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ADTRAN

Interoperability Guide

ADTRAN SBC and Cisco Unified Call Manager SIP Trunk Interoperability

This guide describes an example configuration used in testing the interoperability of an ADTRAN session border controller (SBC) and the Cisco Unified Call Manager (CUCM) private branch exchange (PBX) using a Session Initiation Protocol (SIP) trunk to provide a SIP trunk gateway to the service provider network. This guide includes the description of the network application, verification summary, and example individual device configurations for the ADTRAN SBC and the CUCM PBX products.

For additional information on configuration of the ADTRAN products, please visit the ADTRAN Support Community at https://supportforums.adtran.com

This guide consists of the following sections:

Application Overview

Increasingly, service providers are using SIP trunks to provide Voice over IP (VoIP) services to customers. ADTRAN SBCs terminate the SIP trunk from the service provider and operate with the customer's IP PBX system. A second SIP trunk from the gateway connects to the IP PBX. The SBC operates as a SIP back-to-back user agent (B2BUA). The ADTRAN SBC features normalize the SIP signaling and media between the service provider and the customer IP PBX. *Figure 1* illustrates the use of the ADTRAN SBC in a typical network deployment.

Additional information is available online at ADTRAN's Support Community, https://supportforums.adtran.com. Specific resources are listed in *Additional Resources on page 16*.

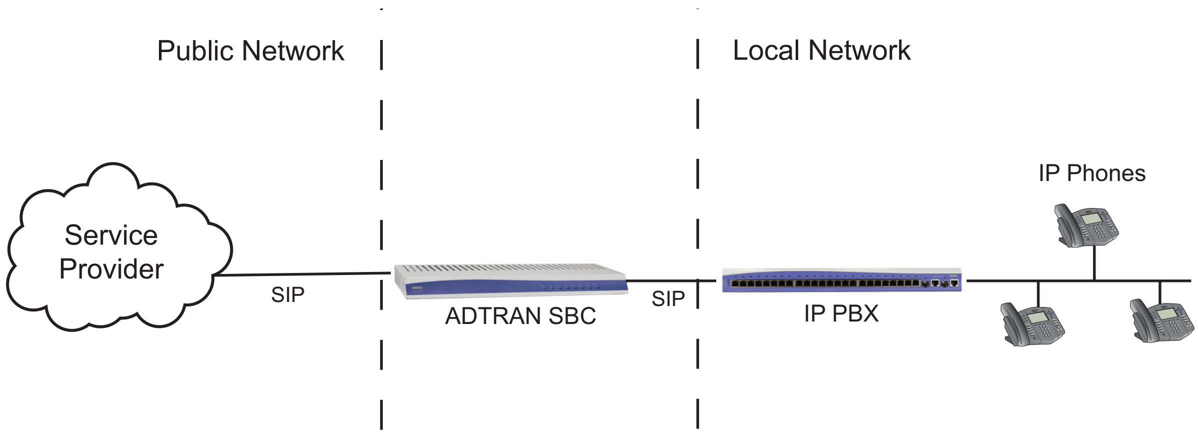


Figure 1. ADTRAN SBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN SBC and the CUCM PBX. The configuration is a typical SIP trunking application, where the ADTRAN gateway Ethernet interface provides the Ethernet wide area network (WAN) connection to the service provider network. A second Ethernet interface connects to the customer local area network (LAN). The CUCM PBX LAN interface connects to the customer LAN. Two SIP trunks are configured on the ADTRAN SBC gateway: one to the service provider and the second to the CUCM PBX. The ADTRAN SBC gateway operates as a SIP B2BUA, and outbound and inbound calls to the public switched telephone network (PSTN) are routed through the ADTRAN SBC.

The ADTRAN SBC provides SIP trunk registration to the service provider if required. Some service providers have different requirements. Consult your service provider for specific SIP trunking configuration information.

The CUCM PBX supports various phone types (including digital, H.323, and SIP IP phones). The phones register locally to the CUCM PBX. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN SBC gateway.

