



Interoperability Guide

ADTRAN SBC and Cisco UC520 SIP Trunk Interoperability

This guide describes an example configuration used in testing the interoperability of an ADTRAN session border controller (SBC) and the Cisco UC520 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 UC520 PBX products.

For additional information 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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- *Additional Resources on page 13*

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 interoperate 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 13*.

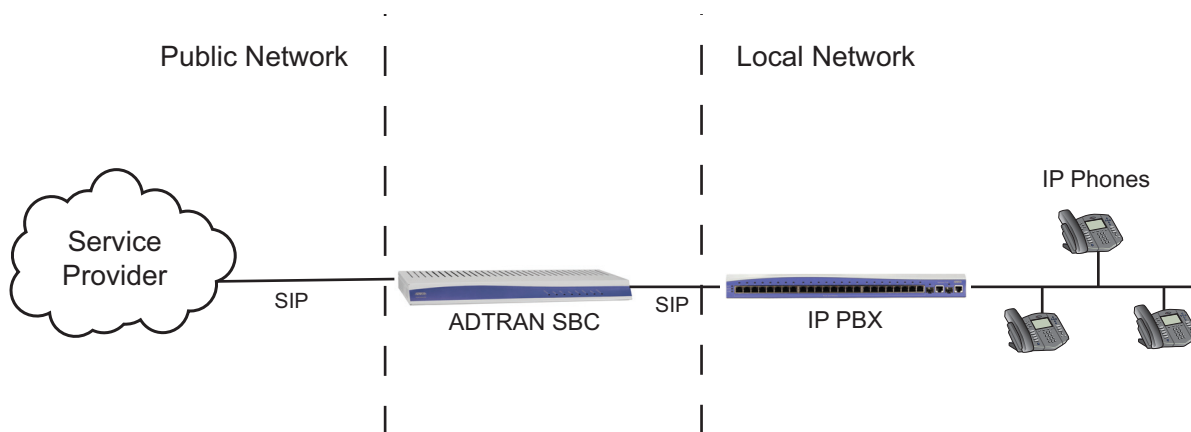


Figure 1. ADTRAN SBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN SBC and the UC520 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 UC520 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 UC520 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 UC520 PBX supports various phone types, including SIP IP phones. The phones register locally to the UC520 PBX. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN SBC gateway.

