Cisco V.150.1 Minimum Essential Requirements (For Cisco IOS Release 15.1(4)M Only)

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Note The information in this document applies to the Cisco V.150.1 Minimum Essential Requirements feature beginning in Cisco IOS Release 15.1(4)M (dated March 28, 2011). This feature was originally released as the Secure Communication Between IP-STE Endpoint and Line-Side STE Endpoint feature in Cisco IOS Releases 12.4(4)T and 12.4(9)T. For reference purposes, there is some "legacy" information provided here about the original feature. For more detailed information about the original feature, see the *Secure Communication Between IP-STE Endpoint and Line-Side STE Endpoint* document.

The Cisco V.150.1 Minimum Essential Requirements feature complies with the requirements of the National Security Agency (NSA) *SCIP-216 Minimum Essential Requirements (MER) for V.150.1* recommendation. The SCIP-216 recommendation has simplified the existing V.150.1 requirements. Beginning in Cisco IOS Release 15.1(4)M, the Cisco V.150.1 MER feature adds negotiation support to the following interfaces:

- Skinny Client Control Protocol (SCCP) for analog gateway endpoints and Secure Communication Interoperability Procol—End Instruments (SCIP—EI)
- Media Gateway Control Protocol (MGCP) T1 (PRI and channel-associated signaling [CAS])
- E1 (PRI) trunks
- Cisco Unified Communications Manager (Cisco UCM) Session Initiation Protocol (SIP) trunks

This feature also provides support for Unified Capability Requirement (UCR) 2008 Modem over IP (MoIP) and Fax over IP (FoIP).

The V.150.1 is an ITU recommendation for using a modem over IP networks that support dialup modem calls for large installed bases of modems and telephony devices operating on a traditional public switched telephone network (PSTN). The V.150.1 recommendation specifically defines how to relay data from modems and telephony devices on a PSTN into and out of an IP network via a modem.

In Cisco IOS Release 12.4(4)T, Cisco developed the Secure Communication Between IP Secure Endpoint and Trunk-Side Secure Terminal Equipment (STE) Endpoint feature, and in Cisco IOS Release 12.4(9)T, the Secure Communication Between IP Secure Endpoint and Line-Side STE Endpoint feature to meet the requirements of this standard. In this document, these features are referred to as "Cisco Legacy V.150.1."

This document focuses primarily on the capabilities of the Cisco V.150.1 MER feature in Cisco IOS Release 15.1(4)M, but also provides some information for the Cisco Legacy V.150.1 feature.

Contents

Prerequisites for Cisco V.150.1 MER

- You must have Cisco IOS Release 15.1(4)M and Cisco UCM 8.6 or later releases installed on your network.
- You must have the following images and licenses installed and running:
  - The adventerprisek9-mz image is needed for Integrated Services Routers (ISRs)
  - The universalk9-mz image in needed for ISR Generation 2s (ISR G2s)
  - UC and security feature licenses are needed for ISR G2s

Restrictions for Cisco V.150.1 MER

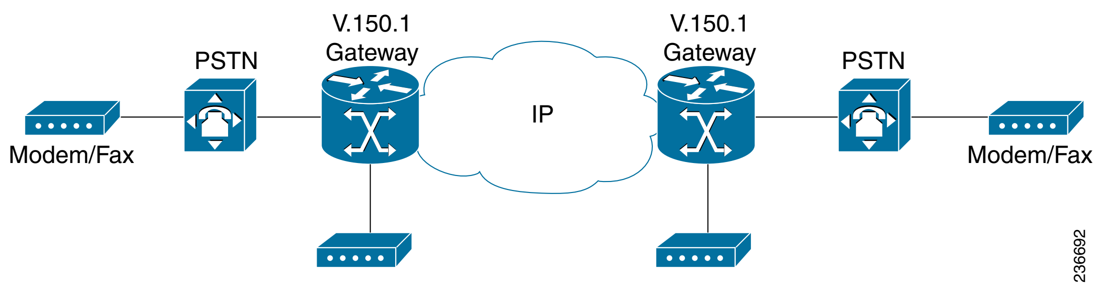
- V.90 and V.92 are not supported in Cisco Legacy V.150.1 or in Cisco V.150.1 MER modem relay.
- Only Cisco UCM 8.6 or later as the call agent.
- ISRs and ISR G2s require Cisco IOS Release 15.1(4)M.
- Cisco V.150.1 MER cannot operate with modem relay that is supported on C542 or C549 DSP technology.
- FoIP implementation cannot interoperate with the non-State Signaling Event (SSE)-based T.38 fax relay protocol.
- RFC 2833 support for modem events is limited to the Cisco V.150.1 MER implementation.
- The Cisco VGD-1T3 platform has Cisco UCM MGCP support, but Cisco V.150.1 MER SCCP Telephony Control Application (STCAPP) support is not available.

Information About Cisco V.150.1 MER

Cisco Legacy V.150.1

In Cisco IOS Release 12.4(4)T, the Secure Communication Between IP Secure Endpoint and Line-Side STE Endpoint feature enabled V.150.1 for STCAPP-control voice ports and allowed an on-network secure terminal equipment (STE), connected directly to a Cisco IOS gateway, to establish a secure call to an IP secure endpoint. Figure 1 shows a basic topology for the V.150.1 standard.

Figure 1 Standard Topology for the V.150.1 Standard



In Cisco IOS Release 12.4(9)T, the Secure Communication Between IP Secure Endpoint and Trunk-Side STE Endpoint feature implemented V.150.1 for the Cisco IOS gateway. The capability was implemented only on MGCP gateways for placing secure calls between the IP secure endpoints and off-network STE devices via MGCP-controlled time-division multiplexing (TDM) trunks.

STE utilizes both modem pass-through and modem relay for secure phone calls. Cisco and another company implemented V.150.1 to carry SCIP (formerly known as Future Narrow Band Digital Terminal [FNBDT]) data to meet the DoD requirements of STE. There is also a VoIP STE that uses only modem relay for secure phone calls.

Cisco's Legacy V.150.1 implementation contains the following features:

- Cisco Legacy V.150.1 supports registration of device capabilities to the Cisco UCM.
- Cisco Legacy V.150.1 enables either V.150.1 modem relay or passthrough on the Cisco UCM-controlled line-side and trunk-side gateway endpoints. Modem relay and modem pass-through using g.711 and g.729, is implemented as nonstandard codecs in the Cisco UCM.
- Cisco Legacy V.150.1 falls back to modem pass-through when the Cisco UCM does not provide modem transport directive, allowing compatibility with earlier Cisco IOS releases. (Secure Communication Between IP Secure Endpoint and Line-Side STE Endpoint analog/BRI only).
- With Cisco Legacy V.150.1, STU devices do not use FNBDT. STU devices use a proprietary STUIII signaling/datapump that is not compatible with Cisco Legacy V.150.1. A STU cannot be used to place a secure call to an IP secure endpoint.

Differences Between Cisco V.150.1 MER and Cisco Legacy V.150.1

Table 1 summarizes the differences and advantages of Cisco V.150.1 MER over the Cisco Legacy V.150.1.

Table 1 Differences Between Cisco Legacy V.150.1 and Cisco V.150.1 MER

Cisco Legacy V.150.1	Cisco V.150.1 MER Modem Relay (SCIP-216 Compliant)
Simple Packet Relay Transport (SPRT) (but not all SPRT messages)	New SPRT Cleardown MR (CM) messages.
Uses SSE for ANSwering tone/ANSwering tone with amplitude modulation (ANS/ANSam)	Move from a proprietary Modem Relay transition to standards-based modem relay transition, using RFC 2833 ANS/ANSam signaling.
Proprietary SSE messages	New Reason Identifier Codes (RICs) and SSEs. Call setup protocol requirements for negotiating specific V.150.1 capabilities.
T.38 (non-SSE)	T.38 fax relay SSE version 3.
Audio codec support only	Both Audio codec and “NoAudio” codec support for interworking with Modem Relay Preferred Devices plus audio codec.
Requires configuration on the gateway to turn on V.150.1 modem relay line-side parameters	Autoconfiguration of V.150.1 controlled at the Cisco UCM device configuration page for SCCP gateway analog phone ports.
No support for MoIP	MoIP—Modem Relay and audio passthrough.
H.323/SIP/TI/E1	V.150.1 MER can be used over SIP and T1/E1 trunks. V.150.1 MER is not supported over H.323 trunks (only Cisco Legacy V.150.1 is supported over H.323 trunks).

Note When endpoints are capable of both modem relay and modem pass-through, Cisco UCM uses MER modem relay as first preference.

Table 2 summarizes the hardware and software compatibility information for Cisco Legacy V.150.1 and Cisco V.150.1 MER.

Table 2 Compatibility Matrix Contrasting Cisco Legacy V.150.1 and Cisco V.150.1 MER

Voice Card	Platform	Digital Signal Processor (DSP)/DSP Module	Original Software Releases for Legacy V.150.1	Original Software Releases for V.150.1 MER
NM-HD-2V	2811/2821/2851 3825/3845	PVDM2s (Built-in DSP)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2911/2921/2951 3925/3945/3925E/3945E	PVDM2s (Built-in DSP)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M

Table 2 Compatibility Matrix Contrasting Cisco Legacy V.150.1 and Cisco V.150.1 MER (continued)

NM-HD-2VE	2811/2821/2851 3825/3845	PVDM2s (Built-in DSP)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2911/2921/2951 3925/3945/3925E/3945E	PVDM2s (Built-in DSP)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
NM-HDV2	2811/2821/2851 3825/3845	PVDM2s (Built-in DSP)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2911/2921/2951 3925/3945/3925E/3945E	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
NM-HDV2-2T1/E1	2811/2821/2851 3825/3845	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2911/2921/2951 3925/3945/3925E/3945E	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
NM-HDV2-1T1/E1	2811/2821/2851 3825/3845	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2911/2921/2951 3925/3945/3925E/3945E	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
VIC-4FXS/DID	2811/2821/2851 3825/3845	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
VIC2-2FXS	2811/2821/2851 3825/3845	PVDM2 (Onboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
VIC3-2FXS/DID	2811/2821/2851 3825/3845	NMs 5510/PVDM2 PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS15.1(4)M
	2901/2911/2921/2951 3925/3945/3925E/3945E	NMs 5510/PVDM2 PVDM3 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
VIC3-4FXS/DID	2811/2821/2851 3825/3845	NMs 5510/PVDM2 PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2901/2911/2921/2951 3925/3945/3925E/3945E	NMs 5510/PVDM2 PVDM3 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M

Table 2 Compatibility Matrix Contrasting Cisco Legacy V.150.1 and Cisco V.150.1 MER (continued)

VWIC2-1MFT-T1/E1	2811/2821/2851 3825/3845	NMs 5510/PVDM2 PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2901/2911/2921/2951 3925/3945/3925E/3945E	NMs 5510/PVDM2 PVDM3 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
VWIC2-2MFT-T1/E1	2811/2821/2851 3825/3845	NMs 5510/PVDM2 PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2901/2911/2921/2951 3925/3945/3925E/3945E	NMs 5510/PVDM2 PVDM3 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
EVM-HD-8FXS/DID	2821/2851 3825/3845	PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
EM3-HDA-8FXS	2911/2921/2951 3925/3945/3925E/3945E	PVDM3 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 15.0(1)M	Cisco UCM 8.6 Cisco IOS 15.1(4)M
	2811/2821/2851 3825/3845	PVDM2 (Motherboard DSP Slot)	Cisco UCM 4.2 Cisco IOS 12.4(4)T	Cisco UCM 8.6 Cisco IOS 15.1(4)M
Built in	VG 202/204/224	PVDM2	Cisco UCM 6.1.3, 7.0.1 or higher Cisco IOS 12.4(22)T or later	Cisco UCM 8.6 Cisco IOS 15.1(4)M

Advantages of Modem Relay Over Modem Pass-through

The advantages of modem relay over modem pass-through are:

- Consumes less bandwidth
- Uses error correction mechanism rather than redundancy
- Specifically designed to transport modem communication over IP whereas modem pass-through adapts a voice codec
- More efficient and robust in maintaining transmissions over IP

For more information, see *Fax/Modem over IP*.

SCIP—EI, Modem over IP, and Fax over IP Interfaces

The following interfaces are supported for SCIP and MoIP:

- MGCP T1 (PRI and CAS) and E1 PRI endpoints subtending MGCP Cisco IOS gateways.
- SCCP Analog FXS SCIP-compliant endpoints subtending SCCP Cisco IOS gateways.

- SCIP-EI V.150 IP endpoints running the SCCP protocol version 21 and later.
- AS-SIP Trunk and SIP ICT.

