# Avaya IP Office™ Platform in a Branch Environment Reference Configuration

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# Chapter 1: Introduction

## Purpose

This document describes the IP Office Branch solution architecture, topology, deployment options, and interoperability. It also describes the functional limitations of the solution and customer requirements that are met by the solution. The document helps sales design specialists to determine the best design options for a customer.

This document briefly describes solution features. For more information about features and functionality, see Avaya IP Office™ Platform Solution Description.

For more information about deploying and administering the solution, see the following documents:

- Administering Centralized Users for an IP Office™ Platform Enterprise Branch
- Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager
- Deploying Avaya IP Office™ as a Distributed Enterprise Branch in a Communication Server 1000 Environment with Avaya Aura® Session Manager

## Document conventions

The following table shows the terminology used in the IP Office Branch documentation for Centralized users with SIP and analog extensions:

**Table 1: Documentation terminology**

| Terminology used | Definition |
|---|---|
| Centralized SIP user | Centralized user in the IP Office Branch with a SIP extension. |
| ATA user | Centralized user in the IP Office Branch with an analog extension or an analog fax device. |

# Change history

The following table describes major changes made in this document for each release:

| Issue | Date | Summary of changes |
|---|---|---|
| Release 10.0, Issue 1 | July 2016 | • Added information on Communication Manager and Session Manager trunk configuration.<br>• Moved the Documentation terminology table from chapter two to chapter one.<br>• Moved Resources out of Introduction to an independent chapter called Resources. |
| Release 11.0, Issue 1 | June 2018 | Added support for new phones:<br>• Avaya J100 series IP desk phones:<br>- Avaya J129 series desk phones.<br>- Avaya J169/J179 phones (standard SIP phones only) |
| Release 11.0 SP1 | August 2018 | Added support phone:<br>J139 support as standard SIP phone |
| Release 11.0.4.2 | December 2019 | Added support phone:<br>J159 support as standard SIP phone |
| Release 11.1 FP1 | January 2021 | Added support phone:<br>J189 support as standard SIP phone |

# Chapter 2: IP Office as an enterprise branch overview

You can deploy IP Office as an enterprise branch to provide a communications solution that is adaptable to meet the growing needs of an enterprise branch network while providing investment protection of the installed hardware platform and phones. You can implement an IP Office enterprise branch on an IP Office Standard Mode, Essential, or Preferred system. The IP Office system can be installed as an independent, standalone branch, or be connected to the Avaya Aura® network and migrated to a Distributed, Centralized, or Mixed enterprise branch to provide specific features and applications to meet the needs of individual employees in each branch location.

In addition to centralized SIP endpoints or centralized analog devices configured as ATA, IP Office can concurrently support other IP and TDM endpoints for a community of Centralized users and IP Office users in the same branch. Ideal for enterprises wanting applications deployed in customer data centers or in the branch, an IP Office Branch can effectively deliver a range of communication tools without complex infrastructure and administration.

For more information on how to add Centralized users to an IP Office enterprise branch, see Administering Centralized Users for an IP Office™ Platform Enterprise Branch

IP Office is also supported in an Avaya Communication Server 1000 (CS 1000) branch environment. Only the Distributed enterprise branch option is supported. IP Office can be deployed as a new branch in an existing CS 1000 configuration with the addition of Avaya Aura® Session Manager to which IP Office connects through the SM Line for branch connectivity. Providing phone investment protection, IP Office can also be deployed as a replacement for Business Communications Manager (BCM) or Norstar in a branch office and connect to CS 1000 through Avaya Aura® Session Manager. IP Office cannot operate as a survivable gateway to CS 1000 endpoints as similar to Survivable Remote Gateway (SRG).

Integration of IP Office with CS 1000 is provided in a separate document. See Deploying Avaya IP Office™ as a Distributed Enterprise Branch in a Communication Server 1000 Environment with Avaya Aura® Session Manager.

## New in this release

IP Office Branch supports the deployment of the following as Centralized users:

- Avaya J129 IP Phone

- Avaya J139 IP Phone
- Avaya J159 IP Phone
- Avaya J169 IP Phone
- Avaya J179 IP Phone
- Avaya J189 IP Phone

Support for Avaya Aura® System Manager Release 8.1.

# Topology

The IP Office Branch solution provides the flexibility to support independent, stand-alone IP Office branches as well as IP Office branches connected to an Avaya Aura® system. The Branch solution also supports CS 1000 integration. The following deployment options are available for the solution architecture:

- **Stand-alone IP Office branch option**: Independent IP Office systems are deployed within the network. These IP Office systems are not connected to each other or to anything else in the network. With this option, IP Office Branches are not connected to an Avaya Aura® system and users cannot access any Avaya Aura® services.
- **Distributed enterprise branch deployment option**: All users in this deployment option are IP Office users. These IP Office users obtain telephony services from the local IP Office and not from Avaya Aura®. The IP Office systems in this deployment option can be connected to Avaya Aura® Session Manager and administrators can obtain Centralized management services through Avaya Aura® System Manager. The enterprise can choose to connect IP Office users in this deployment option to an IP Office voice mail system, Embedded Voicemail or Voicemail Pro, or a Centralized voice mail system, such as Avaya Aura® Messaging or Avaya Modular Messaging. IP Office users in this deployment also have access to some Centralized Avaya Aura® applications and services.

  With the Distributed branch deployment option, you can also connect CS 1000 to IP Office in the branch through Avaya Aura® Session Manager. Users still obtain telephony services from the local IP Office, but can use Avaya CallPilot® as their voice mail system. When connected to CS 1000, IP Office and CS 1000 interoperate as peers through Avaya Aura® Session Manager.
- **Centralized enterprise branch deployment option**: All users in the enterprise are Centralized users.

  Centralized users register to Avaya Aura® Session Manager and obtain telephony services from the Avaya Aura® Communication Manager Feature Server or Evolution Server in the enterprise core. If WAN connectivity to Avaya Aura® Session Manager is lost, the user automatically gets basic telephony services from the local IP Office. The telephony features provided by IP Office in survivability mode is limited compared to the features that are normally provided to the Centralized phone.

  Centralized users must be configured on the Avaya Aura® Session Manager, Communication Manager, and IP Office. A Centralized user must be configured on the Avaya Aura® Session

Manager and Avaya Aura® Communication Manager as a SIP user. On IP Office, the Centralized user must have either a SIP extension, an analog extension, or an analog fax device.

- **Mixed enterprise branch deployment option**: An enterprise branch with both Centralized users and IP Office users. Centralized users and IP Office users obtain the same telephony services described above. All users in this deployment option must use a Centralized voice mail system: Avaya Aura® Messaging or Avaya Modular Messaging.

The deployment options in the Branch solution allow you to start off with stand-alone IP Office systems and then slowly evolve the solution architecture into a Centralized environment as your enterprise grows.

The following image shows the topology of the solution architecture with the deployment options described above. This image does not show CS 1000 in the Distributed branch deployment.

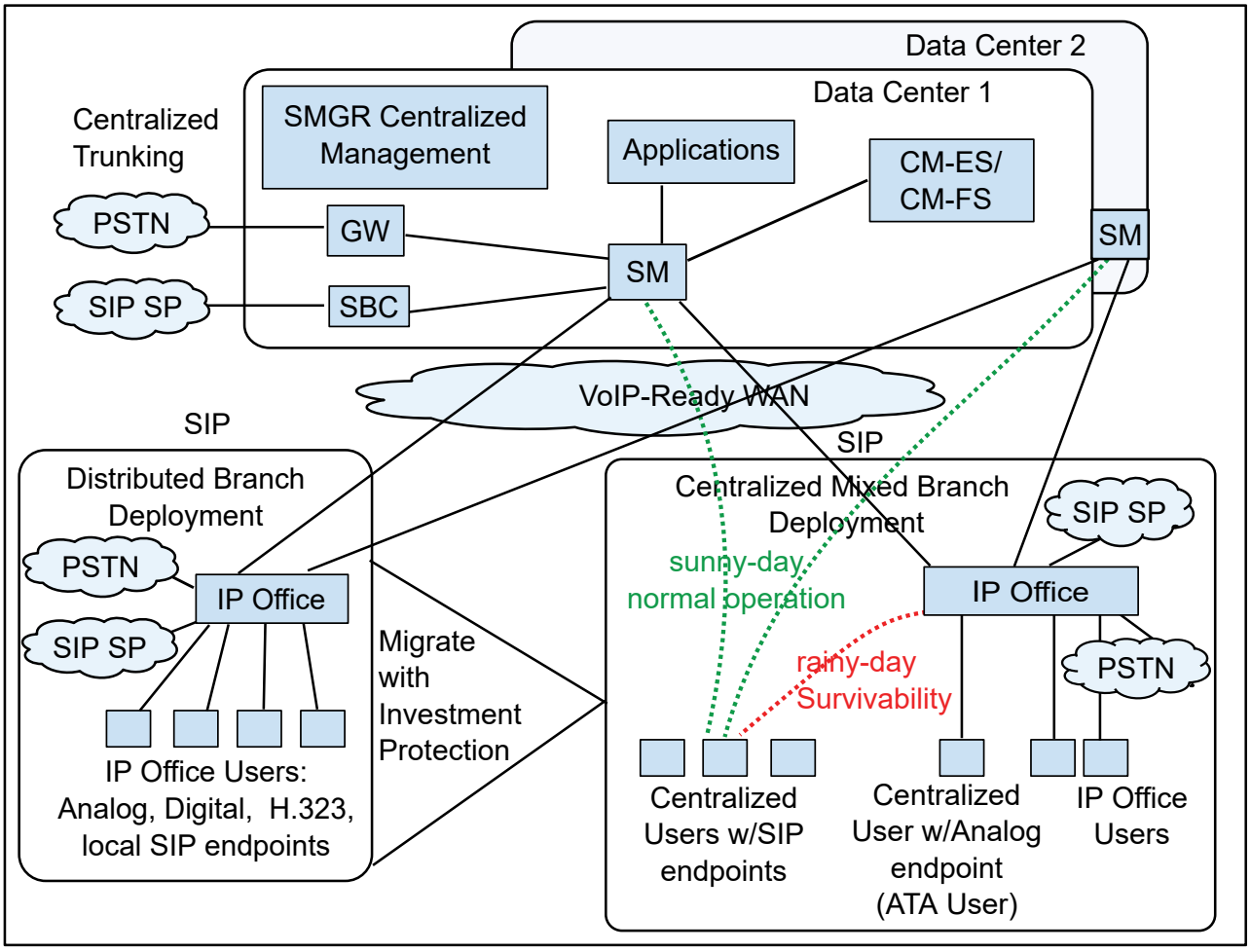


**Figure 1: Topology of solution architecture**

## Related links

Components on page 11
Redundancy and high availability configuration on page 48

# Components

The following table briefly describes the components in the IP Office Branch solution and how these components fit into the Branch architecture. Some of these components are only implemented in certain branch deployment environments. For more information about supported Avaya and third-party components, see https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml?name=IP+Office+Platform.

**Table 2: Branch components**

| Component | Reference Configuration Characteristics |
|---|---|
| IP Office | Includes hardware and software components. In this solution, there is an IP Office in each branch. IP Office systems can be deployed as stand-alone branches or connected to an Avaya Aura® network. |

*Table continues…*

| Component | Reference Configuration Characteristics |
|---|---|
| Main Avaya Aura® network | Avaya Aura® provides access to centralized services.<br>• Avaya Aura® System Manager provides administrative functions that administrators can use to easily manage all users and IP Office systems in the enterprise. In a Distributed, Mixed, or Centralized environment, IP Office branches are connected to the Avaya Aura® System Manager server.<br>**Note:**<br>Avaya Aura® System Manager does not manage the following applications: Voicemail Pro, Avaya one-X® Portal for IP Office, CCR, and Unified Communications Module.<br>• Avaya Aura® Session Manager connects branches to each other and to other applications and services in the enterprise center. For IP Office users in Distributed and Mixed deployment environments, Avaya Aura® Session Manager acts as a SIP proxy to route SIP sessions to and from the SIP connections to the IP Office. For centralized users, Avaya Aura® Session Manager is also the main interface that handles user registration and call routing.<br>• Avaya Aura® Communication Manager provides telephony services to centralized users through Avaya Aura® Session Manager.<br>• Avaya WebLM is included within the Avaya Aura® System Manager and provides centralized licensing management.<br>• Secure Access Link (SAL) provides remote access and alarm reception capabilities. The SAL Gateway is installed on a Red Hat Enterprise Linux host in the customer network and acts as an agent on behalf of several managed elements. All communication is outbound from the environment of the customer and uses encapsulated Hypertext Transfer Protocol Secure (HTTPS).<br>Other centralized services such as conferencing and messaging are available to centralized users. Some of these services are also available to IP Office users deployed in Distributed and Mixed branch environments. For more information about applications that interoperate with the IP Office branch, see the IP Office branch interoperability section. |

*Table continues…*

| Component | Reference Configuration Characteristics |
|---|---|
| CS 1000 | In a Distributed branch environment, you can connect CS 1000 to the IP Office in the branch through Avaya Aura® Session Manager. Users still obtain telephony services from the local IP Office, but can use Avaya CallPilot® as their voice mail system.<br><br>CS 1000 provides telephony interoperability between IP Office users in the branch and CS 1000 users at other sites. Avaya Aura® System Manager is used to centrally manage CS 1000.<br><br>In branch deployments with CS 1000, all IP Office endpoints are supported. Some CS 1000 endpoints can also be migrated to IP Office endpoints. For more information about supported endpoints, see the following documents:<br><br>• Deploying an IP500 V2 IP Office Essential Edition System<br>• *Deploying IP Office™ Platform as a Distributed Enterprise Branch in a Communication Server 1000 Environment with Avaya Aura® Session Manager*<br><br>**Important:**<br><br>IP Office does not support Meridian Customer Defined Networking (MCDN), but all major Branch features, including dial plans and centralized voice mail, are supported in Distributed branch environments with CS 1000. |

*Table continues…*

| Component | Reference Configuration Characteristics |
|---|---|
| Supported endpoints | All IP Office endpoints are supported as IP Office users. For a list of IP Office endpoints, see phones in Deploying an IP500 V2 IP Office Essential Edition System.<br><br>The following SIP endpoints are supported as Centralized users. For information about centralized phones and adding Centralized users to an enterprise branch, see Administering Centralized Users for an IP Office™ Platform Enterprise Branch.<br><br>• 9620 SIP<br>• 9630 SIP<br>• 9640 SIP<br>• 9650 SIP<br>• 9601 SIP<br>• 9608 SIP<br>• 9611G SIP<br>• 9621G SIP<br>• 9641G SIP<br>• 1120E<br>• 1140E<br>• 1220<br>• 1230<br>• Avaya B179<br>• Avaya J159 IP Phone<br>• Avaya J169 IP Phone<br>• Avaya J179 IP Phone<br>• Avaya J129 IP Phone<br>• Avaya J139 IP Phone<br>• Avaya J189 IP Phone<br>• Avaya Workplace Client for Windows<br><br>**Note:**<br><br>The 9600 series SIP phones and Avaya one-X® Communicator SIP are supported only as Centralized phones for use by Centralized users. They are not supported as IP Office phones for use by IP Office users.<br><br>The following endpoints are supported as IP Office or Centralized users:<br><br>• Avaya H175 Video Collaboration Station<br>• The 1100 and 1200 series phones |

*Table continues…*

| Component | Reference Configuration Characteristics |
|---|---|
| | • B179 series phones<br>• J100 series for J129, J139, J159, J169/179, J189 phones (Supported only as Standard SIP phone)<br>IP Office Branch also supports video endpoints connected to IP Office or to the Avaya Aura® infrastructure. |

**Related links**

IP Office branch interoperability on page 15
Video configuration on page 38

# IP Office branch interoperability

The IP Office in the branch can interoperate with other applications and services. For more information about IP Office branch interoperability, see https://secureservices.avaya.com/compatibility-matrix/menus/product.xhtml?name=IP+Office+Platform.

Centralized SIP users in the branch connect directly to Avaya Aura® Session Manager in sunny-day and have full access to all central applications and services that the Avaya Aura® solution provides to SIP users. IP Office users in the branch cannot access all the features provided through the Avaya Aura® solution. Users in a stand-alone IP Office branch deployment do not have access to any of the centralized applications and services.

## Differences in service for IP Office users and Centralized users

IP Office users and Centralized users cannot access all of the services offered through the products listed above. The following table describes the services available to the different users.

| Products | Description |
|---|---|
| Avaya Aura® Session Manager | Used by Centralized users and IP Office users, but plays an enhanced role for Centralized users. For Centralized users, Session Manager is the main interface that handles user registration and call routing. |
| Avaya Aura® Conferencing | Avaya Aura® Conferencing services can be accessed by IP Office users and Centralized users, but certain services are not available to IP Office users.<br>IP Office users can dial in to an Avaya Aura® Conferencing bridge and be participants in an Avaya Aura® ad-hoc conference. However, IP Office users cannot start ad-hoc Avaya Aura® conferences or perform conference control functions. |

*Table continues…*

| Products | Description |
|---|---|
| Avaya Experience Platform™ (AXP) On-Prem (formerly Avaya Call Center Elite) | IP Office users cannot have agent roles. Only Centralized users can perform agent roles within the branch. In the Sunny day, the Centralized user is registered to Session Manager and has full agent access. In the Rainy day, the Centralized user in the branch is unavailable and no longer accessible to the Avaya Call Center. The Centralized user's phone registers to the IP Office in the Rainy day and obtains access to the survivable telephony features supported by IP Office. When the Sunny day returns, the Centralized user will reregister to Session Manager and again be available to the Avaya Call Center. |
| Avaya Aura® Presence Services | Centralized users can use Avaya Aura® Presence Services.<br>IP Office users cannot use Avaya Aura® Presence Services. IP Office can only use IP Office Presence, which requires Avaya one-X® Portal to be deployed in each branch on Unified Communications Module or on an external server. No interaction is currently supported between IP Office and Avaya Aura® Presence Services. |
| Avaya Aura® Messaging | When the connection to the Avaya Aura® network is down, the **Message Waiting Indicator** button does not light up to indicate a new message. When the connection to the Avaya Aura® network is restored, the **Message Waiting Indicator** continues working as expected for Centralized users. However, for IP Office users, the Avaya Aura® Messaging server does not refresh immediately even after the connection to the Avaya Aura® network is restored, the **Message Waiting Indicator** does not light up immediately to indicate a new message. |
| Lync Integration client plug-in | The IP Office Lync Integration client plug-in is available to IP Office users. It requires Avaya one-X® Portal to be installed in each branch on a Unified Communications Module or an external server. The IP Office Lync Integration client plug-in cannot be used by Centralized users.<br>Centralized users in the Sunny day can use Lync Integration, which supports integration with the Avaya SIP phones using Session Manager. This sunny-day support does not involve IP Office. In the Rainy day, when the branch loses connection to the center, the Lync integration of the Centralized users through ACA will not be available. |
| CS 1000 and Avaya CallPilot® | Only available in a distributed branch deployment connected to CS 1000. With this option, IP Office users can use Avaya CallPilot® as their voice mail system. |

*Table continues…*

| Products | Description |
|---|---|
| Radvision Scopia | Radvision Endpoints are video endpoints for IP Office. Using this option, IP Office devices can join or dail out from a meeting. NAT Firewall traversal is offered to enable outside to/from inside video calls. This enables Internet based endpoints to make calls to video endpoints or meeting rooms in a company network.<br>The Elite MCU series supported with IP Office are Elite 5100 MCU Series and Elite 5200 MCU Series. |
| Avaya Session Border Controller | This is applicable to Distributed, Mixed, and Centralized branch deployments and includes the following types of calls:<br>• Local SIP trunk - Session Border Controller - IP Office - IP Office user<br>• Local SIP trunk - Session Border Controller - IP Office - Session Manager - Centralized users (in sunny-day)<br>• Local SIP trunk - Session Border Controller - IP Office - Centralized users (in rainy-day)<br>• Local SIP trunk - Session Border Controller - IP Office - Session Manager - users in headquarters or other enterprise sites<br>• Central SIP trunk - Session Border Controller - Session Manager - IP Office - IP Office user<br>• Central SIP trunk - Session Border Controller - Session Manager - Centralized users<br>• IP Office remote worker - Session Border Controller - IP Office - Session Manager - users in headquarters or in other enterprise sites<br>• IP Office remote worker - Session Border Controller - IP Office - Session Manager - Centralized users in a Mixed branch (in sunny-day)<br>• IP Office remote worker - Session Border Controller - IP Office - Centralized users in a Mixed branch (in rainy-day)<br>• IP Office remote worker - Session Border Controller - IP Office - Session Manager - voice mail services on Avaya Aura® Messaging/ Modular Messaging.<br>• Avaya Aura® remote worker - Session Border Controller - Session Manager - IP Office - IP Office user |
| Avaya BCM | Positions IP Office as a SIP gateway between BCM and Session Manager. |
| Avaya Experience Portal | IP Office supports basic interoperability with Avaya Experience Portal for SIP calls sent through the Avaya Aura® Session Manager. |
| Voice Portal | Provides only basic connectivity support with IP Office. |

*Table continues…*

| Products | Description |
|---|---|
| Avaya Aura® Communication Manager | IP Office supports interoperability with Avaya Aura® Communication Manager for SIP calls sent through the Avaya Aura® Session Manager. In addition, Centralized users use the Communication Manager as their telephony server that provides their features and handles all their calls in Sunny day. |

# Customer requirements

The IP Office Branch solution supports the following customer requirements:

- Flexible deployment options with autonomous IP Office branches as well as Distributed, Mixed, and Centralized Branch environments.
- Support for multiple voicemail options depending on the deployment model.
- Freedom to evolve to different deployment phases as the needs of your enterprise change. You do not have to follow the evolution phases in any particular order. For more information about evolution phases, see Avaya IP Office™ Platform Solution Description.
- The flexible deployment options offered with the architecture provide guaranteed investment protection within the hardware platform and phones.
- Centralized management services available in Distributed, Mixed, and Centralized Branch environments with the Avaya Aura® network.
- Access to PSTN trunking and a variety of other services, such as messaging and conferencing, for centralized users with the Avaya Aura® network. IP Office users in the Distributed and Mixed Branch environment can also access some of these services.
- Support for CS 1000 configurations connected to the IP Office in the branch with Avaya Aura® Session Manager. This provides telephony interoperability and phone investment protection.
- Support for the Rainy day mode, guaranteeing that centralized users still have access to telephony services through IP Office when the WAN connection between the branch and the enterprise center is down or if the core Avaya Aura® server is down.

**Related links**

Topology on page 9
IP Office branch interoperability on page 15

# Voicemail options

The following IP Office voicemail options are supported:

## Local voicemail options

IP Office systems deployed as Distributed or stand-alone enterprise branches can be configured to use the following local voice mail options:

- **Embedded Voicemail** — Embedded Voicemail is the IP Office default voicemail option. It uses the system SD card in the IP Office system control unit for storage of prompts and messages. Embedded Voicemail supports mailboxes for all local extension numbers, announcements to waiting callers, and auto attendants (up to 40) for external calls. The capacity is limited to 15 hours of recorded messages, prompts and announcements.
- **Voicemail Pro** — Voicemail Pro runs on an external server connected to IP Office or on Unified Communications Module and provides a wide range of features. Voicemail Pro supports 40 voice channels, which includes:
- Message/greeting recording
- Message playing
- System/custom IVR prompt playing
- Call recording and prompt playing to outcalls made from VoicemailPro

Mailbox messages, greetings, recordings, and announcements are stored locally on Unified Communications Module or on a standalone computer where Voicemail Pro is running. Management of the Voicemail Pro call flows in the branches can be done from a remote central Voicemail Pro client. Use of the Voicemail Pro client to remotely manage the calls flows is available only when Voicemail Pro is running on an external server, not when it is deployed on Unified Communications Module. The Voicemail Pro client can also connect to Voicemail Pro that runs the Unified Communications Module server and can therefore, be used in the management. For enterprise branch deployments with multple Voicemail Pro servers, the Voicemail Pro client provides an export or import feature that allows the same Voicemail Pro configuration including call flows to be deployed on multiple Voicemail Pro servers.

**Note:**

If the IP Office 500v2 systems are present in the enterprise deployment with System Manager, you can manage Voicemail Pro callflow.

Voicemail Pro is the only option that supports manual call recording for IP Office users and automatic call recording for the IP Office system.

**Note:**

If security of signaling links is a concern, the Voicemail Pro application should run on a co-resident Unified Communications Module blade. Security enhancements such as TLS are not applicable to links with external servers running Voicemail Pro. You can now manage Voicemail Pro from Avaya Aura® System Manager.

The local voicemail options can be configured only in Distributed enterprise branch deployments where there are only IP Office users. They cannot be configured in Centralized or Mixed enterprise branch deployments with Centralized users. Only one type of voicemail system can be configured for an IP Office branch. This does not preclude the use of local auto-attendant in Centralized or Mixed enterprise branch deployments.

