CHAPTER 6

Media Resources

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A media resource is a software-based or hardware-based entity that performs media processing functions on the data streams to which it is connected. Media processing functions include mixing multiple streams to create one output stream (conferencing), passing the stream from one connection to another (media termination point), converting the data stream from one compression type to another (transcoding), echo cancellation, signaling, termination of a voice stream from a TDM circuit (coding/decoding), packetization of a stream, streaming audio (annunciation), and so forth.

Use this chapter to determine which of the media resources described below are needed for your deployment. Also determine whether the resources that are required can be provided by software-based features or whether it is necessary to provision digital signal processors (DSPs) to implement the resources. Although the resources are discussed in individual sections, the same basic resources (DSPs and Cisco IP Voice Media Streaming Application) may be shared to implement higher-level functions.

This chapter focuses on the following features:

- Voice Termination, page 6-2
- Audio Conferencing, page 6-8
- Transcoding, page 6-11
- Media Termination Point (MTP), page 6-13
- Annunciator, page 6-18
- Cisco RSVP Agent, page 6-19
- Cisco IP Voice Media Streaming Application, page 6-19

For more details about the following features, refer to their respective sections:

- Music on Hold, page 7-1
- Unified CM RSVP-Enabled Locations, page 9-16

For details about the dependencies on hardware and software, see the section on Hardware and Software Capacities, page 6-20.

You can control media resources on Cisco Unified Communications Manager (Unified CM) through the use of media resource groups and media resource group lists. You can create pools of resources to control the specific hardware or software that is used. Cisco recommends that you use pools to group resources based on physical location. For design guidelines based on the various call processing models, see the section on General Design Guidelines, page 6-22.

What's New in This Chapter

Table 6-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 6-1 New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in:
Dynamic allocation of media termination points (MTPs)	When are Media Termination Points Required for Named Telephony Events?, page 6-14

Voice Termination

Voice termination applies to a call that has two call legs, one leg on a time-division multiplexing (TDM) interface and the second leg on a Voice over IP (VoIP) connection. The TDM leg must be terminated by hardware that performs coding/decoding and packetization of the stream. This termination function is performed by digital signal processor (DSP) resources residing in the same hardware module, blade, or platform. All DSP hardware on Cisco TDM gateways is capable of terminating voice streams, and certain hardware is also capable of performing other media resource functions such as conferencing or transcoding (see Audio Conferencing, page 6-8 and Transcoding, page 6-11).

Table 6-3 through Table 6-7 show the number of calls that each hardware platform can support, which is determined by the type of DSP chipset and the number of DSPs on the hardware. The hardware has either fixed DSP resources that cannot be upgraded or changed, or it has modular DSP resources that can be upgraded. For modular (upgradeable) hardware, Table 6-3 through Table 6-7 also list the maximum number of DSPs per hardware module.

The number of supported calls depends on the computational complexity of the codec used for a call and also on the complexity mode configured on the DSP. Cisco IOS enables you to configure a complexity mode on the hardware module. Some hardware platforms have two complexity modes, medium complexity and high complexity, while other hardware platforms have medium and high complexity as well as flex mode.

Medium and High Complexity Mode

To determine how many calls a module can support, locate the module in Table 6-3 through Table 6-7, and determine the number of DSPs it can provide as well as the desired codec type. For example, an NM-HD-2VE module with three C5510 DSPs configured in flex mode can support eight G.729 calls on each DSP, for a total of 24 calls using flex mode and G.729 codecs. Using G.711 codecs in flex mode, the same hardware could support 48 calls.

As indicated in Table 6-2, if a codec is supported in medium complexity mode, it is also supported in high complexity mode, albeit with fewer supported calls.

You can configure each DSP separately as either medium complexity, high complexity, or flex mode (C5510 only). The DSP treats all calls according to its configured complexity, regardless of the actual complexity of the codec of the call. A resource with configured complexity equal or higher than the actual complexity of the incoming call must be available, or the call will fail. For example, if a call requires a high-complexity codec but the DSP resource is configured for medium complexity mode, the call will fail. However, if a medium-complexity call is attempted on a DSP configured for high complexity mode, then the call will succeed and Cisco IOS will allocate a high-complexity mode resource.

To determine the maximum number of calls supported, find the appropriate row in Table 6-3 through Table 6-7 that contains the desired hardware. Check the medium and high complexity columns in Table 6-2 to verify which complexity modes are capable of handling the desired codecs. Then read the maximum number of supported calls for each DSP at the desired complexity mode.

Flex Mode

Flex mode, available only on hardware platforms that use the C5510 chipset, eliminates the requirement to specify the codec complexity at configuration time. A DSP in flex mode accepts a call of any supported codec type, as long as it has available processing power. The overhead of each call is tracked dynamically via a calculation of processing power in millions of instructions per second (MIPS). Cisco IOS performs a MIPS calculation for each call received and subtracts MIPS credits from its budget whenever a new call is initiated. The number of MIPS consumed by a call depends on the codec of the call, as shown in the Flex Mode column of Table 6-2. The DSP will allow a new call as long as it has remaining MIPS credits greater than or equal to the MIPS required for the incoming call. The Flex Mode column of Table 6-2 groups the supported codecs by the number of MIPS for a single call (either 15, 30, or 40 MIPS per call) and shows the MIPS budget available for various hardware.

Flex mode has an advantage when calls of multiple codecs must be supported on the same hardware because flex mode can support more calls than when the DSPs are configured as medium or high complexity. However, flex mode does allow oversubscription of the resources, which introduces the risk of call failure if all resources are used. With flex mode it is possible to have fewer DSP resources than with physical TDM interfaces.

For example, each DSP has a budget of 240 MIPS, for a total budget of 720 MIPS per NM-HD-2VE module. For an NM-HDV2 module, the budget per DSP is also 240 MIPS, but refer to Table 6-3 to determine the total number of MIPS available based on your choice and the number of PVDMs.

Compared to medium or high complexity mode, flex mode has the advantage of supporting the most G.711 calls per DSP. In medium complexity mode a DSP can support 8 G.711 calls, while flex mode supports 16 G.711 calls.

DSP Resources for Voice Termination

Table 6-3 through Table 6-7, organized by DSP chipset, provide information on DSP support by platform, DSP density, and the number of voice terminations (or calls) supported per DSP. Table 6-2 lists the codecs supported by the hardware modules in each complexity mode.

Table 6-2 Codecs Supported by Each Complexity Mode

Medium Complexity	High Complexity	Flex Mode
G.711 (a-law, mu-law) Fax/modem pass-through Clear channel G.726 (32K, 24K, 16K) GSM-FR Fax relay G.729 (a, ab)	G.711 (a-law, mu-law) Fax/modem pass-through Clear channel G.726 (32K, 24K, 16K) GSM-FR Fax relay G.729 G.729 (a, b, ab) G.728 G.723.1 (32K, 24K, 16K) G.723.1a (5.3K, 6.3K) GSM-EFR Modem relay	*At 15 MIPS per call:* • G.711 (a-law, mu-law) • Fax/modem pass-through • Clear channel *At 30 MIPS per call:* • G.726 (32K, 24K, 16K) • GSM-FR • Fax relay • G.729 • G.729 (a, b, ab) *At 40 MIPS per call:* • G.728 • G.723.1 (32K, 24K, 16K) • G.723.1a (5.3K, 6.3K) • GSM-EFR • Modem relay

Hardware based on the C5510 chipset supports medium and high complexity modes as well as flex mode, as indicated in Table 6-3.

