



Avaya Call Center

Release 5.0

Automatic Call Distribution (ACD) Guide

07-602568
Release 5.0
January 2008

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Preface

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- [Audience](#) on page 15
- [Reasons for reissue](#) on page 16
- [Related documents](#) on page 16
- [Availability](#) on page 17

Purpose

This document explains the features that comprise the Call Center Automatic Call Distribution (ACD). It provides an introduction to each call center feature and presents required forms for administration, detailed descriptions, considerations, and interactions between call center features. This document provides an overall reference for planning, operating, and administering your ACD Call Center.

Audience

This document is intended for communication server system administrators and managers, end-users interested in information about specific features, and Avaya support personnel responsible for planning, designing, configuring, selling, and supporting the system.

Reasons for reissue

The following features have been added to this document to support Release 5.0:

- NCR and UUI over SIP trunks
- Avaya Agent Deskphone 16CC

Related documents

You might find the following Avaya documentation useful. This section includes the following topics:

- [Other Call Center documents](#) on page 16
- [Associated application documentation](#) on page 16

Other Call Center documents

These additional documents are issued for Avaya Call Center applications:

- *What's New for Avaya Call Center 5.0* - Provides a high-level overview of the new features available for the most-current release.
- *Avaya Call Center Call Vectoring and EAS Guide* - Provides information on how to write, use, and troubleshoot vectors, which are command sequences that process telephone calls in an Automatic Call Distribution (ACD) environment.
- *Avaya Communication Manager Call Center Software - Basic Call Management System (BCMS) Operations* - Provides information on the use of the BCMS feature for ACD reporting.
- *Avaya Business Advocate User Guide* - Provides a general understanding of how Avaya Business Advocate can be used for call and agent selection.
- *Avaya IQ Documentation DVD* - Provides information about Avaya's software-only reporting solution for its contact center portfolio.

Associated application documentation

The most recent application documentation for Avaya Communication Manager and Avaya Call Management System is available on the Avaya Support web site: <http://support.avaya.com>.

Availability

Copies of Avaya Call Center documentation are available from The Avaya Support Web site, <http://support.avaya.com>.

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ACD basics

This section includes the following topics:

- [Communication server features](#) on page 19
- [What the ACD does](#) on page 23
- [Split queues](#) on page 24
- [Distributing and handling calls](#) on page 32
- [ACD and call management systems - BCMS and Avaya CMS](#) on page 44

Communication server features

This section includes the following topics:

- [What a communication server does](#) on page 19
- [Trunks, trunk groups, and extensions](#) on page 20
- [Automatic-in processing](#) on page 20
- [Communication server attendant](#) on page 21
- [Direct inward dialing processing](#) on page 21
- [DID processing example](#) on page 22

What a communication server does

A communication server is an electronic device that processes incoming, outgoing, and internal calls and connects them to the proper destinations. The telephone company communication server in your local area is called a Central Office (CO). A communication server owned by a company or organization processes incoming, outgoing, and internal calls. Throughout this section, the term communication server is used to refer to a company or organization's communication server.

Trunks, trunk groups, and extensions

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

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

Since a trunk can carry only one call at a time, trunk groups are usually created. A trunk group is a group of trunks that are assigned to the same digits. With a trunk group, the CO receives the digits of a dialed phone number and checks the trunk group assigned to that number to see if any of the trunks are available. The CO then seizes an available trunk. As many simultaneous calls can be made over a trunk group as there are trunks in that trunk group. 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's communication server location) to the communication server, the communication server then connects the calls to their proper destinations within the company.

The communication servers previously listed, in addition to connecting incoming calls to the proper destinations, are also like private COs for company employees. Employee phones are connected to a communication server by telephone lines called extensions. Extensions are then assigned numbers, and these numbers become the employee phone numbers for internal (intra-company) calls.

Automatic-in processing

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

Communication server attendant

Incoming calls can also go to a communication server attendant. A communication server attendant is a person 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. Often, a communication server will have more than one attendant, and all of the communication server's attendants will answer calls directed to the attendant queue, which holds calls until an attendant is available. The attendant queue receives internal calls made from employee extensions, and also receives incoming calls through DID processing and automatic-in processing. Attendant call handling varies, depending on the company's needs. However, if the attendant has an automatic-in number, it will normally be the number published in the phone book, and the DID number will most likely be used by off-site employees who know only the attendant's extension number.

Centralized Attendant Service (CAS) is a communication server feature that enables attendants to be consolidated 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.

Direct inward dialing processing

With Direct Inward Dialing (DID) processing, incoming trunks do not connect the CO directly to an employee's phone; instead, the incoming trunks are pooled by the communication server, and this pool of trunks is then shared by employee phones. Extension numbers may serve as the final digits of employee phone numbers for incoming calls. That is the CO may assign a 2-3- or 4- digit prefix to a trunk group. Then, when a 7-digit employee phone number is dialed, the call is processed as follows:

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

DID processing example

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

The steps in completing calls to Employees A and B might be 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 will not accept any more calls, so another call beginning with the digits 538 will seize yet another trunk in the trunk group.

What the ACD does

This section includes the following topics:

- [About the ACD](#) on page 23
- [Things to know before you start](#) on page 23
- [Automatic-in processing of ACD calls](#) on page 24

About the ACD

Automatic Call Distribution (ACD) is a communication server software feature that processes high-volume incoming, outgoing, and internal calls and distributes them to groups of extensions called hunt groups or splits. The communication server also sends information about the operation of the ACD to the CMS which stores and formats the data and produces real-time and historical reports on ACD activity.

ACD is used by a call center to route incoming calls to specifically assigned splits/skills and agents. ACD allows a system administrator to create an efficient call management environment. This administrator can add or remove splits/skills from the system, add or remove announcements, add or remove agents, add trunk groups and route calls to the appropriate splits/skills. The administrator can also specify ACD measurement criteria and use an optional CMS package to provide reports on ACD efficiency.

Things to know before you start

A voice response port or a person who answers ACD calls is called an agent. Companies that operate high-volume call-answering centers, for example, a catalogue sales center, a reservations center, or a customer service center, use the ACD feature to process incoming calls and distribute them to agents. In addition to agents, each ACD split can be assigned a split supervisor. The split supervisor uses various communication server and CMS features to monitor split and agent performance and to provide assistance if necessary. Maintaining trunks from the CO to the communication server and hiring agents to answer calls costs money. However, if customers who call to purchase goods or services have difficulty reaching an agent and, therefore, stop trying to get through, the call center loses revenue. Call center management needs, therefore, to determine how many trunks and agents are necessary to minimize costs and maximize the ability of customers to purchase goods or services. Management can then set up and maintain the ACD accordingly.

Automatic-in processing of ACD calls

Through communication server administration, each automatic-in trunk group is assigned to an ACD split. All calls that come in on an automatic-in trunk group are directed to the assigned split. Then the ACD software distributes the calls to the agent extensions assigned to the split according to the assigned call distribution method (described later).

DID processing of ACD calls

The communication server enables you to dial directly to various extensions such as a VDN, a hunt group, an agent, or a login ID. Each extension can be assigned to a split as a DID extension.

For DID processing, trunk groups are not assigned to the split. The creation of associated extensions is sufficient to send calls arriving over DID trunk groups to the appropriate split. Each split can receive incoming calls through DID processing, automatic-in processing, or both. Automatic-in trunk groups carry calls only to the split, whereas DID trunk groups carry calls to any extension identified in the communication server software, not just a split.

Split queues

This section defines a split queue and explains how to set up call processing to a split. This section includes the following topics:

- [Split queue call processing](#) on page 24
- [Announcements for calls in a split queue](#) on page 26
- [Answer supervision and abandoned calls](#) on page 28
- [Intraflow and interflow](#) on page 28
- [Night Service](#) on page 31

Split queue call processing

A split queue is a holding area for calls waiting to be answered, usually in the order in which they were received. When a call is put into queue, the caller may hear one or more delay announcements, music, and/or silence, depending on the treatment assigned for the split. (Treatment of calls in queue is assigned through communication server administration.)

Things to know before you start

Calls enter the queue at the bottom and move toward the top or head of the queue. After a call reaches the head of the queue, it connects to the next available agent.

For communication servers with the Call Vectoring feature, all call treatment including routing, queuing, announcements, and music is specified by call vectors. When a call arrives at a split, the ACD software checks to see if an agent is available to handle the call. If an agent is not available, or busy, the call enters the split's queue.

Calls queue only if no agents are available, a queue is assigned to the split, and the queue is not full. If the queue is full, the caller hears a busy tone or the call goes to coverage. If the split is vector controlled, then this step will fail. Furthermore, if no agents are logged into the split or if all agents are in AUX work mode (described later), calls do not queue.

Priority and normal split queues

Each split can have two queues: a normal queue and a priority queue. A split always has a normal queue and can also be assigned a priority queue. The ACD distributes all calls in the priority queue before it distributes any calls in the normal queue. Therefore, the priority queue, if one exists, must be empty before the ACD distributes calls in the normal queue.

Priority queuing may be assigned in the Class of Restriction (COR) associated with the split extension number. A split may also be assigned Priority Queuing on Intraflow, which means that calls to that split, if rerouted to another local split, will enter the destination split's priority queue.

Dynamic queue slot allocation

Beginning with Communication Manager 2.1, Communication Manager dynamically allocates queue slots to hunt groups or skills. You no longer have to estimate and administer queue slots. The system dynamically allocates hunt group or skill queue slots on an as needed basis. When a queue slot resource is needed, it is extracted from a common pool. When the call is removed from queue, the queue slot resource is relinquished and returned to the common pool. There are enough queue slots to allow all possible calls to queue.

Dynamic queue slot allocation has the following advantages:

- Reduced administration
- Expanded capacities, such as increased skill availability in your 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.

Use the **Queue Limit** field to specify the maximum number of calls that can be queued to the hunt group. For more information about this field, see [Hunt Group screen field descriptions](#) on page 358.

Announcements for calls in a split queue

When a call enters a split queue, the caller hears ringing until the call is connected to an agent or an announcement. Depending on the treatment assigned to a split, the caller may hear one or two announcements, music, or silence. An announcement is a recorded message that provides information such as the destination the call has reached or a company's business hours, or it tries to persuade the caller to stay on the line.

Things to know before you start

Announcements and delay time are assigned to splits through communication server administration. Delay time is the amount of time a call will wait 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 plays, the caller hears music or silence until the second announcement plays or the call connects to an agent. The type of caller feedback (music or silence) is also assigned to a split through communication server administration.

For communication servers with the Call Vectoring feature, announcement capabilities are more flexible than those described in this section. See [Call Vectoring](#) on page 137.

Related features

The communication server supports both internal and external announcement devices.

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

Rules for announcements

The announcement is played from beginning to end unless an agent becomes available. In such a case, the announcement is interrupted and (if manual answering operation is assigned to the agent, or if calls are delivered to the agent on a manual answering basis) ringback is provided. If the call is queued, the call remains as such while the announcement is played. Any feedback that is provided before an announcement (for example, a wait with music or ringback) continues until the announcement is played.

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, connects to an announcement, or enters the announcement queue. The following rules apply to announcements without vectoring implemented:

- Calls directly entering a split queue always receive a forced first announcement if assigned. The caller also hears first and second delay announcements if administered and 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's queue is full, the call retries the announcement step for an indefinite period of time 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

External and internal announcement units are available. The number of calls that can be queued to an announcement depends on the size of the communication server you have. The capacity tables in the System Description have details for each communication server model. Queuing for internal announcements is quite different. Internal announcements are delivered by a multi-port/channel announcement board, and a call receives an announcement only when it connects to one of the announcement ports or channels. Therefore, all calls wait in a single queue to access a channel on the announcement board regardless of the split announcement they are waiting to receive. 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 and abandoned calls

Answer supervision is a signal sent by the communication server to the serving Central Office (CO). This signal tells the CO that an incoming call has been answered and that the CO should begin tracking toll charges for the call (if they apply). Answer supervision is sent immediately before a call connects to an agent's telephone, to music, or to an 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. A call can abandon while in queue or while ringing at an agent position. Abandoned calls represent lost sales or lost good will. Adequate split staffing and effective use of announcements can reduce the number of abandoned calls. Splits should be staffed so that calls do not have to wait in queue for an unreasonable amount of time, and announcements can be used to persuade the caller to wait until someone answers the call.

Abandoned call search

If answer supervision is sent before a caller abandons, ghost calls can occur. A ghost call is a call that is sent to an agent after the caller hangs up. Ghost calls occur because, after a caller hangs up, some COs wait 2 to 25 seconds before sending a disconnect signal to the communication server. Ghost calls are a problem because they waste agents' time, and they can delay or prevent other calls from connecting to an agent. To minimize this problem, Abandoned Call Search can be assigned to specific trunk groups for the communication server.

With Abandoned Call Search, the communication server checks the incoming trunk before delivering an ACD call to an agent. If the trunk is on-hook at the CO (the call has been abandoned), the communication server releases the trunk and does not deliver the call. If a call is still in progress on the trunk, the communication server delivers the call to an agent.

Intraflow and interflow

Intraflow and interflow allows you to redirect ACD calls to another split or other local or remote destinations. Redirecting calls to a local destination is called intraflow. Redirecting calls to a destination outside the communication server is called interflow.

Things to know before you start

Intraflow and interflow are set up differently on the Generic 3 and newer communication servers. If Call Vectoring is active on the communication server, redirection of calls differs significantly from the following intraflow/interflow descriptions.

Description

As many as three intraflow destinations OR one interflow destination can be established for a split through communication server administration. Intraflow uses the Call Coverage feature to redirect ACD calls to a coverage path that contains one, two, or three of the following internal destinations:

- An extension
- An ACD split (including messaging system and message center splits) or Hunt Group. The term *Hunt Group* refers to groups of extensions that receive distributed calls. The term *split* refers to a hunt group that is measured by CMS.
- An attendant group
- An announcement followed by a forced disconnect

Call Forwarding and ACD splits can be set up to intraflow calls unconditionally.

Interflow destinations are the same as those listed above for intraflow (plus the CAS attendant), except interflow sends calls to destinations outside the communication server.

Setting up splits

If a split is assigned more than one intraflow destination, the communication server tries each destination in the order in which it was assigned. If no destination can accept the call, the communication server leaves the call in the original split's queue. If an interflow destination is specified and activated, the communication server tries only that destination. If the interflow destination cannot accept the call, the caller hears a busy signal. ACD splits can be set up to intraflow calls unconditionally. Unconditional intraflow redirects all calls to the specified destination. Unconditional intraflow is normally used to redirect calls when a split is not staffed.

Splits can also be set up to intraflow calls when one or all of the following criteria are met:

- Don't Answer - Calls redirect if not answered within the assigned Don't Answer Interval (1 to 99 ringing cycles).
- No Agents Staffed or All Agents in AUX Mode - Call redirect if there are no agents staffed or if all agents are in the AUX work mode.

Assigning queue status

If an intraflow destination has a queue, that queue may be assigned an inflow threshold. The inflow threshold, which is established through communication server administration, is the length of time the oldest call in queue has waited. Once the inflow threshold is reached, that queue does not accept intraflowed calls and the communication server tries the next administered destination.

Through communication server administration, a split can be assigned Priority Queuing on Intraflow which allows intraflowed calls to enter the priority queue at the destination split.

Types of calls for a split

The following types of intraflow/interflow can be used for a split:

- Don't Answer Time Interval intraflow (using the Call Coverage feature)
- Busy intraflow (using the Call Coverage feature)
- Unconditional intraflow (using the Call Forwarding-All feature).

When calls are intraflowed using the Call Coverage feature, CMS only reports inflowed and outflowed calls if the call queues to the original split. For example, a call that covers using the busy criterion will not be recorded as in/outflowed since it could not queue to the original split. Calls that queue before covering using the Don't Answer criteria are recorded as in/outflowed calls.

Setting up intraflow/interflow

A split can have either intraflow or interflow active, but not both. However, both conditional (Call Coverage) and unconditional (Call Forwarding) intraflow can be active for a split at the same time. In this case, unconditional intraflow is first invoked for the split's incoming calls. Then, after the communication server forwards a call to the unconditional destination, the communication server uses the conditional intraflow criteria to determine whether to redirect the call to the next destination. Thus, when unconditional and conditional intraflow are used together, the conditional intraflow criteria are applied to the forwarded-to destination, not to the original split.

This combination of unconditional and conditional intraflow allows Dialed Number Identification Service (DNIS) numbers to appear on agent display telephones. In this case, the DNIS number is actually a dummy split extension (that is, the split extension has no assigned agent extensions). The intraflow destinations are the real splits (with staffed agents). With such a configuration, CMS will count incoming calls for the DNIS number (that is redirected using unconditional intraflow to real splits) as outflows. CMS will also count the calls to the destination splits as ACD calls and inflowed calls. And regardless of the split where calls actually connect to agents, the agents will see the DNIS (dummy split) number on their display telephone.

The intraflow criteria and destinations are assigned through communication server administration. Console permissions and the Call Forwarding dial access code are also assigned through communication server administration. Unconditional intraflow or interflow can be activated by entering the Call Forwarding dial access code from a station with console permission, the split's extension, and the interflow or intraflow destination number.

The split supervisor cannot establish conditional intraflow from a telephone. Furthermore, CMS cannot be used to set up or activate intraflow/interflow.

Night Service

Night Service redirects all calls to one of the following internal destinations:

- An ACD split
- An extension
- An attendant group
- An announcement with forced disconnect.

Night service is available for a hunt group, a trunk group, or a system. These types of night service are explained below.

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, its destination will be used for calls that come in over the trunk group even if they 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. To avoid dissatisfied callers, agents should continue to staff the split until the queue is empty.

Distributing and handling calls

This section describes how calls are distributed to agents and how agents handle the calls.

This section includes the following topics:

- [About call distribution](#) on page 32
 - [Call distribution methods without EAS](#) on page 32
 - [Call distribution methods with EAS](#) on page 34
 - [How agents handle calls](#) on page 36
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About call distribution

ACD calls are delivered to agents according to the type of call distribution (also known as hunting) that is assigned to the split or skill. When the ACD and Queue field on the Hunt Group screen are set to y, queues for agents and queues for calls are established in the communication server. These queues are used to handle incoming calls based on the type of call distribution that is implemented. The types of call distribution available for use depend on whether or not EAS is used by the call center.

Call distribution methods without EAS

Without EAS, the following call distribution methods are available:

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

Note:

The following descriptions of ACD call distribution assume that the Multiple Call Handling (MCH) feature is not assigned. Agent availability is different for splits assigned the MCH feature.

Direct

ACD software searches for an available agent in the order that extensions were assigned to the split (through communication server administration), starting with the first extension assigned to the split. This type of call distribution is most useful when management wants the most effective or most experienced agents to handle more calls. Agents are rank-ordered from most to least effective and then are assigned to the split in that order. Direct call distribution is called Direct Department Calling (DDC).

If you administer a split for DDC, an incoming call is routed to the first available agent extension in the administered sequence. If the agent is not available, the call routes to the next available agent, and so on. Incoming calls are always routed to the first agent in the sequence, so calls are not evenly distributed among agents.

UCD-MIA

When the UCD-MIA call distribution method is used, the communication server searches for the agent extension that has been idle (waiting) the longest and delivers the call to that extension if the agent is available to handle an ACD call. This type of call distribution ensures a high degree of equity in agent workloads even when call-handling times vary.

The ACD software determines which agent extension has been idle the longest by maintaining an ordered list (queue) of agents who are eligible to receive the next ACD call. Eligible agents enter the queue at the bottom and move toward the top of the queue. The agent who has been in queue the longest receives the next ACD call unless the agent is not available at the time the call is to be distributed. If the agent at the top of the queue is not available, the ACD software 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 or skill associated with the call. The *MIA across splits/skills* option is used to put an agent at the bottom of all split or skill queues that the agent is logged in to when the agent completes any ACD call. Agents move toward the top of the eligible-agent queue as long as they 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/skills* option is turned on). Agents in After Call Work (ACW) mode are in eligible agent queues on Generic 3 communication servers. You can choose whether these agents are or are not in the eligible-agent queues for the communication server.

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

- In ACW
- 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 agents' progress in each queue is independent of any activity in other queues. Agents in the AUX state are not in the eligible-agent queue.

You can set the communication server 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/skills. If the *MIA Across Splits/Skills?* field on the Feature-Related System Parameters screen is set to n, the communication server maintains available agent queues for each split or skill. When agents answer a call, they are only removed from the available agent queue for the split or skill at which that call arrived. If the field is set to y, then the agent is removed from all split or skill queues that the agent is logged in to whenever they answer a call for any of their assigned splits/skills.

ACD basics

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? on the Feature-Related System Parameters screen is y, the agent is queued when the call completes.
- If ACW Agents Considered Idle? is n, the agent is queued when ACW completes.

Note:

If you are using an Expert Agent Distribution method (EAD-MIA or EAD-LOA), then the agent is put back in queue(s) after completing an ACD call based on skill level.

If you are not using an EAD call distribution method, then the agent is put at the bottom of the queue(s) after completing an ACD call.

Call distribution methods with EAS

With EAS, the following call distribution methods are available:

- 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.

Agents available, call arrives, and agent selection method is:	THEN the communication server selects:
EAD-MIA	the highest skill level, 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 the same in the EAS environment as it does without EAS, except that the communication server searches for the most idle agent with the required skill.

UCD-MIA does not select an agent based on 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 that agent.

EAD-MIA

The EAD-MIA call distribution method selects the most idle agent with the required skill to handle the call and the highest skill level.

This method of call distribution adds a layer of processing on top of the Most Idle Agent distribution call processing. EAD-MIA sorts the agents in the eligible-agent queue into multiple queues based on 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, most expert agent available.

When you are using EAS Preference Handling Distribution (EAS-PHD), the agent can enter the MIA queue at one of 16 levels. The lower the level, the higher the level of expertise; so 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 or level 2 agents. When agents with a lower skill level become idle, they enter the MIA queue in front of agents with a higher skill level. See [Expert Agent Selection](#) on page 153 for more information about EAS Call Distribution.

UCD-LOA

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

The least occupied agent is the agent who has spent the lowest percentage of their time on ACD calls since logging in. The agent's place in the queue of available agents is determined by this percentage. The agent occupancy (the percentage of time on calls) is always calculated separately for each skill an agent is logged into, so there is an available agent queue for each skill.

EAD-LOA

When the EAD-LOA call distribution method is in use, the communication server 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.

How agents handle calls

An agent can receive split calls and, in most cases, personal calls that are not related to a split. Calls distributed to an agent's telephone by the ACD feature on the communication server are considered ACD calls. Calls dialed directly to an individual agent using the agent's extension number (such as internal calls and DID extension calls) are called extension-in (EXT-IN) calls. Outgoing calls the agent makes are called extension-out (EXT-OUT) calls. EXT-IN and EXT-OUT calls are considered non-ACD calls.

Things to know before you start

The capability of a telephone to receive EXT-IN calls or to make EXT-OUT calls can be restricted through communication server administration. The following descriptions of agent call handling assume that the Multiple Call Handling (MCH) feature is not assigned. Agent availability and call handling are different for splits assigned the MCH feature.

ACD calls are distributed only to available agent extensions. To be considered available, an agent must first staff an agent extension and then select a call-answering mode (automatic in or manual in).

Staffing agent extensions without EAS

To staff an agent extension on the communication server without the EAS feature, an agent must dial a login access code or press the LOGIN button on the agent's telephone. The agent must then dial a split number and a login ID. The login ID length, the login dial access code, and, if desired, the LOGIN button are assigned through communication server administration. The split number may also be assigned to the LOGIN button or to another telephone button.

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 an extension is active and logged into the system (AUX work mode). Pressing the login button and then following the appropriate system login procedure makes the extension staffed in AUXWORK. This procedure varies with the type system you have.

Agent logout

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

Agent request for supervisor assistance

When supervisor assistance is needed, an agent can press the ASSIST button or dial the ASSIST feature access code and the split or skill group number brings the designated person on line. On G2 the agent must place the current call on hold before pressing ASSIST. On G3 pressing ASSIST 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 function(s) the agent is performing at a given time.

When the agent is engaged in an ACD call, the agent is in the ACD agent state.

After staffing an extension, the agent is in the auxiliary work (aux-work) mode, which is considered non-ACD work.

Auto-in versus manual-in

In aux-work mode, the agent is not yet available to receive ACD calls. To become available for ACD calls, the agent must press the manual-in or auto-in button to select a call answering mode.

<p>Auto-in</p>	<p>Like the manual-in button, the auto-in button tells the ACD that the agent is available for an ACD call. However, when the call ends, the agent is immediately available for another ACD call according to the established call distribution method. The agent does not have to press any buttons to receive another ACD call. This type of call answering increases the number of calls that agents can answer in a given period of time and is most effective if agents have little or no call-related work to do after finishing each ACD call. The communication server has a timed ACW feature for auto-in operation. This option automatically puts the agent into ACW for a preset length of time at the end of an auto-in call. When the time is up, the agent automatically becomes available to take an ACD call. manual-in and auto-in dial access codes and telephone buttons are assigned through communication server administration.</p>
<p>MANUAL-IN</p>	<p>The MANUAL-IN button tells the ACD that the agent extension is available for an ACD call. The ACD then distributes a call to the agent according to the established call distribution method. When the call ends, the agent automatically enters the ACW state. While in ACW, the agent is not available 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 telephone buttons are assigned through communication server administration.</p>

Ringling versus zip tone for incoming calls

When a call arrives at a telephone, the agent may hear ringing or zip tone (beeping), depending on how the telephone is administered. Ringing is recommended when an agent answers calls using the handset. When a call connects to the agent telephone, the telephone rings, and the agent picks up the handset to answer the call.

Zip tone is recommended when the agent uses a headset to answer calls. Zip tone can also be used with a handset, but the agent must hold the handset and listen for the zip tone. When a call connects to an agent telephone, the agent hears one burst of zip tone for calls dialed directly to the split (or agent extension on the communication server) and, without pushing any buttons, the agent greets the caller.

Ringing or zip tone is established on a per-telephone basis through communication server administration.

Ringing is also called *manual answer* and zip tone is also called *automatic answer*.

Auxiliary Work (AUXWORK) and ACW

To temporarily stop ACD calls from arriving at an agent's telephone, an agent can press the Auxiliary Work (AUXWORK) or After Call Work (ACW) button.

AUXWORK: The agent is involved in non-ACD work, is on break, in a meeting or at lunch. CMS 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 AUX. AUX-OUT implies that the agent placed an outgoing call while in AUX.

The AUXWORK button temporarily stops ACD calls from arriving at the agent's telephone. 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 this button which places the agent in the auxiliary-work state. To receive ACD calls again, the agent presses the manual-in or auto-in button.

The AUXWORK button (or the dial access code, if no button is available) is assigned through communication server administration. If an agent is normally logged into more than one split, an AUXWORK button for each split may be assigned. Then, when the agent presses the AUXWORK button for a particular split, the agent will not receive calls from that split. However, the agent will still be available for calls from the other splits the agent is logged into.

Also, if an agent is logged into more than one split or skill and receives an ACD call for one split or skill, the agent is unavailable for calls for other splits/skills.

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 in ACW. ACW-OUT implies that the agent made an outgoing call while in ACW.

The ACW button temporarily stops ACD calls from arriving at the agent's telephone. An agent who is in auto-in mode presses this button during a call so that when the call is finished, the agent will 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 the 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 into more than one split, pressing the ACW button makes the agent unavailable for calls in all splits. CMS considers the agent to be in the OTHER state for all splits other than the split in which the agent is currently in ACW.

ACD basics

The following table lists additional agent states/work modes that may display.

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

Trunk states

Trunk State indicates the current status of a specific trunk, or the ability to change that state. Trunk states are described in the following table.

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.
Fwrđ	A queued call has been intraflowed outside the ACD or has been interflowed to another PBX/communication server.
Mbusy	Maintenance Busy, or out of service for maintenance purposes.
Hold	The agent has put the call on hold.

Other telephone buttons

This section describes other buttons that can be assigned to an agent's telephone.

CALL APPEARANCE	<p>These buttons are used to place (originate) and answer calls. Two status lamps (red and green) are next to each call appearance button. The red lamp lights when an agent presses an appearance button to make or answer a call. The green lamp flashes to indicate an incoming call.</p> <p>Except with Multiple Call Handling, incoming ACD calls always arrive at the first call appearance. However, telephones may be assigned more appearances to provide additional call-handling capabilities. For example, an agent can use a second call appearance to transfer or place calls since the line will be free of ACD calls. On a two-appearance telephone, the second appearance can only be used to originate calls.</p>
ADD SKILL	<p>For communication servers with EAS, logged-in agents or telephone users with console permissions can press this button to add a skill. This is an Abbreviated Dialing (AD) button programmed with the add skill Feature Access Code (FAC).</p>
ALERT CHANGE	<p>The lamp associated with this telephone button flashes when another user changes an agent's assigned skills or moves an agent from his or her current split to a different split. The lamp does not flash when an agent changes his or her own skills from the telephone.</p>
ASSIST	<p>Press this button to request help from the split supervisor. The ASSIST button automatically dials the split supervisor's extension and connects the agent to the supervisor. Pressing the ASSIST button automatically puts the current call on hold.</p>
AUDIO TROUBLE	<p>Agents press this button to report a call with poor transmission quality to CMS. The message the communication server sends CMS includes the agent's extension, the trunk being used, and the time of day the trouble occurred. This information is reported in CMS exception reports and is useful for trouble-shooting trunk and extension problems. For more information, see <i>Avaya CMS Administration</i>.</p> <p>Stroke count button 0 is used for reporting audio difficulty.</p>
CONFERENCE	<p>Press this button to add another person to a two-person call. An agent with a multi-appearance telephone can add up to four additional people to a 2-person call. For single-appearance telephones, only one person can be added. Single appearance telephones do not have a CONFERENCE button. Agents must use the RECALL button to conference a call. If an agent adds another agent into a conference call, the resulting conference is not considered an ACD call for the added agent. The ACD considers the added agent to be on an extension-in call.</p>
CALL WORK CODE	<p>Agents press this button and enter up to 16 digits to record the occurrence of a customer-defined event. Call Work Codes are stored on CMS, not on the communication server.</p>
CALLER-INFO	<p>With the Call Prompting feature, agents press this button to display the digits collected by the last collect digits vector command.</p>

EMERGENCY	Press this button to report a malicious call to the controller. The controller can then trace the call.
HOLD	<p>Press this button to put a call on hold. The ACD will not send any more calls to an agent who has a call on hold. For communication server with Multiple Call Handling, an agent can put an ACD or non-ACD call on hold and receive an ACD call by pressing the auto-in or manual-in button. With Multiple Call Handling, multiple ACD calls can be delivered automatically to an agent in auto-in or Manual-In work mode, provided that an unrestricted line appearance is available on the telephone.</p> <p>Single appearance telephones do not have a HOLD button. Agents must use the RECALL button or the telephone's switchhook to put a call on hold. A single appearance telephone cannot be used to handle multiple ACD calls.</p>
LOGIN	Press this button to staff the extension and start CMS collection of agent data. This is an Abbreviated Dialing (AD) button programmed with the login Feature Access Code (FAC).
LOGOUT	Press this button to unstaff the extension and end CMS collection of agent data. This is an Abbreviated Dialing (AD) button programmed with the logout Feature Access Code (FAC).
RECALL/Flash switchhook	To put calls on hold, transfer calls, and create conference calls, agents using single-appearance telephones press the RECALL button or flash the switchhook if the phone is not equipped with a RECALL button.
RELEASE	Press this button to disconnect a call. Do not use the DROP button.
REMOVE SKILL	With EAS, logged-in agents or telephone users with console permissions can press this button to remove a skill. This is an Abbreviated Dialing (AD) button programmed with the remove skill Feature Access Code (FAC).
STROKE COUNT	As many as nine STROKE COUNT buttons can be assigned. Agents press these buttons to record call events of interest. CMS records and reports stroke-count information. Stroke count button 0 is reserved for audio difficulty.
TRANSFER	<p>Agents normally press the TRANSFER button to transfer calls to other agents or the split supervisor. This button is only available on multi-appearance telephones. Single-appearance telephone users must use the telephone's switchhook.</p> <p>Agents can also use the TRANSFER button to transfer calls to external destinations. External transfer must be assigned to a telephone as a feature over and above the normal transfer feature. If an agent transfers a call to another agent, the call is not considered an ACD call for the agent receiving the call unless the transferring agent dialed a split extension, VDN, or agent login ID, an EAS capability known as Direct Agent Calling (DAC). The ACD considers the agent receiving the transfer to be on an extension-in call. For the agent transferring a call, the call is counted as an EXT-OUT call.</p>
VUSTATS	Agents with display telephones press this button to display agent, split or skill, VDN, or trunk group data similar to that reported by CMS.

Queue status lamps

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

NQC	The lamp associated with the Number of Queued Calls (NQC) button tells the agent that calls are in queue and when the number of calls in queue has met or exceeded the assigned queue threshold for the split. If no calls are in the split's queue, the status lamp associated with the button is dark. When one or more calls are 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 lamp associated with the Oldest Queued Time (OQT) button tells the agent that calls are in queue and when the oldest call in queue has been waiting longer than the assigned wait time threshold (0 to 999 seconds) for the split. If no calls are in the split's 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 by the oldest call in queue, the lamp flashes on and off. A flashing queue status lamp tells agents they need to handle calls more quickly. The thresholds that cause the lamps to flash and the telephone buttons are assigned through communication server administration.
Auxiliary queue status lamps	An auxiliary queue status lamp indicates that either the NQC threshold or the OQT threshold has been reached. The lamp lights when the assigned threshold is met or exceeded. Unlike the lamps on a telephone, the auxiliary queue status lamp does not indicate when calls queue to the split.

Display buttons

The following telephone buttons control the information that appears on the display.

NORMAL	Press this button to display information about the active call appearance. Press this button to display incoming call information (either an extension-in call or an intraflowed/interflowed call) for a different call appearance.telephone
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Split supervisor telephone 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 server administration procedures.

Telephone button definitions

The following telephone buttons are available only to the split supervisor's extension.

NIGHT SERVICE	The split supervisor presses this button to send all calls to night service. The Night Service may be Trunk Group Night Service or Split Night Service. Also, a separate button for each type of night service may be available.
RECORD ANNCT	The supervisor presses this button to either listen to or to record an announcement for the split.
SERVICE OBSERVE	<p>The supervisor presses this button and dials an agent extension number to listen to conversations on the telephone. The Service Observe feature permits the supervisor to check an agent's call-handling technique. An agent's telephone may also be assigned the SERVICE OBSERVE button so that the agent can listen to another agent's conversations. This capability is especially useful for agent training. Service Observing can be set up for listening only or for both listening and talking.</p> <p>For communication servers with EAS, a logical agent ID, which is associated with an agent, not the telephone the agent is currently using, can be service observed. For communication servers with Call Vectoring, VDNs can be service observed. Feature Access Codes which allow Service Observing from an external location or from a telephone that does not have feature buttons can be assigned through communication server administration.</p>
VU STATS	Split supervisors and agents with display telephones press this button to display agent, split or skill, VDN, or trunk group data similar to that reported by CMS.

ACD and call management systems - BCMS and Avaya CMS

This section includes the following topics:

- [About BCMS](#) on page 44
- [About CMS](#) on page 45

About BCMS

The Basic Call Management System (BCMS) is a software package, residing on the PBX/communication server, used to provide real-time and historical reports to assist in managing ACD splits/skills, agents (extensions), trunk groups and VDNs (G3 only). These reports, provided by the system, are a subset of those reports available with the CMS adjunct.

About CMS

The Avaya Call Management System (CMS) is an adjunct that collects specific ACD data on measured splits/skills, measured agents, measured extensions, measured trunks and measured trunk groups for reporting purposes. If Call Vectoring is purchased, ACD will report on measured VDNs and Vectors. CMS provides call management performance recording and reporting. It can also be used to perform some ACD administration. CMS is used by customers to determine how well their customers are being served (in other words., speed of call answers, number of calls) and how efficient their call management operation is (in other words, agents versus traffic requirements).

This section includes the following topics:

- [How CMS works with ACD](#) on page 46
- [Data measured by CMS](#) on page 46
- [Assigning CMS measurement of the ACD](#) on page 46
- [Things to know before you start](#) on page 47
- [Communication server features that affect CMS data](#) on page 47
- [Hold, Conference, and Transfer](#) on page 47
- [About MCH](#) on page 47
- [Call Pickup](#) on page 48
- [Intraflow and interflow](#) on page 48
- [About RONA](#) on page 48
- [Phantom abandon call timer](#) on page 49
- [About moving an agent while staffed](#) on page 49
- [Expanded agent capabilities](#) on page 50
- [About BSR](#) on page 50
- [About UCID](#) on page 51
- [Avaya Business Advocate](#) on page 51

How CMS works with ACD

To collect information on ACD, CMS must be able to communicate with the ACD resident in the communication server. The communication server to CMS Platform communication consists of electronic messages sent back and forth between the ACD communication server and the CMS Platform using a data link. There are two types of messages:

- Translations - Tells CMS the configuration of the ACD. This includes what data is measured (to be collected) and the ACD assignments.
- Status Changes - Tells CMS when the states of agents or trunks change due to call activity. Occurrences are counted and durations are tracked.

Note:

CMS can also be used to change configurations within the ACD. Therefore, CMS can at times send translations back to the PBX.

Data measured by CMS

CMS measures data on ACD splits, agents, extensions, trunks and trunk groups. If Call Vectoring is purchased, ACD will report on VDNs and Vectors. The following table summarizes the types of data generated by the ACD and measured by the CMS.

Data type	Parameters
Agents	Agent States
Splits/Skills	Events Workload Distribution Split/Skill call totals
Trunks	Trunk states
Trunk Groups	Events Workload Distribution Trunk call totals

Assigning CMS measurement of the ACD

CMS collects data on splits/skills, agents, extensions, trunks, trunk groups, VDNs, and vectors. However, for CMS to collect data, the appropriate items (splits/skills, extensions, and so on) must be identified as measured on the communication server.

Individual splits and trunk groups are assigned to CMS measurement through communication server administration. Extensions are measured by virtue of their assignment to measured splits. Trunks are measured by virtue of their assignment to measured trunk groups. The number of measured splits cannot be changed using the CMS ACD Status screen.

Measured splits need not be numbered sequentially. VDNs are measured individually. All vectors are measured.

Things to know before you start

The CMS ACD Status window lists the total number of measured splits/skills, extensions (Agent Positions), trunks, and trunk groups established in an ACD. For more information, see *Avaya CMS Administration*.

You can add, delete, or change measured trunks, trunk groups, agent extensions, agent login IDs, VDN extensions, splits, and skills without busying out the link to CMS and losing CMS data.

Communication server features that affect CMS data

There are several communication server features that affect CMS data, such as Conference, Transfer, Multiple Call Handling, Call Pickup, Intraflow, Interflow, Redirection on No Answer, Phantom Abandon Call Timer, Move Agent While Staffed, Expanded Agent Capabilities, Best Service Routing, and Universal Call ID. These features are explained below.

Hold, Conference, and Transfer

CMS tracks any type of call an agent puts on hold by pressing the Hold button, dialing the hold access code, pressing the Conference or Transfer button, or flashing the switchhook.

Information on all calls (split or skill ACD, direct agent ACD, and extension calls) and the time spent on hold is stored in agent database tables. Information on split or skill calls only and the time spent on hold is stored in split or skill tables.

About MCH

Avaya communication servers have options to the Multiple Call Handling (MCH) feature that can force agents to receive one or more ACD calls with other ACD calls or extension (non-ACD) calls on hold or active. For these forced options, talk time (and not ringing time) accumulates until the agent puts the current call on hold or releases it.

With Multiple Call Handling, an agent can put a call on hold and press the manual-in or auto-in button to receive another ACD call. When multiple calls are on hold at the same time, hold time accumulates for each call on hold, and the total hold time can exceed clock time. For example, if two calls are on hold for 5 minutes each, 10 minutes of hold time accumulates.

Call Pickup

When an agent uses the Call Pickup feature to pick up an ACD call that rings at another agent's extension, CMS tracks the call as an AUX-IN call for the agent picking up the call. The split or skill of the agent originally called is credited with an outflow call, even if the agent who picked up the call is in the same split or skill. If an agent is logged into more than one split or skill, the call is counted for the split or skill the agent has been logged into the longest. Thus, when Call Pickup is used, CMS does not count the call as an ACD call, even though the call queued to a split or skill and was answered. Various other types of data associated with ACD calls (for example, *Percent Answered Within Service Level* and *Average Speed of Answer*) will also not include data on calls answered using the Call Pickup feature. Because the split or skill of the agent originally called is credited with an outflow call, the call counts against the *Percent Answered Within Service Level* for that split or skill.

Intraflow and interflow

When a call is intraflowed or interflowed from a split or skill, CMS counts the call as an outflow call for the split or skill. If a call is intraflowed into a split or skill, CMS counts the call as an inflow call for the split or skill. CMS counts interflowed calls as ordinary incoming calls for the split or skill. However, because calls can be intraflowed/interflowed to destinations that are not splits/skills or are not measured by CMS, an outflow call from a split or skill will not always show a corresponding inflow call for another split or skill. Conversely, because calls can be intraflowed/interflowed into a split or skill from originating locations that are not measured by CMS, an inflow call to a split or skill may not show a corresponding outflow from another split or skill.

If an intraflowed/interflowed call connects to an agent in the destination split or skill, that call is counted as an ACD call for the split or skill.

A dummy split or skill may be established which intraflows calls to another split or skill. For CMS to count outflow calls for dummy splits/skills, intraflow should be established using the Call Forwarding feature. If Call Coverage is used to intraflow calls, at least one agent must log into the dummy split or skill and go into ACW, and the call must queue to the dummy split or skill for at least one ring cycle for an outflow call to be counted.

For communication servers with the Call Vectoring feature, intraflow and interflow work differently, and CMS data related to intraflow and interflow are recorded differently.

About RONA

When a ringing call times out and is requeued to the same split or skill by the Redirection On No Answer (RONA) feature, Avaya CMS counts an outflow and an inflow for the split or skill. That is, the redirected call appears as two offered calls to the split or skill. If the call redirects from ringing to a VDN, there is outflow from the initial VDN and from the split or skill. If the call was in another VDN prior to redirection to another VDN, then there is inflow to that VDN.

Also, NOANSREDIR is incremented for the split or skill and the VDN. For CMS R3V2 and newer, the database item NOANSREDIR is also incremented for split or skill and for VDN, if the call is in a VDN. If a split or skill is set up so that split or skill calls do not redirect back to the split or skill except by way of the Redirection On No Answer feature, the unique calls offered to the split or skill can be calculated by subtracting the value of NOANSREDIR from CALLSOFFERED.

If a call redirects from ringing to a VDN, there is outflow from the split or skill and, if the call was in another VDN, there also is inflow to the new VDN and outflow from the initial VDN. The NOANSREDIR is incremented for split or skill and VDN.

Phantom abandon call timer

CMS can collect information about phantom abandon calls. When this capability is enabled, calls with a talk time (duration) shorter than the administered value (1 - 10 seconds) are counted as phantom abandon calls. Setting the timer to zero disables it. CMS uses the PHANTOMABNS database item to store the number of phantom abandon calls.

This capability is important in areas where the public network switches do not provide disconnect supervision. Without this capability, short-duration calls that queue to a split or skill and are answered by an ACD agent or other answering position are counted as ACD calls, even if the calling party hangs up before the call is answered. This type of call is called a phantom or ghost call.

About moving an agent while staffed

A staffed agent can be moved between splits or the skill assignments for staffed agents can be changed. If the agent has any call on the telephone or is in ACW, then the move cannot take place immediately, but is pending the agent telephone going idle (all calls have been terminated), or the agent changing out of the ACW mode.

CMS provides two real-time database items in the agent data, MOVEPENDING and PENDINGSPILT, that can be accessed by using custom reports to provide information about whether agents have moves pending and, if so, the split or skill to which they are being moved. Note that in the case that the agent's skills are being changed and the change adds more than one skill, the PENDINGSPILT item will show the first skill that is being added. It is also possible for MOVEPENDING to be set, but for PENDINGSPILT to be blank (or 0). This can happen, for example, when the link to the communication server comes up and a move is pending for an agent. CMS will be notified by the communication server that the move is pending, but PENDINGSPILT will not be set.

Expanded agent capabilities

Expanded Agent Capabilities allow EAS agents to have up to 20 or 60 skills (depending on platform) assigned. Each skill may be assigned a level from 1 to 16, where Reserve 1 and Reserve 2 are the highest levels and 16 is the lowest. (The numeric level replaces the skill type p or s used in earlier G3 EAS releases.) Agents may have a call handling preference based either on the skill level, meaning that the agent will serve calls waiting for his or her highest level skill before serving calls waiting for any lower level skills; or based on greatest need, meaning that the agent will serve the highest-priority, oldest call waiting for any of his or her skills, or percent allocation, based on the percent distribution of calls among the agent's skills.

The expanded agent capabilities feature also allows the specification of the skill to be used for the agent's direct calls. This also allows specification of the level for the direct agent skill, which, in conjunction with the agent's call handling preference, may affect the order in which a direct agent call is delivered to an agent. That is, direct agent calls need to be delivered for all skill ACD calls. A concept introduced in R3V5 CMS, that of the top skill, can be useful in EAS implementations that use skill level call handling preference for agents. An agent's first administered, highest level skill is the agent's top skill, since it is for this skill that the agent is most likely to handle calls. This is the skill that can count on the agent.

Database items track the number of top agents in skills, as well as the time top agents spent available and in AUX.

The expanded agent capabilities on the communication server include an increased number of measured splits/skills to 600 and an increase in the number of measured agent/split or agent/skill pairs to 10,000 for the G3r processor, as well as new options for Most Idle Agent (MIA) call distribution. The new options allow selection of MIA distribution across skills, rather than for each skill, and selection of whether agents in ACW are or are not included in the agent free list. These options have no direct impact on CMS, since CMS does not keep track of the most idle agent.

About BSR

Best Service Routing (BSR) allows calls to be balanced at a single site or between multiple sites. BSR is enhanced multi-site routing that provides new call vectoring functions that build upon the Look-Ahead Interflow feature to route a call to the best skill on a single ECS or to the best skill in a network of Avaya communication servers.

The best skill is defined as the local skill or remote ECS that offers the shortest waiting time for the call in a call surplus (calls queued) situation for the application. The waiting time is calculated using the Expected Wait Time (EWT) predictor, and can be adjusted by the user. In an agents available situation, the best skill is determined based on the assigned available agent strategy. BSR data is tracked in the vector, VDN, and call history tables.

About UCID

Universal Call ID (UCID) is a unique tag that is assigned to a call. The tag allows call-related data to be collected and aggregated from multiple sources and multiple sites. The UCID may then be used to group all the data from various sources about a particular call.

CMS will receive the UCID assigned to calls by a communication server when this feature is enabled. The UCID is then stored, along with data about the call itself, by the call history feature (which includes both internal and external call history). The data will be available to both Custom Reports and the Report Designer. UCID data is stored in the call history and agent trace tables.

Avaya Business Advocate

Avaya Business Advocate is a collection of features that provide flexibility in the way a call is selected for an agent in a call surplus situation and in the way that an agent is selected for a call in an agent surplus situation. Advocate also includes methods for automating staffing adjustments.

ACD call center features

This section describes the following Avaya Call Center features that are administered on the Avaya communication server. This section includes the following topics:

- [Abandoned Call Search](#) on page 55
- [ACD options by agent](#) on page 57
- [Add/Remove Skills](#) on page 58
- [Agent Call Handling](#) on page 60
- [Auto-Available Split/Skill](#) on page 78
- [Automatic Call Distribution](#) on page 82
- [Avaya Agent Deskphone 16CC](#) on page 93
- [Avaya Business Advocate](#) on page 98
- [Basic Call Management System](#) on page 107
- [Best Service Routing](#) on page 110
- [Call Prompting](#) on page 134
- [Call Vectoring](#) on page 137
- [Direct Agent Calling](#) on page 145
- [Expert Agent Selection](#) on page 153
- [Forced Agent Logout from ACW mode](#) on page 162
- [Forced Agent Logout by Clock Time](#) on page 165
- [Inbound Call Management](#) on page 171
- [Information Forwarding](#) on page 178
- [Intraflow and Interflow](#) on page 186
- [Location Preference Distribution](#) on page 189
- [Look-Ahead Interflow](#) on page 196
- [Multiple Call Handling](#) on page 202
- [Network Call Redirection \(NCR\)](#) on page 208
- [Queue Status Indications](#) on page 209
- [Proactive Contact outbound calling improved reporting](#) on page 212
- [Reason codes](#) on page 218
- [Redirection on No Answer](#) on page 222

ACD call center features

- [Redirection on IP Failure](#) on page 235
- [Redirection on OPTIM Failure](#) on page 239
- [Remote Logout of Agent](#) on page 242
- [Reporting adjuncts](#) on page 247
- [Service Level Maximizer](#) on page 256
- [Service Observing](#) on page 257
- [Service Observing with Multiple Observers](#) on page 274
- [Universal Call ID](#) on page 277
- [Variables in Vectors](#) on page 292
- [VDN in a Coverage Path](#) on page 293
- [VDN of Origin Announcement](#) on page 297
- [VDN Time Zone Offset](#) on page 303
- [VDN Variables](#) on page 306
- [Voice Response Integration](#) on page 307
- [VuStats](#) on page 314

Related feature or screen: Refer to the *Feature Description and Implementation for Avaya Communication Manager*, for more information about the following related features or forms:

- Announcements/Audio Sources
- Calling Party/Billing Number
- CallVisor Adjunct-Switch Application Interface
- Class of Restriction
- Hunt Groups
- Malicious Call Trace
- Recorded Announcements (additional information also available in [Recorded announcements](#) on page 415)
- Service Observing
- Callmaster™ phones
- 500, 2500, K2500, 7101A, 7102A, 7103A, 7104A, 8110, OPS, DS1FD, DS1SA, and VRU phones

Abandoned Call Search

This section includes the following topics:

- [About Abandoned Call Search](#) on page 55
- [Administering abandoned call search](#) on page 55
- [Abandoned Call Search considerations](#) on page 56

About Abandoned Call Search

Abandoned Call Search allows the communication server to identify abandoned calls if the Central Office (CO) does not provide timely disconnect supervision. An abandoned call is one in which the calling party hangs up before the call is answered. Note that Abandoned Call Search is suitable only for older COs that do not provide timely disconnect supervision. Most COs provide timely disconnect supervision and do not require Abandoned Call Search.

Before an incoming Automatic Call Distribution (ACD) call rings a hunt group member or agent, the system checks to make sure that the calling party has not abandoned the call. If the calling party has abandoned the call, the call does not ring the hunt group member or agent.

If a call has been abandoned, the system determines if the calling party is still connected to the ground-start trunk at the CO. To do this, the system flashes (that is, opens the tip-ring loop for 150 to 200 ms) the CO end of the trunk. If the calling party is still connected, the CO does not respond. If the calling party has abandoned the call, the CO sends the system a disconnect signal within 800 ms. The system interprets this as an abandoned call, releases the trunk, and the call does not ring the hunt group member or agent.

Outside of the U.S., a flash of this duration may be handled differently. For more information about trunk flash, see *Feature Description and Implementation for Avaya Communication Manager*.

Administering abandoned call search

The following forms and fields are required to administer the Abandoned Call Search feature.

screen	Field
Trunk Group <ul style="list-style-type: none"> • CO • FX • WATS 	Abandoned Call Search

ACD call center features

You administer Abandoned Call Search on a per-trunk-group basis. Administer each ground-start CO, Foreign eXchange (FX), and WATS trunk group either having Abandoned Call Search or not having it. Abandoned Call Search is not supported for tie trunks.

Abandoned Call Search considerations

- Abandoned Call Search works with ground-start analog trunks that do not provide disconnect supervision and that do react to a 500-ms break.
- Some older COs can take as long as two minutes to notify the communication server of a disconnect. Thus, the communication server must determine within one second whether the call has been abandoned, before extending the call. Even with Abandoned Call Search or disconnect supervision, there is a small probability that a call will be extended to the destination hunt group after the caller has hung up. Abandoned Call Search and disconnect supervision significantly reduce that probability.
- Abandoned Call Search allows agents and hunt group members to answer more calls because time is not wasted on abandoned calls. In addition, call-handling statistics that the Call Management System (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 options by agent

This section includes the following topics:

- [ACD options by agent description](#) on page 57
- [Reason to use](#) on page 57
- [ACD options by agent considerations](#) on page 57

ACD options by agent description

You can now set the following ACD options for individual agents:

- ACW Agent Considered Idle
- AUX Work Reason Code Type
- Forced Agent Logout from ACW
- MIA Across Skills
- Logout Reason Code Type

In previous releases, you could set these options only system-wide.

Reason to use

Customers may want to set the ACD options differently for each agent.

ACD options by agent considerations

Use the Login ID screen to set the ACD options for each agent. The ACD option fields currently on the Feature-Related System Parameters screen remain the same. Consider the following:

- The ACD option settings on the Agent Login ID screen take precedence over the system-wide settings on the Feature-Related System Parameters screen.
- If any options are set to **system** on the Login ID screen, the system-wide setting is applied.
- The **system** setting is the default setting for both new and upgraded systems.
- The settings on the Feature-Related System Parameters screen or the Login ID screen override the system option and apply to both ACD calls and Direct Agent calls.
- In order for these changes to take effect, the agent must log out and log back in.

Add/Remove Skills

This section includes the following topics:

- [About Add/Remove Skills](#) on page 58
- [Forms and fields required for the Add/Remove Skills](#) on page 59
- [Add/Remove Skills feature considerations](#) on page 59
- [Interactions with other features and systems](#) on page 59

About Add/Remove Skills

Add/Remove Skills allows an agent using Expert Agent Selection (EAS) to add or remove skills. A skill is a numeric identifier in the communication server that refers to an agent's specific ability. For example, an agent who is able to speak English and Spanish could be assigned a Spanish-speaking skill with an identifier of 20. The agent then adds skill 20 to his or her set of working skills. If a customer needs a Spanish-speaking agent, the system routes the call to an agent with that skill.

Agents can dial feature access codes (FACs) to add or remove a skill. Or a supervisor with console permission can enter an agent's login ID and add or remove an agent's skill. If a supervisor adds or removes a skill for an agent, the agent receives a change notification.

To determine if they need to add or remove a skill, agents and supervisors can use:

- Queue-status indications
- Avaya Basic Call Management System Reporting Desktop VuStats
- Avaya Call Management System (CMS) or Basic Call Management System (BCMS) information

When adding a skill, the agent must specify the skill priority level (1 - 16).

On phones with displays, the system prompts the agent through the process of adding or removing a skill and displays the updated set of skills.

Forms and fields required for the Add/Remove Skills

The following forms and fields are required to administer the Add/Remove Skills feature.

screen	Field
Class of Restriction (COR)	Add/Remove Agent Skills
Feature Access Code (FAC)	Add Agent Skills Remove Agent Skills
Language Translations	41-44 on Page 5
Hunt Group	Skill
Class of Service	Administer console permissions for supervisors

Add/Remove Skills feature considerations

Consider the following when using the Add/Remove Skills feature:

- A skill cannot be removed from an agent's skill set if the agent is on a call for that skill or in the After Call Work (ACW) state for that skill.
- With EAS, agents cannot remove their direct agent skill.

Interactions with other features and systems

The Add/Remove Skills feature has the following interactions with other features and systems:

Auto-Available Skills (AAS). If an agent adds a skill that is administered as Auto-Available, you must set the AAS field to y for that agent's login ID on the Agent Login ID screen.

BCMS. BCMS begins tracking a new skill as soon as it is added. When an agent removes a skill, the real-time agent information specific to that skill is removed from the real-time reports, but it still appears on the historical reports.

EAS-PHD. When EAS-PHD is set as an option, agents cannot remove their direct agent skill. In an EAS environment, agents must have at least one skill assigned to them during a login session. With EAS-PHD, agents can specify up to 20 or 60 skills (depending on platform).

Note:

If EAS-PHD is not enabled, agents can specify only four skills.

VuStats: Because VuStats displays information gathered by BCMS whether BCMS is enabled or not, the BCMS interaction above applies to VuStats.

Agent Call Handling

This section includes the following topics:

- [About Agent Call Handling](#) on page 60
- [Administering Agent Call Handling](#) on page 61
- [Communication server controls](#) on page 62
- [Agent Call Handling considerations](#) on page 73
- [Agent Call Handling interactions](#) on page 75

About Agent Call Handling

Agent Call Handling allows you to administer functions that Automatic Call Distribution (ACD) agents use to answer and process ACD calls.

You define the following agent capabilities:

- Agent login and logout
- Agent answering options: Automatic Answer (zip tone) or Manual Answer
- ACD work modes: Auxiliary Work (AUX Work), auto-in, manual-in, or ACW
- Timed ACW
- Agent request for supervisor assistance
- ACD call disconnect (Release button)
- Stroke counts
- Call work codes
- Forced entry of stroke counts and call work codes

[Agent capacity and related limits](#) on page 70 describes agent-capacity planning.

Note:

All of these agent capabilities are also supported through the CallVisor Adjunct/Switch Applications Interface (ASAI). For more information about the CallVisor Adjunct-Switch Application Interface, see *Feature Description and Implementation for Avaya Communication Manager*.

Administering Agent Call Handling

The following forms and fields are required to administer the Agent Call Handling feature.

screen	Field
Feature Access Code (FAC)	Feature Access Codes for ACD features
Hunt Group	Forced Stroke Count or Call Work Code Timed ACW Interval
Vector Directory Number	VDN Timed ACW Interval
Station (multi-appearance)	Button/Feature Button Assignments: <ul style="list-style-type: none"> ● manual-in ● auto-in ● aux-work ● after-call ● assist ● release ● work-code ● stroke-cnt Active Station Ringing (DCP, Hybrid) Idle/Active Ringing (Callmaster) VuStats
Stations (all)	Auto Answer
Attendant Console	Feature Button Assignments <ul style="list-style-type: none"> ● after-call ● assist ● auto-in ● aux-work ● manual-in ● release ● work-code ● stroke-cnt Auto Answer
Agent LoginID (EAS only)	All

Communication server controls

This section describes how the communication server controls agent work.

Note:

This information applies generally to ACD; see [Expert Agent Selection](#) on page 153 for more information on EAS.

This section includes the following topics:

- [Agent login and logout](#) on page 63
- [Login to a split \(non-Expert Agent Selection\)](#) on page 63
- [Login to a skill \(Expert Agent Selection\)](#) on page 63
- [Logout](#) on page 64
- [Agent answering options](#) on page 65
- [Automatic Answer](#) on page 65
- [Manual Answer](#) on page 65
- [ACD work modes](#) on page 65
- [Auxiliary Work mode](#) on page 66
- [Auto-in mode](#) on page 66
- [Manual-In mode](#) on page 66
- [After Call Work mode](#) on page 66
- [Timed After Call Work](#) on page 67
- [Timed ACW and VDN](#) on page 67
- [Cancelling Timed ACW](#) on page 67
- [Agent request for supervisor assistance](#) on page 68
- [Stroke counts](#) on page 68
- [Call work codes](#) on page 69
- [Forced entry of stroke counts and call work codes](#) on page 69
- [Agent capacity and related limits](#) on page 70
- [Callr-info display options](#) on page 72
- [Ringer-off control of auto-answer for non-ACD calls](#) on page 73

Agent login and logout

To receive ACD calls, an agent must log into the system. An agent can be logged into multiple splits/skills. If a hunt group is measured by Call Management System (CMS) or Basic Call Management System (BCMS) or is a skill, an agent must enter a login ID.

Login to a split (non-Expert Agent Selection)

To log in, an agent goes off-hook and dials the login feature access code (FAC), followed by the split number and the login ID, if required. If login is successful, the agent automatically enters Auxiliary Work mode for that split. The Auxiliary Work button lamp for that split lights steadily and the agent hears the confirmation tone.

If the split is measured, the system sends messages to CMS or BCMS that the agent (identified by login ID) has logged in and has entered Auxiliary Work mode.

Login to a skill (Expert Agent Selection)

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

Cancelled logins

Login is cancelled 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 a split number the agent is not assigned for.
 - Dials a split number the agent is already 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 cancelled and the agent receives a reorder tone if the system maximum number of agents are already logged in.

An EAS agent can be denied login to some of his assigned skills if the system maximum number of agent-skill pairs has been reached. The display of skills will show a * for each skill not logged in.

Logout

The agent should log out when he or she leaves for an extended period of time and is unavailable for ACD calls. If the split or skill is measured by CMS or BCMS and an agent logs out, a message is sent to the CMS or BCMS so that the agent's status is no longer measured. In a non-EAS environment, if an agent is logged into multiple splits, the agent should log out of each split.

When temporarily unavailable for calls, an agent should use Auxiliary work mode, rather than logging out. CMS or BCMS can continue tracking the agent's auxiliary work time.

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 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 he or she is not logged into.

If an agent is using a handset in Automatic Answer mode, the agent can log out simply by hanging up or turning off the headset. (This 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 he or she has logged into.

Agent answering options

An agent can answer ACD calls by using either a headset, handset, or speakerphone. You can assign an agent as either Automatic Answer or Manual Answer.

Note:

Use Automatic Answer with a headset. See [Agents with Automatic Answer](#) on page 75 for more information.

Automatic Answer

The information in this section applies to ACD and EAS environments.

An agent assigned to Automatic Answer hears zip tone and connects directly to incoming calls without ringing.

Note:

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

Manual Answer

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

ACD work modes

At any given time, an agent can be in one of four work modes:

- Auxiliary Work (AUX)
- auto-in
- manual-in
- After Call Work (ACW)

An agent can change work modes at any time.

To enter any work mode, an agent presses the button or dials the FAC for that mode, depending on what you have administered. If the agent has no active or held calls, the work-mode button lamp lights steadily and CMS or BCMS is informed of the agent's mode change. If the agent has active or held calls, the lamp flashes until all calls are dropped, then the new work mode's lamp lights steadily and CMS or BCMS is informed of the agent's mode change.

ACD call center features

The attempt is cancelled and the agent receives an intercept if the agent:

- Tries to enter a work mode for an invalid split or skill
- Tries to enter the work mode for a split or skill of which he or she is not a member
- Dials an invalid FAC

Auxiliary Work mode

An agent should enter Auxiliary Work mode whenever taking a temporary break. This makes the agent unavailable for ACD calls and removes them from the most-idle-agent queue. CMS and BCMS can continue to track the agent.

In a non-EAS environment, when an agent is in AUX Work mode for a particular split, the agent may be available for ACD calls to other splits that the agent is logged into, depending on the agent's state in those splits. Even in AUX, the agent is still available for non-ACD calls. CMS/BCMS is notified whenever an agent in AUX Work mode receives an incoming non-ACD call or makes an outgoing call. When an agent logs into a split, he or she automatically enters AUX Work mode for that split.

Note:

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

Auto-in mode

In auto-in mode, the agent automatically becomes available for answering new ACD calls upon disconnecting from an ACD call.

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.

After Call Work mode

An agent should enter ACW mode when he or she needs to perform ACD-related activities, such as filling out a screen as a result of an ACD call. The agent is unavailable for ACD calls to all splits/skills while in ACW mode. Communication server administration determines whether the agent remains in the Most Idle Agent queue while in ACW.

When an agent is in the Manual-In mode and disconnects from an ACD call, he or she automatically enters ACW mode. Although no longer available for ACD calls, the agent is still available for non-ACD calls. CMS or BCMS is notified whenever an agent in ACW mode receives an incoming non-ACD call or makes an outgoing call.

Timed After Call Work

With Timed ACW administered, an auto-in agent is immediately placed in ACW mode for a specific length of time after completing the currently-active ACD call. When the Timed ACW interval expires, the agent automatically returns to the auto-in work mode. If the agent activates ACW while not on a call, the agent is placed in ACW (not timed) mode regardless of whether the agent is in auto-in or manual-in mode.

Use Timed ACW to allow agents to rest between incoming ACD calls, or to pace agents when they have to complete work from the previous call within an allotted time. In addition, if you have Home Agent, use Timed ACW to allow agents additional time to dial a FAC to place themselves in a different work mode after the current ACD call is completed.

