Avaya Aura® Communication Manager Special Application Features

Overview

The *Avaya Aura® Communication Manager Special Applications Features d*ocument describes the special application features that can be administered with 6.3 release of Avaya Aura® Communication Manager. This document serves as an overview of the available features, for more detailed information the individual feature documents that can be found by doing a search on the SA number from support.avaya.com.

Prior to Communication Manager 5.2.1, customers could activate Communication Manager special applications (SAs), aka green features, only via an RFA-generated license file and most features carried a per-system RTU fee. Beginning with Communication Manager 5.2.1, customers, business partners and Avaya support personnel will have access to the form to activate the unrestricted special applications. Customers can activate most unrestricted special applications (green features) without intervention and the features no longer carry a RTU fee. Due to increased administration, risk, or specific configurations some special applications are restricted to either a) Business Partner/Avaya or b) Avaya only. For those features noted below as restricted, please contact your Business Partner, Avaya support or the Avaya Custom Development team.

Warning: Special Application features are intended to serve specific needs and typically are not recommended for general use. Special App features have limited testing and may be applicable only to specific configurations. Activating one or more of these features may result in unpredictable system behavior or malfunction. Before activating a feature please read the associated documentation at http://support.avaya.com**, search for SA number**, **to determine if the feature is applicable to your needs and system configuration. Before activating two or more features please contact CustomCommunication Manager@avaya.com for potential risks related to feature interactions.**

Contents

What is a Special Application feature?

A custom feature requested by a customer for their specific need. Special Applications are
included in the Communication Manager software base, however additional administration is
required.

Before activating a Special Application feature please review the documentation carefully. If you
have questions please contact CustomCM@avaya.com.

How do I know if a Special Application feature will work for my customer?

Every attempt is made to offer Special Application features to all customers however, note that
requirements are based on the original customer's configuration and Communication Manager
software release. The related requirements document and/or MR should be reviewed before
ordering any Special Application feature and Custom Development does not guarantee the
feature functionality and interactions outside of what is described in the related documentation.
Any change to the functionality of the Special Application feature requires a GRIP request to
be submitted before an assessment of the requested change can be made.

How do I activate a Special Application feature?

Most Special Application features can be activated by executing "change system special-
applications" from the SAT screen. Some features are restricted to Business Partners and/or
Avaya personnel due to complexity and/or restrictions. Those restricted features will be noted in
this document. If you believe a restricted feature is needed, you should contact the Custom
Development Group at CustomCM@avaya.com.

Do Special Applications carry forward after an upgrade?

Yes. Once you activate a feature it will remain activated after upgrade. The only exception to this
is if you change platforms and have to re-translate the system.

Can I activate multiple Special Applications?

Every attempt is made to run tests between features when possible interactions are identified however, Custom Development does not guarantee that multiple features will co-exist on a single system.

Special Application Features

SA7161 NORTEL SL1 PRI and DMS Names Display

This special application contains a modification to the ISDN PRI operation on the Communication Manager switch. The current specification of the AT&T PRI standard that is implemented in Communication Manager does not include some modifications to the AT&T PRI standard that would allow complete operation with the Northern Telecom DMS250/100 product

SA7291 - TAAS Pickup During Day

This feature allows any voice terminal to pickup a call from the attendant queue when the number of calls in queue has reached the administered warning level by dialing the TAAS access code regardless of whether the system is in night service or not.

SA7311 - CDR Record Answering Party

This feature will change the operation of CDR (Call Detail Recording) such that the actual party that answers a call is considered to be the destination party on the record. In the case where the dialed party answers the call, then no change is seen from the standard operation.

SA7409a - Flash to Answer CW/Hold

For analog stations with the call waiting feature(s) activated and the Flash for Answer/Hold field active, this Special Application provides automatic hard-hold activation when the called-party flash-hooks upon receiving the call-waiting indication. It also provides automatic unhold of hard held calls for those same stations.

SA7409b - Cancel ARS using '*'

This Special Application will allow stations to access station features administered on the system, e.g. Call Forwarding, Abbreviated Dialing etc., by dialing the digit '*' as the first digit after the hotline station goes off-hook or after a non-hotline station has dialed the ARS feature access code.

SA7474 - Cover Internal on Night Service

The current implementation of Communication Manager provides separate sets of coverage path criteria depending on whether the call is internal or external. The call is considered to be an external call if the originator of the call was a trunk (except for tie trunks administered as internal tie trunks) or an attendant. If the call meets the criteria for a specific coverage path, then the call is routed to the parties administered for that coverage path. If the call does not meet the criteria for that coverage path, the next coverage path in the path list is attempted until either a path meets the criteria or there are no more paths.

This feature provides a better operation in situations where the coverage changes between day and night schedules. This feature is not intended to be a full time of day call coverage feature, but is tied to the manual selection of night or day service by the principal attendant console operator.

SA7483 - Enhanced Emergency Alert

The following feature changes have been made, and will only be activated when the SA is activated:

- When an Emergency call comes in to a station, the station will receive priority (usually triple, it is now administrable) ringing. Currently it receives single ringing. If a station is off-hook when the emergency call comes in, it will receive only 1 cycle of priority ringing.
- When an Emergency call comes in to a station, the station display will indicate an emergency. Currently it indicates a redirected call. If the Emergency Station is equipped with a display it will show a message of the format:
  n=<calling party name> <calling party extension> EMERGENCY
  where "n" is the appearance (a through j) and "calling party name" is the name assigned to the station through standard administration. For example, if an Emergency Call rings on the first appearance, a display might show:
  a=Cafeteria 4701 EMERGENCY

SA7487 - Cover External on Trunk on Hold

The coverage feature in the standard Communication Manager has an option to route calls differently depending on whether the call being covered is an internal (station) call versus an external (trunk) call. This decision is based on the originator of the call. If the originator of the call is a trunk or attendant, the call will use the external coverage criteria.

SA7491 - Expand DS1 Circuit Packs to 332 Linux only

This feature was developed for a very specific configuration. It is not available for customer activation at this time. Contact the Avaya Custom Development Group, CustomCM@avaya.com, for more information.

SA7578 - Integrated Directory Service over DCS

This feature provides the user with the capability to use the directory feature over a DCS network. Currently directory will only look at extensions on the local switch. This new feature allows directory to display extensions on other switches in the DCS network.

SA7666 - COS Conference Tone Check

The special application will check to see if any member of a conference call has the "Conference Tone" flag set to "y" on their COS before applying the conference tone to the call. If the conference call has only trunks on the call, then conference tone will always be applied.

SA7710 - Enhanced Display on Redirected Calls

This feature provides the user with the capability to display up to 27 characters for both the name of the called party and calling party along with the extensions on a bridge appearance of the called party's terminal. The 27-character display is determined by the called party's terminal type, which must be a 2x40 2-line display terminal. Terminals that are 2x40 2 line display sets include: 8434D, 7444D, 7407+, 7407D, and Callmasters.

SA7776 - Display Incoming Digits for ISDN Trunk Groups

This feature is to allow displaying the in-dialed digits for isdn-pri trunk groups. The customer currently can route all calls to an isdn-pri trunk group to a VDN by using the call by call features on the 2nd page of the isdn-pri trunk group form. They delete the in-dialed digits and insert the VDN. By displaying the in-dials to the agent, the agent can quickly type in the dialed number and bring up the customer account.

SA7777 - Night Service on DID Trunk Groups

This feature is to allow a call center to close for the night and direct all calls to the Night Service destination. This Night service destination will probably be to another switch within the customer's network. The customer will do this by inserting the aar access code and a prefix in front of the dialed digits by using the digit insertion on page 1 of the DID trunk group form. The "Night Service:" destination will be administered to a VDN. The vector that the VDN points to will "route to digits". This will send the call off according to the AAR digit analysis administration, thus routing the call to another switch within their network.

SA7778 - Display UUI Information

This feature is to provide a station button that will display certain information contained in the UUI field of an ISDN setup message. This will be used when the customer networks their call centers together via an ISDN private network

SA7779 - Enhanced DID Routing

The Enhanced DID Routing feature provides for a total Data Base bypass for incoming DID calls, where all 1-16 digit in-dials are valid numbers if the DID trunk group is administered to redirect to a VDN, Hunt Group or Station Extension.

The Enhanced DID Routing feature allows all calls to a specified DID trunk group to be routed to the same destination independent of the number dialed. The Enhanced DID Routing operation is initiated by administering the "Incoming Destination:" and the "Expected Digits:" fields on the DID trunk group form. If the "Incoming Destination:" field is NOT administered on the DID trunk group form, then the trunk group will work as a normal DID trunk group. The "Incoming Destination:" field may be administered to a Hunt Group, Station extension or a VDN. The "Expected Digits:" field is to be administered to the number of digits expected from the Central Office (CO) and may contain a value from 1 - 16

SA7852 - # and * in Vector Collect Step?

This feature gives options for Call Vectoring to allow the user to enter data on a Vector Collect step using the # and/or * keys in addition to the digits 0 through 9. . Without these options, the * key causes entry to start over after dripping previously entered digits, and the # indicates end-of-entry. There are system-wide options, applying only to "Vector Collect" call vectoring steeps, but applying to all "Vector Collect's" in all vectors. The digits collected are made available to the ASAI interface via the "adjunct" step. The # will have the value Hex23 (0x23) and the * will have the value Hex 2a (0x2a). These options are available so that customers may prompt for entry of account codes or other information which contains # or *, and have that information transmitted over ASAI to a host. These options will be referred to below as the Collect Literally Options. There will be an option for * and an option for #, both options may be active together.

SA7880 - ASAI Internally Measured Data

This feature was developed for a very specific configuration. It is not available for customer activation at this time. Contact the Avaya Custom Development Group, CustomCM@avaya.com, for more information.