Table 6-3 DSP Resources on Cisco IOS Hardware Platforms with C5510 Chipset

Hardware Module or Chassis	DSP Configuration	Maximum Number of Voice Terminations (Calls) per DSP and per Module		
		Medium Complexity (8 calls per DSP)	**High Complexity (6 calls per DSP)**	**Flex Mode[1] (240 MIPS per DSP)**
VG-224	Fixed at 4 DSPs	N/A	24 calls per platform Supported codecs: • G.711 (a-law, mu-law) • G.729a	N/A
NM-HD-1V[2]	Fixed at 1 DSP	4 calls per NM	4 calls per NM	240 MIPS per NM
NM-HD-2V	Fixed at 1 DSP	8 calls per NM	6 calls per NM	240 MIPS per NM
NM-HD-2VE	Fixed at 3 DSPs	24 calls per NM	18 calls per NM	720 MIPS per NM

Table 6-3 DSP Resources on Cisco IOS Hardware Platforms with C5510 Chipset (continued)

Hardware Module or Chassis	DSP Configuration	Maximum Number of Voice Terminations (Calls) per DSP and per Module		
		Medium Complexity (8 calls per DSP)	**High Complexity (6 calls per DSP)**	**Flex Mode[1] (240 MIPS per DSP)**
NM-HDV2 NM-HDV2-2T1/E1 NM-HDV2-1T1/E1	1 to 4 of: PVDM2-8[3] (½ DSP) PVDM2-16 (1 DSP) PVDM2-32 (2 DSPs) PVDM2-48 (3 DSPs) PVDM2-64 (4 DSPs)	Calls per PVDM: 4 8 16 24 32	Calls per PVDM: 3 6 12 18 24	MIPS per PVDM: 120 240 480 720 960
2801 2811	1 to 2 of: PVDM2-8[3] (½ DSP) PVDM2-16 (1 DSP) PVDM2-32 (2 DSPs) PVDM2-48 (3 DSPs) PVDM2-64 (4 DSPs)	Calls per PVDM: 4 8 16 24 32	Calls per PVDM: 3 6 12 18 24	MIPS per PVDM: 120 240 480 720 960
2821 2851	1 to 3 of: PVDM2-8[3] (½ DSP) PVDM2-16 (1 DSP) PVDM2-32 (2 DSPs) PVDM2-48 (3 DSPs) PVDM2-64 (4 DSPs)	Calls per PVDM: 4 8 16 24 32	Calls per PVDM: 3 6 12 18 24	MIPS per PVDM: 120 240 480 720 960
3825 3845	1 to 4 of: PVDM2-8[3] (½ DSP) PVDM2-16 (1 DSP) PVDM2-32 (2 DSPs) PVDM2-48 (3 DSPs) PVDM2-64 (4 DSPs)	Calls per PVDM: 4 8 16 24 32	Calls per PVDM: 3 6 12 18 24	MIPS per PVDM: 120 240 480 720 960

1. In flex mode, the maximum number of supported calls depends on the number of MIPS used per call (see Table 6-2).
2. With the NM-HD-1V module, the number of voice terminations (calls) is limited by the number of physical ports on the module.
3. The PVDM2-8 has a single half-capacity C5510.

Hardware that is based on the C5421 chipset may have DSPs configured as either medium complexity or high complexity. Table 6-4 lists the call density per DSP, and Table 6-2 lists the codecs supported in each complexity mode.

Table 6-4 DSP Resources on Cisco IOS Hardware Platforms with C5421 Chipset

Hardware Module	DSP Configuration	Maximum Number of Calls per DSP and per Module	
		Medium Complexity (8 calls per DSP)	**High Complexity (8 calls per DSP)**
NM-HDA-4FXS	Fixed at 2 DSPs *or* Fixed at 1 of DSP-HDA-16 (4 DSPs)	16 calls per NM 16 calls per NM	8 calls per NM 16 calls per NM
AIM-VOICE-30 AIM-ATM-VOICE-30	Fixed at 4 DSPs	30 or 60 calls per AIM	16 or 30 calls per AIM

Hardware that is based on the C549 chipset may have DSPs configured as either medium complexity or high complexity. Table 6-5 lists the call density per DSP, and Table 6-2 lists the codecs supported in each complexity mode.

Table 6-5 DSP Resources on Cisco IOS Hardware Platforms with C549 Chipset

Hardware Module	DSP Configuration	Maximum Number of Calls per DSP and per Module	
		Medium Complexity (4 calls per DSP)	**High Complexity (2 calls per DSP)**
NM-HDV NM-HDV-FARM	1 to 5 of PVDM-12 (3 DSPs per PVDM-12)	12, 24, 36, 48, or 60 calls per NM	6, 12, 18, 24, or 30 calls per NM
1751[1] 1760	1 to 2 of: PVDM-256K-4 (1 DSP) PVDM-256K-8 (2 DSPs) PVDM-256K-12 (3 DSPs) PVDM-256K-16HD (4 DSPs) PVDM-256K-20HD (5 DSPs)	Calls per NM: 4 or 8 8 or 16 12 or 24 16 or 32 20	Calls per NM: 2 or 4 4 or 8 6 or 12 8 or 16 10
 PA-VXA-1TE1-24+ PA-VXA-1TE1-30+ PA-VXB-2TE1+ PA-VXC-2TE1+	Fixed at: 7 DSPs 8 DSPs 12 DSPs 30 DSPs	Calls per PA: 28 32 48 120	Calls per PA: 14 16 24 60
PA-MCX-2TE1 PA-MCX-4TE1 PA-MCX-8TE1	Fixed (No on-board DSPs)	Depends on PA-VX(*x*)[2]	Depends on PA-VX(*x*)[2]

1. The 1751 supports a maximum of 8 DSPs (32 channels), and you can order these modules in multiples of 2 PVDMs as long as the total is less than 32 channels. The part number indicates the number of channels.
2. The multichannel port adaptors utilize unused DSPs from PA-VXAs, PA-VXBs, or PA-VXCs across the mix backplane.

Hardware that is based on the C542 chipset supports the following codecs:

- G.711 (a-law, mu-law)
- Fax/modem pass-through
- Clear channel
- G.726 (32K, 24K, 16K)
- GSM-FR
- Fax relay
- G.729
- G.729 (a, b, ab)
- G.728
- G.723.1 (32K, 24K, 16K)
- G.723.1a (5.3K, 6.3K)
- GSM-EFR
- Modem relay

Table 6-6 lists the call density per DSP.

Table 6-6 DSP Resources on Cisco IOS Hardware Platforms with C542 Chipset

Hardware Module[1]	DSP Configuration	Maximum Number of Calls per DSP and per Module
NM-1V	Fixed at 2 DSPs	1 call per DSP 2 calls per NM
NM-2V	Fixed at 4 DSPs	1 call per DSP 4 calls per NM

1. These modules do not have any complexity mode but support all codecs equally.

Table 6-7 lists non-IOS hardware for DSP resources. All non-IOS hardware platforms have a fixed configuration of DSPs, as indicated in Table 6-7.

Table 6-7 DSP Resources on Non-IOS Hardware Platforms

Hardware Module or Platform	DSP Configuration	Maximum Number of Calls per DSP and per Module	Codecs Supported
WS-6608-T1 WS-6608-E1	Fixed at 64 of C549 (8 DSPs per port)	2 calls per DSP 256 calls per module[1]	G.711 a-law, mu-law G.729a
WS-6624-FXS	Fixed at 12 of C549	2 calls per DSP 24 calls per module	G.711 a-law, mu-law G.729a
VG-248	Fixed at 12 of C5409	4 calls per DSP 48 calls per platform	G.711 a-law, mu-law G.729a
WS-SVC-CMM-ACT	Fixed at 4 of Broadcom 1500	32 calls per DSP 128 calls per module	G.711 (10-30 ms) G.729 (10-60 ms) G.723 (30-60 ms)

Table 6-7 DSP Resources on Non-IOS Hardware Platforms (continued)

Hardware Module or Platform	DSP Configuration	Maximum Number of Calls per DSP and per Module	Codecs Supported
WS-SVC-CMM-6T1	Fixed at 12 of C5441	15 calls per DSP 144 calls per module	G.711 (10, 20, 30 ms) G.729 (10, 20, 30, 40, 50, 60 ms)
WS-SVC-CMM-6E1	Fixed at 12 of C5441	15 calls per DSP 180 calls per module	G.711 (10, 20, 30 ms) G.729 (10, 20, 30, 40, 50, 60 ms)
WS-SVC-CMM-24FXS	Fixed at 3 of C5441	15 calls per DSP 24 calls per module	G.711 a-law, mu-law G.729 G.729a
ATA-188[2]	Fixed at 1 of Komodo 3880	2 calls per platform	G.711 a-law, mu-law G.729

1. A maximum of 192 calls are possible with T1 and 240 calls with E1, based on the number of physical ports. If none of the DSPs are configured for T1 or E1, then a maximum of 256 DSP resources are available.
2. The ATA module does not define complexity. It supports G.711, G.729, and G.723 only.

Audio Conferencing

A conference bridge is a resource that joins multiple participants into a single call. It can accept any number of connections for a given conference, up to the maximum number of streams allowed for a single conference on that device. There is a one-to-one correspondence between media streams connected to a conference and participants connected to the conference. The conference bridge mixes the streams together and creates a unique output stream for each connected party. The output stream for a given party is the composite of the streams from all connected parties minus their own input stream. Some conference bridges mix only the three loudest talkers on the conference and distribute that composite stream to each participant (minus their own input stream if they are one of the talkers).