Timed ACW and VDN

You can administer Timed ACW for all calls to a split or skill and/or to a VDN. Any completed calls to the split or skill or to the VDN, including direct agent calls, are followed by a timed ACW when the answering agent is in auto-in work mode. If a VDN call routes to a split or skill, the Timed ACW interval for the VDN overrides the Timed ACW interval for the hunt group. VDN override applies to VDN-Timed ACW.

Cancelling Timed ACW

Timed ACW is cancelled 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 cancelled. 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 he or she 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 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 cancelled.

Ringling or held ACD call is dropped by the caller: If a ringing or held ACD call to an auto-in agent is dropped by the caller, Timed ACW is not activated.

Agent request for supervisor assistance

To request assistance from the split or skill supervisor, an agent, with or without an active ACD call, presses the Assist button or puts the call on hold and dials the Assist FAC plus the split or skill number. The agent must be logged into the split or skill. Assist generates 3-burst ringing at the supervisor's station. If a split or skill supervisor is not assigned, the agent receives intercept tone.

Attendants should press the Start button before pressing the Assist button. This allows them to later transfer the call. This rings like a priority call at the supervisor's set.

When the agent presses the Assist button, the following happens:

1. If the agent is active on an ACD call, the ACD call is automatically placed on hold and a call is placed to the split or skill supervisor. If the agent is not active on an ACD call, a call is automatically placed to the supervisor.
2. CMS or BCMS is notified of the request and the supervisor's display shows that the call is a request for assistance. This rings like a priority call at the supervisor's set.
3. The caller hears silence or music on hold.
4. After the agent has talked to 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 plus the split or skill number, the system handles the request as if the agent pressed the Assist button, except that the Assist call does not follow the supervisor's coverage path.

Stroke counts

Stroke counts allow you to record in CMS the number of times that a particular customer-related event occurs. For example, agents could press a button each time a customer requests information on a certain item.

Stroke counts are reported to CMS in real time. The communication server does not store stroke counts. Use stroke counts only when CMS is connected and you have defined ACD splits/skills to be measured by CMS.

Stroke counts allow agents to record up to nine administrator-defined events on a per-call basis. You can assign 10 Stroke Count button types. Stroke Count 0 is reserved for tracking Audio Difficulty or poor transmission quality.

For troubleshooting purposes, CMS records the equipment location of the trunk that the agent was using when he or she pressed the Audio Difficulty button. Make sure that agents are aware that pressing this does not improve audio transmission quality.

To enter a stroke count, an ACD agent presses a Stroke Count button while off-hook. The system validates that the agent is either active on an ACD call or in the ACW mode for an ACD split or skill. If yes, the feature lamp lights steadily for two seconds to indicate activation and the stroke count is sent to CMS. If not, the feature lamp flutters and no message is sent.

Call work codes

Call work codes are up to 16-digit sequences that ACD agents enter to record customer-related information. You define the codes for your site. Codes that agents enter are sent to CMS for storage for splits/skills measured by CMS and only when the link to the CMS is up. Agents must have multiappearance phones (for example, Callmaster) to enter call work codes.

To enter call work codes, the agent must be off-hook and either:

- On an ACD call
- In ACW mode after disconnecting from a call while in Manual-In mode remaining off-hook
- In Timed ACW after disconnecting from a call while in auto-in mode
- In auto-in mode and pending for ACW mode

The sequence of event is as follows:

1. The agent select Call Work Code (CWC) button.
2. The CWC lamp lights steadily and a C: prompt appears on the agent's display. The agent must wait for the ready indication before entering the call work code or the caller hears the touch-tone digits being dialed.
3. Agent enters up to 16 digits on the dial pad. The agent can press * to erase digits.
4. The agent presses # to send the code entry to CMS.
5. The Call Work Code lamp goes dark and the display returns to normal.
6. If the agent presses any feature button or hangs up during digit collection, the code entry is cancelled and data is not sent to CMS. The CWC lamp goes dark and the display is cleared.

Call work codes may be used by as many as 100 agents simultaneously. If 100 agents are simultaneously using this function, and another agent attempts to enter a call work code, the agent receives a display message to try again later.

Forced entry of stroke counts and call work codes

You can administer a split or skill so that agents must enter a stroke count and/or a call work code before becoming available for another call using Manual-In mode.

Note:

Multi-appearance phones or an attendant console are required for agents to enter stroke counts or call work codes.

To enter a stroke count and/or call work code, the agent must be on a call, or in ACW mode after releasing a call in Manual-In mode.

After releasing a call, the agent automatically enters ACW mode and cannot return to Manual-In mode until entering a stroke count or call work code. If the agent presses the Manual-In button or FAC before entering a stroke count or a call work code, the Manual-In lamp flutters or intercept tone is given.

ACD call center features

Once the agent enters a stroke count or call work code and presses the Manual-In button or FAC, he or she returns to Manual-In mode and the Manual-In lamp lights.

Any of the agent's splits/skills can have Forced Entry assigned. If the agent goes into Auxiliary Work mode in any split or skill, the Forced Entry requirement for all other splits/skills is removed.

Agent capacity and related limits

Agent Sizing adds an overriding capacity limit to the number of logged-in ACD agents. It can be used 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 agents limit applies to ACD agents in traditional or non-EAS ACD splits or in Expert Agent Selection (EAS) skills. Auto-Available split or skill (AAS) agent ports are logged in and counted when they are first assigned, while the non-AAS agents are counted when they actually log in. Each logged-in agent is counted as a single agent independent of the number of splits/skills logged in to for the Logged-in ACD agents limit. AAS and non-AAS agents are counted towards this limit whether they are BCMS/CMS measured or not.

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.

Avaya Business Advocate licensing

When an agent logs in, the *Logged-In Advocate Agents* license setting is counted only if *any* of the following fields are set as described in the table.

screen	Field	Set to
Login ID for the agent	Service Objective	y
	Call Handling Preference	percent-allocation
	Reserve Level	1 or 2
hunt group for the skill the agent logs into	Service Level Supervisor	y
	Group Type	pad
	Dynamic Queue Position	y

The **Service Objective** field setting on the hunt group screen is not used for Avaya Business Advocate agent counting. Only agents whose Login ID screen have the **Service Objective** field set to **y** are counted. Skills with Least Occupied Agent assignments of type ucd-loa or ead-loa are *not* counted as Avaya Business Advocate agent types starting with Communication Manager Release 9.

The agent sizing license limit is administered by authorized Avaya personnel. The Logged-in ACD Agents option (and Logged-in Advocate Agent Count) on the System-Parameters Customer-Options screen are set by the loaded license file. The maximum number of allowed logged-in ACD and Avaya Business Advocate agents is set to correspond to the configuration you purchase.

Agent sizing when agents work in shifts

For agent sizing, if you have agents working in shifts, you should purchase enough 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 could be denied login because too many agents are currently logged in. Additionally, the non-ACD and/or non-agent (AAS/VRU) use of Hunt Group resources must be considered. Call center managers need to be aware of their logged-in ACD agent and other related limits when adding agents to handle a traffic peak or when planning a special campaign. Some of the 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 supported is dependent on the upper limits that the system platform supports. The following limits must also be considered.

- Maximum Hunt Group members
 - Non-ACD members include hunting 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/skills, each combination (pair) is counted as a member (e.g., an EAS agent logged into 4 skills or a non-EAS agent assigned to 4 splits counts as 4 members). Non-EAS ACD members are counted when assigned (note that many more splits can be assigned to an agent than can be logged into but each agent-split pair is still counted towards the limit). EAS ACD members are counted when they 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 done as above for maximum Hunt Group members.
- Additional traditional ACD (non-EAS) agents limits:
 - Maximum logged-in agents system limit
 - Maximum splits an agent can log into

ACD call center features

- Additional EAS limits:
 - **ACD members (skill pairs) 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 pre-assigned
 - **Agent login IDs logged-in (staffed) 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
- Call Management System (CMS) logged in ACD members (agent-split or skill pairs) limits assigned. Both a Avaya setup and customer-administered limit is assigned in CMS. These limits are related to the CMS memory/hardware configuration equipped and are passed over the link to the communication server to reduce/set 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 reorder tone or is otherwise denied log in. Also with EAS, an agent logging in may 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.

Callr-info display options

This feature allows administrators to decide when an agent's station display is cleared of caller information (Callr-info). Options include:

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

Reason to use. This feature is designed to meet U.S. government privacy requirements as specified in the Health Insurance Portability and Accountability Act (HIPAA). HIPAA has a specification that medical records cannot be left where they can be viewed by others.

Call centers also have a requirement that agents can see the data on the station display when the agent goes into ACW mode. Agents must be able to see the data in order to use it for other purposes without having to write it down.

Administering Callr-info. To administer the Callr-info display options:

1. Go to the **feature related system parameters** screen.
2. Set one of the following options on the **Clear callr-info** field.

Option	Clears the display of the existing call information
next-call	When the next call is received. This is the default setting.
on-call-release	As soon as the call is released. A call is released when the agent presses the release button or when the caller disconnects from the call.
leave-ACW	As soon as the call is released except when the agent is put into the ACW state. The station display is cleared when the agent leaves ACW to go into any other state including when the agent is forced out of ACW.

This feature applies only to those stations supporting a two-line display, such as Callmaster IV, Callmaster VI, or 8434D stations. It also applies to an IP Softphone or IP Agent that is emulating a two-line display telephone.

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

Starting in Release 3.1, administrators can allow agents to use the ringer-off button to prevent non-ACD auto-answer calls from ringing.

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

- The **Auto-Answer** field on the **Agent LoginID** screen was set to **all**, or
- The **Auto-Answer** field on the **Agent LoginID** screen was set to **station** and the **Auto-Answer** field on the **Station** screen was set to **all**

Agents could not use the ringer-off button to prevent this single ring from occurring.

Reason to use: Some call centers prefer quiet environments where they do not want to hear the audible ring for every non-ACD call that is received.

Administration: Set the **Allow Ringer-off with Auto-Answer** field on the **feature-related system parameters** screen to **y**.

Agent Call Handling considerations

This section includes the following topics:

ACD call center features

- [Release button](#) on page 74
- [Timed ACW](#) on page 74
- [Non vector-controlled splits/skills](#) on page 74
- [Agents logged into multiple splits/skills](#) on page 75
- [Agents with Automatic Answer](#) on page 75
- [Callmaster telephones](#) on page 75
- [Agents assigned to hunt-group and ACD calls](#) on page 75

Release button

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

Timed ACW

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.

Non vector-controlled splits/skills

For non vector-controlled splits/skills, the last available agent in a split or skill cannot enter Auxiliary Work mode if any calls remain in the queue. (However, the agent can log out.)

When the last available agent tries to enter Auxiliary Work mode, the following occurs:

- The Auxiliary Work button flashes indicating the change is pending.
- New calls on the ACD split or skill either receive 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 Auxiliary Work button lamp flashes until the queue is empty. The telephone then enters Auxiliary Work mode and the associated lamp lights steadily.

Agents logged into multiple splits/skills

If an agent is logged into multiple splits/skills, the agent may become unavailable for calls to one split or skill because of activity at another split or skill. For example, if an agent enters After Call Work mode for one split or skill, the agent becomes unavailable for calls to other splits/skills.

An agent should not log into a split or skill while a call is on hold at the extension.

Agents with Automatic Answer

Agents who use Automatic Answer should use a headset. The agent hears zip tone through the headset and automatically connects to a call.

If either the incoming trunk group or the agent's extension is data-restricted, the agent does not hear zip tone. Therefore, do not assign data-restriction to a headset user's extension.

It is not recommended that you use Automatic Answer with a handset or speakerphone. The handset or speakerphone must be off-hook (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, not ringing.

Callmaster telephones

Calls for Callmaster digital phones and attendant stations are announced by double tones. The tones that are doubled are zip (Auto-Answer ACD agent calls) and Incoming Call ID (for End of VDN of Origin announcements and all other Auto-Answer calls). The user hears part of the first tone and all of the second tone.

Agents assigned to hunt-group and ACD calls

Do not use agents for hunt-group calls and ACD split or skill calls simultaneously. Otherwise, all of the calls from one split or skill (either ACD or hunt-group) are answered first.

The oldest call-waiting termination is supported only for agents who are servicing ACD calls only.

Agent Call Handling interactions

Abbreviated Dialing: Assign Abbreviated Dialing buttons to make agent login easier. You can program an Abbreviated Dialing button to dial an access code, split number, and/or agent login ID. You can use Autodial feature buttons to assign login and logout feature buttons.

Auto-Available split or skill (AAS): An AAS cannot be administered for Timed ACW.

ACD call center features

Bridging: ACD split or skill calls are not bridged.

Station calls are bridged and agents are able to bridge onto them. If an agent bridges onto a call, the call is considered a non-ACD extension-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/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 or 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 or split or skill.

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.

Call Work Codes: The CWC 100-agent limit is shared with reason codes. Therefore, no more than 100 agents can simultaneously enter either a call work code or reason code.

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.

Avaya 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.

Conference: If an agent receives an ACD call through a VDN and then conferences in other agents, the agents added to the call use the Timed ACW interval associated with the number dialed to conference them. An ACD agent on conference with more than three parties may cause inaccurate CMS measurements.

Expert Agent Selection: 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 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.

Multiple Call Handling: If MCH calls are on hold at an agent's 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. See [Multiple Call Handling](#) on page 202 for detailed information.

Transfer: If an agent receives an ACD call through a VDN and then 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 his or her 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 or split or skill that first received the call.

VDN Override: If a VDN has VDN Override set to no and the vector routes a call to a second VDN, the first VDN's Timed ACW interval is used for Timed ACW. If VDN Override is set to yes, the second VDN's Timed ACW interval is used.

If no interval is set for the second VDN, no Timed ACW is associated with the call.

Voice Response Integration: 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.

Auto-Available Split/Skill

This section includes the following topics:

- [About AAS](#) on page 78
- [Administering AAS](#) on page 78
- [AAS detailed description](#) on page 79
- [AAS considerations](#) on page 80
- [AAS interactions](#) on page 80

About AAS

Auto-Available Split/Skill (AAS) allows members of an ACD split or skill to be in auto-in work mode continuously. An agent in auto-in work mode becomes available for another ACD call immediately after disconnecting from an ACD call.

Use AAS to bring ACD agents back into auto-in work mode after a system restart. Although not restricted to such, this feature is intended to be used for splits/skills containing only nonhuman members - for example, recorders or voice response units (VRUs).

Administering AAS

The following forms and fields are required to administer the AAS feature.

Screen	Field
Hunt Group	AAS
Agent LoginID (EAS only)	AAS

Procedure

To administer AAS:

1. Verify that the ACD field is set to y on the System Parameters Customer-Options screen.
If this field is not set to y, contact your Avaya Services representative.
2. Verify that the ACD field screen is set to y on the Hunt Group.

AAS detailed description

This section includes the following topics:

- [Agent login with AAS](#) on page 79
- [Agent logout with AAS](#) on page 79

Agent login with AAS

With AAS, ACD splits or skills generally operate as usual. The major difference is in how work modes are handled.

For splits/skills with AAS, agents are automatically logged in under the following circumstances:

- Call Management System (CMS) completes an Agent Move request into an Auto-Available split or skill.
- A maintenance-busied-out port, which is defined as an agent in an Auto-Available split or skill, is released.
- The system reinitializes and requires agents to log in again.
- You administer a split or skill on the Hunt Group screen as AAS = y.
- You administer an agent into an existing AAS split or skill.

Once an agent is logged into an Auto-Available split or skill, it is immediately moved to the auto-in work mode and subsequent requests to change out of that mode are denied.

Agent logout with AAS

For splits/skills with AAS, agents are automatically logged out under the following circumstances:

- CMS completes an Agent Move request out of an Auto-Available split or skill.
- The Auto-Available agent's port is unavailable because maintenance is being performed.
- You administer a split or skill as AAS = n.
- You remove an agent from an existing AAS split or skill.
- Redirection on No Answer (RONA) redirects a call that the agent has not answered after an administered number of rings.

AAS considerations

- AAS is intended primarily 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 restart. AUDIX is incompatible with AAS because it uses BX.25 messages to automatically activate its ACD agent ports after a PBX restart.
- Because AAS is intended for nonhuman agents, do not administer an Auto-Answer telephone as a member of an AAS.
- AAS is not intended for any agent port hardware that can change its work mode state since a request to move to any state other than auto-in is denied; however, administration of such telephones is not blocked.

AAS interactions

This section includes the following topics:

- [Auto-Answer](#) on page 80
- [CMS](#) on page 81

Auto-Answer

Do not administer an Auto-Answer telephone as a member of an AAS.

Auto-Answer was originally implemented for human agents. If a non analog telephone is administered as Auto-Answer and that telephone is logged into a split or skill, when the telephone goes on-hook, it is logged out.

Agents at analog telephones defined as Auto-Answer who are logged into a split or skill must dial a log-out FAC to log out. If a telephone is a member of an AAS, a log-out FAC is denied. To log the agent out, you must either remove the agent from the split or skill when not active on a call or busy-out the physical extension.

If an agent in an AAS with an Auto-Answer telephone goes off-hook, the telephone is logged into any Auto-Available splits of which it is a member. To log out of the AAS splits/skills, the agent goes on-hook, is placed in AUX work mode, and then presses the RELEASE button on non analog sets or disconnects on analog sets. Because agents are not placed immediately in auto-in work mode, they may place personal or emergency calls rather than answering ACD calls that may be in queue.

CMS

For each agent, AAS notifies CMS of any login, logout, or change into the auto-in work mode. In a non-EAS environment, an AAS agent is identified to CMS with a login ID equivalent to the agent's administered extension. With EAS, the AAS login ID and port are assigned on the Login ID screen.

With CMS Move Agent, you can move a member from one AAS split or skill to another while that member is logged in.

Automatic Call Distribution

This section includes the following topics:

- [About ACD](#) on page 82
- [Administering ACD](#) on page 86
- [ACD detailed description](#) on page 88
- [Avaya Business Advocate](#) on page 98
- [DAC considerations](#) on page 147
- [Direct Agent Call \(DAC\) interactions](#) on page 150

About ACD

Automatic Call Distribution (ACD) allows incoming calls to connect automatically to specific splits/skills. An ACD split or skill is simply a hunt group that is designed to receive a high volume of similar calls. ACD hunt groups with basic ACD, but non-EAS, are called splits. ACD hunt groups with basic ACD with EAS are called skills. Calls to a specific split or skill are automatically distributed among the agents, or hunt group members, assigned to that split or skill. Calls queue to the split or skill until an agent is available.

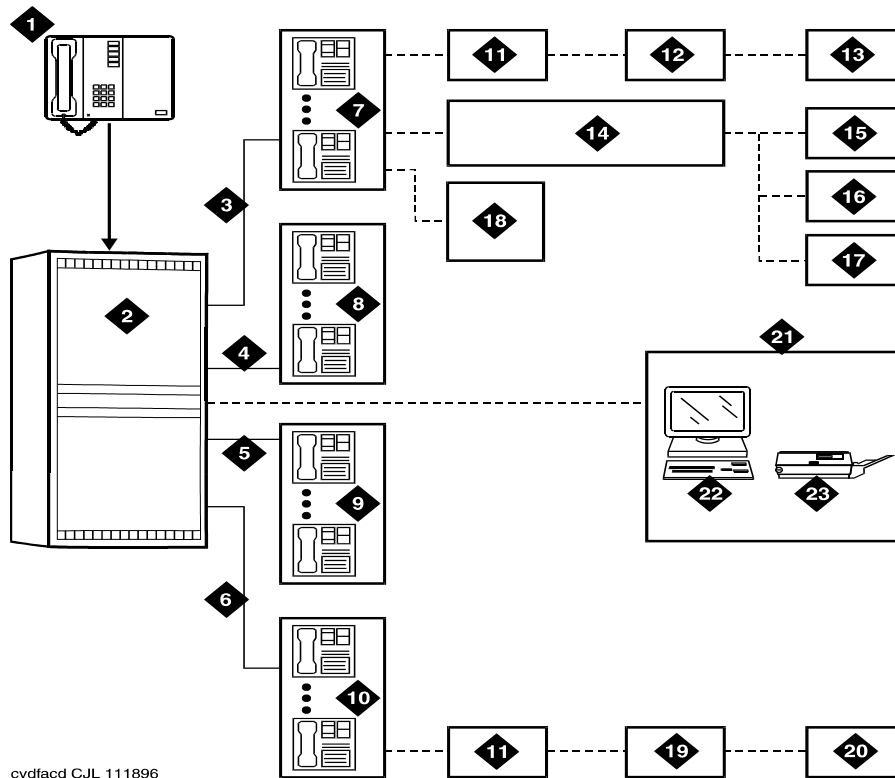
An ACD agent can be a physical telephone extension, an individual attendant extension, or, in an Expert Agent Selection (EAS) environment, an agent login ID. An agent can be logged into multiple splits/skills. However, in a non-EAS environment, agents can be logged into only one split if that split is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

You can assign a supervisor to each split or skill. The split or skill supervisor can listen in on agent calls, monitor the split or skill queue status, and assist agents on ACD calls. Although supervisors can assist agents on ACD calls, the supervisors do not normally receive ACD calls unless they are also members of the split or skill.

If you have Call Management System (CMS) or Basic Call Management System (BCMS), you can measure and create reports on the status of ACD agents, splits/skills, and trunks. See [Agent Call Handling](#) on page 60 and [Reporting adjuncts](#) on page 247 or [Basic Call Management System](#) on page 107 before setting up your ACD splits. See [Agent Call Handling](#) on page 60 for detail on administering agent functions and operations.

The following figure depicts a typical ACD arrangement.

Typical ACD Arrangement



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- | | |
|---|--|
| 1. Incoming calls | 13. Announcement 2 |
| 2. ACD switch | 14. Intraflow (Call Coverage) |
| 3. Trunk group 1 | 15. Split 2 Personal Travel (3rd choice) |
| 4. Trunk group 2 | 16. Split 3 Group Travel (2nd choice) |
| 5. Trunk group 3 | 17. Split 4 General Information (1st choice) |
| 6. Trunk group 4 | 18. Supervisor (with Service Observing) |
| 7. Split 1 Business Travel (10 agents) | 19. Announcement |
| 8. Split 2 Personal Travel (8 agents) | 20. Disconnect |
| 9. Split 3 Group Travel (5 agents) | 21. Call Management System (CMS) |
| 10. Split 4 General Information (15 agents) | 22. Terminal |
| 11. Queues | 23. Printer |

Configuring a call center or call center with EAS checklist

This section includes the following topics:

- [Step 1: Configure the basics](#) on page 84
- [Step 2: Define the applications](#) on page 84
- [Step 3: Define trunks](#) on page 85
- [Step 4: Define hunt groups \(skills\)](#) on page 85
- [Step 5: Define agents](#) on page 85
- [Step 6: Define caller treatments and backup treatments](#) on page 85
- [Step 7: Configure and record announcements](#) on page 85
- [Step 8: Define vectors](#) on page 85

Step 1: Configure the basics

To configure the basics:

1. Confirm the configured options for the license file.
2. Set the system date and time.
3. Configure the dial plan.
4. Define the Feature Access Codes.
5. Define the Class of Service (COS).
6. Define the Class of Restrictions (CORs).
7. Configure Abbreviated Dialing.

For detailed information about how to perform these steps, see *Feature Description and Implementation for Avaya Communication Manager*.

Step 2: Define the applications

To define the applications:

1. Define caller types.
2. Define incoming called numbers.
3. Define the corresponding VDNs.
4. Define the skills needed to support call types.

For detailed information about how to perform these steps, see *Avaya Call Center Call Vectoring and EAS Guide*.

Step 3: Define trunks

For detailed information about how to perform this step, see *Feature Description and Implementation for Avaya Communication Manager*.

Step 4: Define hunt groups (skills)

Defining hunt groups (skills) includes all of the attributes. For detailed information about how to perform this step, see *Feature Description and Implementation for Avaya Communication Manager*.

Step 5: Define agents

To define agents:

1. Determine station types.
2. Define stations using the station screen. Also select auto-answer or manual-answer.
3. Define work modes. For example, auto-in, manual-in, or mixed environment.
4. Assign feature or work buttons.
5. Administer the Login ID forms.
 - a. Define the skills.
 - b. Define the attributes.
6. Determine what options (system, VDN, hunt group, or agent, and so on) are required for the call center, and what applications you want supported.

For detailed information about how to perform these steps, see *Feature Description and Implementation for Avaya Communication Manager* and *Avaya Call Center Call Vectoring and EAS Guide*.

Step 6: Define caller treatments and backup treatments

For detailed information about how to perform these steps, see *Avaya Call Center Call Vectoring and EAS Guide*.

Step 7: Configure and record announcements

For detailed information, see [Recorded announcements](#) on page 415.

Step 8: Define vectors

For detailed information about how to perform these steps, see *Avaya Call Center Call Vectoring and EAS Guide*.

Administering ACD

The following forms and fields are required to administer a call center.

Screen	Items or fields
System-Parameter Customer-Options	<p>Use the display system-parameters customer-options command to verify that the options are active for the ACD and any of the following:</p> <ul style="list-style-type: none"> ● Features provided in the call center software package ● Any purchased a la carte features
Date and Time	<p>Use the set time command. If you need to configure the rules, use the change daylight-savings rules command. For Linux systems, use the Integrated Management Maintenance Web Interface to configure time.</p>
Dial Plan	<ul style="list-style-type: none"> ● Feature Access Codes ● Extensions for VDNs, hunt groups, agent stations and logins, announcements, and so on.
Feature-Related System-Parameters	<ul style="list-style-type: none"> ● System options for general PBX features ● UCID ● ISDN and SIP system options for trunks ● Path replace while in queue or vectoring ● CPN/ANI/ICLID ● Vector Disconnect Timer ● Zip Tone options ● Station display options ● EAS activation and related options ● Vectoring options ● Call Prompting options ● BSR options ● OCM options - Outbound Call Management ● Agent and Call Selection options ● CMS release ● BCMS options

Screen	Items or fields
Feature Access Codes (FACs)	<ul style="list-style-type: none"> ● Announcement Access Code ● Login Access Code ● Logout Access Code ● After Call Work Access Codes and other work mode codes ● Service Observing codes ● Add/remove agent skill ● Remote logout of agent ● ...and so on
Abbreviated Dialing	For login and logout buttons and other agent-related buttons only defined using FACs.
Class of Service	Feature access capabilities for agent and supervisor stations (recording announcement using telephone required console permissions)
Trunk Groups	<p>All</p> <p>In the Group Number field, assign consecutive Group Numbers to trunk groups when you are using two or more trunk groups to support ACD applications.</p>
Stations	<p>Feature buttons</p> <p>Agent answering options, and so on</p>
Hunt Groups	<p>All</p> <p>When the ACD field is set to y, complete the fields that apply. In the AAS field, enter y to enable Auto-Available Split/Skill. See Auto-Available Split/Skill on page 78 for more information.</p>
Agent Login ID (EAS only)	All
Media Gateway	Administer gateway-announcement board
Enable announcement-board	As required
Announcement/Audio Sources	All
VDNs	<ul style="list-style-type: none"> ● Vector numbers ● VDN-related parameters
Call Vectors	Programming for call handling

ACD detailed description

For a detailed description of call distribution methods, see [ACD basics](#) on page 19.

This section includes the following topics:

- [Table of call distribution methods](#) on page 88
- [Basic ACD \(non-EAS without vectoring\) queuing and announcements](#) on page 89
- [First announcement](#) on page 89
- [Forced first announcement](#) on page 90
- [Second announcement](#) on page 90
- [Forced disconnect](#) on page 91
- [Announcement rules](#) on page 91
- [Entering the queue](#) on page 91
- [Priority queuing](#) on page 92
- [Queue status indications](#) on page 92

Table of call distribution methods

The following table summarizes the different call distribution methods.

WHEN agents are available, a call arrives, and the agent selection method is:	THEN the communication server selects:	This distribution method is available with:
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 (non-EAS without vectoring) queuing and announcements

Starting with Communication Manager 2.1, the system automatically allocates queue slots. The queue slot pool allows callers to wait for an agent to become available. 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 Auxiliary Work mode

The caller gets a busy signal (or busy coverage, if administered) unless a call comes in using an automatic-in Central Office (CO) facility. In this case, the caller hears ringback from the CO and the system continues trying 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 on how announcements are affected by call forwarding and call coverage, see *Feature Description and Implementation for Avaya Communication Manager*.

Related topic

For more information, see [Dynamic queue slot allocation](#) on page 25.

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 this 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 then the first announcement. The system then tries to connect the call to an agent.
- If the announcement is busy and has no queue, the system waits 10 seconds and then tries to access the announcement.
- If the announcement is busy and has a queue, then:
 - If the queue is full, the system waits 10 seconds, then tries to access the announcement.
 - 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 (for example, it may have been deleted), then the system tries 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 announcements 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 these calls queue long enough, they hear first and second announcements.
- 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. These calls can receive a forced first announcement at the original split, if administered, but not at the split they 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:

ACD call center features

- 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 a calling party's Class of Restriction (COR).
- 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. For more information, see [Information Forwarding](#) on page 178.

Queue status indications

You can assign queue status indications on agent or supervisor telephones or consoles for ACD calls in queue. For more information, see [Queue Status Indications](#) on page 209.

Avaya Agent Deskphone 16CC

This section provides an overview of the Avaya Agent Deskphone 16CC. For more information, refer to *Avaya Agent Deskphone 16CC User Guide* and *Avaya Agent Deskphone 16CC Administrator Guide*.

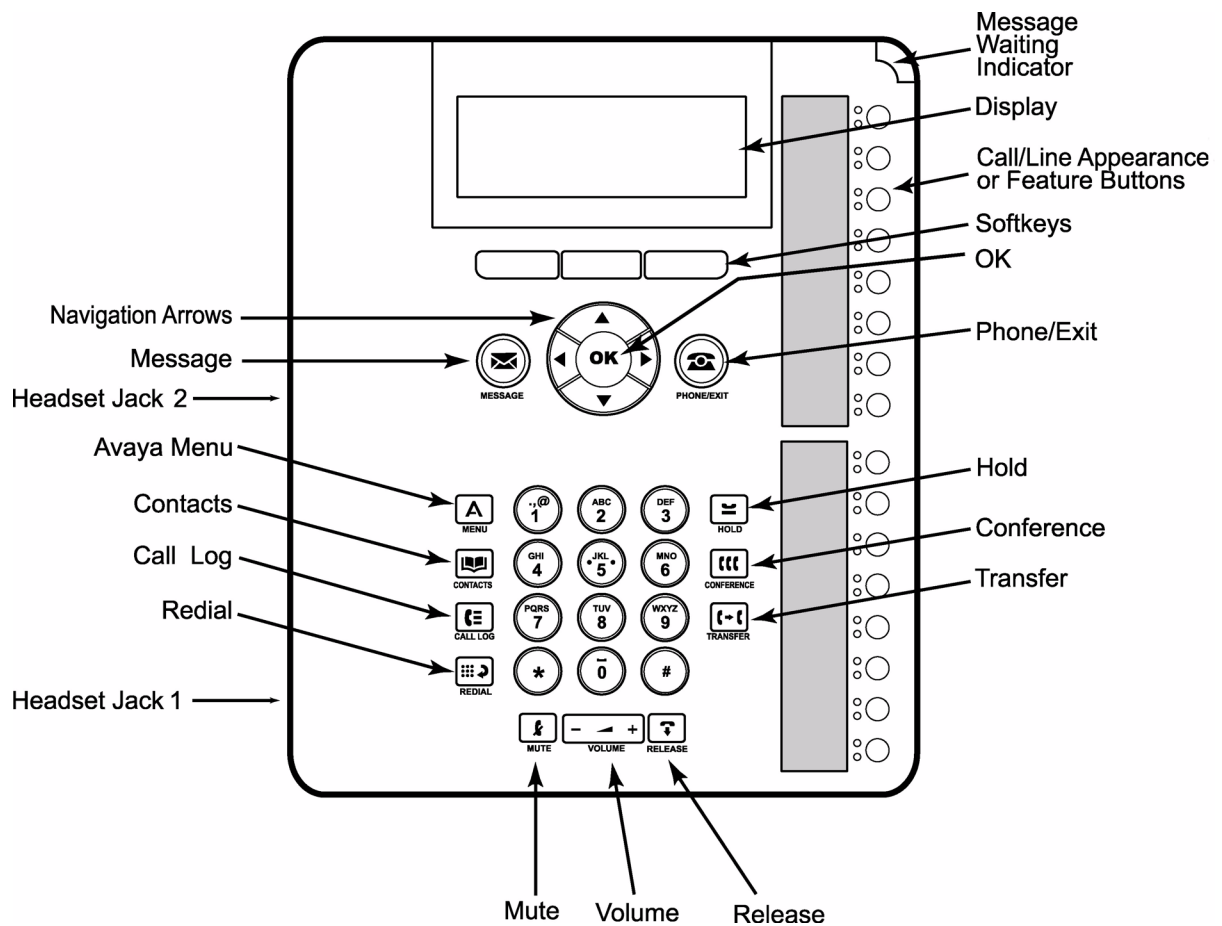
This section includes the following topics:

- [About the Agent Deskphone 16CC](#) on page 93
- [16CC basic feature support](#) on page 94
- [16CC support details](#) on page 95
- [16CC value-added capabilities](#) on page 96
- [16CC hardware features](#) on page 96
- [16CC software features](#) on page 97
- [Administering the 16CC](#) on page 97

About the Agent Deskphone 16CC

The Avaya Agent Deskphone 16CC is a new multiline SIP deskphone for Expert Agent Selection (EAS) agents that works with Avaya Communication Manager call center systems. The 16CC has many features and capabilities including a Phone screen to view and manage calls, a Contacts list, a Call Log, feature buttons to change agent work mode and status, and a menu of options and settings to customize the phone.

The following illustration shows the layout of the 16CC phone features and functions.



⚠ Important:
 The following sections list all the phone and Call Center features and functions that are supported on the 16CC phone. These lists are intended to be complete. No other Call Center features or functions, available with non-SIP phones, are supported with Call Center Release 5.0 and the 16CC phone.

16CC basic feature support

- EAS Agent log in and log out
- Manual-In and Auto-In work modes
- Auxiliary (Aux) work and After Call Work (ACW) feature buttons
- Reason codes (Aux and log out)
- Direct agent calling support
- Auto answer and manual answer

- VDN of Origin (Whisper) announcements
- Message Waiting Indication (MWI) for the phone physical extension only
- Multiple Call Handling
- Headset only
- IP connectivity failure detection (ROOF)

16CC support details

- Agent log in with optional password entry and loginID message waiting confirmation tone implemented using a specific Login softkey on the phone
- Agent log out with optional Logout Reason Code entry implemented using a specific Login softkey on the phone
- Options, skill assignments, and so forth defined for the agent are downloaded to the phone at login
- Change of work mode, Auto-In, Manual-In, or both using assigned feature buttons
- Change of work state, Auxiliary (Aux) work with optional Aux Work Reason Code entry and ACW, using assigned feature buttons
- Support of forced or requested Reason Code options
- Pre-assigned Reason Code Aux Work buttons as well as the generic unassigned Aux Work feature button with prompted entry of Reason Code
- Fixed (hard) Release button as well as soft Drop button for dropping the last added party during a conference operation
- Manual answer by way of appearance alerting or optional auto answer with zip tone call delivery with support of auto answer options
- Support of existing incoming call displays or equivalent, for example, a= 303-536-1234 -> *vdn_name*
- Indication of waiting Direct Agent Calls
- Visual and audible indications to agent with most being provided inband, in the audio path from Communication Manager, except for log in confirmation and denial indications, which are provided by the phone
- Access to some Call Center functions using Feature Access Codes with tones and other feedback provided inband
- VDN of Origin Announcements option support
- Communication Manager forced change of state to ACW (Manual-In or Timed ACW) or Aux Work (Maximum Agent Occupancy and so forth) or log off (Remote Logout, forced logout from Aux or by clock time and so forth)

ACD call center features

- Currently supported Advanced SIP Telephony endpoint feature capabilities including Hold, Conference, Transfer, Call Pickup, Call Park, Mute, Date and Time display, Message Waiting Indication (for station extension only), Send-All-Calls, Elapsed Call Timer, and multiple call appearances
- Up to 13 digit extension support (Communication Manager R4.0 Expanded Dial Plan)
- Agents using 16CC phones can be service observed by someone using a non-SIP phone
- Connectivity failure for SIP stations can be detected using the Redirection on OPTIM Failure (ROOF) feature, which is similar to Redirection on IP Failure (ROIF) for H.323 IP stations. For more information, see [Redirection on No Answer](#) on page 222, [Redirection on IP Failure](#) on page 235, and [Redirection on OPTIM Failure](#) on page 239.
- ASAI/CTI change of agent work mode or work state including login and logout
- Third party call control using ASAI/CTI support for the 4620SIP and 4620SIPCC set types
- Full reporting adjunct agent tracking support, CMS, Avaya IQ, or both

16CC value-added capabilities

- Button operation for log in and log out
- Text display, in addition to lamp indications, of work mode, work state, and type of call (rc is the entered reason code)
 - Auto-In versus Manual-In (Auto or Man)
 - Agent or call state (Avail, Incoming, Incoming-ACD, Incoming-DAC, On-ACD, On-DAC, Hold, On-ACD-Hold, On-DAC-Hold, Aux, Aux rc, Aux Pend, Aux rc Pend, ACW, and ACW Pend)
 - Waiting DAC call indication (DAC)
- Ability to display a full list of assigned skill numbers (initial and after changes) with alert when skills change
- Text display of denial or other error conditions
- Ability to work from home using a separate VPN hardware device called a VPN brick
- VPN brick can be daisy-chained behind a personal computer so computer and phone can share a single LAN connection

16CC hardware features

- Backlit display that measures 3.5 inches diagonally with an adjustable display angle
- Four line display with up to 24 characters per line
- Sixteen line appearance or feature buttons (in two groups of eight) with red and green LEDs and paper labels
- Four-way navigation cluster and OK button

- Three context sensitive softkeys
- Twelve dedicated function buttons
- Application access for Messages, Contacts, Call Log, and telephony functions
- Options and settings: Avaya Menu button
- Telephony functions: Redial, Release, Mute, Hold, Conference, Transfer, Volume control
- One way, receive-only internal speaker for alerting and error indications
- Message waiting indicator (for physical extension only)
- Fixed position wedge stand
- Ethernet 10/100 line interface with a switched 10/100 port for co-located laptop or personal computer
- PoE 802.3af class 2 device, which also supports a local power supply
- Two headset jacks, agent and supervisor, conveniently located on side of phone

16CC software features

- Contacts application
- Call log
- SIP protocol support
- Standards-based codec support for G.711, G.726, and G.729A/B
- User interface support for the following languages: English, Parisian and Canadian French, Latin American and Castilian Spanish, German, Italian, Dutch, and Brazilian Portuguese

Administering the 16CC

The 16CC phone uses Avaya Outboard Proxy SIP (OPS) features on the trunk side of the Avaya Communication Manager whereas H.323 phones use features on the line side of Communication Manager. The 16CC is configured as a 4620SIPCC phone type. It uses the existing 4620 station forms along with the AST OPTIM off-PBX telephone integration screen (OPS type only) in the same manner as other SIP phones, except that multiple assignments such as OPS plus EC500 are not supported. Communication Manager R5.0 also has a new 4620SIP alias station type for non-Call Center SIP phone support.

The server requirements for the 16CC are Avaya Communication Manager R5.0 or later, SIP Enablement Server R5.0 or later, and an HTTP file server. For more information about setting up and using the 16CC phone, see *Getting Started with Avaya Call Center 5.0 and Avaya Agent Deskphone 16CC* and *Avaya Agent Deskphone 16CC User Guide*.

Avaya Business Advocate

This section provides an overview of Avaya Business Advocate. For extensive information on implementing and using Avaya Business Advocate, refer to the *Avaya Business Advocate User Guide*.

This section includes the following topics:

- [About Avaya Business Advocate](#) on page 98
- [Administering Avaya Business Advocate](#) on page 99
- [Call and agent selection](#) on page 101

About Avaya Business Advocate

Avaya Business Advocate is a collection of features that provide flexibility in the way a call is selected for an agent in a call surplus situation and in the way that an agent is selected for a call in an agent surplus situation.

Note:

Avaya Business Advocate requires the Expert Agent Selection (EAS) feature to be enabled.

Avaya Business Advocate provides predictive and adaptive methods for call centers that address three fundamental questions in terms of how the most expensive resource of the center, its agents, are used every time a call is handled.

What should this agent do next?

Avaya Business Advocate decides what the agent should do after he or she becomes available and calls are waiting in queue. With Avaya Business Advocate, this decision does not come from executing a set of pre-programmed directives such as, the highest priority or oldest-waiting call. Such a fixed plan of attack considers nothing in terms of consequences. Instead, Avaya Business Advocate understands the consequences of its choices and the business objectives for each type of call.

Which agent should take this call?

Avaya Business Advocate decides which agent should take a call when there is more than one agent waiting for the call. Avaya Business Advocate can make this choice so that workloads are distributed fairly across the agents to eliminate hot seats. Avaya Business Advocate can also promote fairer opportunities for compensation by delivering a predetermined mix of calls to agents.

Does the call center need to adjust its operations?

Avaya Business Advocate continuously evaluates the performance of the call center and makes adjustments accordingly. Avaya Business Advocate can prevent callers from waiting too long, and makes sure that the call center consistently meets service-level goals.

Administering Avaya Business Advocate

Based on the needs and challenges of your call center, you will determine which combination of call and agent selection will give you the best results and administer those methods on the communication server. See the [Call and agent selection](#) on page 101 for information about these methods.

You need to make several decisions about how to implement Avaya Business Advocate. Some of these decisions affect your call center system wide, while others affect particular Vector Directory Numbers (VDNs), skills, or agents.

The following table lists the features that are available with Avaya Business Advocate and Dynamic Advocate, the level of impact for implementing those features, and where the features are administered on the communication server.

Feature	Decision level	Administered screen
Least Occupied Agent:		
<ul style="list-style-type: none"> LOA (Group Type) 	Skill	Hunt Group screen
<ul style="list-style-type: none"> ACW Considered Idle 	System	Feature-Related System Parameters screen
Percent Allocation		
Percent Allocation (call handling preference)	Agent	Agent LoginID screen
PAD (group type)	Skill	Hunt Group screen
<ul style="list-style-type: none"> Expected Call Handling Time 	Skill	Hunt Group screen
<ul style="list-style-type: none"> Dynamic Percentage Adjustment 	Skill	Hunt Group screen
Service Level Target	Skill	Hunt Group screen
ACW Considered Idle?	System	Feature-Related System Parameters screen

ACD call center features

Feature	Decision level	Administered screen
Auto Reserve Agents	System	Feature-Related System Parameters screen
Dynamic Queue Position		
Dynamic Queue Position	Skill	Hunt Group screen
Service Objective	VDN	Vector Directory Number screen
Service Objective		
Service Objective (activate for agent)	Agent	Agent LoginID screen
Service Objective (set target objective)	Skill	Hunt Group screen
Call Selection Measurement (CWT or PWT)	System	Feature-Related System Parameters screen
Service Level Supervisor		
Service Level Supervisor (administer for skill)	Skill	Hunt Group screen
Activate on Oldest Call Waiting	Skill	Hunt Group screen
Call Selection Override	System	Feature-Related System Parameters screen
	Skill	Hunt Group screen
Other		
Overload Thresholds	Skill	Hunt Group screen
Dynamic Threshold Adjustment	Skill	Hunt Group screen
Service Level Target	Skill	Hunt Group screen
Reserve Agents	Agent	Agent LoginID screen
Predicted Wait Time (PWT)	System	Feature-Related System Parameters screen
Call Handling Preference (call selection method: Greatest Need, Skill Level, Percent Allocation)	Agent	<ul style="list-style-type: none"> ● Agent LoginID screen
Group Type (agent selection method: UCD-MIA, EAD-MIA, UCD-LOA, EAD-LOA, PAD)	Skill	<ul style="list-style-type: none"> ● Hunt Group screen

Call and agent selection

This section includes the following topics:

- [Call selection](#) on page 101
- [Agent selection](#) on page 101
- [Automated agent staffing adjustments](#) on page 101
- [Call selection at a glance](#) on page 102
- [Agent selection at a glance](#) on page 102
- [Combining agent and call selection methods](#) on page 103
- [Different needs within a call center](#) on page 104
- [Feature compatibility](#) on page 105
- [Call selection methods \(call handling preferences\)](#) on page 105
- [Agent selection methods \(hunt group types\)](#) on page 106
- [Feature combinations to avoid](#) on page 106

Call selection

Call selection methods are used when calls are in queue and an agent becomes available. This is known as a call surplus condition. During such conditions, the communication server considers the call selection method that is administered for the agent on the Agent LoginID Screen to determine which skill to serve. Once a skill is identified, the call at the head of that queue is selected and delivered to the agent. Call selection is based on such things as call handling preference, call selection measurement, and the use of service objectives.

Agent selection

Agent selection methods are used when there are one or more available agents for an incoming call. This is known as an agent surplus condition. Agent selection methods are administered as a hunt group type for the skill. Avaya Business Advocate allows you to select agents according to occupancy, idleness, individual skill level, and the percentage of time that you want the agent to spend serving each skill.

Automated agent staffing adjustments

Avaya Business Advocate provides you with options that automate staffing during call center operation. These methods simplify call center management and eliminate the need for moving agents from skill to skill to ensure coverage as call conditions change.

ACD call center features

Avaya Business Advocate offers you the ability to assign reserve agents and set overload thresholds to determine when those reserve agents will be engaged. The Dynamic Advocate feature, known as Dynamic Threshold Adjustment, takes this a step further by automatically adjusting the thresholds as needed to help maintain the service levels you defined.

The Dynamic Percentage Adjustment feature, gives you the ability to automate adjustments to predefined allocations for your agents' time to maintain defined service levels. Auto Reserve Agents, another feature that is new with R9, allows you to intentionally leave an agent idle in a skill when the agent's adjusted work time has exceeded the percentage that you administered for that skill.

Call selection at a glance

The following table shows what happens during call surplus conditions, according to the call selection methods that have been administered on the communication server.

IF calls are waiting when an agent becomes available and the agent's selection method is:	THEN the communication server takes the highest priority call:
Skill Level without Service Objective	With the highest skill level and the longest CWT or PWT.
Skill Level with Service Objective	With the highest skill level and the highest ratio of CWT/SO or PWT/SO.
Greatest Need without Service Objective	With the longest CWT or PWT.
Greatest Need with Service Objective	With the highest ratio of CST/SO or PWT/SO.
Percent Allocation	That is the oldest call waiting that best maintains the administered target allocations for all skills.

Agent selection at a glance

The following table shows what happens during agent surplus conditions, according to the agent selection method that has been administered.

WHEN agents are available, a call arrives, and the agent selection method is:	THEN the communication server selects:
EAD-MIA	The highest skill level, most idle agent.
UCD-MIA	The most idle agent, without regard to skill level.

WHEN agents are available, a call arrives, and the agent selection method is:	THEN the communication server selects:
EAD-LOA	The highest skill level agent with the lowest occupancy.
UCD-LOA	The least occupied agent, without regard to skill level.
PAD	The agent with the lowest ratio of adjusted work time and target allocation for the skill.

Combining agent and call selection methods

Avaya Business Advocate provides a variety of features to help meet your business goals and to help you manage your agent resources. The table below shows some of the ways you can combine call and agent selection methods to meet your company's specific needs.

IF your goal is to:	THEN consider:
Maintain service levels while controlling the time agents spend serving each of their skills	<ul style="list-style-type: none"> ● Percent Allocation ● Dynamic Percentage Adjustment ● PAD
Maintain service levels using more or less time from reserve resources to supplement staffing as needed	<ul style="list-style-type: none"> ● Greatest Need ● Service Level Supervisor ● Dynamic Threshold Adjustment ● UCD-LOA
Add customer segmentation with differentiated levels of service while routing all segments to the same skill to simplify staffing	<ul style="list-style-type: none"> ● Greatest Need ● Dynamic Queue Position ● UCD-LOA
Increase revenue by assigning agents their best skills as primary skills and limiting the use of reserve skills to eliminate long call wait times	<ul style="list-style-type: none"> ● Greatest Need ● Service Objective ● Service Level Supervisor ● UCD-LOA
Ensure that critical skills are covered, regardless of caller wait time in other skills	<ul style="list-style-type: none"> ● Greatest Need ● Service Level Supervisor ● Call Selection Override ● Oldest Call Waiting ● UCD-LOA

ACD call center features

IF your goal is to:	THEN consider:
Control the time your agents spend serving their assigned skills while maintaining the ability to change to meet service level requirements for the center	<ul style="list-style-type: none"> ● Percent Allocation ● Dynamic Percentage Adjustment ● Call Selection Override ● Service Level Supervisor ● PAD
Automate agent staffing to activate back up agents a little sooner or a little later to meet service level goals	<ul style="list-style-type: none"> ● Greatest Need or Skill Level ● Service Level Supervisor ● Dynamic Threshold Adjustment ● UCD-LOA or EAD-LOA
Minimize the complexity of differentiating service levels for different types of calls that require similar agent abilities	<ul style="list-style-type: none"> ● Greatest Need or Skill Level ● Dynamic Queue Position ● UCD-LOA or EAD-LOA
Maximize the amount of time that agents spend in high contribution roles while limiting their use of lesser skills to address wait time problems	<ul style="list-style-type: none"> ● Greatest Need ● Service Objective ● UCD-LOA
Spread calls more evenly among agents while delivering the right level of service to each skill	<ul style="list-style-type: none"> ● Greatest Need ● Service Objective ● UCD-LOA
Use agents in their most proficient skills while minimizing the hot seat problem to some extent	<ul style="list-style-type: none"> ● Skill Level ● EAD-LOA

Different needs within a call center

You may find that one Avaya Business Advocate solution does not fit for your entire organization. Your call center may have different needs within particular areas or departments, and Avaya Business Advocate can help to meet these varying needs. A sales department, for example, may choose to use Dynamic Queue Position to create differentiation among various types of customer without creating a different skill for each type of sales call. A service department, on the other hand, may be more interested in working toward similar goals for each technical support skill, while eliminating the hot seats often experienced by the well trained, multi-skilled agents.

Avaya offers a subscription service for Avaya Business Advocate customers that provides access to skilled consultants with expertise in understanding how Avaya Business Advocate helps to solve business problems. For more information, please contact the Advisory Team at 877-977-0078 or by e-mail at advisoryhelp@avaya.com.

Feature compatibility

It is important to choose the right combination of features to meet your organization's needs and ensure that Avaya Business Advocate is set up to work most effectively. This section summarizes the features that provide the best results when used together and also lists those that are not designed to work together.

Call selection methods (call handling preferences)

The following table shows the features that work effectively with the various Avaya Business Advocate call selection methods.

Call selection method	Recommended to work with
Greatest Need	<ul style="list-style-type: none"> ● Predicted Wait Time ● Service Objective ● Service Level Supervisor ● UCD-MIA ● UCD-LOA
Skill Level	<ul style="list-style-type: none"> ● Predicted Wait Time ● Service Objective ● Service Level Supervisor ● EAD-MIA ● EAD-LOA
Percent Allocation	<ul style="list-style-type: none"> ● Dynamic Percentage Adjustment ● Auto Reserve Agents ● Service Level Supervisor ● PAD

Agent selection methods (hunt group types)

The following table shows which features work with the various agent selection methods.

Agent Selection Method	Recommended to work with
UCD-MIA	<ul style="list-style-type: none"> ● Greatest Need ● Predicted Wait Time ● Service Objective ● Service Level Supervisor
EAD-MIA	<ul style="list-style-type: none"> ● Skill Level ● Predicted Wait Time ● Service Objective ● Service Level Supervisor
UCD-LOA	<ul style="list-style-type: none"> ● Greatest Need ● Predicted Wait Time ● Service Objective ● Service Level Supervisor
EAD-LOA	<ul style="list-style-type: none"> ● Skill Level ● Predicted Wait Time ● Service Objective ● Service Level Supervisor
PAD	<ul style="list-style-type: none"> ● Percent Allocation ● Dynamic Percentage Adjustment ● Auto Reserve Agents ● Service Level Supervisor

Feature combinations to avoid

The PAD agent selection method should not be used with Greatest Need or Skill Level call selection methods.

Basic Call Management System

This section includes the following topics:

- [About BCMS](#) on page 107
- [Administering BCMS](#) on page 108
- [BCMS interactions](#) on page 109

About BCMS

Basic Call Management System (BCMS) provides real-time and historical reports to assist you in managing agents, ACD splits/skills (hunt groups), VDNs, and trunk groups. You can display BCMS reports on a terminal or print a paper copy.

BCMS provides the following reports:

- Real Time Reports
 - Split/Skill Status
 - System Status
 - VDN Status
- Historical Reports
 - Agent
 - Agent Summary
 - Split/Skill
 - Split/Skill Summary
 - Trunk Group
 - Trunk Group Summary
 - VDN
 - VDN Summary

For a detailed description of BCMS and the reports it provides, see *Avaya Communication Manager Call Center Software - Basic Call Management System (BCMS) Operations*.

Administering BCMS

The following forms and fields are required to administer the BCMS feature.

Screen	Field
System Parameters Customer-Options	Enable the following options: <ul style="list-style-type: none"> ● ACD ● BCMS (Basic) ● BCMS/Service Level ● VuStats
Agent LoginID	When BCMS is being used with EAS, complete all fields for each agent
BCMS/VuStats Login ID	When BCMS is being used without EAS, enter a login ID and name for each agent.
Feature-Related System Parameters	Administer the following options: <ul style="list-style-type: none"> ● Minimum Agent-Login Password Length ● BCMS/VuStats Measurement Interval ● BCMS/VuStats Abandon Call Timer ● Validate BCMS/VuStats Login IDs ● Remove Inactive BCMS/VuStats ● System Printer Endpoint ● Lines Per Page ● EIA Device Bit Rate
Hunt Groups	Measured Acceptable Service Level
Trunk Groups	Measured
Vector Directory Number (VDN)	Measured Acceptable Service Level

All agents should log off before any changes are made to the BCMS/VuStats Login ID screen.

Measurements can be turned off for a split or skill while agents are logged in, but agents must be logged off to start measurements for a split or skill.

BCMS interactions

Call redirection and conference calls: For information about how BCMS records redirects and conferences calls, see *Avaya Communication Manager Call Center Software - Basic Call Management System (BCMS) Operations*.

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 agent's *from* split or skill and started for the agent's *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 would exceed the maximum number of measured agents, the communication server 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, then 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 these 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 them as INFLOW).

System Measurements : The system can simultaneously produce BCMS reports, adjunct CMS reports, and communication server 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.

Best Service Routing

This section includes the following topics:

- [About BSR](#) on page 110
- [Benefits of BSR](#) on page 111
- [BSR requirements](#) on page 112
- [Administering single-site BSR](#) on page 114
- [Administering multi-site BSR](#) on page 114
- [Administering BSR polling over IP without the B-channel](#) on page 115
- [Interactions for BSR polling over IP without the B-channel feature](#) on page 122
- [BSR detailed description](#) on page 122
- [Call surplus situations](#) on page 122
- [Agent surplus situations](#) on page 123
- [Alternate Selection on BSR Ties](#) on page 124
- [Vector commands for single-site BSR](#) on page 126
- [Vector commands for multi-site BSR](#) on page 128
- [BSR considerations](#) on page 129
- [BSR interactions](#) on page 130

About BSR

Best Service Routing (BSR) allows the communication server to compare specified skills, identify the skill that will provide the best service to a call, and deliver the call to that resource. If no agents are currently available in that skill, the call is queued. To respond to changing conditions and operate more efficiently, BSR monitors the status of the specified resources and adjusts call processing and routing as appropriate.

BSR is available in single and multi-site versions. Single-site BSR compares skills on the local communication server to find the best resource to service a call. Multi-site BSR extends this capability across a network of communication servers, comparing local skills, remote skills, or both, and routing calls to the resource that will provide the best service.

Best Service Routing is summarized in this chapter. For complete information on how to get the most from BSR, see *Avaya Call Center Call Vectoring and EAS Guide*.

Benefits of BSR

Both single- and multi-site BSR intelligently compare specific resources to find the one that can best service a call. In addition, multi-site BSR allows you to integrate a network of call centers for better load balancing and optimal agent utilization. Depending on your specific application, BSR can yield a variety of other benefits.

You can benefit by...	As a result of...
Improved customer satisfaction	<ul style="list-style-type: none"> ● Lower average speed of answer (ASA), thus more calls handled¹ ● Greater probability that expert agents will be available for a specific call type (for centers with EAS) ● Lower abandonment rate ● By balancing the load between locations in a network, BSR reduces extremes in wait times between the locations.
Increased revenue	<ul style="list-style-type: none"> ● Lower average speed of answer (ASA), thus more calls handled ● Lower abandonment rate ● Lower network cost ● Greater probability that high-revenue generating agents will be available for a specific call type (for centers with Expert Agent Selection)
Improved productivity	<p>Increased agent occupancy rates:</p> <ul style="list-style-type: none"> ● Improve your service without adding staff, or reduce staff while maintaining your current level of service. ● Faster delivery of interflowed calls ● Agents at one location in a network are less likely to sit idle while calls wait in queue at another location.
Increased operating flexibility, easier staffing and scheduling	<ul style="list-style-type: none"> ● Larger pool of agents available to take calls in a skill ● Spikes in call volume at a single call center can be distributed across all centers. Temporarily understaffed centers can be supported by the other centers in the network.
Improved service levels	Lower average speed of answer (ASA), thus more calls handled
Increased performance	Less messaging and processing required per call
Operating separate sites as an integrated virtual call center	<ul style="list-style-type: none"> ● Ability to compare resources and queue a call to the best one ● Enhanced information forwarding capabilities

1. A location's ASA may be low because the location is under utilized and agents are frequently sitting idle. When BSR is implemented at such a location, ASA may rise because of the rise in incoming call volume.

BSR requirements

For single-site BSR applications, your communication server must meet the requirements shown below (except for LAI). To use multi-site BSR applications, both the communication servers involved and the network connecting them must meet all the requirements described in this section.

This section includes the following topics:

- [Communication server requirements for BSR](#) on page 112
- [Network requirements for BSR](#) on page 113

Communication server requirements for BSR

Your communication server has to meet all of the requirements shown below to support Best Service Routing. Check the settings shown below before you try to use BSR.

Screen	Page	Field	Must be set to...
Software Version	1	Memory Resident	G3V6i.03, G3V6r.03, or higher
System-Parameters Customer-Options ¹	1	G3 Version	V6 or higher
	2	ISDN-BRI Trunks	Y
		ISDN-PRI Trunks ²	Y
	3	Vectoring (G3V4 Advanced Routing)	Y
		Vectoring (Best Service Routing)	Y
Lookahead Interflow (LAI) ³		Y	
Feature-Related System Parameters	8	Adjunct CMS Release	R3V6 or higher, or left blank

1. ISDN connectivity is only necessary if you want to use multi-site BSR. One or both of these fields must be set to Y. Multi-site BSR operates over both BRI and PRI trunks.

2. Or H.323 IP trunking. For more information, see [Administering BSR polling over IP without the B-channel](#) on page 115.

3. Look-Ahead Interflow is only necessary if you want to use multi-site BSR. For status poll and interflow calls used in multi-site BSR applications, set up trunks as you would for LAI. See [Look-Ahead Interflow](#) on page 188, for more information. Information Forwarding is not required for polling calls.

Note:

If you begin using BSR and then decide to turn it off, you will not be able to set Vectoring (Best Service Routing) to n until you remove all BSR commands from vectors.

Network requirements for BSR

To support BSR, networks must meet both the criteria for LAI call control operation over switched networks (see [Look-Ahead Interflow](#) on page 188) and the following criteria:

**CAUTION:**

To ensure your network meets the requirements for BSR support presented below, contact your Account Executive about BSR network certification.

- 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 QSIG Manufacturer Specific Information (MSI IE), in the ISDN SETUP and DISCONNECT messages. The network must also allow User-to-User Information Elements to be transported in the first call-clearing message prior to answer, while a call is still in the call-proceeding state. (For more information, see [Information Forwarding](#) on page 178.)

With BSR poll calls, the information is forwarded back in the DISCONNECT message. In this case the network must support forwarding of UUI in the first call clearing message prior to the active state (in other words, while still in the call proceeding state).

Private networks can be configured for either QSIG (transport using MSI packaged in a codeset 0 Facility IE) or non-QSIG (transport using a codeset 0 UUI IE). Currently, public networks do not support QSIG and user data can only be transported using the UUI IE when supported by the network. Future public network offerings may support QSIG, possibly by Virtual Private Network.

- The communication server must support the ISDN country protocol.
- The network byte limit for user information contents (the user data portion) must be large enough to carry the data needed for the customer application.

Note:

Some public network providers may require service activation and/or fees for user information transport.

- Response time for ISDN signaling should be fast enough that an entire consider series can execute within a single ring cycle.

Enhanced information forwarding has been tested with several major carriers. To find out if these 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 involved with the preferred specific configuration, use of private ISDN trunking between the nodes should be assumed until successful testing has been completed.

Administering single-site BSR

The following forms and fields are required to administer the BSR feature in a single-site configuration.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● G3 Version ● Vectoring (G3V4 Advanced Routing) ● Vectoring (Best Service Routing)
Feature-Related System Parameters	Adjunct CMS Release
ISDN Numbering - Public/Unknown	<ul style="list-style-type: none"> ● Ext Len ● Ext Code ● CPN Prefix
VDN	BSR Available Agent Strategy
Call Vector	Complete a screen for each vector that uses BSR commands

Administering multi-site BSR

The following forms and fields are required to administer the BSR feature in a multi-site configuration.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● G3 Version ● Vectoring (Best Service Routing) ● Vectoring (G3V4 Advanced Routing) ● Lookahead Interflow (LAI)
Feature-Related System Parameters	Adjunct CMS Release
Trunk Group (ISDN-BRI) ¹	<ul style="list-style-type: none"> ● Outgoing Display ● Supplementary Service Protocol ● UUI Treatment
Trunk Group (ISDN-PRI)	<ul style="list-style-type: none"> ● Outgoing Display ● Supplementary Service Protocol ● UUI Treatment

Screen	Field
Trunk Group (SIP)	UI Treatment
Best Service Routing Application Plan	Complete one screen for each BSR application
VDN	<ul style="list-style-type: none"> ● BSR Application ● BSR Available Agent Strategy
Call Vector	Complete a screen for each vector (primary, status poll and interflow vectors) in a BSR application

1. Settings in the fields Codeset to Send TCM, Lookahead and Send Codeset 6/7 LAI IE on the ISDN trunk forms do not affect BSR.

Multi-site BSR requires that Look-Ahead Interflow (LAI) be enabled. See [Administering LAI](#) on page 196 for a list of required forms and fields. Since BSR can forward information such as VDN name, in-VDN time, and collected digits with interflowed calls, also see [Administering User-to-User Information transport for ISDN trunks](#) on page 179 and [Administering User-to-User Information transport for SIP trunks](#) on page 181 for instructions.

Administering BSR polling over IP without the B-channel

This section includes the following topics:

- [About BSR polling](#) on page 115
- [Prerequisites](#) on page 116
- [Administration forms](#) on page 116
- [Operational elements](#) on page 121
- [Interactions for BSR polling over IP without the B-channel feature](#) on page 122

About BSR polling

BSR polling can be configured so that the use of a B-channel is not required when a polling signal is sent over an H.323 IP trunk. This feature offers the following advantages:

- Improved trunk efficiency - This feature uses QSIG CISC/TSCs (Call Independent Signaling Connections / Temporary Signaling Connections) that send BSR polls over D-channel without associated seizure of a B-channel. This polling strategy allows more trunk bandwidth to be available for other forms of voice or data traffic.
- Reduced hardware requirements - If Voice over IP functionality is not required for the trunk, a IP Media Processor circuit pack (TN2302) is not required. In either case, a C-LAN circuit pack (TN799) is required to support D-channel signalling.

ACD call center features

Note:

BSR polling over IP without the B-channel uses non-call associated TSCs.

Prerequisites

The following prerequisite conditions must be satisfied to enable BSR polling over IP without the B-channel.