The following interfaces are supported for FoIP:

- MGCP T1 (PRI and CAS) and E1 PRI endpoints subtending MGCP Cisco IOS gateways.
- Cisco UCM AS-SIP Trunk and Cisco UCM SIP ICT.
- Cisco UCM FoIP is not supported on SCCP analog FXS ports in Cisco IOS Release 15.1(4)M and Cisco UCM 8.6.

The following interface is *not* supported for the UCR 2008 SCIP, MoIP, FoIP functionality, provided by this feature:

- H.323 ICT (not supported in MER—*only* for Cisco Legacy V.150.1).

Cisco V.150.1 MER Network Architecture

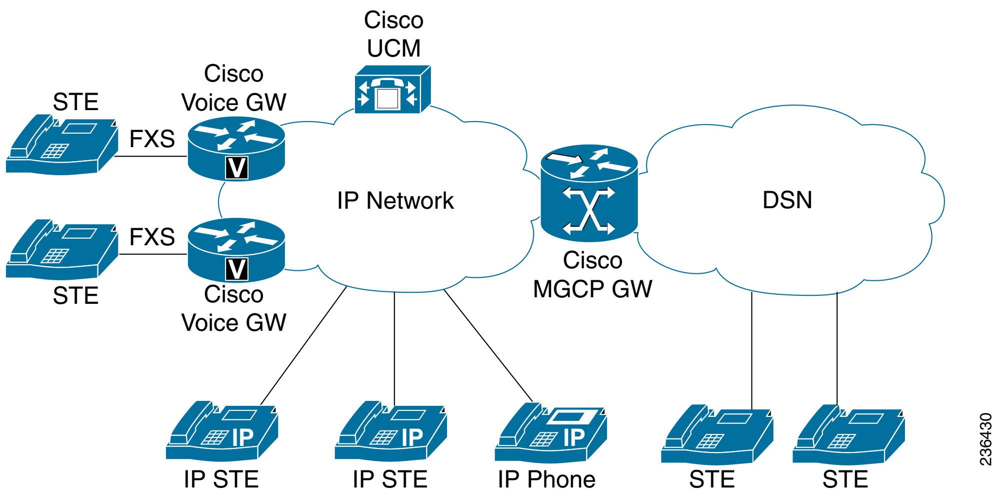
The two types of endpoints in the MER network are:

- SCIP–EI Phone: IP connectivity resides in IP network.
- Analog STE interface, residing in an IP or DSN network.

The Cisco V.150.1 MER network architecture (shown in Figure 2) supports the following:

- Gateway-to-gateway functionality for PSTN-STE endpoints.
- FNBDT traffic for the same topology as in the Secure Communication Between IP Secure Endpoint and Trunk-Side STE Endpoint feature.
- V.150.1 FoIP functionality with MGCP endpoints.
- Voice gateway connectivity between DSN and IP network, and transports encrypted voice and data media.

Figure 2 Cisco V.150.1 MER Network Architecture



How to Configure Cisco V.150.1 MER

To configure the line-side functionality of the Cisco V.150.1 MER feature, perform the following tasks:

- Configuring the Cisco UCM, page 8
- Configuring the Gateway in the Cisco Unified CM Administration, page 9
- Configuring the Phone Settings in the Cisco Unified CM Administration, page 11
- Adding a New Directory Number in the Cisco Unified CM Administration, page 13
- Configuring the Gateway (Line-side), page 15
- Configuring the Gateway (Trunk-side), page 22
- Configuring the SIP Trunk, page 30
- Verifying and Troubleshooting the Cisco V.150.1 MER Configuration, page 34
- Symptoms and Possible Solutions for Cisco V.150.1 MER, page 43

Configuring the Cisco UCM

To configure the Cisco UCM, perform the tasks in this section.

Step 1 Start the web-based application Cisco Unified CM Administration.

Step 2 Enter your username and password, and click **Login**.

Step 3 From the menu, choose **Device**.

Step 4 Click **Add New**.

Step 5 Choose a Gateway Type from the drop-down list.

Step 6 Click **Next**.

Step 7 Choose a protocol in the Protocol drop-down field.

Step 8 Click **Next**.

Configuring the Gateway in the Cisco Unified CM Administration

To configure the gateway, perform the tasks in this section. See Figure 3 for an example screen of gateway configuration settings.

Step 1 Enter a MAC address in the Mac Address field.

Step 2 Choose a UCM group from the Cisco Unified Communications Manager Group field.

Step 3 Configure slots, VICs, and endpoints in the Configured Slots, VICs and Endpoints field.

Step 4 Choose or change other configuration layouts in the Product Specific Configuration Layout section if needed.

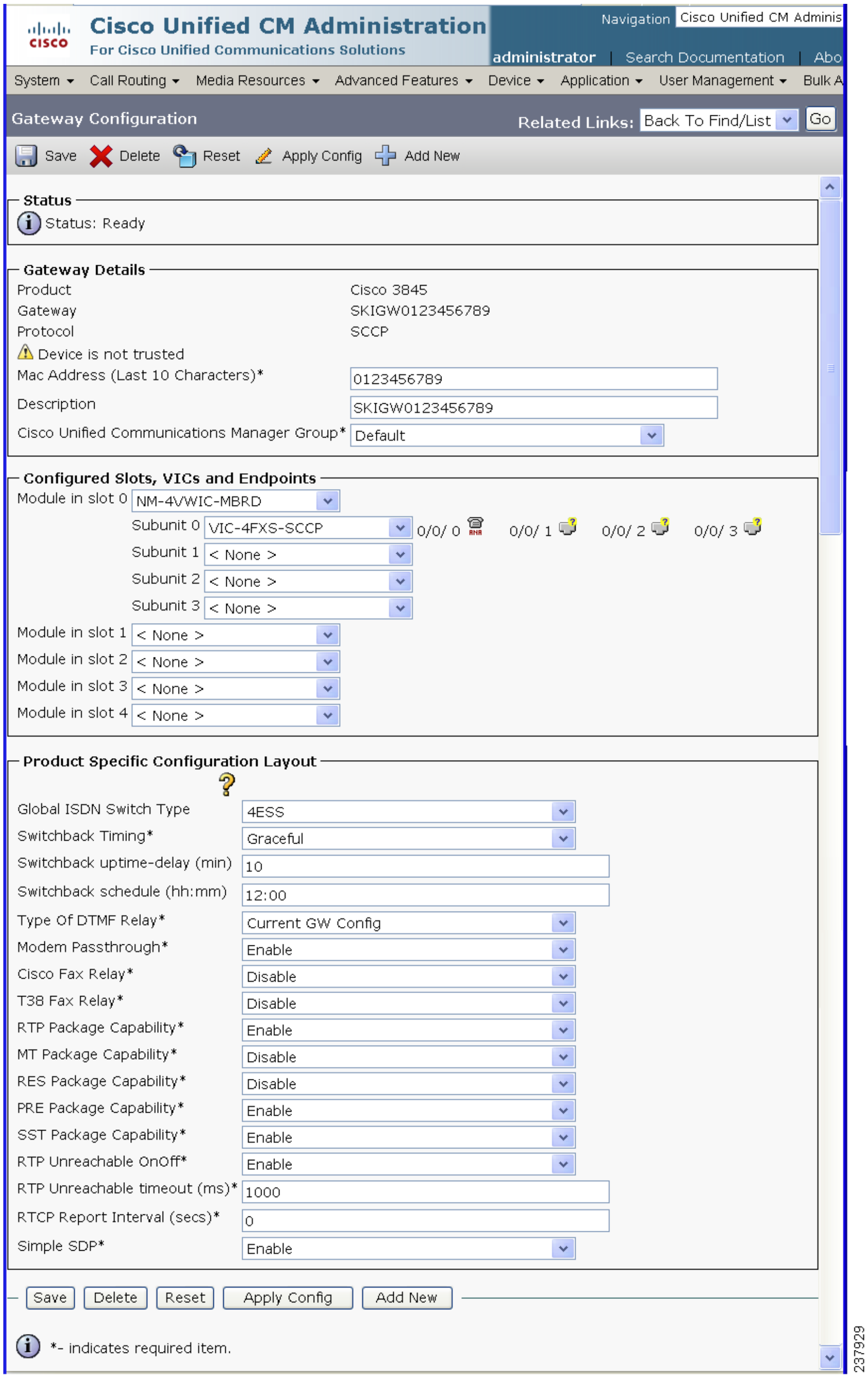
Step 5 Click **Save**. A message appears: "Click the Apply Config button to have the changes take effect."

Step 6 Click **OK**.

Step 7 In the Configured Slots, VICs and Endpoints section, choose a subunit from the drop-down menu if needed.

Step 8 When a Subunit is selected, icons appear to the right of the **Subuni**t field. Click the icons to configure the devices. The Phone Configuration screen displays.

Figure 3 Example of Gateway Configuration Settings



Configuring the Phone Settings in the Cisco Unified CM Administration

To configure the phone settings, perform the following steps. Figure 4 provides an example of the screen for phone configuration.

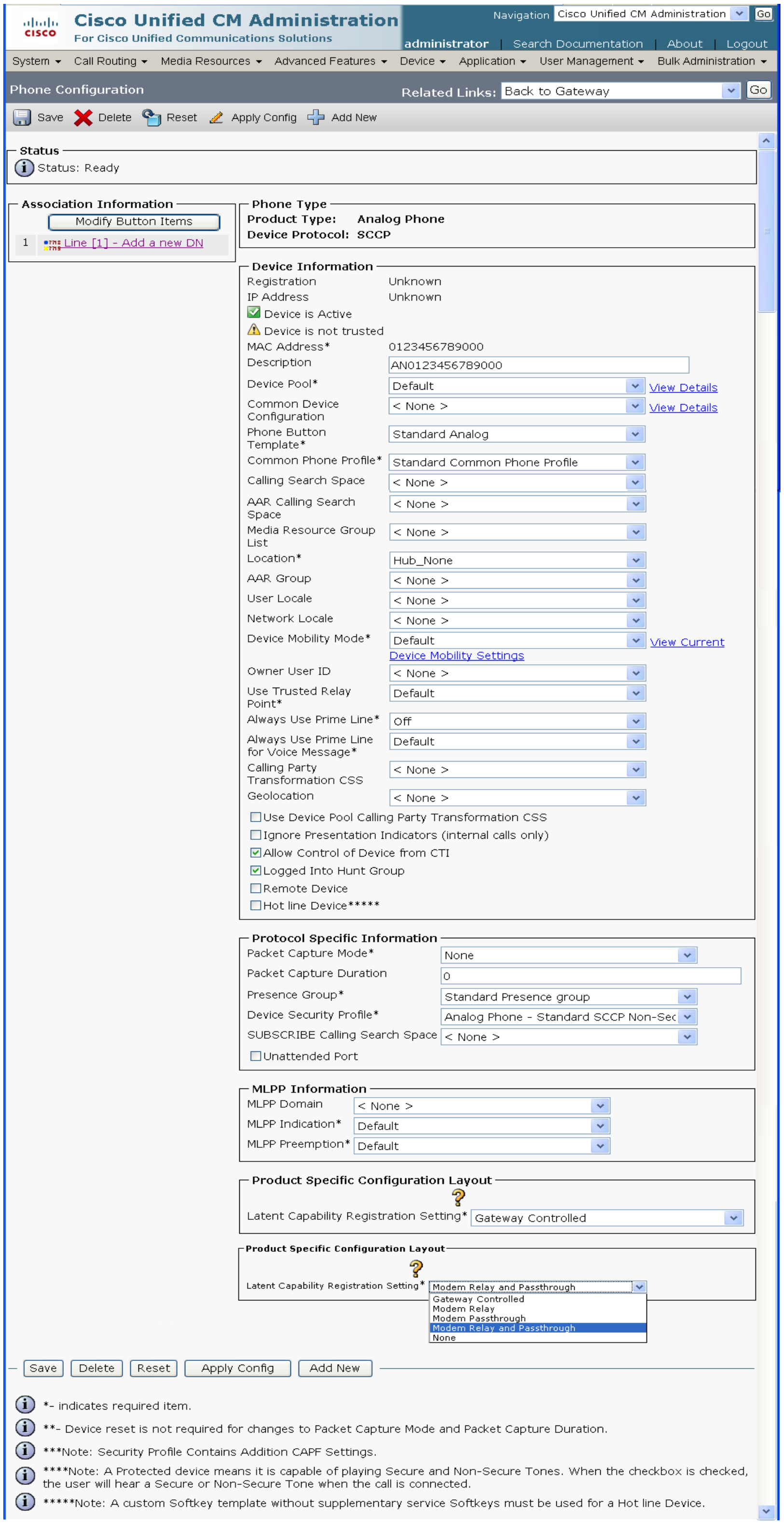
Step 1 Choose desired settings from the drop-down options. For required fields, Default is often the correct choice.

Note From the drop-down list, be sure to choose **Modem Relay** or **Modem Relay and Passthrough**, depending on your environment.

Step 2 Click **Save**.