Audio Conferencing Resources

A hardware conference bridge has all the capabilities of a software conference bridge. In addition, some hardware conference bridges can support multiple low bit-rate (LBR) stream types such as G.729, GSM, or G.723. This capability enables some hardware conference bridges to handle mixed-mode conferences. In a mixed-mode conference, the hardware conference bridge transcodes G.729, GSM, and G.723 streams into G.711 streams, mixes them, and then encodes the resulting stream into the appropriate stream type for transmission back to the user. Some hardware conference bridges support only G.711 conferences.

All conference bridges that are under the control of Cisco Unified Communications Manager (Unified CM) use Skinny Client Control Protocol (SCCP) to communicate with Unified CM.

Unified CM allocates a conference bridge from a conferencing resource that is registered with the Unified CM cluster. Both hardware and software conferencing resources can register with Unified CM at the same time, and Unified CM can allocate and use conference bridges from either resource. Unified CM does not distinguish between these types of conference bridges when it processes a conference allocation request.

The number of individual conferences that may be supported by the resource varies, and the maximum number of participants in a single conference varies, depending on the resource.

The following types of conference bridge resources may be used on a Unified CM system:

- Software Audio Conference Bridge (Cisco IP Voice Media Streaming Application), page 6-9
- Hardware Audio Conference Bridge (Cisco NM-HDV2, NM-HD-1V/2V/2VE, 2800 Series, and 3800 Series Routers), page 6-9
- Hardware Audio Conference Bridge (Cisco WS-SVC-CMM-ACT), page 6-10
- Hardware Audio Conference Bridge (Cisco NM-HDV and 1700 Series Routers), page 6-10
- Hardware Audio Conference Bridge (Cisco Catalyst WS-X6608-T1 and WS-X6608-E1), page 6-10
- Built-in Conference, page 6-11

Software Audio Conference Bridge (Cisco IP Voice Media Streaming Application)

A software unicast conference bridge is a standard conference mixer that is capable of mixing G.711 audio streams and Cisco Wideband audio streams. Any combination of Wideband or G.711 a-law and mu-law streams may be connected to the same conference. The number of conferences that can be supported on a given configuration depends on the server where the conference bridge software is running and on what other functionality has been enabled for the application. The Cisco IP Voice Media Streaming Application is a resource that can also be used for several functions, and the design must consider all functions together (see Cisco IP Voice Media Streaming Application, page 6-19).

Hardware Audio Conference Bridge (Cisco NM-HDV2, NM-HD-1V/2V/2VE, 2800 Series, and 3800 Series Routers)

DSPs that are configured through Cisco IOS as conference resources will load firmware into the DSPs that is specific to conferencing functionality only, and these DSPs cannot be used for any other media feature.

The following guidelines and considerations apply to these DSP resources:

- Based on the C5510 DSP chipset, the NM-HDV2 and the router chassis use the PVDM2 modules for providing DSPs.
- DSPs on PVDM2 hardware are configured individually as either voice termination, conferencing, media termination, or transcoding, so that DSPs on a single PVDM may be used as different resource types. Allocate DSPs to voice termination first, then to other functionality as needed.
- The NM-HDV2 has 4 slots that will accept PVDM2 modules in any combination. The other network modules have fixed numbers of DSPs.
- A conference based on these DSPs allows a maximum of 8 participants. When a conference begins, all 8 positions are reserved at that time.
- The PVDM2-8 is listed as having ½ a DSP because it has a DSP that has half the processing capacity of the PVDM2-16. For example, if the DSP on a PVDM2-8 is configured for G.711, it can provide (0.5 * 8) bridges/DSP = 4 conference bridges.
- Use Table 6-2 and Table 6-3 to determine how many DSPs may be provisioned with specific hardware.
- A DSP farm configuration in Cisco IOS specifies which codecs may be accepted for the farm. A DSP farm that is configured for conferencing and G.711 provides 8 conferences. When configured to accept both G.711 and G.729 calls, a single DSP provides 2 conferences because it is also reserving its resources for performing transcoding of streams.
- The I/O of an NM-HDV2 is limited to 400 streams, so ensure that the number of conference resources allocated does not cause this limit to be exceeded. If G.711 conferences are configured, then no more than 6 DSPs (total of 48 conferences with 8 participants each) should be allocated per

NM because (48 * 8) participants = 384 streams. If you configure all conferencing for both G.711 and G.729 codecs, then each DSP provides only 2 conferences of 8 participants each. In this case, it is possible to populate the NM fully and configure it with 16 DSPs so there would be 256 streams.

- Conferences cannot natively accept calls utilizing the GSM codec. A transcoder must be provided separately for these calls to participate in a conference.
- Any PVDM2-based hardware, such as the NM-HDV2, may be used simultaneously in a single chassis for voice termination but may not be used simultaneously for other media resource functionality. The DSPs based on PVDM-256K and PVDM2 have different DSP farm configurations, and only one may be configured in a router at a time.

Hardware Audio Conference Bridge (Cisco WS-SVC-CMM-ACT)

The following guidelines and considerations apply to this DSP resource:

- DSPs on this hardware are configured individually as either voice termination, conferencing, media termination, or transcoding, so that DSPs on a single module may be used as different resource types. Allocate DSPs to voice termination first.
- This Cisco Catalyst-based hardware provides DSP resources that may provide conference bridges of up to 32 participants per bridge.
- Each module contains 4 DSPs that are individually configurable. Each DSP can support 32 conference bridges.
- The G.711 and G.729 codecs are supported on these conference bridges without extra transcoder resources. However, transcoder resources would be necessary if other codecs are used.

Hardware Audio Conference Bridge (Cisco NM-HDV and 1700 Series Routers)

The following guidelines and considerations apply to these DSP resources:

- This hardware utilizes the PVDM-256K type modules that are based on the C549 DSP chipset.
- Conferences using this hardware provide bridges that allow up to 6 participants in a single bridge.
- The resources are configured on a per-DSP basis as conference bridges.
- The NM-HDV may have up to 5 PVDM-256K modules, while the Cisco 1700 Series Routers may have 1 or 2 PVDM-256K modules.
- Each DSP provides a single conference bridge that can accept G.711 or G.729 calls.
- The Cisco 1751 is limited to 5 conference calls per chassis, and the Cisco 1760 can support 20 conference calls per chassis.
- Any PVDM2-based hardware, such as the NM-HDV2, may be used simultaneously in a single chassis for voice termination but may not be used simultaneously for other media resource functionality. The DSPs based on PVDM-256K and PVDM2 have different DSP farm configurations, and only one may be configured in a router at a time.

Hardware Audio Conference Bridge (Cisco Catalyst WS-X6608-T1 and WS-X6608-E1)

The following guidelines and considerations apply to these DSP resources:

- This hardware has 8 DSPs that are physically associated to each port, and there are 8 ports per card. Configuration of the DSPs is at the port level, so that all DSPs associated to a port perform the same function.
- Conference bridges may have up to 32 participants, and each port supports 32 conference bridges.
- For conferences with G.711 or G.723 there may be 32 conferences per port. If G.729 calls are used, there may be 24 conferences per port.

Built-in Conference

Some phone models have a built-in conference resource that allows a three-way conference. This bridge is invoked only by the Barge feature and is not used as a general conferencing resource. For details on which phones have this bridge, see IP Telephony Endpoints, page 20-1. This bridge accepts only G.711 calls.

Transcoding

A transcoder is a device that converts an input stream from one codec into an output stream that uses a different codec. It may also connect two streams that utilize the same codec but with a different sampling rate. In a Unified CM system, the typical use of a transcoder is to convert between a G.711 voice stream and the low bit-rate compressed voice stream G729a. The following cases determine when transcoder resources are needed:

- Single codec for the entire system

  When a single codec is configured for all calls in the system, then no transcoder resources are required. The G.711 codec is supported by all vendors. A single-site deployment usually has no need for conserving bandwidth, and a single codec can be used. In this scenario, G.711 is the most common choice.

- Multiple codecs in use in the system, and all endpoints are capable of all codec types

  The most common reason for multiple codecs is to use G.711 for LAN calls to maximize the call quality and to use a low-bandwidth codec to maximize bandwidth efficiency for calls that traverse a WAN with limited bandwidth. Cisco recommends using G.729a as the low-bandwidth codec because it is supported on all Cisco Unified IP Phone models as well as most other Cisco Unified Communications devices, therefore it can eliminate the need for transcoding. Although Unified CM allows configuration of other low-bandwidth codecs between regions, the current phone models do not support those codecs and would require transcoders. They would require one transcoder for a call to a gateway and two transcoders if the call is to another IP phone. The use of transcoders is avoided if all devices support and are configured for both G.711 and G.729 because the devices will use the appropriate codec on a call-by-call basis.