SA7900 - Service Observe Physical Set

This feature, when administered, will allow the observer to observe any active call on the set which contains the call appearances of the dialed extension. This allows the service observer to observe a particular physical set, regardless of whether the user is active on their own call appearance or a bridged appearance of some other extension.

SA7933 - Busy Tone with SAC and No Available Cvg Points

This green feature, when administered, will apply busy tone to the calling party and drop the called voice terminal when the called voice terminal has activated the Send All Calls feature and no coverage point is available.

SA7963 - Dial By Name

The Dial By Name feature is a method for incoming calls to a PBX to be routed to a destination based on the name of the person to be reached. Incoming calls will be routed to a vector especially set up to use this feature. The incoming call will receive an announcement prompting them for the various choices that the call can be routed to. This development will allow the inclusion in the vector of the additional option, the Dial By Name feature. When this option is chosen, the user will then be prompted by announcements to instruct them how to dial the name. When a unique match with a name in the database has been found, the user will be routed to the appropriate extension.

SA7991 - Variable Length Account Code

This feature provides the user with the capability to enter a CDR account code with a length variable from 1 to the maximum number of digits assigned in the system parameters cdr form. When fewer than the maximum number of digits is desired, a "#" terminates the account code and dial tone is returned to the user as if the maximum number of digits was dialed.

SA7994 - Incr Station Busy Ind to 25,000 Linux only

This enhancement increases the number of Facility Busy Indicators on the large platforms from 10,000 to 25,000

SA8052 - ISDN Redirecting Number

This feature creates a redirecting number IE, or original called number IE, in the SETUP message for redirecting a call offnet over an isdn trunk to either a *Bellcore* or *Nortel* switch. Redirections in this case are either a call forward or call coverage. This feature is needed to notify these other switches that the call is a redirection and not a direct termination.

This is especially useful if the voice mail system is attached to one of these off net switches in order to receive the correct greeting.

SA8077 - Russian Power Industry Feature

Special equipment is used in the power industry to access "ADASE" tie trunks using carrier systems. The word ADASE (pronounced adacee), a transliteration of the Russian abbreviation "power network equipment for long distance automatic communications", is used to refer to this special equipment. Its purpose is to allow direct high priority access by "dispatchers" to the tie trunks while allowing the PBX low priority use of the trunks when they are not needed by the dispatchers. The TN2209 Circuit Pack, an analog 4-wire tie trunk circuit, developed as an "integrated converter" using the TN760D software interface, can be used for this application.

SA8122 - QSIG CPC Conversion for Code Set 5

This feature identifies the category of a calling party on an outgoing call over a QSIG trunk. By identifying the calling party category, the switch is able to place terminating restrictions on an incoming call based on the category of the calling party.

SA8140 - Attendant Dial 0 Redirect

Communication Manager has a Priority Attendant Queuing feature that allows calls to the attendant to be prioritized by call type (CO Call, Emergency Access Call, Redirected call, etc.). SA8141 enhances the current priority queuing by allowing the customer to assign attendant queue priorities to LDN calls as well. If a calling party makes a call to a Listed Directory Number (LDN) that has a priority level assigned, the call will be queued based on the priority of that LDN. If no priority level is administered for the LDN, the call will be placed in the attendant queue with a priority based on the call type.

SA8141 - LDN Attendant Queue Priority

Communication Manager has a Priority Attendant Queuing feature that allows calls to the attendant to be prioritized by call type (CO Call, Emergency Access Call, Redirected call, etc.). SA8141 enhances the current priority queuing by allowing the customer to assign attendant queue priorities to LDN calls as well. If a calling party makes a call to a Listed Directory Number (LDN) that has a priority level assigned, the call will be queued based on the priority of that LDN. If no priority level is administered for the LDN, the call will be placed in the attendant queue with a priority based on the call type.

SA8143 - Omit Designated Extensions From Displays

This feature is used to suppress a stations' extension number when calling other stations. Stations will have this feature active by a new field on the COR form that indicates whether this stations' number should be suppressed. However, if one of these stations calls a station that has console permissions COS, and the "override by COS" feature bit is set, then the stations' number will be displayed.

SA8146 - Display Update for Redirected Calls

This feature enables an update of the originator's display for calls that are redirected. The originator's display will be updated to contain the principal station's name, the redirection reason, and the name and extension of the "redirected to" station for intraswitch redirections.

For interswitch redirections, connected via isdn-pri, the originator's display will be updated with the name and number of the "redirected to" station and the principal station's name if the principal is on the same switch.

SA8156 - Attendant Priority Queuing by COR

This feature allows calls to the attendant that are queued to be answered according to priority levels associated with the calling party's Class of Restriction (COR). (The calling party may be a station or a trunk.) Each COR may be assigned one priority level.

Communication Manager has a Priority Attendant Queuing feature that allows calls to the attendant to be prioritized by call type (CO Call, Emergency Access Call, Redirected call, etc.). SA8156 enhances the current priority queuing by allowing the customer to assign attendant queue priorities by COR as well.

SA8157 - Toll Free Vectoring until Answer

This feature provides the user with the capability to be given announcements and music in Call Vectoring without returning network answer supervision (i.e., an ISDN CONNECT message) over ISDN trunk facilities and starting CDR call duration timing.

SA8201 - Start Time and 4-Digit Year CDR Custom Fields

This feature provides the user with the capability to customize the CDR (Call Description

Record) using five new keywords. These five new keywords allow the user to add start date and end date in one of two formats either DDMMYY or DDMMYYYY format and to add start time and end time in HHMMSS format.

SA8202 - Intra-switch CDR by COS

This feature provides the user with the capability to define an unlimited number of stations to generate CDR records.

This feature allows the customer to designate which stations will generate intra-switch CDR records based on the station's COS. This feature will override the current intra-switch CDR table that required each station to be administered individually. Using the new feature (SA# 8202) each COS (0-15) will have a "n/y" field titled "Intra-switch CDR". If this field is set to yes for a given station's COS, station-station calls in which that station is the calling or the called party will generate a CDR record.

SA8211 - Prime Appearance Preference

This feature provides the capability to specify one call or bridged appearance on a non-BRI, multi-appearance telephone as the prime appearance. The prime appearance will be automatically selected for the user when the user goes on-hook, giving that appearance preference the next time the user goes off-hook to make a call.

SA8240 - Station User Admin of FBI

This feature provides the user with the capability to administer a facility busy indication (FBI) button from the station. An FBI button is normally administered from a System Access Terminal (SAT) as button type 'busy-ind'. An FBI provides a visual indication of the busy or idle status of an extension number, trunk group, terminating extension group, hunt group, or loudspeaker paging zone.

The feature will be activated by either feature access codes or a button on the set.

The class of restriction will be used to allow users to use the feature.

SA8312 - Meet-Me Paging

This feature allows extension numbers that are translated as "PAGING" station types to activate tone pagers. When one of these "PAGING" stations is called, it causes a special Meet-Me page message to be sent to a paging adjunct which, in turn, causes the pager to alert. The owner of the pager may then be connected to the calling party by dialing a feature access code and the page extension number.

SA8323 - Idle Call Preference Display

This feature provides users with the Idle Appearance Preference option set to receive a display of incoming call information without answering the call.

SA8339 - PHS X-Station Mobility

The X-Station Mobility Feature will allow phones remote over a trunk interface to be controlled by Communication Manager as if they were directly connected to it. This is accomplished by administering a station record for each station of type "X-Mobile."

SA8348 - Map NCID to Universal Call ID

The purpose of this Special Application is to enable the mapping of the MCI*WorldCom* Network Call Identifier, the Generic Digits Information Element in codeset 6 in the SETUP message, to the internal CM Universal Call ID (UCID) for call reference. This UCID will be utilized for the call reference identification while the call is being processed in the Communication Manager that received the call and may be passed on to another Communication Manager in a private network as UCID in the UUI message content. This is to enable call and data record association within the call receipt Communication Manager and at the privately networked Communication Manager. It also provides the end user with a related call reference identification for the purpose of reporting trouble to the Service Provider.

SA8428 - Station User Button Ring Control

This feature provides the user with the capability to control the ring behavior for each line appearance and bridged appearance from the station.

Activation or deactivation of the ringer will be controlled by a feature access code or a new feature button btn-ring. Also, this feature will add a new feature button, ring-stat, which will be used to display the ringer status for line and bridged appearances.

SA8434 - Delay PSTN Connect on Agent Answer

This feature allows an incoming ISDN trunk call to be routed to a logged-in Avaya Communication Manager EAS (Expert Agent Selection) agent without sending an ISDN CONNECT message to the Public Switched Telephone Network (PSTN). By doing this, there is no PSTN switch billing for the call as required initially for the operation of the Telecom Italia Directory Assistance systems in Italy, and subsequently has been used for Avaya-based Directory Assistance systems in Germany, Hungary, Russia, United Arab Emirates, Philippines, and other countries. This Green feature is activated for an incoming ISDN call that is routed through a VDN whose "Delay ISDN Connect message on Agent Answer" field is set to "y".

SA8439 - Forward Held-Call CPN

SA8439 provides the forwarding of the Calling Party Number (CPN) associated with a held ISDN trunk call to the ISDN SETUP message created for the second leg of an outgoing call transfer or call conference operation. This change to normal CPN handling for a transferred or conferenced ISDN call is required by the Nortel BLU Service provider in Italy to cause billing of the PSTN charges to apply to an incoming ISDN call that is transferred to (or conferenced with) an outgoing ISDN trunk. (Currently, the SETUP message for the outgoing second leg of a call transfer or call conference to an ISDN trunk contains the CPN of the station used to perform the call transfer or call conference operation.)