Step 3 The following message displays: Click the Apply Config button to have the changes take effect. Click **OK**. The Phone Configuration page refreshes, and the Add a new DN field appears on the left of the screen.

Figure 4 Example of Phone Configuration Settings



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Adding a New Directory Number in the Cisco Unified CM Administration

To add a new directory number (DN), perform the task in this section. Figure 5 and Figure 6 provide examples of the screens for DN settings.

Note You must have performed the tasks in the "Configuring the Phone Settings in the Cisco Unified CM Administration" section on page 11 for this field to appear on the screen.

Step 1 Find the Add a new DN field on the left of the refreshed Phone Configuration page.

Step 2 Click **Add a new DN**. The Directory Number Configuration page displays.

Step 3 In the Directory Number field, add a directory number.

Step 4 In the section Multiple Call/Call Waiting Settings on Device [Device Name], set Maximum Number of Calls and Busy Trigger at **1** for V.150.1 endpoints.

Step 5 Enter or choose values in the remaining fields that are required or desired for your particular network environment.

Figure 5 Example of Directory Number (DN) Settings (Part 1)

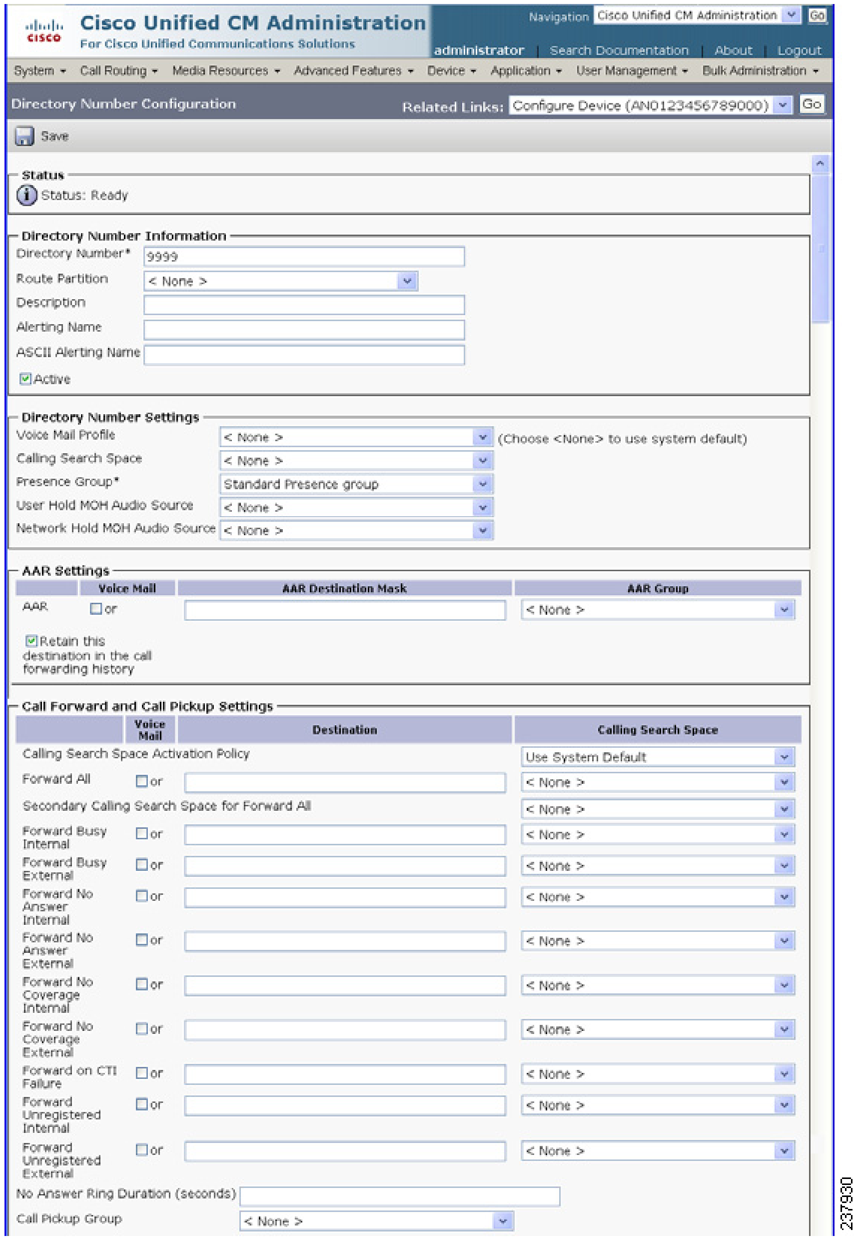
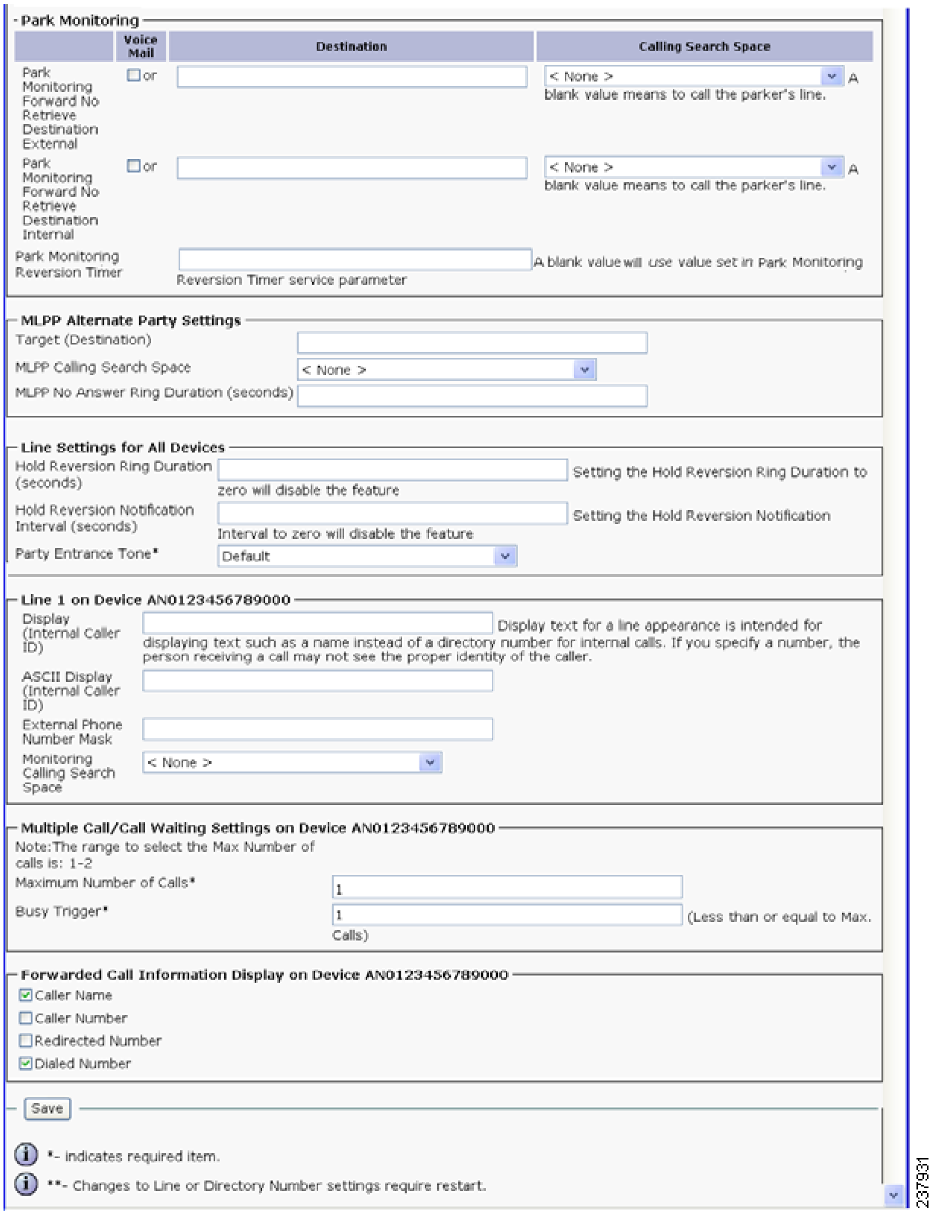


Figure 6 Example of Directory Number Settings (Part 2)



Configuring the Gateway (Line-side)

To configure the line-side gateway, perform the following tasks (in some of these tasks, the command syntax has been abbreviated for clarity):

- Configuring SCCP on Cisco IOS Gateways, page 16 (required)
- Configuring Modem Transport Methods for STCAPP Devices, page 17 (required)
- Configuring Modem Pass-through Calls, page 19 (required)

- Configuring V.150.1 Modem Relay Parameters, page 20 (optional)

Configuring SCCP on Cisco IOS Gateways

SCCP messaging enables Cisco Unified Communications Manager endpoint call control using the STCAPP. To configure SCCP on the Cisco IOS gateway, perform the tasks in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sccp local** *interface-type interface-number*
4. **sccp ccm** {*ip-address* | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*]
5. **sccp**
6. **sccp ccm group** *group-number*
7. **associate ccm** *identifier-number* **priority** *priortiy-number*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**sccp local** *interface-type interface-number* **Example:** `Router(config)# sccp local fastethernet 0/0`	Selects the local interface that the SCCP application uses to register with Cisco Unified Communications Manager. • This is the interface whose MAC address is specified for SCCP gateway registration using the Cisco Unified Communications Manager autoconfiguration in the "Configuring the Gateway in the Cisco Unified CM Administration" section on page 9. • *interface-type*—Specifies the interface type that the SCCP application uses to register with the Cisco UCM. • *interface-number*—Specifies the interface number that the SCCP application uses to register with the Cisco UCM.

	Command or Action	Purpose
Step 4	**sccp ccm** {*ip-address* | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*] **Example:** `Router(config)# sccp ccm 10.1.1.1 version 8`	Adds a Cisco UCM server to the list of available servers and sets various parameters. • *ip-address*—Specifies the IP address of the Cisco UCM server. • *identifier-number*—Identifies the Cisco UCM associated with the *group-number* value configured in Step 6. Valid entries are from 1 to 65535. There is no default value. • **version**—Identifies the version number of the Cisco UCM.
Step 5	**sccp** **Example:** `Router(config)# sccp`	Enables SCCP and its related applications.
Step 6	**sccp ccm group** *group-number* **Example:** `Router(config)# sccp ccm group 1`	Creates a Cisco UCM group. • *group-number*—Associates the Cisco UCM group with the Cisco UCM group *identifier-number* configured in Step 3. Range is 1 to 65535. There is no default value.
Step 7	**associate ccm** *identifier-number* **priority** *priority-number* **Example:** `Router(config)# associate ccm 1 priority 1`	Associates a Cisco UCM with a Cisco UCM group. • *identifier-number*—Identifies the Cisco UCM associated with the Cisco UCM *group-number* configured in Step 6. Valid entries are from 1 to 65535. There is no default value. • *priority-number*— Priority of the Cisco UCM within the Cisco UCM group. Range is 1 to 4. There is no default value. The highest priority is 1.
Step 8	**exit** **Example:** `Router(config)# exit`	Exits the current configuration mode.

Configuring Modem Transport Methods for STCAPP Devices

This task configures modem transport methods for STCAPP devices. Perform this task to specify modem transport capability.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **stcapp register capability** *voice-port* **modem-relay**
4. **stcapp register capability** *voice-port* **modem-passthrough**
5. **stcapp register capability** *voice-port* **both**
6. **stcapp ccm group** *group-id*

7. **stcapp**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`stcapp register capability`** *`voice-port`* **`modem-relay`** **Example:** `Router(config)# stcapp register capability 1/1/0 modem-relay`	Specifies device modem transport capability. • *voice-port*—Specifies the voice interface slot number. • **modem-relay**—Specifies that the device supports V.150.1 modem relay. **Note** Beginning with Cisco IOS Release 15.1(4)M, the **modem-relay** or **both** option enables both the V.150.1 MER latent caps and V.150.1 virtual codec caps to be reported to Cisco Unified Communications Manager via the StationCapabilitiesResMessage. When both sides of the call support V.150.1 MER caps and V.150.1 virtual codec caps, the V.150.1 MER cap is chosen and sent to the SCCP gateway during the call setup via ORC/SMT. Otherwise, the V.150.1 virtual codec caps are used. **Note** The **stcapp register capability** command has three options: – **modem relay** – **modem-passthrough**, which limits codec capabilities when registering – **both**
Step 4	**`stcapp register capability`** *`voice-port`* **`modem-passthrough`** **Example:** `Router(config)# stcapp register capability 1/1/1 modem-passthrough`	Specifies device modem transport capability. • *voice-port*—Specifies the voice interface slot number. • **modem-passthrough**—Specifies the device supports modem pass-through (voice band data).

