Avaya Aura® with 9600-Series IP Deskphones and J100-Series IP Phones

July 2019 - Avaya Aura® Platform Release 8.1.0

Introduction

The Avaya Aura® solution is a rich, highly interoperable set of SIP components that takes enterprise communications architecture to the next level. At the core of the solution is the Avaya Aura® Session Manager providing SIP interoperability, dial plan generation, SIP normalization, SIP routing, and many other SIP services to create a secure, centralized and easy-to-manage enterprise backbone network.

A chief advantage of the Avaya Aura® solution is the ability to deliver the right features to the right users across an enterprise regardless of protocol or endpoint device. System architects and planners have the advantage with the Avaya Aura® solution to choose the protocol and endpoint that satisfies the user's needs and deliver these services using the delivery mechanism and network of choice. In addition, the Avaya Aura solution allows enterprises to mix and match not only TDM and IP, but SIP and H.323 endpoints anywhere in the customer network.

Avaya Aura Protocol Support

The Avaya Aura® solution supports the largest variety of technologies in the industry including analog, Digital (the Avaya "DCP" protocol), H.323 Internet Protocol (IP), and Session Initiation Protocol (SIP) handsets and auxiliary equipment. Each technology is unique and offers differing levels of functionality based on the technology supported with analog devices providing the lowest capabilities, and H.323 and SIP the richest. These notes document the available services for the H.323 and SIP protocols and are valid for Avaya Aura® Platform 8.1.0.

The information is this document is accurate as of the issue date and subject to change.

Avaya Aura® with 9600-Series IP Deskphones – Available Services

The following table shows each relevant Avaya Aura® service in alphabetical order with the support for each configuration for comparison. The associated deskphones are the Avaya 9600-Series IP Deskphones.

- The "H.323" columns are based upon Avaya Aura® Communications Manager 8.1.0.
  - The two columns distinguish between Avaya one-X® Deskphone H.323 3.2.8 software which is used on the 9620L/9620C/9630G/9640/9640G/9650/9650C IP Deskphones,and the Avaya Deskphone H.323 6.8.2 software which is used on the 9608/9608G/9611G/9621G/9641G/9641GS IP Deskphones and J169/J179 IP Phones.

- The "SIP" columns are based upon Avaya Aura® Platform 8.0.1 (Avaya Aura® Communications Manager 8.1.0 with Avaya Aura® Session Manager 8.1.0).
  - The three columns distinguish between Avaya one-X® Deskphone SIP 2.6.17 software which is used on the 9620L/9620C/9630G/9640/9640G/9650/9650C IP Deskphones, the Avaya Deskphone SIP 7.1.6 software which is used on the 9601/9608/9608G/9611G/9621G/9641G/9641GS IP Deskphones, and the J100 SIP 4.0.2 software which is used on the J129/J139/J169/J179 IP Phones.

Table entries which have been updated with this version of the document are identified in *italic* text.

The term "AST" is used to identify a set of SIP features that use standards IETF signaling to accomplish, but are not contained in the "SIPPING-19" specifications. These items are "Advanced SIP Telephony" and are identified in the following table by orange coloring.

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Abandoned Call Logging	Supported	Supported	Supported	Supported	Supported
Abbreviated Dialing	Supported	Supported	Partially Supported (no button)	Partially Supported (no button)	Partially Supported (no button)
Abort Transfer	Supported	Supported	Supported	Supported	Supported
Account Codes	Supported	Supported	Supported	Supported	Supported
Advice of Charge	Supported	Supported	Not Supported	Not Supported	Not Supported
Announcements	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Attendant Console	Supported	Supported	Not Supported	Not Supported	Not Supported
Authorization Codes	Supported	Supported	Supported	Supported	Supported
Automatic Answer Intercom	Supported	Supported	Supported	Supported	Supported
Automatic Callback	Supported	Supported	Supported	Supported	Supported
Automatic Dial Buttons	Supported	Supported	Supported	Supported	Supported
Automatic Exclusion	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Automatic Hold	Supported	Supported	Supported	Supported	Supported
Automatic Route Selection (ARS)	Supported	Supported	Supported	Supported	Supported
Aux-work for a Hunt Group	Supported	Supported	Not Supported	Supported via “Hunt Group Busy Button”	Supported on J169/J179. Not supported on J129/J139.
Bridged Line (Call) Appearances	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Bridged Line Appearances - Analog	Supported	Supported	Not Supported	Not Supported	Not Supported
Busy Line Indicator	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Button Module (12,24)	Supported (Model specific)	Supported (Model specific)	Supported (Model specific)	Supported (Model specific)	Supported (Model specific)
Call Coverage (6 levels)	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Caller ID (Name and number)	Supported	Supported	Supported	Supported	Supported
Call Forward (All, busy, don’t answer, disable)	Supported	Supported	Supported	Supported	Supported
Call Hold, Resume	Supported	Supported	Supported	Supported	Supported
Call Log (Missed/Answered/Outgoing calls, Call/Delete/Details)	Supported (HTTP backup)	Supported (HTTP backup)	Supported (local)	Supported (local)	Supported (local)
Call Log (Busy)	Supported	Supported	Not Supported	Supported	Supported
Call Log (Offline)	Supported with CES	Supported	Not Supported	Supported	Supported
Call Park, Answer Back	Supported	Supported	Supported	Supported	Supported
Calling Party Number Block, Unblock	Supported	Supported	Supported	Supported	Supported
Calling Party Number Block, Unblock of Internal Numbers	Supported	Supported	Partially Supported	Partially Supported	Partially Supported
Call Pickup	Supported	Supported	Supported	Supported	Supported on J139/J169/J179. Not supported on J129.
Class of Service	Supported	Supported	Supported	Supported	Supported
Click to Conference	Supported	Supported	Not Supported	Not Supported	Not Supported
Code Calling	Supported	Supported	Supported	Supported	Supported
Conference (Ad-Hoc – 6 Party)	Supported	Supported	Supported	Supported	Supported
Conference (No Hold)	Supported	Supported	Not Supported	Not Supported	Supported on J169/J179. Not supported on J129/J139.