### Centralized voicemail options

IP Office systems deployed in Distributed, Mixed, or Centralized enterprise branch environments can be configured to use the following Centralized voicemail options. Stand-alone branches cannot use Centralized voicemail systems.

- **Avaya Aura Messaging** — An IP Office system can be configured to use Avaya Aura® Messaging as its voicemail server when Session Manager is used as the core SIP router.
- **Modular Messaging** — An IP Office system can be configured to use Modular Messaging as its voicemail server when Session Manager is used as the core SIP router.
- **CallPilot** — An IP Office system deployed as a Distributed enterprise branch in a Avaya Communication Server 1000 branch environment can be configured to use Avaya CallPilot® as the voicemail system. In this deployment, the Distributed enterprise branch configuration includes Session Manager as part of the CS 1000 core. All users in this configuration are IP Office users.

When a central voicemail system is configured for the branch, the mailboxes of the branch users are on the central voicemail system. However, you still have the option to use the automated attendant of the IP Office local Embedded Voicemail or the call flows of a local Voicemail Pro, even though the embedded voicemail or Voicemail Pro are not used for the voicemail messages of the branch users. This configuration requires the use of Embedded Voicemail or Voicemail Pro port licenses. If the configuration requires more ports than the number of ports that are licensed and come with the IP Office system, additional port licenses are required.

When Messaging or Modular Messaging is used as the central voicemail system, IP Office users and Centralized users in Rainy day can leave and retrieve voicemail over the PSTN when the SM Lines are unavailable. DTMF digits are used to indicate a specific mailbox. Message Waiting Indication is not provided.

# Test strategy summary

Avaya conducted thorough Solution Verification (SV) testing for all the features available in the Branch deployment options and for IP Office interoperability with CS 1000 and major Avaya Aura® components. Once the SV testing was complete, Avaya Solution and Interoperability (SIL) teams also re-tested features, interoperability, failover, and deployment to ensure solution stability and resolve any remaining issues.

This topic describes the Avaya Solution and Interoperability test strategy for the IP Office Branch solution in distributed, mixed, and centralized enviroments connected to the Avaya Aura® infrastructure. Avaya did not conduct any separate tests as part of this solution for stand-alone IP Office branch deployments. Stand-alone IP Office branches operate in the same way as the standard IP Office system.

The goal of the Avaya Solution and Interoperability tests conducted was to achieve solution stability for:

- Installation, configuration, and maintenance
- Feature functionality in all test conditions as well as in standard and stretched network conditions

- Failover and failback scenarios

# Chapter 3: Design considerations

## Caveats and limitations

The following general caveats and limitations exist for the Avaya IP Office Branch solution.

### Solution not supported in all IP Office editions

Branch functionality is supported in IP Office Essential and Preferred Editions. The following limitations exist for branch support in other IP Office editions:

- The IP Office Basic Edition does not support branch functionality.
- The IP Office Server Edition is not positioned as a branch edition. The Server Edition had the ability to support certain call features and interoperability with the following products using SIP through an SM Line interface:
  - Avaya Aura® Session Manager
  - Avaya Aura® Communication Manager or Avaya Communication Server 1000 through the Session Manager

  However, the Server Edition does not support the following centralized services available in Branch:
  - Centralized management through Avaya Aura® System Manager
  - Centralized licensing through Avaya WebLM
  - Centralized users
  - Centralized voice mail over the SM Line
- Co-existence of Branch functionality with the IP Office Small Community Network (SCN) is also limited. SCN can connect to the Session Manager through the SM Line. However SCN does not support centralized users, centralized voice mail over the SM Line, centralized management through System Manager, and centralized licensing through Avaya WebLM.

### IP Office Branch positioning with Avaya Aura®

IP Office is not an H.248 gateway under the control of Communication Manager. IP Office is a telephony system and SIP gateway connected through SIP to Session Manager. IP Office as a Branch solution has the following limitations:

- A Session Manager and SIP gateway solution with IP Office Branch does not support the Communication Manager Inter-Gateway Alternate Routing (IGAR) feature.

- Communication Manager cannot directly control the media resources of the IP Office. In Sunny day mode, Communication Manager based media services, such as conferencing and Music on Hold, deliver media to the branch from a central Communication Manager gateway resource.

## IP Office administration

When System Manager is configured to centrally manage IP Office systems, you cannot use the IP Office Manager to administer the IP Office systems. If you need to administer the IP Office using the IP Office Manager that is directly connected to the branch, you must first disable the System Manager administration feature for the Branch. For more information, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

## User functionality

Mixed enterprise branch deployments support both Centralized users and IP Office users. Centralized users obtain telephony services from the Avaya Aura® Communication Manager Feature or Evolution Serve in the enterprise core, and IP Office users obtain telephony services from the local IP Office. As a result, when the Mixed enterprise branch is deployed, the IT department should identify and record which telephones are associated with a Centralized user experience and which telephones are associated with an IP Office user experience.

## No support for Communication Manager Messaging

The IP Office Branch solution does not support Communication Manager Messaging. Avaya Aura® Messaging and Avaya Modular Messaging are supported as Centralized voice mail systems.

## Mixed conferences

Merging a local IP Office conference with a conference on Avaya Aura® Conferencing is not supported on an IP Office phone.

- A SIP trunk call connected to an Avaya Aura® Conferencing network cannot be added to an IP Office conference. The calls will not merge if you try to create a local IP Office ad-hoc conference.
- A SIP trunk call added to a local IP Office conference cannot process conferencing data from the Avaya Aura® Conferencing network.

## One voice mail type per branch

IP Office only supports one voice mail type for all users in the same branch. In a Mixed branch with both IP Office users and Centralized users, all users must use a Centralized voice mail system. IP Office users in the Mixed branch cannot select a local IP Office voice mail system.

## 1100 and 1200 series phones deployed as Centralized users

- Ensure that the 1100 and 1200 series phones within a branch are all Centralized or all IP Office users, otherwise you must configure each phone manually.
- In the Rainy day mode, direct media with SRTP is not supported. In this scenario, media is anchored on IP Office.
- The address book feature of the phone varies for different servers. Hence, the contacts added to the phones while the phones are connected to Session Manager are unavailable when the phones are connected to IP Office, and vice versa.
- Install the security certificate manually. You must accept the authorization prompts from the phones to go ahead and install the certificate. If you do not provide any input, then the certificate does not get installed.
- 1100 and 1200 series phones support only two servers. You can deploy this phone as Centralized users in branches that connect to a single Session Manager and configure it with the local branch IP Office as the secondary server. However, you cannot deploy this phone as Centralized users in a branch that is configured to two Session Manager. If there are Centralized 1100 series or 1200 series phones in the branch, then you must not configure the IP Office and other Centralized phones in that branch to two Session Manager. If you configure the IP Office with two Session Manager, then the Centralized 1100 series and 1200 series phones will receive no service. This occurs if the primary Session Manager is down and the secondary Session Manager is still reachable from the IP Office. The Centralized 1100 series and 1200 series phones will not be able to register to the IP Office that will still be in Sunny day mode.

## B.179 series phones deployed as Centralized users

IP Office deployed as a Branch supports B179 R2.4 phones as Centralized users. The configuration of the Centralized B179 phone is done through the phone's built-in configuration tool that is available through a web browser and not through a configuration file. The B179 phone does not support manual failback. Therefore, when a deployment includes a B179 phone as a Centralized User, the failback policy must be automatic.

## Avaya H175 Video Collaboration Station endpoints deployed as Centralized users

The following is not supported on the Avaya H175 Video Collaboration Station in Rainy day mode:

- Message Waiting Indication (MWI).
- one-X Portal and Voicemail Pro.
- Ad-hoc conferencing because local three-way conferencing is not supported.
- Deploying an Avaya H175 Video Collaboration Station as a Centralized user and another Avaya H175 Video Collaboration Station as an IP Office user in the same branch.

The following behaviors apply when the Avaya H175 Video Collaboration Station is deployed as a Centralized user:

- Rainy day mode with IP Office is treated in the same way as it is with a third party SIP survivability server.
- The Avaya H175 Video Collaboration Station is not considered an IP Office user when it fails over to IP Office in Rainy day mode.
- IP Office treats an Avaya H175 Video Collaboration Station Centralized user and 9608, 9611, 9621, and 9641 SIP Centralized users in the same way.

# Capacity and scalability

The IP Office Branch solution is flexible and scalable to meet the needs of small, medium, and large enterprises. You can also evolve the architecture of the solution as your enterprise grows.

The following table defines general capacity information for all deployment options in the IP Office Branch solution. See the IP Office Knowledge Base for detailed capacity information.

For a list of the maximum capacity limits for each Avaya Aura® System Manager in the solution, see **Performance specifications** in *Avaya Aura® System Manager Overview and Specification*. System Manager limits only apply if your solution uses a Distributed, Mixed, or Centralized deployment connected to the Avaya Aura® infrastructure.

**Table 3: IP Office Branch general capacity information**

| Description | Capacity |
|---|---|
| Number of users | Solution is ideal for a small or large number of users.<br>Up to 384 users can be deployed on each IP Office branch, and up to 250 000 users can be deployed on Avaya Aura® System Manager. |
| Number of branches | Up to 2000 IP Office branches can be connected to an Avaya Aura® Session Manager. Up to 2000 Unified Communications Module or Application Server can also be supported in addition to the 2000 IP Office branches. The IP Office Branch solution is not limited to 250 gateways per Communication Manager. |
| Number of SIP sessions | Up to 128 calls signalled through SIP are supported between the IP Office branch and the Avaya Aura® Session Manager. |

*Table continues…*

| Description | Capacity |
|---|---|
| Number of trunks and channels | The Branch solution supports up to:<br>• 16 analog trunks<br>• 16 ISDN BRI trunks (32 channels)<br>• 8 digital PRI trunks (up to 192 T1 channels or 240 E1 channels) |
| IP Office voice mail | The IP Office system uses Embedded Voicemail by default. Embedded Voicemail supports up to:<br>• 6 ports<br>• 40 auto-attendants<br>• 15 hours of recorded messages, prompts, and announcements.<br>With the IP Office Preferred or Advanced editions, VoiceMail Pro is also supported. The VoiceMail Pro system supports up to 40 ports. |

The capacity information presented in Table 3: IP Office Branch general capacity information on page 25 describes general maximums supported by the IP Office system, but these maximums are not automatically available to all enterprises that purchase IP Office systems. You must obtain licenses and hardware modules to deploy VCM channels, SIP sessions to and from Avaya Aura® Session Manager, and voice mail ports (if applicable). Use the information in the following sections to calculate what you need to obtain to meet the branch requirements for your enterprise.

## Avaya WebLM capacity

The Avaya WebLM on the Avaya Aura® System Manager supports licensing of up to 2000 IP Office branches from Avaya Aura® Release 6.2 Feature Pack 3 onwards.

**Note:**

The previous releases of the WebLM server, prior to Avaya Aura® 6.2 Feature Pack 3 has limited capacity. The server could handle approximately 300 to 400 IP Office systems in a typical Branch deployment. The number depends on the server load and on the number of different license types requested by the IP Office systems. If your deployment has a large number of branches, then you must upgrade your System Manager to Avaya Aura® Release 6.2 Feature Pack 3 or later.

## VCM channels

IP Office uses voice compression modules (VCMs) to convert media traffic between IP and TDM. Some examples of when VCM channels are required in the branch are:

- Each call between an IP endpoint or trunk and a TDM endpoint or trunk

- Each call between an IP Office analog or digital endpoint and a device behind the SM Line, including calls to phones in other sites, centralized phones within the branch, and centralized trunks or voice mail systems through the Avaya Aura® Session Manager
- Each call between an IP Office ISDN or analog PSTN trunk and a device behind the SM Line, including calls between centralized SIP users within the branch local ISDN or analog PSTN trunks.

Calls between IP Office users with IP endpoints and entities behind the SM Line are considered IP device to IP device calls. These calls do not typically require VCM channels unless no common codec can be selected on the two sides.

The following table describes general usage of VCM channels within IP Office environments.

| Call type | Voice compression channel usage |
|---|---|
| IP device to non-IP device | These calls require a voice compression channel for the duration of the call. If no channel is available, busy indication is returned to the caller. |
| IP device to IP device | Call progress tones (for example dial tone, secondary dial tone, and so on) do not require voice compression channels with the following exceptions:<br>• Short code confirmation, ARS camp on and account code entry tones require a voice compression channel.<br>• Devices using G723 require a voice compression channel for all tones except call waiting.<br>When a call is connected:<br>• If the IP devices use the same audio codec no voice compression channel is used.<br>• If the devices use differing audio codecs, a voice compression channel is required for each. |
| Non-IP device to non-IP device | No voice compression channels are required. |
| Music on Hold | This is provided from the IP Office TDM bus and therefore requires a voice compression channel when played to an IP device. |
| Conference resources and IP devices | Conferencing resources are managed by the conference chip which is on the IP Office TDM bus. Therefore, a voice compression channel is required for each IP device involved in a conference. This includes services that use conference resources such as call listen, intrusion, call recording and silent monitoring. |
| Page calls to IP device | For page calls, IP Office 4.0 and later can use any codec, except G722. IP Office requires one channel for each codec type used by the target phones. |

*Table continues…*

| Call type | Voice compression channel usage |
|---|---|
| Voice mail services and IP devices | Calls to the IP Office voice mail servers are treated as data calls from the TDM bus. Therefore calls from an IP device to an IP Office voice mail system require a voice compression channel. This does not apply to central voice mail systems over the SM line. |
| T38 fax calls | In order to use T38 fax connection, fax tone detection is performed if the analog extension connected to the fax machine is set as “Standard telephone.” If the fax machine does not include an attached handset that is used to make/receive voice calls, then the Equipment Classification of an analog extension connected to the fax machine can be set to **Fax Machine**, which will result in T38 fax connection without fax tone detection and respective signaling renegotiation. Additionally, a new short code feature, Dial Fax, is available. |

**Note:**

T3 IP devices must be configured to 20ms packet size for the above conditions to apply. If left configured for 10ms packet size, a voice compression channel is needed for all tones and for non-direct media calls.

### Measuring channel usage

The IP Office System Status Application can be used to display voice compression channel usage. Within the **Resources** section it displays the number of channels in use. It also displays how often there have been insufficient channels available and the last time such an event occurred.

For the VCM cards, the level of channel usage is also indicated by the LEDs (1 to 8) on the front of the VCM card.

### Calculation guidelines

On the IP Office, you can configure the following number of VCM modules:

- Up to 2 optional VCM 32 modules or VCM 64 modules, providing 32 and 64 VCM channels each
- Up to 2 combination cards, providing 10 VCM channels each

With 4 modules, the maximum VCM channel capacity is 148.

The calculation of how many VCM channels are required in a particular deployment depends on the type and number of phones in the branch, the types of trunks used, the traffic patterns or typical call flows, as well as typical call frequency and length. For example, if most phones are centralized SIP phones and most calls are made to local ISDN or analog PSTN trunks, then VCM channels will be needed for most calls. However, VCM channels will not be needed as often if most phones are analog or digital phones and most calls are made to local ISDN or analog PSTN trunks.

As a general rule, the number of required VCM channels is equivalent to the number of IP Office IP phones (not including centralized phones) divided by three, plus the number of SIP trunk channels used for simultaneous calls over the SM Line. However, this calculation rule can vary depending on the deployment and traffic patterns of the enterprise.

# SM trunk channel licenses

You must obtain a SM trunk channel license for each simultaneous SIP sessions to the Session Manager or call required in your IP Office branch. For example, if your system needs to support 5 simultaneous SIP sessions, you must obtain 5 SM trunk channel licenses. Calls over the SM trunk use only SM trunk channel licenses. Regardless of the call media, an SM trunk channel license is required for each call with signaling over the SM Line.

Examples of calls over the SM Line include:

- Calls between an IP Office user and a device behind the SM Line. This includes calls to users in other sites, to centralized users within the branch in sunny-day, and centralized trunks or voice mail through Avaya Aura® Session Manager
- Calls between an IP Office PSTN trunk and a device behind the SM Line

### Calculation guidelines

The calculation of how many SM trunk channel licenses are required depends on the expected traffic patterns in the enterprise deployment. If your deployment uses Avaya WebLM, the number of SM trunk channel licenses required by the IP Office must be specified in the IP Office licensing configuration.

The ratio of the average number of users to each SM trunk channel license must be calculated based on traffic patterns and media call flows. Do not assume that the user to license ratio needs to be 1 to 1. To determine the appropriate ratio, look at patterns such as the average number of SIP calls over the SM Line, the call duration, and the destination of incoming calls.

**Related links**

IP Office SIP connection to Avaya Aura Session Manager on page 36

# Voice mail port licenses: Embedded Voicemail and Voicemail Pro

You must obtain a voice mail port license for each simultaneous call you want to allow to your IP Office voice mail system (Embedded Voicemail or Voicemail Pro). Voice mail port licenses for Embedded Voicemail are required in the following two cases:

- Branch users use Embedded Voicemail as their voice mail system.
- Branch users use a centralized voice mail system through Avaya Aura® Session Manager (Avaya Aura® Messaging, Modular Messaging, or CallPilot®), but use Embedded Voicemail for auto-attendant features and announcements.

Voice mail port licenses for Voicemail Pro are required in the following two cases:

- Branch users use Voicemail Pro as their voice mail system.
- Branch users use a centralized voice mail system through Avaya Aura® Session Manager (Avaya Aura® Messaging, Modular Messaging, or CallPilot®), but use Voicemail Pro for local call flow processing.

### Calculation guidelines

The following table provides information about:

- The number of voice mail ports provided as entitlements with the IP Office Essential, Preferred, and Advanced packages.
- The maximum number of ports that can be deployed on IP Office voice mail systems.

**Table 4: Ports entitlements with IP Office package editions**

| Ports provided as entitlements | |
|---|---|
| IP Office Essential edition | 2 ports provided as entitlements |
| IP Office Preferred and Advanced editions | 4 ports provided as entitlements |

**Table 5: Maximum number of ports supported with IP Office voice mail systems**

| Maximum number of ports supported | |
|---|---|
| Embedded Voicemail | up to 6 ports |
| Voicemail Pro | up to 40 ports |

If the enterprise deployment uses Avaya WebLM licensing, the required number of IP Office voice mail port licenses for Embedded Voicemail or Voicemail Pro must be specified in the IP Office licensing configuration.

# Migration roadmap and limitations

You can upgrade from a B5800 Branch Gateway to the current release of IP Office, or from a previous release of IP Office to the current release of IP Office. The upgrade process for the IP Office Branch solution varies depending on how the architecture is deployed.

If the B5800 Branch Gateway or IP Office release you are upgrading from is connected to the Avaya Aura® infrastructure and uses Avaya Aura® System Manager for centralized management, use Avaya Aura® System Manager to upgrade the firmware and software for the system.

If the architecture is deployed in a stand-alone IP Office branch environment with no centralization or connection to the Avaya Aura® network, you must use IP Office Manager for upgrades. From IP Office Manager, you can perform an upgrade using the upgrade wizard or the System SD card.

For procedures on how to perform System Manager or IP Office Manager upgrades for your Branch solution, see Migrating an IP Office or B5800 Branch Gateway to an IP Office Enterprise Branch.

# Dial plan considerations

A uniform dial plan greatly simplifies configuration, management and phone calls within the network branch sites. For example, if each branch has similar roles such as reception, manager and warehouse, using the same extension number for each role and a unique prefix for each branch allows calls between sites with little need for directory lookups. It also means a standard configuration can be used at branches which simplifies installation, user training and maintenance.

IP Office supports dial plans comprised of the branch prefix and local number length for IP Office users and enterprise-wide extensions for Centralized users.

## Branch prefix and local number length for IP Office users

Dial plans comprised of the branch prefix and local number length for IP Office users should not exceed 15 digits. The **Branch Prefix** field and the **Local Number Length** field appear in IP Office Manager under **System** > **Telephony** > **SM** tab.

The branch prefix enables IP Office users to have short extension numbers within the local branch but appear to the rest of the enterprise to have unique full enterprise-wide numbers in enterprise canonical format. If a number is configured in the **Branch Prefix** field, it causes automatic conversion of the IP Office users' extension numbers in calls to and from the SM Line. The branch prefix is added as a prefix to the IP Office user's extension number when it appears as the calling number in calls sent to the SM Line. Similarly, it is added to the extension number when it appears in Diversion-Header or History-Info in calls sent over the SM Line to centralized voicemail.

In calls received from the SM Line, if the called number starts with the branch prefix, the prefix is removed and the IP Office will try to target the remaining number locally, to a non-centralized extension or hunt group. If there is no match, IP Office will target it to any matching system short code. If the called number does not start with the branch prefix, the whole number is checked for a match against system short codes.

Centralized users (SIP or ATA) can call local IP Office users in the same branch by dialing the full enterprise number composed of the branch prefix plus local extension number, and IP Office will remove the branch prefix and target the local extension. This may be needed if short-form dialing is not set up on the centralized Avaya Aura® Communication Manager for Sunny day and the Centralized users are accustomed to dialing the full number.

IP Office users cannot call other local IP Office users in the same branch by dialing the full enterprise number composed of the branch prefix plus local extension number.

**Note:**

This could be enabled in Sunny day if there is a short code in the IP Office configuration that matches the branch prefix and routes it to the SM Line. But it cannot be enabled in Rainy day. Users should not be instructed to dial differently in Sunny day and Rainy day. Therefore, enabling IP Office users to be able to call other local IP Office users in the same branch by dialing the full enterprise number should not be enabled for Sunny day.

When a call is made by any source on the IP Office system and short code matching targets the SM Line, if the target (the dialed number) starts with the branch prefix, the branch prefix is removed and the call is targeted locally. This does not apply if the target is the extension number of a Centralized user or Centralized group configured on the IP Office. In this case, the call is handled as specified below and in Sunny day it will be sent to Session Manager.

The branch prefix is expected to be used in Distributed enterprise branch deployments and it is expected to be left blank in Centralized enterprise branch deployments. In Mixed enterprise branch deployments, the branch prefix can be left blank or used. If left blank, the IP Office users' extension numbers will have to be the full enterprise number. If used, the IP Office users' extension numbers will be shorter than the Centralized users' extension numbers.

The **Local Number Length** field sets the default length for extension numbers. If an extension number of a different length is configured, a warning appears. The **Local Number Length** field can be left blank. In Mixed enterprise branch deployments, if the **Branch Prefix** field is not blank, and the Centralized users and IP Office users have extension numbers of different lengths, then it is recommended to leave the **Local Number Length** field blank.

### Enterprise-wide extensions for Centralized users

The extension number configured on IP Office for the Centralized user is the user's enterprise-wide number. This is the same number that is configured for that user on Session Manager, which is the number that the centralized phone uses when registering to Session Manager in Sunny day and when registering to IP Office in Rainy day.

Centralized users' extension numbers can be up to 13 digits in length. IP Office users' extensions can be up to 9 digits in length.

**Note:**

Although IP Office deployed as a Centralized branch supports extension numbers up to 15 digits, the 13-digit length is determined by the maximum extension number length allowed for provisioning Centralized users in Communication Manager.

The IP Office **Branch Prefix** field, which causes number modifications for IP Office users' extension numbers, does not impact Centralized users' extension numbers when they appear as calling number or as called number.

Centralized users may have extension numbers that are not related to the local IP Office branch prefix or numbers that begin with the same digits as the IP Office branch prefix. If Centralized users have extension numbers that begin with the same digits as the IP Office branch prefix, the Centralized users' extension numbers look like the IP Office users' extension numbers to the rest of the enterprise. This enables users to keep their numbers when migrating from a Distributed enterprise branch to a Centralized enterprise branch.

### Targeting/routing to Centralized users

Calls to centralized users, which can arrive from different sources, are sent to the SM Line in Sunny day and are targeted to the Centralized user's extension locally in Rainy day. Both Sunny day and Rainy day call handling are done automatically by matching the called number to the Centralized user's extension number.

IP Office can manage calls to Centralized users that are dialed to the Centralized user's full extension number, or calls that are dialed using short-form dialing. To support short-form dialing, the global parameter Short Form Dialing Length must be configured. The **Short Form Dialing Length** field appears in IP Office Manager under **System > Telephony** tab **> SM** tab. Configuration of this feature allows IP Office to treat the last *N* digits (where *N* is the number configured for the Short Form Dialing Length) as an alias to that user's extension number.

### Short-form dialing from Centralized phones

The Sunny day conversion from the dialed short-form number (for example, 1111) to the enterprise canonical extension number (for example, 5381111) is done by the Communication Manager

Feature Server (CM-FS) or Communication Manager Evolution Server (Communication Manager-ES) based on the caller's location. When a Centralized user makes a call in Sunny day, the call goes directly from the Centralized phone to Session Manager, and IP Office is not involved at this stage. Session Manager first sends the call to the Communication Manager-FS or Communication Manager-ES responsible for calling the Centralized user for origination-side features. That Communication Manager performs the called-number conversion from the dialed short-form to the enterprise canonical number, and sends the call back to Session Manager with the called number modified to the enterprise-canonical number. Session Manager then sequences the call to other applications, if any, and then routes the call based on the enterprise-canonical number. This conversion in Communication Manager is based on Communication Manager configuration of a per-branch location via ip-network-map and ip-network-region forms, as well as Communication Manager per-location AAR analysis to identify the short-form dialed number and per-location route pattern which modifies the called number before sending to Session Manager.