	Command or Action	Purpose
Step 5	**stcapp register capability** *voice-port* **both** **Example:** `Router(config)# stcapp register capability 1/1/2 both`	Specifies device modem transport capability. • *voice-port*—Specifies the voice interface slot number. • **both**—Specifies the device supports both modem relay and modem pass-through.
Step 6	**stcapp ccm-group** *group-id* **Example:** `Router(config)# stcapp ccm-group 1`	Configures the Cisco UCM group number for use by the STCAPP.
Step 7	**stcapp** **Example:** `Router(config)# stcapp`	Enables the STCAPP.
Step 8	**exit** **Example:** `Router(config)# exit`	Exits the current configuration mode.

Configuring Modem Pass-through Calls

This task configures modem pass-through calls on the gateway. Perform this task to enable interoperation with the SCCP gateway running versions of Cisco IOS software prior to Cisco IOS Release 12.4(4)T that are not V.150.1-capable.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **modem passthrough nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy** [**maximum-sessions** *sessions*]]
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice-service configuration mode and specifies VoIP encapsulation.
Step 4	**modem passthrough nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy** [**maximum-sessions** *sessions*]] **Example:** `Router(config-voi-serv)# modem passthrough nse codec g711ulaw`	Configures modem pass-through over VoIP globally for all dial peers.
Step 5	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits the current configuration mode.

Configuring V.150.1 Modem Relay Parameters

This task configures optional V.150.1 modem-relay parameters. Configure these parameters to address specific network conditions for latency, redundancy, and V.14 parameters.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **modem relay nse codec g711ulaw**
5. **modem relay latency** *milliseconds*
6. **modem relay sse redundancy interval** *milliseconds*
7. **modem relay sse redundancy packet** *number*
8. **modem relay sse t1** *milliseconds*
9. **modem relay sse retries** *value*
10. **modem relay sprt retries** *value*
11. **modem relay sprt v14 receive playback hold-time** *milliseconds*
12. **modem relay sprt v14 transmit hold-time** *milliseconds*
13. **modem relay sprt v14 transmit maximum hold-count** *characters*
14. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice service voip`** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode and specifies VoIP encapsulation.
Step 4	**`modem relay nse codec g711ulaw`** **Example:** `Router(config-voi-serv)# modem relay nse codec g711ulaw`	Specifies that the named signaling event (NSE) codec type is G.711 mu-law.
Step 5	**`modem relay latency`** *`milliseconds`* **Example:** `Router(config-voi-serv)# modem relay latency 250`	Specifies the estimated one-way delay across the IP network. • Range is 100 to 1000. Default is 200.
Step 6	**`modem relay sse redundancy interval`** *`milliseconds`* *Example:* `Router(config-voi-serv)# modem relay sse redundancy interval 25`	Specifies the timer value for redundant transmission of SSEs. • Range is 5 to 50 ms. Default is 20.
Step 7	**`modem relay sse redundancy packet`** *`number`* **Example:** `Router(config-voi-serv)# modem relay sse redundancy packet 2`	Specifies the SSE packet transmission count before disconnecting. • Range is 1 to 5 packets. Default is 3.
Step 8	**`modem relay sse t1`** *`milliseconds`* **Example:** `Router(config-voi-serv)# modem relay sse t1 2100`	Specifies the repeat interval, in milliseconds (ms), for initial audio SSEs used for resetting the SSE protocol state machine (clearing the call) following error recovery. • Range is 500 to 3000 ms. Default is 1000.
Step 9	**`modem relay sse retries`** *`value`* **Example:** `Router(config-voi-serv)# modem relay sse retries 5`	Specifies the number of SSE packet retries, repeated every t1 interval, before disconnecting. • Range is 0 to 5. Default is 5.

	Command or Action	Purpose
Step 10	**modem relay sprt retries** *value* **Example:** `Router(config-voi-serv)# modem relay sprt retries 10`	Specifies the number of SPRT packet retries, repeated every t1 interval, before disconnecting. • Range is 0 to 10. Default is 10.
Step 11	**modem relay sprt v14 receive playback hold-time** *milliseconds* **Example:** `Router(config-voi-serv)# modem relay sprt v14 receive playback hold-time 32`	Configures the time, in ms, to hold incoming data in the V.14 receive queue. • Range is 20 to 250. Default is 50.
Step 12	**modem relay sprt v14 transmit hold-time** *milliseconds* **Example:** `Router(config-voi-serv)# modem relay sprt v14 transmit hold-time 12`	Configures the time to wait, in ms, after the first character is ready before sending the SPRT packet. • Range is 10 to 30. Default is 20.
Step 13	**modem relay sprt v14 transmit maximum hold-count** *characters* **Example:** `Router(config-voi-serv)# modem relay sprt v14 transmit maximum hold-count 22`	Configures the number of V.14 characters to be received on the ISDN PSTN interface that will trigger sending the SPRT packet. • Range is 8 to 128. Default is 16.
Step 14	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits the current configuration mode.

Configuring the Gateway (Trunk-side)

To configure the trunk side of the gateway, perform the following tasks:

- Configuring the T1 Controller and Operating Parameters, page 22 (required)
- Configuring MGCP for Compatibility with Cisco UCM, page 24 (required)
- Configuring MGCP Parameters for Modem Relay, page 28 (optional)

Configuring the T1 Controller and Operating Parameters

To configure the T1 controller and operating parameters, perform the tasks in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **controller** {**t1** | **e1**} *slot*/*port*
4. **framing** {**sf** | **esf**}

5. **clock source** {**line** {**primary** | **secondary**} | **internal**}
6. **linecode** {**ami** | **b8zs**}
7. **cablelength short** *length*
8. **pri-group timeslots** *timeslot-range* **service mgcp**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`controller`** {**`t1`** | **`e1`**} *`slot`*/*`port`* **Example:** `Router(config)# controller t1 1/0`	Configures a T1 or E1 controller and enters controller configuration mode. • Specify **t1** for a T1 controller. • *slot*/*port*—Backplane slot number and port number on the interface. See your hardware installation manual for the specific values and slot numbers.
Step 4	**`framing`** {**`sf`** | **`esf`**} **Example:** `Router(config-controller)# framing esf`	Selects the frame type for the T1 data line. • **sf**—Specifies super frame as the T1 frame type. • **esf**—Specifies extended super frame as the T1 frame type.
Step 5	**`clock source`** {**`line`** {**`primary`** | **`secondary`**} | **`internal`**} **Example:** `Router(config-controller)# clock source internal`	Sets the T1 line clock source. • **line**—Specifies that the interface will clock its transmitted data from a clock recovered from the line's receive data stream. This is the default. • **primary**—Primary TDM clock source. • **secondary**—Secondary TDM clock source. • **internal**—Selects the free running clock (also known as the internal clock) as the clock source.
Step 6	**`linecode`** {**`ami`** | **`b8zs`**} **Example:** `Router(config-controller)# linecode b8zs`	Selects the line code for the T1 line. • **ami**—Specifies alternate mark inversion (AMI) as the line code. • **b8zs**—Specifies binary 8-zero substitution (B8ZS) as the line code. This is the default.

	Command or Action	Purpose
Step 7	**`cablelength short`** *`length`* **Example:** `Router(config-controller)# cablelength short 133`	Sets the cable length 655 feet or shorter for Cisco routers. • **133**—Specifies a cable length from 0 to 133 feet.
Step 8	**`pri-group timeslots`** *`timeslot-range`* **`service mgcp`** **Example:** `Router(config-controller)# pri-group timeslots 1-24 service mgcp`	Specifies an ISDN PRI group on the channelized T1 controller, and configures service type **mgcp** for Media Gateway Control Protocol service.

Configuring MGCP for Compatibility with Cisco UCM

To ensure proper operation of the Cisco V.150.1 MER feature on the Cisco UCM, peform the MGCP CLI configuration steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp**
4. **mgcp call-agent** [*ipaddr* | *hostname*] [*port*] **service-type mgcp** {**version** *version-number*]
5. **mgcp dtmf-relay voip codec** {**all** | **low-bit-rate**} **mode** {**cisco** | **nse** | **out-of-band** | **nte-gw** | **nte-ca**}
6. **mgcp rtp unreachable timeout** *timeout-value* [**action notify**]
7. **mgcp modem passthrough** {**voip** | **voaal2**} **mode** {**cisco** | **nse**}
8. **mgcp package-capability rtp-package**
9. **no mgcp package-capability res-package**
10. **mgcp package-capability sst-package**
11. **no mgcp package-capability fxr-package**
12. **mgcp package-capability pre-package**
13. **mgcp package-capability mdste-package**
14. **no mgcp timer** {**receive-rtcp** | **net-cont-test** | **nse-response t38}** *timer*
15. **mgcp sdp simple**
16. **mgcp rtp payload-type g726r16 static**
17. **mgcp rtp payload-type nte** *number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mgcp** **Example:** `Router(config)# mgcp`	Initiates the MGCP application.
Step 4	**mgcp call-agent** [*ipaddr* | *hostname*] [*port*] **service-type mgcp** {**version** *version-number*] **Example:** `Router(config)# mgcp call-agent cisco-cm1 2427 service-type mgcp version 0.1`	Specifies the call agent's IP address or domain name, the port, and gateway control service type.

	Command or Action	Purpose
Step 5	**mgcp dtmf-relay voip codec** {**all** | **low-bit-rate**} **mode** {**cisco** | **nse** | **out-of-band** | **nte-gw** | **nte-ca**} **Example:** `Router(config)# mgcp dtmf-relay voip codec all mode nte-gw`	Ensures accurate forwarding of digits on compressed codecs. • **all**—Configures dual-tone multifrequency (DTMF) relay to be used with all voice codecs. • **low-bit-rate**—Configures DTMF relay to be used with only low-bit-rate voice codecs, such as G.729. • **cisco**—Real-time Transport Protocol (RTP) digit events are encoded using a proprietary format similar to Frame Relay as described in the FRF.11 specification. The events are transmitted in the same RTP stream as nondigit voice samples, using payload type 121. • **nse**—RTP digit events are encoded using the format specified in RFC 2833, Section 3.0, and are transmitted in the same RTP stream as nondigit voice samples, using the payload type that is configured using the **mgcp tse payload** command. • **out-of-band**—MGCP-digit events are sent using NTFY messages to the call agent (CA), which plays them on the remote gateway using RQNT messages with S: (signal playout request). • **nte-gw**—RTP digit events are encoded using the format specified in RFC 2833, Section 3.0, and are transmitted in the same RTP stream as nondigit voice samples. The payload type is negotiated by the gateways before use. The configured value for the payload type is presented as the preferred choice at the beginning of the negotiation. • **nte-ca**—Identical to the **nte-gw** keyword behavior except that the CA's local connection options a: line is used to enable or disable DTMF relay.
Step 6	**mgcp rtp unreachable timeout** *timeout-value* [**action notify**] **Example:** `Router(config)# mgcp rtp unreachable timeout 1000 action notify`	Enables detection of an unreachable remote VoIP endpoint. • *timeout-value*—Time, in milliseconds, that the system waits for voice packets from the unreachable endpoint. Range is 500 to 10000. • **action notify**—Sends a notification when the timeout value has been exceeded.

	Command or Action	Purpose
Step 7	**mgcp modem passthrough** {**voip** | **voaal2**} **mode** {**cisco** | **nse**} **Example:** `Router(config)# mgcp modem passthrough voip mode nse`	Sets the method for changing speeds that enables the gateway to send and receive modem and fax data in VoIP and Voice over ATM (VoATM) adaptation layer 2 (VoAAL2) configurations. • **voip**—VoIP. • **voaal2**—Voice over AAL2 calls using Annex K type 3 packets. • **cisco**—Cisco-proprietary method for changing modem speeds, based on the protocol. • **nse**—NSE-based method for changing modem speeds. For VoAAL2 configurations, AAL2 Annex K (type 3) is used.
Step 8	**mgcp package-capability rtp-package** **Example:** `Router(config)# mgcp package-capability rtp-package`	Enables the MGCP package capability type for RTP packages on the gateway.
Step 9	**no mgcp package-capability res-package** **Example:** `Router(config)# no mgcp package-capability res-package`	Disables the MGCP package capability type for RSVP packages on the gateway.
Step 10	**mgcp package-capability sst-package** **Example:** `Router(config)# mgcp package-capability sst-package`	Enables the MGCP package capability type for SST packages on the gateway.
Step 11	**no mgcp package-capability fxr-package** **Example:** `Router(config)# no mgcp package-capability fxr-package`	Disables the MGCP package capability type for FXR packages for fax transmissions on the gateway.
Step 12	**mgcp package-capability pre-package** **Example:** `Router(config)# mgcp package-capability pre-package`	Enables the MGCP package capability type for PRE packages on the gateway.
Step 13	**mgcp package-capability mdste-package** **Example:** `Router(config)# mgcp package-capability mdste-package`	Enables the MGCP package capability type for modem relay STE packages on the gateway. • Enables events and signals for modem connections enabling a secure communication path between IP-STE and STE.