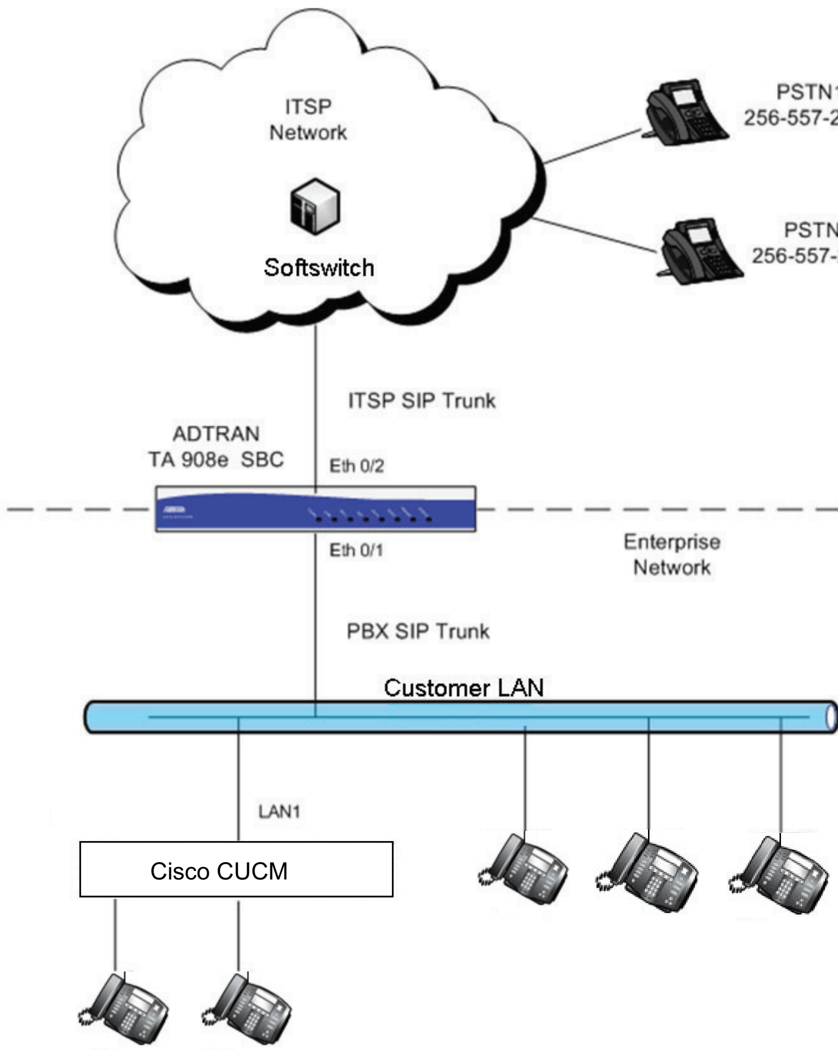


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the Cisco Unified Call Manager PBX is available on ADTRAN products with the SBC feature code as outlined in the *AOS Feature Matrix*, available online at ADTRAN's Support Forum, https://supportforums.adtran.com. The test equipment, testing parameters, and associated caveats are described in the following sections.

Equipment and Versions

The following table outlines the equipment and firmware versions used in verification testing.

Table 1. Verification Test Equipment and Firmware Versions

Product	Firmware Version
ADTRAN Total Access 908e IP Business Gateway SBC (P/N 424908L1SBC)	R10.1.0
Cisco Unified Call Manager PBX	8.6.2

Verification Performed

Interoperability verification testing focused on SIP trunk operations between the ADTRAN SBC gateway and the CUCM PBX. Other PBX features not specific to basic SIP trunking were not included in this verification. Verification testing included the following features:

- CUCM PBX SIP trunk operation with the ADTRAN SBC gateway.
- CUCM PBX SIP OPTIONS message for SIP trunk keepalive.
- Basic inbound and outbound calling with the PSTN using SIP trunking.
- Dial plan operation with the PSTN.
- Dual tone multifrequency (DTMF) operation (both RFC 2833 and in-band signaling).
- Coder-decoder (CODEC) negotiation using both G.711u and G.729.
- Call forwarding (local and external) with the PSTN.
- Call hold and retrieval with the PSTN.
- Call transfers (consultative and unassisted) with the PSTN.
- Three-way conferencing with the PSTN.
- Caller ID presentation and privacy with the PSTN.
- Voicemail operation with the PSTN.

Configuring the ADTRAN SBC Using the CLI

The SBC can be configured using either the command line interface (CLI) or the web-based graphical user interface (GUI). The following sections describe the key configuration settings required for this solution using the CLI. Refer to *Additional Resources on page 16* for more information about SBC GUI configuration.

To configure the SBC for interoperability with the CUCM PBX, follow these steps:

- *Step 1: Accessing the SBC CLI on page 5*
- *Step 2: Configuring the Basic Network Settings on page 5*
- *Step 3: Configuring Global Voice Modes for Local Handling on page 6*
- *Step 4: Configuring the Service Provider SIP Trunk on page 6*
- *Step 5: Configuring the CUCM PBX SIP Trunk on page 6*
- *Step 6: Configuring a Trunk Group for the Service Provider on page 7*

Step 1: Accessing the SBC CLI

To access the CLI on your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet** <*ip address*>), for example:

   telnet 10.10.10.1.

> NOTE: *If during the unit's setup process you have changed the default IP address (10.10.10.1), use the configured IP address.*

3. Enter your user name and password at the prompt.

> NOTE: *The AOS default user name is **admin** and the default password is **password**. If your product no longer has the default user name and password, contact your system administrator for the appropriate user name and password.*

4. Enable your unit by entering **enable** at the prompt as follows:

   >**enable**

5. If configured, enter your Enable mode password at the prompt.
6. Enter the unit's Global Configuration mode as follows:

   #**configure terminal**
   (config)#

Step 2: Configuring the Basic Network Settings

Basic network configuration includes setting up two Ethernet interfaces, one for the Ethernet WAN interface to the service provider, and the second for the Ethernet LAN interface to the CUCM PBX. Both interfaces are configured using the **ip address** <*ipv4 address*> <*subnet mask*> and **media-gateway ip primary** commands. The **ip address** command configures a static IP address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic. Enter the commands from the Ethernet interface configuration mode as follows:

For the LAN interface:
(config)#**interface ethernet 0/1**
(config-eth 0/1)#**description CUSTOMER LAN**
(config-eth 0/1)#**ip address 10.66.0.30 255.255.255.0**
(config-eth 0/1)#**media-gateway ip primary**

For the WAN interface:

(config)#**interface ethernet 0/2**
(config-eth 0/2)#**description PROVIDER WAN**
(config-eth 0/2)#**ip address 192.0.2.3 255.255.255.248**
(config-eth 0/2)#**media-gateway ip primary**
(config-eth 0/2)#**no shutdown**

Step 3: Configuring Global Voice Modes for Local Handling

Configure the ADTRAN SBC to use the local mode for call forwarding and transfer handling. By default, both of these functions are handled by the network. To change these settings, use the **voice transfer-mode local** and **voice forward-mode local** commands. Enter these commands from the Global Configuration mode. By using the **local** parameter, both commands specify allowing the unit to handle call forwarding and transfers locally.