For Rainy day calls made from one Centralized user to another Centralized user, the IP Office Short Form Dialing Length has to be set.

# Security considerations

The IP Office Branch solution provides security to prevent unauthorized users from accessing the system and to prevent unwanted traffic from entering. This section provides an overview of the security for all deployments in the Branch architecture. For detailed information about Avaya Aura® System Manager security, see "Security considerations" in *Avaya Aura® System Manager Overview and Specification*.

## System security settings

Update security settings to prevent unauthorized users from changing system settings. You can edit the access privileges for each user and update security passwords. By default, security settings for a centralized, mixed, or distributed deployment connected to the Avaya Aura® infrastructure, are managed in the System Manager web interface. However, you can choose to disable centralized administration and update the settings for each IP Office using IP Office Manager.

If autonomous branches with no centralization are deployed, you must update security settings with IP Office Manager.

## Secure TLS and SRTP signalling

When the Branch solution is deployed in a centralized, mixed, or distributed environment connected to the Avaya Aura® infrastructure, you can use secure transport methods and media encryption to maintain the confidentiality of media communications. You can select the secure TLS, instead of TCP, as the transport method for SIP phones deployed as Centralized users and for many types of SIP endpoints that are deployed as IP Office users. You can also configure TLS as the SIP connection between Session Manager and IP Office.

When TLS is activated, you can enable Secure Real-time Transport Protocol (SRTP) for media encryption. SRTP is supported by IP Office with many types of endpoints deployed as IP Office users and with all phones that are deployed as Centralized users. SRTP interoperability is also supported between IP Office and other components in the solution on calls through the SM Line.

In cases where there is no SRTP compatibility between one call leg and the other, IP Office does not set up direct media for the call. Instead, IP Office relays the media and converts it as required. An example for such an SRTP call flow is described in Appendix A.

IP Office supports SRTP interoperability with the following products:

- The following endpoints deployed as IP Office users:
  - 9608, 9611, 9621 and 9641 H323 phones (also known as 96x1 H.323 phones)
  - Avaya Communicator for iOS and Avaya Communicator for Windows
  - one-X Mobile Preferred iOS and one-X Mobile Preferred Android
  - Avaya H175 Video Collaboration Station
  - 11xx/ 12xx series phones
  - B179 phone
  - Radvision XT series
  - J100 series for J129, J139, J159, J169/J179, J189 phones (Supported only as Standard SIP Phone)
- Avaya Aura® Session Border Controller
- The following endpoints deployed as Centralized users, both in Sunny day and in Rainy day:
  - 9600 series SIP phones (both 96x1 and 96x0)
  - Avaya H175 Video Collaboration Station
  - Avaya Communicator for Windows
  - Avaya one-X® Communicator
  - 11xx/ 12xx series phones
  - B179 phone
  - J100 series for J129, J139, J159, J169/J179, J189 phones (Supported only as Standard SIP Phone)
- Communication Manager and its endpoints in other sites with signaling through Session Manager that includes the following phones:
  - G450 Branch Gateway used by the central Communication Manager to handle any calls that are not shuffled.
  - Phones, such as, 9600, J100 series phones, and Avaya H175 Video Collaboration Station, which are registered to Communication Manager. IP Office exchanges media directly when Communication Manager sets up direct media or shuffles the call.
- The following other applications and products are behind Session Manager:
  - Avaya Aura® Messaging
  - Avaya Aura® Conferencing
  - Avaya Aura® Session Border Controller

Certificate configuration

Establishment of TLS connections depends on the successful exchange and verification of certificates. Certificates must be set up appropriately on the different products involved. In branch

deployments, when IP Office is managed by System Manager, each IP Office must obtain an identity certificate that is generated and signed by System Manager. Each IP Office must be added to the System Manager Certificate Authority and the Simple Certificate Enrollment Protocol (SCEP) must be configured to enable IP Office to obtain required certificates from System Manager. For more information, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

If the phones register using the TLS protocol, in order to register to IP Office in Rainy day, the System Manager CA root certificate must be included in the list of files installed on the file server. This is required because the phones must trust the System Manager CA root certificate so that they can verify the IP Office Identity Certificate that is signed by the System Manager CA.

### Ports and firewalls

If the network includes any firewalls between the branch and the rest of the enterprise, then the appropriate ports must be opened on those firewalls to enable necessary traffic to enter.

**Related links**

Port assignments on page 40

Caveats and limitations on page 22

# Chapter 4: Configuration details

## Network configuration

IP Office is a converged telephony system that combines aspects of a traditional PABX telephone system and IP data and telephony systems. The Branch solution supports VoIP that includes IP trunks, SIP 9600 series telephones for centralized users, as well as H.323 and SIP IP telephones for local IP Office users. In a Distributed, Mixed, or Centralized environment connected to the Avaya Aura® infrastructure, the VoIP connection is established between the IP Office branch and the enterprise center over WAN. VoIP signaling is based on SIP signaling between the IP Office in the branch and the Avaya Aura® Session Manager in the enterprise center.

Regardless of the deployment architecture, you must consider the following network configuration factor:

- A network assessment is a mandatory requirement for all systems using VoIP. For support issues with VoIP, Avaya may request access to the network assessment results and may refuse support if those are not available or satisfactory.

If you need to update the default timer settings for the solution, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

### IP Office SIP connection to Avaya Aura® Session Manager

When connecting the IP Office in the branch to Avaya Aura® Session Manager, you must configure the SIP connection on both systems as follows:

- On the IP Office, the connection to Session Manager is configured as an SM Line. This is a special type of SIP trunk on the IP Office.
- On the Session Manager, the connection to the IP Office is configured as a SIP Entity Link.

SM Lines provide the following additional capabilities in the Branch solution:

- Ability to monitor the SIP connection to Avaya Aura® Session Manager and determine whether the connection is up or down.
- Ability to configure dial plans and prefixes for the branch.

You must obtain SM Trunk Channel licenses for the IP Office to make or receive calls over the SM Line.

IP Office supports two Sessions Managers, a primary Session Manager and a secondary Session Manager, which are each connected through an SM Line. To route calls, you must configure the following routing rules on IP Office and on Avaya Aura® Session Manager:

- On the IP Office, you must configure appropriate IP Office Short Codes and optionally IP Office ARS entries to route calls to the rest of the enterprise through the SM Line. This is not needed to route calls to Centralized users in the branch. IP Office automatically routes these types of calls within the branch in sunny-day mode through the SM Line.
- On the Session Manager, you must configure appropriate Routing Policies and Dial Patterns to route calls to the IP Office in each branch.

Each IP Office system has 2 LAN interfaces, labelled as LAN (LAN 1) and WAN (LAN 2). The SM Lines, as well as connections to IP phones, should be configured to use LAN 1. LAN 2 is primarily intended to connect to the Internet or to public SIP trunks from carriers. You must enable SIP trunks on the LAN 1 configuration in order for the SM Line to use that LAN interface. For more information about configuring SM Lines and other SIP connections, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

**Related links**

SM trunk channel licenses on page 29
Dial plan considerations on page 31

## Fax services through the SM line

You can deploy fax machines as IP Office analog endpoints. The Branch solution supports fax machines deployed as centralized ATA endpoints.

In the IP Office branch, you can send and receive faxes from the enterprise headquarters or other enterprise sites. Faxes are sent over the SM line using the T38 protocol and consume WAN bandwidth. The settings for T38 are set on the T38 Fax tab.

# JRE requirement for client computers

When launching IP Office Manager, client computers need Java Runtime Environment (JRE). JRE is required to open IP Office Manager through the Java Applet.

As an System Manager administrator, you must install JRE 1.7+ on your client machine to manage IP Office users, system configuration, and security configuration.

If JRE 1.7+ is not installed, the system displays the following message:

```
Failed to launch IP Office Manager.

IP Office Manager requires Java Runtime Environment to launch, System
has detected that there is no Java Runtime Environment present or
version present is below recommended Java Runtime Environment version
1.7+. Download and install latest Java Runtime Environment version for
Windows operating system from the Oracle site http://www.oracle.com/technetwork/
java/javase/downloads/index.html.
```

You can download the latest version of JRE from http://www.oracle.com/technetwork/java/javase/downloads/index.html.

**Note:**

- Upgrade JRE to JRE 1.7.0_51+ and upgrade JDK plugin in the browser to JDK 7.0.510+. Because JRE 1.7 introduced security settings changes, you must clear the browser cache and temporary internet files of Java from Java Control Panel. To delete the cache of applications and applets, when you delete the temporary internet files from Java Control Panel, click **Installed Applications and Applets**.
- The Java version on the client computer must correspond to the supported browser type. For example, a 32-bit browser requires a 32-bit Java version, and a 64-bit requires a 64-bit Java version.

# Video configuration

You can connect video endpoints in the Branch to IP Office or to the centralized Avaya Aura® infrastructure.

**Note:**

The supported video endpoints in IP Office Branch deployment are Avaya Scopia® and Avaya Workplace Client.

### Deployment option 1: Connect video endpoints to IP Office

In this type of deployment, the signaling call flow for video calls goes through the IP Office. The RTP media of the video stream does not go through the IP Office if direct media is appropriately set up on the IP Office. If the WAN connection between the branch and the center is lost, the video endpoints are still connected to the IP Office. Video calls to other sites will not be possible if the WAN is disconnected, but local video calls within the branch or audio-only calls to the PSTN through the IP Office can still be made. For more information about connecting the video endpoints to IP Office, see *IP Office: Avaya Scopia® Installation Notes*.

**Note:**

You must set up video to use Direct Media on IP Office.

IP Office video endpoints registered to IP Office in the branch can have video calls with the headquarters or other sites of the enterprise. You can use these endpoints to make and receive video calls to and from any of the following:

- A video multipoint control unit (MCU) located centrally such as in the headquarters, which is connected via the central Avaya Aura® Session Manager.
- A video endpoint registered to the central Session Manager.
- A video endpoint registered to an IP Office in another branch, which is connected via the central Session Manager

### Deployment option 2: Connect video endpoints to the Avaya Aura® infrastructure

In a typical branch deployment using this option, the video endpoint is deployed within the branch and connected through the WAN to the central Avaya Aura® infrastructure. In this deployment option, IP Office is not involved in either the signaling or the media flow between the video endpoint in the branch and endpoints in other sites. If the WAN connection between the branch and the center is lost, the video endpoints do not receive service. Failover of these endpoints to IP Office is not supported.

For more information about connecting video endpoints to the Avaya Aura® infrastructure, see *Avaya Aura® Communication Manager and Avaya Scopia® Interoperability Day 180 Solution Quick Setup*.

**Related links**

Network impact on video on page 43

# Packaging and order codes

Many solutions, applications, and features on IP Office are licensed and only operate when a valid license is acquired. IP Office can obtain licenses through one of the following sources:

- Product Licensing and Delivery System (PLDS) Nodal license files
- Interactions with a central Avaya WebLM server

**Note:**

ADI licenses are not supported on IP Office.

The remote PLDS license on WebLM is the recommended method for an IP Office deployed as an enterprise branch. PLDS licenses for IP Office are available for all supported features. For more information about IP Office licenses, see Avaya IP Office™ Platform Solution Description.

The following table describes key IP Office licenses that are of particular interest to branch deployments.

| Display name | Description |
|---|---|
| Avaya WebLM Model | This license is required by IP Office in branch deployments in order to connect to the WebLM server and use WebLM centralized licensing. |
| Branch System | This license is required for IP Office in branch deployments. It is used for tracking purpose only. |
| IP Office Centralized Endpoints license | One license is required for each Centralized user configured on the IP Office with a SIP extension or an analog extension.<br>**Important:**<br>In addition to this IP Office license, each centralized user also requires a license on the central Avaya Aura® system. |
| SM Trunk Channel | This license is required with sufficient quantity to allow for the maximum number of concurrent SM Line calls to be enabled on the system. The total number of SM Trunk Channel licenses requirements depend on the expected traffic patterns in the particular deployment. |

**Related links**

SM trunk channel licenses on page 29

# Port assignments

The IP Office Branch solution uses protocol interactions between the IP Office in the branch and solution components in the Avaya Aura® core as well as in other parts of the enterprise network. If the network includes any firewalls between the branch and the rest of the enterprise, then the appropriate ports must be opened on those firewalls to enable necessary traffic to enter.

If the deployment includes centralized users, you must also open the appropriate ports on the firewall to enable interactions between centralized SIP phones and other sites.

The following image shows the main ports used in the IP Office Branch solution.

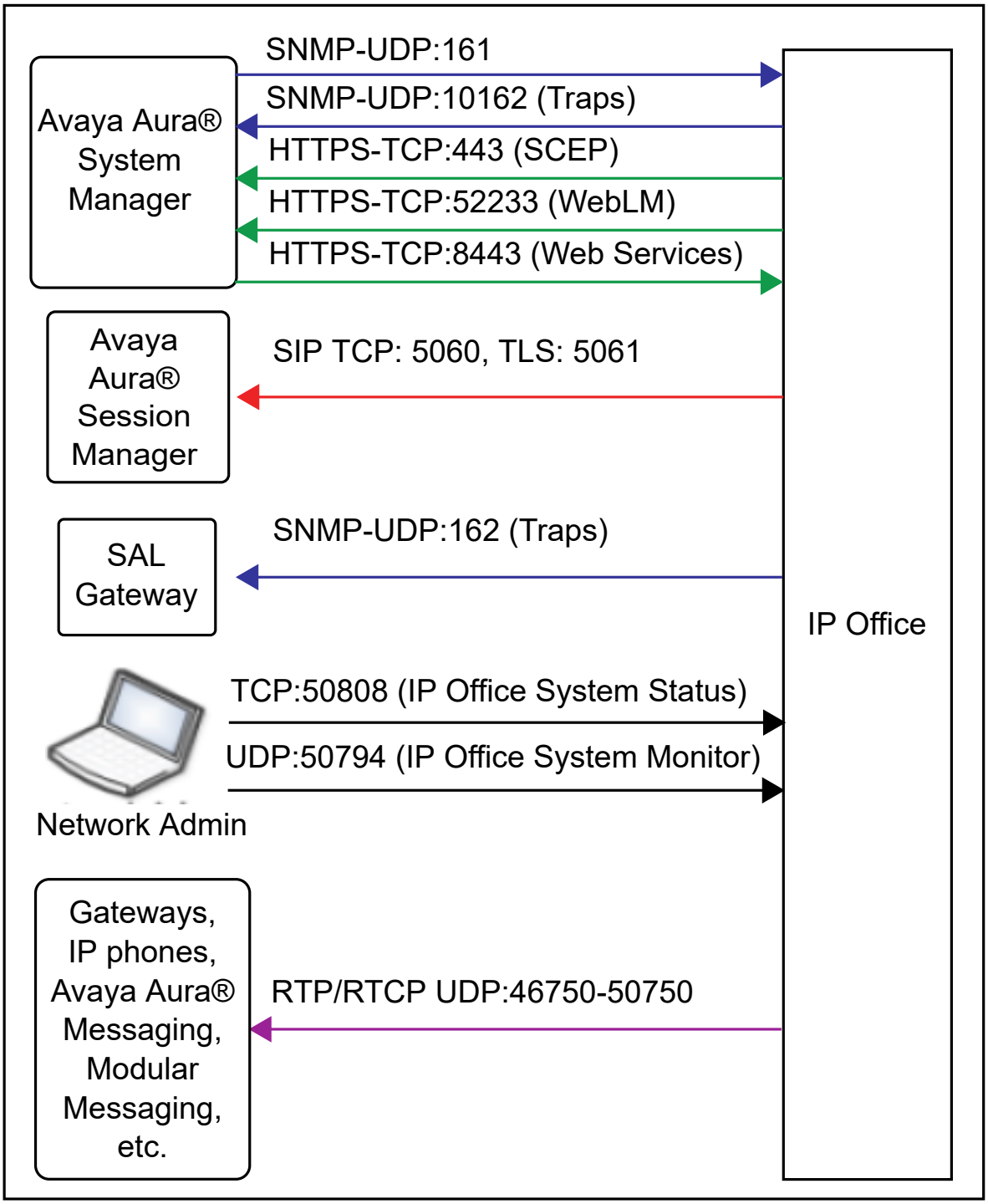


**Figure 2: Main ports in IP Office Branch solution**

For a detailed list of IP Office and SIP phone ports, see the following part matrix documents at http://support.avaya.com/security:

- Avaya one-X® Deskphone 96x1 SIP Release 6.2 Port Matrix

- IP Office Port Matrix

# Traffic and quality of service configuration

## Quality of Service requirements

To achieve good voice quality, the enterprise network must meet certain requirements. The terms used to describe acceptable voice quality are toll quality and business communication quality. Optimal voice quality is toll quality, but business communication quality is well suited for most enterprises. Business communication quality is not as high as toll quality, but is still much better than cell phone quality.

The following table summarizes the network delay, jitter, and packet loss requirements that the network must meet. Even if these requirements are met, other factors might still prevent you from achieving optimal voice quality.

**Table 6: Quality of Service (QoS) requirements**

| Requirements | Description |
|---|---|
| Network delay | Voice quality:<br>• To obtain toll quality, the delay cannot exceed 80 millseconds (ms).<br>• To obtain business communication quality, the delay must be between 80 to 180 ms. Business communication quality is suitable for most enterprises.<br>• Delays exceeding 180 ms provide a lower quality than business communication quality, but this might still be acceptable for some enterprises. |
| Network jitter | To achieve optimal voice quality, the average jitter must be less than half the network packet payload. This value can vary depending on the type of service the jitter buffer has in relation to other buffers and to the packet size used.<br>Assuming the packet size is 20 ms, to prevent problems with voice quality, the network jitter must not exceed 20 ms. |
| Network packet loss | Voice quality:<br>• To obtain toll quality, the packet loss cannot exceed 1%.<br>• To obtain business communication quality, the packet loss cannot exceed 3%.<br>• Packet losses exceeding 3% might result in signalling interferences. |

When transporting voice over low speed links, normal data packets can prevent or delay voice packets from getting across the link. The voice transportation can result in unacceptable speech

quality. To ensure low speech latency and help maintain sufficient voice quality, implement another Quality of Service (QoS) mechanism, such as a QoS router, on the traffic routers and switches in your network. QoS mechanisms must be implemented on the WAN routers that connect the branch with the rest of the enterprise network.

QoS must be based on DSCP tagging. Use the DSCPs to determine the appropriate priority queuing and handling to implement on the network routers. DSCP tagging should be then be configured consistently on the IP Office and on the Avaya Aura® Communication Manager for the telephones connected to the Avaya Aura® system.

## Network impact on video

Multiple network parameters affect the quality of video, such as the bit rate, the frame rate, the packet loss ratio, the content type, the compression method, jitter, and intra-frame coding. Each of these parameters might impact the quality of experience of a user.

### Recommended specifications

The following table lists the recommended values for the network parameters that impact the video quality:

| Network parameter | Recommended value |
|---|---|
| One-way network delay | <= 80 ms to 180 ms |
| Network packet loss for video | <= 0.2 % |
| Latency | < 300 ms |
| Jitter | <= 20 ms |
| Frame rate | >= 26 fps |
| MOS | 4 to 5 |

# Bandwidth requirements

For local branch calls involving Centralized users, the signaling path goes over the WAN to the Avaya Aura® Session Manager and back to the branch. These types of local branch call scenarios include:

- Calls between two Centralized users in the same branch
- Calls between a Centralized user and an IP Office user in the same branch
- Calls between a Centralized user and the PSTN via a PSTN trunk on the local IP Office in the branch

While the signaling path goes over the WAN, the media path for such calls remains within the branch and does not go back and forth across the WAN. This depends on the central Avaya Aura® Communication Manager being appropriately configured for direct IP-IP audio.

- If Avaya Aura® Communication Manager is configured for Initial Direct Media, then no media will be sent over the WAN for the call.
- If Avaya Aura® Communication Manager is not configured for Initial Direct Media, but only for direct IP-IP audio or shuffling, then the first few media packets of each local call are relayed across the WAN through a Communication Manager central media resource, such as the

G450 Media Gateway in the enterprise center. This occurs until Avaya Aura® Communication Manager shuffles the call so the media goes directly within the branch.

### Exceptions for local calls

Exception cases where the media streams of local calls consume WAN bandwidth are as follows:

- If the Centralized user initiates a conference on Avaya Aura® Communication Manager, then the different media streams are pulled by Communication Manager across the WAN to the central G4xx gateway. There is one media stream between the central G4xx gateway (where the Communication Manager conference is hosted) and each of the conference participants.

  For example, in a three-way sunny-day Communication Manager conference that includes two Centralized users in the branch and a PSTN caller through a local PSTN trunk on the IP Office in the branch, there will be three media streams across the WAN between the central G4xx gateway and the branch.
- If the Centralized user puts the call on hold, then Avaya Aura® Communication Manager plays music-on-hold from the central G4xx gateway across the WAN to the branch.

### Other call scenarios that consume WAN bandwidth

Calls between IP Office users and users in other sites of the enterprise are routed either through the Avaya Aura® Session Manager or through the PSTN, depending on how the solution is configured. If configured to route the calls across the SIP connection between the IP Office and the Avaya Aura® Session Manager, then both the signaling and the call media go across the WAN.

For site-to-site calls between Centralized SIP users in sunny-day, the signaling goes through the Avaya Aura® Session Manager and the media goes between the endpoints across the WAN. The IP Office in the branch is not involved in such calls. The Communication Manager Inter-Gateway Alternate Routing (IGAR) feature is not supported with IP Office branch deployments and the media flow of such site-to-site calls between Centralized SIP users cannot be sent through the PSTN instead of the WAN.

## Codec bandwidth usage

By default, all IP Office IP trunks and extensions use automatic codec negotiation. The default negotiation order varies depending on the type of SD card configured.

- With an IPO MU_LAW SD card, the default negotiation order is G711.MU_LAW, G711 A_LAW, G729a, and G723.1.
- With an IPO A_LAW SD card, the default negotiation order is G711.A_LAW, G711.MU_LAW, G729a, and G723.1.

Calls to and from IP devices can require conversion to the audio codec format being used by the IP device. For IP Office systems, this conversion is done by voice compression channels. The voice compression channels support the common IP audio codecs G711, G723 and G729a. For more information about configuring codecs and voice compression channels, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

The following table lists the default codecs available and the other options you can select for each codec.

| Setting | Selected Preference | 2nd Preference | 3rd Preference | 4th Preference |
|---|---|---|---|---|
| **G.729** | G729(a) 8K CS-ACELP | G711 U-Law 64K | G711 A-Law 64K | G723.1 6K3 MP-MLQ |
| **G.723** | G723.1 6K3 MP-MLQ | G729(a) 8K CS-ACELP | G711 U-Law 64K | G711 A-Law 64K |
| **G.711 U-Law** | G711 U-Law 64K | G711 A-Law 64K | G729(a) 8K CS-ACELP | G723.1 6K3 MP-MLQ |
| **G.711 A-Law** | G711 A-Law 64K | G711 U-Law 64K | G729(a) 8K CS-ACELP | G723.1 6K3 MP-MLQ |

Some calls and media flows do not consume WAN, but others do. For calls that consume WAN bandwidth, codec selection is an important factor in determining the amount of bandwidth consumed and the audio quality of the call. The G729 codec consumes less bandwidth but can result in reduced audio quality. Depending on the needs of the enterprise, the audio quality with the G729 codec may not be acceptable. The G711 codec provides optimal audio quality, but consumes more bandwidth. The amount of bandwidth consumed by different codecs is the same for Avaya Aura® Communication Manager and IP Office.

### Typical bandwidth usage

The following table can be used to help assess how much bandwidth (in Kbits/sec) is used for various types of codecs and packet sizes. The values shown have a 7–byte L2 WAN header and are rounded up.

**Table 7: Bandwidths used for codecs and packet sizes**

| Packet Size | 10 ms | 20 ms | 30 ms | 40 ms | 50 ms | 20 ms |
|---|---|---|---|---|---|---|
| G.711 | 102 | 83 | 77 | 74 | 72 | 71 |
| G.729 | 46 | 27 | 21 | 18 | 16 | 15 |
| G.723-6.3 | NA | NA | 19 | NA | NA | 13 |
| G.723-5.3 | NA | NA | 18 | NA | NA | 12 |
| G.722.2 | NA | 43 | NA | 34 | NA | 31 |

**Related links**

VCM channels on page 26
Bandwidth management on page 46

## Direct media setting in Avaya Aura® Communication Manager

To prevent media flow from unnecessarily crossing the WAN to a central Communication Manager media resource, you must enable the Initial IP-IP Direct Media parameter in Avaya Aura® Communication Manager. This must be done in all IP Office branch deployments with Centralized users. Enabling this parameter is especially important for the following types of calls:

- Calls between Centralized users within the branch
- Calls between Centralized users and local IP Office trunks

## WAN bandwidth usage for signaling traffic

Determining the WAN bandwidth needed for signaling traffic requires network engineering calculations based on the user's expected call patterns. Signaling bandwidth is impacted by the average frequency of calls, while media bandwidth is also impacted by the average length of calls.