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Contact Center (General)	Supported	Supported	Not Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Contacts (Add/Edit/Delete/Details)	Supported	Supported	Supported	Supported	Supported
Core Redundancy	Supported	Supported	Supported	Supported	Supported
Crisis Alert (Dial/View)	Supported	Supported	Partially Supported (Dial only)	Partially Supported (Dial only)	Supported on J169/J179. Not supported on J129/J139.
Directed Call Pickup	Supported	Supported	Supported	Supported	Supported on J139/J169/J179. Not supported on J129.
Directory (Aura Integrated)	Supported	Supported	Not Supported	Supported	Supported
Directory (LDAP)	Supported	Supported	Supported	Supported	Supported
Distinctive Ringing	Supported	Supported	Supported	Supported	Supported
Emergency Button (one touch access)	Supported	Supported	Supported	Supported	Supported
Enhanced Call Forwarding	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Enhanced Group Call Pickup Alerting	Supported	Supported	Supported	Supported	Supported
Exclusion	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Extended Call Pickup	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Extension to Cellular (EC500)	Supported	Supported	Supported	Supported	Supported
Favorite Button	Not Supported	Not Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Feature Named Extensions	Supported	Supported	Supported	Supported	Supported
Flexible Language Displays	Supported	Supported	Supported	Supported	Supported
Forced Entry of Account Codes	Supported	Supported	Supported	Supported	Supported
Guest Login	Supported	Supported	Not Supported	Not Supported	Supported on J139/J169/J179. Not supported on J129.
Group Paging	Supported	Supported	Partially Supported (initiate only)	Supported	Supported
Hold Recall	Supported	Supported	Supported	Supported	Supported
Hospitality (general)	Supported	Supported	Partially supported	Partially supported	Partially supported
Hotline	Supported	Supported	Not Supported	Supported (settings file)	Supported (settings file)
Hunt Groups	Supported	Supported	Supported	Supported	Supported
Hunt Group Busy Button	Not Supported	Not Supported	Not Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Idle Line Appearance Select	Supported	Supported	Supported	Supported	Supported
Intercom - Automatic	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Intercom - Dial	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Last Number Dialed	Supported	Supported	Supported	Supported	Supported
Limit Number of Concurrent Calls	Supported	Supported	Not Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Local Survivability with Survivable Remote	Supported	Supported	Supported	Supported	Supported
Local Survivability with Third Party Gateways	Not Supported	Not Supported	Supported	Supported	Supported
Local Survivability with IP Office Centralized	Not Supported	Not Supported	Supported	Supported	Supported
Loudspeaker Paging	Supported	Supported	Supported	Supported	Supported
Loudspeaker Paging - Deluxe	Supported	Supported	Supported	Supported	Supported
Malicious Call Trace (MCT) - Activation	Supported	Supported	Supported	Supported	Supported
Malicious Call Trace (MCT) - Controller	Supported	Supported	Not Supported	Not Supported	Not Supported
Manual Signaling	Supported	Supported	Not Supported	Not Supported	Not Supported
Meet-Me Conferencing - CM	Supported	Supported	Supported	Supported	Supported
Message Retrieval (one button)	Supported	Supported	Supported	Supported	Supported
Message Waiting Indication (own number)	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Message Waiting Indication (third party number)	Supported	Supported	Supported	Supported	Supported
MLPP (Multiple Level Precedence and Pre-emption) TDM Trunking	Supported	Supported	Supported	Supported	Supported
Multiple Call Handling, Multiple Lines, Multiple Call Appearances	Supported	Supported	Supported	Supported	Supported
Multiple Device Access	Not Supported	Not Supported	Partially Supported	Supported	Supported
Multi-Language (Input and Display)	Supported	Supported	Supported	Supported	Supported
Multi-Language Input (Korean, Hebrew)	Not Supported	Not Supported	Supported	Not Supported	Not Supported
Multi-Language Input (Arabic, Japanese, Chinese)	Not Supported	Not Supported	Not Supported	Not Supported	Not Supported
Night Service	Supported	Supported	Not Supported	Not Supported	Not Supported
One Touch Recording for Modular Messaging	Supported	Supported	Supported	Supported	Supported
one-X Communicator / Equinox – Deskphone Mode	Supported	Supported	Not Supported	Supported	Supported
one-X Mobile Integration	Supported	Supported	Supported	Supported	Supported
one-X Portal Integration	Supported	Supported	Partially Supported	Partially Supported	Partially Supported
Pin Checking	Supported	Supported	Not Supported	Not Supported	Not Supported
Presence - Advertise	Supported	Supported	Supported	Supported	Supported

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Presence - Display	Not Supported	Not Supported	Supported	Supported	Supported on J169/J179. Not supported on J129.
Priority Calling	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Pull Transfer	Supported	Supported	Not Supported	Not Supported	Not Supported
Remote Worker - SBC	Not Supported	Not Supported	Supported	Supported	Supported
Remote Worker – VPN Integrated	Supported	Supported	Not Supported	Not Supported	Not Supported
Ring Tones – “Classic” or “European”	Supported	Supported	Partially Supported (Classic only)	Supported	Supported
Ring Tones - Rich	Supported (9670G only)	Supported	Not Supported	Supported	Supported
Ring Tones - Downloaded	Not Supported	Not Supported	Not Supported	Supported	Supported
Ringing Control - Bridged Line	Supported	Supported	Supported	Supported	Supported
Ringing Control – Per Contact	Not Supported	Not Supported	Not Supported	Supported	Supported
Ringing - Abbreviated and Delayed	Supported	Supported	Not Supported	Not Supported	Not Supported
Send All Calls	Supported	Supported	Supported	Supported	Supported
Service Observing	Supported	Supported	Partially Supported (observed only)	Supported	Supported on J169/J179. Not supported on J129/J139.

Service/Feature	H.323 (H.323 3.2.8)	H.323 (H.323 6.8.2)	SIP (SIP 2.6.17)	SIP (SIP 7.1.6)	SIP (SIP 4.0.2)
Simulated Bridged Appearance	Supported	Supported	Supported	Supported	Supported
Station On-Hook Dialing	Supported	Supported	Supported	Supported	Supported
Team Button	Supported	Supported	Not Supported	Supported	Supported on J169/J179. Not supported on J129/J139
Temporary Bridged Appearance	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Time of Day Routing	Supported	Supported	Supported	Supported	Supported
Traffic Measurements	Supported	Supported	Supported	Supported	Supported
Transfer (Attended, Unattended)	Supported	Supported	Supported	Supported	Supported
Transfer on Hang-up	Supported	Supported	Not Supported	Supported	Supported
Transfer to Voicemail (Answered)	Supported	Supported	Supported	Supported	Supported
Transfer to Voicemail (Alerting)	Supported	Supported	Not Supported	Not Supported	Not Supported
VIP Calling	Supported	Supported	Supported	Supported	Supported
Voice Initiated Dialing	Supported	Not Supported	Not Supported	Not Supported	Not Supported
Web Browser	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.
Whisper Page Activate	Supported	Supported	Supported	Supported	Supported on J169/J179. Not supported on J129/J139.