SA8440 - Unmodified QSIG Reroute Number

This is a modification to the QSIG call forwarding feature currently supported by Communication Manager. When the QSIG call forwarding feature is active, a call that arrives on an ISDN trunk administered for supplementary service option B and is then forwarded offnet will cause a facility message to be sent back toward the switch that originated the call. The purpose of this message is to allow the originating switch to pick a better route to reach the forwarded-to party. The facility message contains the number of the forwarded-to party. This number is normally processed by Communication Manager so that the digits in the facility message are not the same digits as those entered when the call forwarding feature was activated. When the Unmodified QSIG Reroute Number feature is active, the number in the facility message will not be processed by Communication Manager so it will exactly match the number entered when call forwarding was activated.

SA8475 – SOSM

This feature provides the capability to observe incoming and outgoing call of Communication Manager endpoints (stations and extensions) and observe outgoing calls from Avaya Communications Manager endpoints to predefined set of telephone numbers within Public Switched Telephone Network.

NOTE: This feature is restricted and can only be activated for specific countries. Contact the Avaya Custom Development group, CustomCM@avaya.com, **for additional information**

SA8481 - Replace Calling Party Number with ASAI ANI

This green feature allows an ASAI application that is placing an outgoing ISDN call via the third party make call operation to supply information that will be used by Communication Manager to build the calling party information associated with the outgoing call. The ASAI application does this by including the desired calling party information in the User to User information element portion of the third party make call request. In order to use this green feature, a customer-supplied ASAI application is required along with the standard ASAI libraries.

new Beginning with Communication Manager 6.0.1 SIP trunks are supported.

SA8500 - Expanded UUI Display Information

This special application will provide for a new button on Callmaster IV display terminals. This new button will enhance the User-to-User Information (info button). The button display information will contain thirty-seven (37) characters for the following:

 10 - Digit ANI
 4 - Digit DNIS
 4- Last four digits of SSN
 16 - Digit account number
 3 - spaces

 This information will be displayed on the second line on the Callmaster IV

SA8506 - Altura Interoperability FIPN

This feature is to provide support for customer's that have Fujitsu switches. By providing an interface to the Fujitsu protocol, this will allow Fujitsu customers to not only expand or upgrade their networks, but to eventually transition their networks to Avaya's Communication Manager.

NOTE: This feature is restricted and can only be activated by the Avaya Custom Development group, CustomCM@avaya.com

SA8507 - H245 Support With Other Vendors

Currently, when a Communication Manager is connected to a third party vendor H323 VOIP gateway, sometimes no DTMF tones were passed through from Communication Manager to the other vendors and vice versa. Vendors like CISCO and Clarent are known to have this problem when they interop with Communication Manager.

In H323 standard, DTMF tones are passed via the H245 link. When Communication Manager connects with vendors like CISCO or Clarent, neither side of the call tried to bring up the H245 channel and no H245 channels were established, thus no DTMF were passed through. Between Communication Manager and Communication Manager, this was not a problem. Communication Manager has its own proprietary way of passing DTMF tones. This problem is only seen when

other vendors support FastStart call setup procedure, which is a procedure to setup call without establishing the H245 link

SA8508 - Multiple Emergency Access Codes

This feature is to provide customers two types of emergency calls, for example medical emergencies and all other emergencies. Each emergency type needs to be handled differently by the attendant, so each emergency type is provided with an unique display. Additionally, there is a need to be able to route emergency calls across the network to the location of the attendants.

SA8510 - NTT Mapping of ISDN Called-Party Subaddress IE

This Green feature maps Called Party "Sub address IE" digits sent with an incoming call's ISDN SETUP message into the ISDN UUI IE information stored by the Communication Manager call-processing for the call. When this call is routed to a VDN or station where ASAI event-notification has been set up by a CTI application, then the call's ISDN UUI IE information (i.e., the Sub address IE digit information) is sent to the CTI application in the "UUI data IE" field of an ASAI event message. This feature is supported for PRI, BRI, and IP trunks only.

SA8517 - Authorization Code By COR

SA8517 provides an enhancement to the standard MV Authorization Code Feature. The enhancement will give an extra level of control of system users' calling privileges by imposing a permission validation of the calling user's authorization codes usage.

SA8520 - Hoteling Application for IP Terminals

This feature is designed to work with customer installations which use a computer based scheduling system to reserve office space for their mobile workforce. Currently Avaya supports these applications with DCP digital terminals but not with IP terminals. The feature will allow the customer to reassign the extension number for a registered IP telephone using only the SAT interface.

SA8558 - Increase Automatic MWI & VuStats S8700 only

Currently, Communication Manager is limited to 1800 Remote Message Waiting Indication and 1000 VuStats buttons on the S8700 platform. This enhancement will expand the capabilities of the S8700 platform to allow for up to 5000 Automatic Message Waiting Indication (MWI) and up to 5000 VuStats buttons per system.

SA8567 - PHS X-Station Mobility over IP

This feature aims to provide user with PHS wireless phone solution over IP network. Integration of the Communication Manager. and the PRIMANET shall provide user with PHS handset that could have bridged appearances on wired terminals and would be able to use Communication Manager features like call forwarding and call pickup. Group Call Pickup, Transfer Recall, Circular Station Hunting,
and Reset Shift Dial capabilities will also be available to the set of features supported from X-mobile Stations.

SA8569 - No Service Observing Tone Heard by Agent

The Service Observing feature allows a supervisor to monitor the performance of a call center agent by listening to his/her calls. A warning tone is normally played to all parties connected to a call that is being service observed. This tone lets everyone know that the call is being observed. An agent could use the presence of the warning tone to modify his/her behavior, based on whether a particular call is being observed. If the 'No Service Observing Tone Heard by Agent' feature is active, the service observing warning tone will be played to all parties on the call except

for the call center agent. This fulfills the requirement of letting the caller know that a call is being monitored without letting the agent know.

SA8573 - Call xfer via ASAI on CAS Main

This feature provides the user with the capability to allow CAS Main calls to route to IVR ports or live agents, and then redirect this call via ASAI message to an extension/live agent either on the CAS Main or CAS Branch location.

SA8582 - PSA Location and Display Enhancements

The (SA8582) PSA Location and Display Enhancements green feature enhances the existing PSA feature. The PSA feature, in general terms, allows a user to associate a compatible terminal to the user's station-translated preferences and permissions. PSA makes it convenient for users to use a pool of phones at different times and locations.

This feature adds three enhancements to the present PSA feature. First, new terminal display messages and a special dial tone will be added so that the status of a PSA station (associated or disassociated) can be readily identified at the terminal. Second, a new port-location form will allow endpoint information to be associated with individual ports. This helps to alleviate the problem of incorrect information (e.g., room number from the last PSA-associated station's record) appearing on an attendant's display when a PSA-disassociated stations calls in a critical situation. Finally, this feature adds the capability of setting the COR for a PSA-disassociated stations on a per-port basis. This adds considerable flexibility to the present system where all PSA-disassociated stations share a single COR.

SA8587 - Networked PSA via QSIG Diversion

For **multi-server** systems that are networked via ISDN Media Transport facilities, Networked PSA will allow a station user to forward/Divert their number to another server and receive calls as if they were on their "home" server. Networked PSA will take advantage of QSIG Call Diversion and MSI messages to accomplish redirecting the call to the proper server. This feature will enhance the mobility of users between locations/servers and will improve business continuity.

SA8589 - Background BSR Polling

SA8589 is intended to provide a solution for Avaya call center customers who wish to use Best Service Routing (Avaya™ Virtual Routing) to poll a large number of Communication Manager sites via a "star" polling strategy, and then interflow the call to one of these locations with a less than 1-second delay. The BBP feature accomplishes this by constantly updating current "BSR best" location information via a continually-executing "background" Communication Manager call-processing task that loops through a sequence of "consider" vector steps. The BBP task stores the results of the its BSR polling in a "global" BSR status information table accessible to a new incoming call doing BSR call-routing to allow immediate interflow of the call.

SA8608 - Increase Crisis Alert Buttons S8700 only

Currently, Communication Manager is limited to 10 Crisis Alert stations on the S8700 platform. (This is in addition to the 28 Attendant Console capacity), each of whom may have a Crisis Alert button. This enhancement will expand the capabilities of the S8700 platform to allow for up to 250 total Crisis Alert stations. In Communication Manager 6.2 release, this limit is again increased to a new limit 750.

SA8621 - SCH Feature Enhancements

This feature was developed for a very specific configuration. It is not available for customer activation at this time. Contact the Avaya Custom Development Group, CustomCM@avaya.com, for more information.

SA8652 - No Hold Consult

This feature allows a party to consult another user without putting the current call on hold. The consult originator will have a two way talk path to all users in the call, whereas the consulted party will only be able to talk to the consult originator, but hear everyone.
The addition of another party i.e. the consulted party will be transparent to the other users already present in the call

SA8654 - Crisis Alert Call Monitoring and Recording

Current Communication Manager has the capability of providing emergency alert signal and display messages to designated terminals (attendant consoles & stations) when an emergency call is made from any station. However, the alerted terminals have no capability to monitor the conversation and/or record of such emergency calls.

SA8654 will enhance the standard Communication Manager software to have the capability of monitoring and/or recording such emergency calls with single button push from the designated **station terminals**. *Please note: the SA8654 capability is NOT available for the attendant consoles*. The capability is provided via **implicit** integration with barge-in capabilities from existing Communication Manager feature (e.g. Service Observing) for call monitoring and **explicit** integration with the Audix One-Step recording feature for call recording. In addition, new display messages will be provided to show the status (e.g. call is monitored by other stations etc.) of the emergency call.

SA8661 - Increased Automatic Wakeup Calls

Currently, Communication Manager in a S8700 configuration can deliver up to 950 wake-up calls in a 15-minute interval. SA8661 will enhance the wake-up call delivery capability up to 200 calls per minute. However, the enhancement **neither** increases the system capacity of the wakeup call allowed, i.e. 15000 calls, in the standard Communication Manager software, **nor** changes the delivery mechanism of wakeup calls.

