for the same extension, allowing the user to handle more than one call on that same extension at the same time.
network region	A group of IP endpoints and Communication Manager IP interfaces that are interconnected by an IP network.
Network Specific Facility (NSF)	An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call Service Selection is used to access a public-network service.
node	A network element that connects more than one link and routes voice or data from one link to another. Nodes are either tandem or terminal. Tandem nodes receive and pass signals. Terminal nodes originate a transmission path or terminate a transmission path. A node is also known as a switching system.
non switch-classified outbound calls	Proactive Contact outbound calls that are automatically launched by Communication Manager.
Non-Facility Associated Signaling (NFAS)	A method that allows multiple T1 or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel

	on each facility), and the other facilities that share the D-channels are configured without D-channels.
pickup group	A group of individuals authorized to answer any call directed to an extension within the group.
poll suppression	An automatic feature of BSR that significantly reduces the number of status polls executed. When a remote location cannot be the best resource at a given moment in time, the intelligent polling feature temporarily suppresses polls to that location. See status poll on page 460.
primary extension	A main extension associated with the physical telephone or data terminal.
principal	A terminal that has the primary extension bridged on other terminals.
principal (user)	A person to whom a telephone is assigned and who has message-center coverage.
private network	A network used exclusively for the telecommunications needs of a particular customer.
Processor Port Network (PPN)	A port network (PN) controlled by a switch-processing element that is directly connected to that PN's TDM bus and LAN bus.
public network	A network that can be openly accessed by all customers for local and long-distance calling.
queue	An ordered sequence of calls waiting to be processed.
redirection criteria	Information administered for each telephone's coverage path that determines when an incoming call is redirected to coverage.
Redirection on No Answer	An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the agent.
reorder tone	A tone to signal that one of the facilities such as a trunk or a digit transmitter, was not available.
Service Level Maximizer (SLM)	An agent selection strategy that ensures that a defined service level of X% of calls are answered in Y seconds. When SLM is active, the software verifies that inbound calls are matched with agents in a way that makes sure that the administered service level is met. SLM is an optional Call Vectoring feature that is used with Expert Agent Selection (EAS), and without Business Advocate.
simulated bridged appearance	A feature with which a terminal user can bridge onto a call answered by another user. A simulated bridged appearance is a temporary bridged appearance.

SIP	Session Initiation Protocol (SIP) is an application-layer control signaling protocol for creating, modifying, and terminating sessions with more than one participant using http like text messages.
split (agent) status report	A report that provides real-time status and measurement data for internally-measured agents and the split to which agents are assigned.
split condition	A condition whereby a caller is temporarily separated from a connection with an attendant. A split condition automatically occurs when the attendant, active on a call, presses the start button.
split number	An identification of the split to Communication Manager and to BCMS.
split report	A report that provides historical traffic information for internally measured splits.
SSC	Single Step Conference (SSC) is a client-side IP call recording method that uses: • The AES DMCC service to provide the required media control by registering standalone recording devices. • The AES TSAPI service to provide call information, call monitoring, and third-party call control functionality.
staffed	An indication that an agent position is logged in. A staffed agent functions in one of four work modes: auto-in, manual-in, ACW, or Aux.
status lamp	A green light that shows the status of a call appearance or a feature button by the state of the light (lit, flashing, fluttering, broken flutter, or unlit).
status poll	A call that Communication Manager makes to gain status data from a remote place in a multisite BSR application plan.
stroke counts	A method used by ACD agents to record up to nine customer-defined events per call when a reporting adjunct is active.
switch-classified outbound calls	Outbound calls placed by the Proactive Contact dialer and connected to the agents.
system printer	An optional printer that used to print scheduled reports using the report scheduler.
system report	A report that provides historical traffic information for internally-measured splits.
system-status report	A report that provides real-time status information for internally-measured splits.