Service Descriptions

The more detailed descriptions of the services listed in the previous table are described here in alphabetical order. Differences between H.323 and SIP operation are highlighted.

Abandoned Call Logging

Calls which arrive, are not answered, and go to voicemail are added to the deskphone call log to notify the user a call was received, but not answered.

Abbreviated Dialing

Abbreviated Dialing (AD) is used to reduce the number of digits needed to dial to place a call. Instead of dialing the entire number, a short code is dialed to access the number. The system then dials the stored number automatically. You can also assign abbreviated dialing buttons to H.323 deskphones, so that you press a single button to dial frequently called numbers.

You cannot assign abbreviated dialing buttons to SIP deskphones. SIP deskphone users can program the FAC and dial code number against a Contact or against a Speed Dial entry.

With the J169/J179 and J100 SIP 2.0.0 software, users can assign contacts to up to 10 “Speed Dial entries”. An elongated press of the digit key dials the associated “Speed Dial” number.

Abort Transfer

Abort Transfer is used to stop the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when the user hangs up.

Account Codes

Account Code Dialing associates a call with an account number. A user enters a FAC for Account Code Dialing before the user dials a deskphone number. You can specify the use of the FAC is mandatory or optional for the user. When a user dials a deskphone number and the FAC, the system records the:

- Deskphone number
- Account code
- Trunk Access Code (TAC), or the Automatic Route Selection (ARS) access code

The system does not record the FAC for Account Code Dialing.

Advice of Charge

Users can view call charges on deskphone displays. From a display, a user can see the cost of an outgoing call, both while the call is in progress and at the end of the call. If users are to control when the display of the call charge information is displayed, a display button can be administered. The system can also be administered so that the system displays call charges automatically whenever a user places an outgoing call.

This feature is not supported on SIP deskphones.

Announcements

"Announcements" is a feature to play recordings for callers in the enterprise. For example, you can inform callers that:

- The call cannot be completed as dialed
- The call is in a queue
- All lines are busy

Announcements are often used in conjunction with music. The source for announcements can be either integrated or external.

Attendant Console

An attendant console is a terminal that has access to all the attendant features – Auto-Manual Splitting, Attendant Conferencing, Attendant Trunk Group Access, Attendant Intrusion, Attendant Overflow, etc. SIP deskphones can be "seen" (as extensions, busy line indicators, bridged appearances, etc.) and connected with an H.323 or DCP attendant console. **SIP deskphones cannot function as full-featured attendant consoles**.

Authorization Codes

Authorization Codes allows a deskphone user to input a personal identification code as a means for extending the control of system users' privileges and security for remote access callers. Authorization codes may be used for any or all of the following reasons:

- allow a calling user to override the Facility Restriction Level (FRL) assigned to the originating station or trunk
- restrict individual incoming tie trunks and remote access trunks from accessing an outgoing trunk
- identify certain calls on call detail records for cost-allocation purposes
- and provide additional security control for the system.

Automatic Answer Intercom

Intercom calls to SIP deskphones when set for automatic answer will immediately go off hook on speaker when an intercom call is received. Also, SIP deskphones may make intercom calls to other deskphones set for auto answer for intercom calls. Incoming automatic answer intercom calls do not alert audibly or visually, the receiving deskphones automatically answer the call, place the talkpath automatically on speakerphone, and display the caller's name on the active line appearance.

Automatic Callback

Automatic Callback (ACB) allows internal users who place calls to busy or unanswered internal deskphones to be called back when the called deskphone becomes available. When a user activates Automatic Callback, the system monitors the called deskphone and when the called deskphone becomes available to receive a call, the system automatically originates the Automatic Callback call. The originating party receives priority ringing. The calling party then lifts the handset, and the called party receives the same ringing that the system provided on the original call.

SIP deskphones can make and receive ACB calls, but only when both deskphones are controlled by the same Communication Manager.

Automatic Dial Buttons

Automatic dial buttons can be administered by the administrator or the user and appear on the button list of the deskphone. SIP users can have the auto dial buttons administered with names and numbers by the administrator or the individual users just like H.323.

Automatic Exclusion

Automatic exclusion allows a user of a deskphone to keep others with bridged line appearances of the same extension from bridging onto an existing call. Automatic exclusion is administered on a per station basis. Users may turn off automatic exclusion by pressing the exclusion button.

Automatic Exclusion is not supported on the J129/J139 IP Phones.

Automatic Hold

With automatic hold, a user can also press a second call appearance to put an active call on hold. With deskphones, the user can choose with each call if hold or automatic hold is to be selected by selecting the “hold and answer” or “drop and answer” options presented on the screen.

Automatic Route Selection (ARS)

Automatic Route Selection routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed. ARS provides a choice of routes for any given public network call.

Aux-work for a Hunt Group

If an agent is an ACD split and a hunt group member, the agent in the split usually has an AUX work button that also activates or deactivates the Hunt Group Busy option. If an agent is the last available member and pushes the AUX work button, the lamp on the button flashes until the queue is empty. The flashing light means that the agent is still available. When the queue is empty, the lamp lights but does not flash, and the system activates Hunt Group Busy. **This capability is not supported on SIP deskphones (refer to “Hunt Group Busy Button” for an alternative).** *As a workaround, the Hunt Group Busy Activation/Deactivation FACs can be dialed from the SIP deskhone.*

Bridged Line (Call) Appearances

The bridged line appearance feature is used to give multi-appearance deskphones an appearance of another deskphone number. With a bridged line appearance, the user can originate, answer, and bridge onto calls to or from the deskphone number of another user. ***The J129/J139 IP Phones do not support Bridged Line (Call) Appearances.***

Bridged Line Appearances - Analog

Analog Bridged Line Appearances allow an analog phone to be bridged with a digital/IP phone. **This feature is not supported on SIP deskphones.**

Busy Line Indicator

The busy line indicator provides deskphone users with a visual indicator of the busy or the idle status of one of the following system resources:

- An extension number
- A trunk group
- A terminating extension group (TEG)
- A hunt group, either direct department calling (DDC) or uniform call distribution (UCD)
- Any loudspeaker paging zone, including all zones

J100 SIP 4.0.2.0 adds support for Busy Indicator on Avaya Aura® with the J169 and J179 IP Phones. This functionality will work on Avaya Aura® 7.x when the J169/J179 are aliased as 9611GSIP.hold This functionality will NOT work on Avaya Aura® 8.x when the J169/J179 are natively supported. It is expected that this will be resolved with Avaya Aura® 8.1.1.