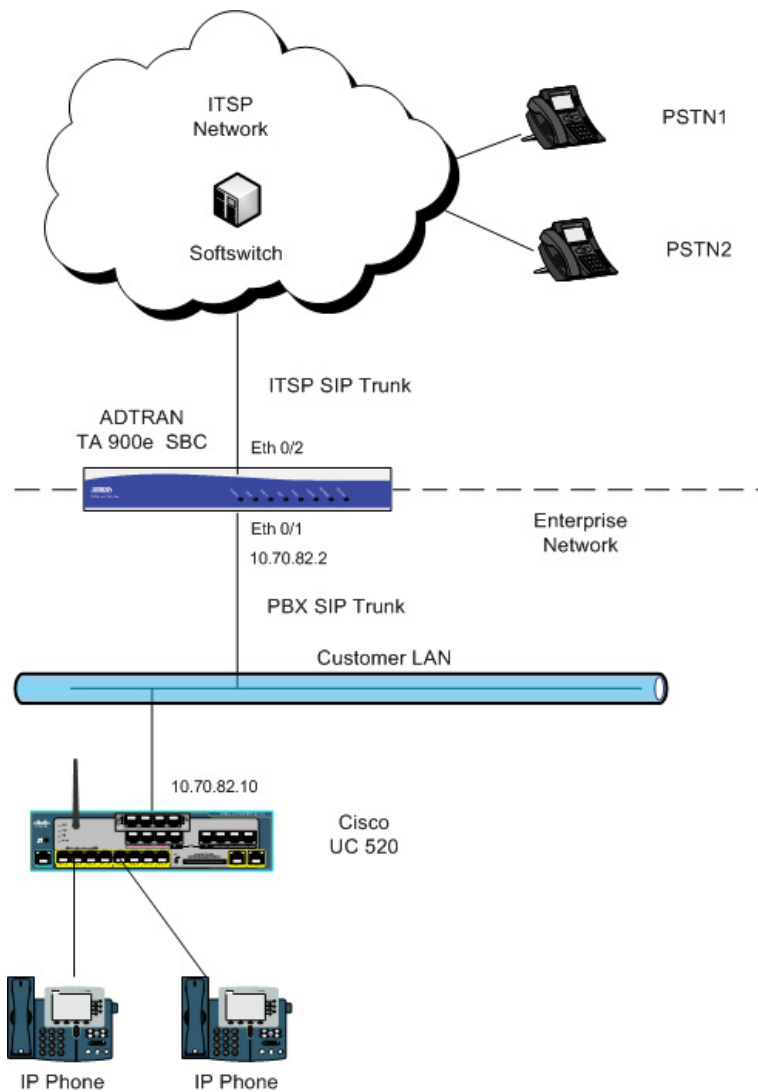


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the Cisco Call Manager IP 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 UC520	15.1

Verification Performed

Interoperability verification testing focused on SIP trunk operations between the ADTRAN SBC gateway and the UC520 PBX. Other PBX features not specific to basic SIP trunking were not included in this verification. Verification testing included the following features:

- UC520 PBX SIP trunk operation with the ADTRAN SBC gateway.
- 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 13* for more information about SBC GUI configuration.

To configure the SBC for interoperability with the UC520 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 UC520 PBX SIP Trunk on page 6*
- *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 8](#)
- [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:
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 UC520 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 10.70.82.2 255.255.255.0  
(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). Check with 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 UC520 PBX SIP Trunk

The second of two voice trunks that must be configured is the SIP trunk to the UC520 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 UC520 PBX IP address. In addition, the UC520 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 10.70.82.10
(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 UC520 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** 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](#), 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 UC520 PBX. This configuration was used to validate the interoperability between the ADTRAN SBC and the UC520 PBX. 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 10.70.82.10  
  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 UC520 PBX

The Cisco UC520 system supports many features, and is configured using the Cisco Configuration Assistant (CCA) running on an administrative workstation or the command line interface (CLI). Refer to the Cisco documentation for detailed instructions about configuring the UC520. The following is an example of the CLI configuration of the Cisco UC520 used for this interoperability test. This is not a complete configuration. Only the commands necessary for configuring the SIP trunk to the ADTRAN SBC are shown.

! Configure Ethernet WAN interface to TA908e

```
interface FastEthernet0/0  
  description $FW_OUTSIDE$
```

```
ip address 10.70.82.10 255.255.255.0
ip access-group 104 in
ip nat outside
ip inspect SDM_LOW out
ip virtual-reassembly in
load-interval 30
duplex auto
speed auto
!
! Configure routing and access lists for voice traffic
!
ip nat inside source list 1 interface FastEthernet0/0 overload
ip route 0.0.0.0 0.0.0.0 10.70.82.2
!
!
access-list 104 permit udp host 10.70.82.2 eq 5060 any
access-list 104 permit udp host 10.70.82.2 any eq 5060
!
!
voice service voip
ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
sip
  registrar server expires max 3600 min 3600
  asserted-id pai
  no update-callerid
  sip-profiles 1000
!
voice class codec 1
  codec preference 1 g711ulaw
!
voice class sip-profiles 1000
  request ANY sdp-header Connection-Info remove
  response ANY sdp-header Connection-Info remove
!
!
dial-peer cor list call-local
  member local
!
!
dial-peer voice 1000 voip
  permission term
  description ** Incoming call from SIP trunk (Generic SIP Trunk Provider) **
  session protocol sipv2
```

```
session target sip-server
incoming called-number .%
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
dtmf-relay rtp-nte
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
!
dial-peer voice 1033 voip
corlist outgoing call-local
description **CCA*North American-10-Digit*10-Digit Local**
preference 1
destination-pattern 9[2-9]..[2-9].....
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
dtmf-relay rtp-nte
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
!
!
sip-ua
no remote-party-id
retry invite 2
retry register 10
timers connect 100
sip-server ipv4:10.70.82.2:5060
host-registrar
!
```

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>