SA8662 - Expanded PMS Name & Number

In order to support the 7-digit and 27-character name message, the Communication Manager software will be enhanced to process the existing PMS messages where the room extension number and/or guest name are normally embedded. These PMS messages are House-Keeper Status (**HKR**), House-Keeper Status from Designated Station (**HKS**), Message Waiting (**MSG**), Controlled Restriction (**CR**), Check-in (**CKI**), Check-out (**CKO**), **VIP**, Room Data Image (**RMI**), Room Change/Swap (**RMC**), and Guest Information Input/Change (**GIC**).

SA8684 - PMS Wakeup Message

Currently, an attendant or a guest requests an Automatic Wakeup call by dialing a feature access code (FAC) or through a wakeup button. This feature adds a third method in which the PMS can set or cancel an Automatic Wakeup call through a PMS message. In addition, the new Automatic Wakeup message allows the PMS to query for the current wakeup information for a particular guest.

SA8693 - Connectivity Check for Direct IP Shuffling

This feature "SA8693 Connectivity Check for Direct IP Shuffling"   green feature will execute an on demand ping test between the two IP stations on the call before allowing a direct IP shuffle. For the ping test to execute each IP station must be in different network region, the prowler board supplying the VOIP resource must be in the same network region as the IP station, and the port network of the prowler boards must be in the same port network group. A port network group is a group of port networks that have an EI and are administered with a set of fibers and SNI's for connectivity between each PN. If the ping test is successful the call is allowed to shuffle ip-direct if all other criteria are met. If the ping test fails, the call will stay on the TDM and not shuffle.

SA8697 - 3rd Party H.323 Endpoint Support

This feature allows third party H.323 IP endpoints to register with a Communication Manager gatekeeper and receive call features from the Communication Manager server in a similar manner to the calling features provided to an analog phone. These third party H.323 endpoints no longer need to authenticate with Communication Manager, as they did in past releases of Communication Manager. The number of endpoints allowed to register without authenticating will be controlled by a separate new entry in the license file for non-authenticating endpoints.

SA8701 - Net Region Support H.323 Endpoints Behind ALG

With the introduction of Application Layer Gateway (ALG) devices such as Kagoor VoiceFlow in the hosted environment and with the existing Communication Manager software, network regions can no longer be associated correctly with H.323 endpoints. The VoiceFlow device is shared among multiple enterprises or multiple locations within an enterprise. All the endpoints are behind NAT/NAPT devices and configured with the VoiceFlow IP address as their gatekeeper (GK). The VoiceFlow will open H.323 signaling connections to the Avaya GK on behalf of the endpoints. From Communication Manager's point of view, all H.323 registrations come from the VoiceFlow and thus network regions can no longer be correctly associated with the endpoints. To solve this problem, the VoiceFlow will include a new H.460.1 element which has the NAT WAN IP address of the endpoint in GRQ/RRQ registration messages. This new green feature will make changes on Communication Manager so that Communication Manager will parse the H.460.1 element in the GRQ/RRQ registration messages from the VoiceFlow and associate the network region with the endpoint based on the corresponding NAT WAN IP address.

SA8702 - CDR Enhancements for Network

This special application is developed and tested only for specific configuration and needs an explicit approval from Avaya for activation. Activating this special application might not produce desirable results in some scenarios. If you need any assistance for configuring the feature, go to the Avaya Support website at *http://support.avaya.com* and open a service request.

SA8731 - Block Outgoing Bridged Call Display

SA8731 will modify display for standard Communication Manager Bridge Appearance feature behavior such that for a given call appearance on a station which is bridge on to multiple stations, only the station that originated the call and the station that has the primary call appearance (which may be the same as the station that originated the call) will have access to the display information for that call. SA8731 will have no impact on the ability to bridge-on to calls or on the ability to originate or answer a call. This feature works for outgoing calls only there is no change to the display behavior for incoming calls.

SA8734 - Enhanced Extension Display

This feature provides the user with the capability to display up to 16 character Complete Phone Number with punctuations in the registration definition display area. It also provides the user with

the capability to display up to 11 digit long unpunctuated complete phone number for the Call Appearance and the Bridge Appearance Buttons. The registered Complete Phone Number is intended to be displayed at the registration definition display area.

SA8741 - CDR Identifier for IP Station Calls

This feature provides the user with the capability to identify the calls originated and terminated for IP stations in account-code field with an identifier 66666 for IP soft phone calls and 77777 for IP hard phone in CDR.

SA8744 - Block Name for Room to Room Calls

For room-to-room calls, the guest's name will be replaced with the administered room information from the site data section of the station form. A call is considered to be room-to-room call when both the caller and called party are administered as a "client room" COS. If nothing is administered on the station form the display will show the extension number

SA8747 - Softphone Indication on DCP Terminals

This feature provides the user with the capability to display the message "Extension in Use by Softphone" on a DCP terminal that is being controlled by an IP softphone. This feature is applicable only when TTI is enabled. Currently, for the DCP Terminals when the IP softphone takes control, the display on the DCP terminal gives no indication.

SA8758 - Auto Exclusion for Analog Bridged Extensions

This feature modifies automatic exclusion to block the principal from bridging onto an active call. Currently, auto exclusion will block any station that has a bridged appearance from joining an existing call but the principal can always join.

SA8759 - Invoke NCR Transfer on Answer Only

For the Nortel Network Call Transfer (Nortel-Transfer), ANSI Network Call Transfer (ANSI-Transfer), Enhanced ANSI Network Call Transfer (Enhanced-ANSI-Transfer), Telcordia Two B-Channel Transfer (Telcordia-TBCT), and ETSI Explicit Call Transfer (ETSI ECT) protocols supported by the standard Network Call Redirection (NCR) feature, the switch invokes the call-redirection operation by the PSTN switch for a call transfer or call-conference/drop when an ALERTing or CONNECT ISDN message is received from the far-end switch over the second call leg. This feature allows the Avaya customer to specify that NCR will be invoked only when a CONNECT ISDN message is received over the second call-leg. This means that when the second call-leg call is in the ALERTing ISDN call-state, the transferring party still has control of the call and can drop the second call-leg to route the call to a different PSTN endpoint before NCR is invoked. Note that this Green feature has no effect on the invocation of NCR by call vectoring (i.e., via the "route-to number ~r" or "queue-to best" vector steps).

SA8796 - Incoming Call Display with No/Delayed Ringing

The feature, when enabled, will provide the capability of displaying incoming call information on the terminating station, which has the call/bridged appearance's ring option set to no-ring or delayed-ring.

SA8797 - CTI Agent Call Capture by FAC

SA8797 provides an agent swap capability so that the logged in agent associated with an analog port can be changed while the TTY/TDD modem call remains connected without holding or otherwise interrupting the modem carrier and maintain CMS tracking of the call.

SA8835 - Conference to VDN

This special application is developed and tested only for specific configuration and needs an explicit approval from Avaya for activation. Activating this special application might not produce desirable results in some scenarios. If you need any assistance for configuring the feature, go to the Avaya Support website at http://support.avaya.com and open a service request.

SA8847 - Forced Disconnect of Diverted Predictive Calls

This feature provides an outbound dialer ASAI/CTI application with the capability to drop an outbound call when the call is diverted to a PSTN Voice Mail service or to another PSTN endpoint number by the PSTN switch. This Green feature is intended for use in Germany initially, but it can be used in any country where the "call diversion" NOTIFY message is supported by PSTN switches conformant with ETSI standard (EN 300 207-1 V3.1.1 (2001-(06)) and ITU-T Recommendation Q.699 (09/97). The SA8847 Green feature is not designed to work with ISDN-BRI trunks for outbound dialer calls from the Communication Manager, because we do not expect BRI trunks to be used by customers that use predictive dialing.

SA8851 - Remove Caller Id from Set Display

An incoming call from PSTN will generally provide calling party number and an administered text name string, which are displayed on the called party's terminal. This Special Application SA8851 provides a capability of blocking such information and replacing it with the incoming trunk group name and trunk access code. The RCISD (Remove Caller-ID from Set Display) feature will be based on the special application and administered station COR to determine whether the party's number/name will be displayed for external calls.

SA8852 - Display VDN Information on Route-To Calls

Currently, if a call routes to a station using the route-to number with cov=y vector command, then the call is treated as though the destination is directly dialed. Specifically, with respect to displays, the terminating station's display only shows the originating station information and does not show the VDN information. Similarly, the caller's display shows the name and number from the station where the call is routed, not the VDN information. SA8852 Display VDN Information on Route-To Calls allows the calling and called party to see the VDN information associated with the call on their station display. This feature will not affect the displays associated with the route-to number with cov=n vector command.

SA8853 - Support of LSPs Behind NAT

The hosted solutions offered by the Service Provider Organization (GMS On Demand) are facing Local Survivable Processor (LSP) registration and filesync problems due to Communication Manager (Communication Manager)'s current design of distinguishing LSPs by IP addresses only. To overcome the problems, the solution provided by this green feature is to slightly modify the configurations of enterprise sites and data center networks to let LSPs and Communication Manager communicate without Juniper VoiceFlow (VF) in the path (thru Network Address Translation (NAT) routers though), and make some Communication Manager code changes to enable Communication Manager to accept registrations from LSPs in enterprise sites in the hosted solutions. Filesync is made work by the network configuration changes. No Communication Manager code changes are necessary in order for filesync to work. The Alternate Gatekeeper List (AGL) element in GCF/RCF messages sent from active LSPs to IP phones will be omitted. This allows
IP phones in hosted solutions to retain the good AGL list sent from VF for the recovery purpose.