Button Module (12, 24)

The 12 button module can be used with the 9608/9608G/9611G/9641G/9641GS IP Deskphone and J169/J179 IP Phones. The 24 button module can be used on all models except the 9601/9620L/9620C/9621G/9670G/J129/J139/J169/J179. They connect to either SIP or H.323 phones and provides up to 24 simultaneously visible and accessible buttons and lamps to support bridged call appearances, busy line indicators and one-touch access to CM features and visible feature status indications.

Call Coverage (6 levels)

Call Coverage provides automatic redirection of calls that meet specified criteria to alternate answering positions in a call coverage path. Lead coverage paths can be administered to apply to all calls at the same time, internal or external calls, or to apply to a specific day of the week or a specific time of day. Different coverage paths are administered based on incoming call origination, type or time. Building on the concept of call forwarding, personal call coverage programming redirects to a defined path of answering deskphones and will default to the called party's voice mailbox only as a last resort. Calls will not be redirected to the forwarding position or voice mailbox of a station user defined in the call coverage path; the originally called party's coverage path overrides intermediary station user call forwarding commands. The objective of personalized call coverage features is to reduce dependency on voice mail systems because a human answering station rather than a non-interactive machine might be preferred by the caller.   Call coverage is supported for 6 levels in CM, and is available for both the deskphones covered for and as coverage points.

Caller ID (Name and number)

Caller ID is used to interpret calling party information that is signaled over ISDN, H.323 or SIP trunks, and displaying the calling party number and calling party name on the deskphone. Caller ID is also known as Incoming Call Line Identification (CLID). Name and number is also displayed for internal callers as well.

Call Forward (All, busy, don't answer, disable)

Call forwarding redirects any incoming calls to another destination. Deskphones can forward calls for the following: all calls, calls when station is in use (busy), and calls that go unanswered. In addition, users can enable or disable call forwarding from their deskphones.

The system forwards a call only once. For example, assume that extension A designates extension B as its forwarded-to destination, and that extension B designates extension C as it's forwarded-to destination. When someone calls extension A, the system first attempts to ring the call at extension A. If the system is unable to ring the call at extension A, the system attempts to ring the call at extension B. If the system is unable to ring the call at extension B, the system redirects the call to the coverage path of extension A, if a coverage path is available at extension A, and if the coverage criteria of extension A are satisfied when applied at extension B. The system does not forward the call to extension C under any circumstances.

Call Hold, Resume

Deskphone users use the "Hold" soft button to hold a call and the "Resume" soft button to un-hold the call. The system holds the call at the call appearance that is used for the call. Deskphone users can hold a call on each call appearance.

Call Log (Missed/Answered/Outgoing calls, Call/Delete/Details)

Call Log stores dialed station numbers and incoming identification numbers [internal CLID (Calling Line Identification), ANI (Automatic Number Identification)]. The numbers that are stored are those of the most recently dialed and incoming calls. Pressing a call log button brings up the display. Calls to numbers appearing in the call log display field can be dialed automatically through menu control keys and entries may also be deleted via the menu control keys.

Users can also add a call log entry to contacts/phonebook. The call log stores up to 100 calls. For H.323 deskphones, the call log can be backed up to an HTTP server and loaded whenever a user logs in, providing access to the same call history information from any H.323 deskphone in the enterprise. For SIP deskphones, the call log is maintained locally.

Call Log (Busy)

Call logs for H.323 deskphones and SIP 6.5.x /J100 SIP deskphones will show missed calls for calls which are not presented to the deskphone due to all appearances being busy. **Call logs for SIP 2.6.x deskphones do not contain such entries.**

Call Log (Offline)

Call logs for H.323 6.6.x and SIP 6.5.x / J100 SIP deskphones which were off-line when a call was received will appear in the log when the deskphone logs back in. Call logs for H.323 3.2.x deskphones which were off-line when a call was received will appear in the log when the H.323 deskphone logs backs in if CES is used. **Call logs for SIP 2.6.x deskphones do not contain such entries.**

Call Park, Answer Back

Call park and answer back are used to park a held call and retrieve it later from any other deskphone within the system. For example, a user can answer a call at one extension, put the call on hold, and then retrieve the call at another extension. Or the user can answer a call at any deskphone after an attendant or another user pages the user. Deskphones can park a call either using the FAC or the “call-park” feature button. Calls are un-parked using the answer back FAC, or the “call-park” button.

Calling Party Number Block, Unblock

Calls from to outside trunks can block the calling party number for outside calls using either a FAC or feature buttons on the set. Deskphone users can invoke either a block or unblock operation before each call made.

Calling Party Number Block, Unblock of Internal Numbers

A typical Communication Manager internal call generally provides calling/called party numbers and an administered text name string which are displayed on the involved parties’ terminals. The internal calling party block feature provides stations the capability (via proper COR administration) of masking off the calling party name and number and replacing it with a “hard-coded,” system-wide text string, "Info Restricted," which will be displayed on the called party‘s terminal. In addition, the feature also provides a way of administering so the CPN/Name information not be masked-off when the call is made. Deskphone users can either block or unblock the CPN and name information with either the FAC codes or feature buttons on the set.

Call Pickup

Call Pickup allows users to answer calls for one another. The Call Pickup feature requires that users be members of the same pickup group. With Call Pickup, the administrator creates one or more pickup groups where a pickup group is a collection, or list, of individual deskphone extensions.
A user extension can belong to only one pickup group.

To pick up a call, the user enters the feature access code (FAC) of the call pickup feature and then the call pickup group number of the ringing call. Alternatively, the user can press the “call-pkup” call pickup button of the group that has been assigned to his/her deskphone to pickup the call.

The J129 IP Phone do not support Call Pickup.

Class of Service

COS is used to allow or deny user access to some system features for deskphones, such as:

- Automatic Callback
- Call Forwarding
- Data Privacy
- Trunk-to-Trunk Transfer Override
- QSIG Call Offer Originations
- Contact Closure Activation
- Console Permission

Click to Conference

Click to conference is only supported for H.323 deskphones. Conferencing is accomplished by highlighting the appropriate contact in the contact list.