- Multiple codecs in use in the system, and some endpoints support or are configured for G.711 only

  This condition exists when G.729a is used in the system but there are devices that do not support this codec, or a device with G.729a support may be configured to not use it. In this case, a transcoder is also required. Devices from some third-party vendors may not support G.729. Another example where G.729 is supported but might not be configured is with Cisco Unity. Cisco Unity has support for accepting calls with G.729a, but the codec is implemented in software and is CPU-intensive. Because as few as 10 simultaneous calls can cause significant CPU utilization, many deployments choose to disable G.729 on Cisco Unity and off-load the transcoding function to a dedicated transcoding resource external to the Unity server. If your system includes Cisco Unity, determine whether Unity will accept G.729a calls or whether it will be configured for G.711 only.

  Note Cisco Unified MeetingPlace Express currently supports G.711 only. In an environment where G.729 is configured for a call into Cisco Unified MeetingPlace Express, transcoder resources are required.

To finalize the design, it is necessary to know how many transcoders are needed and where they will be placed. If multiple codecs are needed, it is necessary to know how many endpoints do not support all codecs, where those endpoints are located, what other groups will be accessing those resources, how many maximum simultaneous calls these device must support, and where those resources are located in the network.

Transcoding Resources

DSP resources are required to perform transcoding. Those DSP resources can be located in the voice modules and the hardware platforms for transcoding that are listed in the following sections.

Hardware Transcoder (Cisco NM-HDV2, NM-HD-1V/2V/2VE, 2800, and 3800 Series Routers)

The following guidelines and considerations apply to these DSP resources:

- Transcoding is available between G.711 mu-law or a-law and G.729a or G.729ab. Each DSP can support 8 sessions.
- Cisco Unified IP Phones use only the G.729a variants of the G.729 codecs. The default for a new DSP farm profile is G.729a/G.729ab/G.711u/G.711a. Because a single DSP can provide only one function at a time, the maximum sessions configured on the profile should be specified in multiples of 8 to prevent wasted resources.
- Transcoding is also available between G.711mu-law/G.711a-law and G.729/G729b, but is not usually used in a Unified CM system. Each DSP can support 6 sessions.
- Refer to the section on DSP Resources for Voice Termination, page 6-4, to determine numbers of DSPs available on a particular platform or network module.

Hardware Transcoder (Cisco WS-SVC-CMM-ACT)

The following guidelines and considerations apply to this DSP resource:

- Transcoding is available between G.711 mu-law or a-law and G.729, G.729b, or G.723.
- There are 4 DSPs per ACT that may be allocated individually to DSP pools.
- The CCM-ACT can have 16 transcoded calls per DSP or 64 per ACT. The ACT reports resources as streams rather than calls, and a single transcoded call consists of two streams.

Hardware Transcoder (Cisco NM-HDV and 1700 Series Routers)

The following guidelines and considerations apply to these DSP resources:

- This hardware utilizes the PVDM-256K type modules that are based on the C549 DSP chipset.
- The NM-HDV may have up to 4 PVDM-256K modules, and the Cisco 1700 series routers may have 1 or 2 PVDM-256K modules.
- NM-HDV and NM-HDV2 modules may be used simultaneously in a single chassis for voice termination but may not be used simultaneously for other media resource functionality. Only one type of DSP farm configuration may be active at one time (either the NM-HDV or the HM-HDV2) for conferencing, MTP, or transcoding.
- Transcoding is supported from G.711 mu-law or a-law to any of G.729, G.729a, G.729b, G.729ab, or GSM codecs.
- Each DSP can provide 2 transcoding sessions.
- The Cisco 1751 has a chassis limit of 16 sessions, and the Cisco 1760 has a chassis limit of 20 sessions.

Hardware Transcoder (Cisco WS-X6608)

The following guidelines and considerations apply to this DSP resource:

- DSPs are allocated to functions at a port level. Each port can provide 24 transcoding sessions.
- There are 8 ports per blade.
- Transcoding is available between G.711 mu-law or a-law and G.729a, G.729ab, G.729, or G.729b.

A transcoder is also capable of performing the same functionality as a media termination point (MTP). In cases where transcoder functionality and MTP functionality are both needed, a transcoder is allocated by the system. If MTP functionality is required, Unified CM will allocate either a transcoder or an MTP from the resource pool, and the choice of resource will be determined by the media resource groups, as described in the section on Media Resource Groups and Lists, page 6-23.

Media Termination Point (MTP)

A media termination point (MTP) is an entity that accepts two full-duplex streams. It bridges the media streams together and allows them to be set up and torn down independently. The streaming data received from the input stream on one connection is passed to the output stream on the other connection, and vice versa. MTPs have many possible uses, as described in the following sections.

Re-Packetization of a Stream

An MTP can be used to transcode a-law to mu-law and vice versa, or it can be used to bridge two connections that utilize different packetization periods (different sample sizes).

H.323 Supplementary Services

MTPs can be used to extend supplementary services to H.323 endpoints that do not support the H.323v2 OpenLogicalChannel and CloseLogicalChannel request features of the Empty Capabilities Set (ECS). This requirement occurs infrequently. Cisco H.323 endpoints support ECS, and most third party endpoints have support as well. When needed, an MTP is allocated and connected into a call on behalf of an H.323 endpoint. Once inserted, the media streams are connected between the MTP and the H.323 device, and these connections are present for the duration of the call. The media streams connected to the other side of the MTP can be connected and disconnected as needed to implement features such as hold, transfer, and so forth.

When an MTP is required on an H.323 call and none is available, the call will proceed but will not be able to invoke supplementary services.

H.323 Outbound Fast Start

H.323 defines a feature called Fast Start, which reduces the number of packets exchanged during a call setup, thereby reducing the amount of time for media to be established. It is useful when two devices that are utilizing H.323 have high network latency between them because the time to establish media is dependant on that latency. Unified CM distinguishes between inbound and outbound fast start based on the direction of the call setup, and the distinction is important because the MTP requirements are not equal. For inbound fast start, no MTP is required. Outbound calls on an H.323 trunk do require an MTP when fast start is enabled. Frequently it is only inbound calls that are problematic, and it is possible to use inbound fast start to solve the issue without also enabling outbound fast start.

Named Telephony Events (RFC 2833)

DTMF tones can be used during a call to signal to a far-end device for purposes of navigating a menu system, entering data, or other manipulation. They are processed differently than DTMF tones sent during a call setup as part of the call control.

Named Telephony Events (NTEs) defined by RFC 2833 are a method of sending DTMF from one endpoint to another after the call media has been established. The tones are sent as packet data using the already established RTP stream and are distinguished from the audio by the RTP packet type field. For example, the audio of a call can be sent on a session with an RTP packet type that identifies it as G.711 data, and the DTMF packets are sent with an RTP packet type that identifies them as NTEs. The consumer of the stream utilizes the G.711 packets and the NTE packets separately.

When are Media Termination Points Required for Named Telephony Events?

For Cisco Unified CM 4.x, only SIP trunks are supported, and they require that an MTP be allocated for all calls using the SIP trunk. The SIP trunk has a configuration parameter, "MTP required," which is selected by default and cannot be changed.

Cisco Unified CM 5.0 adds support for SIP on line devices and also removes the requirement that MTPs be allocated for all calls using SIP trunks. In Unified CM 5.0, an MTP is required when two endpoints do not have a method in common for sending DTMF between them or when the system configuration specifies that one must be allocated. The following description expands on the details and applies only to Unified CM 5.0.

Verify the types of endpoints planned for the system based on the following rules:

1. Two non-SIP endpoints do not require MTPs.

   All Cisco Unified Communications endpoints other than SIP send DMTF to Unified CM via various signaling paths, and Unified CM forwards the DTMF between dissimilar endpoints. For example, an IP phone may use SCCP messages to Unified CM to send DTMF, which then gets sent to an H.323 gateway via H.245 signaling events. The two endpoints have a common method by sending DTMF to Unified CM. If no SIP endpoints are used, stop here.