Software: All locations that use the BSR polling over IP without the B-channel feature must use Communication Manager R11 or later. The Call Center version must be V6 or later and multi-site BSR must be administered. For more information, see [Administering multi-site BSR](#) on page 114.

Note:

If the remote communication server does not have R11 installed, the CISC (Call Independent Signalling Connection) SETUP does not start vector processing and the poll operation ends in a timeout, logging a vector event. The next step in the vector is then executed.

License file options: To use the BSR polling over IP without the B-channel feature, the IP Trunk, QSIG Basic Call Setup, and QSIG Basic Supplementary Services customer options must be enabled.

C-LAN connectivity : A C-LAN circuit pack (TN799) is required for this feature.

Administration forms

To enable BSR polling over IP without the B-channel, you must:

- Set up the signalling group for H.323, QSIG, and NCA-TSCs
- Set up the designated trunk group for ISDN and IP, with at least one trunk assigned.

 **Important:**

If an IP Media Processor circuit pack is not installed, the Trunk Group Status screen indicates the trunk as `out-of-service`. However, status poll signaling over D-channel is not affected.

The forms that you must administer to enable BSR polling over IP without the B-channel are described in the following sections:

- [Pattern screen](#) on page 117
- [Signaling Group screen](#) on page 118
- [Trunk Group screen](#) on page 118
- [Feature-Related System Parameters screen \(ISDN\)](#) on page 121

Pattern screen

Use the `display route-pattern xx` command (where `xx` is the route pattern used by the trunk group that supports the TSCs) to display the Pattern screen. Verify that your administration settings comply with the following requirement:

- In the TSC column, set the value to `y`.

An example route-pattern screen that is configured for BSR polling over IP without the B-channel is shown below.

```

display route-pattern 32                                     Page 1 of 3
                                     Pattern Number: 32

  Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC
  No           Mrk Lmt List Del  Digits          QSIG
                                     Dgts          Intw
1: 32   0                3                n  user
2:                n  user
3:                n  user
4:                n  user
5:                n  user
6:                n  user

  BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature BAND  No. Numbering LAR
  0 1 2 3 4 W      Request      Subaddress
1: y y y y y n  y  as-needed rest                none
2: y y y y y n  n           rest                none
3: y y y y y n  n           rest                none
4: y y y y y n  n           rest                none
5: y y y y y n  n           rest                none
6: y y y y y n  n           rest                none

```

Best Service Routing screen

Use the `display best-service-routing xx` command (where `xx` is a BSR application plan number) to display the Best Service Routing screen. Verify that your administration settings comply with the following requirement:

- The Status Poll VDN field must specify an AAR or AAS pattern that routes over an IP trunk.



Important:

Do not specify a TAC in the Status Poll VDN field. If you do so, the poll will route through a B-channel, if one is available.

Signaling Group screen

Use the `display signaling-group xx` command (where `xx` is the signaling group number) to display the Signaling Group screen. Verify that your administration settings comply with the following requirements:

- You must specify the TSC-related fields in the upper-right corner of the field. The relevant fields include:
 - Max number of NCA TSC:
 - Trunk group for NCA TSC:
- The Supplementary Services Protocol: field must be set to `b`.

An example signaling-group screen configured for BSR polling over IP without the B-channel is shown as follows.

```

display signaling-group 32
                                SIGNALING GROUP

Group Number: 32                Group Type: h.323
                                Remote Office? n
                                Max number of NCA TSC: 10
                                Max number of CA TSC: 10
                                Trunk Group for NCA TSC: 32

Trunk Group for Channel Selection: 32
Supplementary Service Protocol: b
                                Network Call Transfer? n

Near-end Node Name: clan-01D12   Far-end Node Name: cland12-loop
Near-end Listen Port: 1720       Far-end Listen Port: 1720
Far-end Network Region:
LRQ Required? n                  Calls Share IP Signaling Connection?y
RRQ Required? n
                                Bypass If IP Threshold Exceeded?n

                                Direct IP-IP Audio Connections?y
                                IP Audio Hairpinning?y
                                Interworking Message: PROGRESS
    
```

Trunk Group screen

Use the `display trunk-group xx` command (where `xx` is the signaling group number) to display the Trunk Group screen. Verify that your administration settings comply with the requirements listed below each of the following example screen pages.

Examples for the key pages in the Pattern screen that must be configured for BSR polling over IP without the B-channel are shown as follows.

Trunk group - Page 1

```

display trunk-group 32                                     Page 1 of 22
                                                         TRUNK GROUP

Group Number: 32                Group Type: isdn                CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1                TN: 1                TAC: 732
  Direction: two-way                Outgoing Display? n                Carrier Medium: IP
  Dial Access? y                Busy Threshold: 255                Night Service:
Queue Length: 0
Service Type: tie                Auth Code? n                TestCall ITC: rest
                Far End Test Line No:

TestCall BCC: 4
TRUNK PARAMETERS
  Codeset to Send Display: 6                Codeset to Send National IEs: 6
  Max Message Size to Send: 260                Charge Advice: none
  Supplementary Service Protocol: b                Digit Handling (in/out): enbloc/enbloc

                Trunk Hunt: cyclical

                Digital Loss Group: 13
Calling Number - Delete:                Insert:                Numbering Format:
                Bit Rate: 1200                Synchronization: async                Duplex: full
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0

```

In the page example shown above, the administrations settings must conform to the following requirements:

- The **Group Type:** field must be set to `isdn`
- The **Carrier Medium:** field must be set to `IP`
- The **Supplementary Service Protocol:** field must be set to `b`

Trunk group - continued

```

display trunk-group 32                                     Page 2 of 22
TRUNK FEATURES
    ACA Assignment? n           Measured: none           Wideband Support? n
                                Internal Alert? n         Maintenance Tests? y
                                Data Restriction? n       NCA-TSC Trunk Member: 1
                                Send Name: y             Send Calling Number: y
    Used for DCS? n             Hop Dgt? n
    Suppress # Outpulsing? n    Numbering Format: public
    Outgoing Channel ID Encoding: preferred    UUI IE Treatment: service-provider

Replace Restricted Numbers? n

                                Replace Unavailable Numbers? n
                                Send Connected Number: n

    Send UCID? y
    Send Codeset 6/7 LAI IE? n

    Path Replacement with Retention? n
    Path Replacement Method: better-route
                                Network (Japan) Needs Connect Before Disconnect? n
    
```

In the page example shown above, the administrations settings must conform to the following requirements:

- The `NCA-TSC Trunk Member:` field must specify a trunk group member.

Trunk group - continued

```

display trunk-group 32                                     Page x of 22
                                TRUNK GROUP
                                Administered Members (min/max): 1/4
GROUP MEMBER ASSIGNMENTS    Total Administered Members: 4

    Port    Code Sfx Name    Night    Sig Grp
1: T00089           32
2: T00090           32
3: T00091           32
4: T00092           32
5:
6:
    
```

In the page shown above, the administrations settings must conform to the following requirements:

- The group member used to make the BSR status polls (which is also specified in the `NCA-TSC Trunk Member:` field on page 2 of this screen) must be associated with the appropriate signaling group. The signaling group is specified in the `Sig Grp` column.

Feature-Related System Parameters screen (ISDN)

Use the **change system-parameters feature** command to display the Feature-Related System Parameters screen. Verify that your administration settings comply with the following requirements:

- The **QSIG TSC Extension:** field must specify an unassigned extension number that is valid for the dial plan.

An example Feature-Related System Parameters screen configured for BSR polling over IP without the B-channel is shown below.

```

change system-parameters features                                     Page 7 of 12
                        FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Name as Connected Name? n
Display Connected Name/Number for ISDN DCS Calls? n
    Send ISDN Trunk Group Name on Tandem Calls? n

                        QSIG TSC Extension: 3999
MWI - Number of Digits Per Voice Mail Subscriber: 5

                        National CPN Prefix:
                        International CPN Prefix:
                        Pass Prefixed CPN to ASAI? n
Unknown Numbers Considered Internal for AUDIX? n
    USNI Calling Name for Outgoing Calls? n
        Path Replacement with Measurements? y
            QSIG Path Replacement Extension: 2999
                Path Replace While in Queue/Vectoring? y

```

Operational elements

The BSR polling over IP without the B-channel feature is associated with the following operational elements:

- The VDN for the BSR status poll contains a number (AAR/ARS access code plus the remote poll VDN address) that routes over a QSIG trunk group. A TAC can not be used for this purpose.
- When a **consider location** vector step routes a call over the IP trunk group with QSIG signalling, the following is true:
 - A TSC **SETUP** message is sent over emulated D-Channel
 - The **reply-best** data is returned with a TSC **RELEASE** message
- This feature works with tandem trunks, if all trunks in the tandem are configured to meet the requirements described in this section. However, tandem configurations are not recommended for the BSR polling over IP without the B-channel feature.

Interactions for BSR polling over IP without the B-channel feature

The BSR polling over IP without the B-channel feature is associated with the following interactions:

- CMS records poll attempts, but not trunk measurements, since no B-Channel trunk facility is used.
- If sufficient trunks are assigned, the trunk group can also be used for H.323 IP voice calls, but an IP Media Processor must be installed.

BSR detailed description

To use Best Service Routing (BSR) on a single communication server, you simply use special commands and command elements that are part of the call vectoring language. As a result, BSR for a single location can be easily added to existing vectors without modifying other parts of the communication server.

Multi-site applications work similarly, but additional administration is required. Since steps in a multi-site BSR vector will contact one or more remote locations, you need to define these locations, tell the communication server how to contact each one, and set up VDNs and vectors to handle communications between the sending communication server and each remote communication server.

Three VDN/vector pairs must be used in every multi-site BSR application. The Primary VDN/vector pair, on the sending communication server, contacts the specified remote communication server, collects information, compares the information, and delivers or queues the call to the resource that is likely to provide the best service. Two VDN/vector pairs are needed on each remote communication server. A Status Poll VDN/vector pair provides information about the best resource at its location in response to inquiries from BSR applications on other communication servers. Finally, an Interflow VDN/vector pair is needed to receive and process the calls interflowed from BSR applications on other communication servers.

Call surplus situations

Every BSR application compares a set of predetermined resources (skills) and selects the best resource to service the call. In a call surplus situation (no agents available), the best resource is the skill with the lowest Expected Wait Time (EWT). For purposes of calculating the best resource in a call surplus situation, BSR allows you to adjust the EWT figure for any skill. The actual EWT for calls in queue is not changed, of course; 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 skills allows you to program preferences in vectors. Because of agent expertise, for example, or the availability or cost of tie trunks, you might prefer that some resources not service a call unless doing so significantly decreases the call's time in queue.

It is possible for you to make adjustments to agent availability using the **consider** step.

Agent surplus situations

In an agent surplus situation (one or more agents available to take incoming calls), BSR will deliver a new call according to the BSR Available Agent Strategy specified on the VDN screen. The best resource will be the skill that meets the criteria defined by the strategy you've chosen for that VDN.

BSR can use any of the five strategies shown in the following table to select an agent when agents are available.

If BSR Available Agent Strategy is set to...	The call will be delivered to...
1st-found	The first available agent. BSR will not consider any other resources as soon as it finds an available agent.
ucd-mia	The resource with an agent who has been idle the longest. BSR will compare all the skills specified in the vector before delivering the call.
ead-mia	The resource with an agent with the highest skill level relevant to the call who has been idle the longest. BSR will compare all the skills specified in the vector before delivering the call.
ucd-loa	The resource with a least-occupied agent. BSR will compare all the skills specified in the vector before delivering the call.
ead-loa	The resource with an agent with the highest skill level relevant to the call who is the least occupied. BSR will compare all the skills specified in the vector before delivering the call.

When agents are available in one or more of the specified resources, BSR does not consider resources (local or remote) that return an EWT (call queue/call surplus situation) in selecting the best place to send the call.

Note:

The BSR Available Agent Strategy assigned to a VDN should match the agent selection method used in the skills considered by a BSR application.

The BSR adjust-by value can be used 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 communication server applies the agent adjustment in the same manner as the calls in queue/call surplus (lowest EWT) situation.

ACD call center features

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.

Alternate Selection on BSR Ties

This section includes the following topics:

- [Understanding Alternate Selection on BSR Ties](#) on page 124
- [Administering Alternate Selection on BSR Ties](#) on page 124

Understanding Alternate Selection on BSR Ties

The Best Service Routing (BSR) feature 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 results are equal in value), the Alternate Selection on BSR Ties determines how BSR chooses which agent, skill, or location to select. The Alternate Selection on BSR Ties chooses between:

- Skills or locations with the same expected wait time (EWT)
- Available agents that are weighted with the same criteria (most idle agent or least occupied agent) in a consider series that is designed to locate the best skill or location

You can set the Alternate Selection on BSR Ties for the system or on a per Vector Directory Number (VDN) basis. Each consider skill or location step applies the assigned strategy by comparing the current best choice from a previous consider step to the value obtained from the current consider step.

If the Alternate Selection on BSR Ties on the active VDN for the call is 1st-found, when the algorithm finds an available agent at a consider step, the processing of the remaining consider steps stops. The Alternate Selection on BSR Ties does not apply in this case.

Administering Alternate Selection on BSR Ties

To administer Alternate Selection on BSR Ties:

1. Enter `change system-parameters customer-options`.

2. On the System-Parameters Customer-Options screen, verify that **Vectoring (Basic)** or **Vectoring (Prompting)** is enabled.
3. On the Feature Related System Parameters screen enter one of the following values in the **BSR Tie Strategy** field:
 - 1st-found (default - results in existing operation)
 - alternate
4. Enter **change vdn <vdn number>** or **add vdn <vdn number>**.
5. On the VDN screen, verify that the **Attendant Vectoring?** and **Meet-me Conferencing?** fields are both set to **n**.

Note:

You cannot see the **BSR Tie Strategy** field if either Attendant Vectoring or Meet-me Conferencing is enabled.

6. Enter one of the following values in the **BSR Tie Strategy** field:
 - system (default - use the Feature Related System Parameter setting)
 - 1st-found
 - alternate

Related topic

For a description of the values used for the **BSR Tie Strategy** fields, see [Call Center System Parameters field descriptions](#) on page 348 and [Vector Directory Number field descriptions](#) on page 362.

Vector commands for single-site BSR

The following table shows the vector commands and command elements used in single-site BSR applications.

Commands and command elements		Use this ...
Commands	<code>consider split/skill</code>	to obtain the Expected Wait Time or agent data needed to identify the best local resource. One consider step must be written for each skill you want to check. ¹
	<code>queue-to</code>	with the best keyword to queue calls to the best resource identified by the consider sequence.
	<code>check</code>	with the best keyword to queue calls to the best resource identified by the consider sequence if the resource meets certain conditions.
Key word	<code>best</code>	in queue-to , check , and goto commands that refer to the resource identified as best by a series of consider steps

Commands and command elements		Use this ...
Conditional	<code>wait-improved</code>	Prevents calls from being queued to an additional skill when the reduction in Expected Wait Time is not enough to be useful. <i>Wait-improved</i> means that a call's EWT must be improved by a specific amount (a figure you specify in seconds) over its current EWT or the communication server will not queue it to the additional skill.
User adjustment	<code>adjust-by</code>	<p>To specify your preferences for the skills that might handle the calls for a particular application, reflecting factors such as agent expertise or reducing calls to a backup skill. When a vector considers a local resource you can make the selection of that skill less desirable. The higher the setting, the less chance that resource will be selected over another with a lower setting (for example, set to 30 makes that choice 30% less desirable). With EWT returned, the setting increases the returned expected wait time for comparison with other returned EWTs. As a result, this skill is less likely to service the call unless its EWT is significantly less than that of any other available skill.</p> <p>Optionally, the <code>adjust-by</code> 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 (busier). 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 (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.</p>

1. Since the **consider** command is designed to compare two or more resources, **consider** commands are typically written in sequences of two or more with the sequence terminating in a **queue-to best** step. This set of **consider** commands and a **queue-to best** step is called a consider series.

Vector commands for multi-site BSR

The following table summarizes the vector commands and command elements that support multi-site BSR applications.

Commands and command elements		Use this...
Commands	consider split/skill	To obtain the Expected Wait Time or agent data needed to identify the best local resource. One consider step must be written for each skill you want to check. ¹
	consider location	To obtain the Expected Wait Time or agent data needed to identify the best resource at a remote communication server. One consider step must be written for each location you want to check. Routing information is obtained from the BSR Application plan for the active VDN.
	reply-best	To return data to another communication 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

Commands and command elements		Use this...
Conditional	<code>wait-improved</code>	To prevent calls from being queued to an additional skill - local or remote - when the reduction in Expected Wait Time is not enough to be useful. <i>Wait-improved</i> means that a call's EWT must be improved by a specific amount (a figure you specify in seconds) over its current EWT or the communication server will not queue it to the additional skill.
User adjustment	<code>adjust-by</code>	<p>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 less chance that resource will be selected over another with a lower setting. With EWT returned, the setting increases the returned expected wait time for comparison with other returned EWTs.</p> <p>Optionally, the <code>adjust-by</code> 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 (busier). 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 (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.</p>

1. Since the **consider** command is designed to compare two or more resources, **consider** commands are typically written in sequences of two or more with the sequence terminating in a **queue-to best** step. This set of **consider** commands and a **queue-to best** step is called a consider series.

BSR considerations

- If one or more of the resources considered 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` 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, then `adjust-by` is applied as part of the algorithm to select the best of the possible choices.

BSR interactions

Agent Telephone Display: If collected digits are forwarded with an interflowed call, the forwarded digits are displayed on the answering agent's telephone display (unless they're overridden with newly collected digits).

Best Service Routing (BSR)/LAI : Restrictions and interactions that apply to LAI also apply to BSR status poll and interflow calls. See *Avaya Call Center Call Vectoring and EAS Guide*, or [Look-Ahead Interflow](#) on page 188 for more information.

BCMS : BCMS does not report accumulated in-VDN time. BCMS does not log LAI attempts and therefore will not log BSR status polls, which are treated as LAI attempts.

Call Vectoring: The following considerations apply to ALL vectors when BSR is enabled on your communication server.

Call Vectoring considerations when BSR is enabled	
<code>route-to VDN</code>	If a call is routed to a new VDN, any best resource data defined by a series of <code>consider</code> steps in the previous VDN will be initialized (cleared)
<code>goto vector</code>	If a <code>goto vector</code> command is executed, any best resource data produced by a series of <code>consider</code> steps in the original VDN will remain with the call and can be used in the subsequent vector.
<code>consider</code>	<ul style="list-style-type: none"> Do not use other commands within a series of <code>consider</code> steps, since these may delay the execution of the series. Skills used in <code>consider</code> commands must be vector controlled.
<code>converse</code>	Collected digits forwarded with the call will be passed to VRU using the digits data passing type.
<code>best</code> (keyword)	<p>The <code>best</code> keyword can be used in the following commands, but only with the conditionals listed:</p> <ul style="list-style-type: none"> <code>goto</code> step or <code>goto vector</code> commands using the <code>expected-wait</code> or <code>wait-improved</code> conditionals <code>check</code> commands using the <code>unconditional</code>, <code>expected-wait</code>, or <code>wait-improved</code> conditionals <p>The <code>best</code> keyword can not be used as a replacement for <code>split</code> or <code>skill</code> in the following vector commands:</p> <ul style="list-style-type: none"> <code>converse-on split/skill</code> <code>messaging split/skill</code>

Direct Department Calling: BSR will function when the considered splits use DDC call distribution. Once the best resource is determined, the actual call distribution will follow the split's DDC setting regardless of the BSR Available Agent Strategy. DDC may not be used as a BSR Available Agent Strategy.

Distributed Networking using QSIG Manufacturers Specific Information (MSI) : BSR will not function with systems from other vendors (unless that vendor develops a corresponding capability that works with the Avaya communication server).

Expert Agent Selection: EAS is required to use the EAD-MIA or EAD-LOA Available Agent Strategy. EAS VDN skills (1st, 2nd, 3rd) can be used in **consider skill** commands.

Facility Restriction Levels: The FRL applies to status poll and interflow calls in the same way it works with the **route-to number** command.

ISDN: Best Service Routing and globally supported information transport are fully functional over ISDN PRI or ISDN BRI trunking facilities.

Note:

Asynchronous Transfer Mode (ATM) trunking and IP trunking can be set up to emulate ISDN PRI. For information on setting this up, see *Administration for Network Connectivity for Avaya Communication Manager*, and *ATM Installation, Upgrades and Administration using Avaya Communication Manager*.

Location Preference Distribution: Local Preference Distribution is used to select an available agent within the call center during **consider** and **queue-to best** step operations. Local Preference Distribution is not used across system sites. In this case, there is no notion of a multi-site network region.

Look Ahead Routing (LAR) - BSR incompatibility: Look Ahead Routing (LAR) and BSR are incompatible. If a trunk is not available at the site being polled, an alternative route (as a secondary route using an ARS pattern) can be used to poll, assuming 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, then BSR processing handles the situation and after a period of 30 seconds, subsequent calls will try to poll that location again.

The use of alternative routes for polling only works if there are alternative routes for the interflow path, regardless of whether LAR or BSR is in use.

Multi-Split/Skill Queuing: A call may be queued up to three times by **queue-to** or **check** commands in the same vector. One vector may therefore contain up to three series of **consider** steps. Each series must be followed by a **queue-to best** step. Each consider series will select the best remote resource from the options you specify and queue the call to that resource.

BSR can only queue simultaneously on the origin communication server.

Network Access : BSR interflow operates over public, private, or virtual private (for example, SDN) ISDN-BRI and ISDN-PRI networks that meet the criteria explained in [Network requirements for BSR](#) on page 113. BSR interflow can also operate over SIP trunks. Best Service Routing requires that the network support transport of user-to-user data using MSI or UUI as a codeset 0 Information Element, or as shared user-to-user information 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.

ACD call center features

Administration or call processing will not prevent access to other types of trunks or to destinations that are not VDNs. However, BSR is only intended to support the types of applications described in this section. Attempts to use the BSR feature for any other purposes may not work.

Operating Support System Interface (OSSI) : The new administration commands, conditionals, keywords and forms are available using OSSI.

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 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 Look-Ahead Interflow `route-to number` vector step, BSR `queue-to best` vector step, and the Adjunct Routing vector steps.

The ability to track a measured ACD call after a path replacement has taken place is available for CMS versions r3v9ai.o or later. Starting with the r3v12ba.x release, CMS reports a path replacement as a *rename* operation rather than a path replacement. The *rename* operation properly reports scenarios where a path replacement takes place from a measured to an unmeasured trunk facility. Avaya recommends that you upgrade CMS to r3v12a.x or later and administer all trunks associated with path replacement as *measured* by CMS to ensure better CMS tracking of path-replaced calls.

Note:

Path replacement is not supported for BSR Local Treatment calls. Both ends of the connection must be answered for path replacement to work. When BSR local treatment is enabled, the local VDN has answered, but the remote VDN where the call is queued has not answered. Therefore, path replacement can not occur when a call is queued remotely by local treatment VDNs.

For more information on path replacement, see *Avaya Call Center Call Vectoring and EAS Guide*.

QSIG : LAI, BSR, and information forwarding function over QSIG trunk facilities if the remote locations are Avaya communication servers.

Redirection on No Answer (RONA) : Calls redirected to a VDN by RONA can be subsequently processed by BSR or LAI applications. When the RONA feature redirects a call to a VDN, any best resource data defined in a previous vector will be initialized (cleared).

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 multi-site 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 will be initialized (cleared). Transferred calls do not forward any of the information that is forwarded with interflows (previously collected digits, In-VDN time, etc.).

Trunk Access Code (TAC): Use of routing numbers (status poll or interflow) that utilize TACs is not recommended 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 option assigned on the VDN screen. It 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 will be used.

VDN Return Destination : The best resource data for a call is initialized when the call first leaves vector processing and therefore will not be available should the call return to vector processing.

VuStats: No enhancements have been added for BSR.

Call Prompting

This section includes the following topics:

- [About Call Prompting](#) on page 134
- [Administering Call Prompting](#) on page 135
- [Call Prompting considerations](#) on page 136
- [Call Prompting interactions](#) on page 136

About Call Prompting

The guide contains information that is critical to the effective and efficient use of this feature. For more information about Call Prompting, see *Avaya Call Center Call Vectoring and EAS Guide*.

Call Prompting uses specialized vector commands to process incoming calls based on information collected from the caller or from an ISDN-PRI message. It can be used in various applications to better handle incoming calls. The following list gives a brief description of some Call Prompting applications.

- Automated Attendant - Allows the caller to enter the extension of the party that he or she would like to reach. The call is routed to that extension.
- Data In/Voice Answer (DIVA) Capability - Allows the caller to hear an announcement based on the digits that he or she 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/adjunct to assist in call handling. This data, for example, may be the caller's account number.
- CINFO (Caller Information Forwarding) Routing - Allows a call to be routed based on digits supplied by the network in an ISDN-PRI message.
- Message Collection - Gives the caller the option of leaving a message or waiting in queue for an agent.

Administering Call Prompting

The following forms and fields are required to administer the Call Prompting feature.

Screen	Field
System Parameters Customer-Options	Vectoring (Prompting) Vectoring (CINFO) ISDN-PRI - for CINFO only This screen describes other vectoring options that may be required depending upon the application.
Feature-Related System Parameters	Prompting Timeout
Vector Directory Number	All
Announcements/Audio Sources	Complete all fields for each extension that provides a Call Prompting announcement
Hunt Group	Vector
Call Vector	Complete a screen for each Call Prompting vector
Station (multi-appearance)	Button/Feature Button Assignments - callr-info
Attendant Console	Feature Button Assignments -callr-info

If Vectoring (Basic) is not enabled on the System-Parameters Customer-Options screen, the Call Prompting feature cannot queue calls or make conditional checks based on queue or agent status, time of day, or day of week.

CINFO requires the AT&T Intelligent Call Processing (ICP) service, ISDN-PRI, and Vectoring (Prompting).

You can administer any display-equipped phone or attendant console with a Caller Information CALLR-INFO button. The button displays digits collected for the last `collect digits` command.

You must have Call Vectoring software for CMS to use Call Prompting (with or without Call Vectoring) with CMS.

Call Prompting considerations

Call prompting, with the exception of CINFO, competes with several features for ports on the call classifier - detector circuit pack or equivalent.

Call Prompting interactions

The following interactions apply specifically to Call Prompting. For general Call Vectoring interactions that may affect Call Prompting applications see [Call Vectoring](#) on page 137.

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 may later 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.

Call Vectoring

This section includes the following topics:

- [About Call Vectoring](#) on page 137
- [Administering Call Vectoring](#) on page 137
- [Call Vectoring interactions](#) on page 140

About Call Vectoring

Call Vectoring processes incoming and internal calls according to a programmed set of commands. These commands, called vector commands, determine the type of processing that calls receive. For example, vector commands can direct calls to on-premise or off-premise destinations, to any hunt group, split or skill, or to a specific call treatment such as an announcement, forced disconnect, forced busy, or delay. Vectors can queue or route calls based on a variety of different conditions.

There are many different applications for Call Vectoring. However, it primarily is used to handle the call activity of ACD splits/skills.

For more information about administering call vectoring, see [Best Service Routing](#) on page 110, [Network Call Redirection \(NCR\)](#) on page 208, and [Look-Ahead Interflow](#) on page 188.

For a detailed description of Call Vectoring, see *Avaya Call Center Call Vectoring and EAS Guide*.

Administering Call Vectoring

This section includes the following topics:

- [Required forms and fields](#) on page 138
- [Fields that do not allow VDN extensions](#) on page 139
- [Fields that allow VDN extensions](#) on page 140

Required forms and fields

The following forms and fields are required to administer the Call Vectoring feature.

Note:

Do not change a vector while it is processing calls since calls already in the vector could experience problems. Instead, add a new vector and change the VDN to point to the new vector.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● Vectoring (Basic) ● Vectoring (G3V4 Enhanced) ● Vectoring (G3V4 Advanced Routing) ● Vectoring (ANI/II-Digits Routing) ● Vectoring (Attendant Vectoring) ● Vectoring (Holiday Vectoring) ● Vectoring (Variables) ● Vectoring (3.0 Enhanced)
Vector Directory Number	All
Announcements/Audio Sources	Complete all fields for each extension that provides a vectoring announcement
Hunt Group	<ul style="list-style-type: none"> ● Vector ● ACD
Call Vector	Complete a screen for each vector
Feature-Related System Parameters	<ul style="list-style-type: none"> ● Vector Disconnect Timer ● Music/Tone on Hold ● Port ● Music (or Silence) on Transferred Trunk Calls
Vector Routing Table	All
Holiday Table	All

Fields that do not allow VDN extensions

You can not enter a VDN extension in the fields listed in the following table.

Screen	Field
Announcements/Audio Sources	Extension Number
Call Coverage Answer Group	Group Member Assignments
Call Coverage Paths	Coverage Point Assignments, other than the last coverage point
Console Parameters	CAS Back-up Extension
Feature-Related System Parameters	<ul style="list-style-type: none"> ● ACA Long Holding Time Originating Extension ● ACA Short Holding Time Originating Extension ● Extensions With System wide Retrieval Permission ● Controlled Outward Restriction Intercept Treatment ● Controlled Termination Restriction (Do Not Disturb) ● Controlled Station-to-Station Restriction
Hospitality	<ul style="list-style-type: none"> ● Extension of PMS Log Printer ● Extension of Journal/Schedule Printer ● Extension of PMS ● Extension to Receive Failed Wakeup LWC Messages
Hunt Group and Agent LoginID with EAS	<ul style="list-style-type: none"> ● Supervisor Extension ● Member Extensions
Intercom Group	Member Extensions
Listed Directory Numbers	LDN Extensions
Loudspeaker Paging and Code Calling Access	Extension Numbers Assigned to Codes
Pickup Groups	Member Extensions
Remote Access	Remote Access Extension
Station Forms	Hunt to Station
Terminating Extension Group	Member Extensions

Fields that allow VDN extensions

You can enter a VDN extension in the following fields.

Screen	Field
Abbreviated Dialing Lists	
Call Coverage Paths	Allow it as the last coverage point only
Hunt Group	Night Destination
Listed Directory Numbers	Night Destination
Trunk Groups	<ul style="list-style-type: none"> ● Night Destination ● Incoming Destination

You cannot enter a VDN extension as auxiliary data for the following buttons:

- Bridged Appearance (brdg-app)
- Data Call Setup (data-ext)

You can enter a VDN extension as auxiliary data for the following buttons:

- Remote Message Waiting Indicator (aut-msg-wt)
- Facility Busy Indication (busy-ind)
- Manual Message Waiting (man-msg-wt)
- Manual Signaling (signal)

Call Vectoring interactions

AP Demand Print: A VDN cannot be used as an argument to the feature access code 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 may 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 may also be accessed by the `queue-to split` and `check split` commands. Also, the messaging step may 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 AAR or ARS and the VDN's 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 FAC.

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 may 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 activation/deactivation call forwarding for a VDN.

An attendant or phone with console permission cannot activation/deactivation call forwarding for a vector-controlled hunt group.

Call Detail Recording: 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 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.

ACD call center features

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.

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 considered 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: Each VDN in the system has a 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 may 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 may be obtained using the `check` and `goto` Call Vectoring commands.

Calls may intraflow from an ACD split or skill that is not vector-controlled into one that is vector-controlled.

Leave Word Calling: LWC messages cannot be stored, canceled, or retrieved for a VDN.

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.

Send All Calls: If the destination of a `route-to with coverage no` command has the Send All Calls 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 All 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 AAR or ARS access codes, the TOD routing algorithm can be used to route the call.

Timed After Call Work (ACW): A Timed ACW interval can be assigned to a VDN.

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.

ACD call center features

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 DDD number using AAR/ARS. This TCM is the FRL associated with the VDN COR.

VDN in a Coverage Path: A call covering to a VDN can be routed to any valid destination by the call vectoring command `route-to`. The coverage option for the `route-to digits` command is disabled for covered calls. In other words, the `route-to digits with coverage=y` functions like the `route-to digits with coverage=n` command when processing covered calls. 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 Message Center system.

The class of restriction assigned to a VDN determines the partition group number (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 has no effect on the display shown on an answering display telephone. This station will show the normal display for a covered call.

Direct Agent Calling

This section includes the following topics:

- [About DAC calling](#) on page 145
- [Administering DAC](#) on page 147
- [DAC considerations](#) on page 147
- [Direct Agent Call \(DAC\) interactions](#) on page 150

About DAC calling

This section includes the following topics:

- [What is DAC?](#) on page 145
- [Advantages of DAC](#) on page 146
- [How DAC works](#) on page 146

What is DAC?

Note:

Direct Agent Calling (DAC) requires CallVisor Adjunct-Switch Application Interface (ASAI) or EAS. Both originating and called party Class of Restrictions (CORs) must be set to allow Direct Agent Dialing. See [Expert Agent Selection](#) on page 153 for information on Direct Agent Announcements (DAA).

DAC is an EAS feature that lets a caller:

- Contact a specific agent instead of a skill hunt group
- Queue for the agent if the agent is on a call
- Use Agent LoginID for callbacks and transfers
- Hear system-wide DAC delay announcement while holding
- Follow the agent's coverage path, if the call is not answered immediately.

Advantages of DAC

DAC calls have two important advantages:

- They reduce the need to transfer callers who want or need to speak with a certain agent, such as the agent spoken to on a previous call.
- They provide more accurate reporting of calls, because CMS counts direct agent calls as ACD calls. In this way, agents get proper credit for taking them. By comparison, calls transferred to an agent are not counted as ACD calls.

How DAC works

DAC works as described below:

- Callers can dial the agent's 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's work state, and the current work mode button on the agent's telephone flashes.
- If the agent is on a call, he or she 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, he or she can still answer a direct agent call in the queue by logging back in and becoming available. Agents who have direct agent calls waiting are not allowed to log off using a FAC. If the agent is in Manual In mode or pushes the After Call Work (ACW) button while on a direct-agent call, the agent goes to 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 *skill level* for the Call Handling Preference, direct agent calls must be assigned the highest priority for them to be delivered before other ACD calls. Otherwise, calls with a higher skill level are distributed before direct-agent calls.

Note that you can use Multiple Call Handling (MCH) to allow agents to answer a DAC with another ACD call active.

Direct agent calls follow the receiving agent's coverage and call forwarding paths, if these 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.

Administering DAC

To administer DAC:

- On the Agent LoginID screen, enter the agent's direct agent skill. It is suggested that you use the Hunt Group screen to set up a skill for all direct agent calls. This skill will:
 - Tell the communication server how to handle calls to the skill
 - Show report users how much time each agent has spent on direct agent calls

Note:

Any agent who will receive DACs should have at least one non-reserve skill assigned to the agent loginID.

- Add the skill to the agent's administered skills on this screen.
Whenever an outside caller dials the agent's extension, the communication server looks at the entry in that field to determine the skill for tracking call data.
- On page 8 of this Feature-Related System Parameters screen, you may specify:
 - A Direct Agent Announcement Extension that plays an announcement to DACs waiting in queue.
 - Amount of delay, in seconds, before the announcement.

You also need to administer a Class of Restriction (COR) for direct agent calls.

Direct Inward Dialing (DID) is administered on the Trunk Group screen.

On the second page of the Hunt Group screen, consider administering Multiple Call Handling On-Request for this hunt group. This feature will enable agents to see that the incoming call is a direct agent call and put their current call on hold to answer the direct agent call.

If there is no answer after a certain number of rings, you may use RONA to redirect the caller to a VDN that points to a vector. You can set up the vector to provide appropriate routing and treatment for the call.

On page 3 of the Hunt Group screen, administer messaging for the direct agent hunt group.

Next, you need to assign this hunt group to agents who need to receive direct agent calls.

DAC considerations

This section includes the following topics:

- [Maximum number of agents](#) on page 148
- [MIA across splits/skills](#) on page 148
- [Announcements](#) on page 148
- [Storing and retrieving messages](#) on page 148

ACD call center features

- [Class of Restriction](#) on page 149
- [Changing hunt groups from ACD to non-ACD](#) on page 150

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 splits/skills that the agent will be logged into. For CMS sizing, count one agent for each agent in each split or skill measured by CMS; one agent logged into three splits/skills counts as three agents.

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

MIA across splits/skills

MIA Across Splits/Skills distributes calls more equally to agents with multiple splits/skills. When agents handle a call for one split or skill, they go to the back of all their idle agent lists.

With MIA Across Splits/Skills, agents may not receive calls from all of their splits/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 these skills may never become the most-idle for skill 22 because they continuously take calls for split 20.

Announcements

Announcements can be analog, aux trunk, DS1, or integrated. Integrated announcements use the TN750, TN2501AP, or co-resident announcement board, and queuing is based on whether one of the playback channels is available. When a channel becomes available, any announcements on the board can be accessed, including the announcement already being played. A caller may 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 allow for more announcements.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Storing and retrieving 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 telephone display and proper authorization. You can also assign a remote Automatic Message Waiting lamp to an agent telephone to indicate when a message has been stored for the split or skill.

Class of Restriction

Each ACD split or skill and each individual agent is assigned a Class of Restriction (COR). You can use Miscellaneous Restrictions to prohibit selected users from accessing certain splits/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 telephone 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 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

- Agents should not be used for hunt group calls and ACD split or skill calls simultaneously. Otherwise, all calls from one split or skill (either ACD or hunt group) 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 multiappearance phones can receive only one ACD call at a time unless Multiple Call Handling is active. Without MCH, a phone is available for an ACD call only if all call appearances are idle. The agent may, however, receive non-ACD calls while active on an ACD call.

Vector-controlled splits/skills

- You can enhance ACD by using Call Prompting, Call Vectoring and Expert Agent Selection. For detailed information on vector-controlled splits/skills, see *Avaya Call Center Call Vectoring and EAS Guide*. Vector-controlled splits/skills should 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/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 Auxiliary Work button, the button lamp lights. To become available for calls, the agent presses the Auxiliary Work button or dials the Hunt Group Busy Deactivation FAC followed by the hunt-group number.

Direct Agent Call (DAC) interactions

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. It is not possible to determine which agent extension to intrude upon.

Automatic Callback: Automatic Callback calls cannot be activated toward 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 at least one agent is on an ACD call or in ACW mode. Queued calls remain in queue until the Coverage Don't Answer Interval expires before redirecting to coverage. If any split or skill agent becomes 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 agent's call coverage path. If an agent activates Send All Calls it does not affect the distribution of ACD calls. An ACD split or skill call directed to an agent station follows the split or skill call coverage path, once the agent's 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 at least one call appearance and the queue, if there is one, is full.
- No agents are logged in.
- All agents are in Auxiliary Work mode.

Call Forwarding All Calls: Call Forwarding All Calls activated for an individual extension does not affect the extension's ACD functions.

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). The system reports to BCMS/CMS that calls are queued on the split or skill. The system reports to 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. See [Intraflow and Interflow](#) on page 143 for more information.

Data Call Setup: Telephone 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 activated do not hear the usual zip tone.

DCS: CMS cannot measure ACD splits/skills on a Distributed Communications System (DCS) network as if they were one communication server. Agents for a split or skill must be all on the same communication server. 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 can make and receive dial intercom calls.

Forced Agent Logout from ACW mode: 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.

ACD call center features

Hold: If an agent puts an ACD call on hold, information is reported to the CMS using Personal Call Tracking. CMS records the amount of time the agent actually talks on the call.

Individual Attendant Access: Individual attendant extensions can be assigned to ACD splits. Unlike telephone 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/skills to be redirected to other destinations on and outside the system.

Multiappearance Preselection and Preference: All assigned call appearances must be idle before an ACD call is directed to a phone.

Location Preference Distribution: Direct Agent calls take precedence over Location Preference Distribution.

Night Service - Hunt Group: When Hunt Group 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. If all agents in the hunt group are busy, the caller hears whatever you have assigned.

Terminating Extension Group: 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 re-connected to the call. If an agent presses the Transfer button, dials the hunt-group extension number, and then 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.

Expert Agent Selection

This section includes the following topics:

- [About EAS](#) on page 153
- [Administering EAS](#) on page 153
- [Administering Direct Agent Announcement](#) on page 154
- [Detailed administration for EAS](#) on page 155
- [Call handling preference administration](#) on page 156
- [EAS considerations](#) on page 158
- [EAS interactions](#) on page 158

About EAS

Use Expert Agent Selection (EAS) to route incoming Automatic Call Distribution (ACD) calls to the agent who is best qualified to handle the call. That is, the agent with the specialized skills or experience required to best meet the caller's needs.

In addition, EAS provides the following capabilities:

- You assign all agent functions to the agent login ID and not to a physical phone. Therefore, EAS agents can login to and work at any phone in the system.
- Using the agent login ID, a caller places a call directly to a specific agent. These calls can be treated and reported as ACD calls.

EAS ensures the best possible service to the caller.

Administering EAS

The following forms and fields are required to administer the EAS feature.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● ACD ● Expert Agent Selection (EAS) ● EAS-PHD ● Vectoring (Basic)
Agent LoginID	All

ACD call center features

Screen	Field
Hunt Group	<ul style="list-style-type: none">● Skill● ACD● Vector● Set all three fields to y.¹● Group Type (ucd/ead)
Vector Directory Number	1st/2nd/3rd Skill (optional)
Class of Restriction	Direct Agent Calling
CDR System Parameters	Record Called Agent Login ID Instead of Group or Member
Call Vector	All
Feature-Related System Parameters	<ul style="list-style-type: none">● Expert Agent Selection (EAS) Enabled● Minimum Agent-LoginId Password Length● Direct Agent Announcement Extension/Delay● Message Waiting Lamp Indicates Status For

1. If the Message Center field is set to AUDIX on the Hunt Group screen, the Skill and ACD fields must be set to y, but the Vector field can be set to either y or n.

EAS must be both optioned on the System Parameters Customer-Options screen, and enabled on the Feature-Related System Parameters screen. Once EAS is optioned, you can complete most of the EAS-related administration prior to enabling the feature.

When EAS is optioned on the System-Parameters Customer-Options screen, Skill Hunt Groups replace splits. In addition, help messages, error messages and field titles change from **Split** to **Skill** on various forms.

Any EAS agent login ID must be part of the station numbering plan.

Physical aspects of the phone, such as the set type and button layout, are associated with the phone and not the login ID. On the Station screen with the EAS feature optioned, when a work-mode button is selected, no Gp information can be entered. The assist and queue status buttons require that **Group** be entered.

Administering Direct Agent Announcement

Direct Agent Announcement (DAA) enhances Direct Agent Calling (DAC) capabilities for CallVisor Adjunct-Switch Application Interface (ASAI) and Expert Agent Selection (EAS). It plays an announcement to DAC waiting in a queue. The following forms should be administered for DAA.

You must also have enabled either Expert Agent Selection (EAS) or ASAI Adjunct Routing (or both).

Screen	Field
System-Parameters Customer-Options	<ul style="list-style-type: none"> ● ACD ● Vectoring (Basic) ● Expert Agent Selection (EAS) or ● ASAI Adjunct Routing
Feature-Related System Parameters	<ul style="list-style-type: none"> ● Direct Agent Announcement Delay ● Direct Agent Announcement Extension
Announcements/Audio Sources	All

Detailed administration for EAS

In general, EAS uses vectors to route calls to agents with the required skills. To administer EAS you must:

- Assign skills to VDNs on the Vector Directory Number screen.
- Create vectors that will route a call to the correct skill.
- Assign skills with priority levels to agents on the Agent Login ID screen.

VDN administration: You can administer up to three VDN skill preferences on the Vector Directory Number screen in the 1st Skill, 2nd Skill and 3rd Skill fields. These fields indicate the skills that are required to handle calls to this VDN. All of the VDN skills on the VDN screen are optional. For example, only the first and third, or only the second and third VDN skills might 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 1st on the VDN screen.

Vector administration: When a call routes to a VDN, the VDN directs the call to the 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 either to specific skill numbers or to the skill preferences administered on the Vector Directory Number screen.

Agent administration: Assign skills to each agent. In addition, assign a skill level to each skill for the agent. When a vector routes incoming calls to a skill, the call is delivered to an available agent with the skill assigned. If no agents are available, the call is queued until it can be answered by an agent who has the skill required to handle the call.

Agent selection: The administered agent selection method and Call Handling Preference determine which agent will receive an incoming call.

Agent selection method: EAS can use either Uniform Call Distribution (UCD) or Expert Agent Distribution (EAD) to select agents for calls. Both methods can use the Most-Idle Agent (MIA) or the Least Occupied Agent (LOA) algorithm to select agents.

For more information on agent selection methods, see [Automatic Call Distribution](#) on page 82.

Call handling preference administration

This section includes the following topics:

- [MIA Across Splits/Skills](#) on page 156
- [Additional agent login ID capabilities](#) on page 157
- [DAC](#) on page 157

The call handling preference selected on the Agent Login ID screen can route calls based on either greatest need or agent skill level. The following table summarizes how a call is routed based on greatest need or agent skill level administration with either UCD or EAD distribution.

If:	EAD/UCD with Skill Level	EAD/UCD with Greatest Need
Agents are available. When a new call arrives it is delivered to:	<ul style="list-style-type: none"> ● EAD - Most-idle agent with the highest skill level for the calls' skill. ● UCD - Most-idle agent with the call's skill. 	<ul style="list-style-type: none"> ● 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, calls are in queue. When an agent becomes available, he or she receives:	<ul style="list-style-type: none"> ● EAD - Highest priority oldest call waiting for agent's highest level skill with calls in queue. ● UCD - Highest priority oldest call waiting for the agent's highest level skill with calls in queue. 	<ul style="list-style-type: none"> ● EAD - Highest priority oldest call waiting for any of the agent's skills. ● UCD - Highest priority oldest call waiting for any of the agent's skills.

MIA Across Splits/Skills

In addition, both UCD and EAD can be used in conjunction the MIA Across Splits/Skills option. With MIA Across Splits/Skills, the system has the option to either:

- Retain the agent's position in other splits/skills MIA lists while handling an ACD/DA call (default), or
- Remove the agent from all MIA lists when handling a call from any of the splits/skills.

The distribution is based on total call activity rather than activity in a single skill.

See [Automatic Call Distribution](#) on page 82 for more information about UCD, EAD, and MIA Across Splits/Skills.

Additional agent login ID capabilities

The following capabilities are also associated with agents' login IDs.

Auto-Answer : When EAS is optioned, auto answer settings can be assigned to agents on the Agent LoginID screen. An agent's auto answer setting will apply to the station where the agent logs in. If the auto answer setting for that station is different, the agent's setting overrides the station's.

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 that login ID is logged in. Calls to the login ID reach the agent independent of the phone the agent is currently using. For example, when agents use multiple phones because they have multiple offices or rotate desks, login IDs allow these agents to be reached independent 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 the agent is logged 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 is logged 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.

DAC

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

See [Avaya Business Advocate](#) on page 98 for additional information on how DAC works, is used in the call center, and is administered.

Direct agent calls are treated as ACD calls and receive 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.

ACD call center features

Any of the agent's skills can be the direct agent skill. When greatest need is optioned as the Call Handling Preference, the agent always gets direct agent calls before any skill calls. This is because direct agent calls have a higher priority than skill calls. However, when skill level is optioned as the Call Handling Preference, the agent will get direct agent calls first only if the direct agent skill has the agent's highest skill level. Otherwise calls from a skill with a higher level will be distributed before direct agent calls. If the direct agent skill and another skill are the same skill level, the agent will always receive direct agent calls before the other skill calls because direct agent calls have a higher priority.

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

EAS considerations

Station User records cannot be shared between TTI ports and EAS LoginID extensions. This causes a reduction in the number of possible EAS LoginID 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 LoginIDs 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-into 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/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.

EAS interactions

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

Abbreviated Dialing: Abbreviated Dialing is used to log in or log out EAS agents. Abbreviated Dialing lists/buttons can only be administered for stations.

Add/Remove Skills: In the EAS environment, agents have the ability to add and remove skills during a login session by dialing a FAC. Other phone users with console permissions can add or remove an agent's skill on behalf of the agent. (Note that the ability to add and remove skills depends on whether a user has a class of restriction (COR) that allows adding and removing skills.)

Administration Without Hardware: EAS login ID extensions are extensions without hardware. Login ID extensions require space in the dial plan.

Agent Work Mode States: With EAS, agents can only be in a single work mode at any one time for all their skills.

Assist: The Assist feature can be used with a skill hunt group (for example, where there is one supervisor per skill hunt group). When assist is selected, a call is placed to the supervisor associated with the skill for the active call.

AUDIX: Calls to the EAS agent login ID can cover to AUDIX.

Auto-Available Splits/Skills: If a skill hunt group is administered as an Auto-Available Skill (AAS) the EAS login IDs assigned to this skill must also be administered as Auto-Available. When the communication server reinitializes, these login IDs are automatically logged in with the auto-in work-mode. If any communication server features attempt to change the work-mode to anything except to auto-in, this attempt is denied. This feature is not intended for human agents.

Automatic Answering with Zip Tone: The Automatic Answer option can only be administered for a physical extension.

Automatic Callback: Users cannot activate Automatic Callback to an EAS agent's login ID. They can activate Automatic Callback to the phone where the agent is logged in.

Call Forwarding: Skill hunt groups (since they 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 class of restriction. This 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 phone an EAS agent logs into.

Class of Service (COS): EAS agents do not have a COS associated with their login ID. Therefore, the COS of the telephone is not affected when an EAS agent logs into it.

Directed Call Pickup: An EAS agent can use the Directed Call Pickup feature to pick up a call and/or have his or her calls picked up by another agent. The Class of Restriction of the agent will override the Class of Restriction of the station where the agent is logged in.

If both the station's COR and the logged-in agent's COR allow the call to be picked up using Directed Call Pickup, the user picking up the call can use either the station's extension or the agent's loginID.

ACD call center features

Displays - Phone: When an EAS agent logs in, the display for originators who call the login ID shows the login ID and agent name (as administered using the Agent Login ID screen). Calls that the agent originates show the agent login ID and agent name at the receiving telephone display. However, the user can display 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 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 extension name of the agent is displayed.

Calls to the physical extension show the physical extension's number and name on the originator's display.

Leave Word Calling: When an EAS agent is logged into a station, the agent can only retrieve LWC messages left for that agent's login ID. To retrieve LWC messages left for that station, the agent must log out.

When an EAS agent is logged into a station, its 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's login ID.

Look Ahead Interflow: VDN skills are not sent to another ACD/PBX when a call interflows using Look Ahead Interflow. If skills have the same meaning on both ACDs, then a Look Ahead Interflow command to a VDN with the same skills assigned can provide a mapping of the skills.

Message Waiting Lamp: The Message Waiting Lamp by default tracks the status of messages waiting for the logged in EAS agent LoginID rather than messages for the physical extension. The operation of the Message Waiting Lamp can be changed so that it tracks the status of messages waiting for the physical extension where the agent is logged in. For more information about Feature-Related System Parameters, see *Administrator Guide for Avaya Communication Manager*.

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 needed by agents using that physical extension, given that enough buttons are available.

Service Observing: The Service Observing feature is activated in the EAS environment by dialing either the physical extension of the telephone where an EAS agent is logged in or the login ID of the agent.

Tenant Partitioning and agent skills: The Tenant Partitioning feature was designed to support multiple customers using the same Communication Manager server. Tenant Partitioning separates entities, thereby avoiding or reducing interactions between entities in different partitions.

Assign the same partition number to agents, groups, and entities to avoid blocking calls and to avoid 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 that are involved with ACD agents and hunt groups, such as VDNs and announcements

Note:

An agent's skill set should contain only skills belonging to the *same* tenant partition; not doing so can result in unintended behavior.

VuStats: VuStats displays can show an agent's skill assignments and can show some measurements by skill.

Forced Agent Logout from ACW mode

This section includes the following topics:

- [About Forced Agent Logout from ACW mode](#) on page 162
- [Reason to use](#) on page 162
- [Prerequisites](#) on page 162
- [Administering Forced Agent Logout from ACW mode](#) on page 163
- [Forced Agent Logout from ACW interactions](#) on page 164

About Forced Agent Logout from ACW mode

The Forced Agent Logout from After Call Work (ACW) feature automatically logs out an Expert Agent Selection (EAS) agent who spends too much time in ACW mode. The timeout period is specified on a per system basis and on a per agent basis. The timeout is reported with a customer-assignable reason code set on a system basis.

The per agent timeout setting takes precedence over the per system setting. For Auto-In agents, the Timed ACW feature takes precedence over the Forced Agent Logout from ACW feature.

Reason to use

This feature is typically used when customers want to:

- Require that agents 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 ACW mode

Prerequisites

You can set Forced Agent Logout from ACW only if all of the following conditions are true:

- Expert Agent Selection (EAS) is enabled and active.
- The Reason Codes feature is active. If the Reason Codes feature is not active, you can still set the maximum time the agent can be in ACW on a system-wide and on an agent basis, but you cannot administer a reason for the logout.
- The Call Center release is 3.0 or later.

If any of these values are not true, you will not be able to change the default values on the Forced Agent Logout from ACW fields that enable the feature.

Administering Forced Agent Logout from ACW mode

The following forms are used to administer Forced Agent Logout from ACW mode.

Screen	Field
Page 13 of Feature-Related System Parameters	Maximum time agent in ACW before logout (sec) - Sets the maximum time the agent can be in ACW on a <i>per system</i> basis.
Agent Login ID	Maximum time agent in ACW before logout (sec) - Sets the maximum time the agent can be in ACW on a <i>per agent</i> basis and sets the reason code that explains why the system logged out the agent.

Note:

Changes do not apply until the agent logs out and logs back in again.

Tips for administering Forced Agent Logout from ACW mode

Consider the following tips when administering Forced Agent Logout from ACW mode.

If	Then
You want this feature to apply to all agents, and you want the timeout period to be the same for all agents	<ol style="list-style-type: none"> 1. Set the Maximum time agent in ACW before logout (sec) field on the Feature-Related System Parameters screen to the desired timeout value in seconds. This value can be from 30 to 9999 seconds. 2. Leave the default setting of <i>system</i> on the Agent Login ID forms.
You want different timeout periods assigned to specific agents	Set the Maximum time agent in ACW before logout (sec) field on the Agent Login ID screen to the desired timeout value for each agent.
You do <i>not</i> want the timeout feature to apply to certain agents	Set the Maximum time agent in ACW before logout (sec) field on the Agent Login ID screen to <i>none</i> for those agents.

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.

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: The Timed ACW feature immediately switches an Auto-In agent into ACW mode for a specific length of time after the agent disconnects from a call. If both Timed ACW and Forced Agent Logout from ACW are administered, consider the following:

- 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.

Forced Agent Logout by Clock Time

This section includes the following topics:

- [About Forced Agent Logout by Clock Time](#) on page 165
- [Reason to use](#) on page 165
- [Prerequisites](#) on page 166
- [Cancelling forced logouts](#) on page 166
- [Administering Forced Agent Logout by Clock Time](#) on page 166
- [Forced Agent Logout from Clock Time interactions](#) on page 167
- [Forced Agent Logout from Clock Time example](#) on page 169

About Forced Agent Logout by Clock Time

The Forced Agent Logout from Clock Time feature allows administrators to:

- Set a specific time when the system automatically logs out Expert Agent Selection (EAS) agents.
- Set a logout reason code for agents on a system-wide basis.
- Administer the system so that agents can override this feature when they press a forced logout override button.

For more information, see [Cancelling forced logouts](#) on page 166.

If the agent is still on an ACD call when the forced agent logout time is reached, the agent is put into 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.

Reason to use

This feature allows you to set a pre-determined time to automatically log out agents when the agents forget to log out at the end of their shifts. You will have a more accurate view of staffing if off-shift agents are logged off the system and calls are not delivered to an autoanswer agent position after the agent leaves.

Prerequisites

You can set Forced Agent Logout by Clock Time only if all of the following conditions are true:

- The **Expert Agent Selection (EAS)** field is set to **y** on the System-Parameters Customer-Options screen.
- The **AAS?** field on the Agent LoginID screen is set to **n**.
- The **Call Center Release** field is set to 4.0 or later on the System Parameter Customer-Options screen.

Cancelling forced logouts

An agent can cancel a forced logout by:

- Pressing the flashing forced logout override button during a pending logout. The lamp turns off to indicate that the forced logout will not occur.
- Pressing the forced logout override button (lamp is dark) anytime during the agent's shift. If the override applies, the lamp lights steady. The lamp turns off when the forced logout time is reached, and resets for the next day. Agents can toggle the button to remove the override.
- Logging back in after the forced logout occurs. The forced logout resets for the next day.

If the forced logout override button flutters, one of the following has occurred:

- The agent has pressed the forced logout override button, but a logout time has not been administered.
- The agent pressed the forced logout override button to override the forced logout prior to the logout time, the forced logout time has passed, and the agent has tried to press the forced logout override button again.

Administering Forced Agent Logout by Clock Time

To administer Forced Agent Logout by Clock Time:

1. On the Feature-Related System Parameters screen (page 14), set the **Clock Time Forced Logout Reason Code** field to specify the logout reason and press Enter.
This sets the logout reason code for all agents.
2. For each agent, enter `change agent-loginID xxxx`.

3. On the Agent Login ID screen, set the **Forced Agent Logout Time** field to the time of day when you want to log out the agent and press Enter. You can use the agent's local time. See the Multiple Locations Feature description in [Forced Agent Logout from Clock Time interactions](#) on page 167 for details.

Note:

Changes do not apply until the agent logs out and logs back in again.

4. Enter `change system-parameters features`.
5. If you want to allow the agent to override this feature, administer a forced logout override button (logout-ovr) on the Station screen.

For information about administering feature buttons, see *Administrator Guide for Avaya Communication Manager*.

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.

Call hold: If the agent has ACD, ACDO, 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 has not yet been 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 Caller-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 these conditions exist, the agent is put into pending logout mode while remaining connected to the conference.

DAC: The Forced Agent Logout by Clock Time feature applies 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 (lamp lit), the system will not log the agent out when the forced logout time arrives.

Improved Integration with Proactive Contact: You should 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 will be 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 will not be 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 savings 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, conference tones, and so on. 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.

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 from Clock Time example

The ABC Call Center has groups of agents in several different timezones. 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 timezone. 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 (set on Login ID screen)	Physical Station	Location for Physical Station (used for timezone 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.

ACD call center features

Location	Name	Timezone 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 timezone offset is defined relative to the main location.

Inbound Call Management

This section includes the following topics:

- [About ICM](#) on page 171
- [Administering ICM](#) on page 171
- [ICM detailed description](#) on page 172
- [ICM considerations](#) on page 176
- [ICM interactions](#) on page 177

About ICM

Inbound Call Management (ICM) allows you to integrate features of the communication server with host-application processing and routing, and automate delivery of caller information to agents' displays. You can create a sophisticated system to handle inbound calls for applications such as telemarketing and claims processing.

To implement ICM, you integrate features of the communication server such as Automatic Call Distribution (ACD), Expert Agent Selection (EAS) Call Vectoring, Direct Agent Calling (DAC), and Call Prompting with an application on a host processor. The host application, or adjunct, can be a CallVisor/PC, an IVR voice system, Telephony Services Server serving a local-area network, or a vendor application using the CallVisor Adjunct/Switch Applications Interface (ASAI). A CallVisor ASAI link between the communication server and adjunct allows the adjunct to control incoming call processing and routing.

In addition, you can automate ACD agent telephone displays and associate them 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). Or, you can set up the adjunct to retrieve caller information from a database and display it on a particular agent's screen, based on the service dialed.

See [ICM detailed description](#) on page 172 for more information on applications.

Administering ICM

Display the System-Parameters Customer-Options and ensure that the ACD option is enabled. If you are using CallVisor ASAI or Call Vectoring, ensure appropriate ASAI Capability Groups options or the Vectoring (Basic) and/or Vectoring (Prompting) options are enabled.

ACD call center features

Administer the system as defined in the table at [Administering ACD](#) on page 86 with the following additional considerations.

Required forms - ICM

Screen	Field
display System-Parameters Customer-Options - Verify options are active	<ul style="list-style-type: none">● ASAI Link Core Capabilities● ASAI Link Plus Capabilities
Trunk Group (ISDN-PRI)	Per Call CPN/BN

Trunk Group screen: Administer the Per Call CPN/BN field for the appropriate ISDN-PRI trunk group. The corresponding information is sent with a call-offered event report to the adjunct.

Hunt Group screen: Complete a Hunt Group screen for each split or skill that the ICM adjunct will monitor.

Call Vector screen: If you are using Call Vectoring, an ASAI link interface extension number is required for adjunct routing vector commands. This extension is the same as the one you enter on the Station screen.

See [Automatic Call Distribution](#) on page 82, [Call Vectoring](#) on page 137, [Call Prompting](#) on page 134, and CallVisor Adjunct-Switch Application Interface and any other features you are implementing for ICM for additional administration requirements.

ICM detailed description

This section includes the following topics:

- [Applications](#) on page 173
- [Agent data screen delivery applications](#) on page 173
- [Integration with speech processing adjuncts](#) on page 174
- [Host/adjunct call routing](#) on page 176

Applications

The following are some typical ICM applications:

- The system passes calling party/billing number (CPN/BN) information and the call is routed to an adjunct application for screen pop and supervisory transfers, with screen duplication.
- The system sends to the adjunct application both caller and prompter information about all incoming calls to a particular number. According to caller information in a database, the application directs the communication server to route the call. For example, the call could be routed to a preferred agent, to best customer treatment, or to accounts receivable.
- The system uses Call Prompting to obtain a customer account number and then passes this information to the adjunct for call routing or screen pop.
- The system connects the caller to a voice response unit (VRU), along with caller CPN/BN and DNIS information. The caller then interacts with the VRU to direct how the call is handled. The system can verify a caller's identity and provide access to database information such as claims status or account balance.
- With Direct Agent Calling (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 Call Management System (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 server. For example, an application at one communication server can determine a caller's account or claim number and pass this information to a special list on another communication server, where an application will transfer the call.

For additional application scenarios, see *Avaya Call Center Call Vectoring and EAS Guide*.

Agent data screen delivery applications

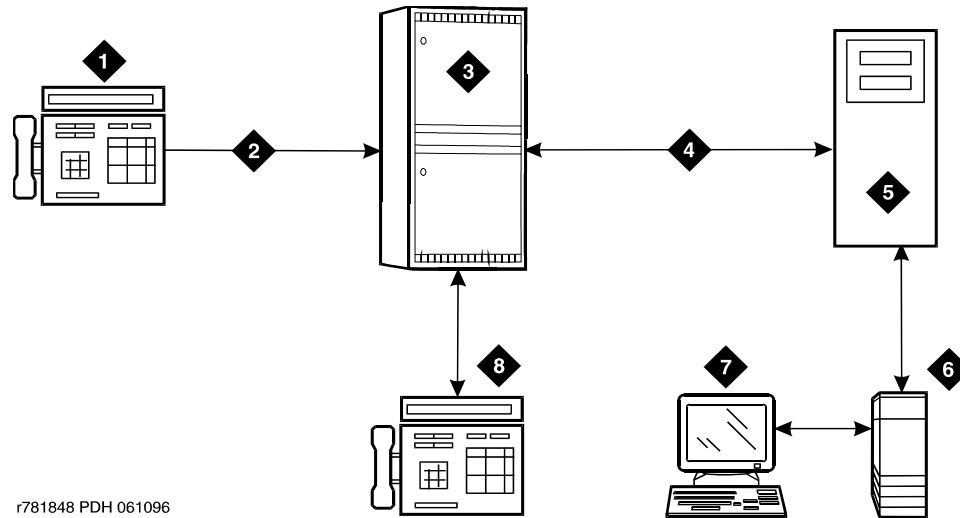
You can use an Interactive Voice Response (IVR) system to deliver appropriate display data about callers to agents. You can pass information such as CPN/BN, DNIS, and Look-Ahead Interflow information, digits collected from Call Prompting, and which agent is selected to an IVR system. The IVR system delivers 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 system 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 should be verified for a particular application. If the host supports ASAI, the IVR system is not needed.

Simplified ICM configuration for data screen delivery



- | | |
|-----------------|------------------------|
| 1. Telephone | 5. IVR |
| 2. ISDN-PRI | 6. Host |
| 3. Avaya switch | 7. Agent data terminal |
| 4. ASAI | 8. Agent telephone |

General processing for this type of application occurs as follows.

1. An Interactive Voice Response (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, when the call drops, and so on.
3. The IVR system sends information to the host application so that it can send a data screen to the agent's data terminal.