Enter the commands as follows:

(config)#**voice transfer-mode local**
(config)#**voice forward-mode local**

Step 4: Configuring the Service Provider SIP Trunk

The first of two voice trunks that must be configured is the SIP trunk to the service provider from the ADTRAN SBC. The minimum amount of configuration is provided in this document; however, your application may require additional settings (depending on your service provider's requirements). Contact your service provider for any specific requirements beyond those listed in this document.

Use the **voice trunk** <*txx*> **type sip** command to define a new SIP trunk and activate the Voice Trunk Configuration mode for the individual trunk. From the Voice Trunk Configuration mode, you can provide a descriptive name for the trunk and define the SIP server's primary IPv4 address (or host name). Use the **description** <*text*> command to label the trunk. Use the **sip-server primary** <*ipv4 address* | *hostname*> command to define the host name or IPv4 address of the primary server to which the trunk sends call-related SIP messages.

Enter the commands as follows:

(config)#**voice trunk T01 type sip**
(config-T01)#**description Provider**
(config-T01)#**sip-server primary 198.51.100.2**

Step 5: Configuring the CUCM PBX SIP Trunk

The second of two voice trunks that must be configured is the SIP trunk to the CUCM PBX from the ADTRAN SBC. The trunk is also configured using the **voice trunk** <*txx*> **type sip**, **description** <*text*>, and **sip-server primary** <*ipv4 address* | *hostname*> commands. Use the **sip-server primary** <*ipv4 address* | *hostname*> command to set the server address to the CUCM PBX IP address. In addition, the CUCM PBX will control call transfers, so enter the **transfer-mode network** command in the trunk's configuration. Use the **grammar from host local** command to specify that the IP address of the interface is used in the SIP FROM field for outbound messages.

Enter the commands as follows:

(config)#**voice trunk T02 type sip**
(config-T02)#**description PBX**
(config-T02)#**sip-server primary 10.66.0.31**
(config-T02)#**transfer-mode network**
(config-T02)#**grammar from host local**

Step 6: Configuring a Trunk Group for the Service Provider

After configuring the two SIP trunks, configure an individual trunk group for the service provider trunk account. The previously created trunks are added to the trunk group, which is then used to assign outbound call destinations (local calls, long distance calls, etc.). A cost is also assigned to each **accept** template in the trunk group.

Use the **voice grouped-trunk** <*name*> command to create a trunk group and to enter the Voice Trunk Group Configuration mode. The **trunk** <*txx*> command adds an existing trunk to the trunk group, so that outbound calls can be placed out of that particular trunk. The <*txx*> parameter specifies the trunk identity where *xx* is the trunk ID number.

Use the **accept** <*template*> command to specify number patterns that are accepted for routing calls out of the trunk. Use the **no** form of this command to remove a configured dial pattern. The <*template*> parameter is specified by entering a complete phone number or using wildcards to help define accepted numbers.

Valid characters for templates are as follows:

0 - 9	Match the exact digit(s) only
X	Match any single digit 0 through 9
N	Match any single digit 2 through 9
M	Match any single digit 1 through 8
$	Match any number string dialed
[]	Match any digit in the list within the brackets (for example, [1,4,6])
,()	Formatting characters that are ignored but allowed
-	Use within brackets to specify a range, otherwise ignored

The following are example template entries using wildcards:

1) NXX-XXXX	Match any 7-digit number beginning with 2 through 9
2) 1-NXX-NXX-XXXX	Match any number with a leading 1, then 2 through 9, then any 2 digits, then 2 through 9, then any 6 digits
3) 555-XXXX	Match any 7-digit number beginning with 555
4) XXXX$	Match any number with at least 5 digits
5) [7,8]$	Match any number beginning with 7 or 8
6) 1234	Match exactly 1234

Some template number rules:

1. All brackets must be closed with no nesting of brackets and no wildcards within the brackets.
2. All brackets can hold digits and commas, for example: [1239]; [1,2,3,9]. Commas are implied between numbers within brackets and are ignored.
3. Brackets can contain a range of numbers using a hyphen, for example: [1-39]; [1-3,9].
4. The $ wildcard is only allowed at the end of the template, for example: 91256$; XXXX$.

Enter the commands as follows:

(config)#**voice grouped-trunk PROVIDER**
(config-PROVIDER)#**trunk T01**
(config-PROVIDER)#**accept N11 cost 0**
(config-PROVIDER)#**accept NXX-XXXX cost 0**
(config-PROVIDER)#**accept NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 1-NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 011-X$ cost 0**

Step 7: Configuring a Trunk Group for the PBX

After configuring a trunk group for the service provider, create a trunk group for the CUCM PBX trunk account. Create the trunk group using the **voice grouped-trunk** <*name*> command. Add an existing trunk to the trunk group using the **trunk** <*txx*> **cost** <*value*> command. The outbound allowed calls are defined using the **accept** <*template*> command, and are assigned a cost using the **cost** <*value*> parameter, as described in *Step 6: Configuring a Trunk Group for the Service Provider on page 7*. Enter the commands from the Global Configuration mode as follows:

(config)#**voice grouped-trunk PBX**
(config-PBX)#**trunk T02**
(config-PBX)#**accept 256-555-01XX cost 0**

Step 8: Enabling Media Anchoring

Media anchoring is an SBC feature that routes RTP traffic through the ADTRAN SBC gateway. Minimum configuration for media anchoring includes enabling the feature using the **ip rtp media-anchoring** command from the Global Configuration mode. The RTP symmetric filter works in conjunction with media anchoring to filter nonsymmetric RTP packets. Enable RTP symmetric filtering using the **ip rtp symmetric-filter** command. Enter the commands as follows:

(config)#**ip rtp media-anchoring**
(config)#**ip rtp symmetric-filter**

> NOTE
>
> *For more information about configuring additional media anchoring settings, refer to the configuration guide Configuring Media Anchoring in AOS, available online at http://supportforums.adtran.com.*

Step 9: Configuring the Double reINVITE Preference

After configuring the trunks, trunk groups, and any media anchoring settings, determine whether a double reINVITE is preferred globally for all calls in the system using the **ip sip prefer double-reinvite** command. Calls that typically require a double reINVITE are forwarded calls and attended transfers. When these calls connect, a double reINVITE is initiated.

By default, the system is configured so that double reINVITEs are preferred. If a transfer call involves a SIP trunk operating in the local transfer mode, a double reINVITE is executed regardless of this preference setting. To avoid extra SIP messaging in situations where it is not necessary, set this feature to not prefer double reINVITEs by entering the **no** version of the **ip sip prefer double-reinvite** command from the Global Configuration mode.

Enter the command as follows:

(config)#**no ip sip prefer double-reinvite**

Step 10: Configuring SIP Privacy (Optional)

The ADTRAN SBC supports SIP user privacy by using the P-Asserted-Identity and Privacy SIP headers. Enable P-Asserted-Identity (PAI) and SIP privacy support by entering the **ip sip privacy** command from the Global Configuration mode and by entering the **trust-domain** command for voice trunks (to add PAI). Enter the commands as follows:

(config)#**ip sip privacy**
(config)#**voice trunk T01 type sip**
(config-T01)#**trust-domain**
(config-T01)#**exit**
(config)#**voice trunk T02 type sip**
(config-T02)#**trust-domain**

ADTRAN SBC Sample Configuration

The following example configuration is for a typical installation of an ADTRAN SBC gateway or router with SIP trunking configured to the service provider and the CUCM PBX. This configuration was used to validate the interoperability between the ADTRAN SBC and the CUCM PBX. Only the commands relevant to the interoperability configuration are shown.

> NOTE
> *The configuration parameters entered in this example are sample configurations only, and only pertain to the configuration of the SIP trunking gateway functionality. This application should be configured in a manner consistent with the needs of your particular network. CLI prompts have been removed from the configuration example to provide a method of copying and pasting configurations directly from this guide into the CLI. This configuration should not be copied without first making the necessary adjustments to ensure it will function properly in your network.*

```
!
interface eth 0/1
  description CUSTOMER LAN
  ip address 10.66.0.30 255.255.255.0
```

```
  media-gateway ip primary
  no shutdown
!
!
interface eth 0/2
  description PROVIDER WAN
  ip address 192.0.2.3 255.255.255.248
  media-gateway ip primary
  no shutdown
!
!
voice transfer-mode local
voice forward-mode local
!
voice trunk T01 type sip
  description Provider
  sip-server primary 198.51.100.2
  trust-domain
!
!
voice trunk T02 type sip
  description PBX
  sip-server primary 10.66.0.31
  trust-domain
  grammar from host local
  transfer-mode network
!
!
voice grouped-trunk PROVIDER
  trunk T01
  accept N11 cost 0
  accept NXX-XXXX cost 0
  accept NXX-NXX-XXXX cost 0
  accept 1-NXX-NXX-XXXX cost 0
  accept 011-X$ cost 0
!
!
voice grouped-trunk PBX
  trunk T02
  accept 256-555-01XX cost 0
!
!
ip sip privacy
!
!
no ip sip prefer double-reinvite
!
```

```
!
ip rtp media-anchoring
ip rtp symmetric-filter
!
end
```

Configuring the Cisco Unified Call Manager PBX

The CUCM PBX system supports many features. The following sections describe the minimum configuration required for SIP trunking interoperability with the ADTRAN SBC. To configure the CUCM PBX using the GUI, follow these steps:

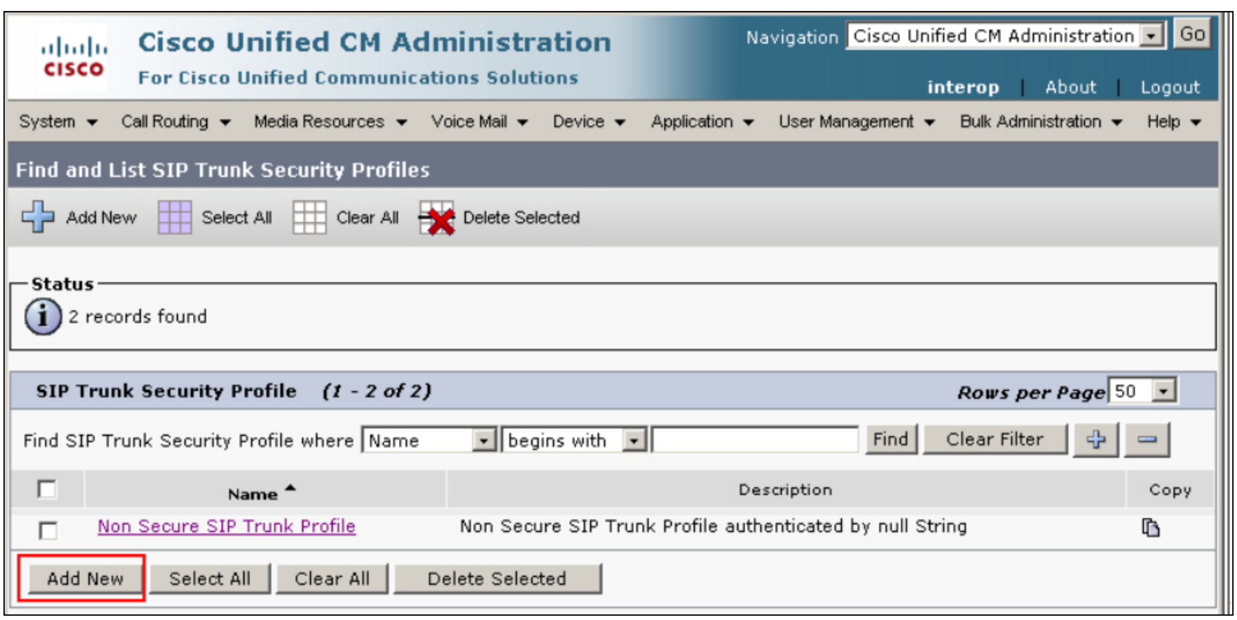
- *Step 1: Connecting to the CUCM PBX GUI on page 11*
- *Step 2: Creating a New SIP Trunk Security Profile on page 11*
- *Step 3: Configuring the SIP Trunk Security Profile on page 12*
- *Step 4: Configuring the CUCM PBX SIP Trunk on page 13*
- *Step 5: Creating a SIP Profile on page 15*

Step 1: Connecting to the CUCM PBX GUI

The CUCM PBX system is configured using the Cisco Unified CM Manager software. Refer to the Cisco documentation for detailed instructions about accessing the GUI.

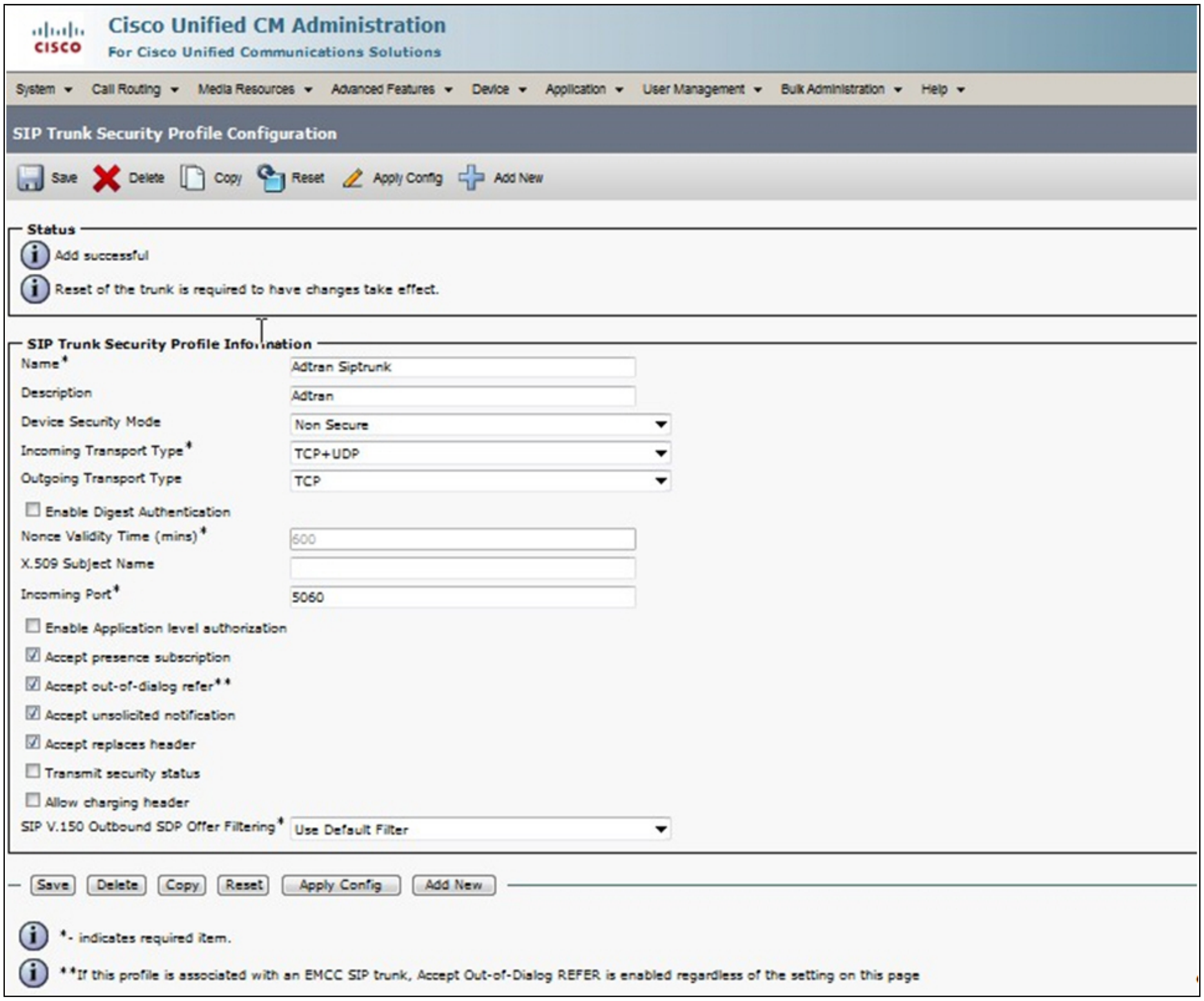
Step 2: Creating a New SIP Trunk Security Profile