Code Calling

One type of loudspeaker paging is “code calling” or chime paging. If frequent voice pages are undesirable, you can assign a unique series of chimes, or a chime code to each extension. The chime code assigned to that extension plays over the speakers when that extension is paged.

Conference (Ad-Hoc – 6 Party)

Ad-hoc conferences of up to 6 parties can be created.

Conference (No Hold)

This feature allows a user to automatically add another party to aconference call while continuing the conversation of the existing call.

Avaya Aura Platform 8.1.0 adds the ability for SIP devices/clients to be able to utilize No Hold Conference. This capability is supported with J100 4.0.2.0 and later on the J169/J179 IP Phones.

Contact Center (General)

Communication Manager has many features that are associated with contact centers or ACD (automatic call distribution). Examples of these are: agent login logout, after call work, aux work, etc. **These features are not supported with SIP 2.6.x software or on the J129/J139 IP Phones.**

Contacts (Add/Edit/Delete/Details)

Deskphones provide access to a maximum of 250 contacts (phonebook entries) with 3 (H.323) or 6 (SIP) numbers each. Changes to contacts are automatically backed up to a server (PPM for SIP, HTTP for H.323 if configured) and loaded whenever a user logs in.

Core Redundancy

Deskphones can connect to multiple IP addresses in the core allowing a core element or network connectivity failure without sacrificing service to the endpoint. H.323 deskphones do this by allowing connections to multiple CLANs or through connecting directly with the Processor Ethernet (PE) server interface. SIP deskphones allow for connection to two core Session Managers, and one Survivable Remote.

Crisis Alert

The Crisis Alert capability notifies the attendant, and up to 10 other designated users when someone dials an emergency number (like 911 in the US). When a user dials an emergency number, the system sends both an audible alert and a visual alert to the attendant console and phones designated for crisis alert displays. Both H.323 deskphones and SIP deskphones can make emergency calls that are viewable using crisis alert. H.323 deskphones can be used to view crisis alert information.

Avaya Aura Platform 8.1.0 adds the ability for SIP devices/clients to be able to receive and view crisis alert information. This capability is supported with J100 4.0.2.0 and later on the J169/J179 IP Phones.

Directed Call Pickup

With Directed Call Pickup, users can specify what other deskphone they want to answer. Call pickup groups are not needed with Directed Call Pickup. With directed call pickup, a user can answer an incoming call on another deskphone by entering the Directed Call Pickup feature access code (FAC) followed by the extension of the ringing call. Alternatively, the user can use the “dir-pkup” directed call pickup button on the deskphone followed by the extension to pickup a call ringing on another extension.

Directory (Aura Integrated)

Avaya Aura® has an integrated directory which contains the names and extensions of all users which are provisioned on the system.

H.323 deskphones access this directory via the “Directory” softkey. Deskphones with SIP 6.5.x software access this directory via the “Contacts” button followed by “Search” softkey. **This capability**

is not supported on Deskphones with SIP 2.6.x software. Once the correct name is found, a call can be made to the associated number or the name/extension can be added to the local contacts.

Directory (LDAP)

9600-Series IP Deskphones and J100-series IP Phones cannot directly access a corporate LDAP directory but can utilize two methods to access that information..

- Avaya distributes a script that can be installed on a standard web server to provide access from the WML browser on the deskphones to an enterprise LDAP directory. This feature is also available from the Utility Server Template as a "Directory Application".

- System Manager can synchronize with an enterprise LDAP directory (refer to this white paper). J100-series IP Phones with SIP software can subsequently search the System Manager Directory.

Distinctive Ringing

Distinctive Ringing provides different ringing patterns for different types of calls. Deskphones configured to use either "Classic" or "European" ringtones will ring with different patterns. Deskphones configured to use either "Rich" or downloaded ringtones will ring with the same pattern irrespective of the type of call.

Emergency Button (one touch access)

The Emergency button on deskphones provides one-touch access to make an emergency call even when the phone is logged out, registered but inactive, or locked.

Enhanced Call Forwarding

Enhanced Call Forwarding allows a deskphone to forward calls to different locations based upon the status of the phone at the time that the call is received as well as the source (internal/external) of the call. There are six types of Enhanced Call Forwarding:

- Unconditional – internal calls
- Unconditional – external calls
- Busy – internal calls
- Busy – external calls
- No Reply- internal calls
- No Reply – external calls

Deskphone users can activate or deactivate any of these types from their phone, and can specify different destinations for each type of call. Deskphone users with SIP 6.5.x software can program all six from a single screen.

Enhanced Call Forward is not supported on the J129/J139 IP Phones.

Enhanced Group Call Pickup Alerting

The enhanced call pickup button on a deskphone can be administered with different alerting options for the group members including, silent, single alert, continuous, delayed, and abbreviated.

This feature only works for the call pickup group button with enhanced call pickup alerting turned on (in the system features form).

Exclusion

Exclusion or "manual" exclusion allows a user of deskphones to keep others with bridged line appearances of the same extension from bridging onto an existing call. The user presses the exclusion button either before the user places the call or when the user is active on the call to keep others from bridging onto the call. If the user presses the exclusion button while others are bridged onto the call, CM will drop the other users from the call. To turn off manual exclusion, the user presses the exclusion button.

Exclusion is not supported on the J129/J139 IP Phones.

Extended Call Pickup

With Extended Call Pickup, users in one pickup group can answer calls that come in for users in another pickup group. Extended Call Pickup allows the administrator to define one or more extended pickup groups and calls are "picked-up" by entering the Extended Call Pickup feature access code (FAC) and the 1-2 digit number to indicate the group of the ringing call to be picked up. There is no feature button that can be administered as an "extended call pickup" button, but an abbreviated dial button can be administered with the Extended Call Pickup FAC code for faster operation.

Extension to Cellular (EC500)

Extension to Cellular (EC500) allows a deskphone and associated cellular phone to ring simultaneously when there is an incoming call. It also allows users to "extend" an active call on their deskphone to their cellular phone.

Favorite Button

On SIP deskphones, buttons that appear on the "features" screen or other screens than the main screen can be programmed at the phone to appear on the main screen. This is quite useful for features that visibly alert or are used often like call pickup, EC500, extend call, etc. **H.323 deskphones do not have this capability**.

Feature Name Extensions (FNE)

Once Extension to Cellular is enabled, a cell phone user can activate certain Communication Manager features by dialing a feature name extension (FNE). FNEs correspond to a direct inward dialing (DID) number for each feature. The administrator creates the FNEs. The FNEs can be administered on a per-CM basis.