2. Two Cisco SIP endpoints do not require MTPs.

   All Cisco SIP endpoints support NTEs, therefore MTPs can be avoided entirely. DTMF is sent directly between the endpoints using NTE. If all endpoints are Cisco SIP devices, then no MTPs are required for DTMF conversion.

3. A combination of a SIP endpoint and a non-SIP endpoint might require MTPs.

   To determine the support for NTE in your devices, see Table 6-8. RFC 2833 is not limited to SIP and can be supported in devices with other call control protocols. For example, Cisco Unified IP Phones that run either an SCCP or SIP stack can support NTEs in both modes. Some devices support DTMF via multiple methods (for example, a Cisco Unified IP Phone 7960 with an SCCP stack can send NTEs to the other device or SCCP to Unified CM). Other devices such as the Cisco Wireless IP Phone 7920 can send only SCCP, and still others can send only NTE (such as the Cisco Unified IP Phone 7960 with a SIP stack). Unified CM has the ability to allocate MTPs dynamically on a call-by-call basis, based on the capabilities of the pair of endpoints. Use Table 6-8 to determine whether MTPs should be provisioned.

4. Dynamic allocation of MTPs is only for outbound calls.

   A SIP trunk that does not require MTPs will use dynamic allocation as described in rule 3. The SIP trunk is capable of accepting calls with or without Session Description Protocol (SDP) in the Invite, therefore it will not allocate an MTP for an inbound call. Only outbound calls may have an MTP allocated as described above. If a particular SIP trunk is used for inbound calls only, then it not necessary to allocate any MTP resources on the system for receiving calls on that trunk.

5. MTPs may be forced by the configuration.

   In Cisco Unified CM 5.0, the SIP trunk parameter **MTP Required** is not selected by default, and the field is unlocked. When a SIP trunk is defined for a Cisco device (such as a SIP gateway), the device should be configured for NTE support. The default setting for MTP Required should normally be used, and MTPs are allocated only when needed.

   There are other reasons for forcing an MTP. The establishment of a SIP dialog behaves differently when an MTP is allocated before call setup starts. SIP uses Session Description Protocol (SDP) for establishing the parameters of a session, and SDP is embedded into SIP messages. When an MTP is forced, the SDP is sent with the Invite message that initiates the call. When an MTP is not forced, it is allocated (if needed) after the Invite message. The Invite message does not contain SDP, and it is sent at a later point in the call setup. If the far-end device supports only Invite messages with embedded SDP, then the MTP Required parameter must be checked (enabled).

   When this configuration parameter forces an MTP, the codec used on the call leg from the MTP to the far-end SIP device must be G.711 mu-law or a-law. There is an additional parameter on the trunk configuration that becomes unlocked when MTP Required is selected. This parameter allows a choice of which variant of G.711 to use. When an MTP is not forced by this configuration parameter, then the normal mechanisms for determining the codec apply. In these cases, an MTP may be inserted dynamically for DTMF relay or other purposes and will not impose any restriction on the codec negotiated.

Table 6-8 Endpoint Support for DTMF Methods

Endpoint Protocol Stack: Endpoints	DTMF Method		
	SCCP	**NTE**	**KPML**[1]
SCCP Stack: 12SP+, 30 VOIP, 7910, 7920, 7935, 7936 VG248, DPA-7610, DPA-7630, CTI Port, First Party Control	Yes	No	No
SCCP Stack: 7902, 7905, 7912,7940, 7941, 7960, 7961, 7970 Future new phone models	Yes	Yes	No

Table 6-8 Endpoint Support for DTMF Methods (continued)

Endpoint Protocol Stack: Endpoints	DTMF Method		
	SCCP	**NTE**	**KPML**[1]
SIP Stack: 7905, 7911, 7912, 7940, 7941, 7960, 7961, 7970 Future new phone models	No	Yes	Yes for: 7911, 7941, 7961, 7970 No for: 7905, 7912, 7940, 7960

1. Key Press Markup Language (KPML)

Note IP Phones play DTMF to the end user when DTMF is received via SCCP, but they do not play tones received by NTE. However, there is no requirement to send DTMF to another end user. It is necessary only to consider the endpoints that originate calls combined with endpoints that might need DTMF, such as PSTN gateways, application servers, and so forth.

Example 6-1 Call Flow that Requires an MTP for NTE Conversion

Assume the example system has CTI route points with first-party control (the CTI port terminates the media), which integrate to a system that uses DTMF to navigate an IVR menu. If all phones in the system are running SCCP, then no MTP is required. In this case Unified CM controls the CTI port and receives DTMF from the IP phones via SCCP. Unified CM provides DTMF conversion.

However, if there are phones running a SIP stack, an MTP is required. NTEs are part of the media stream, therefore Unified CM does not receive them. An MTP is invoked into the media stream and has one call leg that uses SCCP, and the second call leg uses NTEs. The MTP is under SCCP control of Unified CM and performs the NTE-to-SCCP conversion under the control of Unified CM.

Configuration of DTMF on SIP and H.323 Gateways

Configure either **sip-notify** or **rtp-nte** as a method under a SIP dial peer. The best configuration of a SIP dial peer will depend on the mix of endpoints that exist in the system (see Table 6-8). When Unsolicited Notify is used on the gateway, that dial peer may be used by endpoints that support SCCP without the need for MTP resources. Endpoints that support only NTE must invoke an MTP to use the gateway.

Conversely, if NTE is configured on the SIP dial peer, then any endpoint that can use NTE can send DTMF directly to the gateway without need for MTPs. An endpoint that supports only SCCP will need to invoke an MTP.

SIP gateways have the ability to send DTMF using NTEs, Unsolicited Notify, or audio tones in the media stream. Unsolicited Notify is a Cisco proprietary method that sends a SIP Notify message with an event that contains the DTMF tone. This method is also supported on Unified CM.

The following example lists a SIP gateway configuration for Named Telephony Events:

```
dial-peer voice 10 voip
dtmf-relay rtp-nte
```

H.323 gateways have support for H.245 Alphanumeric, H.245 Signal, NTE, and audio in the media stream. The NTE option must not be used because it is not supported on Unified CM for H.323 at this time. The preferred option is H.245 signal. MTPs are required for establishing calls to an H.323 gateway if the other endpoint does not have signaling capability in common with Unified CM. For example, a Cisco Unified IP Phone 7960 running the SIP stack supports only NTEs, so an MTP is needed with an H.323 gateway.

The following configuration is recommended for DTMF methods on an H.323 gateway:

```
dial-peer voice 10 voip
dtmf-relay h.245-signal
```

CTI Route Points

CTI Route Points that have first-party control of a phone call will participate in the media stream of the call and require an MTP to be inserted. When the CTI has third-party control of a call such that the media passes through a device that is controlled by the CTI, then the requirement for an MTP is dependant on the capabilities of the controlled device.

MTP Resources

The following types of devices are available for use as an MTP:

Software MTP (Cisco IP Voice Media Streaming Application)

A software MTP is a device that is implemented by installing the Cisco IP Voice Media Streaming Application on a server. When the installed application is configured as an MTP application, it registers with a Unified CM node and informs Unified CM of how many MTP resources it supports. A software MTP device supports only G.711 streams. The IP Voice Media Streaming Application is a resource that may also be used for several functions, and the design guidance must consider all functions together (see Cisco IP Voice Media Streaming Application, page 6-19).

Software MTP (Based on Cisco IOS)

- The capability to provide a software-based MTP on the router is available beginning with Cisco IOS Release 12.3(11)T for the Cisco 3800 Series Routers and Release 12.3(8)T4 for other router models.
- This MTP allows configuration of any of the following codecs, but only one may be configured at a given time: G.711 mu-law and a-law, G.729a, G.729, G.729ab, G.729b, GSM, and pass-through. Some of these are not pertinent to a Unified CM implementation.
- The router configuration permits up to 500 individual streams, which support 250 transcoded sessions. This number of G.711 streams generates 5 Mbytes of traffic.

Hardware MTP (Cisco NM-HDV2, NM-HD-1V/2V/2VE, 2800 and 3800 Series Routers)

- This hardware uses the PVDM-2 modules for providing DSPs.
- Each DSP can provide 16 G.711 mu-law or a-law MTP sessions or 6 G.729, G.729b, or GSM MTP sessions.

Hardware MTP (Cisco WS-SVC-CMM-ACT)

- This module has 4 DSPs that may be configured individually.
- Each DSP can support 128 G.729, G.729b, or GSM MTP sessions or 256 G.711 mu-law or a-law MTP sessions.