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

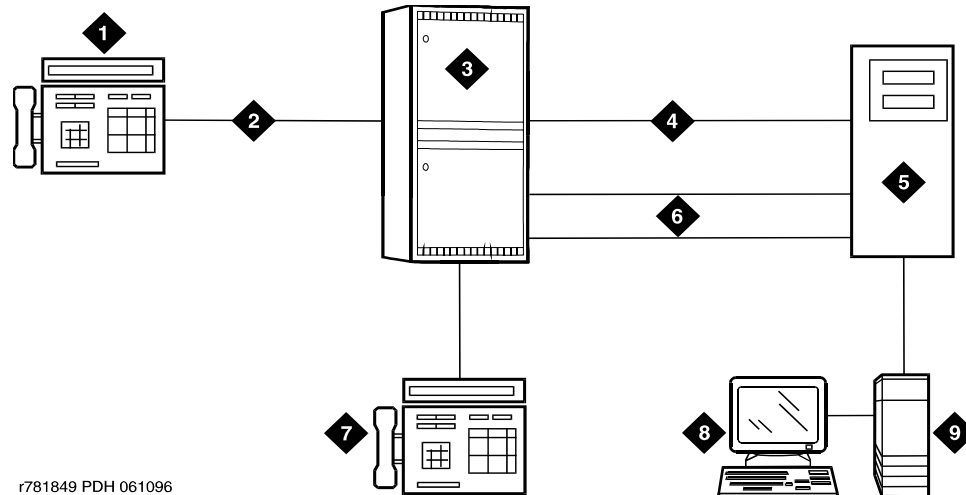
ICM can be used to provide integration with VRUs. The advantages of using ICM with CallVisor ASAI in addition to tip/ring interfaces are as follows:

- Data-screen integration is provided on transferred calls.
- Answer notification is provided on internal calls (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.

Simplified ICM configuration for speech processor integration



- | | |
|---------------------|------------------------|
| 1. Phone | 6. Tip/ring lines |
| 2. ISDN-PRI | 7. Agent phone |
| 3. Avaya switch | 8. Agent data terminal |
| 4. ASAI | 9. Host |
| 5. Speech processor | |

General processing for this type of application occurs as follows:

1. The communication server uses CallVisor ASAI link to pass incoming call information to the IVR voice system.
2. The split or skill on the communication server distributes the call to an available voice line.
3. After digits are collected using a DTMF keypad, the IVR system transfers the call back to a split or skill or specific agent on the communication server using CallVisor ASAI.
4. If the call is transferred to an agent, the communication server uses CallVisor ASAI link to pass an event report on which agent receives the call.
5. The IVR system forwards the agent ID to the host application, which delivers a data screen to the agent.
6. Agents can display collected digits on their data terminals. Except for the dialed number, information from an IVR system cannot be carried with the call and displayed on a phone. For example, digits collected in an IVR system adjunct cannot be passed to the communication server for display.

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7. If the collected digits are the extension where the call is being routed, these routing digits are passed to the communication server as the destination in the CallVisor ASAI third-party make-call request. The IVR system uses the request to set up various types of calls.

Host/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

Administrators and planners must consider:

- ICM traffic
- Rated communication server capacity
- CallVisor ASAI interface traffic
- Rated capacity of the adjunct application processor

Avaya Technical Design Center can provide planning assistance.

In addition, you must consider the following:

- 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](#) on page 82 for more information on direct agent calls.

Priority Calling: CallVisor ASAI allows both Priority Calling and DAC for the same call.

Information Forwarding

This section includes the following topics:

- [About Information Forwarding](#) on page 178
- [Administering User-to-User Information transport for ISDN trunks](#) on page 179
- [Administering User-to-User Information transport for SIP trunks](#) on page 181
- [Information Forwarding detailed description](#) on page 182
- [Administering the UUI station button](#) on page 183
- [Information Forwarding considerations](#) on page 183
- [Information Forwarding interactions](#) on page 185

About Information Forwarding

Whenever the communication server interflows a call over ISDN (for example, PRI or BRI) or SIP trunk facilities 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.

Administering User-to-User Information transport for ISDN trunks

This section outlines the procedure to administer incoming and outgoing ISDN trunk groups to send user data over the network. Before administering trunk groups, review these guidelines:

- If you are using shared UUI (any Supplementary Service other than b), then you must administer the `UUI Treatment` for the trunk groups (both outgoing and incoming at the remote end) as shared. Use this option when you want to forward information to the communication server in non-QSIG networks.
- With QSIG (Supplementary Service b), you need to administer Shared UUI to include ASAI user information with MSI transport. If `UUI Treatment` is the default service-provider, the communication server forwards the ASAI user data (if provided) in a non-shared codeset 0 UUI IE while forwarding the other data as MSI.

Note:

You do not need to complete this procedure if you do not intend to send user data over the network.

The following table lists the screen and fields needed to administer information transport on ISDN trunk groups.

Screen	Fields	Why is this field needed?	Optional field?
ISDN Trunk Group (BRI or PRI)	UUI IE Treatment	Set field to either: <ul style="list-style-type: none"> • Shared (for trunk groups connected to the communication server, if you want shared data). • Service-provider (for trunk groups connected to communication server releases prior to R6.3, or if you want service provider functionality). 	Yes However, this field cannot be blank.
	Maximum Size of UUI IE Contents	Set according to what the network supports. 128 (default) is recommended for private networking.	Yes
	Shared UUI Feature Priority fields	Set the priority for each type of user data such as UCID, ASAI, and other application information. Only needed for non-QSIG trunk groups.	Yes If blank, information not sent.

To administer Shared UUI information transport:

1. In the command line, enter:


```
change trunk-group n
```

 where `n` is the number of the trunk group you want to administer.
 The Trunk Group administration screen is displayed.

ACD call center features

2. Go to page 3 of the screen.



CAUTION:

If you want service provider functionality, do not enter **shared** in page 3 in the `UUI Treatment` field. Instead, leave the default **service-provider** in this field.

3. If the trunk group is not connected to an early version communication server, or if you otherwise do not want service provider functionality, enter **shared** in the `UUI Treatment` field.

The `Maximum Size of UUI IE Contents` field appears.

Note:

If you enter **shared** and the `Send Codeset 6/7 LAI` trunk group option is on, you send the LAI information twice (unless the LAI Name and Other LAI data items' priorities are blank with non-QSIG - with QSIG, both are always sent), and you may exceed the maximum ISDN message size.

4. If you want to change the default size of 128 in the `Maximum Size of UUI IE Contents` field, then enter the number for the maximum UUI size. If you want to keep the default size, go to Step 5.

You must administer the trunk groups to send the appropriate amount of user information over the connected network. For example, if the public network only supports 32 bytes of user information, and you enter a number larger than 32, the network may reject the entire UUI IE.

The communication server accepts a range from 32 to 128.

5. Go to page 5 (the Shared UUI Feature Priorities page which displays only when the `UUI IE Treatment` field is set to **shared**).

Notice that all feature names (whether enabled or not) appear on this page. The default values were assigned when Shared UUI was enabled.

6. Either leave the default settings, or reassign numbers from 1 to 6 (1 is the highest priority) to each feature. For more information about user needs, see [Determining user information needs](#) on page 183.

Note:

If you leave a feature field blank, information about that feature will not transport in the UUI IE. If the public network supports less than 128 bytes, you need to choose what feature information you want to send, and give that feature field a higher priority.

7. Press **Enter** to save your changes.

Administering User-to-User Information transport for SIP trunks

The following table lists the screen and fields needed to administer information transport on SIP trunk groups.

Screen	Fields	Why is this field needed?	Optional field?
SIP Trunk Group	UUI Treatment	Set field to Shared.	Yes However, this field cannot be blank.
	Maximum Size of UUI Contents	Set according to what the network supports. 128 (default) is recommended for private networking.	Yes
	Shared UUI Feature Priority fields	Set the priority for each type of user data such as UCID, ASAI, and other application information.	Yes If blank, information not sent.

To administer Shared UUI information transport:

1. In the command line, enter:

```
change trunk-group n
```

where *n* is the number of the trunk group you want to administer.

The Trunk Group administration screen is displayed.

2. Go to page 3 of the screen.
3. Enter **shared** in the UUI Treatment field.

The Maximum Size of UUI IE Contents field appears.

4. If you want to change the default size of 128 in the Maximum Size of UUI IE Contents field, then enter the number for the maximum UUI size. If you want to keep the default size, go to Step 5.

You must administer the trunk groups to send the appropriate amount of user information over the connected network. For example, if the public network only supports 32 bytes of user information, and you enter a number larger than 32, the network may reject the entire UUI.

The communication server accepts a range from 32 to 128.

5. Go to page 4 (the Shared UUI Feature Priorities page which displays only when the UUI Treatment field is set to shared).

Notice that all feature names (whether enabled or not) appear on this page. The default values were assigned when Shared UUI was enabled.

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6. Either leave the default settings, or reassign numbers from 1 to 6 (1 is the highest priority) to each feature. For more information about user needs, see [Determining user information needs](#) on page 183.

Note:

If you leave a feature field blank, information about that feature will not transport in the UUI. If the public network supports less than 128 bytes, you need to choose what feature information you want to send, and give that feature field a higher priority.

7. Press **Enter** to save your changes.

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.

Beginning with Call Center 5.0, user-to-user information can also be transported over SIP trunks.

This feature:

- Enables multiple applications on the communication server to share the contents of the UUI IE or MSI
- Allows for 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 (see [Look-Ahead Interflow](#) on page 188).

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 wherever they are (see [Best Service Routing](#) on page 110).
- Universal Call ID - provides a means to collect and trace call data from multiple call centers (see [Universal Call ID](#) on page 277).

Determining user information needs

The network byte limit on user information contents (the user data part of the UUI) must be large enough to carry the data needed for the customer application.

If you want to forward information over a network that does not support at least 82 bytes of user data, you must determine the space required for the application and adjust priorities accordingly.

Administering the UUI station button

Administrators can set it up so that agents can press the User-to-User Information (UUI) button to display call-related information, such as customer account numbers. This includes:

- Information that an adjunct provides to Communication Manager
- Information from a remote site, such as data forwarded with the call

The station displays up to 32 characters of Adjunct Switch Application Interface (ASAI) user data associated with the call or inserted by the ASAI.

Reason to use. The ability to display the UUI data using a station set button provides another way to provide the agent with caller information.

Administration.

1. Administer the uui-info button as a feature button using the `change station x` or the `change attendant x` commands.

For more information about how to set up feature buttons, see *Administrator Guide for Avaya Communication Manager*.

2. Use the `display cor x` command to get to the **CLASS OF RESTRICTION** screen. There is a new field on page 2 called **Station-Button Display of UUI IE Data?**
3. Type `y` to allow the use of this feature.

This Class of Restriction (COR) is then assigned to the agents equipped with the button.

Information Forwarding considerations

- Enhanced information forwarding has been tested with several major carriers. To find out if these 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 involved with the preferred specific configuration, use of private trunking between the nodes should be assumed until successful testing has been completed.

ACD call center features

- 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. For more information, see the [Troubleshooting](#) on page 184 section.
- The `Send codeset 6/7 LAI` trunk group option operates independently of the `UUI IE Treatment` trunk group option. However, if you turn both of these options on, you'll 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:
 - put a blank in the priority fields for `VDN Name` and `Other LAI Information` on the Shared UUI Feature Priorities screen, or
 - disable the `Send codeset 6/7 LAI` option.
- For non-QSIG or QSIG trunk groups to the communication server that require information forwarding, the `UUI IE Treatment` should be shared and the `Send Codeset 6/7 LAI IE` should be n.
- Information transported using the Shared UUI will not work with non-Avaya switches unless they adhere to the proprietary encoding.

Troubleshooting

The following troubleshooting hints should be reviewed when information is not forwarded, even though you received no error messages while administering the Shared UUI feature, and all software and connections meet the minimum requirements:

- If DCS is used, make sure all ISDN trunks between the communication server used for DCS or remote AUDIX are configured in the D-channel mode.
- For each ISDN trunk administered with the Shared UUI option, make sure the UUI size does not exceed the UUI IE size that the network can support.
- For all non-QSIG ISDN trunks and SIP trunks, make sure the `UUI Treatment` field is set to shared.
- Make sure trunk group options are set correctly for the application and configuration.
- Applications may fail on networks supporting limited UUI transport. Administration determines which application's UUI will be transported in these cases. If a given application is failing, first check the administration to determine if the application in question has the highest priority. This applies to tandem nodes as well as originating nodes.

Applications that originate UUI on tandem nodes can request that assigned priorities at the tandem node be applied to the resulting UUI. Therefore, it is possible for a tandem node to erase UUI information received from the originator. Passing UUI through a tandem node transparently, as required for UUS Service 1, does not apply to the proprietary shared UUI procedures of the communication server.

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.

Intraflow and Interflow

This section includes the following topics:

- [About Intraflow](#) on page 186
- [About Interflow](#) on page 186
- [Administering Intraflow and Interflow](#) on page 187
- [Intraflow and Interflow detailed description](#) on page 187
- [Interflow and Intraflow considerations](#) on page 188
- [Interflow and Intraflow interactions](#) on page 188

About Intraflow

Use Call Coverage with Intraflow to redirect ACD calls from one split or skill to another conditionally, according to the coverage path's redirection criteria. For example, you can define a split or skill's coverage path to automatically redirect incoming ACD calls to another split or skill when a telephone is busy or unanswered. You can redirect calls to less busy splits/skills, for more efficient call handling.

Use Call Forwarding with Intraflow to unconditionally forward calls for a split or skill.

About Interflow

Interflow allows you to redirect ACD calls from a split or skill on one communication server to a split or skill on another communication server or external location. Use Call Forwarding All Calls with Interflow to unconditionally forward calls directed to a split or skill to an off-premises location. Calls can be forwarded to destinations off the communication server (that is, phone numbers on the public telephone network). You cannot use Call Coverage with Interflow. If a coverage point station or split or skill is forwarded/interflowed, it is taken out of the coverage path.

For details on how to forward calls to an external extension and on Call Coverage redirection criteria, see *Feature Description and Implementation for Avaya Communication Manager*. See [Call Vectoring](#) on page 137 and [Look-Ahead Interflow](#) on page 188 for information on advanced Interflow capabilities.

Administering Intraflow and Interflow

The following forms and fields are required to administer the Intraflow and Interflow feature.

Screen	Field
Feature-Related System Parameters	Coverage - Don't Answer Interval for Subsequent Redirection
Feature Access Code (FAC)	<ul style="list-style-type: none"> ● Call Forwarding Activation ● Call Forwarding Deactivation
Hunt Group	<ul style="list-style-type: none"> ● Inflow Threshold ● Priority on Intraflow
Coverage Paths	<ul style="list-style-type: none"> ● Don't Answer ● Busy ● Number of Rings

Intraflow and Interflow detailed description

Assign an inflow threshold for each split or skill receiving Intraflow and Interflow calls. This threshold prevents a split or skill from receiving new ACD calls if the oldest call in the queue has been there longer than the threshold. If an ACD call is forwarded or redirected through Call Coverage, but cannot be routed to another split or skill or coverage path point, it remains in queue at the original split or skill even though coverage tone may be heard.

For a split or skill with Intraflow and Call Coverage assigned, 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, that call is placed ahead of non priority calls but behind other priority calls already in the queue. All priority calls are answered before any non priority calls.

Calls intraflowed using Call Coverage to a covering split or skill are never connected to the first delay announcement at the covering split or skill. Calls redirected using Call Forwarding receive the delay first announcement at the forwarded-to split or skill, but never receive a forced first announcement.

As an illustration of how Intraflow works, assume the following:

- 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.

ACD call center features

- 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. Then 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's multiappearance phone. The agents use the button to identify a call that is intraflowed from another split or skill. When an agent receives such a call, the button lamp lights.

Interflow and Intraflow considerations

The same coverage path can be used for as many splits/skills as desired. You should 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

Call Coverage: All splits/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 may then 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's 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. For more information, see [About LAI](#) on page 196.

Location Preference Distribution

This section includes the following topics:

- [Description of Location Preference Distribution](#) on page 189
- [Reasons to use](#) on page 189
- [Prerequisites](#) on page 190
- [About location numbers](#) on page 190
- [Call-handling conditions](#) on page 192
- [How to administer Location Preference Distribution](#) on page 193
- [Local Preference Distribution interactions](#) on page 193

Description of Location Preference Distribution

Location Preference Distribution tries to route incoming Automatic Call Distribution (ACD) calls to agents located in the same location as the incoming trunk on which the call originated whenever possible. If an incoming caller cannot be matched with an agent in the same location, calls are routed to agents at different locations. In this case, the routing is determined by administered distribution algorithms 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.

Reasons to use

Customers can use this feature to:

- Lower customer networking costs by reducing the amount of intra-switch network traffic
- Improve audio quality

Prerequisites

You can set Location Preference Distribution only if all of the following conditions are true:

- Expert Agent Selection (EAS) is enabled and active.
- The **Multiple Locations** field is set to **y** on Page 4 of the Systems-Parameters Customer-Options screen.
- The Call Center release is 3.0 or later.

About location numbers

Agents obtain a location number when they log in. The location number remains with the agent until the agent logs out. Local Preference Distribution uses location numbers to match the incoming trunk and the agent.

This section includes the following topics:

- [The Multiple Locations feature](#) on page 190
- [Changes to the location number](#) on page 190
- [How trunks, stations, and agent endpoints obtain location numbers](#) on page 191
- [How to set up a location number](#) on page 191

The Multiple Locations feature

The location number used by Location Preference Distribution is defined by the Multiple Locations feature. The Multiple Locations feature was originally developed to display local time on station sets that are located in a different time zone from their connecting switch. Call Center added the ability to have the Call Management System (CMS) track and report on agents and trunks using a location number, or location ID. The location number for the agent is provided to CMS when the agent logs in. The location number for a trunk is provided to CMS when a measured incoming call is received.

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's ID obtained when the agent logged in and the call's ID when the call was originally received. During such a change, the Location Preference Distribution matching may not be appropriate until all agents log out and log back in again. The software does not check for these 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 that IP region.

If the location field is left blank on the Network Region screen, the IP phone derives its location from the CLAN board located on the cabinet or gateway where the phone is registered.
- IP and SIP trunks obtain their location from the cabinet containing the CLAN/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 details on how to use these forms and the commands associated with these forms, see *Administrator Guide for Avaya Communication Manager*.

Step 1: Establish location numbers

Use the Locations screen to establish location numbers. This will define the characteristics of the location that can include:

- Time zone offset between local standard time and the remote server location
- Daylight savings rules used by any expansion port networks (EPNs) located in different time zones
- Number plan area codes
- 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

Step 2: Assign location numbers to EPN cabinets or to the Media Gateway

Do one of the following tasks:

- Use the Cabinet Description screen to assign location numbers to the appropriate EPN cabinets.

Use the `change cabinet xx` command.
- Use the Media Gateway screen to assign location numbers to the Media Gateway.

ACD call center features

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.

Step 3: Assign the location by IP network region

Use the Change IP-Network-Region screen to administer the location by IP network region.

This sets the following conditions:

- The correct date and time information and trunk routing based on the IP network region.
- 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 Avaya Communication Manager endpoints, and in the same location as the S8300 Media Server or remote office users. This allows IP endpoint users the ability to move from location to location and always have correct display information. Remote users are identified in a network region and location that routes them to correct 911 services or notifies them through announcements that they are in a different 911 jurisdiction than where they are registered.

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 is 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 calling party's trunk or station. If there is no match for an idle agent, the incoming ACD call is routed to the agent at the top of the skill's free-agent list based on the administered selection criteria.

Call-surplus conditions

A call-surplus condition is 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.

How to administer Location Preference Distribution

The following forms are used to administer Location Preference Distribution. You can administer Local Preference Distribution to handle agent-surplus conditions, call-surplus conditions, or both.

Use the following screen	With the following field set to y	To set up an algorithm designed for
Hunt Group	Local Agent Preference	Agent-surplus conditions
Agent Login ID	Local Call Preference	Call-surplus conditions

Local Preference Distribution interactions

Only the features that are impacted by Local Preference Distribution are described in this section.

Avaya Business Advocate: Avaya Business Advocate provides call handling preferences based on:

- A service objective that is assigned on the Hunt Group screen.
- Agent percentage allocation that is assigned on the Agent Login ID screen on a per skill basis.

Local Preference Distribution takes precedence over any Avaya Business Advocate call handling preferences.

ACD call center features

Best Service Routing (BSR): Local Preference Distribution is used to select an available agent within the Call Center during **consider** and **queue-to best** step operations. Local Preference Distribution is not used across system sites. In this case, there is no notion of a multi-site network region.

Call Admission Control: Location Preference Distribution does not interact directly with the Call Admission Control 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 inter-node 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 multi-site feature - 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, as well as other types of networks.

Call handling conditions: For information about how call handling conditions are used with Location Preference Distribution, see [Call-handling conditions](#) on page 192

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: 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 (PA) assignment for the skill assigned on the Agent Login ID screen.
- The Service Objective 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.

EAS: Expert Agent Selection (EAS) must be enabled and active before you can assign Local Preference Distribution.

Inter-Gateway Alternate Routing (IGAR): IGAR provides the ability to alternately use the Public Switched Telephone Network (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.

IP hard and soft phones: For more information about how IP phones interact with Location Preference Distribution, see [How trunks, stations, and agent endpoints obtain location numbers](#) on page 191.

Multiple Locations: For information about how the Multiple Locations feature interacts with Location Preference Distribution, see [The Multiple Locations feature](#) on page 190.

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.

Percent Allocation: See [Reserve agents](#) on page 195.

Reserve agents: You can assign reserve agents using any of the following features:

- Service Level Maximizer (SLM)
- Avaya Business Advocate Service Level Supervisor (SLS)
- Avaya Business Advocate 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.

For more information about reserve agents, see *Avaya Business Advocate User Guide*.

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 [Reserve agents](#) on page 195.

Service Level Supervisor (SLS): See [Reserve agents](#) on page 195.

Look-Ahead Interflow

This section includes the following topics:

- [About LAI](#) on page 196
- [Administering LAI](#) on page 196
- [LAI considerations](#) on page 197

About LAI

With Look-Ahead Interflow (LAI), you can optionally route a call to a backup location based on your system's ability to handle the call within parameters defined in a vector. In turn, the backup system can accept or deny the call also based on parameters defined in a vector.

LAI requires end-to-end ISDN connections, and it works over private and public networks.

LAI can:

- Produce First in First Out (FIFO) or near FIFO call routing
- Provide globally-supported information forwarding over public or private ISDN (PRI and BRI) networks using non-QSIG or QSIG protocols. For more information, see [Information Forwarding](#) on page 178.

Administering LAI

The following forms and fields are required to administer the LAI feature.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● Vectoring (Basic) ● ISDN-PRI ● Lookahead Interflow
Trunk Group (ISDN)	<ul style="list-style-type: none"> ● Outgoing Display ● Codeset to Send TCM, Lookahead ● Supplementary Service Protocol ● UUI Treatment
Feature-Related System Parameters	Interflow-Qpos EWT Threshold

Screen	Field
ISDN Numbering - Public/ Unknown	<ul style="list-style-type: none"> ● Ext Len ● Ext Code ● CPN Prefix
Call Vector	Complete a screen for each Look-Ahead Interflow vector

See [Call Vectoring](#) on page 137 for associated Call Vectoring administration.

- System-Parameters Customer-Options - For full functionality, options must be enabled at both the sending and receiving communication servers. If Look-Ahead Interflow is not optioned on the receiving communication server, interflow still results on a look-ahead basis. However, the forwarded Dialed Number Identification Service (DNIS) (sending communication server VDN name) information is ignored and tandem Look-Ahead Interflow is not provided.
- Trunk Group Screen (ISDN) - If you do not want the call originator's display to update on each Look-Ahead Interflow call attempt, look-ahead calls should be routed over trunk groups with the `Outgoing Display` field set to n.
- Feature-Related System Parameters Screen - Administer the `Interflow-Qpos EWT Threshold` field when working with enhanced Look-Ahead Interflow. Any calls that will be answered before this threshold will not be interflowed (therefore saving CPU resources).
- ISDN Numbering - Public/Unknown Screen - Administer a CPN Prefix for each Vector Directory Number (VDN) that maps to a vector used to place Look-Ahead Interflow calls. If you do not, a Look-Ahead Interflow DNIS of all blanks displays on the answering agent's phone.

For private network non-QSIG connectivity with direct facilities between the communication servers, administer Look-Ahead Interflow DS1/E1 circuit packs with Country Protocol Option 1 independent of the country where the system is located.

LAI considerations

Carrier compatibility: LAI has been tested with several major carriers. To find out if these 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 involved with the preferred specific configuration, use of private ISDN trunking between the nodes should be assumed until successful testing has been completed.

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 Look-Ahead Interflow is enabled are treated as Look-Ahead Interflow call attempts.

ACD call center features

A vector may route a call over an ISDN facility to a destination that is not a VDN. The sending communication server processes this call as a Look-Ahead Interflow call even though it is not. ISDN processing at the receiving communication server causes the call to always be accepted. However, the DNIS and any other information in the Look-Ahead Interflow information forwarded with the call are ignored.

Interim call handling before LAI is accepted by receiving communication server: Until the look-ahead attempt is accepted by the receiving communication server, the caller continues to hear any feedback applied by the sending communication server vector and will remain in any split or skill queues.

Call handling with Route-to number or Route-to digits handling with coverage: Route-to number with coverage `y` or route-to digits with coverage `y` commands never result in a Look-Ahead Interflow call attempt. The sending end assumes the call is always going to be accepted. This command always completes the call. Moreover, the command should not be used if the vector at the receiving communication server might deny the call, since the caller in this case would be given a busy signal or would be disconnected. Use this command with coverage `y` only when you want unconditional interflow (with Look-Ahead Interflow active) and the terminating communication server is set up accordingly.

Continuity during call transfer between communication servers: Audible feedback may be provided to the caller before interflow is attempted. Therefore, another audible feedback from the receiving communication server may confuse the caller. For example, a caller hearing ringback on the sending communication server may be confused if music is applied suddenly when the call interflows to the receiving communication server.

Backward compatibility of LAI applications: For backward compatibility of LAI applications between Avaya communication servers, leave the **Send Codeset 6/7 LAI IE** option on the Trunk Group screen set to its default `y`. Existing LAI applications will 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 (the conditional applies only to calls in queue).

AAR/ARS: ISDN facilities used to provide Look-Ahead Interflow 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 (unless they're overridden with newly collected digits) on the telephone display.

Attendant Control of Trunk Group Access: Calls will not route over a trunk with Attendant Control of Trunk Group Access set.

Authorization Codes: Authorization Codes must not be 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 considered an invalid destination (rejected for Look-Ahead Interflow or not available for standard interflow) without the application of recall dial tone.

BCMS: BCMS does not log LAI attempts, nor does it report accumulated in-VDN time.

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 Look-Ahead Interflow attempts.

If a local (on-communication server) call to a VDN generates a Look-Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving communication server, then one Outgoing Call CDR record is generated with the originating extension as the calling number.

If an incoming (off-communication server) call to a VDN generates a Look-Ahead Interflow call attempt that is accepted, and no answer supervision is returned from the receiving communication server, then 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 (off-communication server) call to a VDN generates a Look-Ahead Interflow call attempt that is accepted, and answer supervision is returned from the receiving communication server, then 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 Look-Ahead Interflow call is treated like any other incoming vector call.

If answer supervision is returned by the vector, and the call is never terminated to another destination, 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. If the `Record VDN in Record` field of the Feature Related System Parameters is `y`, then the VDN extension overrides the `Call to Hunt Group - Record` administration option for vector calls.

Call Prompting: Digits collected at the sending communication server, no matter how they are collected (caller-entered, ASAI provided, CINFO provided, etc.) are forwarded with interflowed calls and available at the remote communication server using information forwarding. For more information, see [Information Forwarding](#) on page 178.

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 a Look-Ahead Interflow destination.

VDN Name Display: The VDN name (part of the LAI information forwarded with calls) can be up to 15 characters long. Any characters over this limit will be 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 not) may 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: The adjunct routing capabilities of vectoring can be used at the sending communication server to determine if a call should be interflowed. Adjunct routing at the receiving communication server can be used 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 (for example, SDN) ISDN-BRI and -PRI networks that meet minimum network requirements.

The sending of a Look-Ahead Interflow 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 Look-Ahead Interflow `route-to number` vector step, BSR `queue-to best` vector step, and the Adjunct Routing vector steps.

The ability to track a measured ACD call after a path replacement has taken place is available for CMS versions r3v9ai.o or later. Starting with the r3v12ba.x release, CMS reports a path replacement as a *rename* operation rather than a path replacement. The *rename* operation properly reports scenarios where a path replacement takes place from a measured to an unmeasured trunk facility. Avaya recommends that you upgrade CMS to r3v12a.x or later and administer all trunks associated with path replacement as *measured* by CMS to ensure better CMS tracking of path-replaced calls.

QSIG: LAI and information forwarding function over QSIG trunk facilities if the remote locations are Avaya communication servers. You may get LAI call control functionality with other vendors if an Avaya communication server is the starting point.

Redirect on No Answer (RONA): Calls redirected to a VDN by RONA can be subsequently 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.

Trunk-to-Trunk Transfer: Interflowed calls may 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.

Multiple Call Handling

Multiple Call Handling (MCH) allows agents to receive an ACD call while other types of calls are alerting, active, or on hold.

This section includes the following topics:

- [Administering MCH](#) on page 202
- [MCH applications](#) on page 202
- [MCH settings](#) on page 203
- [MCH considerations](#) on page 205
- [MCH interactions](#) on page 206

Administering MCH

The following forms and fields are required to administer the MCH feature.

Screen	Field
Hunt Group	<ul style="list-style-type: none">● ACD● Multiple Call Handling

The MCH column on the List Hunt Group screen contains the value that you enter in Multiple Call Handling.

MCH applications

Use Multiple Call Handling in applications where you want agents to take additional calls without dropping the active call. Examples of applications include:

- An agent and a caller may need to wait on a call for information. MCH allows the agent to put the call on hold and handle other ACD calls until information is available.
- ACD calls may be more important to your business than non-ACD calls. Use MCH to interrupt agents on non-ACD calls with an ACD call.
- In an EAS environment, calls from one skill may be more important than calls from another skill. Use MCH to interrupt an agent who has a call from the less-important skill with a call from the more-important skill.

You can use MCH in an Expert Agent Selection (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, he or she can handle multiple simultaneous ACD or direct agent calls only in the MCH skills.
- Without EAS, agents can be logged into only one split if it is an MCH split. Similarly, an agent logged in to a non-MCH split cannot log into an MCH split.

MCH settings

This section includes the following topics:

- [On request](#) on page 203
- [One forced](#) on page 203
- [One per skill](#) on page 204
- [Many forced](#) on page 204
- [MCH example](#) on page 205

On request

In on-request splits/skills, the following is true:

- If an agent goes into 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 their skill level in the EAD queue, or is made available in the DDC queue.
- Agents must select auto-in or manual-in work mode for each new ACD call they 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/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.

ACD call center features

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 could be forced to take an ACD call for one of his or her other skills, depending on that skill's MCH settings.

Because one-forced MCH forces an ACD call to alert an agent who is not on an ACD call, use it 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 this 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 that 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 could be forced an ACD call for one of his or her other skills if those skills are many-forced or one-per-skill MCH.

Use one-per-skill MCH when calls from one skill are 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 they must put equally important calls on hold. It can also be used 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).

Condition	Calls Delivered?			
	Skill 1 (MCH=one-req uest)	Skill 2 (MCH=one- forced)	Skill 3 (MCH=one- per-skill)	Skill 4 (MCH=many-for ced)
No calls on set	yes	yes	yes	yes
One active extn call	no	yes	yes	yes
Skill 1 call active	no	yes	yes	yes
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/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 auto-in or manual-in work mode. All forced MCH calls are delivered with ringing at the agent's station, not with zip tone. Requested MCH calls are delivered with ringing or zip tone.
- Agents can toggle between auto-in and manual-in work mode.

ACD call center features

- If an agent selects ACW or AUX work mode with calls on hold, the work mode is pending until all calls complete or until a manual-in call completes. New ACD calls are not delivered when AUX work is pending. When an ACD or direct agent 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 forced MCH calls.
- If an agent is either in 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 direct agent call abandons from hold, the agent is pending for ACW work mode and the after-call button lamp flashes.
- If an agent reconnects to an ACD or direct agent call on hold, his or her work mode changes to the call's work mode (auto-in or manual-in).
- Do not use forced MCH with DDC distribution because the first agent continues to receive calls until all line appearances are busy.

MCH interactions

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 and Stroke Counts: 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, he or she cannot request another call.

If agents with many-forced MCH are in a split or skill with forced entry of CWC or stroke counts, they are forced to handle an ACD call even if they have not entered a code/count.

Direct Agent Calling: 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 auto-in or manual-in work mode with calls on hold, he or she 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.

Queueing: 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), he or she 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.

Redirection on No Answer: 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's telephone, the agent does not receive additional ACD calls because the appearance is reserved for making conference or transfer calls.

Network Call Redirection (NCR)

Network Call Redirection (NCR) offers an optional call redirection method between sites on a public network or a Virtual Private Network (VPN) that can reduce trunking costs. These cost reductions are particularly valuable in enterprises or multi-site call centers environments where network costs are typically high.

When an incoming call arrives at an Avaya communication server that has the NCR feature enabled, call redirection is managed by the PSTN instead of the local server. As a result, ISDN or SIP trunks that the server would otherwise retain to accomplish the redirection are not required.

The cost reductions associated with reduced trunk use can be significant, particularly when other Avaya call vectoring features, such as Best Service Routing (BSR) and Expected Wait Time (EWT) are implemented.

For a detailed description of the NCR feature, see *Avaya Call Center Call Vectoring and EAS Guide*.

Queue Status Indications

This section includes the following topics:

- [About Queue Status Indications](#) on page 209
- [Administering Queue Status Indications](#) on page 209
- [Queue Status Indication detailed description](#) on page 210
- [Queue Status Indication interactions](#) on page 211

About Queue Status Indications

Queue Status Indications allows you to assign queue-status indicators for Automatic Call Distribution (ACD) calls based on the number of split or skill calls queued and time in queue. You can assign these indications to lamps on agent, supervisor, or attendant telephones or consoles to help users monitor queue activity.

In addition, you can define auxiliary queue warning lamps to track queue status. On telephones and consoles with displays, you can display the number of calls queued and time in queue of the oldest call in the split or skill.

Administering Queue Status Indications

The following forms and fields are required to administer the Queue Status Indications feature.

Screen	Field
Hunt Groups	<ul style="list-style-type: none"> ● Queue Limit ● Calls Warning Threshold (per split or skill or attendant group) ● Time Warning Threshold (per split or skill or attendant group) ● Time Warning Port (per split or skill) ● Calls Warning Port (per split or skill)
Station (multi-appearance)	Button/Feature Button Assignments: <ul style="list-style-type: none"> ● q-calls ● q-time

Screen	Field
Attendant Console	Feature Button Assignments: <ul style="list-style-type: none">● atd-qcalls● atd-qtime● q-calls● q-time
System Capacity	Queue Status Buttons

Queue Status Indication detailed description

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 telephone 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

You can use Queue Status Indications to provide status information for attendant groups or other hunt group types (DDC and UCD). With attendant groups, the button names (AQT and AQC) are different than 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.

If you need to know how many queue status buttons have been administered, or how many your system will allow you to administer, check page 5 of the System Capacity screen.

Queue Status Indication interactions

Attendant and Telephone Display Timers: The timer and the queue status information can be displayed at the same time. On 1-line displays, the timer is displayed in the last eight display positions and the number of queued calls is not displayed. On 2-line displays, 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.

Proactive Contact outbound calling improved reporting

This section includes the following topics:

- [About Proactive Contact outbound calling improved reporting](#) on page 212
- [Reason to use](#) on page 212
- [Prerequisites](#) on page 213
- [Switch-classified and non switch-classified calls](#) on page 213
- [The Proactive Contact outbound calling process](#) on page 213
- [Administering PC switch-classified calls for improved reporting](#) on page 216
- [Administering PC non switch-classified calls for improved reporting](#) on page 216
- [Proactive Contact improved reporting interactions](#) on page 217

About Proactive Contact outbound calling improved reporting

Avaya Call Center customers using a Proactive Contact integration can track and analyze the time agents in AUX Work mode spend on outbound calls using the same real-time and historical Call Management System (CMS) reports used by customers with other Outbound Call Management (OCM) applications. This feature:

- Delivers outbound Proactive Contact calls to agents that are in the AUX Work mode as though the calls were Automatic Call Distribution (ACD) calls associated with a skill specified for reporting.
- Includes these calls in Least Occupied Agent (LOA) occupancy calculations since they are treated as ACD calls by the Avaya Communication Manager software.
- Tracks switch-classified and non switch-classified outbound calls on CMS as ACD-OUT calls instead of as AUX-IN calls.
- Allows CCR to track these calls using the Proactive Contact event stream instead of the Management Information System (MIS) link SPI events.

Reason to use

For customers using Proactive Contact, this feature improves reporting capabilities and provides fair treatment of agents.

Prerequisites

You can use the improved integration with Proactive Contact outbound calling only if all of the following Avaya 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

This section includes the following topics:

- [Acquiring agents for outbound calling](#) on page 213
- [Launching switch-classified outbound calls](#) on page 214
- [Launching non switch-classified outbound calls](#) on page 215

Acquiring agents for outbound calling

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 acquires them for outbound calls as follows:

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 Auto-In or Manual-In mode.
3. Agents select an outbound campaign (application) using the Proactive Contact terminal.

ACD call center features

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 AUX 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.

For more information, see:

- [Launching switch-classified outbound calls](#) on page 214
- [Launching non switch-classified outbound calls](#) on page 215

Launching switch-classified outbound calls

Switch-classified outbound calls are launched using the following process:

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.

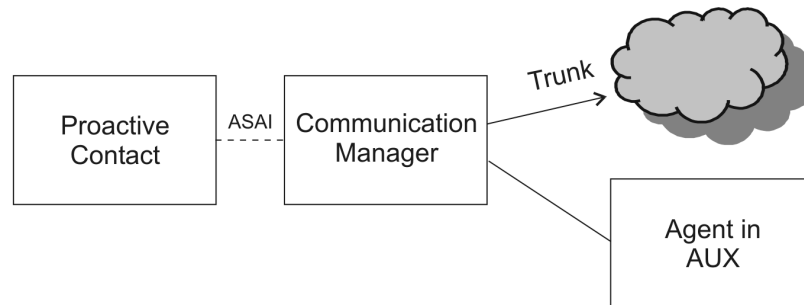
For more information about switch-classified calls, see *Avaya Call Center Automatic Call Distribution (ACD) Guide*.

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.

- CCR (if connected) receives a message to ignore these events.

Example

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. announcement 42015
4. wait-time 10 secs hearing silence
5. goto step 3 unconditionally
6. disconnect after announcement 42020
  
```

Launching non switch-classified outbound calls

Non switch-classified outbound calls are launched using the following process:

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 yes.
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](#) on page 216.

- CCR receives a message to ignore these events.

Administering PC switch-classified calls for improved reporting

To administer improved reporting for Proactive Contact switch-classified calls:

1. Enter `change vdn xxxxx`.
2. On the Vector Directory Number screen (page 2), set the options for the following fields:
 - **Reporting for PC Predictive Calls?**
 - **PC Predictive Reports Skill:** (appears after the **Reporting for PC Predictive Calls?** field is set to **y**.)

For more information about these fields, see [Vector Directory Number field descriptions](#) on page 362.

3. Press Enter.

Administering PC non switch-classified calls for improved reporting

To administer improved reporting for Proactive Contact non switch-classified calls:

1. Enter `system-parameters features`.
2. On the Feature-Related System Parameters screen (page 13), set the options for the following fields:
 - **Report for PC Non-Predictive Calls?**
 - **PC Non-Predictive Reports Skill:** (appears after the **Report for PC Non-Predictive Calls?** field is set to **y**.)

For more information about these fields, see [Call Center Miscellaneous field descriptions](#) on page 347.

3. Press Enter.

Proactive Contact improved reporting interactions

BCMS: You can use Basic Call Management System (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 Proactive Contact calls since Proactive Contact agents are logged in to auto-answer stations.

Forced Agent Logout by Clock Time: You should 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 will be logged out immediately.

Reason codes

This section includes the following topics:

- [About reason codes](#) on page 218
- [Administering reason codes](#) on page 218
- [Reason code detailed description](#) on page 219
- [Reason code considerations](#) on page 220
- [Reason code interactions](#) on page 221

About reason codes

Reason codes allow agents to enter a numeric code that describes their reason for entering Auxiliary (AUX) work mode or for logging out of the system. Reason codes give call center managers detailed information about how agents spend their time. Use this data to develop more precise staffing forecasting models or use it with schedule-adherence packages to ensure that agents are performing scheduled activities at the scheduled time.

You can administer the codes so that entry of the code is forced or optional. Beginning with 3.0, you can have up to one hundred AUX reason codes, including a default code (0).

You can use VuStats to display the reason code name or number. Use VuStats or CMS to gather historical and real-time reason-code statistics.

You must have Expert Agent Selection (EAS) enabled to use reason codes.

Administering reason codes

The following screens and fields are required to administer reason codes.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none">● ACD● EAS● Reason Codes
Attendant Console	RC
Agent LoginID	RC

Screen	Field
Feature-Related System Parameters	<ul style="list-style-type: none"> ● EAS ● Aux Work Reason Code Type ● Logout Reason Code Type ● CMS Release (Release 5 and later configurations, or none) ● Clock Time Forced Logout Reason Code
Feature Access Code (FAC) (optional)	<ul style="list-style-type: none"> ● Aux Work Reason Code Type ● Logout Reason Code Type
Language Translations	Line 49
Reason Code Names	All
Station	RC (for the Aux Work button)

Reason code detailed description

Forced reason codes: If you have administered forced reason codes, agents cannot enter AUX work mode or log out until they enter a code. Agents can enter codes 1 - 99, but not default code 0.

If an agent enters an invalid code or fails to enter a code within the 10-second timeout interval, the change is denied and the agent remains in the current work mode. If the agent dialed a FAC, the agent hears an intercept tone. If the agent pressed the AUX button, the AUX lamp flutters and then goes dark or lights steadily if the agent was already in AUX with a different reason code.

Requested reason codes: 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 timeout interval, the agent enters AUX work mode or logs out with default code 0.

Entering AUX work mode: 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
- Dialing an AUX work FAC and responding to the dial tone prompt for a reason code

If there are no calls ringing, active, or held at the telephone, agents enter AUX work mode immediately and the AUX lamp lights steadily. Otherwise, the AUX lamp blinks until the agent completes all calls at the telephone.

ACD call center features

If a button for AUX work is associated with the reason code that the agent entered, the button lamp lights. If no such AUX button exists, the system lights the first AUX button lamp with no administered reason code.

You can assign an AUX button without a reason code to an agent's phone. This allows agents with a limited number of buttons to use all 100 reason codes.

Logging out: To log out with a reason code, the agent dials the logout FAC, hears a second dial tone and enters a reason code. The agent hears confirmation tone and is logged out.

Default code: Default code 0 is used when the system puts an agent into AUX work mode or logs the agent out without the involvement of the agent. For example:

- When an agent logs in and is put into AUX mode
- When an agent makes or receives a non-ACD call from the available state
- When a call is redirected as a result of Redirection on No Answer (RONA) and the agent is logged out or put into AUX mode
- When 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 timeout period
- When an agent with requested reason codes enters # or *

Reason code considerations

- If an agent in auto-in or manual-in work mode dials the logout FAC but fails to enter a reason code and logout reason codes are forced, the agent is returned to the available state. ACD calls are delivered even if the agent has left the phone. To prevent this, be certain that agents enter AUX or ACW work mode before logging out.
- When an agent changes to AUX work mode and the AUX Work Reason Code Type is set to none, the agent is put into AUX work mode with the default reason code even if you have administered a different reason code for the AUX button. Setting AUX Work reason code in this way allows you to complete button administration before activating the feature.
- Do not administer AUX buttons without a reason code for hybrid station sets.
- When an agent in AUX work mode is active on a non-ACD call, the agent cannot immediately change the reason code. A change is pending until the call drops.
- There is a limit to the number of agents who can simultaneously be entering either a reason code or a Call Work Code.

Reason code interactions

Abbreviated Dialing: You can program FACs for 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 AUX work mode with a reason code, the agent enters AUX work for all of his or her skills with the same reason code.

ASAI: ASAI allows a host to log an agent out and place an agent in AUX work mode with a reason code. The host can query the agent's current work mode and receive the reason code associated with the AUX work mode.

Auto-Available Split/Skill : The system logs AAS agents out for Redirection on No Answer with the default reason code.

Basic Call Management System: Statistics about AUX work mode by reason code are not available in BCMS reports.

CMS: CMS tracks time in AUX work mode by reason code and displays 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 AUX work mode when moved, the agent is returned to AUX work mode with the same reason code when the move is completed.

Direct Agent Calling: When a direct agent call 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 (ring-ping or call-waiting tone). If there is an AUX button with no assigned reason code administered, then that lamp flutters.

Redirection on No Answer: When a call is redirected using RONA, an agent is placed into AUX work mode with the default code or is logged out with the default code if the agent is in an auto-available skill.

Redirection on IP Failure and Redirection on OPTIM Failure: The ROIF and ROOF features use the same reason code.

Redirection on No Answer

This section includes the following topics:

- [About RONA](#) on page 222
- [Administering RONA](#) on page 223
- [RONA detailed description](#) on page 223
- [RONA application examples](#) on page 224
- [RONA considerations](#) on page 227
- [RONA interactions](#) on page 228
- [Interactions with other ringing call timers](#) on page 233

About RONA

Redirection on No Answer (RONA) redirects a ringing ACD split or skill call or direct agent call after an administered number of rings. RONA prevents an unanswered call from ringing indefinitely. The call can redirect either to the split or skill to be answered by another agent or to a VDN for alternative call handling. Direct agent calls route to the agent's coverage path, or to a VDN if no coverage path is administered.

You must have ACD enabled to use RONA. Administer RONA for each ACD hunt group as required. RONA can be used in Auto-Available Splits/Skills (AAS), or in splits/skills with agents operating in Manual-In work mode. You can administer RONA for vector-controlled or non vector-controlled splits/skills.

Do not administer RONA for splits/skills controlled by adjuncts or AUDIX or for auto-answer agents assigned splits/skills because calls must ring at a telephone to be redirected.

Administering RONA

The following forms and fields are required to administer the RONA feature.

Screen	Field
Hunt Group	<ul style="list-style-type: none"> ● ACD ● AAS ● Vector ● Controlling Adjunct ● Message Center ● Redirect On No Answer (Rings) ● Redirect to VDN
Station (multifunction)	Button Assignments: <ul style="list-style-type: none"> ● noans-ahrt

Note:
You must set Controlling Adjunct to none.

RONA detailed description

When RONA is invoked for a call, the system:

- Places an agent in AUX work mode, and thus unavailable to receive calls from other splits/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 queued at the highest priority so that they are distributed before any other split or skill calls. See [RONA routing sequences](#) on page 225 for more information about call redirection.
- Sends a message to CMS. When a RONA timeout occurs, the Noans-ahrt lamp for the split or skill lights steadily. The supervisor presses the Noans-ahrt button to display the login ID or the extension and name of the last agent timed out with RONA.
- Records the redirection in BCMS or CMS. See [Using BCMS/CMS reports with RONA](#) on page 226 for additional information.

RONA application examples

This section includes the following topics:

- [VRU applications](#) on page 224
- [Other applications](#) on page 225
- [RONA routing sequences](#) on page 225
- [Using BCMS/CMS reports with RONA](#) on page 226
- [Returning AAS agents to service](#) on page 227

VRU applications

Typically, RONA is used with 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 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 a VRU port, the agent is logged out and the call is redirected back to the converse split or skill.

Note:

RONA can not redirect a call to an administered VDN from a **converse** step.

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 timeout, the **converse-on** command step (Step 2) is skipped, and no digits are collected by Step 3 (after the 10-second timeout). 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 may not be answered because an agent left without entering 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 direct agent calls

Redirection Destination	Explanation
Coverage path	Direct agent calls redirect to a coverage path, if one exists. Priority calls do not route to coverage.
RONA VDN	If no coverage path exists but a VDN is administered for RONA, direct agent calls redirect to the VDN.
VDN return destination	For external calls, if neither a coverage path nor a RONA VDN are administered, then IC Email calls 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.
Requeue to split or skill	If a RONA VDN is not administered, calls redirect to the split or skill.

RONA Routing Sequence for Split Or Skill Calls

Redirection Destination	Explanation
Coverage path	In non vector-controlled splits, if calls cannot requeue to the split, they redirect to the split's 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 can not requeue, they 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 timeouts and how calls were redirected.

With R3V2 and later releases of CMS, the exception report lists agents who were timed out and made unavailable. With BCMS and earlier releases of CMS, you can determine which agents were in 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 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 = yes.

With BCMS, agents' changes to AUX work mode appears in the BCMS Split (AGENT) Status report. In an AAS split, agents log out, so they 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.

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

- RONA can timeout while an agent is actually at the station if the agent does not answer soon enough or has selected another work mode while a call is ringing. RONA handles the call as usual, making the agent unavailable. With ACD splits, agents at multifunction telephones know that they have been made unavailable when they see the aux-work lamp lit. They 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-ahrt button can be assigned to non-SIP agent or supervisor phones to indicate that a call has been redirected. When the noans-ahrt button is pressed, the phone display shows the login ID or extension and name of the last agent timed out with RONA.

RONA interactions

AAS: Use AAS with RONA for 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 considered 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 may be left at this split or skill. When the system reinitializes, all busied-out ports are automatically logged back into the AAS splits. New calls cause a RONA timeout 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 timeout occurs, an agent is placed in AUX work mode with notification to CMS for all splits that the agent is logged into. The agent is responsible for becoming available in each split. In an AAS, agents are logged out of all splits that they 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 timeout occurs on a ringing ACD call, RONA timing continues. At timeout, the call is redirected, CMS is notified, and the agent is placed in AUX work (overriding the pending ACW request).

If an agent presses the aux-work button with an ACD call ringing, the change request is pending. With ACD splits/skills, since the RONA time-out changes the state to aux-work, there is no conflict with the pending aux-work change request. With AAS splits/skills, an agent-initiated aux-work change is denied per existing operation.

ASAI: RONA applies to vector-processed calls that are routed by an adjunct to a split or 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 timeout puts an agent into AUX work mode.

The adjunct makes an agent query (as part of the value query capability group) to determine the agent's 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 timeout 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 IVR VRU applications are configured with non vector-controlled splits/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/skills are not used for this application because the ASAI link controls the login or logout).

You can assign RONA to these splits/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 (assuming that they are active) 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.

RONA/ASAI events

Stage of Call	ASAI Event	ASAI Associations
1. RONA timeout	Logout (for AAS)	Domain (agent) control
2. Call redirected to split	Call redirected	Domain (station) control (for agent ext call is leaving)
	Call queued (only if the call queues)	Domain (station) control, (for new agent & for internal originator) call control, notification
3. Call redirected to VDN processing	Call redirected	Domain (station) control (for agent ext that call is leaving)
	Call redirected (only if call is redirected to a VDN with Notification active)	Notification
	Call offered to Domain	Notification (VDN)
4. Call delivered to agent	Alerting	Domain (station) control, (for new agent & for internal originator) call control, notification
5. Call routed to split's coverage path	Call redirected followed by existing operation of ASAI Events	Domain (station) control (for agent ext that call is leaving)
6. Infinite feedback to caller	Call redirected	Domain (station) control (for agent ext that call is leaving)

RONA/ASAI events

Stage of Call	ASAI Event	ASAI Associations
7. Continue vector processing	Call redirected followed by existing ASAI events	Domain (station) control (for agent ext that call is leaving redirecting to VDN)
8. Call routed to direct agent's coverage path	Call redirected followed by existing operation of ASAI events	Domain (station) control (for agent ext that call is leaving)

Note:

When a call is redirected using ASAI Redirect Call, the RONA timer is canceled.

Attendant return call: 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.

AUDIX Transfer: 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.

Automatic answering: If an agent with automatic answering receives a call with zip tone instead of ringing, RONA timing is canceled.

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-redirectioned coverage call (for example, the RONA split was the second point in the coverage path), the call is redirectioned 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-redirectioned handling.

Call Detail Recording (CDR): When an agent is assigned to be recorded on the CDR record as the called number, the RONA redirectioned-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 redirectioned 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.

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-redirectioned calls involving vector-controlled splits.

Calling/Called Number Display: A call to a split or skill that RONA redirections 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.

ACD call center features

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 AUX work mode or logged out even if they are the last agent 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.

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.

ACD call center features

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 may later be reset. RONA interactions with other timers are summarized in the following table.

Summary of RONA-Timer interactions

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 DA - Subsequent Redirection No Answer	Stopped	If redirects to covering point
Agent DA	Agent DA Coverage (Direct Agent Calls)	Stopped	If covers to direct agent with coverage
NATO	DID/CO Trk No Answer Timeout	Continues	N/A
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 timeout to shorter than other timeout 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 10 or more rings.

Since queue time is variable, assign a coverage timeout period that is greater than the longest expected queue time plus three or four rings (the time the call could ring the agent).

The NATO timer starts when the call seizes the incoming trunk. The timer could thus be timing before the call is queued by vector processing. Therefore, set the NATO timer to greater than the longest expected time before the call rings the agent (including time before and after being queued) plus 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 IP Failure

This section includes the following topics:

- [ROIF description](#) on page 235
- [How ROIF works](#) on page 235
- [Administering ROIF](#) on page 236
- [ROIF considerations](#) on page 236
- [ROIF interactions](#) on page 237

ROIF description

The Redirection on IP Failure (ROIF) feature applies only to agents using ACD auto-answer and IP hard- or softphones.

For releases prior to 2.1, calls could sometimes be lost when delivered by Avaya Communication Manager to auto-answer ACD agents equipped with IP phones. ROIF provides redirection of calls back into queue or to the specified VDN when calls to auto-answer ACD stations cannot be connected due to loss of IP connectivity.

How ROIF works

ROIF works as follows:

- When the system option (Switch Hook Query Response Timeout) is active, ROIF checks IP connectivity before delivering a call to the auto-answer agent using an IP phone.
- Invokes RONA if IP connectivity is not acknowledged. The agent is taken out of service and the call is put back in queue or forwarded to a RONA-specified VDN.
- Prevents a lost call during the period when IP connectivity failure has not been detected by Communication Manager maintenance.
- Puts the non Auto-Available Split/Skill (AAS) agent into Aux Work mode, then redirects the call to the split or skill queue or RONA-specified VDN if an IP connectivity failure is detected while that call is being delivered. If the Reason Codes feature is active, the change to Aux Work is reported with the ROIF reason code.
- Logs out the AAS agent instead of putting the agent into AUX Work.

Administering ROIF

This section includes the following topics:

- [Required forms for ROIF](#) on page 236
- [About setting the switch hook query timeout value](#) on page 236

Required forms for ROIF

The following forms and fields are required to administer ROIF.

Screen	Field
System Parameters Features (page 12 for Release 2.1)	Switch Hook Query Response Timeout
	Auto-answer IP Failure Aux Reason Code

About setting the switch hook query timeout value

When you set the switch hook query 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 the communication server 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.



CAUTION:

If the switch hook query timeout value is set too low, agents will be erroneously taken out of service when they are available.

ROIF considerations

This section includes the following topics:

- [About setting reason codes for ROIF](#) on page 237
- [Auto-in or manual-in button](#) on page 237
- [Manual answer agents](#) on page 237
- [Call redirection alert button](#) on page 237

About setting reason codes for ROIF

Set the reason code to a non zero number not currently being used in order for the system to distinguish between Aux Work changes that have been caused by miscellaneous changes and those caused by a loss of IP connectivity. ROOF uses the same reason code as ROIF.

For more information, see [Reason codes](#) on page 218.

Auto-in or manual-in button

If an agent has not received a call within the usual timeframe, the agent should 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.

Manual answer agents

The manual answer station is always on-hook until answered and the call is delivered by ringing the station. With IP, the system detects that the phone is not ringing and invokes existing IP failure mechanisms so that the call is not lost and delivered to the next available agent.

Call redirection alert button

A noans-ahrt 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 with RONA. ROIF first attempts to redirect DAC to the agent's coverage path. If the call cannot go to coverage, the call is redirected to the *Redirect to VDN* if it is assigned to the direct agent skill group. If *Redirect 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 will hear a busy signal. If all fails, the caller receives ringback until the system receives a caller disconnect. This also applies to priority direct agent calls in which *Redirect to VDN* is not specified.
- The agent will not be aware that the line is in Aux Work during an IP connectivity failure. If connectivity is restored during the TCP retry period, the lamp will indicate that the line is in the Aux Work mode.

ACD call center features

- The only indication that CMS receives after an 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 not specifically identified, other than the reason code.
- As with RONA, calls that are redirected due to ROIF are given uninterrupted ringback when re-queued. Calls that are 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 abandons the call. In this case, a DFWD-unknown message is sent to CMS to decrease the tracking of call ringing.
- ROIF does not apply to manual ringing agents. Therefore, there are no interactions with ringing call timers for ROIF.
- ROIF does not provide a lamp indication to the call center supervisor as is done for RONA.
- ROIF applies to AAS agents/VRU/IVR ports if they are connected through IP, and auto-answer is active. AAS lines are logged out if an IP failure is detected during call delivery.

For more information, see [RONA interactions](#) on page 228.

Redirection on OPTIM Failure

This section includes the following topics:

- [ROOF description](#) on page 239
- [How ROOF works](#) on page 239
- [Administering ROOF](#) on page 239
- [ROOF considerations](#) on page 240
- [ROOF interactions](#) on page 241

ROOF description

The Redirection on OPTIM Failure (ROOF) feature applies only to ACD agents using SIP hard- or softphones administered as Off-PBX Telephone Integration and Mobility (OPTIM) endpoints.

As with ROIF, ROOF provides redirection of calls back into queue or to the specified VDN when calls to ACD stations cannot be connected due to loss of IP connectivity.

The ROOF feature is different from the ROIF feature in the following ways:

- ROOF applies to agents using auto answer or manual answer. ROIF applies only to agents using auto answer.
- ROOF applies to SIP IP endpoints, for example, the 16CC SIP telephone. ROIF applies to H.323 IP endpoints, for example, the 4622SW IP telephone.

How ROOF works

ROOF works as follows:

- Before delivering a call to an OPTIM endpoint, the system checks for IP connectivity.
- Invokes RONA if IP connectivity is not acknowledged. The agent is placed in AUX work mode and the call is put back in queue or forwarded to a RONA-specified VDN.
- Prevents a lost call during the period when IP connectivity failure has not been detected by Communication Manager maintenance.

Administering ROOF

This section includes the following topics:

ACD call center features

- [Required forms for ROOF](#) on page 240
- [About setting the Redirect On OPTIM Failure value](#) on page 240

Required forms for ROOF

The following forms and fields are required to administer ROOF.

Screen	Field
System Parameters Features	Auto-answer IP Failure Aux Reason Code
Trunk Group	Redirect On OPTIM Failure

About setting the Redirect On OPTIM Failure value

The range for the Redirect On OPTIM Failure field is 250 - 32000 ms. The default and recommended value for this field is 5000. If agents are erroneously being placed in Aux work mode when they are available, the Redirect On OPTIM Failure value is set too low and should be increased.

ROOF considerations

This section includes the following topics:

- [About setting reason codes for ROOF](#) on page 240
- [Auto-in or manual-in button](#) on page 241
- [Call redirection alert button](#) on page 241

About setting reason codes for ROOF

Set the reason code to a non zero number not currently being used in order for the system to distinguish between Aux Work changes that have been caused by miscellaneous changes and those caused by a loss of IP connectivity. ROOF uses the same reason code as ROIF.

For more information, see [Reason codes](#) on page 218.

Auto-in or manual-in button

If an agent has not received a call within the usual time frame, it is possible that a network disruption has occurred, causing the communication server to change the agent's state to Aux Work. The agent should press the auto-in or manual-in button to ensure IP connectivity. The lamp update for the Aux Work button may not be received if the SIP station has lost IP connectivity.

Call redirection alert button

A noans-ahrt button can be assigned to non-SIP agent or supervisor phones to indicate that a call has been redirected.

ROOF interactions

The ROOF call and agent interactions are the same as for RONA, with the following additions:

- ROOF is applied system-wide. The default for the system is *active*.
- ACD calls delivered from the split or skill queue and Direct Agent Calling (DAC) work the same as with RONA. ROOF first attempts to redirect DAC to the agent's coverage path. If the call cannot go to coverage, the call is redirected to the *Redirect to VDN* if it is assigned to the direct agent skill group. If *Redirect 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 will hear a busy signal. If all fails, the caller receives ringback until the system receives a caller disconnect. This also applies to priority direct agent calls in which *Redirect to VDN* is not specified.
- The agent will not be aware that the line is in Aux Work during an IP connectivity failure. If connectivity is restored during the TCP retry period, the lamp will indicate that the line is in the Aux Work mode.
- The only indication that CMS receives after a ROOF 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 not specifically identified, other than the reason code.
- As with RONA, calls that are redirected due to ROOF are given uninterrupted ringback when re-queued. Calls that are 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 abandons the call. In this case, a DFWD-unknown message is sent to CMS to decrease the tracking of call ringing.

For more information, see [RONA interactions](#) on page 228.

Remote Logout of Agent

This section includes the following topics:

- [Description of Remote Logout of Agent](#) on page 242
- [Reason to use](#) on page 242
- [Prerequisites](#) on page 243
- [Administering Remote Logout of Agent](#) on page 243
- [Locally logging out an agent](#) on page 243
- [Administering Remote Logout of Agent using a VDN](#) on page 244
- [Remotely logging out an agent using the assigned VDN](#) on page 245
- [Remote Logout of Agent failures](#) on page 245
- [Remote Logout of Agent interactions](#) on page 245

Description of Remote Logout of Agent

The Remote Logout of Agent feature allows a user to logout an idle ACD or EAS agent without being physically present at the agent's station. The user who is logging out the agent can be locally or remotely located.

The Remote Logout of Agent is similar to the Add/Remove Skills feature.

Reason to use

If an agent walks away from his or her station without logging out, ACD calls are sent to the station without being serviced. Without the Remote Logout of Agent feature, supervisors or other agents had to walk over to the agent's station, enter the logout FAC to change the agent's work mode or log out. Customers could also *busy-out* the station from the switch room or use the RONA feature to put the station in aux work mode without logging out the agent.

The Remote Logout of Agent feature makes it simpler to logout the agent from the user's station.

Prerequisites

The Remote Logout of Agent feature can only be used if user permissions are administered appropriately for the person or VDN attempting to use the feature. The communication server administrator must ensure that the appropriate users have permissions administered so that they can use this feature.

The feature user must:

- Be in the same Tenant Partition as the agent as set on the Tenant screen
- Have Remote Logout COR permissions set on the Class of Restriction screen
- Have console permissions set on the COS screen for local users
- Have a Call Center release set to 9.1 or later set on the System Parameters Customer-Options screen by the RFA license file

For a description of how to check these values, see *Administrator Guide for Avaya Communication Manager*.

Administering Remote Logout of Agent

The following forms and fields are required to administer the Remote Logout of Agent feature.

Screen	Field
Class of Service (COS)	Console Permissions - set to y
Class of Restriction (COR)	Remote Logout of Agent - set to y
Feature Access Code (FAC)	Remote Logout of Agent Access Code - set to the desired dial code

Locally logging out an agent

To log out an agent from a local site:

1. Use a local station assigned with the COS and COR to logout an agent locally within the communication server.
2. Enter the FAC that was established to activate this feature followed by the agent's login ID or physical station extension.
 - You must be in the same Tenant Partition as the agent.
 - The physical extension is used only in non-EAS systems.

Administering Remote Logout of Agent using a VDN

Many call centers are geologically dispersed, but the reporting and tracking of agent activity takes place from one main location. Or, agents can log into 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's 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 may 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 might not hear confirmation tone (it varies with trunk type and trunk administration) and the user's phone will continue 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

Note:

This procedure assumes you have used the example in [Administering Remote Logout of Agent using a VDN](#) on page 244.

To log out an agent from a remote location using the assigned VDN:

1. Dial into 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 [Administering Remote Logout of Agent using a VDN](#) on page 244.

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 Agent A's login ID or name.

Remote Logout of Agent failures

In addition to failures that occur due to permission requirements, see [Prerequisites](#) on page 243. The logout fails if a Remote Logout of Agent 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

Auto-Available Split/Skill : If an agent login ID is assigned to an Auto-Available split or skill, 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.

ACD call center features

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 businesses and organizations that use Avaya Communication Manager servers to process large volumes of telephone calls using the Automatic Call Distribution (ACD) feature and (optionally) the Avaya Proactive Contact outbound dialer. Reporting adjuncts include Avaya Call Management System (CMS) and CCR.

Reporting adjuncts support solutions for:

- Routing and agent selection tracking
- Multi-site contact centers
- Remote agents
- Reporting
- Interfaces to other systems
- Workforce management
- Desktop applications
- System recovery
- Quality monitoring

This section provides information about administering the reporting adjuncts.

This section includes the following topics:

- [Considerations for interfacing with the reporting adjuncts](#) on page 247
- [Administering reporting adjuncts on CM](#) on page 250
- [Adding the reporting adjunct nodes](#) on page 250
- [Administering Avaya IQ](#) on page 251
- [Administering CMS](#) on page 252
- [Administering both reporting adjuncts at the same time with CM 4.0 or later](#) on page 254

Considerations for interfacing with the reporting adjuncts

You must consider the following factors when interfacing between the Communication Manager server and the reporting adjuncts:

- [IP trunk groups and SIP trunk groups](#) on page 248
- [Reporting adjunct representation of IP or SIP trunk member port-IDs](#) on page 248
- [Display of physical \(non-IP\) member port-IDs](#) on page 249

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.

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.

Reporting adjunct representation of IP or SIP trunk member port-IDs

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.

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	000000001 through 000000499
T00500 through T00999	000001000 through 000001499
T01000 through T01499	000002000 through 000002499
T01500 through T01999	000003000 through 000003499
T02000 through T02499	000004000 through 000004499
T02500 through T02999	000005000 through 000005499
T03000 through T03499	000006000 through 000006499
T03500 through T03999	000007000 through 000007499
T04000 through T04499	000008000 through 000008499
T04500 through T04999	000009000 through 000009499
T05000 through T05499	000010000 through 000010499
T05500 through T05999	000011000 through 000011499
T06000 through T06499	000012000 through 000012499
T06500 through T06999	000013000 through 000013499

Communication server representation	Reporting adjunct representation
T07000 through T07499	000014000 through 000014499
T07500 through T07999	000015000 through 000015499
T08000 through T08499	000016000 through 000016499
T08500 through T08999	000017000 through 000017499
T09000 through T09499	000018000 through 000018499
T09500 through T09999	000019000 through 000019499
T10000 through T10499	000100000 through 000100499
T10500 through T10999	000110000 through 000110499
T11000 through T11499	000120000 through 000120499
T11500 through T11999	000130000 through 000130499
T12000	000140000

Display of physical (non-IP) member port-IDs

Port IDs for physical circuit-switched trunks that terminate on an H.248 Media Gateway such as the G700, 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 that port IDs are listed in the reporting adjunct.

The following table compares the standard trunk equipment 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	gggVssc
where: bb = cabinet (1-64) X = carrier (A-E) ss = slot (1-25) ccc = circuit (1-256)	where: ggg = gateway (replaces cabinet) (1-250) V = indicates H.248 media gateway (replaces carrier) s = slot (1-4) cc = circuit (1-32)

ACD call center features

In reports, gateway-terminated trunk equipment locations are displayed in a slightly different format than that used for Avaya communication servers.

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.

Carrier number: The carrier number shows as the letter V followed by two digits (01-08) 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.

H.248 Media Gateway port ID representations on communication servers and reporting adjuncts

Avaya communication server	On CMS, CMS Supervisor, and CCR
Example 1: gateway=12, slot number=2, circuit number=16	
012V216	012V02016
Example 2: gateway=130, slot number=2, circuit number=16	
130V216	130V02016
Example 3: gateway=240, slot number=2, circuit number=16	
240V216	240V02016

Administering reporting adjuncts on CM

If you are using Call Center Release 4.0 or later, there are two fields dedicated to each of the reporting adjuncts, **CMS** and **CCR**. Each field can be set to just **CMS** or just **CCR**, or both fields can be set to **CMS** or **CCR**. Configurations with both reporting adjuncts are supported for an R14 CMS and an R4.0 CCR or an R13.1 CMS and an R4.0 CCR. High availability configurations are also supported.

Adding the reporting adjunct nodes

To add the name and IP address of each reporting adjunct node:

1. Enter **change node-names ip**

2. Enter the host name and IP address of each reporting adjunct. For CCR, enter the host name and IP address for the Data Collection Host that will be monitoring this communication server.

Administering Avaya IQ

This section describes administration tasks specifically for CCR. This section includes the following topics:

- [Communication Manager server-to-Avaya IQ interface](#) on page 251
- [Avaya IQ measurements](#) on page 252
- [Measured extensions and multiple splits](#) on page 252
- [Measured trunks versus unmeasured facilities](#) on page 252

Communication Manager server-to-Avaya IQ interface

When using CCR as a reporting adjunct, Communication Manager must be running Expert Agent Selection (EAS). That is, EAS must be active and enabled on the Feature-Related System Parameters screen.

Also, you must administer the communication servers to create and send UCIDs.

To administer the communication server:

1. Enter **change system-parameters features**
2. On page 5 of the Feature-Related System Parameters screen, set **Create Universal Call ID (UCID)?** to **y**.
3. On page 5 of the Feature-Related System Parameters screen, set the **UCID Network Node ID** to a 5-digit value from 1 to 32,767 that is unique among your communication servers.
4. On page 12 of the Feature-Related System Parameters screen, set **Send UCID to ASAI?** to **y**.

Note:

Detailed information on administering the interface between the communication server and CCR is available in the *Avaya IQ Administration*.

Avaya IQ measurements

The following forms and fields are required to enable CCR measurements:

Screen	Field
Hunt Group	Measured
Trunk Group (All)	Measured
Vector Directory Number (VDN)	Measured

Measured extensions and multiple splits

CCR can be used only when Expert Agent Selection (EAS) is administered. A non-EAS environment is not supported.

To enable EAS:

1. Enter change system-parameters features.
2. On page 11 of the Feature-Related System Parameters screen, set **Expert Agent Selection (EAS) Enabled?** to **y**.

Measured trunks versus unmeasured facilities

Unmeasured facilities are not actual trunks, but are tracking records on the CMS. See [Administering CMS](#) on page 252.

CCR does not require data allocation for these unmeasured facilities. Rather, the number of measured trunks is based on the capacity of the communication server that CCR is monitoring. CCR does not require data storage allocation for unmeasured facilities.

Administering CMS

This section describes administration tasks specifically for CCR. This section includes the following topics:

- [Communication server-to-CMS interface](#) on page 253
- [CMS measurements](#) on page 253
- [Measured extensions and multiple splits](#) on page 253
- [Measured trunks versus unmeasured facilities](#) on page 253
- [Allocation of measured trunks and unmeasured facilities](#) on page 253

Communication server-to-CMS interface

For detailed information on administering the interface between the communication server and the CMS, refer to *Avaya CMS Switch Connections and Administration*.

CMS measurements

The following forms and fields are required to enable CMS measurements:

Screen	Field
Hunt Group	Measured
Trunk Group (All)	Measured
Vector Directory Number (VDN)	Measured

Measured extensions and multiple splits

On a non EAS configuration, set the Installation, Authorization, and Data Storage Allocation to the number of CMS agents assigned on the Hunt Group screen.

Measured trunks versus unmeasured facilities

CMS requires data allocation for unmeasured facilities trunk data. Unmeasured facilities are not actual trunks, but are tracking records on the CMS.

Unmeasured facilities on CMS are required for:

- Internal calls (intra-communication server) to a measured split or agent
- Internal calls to VDNs
- Calls made by agents to internal destinations or on an unmeasured facility group
- Transfers and conferences until the transfer/conference is complete.

Allocation of measured trunks and unmeasured facilities

Maximum values for measured trunks and unmeasured trunk facilities are specified for each ACD in the CMS **Data Storage Allocation** window. Prior to CMS R14, the unmeasured trunks were counted along with the measured trunks toward the system and ACD allocated trunk limit of 40,000 across all ACDs. Also, the recommended assignment of unmeasured facilities per ACD was 25% of the total number of trunks allocated for the ACD.

ACD call center features

CMS R14 treats the measured and unmeasured trunks separately, meaning that the unmeasured trunks are not subtracted from the maximum measured trunks. Data Storage Allocation Help suggests that unmeasured trunks be set at 50% of the measured trunks allocated. The unmeasured trunks system limit is now 20,000 (50% of the measured trunks system limit of 40,000) and the unmeasured limit for an ACD is 6,000 (50% of the measured trunk ACD limit).

Administering both reporting adjuncts at the same time with CM 4.0 or later

To administer both reporting adjuncts on Communication Manager at the same time:

1. Enter **change system-parameters features**
2. On page 12 of the Feature-Related System Parameters screen, enter the appropriate value for the fields under **Reporting Adjunct Release**.

If the CMS (appl mis) field is set to:	You can set the CCR (appl ccr) field to:
R12	Not supported - must be blank
R13	Not supported - must be blank
R13.1	4.0 (for SPI L20)
R14	4.0 (for SPI L21)
blank (This is the only allowed entry if the dial plan is set to more than 7-digits for any entity.)	4.0 (for SPI L21)

Default: Both fields are blank.

3. Enter **change communication-interface processor-channels**
4. In the next available **Proc Chan** row, set **Enable** to **y**.
5. Set **Appl.** to **mis** (for the CMS system).
6. Set **Mode** to **s**.
7. Set **Link** to the data link that is administered to communicate with the adjunct.
8. Set **Chan** to an unused value in the range allotted to the link type being used for this channel.
9. Select the **Node** name from the drop down list.
10. Set **Port** to **0**.
11. Set both **Local/Remote** to **1**.

12. To administer the CCR system, repeat Steps 1 through 11, except in Step 5 set **Appl.** to **ccr**.

Service Level Maximizer

Service Level Maximizer (SLM) is an optional Avaya Communication Manager Call Vectoring feature that is used with Expert Agent Selection (EAS), and without Avaya Business Advocate.

SLM 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.

For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

MAO

When SLM is used, an optional feature called Maximum Agent Occupancy (MAO) can be used to set thresholds on the amount of time an agent spends on a call. MAO is used to avoid agent burnout.

For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

Auto-reserve agents

Auto-reserve is an added feature that can be used to ensure that the service level is met in critical skills. When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

Service Observing

This section includes the following topics:

- [About Service Observing](#) on page 257
- [Administering Service Observing](#) on page 258
- [Service Observing detailed description](#) on page 259
- [Service Observing indicators](#) on page 262
- [Service Observing considerations](#) on page 268
- [Service Observing interactions](#) on page 270

About Service Observing

Service Observing allows a specified user, such as a supervisor, to observe or monitor another user's calls. In this section, *observer* refers to the supervisor who is observing calls. *Agent* refers to the extension, attendant, or logical agent being observed. A vector directory number (VDN) call can also be observed. Observers can observe in listen-only or listen-and-talk mode.

Note that you set up Service Observing to observe a particular extension, not all calls to all extensions at a station.

Service Observing may be subject to federal, state, or local laws, rules, or regulations or require the consent of one or both of the call parties. Familiarize yourself and comply with all applicable laws, rules, and regulations before using this feature.

Administering Service Observing

The following forms and fields are required to administer the Service Observing feature.

Screen	Field
System Parameters Customer-Options	<ul style="list-style-type: none"> ● Service Observing (Basic) - for basic or Logical Agent ID observing ● Service Observing (Basic) and Service Observing (Remote/By FAC) - for remote observing or observing using feature access code ● Service Observing (Basic) and the Service Observing (VDNs) - for VDN observing ● Vectoring (Prompting) - for vector-initiated observing
Class of Restriction	<ul style="list-style-type: none"> ● Can Be Service Observed ● Can Be Service Observer ● Service Observing COR Table
Feature-Related System Parameters	<ul style="list-style-type: none"> ● Service Observing Warning Tone ● Expert Agent Selection (Logical Agent ID) ● Service Observing Allowed with Exclusion
Station (multiappearance)	Button/Feature Button Assignment (basic, VDN, Logical Agent ID): <ul style="list-style-type: none"> ● serv-obsrv
Feature Access Code (FAC)	<ul style="list-style-type: none"> ● Service Observing Listen Only Access Code (remote by FAC, VDN, Logical Agent ID) ● Service Observing Listen/Talk Access Code (remote by FAC, VDN, Logical Agent ID) ● Service Observing No Talk Access Code (remote by FAC, VDN, Logical Agent ID)
Vector Directory Number (VDN)	Observe on Agent Answer
Class of Restriction forms	
Agent COR screen	Can Be Service Observed field set to yes (if extension is to be observed)
Observer COR screen	<ul style="list-style-type: none"> ● Can Be Service Observer field set to yes (if extension is to act as observer) ● In Service Observing COR table, enter y next to the CORs to be observed

Service Observing detailed description

This section includes the following topics:

- [About using Service Observing](#) on page 259
- [Service Observing with Exclusion](#) on page 260
- [No-talk FAC for Service Observing](#) on page 260
- [Observing Logical-Agent IDs](#) on page 260
- [Observing VDNs](#) on page 261
- [Observing Remotely or by FAC](#) on page 261

About using Service Observing

To begin observing, the observer presses the Service Observing button plus the agent's extension number. Initially, the observer is in listen-only mode. The observer presses the Service Observing button to toggle between listen-only and listen/talk mode. The lamp indicates which mode the observer is in.

To deactivate Service Observing, the observer hangs up, selects another call appearance, or presses the Release button.

An observer can observe an agent who is not active on a call. The observer is in the wait state until the agent receives a call, then the observer is bridged onto the call.

You can administer a warning tone on each system to let agents and callers know when someone is observing a call. Before connection, the conference tone may add 2-3 seconds delay if enabled. The parties hear a 2-second, 440-Hz warning tone before an observer connects to a call, followed by a half-second burst of this tone every 12 seconds during observation.

Service Observing with Exclusion

Starting with Release 2.2, an option is available on the Feature Related System Parameters screen called **Service Observing Allowed with Exclusion?**.

Release	Service Observing Allowed with Exclusion field	Result
Release 2.2 and later	Set to y	Allows Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
	Set to n (default)	Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped.
Earlier than Release 2.2	NA	

No-talk FAC for Service Observing

Starting with Call Center Release 3.1, a station user or a Service Observing **route-to number** vector operation can optionally activate Service Observing in a listen-only/no-talk mode that does not reserve a second time slot in port network gateway configurations. This does not apply to H.248 Media Gateway configurations. With this option, the ability to switch to the Service Observing talk mode while observing stations or ACD agents is denied. By not reserving the extra time slot, call recording applications that use Service Observing in a listen-only/no-talk mode have greater recording capacity since time slot usage is reduced.

This feature is available only through the Call Center release 3.1 Service Observing No Talk Feature Access Code (FAC), where a Service Observing listen-only/no-talk session cannot be activated or toggled to different modes using a Service Observing station button.

To use this option, first assign an appropriate dial code in the **Service Observing No-Talk Access Code** field on the Feature Access Code (FAC) screen. Then activate observing associations for call recording using the defined No-Talk FAC.

Observing Logical-Agent IDs

With EAS, an observer can observe agents based on their logical-agent ID rather than their physical phone. The observer enters the logical-agent ID extension number of an agent, who must be logged in to a phone. The observer can monitor every ACD, personal, and direct agent call delivered to or placed by the agent, including calls placed to the physical extension.

Only one observer can observe an extension at one time. An observer cannot observe a logical agent ID extension at a physical telephone that is already being observed. Likewise, an observer cannot observe a physical extension that is being observed as a logical-agent ID extension.

Observing VDNs

To observe a VDN, the observer enters a specific VDN extension and bridges onto calls (one call at a time) that have started vector processing for that VDN. The observer hears all tones, call prompting, caller dialing, announcements, music, and speech that the agent and caller hear. If an observer is in a COR administered to hear VDN of Origin announcements and has a VOA Repeat button, he or she can hear and replay VDN of Origin announcements.

Service observing of VDNs is enhanced to (optionally) start observation of a call to the VDN when the call is delivered 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 must first pass through vector processing before the observing is enabled. When the observing connection is completed for the first call (the call is released), the observer is bridged on a subsequent call to the VDN (which has also been through vector processing) when the call is answered by an observable agent/station. This ability saves time for the observer because, after observing of the VDN has been activated, the observer does not have to wait (and listen) for each subsequent call to go through vector processing and for the agent to answer.

The ability to observe VDNs when the call is delivered to an agent/station is activated by setting the **Observe on Agent Answer** field on the VDN screen to **y**.

The observer sees the name of the VDN, agent, or trunk as each is accessed in sequence by the VDN. For example, during vector processing the VDN name is displayed, but when the call connects to an agent, the agent name is displayed.

When the observer connects to a call in vector processing, the system 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 listen-only as long as the call is in vector processing. Once the call is out of vector processing, an observer with listen/talk capability can talk as well as listen.

Observing Remotely or by FAC

Observers can observe calls from a remote location or locally using Service Observing FACs. When observing remotely, observers must use FACs. Different FACs are required for listen-only and listen/talk modes. When observing locally or remotely by FAC, the observer cannot toggle between modes. Physical extensions, logical-agent ID extensions, and VDNs can be observed remotely.

Remote observing is initiated through Remote Access or Call Vectoring.

- With Remote Access, an observer accesses a communication server using a trunk group dedicated to Remote Access or using a 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.

ACD call center features

- With Call Vectoring, an observer accesses a communication server by dialing a VDN extension or a Central Office (CO) trunk that has a VDN extension as its incoming destination. Using route-to commands, you can design a Service Observing vector to allow a VDN call to directly access a specific extension to be observed or a Service Observing dial tone. At the dial tone, observers can enter any extension that they are authorized to observe. The following is a simple example of a Service Observing vector.

