



Interoperability Guide

ADTRAN SBC and Cisco Call Manager Express SIP Trunk Interoperability

This guide describes an example configuration used in testing the interoperability of an ADTRAN session border controller (SBC) and the Cisco Call Manager Express 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 Cisco Call Manager Express products.

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

This guide consists of the following sections:

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- *Verification Performed on page 4*
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- *Additional Resources on page 17*

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 17*.

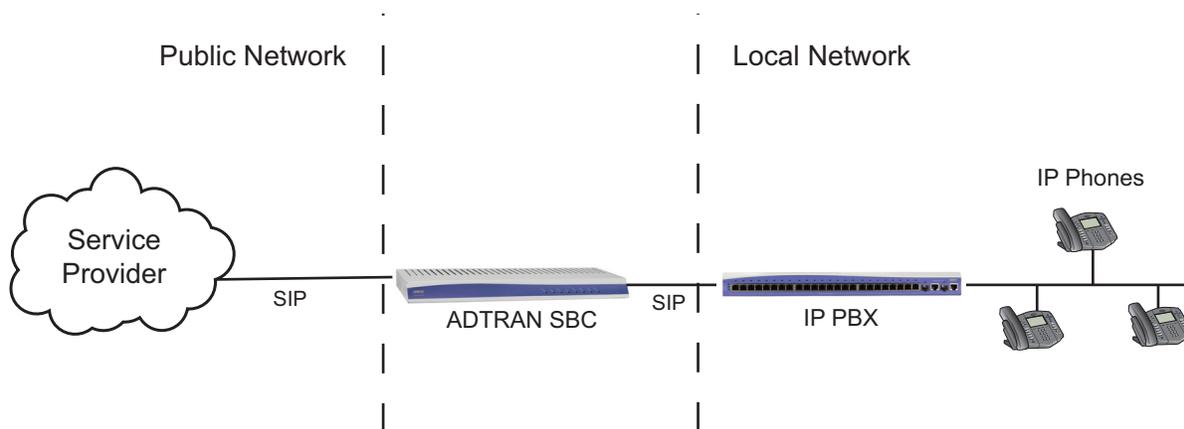


Figure 1. ADTRAN SBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN SBC and the Cisco Call Manager Express. 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 Cisco Call Manager Express 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 Cisco Call Manager Express. 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 Cisco Call Manager Express supports various phone types, including SIP IP phones. The phones register locally to the Cisco Call Manager Express. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN SBC gateway.