	Command or Action	Purpose
Step 14	**no mgcp timer** {**receive-rtcp** *timer* | **net-cont-test** *timer* | **nse-response t38** *timer*} **Example:** `Router(config)#no mgcp timer receive-rtcp`	Configures how a gateway detects the RTP stream host. • The **no** form of this command resets the default values.
Step 15	**mgcp sdp simple** **Example:** `Router(config)# mgcp sdp simple`	Specifies use of a subset of the Session Description Protocol (SDP). • Some call agents require this subset to send data through the network.
Step 16	**mgcp rtp payload-type g726r16 static** **Example:** `Router(config)# mgcp rtp payload-type g726r16 static`	Specifies use of the G.726r16 codec for the RTP payload type for backward compatibility in MGCP networks. • **g726r16**—Payload type for the G.726 codec at 16K. • **static**—Static payload type.
Step 17	**mgcp rtp payload-type nte** *number* **Example:** `Router(config)# mgcp rtp payload-type nte 101`	Configures the dynamic RTP payload type for RFC 2833 named telephone event (rtp-nte) packets when doing DTMF interworking. • The payload type is used for both transmitting and receiving, therefore it must be the same value that is used on the peer. • The *number* argument must be in the range of 96 to 127.

Configuring MGCP Parameters for Modem Relay

To configure MGCP parameters for modem relay, perform the tasks in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp modem relay mode voip sse [redundancy {interval** *number* | **packet** *number*}] [**retries** *value*] [**t1** *time*]
4. **mgcp modem relay voip sprt v14** {**receive playback hold-time** *milliseconds* | **transmit hold-time** *milliseconds* | **transmit maximum hold-count** *characters*}
5. **mgcp package-capability** *package*
6. **mgcp dtmf-relay voip codec all mode nte-gw**
7. **mgcp rtp payload-type nte 101**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mgcp modem relay mode voip sse** [**redundancy** {**interval** *number* | **packet** *number*}][**retries** *value*] [**t1** *time*] **Example:** `Router(config)# mgcp modem relay mode voip sse redundancy packet 5`	Specifies SSE modem-relay parameters. • **redundancy**—(Optional) Packet redundancy for modem traffic during modem pass-through. By default redundancy is disabled. • **interval** *milliseconds*—Specifies the timer in milliseconds (ms) for redundant transmission of SSEs. Range is 5 to 50 ms. Default is 20. • **packet** *number*—Specifies the SSE packet retransmission count before disconnecting. Range is 1 to 5. Default is 3. • **retries** *value*—(Optional) Specifies the number of SSE packet retries, repeated every t1 interval, before disconnecting. Range is 0 to 5. Default is 5. • **t1** *milliseconds*—Specifies the repeat interval, in ms, for initial audio SSEs used for resetting the SSE protocol state machine (clearing the call) following error recovery. Range is 500 to 3000. Default is 1000.
Step 4	**mgcp modem relay voip sprt v14** {**receive playback hold-time** *milliseconds* | **transmit hold-time** *milliseconds* | **transmit maximum hold-count** *characters*} **Example:** `Router(config)# mgcp modem relay voip sprt v14 transmit hold-time 250`	Specifies SPRT modem-relay parameters. • **receive playback hold-time** *milliseconds*—Configures the time in ms to hold incoming data in the V.14 receive queue. Range is 20 to 250. Default is 50. • **transmit hold-time** *milliseconds*—Configures the time to wait, in ms, after the first character is ready before sending the SPRT packet. Range is 10 to 30. Default is 20. • **transmit maximum hold-count** *characters*—Configures the number of V.14 characters to be received on the ISDN public switched telephone network (PSTN) interface that will trigger sending the SPRT packet. Range is 8 to 128. Default is 16.

	Command or Action	Purpose
Step 5	**`mgcp package-capability`** *`package`* **Example:** `Router(config)# mgcp package-capability mdste-package`	Specifies the MGCP package capability type for the media gateway. • When the package type is entered as **mdste-package**, NoAudio Codec support is implicitly enabled because NoAudio codec can be used only when V.150.1 MER modem relay is also used in the call. NoAudio codec is not supported when there is no support for MER modem relay from the remote end. SSE-based T.38 support is implicitly enabled. **Note** If the **mgcp package-capability mdste-package** command is not entered, NoAudio support and SSE-based T.38 support are implicitly disabled. • To have the secure RTP session, you must enter the *package* argument as **srtp-package**.
Step 6	**`mgcp dtmf-relay voip codec all mode nte-gw`** **Example:** `Router(config)# mgcp dtmf-relay voip codec all mode nte-gw`	Specifies that RTP digit events are encoded using the named telephony event (NTE) format specified in RFC 2833, Section 3.0, and are transmitted in the same RTP stream as nondigit voice samples. • The payload type is negotiated by the gateways before use. The configured value for the payload type is presented as the preferred choice at the beginning of the negotiation.
Step 7	**`mgcp rtp payload-type nte 101`** **Example:** `Router(config)# mgcp rtp payload-type nte 101`	Specifies use of NTE as the payload type and 101 is the value for the NTE payload for backward compatibility in MGCP networks.
Step 8	**`exit`** **Example:** `Router(config)# exit`	Exits the current configuration mode.

Configuring the SIP Trunk

SIP SDP content includes information from both Legacy Cisco V.150 and V.150.1 MER, and SIP options include the Profile-level V.150.1 Filter and Service Parameter-level V.150.1 Filter. For a chart showing modem transport methods, see Table 3 in the "Troubleshooting Tips" section on page 32. To configure the SIP trunk, perform the following tasks:

- Configuring the Profile-level V.150.1 Filter, page 31
- Associating a SIP Trunk Security Profile with a Trunk, page 31
- Setting the Service Parameter-level V.150.1 Filter, page 32

Configuring the Profile-level V.150.1 Filter

To configure the profile-level V.150.1 filter, perform the tasks in this section. Figure 7 provides a sample screen of this configuration procedure.

Step 1 From the Cisco UCM Administration page, choose **System**.

Step 2 Choose **Security**.

Step 3 Choose **SIP Trunk Security Profile**.

Step 4 Choose **Find**.

Step 5 Choose **Add New**. The SIP Trunk Security Profile Configuration page displays in which to create a new profile.

Step 6 Verify that the SIP V.150.1 SDP Offer Filtering drop-down list exists within the Profile and has a setting of Use Default Filter.

Step 7 Enter a name in the Name field.

Step 8 Choose an appropriate value in the Incoming Transport Type field.

Step 9 Type in an appropriate value in the Incoming Port field.

Step 10 In the SIP V.150.1 SDP Offer Filtering drop-down list, select the desired filtering action.

Step 11 Click **Save**.

Associating a SIP Trunk Security Profile with a Trunk

To associate a SIP trunk security profile with a trunk, complete the tasks in this section. Figure 7 provides a sample screen of the SIP trunk profile configuration.

Step 1 From the Cisco Unified CM Administration page, choose **Device**.

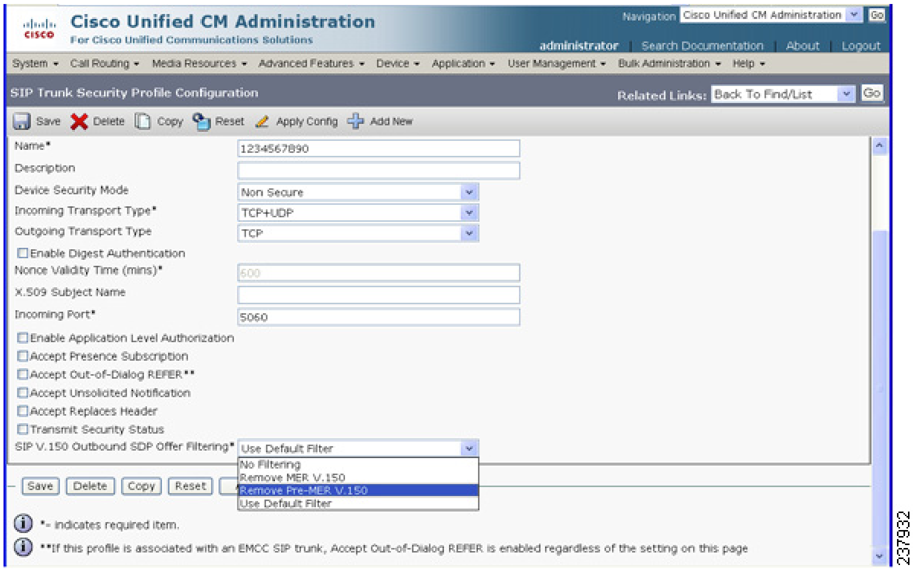
Step 2 Choose**Trunk**.

Step 3 Click **Find**.

Step 4 Choose the desired trunk.

Step 5 Find the SIP Trunk Security Profile option and choose the profile that you just created.

Figure 7 Example of SIP Trunk Security Profile Configuration



Setting the Service Parameter-level V.150.1 Filter

To set the service parameter-level V.150.1 filter, perform the tasks in this section.

Note In order for this parameter to be used by a trunk, set the SIP SDP Outbound Offer Filtering parameter of the SIP Trunk Security Profile associated with that trunk to **Use Default Filter**.

Step 1 On the Cisco Unified CM Administration page, choose **System**.

Step 2 Choose **Service Parameters**.

Step 3 Choose the **Active** server.

Step 4 Choose **Cisco CallManager Service**.

Step 5 In the Clusterwide Parameters (Device—SIP) section, verify the SIP V150 SDP Offer Filtering drop-box exists, with a default setting of No Filtering.

Step 6 Choose the **SIP V150 SDP Offer Filtering** drop-down list.

Step 7 Choose the desired filtering action.

Step 8 Choose **Save**.

Troubleshooting Tips

The following options are provided to fix interoperability issues that may arise due to some additions made to the SDP content to ensure backward compatibility with existing Cisco UCMs running Cisco Legacy V.150.1. Although according to the V.150.1 specification these additions should not impact SDP parsing, the fail-safe option to remove them is provided. These options can be configured on a per-trunk or per-cluster basis:

- No Filtering (Default)—No filtering is performed on SIP SDP content. This is the default option.
- Remove V.150.1 MER—The SIP trunk removes MER lines in outbound SDP offers. Use this value to reduce ambiguity when a trunk is connected to a pre-V.150.1 MER Cisco UCM. On the legacy Cisco UCM versions used by Cisco internally during development testing, backward compatibility with legacy V.150.1 functionality worked without this option. However, it may be needed on older Cisco UCM versions.
- Remove Pre-MER V.150.1—The SIP trunk removes any lines in outbound SDP offers that are not MER-compliant. If the trunk is to a MER-compliant LSC that cannot process an offer with pre-MER lines, choose this value. This option should be selected only when a non-Cisco LSC is misinterpreting or failing to operate on either a legacy V.150.1 offer or a MER+Legacy V.150.1 offer. A MER+Legacy V.150.1 offer can be identified by the presence of an "a=vndpar 2 15 2 ##" line at the end of the SDP. If third parties have coded their parsers appropriately, this option should not need to be used; it is mentioned here as a precaution.

Table 3 Chart of Modem Transport Methods

	Secure Terminal Unit (STU)	**On-net STE (Secure Communication between IP Secure Endpoint and Line-Side STE Endpoint Gateway)**	**Off-net STE (PSTN)**	**IP Secure Endpoint**
Secure Terminal Unit	voice band data [1]	voice band data	voice band data	None
On-net Secure Terminal Equipment (Secure Communication Between IP Secure Endpoint and Line-Side STE Endpoint Gateway)	voice band data	voice band data or V.150.1 modem relay	voice band data or V.150.1 Modem Relay	V.150.1 modem relay
Secure Terminal Equipment (STE) (PSTN)	voice band data	voice band data	voice band data	V.150.1 modem relay
IP Secure Endpoint	None	V.150.1 modem relay	V.150.1 modem relay	IP

1. voice band data (VDB) = modem Pass-through

Note The type of V.150.1 negotiated is determined by the parties involved in the call. If all components (Cisco UCM, gateways, endpoints) are SCIP-216 (MER)-compliant, the SCIP-216 (MER) implementation of V.150.1 will be used. If one or more of the components are using a pre-SCIP -216 implementation of V.150.1 (legacy), the pre-SCIP implementation of V.150.1 will be used. This also will be the case for the MoIP call.