```
1.wait-time 0 seconds hearing ringing
2.collect 5 digits announcement 2300
  ("please dial your 5- digit security code")
3.goto step 5 if digits = 12345
4.disconnect after announcement 2000
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
```

You can combine Call Prompting and Call Vectoring to provide security and to limit observation.

Service Observing indicators

This section includes the following topics:

- [General indications to observer](#) on page 262
- [Tables showing what observers receive when using Service Observing](#) on page 263
- [General security](#) on page 267
- [VDN-call security](#) on page 267
- [Vector-initiated security](#) on page 267
- [Remote-access security](#) on page 268

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

Condition	Button lamp	Tone
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 show the indicators that observers receive when they activate and use Service Observing. In these 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. See [Service Observing considerations](#) on page 268 for conditions that make a call ineligible. 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 ¹	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

Feedback to observers when activation denied

Condition	State	Lamp	Tone
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

1. Extension COR cannot be observed or COR for observer calling permission does not allow observing the COR of extension to be observed.

Feedback to observer when activation allowed - at time of activation

Condition	State	Lamp	Tone
Active-eligible call	observing	steady/ winking	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
No active/eligible Call	wait state	flash	silence
Call in 6-party conference	wait state	flash	silence
Call already being observed	wait state	flash	silence
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

Feedback to observer when activation allowed - after observe activated

Condition	State	Lamp	Tone
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 Telephones

Action by observer	What happens	Condition
No action.	The Service Observing button is not highlighted.	The telephone is idle.
The observer presses the Service Observing button.	The Service Observing button is highlighted.	The system is waiting for dialing to begin.
The observer dials a station extension, Agent ID extension, or a VDN extension.	The Service Observing button remains highlighted and displays two down arrows.	The observer has successfully established a session and is "awaiting a call to observe".
No action.	The Service Observing 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 the Service Observing button 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 Service Observing button.	The observer has actively changed their Service Observing mode.

While observing, the observer should press only the following buttons:

- Call Appearance
- Service Observing
- Position Busy
- Auto-ckt Assure
- Release (ACD) (This will end Service Observing)
- Bridged Appearance
- Auxiliary Work
- Queue Status (NQC, OQT, AQC, and AQT)
- System Night Service
- Hold (ignored)

General security

Use the following COR restrictions to prevent unauthorized observing.

- For the observer, set the `Can Be An Observer` field on the COR screen to `y`.
- For the agent to be observed, set the `Can Be Observed` field on the COR screen to `y`.
- For the observer, grant permissions to all CORs to be observed on the Service Observing Permissions COR table.

VDN-call security

Use the following COR restrictions for VDN-call observing.

- For the VDN extension to be observed, set the `Can Be Observed` field on the COR screen to `y`.
- For the VDN destination, set the `Can Be Observed` field on the COR screen to `y`.
- Enter the VDN extensions to be observed in the observer's Service Observing Permissions COR table.

Vector-initiated security

Use the following guidelines for vector-initiated observing.

- Use Call prompting commands in Service Observing vectors to provide passcode protection and limit access to specific destinations or vector-verified, caller-entered digits.
- Use Time of Day/Day of Week checks in Service Observing vectors.
- Create a vector used exclusively for Service Observing.
- If you use route-to commands to observe a VDN extension, ensure the extension has an observable COR.
- If the observer is observing locally, grant calling permission to the observer on the VDN's COR.

In vector-initiated Service Observing, the COR assigned to the VDN used to initiate Service Observing, the COR assigned to the internal caller extension, and the COR assigned to agent to be observed are used to determine if Service Observing will be allowed. If the agent's COR is not observable, observation fails regardless of the VDN or caller COR. When a call routes through multiple VDNs, the COR of the last VDN is used for calling/observing permissions regardless of VDN Override settings.

If you have administered the optional warning tone, the caller and the observer hear the tone only when the system connects the call to the answering or routed-to destination after vector processing is finished. The periodic tone is heard during the call even if the call is transferred off-communication server. Use a warning announcement at the beginning of vector processing to inform the caller of observation since the system cannot give a warning tone until the call is out of vector processing.

Remote-access security

Use the following guidelines for remote observing.

- Use barrier codes and authorization codes to limit the use of Remote Access to authorized users. Refer to *Feature Description and Implementation for Avaya Communication Manager* for information about these codes and other remote access security measures.
- Use different authorization codes for different service observing permissions.
- Use Facility Restriction Levels (FRLs) and restrictions such as the authorization code COR to restrict Remote Access service observer access to other destinations (for example, stations or trunks).
- Use Call Prompting to create additional access security.

Assign the VDN, barrier code, and authorization code calling and service observing permissions and set Can Be Observer to yes on the associated COR screen. The last COR encountered is used to determine observer permissions.

Service Observing considerations

This section includes the following topics:

- [Observability](#) on page 268
- [Ineligibility](#) on page 268
- [Trunk calls](#) on page 269
- [Multiple observers](#) on page 269
- [Conferenced calls](#) on page 269
- [Transferred calls](#) on page 270

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.

Ineligibility

A call to an agent extension or VDN is ineligible for observing when the call:

- Is already being observed

- Is being busy-verified
- Has Data Privacy active
- Has Data Restriction active, 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, 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 Service Observing with Exclusion is active, observing is allowed when manual exclusion is active.

- Is in a conference where adding the observer results in more than six parties (see [Conferenced calls](#) on page 269 for more detail on conferences)
- Is a VDN-observed call that reaches an unobservable extension or VDN. (Note that the COR of the hunt group split or skill used to distribute the call to the station/agent is not checked. The CORs of stations/agents conferenced with the call are not checked.)

Trunk calls

If an agent being observed makes a trunk-call, observation starts after the agent finishes dialing. For Central Office (CO) trunks, dialing is considered complete when answer supervision is returned or when answer supervision timeout occurs.

Multiple 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 50.

When the **Allow Two Observers in the Same Call** option is active, the VDN observed call could connect to:

- An agent being observed by an agent
- A station observer
- A conference with another observer.

See [Service Observing with Multiple Observers](#) on page 274 for details.

Conferenced calls

An observer cannot initiate a conference while observing.

ACD call center features

If an observed agent conferences a call and the number of conferenced 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 conferenced 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.

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 interactions

ASAI: A call to an observed VDN continues to be observed after it routes to an adjunct. A call can be routed to a Service Observing FAC by the adjunct routing command in the same way that it can be 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 Call Center Release 3.1 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 Feature Access Code 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. If you need to notify the parties in a call that the call may be recorded, design your vector to execute an **announcement** vector step before queuing or routing the call that provides an audible warning message. As another option, you can have the Communication Manager provide the warning tones by enabling the Service Observing warning tones on the system-parameter features screen.

BCMS: BCMS does not report on Service Observing. BCMS reports show normal measured-call and agent activity related to Service Observing calls. When a physical agent (non-EAS) is observed, the BCMS Report By Login ID shows the physical extension along with the login ID.

Bridged appearances: If an observer observes 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.

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.

ACD call center features

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: Observing by VDN provides monitoring by DNIS since the VDNs represent the DNIS of the service dialed.

Direct Agent Calling: 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 or 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.

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 or assumed when the **Observe on Agent Answer** option is set to **y**. This includes routing to internal destinations (stations, hunt groups, ACD splits/skills, the attendant, etc.) or to external destinations (using trunk facilities).

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** option 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 match 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 with Multiple Observers

This section includes the following topics:

- [Service Observing with Multiple Observers description](#) on page 274
- [Prerequisites](#) on page 274
- [Administering Service Observing with Multiple Observers](#) on page 275
- [Service Observing with Multiple Observers interactions](#) on page 275
- [Service Observing with Multiple Observers example](#) on page 276

Service Observing with Multiple Observers description

Service Observing with Multiple Observers means that:

- Up to two observers can monitor the same agent Login ID or station extension using the Service Observing station button or using any of the following Feature Access Codes (FACs):
 - Service Observing Listen-Only
 - Service Observing Listen/Talk
 - Service Observing No-Talk
- Two separate calls, each with an associated service observer, can be conferenced together with both service observers included in the merged conferenced call except when both observers are VDN observers. In this case one VDN observer will be dropped.
- Customers who use call recording products, such as the Avaya Witness Call Recording or NICE can connect a voice-storage server to a station or Login ID extension in order to record agent-to-customer transactions acting as an observer.
- Customers who use call recording products can also allow an observer to monitor a station or Login ID extension and record the transaction at the same time.

Note:

This feature does not allow multiple observers on the same call for the Service Observing by VDN feature.

Prerequisites

You can use the Service Observing with Multiple Observers only if all of the following fields are set as follows:

- The **Call Center Release** field is set to 4.0 or later on the System Parameter Customer-Options screen.
- The **Service Observing (Basic)** field is set to **y** on the System Parameter Customer-Options screen.
- The **Can Be A Service Observer?** field is set to **y** on the Class of Restriction screen to be an observer.
OR
- The **Can Be Service Observed?** field is set to **y** on the Class of Restriction screen to be an observee.

Administering Service Observing with Multiple Observers

To administer Service Observing with Multiple Observers:

1. Enter `change system-parameters features`.
2. On the Features-Related System Parameters screen (page 11), enter **y** in the **Allow Two Observers in Same Call?** field and press Enter.
3. If you are going to use a recording device, do the following steps:
 - a. Enter `change cor x`.
 - b. On the Class of Restriction screen (page 2), enter **y** in the **Service Observing by Recording Device?** field for the COR to be used for the recording device line appearance and press Enter.

Service Observing with Multiple Observers interactions

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 Malicious Call Trace (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.

Service Observing with Multiple Observers example

A typical user of the Service Observing with Multiple Observers feature is Call Center in a financial institution like a bank where all the calls need to be recorded for security and quality control purposes. The Call Center may use a call recording facility like Avaya Witness Call Recording to record the calls to each agent. Additionally a Call Center manager or supervisor may simultaneously use temporary service observing of an agent to monitor how the agent handles calls.

The benefit provided by the Service Observing with Multiple Observers feature 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 (service observing association) 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.

Universal Call ID

This section includes the following topics:

- [About UCID](#) on page 277
- [How are UCIDs tracked?](#) on page 279
- [UCID interactions](#) on page 285
- [Administering the UCID feature](#) on page 286
- [UCID considerations](#) on page 290

About UCID

This section includes the following topics:

- [UCID definition](#) on page 277
- [What is UCID's purpose?](#) on page 277
- [What does UCID look like?](#) on page 278
- [How does UCID work?](#) on page 278
- [What creates UCIDs?](#) on page 278
- [When are UCIDs created?](#) on page 278
- [How are UCIDs transmitted?](#) on page 279

UCID definition

Universal Call ID (UCID) is a unique tag assigned to a 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.

What is UCID's purpose?

The purpose of UCID is to tag a call with a unique identifier.

ACD call center features

UCID provides a way to track calls across multiple communication servers and Voice Response Units.

Call centers can use UCID to track call history. Because UCID can uniquely identify every call in a network of any size, it is possible to track call-related data from multiple sources and multiple sites. For example, you can combine data from many locations and print reports that enable you to track a call throughout its lifecycle. For information about how to create reports, see *Avaya CMS Supervisor Reports*.

Note:

Although UCID is intended for call centers, a communication server configured to create UCIDs will assign one to every call - not just to Automatic Call Distribution (ACD) calls.

What does UCID look like?

The Universal Call ID is an 8-byte data element that displays as a 20-character number.

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 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 ¹
		ISDN (BRI or PRI) trunks using Shared UUI service protocol	codeset 0 shared user-to-user information (UUI) IE ¹
		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 ¹
Switch	CMS	X.25	SETUP5 CMS message
Switch	CTI adjunct	ASAI	various ASAI messages

1. Refer to [Information Forwarding](#) on page 178.

How are UCIDs tracked?

This section includes the following topics:

- [Overview of UCID tracking](#) on page 280
- [Station-to-station calls](#) on page 280
- [Incoming trunk calls](#) on page 280
- [Outgoing trunk calls](#) on page 280
- [Simple transfer or conference](#) on page 280
- [Complex conference](#) on page 281
- [Configuration - communication server before the IVR system](#) on page 282
- [Simple call tracking](#) on page 283
- [An IVR system transfers a call](#) on page 283
- [Configuration - IVR before the communication server](#) on page 284

Overview of UCID tracking

The way a network maintains and tracks a UCID depends on the call path. To illustrate UCID transport throughout a call's life cycle, this section describes several call scenarios:

- 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 either

- 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:

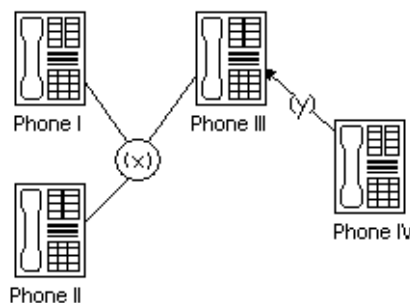
If, during the conference or transfer, the incoming call drops before the operation is complete, the two UCIDs will not appear to be associated because no merge of the two parts of the call was done.

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).



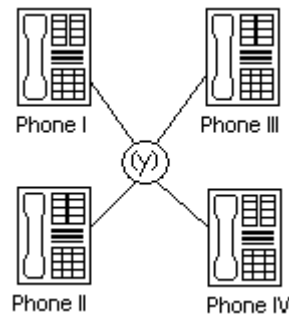
3. The person at Phone III puts the conference call on hold and answers the incoming call from Phone IV.

ACD call center features

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:

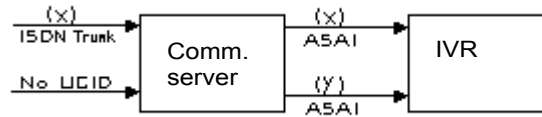
This configuration is more common than a call coming in to the IVR system before reaching the communication server.

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.



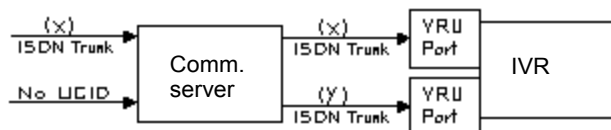
When the communication server is before the IVR system:

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

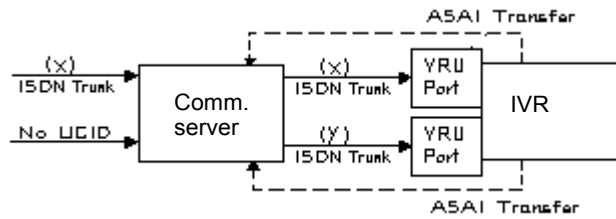
The following call scenario involves an IVR system behind the communication server configuration when the IVR system initiates a call transfer after the call is answered by a port on the IVR system that serves as an ACD agent.

1. Call is directed to the IVR system VRU port (typically by call vectoring) with UCID information (UCID x or UCID y).

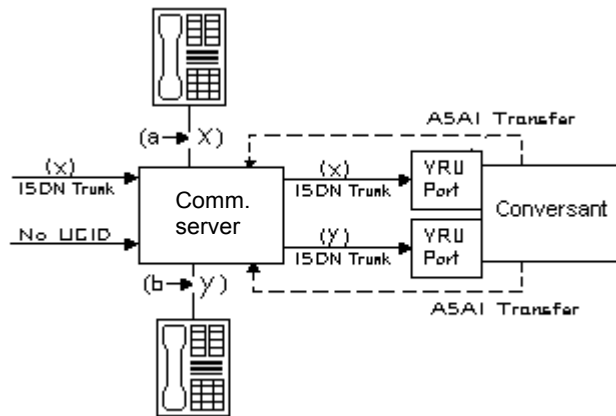


ACD call center features

- The IVR system determines the call's destination and transfers the call (using an ASAI third-party transfer operation).



- 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).



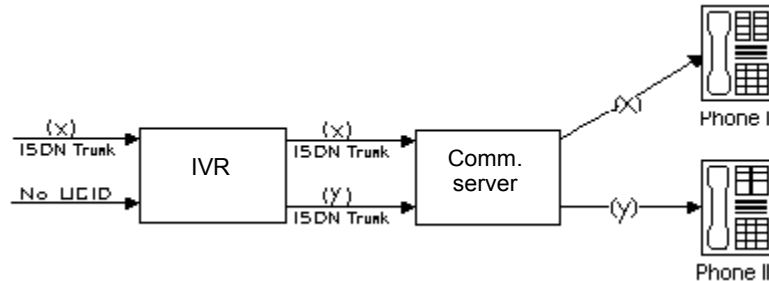
- The UCIDs of the transfer segment and merged call are returned to the IVR system in ASAI acknowledgment messages.
- The communication server sends UCID information to CMS if trunk, VDN(s), and/or split or skill(s) involved in the call are measured.

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 interactions

Distributed Communications System (DCS): If DCS is used in a network of communication servers where UCIDs are tracked, the DCS feature must be configured with ISDN or SIP trunks having the Shared UUI service protocol. Otherwise, calls that are handled through one of the many DCS features (such as DCS Coverage) will not retain the UCID initially assigned to the call.

Remote messaging system: For a remote messaging system over DCS, the DCS trunks used to accomplish the remote messaging system operation must be configured (as described previously in Distributed Communications System) to retain the UCID associated with a call.

Tandem Calls: When a tandem call is made through the communication server, the UCID information may be blocked or passed through the tandem communication server. To pass a UCID through a tandem communication server, both the incoming and outgoing trunks at the tandem communication server must be configured to handle UCIDs. See [Information Forwarding](#) on page 178 for proper private and public network information forwarding administration.

Administering the UCID feature

This section includes the following topics:

- [Before you start](#) on page 286
- [Administering UCID](#) on page 287

Before you start

Before you start to administer the UCID feature, be aware that UCIDs are successfully transmitted only when all of the communication server network components meet the requirements (software and connections) specified below.

Some requirements vary, depending on how you want to use UCID. The following table lists requirements for different applications of UCID.

Operation	Requirement
UCID sent from communication server to communication server	ISDN (BRI/PRI) trunks with Shared UUI or QSIG service protocol, or SIP trunks with Shared UUI
UCID sent from communication server to the IVR system	ASAI link to the IVR system
Communication server receives UCID from the IVR system	ISDN-PRI connection (with shared UUI) between communication server and the IVR system
Switch sends UCIDs to CMS	Ethernet connection from switch to CMS
Switch sends UCIDs to a CTI Application	ASAI link to adjunct

To maximize the benefits of UCID, it is recommended that you have an updated version of the Avaya communication server.

In the case of a communication server network component that cannot support UCIDs, it is recommended that the component (trunk group, ASAI connection, or CMS software) be administered to disallow the sending or receiving of UCID. For example, if an Avaya communication server is connected to a non-Avaya switch, then the connecting trunk must be administered to not send UCID over that trunk for outgoing calls.

Administering UCID

These instructions assume you're using the System Administration Terminal (SAT) screen or a terminal emulator to access the communication server software.

There are three tasks involved in administering UCID on the communication server. Complete these tasks in the following order:

Task A: Check ASAI Interface before enabling UCID

Task B: Set the communication server to create and send UCIDs

Task C: Enable UCID Transmission on Trunk Groups

Task A: Check ASAI interface before enabling UCID

If this procedure is not performed, you may encounter the *ASAI Interface feature not assigned* error message in later steps.

Screen	Field	Why?	Optional?
Optional Features	ASAI Interface	Global setting to send UCIDs to ASAI	Y

To check the ASAI interface:

1. In the command line, enter `change system-parameters customer-options` and press RETURN.
2. On page 1, enter `y` in the `ASAI Interface` field.
3. Log out and log back in if this and any other fields in this screen have changed.

Task B: Set the communication server to create and send UCIDs

You must administer each communication server that you want to create UCIDs. If you do not administer a communication server, it will pass along an already-created UCID, but it won't create one if a call comes to it first.

Screen	Field	Why?	Optional?
Feature-Related System Parameters	Create Universal Call ID (UCID)?	To generate a UCID for each call when necessary.	N
	UCID Network Node ID	Important component of the UCID tag.	N
	Send UCID to ASAI?	So that ASAI receives UCIDs.	Y
	Adjunct CMS Release	So that CMS receives UCIDs.	Y

For the communication server to create and send UCIDs:

1. If your network includes CMS and you want CMS to track UCIDs, then enter **busyout mis** in the command line. If not, go to Step 2.
2. In the command line, enter change system-parameters feature and press RETURN.
3. The Feature-Related System Parameters screen displays.
4. Go to page 4 of the screen.
5. In the Create Universal Call ID (UCID)? field, enter y.
6. In the UCID Network Node ID field, enter the node ID number.
Valid numbers are from 1 to 32,767.



CAUTION:

The UCID Network Node ID *must* be unique for every communication server and IVR in the system. If it is not unique, the integrity of the UCID is compromised.

7. If your network includes ASAI, go to page 7 of the screen. If not, go to Step 8.
8. In the Send UCID to ASAI? field, enter y.
9. If you have performed the **busyout mis** command, go to Step 9. If not, you are done with this task (Task b), so press **Enter** to save your work and go to Task c.
10. Go to page 8 of the screen.
11. In the Adjunct CMS Release field, enter R3V6.
12. Press **Enter** to save your work.
13. In the command line, enter release mis.

Task C: Enable UCID transmission on trunk groups

ISDN trunk groups

When you send UCIDs over ISDN trunks, it is administered on a trunk group basis.

The following table provides the screen and field information that you need to perform this task.

Screen	Field	Why?	Optional?
Trunk Group	Group Type	Specify the group type as ISDN.	N
	UUI IE Treatment	Must be set to Shared to send UCID for protocols other than QSIG.	N
	Supplementary Service Protocol	Specify correct service type. b is for QSIG, others are for UUI.	N
	Send UCID?	Allows or blocks UCID transmission.	N

To enable UCID transmission on an ISDN trunk group:

1. In the command line, enter **change trunk-group n** and press RETURN.
n is the number of the trunk group you want to administer.
2. The Trunk Group administration screen displays.
3. On page 1, enter **ISDN** in the Group Type field.
4. Go to page 2 of the screen.
5. In the `Supplementary Service Protocol` field, enter the letter of the service protocol you want for this trunk group.
b is for QSIG, other protocols are for UUI. For information about administering UUI, see [Administering User-to-User Information transport for ISDN trunks](#) on page 179
6. Go to page 3 of the screen.
7. In the **Send UCID?** field, enter **y**.
8. Press **Enter** to save your changes.

SIP trunk groups

When you send UCIDs over SIP trunks, it is administered on a trunk group basis. The following table provides the screen and field information that you need to perform this task.

Screen	Field	Why?	Optional?
Trunk Group	Group Type	Specify the group type as SIP.	N
	UUI Treatment	Must be set to Shared to send UCID.	N
	Send UCID?	Allows or blocks UCID transmission.	N

To enable UCID transmission on a SIP trunk group:

1. In the command line, enter **change trunk-group n** and press RETURN.
n is the number of the trunk group you want to administer.
2. The Trunk Group administration screen displays.
3. On page 1, enter **SIP** in the Group Type field.
4. Go to page 3 of the screen.
5. In the **UUI Treatment** field, enter **Shared**.
6. In the **Send UCID?** field, enter **y**.
7. Press **Enter** to save your changes.

You also need to administer your trunk groups to send user data over your private and public networks. To administer the trunk groups, see [Information Forwarding](#) on page 178.

UCID considerations

UCID has been tested with several major carriers. To find out if these 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 involved with the preferred specific configuration, use of private trunking between the nodes should be assumed until successful testing has been completed.

Troubleshooting

The following troubleshooting hints should be reviewed when UCIDs are not transmitted, even though you received no error messages while administering the UCID feature, and all software and connections meet the minimum requirements:

- A tandem communication server has the `Send UCID?` option set to `y` for all trunk groups that AAR/ARS or station users may use to tandem an incoming call.

- If DCS is used, make sure all ISDN trunks between the communication servers used for DCS or remote AUDIX are configured in the D-channel mode.
- For CMS tracking purposes, make sure all trunks, VDN, and split or skills that handle calls for which UCIDs are tracked are administered as measured (either both or external).

Variables in Vectors

Variables in Vectors (VIV) is a Call Vectoring feature introduced in Avaya Communication Manager 2.0. The VIV feature allows you to create variables that can be used in commands to:

- Improve the general efficiency of vector administration.
- Provide increased manager and application control over call treatments.
- Allow you to create more flexible vectors that better serve the needs of your customer and call center operations.

The 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 (Feature Access Codes) 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.

For more information about VIV capabilities, administration requirements, and vector examples, see *Avaya Call Center Call Vectoring and EAS Guide*.

VDN in a Coverage Path

This section includes the following topics:

- [About VICP](#) on page 293
- [Administering VICP](#) on page 293
- [VICP considerations](#) on page 293
- [VICP interactions](#) on page 294

About VICP

VDN in a Coverage Path (VICP) enhances Call Coverage and Call Vectoring. If Basic Call Vectoring or Call Prompting is enabled on your communication server, you can assign a Vector Directory Number (VDN) as the last point in a coverage path. Calls that reach this coverage point can be processed by a vector or by Call Prompting.

Administering VICP

The following forms and fields are required to administer the VICP feature.

Screen	Field
Coverage Paths	All
Call Vector	All
Vector Directory Number	All

Call Coverage Path screen - Set one of the following to the extension of the VDN you want to use as a coverage point: Point 1, Point 2, Point 3, Point 4, Point 5, Point 6.

VICP considerations

Once a call has covered to a VDN, it cannot be further redirected by features such as Call Coverage, Call Forwarding, or Night Service.

A VDN is not allowed to be a member of a coverage answer group. A vector cannot route a covered call to a coverage answer group - a coverage answer group can only be a point in a coverage path.

Removing a VDN from the system with the `remove vdn <extension>` command automatically removes the VDN from any coverage paths.

VICP interactions

AAR/ARS Partitioning: The class of restriction assigned to the VDN determines the partition group number (PGN). The PGN in turn determines the AAR or ARS routing tables used by `route-to` commands.

ASAI: For direct calls to a VDN, the `adjunct routing` command operates like the command `route to digits with coverage=y`. For calls that cover to a VDN, however, the `adjunct routing` command operates the same as a `route to digits with coverage=n` command. Since calls redirected once to coverage should not be redirected again, the coverage option is disabled for the `adjunct routing` command in this situation.

Attendant: A call covering to a VDN can be connected to an attendant queue or hunt group by a vector. Internal calls that route to an attendant display the class of restriction 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, the conference will be denied until the call has completed vector processing.

An attendant-extended call that covers to a VDN will not return. If the attendant extends a call to a local destination that covers the call to a VDN, the attendant's return call timer is canceled when vector processing begins and the Return Call button will 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 may be subsequently transferred to AUDIX. Calls may also be transferred out of 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.

Calls that have covered to a VDN cannot be redirected again by Call Coverage.

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.

Call Park: A parked call will not cover to a VDN. When a call is parked at an extension with a VDN in its coverage path, the call will continue ringing the extension. If the call is parked to a hunt group extension and the call is in queue, the call will remain in the queue until it is retrieved, or answered by an agent, or abandoned by the caller. A vector event is generated for these 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.

Call Vectoring. The class of restriction assigned to a VDN determines the partition group number (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 has no effect on the display shown on an answering display. This station will show 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 with coverage=y` command. For calls that cover to a VDN, however, the `adjunct routing` command operates the same as a `route to digits with coverage=n` command. Calls redirected once to coverage should not be redirected again, however, so in this situation the coverage option is disabled for the `adjunct routing` command.

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 about 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 call vectoring command `route-to`. The coverage option for the `route-to digits` command is disabled for covered calls. In other words, the `route-to digits with coverage=y` functions like the `route-to digits with coverage=n` command when processing covered calls. 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 Message Center system.

Class of Restriction (COR): The COR assigned to the covering VDN governs the vector routing of the call.

Conference: Calls in an established conference will 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 Consult 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.

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 will 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 will 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 will 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. Note that this interaction can also occur when a `messaging split` or `route-to` command routes a covered call to a split that isn't vector-controlled.

Look-Ahead Interflow: For calls that have covered to a VDN, LAI works like a `route-to digits/number with cov=n` vector command. Any Dialed Number Identification Service (DNIS) digits sent with the interflowed call will indicate 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 may 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 or 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 called station B. Station B was busy, and the call covered 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.

Terminating Extension Groups: A VDN may be assigned as the last point in the coverage path for a Terminating Extension Group.

Transfer: Calls may be transferred to extensions that cover to a VDN. Users who receive a covered call may transfer it 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 may be subsequently transferred to AUDIX. Calls may also be transferred out of AUDIX to a VDN.

VDN of Origin Announcement

This section includes the following topics:

- [About VOA](#) on page 297
- [Administering VOA](#) on page 297
- [VOA detailed description](#) on page 298
- [VOA considerations](#) on page 299
- [VOA interactions](#) on page 300

About VOA

VDN of Origin Announcement (VOA) provides agents with a short message about a caller's city of origin or 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 800 numbers, one for placing orders and one for technical support, you can administer two 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. When an agent answers the call, he or she hears the VOA message and can respond appropriately to the caller's request.

Administering VOA

The following forms and fields are required to administer the VOA feature.

Screen	Field
Attendant Console	Feature Button Assignments: <ul style="list-style-type: none"> ● VOA Repeat
Class of Restriction (COR)	VDN of Origin Announcements
Feature-Related System Parameters	Hear Zip Tone Following VOA?
Announcements/Audio Sources	All
Vector Directory Number	VDN of Origin Annc. Extension
Phones	Feature Button Assignments: <ul style="list-style-type: none"> ● VOA Repeat

ACD call center features

- Announcements/Audio Sources - Assign each VOA you want to use. You can administer aux-trunk types with queue, without queue, and with barge-in. You can administer integrated types with queue and without queue. Do not administer analog and integrated repeating announcement types as VOAs.

Note:

The VDN for which you are administering a VOA must be in a vector command line.

VOA detailed description

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.

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

- Because callers are kept waiting while a VOA plays, messages should be kept very brief - no more than 1.5 seconds in length. Agents should use a speakerphone or headset, so they do not miss the VOA while they are picking up the handset. If agents cannot use a speakerphone or headset, administer phones with a VOA Repeat button.
- If you have multiple announcement boards, you should place shorter VOAs on one board and longer recorded announcements on the other to avoid delaying delivery of VOAs. If you have only one announcement board, place VOAs on the integrated board and consider installing 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 the TN750 circuit board for 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 on the TN750. A burst of VOAs can delay other announcements. Therefore, record non-VDN of Origin Announcements as auxiliary or analog.
- 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 should consist of one short announcement that repeats during the full announcement time. For example, you might want to 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 cannot play an announcement. If you use aux-trunk or integrated announcement with queue, the system plays the current announcement for an agent and then connects the next agent in the queue.

VOA interactions

Agent Call Handling - Automatic Answer:

- ACD agents at phones in Auto Answer mode hear a zip tone, then the VOA. You can also administer a zip tone after the VOA completes, to alert agents that an announcement is complete and a caller is connected.
- Non-ACD agents can receive a VOA if a call is routed to them using vector processing. When non-ACD agents at phones in Automatic Answer mode receive calls, they hear a call ID tone then the VOA. Agents hear a second zip tone after the VOA indicating connection to the caller.

Agent Call Handling - Manual Answer: When non-ACD agents at phones in Manual Answer mode receive calls they hear ringing, answer the call, and hear the VOA.

ASAI Adjunct Routing: If a vector step includes Adjunct Routing, the VOA is played for the agent to whom the call is routed.

Auto-Available Split/Skill (AAS): AAS is intended to be used for splits/skills containing only nonhuman adjuncts such as a voice messaging system or an IVR system. However, VOAs can be directed to Auto-Available splits/skills.

Call Forwarding: VOAs apply to forwarded calls, including those forwarded to a hunt group. The answering station must be on the same communication server. If a VOA is forwarded, the message is played only if the destination extension is administered with a COR that allows VOA.

Call Pickup: Call Pickup allows an agent to pick up a ringing call on another extension. If the pick-up extension has COR permissions for VOA, the agent can receive a 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 the VOA. Only the person using the button can hear the 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, a VOA is not applied. After returning to the vector, the call can be routed to a station or VDN where the answering agent receives the 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 are not played 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. The VOA only applies to direct agent calls if the calls reach an agent through vector processing. Direct agent calls from a phone on a communication server are not vector-processed and cannot cause a VOA to be played.

Enhanced Automatic Wake-up: If you are using enhancements to Automatic Wake-up with integrated announcements, there can be contention for integrated announcement ports. VOAs have priority over Automatic Wake-Up announcements.

EAS: When you are using Expert Agent Selection (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 telephone the agent is using.

Hold: Agents cannot use the VOA Repeat button if their calls are all on hold. The VOA Repeat button only applies to active calls.

Home Agent: You can assign an initial VOA to a home-agent port on the communication server. 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 routed to a hunt group. The COR for the answering station's extension determines whether the station can receive a VOA.

Look-Ahead Interflow: VOAs apply only to the communication server where the VDN is defined. If a call interflows to another communication server, the VOA is lost. You can have the interflow to another communication server access a VDN with the same VOA message as on the original communication server.

Redirection on No Answer (RONA): If a call re-queues to a split or skill because the RONA timer expired, the 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 he or she hears the announcement. However, if another party on the call presses the VOA Repeat button, the user and the observer hear the VOA.

Supervisor Assist: If an agent requests supervisor assistance and conferences the supervisor into a call, either the agent or the supervisor can use their VOA Repeat button to replay the VOA, but only the person who presses the button hears the VOA.

ACD call center features

Transfers: If an agent receives a VDN call and transfers the call, the answering party can use the VOA Repeat button to replay the message.

VOA distribution: If you use long VOAs or multiple VOAs, there may be a delay between the zip tone and the announcement. The system provides multiple announcement circuit packs to help prevent announcement delays. Contact your Avaya representative for more information.

VDN Time Zone Offset

This Vector Directory Number (VDN) feature is designed for call centers with locations in different time zones. You can program a single vector with TOD conditional steps that handle each time zone based on the active VDN for the call.

This section includes the following topics:

- [Reason to use VDN Time Zone Offset](#) on page 303
- [Operation](#) on page 303
- [Interactions with other features](#) on page 304
- [Example of VDN Time Zone Offset](#) on page 304
- [How to administer VDN Time Zone Offset](#) on page 305

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 a time in all vectors without converting to the local time reflected on each switch clock.

Operation

Call Vectoring time-of-day (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 Daylight Savings Time (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 TOD conditional calculation based on the active VDN for the call. This way 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.

Administering the offset

The offset is assigned on the VDN screen as follows:

[+ or -] [0 - 23] : [0 - 59]

0 - 23 = hours

0 - 59 = minutes

The offset changes the server local time used for the TOD conditionals in the vectors processed for the VDN to the VDN local time.

- Use the - sign if the VDN local time is earlier than the server local time.
- Use the + sign if the VDN local time is later than the server local time.

For example, if the server local time is in Eastern Standard Time and the local time used for calls to the VDN is Mountain Standard Time, use - 2:00.

Interactions with other features

VDN Time Zone Offset interacts with other features as follows:

- VDN Time Zone Offset does not apply to time calculations associated with the do, dow and tod Variables in Vectors variable types. Those 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. For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.
- The LSP and ESS servers should be synchronized with the main server system time. This ensures that if a switchover occurs to the survivability server, the VDN offset is applied consistently when the vector TOD conditional steps are being processed by the survivability server.

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. They can do this by using the VDN Time Zone Offset feature along with skill preferences. This company uses Expert Agent Selection (EAS) and the system switch clock is set to GMT.

Assuming Daylight Savings 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 (Greenwich Mean Time). 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	1st = 51
VDN2	New York	10002	-5:00	1st = 60
VDN3	Denver	10003	-7:00	1st = 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.*]

How to administer VDN Time Zone Offset

Administer this feature using the **VDN Time-Zone Offset** field on the **VDN** screen. For more information about this field, see [VDN Time-Zone Offset](#) on page 367.

VDN Variables

VDN Variables provide more opportunities for VDNs to use a smaller set of vectors.

You can:

- Assign up to five variable fields, V1 through V9, on the **VDN** screen
- 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 use 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

Reason to use

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 *Avaya Call Center Call Vectoring and EAS Guide*.

Voice Response Integration

This section includes the following topics:

- [About VRI](#) on page 307
- [Administering VRI](#) on page 308
- [VRI detailed description](#) on page 308
- [VRI interactions](#) on page 310

About VRI

Voice Response Integration (VRI) integrates Call Vectoring with the capabilities of voice response units (VRUs). You can:

- Run a VRU script while retaining control of a call in vector processing
- Run a VRU script while a call is queued, retaining its position in the queue
- Pool IVR ports for multiple applications
- Use a VRU as a flexible external-announcement device
- Pass data between the system and a VRU
- Tandem VRU data through a communication server to an ASAI host

The **converse-on** command, which is part of Basic Call Vectoring, provides these capabilities. Use a **converse-on** call-vector step to integrate a VRU with Automatic Call Distribution (ACD). VRI allows you to use VRU capabilities while controlling a call in ACD.

Include VRUs with vector processing to take advantage of the following:

- Access to local and host databases
- Validation of caller information
- Text-to-speech capabilities
- Speech recognition
- Increased recorded announcement capacity
- Audiotex applications
- Interactive voice-response (IVR) applications
- Transaction-processing applications

VRI allows users to make productive use of queuing time. For example, while a call is queued, a caller can listen to product information using an audiotex application or can complete an interactive voice-response transaction. It may be possible to resolve the caller's questions while the call is queued, which helps reduce queuing time for other callers during peak times.

Administering VRI

Enable Call Prompting to allow the system to collect digits from the caller and an IVR system to return data. You must have Call Prompting to administer the Converse Data Return Code and use the digits keyword for the <data_1> or <data_2> fields on the `converse-on` command.

Screen	Field
System Parameters Customer-Options	Call Prompting
Feature Access Code (FAC)	Converse Data Return Code
Feature-Related System Parameters	Converse Delay Data1/Data2 Converse Signaling Tone/Pause
Call Vector	All

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/skills. Non converse splits/skills are splits/skills that are accessed by `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 may pass 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 may return 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 its 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.

Call Prompting allows you to collect and use digits that the VRU returns. These 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's (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, thus 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. Take the following precautions:

- If the collected digits are used to route calls internally, ensure that the Class of Restriction (COR) for the vector directory number (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
```

VRI interactions

Converse splits interact like other vector-controlled splits unless noted here.

Adjunct Switch Applications 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 do this.

Agents: Although not recommended, 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 they 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 it hasn't previously been sent. If digits are passed to the VRU, answer supervision is sent after 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 may 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/skills for VRI except when ASAI controls the converse split or skill.

Automatic answering: When you administer ports on your IVR system as agents of a converse split or skill, do not administer agents as automatic answer. The system-provided zip tone may interfere with the interaction between the IVR system and the calling party.

BCMS/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 considered 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 may no longer match, VDN and trunk totals match.

Call Detail Recording: 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; these 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: 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: You can administer a converse split or skill as a DDC split or skill.

Distributed Communications System: If an incoming DCS call is placed to a vector with a **converse-on split/skill x pri m passing ani...** step, the caller's DCS extension is sent to the VRU.

Expert Agent Selection: **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.

ISDN: You can administer a **converse-on** step to send a caller's calling party/ billing number (CPN/BN) to the IVR system using the caller keyword.

Intrswitch CDR: If a converse-on call is answered and either the caller or the VDN associated with the call is administered for intraswitch recording, timing for the call is started and the CDR record shows **calling party to VDN** as the originating and answering parties.

Line-side T1 connectivity: T1 connectivity between the switch 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.

ACD call center features

Look-Ahead Interflow: 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 a Look-Ahead Interflow 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 it to a message split, it 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: 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 they are 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 ANI to VRUs using the ani data item on the **converse-on** step.

Recorded announcement: Use VRI to increase the system's 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 timeout* administered, and the call rings an agent/port for longer than the timeout interval, the call redirects and the agent/port is put into 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 may 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's 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.

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/skills: `Converse-on` steps can deliver calls only to skills or vector-controlled splits.

VuStats

This section includes the following topics:

- [About VuStats](#) on page 314
- [Administering VuStats](#) on page 315
- [VuStats detailed description](#) on page 318
- [VuStats considerations](#) on page 322
- [VuStats interactions](#) on page 322
- [VuStats display interactions and modifications for IP telephones](#) on page 323

About VuStats

VuStats presents call center statistics on phone displays. Agents, supervisors, call center managers, and other users can press a button and view statistics for agents, splits or skills, VDNs, and trunk groups.

These statistics reflect current information collected during the current BCMS interval, information collected since the agent logged in or 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 on demand or update periodically.

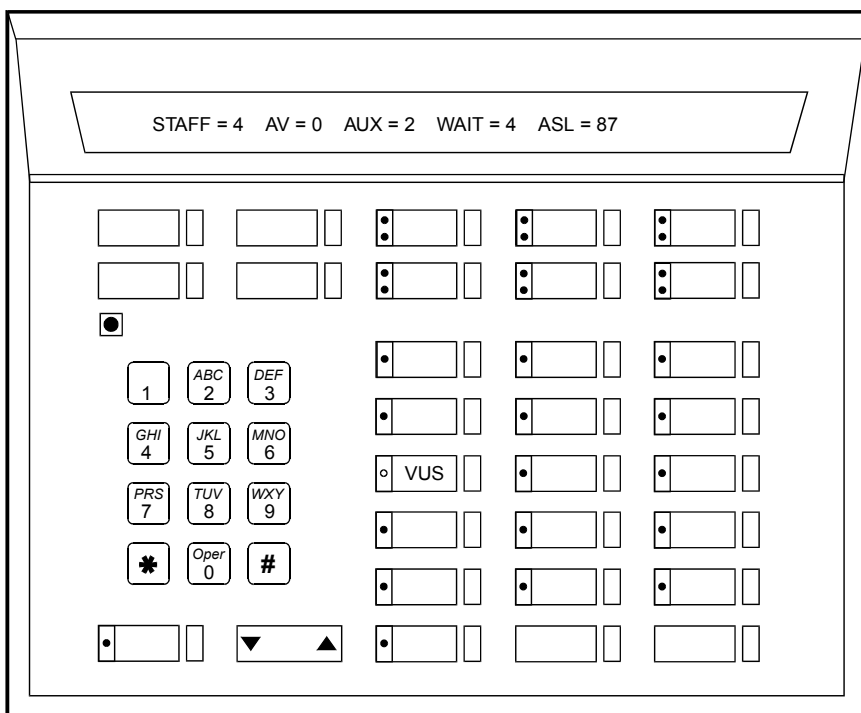
With VuStats, anyone who is using a telephone with digital display can view BCMS statistics, which are otherwise available only on BCMS reports or management terminals. These statistics can help agents monitor their own performance or can be used to manage splits, skills, or small call centers.

Note:

Although VuStats can run with either BCMS or CMS enabled, neither is required.

The following figure illustrates a Callmaster with a VuStats display.

Callmaster with VuStats display



callmstr CJL 061896

Administering VuStats

The following forms and fields are required to administer the VuStats feature.

Screen	Field
System Parameter Customer-Options	ACD BCMS/VuStats Login ID BCMS/VuStats Service Level VuStats or VuStats (G3V4 Enhanced)
Feature-Related System Parameters	ACD Login Identification Length BCMS/VuStats Measurement Interval BCMS/VuStats Abandoned Call Timer Validate BCMS/VuStats Login IDs Clear VuStats Shift Data
Trunk Group	Measured

ACD call center features

Screen	Field
Attendant Console	Feature Buttons
BCMS/VuStats Login ID	Login ID, Name
Hunt Group	ACD Acceptable Service Level Measured Objective
Station	Feature Buttons
Vector Directory Number	Acceptable Service Level Measured
VuStats	All

Display the System-Parameter Customer-Options screen and ensure that ACD, BCMS/VuStats Login IDs, BCMS/VuStats Service Level, and VuStats or VuStats (G3V4 Enhanced) are set to y.

Feature-Related System-Parameters screen

ACD Login Identification Length: If you are not using EAS, enter a number (1-9) that identifies the length of agent login IDs used by BCMS/VuStats. If you are not using BCMS/VuStats login IDs, accept the default 0. This field defines the ACD login ID length and the BCMS login ID length, so you must coordinate with the BCMS administrator before changing this field.

BCMS/VuStats Measurement Interval: This interval determines how frequently BCMS polls and records data for BCMS reports and VuStats displays. Set this field to half-hour or hour.

If you specify hour, an entire day of traffic information is available for BCMS history reports. Otherwise, only half a day is available. There is a maximum of 25 measurement intervals, including the current interval.

BCMS/VuStats Abandon Call Timer: Set this field to 1-10, or leave blank. This value is the number of seconds a call can last and still be recorded as an abandoned call. For example, if you set this field to 5, a call could last up to 5 seconds and be recorded as abandoned. Thus, very short calls are not included as ACD calls in BCMS and VuStats statistics. Abandoned time is measured from the time the call is answered until the agent hangs up. Any time an agent is on a call that is within the abandon call timer value is recorded as total AUX time with the default reason code. Use this timer if your central office does not provide disconnect supervision.

Validate BCMS/VuStats login IDs: Set to n to allow entry of any ACD login of the proper length. Set to y to allow entry only of login IDs that have been entered on the BCMS/VuStats Login-ID screen.

Clear VuStats Shift Data: Set to on-login or at-midnight to specify when shift data for an agent is cleared.

Agent Login ID screen

Administer agent login IDs for EAS. With EAS, VuStats accesses agent and agent-extension object type information based on agent login ID. Agents logging in agent IDs (administered on this screen or BCMS/VuStats Login ID screen) can view their own statistics on any VuStats phone they are using. If agent IDs are not administered, VuStats displays only statistics collected for the agent's extension.

Trunk Group screen

For each trunk group that will have VuStats display statistics, set Measured to internal or both. Specify internal to record statistics for BCMS/VuStats. Specify both to record statistics for BCMS/VuStats and CMS.

Attendant Console screen

Administer a VuStats feature button (vu-display) to allow an attendant to display VuStats statistics. There is no limit to the number of VuStats buttons that can be administered.

Fmt : When you assign VuStats feature buttons, an Fmt field appears. You can associate a VuStats feature button with a particular display format. The Fmt value identifies the VuStats format used to display the information. Specify 1 - 50 in the Fmt (1 is the default format).