SA8854 - NCR OOB Transfer & Connect/Courtesy Transfer

This enhancement will allow the use of the Avaya Communication Manager Network Call

Redirection (NCR) feature with the AT&T 4ESS Out-of-Band Transfer & Connect (OOB T&C) protocol. Invocation of the 4ESS OOB T&C protocol is possible via Communication Manager call vectoring (i.e., "route-to number ~r", "queue-to best", and "adjunct-route" vector steps), as well as by IVR- or Voice Portal-initiated Lineside-E1/T1 call-transfer. The Avaya implementation of the NCR/OOB

T&C protocol will conform to AT&T "Toll Free Transfer and Connect" standard TR50075.

SA8859 - Default PSA Station

Special Application SA8859 introduces a system wide PSA/EMU logoff timer ranging from 15-X minutes.   This timer shall be used to determine whether an inactive PSA station should revert back to default or not upon off-hook. This timer typically is reset upon completion of a call, i.e. on-hook. This feature also introduces a per-station option to indicate the station can be used as a default station when the PSA session expires. A particular phone with the default option set, when used for PSA/EMU operation will revert back the pre-PSA station profile upon the PSA timer expiration. Stations without the option set, existing PSA operation will apply. For implementation simplicity, certain station administration restriction may be imposed when setting the option, e.g. the station must have physical port associated.

SA8869 - SIP Caller-ID Blocking in a Hosted Environment

This feature allows OPTIM SIP endpoints to block caller-id on calls placed from an OPTIM station. Normal Communication Manager operation considers a call originated by the OPTIM endpoint as an internal call even though the call is originated over a SIP trunk. This development allows an OPTIM originated call to have its caller ID blocked to other OPTIM and non-OPTIM endpoints and over trunks.

SA8870 - Tandem Network Call Redirection

This feature enables Avaya customers to use the NCR feature for a configuration where the customers switch is connected via a tandem Communication Manager switch to a PSTN switch and reduces the need for PRI B-channels between these switches when NCR is invoked at the customers switch, hence providing cost benefits. The scope for the SA8870 Green feature currently only covers tandemed NCR invocation support only for the Telcordia Two B-Channel Transfer (TBCT) call redirection protocol that is provided by the PSTN.

SA8874 - Call Status Messages for 7434ND IP Softphone

This Green feature extends the endpoints supported for call-progress states reported by Communication Manager to the station endpoint type of 7434ND configured as an "IP softphone" as used for an "IP line-side connection" between a Communication Manager and the Voice Portal 3.0 or Interactive Response 2.0 adjunct. The additional call-progress states that will be reported for the 7434ND "IP Softphone" station are "ringback connected", 'busy tone connected", "intercept tone connected", and "reorder tone connected".

SA8876 - Expanded Holiday Table

SA8876 expands the holiday table to accommodate up to 99 holidays in one holiday table. SA8876 is based on the standard Holiday Coverage feature provided in a previous Communication Manager release that uses the standard Call Center "Holiday Vectoring" form (with 15 entries per form). With the introduction of the SA8876 Holiday Coverage Table feature, the new Holiday Coverage Table is

provided for administration in the same way as previously was done with the "Holiday Vectoring" form, but the number of entries per Holiday Coverage Table is increased to 99.

SA8879 - DCP Xfer Lamp Control/Buttonless Auto Exclusion

Button-less Auto Exclusion

When this new exclusion system wide parameter is set, the feature can be activated on all Bridge Appearances on a COS basis. All bridged appearances which have this feature activated on their COS will operate in exclusion mode regardless of the existence of an administered exclusion button. Currently this feature is only administrable on a per-station basis by administering a feature exclusion button. This feature relaxes the requirement to use a feature button.

DCP Transfer Lamp Control:

When a user (principal/Bridge station) initiates transfer or conference (places the appearance on soft-hold), the bridged appearances of that line appearance will follow the active call lamp settings. This is important so that the other station users that have a bridged appearance of the line will not think the line is on hold, while the call is transitioning to a transfer/conference. The behavior of the lamps upon activation of this feature will be independent of DCP LED settings. The appearance of the principal on the transferring/conferencing station will follow the soft hold call lamp settings, which is the same as that happens today.

SA8885 - US Time for User-Defined language

Special Application SA8885 allows a customer to choose "user-defined" as their station language while keeping the station's clock to display the 12- hour clock.

SA8886 - ISDN Incoming Calling Party Number Conversion

This feature will enable customers to support a more flexible Incoming Calling Party Conversion on ISDN trunks. With this feature Communication Manager will be able to support conversions for 24 digit **incoming** calling party number, increased from the current limit of 15.

When you set the **SA8886 - ISDN Incoming Calling Party Number Conversion** field to y, the system displays the Calling Party Number Conversion screen. For complete description of the fields on the Calling Party Number Conversion screen, see APPENDIX A

Communication Manager converts the incoming calling-party number to an appropriate numbering format. Also, Communication Manager converts a calling-party number that has more than 15 digits to the supported length of 15 digits.

Minimum load – 6.3.2

SA8887 - Hotline for IP Telephones

With the SA8887-Hotline for IP telephones feature, IP stations can automatically place a call to a predetermined number inside or outside the PBX when the IP station goes off -hook. This predetermined number is stored in Communication Manager as part of the abbreviated dialing list. Unless prompted for authorization, the hotline user receives calls normally. The term *Hotline Station* refers to the IP station configured with the hotline feature. The term *Hotline Destination* is the predetermined number. In Communication Manager Release 6.2 and later, the SA8887-Hotline for IP Telephones feature supports overlapped signaling on QSIG trunk.

Note: The Hotline feature does not work for SIP endpoints.

SA8888 - Per Station Music On Hold

Special Application 8888 will allow assigning different music sources to individual stations without the Tenant Partitioning system custom option enabled. SA8888 will introduce a new field, "**Music Source**", in the station administration.

SA8889 - Verizon VoiceGenie SIP MIME Message Bodies

Verizon requires their inbound directory assistance and other applications' calls to be filtered via their VoiceGenie (brand name) IVR before being sent to Avaya Communication Manager (Communication Manager) via a SIP trunk. The incoming calls to Communication Manager are directed to Vector Directory Numbers (VDNs) to be distributed by call vectoring to Verizon EAS (Expert Agent Selection) agents within the same Communication Manager server.

The incoming SIP calls will contain a proprietary SIP MIME attachment message body of customer information added by the IVR. Communication Manager will store that information. The AE Services server will send a message to Communication Manager for each extension that it wants Communication Manager to monitor. AE Services will do so for every agent, when the agent logs in. Communication Manager will keep track of the list of extensions to monitor and will automatically send the SIP MIME attachment message body data to the AE Services server whenever a call terminates to one of the monitored stations. This is the normal method of operation.

Communication Manager will also be able to send the SIP MIME attachment message body data to the AE Services server whenever Communication Manager is queried by AE Services. This method will be used after the call is answered. This method won't be used on every call, but rather to handle complex scenarios.

The query will be used to refresh data after it was sent automatically.

SA8891 - Verizon VoiceGenie SIP Headers

Verizon requires their inbound directory assistance and other applications' calls to be filtered via their VoiceGenie (brand name) IVR before being sent to Avaya Communication Manager (Communication Manager) via a SIP trunk. The incoming calls to CM are directed to Vector Directory Numbers (VDNs) to be distributed by call vectoring to Verizon EAS (Expert Agent Selection) agents within the same Communication Manager server.

The incoming SIP calls will contain a proprietary SIP Header of customer information added by the IVR. Communication Manager will store that information. The AE Services server will send a message to Communication Manager for each extension that it wants Communication Manager to monitor. AE Services will do so for every agent, when the agent logs in. Communication Manager will keep track of the list of extensions to monitor and will automatically send the SIP Header data to the AE Services server whenever a call terminates to one of the monitored stations. This is the normal method of operation.

Communication Manager will also be able to send the SIP Header data to the AE Services server whenever Communication Manager is queried by AE Services. This method will be used after the call is answered. This method won't be used on every call, but rather to handle complex scenarios. The query will be used to refresh data after it was sent automatically.

SA8893 - Blast Conference

This special application allows an endpoint to dial a single number and have CM call to up to 8 other pre-administered endpoints, conferencing them all together.

Note: This special application has not explicitly been designed or tested to work with SIP-based deskphones or clients.

SA8896 - IP Softphone Lamp Control

SA8896 will control the lamp updates passed to IP softphones. If SA8896 is turned on, Communication Manager will send lamp updates to the IP softphones according to the Communication Manager default LED settings specified bellow. This feature does not affect any terminal types other than IP softphones. The feature will be active system wide if the SA8896 green bit turned on.

SA8900 - Support for NTT Call Screening

Special Application "(SA8900) – NTT call screening" allows a user to block malicious calls, clear the list of the screened calls and clear the last added entry in the screened list. The blocking of the malicious call is provided by the NTT PSTN service provider. The SA8900 Green Feature is intended for use only in Japan and is designed to work with ISDN-PRI and ISDN-BRI trunks for outbound calls from the Communication Manager.

SA8904 - Location Based Call Type Analysis

This Special Application SA8904 provides an ability for a customer to dial short-extension to reach other extension with in their site or location. With the introduction of SA8904 feature customer can reduce their multiple switches to single switch without changing their current extensions. User can dial the site based extension to call with in site or location. When SA8904 is administered the calls are routed based on Communication Manager 4.0 standard feature Enbloc Dialing and Calltype Analysis Table. This feature will be available in all Communication Manager 4.0 platforms and beyond.

SA8911 - Expanded Public Unknown Table

This Special Application SA8911 provides a capability to administer a maximum of 20,000 entries in public unknown numbering table on S87XX/S85XX servers and a maximum of 9999 entries on S84XX/ S83XX

SA8917 - LSP Redirect using special coverage point

In general stations on inaccessible port networks or media gateway controlled by local survivable processor (LSP) are considered busy and any calls to those stations whether internal or external are treated with busy criteria. In service provider network when the WAN link fails, all the incoming PSTN calls reach primary Communication Manager. Communication Manager treats IP phones registered to LSP as out of service and call fails with user busy release cause if no coverage path is administered or calls are covered to Voicemail if coverage path is administered. This is unacceptable scenario for service providers and so this green feature is needed. This feature gives the ability of routing calls to stations controlled by local survivable processor (LSP) over public network /PSTN. The call is routed by administration of special coverage point and using remote coverage feature.