What to Do Next

For more information on configuring SIP trunks in Cisco Unified Communications Manager 8.0(2), see *Understanding Cisco Unified Communications Manager Trunk Types.*

For additional information about SIP and configuring SIP trunks, see *Understanding Session Initiation Protocol.*

Verifying and Troubleshooting the Cisco V.150.1 MER Configuration

To verify and troubleshoot the configuration of the Cisco V.150.1 MER feature, perform the steps in this section. The **show** commands provide information about the configuration. The **debug** commands are useful when problems are apparent in the system. The information in Step 10 provides guidelines for ensuring a correct configuration. Table 4 in Step 9 provides a list of symptoms that may occur and possible resolutions to those problems.

SUMMARY STEPS

1. **show voice dsp active**
2. **show call active voice**
3. **show stcapp device voice-port 1/0/0**
4. **debug voice application stcapp all (device registration)**
5. **debug voice application stcapp all (line-side call setup)**
6. **debug voip rtp session named**
7. **debug mgcp packets (registration)**
8. **debug mgcp packets**
9. **debug mgcp all (MGCP trunk)**
10. Review the information for compliance of your configuration.

DETAILED STEPS

Step 1 **show voice dsp active**

Use the **show voice dsp active** command to display status information for all DSP voice channels:

```
Router# show voice dsp active

----------------------------FLEX VOICE CARD 1 -----------------------
                        *DSP ACTIVE VOICE CHANNELS*
DSP    DSPWARE            VOX DSP                 SIG DSP          PAK   TX/RX
TYPE   VERSION    CODEC     NUM CH TS VOICEPORT SLT NUM CH TS RST AI ABRT PACK COUNT
====== ========== ========  === == == ========= === === == == === == ==== ===
C5510    28.0.136 modem-rel 001 01 05 1/0/0     001 001 04 06   0  0    0    2912/3533
C5510    28.0.136 modem-rel 001 02 23 1/0:23    001 002 11 23   0  0    0    3458/3021
------------------------END OF FLEX VOICE CARD 1 --------------------
```

Step 2 **show call active voice**

Use the **show call active voice** command to display call information for voice calls in progress:

```
Router# show call active voice

.
.
.

Modem Relay Mode = signaling-assisted
Modem Relay Local Rx Speed=9600 bps
```

```
Modem Relay Local Tx Speed=9600 bps
Modem Relay Remote Rx Speed=19200 bps
Modem Relay Remote Tx Speed=19200 bps
Modem Relay Phy Layer Protocol=v32
Modem Relay Ec Layer Protocol=v14
SPRTInfoFramesReceived=0
SPRTInfoTFramesSent=0
SPRTInfoTFramesResent=0
SPRTXidFramesReceived=0
SPRTXidFramesSent=1
SPRTTotalInfoBytesReceived=806778
SPRTTotalInfoBytesSent=806562
SPRTPacketDrops=0
```

Step 3 **show stcapp device voice-port 1/0/0**

Use the **show stcapp device voice-port 1/0/0** command to display call information for voice calls on a specific port:

```
Router# show stcapp device voice-port 1/0/0

Port Identifier:  1/0/0
Device Type:      ALG
Device Id:        6
Device Name:      AN1A6D001760200
Device Security Mode : None
Modem Capability: Both
Device State:     IS
Diagnostic:       None
Directory Number: 2011
Dial Peer(s):     100
Dialtone after remote onhook feature: activated
Busytone after remote onhook feature: not activated
Last Event:       STCAPP_CC_EV_CALL_FEATURE
Line State:       ACTIVE
Line Mode:        CALL_BASIC
Hook State:       OFFHOOK
mwi:              DISABLE
vmwi:             OFF
mwi config:       Both
Privacy:          Not configured
PLAR:             DISABLE
Callback State:   DISABLED
CWT Repetition Interval: 0 second(s) (no repetition)
Number of CCBs:   1
Global call info:
    Total CCB count      = 1
    Total call leg count = 2

Call State for Connection 1 (ACTIVE): TsConnected
Connected Call Info:
   Call Reference: 28055870
   Call ID (DSP):  52
   Local IP Addr:  10.10.10.139
   Local IP Port:  18258
   Remote IP Addr: 10.10.10.139
   Remote IP Port: 17748
   Calling Number: 2011
   Called Number:  3011
   Codec:          g711ulaw
   SRTP:           off
MER Capabilites Active:
Capability and Version : 0x20110000
Modulation and RFC2833 : 0xF0000005
```

```
SPRT Max Payload Chan0 : 0
SPRT Max Payload Chan2 : 0
SPRT Max Payload Chan3 : 0
SPRT Max WinSize Chan2 : 0
SSE Standard Support   : 0x5
SSE Vendor Support     : 0x5
NSE Payload Value      : 0
RFC2833 Payload Value  : 101
SSE Payload Value      : 0
SPRT Payload Value     : 0
NoAudio Payload Value  : 0
```

Step 4 **debug voice application stcapp all (device registration)**

Use the **debug voice application stcapp all** command to display debugging information for the components of the STCAPP:

```
Router# debug voice application stcapp all

*Jan  4 20:45:50.877: 1/0/0:     Registering device
*Jan  4 20:45:50.877: 1/0/0: stcapp_register_device

.
.
.

*Jan  4 20:45:51.881: sccp_parse_control_msg: glob_ccm->version 9
*Jan  4 20:45:51.881: SCCP(AN43E17E8B90200)rcvd RegisterAckMessage
*Jan  4 20:45:51.881: sccp_appl_service_stop_timer: Stop A69DA3C timer
*Jan  4 20:45:51.881: sccp_parse_control_msg_v1: rcvd register ack, ka_interval 30, for
prof_id 0, appl_type 4 negotiated sccp version 21
*Jan  4 20:45:51.881:  RegisterAck msg rcvd in hex -
81 0 0 0 1E 0 0 0 4D 2F 44 2F 59 0 0 0 3C 0 0 0 15 20 F1 FF

.
.
.

Jan  4 20:45:51.881: sccp_parse_control_msg: glob_ccm->version 9
*Jan  4 20:45:51.881: SCCP(AN43E17E8B90200)rcvd CapabilitiesReqMessage
*Jan  4 20:45:51.881: sccp_generate_msg: msg_id 16 msg_len 296 pak_size 304
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: Codec list with pkt_period (cnt
16) -
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 257
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 257257
30,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 112
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 112112
20,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 114
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 114114
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 299
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 299299
20,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 300
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 300
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: v150_mr.cap_n_ver: 0x1120
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: v150_mr.mod_n_2833:
0xFF0F00F0300 0,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 301
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 301
```

```
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21:
v150_sprt_payload.chan0_max_payload: 35840
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21:
v150_sprt_payload.chan2_max_payload: 33792
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21:
v150_sprt_payload.chan3_max_payload: 35840
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21:
v150_sprt_payload.chan2_max_windows: 2048301 0,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 302
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 302
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: v150_sse.standdard_field
0x5000000302 0,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 111
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 111111
20,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 113
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 113113
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 4
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 44 20,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 2
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 22 20,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 11
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 1111
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 12
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 1212
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 15
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 1515
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 11
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 1111
220,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_rec->codec = 86
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: msg_cap->payload_caps = 86
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: codec_params=0300000086 120,
*Jan  4 20:45:51.881: sccp_send_capabilities_rsp_msg_v21: CapRes msg txed in hex(including
header) - pak->datagramsize 304, actual_len 272

*Jan  4 20:45:51.881: sccp_print_hex_msg: Len:272 Hex:
28 01 00 00 15 00 00 00 10 00 00 00 10 00 00 00 01 01 00 00 1E 00 00 00 00 00 00 00 00
00 00 70 00 00 00 14 00 00 00 00 00 00 00 00 00 00 00 72 00 00 00 DC 00 00 00 00 00 00 00
00 00 00 00 2B 01 00 00 14 00 00 00 00 00 00 00 00 00 00 00 2C 01 00 00 00 00 00 00 00
11 20 FF 0F 00 F0 2D 01 00 00 00 00 00 00 8C 00 84 00 8C 00 08 00 2E 01 00 00 00 00 00 00
05 00 00 00 00 00 00 00 6F 00 00 00 14 00 00 00 00 00 00 00 00 00 00 00 71 00 00 00 DC 00
00 00 00 00 00 00 00 00 00 00 04 00 00 00 14 00 00 00 00 00 00 00 00 00 00 00 02 00 00 00
14 00 00 00 00 00 00 00 00 00 00 00 0B 00 00 00 DC 00 00 00 00 00 00 00 00 00 00 00 0C 00
00 00 DC 00 00 00 00 00 00 00 00 00 00 00 0F 00 00 00 DC 00 00 00 00 00 00 00 00 00 00 00
0B 00 00 00 DC 00 00 00 00 00 00 00 00 00 00 00 56 00 00 00 78 00 00 00 03 00 00 00 00 00
00 00
```