ID number : Optionally administer an ID number for each vu-display button. Use the ID number to define the agent, split or skill, trunk group, or VDN that the VuStats display will describe. The ID can be an agent login ID or extension number, a split or skill or trunk group number, or a VDN extension. For example, a vu-display button administered with split or skill ID 6 is used to view statistics for split or skill number 6.

Do not administer IDs for VuStats displays with the agent object type. Agent object type displays are limited to statistics for the logged-in agent.

IDs allow supervisors and agents to bypass entering an agent extension, split or skill, or VDN number when viewing statistics. IDs can also be used to limit access to certain statistics to designated phones.

BCMS/VuStats Login ID screen

Administer agent login IDs if you do not have EAS. BCMS/VuStats login IDs can be used to track statistics by specific agent rather than extension number. Use any character, except a space, as a placeholder for data in Format Description text. The default is \$. Each character holds a place for one character of data.

Hunt Group screen

ACD: Set this field to y.

Acceptable Service Level: Specify the number of seconds within which calls to this hunt group are answered. Calls answered within this time are considered acceptable. BCMS and VuStats use this value to determine the percentage of calls that meet the acceptable service level.

Measured: Set this field to internal or both. Specify internal to record statistics for BCMS/VuStats. Specify both to record statistics for BCMS/VuStats and CMS.

Objective: Specify an objective, or goal, for the split or skill. Examples include an agent objective of a number of ACD calls to be handled, an average talk time, or a percent of calls to be answered within the acceptable service level.

Station screen

Administer a VuStats feature button (vu-display) to allow agents to display VuStats statistics. For more information, see [Attendant Console screen](#) on page 317.

Vector Directory Number screen

For each VDN that has statistics displayed by VuStats, administer the following fields:

Acceptable Service Level: Specify the number of seconds within which calls to this VDN are answered. Calls answered within this time are considered acceptable. BCMS and VuStats use this value to determine the percentage of calls that meet the acceptable service level.

Measured: Set this field to internal or both. Specify internal to record statistics for BCMS/VuStats. Specify both to record statistics for BCMS/VuStats and CMS.

VuStats Display Format screen

For definitions related to completing this screen, see the [VuStats Display Format screen](#) on page 370.

VuStats detailed description

This section includes the following topics:

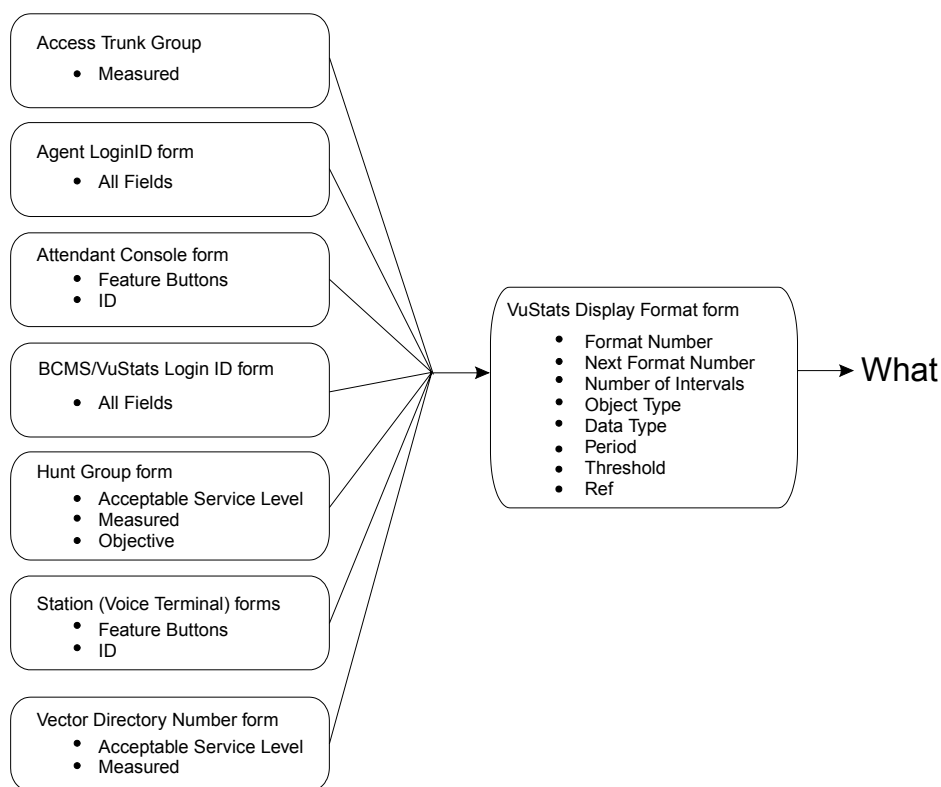
- [VuStats forms and fields](#) on page 319
- [Data type](#) on page 319
- [Period](#) on page 320
- [Threshold](#) on page 320

- [Format description](#) on page 320
- [Display linking](#) on page 320
- [How the information looks](#) on page 321
- [When the information updates](#) on page 321

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 can display information agents are interested in, such as the total number of calls the agent has answered since login, the average time 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 level.

ACD call center features

For split or skill object types, VuStats can display split or skill description and performance information, such as average speed of answer, number of calls waiting, and agent work states. VuStats can also display an objective, acceptable service level, or percent of calls answered within the acceptable service level for a split or skill.

For more information, see the data types tables in [ACD Call Center screens](#) on page 325.

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 4 completed intervals, it displays the number of ACD calls received in the past 2 hours (1/2-hour intervals) or 4 hours (1-hour intervals) plus those completed during the current interval. Using historical data can affect processor occupancy, depending upon the number of active users, their 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 \geq (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](#) on page 315, *AUX=* identifies the data type *split-agents-in-aux-all* (that is, the number of agents currently in AUX work mode for a specified split or skill). Text appears on the display exactly as you enter it in the field. Text is optional.

Because of the 40-character limit, use abbreviations when possible. For example, use **S=** to indicate split number.

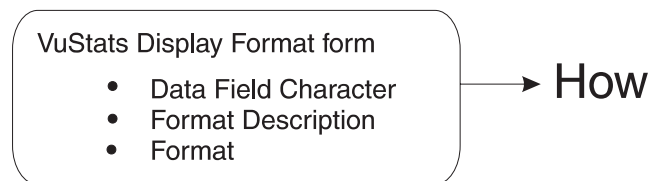
Display linking

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 telephones.

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	<ul style="list-style-type: none"> ● 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.

ACD call center features

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 an agent's login session. This shift data clears either at midnight or the next time the agent logs in depending upon how the system is administered. If the data clears at login and agents log out to go to lunch, the system clears their accumulated data when they log back in after lunch.

To accumulate a full day's statistics, you can require agents and supervisors to keep a running total of all their login sessions, or, to avoid this, use historical data, require agents to use AUX work mode when temporarily unavailable, or administer the system to clear shift data at midnight.

VuStats interactions

BCMS: You must have BCMS activated to receive BCMS reports. VuStats displays data collected by BCMS, but BCMS need not be enabled for you to use VuStats.

Call Prompting: When Call Prompting digits are displayed, VuStats is canceled. When an agent reactivates VuStats, the VuStats display overwrites the Call Prompting display.

Call Work Codes (CWC): The CWC-display prompt suspends VuStats, so when the CWC prompt is removed, the VuStats display reappears.

If VuStats is activated while a CWC is being entered (that is, the pound (#) sign is not yet dialed), the CWC display is overwritten. The CWC must be reentered.

Change skills: An agent changing skills automatically cancels VuStats. Display of the new skills overwrites the VuStats display. When the agent reactivates VuStats, the VuStats display overwrites the new skills display.

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 agent's previous split or skill to the new split or skill. Therefore if you must frequently move agents between splits/skills, do not associate agents' VuStats buttons with a specific split or skill. Instead, associate the VuStats button with the agent format (without an ID) on each agent's phone and use a split or skill reference to view the agent's split or skill.

EAS-PHD: When you have EAS-PHD enabled, VuStats can provide statistical data for all twenty skills. However, agent statistics by skill (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 cancelled. The Integrated Directory display overwrites the VuStats display and the VuStats button extinguishes. When VuStats is reactivated, the VuStats display overwrites the Integrated Directory display.

Queue-Status Indications: The queue-status button display automatically cancels VuStats. When VuStats is reactivated, the VuStats display overwrites the queue-status display.

Reason Codes: Using certain VuStats data types, you can report 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; when the reason codes prompt is removed, the VuStats display reappears.

Service Observing: On telephones with a 1-line display, the Service Observing button display automatically cancels VuStats. When VuStats is reactivated, the VuStats display overwrites the Service Observing display.

VuStats display interactions and modifications for IP telephones

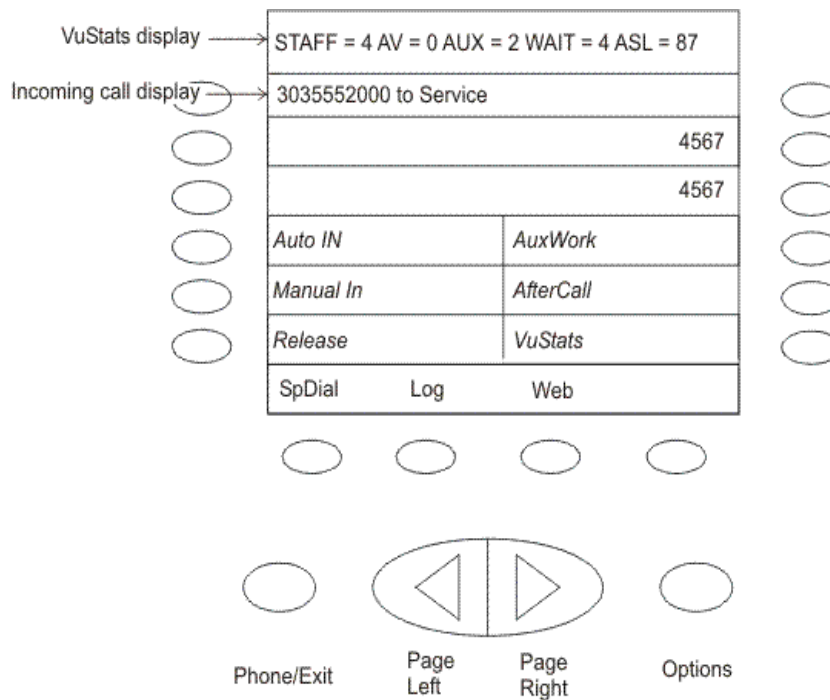
If you have Avaya Communication Manager Release 3.1.2 (load 632) or later and are using a 4610, 4620, 4621, 4622, or 4625 type IP telephone:

- For incoming calls, the incoming call is seen automatically in 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) still function in the same way as they did prior to Release 3.1.2 (load 632).

ACD call center features

The VuStats and incoming call information displays on these types of telephones are depicted in the following diagram of a 4622 telephone display.



Some situations could result in VuStats not being displayed, even though the **VuStats** button is highlighted for active VuStats mode. With certain VuStats formats, the VuStats display line does not display 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, VuStats needs to be deactivated to see the Caller-Info (collected digits) display.

ACD Call Center screens

This section defines the screens you use to administer the ACD Call Center features. This section describes the following administrative screens:

- [Agent LoginID screen](#) on page 325
- [Best Service Routing Application Plan](#) on page 331
- [BCMS/VuStats Login ID screen](#) on page 333
- [Call Classification screen](#) on page 336
- [Call Vector screen](#) on page 341
- [Duplicate Vector screen](#) on page 344
- [Feature-Related System Parameters screen](#) on page 345
- [Holiday Table screen](#) on page 355
- [Hunt Group screen](#) on page 357
- [Reason Code Names screen](#) on page 359
- [Service Hours Table screen](#) on page 360
- [Vector Directory Number screen](#) on page 361
- [Vector Routing Table screen](#) on page 368
- [VuStats Display Format screen](#) on page 370

Agent LoginID screen

Use this screen to administer agent login IDs for the Expert Agent Selection (EAS) feature.

This section includes the following topics:

- [Agent LoginID administration commands](#) on page 326
- [Agent LoginID field descriptions](#) on page 326
- [List Agent-LoginID field descriptions](#) on page 331

Agent LoginID administration commands

Use the following commands to administer the Agent LoginID screen.

Action	Object	Qualifier ¹
add	agent-loginid	xxxxx (extension) or next
change	agent-loginid	xxxxx (extension)
display	agent-loginid	xxxxx (extension) [print or schedule]
duplicate	agent-loginid	xxxxx (extension) start xxxxx (starting extension number) count x
remove	agent-loginid	xxxxx (extension)
list	agent-loginid	[staffed unstaffed [name x][aas y/n]]

1. Brackets [] indicate the qualifier is optional. Enter the text depicted inside single quotes (' ') exactly as shown or enter an abbreviated screen of the word.

Agent LoginID field descriptions

Make assignments as required for the following fields on the screen:

AAS?: Enter y if this extension is used as a port for an Auto Available Split/Skill. Default is n.



Important:

Entering y in the AAS field clears the password and requires execution of the **remove agent-loginid** command. To set AAS to n, remove this logical agent and add it again.

This option is intended for communication server adjunct equipment ports only, not human agents.

ACW Agent Considered Idle: Enter y to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter n to exclude ACW agents from the queue. Valid entries are system (default), n, and y.

AUDIX?: Enter y if this extension is used as a port for AUDIX. Default is n.

Note:

The AAS and AUDIX fields cannot both be y.

AUDIX Name for Messaging: Do one of the following actions:

- Enter the name of the messaging system used for LWC Reception

- Enter the name of the messaging system that provides coverage for this Agent LoginID
- Leave blank (default)

Auto Answer: When using EAS, the agent's auto answer setting applies to the station where the agent logs in. If the auto answer setting for that station is different, the agent's setting overrides the station's setting.

The following entries are valid:

Valid entries	Usage
all	Immediately sends all ACD and non ACD calls to the agent. The station is also given a single ring while a non-ACD call is connected. You can use the ringer-off button to prevent the ring when the feature-related system parameter, Allow Ringer-off with Auto-Answer is set to y .
acd	Only ACD split /skill calls and direct agent calls go to auto answer. If this field is <i>acd</i> , non ACD calls terminated to the agent ring audibly.
none	All calls terminated to this agent receive an audible ringing treatment. This is the default.
station	Auto answer for the agent is controlled by the auto answer field on the Station screen.

Aux Work Reason Code Type: Determines how agents enter reason codes when entering AUX work.

Valid entries	Usage
system	Settings assigned on the Feature Related System Parameters screen apply. This is the default.
none	You do not want an agent to enter a reason code when entering AUX work.
requested	You want an agent to enter a reason code when entering AUX mode but do not want to force the agent to do so. To enter this value, the reason codes and EAS on the System-Parameters Customer-Options screen must be set to y .
forced	You want to force an agent to enter a reason code when entering AUX mode. To enter this value, the Reason Codes and EAS on the System-Parameters Customer-Options screen must be set to y .

Call Handling Preference: Options are skill-level (default), greatest-need, and percent-allocation. When calls are in queue and an agent becomes available, skill-level delivers the oldest, highest priority call waiting for the agent's highest level skill. Greatest-need delivers the oldest, highest priority call waiting for any of the agent's skills. Percent allocation delivers a call from the skill that will otherwise deviate most from its administered allocation. Percent-allocation is available only with Avaya Business Advocate software. For more information, see *Avaya Business Advocate User Guide*.

COR: Enter the Class of Restriction for the agent. Valid entries are 0-995. Default is 1.

Coverage Path: Enter the number of the Coverage Path used by calls to the LoginID. Valid entries are a path number between 1-999, time of day table t1-t999 or blank (default). This is used when the agent is logged out, busy, or does not answer.

Direct Agent Calls First (not shown): This field replaces the Service Objective field when percent-allocation is entered in the Call Handling Preference field. Enter y if you want direct agent calls to override the percent-allocation call selection method and be delivered before other ACD calls. Enter n if you want direct agent calls to be treated like other ACD calls. For more information, see *Avaya Business Advocate User Guide*.

Direct Agent Skill: Enter the number of the skill used to handle Direct Agent calls. Valid entries are 1-2000, or blank (default).

Forced Agent Logout Time: Enables the Forced Agent Logout by Clock Time feature by administering a time of day to automatically log out agents using an hour and minute field. Valid entries for the hour field are 01-23. Valid entries for the minute field are 00, 15, 30, and 45. The default is blank (not administered).

Examples: 15:00, 18:15, 20:30, 23:45.

For more information about this feature, see [Forced Agent Logout by Clock Time](#) on page 165.

Local Call Preference: You can administer Local Preference Distribution to handle agent-surplus conditions, call-surplus conditions, or both. Use this field to administer call-surplus conditions. To set up an algorithm for agent-surplus conditions, set the Local Agent Preference field on the Hunt Group screen.

Valid entries are y or n. The default is n. You can set this field to y only if the Call Center Release field is set to 3.0 or later and the Multiple Locations customer option is active.

For more information, see [Location Preference Distribution](#) on page 189.

Login ID: Display-only field. Contains the identifier for the Logical Agent as entered on the command line.

LoginID for ISDN/SIP Display: Enter y if the Agent LoginID CPN and Name field is to be included in ISDN and SIP messaging over network facilities. If set to n (the default), the physical station extension CPN and Name is sent. The Send Name on the ISDN Trunk Group screen prevents sending the calling party name and number if set to n and may prevent sending it if set to r (restricted).

Logout Reason Code Type: Determines how agents enter reason codes.

Valid entries	Usage
system	Settings assigned on the Feature Related System Parameters screen apply. This is the default.
none	You do not want an agent to enter a reason code when logging out.
requested	You want an agent to enter a reason code when logging out but do not want to force the agent to do so. To enter this value, the reason codes and EAS on the System-Parameters Customer-Options screen must be set to y.
forced	You want to force an agent to enter a reason code when logging out. To enter this value, the Reason Codes and EAS on the System-Parameters Customer-Options screen must be set to y.

LWC Reception: Valid entries are audix, msa-spe (default), and none.

Maximum time agent in ACW before logout (sec): This field is used for setting a maximum time the agent can be in ACW on a per agent basis.

Valid entries are:

- system - This is the default. Settings assigned on the Feature Related System Parameters screen apply.
- none - ACW timeout does not apply to this agent.
- 30-9999 sec - Indicates a specific timeout period. This setting will take precedence over the system setting for maximum time in ACW.

For more information, see [Tips for administering Forced Agent Logout from ACW mode](#) on page 163.

Messaging Server Name for Messaging: Do one of the following actions:

- Enter the name of the Messaging Server used for LWC Reception
- Enter the name of the Messaging Server that provides coverage for this Agent LoginID
- Leave blank (default)

MIA Across Skills: Enter y to remove an agent from the MIA queue for all the splits/skills/hunt groups that he or she is available in when the agent answers a call from any of his or her splits/skills/hunt groups.

Valid entries are system, n, and y. The default is system.

Name: Enter up to a 27-character string naming the agent. Any alpha-numeric character is valid. Default is blank.

PA (Percent Allocation): If the call handling preference is percent-allocation, you must enter a percentage for each of the agent's skills. Enter a number between 1-100 for each skill. Your entries for all the agent's skills together must add up to 100%. Do not use target allocations for reserve skills. Percent Allocation is available as part of the Avaya Business Advocate software.

Password: Only displayed if both the AAS and AUDIX fields are n. Enter up to nine digits as the password the Agent must enter upon login. Valid entries are the digits 0 through 9. Enter the minimum number of digits in this field specified by the Minimum Agent-LoginID Password Length field on the Feature-Related System Parameters screen. Default is blank.

Password (enter again): Only displayed if both the AAS and AUDIX fields are n. Reenter the same password exactly as it was entered in the Password field. Default is blank.

Note:

Values entered in this field are not echoed to the screen.

Port Extension (not shown): Only displayed if either the AAS or AUDIX field is y. Enter the assigned extension for the AAS or AUDIX port. This extension cannot be a VDN or an Agent LoginID. Default is blank.

RL (Reserve Level): Enter any reserve levels assigned to this agent with the Service Level Supervisor feature. You can assign a reserve level of 1 or 2. When this skill reaches the corresponding EWT threshold set on the Hunt Group screen, the agent automatically logs into the skill. The agent takes calls until the skill's EWT drops below the preassigned overload threshold. Service Level Supervisor is available as part of the Avaya Business Advocate software.

Security Code: Enter the 4-digit security code (password) for the Demand Print messages feature. This field can be blank (default).

Service Objective: This field is displayed only when the call handling preference is greatest-need or skill-level. Enter y or n in this field. Service Objective is administered on the Hunt Group screen and the agent LoginID screen. The communication server selects calls for agents according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and the administered service objective for the skill. Service Objective is a feature that is part of the Avaya Business Advocate software.

SN (Skill Number): Enter the Skill Hunt Group(s) that this agent handles. The same skill may not be entered twice. Consider the following options:

- If EAS-PHD is not optioned, enter up to four skills.
- If EAS-PHD is optioned, enter up to 20 or 60 skills depending on the platform.



Important:

Assigning a large number of skills to agents can potentially impact system performance. Review system designs with the ATAC when a significant number of agents have greater than 20 skills per agent.

SL (Skill Level): Enter a skill level for each of an agent's assigned skills. If EAS-PHD is not optioned, 2 priority levels are available. If EAS-PHD is optioned, 16 priority levels are available.

TN: Enter the Tenant Partition number. Valid entries are 1-100. Default is 1.

Note:

Values entered in this field are not echoed to the screen.

List Agent-LoginID field descriptions

The following fields appear only on the List Agent LoginID screen. All these fields are display-only:

AAS/AUD: This field is y if the login ID is assigned as an auto-available split or skill or an AUDIX port.

Agt Pref: Shows the call handling preference assigned to this loginID.

Dir Agt: Shows the entry in the Direct Agent Skill field.

Extn: The physical extension at which this agent is currently logged in. This field is blank if the agent is not logged in.

Skl/Lv: Shows the agent's assigned skills and the skill level for each one.

SO: Shows the entry in the Service Objective field. If you are not using Service Objective, this field is blank.

Best Service Routing Application Plan

Use this screen to identify the remote locations used in each Best Service Routing (BSR) application.

Note:

For an explanation of BSR application plans see [Best Service Routing](#) on page 110.

This section includes the following topics:

- [BSR administration commands](#) on page 332
- [BCMS/VuStats field descriptions](#) on page 334

BSR administration commands

Use the following administration commands to administer the BSR screen.

Action	Object	Qualifier ¹
add	best-service-routing	xxx (application number) or 'next'
change	best-service-routing	xxx (application number)
display	best-service-routing	xxx (application number)
remove	best-service-routing	xxx (application number)
list ²	best-service-routing	

1. Brackets [] indicate the qualifier is optional. Single quotes (' ') indicate that you must enter the text inside the quote exactly as shown. You can also enter an abbreviated screen of the word.

2. The command `List best-service-routing` displays the List Best Services Routing Applications screen. This is a display-only screen.

Note:

If you execute `remove best-service-routing` against a BSR application table with no name assigned, the system will generate error "Identifier not assigned." To resolve the situation, give the table a name and then you will be able to remove it.

BSR application field descriptions

Make assignments as required for the following fields on the screen:

Interflow VDN: Enter the routing number including the dial access code your communication server uses to access the Interflow VDN at the remote location. Valid entries can be up to 16 characters long and contain the following characters:

- 0-9
- * or #
- p (pause)
- w/W (wait)
- m (mark)
- s (suppress)

Location Name: Give each location a name with up to 15 characters.

Lock: When set to y, this field provides extra security by not sending the information over the CMS. When set to n, the system sends the information.

Maximum Suppression Time: Enter the maximum poll suppression time in seconds from 0-60. This value applies when a subsequent Call Vector **consider** command replaces a location as the best. For example, if the poll suppression time is set to 30 seconds, the remote location polling is suppressed for up to 30 seconds if the Expected Wait Time (EWT) is far from being the best.

Name: Give the application plan a name with up to 15 characters.

Net Redir: When set to y, this field enables network call redirection. When set to n, network call redirection is not enabled. Default is n.

Num: Enter the location number. Location numbers are identifiers, and therefore need not be in sequential order. For example, you can assign locations with the identifiers 1, 3, 14 and 89 to one application plan.

Number: This display-only field shows the identifying number of the plan you are working with. Numbered from 1 to 255.

Status Poll VDN: Enter the routing number including the dial access code your communication server uses to access the Status Poll VDN at the remote location. Valid entries can be up to 16 characters long and contain the following characters:

- 0-9
- * or #
- p (pause)
- w/W (wait)
- m (mark)
- s (suppress)

Switch Node: This is an optional field. If you are using Universal Call ID, enter the UCID Network Node ID for each communication server. Valid Network Node IDs range from 1-32,767.

BCMS/VuStats Login ID screen

This section includes the following topics:

- [About the BCMS/VuStats Login ID screen](#) on page 334
- [BCMS/VuStats login ID administration commands](#) on page 334
- [BCMS/VuStats field descriptions](#) on page 334

- [Implementation notes](#) on page 335

About the BCMS/VuStats Login ID screen

You must administer the BCMS/VU-Stats Login IDs screen, if:

- The EAS feature is not optioned.
- The BCMS/VuStats Login ID field on the Feature-Related System Parameters screen is set to y. This field is located in the Call Management System section of the screen.

You do not have to enter agent names to administer the screen. If you choose not to associate names with login IDs, the data you receive from BCMS or VuStats defaults to:

ID xxxxxxxxx

where xxxxxxxxx is an agent login ID.



Important:

Only agents using administered login IDs can successfully log in to a split or skill that is measured by BCMS.

BCMS/VuStats login ID administration commands

Use the following administration commands to administer the BCMS/VuStats Login ID screen.

Action	Object	Qualifier ¹
add	bcms/vustats login IDs	
change	bcms/vustats login IDs	[login ID]
display	bcms/vustats login IDs	[login ID]
list	bcms/vustats login IDs	[login ID] count X

1. Brackets [] indicate the qualifier is optional.

BCMS/VuStats field descriptions

Make assignments as required for the following fields on the screen:

Login ID: This field has the following requirements:

- Make the login ID the same length as the ACD Login Identification Length field that is specified on the Feature Related System Parameters screen. If the length does not match, the system displays an error message and places the cursor at the field that is incorrect.
- If you change the administered login length to a different value, you will change the allowed length for all other IDs entered on this screen. If you do not adjust the login lengths, agents will not be able to log on. Change the ACD login identification length to fit the existing logins or change the logins to match the ACD login identification length.
- If you enter a duplicate login ID, the system displays an error message and places the cursor at the duplicated field.

Name: A name associated with the login ID (optional).

Implementation notes

The screen displays only two pages at a time. This is equivalent to 64 login IDs. If you are adding login IDs, you can enter two pages, and reissue the command to fill an additional two pages, repeating as necessary. When you change or display login IDs, the system displays two pages of login IDs beginning with the ID you specify. If you do not specify a login ID, the display begins with the first login ID. The list command lists all login IDs and may run to 63 pages.

Call Classification screen

This section includes the following topics:

- [ASAI SCC operation](#) on page 336
- [How the call classifier is inserted](#) on page 339
- [How the call attempt is rejected](#) on page 339
- [About the SIT Treatment for Call Classification screen](#) on page 340
- [SIT Treatment for Call Classification administration commands](#) on page 340
- [SIT Treatment for Call Classification field descriptions](#) on page 340

ASAI SCC operation

This section includes the following topics:

- [Description of ASAI SCC](#) on page 336
- [How the SCC software worked prior to Release 2.2](#) on page 337
- [How SCC software works with Release 2.2](#) on page 337
- [How SCC software works with Release 3.0 and later](#) on page 337

Description of ASAI SCC

When an ASAI-connected adjunct places a Third Party Make Call, the adjunct can invoke a Switched Classified Call (SCC). The adjunct uses an ASAI REGISTER message with the Service Circuit parameter set to *classifier*. The ASAI REGISTER message can also include:

- A request to perform Answering Machine Detection (AMD) on the call by including the Answer Machine parameter.
- A ringback timeout value specified by the number of rings that is converted to seconds. The number of rings can be 2 to 15 ring cycles using 6 seconds per ring cycle. The ringback timeout value is used by the classifier to determine how long to wait for a far-end answer after ringing is detected. If this timeout value is not provided in the request message, a default of 60 seconds for 10 rings is used. No answer timeout drops the classifier and ends the call with a cause CS3/19 - No Answer.

SCCs use call-classification resources to place outgoing calls as requested by adjunct predictive dialing and Outgoing Call Management applications. Call classifiers detect:

- Inband call progress signals including network SIT tones
- Live answers

- Modems
- Answers made by an answering machine

How the SCC software worked prior to Release 2.2

Releases prior to 2.2 did not support Switched Classified Calls (SCCs) in S8300 server or S8700 server configurations with outgoing trunks terminated on H.248 Media Gateways (MGs). The SCC software only looked for TN call classifier resources that reside in the MCC, SCC, G600 or G650 Port Network Gateways (PNGs) and did not select classifier resources available in the G700 or G350 H.248 MGs.

In addition, releases prior to 2.2 provided limited classification capability for mixed PNG and MG configurations. This was because outgoing trunks used for SCCs had to be terminated on PNGs. When a classifier resource was obtained, it had to be on the same gateway as the SCC-selected outgoing trunk for IP-connected PNGs.

How SCC software works with Release 2.2

The SCC software is enhanced to select classifier resources on H.248 Media Gateways (MGs) for launching outgoing switch-classified calls. As with IP-connected Port Network Gateways (PNGs), the classifier resource is selected from the same H.248 gateway that the selected outgoing trunk is terminated on. If there are no classifier resources available on that gateway, the switch classifier call is rejected. This is a change from the operation with all TDM or multi-connect PNG configurations where a classifier board can be selected from any PNG.

In a mixed PNG or MG environment, if the selected outgoing trunk is on:

- A TDM multi-connected PNG, the software searches for an available classifier board on the outgoing trunk PNG. If not found, other TDM multi-connected PNGs are searched for an available classifier board.
- An H.248 Media Gateway or IP-connected port network gateway, the classifier resource must be available on the outgoing trunk gateway or the SCC is rejected.

For more detailed information, see *Avaya Communication Manager ASAI Technical Reference*.

How SCC software works with Release 3.0 and later

The Release 3.0 SCC software is enhanced with a hunting algorithm. If the software cannot find an available outgoing trunk member and classifier on the same gateway on the first attempt, the algorithm tries to select other H.248 Media Gateways (MGs) and Port Network Gateways (PNGs). This algorithm also works with IP-connected PNGs and in a mixed PNG and MG configuration.

How the hunting algorithm works

When an SCC has been requested, the algorithm searches the members of the outgoing trunk group for an available trunk on a PNG or MG.

If the trunk member terminates on an H.248 MG: The algorithm checks the gateway for an available classifier or tone detector resource:

- If a classifier or tone detector resource is available, the SCC is launched.
- If a classifier resource is not available, the algorithm checks the outgoing trunk group for an available member on another MG or PNG with an available classifier or tone detector resource.

If the trunk member terminates on a PNG: The algorithm checks the gateway for an available classifier board:

- If a classifier port network board exists in the PNG, the SCC is launched.
- If a classifier board is not available on that PNG, and the PNG is TDM multi-connected with other PNGs, an available classifier board is used on another PNG. The SCC is then launched.
- If the PNG is IP-connected, or there are no available classifier boards among the TDM multi-connected PNGs, the trunk group is checked again for an available member on another PNG or MG with an available classifier or tone detector resource.

If no available classifier is found : If the algorithm does not find a trunk member on either a PNG or H.248 MG with an available classifier or tone detector resource after searching through the entire trunk group, the algorithm uses the first available trunk member found for launching the call. If there is not an available classifier board on that gateway when the classifier is to be connected, the SCC fails.

Note:

The hunting algorithm is applicable only when the outgoing trunk group is distributed across gateways so that a trunk member and classifier resource match can be made on another gateway after the first attempt fails.

How the call classifier is inserted

The **Call Classification After Answer Supervision?** field on the Feature Related System Parameters screen determines when the classifier is inserted in the connection.

If set to	Then the classifier is
n	Always inserted as soon as all digits have been outpulsed or the SETUP message has been sent. This is true even with ISDN trunks. D-Channel call progress indications usually take precedence over classifier-detected progress tones. This setting is the default.
y	Inserted after the answer has been received for the call if Answering Machine Detection is requested on the call. Otherwise, the classifier is not inserted in the connection with this setting. The network needs to reliably determine answer supervision through ISDN-type facilities.

How the call attempt is rejected

If an outgoing trunk could not be selected for the call, when the call attempt is rejected depends on how the **Call Classification After Answer Supervision?** field on the Feature Related System Parameters screen is set.

If set to	Then the call attempt is
n	<ul style="list-style-type: none"> ● Rejected with cause CS3/20 - no trunks ● Rejected with cause CS3/21 - no classifiers - if a classifier could not be obtained
y	<ul style="list-style-type: none"> ● Rejected with cause CS3/20 - no trunks ● Rejected with cause CS3/21 - no classifiers - if Answering Machine Detection is requested for the call and a classifier could not be connected after answer supervision has been received

About the SIT Treatment for Call Classification screen

Use this screen to specify the treatment of Special Information Tones (SITs) used for Outbound Call Management type calls with USA tone characteristics. The port network TN744 Call Classifier circuit pack ports or H.248 Media Gateway internal tone detector resources in classified mode are used to detect SITs. The classifiers are capable of detecting the following six SITs:

- SIT Ineffective Other
- SIT Intercept
- SIT No Circuit
- SIT Reorder
- SIT Vacant Code
- SIT Unknown
- AMD Treatment

SIT Treatment for Call Classification administration commands

Use the following administration commands to administer the SIT Treatment For Call Classification screen. In some cases, just the most commonly used commands are shown. See *Administrator Guide for Avaya Communication Manager*, for a listing of all administration commands, the command structure, and the use of abbreviated command words when entering a command.

Action	Object	Qualifier ¹
change	sit-treatment	-
display	sit-treatment	['print' or 'schedule']

1. Brackets [] indicate the qualifier is optional. Single quotes (' ') indicate the text inside the quote must be entered exactly as shown or an abbreviated screen of the word can be entered.

SIT Treatment for Call Classification field descriptions

Make assignments as required for the following fields on the screen:

In the field following each type of SIT, enter answered to specify that the call is classified as answered, and is therefore sent to an agent; or enter dropped to specify that the call is classified as not answered, and is therefore not sent to an agent.

AMD (Answering Machine Detected): An ASAI adjunct can request AMD for a call. If Answering Machine is detected, one of two treatments is specified. Valid entries are dropped and answered. Default is dropped.

AMD Treatment has two separately administrable subfields. Talk Duration is for full seconds and Pause Duration is for fractions of a second, separated by a display-only decimal point. Talk Duration defaults to 2.0 seconds and allows a range from 0.1 seconds to 5.0 seconds in increments of 0.1 seconds. Pause duration defaults to 0.5 seconds and allows a range from 0.1 seconds to 2.0 seconds in increments of 0.1 seconds.

SIT Ineffective Other: Sample announcement following this SIT - *You are not required to dial a 1 when calling this number.* Valid entries are answered and dropped. Default is dropped.

SIT Intercept: Sample announcement following this SIT - *XXX-XXXX has been changed to YYY-YYYY, please make a note of it.* Valid entries are answered and dropped. Default is answered.

SIT No Circuit: Sample announcement following this SIT - *All circuits are busy, please try to call again later.* Valid entries are answered and dropped. Default is dropped.

SIT Reorder: Sample announcement following this SIT - *Your call did not go through, please hang up and dial again.* Valid entries are answered and dropped. Default is dropped.

SIT Unknown: A situation or condition that is unknown to the network is encountered. Valid entries are answered and dropped. Default is dropped.

SIT Vacant Code: Sample announcement following this SIT - *Your call cannot be completed as dialed, please check the number and dial again.* Valid entries are answered and dropped. Default is dropped.

Call Vector screen

This screen programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN).

This section includes the following topics:

- [Call Vector administration commands](#) on page 342
- [Field descriptions of Call Vector screen](#) on page 342

Call Vector administration commands

Use the following commands to administer the Call Vector screen.

Action	Object	Qualifier ¹
change ²	vector	1-MAX
display	vector	1-MAX ['print' or 'schedule']
list	vector	1-MAX ['count' 1-MAX] ['print' or 'schedule']

1. Brackets [] indicate the qualifier is optional. Single quotes (' ') indicate the text inside the quote must be entered exactly as shown or an abbreviated screen of the word may be entered. MAX is the maximum number available in your system configuration.
2. Do not change a call vector while it is processing a call. It is recommended to add a new vector with the Call Vector screen, and then use the Vector Directory Number screen to point an existing VDN to the new vector.

Field descriptions of Call Vector screen

Make assignments as required for the following fields on the screen:

3.0 Enhanced? Display-only field. This value appears only if Vectoring (3.0 Enhanced) is set to y on the System Parameter Customer Options screen.

ANI/II-Digits? Display-only field. Indicates whether you can use ANI and II-Digits Vector Routing Commands. ANI/II-Digits Routing requires that G3V4 Enhanced be set to y.

ASAI Routing? Display-only field. Indicates whether or not the **CallVisor Adjunct/Switch Applications Interface (ASAI) Routing** option is enabled on the System-Parameters Customer-Options screen. Valid values are y or n.

Attendant Vectoring: Appears only when Attendant Vectoring is optioned. Valid entries are y and n (default). The Attendant Vectoring field defaults to n and changes are allowed to the field. If Basic Vectoring and Vector Prompting are both set to n, then the Attendant Vectoring field defaults to y and no changes are allowed to the field.

To associate VDNs and vectors for attendant vectoring, a field has been added to both the VDN and the call vectoring forms to indicate attendant vectoring. When attendant vectoring is indicated for VDNs and vectors, all call center-associated fields (such as Skills and BSR) are removed.

Basic? Display-only field. Indicates whether the *Vectoring (Basic)* option is enabled on the System-Parameters Customer-Options screen. Valid values are y or n.

BSR? A y in this display-only field indicates that the Vectoring (Best Service Routing) option is enabled on the System-Parameters Customer-Options screen. Thus, you can use BSR commands and command elements in your vectors. An n indicates that the BSR option is not enabled.

CINFO? Display-only field. Indicates whether the **Vectoring (CINFO)** option is enabled on the System-Parameters Customer-Options screen. Valid values are y or n.

EAS? Display-only field. Indicates whether the **Expert Agent Selection (EAS)** option is enabled on the System-Parameters Customer-Options screen. Valid values are y or n.

When Expert Agent Selection (EAS) is enabled, the help messages and error messages associated with this screen will reflect a terminology change from Split to Skill. In addition, the vector commands entered also will be affected by this terminology change (for example, **check backup split** becomes **check backup skill** when EAS is enabled).

G3V4 Adv Route? Display-only field. Indicates whether you can use the G3V4 Advanced Vector Routing commands.

G3V4 Enhanced? Display-only field. Indicates whether you can use G3V4 Enhanced Vector Routing commands and features.

Holidays? Display-only field. The value of y appears if Holiday Vectoring is set to y on the system Parameters Customer Options screen.

LAI? Display-only field. Indicates whether Look-Ahead Interflow is enabled.

Lines 01 through 32: Enter vector commands as required (up to the maximum allowed in your configuration). Valid entries are adjunct, announcement, busy, check, collect, consider, converse-on, disconnect, goto, messaging, queue-to, reply-best, route-to, stop, wait-time. Default is blank.

Lock? This field controls access to the vector from the CMS or Visual Vectors. Valid entries are y and n (default). Enter n to give CMS and Visual Vectors users the ability to administer this vector from these client programs. Enter y if you do not want this vector to be accessible to these client programs. Locked vectors can only be displayed and administered through the SAT or a terminal emulator.

Note:

Always lock vectors that contain secure information (for example, access codes).

Meet-me Conf? Appears when the meet-me conference feature is enabled. Valid entries are y and n. For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

Multimedia? Indicates whether the vector should receive early answer treatment for multimedia calls. This only applies if Multimedia Call Handling is enabled. If you expect this vector to receive multimedia calls, set this field to y. Valid values are y or n (default). If this value is y, the call is considered to be answered at the start of vector processing, and billing for the call starts at that time.

Name: Enter up to 27 alphanumeric characters to represent the vector name. This is an optional field. Default is blank.

Number: Display-only field when the screen is accessed using a change or display administration command. Enter a vector number when completing a paper screen.

Prompting? Display-only field. Indicates whether the **Vectoring (Prompting)** option is enabled on the System-Parameters Customer-Options screen. Valid values are y or n.

Variables? Display-only field. The value appears only if **Vectoring (Variables)** is set to y on the System-Parameters Customer-Options screen.

Duplicate Vector screen

Use this screen to duplicate vectors from an existing vector and edit the duplicate vectors to create vectors that are similar to the existing vector. You can use this functionality to configure one vector as a template that can be reused when creating similar vectors.

Note:

See the *Avaya Call Center Release 4.0 Call Vectoring and Expert Agent Selection (EAS) Guide* for details about duplicating vectors.

The following fields are populated in the Duplicate Vector screen:

Count: Displays the number of duplicates created from the master vector.

More? Displays * if there is at least one more VDN assigned to the same vector. For example, if 5555 displays in the **VDN Assigned to** field and an asterick (*) displays in the **More?** field, this means that the master vector you selected is already assigned to VDN 5555 as well as to other VDNs.

Name: Displays the vector name if any of the vectors have an assigned name. The duplicated vectors can already be assigned names but they must be vectors that contain no steps. You can edit the vector name for any of the duplicated vectors.

If you specify a used or out of range vector number, an error message is displayed. You cannot move to the next field until you enter an unused number.

VDN Assigned to: Displays the VDN if a VDN was assigned to the master vector.

Vector: Displays the vector number.

Feature-Related System Parameters screen

Call center system parameters are listed on two pages of the Feature-Related System Parameters screen.

For a description of the rest of the Feature-Related System Parameters screens, see *Administrator Guide for Avaya Communication Manager*.

This section includes the following topics:

- [System Parameter administration commands](#) on page 345
- [Agent and Call Selection screen field descriptions](#) on page 345
- [Call Center Miscellaneous field descriptions](#) on page 347
- [Call Center System Parameters field descriptions](#) on page 348
- [EAS page field descriptions](#) on page 351
- [Forced Agent Logout Parameters field descriptions](#) on page 354

System Parameter administration commands

Use the following administration commands to administer the call center System Parameters.

Action	Object	Qualifier
change	system-parameters	features
display	system-parameters	features
list usage	forced-agent-logout	none

Agent and Call Selection screen field descriptions

ACD Login Identification Length: Enter the number of digits (0 through 9) for an ACD Agent Login ID if **Expert Agent Selection (EAS)** on the System-Parameters Customer-Options screen is n. Default is 0. If **BCMS/VuStats Login IDs** is y, the ACD login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of digits in the agent's login ID. For CMS, this field cannot be 0.

ACW Agents Considered Idle? Enter **y** (default) to have agents who are in After Call Work included in the Most-Idle Agent queue. Enter **n** to exclude agents from the queue while they are in ACW and return them to the bottom of the queue when they become available.

Reporting Adjunct Release: Specifies the release of the CMS reporting adjunct (**CMS (appl mis)**) and the CCR reporting adjunct (**CCR (appl ccr)**) used with the system. Default is blank.

Auto Reserve Agents: Enter **all** if an agent is to be intentionally left idle in a skill if her work time in the skill has exceeded her target allocation for the skill. Enter **secondary-only** to activate this feature only for the agent's non-primary skills (skill levels 2 through 16).

Aux Work Reason Code Type: Enter **none** if you do not want an agent to enter a reason code when entering AUX work. Enter **requested** if you want an agent to enter a reason code when entering AUX mode but do not want to force the agent to do so. Enter **forced** to force an agent to enter a reason code when entering AUX mode. To enter **requested** or **forced**, the **Reason Codes** and **EAS** fields on the System-Parameters Customer-Option screen must be **y**.

BCMS/VuStats Abandon Call Timer (seconds): Enter **none** or 1-10 to specify the number of seconds for calls to be considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

BCMS/VuStats LoginIDs? This field is used to activate login IDs for use with BCMS and VuStats, and is displayed only if:

- EAS is not optioned for the system
- Either BCMS or VuStats is optioned for the system

If EAS is optioned for the system, this field is set to **y** and cannot be changed to **n**.

The default value is **n**. Set this field to **y** if you want to use BCMS or VuStats to monitor agent call activity. Starting with CM 4.0, the login ID can be up to 13 digits.

BCMS/VuStats Measurement Interval: You can enter **half-hour** or **hour** (default) for polling and reporting measurement data if the **BCMS (Basic)** and/or the **VuStats** on the System-Parameters Customer-Options screen is **y**. If neither of these features is optioned, and if you enter a value in the **BCMS Measurement Interval** field, the system displays the following error message:

```
<value> cannot be used; assign either BCMS or VuStats first
```

If you receive this message, see your Avaya representative to turn on **BCMS (Basic)** and/or **VuStats** on the System-Parameters Customer-Options screen.

There are a maximum of 25 time slots available for measurement intervals. If hour is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval may be changed at any time, but will not go into effect until the current interval completes.

Call Selection Measurement: Valid entries are **current-wait-time** (default) and **predicted-wait-time**. This field determines how the CM communication server selects a call for an agent when the agent becomes available and there are calls in queue. **Current Wait Time** selects the oldest call waiting for any of the agent's skills. Predicted Wait Time is a feature of the Avaya Business Advocate software.

Clear VuStats Shift Data: Enter **on-login** to clear shift data for an agent when the agent logs in. Enter **at-midnight** to clear shift data for all agents at midnight.

Logout Reason Code Type: Enter **none** if you do not want an agent to enter a reason code when logging out. Enter **requested** if you want an agent to enter a reason code when logging out but do not want to force the agent to do so. Enter **forced** to force an agent to enter a reason code when logging out. To enter **requested** or **forced**, the `Reason Codes` and `EAS` on the System-Parameters Customer-Option screen must be **y**.

MIA Across Splits or Skills? Enter **y** to remove an agent from the MIA queue for all the splits/skills that the agent is available in when the agent answers a call from any of his or her splits/skills. The default is **n**, which keeps the agent in the MIA queue for the other splits/skills when answering a call from one of the assigned splits/skills.

Service Level Supervisor Call Selection Override? Enter **y** (default) to enable an agent to receive a lower priority call from a skill in an over threshold state before receiving a higher priority call from a skill not in an over threshold state.

Validate BCMS/VuStats login IDs: Enter **n** to allow entry of any ACD login of the proper length. Enter **y** to allow entry only of login-IDs that have been entered on the BCMS Login-ID screen.

Call Center Miscellaneous field descriptions

Clear Callr-info: This field controls when the display of collected digits on the second information line of an agent's two-line display station set (Callmaster IV) is to be cleared. When set to **next-call** (default) the display is cleared when the next call is received. When set to **on-call-release**, the display is cleared when the current call is released by the agent or upon caller disconnect. When set to **leave-ACW**, the display is cleared upon call release except when the agent is to be put into ACW. Then the display is cleared when the agent leaves ACW.

Allow Ringer-off with Auto-Answer? This field provides a system option to allow the Ringer-off button assigned to auto-answer sets to apply to both non-ACD and ACD calls. When set to **n** (default), the Ringer-off button applies only to ACD calls received by the agent. Non-ACD calls delivered to the agent continue to apply a single alerting ring even when automatically connected with the auto-answer option set to **all**.

Reporting for PC Non-Predictive Calls? Activates improved integration with Proactive Contact Outbound Calling for non switch-classified outbound calling. Valid entries are **y** and **n**. The default is **n**.

For more information about this feature, see [Proactive Contact outbound calling improved reporting](#) on page 212.

PC Non-Predictive Reports Skill: Administers a skill hunt group used for reporting associated with Proactive Contact non-predictive switch-classified calls on a per system basis. Reports are generated as though the agent were in the ACD-OUT state. This field is available only if the **Report for PC Non-Predictive Calls?** field is set to **y**. Valid entries are skill numbers from 1-n. The default value is blank. Use 1-99 for S8300 or S8400 servers. Use 1-2000 for all other platforms.

For more information about using Proactive Contact, see [Proactive Contact outbound calling improved reporting](#) on page 212.

Call Center System Parameters field descriptions

Allow Two Observers in Same Call? Provides control of the Multiple Observers with Service Observing feature on a system-wide basis. The feature is enabled when this field is set to **y**. Valid entries are **y** and **n**.

Available Agent Adjustments for BSR: Allows the use of BSR available agent adjustments for available agents. The **Vectoring (Best Service Routing)** field must be **y** on the System-Parameters Customer-Options screen. Valid entries are **y** and **n**.

BSR Tie Strategy field: This field is available only if Vectoring (Best Service Routing) is set to **y** on the System Parameters Customer-Options screen.

Valid entries	Usage
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Converse First Data Delay/Second Data Delay: The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. This field is available only if **Vectoring (Basic)** on the System-Parameters Customer-Options screen is **y**. Valid entries are **0** to **9**.

Converse Signaling Tone/Pause: In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum timer settings for

the CONVERSANT or IR are 60 msec tone and 60 msec pause. This field is available only if **Vectoring (Basic)** and **DTMF** on the System-Parameters Customer-Options screen are **y**.

Valid entries	Usage
40 to 2550 (in increments of 10).	Values entered in the Tone/Pause fields are rounded up or down depending upon the type of circuit pack used to output the digits. <ul style="list-style-type: none"> ● TN742B or later suffix analog board — Tone and pause round up or down to the nearest 25 msec. For example, a 130 msec tone rounds down to 125 msec, a 70 msec pause rounds up to 75 msec for a total of 200 msec per tone. ● TN464F, TN767E or later suffix DS1 boards — Tone and pause round up to the nearest 20 msec. For example, a 130 msec tone rounds up to 140 msec, a 70 msec pause rounds up to 80 msec for a total of 220 msec per tone. If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers might stay in effect. To reset your timers to the system default, pull and reseal the circuit pack.
100	

Direct Agent Announcement Delay: Specifies the number of seconds the caller hears ringback before the Direct Agent Announcement is heard by the calling party. This field is available only if **Expert Agent Selection (EAS)** or **ASAI Link Core Capabilities** on the System-Parameters Customer-Options screen is **y**. Valid entries are **0** to **99** or blank.

Direct Agent Announcement Extension: Specifies the extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled: To enable this field, either no ACD or vectoring hunt groups might exist, or existing ACD or vectoring hunt groups must be skilled. This field is available only if **Expert Agent Selection (EAS)** on the System-Parameters Customer-Options screen is **y**. Valid entries are **n** and **y**.

Interflow-qpos EWT Threshold: Specifies the number of seconds for the EWT threshold. This threshold is part of the enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold are not interflowed, therefore saving CPU resources. This field is available only if **Lookahead Interflow (LAI)** field on the System-Parameters Customer-Options screen is **y**. Valid entries are **0** to **9** or blank.

Message Waiting Lamp Indicates Status For: This field is available only if **Expert Agent Selection (EAS)** on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a telephone, you need to indicate if the message is at the telephone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Minimum Agent-LoginID Password Length: Specifies the minimum number of digits that must be administered as an EAS Agent's LoginID password. This field is available only if **Expert Agent Selection (EAS)** on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
0 to 9	Entering a 0 or blank indicates no password is required.

Prompting Timeout (secs): Specifies the number of seconds before the **Collect Digits** command times out for callers using rotary dialing, or for interdigital or short entry timeout. This field is available only if **Vectoring (Prompting)** on the System-Parameters Customer-Options screen is **y**. Valid entries are 4 to 10. Default is 10.

Reverse Star/Pound Digit for Collect Step: The "*" is interpreted as a "caller end-of-dialing indicator and the "#" is an indicator to clear all digits previously entered by the caller for the current "collect" vector step. Enter **y** to reverse the star and pound digits by the **collect** vector step. If set to **y**, it does not affect any other DEFINITY vector step or other non-ACD DEFINITY feature (such as ARS) in that the "*" and "#" digit-processing is unchanged. Valid entries are **y** and **n**.

Service Observing: Warning Tone: Assigns a warning tone that is given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Conference Tone?** is set to **y**. This field is available only if **Service Observing (Basic)** on the System-Parameters Customer-Options screen is set to **y**. Valid entries are **y** and **n**.



CAUTION:

The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

or Conference Tone: Assigns a conference tone that is given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Warning Tone?** is set to **y**. This field is available only if **Service Observing (Basic)** on the System-Parameters Customer-Options screen is set to **y**. Valid entries are **y** and **n**.

Note:

When the Service Observing feature is deciding which observer to allow onto a conferenced call, the observer chosen could be dependent on the warning tone setting.

Service Observing Allowed with Exclusion: Allows Service Observing of a station with Exclusion active, either by Class Of Service or by manual activation of Exclusion. Default is **n**.

Valid entries	Usage
y	Enter y to allow Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
n	Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped. This is the default.

Store VDN Name in Station's Local Call Log: Specifies if the communication server sends a message telling the telephone to store the VDN name or the calling party's name in the station call log for any of the following telephones:

- 2420
- 4610
- 4620
- 4625

Valid entries	Description
y	The communication server sends a message telling the telephone to store the VDN name in the station call log.
n	The communication server sends a message telling the telephone to store the calling party's name in the station call log. This is the default setting.

EAS page field descriptions

Call Classification After Answer Supervision? For use with ASAI Outbound Call Management (OCM). Enter y to force the communication server to rely on the network to provide answer/busy/drop classification to the communication server. After the call has been answered, a call classifier can be added to perform answering machine, modem and voice answering detection. Enter n for standard operation. For standard operations, a classifier is switched on for every switch-classified outgoing call used for call-progress detection - after all digits have been outpulsed.

Converse First Data Delay/Second Data Delay: Only displays if **Vectoring (Basic)** on the System-Parameters Customer-Options screen is y. The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to a voice response unit (VRU) before the unit is ready. The delay commences when the VRU port answers the call. Enter the number of seconds (0 to 9) for the delay. Default is 0. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to the VRU. The Second Data Delay prevents the second set from being outpulsed before the VRU is ready. The delay commences when the first group of digits has been outpulsed. Enter the number of seconds (0 to 9) for the delay. Default is 2.

Converse Signaling Tone/Pause: Only displays if **Vectoring (Basic)** and **DTMF** on the System-Parameters Customer-Options screen are y. In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to a voice response unit (VRU). In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum setting for the Avaya IVR system is a 60 msec tone and a 60 msec pause.

Note:

Values entered in the **Tone/Pause** fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits.

- For TN742B (or later) suffix analog boards, tone and pause round up or down to the nearest 25 msec. For example a 130 msec tone rounds down to 125 msec, a 70 msec pause rounds up to 75 msec for a total of 200 msec per tone.
- For TN464F, TN767E (or later) suffix DS1 boards, tone and pause round up to the nearest 20 msec. For example a 130 msec tone rounds up to 140 msec, a 70 msec pause rounds up to 80 msec for a total of 220 msec per tone.

If a circuit pack has been used for end-to-end signalling to the VRU, and has then been used to send digits to a different destination, the VRU timers may stay in effect. To reset your timers to the system default, pull and reseal the circuit pack.

Delay: Only displays if **Expert Agent Selection (EAS)** or **ASAI** on the System-Parameters Customer-Options screen is y. Enter the number of seconds (0 to 99) the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension: Only displays if Expert Agent Selection (EAS) or ASAI on the System-Parameters Customer-Options screen is y. Enter a valid announcement extension (consistent with the dial plan).

Expert Agent Selection (EAS) Enabled: Only displays if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is y. Enter y to enable Expert Agent Selection. To enable this field, either no ACD or vectoring hunt groups may exist or, existing ACD or vectoring hunt groups must be skilled. Default is n.

Interflow-qpos EWT Threshold: Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources). Enter the number of seconds for this threshold. The default is 2 seconds.

Message Waiting Lamp Indicates Status For: Only displays if **Expert Agent Selection (EAS)** on the System-Parameters Customer-Options screen is y. If **Expert Agent Selection (EAS) Enabled** is y, you can enter either station or loginID, otherwise you can only enter station.

Minimum Agent-LoginID Password Length: Only displays if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is y. Enter the minimum number of digits that must be administered as an EAS Agent's LoginID password. Valid entries are 0 through 9 or blank. Entering a 0 or blank indicates no password is required. Default is blank.

Prompting Timeout (secs): Only displays if **Vectoring (Prompting)** on the System-Parameters Customer-Options screen is y. Enter the number of seconds, from 4 to 10 (default), before the Collect Digits command times out for callers using rotary dialing, or for interdigital or short entry timeout.

Reverse Star/Pound Digit For Collect Step? Setting this field to y reverses the normal handling of the asterisk (*) and pound (#) digits by the `collect` vector command. With the **Reverse Star/Pound Digit for Collect Step** set to y, the asterisk (*) digit is interpreted as a caller end-of-dialing indicator and the pound (#) digit is interpreted to clear all digits that were previously entered for the current `collect` vector step.

Note:

Any use of the asterisk (*) or pound (#) digits in the `converse` and `adjunct-route` vector commands is not changed by this field.

Note:

The use of Service Observing features may be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Send UCID to ASAI? Enter y to enables transmission of Universal Call ID (UCID) information to ASAI. Enter n (default) to prevent transmission of UCID information to ASAI.

Service Observing Allowed with Exclusion? Allows Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.

For more information, see [Service Observing with Exclusion](#) on page 260.

Service Observing: or Conference Tone: Service Observing (Basic) on the System-Parameters Customer-Options screen must be y before this field may be administered. Enter y to assign a warning tone to be given to the caller and the agent when a call that is being service observed is conferenced with another extension. Default is n.

Service Observing: Warning Tone: Service Observing (Basic) on the System-Parameters Customer-Options screen must be y before this field may be administered. Enter y to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. Default is n.

Forced Agent Logout Parameters field descriptions

Maximum Time Agent in ACW before Logout (secs): Sets the maximum time the agent can be in ACW on a *per system* basis. Valid entries are 30-9999 seconds. The default value is blank.

ACW Forced Logout Reason Code: Sets the reason code for the Forced Agent Logout by ACW mode feature. Valid entries are 0-9. The default value is 0.

For more information about this feature, see [Forced Agent Logout from ACW mode](#) on page 162.

Clock Time Forced Logout Reason Code: Sets the reason code for the Forced Agent Logout by Clock Time feature. Valid entries are 0-9. The default value is 0.

For more information about this feature, see [Forced Agent Logout by Clock Time](#) on page 165.

Holiday Table screen

This screen is used to establish the days/times for which Holiday Vectoring will apply. It is available to you only if Holiday Vectoring is set to y on the Customer Options screen.

This section includes the following topics:

- [Holiday table administration commands](#) on page 355
- [Holiday Table screen field descriptions](#) on page 355
- [Implementation notes for entering dates](#) on page 356
- [About administering a holiday table](#) on page 356

Holiday table administration commands

You can use the commands listed in the table below to administer Holiday Tables.

Action	Object	Qualifier
add	holiday-table	1 through 99, or next
change	holiday-table	1 through 99, or next
display	holiday-table	1 through 99, or next
remove	holiday-table	1 through 99, or next
list	holiday-table	none - all Holiday Tables will display

Holiday Table screen field descriptions

Make assignment as required for the following fields on the screen:

Description: Enter a description of the holiday defined on this line. Default is blank.

End: Enter the Month (1 through 12), the Day (optional, 1 through 31), Hour (optional, 00 through 23), and the Min (minute) (optional, 00 through 59) on which vector processing for this holiday should end. See the [Implementation notes for entering dates](#) on page 356 for additional information.

Name: Enter a 1 to 15-character alphanumeric table name. You may leave this field blank. Default is blank.

Number: This is the table number that you entered on the command line. It is a display only field.

Start: Enter the Month (1 through 12), the Day (optional, 1 through 31), Hour (optional, 00 through 23), and the Min (minute) (optional, 00 through 59) on which vector processing for this holiday should begin. See the [Implementation notes for entering dates](#) on page 356 for additional information.

Implementation notes for entering dates

Note:

When using a range of dates, the end date must be greater than the start date. Ranges must be within one calendar year. In the example above, two entries were made - one for each calendar year.

The Holiday Table Screen can be used for entering individual holiday or for holiday ranges. The following rules apply to entering dates on this screen:

- If a day is entered, the corresponding month must be entered.
- If a month is entered, the corresponding day must be entered.
- If an hour is entered, the corresponding minute must be entered.
- If a minute is entered, the corresponding hour must be entered.
- If an hour/minute is entered, the corresponding month/day must be entered.
- If a month/day is entered, the corresponding hour/minute is not required.
- If an end month/day is entered, the corresponding start month/day must be entered.
- If a start month/day is entered, the corresponding end month/day is not required.
- To enter an individual holiday, enter a start month/day, but do not enter an end month/day.
- To enter a holiday range, enter both a start month/day and an end month/day.
- The start m/d/h/m must be less than or equal to the end m/d/h/m.

About administering a holiday table

Consider the following when administering a holiday table:

- There is no validation that verifies the consistency among the 15 holidays in any table. If the same holiday is entered twice, the system stops checking with the first entry found.
- With holidays that are ranges of dates, the ranges could overlap. When a call is in vector processing, the holidays are checked from top to bottom on the table and the check stops if a match is found. Even though there might be multiple entries that would match, 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 1 - 30. If the month is January, March, May, July, August, October, or December, then the date can be 1 - 31. If the month is February, then a range of 1 - 29 is allowed.

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, the holiday tables must be readministered.

Hunt Group screen

This section includes the following topics:

- [About hunt groups](#) on page 357
- [About the Hunt Group screen](#) on page 357
- [Hunt Group screen field descriptions](#) on page 358
- [Hunt Group administration commands](#) on page 358

About hunt groups

The ACD software directs a high volume of calls to hunt groups that are designed as ACD splits. Each ACD split is created to receive calls for one or more services, such as business travel or billing. The services that are defined are based on the needs of the people who are calling the call center.

About the Hunt Group screen

The Hunt Group screen is used to define how the communication server sends calls to extensions in ACD and non-ACD environments.

Complete one Hunt Group screen for each split or skill.

Hunt Group screen field descriptions

Local Agent Preference: You can administer Local Preference Distribution to handle agent-surplus conditions, call-surplus conditions, or both. Use this field to administer agent-surplus conditions. To set up an algorithm for call-surplus conditions, set the Local Call Preference field on the Agent ID screen.

Valid entries are y or n. The default is n. You can set this field to y only if the Call Center Release field is set to 3.0 or later and the Multiple Locations customer option is active.

For more information, see [Location Preference Distribution](#) on page 189.

For a full discussion of the rest of the fields on the Hunt Group screen, see *Administrator Guide for Avaya Communication Manager*.

Queue Limit: This field is visible only if the **Queue?** field is set to **y**. The **Queue?** field is also located on the Hunt Group screen.

You can set the following values in the **Queue Limit** field.

Value	Usage
unlimited	The queue is allocated dynamically by the system. All calls to this hunt group are put in queue when an agent or a station is not available. This is the default value.
1-999	The system limits the number of calls that queue to this hunt group to the value specified.

Hunt Group administration commands

You can use the commands listed in the table below to administer the Hunt Group screen.

Action	Object	Qualifier
add	hunt-group	1-99 (si/csi), 1-999 (r), or next
change	hunt-group	1-99 (si/csi), 1-999 (r), or next
display	hunt-group	1-99 (si/csi), 1-999 (r), or next
remove	hunt-group	1-99 (si/csi), 1-999 (r), or next
list	hunt-group	1-99 (si/csi), 1-999 (r), or none (all hunt groups display)

Note:

Large Linux platforms (S8700, S8710, and S8500) can support 2000 hunt groups.

Reason Code Names screen

Use the Reason Code Names screen to assign names to reason codes. You can assign a different name to each reason code for Aux Work and for Logout.

This section includes the following topics:]

- [Administration commands for the Reason Code Names screen](#) on page 359
- [Reason Code Names field descriptions](#) on page 359

Administration commands for the Reason Code Names screen

Use the following administration commands to access the Reason Code Names screen.

Action	Object
display	reason-code-names
change	reason-code-names

Reason Code Names field descriptions

Make assignments as required for the following fields on the screen.

Aux Work: For each reason code enter the name to be associated with this reason code when the agent uses this reason code to enter Aux Work mode. Names can be up to 16 characters long. Default is blank.