A call between a Centralized SIP user and a local IP Office PSTN trunk in the same branch typically generates about 50 kilo Bytes of signaling traffic on the WAN link between the branch and the Avaya Aura® core. This number must be multiplied by the expected number of calls to calculate the estimated bandwidth required for signaling in kilo bits per second.

**Calculation example**

If the average number of PSTN calls made or received by all the Centralized SIP users in the branch is 150 calls an hour, then 7,500 kilo Bytes of bandwidth is consumed per hour. On average, if spread evenly, this would mean that signaling for these calls will consume 17 kilo bits per second of the WAN bandwidth.

# Bandwidth management

In IP Office Branch solution, Avaya Aura® Session Manager provides bandwidth management through Call Admission Control (CAC). This service is available is available in Distributed, Mixed, and Centralized deployment options connected to the Avaya Aura® network.

You must configure a location for each branch on Avaya Aura® Session Manager, and make sure that Session Manager detects that the IP Office and the centralized users in the branch are part of the same location. Make sure the IP address pattern configured in the **Location Pattern** section of the Session Manager location represents the IP addresses for the branch. You must select the name of the location in the **Location** field of the SIP Entity configured in Avaya Aura® System Manager for the IP Office in the branch.

You can configure a bandwidth limit for a branch location in Avaya Aura® Session Manager. The bandwidth limit should be configured according to:

- The network bandwidth available for media traffic on the WAN link that connects the branch to the rest of the enterprise network
- Expected traffic patterns to and from the branch

The Session Manager CAC feature inspects the SDP in call signaling to count the total bandwidth of calls. When the bandwidth limit is reached the Session Manager CAC rejects all new calls to and from the IP Office branch location.

The Session Manager CAC is aware that centralized users and IP Office are in the same branch, and can calculate the following call details to determine the bandwidth of calls:

- The number of calls to and from the IP Office
- The number of calls to and from centralized users in the branch

The CAC can determine the types of calls that consume WAN bandwidth. For example, a call between an IP Office user or trunk and a centralized user in the same branch does not consume WAN bandwidth because the media flow is local within the branch. The CAC does not count this type of call with local media flow toward the configured WAN bandwidth limit for the branch.

## Session Manager CAC specifications

Audio and multimedia calls require high bandwidth and low latency for best user experience. Call Admission Control (CAC), also known as Bandwidth Management, provides an efficient means to prevent degradation of quality by limiting the number of concurrent calls over limited bandwidth links. Based on location level settings, Avaya Aura® Session Manager provides the following set of CAC functionality using Session Description Protocol (SDP):

1. Recognizes different types of calls categorized either as audio or multimedia. Multimedia includes video and other forms of non-audio media sent by endpoints as part of a session.
2. Provisions a second, lesser limit on the bandwidth usage permitted for a location. This second limit applies only to multimedia calls (**Multimedia Bandwidth**) and prevents such calls from consuming too large a percentage of the available bandwidth of a location (**Total Bandwidth**).
3. If the **Total Bandwidth** is left “BLANK” then Session Manager does not perform CAC for calls in the Location.
4. Allows audio and multimedia bandwidth sharing at a location by selecting the option **Audio Calls Can Take Video Bandwidth**.
   - If selected, then Session Manager considers only the **Total Bandwidth** when deciding whether a new audio call can proceed.
   - If not selected, then Session Manager considers the **Total Bandwidth** minus the **Multimedia Bandwidth** when deciding whether a new audio call can proceed.
5. Specifies per-call bandwidth limits (as specified in **Per-Call Bandwidth Parameters** section), restricting the size of individual multimedia calls. Session Manager alters the SDP provided by call parties by enforcing the bandwidth limits as follows:
   - Determines how much bandwidth to be reserved for each call and counts the determined value against the provisioned limit.
   - If the multimedia bandwidth is beyond what is provisioned (**Maximum Multimedia Bandwidth (Intra-Location)** or **Maximum Multimedia Bandwidth (Inter-Location)**), Session Manager can reduce the multimedia bandwidth as low as the administered multimedia minimum (**Minimum Multimedia Bandwidth**) limit. In such cases, users experience a reduction in media (usually video) quality. Otherwise, calls are either alternate-routed or denied when limit enforcement cannot be achieved by quality reduction. Audio call quality is not modified by Session Manager.
   - If Session Manager cannot allow the multimedia minimum (**Minimum Multimedia Bandwidth**), then multimedia streams are removed from the call by setting their ports to zero, which results in denying the multimedia portion of the call

For information about administering CAC, see *Administering Avaya Aura® Session Manager*.

# Redundancy and high availability configuration

Redundancy and high availability are supported in the Avaya Aura® network. Centralized users in the architecture can also access the Rainy day mode when all available Avaya Aura® servers are down.

## Redundancy and high availability from the Avaya Aura® network

The following redundancy and high availability options are available through the Avaya Aura® network.

**High availability:**

The Avaya Aura® network uses System Platform high availability. When deploying Avaya Aura® System Manager, you must also install and configure System Platform. For more information about System Platform high availability, see “Installing System Platform” in *Implementing Avaya Aura® System Manager*.

**Session Manager geographic redundancy:**

The IP Office Branch solution supports Avaya Aura® Session Manager geographic redundancy when a back-up Session Manager is deployed in the secondary data center. If for any reason the IP Office system's primary Session Manager goes out of service, the system will automatically attempt to use the secondary Session Manager. Prioritization of the Session Manager is determined by the line number configured for a particular Session Manager. For example, if the first Session Manager is configured with line number 17 and the second Session Manager is configured with line number 18, then line number 17 has the higher priority and is considered the primary Session Manager. If for some reason you want to designate the secondary Session Manager as the primary line, you must change one or both of the line numbers associated with the Session Manager so that the secondary Session Manager number is lower than the primary line number.

**System Manager geographic redundancy:**

In deployments with Avaya Aura® System Manager geographic redundancy set up, the IP Office devices added in the primary System Manager are also visible in the secondary System Manager. To use Avaya Aura® System Manager geographic redundancy in the IP Office Branch solution, make sure:

- The initial interactions through the SCEP protocol are done between the IP Office and the primary System Manager to establish trust.
- The IP address for the primary and secondary System Manager are configured as trap destinations in the IP Office. This way, IP Office sends alarms and traps to both System Manager servers. When the primary System Manager server is down, the secondary System Manager server still receives this information.

If the IP Office is using Avaya WebLM centralized licensing and the connection between the IP Office and the Avaya WebLM server on the primary System Manager is down, then the IP Office goes into WebLM Error Mode. In this mode, all functionality is available for a grace period of 30 days. No action is required if the Avaya WebLM server on the primary System Manager is active

and reachable in less than 30 days. If there is a risk of the System Manager server not being available for 30 days, you must modify the IP Office configuration to use the Avaya WebLM server on the secondary System Manager as soon as possible.

# Redundancy through Rainy day support

When all available Avaya Aura® servers are down, Centralized users can still obtain telephony services through IP Office with Rainy day. This section describes the features of endpoints when registered to IP Office in Rainy day mode.

## 9600 Series SIP phone features available in Rainy day mode

The following features are available on Centralized SIP phones when registered to IP Office in Rainy day mode:

- Make or receive calls to or from other endpoints in the branch and to or from any type of local PSTN trunk
- Caller ID
- Multiple call appearances but not bridged appearances
- Call hold and consultative hold
- Music on hold
- Attended call transfer
- Unattended call transfer
- Three-party ad-hoc conferencing done locally on the phone, as well as capability to dial into Meet-Me conferencing on IP Office up to 64-party conference
- Centralized voice mail coverage and access over PSTN, but no Message Waiting Indication (MWI)
- Automated Attendant
- Survivability mode indication on the phone screen
- Local telephone features: redial, mute, audio selection (speaker / headset / handset), Call Logs, Volume Control, local contacts, speed-dials, auto dials
- Station Message Detail Recording (SMDR) records stored on the IP Office for retrieval after WAN recovery
- Hunt groups

  IP Office can be configured with Centralized hunt groups for which IP Office processing is in effect only in the Rainy day mode. The IP Office administrator must configure the hunt groups on the IP Office consistent with the configuration on the central Avaya Aura® Communication Manager for the Sunny day mode.
- Call Management

  IP Office can be configured with short codes using the Barred feature to restrict in the Rainy day mode what calls the Centralized user can make. The IP Office administrator must

configure this consistent with the Class of Restriction (CoR) configured on Communication Manager, which is applied to the same user in the Sunny day mode.

- Send call to mobile phone

  IP Office can be configured with Mobile Twinning to send calls for the Centralized user in the Rainy day mode to a mobile number. The IP Office administrator must configure this on the IP Office consistent with the EC500 configuration on the central Communication Manager for the same Centralized user.

- Call forwarding

  Local Call Forwarding on the phone in the Rainy day mode can be configured. The Call Forwarding set on Communication Manager in the Sunny day mode has no impact on the local behavior of the phone or on the IP Office behavior in the Rainy day mode. Also, the local Call Forwarding set on the phone works only in the Rainy day mode after failback.

- Authorization codes

  IP Officecan be configured to support authorization codes that Centralized users can use in the Rainy day mode. The IP Office administrator must configure authorization codes consistent with the authorization codes configured on Communication Manager, which are available to the same Centralized users in the Sunny day mode. Centralized SIP phone users in Sunny day will hear 3 beeps to indicate that an authorization code is required. In the Rainy day mode, the Centralized SIP phone users will hear 1 beep that repeats approximately every 5 seconds.

**Note:**

The 9600 Series SIP phones cannot dial the **#** symbol in the Rainy day mode. In the Rainy day mode, all features that require the **#** symbol to be dialed must be redefined.

## Avaya H175 Video Collaboration Station features available in Rainy day mode

The Avaya H175 Video Collaboration Station deployed as a Centralized user provides basic audio telephony functionality when registered to IP Office in Rainy day mode.

**Related links**

Avaya H175 Video Collaboration Station endpoints deployed as Centralized users on page 24

## ATA endpoint features in Rainy Day

The following features are available for ATA endpoints when registered to IP Office in Rainy Day mode.

- Make or receive calls to or from other endpoints in the branch and to or from any type of local PSTN trunk
- Call hold and consultative hold
- Music on hold
- Attended call transfer

- Unattended call transfer
- ATA user can dial into Meet-Me conferencing on IP Office (up to 64–party conference)
- Centralized voice mail coverage and access over PSTN, but no Message Waiting Indication (MWI)

## 1100 and 1200 series phone features in Rainy day mode

The following features are available for 1100 and 1200 series SIP phones with IP Office in Rainy day mode:

- Make or receive calls to or from other endpoints in the branch and to or from any type of local PSTN trunk
- TLS and SRTP without direct media on 1100 and 1200 series phones
- Call forwarding on the phone set instead of on the server

  **Note:**

  Call forwarding is not available in Rainy Day for busy and unanswered calls.
- Speed dial through address book
- Caller ID
- Receive multiple simultaneous calls
- Call hold and retrieve
- Consultative hold
- Music on hold
- Do not disturb status for 1100 and 1200 series phones
- Attended and unattended call transfer
- Ad-hoc conference
- Meet-me conference for 1100 and 1200 series phones
- Centralized voice mail coverage for 1100 and 1200 series phones
- Auto-attendant support
- Hunt group support
- SMDR support
- Call restriction support
- Survivability mode indication on phone set for 1100 and 1200 series phones
- Last number redial
- Mute and unmute
- Incoming and outgoing call logs
- Local address book

- Feature access codes or short codes for 1100 and 1200 series phones
- Centralized voice mail coverage and access over PSTN, but no Message Waiting Indication (MWI)

# Failover for 9600 SIP, J100 series, and Avaya H175 Video Collaboration Station endpoints

When 9600 SIP, J100 series, and Avaya H175 Video Collaboration Station endpoints lose connectivity, they can failover to another SIP controller. While the endpoints are failing over, they display ⚠. The failover process is usually completed within two minutes.

During failover:

- No new calls can be made or received.
- Held calls are lost.
- Transfers are lost.

The following table lists key failover scenarios and the types of messages that you might see on each endpoint.

**Table 8: Failover messages**

| Scenario | Message on 9600 series SIP endpoints | Message on Avaya H175 Video Collaboration Station endpoints | Message on J100 series phones |
|---|---|---|---|
| Extension is idle | The endpoint displays the message `Link recovery. Limited phone service. Calls may be lost.` | The endpoint displays the message `Lost Connection to SIP Server...` | The J100 series phone displays `Acquiring service` during failover when extension is idle. |
| A call is in progress | Both endpoints display ⚠ with the message `Limited phone service`. The only call control feature available is End Call. The phone fails over after the call ends. | | The J100 series phone will display Limited phone service with Link Recovery. Call may be lost. Users can end call. When call ends, phone will go to Acquiring service and failover. |
| The endpoint cannot detect an available controller | The endpoint displays the message `Acquiring Service...` Failover cannot occur until an available SIP controller is detected. | The endpoint displays the message `Lost Connection to SIP Server...` Failover cannot occur until an available SIP controller is detected. | The J100 series phone displays `Acquiring service` during failover when all SIP proxy servers are detected to be unreachable. |

# Appendix A: Overview of IP Office example call flows

This section describes major IP Office call flows, but does not provide an exhaustive list of all possible call flow scenarios.

### Calls from IP Office user phones (i.e. deployed as a Distributed enterprise branch)

1. IP Office user phone to IP Office user phone

   IP Office user phone A > IP Office > IP Office user phone B

2. IP Office user phone to PSTN (using local trunking)

   IP Office user phone > IP Office > IP Office trunk > PSTN

3. IP Office user phone to PSTN (using centralized trunking)

   IP Office user phone > IP Office > Session Manager > central gateway/SBC > PSTN

4. IP Office user phone to headquarters or other enterprise site

   IP Office user phone > IP Office > Session Manager > target phone's controller > target phone

### Calls to IP Office user phones (i.e. deployed as a Distributed enterprise branch)

1. From PSTN to IP Office user phone via IP Office auto-attendant (branch auto-attendant LDN is associated with local IP Office trunk)

   PSTN > IP Office trunk > IP Office auto-attendant > caller enters extension number > IP Office user phone

2. From PSTN direct to IP Office user phone's DID (LDN is associated with local IP Office trunk)

   PSTN > IP Office trunk > IP Office > IP Office user phone

3. From headquarters (or other enterprise site) to IP Office user phone's enterprise number

   Originating phone > originating phone's controller > Session Manager > IP Office > IP Office user phone

4. From headquarters (or other enterprise site) to IP Office user phone via IP Office auto-attendant

   Originating phone > originating phone's controller > Session Manager > IP Office > IP Office auto-attendant > caller enters extension number > IP Office user phone

### Calls from Centralized phones in normal mode (aka Sunny day)

1. Centralized phone A to centralized phone B in the same branch

   Centralized phone A > Avaya Aura core servers (including Session Manager and CM-FS or CM-ES) > centralized phone B

   **Note:**

   The IP Office is not involved in handling the call in this scenario.

   The call flow descriptions in this document are intended to clarify the call flow between the branch and the core with focus on IP Office involvement. For brevity, they do not elaborate on the processing of the call within the core, among the different elements of the Avaya Aura® server infrastructure at the enterprise core. For example, a more complete depiction of the call flow for this scenario would be:

   Centralized phone A > Session Manager > origination-side processing by the CM-FS or CM-ES responsible for user of phone A > Session Manager > termination-side processing by the CM-FS or CM-ES responsible for user of phone B > Session Manager > centralized phone B

   This depiction of the call flow includes the core Avaya Aura® Communication Manager acting as the feature server for the centralized user. The Communication Manager Feature Servers (CM-FSs) or Communication Manager Evolution Servers (CM-ESs) for user A and user B may be the same CM server or two different CM servers, depending on the provisioning of the users on the core CM servers. Furthermore, additional application servers may be sequenced in the call flow by Session Manager if additional core sequenced applications are deployed by the enterprise. See the Avaya Aura® Session Manager documentation for more information regarding the operation of the CM-FS or CM-ES with Session Manager, and regarding Session Manager application sequencing.

2. Centralized phone to IP Office user phone (dialing IP Office user phone's enterprise number)

   Centralized phone > Session Manager > CM-FS or CM-ES > Session Manager > IP Office > IP Office user phone

3. Centralized phone to PSTN (using local trunking)

   Centralized phone > Session Manager > CM-FS or CM-ES > Session Manager > IP Office > IP Office trunk > PSTN

4. Centralized phone to PSTN (using centralized trunking)

   Centralized phone > Session Manager > CM-FS or CM-ES > Session Manager > central gateway/SBC > PSTN

   **Note:**

   The IP Office is not involved in handling the call in this scenario.

5. Centralized phone to headquarters or other enterprise site

   Centralized phone > Session Manager > CM-FS or CM-ES > Session Manager > target phone's controller (if target phone does not register directly to SM) > target phone

   **Note:**

   The IP Office is not involved in handling the call in this scenario.

### Calls to Centralized phones in normal mode (aka Sunny day)

1. From PSTN to centralized phone's DID (LDN is associated with local IP Office trunk)

   PSTN > IP Office trunk > IP Office > Session Manager > CM-FS or CM-ES > Session Manager > centralized phone

2. From PSTN to centralized phone via IP Office auto-attendant (branch auto-attendant LDN is associated with local IP Office trunk)

   PSTN > IP Office trunk > IP Office auto-attendant > caller enters extension number > IP Office modifies extension number to user's enterprise number > Session Manager > CM-FS or CM-ES > Session Manager > centralized phone

3. From PSTN to centralized phone's DID (if LDN is ported to centralized trunks at the core)

   PSTN > central gateway/SBC > Session Manager > CM-FS or CM-ES > Session Manager > centralized phone

   **Note:**

   The IP Office is not involved in handling the call in this scenario.

4. From branch IP Office user phone to centralized phone (dialing phone's enterprise number)

   IP Office user phone > IP Office > Session Manager > CM-FS or CM-ES > Session Manager > centralized phone

5. From headquarters (or other enterprise site) to centralized phone's enterprise number

   Originating phone > originating phone's controller > Session Manager > CM-FS or CM-ES > Session Manager > centralized phone

   **Note:**

   The IP Office is not involved in handling the call in this scenario.

### Calls from Centralized phones in survivability mode (aka Rainy day)

1. Centralized phone A to Centralized phone B in the same branch

   Centralized phone A > IP Office > centralized phone B

   **Note:**

   In Rainy day the centralized phones register to survivability service on the IP Office in the branch.

2. Centralized phone to IP Office user phone (dialing either IP Office user phone's enterprise number or short extension)

   Centralized phone > IP Office > IP Office user phone

3. Centralized phone to PSTN

   Centralized phone > IP Office > IP Office trunk > PSTN

4. Centralized phone to headquarters or other enterprise site

   IP Office can be administered to send such calls over the PSTN. See below.

### Calls to Centralized phones in survivability mode (aka Rainy day)

1. From PSTN to centralized phone's DID

   PSTN > IP Office trunk > IP Office > centralized phone

   **Note:**

   In Rainy day the centralized phones register to survivability service on IP Office in the branch.

2. From PSTN to centralized phone via IP Office auto-attendant

   PSTN > IP Office trunk > IP Office auto-attendant > caller enters extension number > centralized phone

3. From branch IP Office user phone to centralized phone (dialing either IP Office user phone's enterprise number or short extension)

   IP Office user phone > IP Office > centralized phone

**Related links**

Centralized deployment example call flows on page 56
Centralized voice mail example call flows on page 68
ATA example call flows on page 77
SRTP example call flows on page 80
Toll bypass functionality on page 84

# Centralized deployment example call flows

This section provides examples of call flows for a system that is configured in a centralized deployment.

## Routing concepts

In the example network deployment provided in this section, all calls that are routed within Avaya Aura® Session Manager are converted to E.164. The primary function for this is to support Tail-End-Hop-Off (TEHO).

### E.164

E.164 is a numbering format that is recommended by the International Telecommunications Union - Telecommunications (ITU-T). E.164 can have a maximum of 15 digits and is preceded by a +.

- Maximum of 15 digits (not including the preceding +).
- First part is a 1 to 3-digit country code.
- Second part is a national destination code.
- Last part is the subscriber number.
- Second and last parts are collectively known as the national number.

### Tail-End-Hop-Off (TEHO)

TEHO is the process of routing a call through a private network to the closest node of the destination call, and then routing over the public network as a local call.

## Call flows

The IP Office centralized solution revolves around SIP endpoints at a branch location that registers to a Session Manager. Local trunking is provided through the IP Office at the branch location. When connectivity to the Session Manager is lost, the SIP endpoints failover and register to the IP Office for connectivity.

### Solution overview

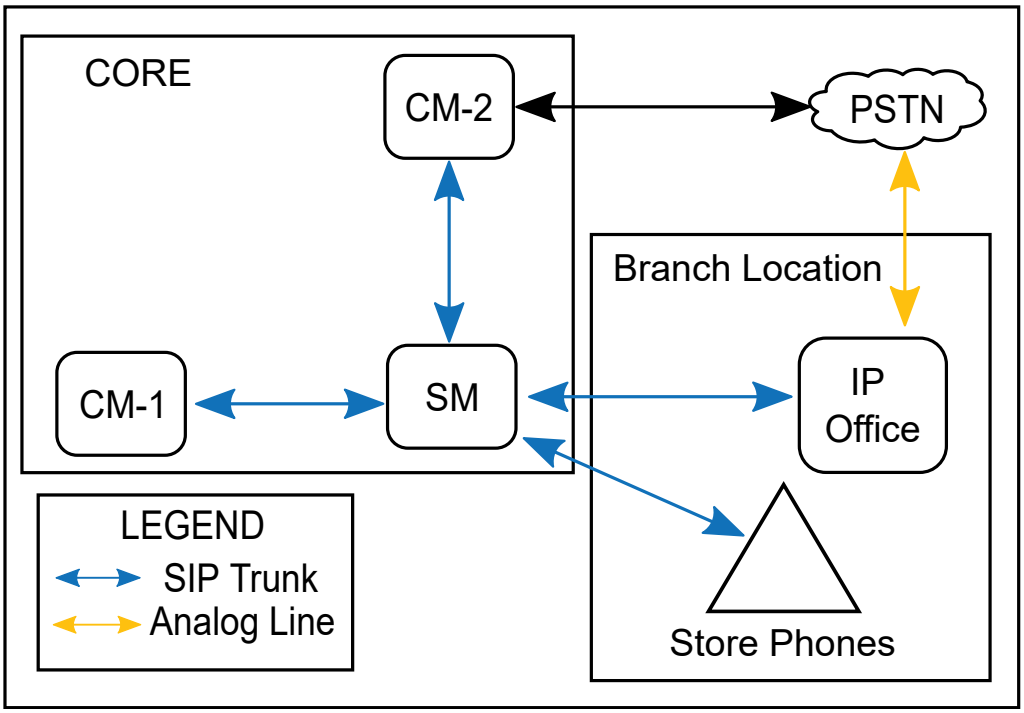


**Figure 3: Solution overview showing the store network**

The Solution overview illustrates the store network.

- **CM-1** represents a Communication Manager used for features for the store phones.
- **CM-2** represents the primary Communication Manager used for corporate endpoints and PSTN trunking.
- **SM** represents all Session Managers.

There must be at least:

- 1 or more Communication Managers
- 1 or more Session Managers
- 1 or more IP Offices

- SIP endpoints at each branch location

## Sunny day call flows

The first set of call flows are for sunny day. This is when all SIP endpoints have full connectivity across the WAN back to the core and are registered to the Session Manager. All SIP trunks are in service.