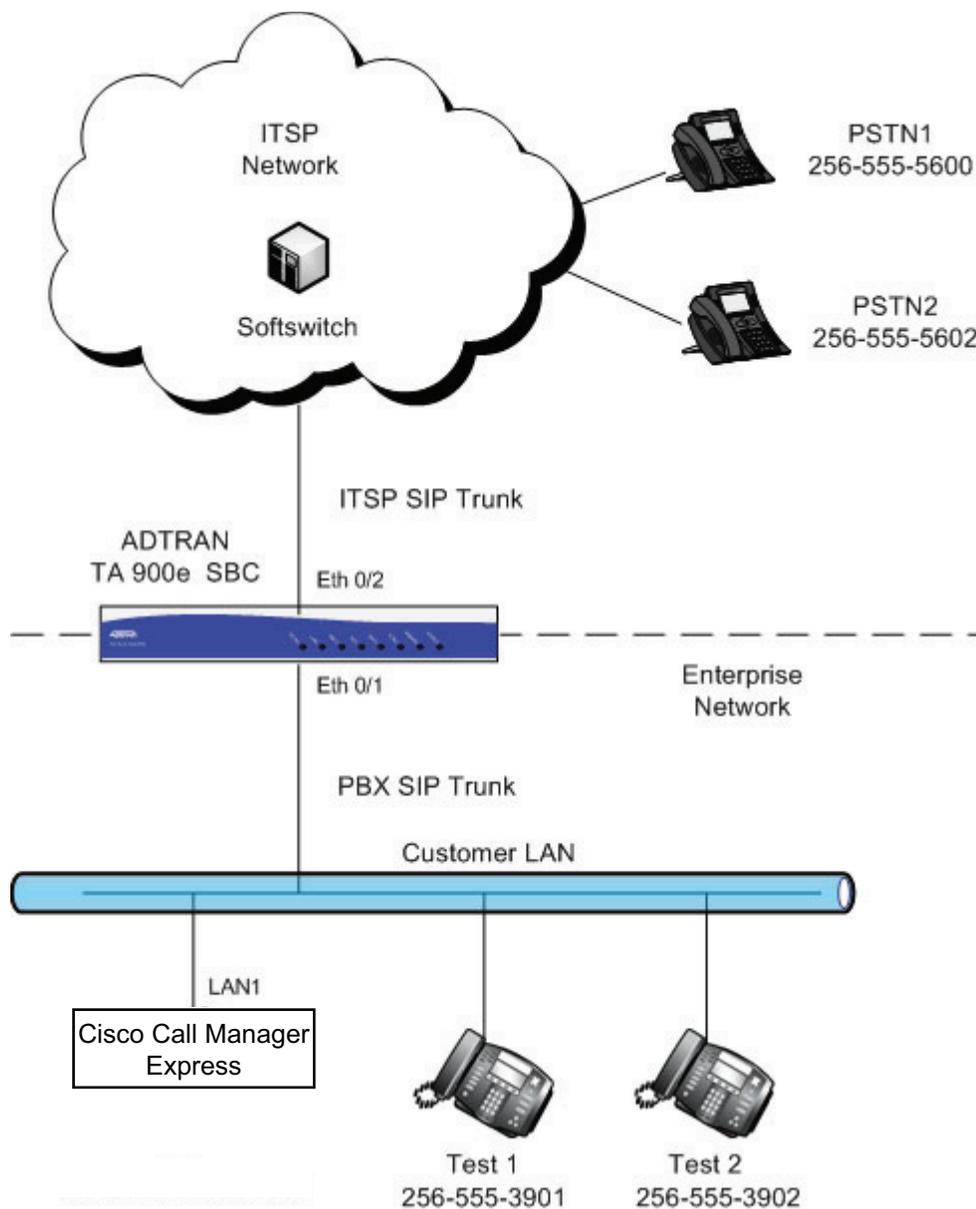


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the Cisco Call Manager Express 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	R10.3.0
Cisco Call Manager Express	15.2

Verification Performed

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

- Cisco Call Manager Express SIP trunk operation with the ADTRAN SBC gateway.
- Cisco Call Manager Express 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 17* for more information about SBC GUI configuration.

To configure the SBC for interoperability with the Cisco Call Manager Express, follow these steps:

- *Step 1: Accessing the SBC CLI on page 5*
- *Step 2: Configuring the Basic Network Settings on page 6*
- *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 Cisco Call Manager Express SIP Trunk on page 7*
- *Step 6: Configuring a Trunk Group for the Service Provider on page 7*
- *Step 7: Configuring a Trunk Group for the PBX on page 8*
- *Step 8: Enabling Media Anchoring on page 9*
- *Step 9: Configuring the Double reINVITE Preference on page 9*

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.



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.



*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 Cisco Call Manager Express. 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 10.70.82.2 255.255.255.0
(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 Cisco Call Manager Express SIP Trunk

The second of two voice trunks that must be configured is the SIP trunk to the Cisco Call Manager Express 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 Cisco Call Manager Express LAN1 IP address. In addition, the Cisco Call Manager Express 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 10.70.82.3  
(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 Cisco Call Manager Express 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** 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
```



For more information about configuring additional media anchoring settings, refer to the configuration guide [Configuring Media Anchoring in AOS](http://supportforums.adtran.com), 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
```

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 Cisco Call Manager Express. This configuration was used to validate the interoperability between the ADTRAN SBC and the Cisco Call Manager Express. Only the commands relevant to the interoperability configuration are shown.



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 10.70.82.2 255.255.255.0  
  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 service provider  
  sip-server primary 198.51.100.2  
!  
!  
voice trunk T02 type sip  
  description PBX  
  sip-server primary 10.70.82.3  
  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  
!  
!  
no ip sip prefer double-reinvite  
!  
!
```

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

Configuring the Cisco Call Manager Express

The Cisco Call Manager Express system supports many features, and is configured using the CLI. Refer to the Cisco documentation for detailed instructions about accessing the CLI. The following is an example of the Cisco Call Manager Express CLI configuration used for this interoperability test. 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.

The following is the CLI configuration of the Cisco Call Manager Express used in the interoperability test with the ADTRAN SBC:

```
CUBE#sh run
Building configuration...

Current configuration : 23630 bytes
!
! Last configuration change at 10:58:03 CST Tue Aug 7 2012 by cisco
! NVRAM config last updated at 11:16:36 CST Tue Aug 7 2012 by cisco
! NVRAM config last updated at 11:16:36 CST Tue Aug 7 2012 by cisco
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.152-4.M1.bin
warm-reboot
boot-end-marker
!
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 $1$RB6P$34wxsXEnEEeG
!
no aaa new-model
clock timezone CST -5 0
!
no ip domain lookup
```

```
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
voice-card 0
dsp services dspfarm
!
voice service voip
ip address trusted list
  ipv4 X.X.X.X           // IP or Network address of devices that need to communicate with
                        CUCME
address-hiding
allow-connections sip to sip
no supplementary-service sip moved-temporarily // Voicemail access. TA900e does not forward a
  DIVERSION header in a 300 message
no supplementary-service sip refer
supplementary-service media-renegotiate
signaling forward none
sip
rel1xx disable
header-passing
asserted-id pai         // Used for privacy calls
early-offer forced
midcall-signaling passthru
!
voice class codec 1
codec preference 1 g729br8
codec preference 2 g711ulaw
!
voice class codec 2
codec preference 1 g711ulaw
!
http client cache memory pool 15000
http client cache memory file 1500
http client cache query
http client cache refresh 864000
no http client connection persistent
http client connection timeout 60
http client response timeout 30
!
hw-module pvdm 0/0
!
vtp domain Corp1
vtp mode transparent
vtp version 2
username cisco secret 5 $1$1FQ1$wmWrRzjU8
```