Logout: For each reason code enter the name to be associated with this reason code when the agent uses this reason code to log out. Names can be up to 16 characters long. Default is blank.

If the trunk member terminates on an H.248 MG: The algorithm checks the gateway for an available classifier or tone detector resource:

- If a classifier or tone detector resource is available, the SCC is launched.
- If a classifier resource is not available, the algorithm checks the outgoing trunk group for an available member on another MG or PNG with an available classifier or tone detector resource.

If the trunk member terminates on a PNG: The algorithm checks the gateway for an available classifier board:

Service Hours Table screen

This screen is used to specify office service hours using up to 99 different tables. It is available to you only if Vectoring (Basic) is enabled.

This section includes the following topics:

- [Service Hours Table administration commands](#) on page 360
- [Service Hours Table field descriptions](#) on page 360

Service Hours Table administration commands

You can use the commands listed in the table below to administer Service Hours Table:

Action	Object	Qualifier
add	service-hours-table	1 through 99
change	service-hours-table	1 through 99
display	service-hours-table	1 through 99
remove	service-hours-table	1 through 99
list	service-hours-table	none
list usage	service-hours-table	none

Service Hours Table field descriptions

Number: Displays the table number that you entered on the command line.

Description: Provides a description for the table. You can enter a 1 to 27-character alphanumeric table name. The default is blank.

Example: Call-ahead Reservations

Use time adjustments from location: Indicates the location number on the Locations screen that specifies how time zone offset and daylight savings time rule time adjustments are to be performed.

Start and End: Defines the range of service hours for each day of the week. Always make sure that the start time is earlier than the end time.

- hour - 0-23

- minute - 0-59

The hour range must be within the specified day, from 00:00 (midnight) until 23:59. If a time range goes past midnight (for example, Friday 19:00 to Saturday 02:00), enter the time in two ranges. Set up the first range as Friday from 19:00 to 23:59 and the second range as Saturday from 00:00 to 01:59.

A time is considered to be in the table from the first second of the start time (for example, 08:00:00). Also, it is still considered to be in the table until the last second of the end time (for example, 17:00:59).

Related topic

For more information about administering this feature, see *Avaya Call Center Call Vectoring and EAS Guide*.

Vector Directory Number screen

This section includes the following topics:

- [About VDNs](#) on page 361
- [VDN administration commands](#) on page 362
- [Vector Directory Number field descriptions](#) on page 362
- [Implementation notes](#) on page 367

About VDNs

This screen is used to define vector directory numbers (VDNs) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector.

VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed using direct dial Central Office (CO) trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN may be Night Destination for LDN.

VDN administration commands

Use the following administration commands to administer the Vector Directory Number screen.

Action	Object	Qualifier ¹
add	vdn	xxxxx (extension number of VDN to be added) or 'next'
change	vdn	xxxxx (extension number of VDN to be changed)
display	vdn	xxxxx (extension number of VDN to be displayed) ['print' or 'schedule']
list	vdn	xxxxx (extension number of VDN to be listed) ('count' 1-MAX) ['print' or 'schedule']
	vdn	bsr xxx (number of a BSR application plan)
remove	vdn	xxxxx (extension number of VDN to be removed)

1. Brackets [] indicate the qualifier is optional. Single quotes (' ') indicate the text inside the quote must be entered exactly as shown or an abbreviated screen of the word can be entered. MAX is the maximum number available in your system configuration. For more information, see *Hardware Description and Reference for Avaya Communication Manager*.

Note:

The asterisks and the footnote are not visible if you have Meet-me Conferencing.

Vector Directory Number field descriptions

On your system, fields display only when certain features are enabled.

1st/2nd/3rd Skill: Only displayed when Expert Agent Selection is enabled on the System-Parameters Customer-Options screen. Enter the desired Skill numbers (or leave blank) in each field. Valid entries are 1-999, or blank (default).

Acceptable Service Level (sec): Only displayed when the BCMS/VuStats Service Level option is enabled on the System Parameters Customer Options screen and the Measured field is internal or both. Enter the number of seconds within which calls to this VDN should be answered. This allows BCMS to print out a percentage of calls that were answered within the specified time. Valid entries are 0 to 9999 seconds. The default is blank.

Allow VDN Override? V This field displays if the **Meet-me Conferencing** field is **n**. Valid entries are **y** and **n**. The default is **n**. This entry changes the *active* VDN for the call.

Valid entries	Usage
n	The <i>active</i> VDN is not replaced by the routed-to VDN.
y	The <i>active</i> VDN is replaced by the routed-to VDN. The routed-to VDN is then used for the parameters associated with the call.

For more information about VDN Overrides and the parameters, see *Avaya Call Center Call Vectoring and EAS Guide*.

Attendant Vectoring? When Attendant Vectoring is optioned, a field on the VDN screen identifies if this is an Attendant Vectoring VDN. If this field is n, there are no changes on the VDN screen. If this field is y, the screen appearance changes (see below).

When removing a VDN, validation verifies that this VDN is not being used on either the Console Parameters screen or the Tenant Partitioning forms.

AUDIX Name: If this VDN is associated with the AUDIX vector, enter the name of the AUDIX machine as it appears in the Adjunct Names screen.

BSR Application: To use multi-site Best Service Routing with this VDN, enter a 1- to 3-digit number to specify an application plan for the VDN. This field only appears if Look-Ahead Interflow (LAI) and Vectoring (Best Service Routing) are enabled on the System Parameters Customer-Options screen.

BSR Available Agent Strategy: The available agent strategy determines how Best Service Routing identifies the best split or skill to service a call in an agent surplus situation. To use Best Service Routing with this VDN, enter an agent selection strategy in this field. Acceptable entries are 1st-found, UCD-LOA, UCD-MIA, EAD-LOA, and EAD-MIA.

This field only appears if Vectoring (Best Service Routing) is enabled on the System Parameters Customer-Options screen and a value other than **1st-found** is set in the **BSR Tie Strategy** field.

BSR Local Treatment? In a multi-site BSR configuration, a call that arrives at a local communication server can be rerouted to a remote server located in a different part of the world. This feature allows you to provide local audio feedback for IP and ISDN calls while a call waits in queue on a remote server.

For more information about this feature, see *Avaya Call Center Call Vectoring and EAS Guide*.

BSR Tie Strategy: This field appears only if Vectoring (Best Service Routing) is set to y on the System Parameters Customer-Options screen.

Valid entries	Usage
system	The setting in the BSR Tie Strategy field on the Feature-Related System Parameters screen applies. This is the default setting.
1st-found	BSR uses the previously selected best choice as the best skill or location.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the consider step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

For more information about the Alternate Selection on BSR Ties feature, see [Alternate Selection on BSR Ties](#) on page 124.

COR: Enter a 1- to 3-digit number that specifies the class of restriction (COR) to be assigned the VDN. The default value is 1. The field cannot be blank and must have an entry in the range from 0-995.

Daylight Savings Rule: This field is used with the **VDN Time-Zone Offset** field to define the daylight saving time rule. The daylight saving time rule and the time zone offset are applied to `goto` time-of-day commands in the vector that is assigned to the VDN. The time-of-day calculations are based on the local time of the receiving call's VDN. The assigned rule number applies start and stop rules that are administered on the system **Daylight Savings Rule** field for that rule number.



Tip:

Use the `list usage vdn-time-zone-offset` command to find VDNs containing an administered daylight saving time rule.

Defines the options available for call centers that use daylight savings time.

Valid entries	Usage
system	The system uses the same daylight saving time rule as the system clock shown in the display/set time field.
0	No daylight saving rule is applied. If the system time has a daylight saving rule specified, this rule is removed before evaluating the <code>goto if time-of-day</code> conditional.
1-15	Indicates the rule as defined on the Daylight Savings Rule field. When you use a number other than 0, the rule associated with the main server clock display time and the main server offset are not used. The offset and rule assigned to the active VDN for the call are applied to the operating system standard time so that local time for the VDN is used to test the time-of-day step.

Display VDN for Route-To DAC? The Display VDN for Route-to DAC option is designed to address situations where one of the following conditions is in effect:

- Either a `route-to number` or `route-to digits` vector command routes an EAS direct agent call, with the coverage option set to `y`
- An adjunct routing step routes a direct agent call, with the coverage option set to `y`

Valid entries are `y` and `n` (default).

For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

Extension: Enter the extension associated with the VDN when completing a paper screen. The extension is a number that starts with a valid first digit and length as defined by the System's dial plan. This is a display-only field when using an administration command such as `add` or `change` to access the screen.

Measured: Used to collect measurement data for this VDN. Valid entries are `internal`, `external`, `both`, or `none`. Data may be collected for reporting by BCMS or CMS. Default is `none`.

Note:

The BCMS feature must be enabled on the System-Parameters Customer-Options screen for the Measured field to be set to `internal` or `both`. In addition, the appropriate CMS release must be administered on the Feature-Related System Parameters screen if the field is being changed to `external` or `both`.

Meet-me Conferencing? This field is available only when the Meet-me Conference feature is enabled. Valid entries are `y` and `n`. The default is `n`.

For more information, see *Avaya Call Center Call Vectoring and EAS Guide*.

Name: Enter up to a 27-character alphanumeric name that identifies the VDN. This is an optional field that need not contain any data. The name may be truncated on agents' displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent. Default is blank.

Observe on Agent Answer? Valid entries are `y` and `n` (default). This entry allows for a service observer to start observing of a call to the VDN when the call is delivered to the agent or the station.

Reporting for PC Predictive Calls? Activates the Improved Integration with Proactive Contact Outbound Calling feature. Use this field if you have switch-classified outbound calling. Valid entries are `y` and `n`. The default is `n`.

For more information about this feature, see [Proactive Contact outbound calling improved reporting](#) on page 212.

PC Predictive Reports Skill: Assigns a skill hunt group to be used for reporting associated with Proactive Contact switch-classified calls on a per VDN basis. Reports are generated as though the agent were in the ACD-OUT state. This field is available only if the **Reporting for PC Predictive Calls?** field is set to **y**. Valid entries are skill numbers from 1-n. The default value is blank. Use 1-99 for S8300 or S8400 servers. Use 1-2000 for all other platforms.

For more information about using Proactive Contact, see [Proactive Contact outbound calling improved reporting](#) on page 212.

Return Destination: The VDN extension number to which an incoming trunk call will be routed if it returns to vector processing after the agent drops the call. Valid entries are the VDN extension, or blank (default).

Service Objective (sec): Displayed in one of two cases:

- When the BCMS/VuStats Service Level option is enabled on the System-Parameters Customer-Options screen and the Measured field is internal or both. Enter the number of seconds within which calls to this VDN should be answered. This will allow BCMS to print out a percentage of calls that were answered within the specified time. Valid entries are 0 through 9999 seconds. Default is blank.
- When the Dynamic Advocate customer option is set on the System-Parameters Customer-Options screen. This field enables the Dynamic Queue Position feature. This feature allows you to queue calls from multiple VDNs to a single skill, while maintaining different service objectives for those VDNs. Enter the service level, in seconds, that you want to achieve for the VDN. Valid entries are 1 through 9999. The default value is 20.

TN: Enter the Tenant Partition number. Valid entries are 1-100. The default value is 1.

VDN of Origin Annc. Extension: Only displayed if VDN of Origin Announcements is enabled on the System-Parameters Customer-Options screen. Enter the extension number of the VDN of Origin announcement. Default is blank.

Vector Number: Enter a vector number that specifies a particular call vector that is accessed through the VDN. This field cannot be blank.

VDN Timed ACW Interval: When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. Enter the number of seconds the agent should remain in ACW following the call. When the administered time is over, the agent automatically becomes available. This field has priority over the Timed ACW Interval field on the Hunt Group screen.

VDN Override for ISDN Trunk ASAI Messages?: The VDN Override for ISDN Trunk ASAI Messages? field is displayed on page 2 of the Vector Directory Number screen only when the following conditions are set by the Communication Manager license file:

- The G3 Version field on the System-Parameters Customer Options screen is set to V10 (or later).
- The ASAI Link Core Capabilities option in the System-Parameters Custom-Options screen is set to **y**.

Note:

The VDN Override for ISDN Trunk ASAI Messages? field can be set to y only if the Allow VDN Override? on page 1 of the Vector Directory Number screen is also set to y.

For more information about this feature, see *Avaya Call Center Call Vectoring and EAS Guide*.

Var Description: This field is displayed only if VDN Variables is active. The description field allows users to describe the VDN variable using up to 15 characters.

Var Assignment: The assignment field assigns an up to 16-digit unvalidated decimal number to the VDN variable. Each digit entry can be:

- 0 - 9
- Left blank

VDN Time-Zone Offset: This field is applied against the communication server clock when a time of day vector command is executed. Daylight savings time changes are handled by the communication server clock using the existing operation.

Based on a syntax of +HH:MM, the valid entries are:

[+ or -] [0 - 23] : [0 - 59]

The default is +00:00. When the default is set, the system communication server time is used without modification.

Valid entries	Usage
+ or -	<ul style="list-style-type: none"> ● Use the - sign if the VDN local time is earlier than the server local time. ● Use the + sign if the VDN local time is later than the server local time. <p>For example, if the server local time is in Eastern Standard Time and the local time used for calls to the VDN is Mountain Standard Time, use - 2:00.</p>
0 - 23	Time in hours
0 - 59	Time in minutes

For more information about this feature, see [VDN Time Zone Offset](#) on page 303.

Implementation notes

- The BCMS feature must have been optioned if the Measured field is set to internal or both. In addition, the appropriate CMS release must be administered on the Feature-Related System Parameters screen if the field is being changed to external or both.

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- The 1st/2nd/3rd Skill fields are only displayed when Expert Agent Selection is enabled on the System-Parameters Customer-Options screen.
- The BCMS Acceptable Service Level (sec) field is only displayed if the BCMS Acceptable Service Level option is enabled on the System-Parameters Customer-Options screen and the Measured field is internal or both.
- Data for the Orig Annc column appears only when VDN of Origin Announcement is enabled on the System-Parameters Customer-Options screen.
- To list all VDNs using the same BSR Application Plan, type the administration command list VDN BSR **xxx** (xxx is the number of the BSR Application Plan used by one or more VDNs).
- To associate VDNs and vectors for attendant vectoring, a field has been added to both the VDN and the Call Vectoring forms to indicate attendant vectoring. When attendant vectoring is indicated for VDNs and vectors, all call center-associated fields (such as Skills and BSR) are removed.

Vector Routing Table screen

This section includes the following topics:\

- [About the Vector Routing Table](#) on page 368
- [Vector Routing Table administration commands](#) on page 369
- [Vector Routing Table field descriptions](#) on page 369

About the Vector Routing Table

This screen is used to store ANI or Digits that you refer to in **goto** vector steps. It is available to you only if the Vectoring (G3V4 Enhanced) field on the System-Parameters Customer-Options screen is set to y.

For more information, refer to [Call Vectoring](#) on page 137.

Vector Routing Table administration commands

You can use the commands listed in the table below to administer Vector Routing Tables. In addition, you can use the List Usage command to see the vectors and digit fields used by a Vector Routing Table.

Action	Object	Qualifier
add	VRT	1 through x, or next
change	VRT	1 through x, or next
display	VRT	1 through x, or next
remove	VRT	1 through x, or next
list	VRT	none - all Routing Tables will display

Vector Routing Table field descriptions

Make assignment as required for the following fields on the screen.

Name: Enter a 1 to 15-character alphanumeric table name. You may leave this field blank. Default is blank.

Number: This is the table number that you entered on the command line. It is a display only field.

Number (1-32): Enter a number. Default is blank. Entries in this field also can include the + and/or ? wildcard. The + represents a group of digits. The ? represents a single digit. The field is limited to 16 characters and these characters are restricted as follows:

- You may enter only a plus sign (+), a question mark (?), or the numbers 0 through 9. No other entries are valid.
- You may enter one plus sign (+) as either the first or last character in the number field. However, you cannot use this character as the sixteenth character of the number field.
- You may use as many question marks (?) as you wish, anywhere in the number field.
- You may not embed blanks in the number field.
- You may leave the field entirely blank. If you leave the field blank, the communication server will store the entry as a null value.

Sort: Enter y if you want the digit fields to be sorted. Default is n. If you elect not to sort the numbers, they will remain in the order that you entered them. If you elect to sort the number fields, they will be sorted as described below. Remember that leading zeros are significant. That means that 02 will sort ahead of a 2 followed by a space.

- Any Plus signs (+) will sort first.
- Any question marks (?) will sort second.
- All numbers (0-9) will sort last.

VuStats Display Format screen

This section includes the following topics:

- [About the VuStats Display Format screen](#) on page 370
- [VuStats Display Format administration commands](#) on page 370
- [VuStats Display Format field descriptions](#) on page 371
- [List VuStats Display Format screen](#) on page 375
- [VuStats fields](#) on page 375

About the VuStats Display Format screen

Use the VuStats Display Format screen to define the content and layout of information on VuStats telephone displays. The system has 50 different display formats; the first display is a predefined example format, which can be changed; displays 2 through 50 are blank. Each display format can contain up to ten data items. However, the amount of data to be displayed is limited to 40 characters, due to the physical limitations on display telephones.

VuStats Display Format administration commands

Use the following administration commands to access the VuStats Display Format screen.

Action	Object	Qualifier ¹
change	vustats-display-format	1-50
display	vustats-display-format	1-50
list	vustats-display-format	1-50 count ##

1. Specify a number from 1 to 50 to indicate the number of the display format to be to changed or displayed. Count **##** is the number of display formats to list.

VuStats Display Format field descriptions

Enter the data as required for the following fields on the screen:

Data Field Character: The character that will be used in the Format Description field to identify the position and length of each data field (see the [Format Description](#) field description). The default is \$. Enter another character if the \$ is needed for fixed text in the Format Description field. Any character is valid except a space.

Data Item fields: On lines 1 through 10, beneath the Data Type field label enter data items for the display format. These data items are associated with the sets of data field characters in the Format Description field. Each data item is defined by one or more of the following fields: Data Type, Format, Period, Threshold, and Reference. Input for these fields is described in more detail below.

Enter each data item in the same order as data fields are defined in the Format Description field. For example, Line 1 of the Data Type field must contain the data item for the first data field (that is, the first set of \$s).

- Data Type - The data item to be included in the current display format. For a complete list of data types available for each object type, see the [Required and allowed fields for split data types](#) on page 379, [Required and allowed fields for agent and agent-extension data types](#) on page 376, [Description of split data types](#) on page 389, [Description of VDN data types](#) on page 392, and [Description of trunk group data types](#) on page 394 tables for a description of data types associated with each object type. The default is blank.
- Format - The format for displaying the data type. The format is required only for a data type with a time value. Enter one of the following Format values. Default is blank.

Valid Input	Description	Minimum Input Length
ccs	Hundred call seconds (CCS) rounded to the nearest CCS	1
h	Hours rounded to the nearest hour	1
h:mm	Hours and minutes rounded to the nearest minute	4
h:mm:ss	Hours, minutes, and seconds	7
m	Minutes rounded to the nearest minute	1
m:ss	Minutes and seconds	4
s	Seconds	1

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- **Period** - Enter the amount of time to be used to collect the historical data for display. If the data type is historical data, you cannot leave this field blank. Valid entries are day (midnight to the current time), interval (the time specified in the Number of Intervals field), or blank. Refer to [VuStats fields](#) to determine if a measurement period is required for a particular Data Type. The default is blank.
- **Threshold** - The threshold field is always an optional field. It contains two subfields, the threshold comparator and the threshold value. The threshold value is used with the threshold comparator to determine if a threshold warning should be generated. A threshold warning is generated if the specified condition is met for one or more of the data items. If the specified condition is not met for any of the data items, then no threshold warning is generated. The threshold value can be any numeric value from 0 to 9999. Default is blank. Valid threshold comparators are:
 - = (equal to)
 - <> (not equal to)
 - < (less than)
 - <= (less than or equal to)
 - > (greater than)
 - >= (greater than or equal to)
- **Ref** - A reference to a split or skill; this field does not appear unless the Object Type is either agent or agent-extension. This field is required only if the data type is an agent-related data type collected on a per-split/skill basis or a split/skill-related data type for one of the agent's logged-in splits/skills. Enter one of the following values:
 - Top references the first-administered, highest-level skill for EAS agents, or the first split or skill logged into for non-EAS agents.

Note:

With EAS, the top skill for VuStats is the first administered, highest level skill measured internally or both. For CMS it is the first-administered, highest-level skill measured externally or both. Therefore, it is possible for the top skill to be a different number skill for CMS than it is for VuStats. To avoid this, measure all skills as both.

- All displays the combined data for all splits/skills the agent is logged into.
- Any number from 1-20. The number represents a split or skill to which the agent has logged in. For example, if the Ref field contains 1, VuStats displays the data for the first split or skill the agent logged into, if the Ref field contains 2, VuStats displays the data for the second split or skill the agent logged into, and so on.

Display Interval: The interval, in seconds, for which data is displayed if no update is entered in the Update Interval field. Enter one of the following values:

- 5 - Display clears after 5 seconds
- 10 - Display clears after 10 seconds

- 15 - Display clears after 15 seconds
- 30 - Display clears after 30 seconds
- not cleared - The display does not clear, and the data appears until the display is used for another operation or until you press the Normal button.

Format Description: The definition of the layout for the 40-character display. Specify the starting position and the length of the data items by entering, for each data field, an optional label for the field followed the appropriate number of data field characters (such as \$s). Each \$ represents one character in the display. For example, if the data will be a maximum of five characters long, enter \$\$\$\$\$.

Some data types have preset maximum field length limits based on the communication server administration. For example, the data type acceptable-service-level is taken from the BCMS Acceptable Service Level field on the Hunt Group and Vector Directory Number forms; on this screen, the field allows a maximum number of four characters. Therefore, for the acceptable-service-level, you should not create a VuStats display field that consists of more than four characters (that is, \$\$\$\$). Other data types have similar limits.

Field lengths for data items that appear as time must match the value in the Format field, which is discussed below. Remember to account for possible colons when the display will be in a time format.

Format descriptions can be all text (such as a message of the day) or they can be all data fields, in which case users will have to memorize the labels or use customer-provided overlays above or below the display.

If the numeric data for a field is too large for the number of data field characters entered, the VuStats display will show asterisks instead of data. If name database items are too large for the number of data field characters, the VuStats display will truncate the data to fit the data field size. The split or skill objective, as entered on the Hunt Group screen, displays as asterisks if the information exceeds the data field size.

If the data for a field is too large for the number of data field characters entered, VuStats displays asterisks. If name database items are too large for the number of data field characters, VuStats truncates the data. VuStats also displays Split/Skill Objective (assigned on the Hunt Group screen) as asterisks if the information exceeds the data field size.

Format Number: The system generates a format number automatically when it creates a VuStats display. You cannot change this number. You can create 50 different display formats. Format Number 1 is a predefined sample format that you can modify.

Next Format Number: To link this display to another display, enter the number of the display format (between 1-50) that should appear when a VuStats user presses the next button, or enter none (default). In general, you only link displays with the same object type.

Number of Intervals: Specify the number of BCMS intervals used to collect data when you have specified interval as the period for a historical Data Type. You can enter a number between 1 and 25, or blank for current interval. The default is blank. If you enter 24, and the BCMS measurement interval on the Feature-Related System Parameters screen is set to 1 hour, you will receive information on the previous 24 hours. If the BCMS measurement interval is set to half-hour, you will receive information on the previous 12 hours. You can also leave this field blank. If you do, you will receive information on the current interval.

Object Type: The type of object for which data will be displayed. Enter one of the following values: agent (for staffed agents to view their own statistics), agent-extension (for other users to view agent statistics), split or skill (default), trunk-group, or vdn.

- Agent - Provides agents with their own statistics, or statistics about the splits/skills they log into.
- Agent-extension - Provides supervisors with statistics about agents or the splits/skills the agents log into. VuStats can automatically display statistics for a specific agent (if you administer agent login ID or BCMS/Vustats login ID). Or, supervisors can enter the ID of any agent they want to review.
- Split/Skill - Displays statistics about a specific split or skill. You must administer the split or skill as Measured (internal or both) on the Hunt Group screen.
- Trunk-group - Displays statistics about a specific trunk group. You must administer the trunk group as Measured (internal or both) on the Trunk Group screen.
- VDN - Displays statistics about a specific VDN. You must administer the VDN as Measured (internal or both) on the Vector Directory Number screen.

On Change: Enter y to update the display whenever the agent's state changes. The update on agent state change is in addition to the update as a result of the value entered in the Update Interval field. If n is entered, an update will only occur based on the Update Interval and not on the agent state change.

Update Interval: The interval, in seconds, between display updates. Enter one of the following values:

- no-update - The display is not updated, and appears only for the interval specified in the Display Interval field
- polled - Updates the display hourly or half-hourly, based on the value in BCMS Measurement Interval (System-Parameters Features screen)
- 10 - Updates every 10 seconds
- 20 - Updates every 20 seconds
- 30 - Updates every 30 seconds
- 60 - Updates every 60 seconds
- 120 - Updates every 2 minutes

List VuStats Display Format screen

A second VuStats Display Format screen is available with the list command. The purpose of this screen is to present the format of all, or a selected number, of VuStats displays. The List VuStats Display Format screen displays the Format Number, Next Format Number, Number of Intervals, Object Type, Update Interval, and Format Description fields, and all designated data items, including the Data Type, Format, Threshold, and Ref (split or skill reference).

Use this screen to compare VuStats displays to each other. This list presentation is most helpful when trying to see how displays are linked to each other. The screen includes the Next field, which contains the number of the next display (if any) to which a display is linked.

VuStats Display Formats field descriptions

Format Description: The definition of the display's layout. The first line of the Format Description contains the text that precedes the data on a display plus the length of each data field (indicated by \$s). The succeeding lines of the Format Description identify the data items, in the order they are to appear. The data items are the actual measurements and other information that tell how agents, splits/skills, vector directory numbers (VDNs), trunk groups, and the call center are performing. The data items are followed by the format (if any), which identifies how the data is to appear in the display, the period and threshold (if any), and the split or skill reference (if any).

Int: Number of Intervals. The number of measurement intervals.

Next: The number of the next display if the current display is linked to (followed by) another display format, or none if the current display format is not linked to another display format.

No: Number. The unique identifying number of each display format.

Object Type: The type of object for which data will be displayed.

Update: Update Interval. The time between display updates.

VuStats fields

The following tables show the required and allowed fields for the following data types:

- [Required and allowed fields for agent and agent-extension data types](#) on page 376
- [Required and allowed fields for split data types](#) on page 379
- [Required and allowed fields for VDN data types](#) on page 380
- [Required and allowed fields for trunk group data types](#) on page 381

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- [Description of agent and agent-extension data types](#) on page 382
- [Description of split data types](#) on page 389
- [Description of VDN data types](#) on page 392
- [Description of trunk group data types](#) on page 394

Required and allowed fields for agent and agent-extension data types

The following table shows the required and allowed fields for the agent and agent-extension data types.

VuStats data type	Format	Period	Threshold	Reference
acd-calls		required	allowed	required
agent-extension				
agent-name				
agent-state				required
average-acd-call-time	required	required	allowed	
average-acd-talk-time	required	required	allowed	required
average-extension-time	required	required	allowed	
call-rate		required	allowed	
current-reason-code			allowed	
current-reason-code-name			allowed	
elapsed-time-in-state				
extension-calls		required	allowed	
extension-incoming-calls			allowed	
extension-outgoing-calls			allowed	
percent-acd-call-time		required	allowed	
shift-acd-calls			allowed	required
shift-aux-time-1	required		allowed	
shift-aux-time-2	required		allowed	
shift-aux-time-3	required		allowed	
shift-aux-time-4	required		allowed	

VuStats data type	Format	Period	Threshold	Reference
shift-aux-time-5	required		allowed	
shift-aux-time-6	required		allowed	
shift-aux-time-7	required		allowed	
shift-aux-time-8	required		allowed	
shift-aux-time-9	required		allowed	
shift-aux-time-all	required		allowed	
shift-aux-time-default	required		allowed	
shift-aux-time-non-default	required		allowed	
shift-aux-time-reason-code	required		allowed	
shift-average-acd-talk-time	required		allowed	required
skill-level				required
split-acceptable-service-level	required			required
split-acd-calls		required	allowed	required
split-after-call-sessions			allowed	required
split-agents-available			allowed	required
split-agents-in-after-call			allowed	required
split-agents-in-aux-1			allowed	required
split-agents-in-aux-2			allowed	required
split-agents-in-aux-3			allowed	required
split-agents-in-aux-4			allowed	required
split-agents-in-aux-5			allowed	required
split-agents-in-aux-6			allowed	required
split-agents-in-aux-7			allowed	required
split-agents-in-aux-8			allowed	required
split-agents-in-aux-9			allowed	required
split-agents-in-aux-all			allowed	required
split-agents-in-aux-default			allowed	required

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VuStats data type	Format	Period	Threshold	Reference
split-agents-in-aux-non-default			allowed	required
split-agents-in-other			allowed	required
split-agents-on-acd-calls			allowed	required
split-agents-on-extension-calls			allowed	required
split-agents-staffed			allowed	required
split-average-acd-talk-time	required	required	allowed	required
split-average-after-call-time	required		allowed	required
split-average-speed-of-answer	required	required	allowed	required
split-average-time-to-abandon	required	required	allowed	required
split-call-rate			allowed	required
split-calls-abandoned		required	allowed	required
split-calls-flowed-in		required	allowed	required
split-calls-flowed-out		required	allowed	required
split-calls-waiting			allowed	required
split-extension				required
split-name				required
split-number				required
split-objective				required
split-oldest-calling-waiting	required		allowed	required
split-percent-in-service-level		required	allowed	required
split-total-acd-talk-time	required	required	allowed	required
split-total-after-call-time	required	required	allowed	required
split-total-aux-time	required	required	allowed	required
time-agent-entered-state				required
total-acd-call-time	required	required	allowed	
total-acd-talk-time	required	required	allowed	
total-after-call-time	required	required	allowed	

VuStats data type	Format	Period	Threshold	Reference
total-aux-time	required	required	allowed	
total-available-time	required	required	allowed	
total-hold-time	required	required	allowed	
total-staffed-time	required	required	allowed	

Required and allowed fields for split data types

The following table shows the required and allowed fields for split data types.

VuStats data type	Format	Period	Threshold
acceptable-service-level	required		
acd-calls		required	allowed
after-call sessions			allowed
agents-available			allowed
agents-in-after-call			allowed
agents-in-aux-1			allowed
agents-in-aux-2			allowed
agents-in-aux-3			allowed
agents-in-aux-4			allowed
agents-in-aux-5			allowed
agents-in-aux-6			allowed
agents-in-aux-7			allowed
agents-in-aux-8			allowed
agents-in-aux-9			allowed
agents-in-aux-all			allowed
agents-in-aux-default			allowed
agents-in-aux-non-default			allowed
agents-in-other			allowed
agents-on-acd-calls			allowed

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VuStats data type	Format	Period	Threshold
agents-on-extension-calls			allowed
agents-staffed			allowed
average-acd-talk-time	required	required	allowed
average-after-call-time	required		allowed
average-speed-of-answer	required	required	allowed
average-time-to-abandon	required	required	allowed
call-rate			allowed
calls-abandoned		required	allowed
calls-flowed-in		required	allowed
calls-flowed-out		required	allowed
calls-waiting			allowed
oldest-calling-waiting	required		allowed
percent-in-service-level		required	allowed
split-extension			
split-name			
split-number			
split-objective			
total-acd-talk-time	required	required	allowed
total-after-call-time	required	required	allowed
total-aux-time	required	required	allowed

Required and allowed fields for VDN data types

The following table shows the required and allowed fields for the VDN data types¹.

VuStats data type	Format	Period	Threshold
acceptable-service-level	required		
acd-calls		required	allowed

1. For a description of VDN data types, see [Description of VDN data types](#) on page 392.

VuStats data type	Format	Period	Threshold
average-acd-talk-time	required	required	allowed
average-speed-of-answer	required	required	allowed
average-time-to-abandon	required	required	allowed
calls-abandoned		required	allowed
calls-flowed-out		required	allowed
calls-forced-busy-or-disc		required	allowed
calls-offered		required	allowed
calls-waiting			allowed
non-acd-calls-connected		required	allowed
oldest-calling-waiting	required		allowed
percent-in-service-level		required	allowed
total-acd-talk-time	required	required	allowed
vdn-extension			
vdn-name			

Required and allowed fields for trunk group data types

The following table shows the required and allowed fields for trunk group data types¹.

VuStats data type	Format	Period	Threshold
average-incoming-call-time	required	required	allowed
average-outgoing-call-time	required	required	allowed
incoming-abandoned-calls		required	allowed
incoming-calls		required	allowed
incoming-usage	required	required	allowed
number-of-trunks			
outgoing-calls		required	allowed
outgoing-completed-calls		required	allowed

1. For a description of trunk group data types, see [Description of trunk group data types](#) on page 394.

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VuStats data type	Format	Period	Threshold
outgoing-usage	required	required	allowed
percent-all-trunks-busy		required	allowed
percent-trunks-maint-busy		required	allowed
trunk-group-name			
trunk-group-number			
trunks-in-use			allowed
trunks-maint-busy		required	allowed

Description of agent and agent-extension data types

The following table describes the agent and agent-extension data types.

VuStats data type	Description	BCMS report: field name/column heading
acd-calls	Split/skill calls and direct agent calls answered by an agent	Split status/VDN Status/ Agent Report: ACD CALLS
agent-extension	The extension for a specific agent; if either BCMS/VuStats login IDs or EAS is optioned, then this shows the agent's login ID.	Split Status: Login ID or EXT
agent-name	The administered name for a specific agent.	Split Status/Agent Report/Agent Summary Report: Agent
agent-state	The agent's current work state	Split Status: STATE
average-acd-call- time	The average of hold-time plus talk-time.	None
average-acd-talk- time	The average time a specific agent has spent talking on completed ACD calls during a specified time period for all internally-measured splits/skills that the agent was logged into. This does not include the time a call was ringing or was on hold at an agent's terminal.	Agent Report/Agent Summary Report: AVG TALK TIME
average-extension- time	The average amount of time an agent spent on non-ACD calls while logged into at least one split/skill during the reported interval. This average does not include time when the agent was holding the EXTN call.	Agent Report/Agent Summary Report: AVG EXTN TIME

VuStats data type	Description	BCMS report: field name/column heading
call-rate	The current rate of ACD calls handled per agent per hour for all split/skills.	None
current-reason-code	The number of the reason code associated with the agent's current AUX work mode, or with the agent's logout.	None
current-reason-code-name	The name of the reason code associated with the agent's current AUX work mode or with the agent's logout.	None
elapsed-time-in-state	The amount of time an agent has been in the current state.	None
extension-calls	The number of incoming and outgoing non-ACD calls that an agent completed while logged into at least one split/skill.	Agent Report: EXTN CALLS
extension-incoming-calls	The number of non-ACD calls that an agent receives while logged into at least one split/skill.	Split Report: EXT IN CALLS
extension-outgoing-calls	The number of non-ACD calls that an agent places while logged into at least one split/skill.	Split Report: EXT OUT CALLS
percent-acd-call-time	<p>The current calculated occupancy for the agent. This data type indicates the percentage of time the agent talks and holds ACD calls, and is calculated as:</p> $\frac{\text{ACDtime} + \text{hold time (ACD calls only)}}{\text{staffed time in interval} + 100}$ <p>The calculation is listed in the <code>occupancy</code> field of the <code>Station Status</code> screen.</p>	None
shift-acd-calls	The number of ACD calls answered by an agent during the administered period.	None
shift-aux-time-1	The amount of time an agent has spent in AUX work mode for reason code 1 during the administered period.	None
shift-aux-time-2	The amount of time an agent has spent in AUX work mode for reason code 2 during the administered period.	None
shift-aux-time-3	The amount of time an agent has spent in AUX work mode for reason code 3 during the administered period.	None

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VuStats data type	Description	BCMS report: field name/column heading
shift-aux-time-4	The amount of time an agent has spent in AUX work mode for reason code 4 during the administered period.	None
shift-aux-time-5	The amount of time an agent has spent in AUX work mode for reason code 5 during the administered period.	None
shift-aux-time-6	The amount of time an agent has spent in AUX work mode for reason code 6 during the administered period.	None
shift-aux-time-7	The amount of time an agent has spent in AUX work mode for reason code 7 during the administered period.	None
shift-aux-time-8	The amount of time an agent has spent in AUX work mode for reason code 8 during the administered period.	None
shift-aux-time-9	The amount of time an agent has spent in AUX work mode for reason code 9 during the administered period.	None
shift-aux-time-all	The amount of time an agent has spent in AUX work mode for all reason codes during the administered period.	None
shift-aux-time-default	The amount of time an agent has spent in AUX work mode for the default reason code (code 0) during the administered period.	None
shift-aux-time-non-default	The amount of time an agent has spent in AUX work mode for reason codes 1 through 99 during the administered period.	None
shift-aux-time-reason-code	The amount of time an agent has spent in AUX work mode for the agent's current reason code during the administered period.	None
shift-average-acd-talk-time	The average talk time for ACD calls for a specific agent during the administered period.	None
skill-level	The skill level at which the skill was assigned to the agent.	None
split-acceptable-service-level	The number of seconds within which calls must be answered to be considered acceptable. Identified on a per-hunt group basis. Timing begins when the call enters the hunt group queue.	Split Status: Acceptable Service Level

VuStats data type	Description	BCMS report: field name/column heading
split-acd-calls	Split/skill calls and direct agent calls answered by an agent.	System Status/Split Report/Split Summary Report: ACD CALLS
split-after-call- sessions	The number of times all agents have entered After Call Work (ACW) for a specific split/skill.	None
split-agents-available	The number of agents currently available to receive ACD calls for a specific split. This includes agents in auto-in or manual-in work mode.	Split Status: Avail
split-agents-in-after- call	For a specific split, the number of agents currently in ACW.	Split Status: ACW
split-agents-in-aux-1	For a specific skill, the number of agents currently in Aux work mode with reason code 1.	None
split-agents-in-aux-2	For a specific skill, the number of agents currently in Aux work mode with reason code 2.	None
split-agents-in-aux-3	For a specific skill, the number of agents currently in Aux work mode with reason code 3.	None
split-agents-in-aux-4	For a specific skill, the number of agents currently in Aux work mode with reason code 4.	None
split-agents-in-aux-5	For a specific skill, the number of agents currently in Aux work mode with reason code 5.	None
split-agents-in-aux-6	For a specific skill, the number of agents currently in Aux work mode with reason code 6.	None
split-agents-in-aux-7	For a specific skill, the number of agents currently in Aux work mode with reason code 7.	None
split-agents-in-aux-8	For a specific skill, the number of agents currently in Aux work mode with reason code 8.	None
split-agents-in-aux-9	For a specific skill, the number of agents currently in Aux work mode with reason code 9.	None

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VuStats data type	Description	BCMS report: field name/column heading
split-agents-in-aux-all	For a specific split/skill, the total number of agents currently in Aux work mode for all reason codes.	Split Status: AUX
split-agents-in-aux-default	For a specific split/skill, the number of agents currently in Aux work mode with the default reason code (code 0).	None
split-agents-in-aux-non-default	For a specific skill, the number of agents currently in Aux work mode with reason codes 1 through 99.	None
split-agents-in-other	The number of agents currently who: are on a call for another split, are in ACW work mode for another split, have a call on hold but are not in another state, or have a call ringing at their terminals, or are dialing a number while in AI/MI.	Split Status: Other
split-agents-on-acd-calls	The number of agents currently on split/skill or direct agent ACD calls for a specific split.	Split Status: ACD
split-agents-on-extension-calls	The number of agents in a specific split who are currently on non-ACD calls.	Split Status: Extn
split-agents-staffed	The number of agents currently logged into a split.	Split Status: Staffed
split-average-acd-talk-time	The average talk time for ACD calls during a specific period/day for a specified split.	System Status/Split Report/Split Summary Report: AVG TALK TIME
split-average-after-call-time	The average time for call-related ACW completed by agents for this split (the same as average-after-call-time, but only available for agent and agent-extension object types). Call-related ACW time is recorded when an agent leaves the ACW state. If an agent is in call-related ACW when an interval completes, all the ACW time will be recorded for the interval in which the agent leaves ACW.	System Status: AVG AFTER CALL
split-average-speed-of-answer	The average speed for answering split and direct agent ACD calls that have completed for a specified split/skill.	System Status/Split Report/Split Summary Report: AVG SPEED ANS
split-average-time-to-abandon	The average time calls waited in queue and ringing before abandoning.	System Status/Split Report/Split Summary Report: AVG ABAND TIME

VuStats data type	Description	BCMS report: field name/column heading
split-call-rate	The current rate of ACD calls handled per agent per hour for a specific split or skill.	None
split-calls-abandoned	The number of calls that abandoned from queue (provided this is the first split/skill queued to) or abandoned from ringing.	System Status/Split Report/Split Summary Report: ABAND CALLS
split-calls-flowed-in	The total number of calls for a specific split/skill that were received as a coverage point (intraflowed) from another internally-measured split/skill, or were call-forwarded (interflowed) to the split/skill.	Split Report/Split Summary Report: FLOW IN
split-calls-flowed-out	The total number of calls for a specific split/skill that successfully extended to the split/skill's coverage point, were call-forwarded out, or were answered using call pick-up.	Split Report/Split Summary Report: FLOW OUT
split-calls-waiting	The number of calls that have encountered a split but have not been answered, abandoned, or outflowed.	System Status: CALLS WAIT
split-extension	The administered extension for a split.	None
split-name	The administered name for a split/skill.	Split Report/Split Status: Split Name System Status: SPLIT
split-number	The administered number for a split/skill.	Split Report/Split Status: Split Name System Status: SPLIT
split-objective	The administered objective for a split/skill.	None
split-oldest-call-waiting	The time the oldest call has been waiting for a specific split/skill.	System Status: OLDEST CALL
split-percent-in-service-level	For a specific split/skill, the percentage of calls answered within the administered service level on the hunt group screen.	System Status/Split Report/Split Summary Report: % WITHIN SERVICE LEVEL
split-total-acd-talk-time	For a specified split/skill, the total time agents spent talking on split/skill calls and direct agent calls for this split.	None
split-total-after-call-time	The total time an agent spent in call-related ACW for this split/skill and non-call-related ACW for any split/skill during a specific time period, excluding time spent on incoming or outgoing extension calls while in ACW.	Split Report/Split Summary Report: TOTAL AFTER CALL

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VuStats data type	Description	BCMS report: field name/column heading
split-total-aux-time	The total time an agent spent in AUX mode for this split/skill.	Split Report/Split Summary Report: TOTAL AUX/OTHER
total-acd-call-time	The total talk time plus the total hold time for split/skill and Direct Agent ACD calls.	None
total-acd-talk-time	The total time agents spent talking on split/skill calls and direct agent calls.	None
total-after-call-time	The total time an agent spent in call-related or non-call-related ACW for any split during a specific time period, excluding time spent on incoming or outgoing extension calls while in ACW. (With EAS, all non-call related ACW time is associated with the first skill logged into.)	Agent Report/Agent Summary Report: TOTAL AFTER CALL
total-aux-time	The total time an agent spent in AUX work for all splits/skills (simultaneously) that the agent was logged into. If an agent entered AUX in one interval, but ended AUX in another, each of the intervals will reflect the appropriate amount of time spent in the interval (agent reports also include OTHER time).	Agent Report/Agent Summary Report: TOTAL AUX/OTHER
total-available-time	The time an agent was available in at least one split/skill.	Agent Report: TOTAL AVAIL TIME
total-hold-time	The total amount of time ACD calls were on hold at a specific agent's phone. This time is the caller's hold time and is independent of the agent's state. This time does not include hold time for non-ACD calls on hold.	Agent Report: TOTAL HOLD TIME
total-staffed-time	The total amount of time an agent was logged into one or more splits/skills during a specific period/day. An agent is clocked for staff time as long as he or she is logged into any split.	Agent Report: TOTAL TIME STAFFED

Description of split data types

The following table describes the split data types.

VuStats data type	Description	BCMS report: field name/column heading
acceptable-service-level	The number of seconds within which calls must be answered to be considered acceptable. Identified on a per-hunt group basis. Timing begins when the call enters the hunt group queue.	Split Status/Split Report: Acceptable Service Level
acd-calls	Split calls and direct agent calls answered by an agent	Split status/VDN Status/ Agent Report: ACD CALLS
after-call sessions	The number of times all agents have entered ACW.	None
agents-available	The number of agents currently available to receive ACD calls. This includes agents in auto-in or manual-in work mode.	Split Status: Avail
agents-in-after-call	The number of agents currently in ACW mode.	Split Status: ACW
agents-in-aux-1	The number of agents currently in Aux work mode for reason code 1 for the referenced skill.	None
agents-in-aux-2	The number of agents currently in Aux work mode for reason code 2 for the referenced skill.	None
agents-in-aux-3	The number of agents currently in Aux work mode for reason code 3 for the referenced skill.	None
agents-in-aux-4	The number of agents currently in Aux work mode for reason code 4 for the referenced skill.	None
agents-in-aux-5	The number of agents currently in Aux work mode for reason code 5 for the referenced skill.	None
agents-in-aux-6	The number of agents currently in Aux work mode for reason code 6 for the referenced skill.	None
agents-in-aux-7	The number of agents currently in Aux work mode for reason code 7 for the referenced skill.	None

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VuStats data type	Description	BCMS report: field name/column heading
agents-in-aux-8	The number of agents currently in Aux work mode for reason code 8 for the referenced skill.	None
agents-in-aux-9	The number of agents currently in Aux work mode for reason code 9 for the referenced skill.	None
agents-in-aux-all	The number of agents currently in Aux work mode for all reason codes for the referenced split/skill.	Split Status: AUX
agents-in-aux-default	The number of agents currently in Aux work mode for the default reason code (code 0) for the referenced split/skill.	None
agents-in-aux-non-default	The number of agents currently in Aux work mode for reason codes 1 through 99 for the referenced skill.	None
agents-in-other	The number of agents who currently: are on a call for another split, are in ACW work mode for another split, have a call on hold but are not in another state, or have a call ringing at their terminal, or are dialing a number from AI/MI mode.	Split Status: Other
agents-on-acd-calls	The number of agents currently on split/skill or direct agent ACD calls for a specific split.	Split Status: ACD
agents-on-extension-calls	The number of agents in a specific split who are currently on non-ACD calls.	Split Status: Extn
agents-staffed	The number of agents currently logged into the specified split.	Split Status: Staffed
average-acd-talk-time	The average talk time for ACD calls during a specific period/day for a specified split.	System Status/Split Report: AVG TALK TIME
average-after-call-time	The average time for call-related ACW completed by agents in this split. Call-related ACW time is recorded when an agent leaves the ACW state. If an agent is in call-related ACW when an interval completes, all the ACW time will be recorded for the interval in which the agent leaves ACW.	System Status: AVG AFTER CALL

VuStats data type	Description	BCMS report: field name/column heading
average-speed-of-answer	The average speed for answering split/skill and direct agent ACD calls that have completed for a specified split/skill during a specified time. This includes queue time and ringing time for this split.	System Status/Split Report: AVG SPEED ANS
average-time-to-abandon	The average time calls waited before abandoning.	System Status/Split Report: AVG ABAND TIME
call-rate	The current rate of ACD calls handled per agent per hour for all split/skills.	none
calls-abandoned	The number of calls that abandoned.	System Status/Split Report: ABAND CALLS
calls-flowed-in	The total number of calls for a specific split that were received as a coverage point (intraflowed) from another internally-measured split, or were call-forwarded (interflowed) to the split. This does not include calls that were interflowed from a remote communication server by the Look Ahead Interflow feature.	Split Report/Split Summary Report: FLOW IN
calls-flowed-out	The number of calls the split extended to its coverage point, calls that call-forward out or are answered by call pickup, calls that queued to this split as a primary split and were answered or abandoned from ringing in another split.	Split Report/Split Summary Report: FLOW OUT
calls-waiting	The number of calls that have encountered a split/skill but have not been answered, abandoned, or outflowed.	System Status: CALLS WAIT
oldest-call-waiting	The time the oldest call has been waiting in the split/skill. Timing begins when the call enters the split/skill.	System Status: OLDEST CALL
percent-in-service-level	The percentage of calls offered to the split that were answered within the service level administered on the hunt group screen.	System Status/Split Report/Split Summary Report: % IN SERV LEVL
split-extension	The administered extension for a split.	None
split-name	The administered name for a split.	Split Report/Split Status: Split Name System Status: SPLIT

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VuStats data type	Description	BCMS report: field name/column heading
split-number	The administered number for a split.	Split Report/Split Status: Split Name System Status: SPLIT
split-objective	The administered objective for a split.	None
total-acd-talk-time	The total time agents spent talking on split/skill calls and direct agent calls for this split.	None
total-after-call-time	The total time agents spent in call-related or non-call-related ACW for any split during a specific time period.	Split Report/Split Summary Report: TOTAL AFTER CALL
total-aux-time	The total time agents spent in AUX work mode for all reason codes for the referenced split/skill during the administered period.	Split Report/Split Summary Report: TOTAL AUX/OTHER

Description of VDN data types

The following table describes the VDN data types.

VuStats data type	Description	BCMS report: field name/column heading
acceptable-service-level	The number of seconds within which calls must be answered to be considered acceptable. Identified on a per-VDN basis. Timing begins when the call enters the vector.	VDN Status/VDN Report: Acceptable Service Level
acd-calls	Split calls and direct agent calls answered by an agent	VDN Status: ACD CALLS
average-acd-talk-time	The average talk time for ACD calls during a specific period/day for a specified VDN.	VDN Status/Split Report: AVG TALK HOLD
average-speed-of-answer	The average speed for answering ACD and CONNect calls that have completed for a specified VDN during a specified time. This includes time in vector processing.	VDN Status/VDN Report/VDN Summary Report: AVG SPEED ANS
average-time-to-abandon	The average time calls waited before abandoning.	VDN Status/VDN Report: AVG ABAND TIME
calls-abandoned	The number of calls that abandoned.	VDN Status/VDN Report/VDN Summary Report: ABAND CALLS

VuStats data type	Description	BCMS report: field name/column heading
calls-flowed-out	The total number of calls for a specific VDN that successfully routed to another VDN or off the communication server.	VDN Status/VDN Report/VDN Summary Report: FLOW OUT
calls-forced-busy-or-disc	The number of calls given forced busy or forced disconnect.	VDN Status/VDN Report/VDN Summary Report: CALLS BUSY/DISC
calls-offered	All calls offered to a VDN, including ACD calls, connected calls, abandoned calls, busy calls (calls that received a busy signal), disconnected calls (calls disconnected by the communication server), and outflow calls (calls directed to another VDN or off-communication server destination).	VDN Status/VDN Report/VDN Summary Report: CALLS OFFERED
calls-waiting	The number of calls that have encountered a VDN, but have not been answered, abandoned, or outflowed.	VDN Status: CALLS WAIT
non-acd-calls-connected	The number of non-ACD calls routed from a specific VDN that were connected to an extension.	VDN Status/VDN Report/VDN Summary Report: CONN CALLS
oldest-calling-waiting	The time the oldest call has been waiting in the VDN. Timing begins when the call enters the vector.	VDN Status: OLDEST CALL
percent-in-service-level	The percentage of calls offered to the VDN that were answered within the service level administered for the VDN.	VDN Status/VDN Report/VDN Summary Report: % IN SERV LEVEL
total-acd-talk-time	The total time agents spent talking on split/skill calls and direct agent calls.	None
vdn-extension	The extension of a vector directory number (VDN).	VDN Status/VDN Report: VDN EXT
vdn-name	The name of a vector directory number (VDN).	VDN Status/VDN Summary Report: VDN NAME

Description of trunk group data types

The following table describes the trunk group data types.

VuStats data type	Description	BCMS report: field name/column heading
average-incoming-call-time	Average holding time for incoming trunk calls.	Trunk Group: INCOMING TIME
average-outgoing-call-time	Average holding time for outgoing trunk calls.	Trunk Group: OUTGOING TIME
incoming-abandoned-calls	Incoming calls abandoned during a specified time period for a specified trunk group.	Trunk Group: INCOMING ABAND
incoming-calls	Incoming calls carried by a specified trunk group.	Trunk Group: INCOMING CALLS
incoming-usage	The total trunk holding time for incoming calls in hundred call seconds.	Trunk Group: INCOMING CCS
number-of-trunks	The number of trunks in a specified trunk group.	Trunk Group: Number of Trunks
outgoing-calls	The number of outgoing calls carried by a specified trunk group.	Trunk Group: OUTGOING CALLS
outgoing-completed-calls	The number of outgoing calls that received answer supervision or answer timeout.	Trunk Group: OUTGOING COMP
outgoing-usage	The total trunk holding time for outgoing calls in hundred call seconds.	Trunk Group: OUTGOING CCS
percent-all-trunks-busy	The percent of time all the trunks in a specified trunk group were busy during a specified period/day. Timing for a call begins when the last trunk is seized.	Trunk Group: % ALL BUSY
percent-trunks-maint-busy	The percent of time trunks were busied out for maintenance during a specified period/day.	Trunk Group: % TIME MAINT
trunk-group-name	The name administered for a specific trunk group.	Trunk Group: Trunk Group Name
trunk-group-number	The number administered for a specific trunk group.	Trunk Group: Trunk Group Number

VuStats data type	Description	BCMS report: field name/column heading
trunks-in-use	The number of trunks currently in use (not idle).	None
trunks-maint-busy	The number of trunks currently busied out for maintenance.	None

Implementing the Time of Day Clock Synchronization feature

The Avaya communication server includes a Time of Day (TOD) Clock Synchronization feature which allows you to maintain synchronous clock times across a multi-site call center network. Maintenance of accurate TOD settings is important for many functions, such as:

- Creation of time stamps for items like error logs, Malicious Call trace records, Avaya BCMS and CMS data
- Scheduling of a large number of diverse task activities on the switch and its adjuncts

This section describes how to design and implement a TOD synchronization strategy that is appropriate for your call center operation. Topics include:

- [TOD synchronization methods](#) on page 397
- [Using NTP/SNTP to enable direct switch synchronization](#) on page 398
- [Using Avaya Site Administration to set up a TOD synchronization schedule](#) on page 399
- [About NTP/SNTP and Internet Time Servers](#) on page 410
- [Setting up ACD offset times for CMS reporting](#) on page 412

TOD synchronization methods

TOD clock synchronization capabilities are available for all Avaya switches running any software release that supports Multiple Locations and Daylight Savings rules. TOD clock synchronization is implemented by either of two methods. The method that you choose depends on the type of Avaya switch platform that you need to synchronize. The two methods are:

- [Using NTP/SNTP to enable direct switch synchronization](#) on page 398
- [Scheduling Time Synchronization tasks through Avaya Site Administration](#) on page 398

Using NTP/SNTP to enable direct switch synchronization

In this method, which is available only for certain Avaya communication servers, individual communication servers use either the Network Time Protocol (NTP) or Simple Network Time Protocol (SNTP) to synchronize their operating system (OS) clocks with highly accurate Coordinated Universal Time (UTC) from an Internet Time Server. The OS clock time is also used for the switch clock time.

This synchronization method is not described in detail in this document. An overview of the method, including information about the Avaya communication servers to which it applies, and where to find more detailed information, is provided in [Using NTP/SNTP to enable direct switch synchronization](#) on page 398.

This method of time synchronization applies to the following communication servers:

- Avaya S8100 Media Server
- Avaya S8300 Media Server
- Avaya S8700 Media Server

Scheduling Time Synchronization tasks through Avaya Site Administration

In this method, which is available only for certain Avaya communication servers, the Avaya Site Administration tool, is used to set up a regularly scheduled synchronization task for switches on a call center network. The time used to synchronize the communication servers is obtained from the system clock of the client PC. The recommended practice is to run NTP or SNTP software on the client so that it can be synchronized with UTC time obtained from an Internet Time Server.

This method of time synchronization applies to the following communication servers:

- DEFINITY G3csi
- Avaya DEFINITY Server CSI
- DEFINITY G3si
- Avaya DEFINITY Server SI
- DEFINITY G3r
- Avaya DEFINITY Server R

Requirements, considerations and procedures associated with this method are described in detail in [Using Avaya Site Administration to set up a TOD synchronization schedule](#) on page 399.

Using NTP/SNTP to synchronize the switch to UTC time

This method applies to certain Avaya switches on which the platform OS (Linux or Windows 2000) uses either the Network Time Protocol (NTP) or Simple Network Time Protocol (SNTP) to obtain highly accurate UTC data from an Internet Time Server. The time on the operating system clock, which is continuously adjusted to match polled UTC time, also provides the basis for the switch clock time. This synchronization method is accurate on the order of fractions of a second.

TOD clock synchronization is enabled on the following communication servers when Communication Manager is installed:

- IP600
- DEFINITY ONE
- Avaya S8100 Media Server
- Avaya S8300 Media Server
- Avaya S8700 Media Server

When a multi-site network includes switches that use this synchronization method, each of these switches maintains its own separate clock time. However, since all of the switches that use this method maintain settings based on UTC time, and thus essentially identical, clock synchronization is still achieved.

For requirements and procedures associated with this screen of clock synchronization, see *Administrator Guide for Avaya Communication Manager*.

For more information about NTP/SNTP software, see [About NTP/SNTP and Internet Time Servers](#) on page 410.

Using Avaya Site Administration to set up a TOD synchronization schedule

This section includes the following topics:

- [Description](#) on page 400
- [Prerequisites](#) on page 400
- [Things to know before you set up a synchronization schedule](#) on page 401
- [Designing a TOD clock synchronization schedule](#) on page 403
- [Creating dedicated switch connections](#) on page 408
- [Setting up a TOD synchronization task schedule in Avaya Site Administration](#) on page 408

Implementing the Time of Day Clock Synchronization feature

For more information about NTP/SNTP see [About NTP/SNTP and Internet Time Servers](#) on page 410.

Description

This method for TOD clock synchronization, which applies to certain Avaya communication servers, uses the Avaya Site Administration tool installed on a client PC to set up a synchronization task schedule.

In the recommended configuration, Avaya Site Administration is installed on the client PC along with NTP or SNTP software. The client PC is also connected to an Internet Time Server that it polls continuously to obtain UTC time for its system clock. The client clock time is then used to synchronize the clock time of switches on the network through the Avaya Site Administration Time Synchronization feature.

This method applies to the following communication servers:

- DEFINITY G3csi
- Avaya DEFINITY Server CSI
- DEFINITY G3si
- Avaya DEFINITY Server SI
- DEFINITY G3r
- Avaya DEFINITY Server R

On applicable communication servers that are installed with R10 or earlier, the synchronization command is ignored if the minute time specified for the incoming time is the same as that currently being counted at the switch. Consequently, this synchronization method is only accurate to within 59 seconds or less on communication servers that are installed with R10 or earlier.

On applicable communication servers that are installed with R11 or later, if the minute time specified for the incoming synchronization command is the same as that being counted at the switch, the minute count on the switch is set back to the 0-second mark for the minute. When potential network delays are factored in, this method is accurate to within 5 seconds or less for communication servers that are R11 or later.

Prerequisites

To implement a TOD clock synchronization schedule through Avaya Site Administration, the following prerequisite conditions must be met:

- Avaya Site Administration must be installed on the client, and it must also be running on the client when synchronization runs are scheduled to occur.

- The client PC must be able to establish a LAN or dialup connection to target communication servers when synchronization runs are scheduled to occur.

Important:

Before you set up TOD synchronization tasks in Avaya Site Administration, it is strongly recommended that you administer dedicated synchronization connections from Avaya Site Administration to each switch, as described in [Creating dedicated switch connections](#) on page 408.

- The client PC must have an appropriate SNTP/NTP software program installed, and be connected to an NTP Time Server over the internet. For more information, see [About NTP/SNTP and Internet Time Servers](#) on page 410.
- The client PC must be configured so that if Daylight Savings Time is in effect at the client location, the same rule also applies to the PC clock.

Things to know before you set up a synchronization schedule

You must understand the following details before you use Avaya Site Administration to set up a TOD clock synchronization schedule for switches on the network.

Specify offset values in Standard Time equivalents

When you set up a regular schedule for a time synchronization task in Avaya Site Administration, you specify an offset value that reflects the difference in local time between the client PC and a target switch location. Because of the way that Avaya Site Administration handles time synchronization settings, the derivation of offset values requires careful consideration.

Avaya Site Administration uses the `set time` command to synchronize the switch clock time to the clock time of the Avaya Site Administration client. Avaya Site Administration always sends the `set time` command in Standard Time. Depending on the switch software version, one of the following results occur:

- if the receiving switch is installed with R11 (or later), the switch checks the Daylight Savings Rule specified in the `set time` command (which is always Standard Time), and compares it to the existing rule in effect for the switch. If a Daylight Savings Rule other than Standard Time is in effect at the switch, the switch adjusts the incoming synchronization time as necessary.
- if the receiving switch is installed with R10 or earlier, the switch checks the Daylight Savings Rule specified in the `set time` command (which is always Standard Time), and compares it to the existing rule in effect for the switch. If a Daylight Savings Rule other than Standard Time is in effect at the switch, the switch sends an error message back to Avaya Site Administration. When Avaya Site Administration receives the error message from the switch, it automatically corrects the synchronization time to comply with the Daylight Savings Rule on the local PC, and resends the adjusted time back to the switch.

Implementing the Time of Day Clock Synchronization feature

When you calculate offset values to use as input in the Avaya Site Administration Time Synchronization feature, you must do the following:

- If either the client or target switches are located in a time zone where Daylight Savings Time rules are in effect, convert local times to reflect what the time would be if the Standard Time rule was in effect. The best practice is to always normalize switch and client times to Standard Time before you calculate offset values.
- After you normalize the location times to their Standard Time equivalents (if necessary), calculate the offset time as the difference between the local Standard Time at the client and the local Standard Time at the switch. This value is the offset between the PC client and switch that you specify when you use the Avaya Site Administration Time Synchronization feature.

For an example scenario that illustrates the offset calculation method, see [Designing a TOD clock synchronization schedule](#) on page 403.

Possible lag times

For possible lag between synchronization start times on the client and actual run times at the switch, you can set up multiple TOD clock synchronization tasks to run simultaneously on the client PC. However, synchronization tasks are actually run in sequence. Actual synchronization run times at the switch may vary from the run time specified on the client by several minutes. However, the accuracy of the synchronization setting is not affected.

EPN locations do not require synchronization

Expansion port network (EPN) cabinets that are located in a different time zone from their connecting switch do not require separate TOD clock synchronization. EPNs obtain the synchronized clock time from the switch, which is adjusted according to any settings that are specified in the `LOCATIONS` administration screen. You can use the `change multiple locations` command to access the `LOCATIONS` screen.

Note:

CMS data is synchronized to the switch time for the ACD from which CMS data is generated. Any Daylight Savings Time rules applied to the switch are also applied to the CMS data.

Run synchronization tasks during low-traffic periods

Avaya Site Administration uses the `set time` command to adjust the switch clock time. Since completion of the `set time` command can be delayed by heavy switch traffic, the clock synchronization task should be scheduled to run during a low-traffic period on each switch.

Run synchronization tasks in the middle of CMS archive intervals

Avaya CMS is designed to run archives at regular intervals of 15 minutes, 30 minutes, or 1 hour. Synchronization tasks should be scheduled so that they run near the middle of an archive interval. This approach minimizes potential redundancies in archive interval records for an ACD.

Consider a case for an archive interval that begins at 09:00. If a synchronization command is received at the switch and changes the switch clock time to 08:59, a second archive interval begins when the switch clock changes to 09:00 again. In this case, two archive intervals that have the same 09:00 time stamp are recorded. One interval extends from 09:00 to 09:01. The other interval also begins at 09:00, and extends for the normal duration that is specified for archive intervals on the ACD, which is either 15, 30, or 60 minutes.

To avoid situations such as that described above, always schedule synchronization tasks to be run near the middle the archive interval specified for each ACD.

To determine the CMS archive interval length specified for an ACD:

1. Start Avaya CMS Supervisor.
2. From the main menu, select:

Tools > System Setup

The **CMS System Setup** dialog is displayed.

3. Do the following:
 - a. In the **Operations** list, select **Storage Intervals**
 - b. From the **ACD:** option box, select an ACD for which you want to obtain archive interval information.
4. Select **OK**.