trunk	A dedicated telecommunications channel between two communications systems or Central Offices (COs).
trunk allocation	The manner in which trunks are selected to form wideband channels.
trunk group	An arrangement of communication channels that carry multiple calls for the same phone number.
UCD-LOA	Uniform Call Distribution-Least Occupied Agent (UCD-LOA) is an agent selection method for delivery of calls under agent surplus conditions. With UCD-LOA implemented, calls are delivered to the available, that is, idle agent with the lowest percentage of work time since login.
UCD-MIA	Uniform Call Distribution-Most Idle Agent (UCD-MIA) is an agent selection method for delivery of calls under agent surplus conditions. With UCD-MIA implemented, calls are delivered to the available agent who has been idle the longest since the last ACD call that the agent received.
UDP	Uniform Dial Plan (UDP) is a feature that allows a unique number assignment for each terminal in multi-switch configurations, such as a Distributed Communications System (DCS) or main-satellite-tributary system.
VDN	Vector Directory Number (VDN) is an extension number that directs calls to a vector. VDNs can represent a call type or a service category, such as Billing or Customer Service.
vector-controlled split	A hunt group that you can gain access to only by dialing a VDN extension.
work mode	A function that an agent performs during the work shift. ACD work modes include AUX work, auto-in, manual-in, and ACW.

Index

Numerics

96X1 SIP phones ... 436
- active VDN ... 436
- agent information line ... 437
- agent login ... 437
- agent logout ... 437
- agent status line ... 437
- customizing agent desktop ... 437
- feature buttons ... 437
- feature support ... 436
- handling denial tones ... 437
- managing calls ... 437
- softkeys ... 437
- UUI ... 436
- viewing calls ... 437

A

AAS interactions ... 62
abandoned call search ... 19, 35
abandoned calls ... 19
abbreviated dialing (AD) buttons ... 27, 30
access codes ... 218
accessing port matrix ... 444
ACD ... 15, 62
- field options
  - considerations ... 36
- methods ... 64

ACD splits ... 20
ACD-related activities ... 41
active VDN ... 436
ACW
- enhanced callr-info ... 50

ACW interactions ... 131
add skill FAC ... 30
adding skills ... 36, 37
adjunct routing ... 256
- considerations ... 257

adjunct switch application interface (ASAI) ... 405
adjunct-controlled skills ... 121
adjust-by ... 293
administering splits ... 20
administering the location number ... 161
advanced multisite routing ... 318
advanced vector routing features
- ASA ... 268
- EWT ... 268
- VDN calls ... 268

after call work (ACW) ... 28
after call work (ACW) mode ... 41
agent AUX work
- forced
- forced *(continued)*
  - forced *(continued)*
    - location ... 127
    - skill ... 127

agent availability query access code
- dial tone ... 68

agent call handling ... 26
agent greeting ... 436
agent login ... 27
agent login ID capabilities ... 98
agent logout ... 27
- forced
  - clock time ... 123
  - location ... 127
  - remotely ... 129
  - skill ... 127

- using VDN ... 201

agent mobility
- features not supported ... 59

agent notifications ... 156
agent occupancy ... 166, 214
agent selection
- adjust-by ... 293

agent skills, tracking ... 47
agent state
- dial tone ... 68

agent states ... 27
agent surplus conditions ... 162
agent surplus situation ... 292
agent TN ... 164
agent work mode ... 41
agent work modes ... 41
agent-skill partitions, interactions ... 165
agent-skill partitions, matching ... 164
allocating dynamic queue slots ... 17
alternate selection ... 294
announcement ... 92
- queuing ... 19
- rules ... 67

announcement for calls in queue ... 18
announcement rules ... 18
answer supervision ... 19
ASA ... 275
- rolling versus interval ... 275
- split calculation ... 276
- VDN calculation ... 276

ASAI interactions ... 120
ASAI message
- contents of ... 259

ASAI messages ... 405
ASAI-requested digit collection ... 84
ASL algorithm
- administering ... 213