Step 5 **debug voice application stcapp all (line-side call setup)**

The **debug voice application stcapp all** can also be used to display debug information for call setup on the line-side:

```
Router# debug voice application stcapp all

.
.
.
```

```
*Jan  4 20:56:33.266: sccp_parse_control_msg: glob_ccm->version 9
*Jan  4 20:56:33.266: SCCP(AN43E17E8B90200)rcvd OpenReceiveChannel
*Jan  4 20:56:33.266: OpenReceviceChannel msg rcvd in hex -
5 1 0 0 32 19 AC 1 35 0 0 1 14 0 0 0 4 0 0 0 0 0 0 0 0 0 0 0 32 19 AC 1 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 65 0 0 0 A 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 A A A 8B 0 0 0 0 0 0 0 0 0 0 0 0 A0 F 0 0 0 0
0 0 0 0 0 0 1 0 0 0 0 0 11 20 FF F 0 F0 0 0 0 0 0 0 0 0 5 0 0 0 0 0 0 0 0 65 0 0 0 0 0 0
*Jan  4 20:56:33.266: OpenReceiveChannelMsg Info:
conference_id = 28055858, pass_through_party_id = 16777269
msec_pkt_size = 20, compression_type = 4
qualifier_in.ecvalue = 0, g723_bitrate = 0, call_ref = 28055858
stream_pass_through_id = 0, rfc2833_payload_type = 101
codec_dynamic_payload = 0, codec_mode = 0
Encryption Info :: algorithm_id 0, key_len 0, salt_len 0
requestedAddrType = 0, source_ip_addr.ipAddrType = 0, source_ip_addr = 10.10.10.139,
source_port_number = 4000,
audio_level_adjustment = 0
*Jan  4 20:56:33.266: v150 latent caps active:
modem relay cap and version: 0x20110000 modulation and rfc2833: 0xF0000FFF
sprt max payload for chan0: 0 chan2: 0 chan3: 0, max window for chan2: 0
sse standard support filed: 0x5 vendor support filed: 0x0
payload nse 0 rfc2833 101 sse 0 v150_sprt 0 noaudio 0
*Jan  4 20:56:33.266:  sccp_dcapi_extract_and_validate_srtp_context
*Jan  4 20:56:33.266: STCAPP:stcapp_get_dcb_and_lcb
*Jan  4 20:56:33.266: 1/0/0: stcapp_get_dcb_and_lcb
*Jan  4 20:56:33.266: 1/0/0: stcapp_screen_api_event
*Jan  4 20:56:33.266: 1/0/0:     event:STCAPP_DC_EV_MEDIA_OPEN_RCV_CHNL received.
*Jan  4 20:56:33.266: 1/0/0: stcapp_screen_open_rcv_chnl
*Jan  4 20:56:33.266: 1/0/0:     active_ccb=0x11544A0, media_state is NO_MEDIA
*Jan  4 20:56:33.266: 1/0/0: ==> Received event:STCAPP_DC_EV_MEDIA_OPEN_RCV_CHNL
*Jan  4 20:56:33.266: 1/0/0:     Call State:PROCEEDING
*Jan  4 20:56:33.266: 1/0/0: stcapp_open_rcv_chnl_eh
*Jan  4 20:56:33.266: 1/0/0:     call_ref=28055858
*Jan  4 20:56:33.266: 1/0/0: stcapp_get_ccb_ptr
*Jan  4 20:56:33.266: 1/0/0:     received ORC: rcv payload=101
*Jan  4 20:56:33.266: 1/0/0: stcapp_set_up_voip_leg
*Jan  4 20:56:33.266: 1/0/0: stcapp_get_ccb_ptr
*Jan  4 20:56:33.266: 1/0/0:     In stcapp_set_up_voip_leg, local port allocated 21240
*Jan  4 20:56:33.266: 1/0/0: stcapp_set_up_modem_parms
*Jan  4 20:56:33.266: STCAPP:Codec: 5 ptime :20, codecbytes: 160
*Jan  4 20:56:33.266: 1/0/0:     CCM directive -> enabling MER modem relay
*Jan  4 20:56:33.266: 1/0/0:     MR parms: sprt_retries=12, sprt_latency=200,
sprt_rx_v14_pb_hold_time=50, sprt_tx_v14_hold_time=20, sprt_tx_v14_hold_count=16,
gw_xid=1, dictsize=1024, stringlen=32, compressdir=3, sse_red_interval=20,
sse_red_pkt_count=3, sse_t1=1000, sse_retries=3, rfc2833_bitmap=0
*Jan  4 20:56:33.266: 1/0/0:     Info provided to RTPSPI - sess_mode:2, desired_qos 0,
codec 5, pkt_period 20,
*Jan  4 20:56:33.266: 1/0/0:     rem_port 4000, lr_port 21240, dtmf_mode 400, rcv_nte 101
nte 0
*Jan  4 20:56:33.266: 1/0/0:     Sending ccIFCallSetupRequest for voip leg
*Jan  4 20:56:33.266: 1/0/0:     ccIFCallSetRequest returned voip call id:12
*Jan  4 20:56:33.266: 1/0/0:     MER modem relay configuration passed down ? call id:12 MR
proto = 4
*Jan  4 20:56:33.266: STCAPP:stcapp_find_ccb_by_call_id:ERROR:Invalid Call ID
*Jan  4 20:56:33.266: 1/0/0: stcapp_conn_db_insert_ccb
*Jan  4 20:56:33.266: 1/0/0:     ccb=0x11544A0
*Jan  4 20:56:33.266: 1/0/0:     call ccCallConnect for voice call_id 11
*Jan  4 20:56:33.266: 1/0/0:     Media state is set to RECV_ONLY
*Jan  4 20:56:33.266: 1/0/0:     Sending dcDeviceOpenReceiveChannelAck
*Jan  4 20:56:33.266: 1/0/0:     ORChnlAck Info: codec:5, loc_ipaddr: 10.10.10.143,
loc_port:21240, chnl_id:16777269
```

```
*Jan  4 20:56:33.266: sccp_spi_orc_ack: enqueue spi evt SCCP_SPI_MEDIA_ORC_ACK,
reg_name=AN43E17E8B90200
*Jan  4 20:56:33.266: 1/0/0:     New State = CONNECTING
*Jan  4 20:56:33.270: STCAPP:Receive CC event:: call_id=12, ccb=0x11544A0
*Jan  4 20:56:33.270: 1/0/0: ==> Received event:STCAPP_CC_EV_CALL_CONNECTED for CallId: 12
*Jan  4 20:56:33.270: 1/0/0:     Call State:CONNECTING
*Jan  4 20:56:33.270: 1/0/0: stcapp_call_connected_eh
*Jan  4 20:56:33.270: 1/0/0: stcapp_create_conference
*Jan  4 20:56:33.270: 1/0/0:     Sending ccConferenceCreate to Symphony
*Jan  4 20:56:33.270: 1/0/0:     Conference created. voice call id:11, voip call id:12
*Jan  4 20:56:33.270: 1/0/0:     No state change
*Jan  4 20:56:33.270: sym_xapp_process_ccapi_events: minor is ZERO - should be non-zero
for CCAPI event
*Jan  4 20:56:33.270: sccp_generate_msg: msg_id 34 msg_len 40 pak_size 48
*Jan  4 20:56:33.270: sccp_open_receive_channel_ack_v14: going to send ack to CCM - status
0, ipaddr 10.10.10.143, port 21240, conn_id 16777269, prof_id 0
*Jan  4 20:56:33.270: sccp_open_receive_channel_ack_v14: OpenRecvChnlAck msg txed in
hex(including header) - len 48

*Jan  4 20:56:33.270: sccp_print_hex_msg: Len:48 Hex:
28 00 00 00 15 00 00 00 22 00 00 00 00 00 00 00 00 00 00 0A 0A 0A 8F F1 1D CE 99 F2 7F
E0 98 50 10 0A F4 F8 52 00 00 35 00 00 01 32 19 AC 01
.
.
.


*Jan  4 20:56:33.270: sccp_transmit_msg: sending on socket 5

*Jan  4 20:56:33.274: sccp_parse_control_msg: msg_ptr 16127364, msg_len 172, msg_id 138
*Jan  4 20:56:33.274: sccp_parse_control_msg: glob_ccm->version 9
*Jan  4 20:56:33.274: SCCP(AN43E17E8B90200)rcvd StartMediaTransmission
*Jan  4 20:56:33.274: StartMediaTrans msg rcvd in hex -
8A 0 0 0 32 19 AC 1 35 0 0 1 0 0 0 0 A A A 8B 0 0 0 0 0 0 0 0 0 0 0 0 A6 41 0 0 14 0 0 0 4
0 0 0 B8 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 32 19 AC 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 65 0 0 0 A 0 0 0 0
0 0 0 0 0 0 0 1 0 0 0 0 0 11 20 FF F 0 F0 0 0 0 0 0 0 0 0 5 0 0 0 0 0 0 0 0 65 0 0 0 0 0 0
*Jan  4 20:56:33.274: StartMediaTransmissionMsg Info:
conference_id = 28055858, pass_through_party_id = 16777269
msec_pkt_size = 20, compression_type = 4
remote_ip_addr = 10.10.10.139, remote_port = 16806
qualifier_out.precedence_value = 184, qualifier_out.ssvalue = 0
qualifier_out.max_frames_per_pkt = 0, g723_bitrate = 0, call_ref = 28055858,
stream_pass_through_id = 0 rfc2833_payload_type = 101
codec_dynamic_payload = 0, codec_mode = 0
Encryption Info :: algorithm_id 0, key_len 0salt_len 0
*Jan  4 20:56:33.274: v150 latent caps active:
modem relay cap and version: 0x20110000 modulation and rfc2833: 0xF0000FFF
sprt max payload for chan0: 0 chan2: 0 chan3: 0, max window for chan2: 0
sse standard support filed: 0x5 vendor support filed: 0x0
payload nse 0 rfc2833 101 sse 0 v150_sprt 0 noaudio 0
```

Step 6 **debug voip rtp session named**

Use the **debug voip rtp session named** command to display debug information for session establishment:

```
Router# debug voip rtp session named

*Jan  4 21:04:25.675:          s=DSP d=VoIP payload 0x65 ssrc 0x1F2A sequence 0x811B
timestamp 0x21C875DF
*Jan  4 21:04:25.675:          Pt:101    Evt:34      Pkt:0B 00 00  <Snd>>>
```

```
.
.
.
*Jan  4 21:04:25.923:          Pt:101    Evt:35      Pkt:0B 07 D0  <Snd>>>
.
.
.
*Jan  4 21:04:29.283:  <<<Rcv> Pt:118    Evt:12      Pkt:01 D8 2C
```

Step 7 **debug mgcp packets (registration)**

Use the **debug mgcp packets** command to display debug registration information for MGCP trunks:

```
Router# debug mgcp packets

*Jan  4 17:50:50.547 EDT: MGCP Packet received from 10.10.10.132:2427--->
AUEP 1581 S1/DS1-0/5@MER-CCM2GW8.cisco.com MGCP 0.1
F: X, A, I
<---

*Jan  4 17:50:50.547 EDT: MGCP Packet sent to 10.10.10.132:2427--->
200 1581
I:
X: 0
L: p:10-20, a:PCMU;PCMA;G.nX64;NoAudio;telephone-event, fmtp:"telephone-event 0-15", b:64,
e:on, gc:1, s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:10-220, a:G.729;G.729a;G.729b;telephone-event, fmtp:"telephone-event 0-15", b:8,
e:on, gc:1, s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:10-110, a:G.726-16;G.728;telephone-event, fmtp:"telephone-event 0-15", b:16, e:on,
gc:1, s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:10-70, a:G.726-24;telephone-event, fmtp:"telephone-event 0-15", b:24, e:on, gc:1,
s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:10-50, a:G.726-32;telephone-event, fmtp:"telephone-event 0-15", b:32, e:on, gc:1,
s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H;telephone-event, fmtp:"telephone-event 0-15",
b:6, e:on, gc:1, s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
L: p:30-330, a:G.723.1-L;G.723.1a-L;telephone-event, fmtp:"telephone-event 0-15", b:5,
e:on, gc:1, s:on, t:10, r:g, nt:IN;ATM;LOCAL, X+mdste/md:V150;V150merrelay,
v:T;G;D;L;H;R;ATM;SST;FXR;PRE;X+mdste;FM
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop, netwtest
<---
```

Step 8 **debug mgcp packets**

Use the **debug mgcp packets** command to display debugging information about call setup on the MGCP trunk:

```
Router# debug mgcp packets

a=cpar: a=T38FaxMaxDatagram:320
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
<---

*Jan  4 17:43:32.611 EDT: MGCP Packet received from 10.10.10.132:2427--->
CRCX 1573 S1/DS1-0/23@MER-CCM2GW8.cisco.com MGCP 0.1
C: D000000001ac193b000000F500000003
```

```
X: 17
L: p:20, a:PCMU;telephone-event, fmtp:"telephone-event 0-15,32-35", s:off, t:b8,
X+mdste/md:v150merrelay
M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
<---

*Jan  4 17:43:32.619 EDT: MGCP Packet sent to 10.10.10.132:2427--->
200 1573 OK
I: 4

v=0
c=IN IP4 10.10.10.139
m=audio 18938 RTP/AVP 0 101 100 118
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3-4
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
a=sqn:0
a=cdsc: 1 audio RTP/AVP 0 101 100 118
a=cdsc: 5 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1,3
a=cdsc: 6 image udptl t38
a=cpar: a=T38FaxVersion:3
a=cpar: a=T38MaxBitRate:33600
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxBuffer:200
a=cpar: a=T38FaxMaxDatagram:320
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
<---

*Jan  4 17:43:32.659 EDT: MGCP Packet received from 10.10.10.132:2427--->
MDCX 1574 S1/DS1-0/23@MER-CCM2GW8.cisco.com MGCP 0.1
C: D000000001ac193b000000F500000003
I: 4
X: 17
L: p:20, a:PCMU;telephone-event, fmtp:"telephone-event 32-35", s:off, t:b8,
X+mdste/md:v150merrelay
M: sendrecv
S:

v=0
o=- 4 0 IN EPN S1/DS1-0/23@MER-CCM2GW8.cisco.com
s=Cisco SDP 0
t=0 0
m=audio 17712 RTP/AVP 0 101 118
c=IN IP4 10.10.10.139
a=rtpmap:101 telephone-event
a=fmtp:101 32-35
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3
a=sqn:0
a=cdsc: 1 audio RTP/AVP 0 101 118
a=cdsc: 4 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
```

```
a=cpar: a=fmtp:120 mr=1;mg=0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1,3
<---

*Jan  4 17:43:32.663 EDT: MGCP Packet sent to 10.10.10.132:2427--->
200 1574 OK
<---

*Jan  4 17:43:38.579 EDT: MGCP Packet sent to 10.10.10.132:2427--->
NTFY 714848268 *@MER-CCM2GW8.cisco.com MGCP 0.1
X: 0
O:
<---

*Jan  4 17:43:38.579 EDT: MGCP Packet received from 10.10.10.132:2427--->
200 714848268
<---
```

Step 9 **debug mgcp all (MGCP trunk)**

Use the **debug mgcp all** command to display session information for debugging the MGCP trunk:

```
Router# debug mgcp all

*Jan  4 17:54:46.499 EDT:
//53/0776534D8005/MGCP|S1/DS1-0/23|-1|-1/<VOICE>/mgcp_xlate_call_feature_type(1062):[lvl=2
]mgcp_xlate_call_feature_type: feature 47
*Jan  4 17:54:46.499 EDT:
//-1/xxxxxxxxxxxx/MGCP/mgcp_cr_and_init_evt_node(4596):[lvl=1]$$$ the node pointer
71E1B348

*Jan  4 17:54:46.499 EDT:
//-1/xxxxxxxxxxxx/MGCP/mgcp_insert_node_to_preprocess_q(4518):[lvl=1]$$$enq to preprocess,
qhead=71E1B348, qtail=71E1B348, count 1, evtptr=71E1B348
*Jan  4 17:54:46.499 EDT:
//53/0776534D8005/MGCP|S1/DS1-0/23|-1|-1/<VOICE>/xlate_ccapi_ev(600):[lvl=1]MGCP APP gets
CC_EV_CALL_FEATURE event: major code=EV_MEDIA_EVT, minor_code(d)=121,
minor_code=v150merrelay, *pkg=67108864

.
.
.


*Jan  4 17:54:54.963 EDT:
//53/0776534D8005/MGCP|S1/DS1-0/23|-1|-1/<VOICE>/mgcp_remove_old_ack(714):[lvl=1]Removing
ack: (trans ID 1600) : 200 1600 OK
I: 5

v=0
c=IN IP4 10.10.10.139
m=audio 17748 RTP/AVP 0 101 100 118
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-35
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=rtpmap:118 v150fw/8000
a=fmtp:118 1,3-4
a=X-sqn:0

a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
```

```
a=sqn:0
a=cdsc: 1 audio RTP/AVP 0 101 100 118
a=cdsc: 5 audio udpsprt 120
a=cpar: a=sprtmap:120 v150mr/8000
a=cpar: a=fmtp:120 mr=1;mg=**MSG 00002 TRUNCATED**
**MSG 00002 CONTINUATION #01**0;CDSCselect=1;jmdelay=no;Versn=1.1;mrmods=1,3
a=cdsc: 6 image udptl t38
a=cpar: a=T38FaxVersion:3
a=cpar: a=T38MaxBitRate:33600
a=cpar: a=T38FaxRateM0anagement:transferredTCF
a=cpar: a=T38FaxMaxBuffer:200
a=cpar: a=T38FaxMaxDatagram:320
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
*Jan  4 17:54:55.047 EDT:
//53/0776534D8005/MGCP|S1/DS1-0/23|-1|-1/<VOICE>/mgcp_remove_old_ack(714):[lvl=1]Removing
ack: (trans ID 1601) : 200 1601 OK
```