Supervisor displays the **Storage Intervals** window, which includes the archive interval for the ACD.

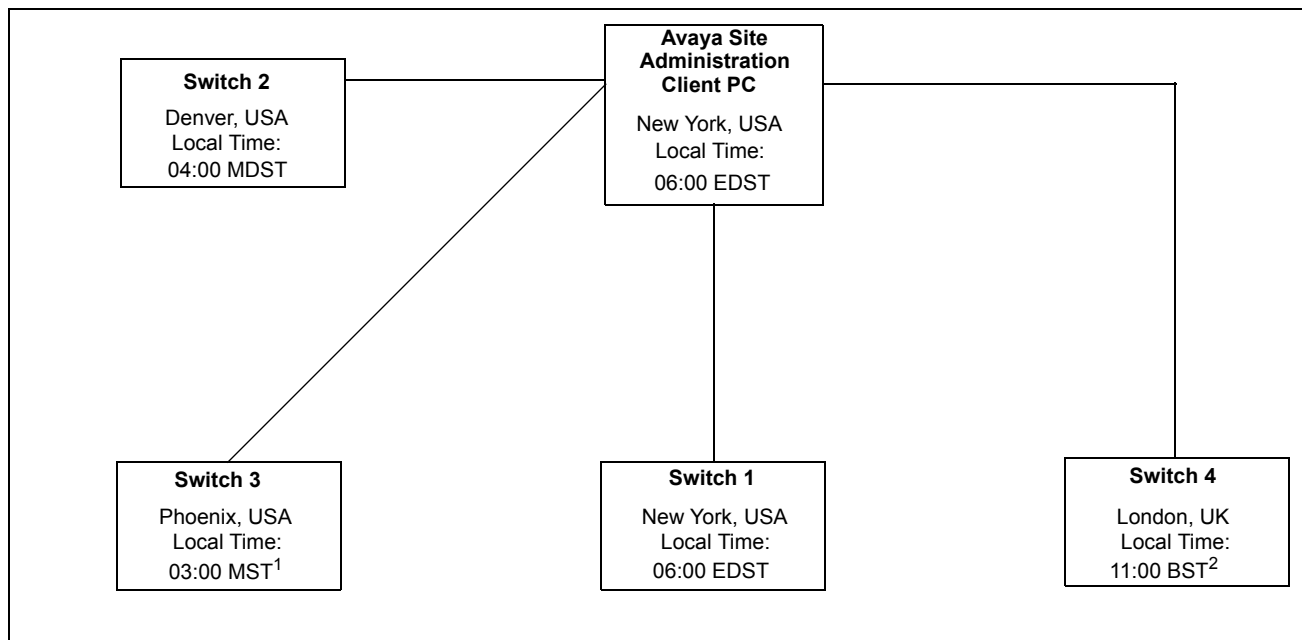
Designing a TOD clock synchronization schedule

This section includes the following topics:

- [Example multi-site call center network](#) on page 404
- [Determining location offset values](#) on page 404
- [Determining synchronization run times](#) on page 406
- [Special considerations for synchronization start times](#) on page 408

Example multi-site call center network

The following example scenario shows how to design an Avaya Site Administration Time Synchronization schedule for a multi-site call network that spans several time zones. The example sites and their respective time zones are shown in the following figure.



Notes:

1. Site is not on Daylight Savings Time
2. British Summer Time (BST) is UK Daylight Savings Time (GMT + 1)

The call center network example in this figure includes four switch locations that are located in different time zones. In this example, all switches and the client are on Daylight Savings Time, with the exception of Switch 2 (Phoenix).

Using Avaya Site Administration to create a TOD synchronization schedule requires careful planning and consideration. The steps described below for the example multi-site scenario represent the most error-free method you can use to design your synchronization schedule.

Determining location offset values

The Avaya Site Administration Time Synchronization feature sends synchronization messages to a switch that are specified in Standard Time using the `set time` command. To calculate offset values that represent the time difference between the Avaya Site Administration client PC and switch locations, all location times should be normalized to their Standard Time equivalents.

⚠ Important:

Always calculate offset values based on comparisons between Standard Time equivalents. Otherwise, if Daylight Savings Time rules are not the same for the Avaya Site Administration PC client location and a target switch location, significant synchronization errors can result.

The following table uses the switch locations described in the example scenario to derive correct offset values for the client PC and switch locations.

Note:

The local times listed in the table are arbitrary in nature, and are intended only to illustrate the time differences between locations. You can use any set of relative location times for this purpose.

Local Time / Normalized Standard Time		Calculated offset value
Avaya Site Administration Client PC (New York)	Switch 1 (New York)	
Local time: 06:00 EDST	Local time: 06:00 EDST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 05:00 EST	0
Client PC (New York)	Switch 2 (Denver)	
Local time: 06:00 EDST	Local time: 04:00 MDST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 03:00 EST	-2
Client PC (New York)	Switch 3 (Phoenix)	
Local time: 06:00 EDST	Local time: 03:00 MST	
Adjusted Standard Time: 05:00 EST (New York)	Adjusted Standard Time: 03:00 MST	-2

Implementing the Time of Day Clock Synchronization feature

Local Time / Normalized Standard Time		Calculated offset value
Client PC (New York)	Switch 4 (London)	
Local time: 06:00 EDST	Local time: 11:00 BST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 10:00 BST	+5

The table shown above demonstrates the importance of normalizing all local times to Standard Time when calculating time offsets. The unadjusted time difference between Denver and New York is 2 hours, and the calculated offset is minus 2 hours.

However, despite the fact that the unadjusted time difference between New York and Phoenix is 3 hours, the calculated offset is also minus 2 hours - the same offset value that is calculated for New York and Denver.

Determining synchronization run times

After you have calculated accurate offset values, you can determine when the synchronization task should be scheduled to run on the Avaya Site Administration client PC. Again, the most reliable method is to normalize all local times to Standard Time equivalents.

The following table shows how to obtain to obtain synchronization task run times from the switch locations used in this example. The task run times represent the values that would be entered in the **Scheduler** dialog in Avaya Site Administration.

Note:

You should always establish clock synchronization run times on the basis of low-traffic time intervals for the switch. In the following example, a single local switch time is used as the synchronization run time for all of the switches. In actual practice, low-traffic periods for switches in a multi-site network may not always be the same for each switch location.

Using Avaya Site Administration to set up a TOD synchronization schedule

Local Time / Normalized Standard Time for synchronization at switch location ¹	Client / Switch Offset ²	Synchronization run time set on client PC ^{3,4}
Switch 1 (New York)		
Local synchronization run time: 03:07 EDST		
Adjusted Standard Time: 02:07 EST	0	2:07 AM
Switch 2 (Denver)		
Local synchronization run time: 03:00 MDST		
Adjusted Standard Time: 02:07 EST	-2	0:07 AM
Switch 3 (Phoenix)		
Local time: 03:07 MST		
Adjusted Standard Time: 03:07 MST	-2	1:07 AM
Switch 4 (London)		
Local time: 03:07 BST		
Adjusted Standard Time: 03:07 BST	+5	8:07 AM

1. Synchronization run times should occur during low traffic periods at the switch. In this example, the simplifying assumption is made that a single low-traffic time (at the switch) is common to all switch locations. This assumption may not be true for all call center operations.
2. Listed offset values are those that were derived in [Determining synchronization run times](#) on page 406.
3. If the PC client clock is currently set to DST rules, add 1 hour to the synchronization start time that you specify in the Avaya Site Administration **Schedule** dialog. If the PC client and switch locations do not use the same DST rules, see [Special considerations for synchronization start times](#) on page 408.
4. Run times entered in the **Schedule** dialog of the Avaya Site Administration Time Synchronization feature must be specified in 12-hour, AM/PM time format.

Special considerations for synchronization start times

If the PC client and switch locations do not use the same DST rules, actual synchronization run times will vary by an hour over the course of the year. Depending on how DST rules between the switch and client PC vary, the actual synchronization run time at the switch will occur either one hour earlier or later than the time specified in Avaya Site Administration. The following basic rules apply:

Note:

The exceptions described below relate to synchronization start times, only. As long as offsets are calculated correctly, as described in [Determining location offset values](#) on page 404, the start time exceptions listed below have no effect on the accuracy of the synchronization.

- When DST is in effect at the switch, but not at the PC client, the actual synchronization run time at the switch will occur 1 hour later than the run time that is specified on the client PC.
- When DST is in effect at the PC client, but not at the switch, the actual synchronization run time at the switch will occur 1 hour earlier than the synchronization run time that is specified on the client PC.

Creating dedicated switch connections

Before you set up a TOD synchronization task, or any other kind of automated tasks, you should create dedicated connections from Avaya Site Administration to each target switch.

Setting up a TOD synchronization task schedule in Avaya Site Administration

To set up a TOD clock synchronization schedule for a switch on your call center network:

1. Start Avaya Site Administration on the client PC, and select a switch from the pull-down list on the main toolbar.
2. In the browser pane located to the right side of the main application window, select the **Fault & Performance** tab.
3. Select the **Time Synchronization** option.
The **Time Synchronization - Properties** dialog is displayed.
4. In the **Time Synchronization - Properties** dialog:
 - If the target switch is located in a different time zone, check the **Offset** option, and specify the time offset between the client PC and the switch.

 **Important:**

When different Daylight Savings Time rules are in effect at the Avaya Site Administration client location and a target switch location, synchronization errors can result if you do not calculate offset values based on Standard Time equivalents. For a description of the method used to calculate accurate offset values, see [Determining synchronization run times](#) on page 406.

Note the offset factor that you specify, which is also used in Step 6.

- Click **Next**.

The **Time Synchronization - Schedule** dialog is displayed.

5. In the **Time Synchronization - Schedule** dialog:

- Check the **Schedule this task to run** option.
- Click the **Schedule** button.

The **Scheduler** dialog is displayed.

6. Do the following in the **Scheduler** dialog:

 **Important:**

The synchronization task should be scheduled to run during a low-traffic period at the switch. Execution of the synchronization command can be delayed by heavy switch traffic.

- a. In the **Date** field, click the arrow to pull down the calendar, and select a day on which the synchronization task will start.
- b. In the **Time** field, enter the time of day that you want the synchronization task to run. The time you specify in this field is the PC client time, and not the time at the target switch. To determine the correct time to enter in the **Time** field, do the following:
 1. Determine what the local time will be at the switch when the synchronization runs. If necessary, adjust this time to its Standard Time equivalent, as described in [Determining synchronization run times](#) on page 406.
 2. Subtract the offset factor that you used in Step 4 from the switch run time that you derived in the preceding substep. The time you calculate is the run time on the client expressed in Standard Time. If Daylight Savings Time is in effect at the client PC, increase the time by 1 hour to account for Daylight Savings Time. For more information, see [Determining location offset values](#) on page 404.

Also, if the client PC and the target switch time use the same DST rules, see [Special considerations for synchronization start times](#) on page 408.
 3. Enter the calculated **Time** field.
- c. Select a **Recurrence Pattern** option (Frequent, Weekly or Monthly) and provide the time parameters specified with that option.

Note:

If you select the **Frequent** option, the recommended practice is to set the task to run at 24 hour intervals.

d. Click **OK**.

The **Time Synchronization - Schedule** dialog is displayed again.

e. Verify the synchronization schedule information that you provided and click **Next**.

The **Time Synchronization - Summary** window is displayed.

f. Click **Finish**.

7. Repeat Steps 1 through 6 for any other switches that need to be synchronized using this method.

About NTP/SNTP and Internet Time Servers

This section includes the following topics:

- [Description of NTP/SNTP and Internet Time Servers](#) on page 410
- [SNTP on switch platforms that support direct synchronization](#) on page 411
- [Platforms that synchronize through an Avaya Site Administration client PC](#) on page 412

Description of NTP/SNTP and Internet Time Servers

The Network Time Protocol (NTP) synchronizes the system time on a computer to that of an Internet Time Server that has been synchronized to a reference source, such as radio, Global Positioning Service (GPS) receiver that provides Coordinated Universal Time (UTC). Communication with the Internet Time Server is maintained either by a dialup modem or direct LAN connection.

The Simple Network Time Protocol (SNTP) is a basic version of NTP that allows for a greater degree of error, but can still deliver time to an accuracy on the order of fractions of a second.

SNTP on switch platforms that support direct synchronization

The following Avaya switch platforms can use SNTP software on the platform operating system to directly synchronize the switch clock to UTC time that is obtained from an Internet Time Server:

- Avaya IP600
- DEFINITY One
- Avaya S8000 Media Server
- Avaya S8300 Media Server
- Avaya S8700 Media Server

The platforms listed above include either Red Hat Linux or Windows 2000 as the platform operating system. The following recommendations and conditions are in effect for SNTP configuration on these systems:

- For Linux platforms, Avaya recommends that the IP addresses for at least three different Internet Time Servers be configured. The following web site provides a list of time servers:

<http://www.eecis.udel.edu/~mills/ntp/servers.htm>

- For Linux platforms, go to the following web sites for information about how to obtain NTP/SNTP software:

<http://www.ubr.com/clocks/timesw/timesw.html>

<http://www.ntp.org/software/index.html>

- Linux platforms support the authentication/encryption mode provided in NTP/SNTP version 3 or later support. This capability is not enabled by default.
- Windows 2000 platforms can use only one SNTP client (W32Time), which limits UTC polling to one Internet Time Server IP address at a time. For more information, see the Microsoft Windows 2000 documentation. Search for the keywords *Window Time Service*.
- Windows 2000 platforms use the W32Time service for SNTP functions. This service does not support an authentication/encryption mode for the SNTP protocol.
- W32Time service allows optional polling of a Microsoft network domain controller as the primary time server. Avaya does not support this configuration.

Platforms that synchronize through an Avaya Site Administration client PC

The following Avaya switch platforms must use the Avaya Site Administration Time Synchronization feature to maintain switch synchronization:

- DEFINITY G3csi
- DEFINITY G3si
- DEFINITY G3r
- Avaya S8100 Media Server
- Avaya S8200 Media Server
- Avaya S8500 Media Server

The following web site provides a list of time servers:

<http://www.eecis.udel.edu/~mills/ntp/servers.htm>

The following web sites provide information about how to obtain NTP/SNTP software for the Avaya Site Administration client PC:

<http://www.ubr.com/clocks/timesw/timesw.html>

<http://www.ntp.org/software/index.html>

Setting up ACD offset times for CMS reporting

This section includes the following topics:

- [About setting up ACD offset times for CMS reporting](#) on page 412
- [Offset procedure](#) on page 413

About setting up ACD offset times for CMS reporting

The time stamp for CMS data is obtained from the local switch on which the data is generated. When a CMS system includes ACDs that are located in different time zones, time zone differences are reflected in CMS reports based on unadjusted data. However, you can use Avaya CMS Supervisor to adjust CMS data derived from remote ACDs in a way that allows you to view data from different time zones in a common time format. This provides you with a more convenient way to view and assess simultaneous call center activity across time zones.

To adjust CMS data to reflect a common time format, you must:

- Designate a master ACD
- Determine the appropriate offsets for each remote ACD, as necessary
- Set the switch time zone offset values for each ACD in the CMS Supervisor **Storage Windows** dialog.

Offset procedure

To use Supervisor to set switch time zone offset values for CMS report times:

1. In the main Supervisor Controller window, select:

Tools > System Setup

The **CMS System Setup** Window is displayed.

2. In the **CMS System Setup** window, do the following:
 - a. Select the **Operations** tab.
 - b. From the displayed list, select **Storage Intervals**.
 - c. In the **ACD** field, select an ACD.
 - d. Select **OK**.

The **Storage Intervals** window is displayed.

3. In the **Switch time zone offset (-23 to +23)** field, enter a an offset value that reflects the time difference between the target ACD and the designated master ACD.

Note:

For instructions on specifying the master ACD, see *Avaya CMS Administration*.

4. From the main menu, select:

Actions > Modify

5. Repeat the procedure for any other ACDs for which a switch time zone offset is required.

Appendix A: Recorded announcements

The Recorded Announcement feature provides an announcement to callers under a variety of circumstances. For example, announcements let callers know that their call is in queue or that the lines are busy.

This appendix gives you extended information about using the recorded announcement feature of the communication server. The main topics in this appendix are:

- [Administering recorded announcements](#) on page 415
- [Recorded announcement types](#) on page 417
- [When to use recorded announcements](#) on page 420
- [About barge-in](#) on page 421
- [Integrated announcements and announcements recorded on external devices](#) on page 422
- [Procedures for recording announcements](#) on page 423
- [Recorded announcements with features](#) on page 427
- [Locally-sourced music and announcements](#) on page 427

Related topics

- For information about administering announcements, see *Administrator Guide for Avaya Communication Manager*.
- For information about adding, recording, saving, copying, restoring, and deleting announcements, see *Feature Description and Implementation for Avaya Communication Manager*.

Administering recorded announcements

The following table lists the forms that you will use to administer announcements.

Screen	Field
Announcements/Audio Sources (includes Integrated Announcement Translations)	All
Feature Access Code (FAC)	Announcement Access Code

Appendix A: Recorded announcements

Screen	Field
Station	COS
Data Modules (for Save/Restore/Copy) Netcon Data Module System Port Data Module (SAP) Announcement Data Module	All
Circuit Packs ¹	All
Feature-Related System Parameters ²	DID/Tie/ISDN Intercept Treatment Controlled Outward Restriction Intercept Treatment Controlled Termination restriction (Do Not Disturb) Controlled Station-to-Station Restriction
Hospitality ²	Announcement Type Length of Time to Remain Connected to Announcement
Trunk Groups (All) ²	Incoming Destination
Coverage Path ²	Coverage Points
Hunt Group ²	First Announcement Extension Second Announcement Extension
Call Vector ²	All fields that require announcements

1. You only need to complete the Circuit Pack screen if you administer the Board Location on the Announcements/Audio sources screen or Data Module screen and do not have the circuit pack plugged in.
2. You only need to complete this screen if you plan to use Recorded Announcements with the screen's associated feature. For example, if you want to use announcements with the Hospitality features, you need to complete the Hospitality screen.

Recorded announcement types

The communication server supports several recorded announcement types. Each support announcement type is described in this section.

This section includes the following topics:

- [Analog line types](#) on page 417
- [DS1 types](#) on page 418
- [Auxiliary trunk types](#) on page 419
- [Integrated types](#) on page 419

Analog line types

External announcement machines for recorded announcements can be interfaced using one of the analog line types. The external announcement machine can then be connected by an analog line port.

Analog

The analog announcement type provides an analog telephone interface using an analog line port for use with an announcement/audio source device that emulates analog telephones. The communication server starts playback by applying ringing; the device indicates playback has stopped by going on-hook (opening the loop). The communication server does not indicate to the device to stop playback. Use the analog type for announcements that play for a specific period and then go on-hook at the end. When the device goes on-hook to indicate that the playback ended, the caller listening to the announcement hears a click. (See ds1, aux-trk, or integrated types for alternative types).

Analog-fd

Like the analog type, analog-fd provides an analog line interface and ringing starts the playback. However, a forward disconnect signal (open loop for about one-half second) is sent to the device to stop playback when there are no callers left to hear it.

Analog-m

Like the analog type, analog-m provides an analog line interface. However, ringing is not applied to start playback. Use this type for continuous playing music or audio sources. The device stays in an off-hook state when active and goes on-hook when it is not playing, is turned off, or is disconnected. This announcement type is used when the `Q` field is set to `b` to provide barge-in repeating or continuous-play announcements.

DS1 types

The DS1 types provide analog-like interfaces with DS1 line ports, which are called Line Side DS1 or Line Side T1. Each of these types indicate to the announcement, music, or audio-source device to start playback using the Line Side T1 equivalent of ringing. The DS1 types also expect off-hook from the device to indicate that the playback is active and on-hook to indicate that the playback is not active.

The `ds1-id` and `ds1-sa` types provide a forward disconnect using transitions of the A signaling bit to the device, which indicates when playback should be stopped. Callers listening to announcements do not hear clicks when the device disconnects (goes on-hook).

`ds1-fd`

The `ds1-fd` announcement type provides a TIA/EIA Foreign eXchange (FX) type DS1 interface. The forward disconnect signal is a toggle of the A bit from 0 to 1 and then back to 0 after 600 msec. This type is used for Line Side T1 ports on the IVR system when they are used as an analog-like announcement device and is the recommended method for interfacing.

`ds1-sa`

The `ds1-sa` announcement type provides a TIA/EIA special-access type DS1 interface. The forward disconnect signal is a toggle of the A bit from 1 to 0 and then back to 1 after 600 msec.

`ds1-ops`

The `ds1-ops` announcement type provides a TIA/EIA off-premises-station type DS1 interface that is used when the device does not support forward disconnect.

Auxiliary trunk types

The Auxiliary Trunk announcement type supports an external announcement machine connected using a 4-wire auxiliary trunk interface, such as a 15A announcement system. The communication server indicates to the device to start or stop the playback on the S lead; the device indicates that the playback is active on the S1 lead.

aux-trunk

Use the aux-trunk (auxiliary trunk) announcement type with a 4-wire interface external device when the playback is to be stopped and started by way of the S1 lead and S1 is used by the device to indicate playback started.

aux-trk-m

Use the aux-trk-m (auxiliary trunk music) with a 4-wire interface device for continuously playing music or audio sources that do not indicate that playback is active on the S1 lead. This announcement type is used when the `Q` field is set to `b` to provide barge-in repeating or continuous-play announcements

Integrated types

The integrated announcement type stores announcements internally on the communication server on an Integrated Announcement circuit pack or embedded gateway processor equivalent. This can include TN2501AP Voice Announcements with LAN (VAL), or an H.248 Media Gateway VAL source.

The TN2501AP circuit pack has 31 ports that are available for playing announcements. The G700 MG VAL source has 15 play ports while the G350 MG VAL source has 6. Integrated announcement sources are recommended for VDN of Origin Announcements and for other general and ACD announcement needs.

The following announcement boards or sources are obsolete and not supported in Communication Manager configurations:

- TN750, TN750B, or TN750C announcement boards
- Co-resident SSP sources (DEFINITY One or S8100)

integrated

Use the integrated announcement type for announcements that are stored on the communication server. This announcement type is recommended for general, ACD, and vectoring announcements and for VDN of Origin Announcements.

integ-rep

The integ-rep (integrated-repeating) announcement type is used to provide integrated, repeating automatic wakeup announcements and is implemented along with the multi-integ hospitality announcement type setting. This type can also be used for call center applications in vectoring where a continuous repeating announcement is required.

integ-mus

The integ-mus announcement type is the same as the integ-rep type except that the Q field is always set to *b* to provide a continuous repeating barge-in operation. This type is typically used to provide music on delay or on hold.

For more information, see [Capabilities of locally-sourced music and announcements](#) on page 429.

When to use recorded announcements

The most common applications for recorded announcement include:

- DID calls cannot be completed as dialed.
- Incoming private-network access calls cannot be completed as dialed.
- Calls enter a split or skill (first announcement).
- DDC, UCD, or direct-agent calls have been in queue for an assigned interval.
- ACD and Call Vectoring calls have been in queue for an assigned interval.
- A call's destination is a recorded-announcement extension.
- A call routes to a vector that contains an announcement step.
- An announcement extension is specified as a coverage point.
- An announcement is the incoming destination of a trunk group.
- A VDN of Origin announcement has occurred.
- A Security violation notification has occurred.
- The Hospitality Automatic Wakeup feature is in use.

About barge-in

Normally, the system connects multiple callers to the beginning of an announcement, regardless of announcement type. However, you can also administer auxiliary trunk announcements, DS1 announcements, and integrated announcement to allow callers to begin listening to an announcement after the system has begun playing its message. This capability is called barge-in.

This section includes the following topics:

- [Barge-in operational details](#) on page 421
- [Non-barge-in operational details](#) on page 421

Barge-in operational details

When you administer barge-in by setting the `Q` field to `b`, only one port plays the announcement at any one time. When the system routes a call to that announcement, the call immediately connects to the port and the caller hears the announcement as it is playing. Most administrators administer barge-in announcements to repeat continually while callers are connected to the port. In this way, the caller listens until the system plays the entire announcement.

Non-barge-in operational details

If an announcement port is available when a call arrives, the system connects the call to the announcement.

If an announcement port is not available and the announcement is administered with *no* as the queue option, the call does not enter the queue for the announcement and the caller hears busy or other feedback, depending upon how the announcement was accessed.

If an announcement port is not available and the announcement is administered with *yes* as the queue option, the call enters the announcement queue. When a port becomes available, the communication server connects the calls waiting in the queue to the beginning of the announcement. The system first connects the call that has been waiting in queue the longest and then connects as many calls as it can.

Integrated announcements and announcements recorded on external devices

Recorded Announcement allows you to administer either integrated announcements or announcements recorded on external devices. The external devices connect to the communication server using analog line circuit packs or auxiliary trunk interfaces, such as a TN2183 or a TN763.

The system stores an integrated announcement on a VAL source (TN2501AP or H.248 MG sources). The system can store multiple announcements on each circuit pack up to the system capacity.

Each TN2501AP integrated-announcement circuit pack allows up to 1 hour of uncompressed voice storage, has 31 playback ports (can play up to 31 simultaneous announcements) and 1 dedicated port for telephone access to be used for recording and playback. The recording time and playback ports for the H.248 MG VAL sources vary by Media Gateway. The TN2501AP and H.248 MG VAL sources are connected to the customer LAN to enable announcement file transfers to take place by way of FTP with a computer. Announcements are wave files that are recorded as CCITT u-law/a-law, 8kHz, 8-bit mono files using a utility such as Microsoft's Sound Recorder on a computer or using an Avaya telephone.

Any announcement stored on a a circuit pack can play through any port on the circuit pack. Any announcement (not administered for barge-in) can be played through multiple ports. For instance, all 31 ports on the TN2501AP can play the same announcement at the same time, each started at a different time.

VAL sources

Type	Storage time	Playback ports
TN2501AP	60 minutes	31
G700 MG	20 minutes	15
G350 MG	10 minutes	6

You must set the Q field to y on the Announcements/Audio Sources screen for each extension that you want to queue for Integrated Announcements. Calls that hear integrated announcements at extensions that have queue assigned only queue when all of the ports on the source that contains the announcement are busy. When a port becomes available, all callers queued to hear a specific announcement up to the maximum supported by the server platform are simultaneously connected to that port to hear the announcement from the beginning. The same queueing pool is used over all integrated sources. The communication server controls the announcement queue length for integrated announcements, but you must set the queue length for analog or aux-trunk announcements.

Procedures for recording announcements

This section includes the following topics:

- [About these procedures](#) on page 423
- [Recording the announcement](#) on page 424
- [Stop recording the announcement](#) on page 425
- [Playing back the announcement](#) on page 425
- [Deleting the announcement](#) on page 426

About these procedures

You can transfer to and from a computer or delete announcement files over the LAN for the TN2501AP and H.248 MG VAL sources using the Voice Announcement Manager (VAM) software or using an FTP client in conjunction with SAT commands.

Announcements for the VAL sources can also be recorded with a telephone using the procedures discussed in this section. For additional information, see *Feature Description and Implementation for Avaya Communication Manager*.

Use these procedures to record announcements on the TN2501AP circuit packs, as well as for G700 and G350 Gateway embedded Media VAL announcements (referred to herein as virtual VAL or VVAL).

For more information and recording tips, see the section about announcement commands in *Avaya Call Center Call Vectoring and EAS Guide*.

About announcement sessions

You can record, play back, or delete integrated announcements by initiating an announcement session. To do this, you must have console permissions assigned to your Class of Service (COS) for the internal station or Remote Access barrier code in order to initiate an announcement session.

Announcement recording

With the VAL announcement sources, recording by telephone always uses port 1, which is dedicated for telephone access with these sources. VAL announcement sources also support recording announcements as .wav files either on a local PC or made by a professional recording studio. The files are moved onto the VAL source using FTP.

Note:

You cannot use a telephone to record an announcement with an audio group assignment. Using FTP, move each pre-recorded file to each of the sources defined for the audio group.

For more information, see *Administrator Guide for Avaya Communication Manager*.

The announcement session process

To begin an announcement session, the user must dial the administered feature access code (FAC) followed by the announcement extension. If an announcement session is already in progress, the user hears reorder tone (fast busy) and the system drops the call.

If the telephone session port to an integrated board is in use, then the user hears reorder tone followed by silence. This indicates that the port will be reserved for an announcement session. The user should redial the FAC and extension every 45 seconds to gain access to the port.

Note:

For Communication Manager Release 1 or later, multiple telephone sessions are allowed with one session associated with each active integrated announcement board.

Once a telephone user accesses an announcement session, the user can dial 1 to record an announcement, 2 to play an announcement, or 3 to delete an announcement. If the circuit pack memory is more than 90% full, then the communication server gives stutter dial tone when the user gains access to an announcement session. Even if the user hears stutter tone, the user can begin speaking to record the announcement.

Note:

Avaya recommends that you use a digital telephone. For more information, see [Stop recording the announcement](#) on page 425.

Recording the announcement

If you dial 1, the communication server attempts to start a recording session and you will receive one of the following outcomes:

- If an announcement already exists and is protected (designated as protect = y), you will hear an intercept tone. Hang up and determine the correct announcement extension to use.
- If the announcement is currently being played to callers, then you will hear the reorder tone.
- If the communication server has started the recording session, then you will hear a record tone and can begin recording the announcement.

Stop recording the announcement

Depending on the type of phone you are using, use one of the following methods to stop the recording after the announcement is complete:

If you are using a hybrid, digital telephone, or IP telephone: Dial # to end the recording. Ending the recording with a # returns you to the dial tone, allowing a playback, delete, or record over operation to be requested. The # tones or a click sound produced when you hang up are not recorded. If the circuit pack memory becomes full during recording, you will hear a reorder tone, the system will drop you, and the announcement is not retained.

If you are using an analog telephone: Hang up. Otherwise, ending with a # puts the tone in the message. If you are using an analog telephone that is not connected with lineside T1 (DS1 type), the system records a click when you hang up. After hanging up, you must redial the FAC plus announcement extension to start a new recording session. If the circuit pack memory becomes full during recording, you will hear a reorder tone, the system will drop you, and the announcement is not retained.

Playing back the announcement

After you complete a recording and hang up, do not immediately dial the extension. The new announcement remains busy for approximately 15 seconds. The new announcement can be played back by dialing the FAC, the announcement extension, and 2 before the 15 second timer expires.

Upon completion of the recording session (drop), the communication server sets a 15-second timer. During this interval, the system restricts you to one of two tasks:

- Listen to the announcement just recorded
- Record another announcement

If you want to listen to the announcement before it is available to others, then dial the FAC, the extension, and 2. The announcement plays and then generates dial tone. You can then perform another operation, such as record a message.

Note:

If a caller attempts to dial an announcement that does not exist, either because it has not been recorded or it has not been sent over FTP to the board, the caller hears silence.

Deleting the announcement

If you dial the FAC, the extension, and then 3, the communication server deletes the announcement and you hear a confirmation tone. If the announcement is protected or is currently being played, then the system does not delete the announcement and you will hear a reorder tone. Avaya recommends that you delete a recorded announcement before re-recording it with the dial 1 function.

Recorded announcements with features

This section includes the following topics:

- [Recorded announcements, the ACD, and other call center features](#) on page 427
- [Recorded announcements and automatic wakeup](#) on page 427

Recorded announcements, the ACD, and other call center features

Recorded announcements are used extensively for ACD, Call Vectoring, Call Prompting, Expert Agent Selection, VDN of Origin Announcement, Direct Department Calling, and UCD features. See the individual features for interaction details.

Recorded announcements and automatic wakeup

Recorded announcements allow Automatic Wakeup to use the built-in integrated announcement circuit pack or sources in place of the Audichron adjunct.

If you use an integrated, multiple integrated, or external type of announcement for Automatic Wakeup, then you can also administer the announcement to repeat (with the integ-rep announcement type) and to allow barge-in as a queue type. The benefit of repeating announcements and barge-in queues is that you do not need to use a separate port for each wakeup announcement. When guests go off-hook to receive an announcement at a particular time, they use only one port and the message repeats on the port until the last guest goes off-hook and the message ends.

Locally-sourced music and announcements

This section includes the following topics:

- [Definitions](#) on page 428
- [About locally-sourced music and announcements](#) on page 428
- [About audio groups](#) on page 428
- [How the algorithm determines the most local source of an audio group](#) on page 429
- [Capabilities of locally-sourced music and announcements](#) on page 429

Definitions

Term	Definition
VAL announcement source	A Voice Announcement with a TN2501AP board.
vVAL announcement source	A vVAL source integrated in a Media Gateway (G700, G250, G350, and so on). The vVAL source is referred to as a virtual VAL source or an embedded VAL source.
announcement file	The recorded announcement file that is played for the specific announcement extension assigned to the audio group. The announcement file is file-transferred (FTP) with the same filename into each of the sources listed for the audio group.

About locally-sourced music and announcements

This feature allows call centers to use any or all of their VAL or vVAL sources in the gateways as sources for the same announcement. A locally-sourced music and announcements architecture:

- Improves the quality of the audio
- Reduces resource usage - for example, VoIP resources, by selecting the nearest available source when playing the announcement
- Provides backup for announcements because a working announcement source with the same announcement file can be selected from the sources if the primary announcement source is not available

About audio groups

The VAL or vVAL sources that contain a particular announcement file are assigned to an audio group - for example, G1. The audio group is then assigned to the announcement or audio extension *port location* as a group-sourced location instead of as a single-sourced location. When the announcement or audio is played to a caller, one of the sources assigned to the group - for example, 025V9 - that is more local to the incoming call trunk facility is selected to play the announcement file.

How the algorithm determines the most local source of an audio group

The algorithm that determines the most local source of an audio group works in the following priority order. The non-working sources are skipped so that a working source in the group can be found. The audio group algorithm:

1. Selects the local source. This is the source local to the trunk or user in the same Media Gateway or in the same group of TDM- or ATM-connected Port Network Gateways (PNGs).
2. Selects a source in a gateway in the same network region.
3. Selects one of the following sources in a gateway in the interconnected network region:
 - Adjacent with the most available bandwidth and highest quality codec
 - Non-adjacent based on the shortest number of hops
4. Selects an Inter-Gateway Alternate Routing (IGAR)-connected source on a gateway interconnected through a PSTN trunk. This selection is based on IGAR % usage.

If the announcement file extension is administered for queuing and the source selected to play that file has no available playback ports, the request to play that announcement is held in queue until a port on the source becomes available. With Call Vectoring, the caller hears the previously-started feedback - such as ringback - until the connection is made to the announcement. If queuing has not been assigned to that announcement file extension, the search continues using the above criteria.

Capabilities of locally-sourced music and announcements

This feature has the following capabilities:

Use single-sourced or group-sourced recorded announcement extensions as Music On Hold (MOH) sources : The integ-mus announcement type can be used to:

- Use single-sourced or group-sourced recorded announcements as MOH sources
- Assign the integ-mus announcement type to a system MOH source instead of to a port location as the system MOH source
- Use integ-mus announcements to play MOH and music in vectors. As with announcements with an audio-group assignment, when the music is played to a caller, the most local available source is used to play the music file. The integ-mus announcement type provides a repeating barge-in operation by combining the integrated repeating type with a forced assignment of barge-in. To assign MOH sources, use the music sources screen or the system-parameters feature-related screen if you are not using tenant partitioning.

Create separate MOH groups with multiple analog or Auxiliary Trunk music source port locations: You can create separate MOH groups that can assign multiple analog or Auxiliary Trunk music source port locations. You can assign these MOH groups - for example, group 1, as system MOH sources instead of a single port location on the music sources screen. When the music is given to the caller, the most local music source of the assigned group is selected.

Use audio groups anywhere a single-sourced announcement or audio source extension is used: You can use an announcement or audio source extension with an assigned audio group anywhere a single-sourced announcement or audio source extension can be used. For example, you can use a group-sourced announcement extension of type *integrated* or *integ-mus* in any of the following Call Vectoring vector commands:

- `announcement xxxxxxxx`
- `collect ... after announcement xxxxxxxx`
- `disconnect after announcement xxxxxxxx`
- `wait-time <time> [secs, mins, hrs] hearing [audio source ext] then [music, ringback, silence, continue]`

Apply a partition-defined system music source as the system music: You can apply a partition-defined system music source with an MOH group or music audio group-sourced extension of type *integ-mus* as the system music using the following commands:

- `wait-time <time> [secs, mins or hrs] hearing music`
- `wait-time <time> [secs, mins or hrs] hearing [audio source ext] then music`

Related topics

For more information about administering this feature, see *Administrator Guide for Avaya Communication Manager* and *Feature Description and Implementation for Avaya Communication Manager*.

Appendix B: Administering VRUs/IVRs as station ports

When Voice Response Units (VRUs) or Interactive Voice Response (IVRs) systems are used in a call center as station ports in a hunt group or in an ACD split or skill, either as a non-vector controlled split or accessed using the **converse-on**, **queue-to**, or **route-to** command, the station ports must be administered on the station screen with the type required by the VRU/IVR ports. The types for VRU/IVR ports supported by Avaya communication servers are listed in the following table.

Type VRU/IVR ports	Forward disconnect?	C&D tones ¹ support ?	Station type	Description
analog T&R	NA	no	2500	Standard station set interface ² .
analog T&R	NA	yes	VRU	Provides standard station set interface ² with C&D tones support using the communication server DTMF Feedback Signals feature.
lineside DS1/DS0 or lineside T1/E1 ³	no	no	ops	OPS is a DS1 type that provides a TIA/EIA off-premises station type DS1 interface used where the device does not require or support forward disconnect.
lineside DS1/DS0 or lineside T1/E1	yes	no	ds1fd	ds1fd provides a TIA/EIA Foreign eXchange (FX) type DS1 interface. The forward disconnect signal is a toggle of the A bit from 0 to 1 and then back to 0 after 600 msecs. This type is used for Line Side T1/E1 ports on the IVR system when used as an analog-like VRU device and is the recommended method for interfacing.
lineside DS1/DS0 or lineside T1/E1	yes	yes	VRUFD	VRUFD is the same as ds1fd, except C&D tone support is provided. This type of administration is not used for an IVR system.

Appendix B: Administering VRUs/IVRs as station ports

Type VRU/IVR ports	Forward disconnect?	C&D tones ¹ support ?	Station type	Description
lineside DS1/DS0 or lineside T1/E1	yes	no	ds1sa	ds1sa provides a TIA/EIA special-access type DS1 interface. The forward disconnect signal is a toggle of the A bit from 1 to 0 and then back to 1 after 600 msec.
lineside DS1/DS0 or lineside T1/E1	yes	yes	VRUSA	VRUSA is the same as ds1sa, except C&D tone support is provided.

1. For more information about support and administration of C&D (Connect and Disconnect) Tones, see the [C and D Tones support and administration](#) on page 432 section that follows this table.
2. Uses Analog Line Circuit Packs, such as TN2135.
3. DS1 Circuit Packs (TN767E (or later) or TN464F (or later) must be equipped.

C and D Tones support and administration

C and D Tones provides DTMF (touch tone) to a VRU port when the incoming caller is on soft hold to indicate the following transfer operation events:

- The VRU-placed call is being Connected to the transferred-to agent.
- The caller has Disconnected while on-hold.

Administration: To administer C and D Tones for a VRU port, complete the following administration steps:

- For the VRU customer option, enable `DTMF Feedback Signals`.
- On page 5 of the Feature-Related System Parameters screen, select values for the Connection and Disconnect fields. Possible values represent the buttons included in the 16-button DTMF array (0-9, *, #, A-D).

Note:

The default Connection and Disconnection tones are C and D, respectively. These tones are applied for 350 msec On with a 100 msec pause period.

- On page 1 of the Station screen for the VRU port, set the type to one of the following:
 - VRU (analog line)
 - VRUFD (Line Side DS1-FD)
 - VRUSA (Line Side DS1-SA)

Appendix C: Avaya reporting adjuncts and Communication Manager compatibilities

The following table describes the compatibility between each DEFINITY or Communication Manager release and each CMS or CCR release.

DEFINITY or Communication Manager release	CMS								IQ
	R14	R13.1	R13	R12	R3V11	R3V9	R3V8	R3V6	R4.x
Avaya Communication Manager 5.0	1	2	2	2	Not supported				1 (R4.1)
Avaya Communication Manager 4.0	1	2	2	2	2	Not supported			1
Avaya Communication Manager 3.1		1	2	2	2	2	2	2	2 ¹
Avaya Communication Manager 3.0		2	1	2	2	2	2	2	N/A
Avaya Communication Manager 2.0, 2.1, and 2.2		2	2	1	2	2	2	2	
Avaya Communication Manager 1.1, 1.2, or 1.3		2	2	2	1	2	2	2	
Avaya Call Processing R10		2	2	2	2	1	2	2	
DEFINITY ECS R9		2	2	2	2	1	2	2	
DEFINITY ECS R8		2	2	2	2	2	1	2	
DEFINITY ECS R7		Not supported				2	2	1	
DEFINITY ECS R6						2	2	2	
DEFINITY ECS R5		Not supported				2	2	2	
DEFINITY G3V4						2	2	2	
DEFINITY G3V3						2	2	2	
DEFINITY G3V2						2	2	2	

1. Aliased as a CMS R13.1. Requires CM 3.1.4 software.

1 = Supported

2 = Supported but not preferred. Some features are unavailable.

Appendix D: Avaya servers and gateways

This section includes the following topics:

- [Avaya media servers](#) on page 436
- [Gateways](#) on page 439

Acronym key

CC: Call Center

CM: Communication Manager

CMC: Compact Modular Cabinet

SCC1: Single Carrier Cabinet gateway

MCC1: Multi-Carrier Cabinet gateway

MG: Media Gateway

MC: Multi-Connect - interconnected with a TDM bus and Center Stage switches

ICC: Internal Call Controller - S8300/S8300B/S8300C processor board plugged into the MG as the resident media server

ECC: External Call Controller - non resident media server providing call control for one or more media gateways

ESS: Enterprise Survivable Server

LSP: Local Survivable Processor - S8300/S8300B processor board plugged into the media gateway

Avaya media servers

The following table describes the Avaya media servers that are certified for Avaya Call Center operation as of Communication Manager 4.0.

Avaya media servers	CC certified?	Communication Manager support	Supported with the following media gateways
DEFINITY CSI	Yes	Not supported starting with 4.0	CMC1
DEFINITY SI	Yes	Not supported starting with 3.0	<ul style="list-style-type: none"> ● SCC1 ● MCC1
DEFINITY R	Yes	<ul style="list-style-type: none"> ● Last supported in 1.3 ● Not supported starting with 2.0 	<ul style="list-style-type: none"> ● SCC1 ● MCC1
S8100	Yes	<ul style="list-style-type: none"> ● Last supported in 2.0.1 ● Not supported starting with 2.1 	G600 IP Connect
S8300/B/C	Yes	Starting with 1.2	<ul style="list-style-type: none"> ● G250 (Basic Call Center only) ● G350 ● G700
		Starting with 3.1	<ul style="list-style-type: none"> ● G150
S8400	Yes	Starting with 3.1	<ul style="list-style-type: none"> ● CMC ● G150 ● G250 ● G350 ● G600 ● G650 ● G700

Avaya media servers	CC certified?	Communication Manager support	Supported with the following media gateways
S8500/S8500B simplex or ESS S8500 is not supported starting with 5.0	Yes	Starting with 2.0	<ul style="list-style-type: none"> ● CMC1 ● SCC1 ● MCC1 ● G350 ● G600 as upgrade ● G650 ● G700
		Starting with 3.0	<ul style="list-style-type: none"> ● G250 (Basic Call Center only)
		Starting with 3.1	<ul style="list-style-type: none"> ● G150
		Starting with 5.0	<ul style="list-style-type: none"> ● G450
S8700	Yes	Starting with 1.1	<ul style="list-style-type: none"> ● SCC1 ● MCC1 ● G600 as upgrade ● G650 ● G700 ● G350
S8710	Yes	Starting with 2.2	Pentium 4 server - gateway support is the same as S8700
		Starting with 3.0	<ul style="list-style-type: none"> ● G250 (Full Call Center)
		Starting with 3.1	<ul style="list-style-type: none"> ● G150
		Starting with 5.0	<ul style="list-style-type: none"> ● G450
S8720	Yes	Starting with 3.1	<ul style="list-style-type: none"> ● Pentium 4 server - gateway support is the same as S8700 ● G150 ● G250 (Full Call Center)
		Starting with 4.0	<ul style="list-style-type: none"> ● G860
		Starting with 5.0	<ul style="list-style-type: none"> ● G450

Appendix D: Avaya servers and gateways

Avaya media servers	CC certified?	Communication Manager support	Supported with the following media gateways
S8720XL equipped with DAL2 shadow memory board	Yes	Starting with 4.0	<ul style="list-style-type: none"> ● Support same as S8720
S8730 equipped with DAL2 shadow memory board	Yes	Starting with 5.0	<ul style="list-style-type: none"> ● Supports all gateways including the G450
LSP in G700	Yes	Starting with 1.2	Up to 50: <ul style="list-style-type: none"> ● G350 ● G700
		Starting with 3.1	<ul style="list-style-type: none"> ● G150
LSP in G350	Yes	Starting with 2.1	One: <ul style="list-style-type: none"> ● G350
		Starting with 3.1	<ul style="list-style-type: none"> ● G150
SBS3000 (VM Blade Server) Hosted Contact Center offer from Global Managed Services	Yes	Starting with 3.0	<ul style="list-style-type: none"> ● G250 (Full Call Center) ● G350 ● G650 (recommended) ● G700
		Starting with 3.1	<ul style="list-style-type: none"> ● G150

Gateways

The following table describes the Avaya gateways that are certified for Call Center operation.

Avaya gateways	Inter-gateway connectivity		
	CCS-certified with MC TDM connect? ¹	CCS-certified with IP connect?	
		ICC as S8300	ECC
Port network media gateways			
CMC1	Yes	N/A	N/A
SCC1	Yes	N/A	N/A
MCC1	Yes	N/A	N/A
G600 - replaced by G650	N/A	N/A	Yes - starting with CM 1.1
G650	Yes - starting with CM 2.0	N/A	Yes - starting with CM 2.0
H.248 media gateways			
G700	N/A ²	Yes - starting with CM 1.2	Yes - starting with CM 1.2
G350	N/A ²	Yes - starting with CM 2.1	Yes - starting with CM 2.0
G250	N/A ²	Basic Call Center only starting with CM 3.0	Yes - starting with CM 3.0
G250-DS1 G250-DCP	N/A ²	Yes - starting with CM 3.1	Yes - starting with CM 3.1
G450	N/A ²	Yes - starting with CM 5.0	Yes - starting with CM 5.0
H.323 media gateways			
G150 remote office	N/A	Yes - starting with CM 3.1	Yes - starting with CM 3.1

Appendix D: Avaya servers and gateways

Avaya gateways	Inter-gateway connectivity		
	CCS-certified with MC TDM connect?¹	CCS-certified with IP connect?	
		ICC as S8300	ECC
High density trunk gateway (connects to CM via SIP trunking)			
G860	N/A	N/A	Yes - starting with CM 4.0

1. See [How the SCC software worked prior to Release 2.2](#) on page 337.

2. H.248 media gateways connect using IP only and can be included in an ECC server configuration. The media gateways are connected with each other and the server using IP. These mixed configurations have been certified for CCS when the server and media gateways have been certified for ECC IP Connect.

Glossary

AAR	See Automatic Alternate Routing (AAR) .
abandoned call	An incoming call in which the caller hangs up before the call is answered.
Abbreviated Dialing	A feature that allows callers to place calls by dialing just one or two digits.
ACA	See Automatic Circuit Assurance (ACA) .
access code	A 1-, 2-, or 3-digit dial code used to activate or cancel a feature, or access an outgoing trunk.
access trunk	A trunk that connects a main communications system with a tandem communications system in an Electronic Tandem Network (ETN) . An access trunk can also be used to connect a system or tandem to a serving office or service node. Also called an <i>access tie trunk</i> .
ACCUNET	A trademarked name for a family of digital services offered by AT&T in the United States.
ACD	See Automatic Call Distribution (ACD) .
ACD agent	See agent .
ACD work mode	See work mode .
ACW	See After Call Work (ACW) mode .
adjunct	A processor that does one or more tasks for another processor and is optional in the configuration of the other processor. See also application .
Adjunct Routing	A means of evaluating calls before the calls are processed by requesting information from an adjunct. The communication server requests instructions from an associated adjunct and makes a routing decision based on agent availability or the caller information.
adjunct-controlled split	An Automatic Call Distribution (ACD) split that is administered to be controlled by another application. Agents logged into such splits must do all telephony work, ACD login/ logout, and changes of work mode through the adjunct (except for auto-available adjunct-controlled splits, whose agents may not log in/out or change work mode).
adjunct-monitored call	An adjunct-controlled call, active-notification call, or call that provides event reporting over a domain-control association.
Adjunct-Switch Application Interface (ASAI)	A recommendation for interfacing adjuncts and communications systems, based on the CCITT Q.932 specification for layer 3.
adjusted EWT	A Best Service Routing (BSR) term for Expected Wait Time (EWT) plus a user adjustment set by a <code>consider</code> command.

administration terminal

administration terminal	A terminal that is used to administer and maintain a system.
Administration Without Hardware (AWOH)	A feature that allows administration of ports without associated terminals or other hardware.
Advocate	See Avaya Business Advocate .
After Call Work (ACW) mode	A mode in which agents are unavailable to receive ACD calls. Agents enter the ACW mode to perform ACD-related activities such as filling out a form after an ACD call. Also see, auto-in work mode , manual-in work mode , and aux-work mode .
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	See Automatic Number Identification (ANI) .
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. See also call appearance .
application	An adjunct that requests and receives ASAI services or capabilities. One or more applications can reside on a single adjunct. However, the communication server cannot distinguish among several applications residing on the same adjunct and treats the adjunct, and all resident applications, as a single application. The terms application and adjunct are used interchangeably throughout this document.
application plan	A plan used only in multi-site Best Service Routing (BSR) applications. The application plan identifies the remote switches that may be compared in a consider series. The plan also specifies the information used to contact each communication server and to interflow calls to the communication 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	See Automatic Route Selection (ARS) .
ASAI	See Adjunct-Switch Application Interface (ASAI) .
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. Also see attendant console .

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. Also see attendant .
Audio Information Exchange (AUDIX)	An Avaya messaging system . AUDIX has been replaced by Message Manager.
AUDIX	See Audio Information Exchange (AUDIX) .
auto-in work mode	A mode in which an agent is ready to process another call as soon as the current call is completed. Auto-in work mode is one of four agent work modes. Also see, aux-work mode , manual-in work mode , and After Call Work (ACW) mode .
Automatic Alternate Routing (AAR)	A feature that routes calls to a different route than the first-choice route when facilities are unavailable.
Automatic Call Distribution (ACD)	A feature that answers calls, and then depending on administered instructions, delivers messages appropriate for the caller and routes the call to an agent when one becomes available.
Automatic Call Distribution (ACD) split	A method of routing calls of a similar type among agents in a call center. Also, a group of extensions that are staffed by agents trained to handle a certain type of incoming call.
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 may signify a faulty trunk.
Automatic Number Identification (ANI)	A display of the calling number so that agents can access information about the caller.
Automatic Route Selection (ARS)	A feature that allows the system to automatically choose the least-expensive way to send a toll call.
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.
auxiliary trunk	A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.
aux-work mode	A mode in which agents are unavailable to receive Automatic Call Distribution (ACD) calls. Agents enter aux-work mode when involved in non-ACD activities such as taking a break, going to lunch, or placing an outgoing call. Also see, auto-in work mode , manual-in work mode , and After Call Work (ACW) mode .

available agent strategy

available agent strategy

A strategy that determines how [Best Service Routing \(BSR\)](#) commands in a vector identify the best split or skill when several have available agents.

Avaya Business Advocate

A product that establishes different levels of service for different types of calls. For example, a company may decide that a premium customer gets faster service than other types of customers.

CCR

CCR is Avaya's next generation reporting platform.

AWOH

See [Administration Without Hardware \(AWOH\)](#).

barrier code

A security code used with remote access to prevent unauthorized access to the system.

Basic Call Management System (BCMS)

An application on the communication server that monitors the operations of an [Automatic Call Distribution \(ACD\)](#) application. BCMS collects data related to the calls on the communication server and organizes the data into reports that help manage ACD facilities and personnel.

BCC

See [Bearer Capability Class \(BCC\)](#).

BCMS

See [Basic Call Management System \(BCMS\)](#).

Bearer Capability Class (BCC)

A code that identifies the type of a call (for example, voice and different types of data).

best

The split, skill, or location that provides the most advantageous service for a caller as determined by [Best Service Routing \(BSR\)](#).

Best Service Routing (BSR)

An Avaya communication server feature based on call vectoring that routes [Automatic Call Distribution \(ACD\)](#) calls to the split, skill, or contact center best able to service each call. BSR can be used on a single communication server, or it can be used to integrate resources across a network of communication servers.

bridge (bridging)

The appearance of a telephone extension at one or more other telephones.

bridged appearance

A call appearance on a telephone that matches a call appearance on another telephone for the duration of a call.

Business Advocate

See [Avaya Business Advocate](#).

call appearance

1. For the attendant console, the six buttons labeled a-f used to originate, receive, and hold calls. Two lights next to the button show the status of the call appearance.
2. For the telephone, a button labeled with an extension and used to place outgoing calls, receive incoming calls, or hold calls. Two lights next to the button show the status of the call appearance.

Call Detail Recording (CDR)

A feature that uses software and hardware to record call data.

Call Management System (CMS)

An application that enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and VDNs. CMS enables customers to partially administer the [Automatic Call Distribution \(ACD\)](#) feature for a communications system.

call vector	A set of vector commands used to process an incoming or internal call.
call work code	A number entered by ACD agents to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers) on ACD calls.
callback call	A call that automatically returns to a voice-terminal user who activated the Automatic Callback feature.
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. If the facility is busy, it is assumed to have been busy for the entire scan interval. 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, it is said to have been busy for 36 CCS.
CDR	See Call Detail Recording (CDR) .
Central Office (CO)	A switch owned by a local telephone company that provides local telephone service (dial-tone) and access to toll facilities for long-distance calling.
Central Office (CO) trunk	A telecommunications channel that provides access from the system to the public network through the local CO.
channel	<ol style="list-style-type: none"> 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	<ol style="list-style-type: none"> 1. An arrangement of electrical elements through which electric current flows. 2. A channel or transmission path between two or more 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.
Class of Restriction (COR)	A feature that allows classes of call-origination and call-termination restrictions for telephones, telephone groups, data modules, and trunk groups. See also Class of Service (COS) .
Class of Service (COS)	A feature that uses a number to specify if telephone users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features. See also Class of Restriction (COR) .
CO	See Central Office (CO) .

communications server

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 a set of two or more. A set of consider commands is called a consider series.
console	See attendant console .
COR	See Class of Restriction (COR) .
COS	See Class of Service (COS) .
coverage answer group	A group of up to eight telephones that ring simultaneously when a call is redirected to it 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	See call work code .
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 domain	<p>An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.</p> <p>A group of VDNs, ACD splits, and stations.</p>
Dynamic Percentage Adjustment	<p>An Avaya Business Advocate feature that makes automatic adjustments to agents' target allocations as needed to help meet the administered service level targets.</p>
Dynamic Queue Position	<p>An Avaya Business Advocate feature that gives you the ability to queue calls from multiple VDNs to a single skill, while maintaining different service objectives for those VDNs.</p>
Dynamic Threshold Adjustment	<p>An Avaya Business Advocate Service Level Supervisor feature that automatically adjusts overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels.</p>
EAD-LOA	<p>See Expert Agent Distribution-Least Occupied Agent (EAD-LOA).</p>
EAD-MIA	<p>See Expert Agent Distribution-Most Idle Agent (EAD-MIA).</p>
Electronic Tandem Network (ETN)	<p>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.</p>
EPN	<p>See Expansion Port Network (EPN).</p>
ETN	<p>See Electronic Tandem Network (ETN).</p>
EWT	<p>See Expected Wait Time (EWT).</p>
Exclusion	<p>A feature that allows multi-appearance telephone users to keep other users with the same extension from bridging onto an existing call.</p>
Expansion Port Network (EPN)	<p>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.</p>
Expected Wait Time (EWT)	<p>A prediction of how long a call waits in queue before the call is answered.</p>
Expert Agent Distribution-Least Occupied Agent (EAD-LOA)	<p>An agent selection method for delivery of calls. With EAD-LOA implemented, calls are delivered to the available agent with the highest skill level and the lowest percentage of work time since login (compared to other available agents with the same skill level).</p> <p>See also Expert Agent Distribution-Most Idle Agent (EAD-MIA), Uniform Call Distribution-Least Occupied Agent (UCD-LOA), and Uniform Call Distribution-Most Idle Agent (UCD-MIA).</p>

Expert Agent Distribution-Most Idle Agent (EAD-MIA)

Expert Agent Distribution-Most Idle Agent (EAD-MIA)	<p>An agent selection method for delivery of calls. With EAD-MIA implemented, calls are delivered to the available agent with the highest skill level who has been idle the longest since their last ACD call (compared to other available agents with the same skill level).</p> <p>See also Expert Agent Distribution-Least Occupied Agent (EAD-LOA), Uniform Call Distribution-Least Occupied Agent (UCD-LOA), and Uniform Call Distribution-Most Idle Agent (UCD-MIA).</p>
extension-in (EXT-IN)	<p>A work state agents go into when they answer 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.</p>
extension-out (EXT-OUT)	<p>A work state that agents go into when they place a non-ACD call.</p>
external call	<p>A connection between a communications system user and a party on the public network, or on another communications system in a private network.</p>
facility	<p>A telecommunications transmission pathway and the associated equipment.</p>
Forced Agent Logout from ACW mode	<p>A feature used to automatically log out an Expert Agent Selection (EAS) agent who spends too much time in After Call Work (ACW) mode.</p>
Forced Agent Logout by Clock Time	<p>A feature used to automatically log out an Expert Agent Selection (EAS) agent at a pre-determined time. This feature is primarily used to automatically log off agents at the end of their shifts.</p>
glare	<p>A simultaneous seizure of a 2-way trunk by two communications systems resulting in a standoff.</p>
ground-start trunk	<p>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.</p>
holding time	<p>A total length of time in minutes and seconds that a facility is used during a call.</p>
intelligent polling	<p>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. Also see status poll.</p>
intercept tone	<p>An tone that indicates a dialing error or denial of the service requested.</p>
interflow	<p>An Automatic Call Distribution (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.</p>
internal call	<p>A connection between two users within a system.</p>
internal measurement	<p>A Basic Call Management System (BCMS) measurement that is made by the system.</p>
intraflow	<p>An Automatic Call Distribution (ACD) term that refers to the ability for calls to redirect to other splits on the same communication server to backup the primary split.</p>

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.
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 line	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 appearance	See appearance .
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.
Location Preference Distribution	A feature used to route incoming Automatic Call Distribution (ACD) calls to agents located at the same location where the trunk is located whenever possible.
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	The terminal that is used by the system administrator to administer the switch. The terminal may also be used to access the Basic Call Management System (BCMS) feature.
manual-in work mode	A mode in which an agent is ready to process another call manually. Also see, auto-in work mode , aux-work mode , and After Call Work (ACW) mode .
Maximum Agent Occupancy (MAO)	A feature used to set thresholds on the amount of time an agent spends on a call. MAO is used to prevent agent burnout. The MAO threshold is a system-administered value that places an agent in AUX 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.
messaging system	A generic name for a system that records, stores, plays, and distributes phone messages. Message Manager is the latest messaging system provided by Avaya.

minor alarm

minor alarm

An indication of a failure that could affect 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, if applicable.

modular processor data module (MPDM)

A Processor Data Module (PDM) that can be configured to provide several kinds of interfaces (RS-232C, RS-449, and V.35) to customer-provided data terminal equipment (DTE).

Modular Trunk Data Module (MTDM)

A trunk-data module that can be configured to provide several kinds of interfaces (RS-232, RS-449, and V.35) to customer-provided data terminal equipment.

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 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.

NFAS

See [Non-Facility Associated Signaling \(NFAS\)](#).

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.

non switch-classified outbound calls

Proactive Contact outbound calls that are automatically launched by Communication Manager. See also, [switch-classified outbound calls](#).

NSF

See [Network Specific Facility \(NSF\)](#).

occurrence

See [appearance](#).

pickup group

A group of individuals authorized to answer any call directed to an extension within the group.

PMS

See [Property Management System \(PMS\)](#).

poll

See [status poll](#).

poll suppression

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. Also see [status poll](#).

polling, intelligent

See [intelligent polling](#).

PPN

See [Processor Port Network \(PPN\)](#).

primary extension

A main extension associated with the physical telephone or data terminal.

principal	A terminal that has its primary extension bridged on one or more 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.
Property Management System (PMS)	A stand-alone computer used by lodging and health-services organizations for services such as reservations, housekeeping, and billing.
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.
queuing	A process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.
R2-MFC signaling	A signal consisting of two frequency components, such that when a signal is transmitted from a switch, another signal acknowledging the transmitted signal is received by the switch.
recall dial tone	A tone signalling that the system has completed a function (such as holding a call) and is ready to accept dialing.
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 at least one of the facilities, such as a trunk or a digit transmitter, needed for the call 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 that allows the terminal user (usually the principal) to bridge onto a call that had been answered by another party on his or her behalf. Also called a <i>temporary bridged appearance</i> .
SLM	See Service Level Maximizer (SLM) .
split (agent) status report	A report that provides real-time status and measurement data for internally-measured agents and the split to which they are assigned.

split condition	
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 the communication server and the Basic Call Management System (BCMS) .
split report	A report that provides historical traffic information for internally measured splits.
staffed	An indication that an agent position is logged in. A staffed agent functions in one of four work modes: auto-in work mode , manual-in work mode , After Call Work (ACW) mode , or aux-work mode .
Station Message Detail Recording (SMDR)	An obsolete term now called Call Detail Recording (CDR) .
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 placed by a consider location vector command to obtain status data from a remote location in a multi-site Best Service Routing (BSR) application.
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. See also, non switch-classified outbound calls .
system printer	An optional printer that may be 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	Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or Central Offices (COs).
UDP	See Uniform Dial Plan (UDP) .
Uniform Call Distribution-Least Occupied Agent (UCD-LOA)	An agent selection method for delivery of calls. With UCD-LOA implemented, calls are delivered to the available agent with the lowest percentage of work time since login. Also see Expert Agent Distribution-Least Occupied Agent (EAD-LOA) , Expert Agent Distribution-Most Idle Agent (EAD-MIA) , and Uniform Call Distribution-Most Idle Agent (UCD-MIA) .

Uniform Call Distribution-Most Idle Agent (UCD-MIA)	<p>An agent selection method for delivery of calls. With UCD-MIA implemented, calls are delivered to the available agent who has been idle the longest since their last ACD call.</p> <p>See also EAD-LOA, EAD-MIA, and UCD-LOA.</p>
Uniform Dial Plan (UDP)	<p>A feature that allows a unique number assignment for each terminal in a multiswitch configuration such as a Distributed Communications System (DCS) or main-satellite-tributary system.</p>
VDN	<p>See Vector Directory Number (VDN).</p>
Vector Directory Number (VDN)	<p>An extension that provides access to the vectoring feature on the switch. Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number.</p>
vector-controlled split	<p>A hunt group or ACD split administered with the vector field enabled. Access to such a split is possible only by dialing a VDN extension.</p>
work mode	<p>A mode that an ACD agent can be in. Upon logging in, an agent enters aux-work mode. To become available to receive ACD calls, the agent enters auto-in work mode or manual-in work mode. To do work associated with a completed ACD call, an agent enters After Call Work (ACW) mode.</p>
work state	<p>An ACD agent may be a member of up to three different splits. Each ACD agent continuously exhibits a work state for every split of which it is a member. Valid work states are Avail, Unstaffed, AUX-Work, ACW, ACD (answering an ACD call), ExtIn, ExtOut, and OtherSpl. An agent's work state for a particular split may change for a variety of reasons. For example, an agent's work state changes when a call is answered or abandoned, or the agent changes work modes. The Basic Call Management System (BCMS) feature monitors work states and uses this information to provide BCMS reports.</p>

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