Hardware MTP (Catalyst WS-X6608-T1 and WS-X6608-E1)

- Codec support is G.711 mu-law or a-law, G.729, G.729b, or GSM.
- Configuration is done at the port level. Eight ports are available per module.
- Each port configured as an MTP resources provides 24 sessions.

Annunciator

An annunciator is a software function of the Cisco IP Voice Media Streaming Application that provides the ability to stream spoken messages or various call progress tones from the system to a user. It is capable of sending multiple one-way RTP streams to devices such as Cisco IP phones or gateways, and it uses SCCP messages to establish the RTP stream. The device must be capable of SCCP to utilize this feature. Tones and announcements are predefined by the system. The announcements support localization and also may be customized by replacing the appropriate .wav file. The annunciator is capable of supporting G.711 a-law and mu-law, G.729, and Wideband codecs without any transcoding resources.

The following features require an annunciator resource:

- Cisco Multilevel Precedence Preemption (MLPP)

  This feature has streaming messages that it plays in response to the following call failure conditions.

  - Unable to preempt due to an existing higher-precedence call.
  - A precedence access limitation was reached.
  - The attempted precedence level was unauthorized.
  - The called number is not equipped for preemption or call waiting.

- Integration via SIP trunk

  SIP endpoints have the ability to generate and send tones in-band in the RTP stream. Because SCCP devices do not have this ability, an annunciator is used in conjunction with an MTP to generate or accept DTMF tones when integrating with a SIP endpoint. The following types of tones are supported:

  - Call progress tones (busy, alerting, and ringback)
  - DTMF tones

- Cisco IOS gateways and intercluster trunks

  These devices require support for call progress tone (ringback tone).

- System messages

  During the following call failure conditions, the system plays a streaming message to the end user:

  - A dialed number that the system cannot recognize
  - A call that is not routed due to a service disruption
  - A number that is busy and not configured for preemption or call waiting

- Conferencing

  During a conference call, the system plays a barge-in tone to announce that a participant has joined or left the bridge.

An annunciator is automatically created in the system when the Cisco IP Voice Media Streaming Application is activated on a server. If the Media Streaming Application is deactivated, then the annunciator is also deleted. A single annunciator instance can service the entire Unified CM cluster if it meets the performance requirements (see Annunciator Performance, page 6-19); otherwise, you must configure additional annunciators for the cluster. Additional annunciators can be added by activating the Cisco IP Voice Media Streaming Application on other servers within the cluster.

The annunciator registers with a single Unified CM at a time, as defined by its device pool. It will automatically fail over to a secondary Unified CM if a secondary is configured for the device pool. Any announcement that is playing at the time of an outage will not be maintained.

An annunciator is considered a media device, and it can be included in media resource groups (MRGs) to control which annunciator is selected for use by phones and gateways.

Annunciator Performance

By default, the annunciator is configured to support 48 simultaneous streams, which is the maximum recommended for an annunciator running on the same server (co-resident) with the Unified CM service. If the server has only 10 Mbps connectivity, lower the setting to 24 simultaneous streams.

A standalone server without the Cisco CallManager Service can support up to 255 simultaneous announcement streams, and a high-performance server with dual CPUs and a high-performance disk system can support up to 400 streams. You can add multiple standalone servers to support the required number of streams.

Cisco RSVP Agent

In order to provide topology-aware call admission control, Unified CM invokes one or two RVSP Agents during the call setup to perform an RSVP reservation across the IP WAN. These agents are MTP or transcoder resources that have been configured to provide RSVP functionality. RSVP resources are treated the same way as regular MTPs or transcoders from the perspective of allocation of an MTP or transcoder resource by Unified CM.

The Cisco RSVP Agent feature was first introduced in Cisco IOS Release 12.4(6)T. For details on RSVP and Cisco RSVP Agents, refer to the chapter on Call Admission Control, page 9-1.

Cisco IP Voice Media Streaming Application

The Cisco IP Voice Media Streaming Application provides the following resources in software:

- Music on Hold (MoH)
- Annunciator
- Software conference bridge
- Media termination point (MTP)

When the Media Streaming Application is activated, one of each of the above resources is automatically configured. If the annunciator, software conference bridge, or MTP are not needed, Cisco recommends that you disable them by disabling the Run Flag service parameter for the Cisco IP Voice Media Streaming Application.

Give careful consideration to situations that require multiple resources and to the load they place on the Media Streaming Application. Each resource has a service parameter that controls the maximum number of connections it can handle, with an associated default setting. You may run limited amounts of all four resources on the same server if you have not changed the default settings. However, if your deployment requires more than the default number of any one resource, configure that resource to run on its own dedicated server (without any other resources or the Cisco CallManager Service running on that server).

The annunciator is the only media resource that is available only through the IP Voice Media Streaming Application. Conferencing, MTPs, and music on hold (MoH) can all reside on servers other than the Unified CM servers. Cisco highly recommends that you disable the MTP and conferencing resources on Unified CM and that you provide external dedicated resources for this functionality.

Cisco also strongly recommends that you run the IP Voice Media Streaming Application on a server other than the publisher or any Unified CM server that provides call processing. The increase in CPU load for media resources might adversely impact call processing performance, and security issues can arise because User Datagram Protocol (UDP) traffic must be received on the Unified CM server.

Hardware and Software Capacities

This section provides data on capacities of the network modules and chassis that carry DSPs, the capacities of the chassis to carry network modules, and software dependencies of the hardware.

PVDMs

Table 6-9 and Table 6-10 show the number of DSPs that can reside on the two models of PVDMs or on fixed-configuration network modules. The PVDM2-*xx* modules are newer than the PVDM-256K-*xx* modules, and the two types are not interchangeable.

Table 6-9 Number of DSPs per PVDM-256K Module

Module	Number of DSPs
PVDM-256K-4	1 DSP
PVDM-256K-8	2 DSPs
PVDM-256K-12	3 DSPs
PVDM-256K-16HD	4 DSPs
PVDM-256K-20HD	5 DSPs

Table 6-10 Number of DSPs per PVDM2 Module or Fixed-Configuration Hardware

Hardware Module or Chassis	Number of DSPs
PVDM2-8	½ DSP
PVDM2-16	1 DSP
PVDM2-32	2 DSPs
PVDM2-48	3 DSPs
PVDM2-64	4 DSPs
NM-HD-1V NM-HD-2V	1 DSP
NM-HD-2VE	3 DSPs

Table 6-11 specifies minimum versions of Cisco IOS software needed to support media resource functionality on these hardware platforms and network modules.

Table 6-11 PVDM2 Slots Available and Cisco IOS Versions Required for Media Support

Chassis or Network Module	Number of PVDM2 Slots	Minimum Cisco IOS Release for Media
2801	2	12.3(11)T
2811	2	12.3(8)T4
2821 or 2851	3	12.3(8)T4
3825 or 3845	4	12.3(11)T
NM-HDV2	4	

Cisco 2800 and 3800 Series Platforms

Although the Cisco 2800 and 3800 Series Routers all have two AIM slots, they do not support the AIM-VOICE-30 or AIM-ATM-VOICE-30 cards because the functionality of these cards is provided instead by PVDM2 modules that are installed on the motherboard.

Network Modules

You can install the NM-HDV2, NM-HD-*xx*, and NM-HDV modules in the Cisco IOS platforms listed in Table 6-12, up to the maximum numbers indicated.

All three families of modules in Table 6-11 may be installed in a single chassis. However, the conferencing and transcoding features cannot be used simultaneously on both the NM-HDV family and either of the other two families (NM-HD-*xx* or NM-HDV2). In addition, the NM-HDV (TI-549), NM-HD-*xx*, and NM-HDV2 (TI-5510) cannot be used simultaneously for conferencing and transcoding within a single chassis.

You can mix NM-HDV and NM-HDV-FARM modules in the same chassis. Not all chassis can be completely populated by these modules. Table 6-11 shows the maximum number of modules that each type of hardware platform can support.

Table 6-12 Number of Module Slots per Platform Type

Cisco IOS Platform	Number of Slots
2691, 2811, 2821, 2851	1
3620[1], 3725, 3825	2
3640	3
3745, 3845	4
3660	6

1. Although it has two NM slots, the Cisco 3620 Router supports only a single NM-HDV module.

Table 6-13 Modules Supported by Platform Type

Cisco IOS Platform	Modules Supported by Platform		
	NM-HDV2	NM-HD-1V NM-HD-2V NM-HD-2VE	NM-HDV NM-HDV-FARM
VG200[1] 2600[2] 3620[3] 3640	No	No	Yes
3660	No	Yes	Yes
2600XM, 2691, 3725, 3745 2811, 2821, 2851 3825, 3845	Yes	Yes	Yes

Note The Cisco VG200, 2620, 2621, and 3620 do not support the NM-HDV-FARM, nor do they support MTP, conferencing, or transcoding. The Cisco 2801 has no NM slot.