Users with SIP deskphones can use Extension to Cellular FNEs to activate features on their SIP deskphones like *active appearance select*, *activate/deactivate SAC*, etc.

Flexible Language Displays

Flexible language displays allows the deskphone to display any one of approximately 15 languages. For H.323, this also includes that ability (with a downloadable tool) to customize the words and phrases displayed by the UI for additional languages.

Forced Entry of Account Codes

Forced Entry of Account Codes requires the entry of an account code for every call. The following can be administered:

- All users enter an account code for all calls
- All users enter an account code for calls that are made on a specific trunk.
- All users enter an account code for calls that are made to a specific deskphone number
- A specific user enters an account for all calls that are made by that user

With Forced Entry of Account Codes (FEAC), the system rejects any call a user makes without an account code FAC, if the call requires an account code. When the system rejects the call, the user hears intercept tone.

Guest Login

Guest Login allows users to temporarily log into an already-logged-in deskphone. When they subsequently log out, the deskphone reverts to the previously logged-in extension. **Guest Login is supported on H.323 deskphones and the J139/J169/J179 IP Phone.**

Group Paging

With the Group Paging feature, you can create a page group, and assign extensions as members of the group. You assign an identifying extension to each page group, which users dial to page the group. When a user dials the extension of the paging group, Avaya Communication Manager activates the speakers on all the deskphones in the group and each user can hear what the “pager” speaks. Group Paging is one-way communication: Group members hear the person place the page, but cannot respond directly.

Deskphones with SIP 2.6.x software can initiate a page but cannot receive a page.

Hold Recall

Calls that are held by a user will return to the deskphone that held the call after a fixed period of time (the long hold recall timer). When the call returns to the deskphone that held the call, the deskphone alerts and the display indicates the call has exceeded the long hold recall timer.

Hospitality

A set of features used in the hotel or hospitality industry. Examples include “Automatic Wakeup” and “Do not Disturb”, etc. These work for H.323 deskphones, **but Aura does not provide the feature access buttons to activate these on SIP deskphones so only FAC access may be possible.** In cases where the feature is activated from an H.323/DCP Deskphone, Digital console, or PMS link, then the feature (DND/AWU) will properly activate and work on a SIP Deskphone.

Hotline

Hotline is the ability to place calls to pre-arranged extensions without entering the extension of the destination with the keypad. H.323 deskphones support this feature. SIP deskphones with SIP 6.5.x / J100 SIP require the use of the 46xxsettings file to administer the deskphone for hotline. **SIP deskphones with SIP 2.6.x software do not support this feature.**

Hunt Group Busy Button

The “Hunt Group Busy Button” allows the user of the deskphone to opt in/out of a hunt group and includes a visual indication of their status in that hunt group. **This button is supported with SIP deskphones running SIP 7.0.1 / J100 2.0.0 or later software. The J129/J139 IP Phones do not support this button.**

Intercom - Automatic

Automatic intercom is a feature where a button is administered that calls a predefined extension when the button is pressed. An intercom call makes a unique alerting sound. If the deskphone has an intercom button with a status lamp, the lamp also flashes.

To control which users can make intercom calls to each other, the user deskphones are placed in groups called "intercom groups." Once you add a set of deskphones to the group, users can make intercom calls by administering an automatic intercom button on their deskphone.

Intercom – Dial

Dial intercom allows one user to call another user in the same intercom group by pressing the dial intercom and one or two other digits. An intercom call makes a unique alerting sound. If the deskphone has an intercom button with a status lamp, the lamp also flashes.

Dial Intercom is not supported on the J129/J139 IP Phones.

Last Number Dialed (LND)

The Last Number Dialed feature redials the same number a user just dialed. LND can be set to 'off', 'redial last number' or 'redial from a list'. If it is set to 'redial last number', pressing the redial softkey will cause the phone to redial the last number dialed; if it is set to 'redial from list', the phone will display a list of the 3 or 6 (a full screen's worth) of the most recently dialed numbers so the user can select one of them to call.

Limit Number of Concurrent Calls

Limit Number of Concurrent Calls (LNCC) is used to keep one Call Appearance idle such that it can be used for outgoing calls. When the LNCC feature is enabled and all but one Call Appearances are busy, subsequent incoming calls receive a busy signal (no coverage path) or follow the coverage path if administered. LNCC is activated and deactivated by a feature button (**limit-call**), or by using two new Feature Access Codes (Limit Number of Concurrent Calls Activation/Deactivation). The **limit-call** button indicates visually whether the LNCC feature is active or not.

LNCC is not supported on SIP 2.6.x deskphones. LNCC is not supported on the J129/J139 IP Phones.

Local Survivability (Survivable Remote)

Local survivability is accomplished with Survivable Remote for full featured survivability for H.323 and SIP deskphones.

Local Survivability with Third Party Gateways

Local survivability is accomplished with third party gateways for SIP deskphones. Avaya SIP deskphones will connect and register alternatively with select third party gateways and receive basic functionality when the branch is disconnected from the core Session Manager servers.

Local Survivability with IP Office Centralized

IP Office 9.0 introduced a Centralized configuration. SIP deskphones will register and receive “sunny day” service from a core Avaya Aura® system, and connect and receive basic “rainy day” functionality from the IP Office when connectivity is lost.

Loudspeaker Paging

Loudspeaker Paging provides deskphone users and/or attendants dial access to voice paging systems. This is useful for paging purposes regardless of the deskphone user’s location within the premises environment. It is often used with the call park feature. The system can provide as many as 9-individual paging zones. In addition, 1-zone can be provided by the system to activate all zones simultaneously. Deskphones users can activate paging by pressing Abbreviated Dialing buttons.

Loudspeaker Paging - Deluxe

Deluxe Paging is similar to Loudspeaker Paging, but Deluxe Paging adds the convenience of automatically parking a call. When one user is away from their desk and receives a call, another user can answer the call, and send out a page. To page and park an active call simultaneously, users press **Transfer**, dial the trunk access code + an extension number where the call will be parked, make the announcement, and press **Transfer** again. The called party dials the Answer-Back TAC to retrieve the call. Deluxe Paging also provides Meet-Me Paging and Meet-Me Conferencing; a user pages another party and adds the party onto a conference call when they call the paging party. Note that without Deluxe Paging, paging and parking are two separate operations.