Once you have accessed the CUCM PBX GUI, you must create a SIP trunk security profile. Navigate to **System** > **Security Profile** > **SIP Trunk Security Profile**. Select **Add New** to add a new SIP trunk security profile.



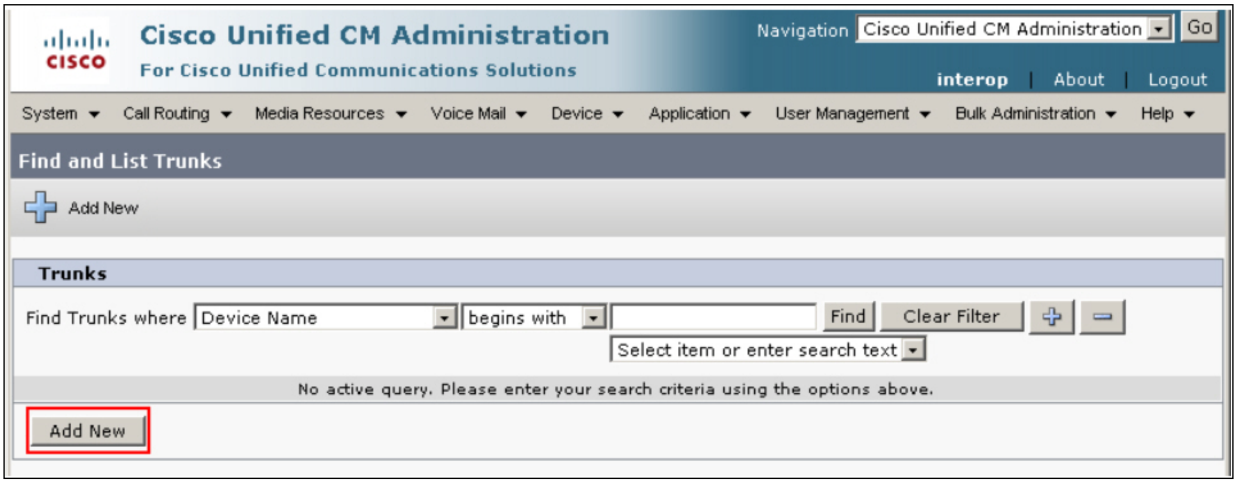
Step 3: Configuring the SIP Trunk Security Profile

In the **SIP Trunk Security Profile Configuration** menu, specify the name of the profile, specify the **Device Security Mode** as **Non Secure** (indicating unencrypted SIP signaling), the **Incoming Transport Type** as **TCP+UDP** (indicating the CUCM PBX listens for both protocols), the **Outgoing Transport Type** as **TCP** (indicating the CUCM PBX only uses TCP to initiate SIP signaling), and the **Incoming Port** as **5060**. Select **Save** in the top right of the menu to save these settings.