SA8927 - Increase Paging Groups

Special Application SA8927 increases the total number of paging groups to 256 for the small platforms and 999 for XL. The total members per group will not change.

SA8928 - Display Names on Bridged Appearance Labels

Currently Communication Manager does not provide an option to display the user name on bridged appearance for phones supporting button labels. The default button label for bridged appearances is the extension of the primary user. The user can display the .user name. by creating custom labels for the bridged appearances via the phones "Options" screen. However, this can become difficult to maintain if there is constant movement of users or bridged appearances.

Special Application SA8928 shall provide an option on the class of restrictions form (COR) to display user name on bridged appearances labels instead of the extension. The end user will still have the option to override the default label with a custom one.

SA8931 - Send IE with EC500 Extension Number

SA8931 will send an ISDN Redirect Number IE (0x74) when a call is extended to EC500 mobile phone (over ISDN trunk or H.323 trunk). The redirect number is the original called PBX EC500 extension. The Redirect Number IE is to be sent along with SETUP message. The calling party number IE (0x6C, if available) and called party number IE (0x70, the cell phone number) are part of the **SETUP** message.

SA8942 - Multiple Unicode Message File Support

Communication Manager software currently supports the install and loading of Communication Manager Phone Display Messages from the external Unicode files "avaya_unicode.txt" and "custom_unicode.txt"
SA8942 will add support for 3 additional Unicode files. After this feature is implemented a customer will have the ability to install 4 translations of the Communication Manager Phone Display Messages.

SA8944 - Multiple Logins for Single IP Address

Currently Communication Manager is able to handle the scenario where an IP endpoint tries to register with an extension that is already in use. Communication Manager is able to make the distinction between an IP endpoint which is recovering and is requesting the same extension from a separate IP endpoint requesting the same extension by comparing IP addresses and MAC addresses. When the IP address or MAC addresses do not match then Communication Manager instructs the IP endpoint to display the message "Extension in Use - # to Register". At this point, the user can press "#" to initiate a forced login. A forced login causes the existing registration to be unregistered so that the new registration can be successful.
Unfortunately Communication Manager cannot make this distinction for Service Provider deployments using network based Session Border Controllers (SBCs) providing an ALG/NAT/NAPT functionality. In this environment, each IP endpoint will have the same SBC proxy IP address. Therefore, Communication Manager treats all registrations attempts with an

already registered extension as if the IP endpoint is recovering. This includes the scenario where a softphone attempts to register in share control mode via the server.

SA8944 will address these issues by only using the MAC address sent in the RRQ's nonStandardData LoginRequest segment.

SA8946 - Site Data Expansion

This Special Application SA8946 gives customers the ability to administer up to 225 Building data. SA8946 increases the existing site-data from 75 to 250 Building Fields. With the introduction of SA8946 feature customer can administer up to 225 Building data, increased site data will work as earlier.

SA8958 - Increase BSR Polling/Interflow Pairs to 40000

Note: Due to the complexity and/or risk involved with activating this feature, it is restricted to Business Partners and Avaya activation. Please contact your Business Partner, Avaya Service Representative or Avaya Custom Development group, CustomCM@avaya.com.

SA8965 - SIP Shuffling with SDP

The new shuffling capability will be restricted to scenarios where both endpoints are SIP and both are not Avaya CMs (CM SIP trunk-to-trunk shuffling) to minimize the development and test effort as well as the risk factor involved with this change.

NOTE: This feature is restricted and can only be activated by the Avaya Custom Development group, CustomCM@avaya.com

SA8967 - Mask CLI and Station Name for QSIG/ISDN Calls

The feature provides a capability of "masking off" the calling party number and name, which is normally sent for outgoing QISG/ISDN calls. It also block the called party's name and connect number off for incoming QSIG/ISDN call.
In Communication Manager 6.2 release, SA8967 behavior is enhanced to support a permanent state of send-nn button. Send-nn button has a new 'call mode' qualifier which can be set to 'permanent' or 'transient'. When set to 'transient', send-nn button retains an existing behavior of needing to press send-nn button every time a dummy identity is to be sent. When set to 'permanent', send-nn button toggles between 2 states – ON, OFF. That means, once the feature is turned ON, it will remain ON until it is turned OFF (by the user). Default is OFF. If turned ON, a dummy identity is sent, otherwise original identity is sent.

SA8972 - Overwrite Calling Identity

The Service Provider requirements for SIP communication during call forwarding are:
1. From header must contain forwarding identity (may or may not be fully qualified E.164)
2. P-Asserted-Identity must contain forwarding identity (may or may not be fully qualified E.164)

Avaya - like all other vendors supplying TI - has been asked to modify the way its equipment behaves so as to update the 'From' and 'PAI' headers in the INVITE message to show the number of the redirecting party for the call forwarding feature. This must be applied to all 3 call forwarding feature scenarios: CFU, CFB, and CFNA.

Note: Due to the complexity and/or risk involved with activating this feature, it is restricted to Business Partners and Avaya activation. Please contact your Business Partner, Avaya Service Representative or Avaya Custom Development group, CustomCM@avaya.com.

SA8983 - Replace CPN when calls forwarded off-net

This Special Application SA8983 modifies the calling party number for outgoing ISDN trunk calls that are the result of either call forwarding to an external number or the EC500 feature. With this feature active, the calling party number for these calls will be that of the call forwarding or EC500 station rather than that of the call originator. This feature works for both station-to-station and for incoming trunk calls to the call forwarding or EC500 station. All types of call forwarding – unconditional, busy/don't answer, and enhanced call forwarding – are supported.

SA8986 - Simultaneous Ring of Attendants

The special application SA8986 provides an attendant the ability to answer a call that is ringing at a different attendant console. An attendant presses the call pickup button to activate the feature.

SA8993 - Increase Tenant Partitions

This Special Application SA8993 provides the ability to customer to administer tenant Partitions up to 250. SA8993 increases the existing Tenant Partitions from 100 to 250. With the introduction of the SA8993 feature customer can administer up to 250 Tenant Partitions, increased Tenant Partitions will work as previously it use to be. In the same way Class of Service, Listed Directory Number and Music sources are also increased from 100 to 250, will work as previously it use to work.

SA9004 –Multi-Location Call Routing for IP-Trunk

Special Application (SA), **'(SA9004) – Multi-Location Call Routing for IP-Trunk'** is used primarily in case of IP-DECT stations administered across multiple location controlled by one Communication Manager and one CLAN terminating IP-Trunks from multiple IP-DECT sites. This result in anomalous routing of call based on the location of CLAN. The objective of this feature is to provide a mechanism by which correct location of IP-DECT station is ascertained irrespective of location of CLAN, thereby providing correct routing of call. The feature is also extended for other applications on H.323 IP-Trunks. This special application is dependent on Multi-Location feature.

SA9009 - Duplicated Processor Ethernet for Sip

Allow sip connections on S87xx processor ethernet using the logical address instead of physical for this connection. This provides connection preserving interchanges for sip calls on that interfaces as well as connection preserving upgrades.

SA9014 – PIN Release of Time of Day Station Lock

SA9014 - PIN Release of TOD Lock is designed for customers who want to release their Time of Day (TOD) locked stations for a period of time. It is an enhancement for TOD Station Lock feature.

SA9016 - G.726 Support on TN2302

This special applications (SA9016) development brings support for G.726 transcoding to the cruiser board (TN2302 V110 and above).

SA9018 - Support 255 Bridge Appearances per Call-appr

This Special Application SA9018 provides a capability of administering a maximum of 255 Bridge Appearances per call appearance of a station. This feature will work on all platforms as it was working with previous limitation of 63 Bridge Appearances per call appearances of a station.

SA9020 - IGAR triggered by IP network performance

Special Application "(SA9020) – IGAR triggered by IP network performance" provides the ability to trigger Inter-Gateway Alternate Routing (IGAR) dynamically, when corporate data network performance degrades to the point where voice call quality becomes unacceptable.

SA9025 - ISDN Subaddress Treatment

This Special Application SA9025 provides the ability to send and receive NSAP subaddress information with network elements. Customer can administer trunk groups to send and receive NSAP subaddress information. With the introduction of the SA9025 feature customer can have tunneling between there networks through public network, subaddress information elements will be part of ISDN SETUP message.

SA9028 - Station TBCT for Heavily Trunked Systems

This feature was developed for a very specific configuration. It is not available for customer activation at this time. Contact the Avaya Custom Development Group, CustomCM@avaya.com, for more information.

SA9035 - Increased Intercom Groups

This Special Application SA9035 provides a capability of administering a maximum of 1024 Intercom Groups on S87xx/S85xx servers and a maximum of 128 Intercom Groups on S84xx/S83xx, instead of 256 on S87xx/85xx and 32 on S84xx/ S83xx.

SA9050 - Increased TOD Routing Tables/Partition Grp Num

This Special Application SA9050 provides a capability of administering a maximum of 32 Time OfDay Routing Tables/Partition Group Numbers on S87xx/S85xx/S8300 in Local survivable processor (LSP) mode and a maximum of 8 on S84xx/ S83xx, instead of 8 on S87xx/85xx/S8300 in LSP and 8 on S84xx/ S83xx.This increase of Time Of Day (TOD) tables/Partition Group Number on respective platforms will be as per the following table.

SA9065 - Crisis Alert to Stations by Location

With this Special Application Communication Manager alerts stations with a crss-alert button only in the same physical location as the calling phone.