### Internal call

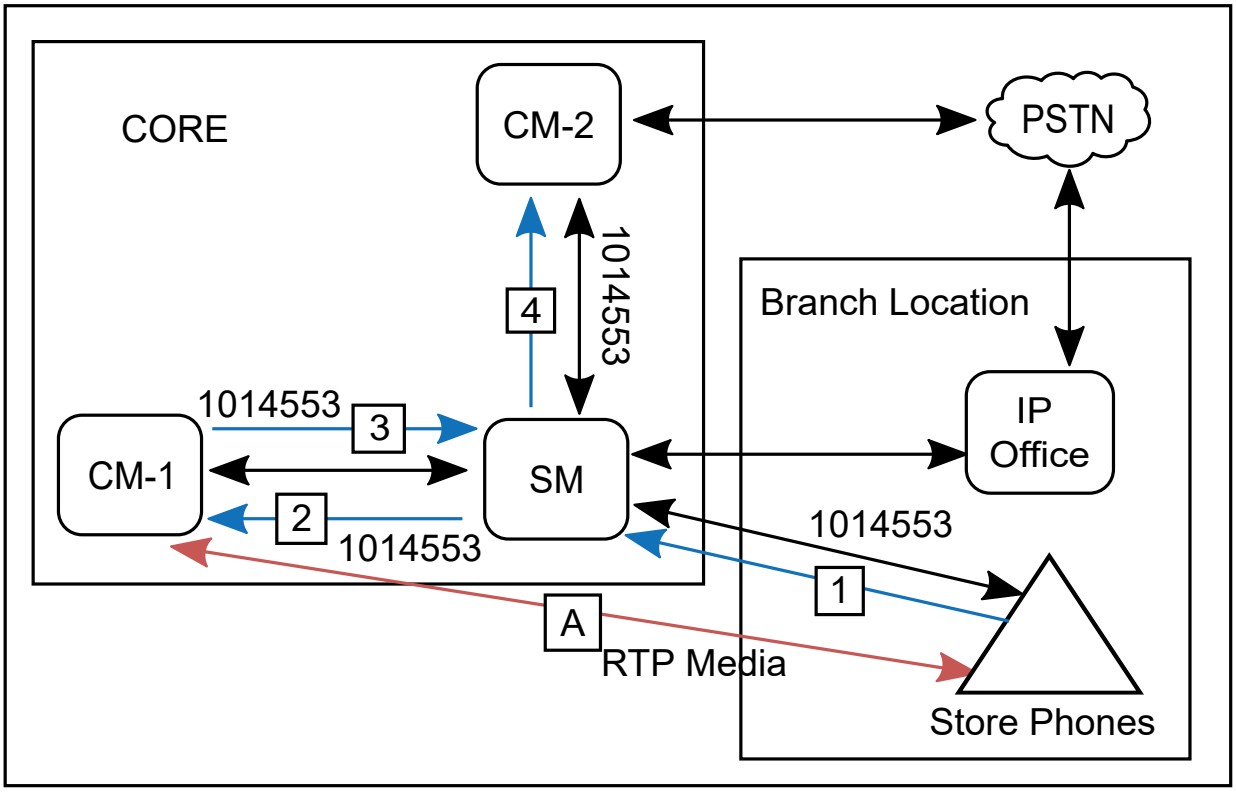


**Figure 4: Sunny day — internal call**

| | |
|---|---|
| **1** | **Store Phone** originates call to a corporate station 1014553. |
| **2** | **SM** sends 1014553 to CM for origination processing. |
| **3** | **CM** identifies call as internal and routes 1014553 back to **SM**. |
| **4** | **SM** sends call to destination phone 1014553 at destination **CM**.<br>This call flow is similar for calls within the same branch or calls to other branches with the exception that the calls terminate at the destined branch. |
| **A** | RTP Media stream is initially between the **Store Phone** and a **CM-1** media gateway. This will shuffle between the **Store Phone** and **CM-2** or another endpoint once established. |

## Local PSTN dialing

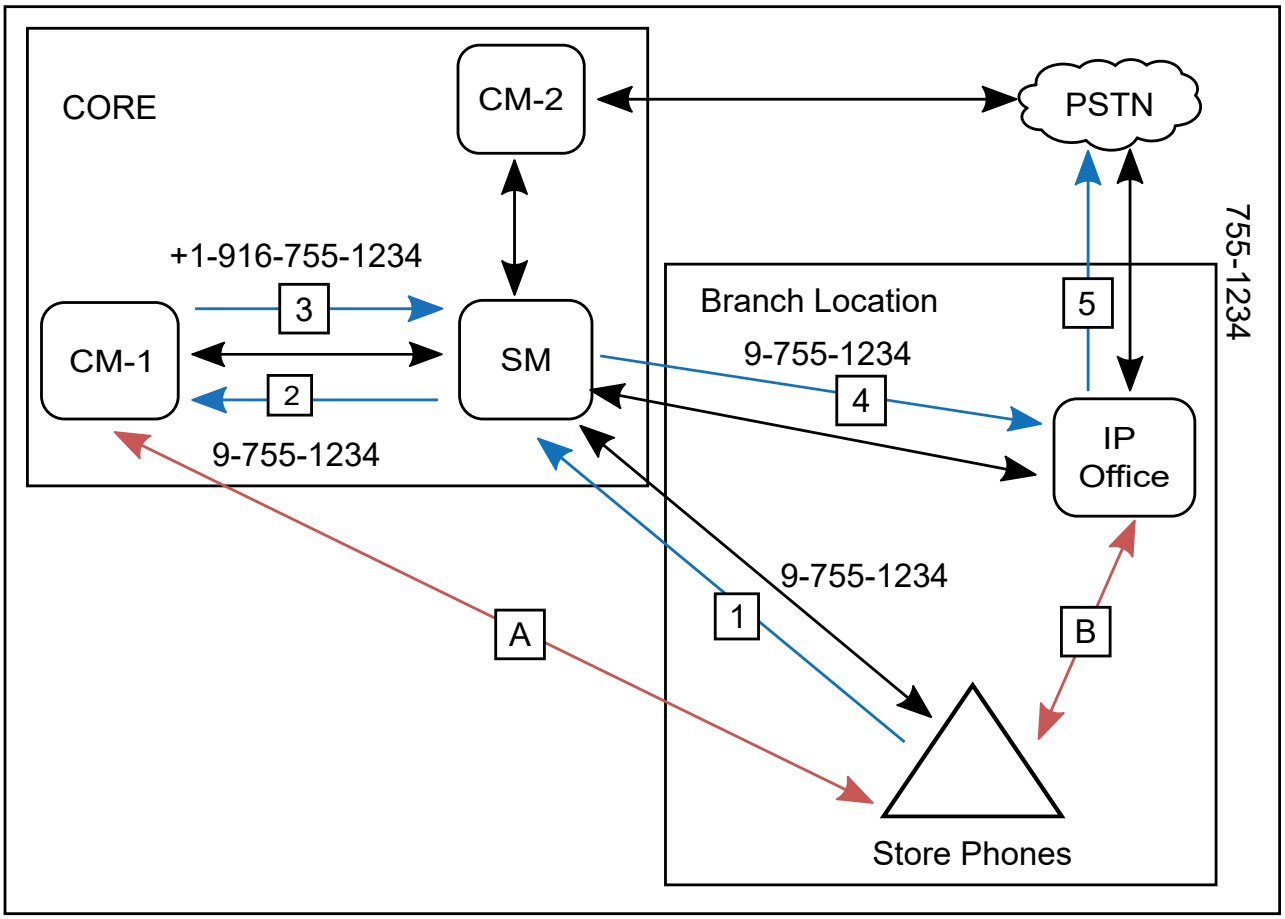


**Figure 5: Sunny day — local PSTN dialing**

| | |
|---|---|
| **1** | **Store Phone** calls local extension 9-755-1234. |
| **2** | **SM** sends 9-755-1234 to **CM** for origination call processing. |
| **3** | **CM** translates the call into E.164 based on callers' location and sends +1-916-755-1234 to **SM**. |
| **4** | **SM** adapts the call for the local carrier and sends 9-755-1234 back to original branch.<br>Session Manager could use the E.164 formatted number to do TEHO routing at this step. |
| **5** | **IP Office Branch** sends 755-1234 to the **PSTN**. |
| **A** | RTP Media stream is initially between the **Store Phone** and **CM-1** media gateway. |
| **B** | Once the call is established, the RTP Media stream will shuffle between the **Store Phone** and the **IP Office Branch**. |

### Long distance and international

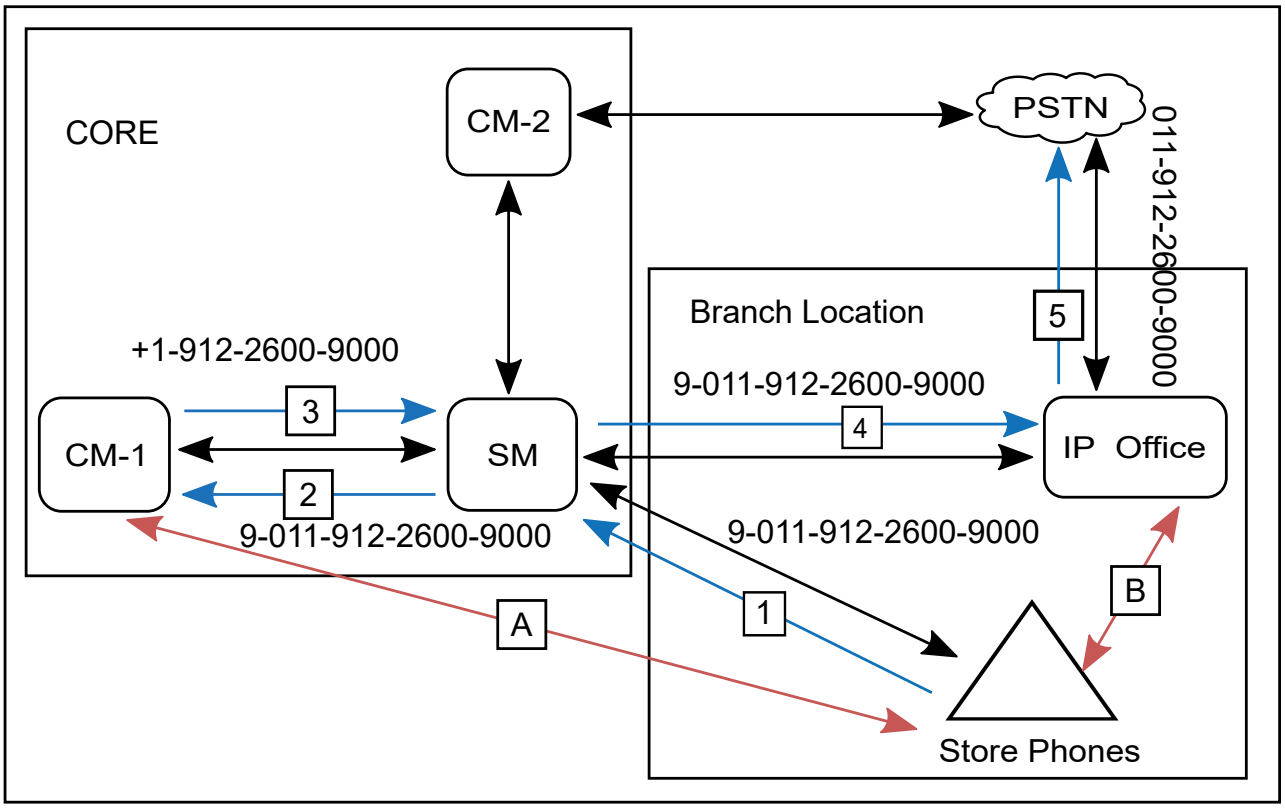


**Figure 6: Sunny day — long distance and international**

| | |
|---|---|
| **1** | **Store Phone** calls long distance number 9-011-912-2600-9000. |
| **2** | **SM** sends 9-011-912-2600-9000 to **CM** for origination call processing. |
| **3** | **CM** translates the call into E.164 based on callers' location and sends +1-912-2600-9000 to **SM**. |
| **4** | **SM** adapts the call for the local carrier and sends 9-011-912-2600-9000 back to original branch.<br>Session Manager could use the E.164 formatted number to do TEHO routing at this step. |
| **5** | **IP Office Branch** sends 011-912-2600-9000 to the CO. |
| **A** | RTP Media stream is initially between the **Store Phone** and **CM-1** media gateway. |
| **B** | Once the call is established, the RTP Media stream will shuffle between the **Store Phone** and the **IP Office Branch**. |

### Incoming call

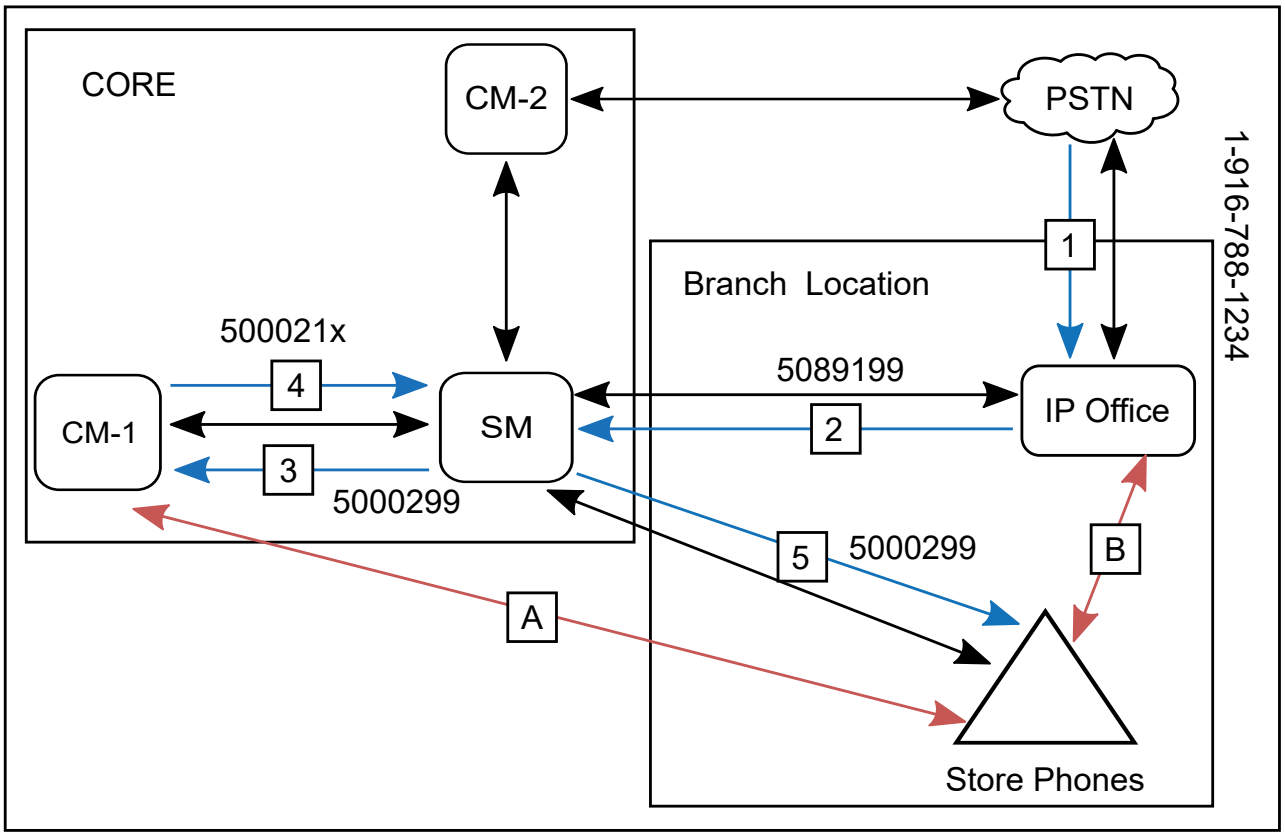


**Figure 7: Sunny day — incoming call**

| | |
|---|---|
| **1** | Customer dials store extension 1-916-788-1234 (the branch phone number). |
| **2** | **IP Office Branch** sends call to VDN 5000299 on **CM** via **SM**. Routing to a VDN allows reporting of branch calls. |
| **3** | **SM** matches call with a Route Pattern and sends to **CM**. |
| **4** | **CM** routes call to x-ported station that is bridged to all branch endpoints.<br>(X-ported refers to AWOH (Admin With Out Hardware) stations that are administered as an endpoint without any physical hardware.) |
| **5** | **SM** sends call to all endpoints at the branch. |
| **A** | RTP Media stream is initially between the **Store Phone** and **CM-1** media gateway. |
| **B** | Once the call is established, the RTP Media stream will shuffle between the **Store Phone** and the **IP Office Branch**. |

## Sunny day: Centralized phone to Centralized phone within the branch

In this scenario, the Centralized user that receives the call answers the call as expected.

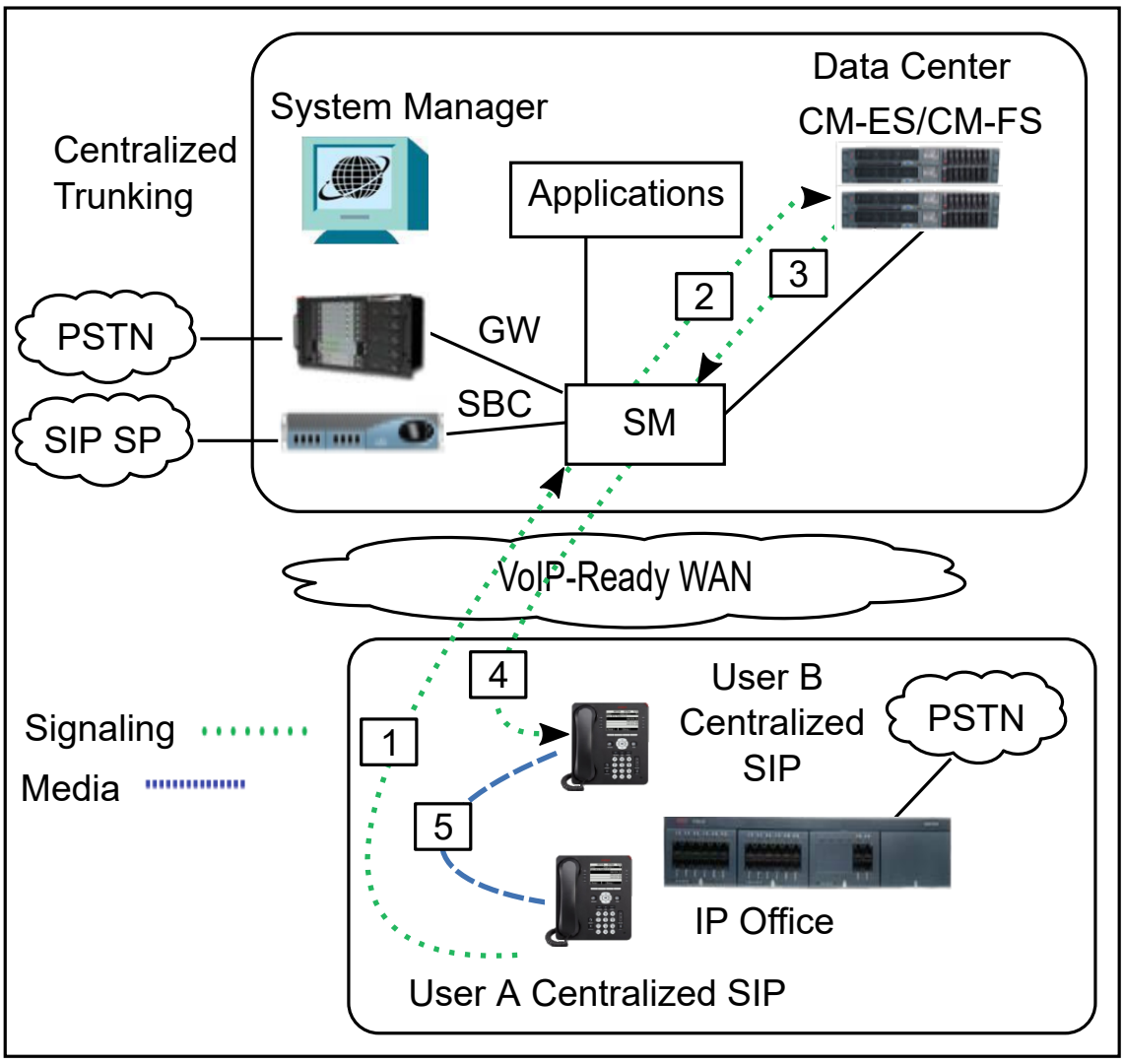


**Figure 8: Media flow for a call from one Centralized user to another Centralized user within the same branch in Sunny day**

IP Office is not involved in this scenario. For the call signaling flow to work as expected, Avaya Aura® Communication Manager must be configured for Initial IP-IP Direct Media.

The following table shows the call signaling and media flow for this call scenario in more detail.

**Table 9: Call signaling flow description for Centralized phone to Centralized phone within the branch in Sunny day**

| Call scenario | Description of call signalling flow |
|---|---|
| User A makes a call to user B. The call connects successfully and user B answers the call as expected. | The following steps outline the call signaling flow shown in the image above.<br>1. Centralized Phone A makes a call to Centralized Phone B.<br>2. Avaya Aura® Session Manager sends Phone B's number to Avaya Aura® Communication Manager for processing.<br>3. Avaya Aura® Communication Manager identifies the call as internal (within the branch) and routes Phone B's number back to Avaya Aura® Session Manager.<br>4. Avaya Aura® Session Manager sends the call invite to Phone B.<br>5. The user of Phone B answers the call. The RTP media stream flows directly between Phone A and Phone B. |

**Related links**

Direct media setting in Avaya Aura Communication Manager on page 45

## Sunny Day: Centralized phone to PSTN number with Music on Hold

The following image shows the call media flow for a call from a centralized phone to the local PSTN trunk in the branch. In this scenario, the call is put on hold with Music on Hold and then the call on hold is retrieved.

The call on hold uses resources from the central gateway for Avaya Aura® Communication Manager and local trunks in IP Office.

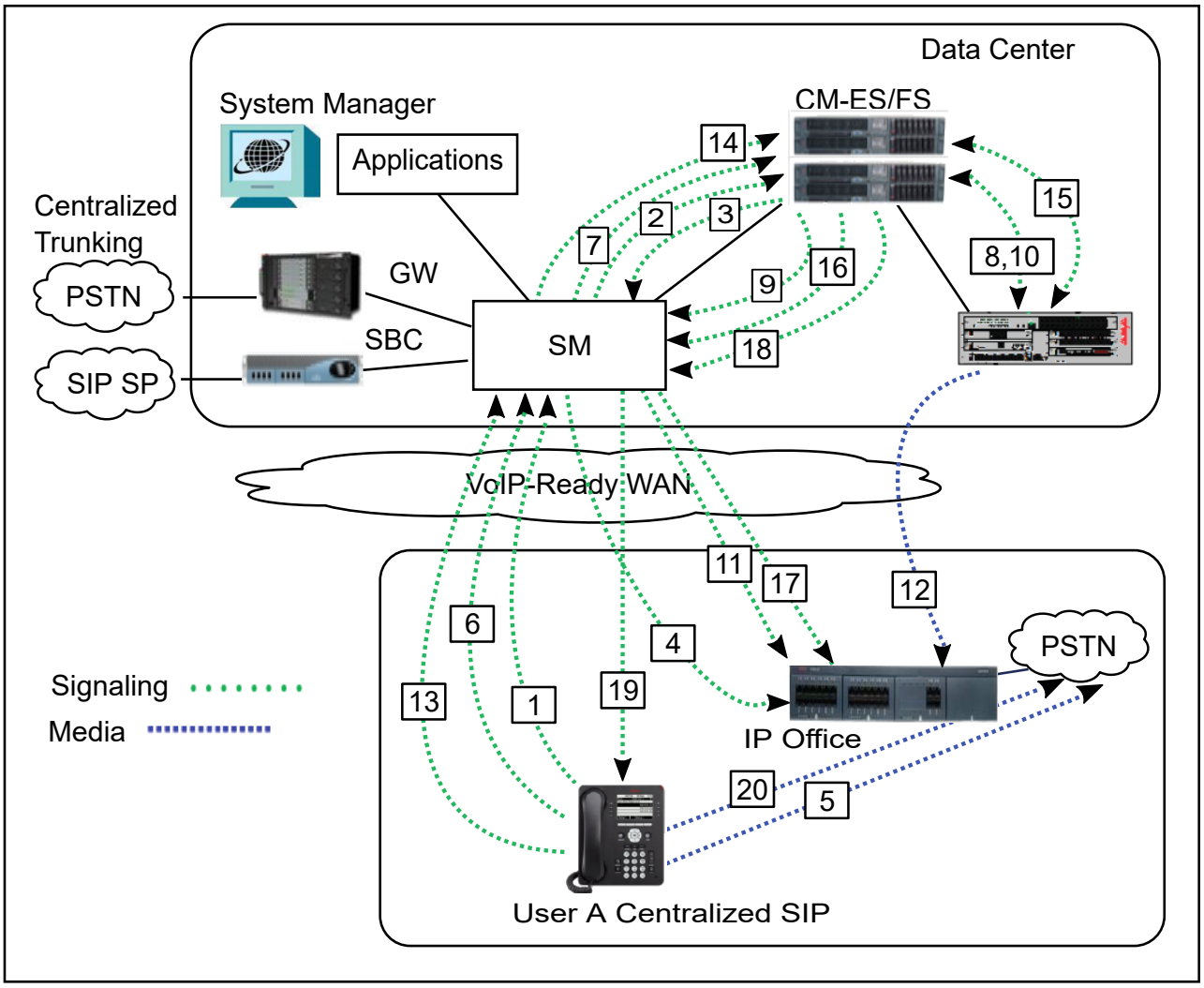


**Figure 9: Call signaling flow for centralized phone to PSTN number with Music on Hold**

**Table 10: Call signaling flow description for Centralized phone to PSTN number with Music on Hold in a Sunny Day mode**

| Call scenario | Description of call signaling flow |
|---|---|
| User A makes a call to user B. The call connects successfully and user A puts the call on hold. Music on Hold plays in the background while the call is on hold. User A then retrieves the call on hold. | 1. Centralized Phone A calls the PSTN number, and sends an INVITE message to Avaya Aura® Session Manager .<br>2. Session Manager sends an INVITE message with the PSTN number to Communication Manager for origination processing.<br>3. Communication Manager identifies the call as a PSTN number and uses the ARS table to route the PSTN number back to Session Manager.<br>4. Session Manager routes call to the destination SIP Entity that is IP Office, and thenIP Office routes the call to the PSTN number.<br>5. After the call is established, RTP media stream is sent directly between Phone A and the PSTN number that is the IP Office number.<br>6. Phone A invokes Music on Hold by sending an INVITE toSession Manager with the PSTN number in the To header<br>7. Session Manager sends an INVITE to Communication Manager for the Music on Hold feature<br>8. Communication Manager exchanges H.248 messages with the gateway for Communication Manager to connect or reserve the Music on Hold resources.<br>9. Communication Manager sends back an INVITE to Session Manager to connect to the central gateway for Communication Manager.<br>10. Communication Manager exchanges more H.248 messages with the gateway for Communication Manager.<br>11. Session Managersends an INVITE to IP Office to connect to the central gateway of Communication Manager and put the call to PSTN with the Music on Hold feature.<br>12. The PSTN number is put on hold taking resources from the central gateway and using local trunks in IP Office.<br>13. Phone A sends an INVITE message to Session Manager to retrieve the call on hold.<br>14. Session Manager sends an INVITE to Communication Manager to retrieve the call from the call on hold status. |

| Call scenario | Description of call signaling flow |
|---|---|
| | 15. Communication Manager exchanges H.248 messages with the central gateway to disconnect or release the Music on Hold resources.<br>16. Communication Manager sends an INVITE message back to Session Manager to disconnect from the central gateway of Communication Manager.<br>17. Session Manager sends an INVITE to IP Office to disconnect to the central gateway of Communication Manager.<br>18. Communication Manager sends an INVITE message to Session Manager to connect to the centralized phone A.<br>19. Session Manager sends an INVITE message to Phone A and the call connects and retrieves from the previous state.<br>20. After the call is established, RTP media stream is set directly between Phone A and PSTN number that is in theIP Office trunk. |

## Rainy day call flows

The next set of call flows are for Rainy day. In a Rainy day scenario, the SIP phones will failover to the IP Office in standalone mode. The standalone mode operates with a reduced SIP feature set which does not support all Sunny day features. In a Rainy day inbound call scenario, the process is as follows: an inbound call rings on all phones until that call is answered. Because bridged line appearances are not supported as a Rainy day feature, once that call is answered, the call will only appear on the active station, and all other stations will show idle. Note that all idle phones will continue to be available to make or answer incoming calls from the local PSTN or make and receive calls from within the store.