AT&T In-Band Transfer and Connect 376
AT&T In-Band Transfer Connect 377, 378
attendant queue 279
attendant vectoring 278
goto step command 365, 396
goto vector command 396
night service 283
restrictions 284
security issues 442
VDNs 279
Attendant Vectoring
announcement Command 363
busy Command 364
disconnect Command 364
Hunt Group Queue 283
route-to number Command 364
stop Command 365
audio difficulty 30
auto reserve
release agents 212
auto reserve agents 211
rules 211
auto-answer
block hang-up 48
auto-in 28
auto-in agents 167
auto-in button 184
auto-in work mode 41
automatic call distribution 62
automatic call distribution (ACD) 15
automatic-in processing 15
AUX work mode 28, 44
AUX work reason code 179
auxiliary (AUX) work 28
auxiliary (AUX) work mode 41
auxiliary work mode
reason codes 178
Avaya support website 447

B

barrier codes 226
basic mode 232
BCMS
interactions 69
reports
BCMS Split Report 433
historical 68
real-time 68
VDN Real-Time Report 433
VDN Summary Report 433
BCMS interactions 425
BCMS reports 188
best service routing 285, 292
available agent 311
considerations 291
interactions 334
writing vectors *(continued)*
local treatment
considerations 291
multisite
poor network response time 316
status poll vector
multisite 315
writing vectors
tips 293
best service routing (BSR) 294
Best Service Routing (BSR)
determining the best resource 291
interactions
Information Forwarding 145
best skill, searching 294
block hang-up
auto-answer agents 48
BSR 285, 290
benefits 286
call interflow with SIP 309
multisite applications 304
network requirements 290
terminology 287
BSR call flow 377
BSR example 297
interflow vector 315
limited trunking 312
primary vector on origin 308
slow networks 316
status poll vector 308, 317
BSR interflow vector
interflow vector
BSR 309
BSR local treatment 328
BSR primary vector
BSR 308
BSR tie strategy 294
Business Advocate 70, 89
Business Advocate licensing 45
button modules 436
button-based deskphones 436

C

call control 120
call coverage 21, 157
call detail recording 370
call distribution methods 23
call flows
answered and abandoned calls 426
busies and disconnects 427
defining and interpreting 426
VDN inflows and outflows 427
vector inflows and outflows 427
call forwarding 21, 157
call handling conditions 161
call origin 406

call processing
split queue ... 17
call processing types
automatic-in ... 15
call prompting ... 70, 71
considerations ... 72
functions
creating service observing vectors ... 79
passing digits to an adjunct ... 79
using digits to collect branching information ... 76
using digits to select options ... 77
interactions ... 85
with VRI ... 70
Call Prompting
interactions
Inbound Call Management (ICM) ... 136
call redirection ... 20
limitations ... 385
call redirection during ROOF ... 199
call surplus condition ... 162
call transfer
IVR system ... 240
Call transfer to VDNs ... 419
call vectoring
description ... 256
interactions ... 420
methods ... 378
removing incorrect digits ... 73
call vectoring interactions ... 425
call work code (CWC) ... 52
call-answering modes ... 27
calling permissions ... 226
callmaster phones ... 254
callr-info
clearing ... 79
calls warning threshold ... 154
canceling
forced logout ... 124
CDR ... 370
central office (CO) ... 14
centralized attendant service ... 14
changing agent skills ... 37
changing skills ... 36
changing vector types ... 370
class of restriction ... 92
class of restriction (COR ... 165
forced agent logout ... 125
CMS ... 122
reports
Split Summary Report ... 433
VDN Report ... 433
Vector Report ... 433
CMS database items ... 167
CMS interactions ... 62, 178, 425
CMS notifications during ROIF ... 184
CMS notifications during ROOF ... 199
CMS reports ... 188
CMS tracking records ... 203
CMS tracking, stroke counts ... 55
coach ... 216
coach mode ... 216
coaching ... 227
collect digits command ... 71
collection
delete ... 445
edit name ... 445
generating PDF ... 445
sharing content ... 445
Combining VDN and ASA
routing example ... 277
communication server attendant ... 14
communication servers ... 14
configuring
Ericsson AXE-10 ... 386
Configuring — communication server before IVR system ... 239
consider split/location adjust-by x ... 293
considerations
call prompting ... 72
Console Parameters
screen example ... 282
console permissions ... 37
content
publishing PDF output ... 445
searching ... 445
sharing ... 445
sort by last updated ... 445
watching for updates ... 445
counted-calls
conditional test when routing calls ... 278
Criteria for choosing algorithms ... 213
CWC
enhanced callr-info ... 49, 50
CWC capacity ... 52