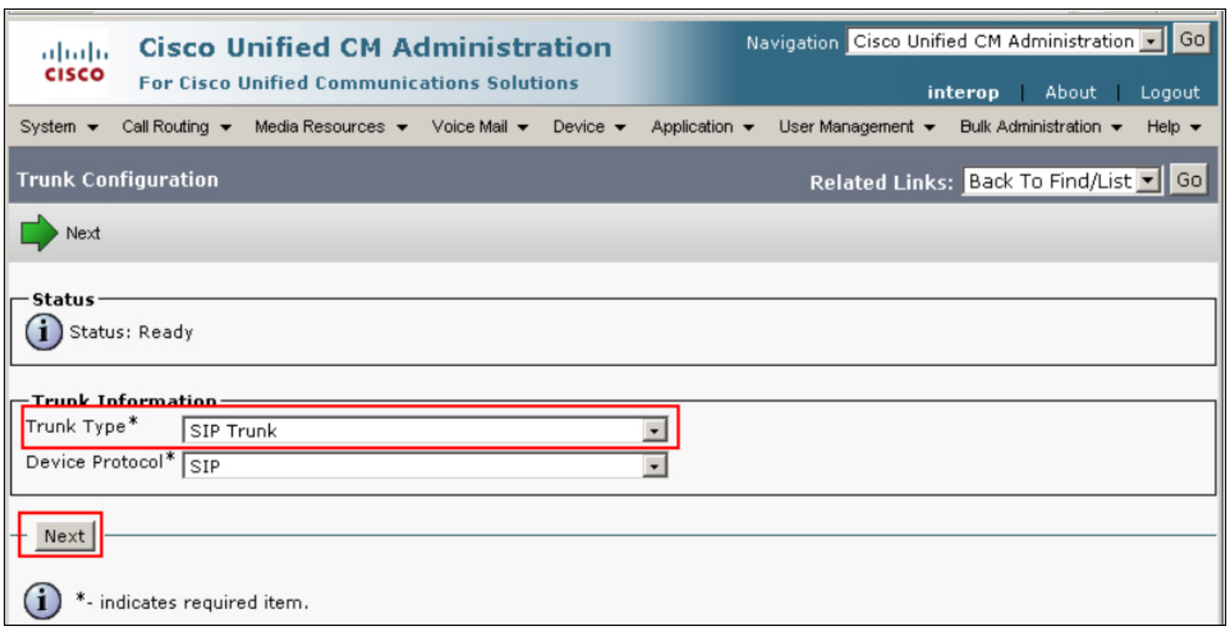


Step 4: Configuring the CUCM PBX SIP Trunk

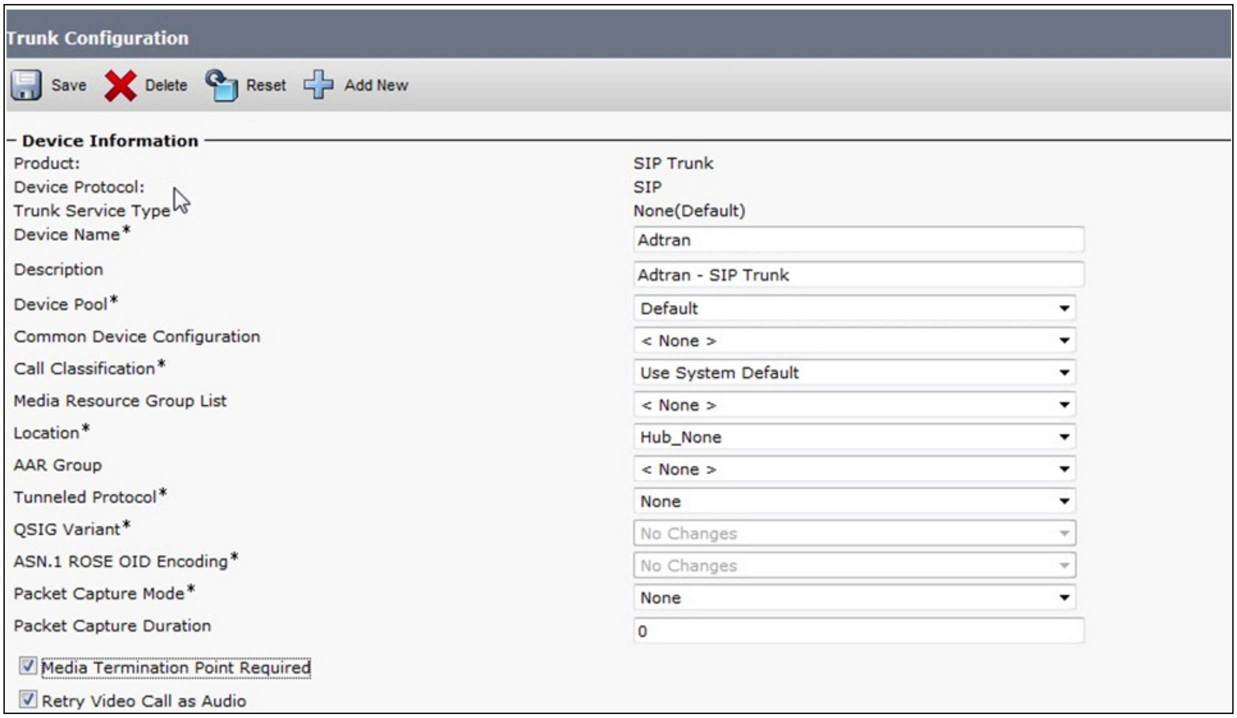
After configuring the CUCM PBX SIP trunk security profile, navigate to **Device** > **Trunk**. Select **Add New** to begin configuring the CUCM PBX SIP trunk.



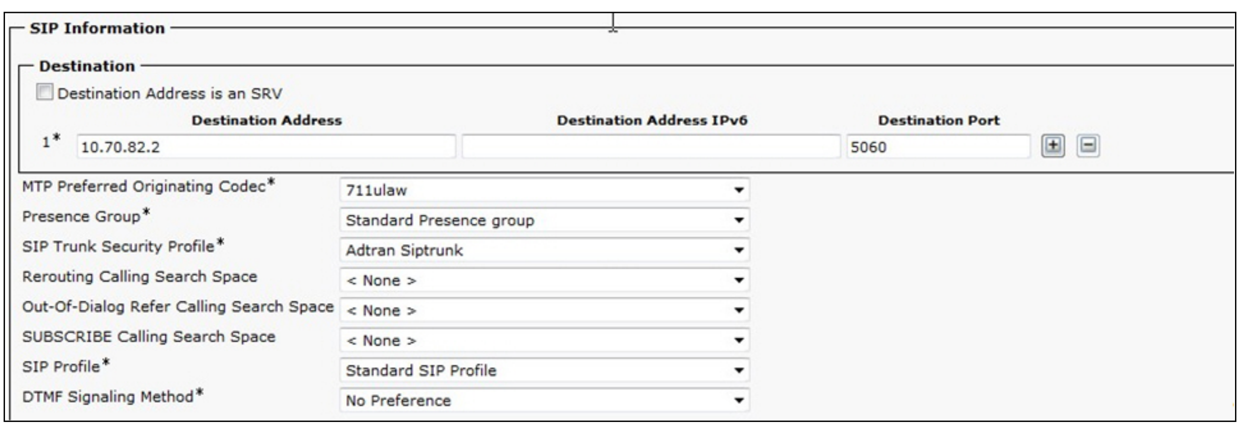
Select **SIP Trunk** from the **Trunk Type** drop-down menu. Once selected, the **Device Protocol** field is automatically specified as **SIP**. Select **Next** to continue.