Step 10 Review the following bullet items to verify compliance of your configuration:

- STE devices operate over V.150.1 and VBD (FNBDT or STUIII).
- IP Secure Endpoint devices operate only over V.150.1—there is no network-side DSP.
- In Cisco Legacy V.150.1, if you configure an SCCP endpoint with the **both** keyword, that endpoint always uses modem pass-through when establishing connections to endpoints supporting both modem-passthrough and V.150.1 modem relay, such as other SCCP ports or MGCP-controlled PSTN trunks. If V.150.1 modem relay is desired, use the **modem relay** keyword when configuring STCAPP ports.
- Use the **modem relay** keyword for STE devices to force V.150.1 when setting up STE-to-STE calls.
- Make sure the global configuration **voice service voip modem passthrough** command is configured. This command provides fallback to VBD mode when your device is communicating with a legacy Cisco SCCP gateway or an STU on a gateway running the Secure Communication Between IP Secure Endpoint and Line-Side STE Endpoint feature.
- Codec capabilities cannot be limited on an MGCP trunk. An MGCP trunk always registers with all supported codec capabilities.

Symptoms and Possible Solutions for Cisco V.150.1 MER

This section provides information about some possible problems or issues that m ay arise when you are configuring and operating the Cisco V.150.1 MER feature. Review the information in Table 4 for symptoms and possible solutions to help ensure operability of the Cisco V.150.1 MER feature in your network.

Table 4 Common Issues and Possible Solutions

Symptom	Possible Solution
STE calls fail to secure	Wrong hardware such as gateway, VICs, or DSPs. Confirm that you have the correct configuration of DSPs (5510 family of DSPs—PVDMs included).
	MGCP gateway is needed for trunks, and the SCCP gateway is needed for line-side devices. Both need to be configured on the Cisco UCM individually, but can run on same the physical gateway. The Cisco UCM supports MGCP version 0.1
	Wrong Cisco IOS software image. Verify that the "adventerprisek9" image is used for trunks.
	Legacy V.150.1 features are available only in Cisco IOS Release 12.4(4)T adventerprisek9 T-images and later releases. Cisco V.150.1 MER features are available beginning in Cisco IOS Release 15.1(4)M adventerprise9 image. Verify that the gateway is running a supported Cisco IOS image.
	Cisco IOS STE/V.150.1 configuration commands are not present.
Trunk-side/off-net calls fail to secure	The MGCP **mgcp package-capability mdste-package** command is missing from the gateway configuration.
	Verify that the MGCP trunk is configured correctly on the Cisco UCM. Verify that Enable V.150.1 subset is checked.
Delay, jitter, loss	Verify the network topology. Verify Quality of Service (QoS) settings. If using high-delay or error-prone links (for example, satellite connections), try configuring optional modem relay parameters such as SSE redundancy.
Dropped secure calls	T1 clocking errors can cause intermittent and dropped secure calls. Verify that the clocking on the T1 is accurate and correct.
On-net STE calls complete/secure, but off-net calls fail to secure	Unsupported PSTN gateway hardware or software.
	Missing PSTN gateway in the MGCP Cisco IOS/Cisco UCM configuration.
	PSTN gateway circuit errors (slips).
Started to configure trunk-side V.150.1, but all trunk-side calls fail	Mismatched MGCP PSTN gateway configuration.
	Make sure the Enable V.150.1 subset checkbox is chosen in the Cisco UCM and the **package-capability mdste-package** command is configured for calls to proceed. If one piece is configured but the other is not, all calls across the MGCP-controlled trunk will fail, not just STE calls.
No audio after transitioning from secure to unsecure mode	MAC calls and busy trigger should be 1-to-1 on analog endpoints. (The symptom is caused whenyou are receiving a second call while in secure mode; you do not hear the call waiting tone.)
The V.150.1 (MER or Legacy) capabilities are lost over the SIP trunk	Ensure that appropriate V.150.1 SDP filtering options are set for the trunk. Filtering options are set via the trunks associated SIP Trunk Security Profile and the SIP V.150 Outbound Offer SDP filtering service parameter.

Note For problems with endpoints, such as phones, see the manufacturer's troubleshooting guide.

Additional References

Related Documents

Related Topic	Document Title
Cisco IOS commands	*Cisco IOS Master Commands List, All Releases*
Voice commands	*Cisco IOS Voice Command Reference*
Information related to MGCP	*Media Gateway Control Protocol Voiceband Data Package and General Purpose Media Descriptor Parameter Package draft-stone-mgcp-vbd-07*
Detailed information about implementing fax/modem over IP	*Fax/Modem over IP*
Information about the Cisco Unified Communications Manager	• *Changes to UCR 2008, Change 1, Section 5.3.2, Assured Services Requirements* • *Cisco Unified Communications Manager (CallManager)*
Information about MGCP	• *Media Gateway Control Protocol (MGCP)* • *Media Gateway Control Protocol Voiceband Data Package and General Purpose Media Descriptor Parameter Package draft-stone-mgcp-vbd-07*
Information about SCCP	*Skinny Client Control Protocol*
Information about using Cisco UCM on SIP trunks	*Understanding Cisco Unified Communications Manager Trunk Types.*

Standards

Standard	Title
ITU-T V.150.1	• *ITU-T V.150.1, Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs, dated 01/2003* • *ITU-T V.150.1, Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs, Amendment 2 – ToIP and New SPRT Data Types Support, dated 05/2006*
SCIP-216	*Minimum Essential Requirements (MER) for V.150.1 Gateways Publication, Revision 2.0, 2 November 2007*
DoD UCR 2008	*Changes to UCR 2008, Change 1, Section 5.3.2, Assured Services Requirements*

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature	To locate and download MIBs for selected platforms, Cisco software releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 2833	*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*

Technical Assistance

Description	Link
The Cisco Support and Documentation website provides online resources to download documentation, software, and tools. Use these resources to install and configure the software and to troubleshoot and resolve technical issues with Cisco products and technologies. Access to most tools on the Cisco Support and Documentation website requires a Cisco.com user ID and password.	http://www.cisco.com/cisco/web/support/index.html

Feature Information for Cisco V.150.1 Minimum Essential Requirements

Table 5 lists the release history for this feature.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 5 lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Table 5 Feature Information for the Cisco V.150.1 Minimum Essential Requirements Feature

Feature Name	Releases	Feature Information
Secure Communication Between IP-STE Endpoint and Line-Side STE Endpoint	12.4(4)T 12.4(9)T	This feature was introduced in Cisco IOS Release 12.4(4)T. In Cisco IOS Release 12.4(9)T, this feature was implemented on the following platforms: Cisco 2801, Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3825, Cisco 3845, Cisco VG 224.
Cisco V.150.1 Minimum Essential Requirements (MER)	15.1(4)M	This feature was renamed and enhanced to provide: • V.150.1 MER modem relay support • RFC 2833 support for Events 32-35 • T.38 Annex F support • No-Audio Codec Support • Backward compatibility to existing V.150.1 implementation

Glossary

ANS—ANSwering tone.

ANSam—ANSwering tone with amplitude modulation.

AS-SIP—Assured Services SIP.

BRI—Basic Rate Interface.

CAS—channel-associated signaling. The transmission of signaling information within the voice channel.

CCM—Cisco CallManager. For updated terminology, see Cisco UCM.

CLI—command-line interface.

CM—Communications Manager.

Cisco UCM—Cisco Unified Communications Manager.

codec—compressor/decompressor.

DoD—Department of Defense.

DN—directory number.

DNS—Domain Name System.

DSP—digital signal processor.

EI—end instrument.

FNBDT—Future Narrow Band Digital Terminal. This protocol is used for transmitting secure calls over V.32 and V.34 datapumps.

FoIP—Fax over IP.

FXS—Foreign Exchange Station.

g.711 and **g.729**—ITU standards for coding analog signals into digital signals, and for audio (speech) compression and decompression.

GW—Gateway (analog endpoints). This includes analog phones, analog secure phones, analog fax machines, and analog modems.

ICT—inter-cluster trunk.

IETF—Internet Engineering Task Force.

IP—Internet Protocol.

IP-STE—Internet Protocol—Secure Terminal Equipment. Specialized encryption-capable IP phones that communicate only over V.150.1 modem relay.

ISDN—Integrated Services Digital Network. A communication protocol offered by telephone companies that permits telephone networks to carry data, voice, and other source traffic.

ISR—Integrated Services Router. Cisco 28*xx* series and 38*xx* series router.

ISR G2—Integrated Services Router Generation 2. Cisco 29*xx* and 39*xx* series routers.

ITU—International Telecommunications Union.

LSC—Local Switch Controller.

MER—Minimal Essential Requirement. This is also referred to as NSA specification SCIP-216.

MGCP—Media Gateway Control Protocol. A control and signal protocol for converting audio signals carried on public switched telephone network (PSTN) circuits to data packets carried over the internet or other packet networks. See also *Media Gateway Control Protocol Voiceband Data Package and General Purpose Media Descriptor Parameter Package* from IETF.

Modem Relay Preferred Endpoint—MER-compatible endpoint that transitions to modem relay with transmitting voice information in the audio state. Example: data-only endpoint that does not support audio capabilities and transitions to modem relay.

MoIP—Modem over IP, also referred to V.150.1 Modem Relay.

NM—network module.

NoAudio—Mechanism for avoiding audio transmission during the audio state. A modem relay preferred endpoint can use NoAudio to identify that it does not support audio capabilities. See section 4.9 in Minimum Essential Requirements (MER) for V.150.1 Gateways Publication, Revision 2.

NSA—National Security Agency.

Passthrough—This term is also referred to as voice band data.

PRI—Primary Rate Interface. An ISDN interface to primary rate access. Primary rate access is a single 64-kbps D channel plus 23 (T1) or 30 (E1) B channels for voice or data.

PSTN—public switched telephone network. A worldwide network based on copper wires, fiber-optic cables, microwave transmissions, cellular networks, communications satellites, and undersea telephone cables connected by switching centers. PSTN originally carried analog voice data, and carries analog and digital data.

PVDM—Packet Voice DSP Module (also referred to as DSP).

PVDM2—Packet Voice DSP Module version 2 (used in ISRs, ISR G2s and PVDM3).

RIC—Reason Identifier Code.

RTP—Real-time Transport Protocol. This protocol is for transmitting real-time data such as audio and video.

SCCP—Skinny Client Control Protocol. Network terminal control messaging protocol between a skinny client and the Cisco Unified Communications Manager.

SCIP–216—Secure Communications Interoperability Protocol (NSA Specification SCIP-216).

SCIP–EI—Secure Communications Interoperability Protocol-End Instrument. This refers to any MER-compliant IP endpoint that conforms to SCIP-215 section 5.3.2.21.3.

SDP—Session Description Protocol. This is the format used to describe streaming media initialization parameters.

SIP—Session Initiation Protocol.

SPRT—Simple Packet Relay Transport.

SRTP—Secure Real-time Transport Protocol.

SSE—State Signaling Event.

STCAPP—SCCP Telephony Control Application.

STE—secure terminal equipment. This refers to specialized encryption-capable BRI/analog phones that can communicate over V.150.1 modem relay or over modem pass-through.

STU—secure terminal unit. This refers to specialized encryption-capable analog phones that operate only over NSE-based modem pass-through connections.

T.38—ITU recommendation for allowing transmission of fax in real time over IP networks.

TDM—time-division multiplexing (see also PSTN).

UCR—Unified Capability Requirement.

V.90—ITU standard for 56-Kbps modems.

V.92—ITU standard providing convenience and performance improvements for dialup modems including faster connect times, faster upload speeds, and V.44 data compression.

VBD—Voice Band Data (also referred to as modem pass-through).

VIC—voice interface card.

VoIP—Voice over IP. Enables a router to carry voice traffic, for example, telephone calls and faxes, over an IP network.