Moving an answered call from one phone to another requires a transfer. Only internal store and PSTN dialing is supported, and any non-PSTN extensions to extension dialing will not be routed. To dial any external number in Rainy day, users will need to dial 9 followed by the complete 7 or 10 digit number.

Additionally, due to the dial plan convention adopted, any store to store calling, or calls from corporate locations, are only supported through the PSTN to the main CO lines provisioned. Without any additional programming, all 7 digit dialing from the corporate location or other retail store to a store operating in Rainy day will receive a ‘wave-off’ tone. In order to mitigate this behavior the recommendation is to provide each station with a point of coverage that directs the calling party to primary DID for store where the extension resides.

### Incoming call — failover, inbound call

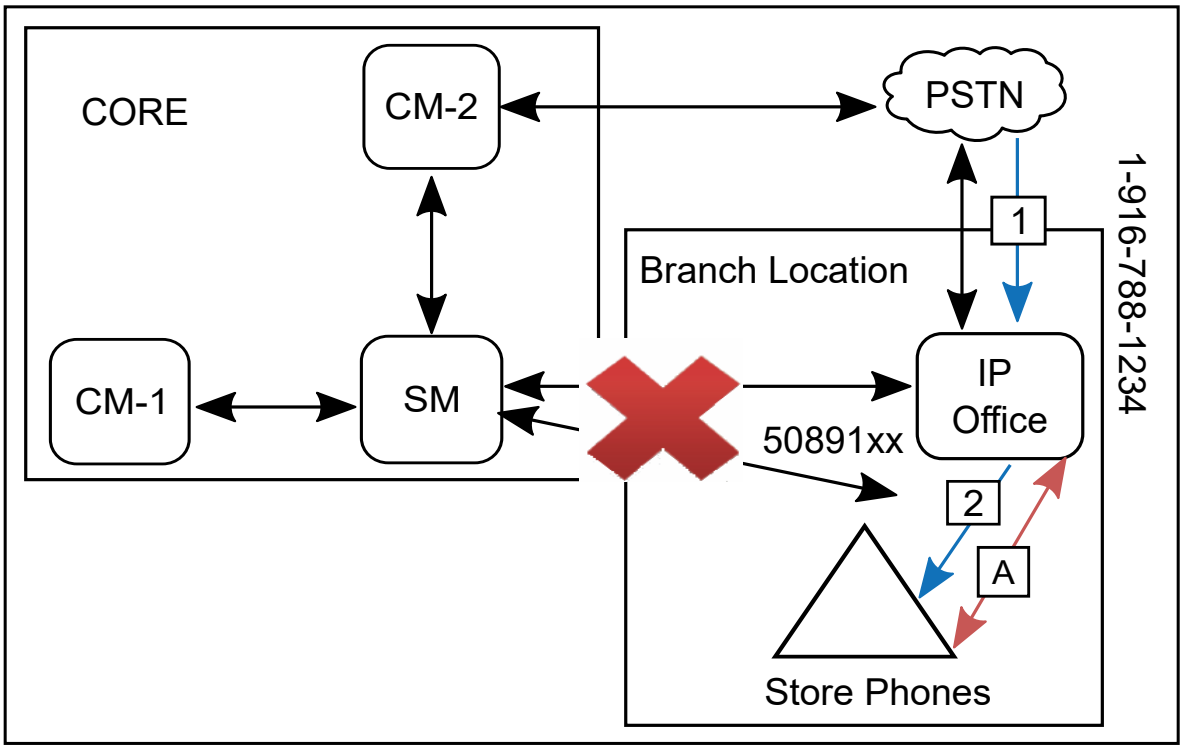


**Figure 10: Rainy day — inbound call**

| | |
|---|---|
| **1** | Customer dials store extension 1-916-788-1234. |
| **2** | **IP Office Branch** sends the call directly to all registered SIP stations in the hunt group. |
| **A** | RTP Media is established between the **Store Phone** and the **IP Office Branch**. |

### Incoming call from corporate – coverage

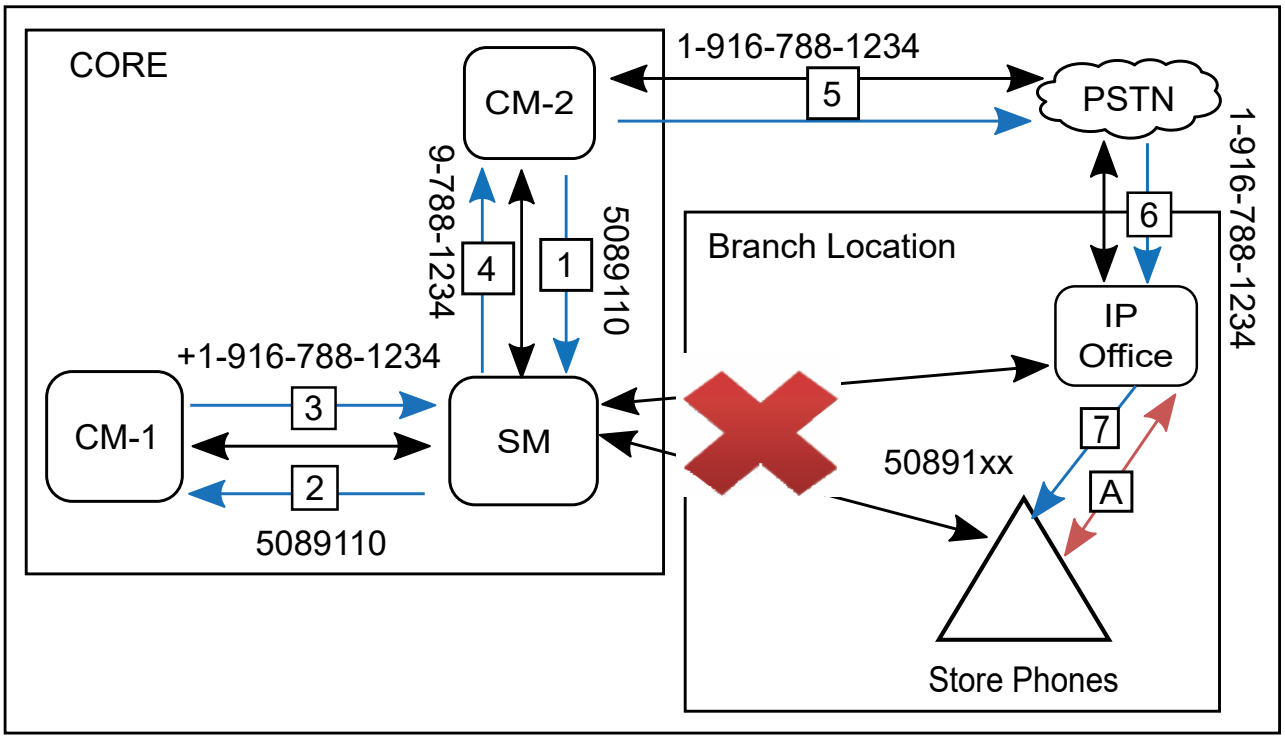


**Figure 11: Rainy day — incoming call from corporate user**

| | |
|---|---|
| **1** | Corporate user dials branch extension 5089110 |
| **2** | **SM** routes call to **CM-1** for origination call processing. |
| **3** | **CM-1** and **SM** identify that branch is unreachable. **CM-1** routes call to coverage path for 5089110. In this case the remote coverage path would be the E.164 formatted DID of the store +1-916-788-1234. |
| **4** | **SM** sends the call back to the originating **CM-2**. **SM** adapts the call to a format that is routable at the local **CM-2**. |
| **5** | **CM-2** sends the call to the PSTN. |
| **6** | Call is routed through the PSTN to the store through the local CO. |
| **7** | **IP Office Branch** routes the call to a hunt group. This hunt group in turn sends the call to all registered SIP stations at the branch. |
| **A** | RTP Media stream is established between the **Store Phone** and the **IP Office Branch**. |

# Centralized voice mail example call flows

The following sections describe call flows for incoming calls transferred to Centralized voice mail systems. The call flow descriptions provided are examples only, and do not cover all possible scenarios.

# Sunny day call flows: Incoming call transferred to Avaya Aura® Messaging or Avaya Modular Messaging

IP Office systems deployed in Distributed, Centralized, and Mixed environments can be configured to use an Avaya Aura® Messaging or Avaya Modular Messaging centralized voice mail system.

The following sections describe Sunny day call and media flow scenarios for:

- Incoming call from centralized SIP phone to H.323 IP Office phone transferred to centralized voice mail
- Incoming call to ATA user in the same branch transferred to centralized voice mail
- Incoming call to centralized SIP user in the same branch transferred to centralized voice mail

## Sunny day: Incoming call from centralized SIP phone to H.323 IP Office phone transferred to centralized voice mail

In this scenario, the call is transferred to a centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system in Sunny day.

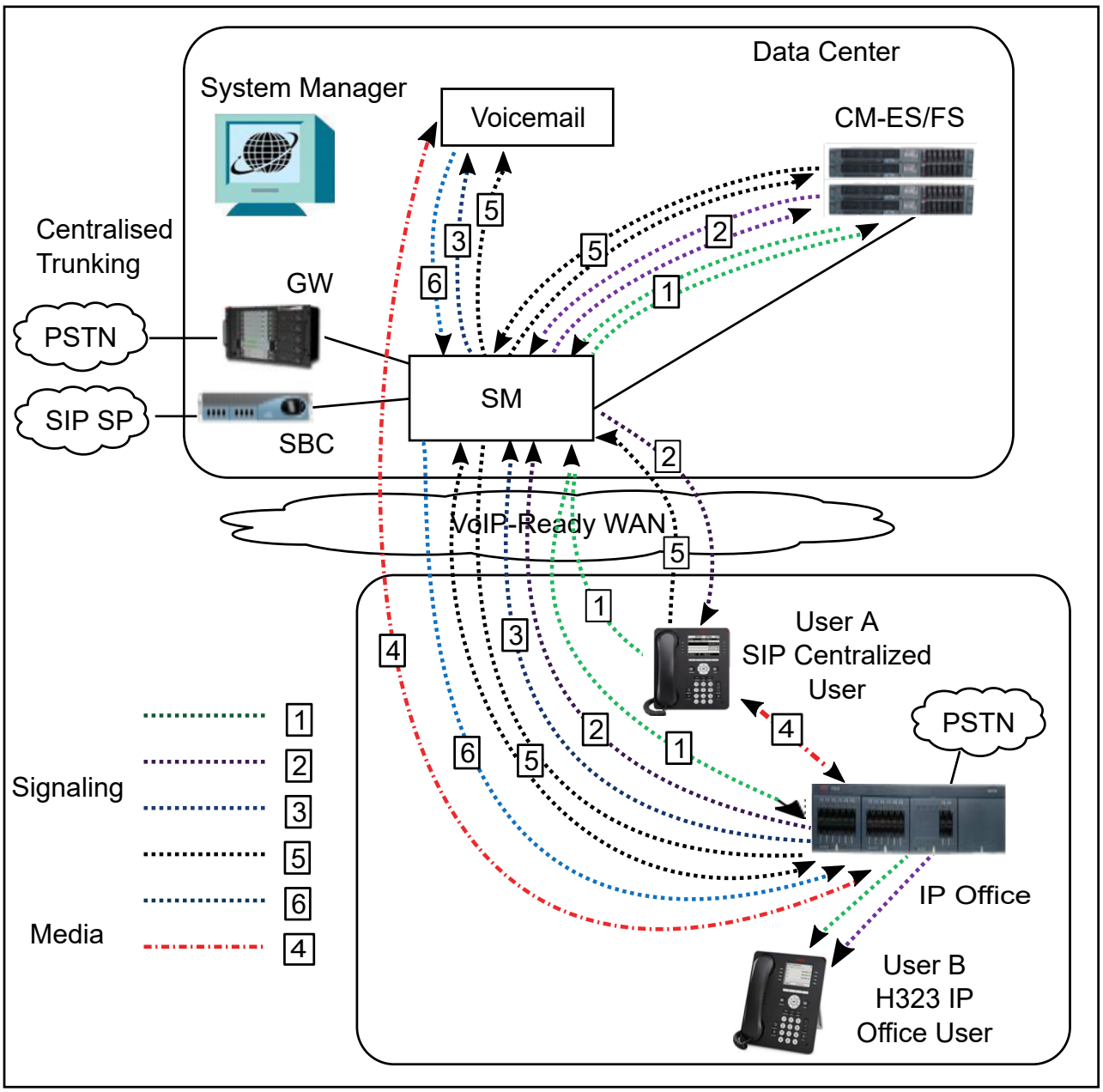


**Figure 12: Call media flow for an incoming call from a centralized SIP phone to an H.323 IP Office phone**

The following table describes the call signaling flow in more detail.

**Table 11: Call signaling flow description for incoming call from a centralized SIP phone to an H.323 IP Office phone transferred to centralized voice mail in Sunny day**

| Call scenario | Description of call signalling flow |
|---|---|
| 1. SIP Centralized user A calls H.323 IP Office user B.<br>2. User B does not answer the call.<br>3. Call is forwarded to an Avaya Aura® Messaging or Avaya Modular Messaging centralized voice mail system.<br>4. User A leaves a voice mail message for user B.<br>5. User A hangs up after leaving the message.<br>6. Message Waiting Indicator lights up on user B's phone to indicate that a new voice message is waiting. | 1. User A SIP Invite ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ H.323 IP Office user B.<br>2. Ringing IP Office phone ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ user A.<br>3. SIP Invite sent from IP Office ▸ Session Manager ▸ centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system.<br>4. RTP is set up between user A, IP Office, and the voice mail system.<br>5. BYE is sent from user A ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ IP Office ▸ Session Manager ▸ voice mail.<br>6. NOTIFY is sent from voice mail ▸ Session Manager ▸ IP Office with Message Waiting Indication for user B. |

## Sunny day: Incoming call to ATA phone transferred to centralized voice mail

The following image shows the call media flow for a call to an ATA phone in the same branch. In this scenario, the call is transferred to a centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system in Sunny day.

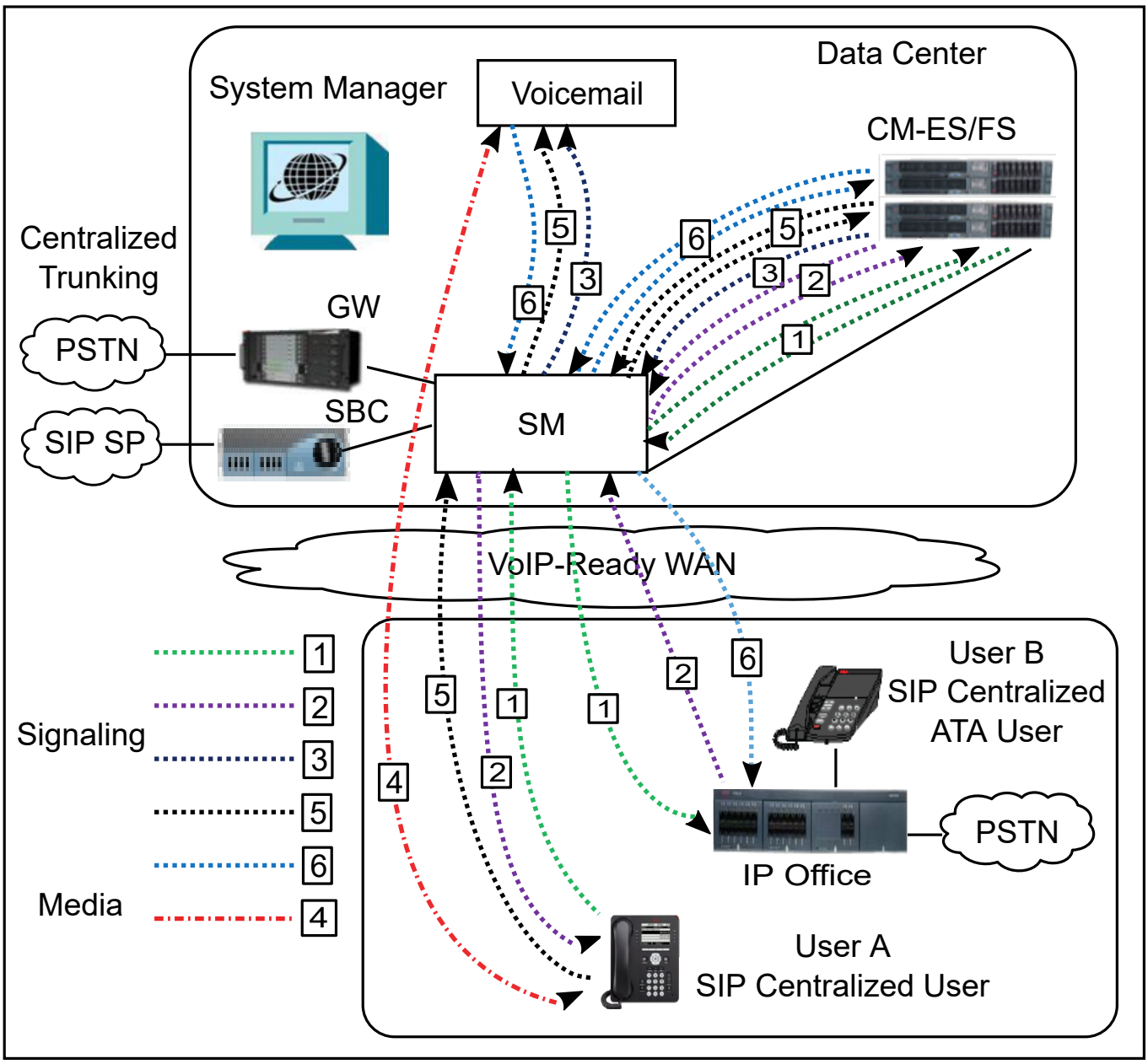


**Figure 13: Sunny day — incoming call to ATA phone transferred to centralized voice mail**

The following table describes the call signaling flow in more detail.

**Table 12: Call signaling flow description for incoming call to ATA phone transferred to centralized voice mail in Sunny day**

| Call scenario | Description of call signalling flow |
|---|---|
| 1. SIP Centralized user A calls SIP Centralized ATA user B.<br>2. User B does not answer the call.<br>3. Call is forwarded to an Avaya Aura® Messaging or Avaya Modular Messaging centralized voice mail system.<br>4. User A leaves a voice mail message for user B.<br>5. User A hangs up after leaving the message.<br>6. Message Waiting Indicator lights up on user B's phone to indicate that a new voice message is waiting. | 1. User A SIP Invite ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ IP Office ▸ user B.<br>2. Ringing ATA phone ▸ IP Office ▸Session Manager ▸ Communication Manager ▸ Session Manager ▸ user A.<br>3. CANCEL is sent for the original invite and a new invite is sent from Communication Manager ▸ Session Manager ▸ centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system.<br>4. Media is set up between user A and the voice mail system.<br>5. BYE is sent from user A ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ voice mail.<br>6. NOTIFY is sent from voice mail ▸ Session Manager ▸ Communication Manager ▸ PUBLISH to Session Manager ▸ NOTIFY ▸ IP Office ▸ user B with Message Waiting Indication. |

## Sunny Day: Centralized SIP phone to Centralized SIP phone transferred to centralized voice mail

In this scenario, User A calls User B. User B does not answer and the call is transferred to a centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system. The following image shows the call media flow for a call from a centralized user to another centralized user with coverage to voice mail.

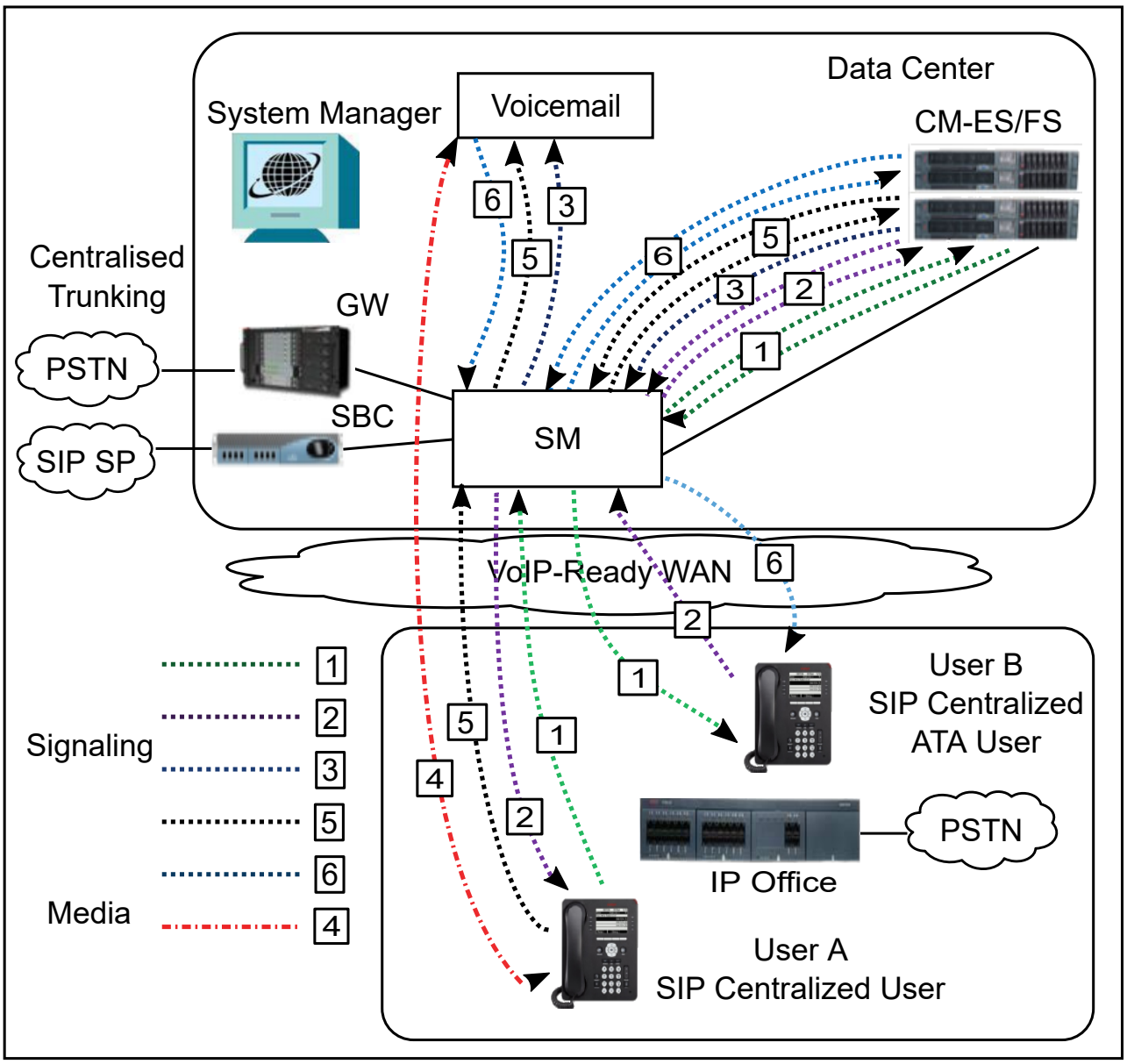


**Figure 14: Centralized user A call to centralized user B with coverage to voice mail**

The following table describes the call signaling flow in more detail.