Malicious Call Trace (MCT) - Activation

A user can use either a feature button (**mct-act**) or a Feature Access Code (FAC) to activate MCT. Either the recipient of the call, or another user or attendant, can activate MCT. When a user or an attendant activates MCT, the system notifies potential MCT controllers. A potential MCT controller is a user or an attendant who is assigned a feature button (**mct-contr**) that can control MCT. The controller can direct the call to be recorded, and Communication Manager will output a record of the malicious call.

Malicious Call Trace (MCT) - Controller

An MCT controller is a user or an attendant who is assigned a feature button (**mct-contr**) that can control MCT. The first user or attendant who presses an **mct-contr** button becomes the MCT controller for the call, and the system stops alerting other potential MCT controllers. The display of the MCT controller shows the information for the traced MCT call.

This feature is not supported with SIP deskphones.

Manual Signaling

The manual signaling feature allows one user to signal another user. When a user presses the manual signaling button, the other user hears a 2-second ring. The status lamp of the user who presses the button lights for two seconds. If the deskphone of the intended recipient of the signal is already alerting, the system:

- Does not generate the 2-second ring
- Causes the manual signaling button lamp of the user who presses the button to flicker briefly

This feature is not supported with SIP deskphones.

Meet-Me Conferencing – CM

The Meet-me Conference feature is used to set up a dial-in conference of up to six parties. Users can dial into the conference.

Message Retrieval (one button)

Pressing the "Message" button on a deskphone will connect the user to the enterprise voicemail system to retrieve their voice mail.

MLPP TDM Trunking

The Multiple Level Precedence and Preemption (MLPP) feature allows users to request priority processing of calls during critical situations and select or pre-empt TDM trunks.

Multiple Call Handling, Multiple Lines, Multiple Call Appearances

Deskphones can have multiple line appearances and handle multiple calls on these appearances simultaneously.

Multiple Device Access

Multiple Device Access (MDA) allows up to ten SIP stations to be registered to the same username and password. A deskphone with SIP 2.6.x software can be at least one of those stations. A deskphone with SIP 7.1.x or J100 SIP software can be at least one of those stations and additionally provides a similar user experience if only one device is registered or if multiple devices are registered. For more details on MDA and the limitations associated with that feature, please refer to the "Avaya Aura® Multi Device Access White Paper".

Multi-Language

The 9600-series IP Deskphones and J100-series IP Phones support an extended list of languages for display and text input. The languages supported for display are: Arabic, Simplified Chinese, Dutch, English, French (Canadian and Parisian), German, Hebrew, Italian, Japanese, Korean, Portuguese, Russian, Spanish (Castilian and Latin American). Arabic is not supported on the J129/9601/9608/9608G.

Night Service

Night Service is used to direct incoming calls to other answering points at night. Night service can be provided for trunk groups, attendants, hunt groups, ACD groups, etc.

One Touch Recording for Modular Messaging (OTR)

OTR is used to record deskphone conversations by pressing a single button on an a deskphone. OTR uses Modular Messaging to record a deskphone conversation. A user needs to press only one feature button on the deskphone to activate OTR. OTR then stores the recorded conversation as a message in the voice mailbox of the user. The system allows OTR only after a call is answered.

one-X Communicator / Equinox – Deskphone Mode

one-X Communicator / Equinox Deskphone Mode allows users of one-X Communicator / Equinox to make or receive phone calls on an associated IP Deskphone.

This feature is not supported on IP Deskphones with SIP 2.6.x software.

one-X Mobile Integration

one-X Mobile allows users of deskphones to add the capabilities of a mobile handset for a single, enterprise extension.

one-X Portal Integration

one-X Portal is a browser-based tool for making and receiving calls remotely, when access to the deskphone is either not possible or not required. one-X portal may be accessed by users of H.323 and SIP deskphones.

PIN Checking

PIN checking requires a deskphone user to enter a PIN before dialing. The redial log does not contain the PIN so that redials will not work to keep users from dialing without knowing the PIN. This feature works only for H.323 deskphones.

Presence - Advertise

SIP and H.323 deskphones provide their presence status to Avaya Aura® Presence Services.

Presence - Display

The display of the presence status of users in the local Contact list is supported on SIP deskphones only (except the J129 IP Phone). The presence status is shown in the Contact list, Call Log, Conference Roster, and Home screen (9621G/9641G/9641GS/J139/J169/J179 only).

Priority Calling

The Priority Calling feature is used by deskphones to provide a special type of call alerting between internal users, including the attendant. The called party hears a distinctive ringing when the calling party uses priority calling.

Priority Calling is not supported on the J129/J139 IP Phones.

Pull Transfer

Pull Transfer is used to allow either the transferring party or the transferred-to party to press the Transfer button to complete the transfer operation. To use Pull Transfer, calling parties and called parties must be on the same Communication Manager.

Pull Transfer is not supported with SIP deskphones.

Remote Worker - SBC

SIP deskphones can be used in remote offices connecting to the main Avaya Aura® system over the Internet by pairing it with an SBC Session Border Controller (either ACME or ASBCE). The SBC allows only authorized users to connect and all signaling and media can be encrypted while it is traversing the Internet.

Remote Worker – VPN Integrated

VPN Remote Access via IP-SEC is supported in H.323 deskphones providing customers the ability to use those phones in the home or small office locations and leverage high speed internet to connect to the main location.

Ring Tones – Classic, European, Rich, Downloaded

All deskphones users can choose between eight ringtones. Deskphones with H.323 6.6.x, SIP 6.5.x, or J100 SIP software can optionally use either eight “Classic” or eight “European” ringtones. 9670G users or Deskphones with SIP 6.5.x/J100 SIP software can additionally choose another six “rich” ring tones. SIP 6.5.x / J100 SIP software also supports the ability to download up to 40 additional ringtones from the HTTP/HTTPS provisioning server.

Ringing Control - Bridged Line

Administrators can individually set the ringer settings for each line bridged line appearance. Ringers can be set to audibly alert or not.

Ringing Control – Per Contact

Users of deskphones with SIP 6.5.x or J100 SIP software can program any of the available ringtones against each of the six numbers associated with their Contacts.