```
!  
redundancy  
!  
interface GigabitEthernet0/0  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
interface GigabitEthernet0/1  
  ip address X.X.X.X 255.255.255.0  
  duplex full  
  speed 100  
!  
interface GigabitEthernet0/2  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface GigabitEthernet0/3  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface FastEthernet0/0/0  
  no ip address  
!  
interface FastEthernet0/0/1  
  no ip address  
!  
interface FastEthernet0/0/2  
  no ip address  
!  
interface FastEthernet0/0/3  
  no ip address  
!  
interface Vlan1  
  no ip address  
!  
ip default-gateway X.X.X.X  
ip forward-protocol nd  
!  
no ip http server  
no ip http secure-server  
!  
ip route 0.0.0.0 0.0.0.0 X.X.X.X
```

```
!  
nls resp-timeout 1  
cpd cr-id 1  
!  
tftp-server flash:P0030801SR02.bin  
tftp-server flash:P0030801SR02.loads  
tftp-server flash:P0030801SR02.sb2  
tftp-server flash:P0030801SR02.sbn  
tftp-server RingList.xml  
tftp-server DistinctiveRingList.xml  
tftp-server its/vrf1/XMLDefault.cnf.xml  
!  
control-plane  
!  
voice-port 0/1/0  
  caller-id enable  
!  
voice-port 0/1/1  
!  
mgcp profile default  
!  
sccp local GigabitEthernet0/1  
sccp ccm 10.70.82.10 identifier 1 priority 1 version 7.0  
sccp  
!  
sccp ccm group 2  
  associate ccm 1 priority 1  
  associate profile 3 register confbridge  
!  
dspfarm profile 3 conference  
  codec g722-64  
  codec g711ulaw  
  codec g711alaw  
  codec g729ar8  
  codec g729abr8  
  codec g729r8  
  codec g729br8  
  maximum sessions 3  
  associate application SCCP  
!  
dial-peer voice 5 voip  
  huntstop  
  max-conn 15  
  destination-pattern .T  
  session protocol sipv2  
  session target ipv4:10.70.82.2  
  incoming called-number .%
```

```
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-nte
!
!
gateway
timer receive-rtp 1200
!
sip-ua
!
gatekeeper
shutdown
!
telephony-service
sdspfarm conference mute-on 111 mute-off 222
sdspfarm units 3
sdspfarm tag 2 confbridge
no auto-reg-ephone
max-ephones 100
max-dn 100
ip source-address 10.70.82.10 port 2000
cnf-file location flash:
cnf-file perphone
load 7960-7940 P0030801SR02
voicemail 9728522606
max-conferences 8 gain -6
call-park system application
moh "flash:/music-on-hold.au"
transfer-system full-consult // full-consult works better than full-blind
transfer-pattern .....
create cnf-files version-stamp Jan 01 2002 00:00:00
!
ephone-template 1
softkeys hold Resume Newcall Select Join
softkeys idle Redial Newcall ConfList RmLstC Cfwdall Join Pickup Login HLog Dnd Gpickup
softkeys seized Endcall Redial Cfwdall Meetme Pickup Callback
softkeys alerting Endcall Callback
softkeys connected Hold Endcall Confrn Trnsfer Select Join ConfList RmLstC Park Flash
!
ephone-dn 1
number 9728522660
label 9728522660
!
ephone-dn 2 dual-line
number 2500
label 2500
!
```

```
ephone-dn 3 dual-line
number 8889
name Conference
conference ad-hoc
!
ephone-dn 5
number 2660
park-slot reserved-for 9728522660
!
ephone-dn 10 dual-line
number 9728522606
no huntstop
!
ephone-dn 11 dual-line
number 9728522606
no huntstop
!
ephone-dn 15
number 1001
mwi on
!
ephone-dn 16
number 1000
mwi off
!
ephone 1
device-security-mode none
mac-address 001A.2F8D.F603
ephone-template 1
button 1:1
!
ephone 2
device-security-mode none
mac-address 0014.A998.C3FC
button 1:2
!
ephone 10
device-security-mode none
vm-device-id CiscoUM1-VI1
button 1:10
!
ephone 11
device-security-mode none
vm-device-id CiscoUM1-VI2
button 1:11
!
line con 0
```

```

login local
line aux 0
line vty 0 4
  exec-timeout 0 0
login local
transport input telnet ssh
!
exception crashinfo dump command sh controllers g0/0
exception crashinfo dump command sh controller g0/0
scheduler allocate 20000 1000
ntp server 10.10.10.5
!
end

```

The Cisco Call Manager Express 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	<i>AOS Command Reference Guide</i>
ANI and DNIS Substitution	<i>Enhanced ANI/DNIS Substitution in AOS</i>
SBC Product Overview	<i>Session Border Controllers in AOS</i>
Media Anchoring	<i>Configuring Media Anchoring in AOS</i>
Configuring SIP Trunks on a Total Access 900 Series Using the GUI	<i>Total Access 900/900e SIP Trunk Quick Configuration Guide</i>