Calculating DSP Requirements for the NM-HDV

There are specific situations where the NM-HDV may not be fully populated. The sampling rate is not normally changed from the system defaults. If you do not require the sampling rate to be changed, then this issue may be ignored.

For sample rates of 20, 30, 40, or 60 ms with voice activity detection (VAD) enabled or disabled (or 10 ms with VAD enabled), it is possible to configure an NM-HDV or NM-HDV-FARM with a full complement of 5 PVDMs, giving 60 usable DSP resources.

For a sampling rate of 10 ms with VAD disabled, it is not possible to utilize all DSPs on a fully populated NM-HDV. The following additional calculation is required to ensure that the packet rate does not exceed 6600 packets per second (pps), which is the capacity of the NM-HDV:

> 100 pps (number of voice terminations) + 600 pps (number of conferences) + 200 pps (number of transcoding sessions) < 6600 pps

General Design Guidelines

The Unified CM constructs of media resource groups (MRGs) and media resource group lists (MRGLs) are used to control how the resources described in this chapter are organized and accessed. This section discusses considerations for how to utilize these constructs effectively. It also discusses specific considerations for the various Unified CM deployment models.

Media Resource Groups and Lists

Media resource groups and lists provide a method to control how resources are allocated that could include rights to resources, location of resources, or resource type for specific applications. This section assumes you have an understanding of media resource groups, and it highlights the following design considerations:

- The system defines a default media resource group that is not visible in the user interface and that all resources are members of when they are created. Consumers of media resources use resources first from any media resource group (MRG) or media resource group list (MRGL) that their configuration specifies. If the required resource is not available, the default MRG is searched for the resource. For simple deployments, the default MRG alone may be used.
- When using MRGs to control access to resources, it is necessary to move the resources out of the default MRG by explicitly configuring them in some other group. If the desired effect is for resources to be available only as a last resort for all calls, then the resources may remain in the default group. Also, if no control over resources is necessary, they may remain in the default group.
- An MRG may contain multiple types of resources, and the appropriate resource will be allocated from the group based on the feature needed. MTPs and transcoders are a special case because a transcoder may also be used as an MTP.
- One use of MRGs is to group resources of similar types. Conference bridge resources vary in the number of participants they support, and MRGs could be used to group the conference resources by conference bridge size.
- Use media resource groups (MRGs) and media resource group lists (MRGLs) to provide sharing of resources across multiple Unified CMs. If you do not use MRGs and MRGLs, the resources are available to a single Unified CM only.
- You can also use MRGs and MRGLs to separate resources based on geographical location, thereby conserving WAN bandwidth whenever possible.
- MRGLs will use MRGs in the order that they are listed in the configuration. If one MRG does not have the needed resource, the next MRG is searched. If all MRGs are searched and no resource is found, the search terminates.
- The algorithm for allocating like resources from an MRG attempts to spread the load across the like resources. When a resource has been used, a pointer for that MRG is incremented to the next device. A device may be present in more than one MRG, which will affect the pointers of all groups in which the device is a member. When MTPs are required and transcoders exist in the same group, the MTPs are always allocated until all MTPs are used, and then transcoders are used as MTPs. When resources of differing capacities exist in the same group, the load sharing attempts to allocate resources based on the capacity. The system will spread the load across resources, but because of the above factors, it frequently will not be round-robin in behavior.
- Unified CM Administration displays the devices in an MRG in alphabetical order, but they are allocated based on their order in the configuration database. You cannot change this order. If you want media resources to be allocated in a specific order, then create a separate MRG for each individual resource and use MRGLs to specify the order of allocation.
- Ensure that the media resources themselves have configurations that prevent further invocation of other media resources. For example, if an MTP is inserted into a call and the codec configured on that MTP does not match the one needed by Unified CM for the call, then a transcoder may also be invoked. A frequent mistake is to configure an MTP for G.729 or G.729b when Unified CM needs G.729a.

Deployment Models

This section examines where and when the MTP and transcoding resources are used within the following three enterprise IP Telephony deployment models plus a fourth application scenario:

- Single-Site Deployments, page 6-24, consist of one or more call processing agents within a single site, with no voice traffic carried over the IP WAN.
- Multisite WAN Deployments with Centralized Call Processing, page 6-24, consist of a single call processing agent that provides service for many sites connected through an IP WAN.
- Multisite WAN Deployments with Distributed Call Processing, page 6-25, consist of call processing agents residing at each of several remote sites connected through an IP WAN.
- IP PSTN Access, page 6-26, is another scenario that requires MTP resources, and it applies to all of the preceding deployment models.

Single-Site Deployments

In a single-site deployment, there is no need for transcoding because there are no low-speed links to justify the use of a low bit-rate (LBR) codec. Some MTP resources might be required in the presence of a significant number of devices that are not compliant with H.323v2, such as older versions of Microsoft NetMeeting or certain video devices. MTP resources may be required for DTMF conversion if SIP endpoints are present (see Named Telephony Events (RFC 2833), page 6-14.)

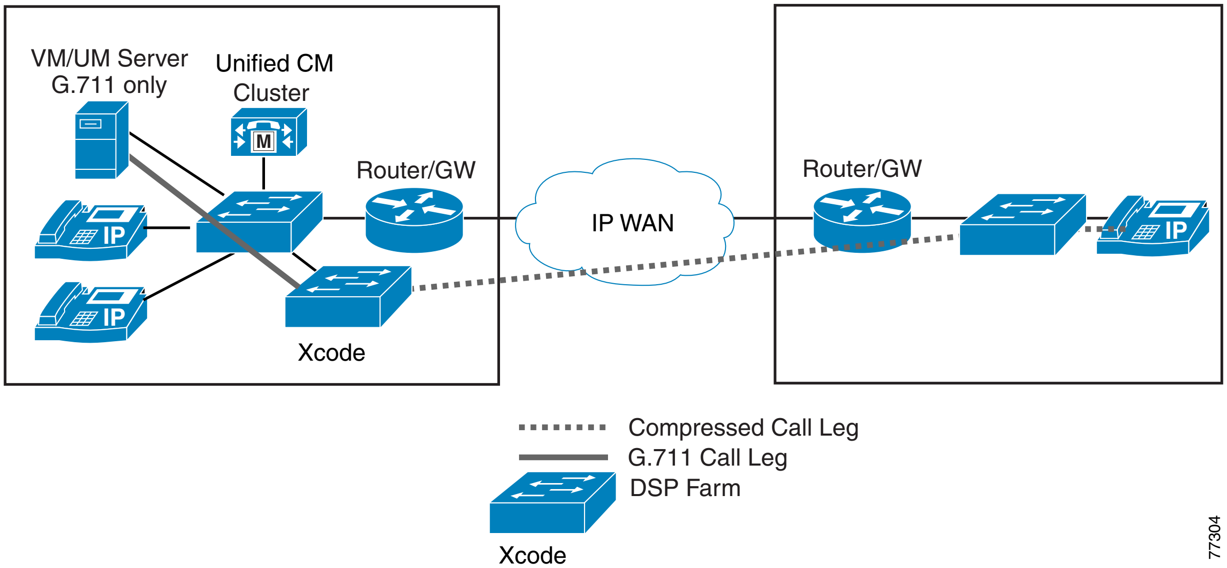
Multisite WAN Deployments with Centralized Call Processing

In a centralized call processing deployment, the Unified CM cluster and the applications (such as voice mail and IVR) are located at the central site, while several remote sites are connected through an IP WAN. The remote sites rely on the centralized Unified CMs to handle their call processing.

Because WAN bandwidth is typically limited, calls are configured to use a low bit-rate codec such as G.729 when traversing the WAN. (See Figure 6-1.)

Voice compression between IP phones is easily configured through the use of *regions* and *locations* in Unified CM. A region defines the type of compression (for example, G.711 or G.729) used by the devices in that region, and a location specifies the total amount of bandwidth available for calls to and from devices at that location.

Figure 6-1 Transcoding for the WAN with Centralized Call Processing



Unified CM uses media resource groups (MRGs) to enable sharing of MTP and transcoding resources among the Unified CM servers within a cluster. In addition, when using an LBR codec (for example, G.729a) for calls that traverse different regions, the transcoding resources are used only if one (or both) of the endpoints is unable to use the LBR codec.