Ringing - Abbreviated and Delayed

The Ringing - Abbreviated and Delayed feature is supported on **H.323 deskphones only** and has two categories of ringing:

- Ringing that alerts consistently and does not change:
  - Ringing, in which the lamp flashes and audible ringing occurs
  - Silent ringing, in which the lamp flashes and audible ringing does not occur
- Ringing that transitions from one ringing state to another:
  - Abbreviated ringing, in which ringing continues for the number of cycles that you specify with the automatic abbreviated transition interval or the delayed transition interval, and then changes to silent alerting
  - Delayed ringing, in which visual alerting continues for the number of cycles that you specify with the automatic abbreviated transition interval or the delayed transition interval, and then changes to ringing

The Ringing - Abbreviated and Delayed feature is most useful in bridging situations in which some users want to:

- Have a call audibly alert as soon as the call arrives

- Be audibly notified if the call is unanswered within a specified number of rings
- Stop the audible alerting

Send All Calls

Send All Calls allows users to temporarily divert all incoming calls to coverage regardless of the assigned call coverage redirection criteria. The feature also allows covering users to temporarily remove their voice terminals from the coverage path. The feature button on the phone will light when the Send All Calls feature is active. A button may be created for the deskphone on which it is administered (own station) or for another station (other station) so that an assistant can activate and monitor Send All Calls for a boss for example.

Service Observing

With Service Observing, designated users who are usually supervisors can listen to other user calls. H.323 and SIP deskphones' calls can be observed. H.323 deskphones or deskphones with SIP 7.0.0 / J100 2.0.0 or later software can be used to observe other calls. Note that Call Center Elite 7.0.1 is also required to allow the use of SIP Service Observe. **Deskphones with SIP 2.6.x software cannot be used to observe other calls.**

Simulated Bridged Appearance

Calls that are redirected to call coverage maintain a simulated bridged appearance on the called deskphone if a call appearance is available to handle the call. The called party can bridge onto the call at any time. The system can be administered to allow a simulated bridged appearance of the call to either remain at, or be removed from, the covering deskphone after the principal bridges onto the call. If two parties are bridged together on an active call with a third party, all three parties hear the bridging tone.

Station On-Hook Dialing

On-hook dialing is the ability to begin a call by merely pressing the dialed digits without going off hook on the receiver or pressing the speaker phone button on the handset. The phone will automatically activate the speakerphone with on-hook dialing. On-hook dialing is useful for users that require a minimum number of button presses to initiate an outgoing call.

Team Button

The Team Button has two generic functions, a display function and an execution function. The display function will allow any member of a team (monitoring station) to observe the station state of other team members (monitored station). The indication is done either via the green and red button LED or as an icon on the display, depending of the type of the station. In addition the ringing of a call on a monitored station may be indicated with audible ringing on the monitoring station. Furthermore active call forwarding from the monitored station to the monitoring station and active send all calls path at the monitored station with the monitoring station as first destination in the coverage path will be displayed.

As an execution function, the Team Button can be used as Speed Dial Button or Pick-Up Button. Depending on the state of the monitored station, when the Team Button on the monitoring station is pushed, a call to the monitored station is established directly or a ringing call is picked from the monitored station.

Team Button is not supported on SIP deskphones with 2.6.x software. Team Button is not supported on the J129/J139 IP Phones.

Temporary Bridged Appearance (TBA)

When a call is made to an individual that is part of a pickup group, one particular member of the group can be the most qualified person to handle the given call. If this individual does not answer the call originally, this individual can bridge onto the call using a temporary bridged appearance. The answering party does not have to transfer the call. A call to an individual can be answered by a member of a call pickup group and while the call is still connected, the called party can bridge onto the call, and the answering party hangs up. TBAs can be administered system-wide to either on or off using the system-parameters features form on CM.

Temporary Bridged Appearance is not supported on the J129/J139 IP Phones.

Time of Day Routing

Time of Day Routing can be used to redirect calls to coverage paths according to the time of the day and the day of the week. Calls can be routed based on the least expensive route according to the time of day and the day of the week that the call is made. You can also deny outgoing long distance calls after business hours to help prevent toll fraud. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks that the system uses for call forwarding to external numbers.

Traffic Measurements

Both H.323 and SIP deskphones can be covered by the traffic measurements used in SM and CM to record traffic levels, usage on trunks, etc.

Transfer (Attended, Unattended)

The Transfer feature is used to allow deskphone users to transfer trunk calls or internal calls to other deskphones or trunks without attendant assistance. Attended transfers are accomplished by making connection with the final destination of the call before completing the transfer; unattended transfers are completed before the final destination station answers the call.

Transfer on Hang-up

Use “Transfer Upon Hang-up” to transfer a call without the need to press the Transfer button a second time. The user presses the Transfer button, dials the number to which the call is being transferred, and then hangs up the receiver. Transfer upon hang-up is an optional capability administered at the system level. Even when transfer upon hang-up is administered, users can still press the transfer button a second time to transfer the call.

This feature is not supported with SIP deskphones with 2.6.x software.

Transfer to Voicemail

Transfer to voicemail allows a user to transfer a live (or alerting) call to a designated voice mail box with the press of a deskphone button. This is useful for assistants who answer calls for others to easily enable callers to leave voice messages for the called parties when they are not available. H.323 deskphones allow the administration of the transfer to voicemail button, but SIP deskphones do not. Also, SIP deskphones do not allow transfer to voicemail while a call is alerting, only after the call is answered can a SIP deskphone use the transfer to voicemail button.

VIP Calling

Administration of the Class of Service on CM enables automatic priority calling when assigned to the originator of the call. Thus, a “VIP” can be set so that every call he/she makes to non VIP deskphones will automatically become a priority call.

Voice Initiated Dialing (VID)

A user can speak a name to search for and call any contact when voice dialing is enabled. A user can optionally add a qualifier like "at home" or "mobile" with the name to get to a specific number for the contact. The first two times a user uses voice dialing, a help screen displays to assist in using the feature. **VID is not supported on SIP deskphones. VID is not supported on H.323 deskphones with 6.3.0 or later software.**

Web Browser

9600-Series IP Deskphones and J169/J179 IP Phones include a WML browser which includes support for the Avaya Web API. The J129/J139 IP Phones do not support the WML browser..

Whisper Page Activation

By activating whisper page, only the person on the paged extension can hear the “page” (an assistant “whispering” a message to the paged party). Other parties on the call cannot hear the page, and the person who activates the page cannot hear anyone on the call. If the paged user has an H.323 or SIP deskphone, the paged user can see who makes the whisper page. Both H.323 and SIP deskphones can activate and receive whisper pages.

Whisper Page is not supported on the J129/J139 IP Phones.

Notice

While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document may be incorporated in future releases.