D

DAC
interactions ... 93
data privacy ... 227
data restrictions ... 227
database items for MAO ... 167
deactivating threshold ... 154
default codes ... 179
default reason codes ... 156
defining location numbers ... 159
dial-ahead digits ... 81, 84
DID processing
example ... 16
digits
ASAI provided dial-ahead digits ... 84
entering
dial-ahead digits ... 75
removing

removing *(continued)*
removing *(continued)*
incorrect digit strings 73
direct agent calling 90
ICM 136
direct agent calling (DAC)
Inbound Call Management (ICM) 136
direct agent calls (DAC) 165
direct department calling 23
direct inward dialing 370
direct inward dialing (DID) 16
disconnect tones 38
display buttons
normal 33
Displaying VDN names
vector-initiated DACs 416
DNIS/ISDN called party 97
documentation center 445
finding content 445
navigation 445
documentation portal 445
finding content 445
navigation 445

E

EAD-LOA 25, 26
EAD-MIA 25
EAS administration from CMS 435
EAS benefits 88
EAS considerations 113
EAS interactions 114
EAS terminology 86
EAS tracking
agents and skills 434
VDN skill preferences 435
EAS-PHD 25, 89
EC500
agent mobility 58
enhanced callr-info 49
96X1 H.323 phones 50
96X1 SIP phones 50
interactions 50
softphones 50
entering
dial-ahead digits 75
establishing location numbers 160
ETSI ECT operations 386
evaluating
split performance 432
evaluating MAO 167
Evaluating target service level compliance 213
event count 30
event counts 55
EWT
split priority level 274, 275
example
example *(continued)*
DID processing 16
FIFO processing with LAI 357
Example
vector for adjunct routing-implemented path replacement 265
Example polling vector for the remote communication server 332
example vector
passing digits to an adjunct 79
remote access service observing vector 80
service observing vector 80, 81
using digits to collect branching information 76
using digits to select options 77
exclusion, service observing 219
exclusion, SSC 220
expected wait time
calculation 270
for individual calls 271
expert agent selection 85, 87, 434
considerations 113
logical agent 85
tracking
non-ACD calls 435
Expert Agent Selection (EAS)
forms 190
EXT-IN 26
EXT-OUT 26
extensions 14

F

facility restriction level (FRL) 165
facility restriction levels (FRLs) 226
failures
VRU 186
feature access code (FAC) 30, 164
finding content on documentation center 445
finding port matrix 444
forced agent logout from ACW 131
forced entry of work codes 52
forced logout override 124
forced reason code 179

G

General indications 222
generic VDN for redirected calls 189
ghost calls 19
greatest need call distribution 89

H

handling wait time 318
holiday table commands
example 341

holiday vectoring
administering 338
description 337
host database lookup 97
How UCID works 235
hunt group 15
hunt group night destination 279
Hunt Group Night Service 22
Hunt Group Queue 283