**Table 13: Call signaling flow description for call from a centralized user to another centralized user with coverage to voice mail**

| Call scenario | Description of call signalling flow |
|---|---|
| 1. SIP centralized user A dials SIP centralized user B<br>2. User at station B does not answer the call.<br>3. Call is forwarded to an Avaya Aura® Messaging or Avaya Modular Messaging centralized voice mail system.<br>4. User A leaves a voice mail message for user B.<br>5. User A hangs up after leaving the message.<br>6. Message Waiting Indicator lights up on user B's phone to indicate that a new voice message is waiting. | 1. User A SIP Invite ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ user B.<br>2. Ringing user B▸ Session Manager ▸ Communication Manager ▸Session Manager ▸▸ user A.<br>3. A CANCEL message is sent for the original invite and a new invite is sent from Communication Manager ▸ Session Manager ▸ centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system.<br>4. RTP media is set up between user A, theCommunication Manager gateway, and the voice mail system.<br>After a series of invites, RTP is set up between User A and the voice mail system.<br>5. BYE is sent from user A ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ voice mail.<br>6. NOTIFY is sent from voice mail ▸ Session Manager ▸ Communication Manager ▸ PUBLISH to Session Manager ▸ NOTIFY ▸ user B with Message Waiting Indication status. |

# Rainy day call flows: Incoming call transferred to Avaya Aura® Messaging or Avaya Modular Messaging

When configuring a branch to use Avaya Aura® Messaging or Avaya Modular Messaging, an internal number is set for calls to the Centralized voice mail system to be automatically routed through the SM Line. You can also configure an additional PSTN number to ensure that calls to Avaya Aura® Messaging or Avaya Modular Messaging are correctly routed to the PSTN trunk when the SM Line is not available. For configuration procedures, see Deploying Avaya IP Office™ Platform as an Enterprise Branch with Avaya Aura® Session Manager.

## Rainy day media flow scenario

The following image shows the media flow for an unanswered call in Rainy day. In this media flow scenario, the unanswered call is transferred to a Centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system.

**Important:**

This type of call flow scenario in Rainy day is not supported on CS 1000 Call Pilot voice mail systems.

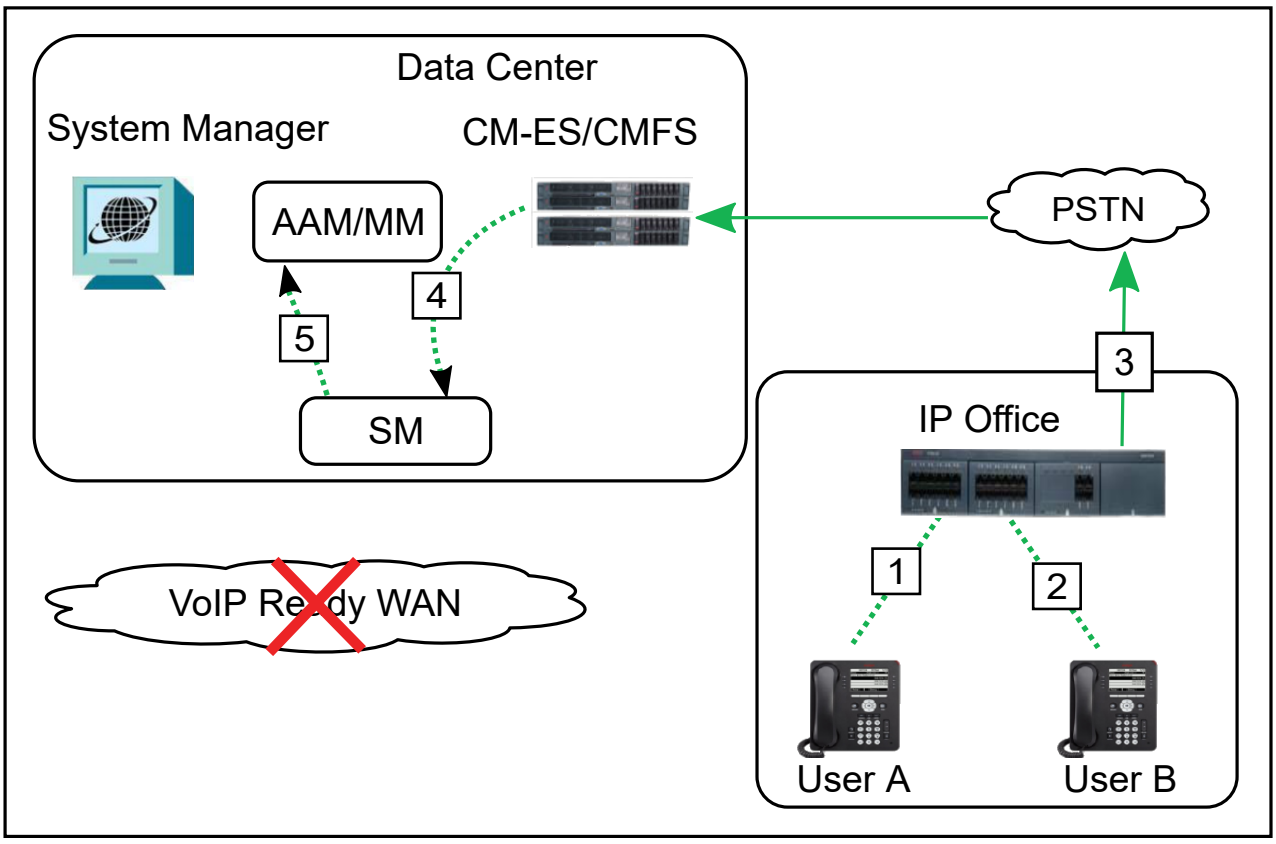


**Figure 15: Incoming call to IP Office phone transferred to Avaya Aura® Messaging voice mail system**

In Rainy day, all station types (SIP, H.323, digital, and analog) are registered to IP Office. IP Office users and Centralized users leave and retrieve voice mails over the PSTN. In Rainy day conditions, DTMF digits indicate specific mailboxes, and the Message Waiting Indicator on the called user's phone does not light up.

The following table describes the call signalling flow in more detail.

**Table 14: Call signaling flow description for incoming calls transferred to Avaya Aura® Messaging in Rainy day**

| Call scenario | Description of call signalling flow |
|---|---|
| 1. User A dials user B. In this example, the call is local (from station to station), but the media flow would be the same for an inbound trunk call.<br>2. User at station B does not answer the call and the process of transferring the call to voice mail begins.<br>3. IP Office determines that the SM Line is not available and routes the call over the PSTN.<br>4. The call is routed from Communication Manager to Session Manager.<br>5. The call is routed from Session Manager to the Avaya Aura® Messaging or Avaya Modular Messaging voice mail system. | 1. Call from user A is routed to user B through the IP Office.<br>2. IP Office determines whether voice mail is enabled.<br>3. The PSTN voice mail number requires a corresponding Short Code or ARS entry for the call to be routed to the correct line in Rainy day. The PSTN number must be a valid DID number from the branch to the core voice mail system.<br>4. Communication Manager processes the incoming voice mail PSTN number and then transfers the number to Session Manager.<br>5. IP Office sends DTMF tones to Session Manager to indicate which mail box the call should be transferred to Session Manager then forwards the call to the Centralized Avaya Aura® Messaging or Avaya Modular Messaging voice mail system.<br>**Note:**<br>IP Office must be administered to send voice mail instructions as DTMF tones. In the IP Office Manager Voicemail tab, you must select **Enable voice mail Instructions Using DTMF**. If this capability is disabled, DTMF digits are not sent automatically. Instead, the caller must manually dial the mail box number. |

# ATA example call flows

The following sections provide general call signaling and media flow information for an ATA phone to an IP Office PSTN trunk. The call flow descriptions provided in this section are examples only. This section does not cover all possible call flow scenarios for ATA users.

## Centralized ATA phone to IP Office PSTN trunk call flow

A Centralized ATA phone is an analog phone for a Centralized user.

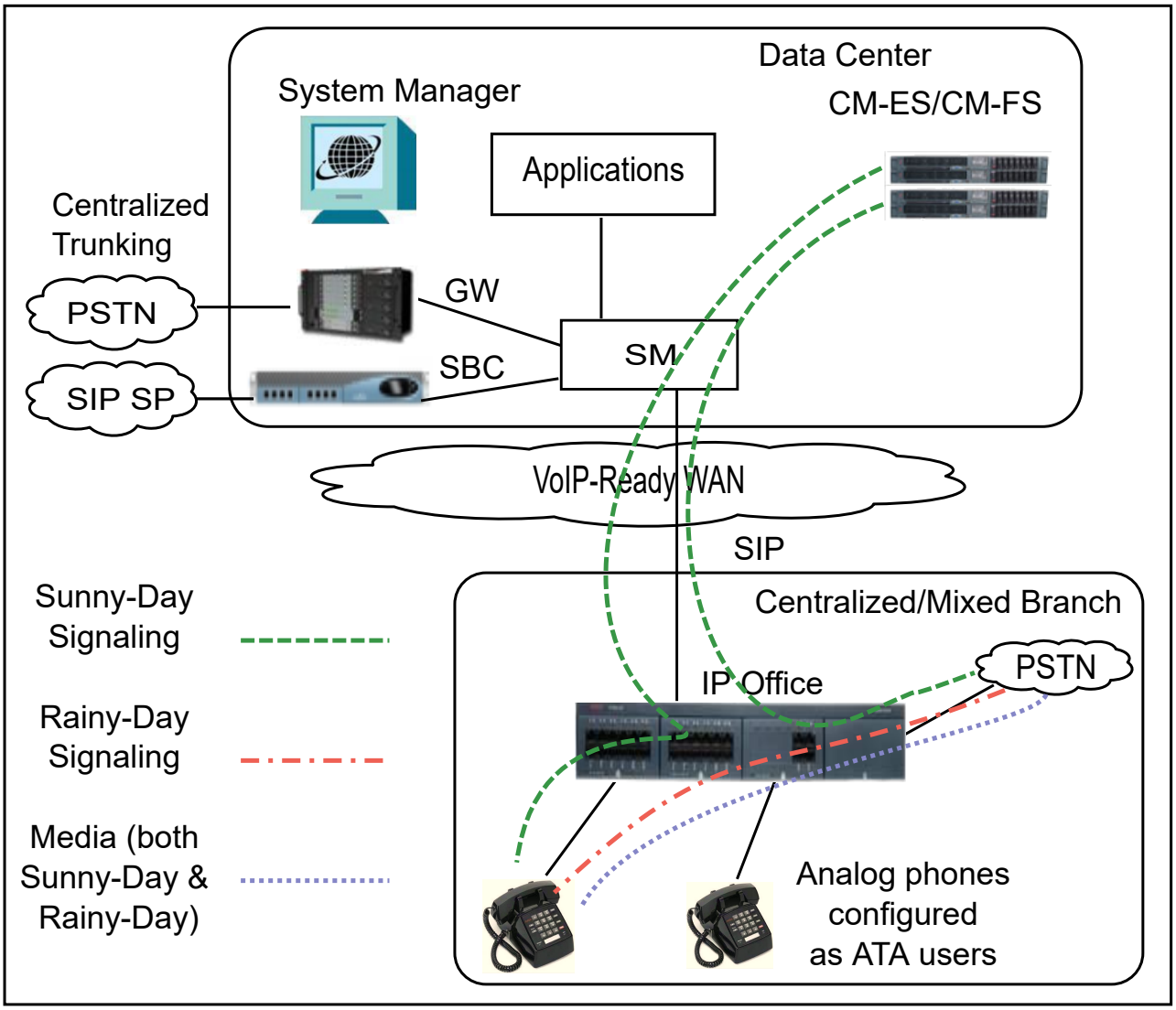


**Figure 16: Call signaling and media flow for calls from ATA phones to the IP Office PSTN trunk in Sunny day and Rainy day**

## Sunny day

In Sunny day, the IP Office transparently relays the dialed string from the ATA user's analog phone through the SIP INVITE sent to Avaya Aura® Session Manager and then to Avaya Aura® Communication Manager. This is shown in more detail in the following table.

**Table 15: Call signaling flow description for Centralized ATA phones**

| Call scenario | Description of call signaling flow |
|---|---|
| Call from ATA phone to IP Office PSTN trunk. The Centralized ATA user is registered with the IP address of IP Office. | SIP INVITE from IP Office ▸ Session Manager ▸ Communication Manager ▸ Session Manager ▸ IP Office ▸ PSTN |

For information about the call signaling and media flow with the Communication Manager call pick-up feature, see ATA phone invoking Communication Manager call pick-up feature call flow on page 79.

### Rainy day

In Rainy day, the SM Line is not available, so all call routing and media is handled directly by the IP Office. IP Office routes the call from the user's Centralized ATA phone to the IP Office PSTN trunk.

## ATA phone invoking Communication Manager call pick-up feature call flow

In Sunny day, users can invoke the Communication Manager call pick-up feature by dialing a Communication Manager Feature Access Code (FAC) or Feature Name Extension (FNE) on the ATA user's phone. IP Office does not process the dialed string or identify the Communication Manager feature accessed.

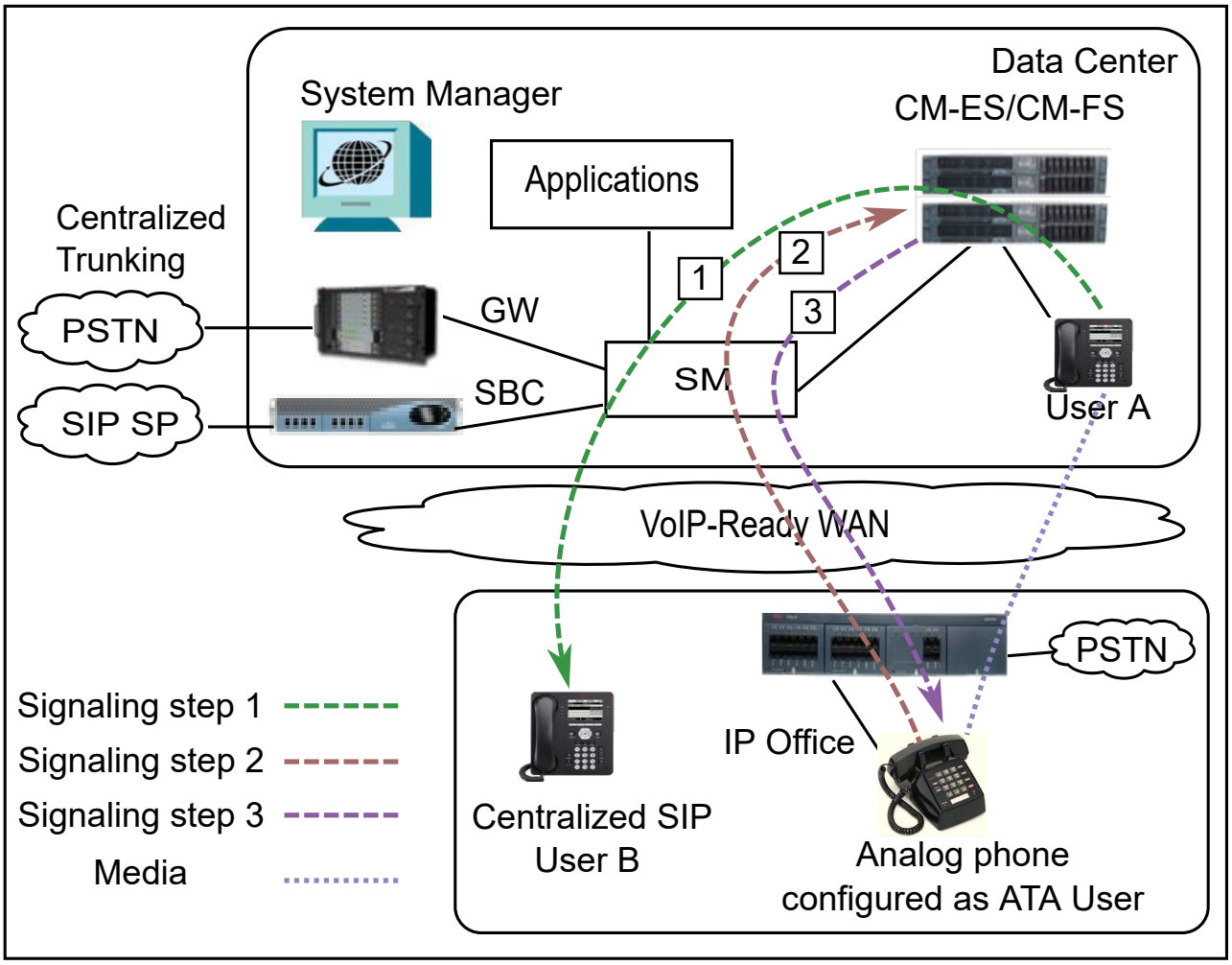


**Figure 17: ATA call media flow with the Communication Manager call pick-up feature**

The media flows between the headquarters user A and the ATA user in the branch through IP Office. IP Office connects and interworks the media flow on the IP network side with the TDM media on the connection to the ATA user's phone. The call signaling and media flow is shown in more detail in the following table.

**Table 16: Call signaling flow description for call pick-up feature with Centralized ATA phones**

| Call scenario | Description of call signaling flow |
|---|---|
| 1. Headquarters user A in the data center makes a call to Centralized SIP user (B) in the branch.<br>2. ATA user hears the Centralized SIP phone (B) ringing and wants to answer the call on their ATA phone. So the ATA user dials a FAC to invoke the Communication Manager call pick-up feature and answer the call.<br>3. Centralized SIP user (B) and the ATA user in the branch are in the same call pick-up group, so Communication Manager delivers the call directly to the ATA user and cancels the call previously delivered to user B. | 1. User A ▸ Communication Manager ▸ SIP INVITE to Session Manager ▸ Centralized SIP user (B)<br>2. SIP INVITE carrying FAC sent from IP Office ▸ Session Manager ▸ Communication Manager<br>3. Communication Manager ▸ SIP `200 OK` message to Session Manager ▸ IP Office ▸ ATA user |

The Communication Manager call pick-up feature is not supported in Rainy day.

# SRTP example call flows

This section describes the following SRTP call flows:

- Centralized SIP phone to IP Office PSTN trunk in Sunny day and Rainy day.
- IP Office user with IP phone to headquarters (HQ) user outside the branch in Sunny day. This call flow option is not available in Rainy day.

This section highlights a few key SRTP call flow examples. Other call flow scenarios not directly described in this section are also possible.

# Centralized SIP phone to IP Office PSTN trunk

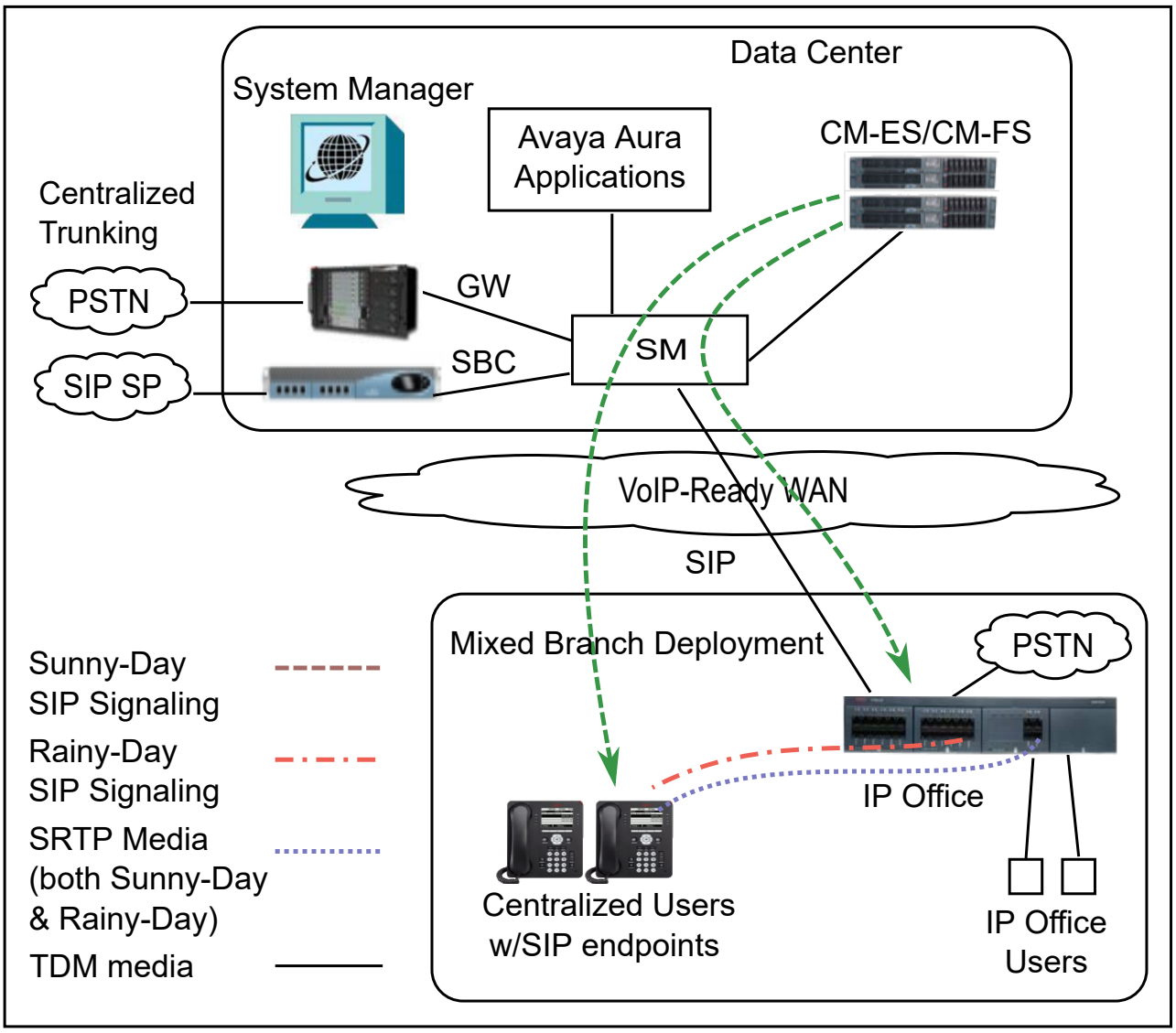


**Figure 18: Call media flow: Centralized SIP phone to IP Office PSTN trunk in Sunny day and Rainy day**

## Sunny day

In Sunny day, the following conditions must be met for the media session to use SRTP between the Centralized SIP phone and the IP Office:

- Signalling is done over TLS via Avaya Aura® Session Manager and the core Avaya Aura® Communication Manager
- Regardless of whether Communication Manager is configured for Initial Direct Media or IP-IP audio (shuffling), signaling and Media Encryption should be configured consistently on Communication Manager, the central Communication Manager Media Gateway, Session Manager, IP Office, and the Centralized SIP phone.

The media session flow uses SRTP between the Centralized SIP phone and the IP Office. IP Office connects and interworks the SRTP flow on the IP network side with the TDM media on the PSTN trunk. The following table describes the media flow for the possible call flow scenarios.

**Table 17: SRTP call flow scenarios**

| Call flow scenario | SRTP media flow |
|---|---|
| Communication Manager is configured for Initial Direct Media | SRTP media flows directly from the Centralized SIP phone ▸ IP Office |
| Communication Manager is configured for direct IP-IP audio (or shuffling), but not for Initial Direct Media | Two stages:<br>1. At the beginning of each call, media initially flows across the WAN through a Communication Manager central media resource, such as the G450 Media Gateway.<br>2. Once Communication Manager shuffles the call, the SRTP media flows directly from the Centralized SIP phone ▸ IP Office. |

### Rainy day

In Rainy day, the following conditions must be met for the media session to use SRTP between the Centralized SIP phone and the IP Office:

- Signalling is over TLS. In Rainy day, this is supported with 9600 Centralized SIP phones, but not with the Centralized SIP Avaya one-X® Communicator.
- Media Encryption is configured on the IP Office and on the Centralized SIP phone.

The media session flow uses SRTP between the Centralized SIP phone and the IP Office. IP Office connects and interworks the SRTP flow on the IP network side with the TDM media on the PSTN trunk.