SA9070 - Increased Logged-In ACD Agents for XL

Special Application "(SA9070) Increased Logged-in ACD Agents for XL allows the Logged-In ACD Agents field on the system parameters customer options for to be set by an RFA/Poetic license file up to a maximum of 10,000 when SA9070 is active with the S8720XL or S8730 Media Servers and associated LSP/ESS platforms (S8300C, S8500X or the S8720XL.

Note: Due to the complexity and/or risk involved with activating this feature, it is restricted to Business Partners and Avaya activation. Please contact your Business Partner, Avaya Service Representative or Avaya Custom Development group, CustomCM@avaya.com.

SA9073 - Use Called Party Location For LWC Time/Date

This Special Application SA9073 – Use Called Party Location for LWC Time/Date provides ability to store the LWC message time stamp depending on the called party's location time zone. SA9073 modifies the time stamp of LWC message system Time/Date to Called Party's Location Time/Date. With the introduction of this feature user can easily to find the LWC message Time/Date depend on his/her location time zone.

SA9075 - Transfer to Any Local Voice Mailbox

This Special Application SA9075 provides the capability to transfer a call to a specified voice mailbox on a Modular Messaging system. The ANI associated with the call being transferred is provided to the Modular Messaging system so that it can be stored along with the message.

SA9077 - Priority Call Follow Coverage

This Special Application SA9077 – Priority Call Follow Coverage provides the ability to follow a cover path for priority call if called number did not answer the call depending on the calling party's Console Permission. SA9077 modifies the priority call to follow Coverage Path. With the introduction of this feature user can make a priority call and leave a voice mail to called party if called party did not answer and if Do-Not-Disturb was enabled.

SA9086 - Mask CLI on PSTN Calls

SA9086 allows replacing the station name and number with an administrable string (which can be name of the enterprise) and number of up to maximum 15 characters on an **outgoing** PSTN call when enabled. For internal calls i.e. trunks that don't have this feature administered the called party will see the name/number of the calling party**.**

SA9089 - Support MCT Recording on H.248 Media Gateway

This feature enables MCT call recording on H.248 media gateways using analog trunk media module with loop-start signaling as an interface between recorder and Communication Manager. This document does not cover administration of basic MCT feature

SA9090 - Increased MIS CMS Applications

This Special Application, SA9090 Increased MIS (CMS) Applications, allows up to 4 CMS applications to be administered. This will allow the ability to separate the adjunct links and reporting capabilities of CMS into two separate HA CMS pairs, one pair for call routing and the other for reporting.

SA9094 - NCR Support of Nortel Release Link Trunk

SA9094 provides for the use of the Network Call Redirection (NCR) feature with the Nortel Release Link Trunk (RLT) protocol. RLT provides the user with cost savings when transferring two calls through the pubic network. Once the calls have been established, the network is instructed to connect the end users in the network and drop the connections to the served user PBX. In other words, the transfer functionality is moved from the PBX to the PSTN, thereby eliminating the need to keep the trunks up through the PBX for the call duration.

SA9095 Hunt Group Modifications

This feature comprises two changes to standard hunt group behavior. Currently, calls to a hunt group will skip over any stations that have call forwarding activated. This feature will change this behavior so that stations in a hunt group with call forwarding active will be available to receive calls made to the hunt group. There will also be a new timer implemented to deal with unanswered calls to a hunt group. This timer will provide a subset of the capabilities in the existing Redirect On No Answer feature. Calls to the hunt group that are not answered within the timeout interval will be redirected to a different station in the hunt group. These new capabilities will be administrable on a per hunt group basis. This feature applies to non-ACD hunt groups only.

Minimum load - Communication Manager 6.0.

SA9096 Increase Paging Group Members

By implementing this feature customer shall be able to administer paging group members up to 127. The call flow and call processing will remain the same for 127 group members as it used to work earlier for 32.

Minimum load - Communication Manager 6.0

SA9099 - Natl/Intl CPN Prefix in CDR Records

Increase the calling number field on the customized CDR form to 20 digits. This will allow CDR records to capture the National/International code.

For all incoming trunk calls over ISDN PRI/H.323, determine if it is a national/international call and implement logic to prefix the national/ international code in the calling party number captured in the CDR.

Calling Party Number: Currently maximum supported length of calling party number is 15 digits and that of National/International CPN Prefix is up to 5 additional characters. So the CPN digits maximum length will be 20 characters.

Minimum load - Communication Manager 6.0.1

SA9104 CDR Answering Party via Call-Pickup

This feature will change the operation of CDR (Call Detail Recording) such that the actual party that answers a call via call pickup is considered to be the destination party on the record. In the case where the dialed party answers the call, then no change is seen from the standard operation.
Minimum load – Communication Manager 5.2.1 SP 4, Communication Manager 6.0 SP2, Communication Manager 6.0.1

SA9105 Modify Authorization code dial sequence

Currently Communication Manager allows authorization code dial sequence as follows
1. Acct-code FAC + Account code
2. ARS FAC + destination Number
3. Auth-code call is made successfully
But, Nortel switch does different way dialing sequence as follows
1. Auth-code FAC + Auth-code, dial tone confirms the validation
2. Acct-code FAC + Account code, dial tone confirms the validation

3. ARS FAC + destination number call is made successfully

This feature will change the dialing sequence to match the existing Nortel sequence.

Minimum load – Communication Manager 5.2.1 SP 4, Communication Manager 6.0 SP2, Communication Manager 6.0.1

SA9106 EC500 Call Pickup and Bridging

This feature will give users additional security on the EC500 feature by disabling bridging on by the principle phone (or any of its bridges) and will also give them the flexibility to join in from the hard phone after dropping off the call from the cell phone, within the administered amount of time. This feature behaves like "Auto Exclusion" by disallowing bridge-on.

Minimum load – Communication Manager 5.2.1 SP 4, Communication Manager 6.0 SP2, Communication Manager 6.0.1

SA9107 Attendant Camp-on to busy station

Currently, if the attendant extends a call to a multi-appearances phone which is busy on the other call but still has at least one call appearance idle, the extended call will behave as the followings:
- If this phone doesn't have any coverage path administered, the extended call rings on one of the idle appearance until either the user answers the call, or the call returns to the attendant after the attendant recall timer expires.
- If this phone has coverage path administered, the extended call rings on one of the idle appearance until either the user answers the call, or the call goes to the coverage.

SA9107 will modify the extended call in the above scenario to always returns to the attendant, if no one answers the call, and the attendant recall timer expires no matter whether there is coverage path administered or not.

Minimum load – Communication Manager 6.0.1

SA9108 - Local Time Support for CDRs

Capture the local time of the endpoints in the CDR record.

Minimum load – Communication Manager 6.0.1

SA9109 - Location of Last Answered Party in CDR

CDR to capture the location of the last party who answered the call if an incoming call involves multiple locations and ITCS is disabled.

Minimum load – Communication Manager 6.0.1

SA9110 - Support of VDN Timezone in CDR

For calls involving VDN, CDR to capture the timezone of the VDN. Without SA9110 CDR record "***"(three asterisks) in the CDR if a call is made to VDN.

Minimum load – Communication Manager 6.0.1

new SA9111 Allow Conference with SA8434

Communication Manager has following special-applications SA8434, when this special-applications is enabled Communication Manager doesn't send ISDN Connect (200OK in case of SIP) message to incoming call answered by an ACD agent. The reason for not sending CONNect/200OK is to prevent the PSTN network from billing the call.

Problem is before Communication Manager.6.2 release, the software **doesn't allow completing the conference** unless CONNect/200OK is sent out to incoming trunk. But the CONNect/200OK is sent out when agent presses the 'start billing' button. Now, with SA9111 activated the agent is able to complete the conference before CONNect/200OK is sent out.

Note: SA9111 is enhanced in Communication Manager 6.2 release to support blind conference scenario as well. Before Communication Manager 6.2, only attended conference is supported to interact with SA8434.

NOTE: Requires activation of SA8434

Minimum load – Communication Manager 5.2.1 SP 5, Communication Manager 6.0 SP2, Communication Manager 6.2

new SA9114 Expand Public Numbers to International for ASAI?

By default the ASAI applications the calling/called party number in the original format as it is generated/received on Communication Manager (national, local, int'l). Due to this, number matching on ASAI applications becomes difficult. If SA9114 is turned on, the number sent to AES from Communication Manager is normalized by Communication Manager to E.164 international.

Minimum load - Communication Manager 6.2, Communication Manager 6.0.1 Service pack 03, Communication Manager 5.2.1 Service pack 08

new SA9115 Increase ip-network-map Table

By default, ip-network-map has 500 entries. SA9115 has increased this limit to 4000 entries. The increased limit is useful for enterprises which have offices distributed across many geographical locations/subnets.

Minimum load – Communication Manager 6.2

SA9119 - Exclusion Event Notification to SIP Phones

The **Exclusion Event Notification to SIP Phones** feature works only with telephones on which you enable **Bridged Call Appearance**.

In an active call, when the principal user or one of the bridge users presses the **Exclusion** button, Communication Manager sends an exclusion notification to all SIP telephones active in the same call. On receiving the notification, all SIP telephones except for the exclusion requestor drop off from the call. However, if you set the **Block Exclusion Event Notification** field on a COR screen to **y**, Communication Manager does not send exclusion notification to SIP telephones that are administered in the COR.

To enable the **Exclusion Event Notification to SIP Phones** feature:

- Set the **Exclusion Event Notification to SIP Phones** field on the Special Applications screen to **y**
- Set the **Block Exclusion Event Notification** field on the Class of Restriction screen to **n**

The following table lists the situations when Communication Manager sends exclusion notification to SIP telephones:

Exclusion Event Notification to SIP Phones	Block Exclusion Event Notification	Exclusion notification sent?
Y	n	yes
Y	y	no
N	n	no
N	y	no

Minimum load – 6.2 SP4

SA9120 - Mute speakerphone for shared control configuration

Using the shared mode of soft phone or ASAI, users can control the desk phone remotely. In such configurations, users can remotely hear the conversations around the desk phone, which may be considered as a privacy breach. SA9120 addresses this potential privacy breach in the following manner.