I

ICI tones 254
ICM 136
ICM applications 132
ICM integration
speech processing adjuncts 134
identifying remote servers 305
trunk states 30
inbound call management
considerations 136
description 132
Inbound Call Management (ICM) 136
interactions
Call Prompting 136
direct agent calling (DAC) 136
Priority Calling 136
incoming call identification (ICI) 254
indicators
service observing 222
information forwarding 136
benefits 137
byte lengths for UUI user data 142
collected digits with interflowed call 145
considerations 144
in-VDN time in interflowed call 146
network requirements 143, 144
support for BSR and LAI 145
Information Forwarding 139
ASAI UUI IE shared format conversion 138
interactions
Best Service Routing 145
information transport over ISDN 146
information transport over SIP 147
InSite Knowledge Base 447
intelligent polling 322
Interactions with other Communication Manager features 418
intercept tones 41
interflow 20, 21, 157
interflow vector 315
interflow-qpos 353
interruptible aux 88, 153, 156
interruptible aux thresholds 154
intraflow 20, 21, 157
calls
increase priority 21
Intraflow and Interflow 145
Intraflow and Interflow *(continued)*
ACD call redirection options 145
interactions 158
intraflow assignment 21
IVR system transfers a call 240

L

LAI backward compatibility 147
LAI command set 349
LAI considerations 345
LAI requirements 344
launching non switch-classified
outbound calls 176
LDN and trunk night destination 279
listen only mode 216
listen/talk mode 216
local call treatment 328
local preference distribution
interactions 162
Local treatment system requirements 329
location ID number 220
location number 159–161, 305
location numbers 159, 160
location numbers, defining 159
location preference distribution 159
logged-in agents 45
logical agent 105
login access code, assigning 27
login ID length, assigning 27
look-ahead interflow 344
enhanced
multisite applications 362
tandem switch configuration
far end switch operation 359
two switch configuration 352
low FRL value 165

M

manual exclusion 227
manual-in 28
manual-in agents 166
manual-in button 184
manual-in work mode 41
MAO 207
MAO interactions 166, 167
MAO reason code 166, 167
MAO threshold 166
MAO, database items 167
MAO, evaluating 167
MAO, overriding 167
matching agent-skill partitions 164
maximum agent occupancy 207
maximum agent occupancy (MAO) 166
MCH
example 170

MCH interactions 172
MCH settings
one forced 169
one per skill 170
measured trunks 203
meet-me conference 363
security issues 441
Meet-me Conference 363, 365
Meet-me Conference capacity issues 368
meet-me conference interactions 369
meet-me conferencing 364
messaging
ASAI
contents of 259
messaging system 122
monitoring locations 220
monitoring VDNs 219
most idle agents 23, 26
multiple call handling
applications 168
considerations 171
multiple call handling (MCH) 30
multiple call handling interactions 172
multiple locations feature 159
multiple monitors 121
multiple observers 227
multiple splits 27
multisite BSR 302
user adjustments 311
multisite BSR application 315
multisite BSR example 312
Music source 282
My Docs 445

N

NCR 371, 385
considerations 385
interactions 387
NCR options 371
NCR protocols unsupported 371
NCT-type 371
network ADR 361
network call deflection (NCD) 373
network call redirection
B-channels 374
SIP
interactions 391
Network Call Redirection
implementation
station call transfer or conference 383
Information Forwarding support 386
route-to command 381, 382
network call transfer (NCT) 371
next call listen only mode 216
next-call mode 232
night service 22, 283
night service button 33
No Talk FAC 217
no talk mode 216
no-talk mode 232
non switch-classified calls 174
non-EAS agent staffing 27
NQC lamp 33

O

observing locations 220
observing VDNs 219
operational differences
working on deskphone 59
working on mobile station 59
OQT lamp 33
switch-classified 175
outbound calling
acquiring agents 174
overriding MAO 167

P

passcodes for service observing 226
path replacement 291, 333
percent allocation 70
percent allocation distribution (PAD) 70
percentage allocation routing 392
considerations 392
rules 394
performance
effects of ASAI link failure 261
evaluating
for split 432
Phantom call administration 267
phantom calls 265
polling guidelines 322
port matrix 444
predicted wait time (PWT) 70
Preference Handling Distribution 105
preventing unauthorized users access 439
Priority Calling
interactions
Inbound Call Management (ICM) 136
Proactive Contact
description 173
Proactive Contact outbound calling 173
protocol
non-QSIG 149
protocols
NCT 372
PSTN support for NCR 371