Enter the appropriate information for the SIP trunk. Enter a name and description for the trunk in the appropriate fields. Specify the **Device Pool** as **Default**, and select the **Media Termination Point Required** check box so that the CUCM PBX includes SDP information in the initial SIP invite message.

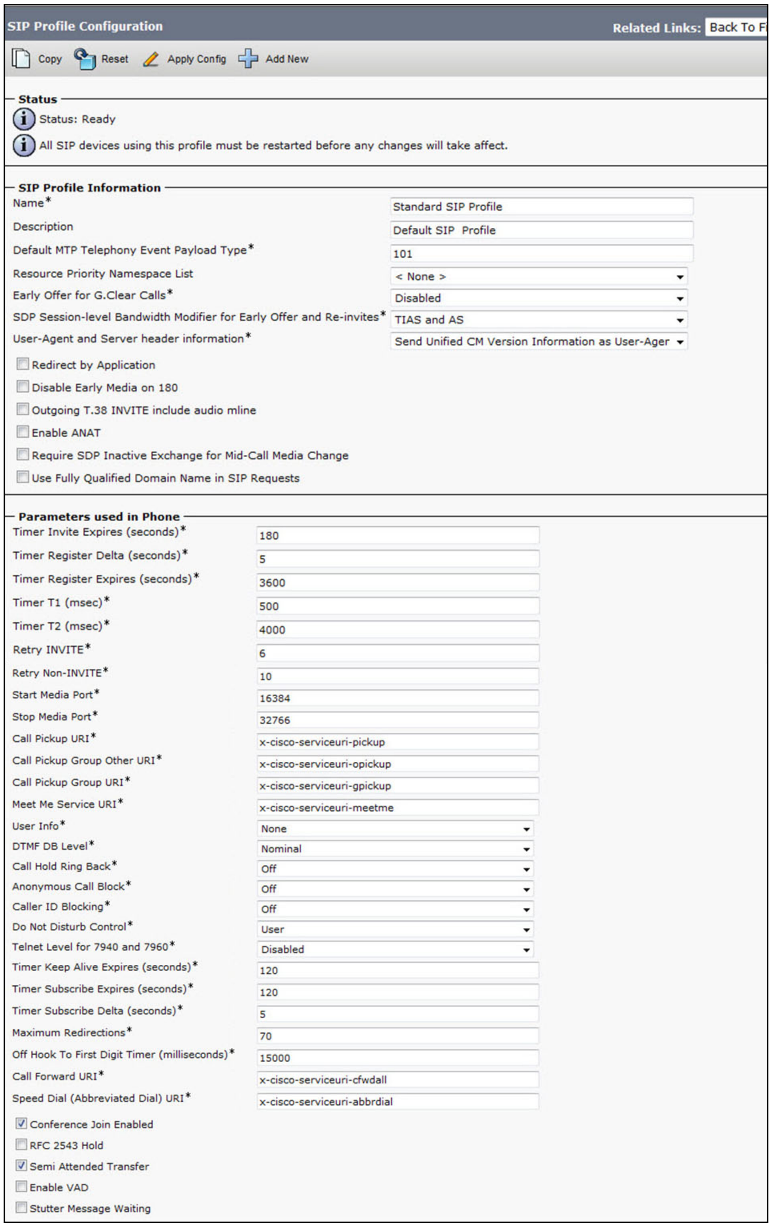


Next, navigate to the **SIP Information** section of the menu. Enter the IP address of the IP office in the **Destination Address** field and port **5060** in the **Destination Port** field. Select the name of the SIP trunk profile created in *Step 2: Creating a New SIP Trunk Security Profile on page 11* from the **SIP Trunk Security Profile** drop-down menu, and specify the **DTMF Signaling Method** as **RFC2833**. Once you have entered these settings, select **Save** to save the configuration.



Step 5: Creating a SIP Profile

After creating the CUCM PBX SIP trunk security profile and the SIP trunk, create a SIP profile on the CUCM PBX. Navigate to **Device** > **Device Settings** > **SIP Profile**. You can choose to use an existing profile, or you can create a new SIP profile for the trunk. In this verification test, the default SIP profile was used. The illustration below shows the default values for a standard SIP profile on a CUCM running firmware version 8.6.2.



NOTE

These settings may vary depending on the service provider requirements.

The CUCM PBX is now configured for interoperability with the ADTRAN SBC gateway.

Additional Resources

There are additional resources available to aid in configuring your ADTRAN SBC unit. Many of the topics discussed in this guide are complex and require additional understanding, such as using the CLI, SBC in AOS, and ANI/DNIS substitution. The documents listed in *Table 2* are available online at ADTRAN's Support Forum at https://supportforums.adtran.com.

Table 2. Additional ADTRAN Documentation

Feature	Document Title
All AOS Commands Using the CLI	*AOS Command Reference Guide*
ANI and DNIS Substitution	*Enhanced ANI/DNIS Substitution in AOS*
SBC Product Overview	*Session Border Controllers in AOS*
Media Anchoring	*Configuring Media Anchoring in AOS*
Configuring SIP Trunks on a Total Access 900 Series Using the GUI	*Total Access 900/900e SIP Trunk Quick Configuration Guide*