**Related links**

Bandwidth requirements on page 43

## IP Office user with IP phone to headquarters user outside the branch

### Sunny day

The call flow is similar if the call is made to the IP Office SIP trunk rather than the IP Office IP phone. This call flow scenario is not supported in Rainy day.

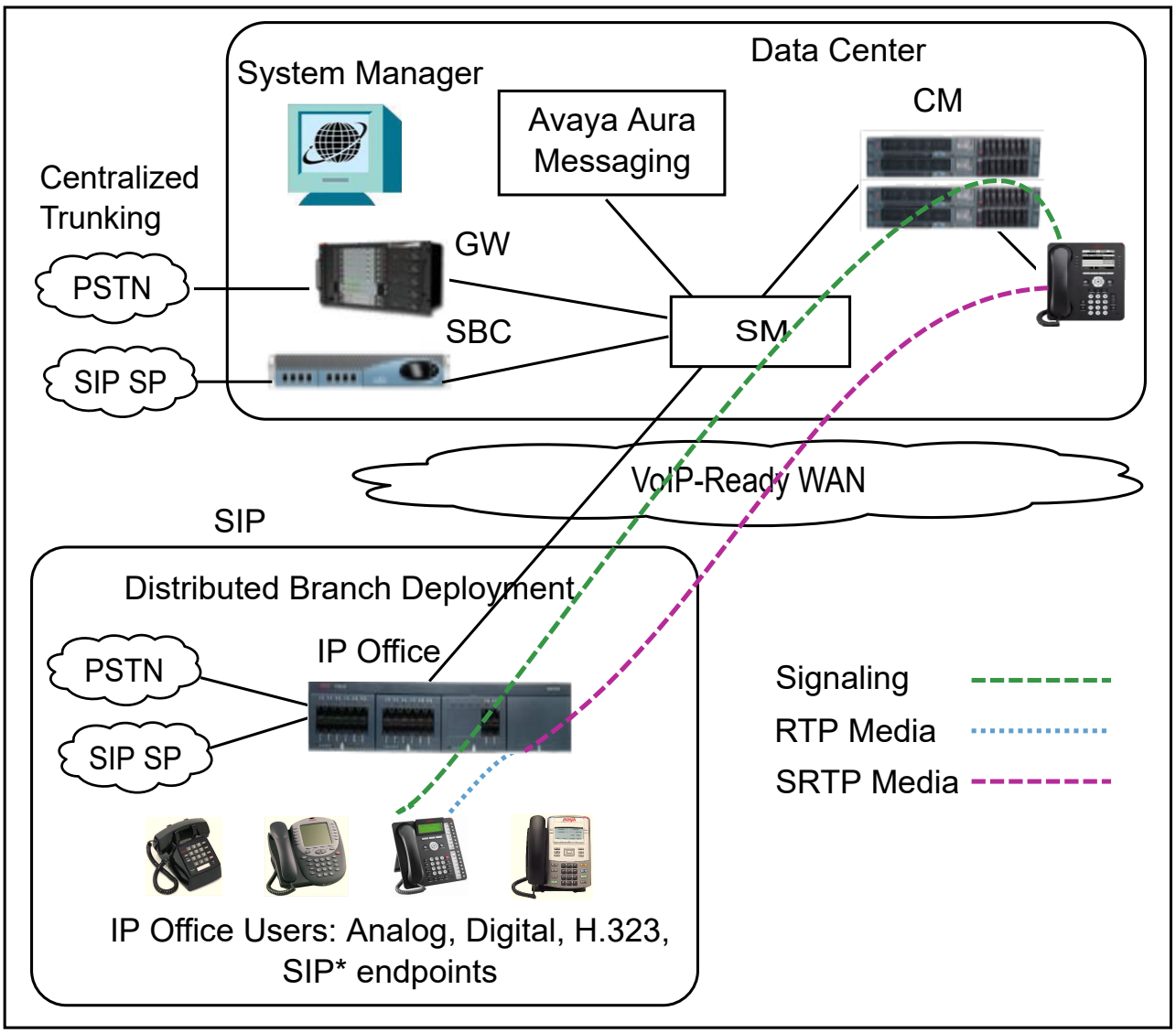


**Figure 19: SRTP call flow: IP Office to HQ user outside the branch in Sunny day**

In Sunny day, the following conditions must be met for the call signalling to negotiate the use of SRTP between IP Office and the HQ user's phone:

- Signalling is done on the SM Line over TLS
- Regardless of whether Communication Manager is configured for Initial Direct Media or IP-IP audio (shuffling), signaling and Media Encryption should be configured consistently on Communication Manager, the central Communication Manager Media Gateway, Session Manager, IP Office, and the Centralized SIP phone.

IP Office negotiates SRTP separately for each call leg. For example, SRTP is disabled for the endpoint of the local IP Office user, or the IP Office user uses an endpoint type in which SRTP is not supported. In such a case, the media flow between IP Office and the IP endpoint of the IP Office user will use RTP.

This call flow scenario has two legs, as described in the following table.

| Description of call leg | Media flow: RTP or SRTP |
|---|---|
| Call Leg 1: IP endpoint (SIP or H.323) of the IP Office user ▸ IP Office | RTP |
| Call Leg 2: IP Office ▸ HQ user's phone | SRTP |

# Toll bypass functionality

The toll bypass functionality is supported only in SCN and Branch deployments. By default, the India Toll Bypass Prevention feature is applied for the India locale and disabled for other locales, but you can apply the feature for any locale.

## System level configuration guidelines

You can create an extension for a new location by selecting **Extension** > **Extn** > **Extension Location**. You can apply the toll bypass functionality at the system level by selecting the **Restrict Network Interconnect** and **Include location specific information** check boxes.

## Trunk level configuration guidelines

The Trunk Level **Include location specific** check box is present on the IP Office line and SM Line or on the **Line** tab. This option is available only if the **Network** type is set to **Private** in an Enterprise trunk. If the line is a PSTN trunk that includes SIP, then you must set the **Network** type of the trunk to **Public**.

**Note:**

- Select the Trunk Level **Include location specific** check box only if the other end is an IP Office, Session Manager R6.3 or later, or Communication Manager R6.3 or later. If the other end of the trunk is not toll compliant, then clear the **Include location specific** check box.
- Configure the Communication Manager-Session Manager trunk as a universal type. On the Trunk Group screen, set **SA9122 Restrict Public-Private Network Connections (Toll Bypass) For Toll Compliance, Treat As:** to universal. For more information about Communication Manager toll bypass configuration, see https://downloads.avaya.com/css/P8/documents/100172354.

## Troubleshooting toll bypass

You can check the call status using SSA. Logs of rejected calls because of the toll bypass restriction are stored on SysMonitor with originating, terminating, and call type (public/private) details. SysMonitor is the primary tool for checking and troubleshooting rejected calls. The following are the recommended trace options:

- Default All
- Call tab, Supplementary services
- SIP, Events Verbose

### Related links

Overview of IP Office example call flows on page 53

Toll bypass call flows on page 85

# Toll bypass call flows

The new enhancements in the **IP Office Restrict Network Interconnect** feature apply to endpoints and trunks. Calls can be made to or from IP phones to local PSTN public trunks only if the location of the IP phone is the same as the system location. No extra licenses are introduced as part of this enhancement. The feature is supported on IP500V2/IP500V2A and IPOL and in all the editions such as Essential, Preferred, or Select.

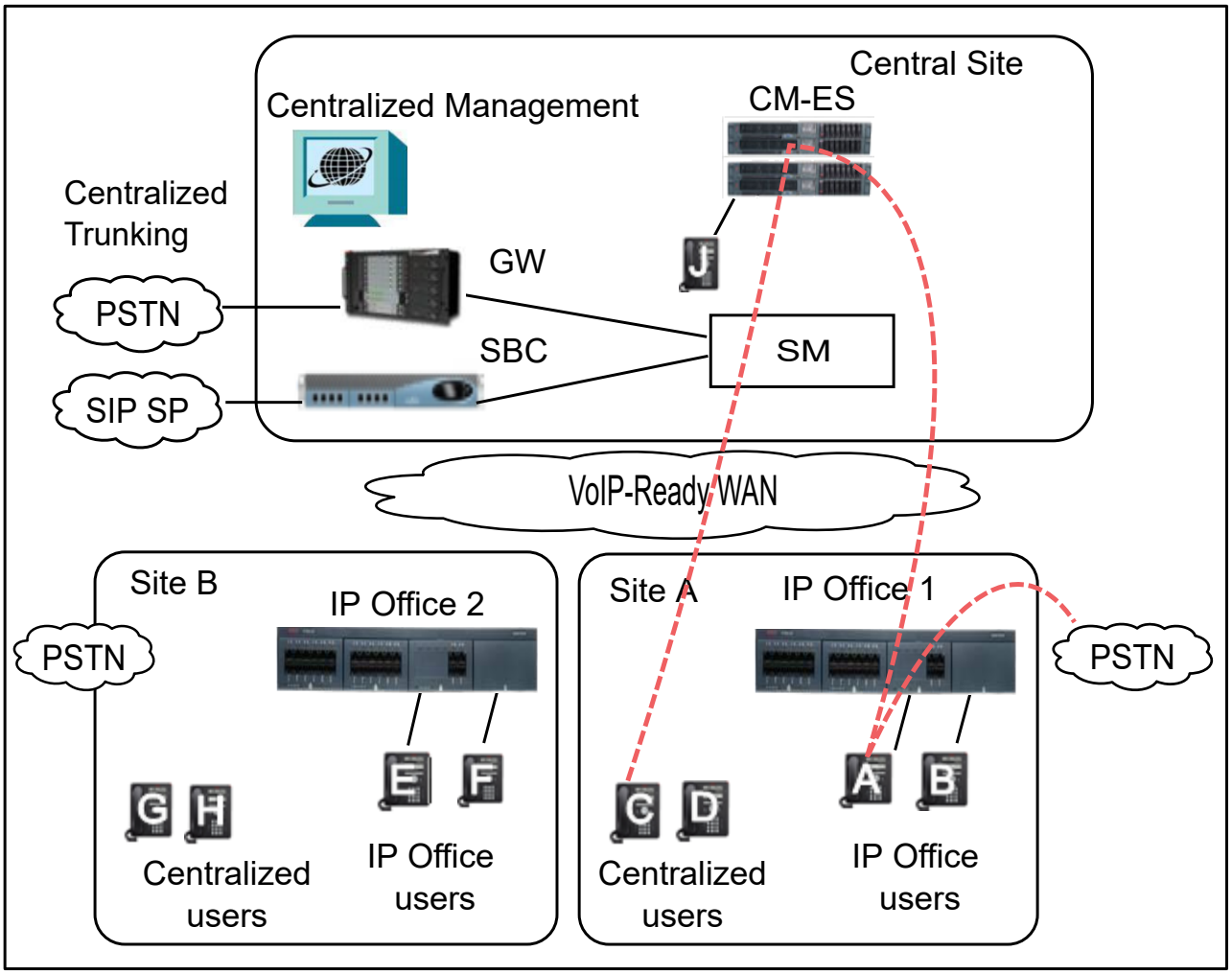


**Figure 20: PSTN call flow between an IP Office users and a Centralized user in the same location**

A Distributed user, A, receives a PSTN call originating from Site A. The Distributed user, A, forwards the call to a Centralized user, C, in Site A. This PSTN call is allowed because the locations of the originating call and the terminating call are the same.

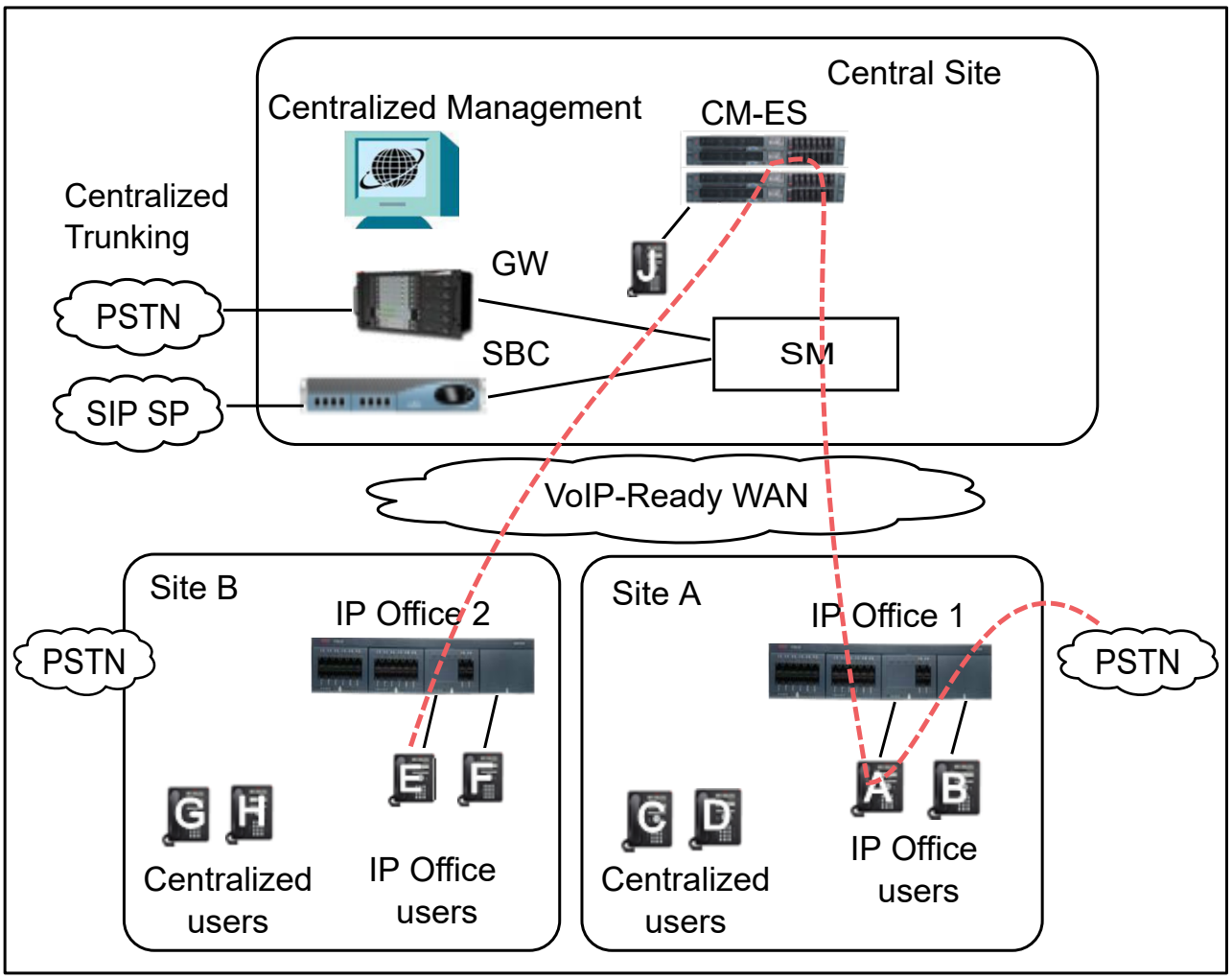


**Figure 21: PSTN call flow between two IP Office users in different locations**

A Distributed user, A, receives a PSTN call originating from Site A. The Distributed user, A, forwards the call to a Distributed user, E, in Site B. This PSTN call is not allowed because the locations of the originating call and the terminating call are different.

**Note:**

There is no restriction on Enterprise calls or non-PSTN calls. The SCN scenario is similar for non-PSTN calls for the Primary IPOL server instead of the Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

**Related links**

Toll bypass functionality on page 84

# Appendix B: Additional Help and Documentation

The following pages provide sources for additional help.

**Related links**

Additional Manuals and User Guides on page 87
Getting Help on page 87
Finding an Avaya Business Partner on page 88
Additional IP Office resources on page 88
Training on page 89

## Additional Manuals and User Guides

The Avaya Documentation Center website contains user guides and manuals for Avaya products including IP Office.

- For a listing of the current IP Office manuals and user guides, look at the Avaya IP Office™ Platform Manuals and User Guides document.
- The Avaya IP Office Knowledgebase and Avaya Support websites also provide access to the IP Office technical manuals and users guides.
  - Note that where possible these sites redirect users to the version of the document hosted by the Avaya Documentation Center.

For other types of documents and other resources, visit the various Avaya websites (see Additional IP Office resources on page 88).

**Related links**

Additional Help and Documentation on page 87

## Getting Help

Avaya sells IP Office through accredited business partners. Those business partners provide direct support to their customers and can escalate issues to Avaya when necessary.

If your IP Office system currently does not have an Avaya business partner providing support and maintenance for it, you can use the Avaya Partner Locator tool to find a business partner. See Finding an Avaya Business Partner on page 88.

**Related links**

Additional Help and Documentation on page 87

# Finding an Avaya Business Partner

If your IP Office system currently does not have an Avaya business partner providing support and maintenance for it, you can use the Avaya Partner Locator tool to find a business partner.

### Procedure

1. Using a browser, go to the Avaya Website at https://www.avaya.com
2. Select **Partners** and then **Find a Partner**.
3. Enter your location information.
4. For IP Office business partners, using the **Filter**, select **Small/Medium Business**.

**Related links**

Additional Help and Documentation on page 87

# Additional IP Office resources

In addition to the documentation website (see Additional Manuals and User Guides on page 87), there are a range of website that provide information about Avaya products and services including IP Office.

- Avaya Website (https://www.avaya.com)

  This is the official Avaya website. The front page also provides access to individual Avaya websites for different regions and countries.

- **Avaya Sales & Partner Portal** *(*https://sales.avaya.com*)*

  This is the official website for all Avaya business partners. The site requires registration for a user name and password. Once accessed, you can customize the portal to show specific products and information type that you want to see.

- **Avaya IP Office Knowledgebase** *(*https://ipofficekb.avaya.com*)*

  This site provides access to an online, regularly updated version of IP Office user guides and technical manual.

- **Avaya Support** *(*https://support.avaya.com*)*

This site provide access to Avaya product software, documentation and other services for Avaya product installers and maintainers.

- **Avaya Support Forums** *(*https://support.avaya.com/forums/index.php*)*

  This site provides forums for discussing product issues.

- **International Avaya User Group** *(*https://www.iuag.org*)*

  This is the organization for Avaya customers. It provides discussion groups and forums.

- **Avaya DevConnect** *(*https://www.devconnectprogram.com/*)*

  This site provides details on APIs and SDKs for Avaya products, including IP Office. The site also provides application notes for third-party non-Avaya products that interoperate with IP Office using those APIs and SDKs.

- **Avaya Learning** *(*https://www.avaya-learning.com/*)*

  This site provides access to training courses and accreditation programs for Avaya products.

**Related links**

Additional Help and Documentation on page 87

# Training

Avaya training and credentials ensure our Business Partners have the capabilities and skills to successfully sell, implement, and support Avaya solutions and exceed customer expectations. The following credentials are available:

- Avaya Certified Sales Specialist (APSS)
- Avaya Implementation Professional Specialist (AIPS)
- Avaya Certified Support Specialist (ACSS)

Credential maps are available on the Avaya Learning website.

**Related links**

Additional Help and Documentation on page 87

# Glossary

| | |
|---|---|
| **9600 series H.323 phones** | This term describes the 9600 series IP Deskphones running H.323 firmware. When running H.323 firmware, these phones are used as IP Office phones in a Distributed enterprise branch deployment. The following 9600 series phones can run H.323 firmware and are supported for use by IP Office users: 9620, 9630, 9640, 9650, 9608, 9611G, 9621G, and 9641G. |
| **9600 series SIP phone** | This term describes the 9600 series IP Deskphones running SIP firmware. When running SIP firmware, these phones are used as Centralized phones in a Centralized enterprise branch deployment. The following 9600 series phones can run SIP firmware and are supported for use by Centralized users: 9620, 9630, 9640, 9650, 9601, 9608, 9611G, 9621G, and 9641G. |
| **Branch office** | A geographic office location for an enterprise other than the main enterprise location. A branch office is typically smaller and has fewer employees than the main office for an enterprise. A branch office is involved in business activities related to the local market's needs. |
| **Centralized enterprise branch deployment option** | This term describes deployments where all users in a branch are Centralized users. See Centralized user. |
| **Centralized management** | This term is used to describe a central management system that delivers a set of shared management services and provides a single access interface to administer multiple branch locations and multiple distributed IP Office users. |
| **Centralized phone** | This term describes a phone that is used by a Centralized user. See Centralized user. |
| **Centralized trunking** | This term describes routing outgoing external calls from the branch sites to the central site in order to utilize the central sites PSTN trunks. The same applies for distributing incoming PSTN calls from the central site to the appropriate branches. |
| **Centralized user** | This term describes a user whose call processing is controlled by Avaya Aura® Communication Manager Feature Server or Evolution Server in the enterprise core. During normal operation, the Centralized user gets their telephony features and services from core applications such as the |

Communication Manager Feature Server or Evolution Server. Through the core Avaya Aura® Session Manager, the Centralized user can also access local PSTN trunks and services, such as local paging, local auto-attendant, and local Meet-me conferencing, on the IP Office in the branch. If WAN connectivity to the Avaya Aura®Session Manager is lost, the Centralized user automatically gets basic services from the local IP Office. When connection to Avaya Aura®Session Manager is available again, failback occurs either automatically or by manual administrative action where call processing for the Centralized user is returned to being controlled by Avaya Aura®Session Manager.

A Centralized user must be configured on the Avaya Aura®Session Manager, on Communication Manager, and on the IP Office. On the IP Office, the Centralized user must have either a SIP extension or an analog extension. There are two types of Centralized users:

- Centralized SIP user — a user configured as a Centralized user whose associated extension is a SIP extension.
- ATA user — a user configured as a Centralized user whose associated extension is an analog extension or analog fax device.

**Note:**

Standard analog phones and fax are supported for use by ATA users.

**Distributed enterprise branch deployment option**

This term describes deployments where all users in a branch are IP Office users. See IP Office user.

**Distributed trunking**

This term describes the scenario where each branch retains and uses its own PSTN trunks for incoming and outgoing external calls.

**E.164 format**

E.164 is a numbering format recommended by the International Telecommunications Union - Telecommunications (ITU-T). E.164 can have a maximum of 15 digits and is preceded by a +.

**Extension**

This term describes a unique number supported within the dial-plan that is assigned to a user. An extension also has associated endpoint(s) configured, where the endpoint can be either a hard device such as a telephone or a soft client running on a computer, mobile device, or tablet.

**Failback**

This term is used for the situation where a centralized extension that is working with a survivability call controller detects that its normal call controller is available again. The extension will go through a process of failback to its normal call controller.

**Failover**

This term is used for the situations where a centralized extension's preferred call controller is no longer available. The extension will go through a process of failover to the first available of its configured

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| | alternate call controllers which then provides survivability services to the extension. |
| **IP Office phone** | This term describes a phone that is used by an IP Office user. See IP Office user. |
| **IP Office user** | This term describes a user who gets their telephony features and services from the local IP Office. IP Office users were formerly referred to as distributed users, local users, or native users.<br>IP Office users with non-IP phones are connected to the IP Office while IP Office users with IP and SIP endpoints can be administered with IP Office as their controller. Access to and from the rest of the Avaya Aura® network is via the IP Office system's SM Line, which connects to Avaya Aura® Session Manager across the enterprise WAN. This connection allows for VoIP connectivity to other sites in the enterprise, to centralized trunking, and to centralized applications such as conferencing and messaging. |
| **Local management** | This term is used to describe managing an IP Office device using the local IP Office Manager application. |
| **Mixed enterprise branch deployment option** | This term describes deployments where there are Centralized users and IP Office users in a single branch. The Centralized users get their telephony services delivered by the Communication Manager Feature Server or Evolution Server in the core and the IP Office users get their telephony services delivered by the local IP Office. |
| **Mixed mode trunking** | The flexibility of Avaya Aura® Session Manager is such that both centralized and distributed trunking can be used. For example, routing all national and international calls via centralized trunking at the headquarters site while still allowing local calls via the branch sites. |
| **Network delay** | Network delay is measured in milliseconds (ms) and occurs one way between network endpoints. |
| **Network jitter** | Jitter is a measure of the variability in network delay between endpoints. |
| **Network packet loss** | Network packet loss is the maximum loss of packets or frames between network endpoints. |
| **PSTN** | Public Switched Telephone Network. The PSTN is the international telephone system. |
| **Rainy day** | This term refers to a loss of network connectivity from the branch to the core data center. |

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| **SM Line** | This term is used to describe a customized type of IP Office SIP trunk that is configured on the IP Office to connect to Avaya Aura® System Manager. |
| **Stand-alone IP Office branch option** | Independent IP Office systems are deployed within the network. These IP Office systems are not connected to each other or to anything else in the network. With this option, there is no Avaya Aura® system deployed in the network and users cannot access any Avaya Aura® services. |
| **Sunny day** | This term refers to full network connectivity from the branch to the core data center. |
| **Tail-End-Hop-Off** | Part of mixed mode trunking, this describes scenarios where certain calls at other branches or the headquarters site are routed to the PSTN of another branch. |

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