Q

QSIG

QSIG *(continued)*
trunk group ... 149
QSIG path replacement ... 333
queue
threshold ... 21
queue status indications ... 177
Queue Status Indications ... 178
interactions ... 178
Queue status lamps ... 33
queuing
priority ... 67

R

reason code
interactions ... 181
reason code types ... 179
reason codes ... 178
considerations ... 180
ROIF ... 183
record annct button ... 33
redirect VDN for ROOF ... 199
redirected calls to attendant VDN ... 283
redirection
OPTIM failure ... 198
redirection criteria ... 157, 158
redirection on IP failure (ROIF) ... 182, 184
redirection on no answer
considerations ... 190
interactions ... 191
Redirection on No Answer (RONA)
returning AAS agents to service ... 190
redirection on OPTIM failure (ROOF) ... 199
release agents
auto reserve ... 212
release button ... 56
remote access ... 226
remote feature activation ... 220
remote logout of agent ... 200
remote logout of agent interactions ... 202
remote logout using a VDN ... 200
remote service observing ... 218
remove skill FAC ... 30
removing incorrect digits strings ... 73
removing skills ... 36, 37
reporting adjuncts
CMS ... 202
considerations ... 203
reports
exception ... 15
historical ... 15
real-time ... 15
requested reason code ... 179
reserve agents ... 210
reserve level ... 153
ringback to callers ... 54
ROIF
ROIF *(continued)*
agent work mode ... 184
auto-in or manual-in buttons ... 184
call priority in queue ... 184
CMS notifications ... 184
interactions ... 184
redirect VDN ... 184
Rolling Average Speed of Answer ... 275
RONA
considerations ... 190
description ... 185
interactions ... 188
routing sequences ... 187
voice response unit applications ... 186
ROOF ... 198
agent work mode ... 199
call priority in queue ... 199
CMS notifications ... 199
interactions ... 199
redirect VDN ... 199
route calls by skill level ... 89
route-to command
~r vector step ... 381, 382
Network Call Redirection ... 381, 382
route-to number commands
using ... 418
route-to vector step ... 165
routing
greatest need ... 90
skill level ... 90
routing calls ... 109, 165
routing commands ... 364
Rules for specifying VDNs ... 276

S

searching for content ... 445
security
main type of problem ... 439
methods for preventing remote access abuse ... 439
preventing unauthorized users access ... 439
with remote access ... 439
with service observing ... 441
with vector initiated service observing ... 441
security alerts
Service Observing ... 225
server requirements ... 290
service hours table
administering ... 395
service hours table routing
goto processing ... 395
overview ... 395
scenario ... 396
Service Hours Table Routing
considerations ... 395
time adjustments ... 395
service level maximizer ... 207

service level maximizer *(continued)*
auto reserve agents ... 210
requirements ... 207
service level supervisor (SLS) ... 70
service level target threshold ... 154
service objective (SO) ... 70
service observing ... 80, 219, 233, 436
agent login IDs ... 217
button restrictions ... 222
description ... 216
ineligibility ... 227
interactions ... 230
monitoring modes ... 216
multiple observers ... 228, 233
remotely ... 218
trunk calls ... 228
VDN ... 219
Service Observing
conferenced calls ... 228
observability ... 227
security alerts
general security ... 225
VDN call security ... 226
skills ... 227
splits ... 227
transferred calls ... 229
service observing FAC ... 217
Service Observing feature key ... 216
Service Observing for AES CTI users ... 229
service observing for SIP phones
description ... 232
service observing permissions ... 226
service observing security ... 226
service observing with multiple observers ... 234
sharing content ... 445
simultaneous call recording ... 217
single-step conference ... 267
singlesite BSR ... 294
example ... 297
planning ... 295
SIP agent login ... 436
SIP agent logout ... 436
SIP NCR and information forwarding ... 389
SIP Network Call Redirection protocols ... 389
SIP phone buttons
after-call ... 436
agnt-login ... 436
agnt-logout ... 436
aut-msg-wt ... 436
auto-in ... 436
aux-work ... 436
call-appr ... 436
manual-in ... 436
q-calls ... 436
stroke-cnt ... 436
uui-info ... 436
vu-display ... 436
SIP phone buttons *(continued)*
work-code ... 436
SIP phones
service observing ... 216
skill
agent ... 104
skill change alert ... 30
skill TN ... 164
skill-based call distribution ... 88
skills
Service Observing ... 227
skip announcements ... 81
SLM
interactions ... 215
low occupancy ... 214
reports ... 213
SLM agent selection ... 207
SLM algorithms ... 212
SLM benefits ... 210
SLM call selection ... 208
SLM. ... 207
SO ... 216
sort documents by last updated ... 445
speech processing adjuncts ... 134
speech-processing adjuncts ... 122
split
queuing ... 64
split queue
types
normal ... 17
priority ... 17
splits
Service Observing ... 227
SSC ... 220
SSC work
Call Vectoring ... 267
staffed agents ... 47
station buttons ... 30
stations
removing ... 371
status polling ... 320
stroke count ... 30
stroke count button ... 55
stroke counts ... 55
supervisor assistance ... 30
supervisor phone buttons ... 33
support ... 447
switch-classified calls ... 174
switches to poll ... 322