In Figure 6-1, Unified CM knows that a transcoder is required and allocates one based on the MRGL and/or MRG of the device that is using the higher-bandwidth codec. In this case it is the VM/UM server that determines which transcoder device is used. This behavior of Unified CM is based on the assumption that the transcoder resources are actually located close to the higher-bandwidth device. If this system was designed so that the transcoder for the VM/UM server was located at the remote site, then G.711 would be sent across the WAN, which would defeat the intended design. As a result, if there are multiple sites with G.711-only devices, then each of these sites would need transcoder resources when an LBR is run on the WAN.

The placement of other resources is also important. For example, if a conference occurs with three phones at a remote site and the conference resource is located in the central (call processing) site, then three media streams are carried over the WAN. If the conference resource were local, then the calls would not traverse the WAN. It is necessary to consider this factor when designing the bandwidth and call admission control for your WAN.

Multisite WAN Deployments with Distributed Call Processing

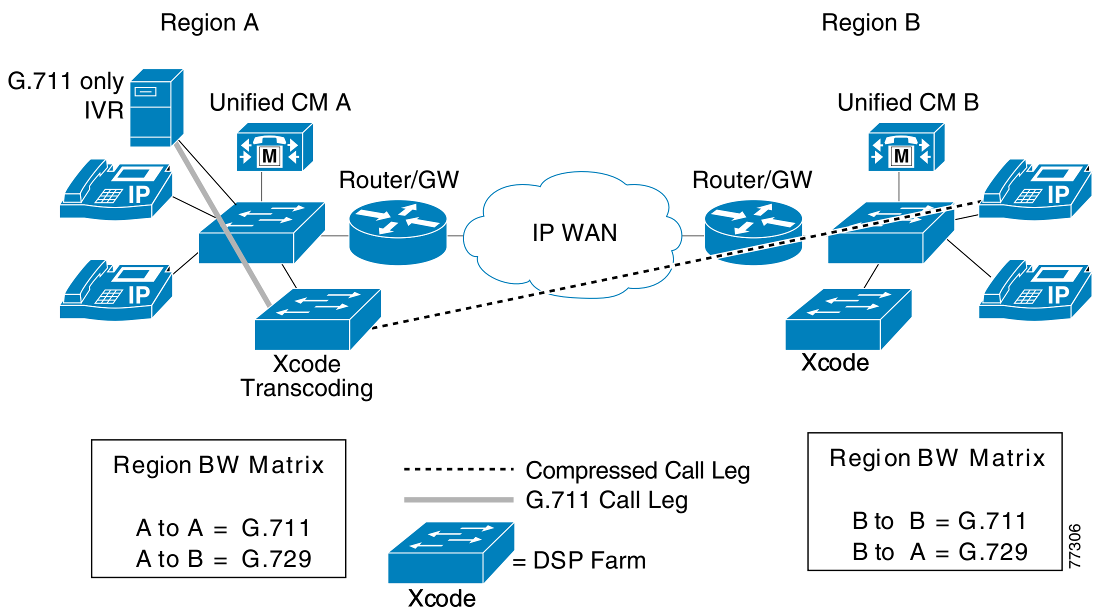
In distributed call processing deployments, several sites are connected through an IP WAN. Each site contains a Unified CM cluster that can, in turn, follow the single-site model or the centralized call processing model. A gatekeeper may be used for call admission control between sites.

Because WAN bandwidth is typically limited, calls between sites may be configured to use an LBR codec (such as G.729a) when traversing the WAN. H.323v2 intercluster trunks are used to connect Unified CM clusters. Unified CM also supports compressed voice call connections through the MTP service if a hardware MTP is used. (See Figure 6-2.)

A distributed call processing deployment might need transcoding and MTP services in the following situations:

- With current versions of Cisco applications, it is possible and recommended to avoid the use of transcoding resources. There might be specific instances where G.711 on a specific device cannot be avoided.
- Some endpoints (for example, video endpoints) do not support the H.323v2 features.

Figure 6-2 Intercluster Call Flow with Transcoding



Unified CM uses media resource groups (MRGs) to enable sharing of MTP and transcoding resources among the Unified CM servers within a cluster. In addition, for calls across intercluster trunks, MTP and transcoding resources are used only when needed, thus eliminating the need to configure the MTP service for applications that do not support LBR codecs.

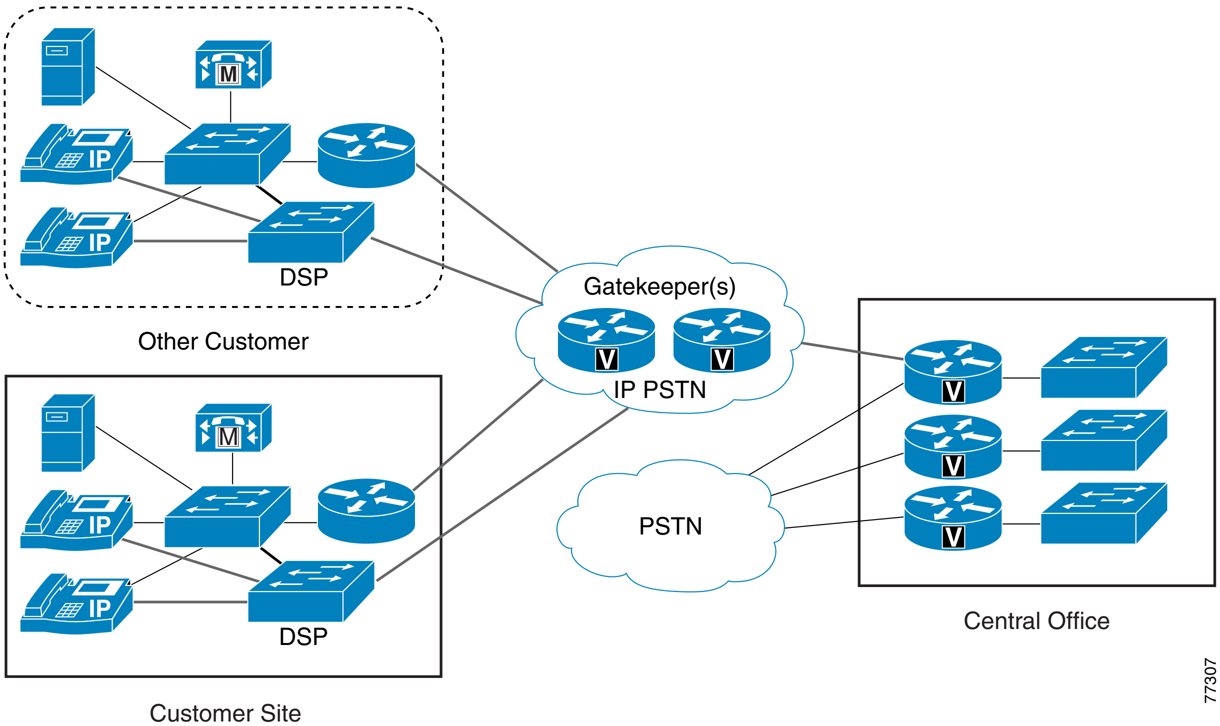
The following characteristics apply to distributed call processing deployments:

- Only the intercluster calls that require transcoding will use the MTP service. For example, if both endpoints of a call are capable of using a G.729 codec, no transcoding resources will be used.
- Sharing MTP resources among servers within a cluster provides more efficient resource utilization.

IP PSTN Access

Another application scenario for MTP and transcoding resources involves a service provider that provides its customers with access to an IP PSTN instead of the traditional PSTN. In such a scenario, the gatekeepers reside in the service provider network. In order to simplify the dial plan, each customer is required to use an MTP to anchor its calls, so that the individual IP addresses assigned to the endpoints can be hidden. The service provider's central office can then relay through the traditional PSTN and/or provide IP connectivity to other customers. Figure 6-3 illustrates this deployment model.

Figure 6-3 IP PSTN Access Example



Note that the customer site in Figure 6-3 can use any of the previous three deployment models: single site, multisite WAN with centralized call processing, or multisite WAN with distributed call processing.

The H.323 trunk from the customer site to the IP PSTN must be configured with MTP so that the endpoint IP addresses remain masked. Thus, all external calls use the MTP resources. However, MTP resources can be shared within the Unified CM cluster to achieve more efficient use of the resources. This technique of utilizing MTPs for address hiding may be used for SIP trunks as well.