When you enable the **Turn on mute for remote offhook attempt** field on the station screen, any off-hook event on the desk phone will result into turning on the mute button automatically. With the **mute** button activated, the user cannot remotely hear conversations around the desk phone. This feature is applicable to the following cases of remotely controlling the desk phone:

- Calls received or originated remotely from soft phones in a shared control mode.
- Calls received or originated remotely by using ASAI. Currently, the following ASAI functionalities are supported: ASAI MakeCall, ASAI AnswerCall, and ASAI Single Step Conference (SSC). Communication Manager turns on the mute button for ASAI MakeCall and AnswerCall and **not** for ASAI Single Step Conference (SSC).

Note: If the call is originated or received from the deskphone, Communication Manager does not turn on the mute button.

In H.323 configuration, Communication Manager controls the signaling of activating the mute button for the off-hook event.
In a SIP configuration, the SIP phone itself controls the functionality to activate the mute button. To activate this feature, set the following fields to *y*:

- **(SA9120) - Automatic Mute for Remote Off-hook Attempts?** field on the change system-parameters special-applications screen
- **Mute on off-hook event in shared control?** field on the station screen.

Caveats

- This feature is not supported for the stations configured with the Auto Answer and the int-auto-answer functionalities.
- This feature is not supported with the softclients that are capable of sending offhook event to Communication Manager, using speakerphone, headset, or handset. Typically, softclients with the Picture Of Phone functionality are capable of sending such offhook events.

- When SA9120 is enabled, Communication Manager gives priority to security over usability. Due to this behavior, in few intricate scenarios, mute button is turned on even though the user does not expect it to be turned on.

Endpoints supported

- **96x0 H.323**: 9620, 9630, 9640, and 9650
- **96x0 SIP**: 9620, 9630, 9640, and 9650
- **96x1 SIP**: 9608, 9611, 9621, and 9641
- **96x1 H.323**: 9608, 9611, 9621, and 9641
- **16xx H.323**: 1603, 1608, and 1616

Minimum load – Communication Manager 6.3 SP2

SA9122 - Restrict Public-Private Network Connections

With the **Restrict Public-Private Network Connections** feature, Communication Manager checks whether a call complies with Indian toll regulations. In case of noncompliance, Communication Manager restricts calls between the enterprise users and public users from external locations.

The **Restrict Public-Private Network Connections** feature is only for audio calls.

To activate this feature, on the change system-parameters special-applications screen, set the **(SA9122) - Restrict Public-Private Network Connections (Toll Bypass)** field to y. The default value of this field is n.

Minimum load – 6.3

SA9123 Re-ring CAG Members in Adjacent Coverage Points

In the absence of the SA9123 feature, Communication Manager does not alert the stations that are common to two adjacent coverage answer groups. The default setting for this feature is *n*. When you enable the feature, Communication Manager alerts the stations that are common to the two adjacent groups.

Minimum load – 6.3

SA9124 - AACC Connected Information Enhancement

When you activate the SA9124 feature, the ASAI protocol interface for link version 5 displays the call information. The processing of call information depends on whether the call is an inbound call or an outbound call.

Inbound calls: When an ASAI CTI application-monitored station or VDN receives a call, the calling-party information is stored in Communication Manager. The ASAI alerting event and the call-offer events use the calling-party information stored in Communication Manager. ASAI

includes the connected-party information in any query or subsequent events, such as connected, transfer, conference, and drop. The connected-party information is the same as the calling-party information.

Outbound calls: When a station makes a call to a trunk destination and Communication Manager receives an alerting event from the trunk, the alerting event might or might not have the connected-party information. If the connected-party information is available in the ISDN-PRI alerting event, then the ASAI alerting event uses the connected-party information to display as the alerting-party information. ASAI includes the connected-party information in the subsequent event, such as connected, transfer, conference, and drop. If the connected-party information is unavailable, Communication Manager sends the default dynamic identifier (#####) as the alerting-party information. When the connected-party information is available, the subsequent events or queries use the connected-party information that is stored in Communication Manager.

In the absence of the connected-party information, ASAI uses the called-party information as the alerting-party information only if both, SA9114 and SA9124, are enabled.

Note: Communication Manager stores only the connected-party information not the called-party information.

Minimum load – 6.3

SA9125 – Select Standard Transfer Dynamically for SEMT

SA9125 is applicable only to Softflare, where the AST2 transfer falls back to AST1 if the SIP REFER message does not contain the Replaces message in the supported header.

Important – Avaya does not recommend the usage of this field to the customers. This field is used by Avaya only for internal testing.

Minimum load – 6.3.2

SA9126 – Additional EC500 FNE Security Check

When the caller makes an EC500 call from a cellular phone, Communication Manager validates the caller ID and the station security code (SSC) assigned to the extension of the user. The Security Code field is on the Station screen.

When you set the **SA9126 - Additional EC500 FNE Security Check** field to **y**, the additional security for an EC500/One-X Mobile Lite call (AEFSC) feature is activated on the system. SA9126 Additional EC500 FNE Security check field is on page 10 of the Special Applications screen. After the feature is activated on the system, you can set the AEFSC Security field to **y** on each station to make the feature available to the EC500 users.

Note: You can change the value of **AEFSC Security** field only when you set the **SA9126 - Additional EC500 FNE Security Check** field on page 10 of the **Special Application** screen to **y**.

AEFSC Security field description

Valid entry	Usage
y	The AEFSC feature is activated for an EC500 user. Communication Manager checks the caller ID and the SSC that is assigned to the extension of the user.
n	The AEFSC feature is deactivated for an EC500 user. Communication Manager checks only the caller ID. The default value is n.

Minimum load – 6.3.2

SA9127 – Drop Ad-hoc Conference when Host drops

When you set the value of **SA9127 – Drop Ad-hoc Conference when Host drops** to y, an active conference call disconnects when the user who initiated the conference call drops from the call.

Communication Manager marks the extension that is used to perform a conference operation on a call as the original ad-hoc conference host. The conference call drops when the original ad-hoc conference host drops from the call.

For example, a user at Station A calls Station B, and the user at Station B conferences Station C into the call. Communication Manager marks Station B as the original ad-hoc conference host. When the user at Station B disconnects the call, the call drops.

Minimum load – 6.3.6

Interactions with other Communication Manager features

This section provides information about how the SA9127 feature interacts with the other features in the system.

Transfer

On a conference call, a user cannot use the extension that is marked as the original ad-hoc conference host to perform a Transfer operation.

Bridged Call Appearance

If a user uses the bridged call appearance of a station to conference a call, Communication Manager marks the station with the bridged call appearance as the original ad-hoc conference host. For example, Station A has a bridged call appearance on Station D. When a user makes a call from Station B to Station A, the user at Station D answers the call. The user at Station D conferences Station C into the call. Communication Manager marks Station D as the original ad-hoc conference host. When the user at Station D disconnects the call, the call drops.

Multiple Device Access

When a user makes a conference call from an MDA device, Communication Manager marks the extension of the device as the original ad-hoc conference host. The call gets

disconnected only when the last connected MDA device that has the same extension as the original ad-hoc conference host drops from the call.

For example, Station A-MDA1 and Station A-MDA2 are registered MDA devices for Extension A. A user makes a call from Station B to Extension A. A user at Station A-MDA1 answers the call and conferences Station C into the call. Communication Manager marks Extension A as the original ad-hoc conference host. The user now uses Station A-MDA2 to bridge onto the call and disconnects the call at Station A-MDA1. The conference call drops only when the user disconnects the call at Station A-MDA2.

Call Park

When the original ad-hoc conference host parks the conference call, the conference call does not drop. Communication Manager does not assign any extension to the original ad-hoc conference host. When the call is unparked, Communication Manager marks the extension that unparks the call as the original ad-hoc conference host. The call drops when the extension that unparks the call disconnects the call.

Note: The extension that unparks the call might not be the same as the extension that had parked the call.

APPENDIX A - Description of fields on the Calling Party Number Conversion screen

LEN

Valid Entry	Usage
From 1 to 24 digits	During conversion, Communication Manager checks the length of the calling-party number. The length of the calling-party number can be of up to 24 digits.
any	During conversion, Communication Manager does not check the length of the calling-party number. Any length of the calling-party number can be converted.

Any CPN

Valid Entry	Usage
x	When looking for a matching entry, Communication Manager does not consider the digits in the calling-party number. Communication Manager disables the **CPN prefix** field.
blank	When looking for a matching entry, Communication Manager compares the digits in calling-party number with the digits entered in the **CPN prefix** field. By default, this field is blank.

CPN Prefix
Use this field to enter a number that is a prefix of the calling-party number. The value in this field can be of up to 24 digits.
Communication Manager disables the **CPN prefix** field if the **Any CPN** field is set to x.

Incoming Trunk Group(s)

Valid Entry	Usage
Trunk group number	Enter a single trunk group or a range of trunk groups.
blank	Any trunk group can match this row.

Delete

Valid Entry	Usage
From 1 to 24 digits	The number of leading digits deleted from the calling-party number. The value in this field can be of up to 24 digits.
all	All digits are deleted.
blank	No digits are deleted from the calling-party number. By default, this field is blank.

Insert

Valid Entry	Usage
From 1 to 15 digits	Digits inserted at the beginning of the calling-party number. The value in this field can be of up to 15 digits.
blank	No digits are inserted at the beginning of the calling-party number. By default, this field is blank.

Final Numb Fmt

Valid Entry	Usage
Intl lev0 lev1 lev2 locl natl pubu unku.	The numbering formats for modifying the calling-party number.
blank	The original numbering format associated with the incoming calling-party number is unchanged. By default, this field is blank.