T

tandem switch
far end operation ... 359
tenant night destination ... 279
tenant number (TN) ... 164
tenant partitioning ... 164

time slot usage, reducing 217
time warning threshold 154
TN assignments 284
touch-based deskphones 436
tracking 434
  agents and skills 434
  for non-ACD calls 435
  VDN skill preferences 435
tracking agent skills 47
tracking stroke counts 55
trunk group
  status 291
trunk group incoming destination 279
trunk group night service 22
trunk groups 14
trunks 14

U

UCD-LOA 25, 26
UCD-MIA 23, 25
UCID 234, 235
UCID requirements 236
UCID tracking 236
UCIDs 235
Universal Call ID
  interactions 242
Universal Call ID (UCID) 241
  creating 235
    tracking .....
      complex conference 238
      incoming trunk calls 237
      outgoing trunk calls 237
      simple transfer or conference 237
      station-to-station calls 237
  transmitting 235
unmeasured facilities 203
UUI 259
    data forwarding
      SIP 390
  enhanced callr-info 50

V

VDN
  generic 189
  multiple observers 228
    override
      application 404
  parameters 404
  service observing 219
  skills 101
VDN Calls 277
  how call counts are calculated 277
VDN for Route-to DAC
  Route-to DAC feature operation 417
VDN in a coverage path 399
VDN in a coverage path *(continued)*
  considerations 399
  interactions 399
VDN observing 220
VDN observing access codes 220
VDN of Origin Announcement 242
  interactions 245
VDN Override
  for ASAI messages 405
VDN Override for ASAI messages 405, 406
VDN Override for call-pickup 406
VDN Override interactions 406
VDN return destination 406
VDN skills 302
VDN time zone offset 247
VDN Time Zone Offset 247
VDN variables 408
VDN Variables 408
vdn-by-loc mode 232
VDN, observing 226
vector
    example
      passing digits to an adjunct 79
      remote access service observing vector . 80
      service observing vector 80, 81
        using digits
          to collect branching
            information 76
          to select options 77
vector command
  advanced vector routing 268
vector directory number (VDN) 165
vector example
  dial-ahead digits 82
vector processing
  for holidays 342
vector routing table 76
VICP 399
  interactions 399
videos 446
view callr-info 49
variables in vectors 398
VOA 242
  considerations 244
VOA interactions 245
voice response integration 442
  interactions 411
voice response integration (VRI) 409
VRI 409
VuStats 248
  button light flashes 250
  considerations 251
  data types 249
  enhanced callr-info 49, 50
  how information appears 250
  information displayed 248
  statistical periods 250

VuStats *(continued)*
thresholds 250
when information updates 251
vustats button 33
VuStats interactions 252

W

wait-time command
meet-me conference 364
watch list 445
wildcards 76
work mode change during ROOF 199

Z

zip tone 254