6AOSSG0027-42A
June 2015

ADTRAN

Interoperability Guide

Integrating an ADTRAN eSBC with an NEC SV8100

This interoperability guide provides instructions for integrating an ADTRAN enterprise session border controller (eSBC) and the NEC SV8100 private branch exchange (PBX) using a Session Initiation Protocol (SIP) trunk to provide a connection to the service provider network. This guide includes the description of the network application, verification summary, and example individual device configurations for the ADTRAN eSBC and the NEC SV8100 PBX products.

This guide consists of the following sections:

Overview

Service providers are increasingly using SIP trunks to provide Voice over IP (VoIP) services to customers. ADTRAN eSBC products provide features that normalize the SIP signaling and media between the customer's PBX and the service provider's SBC and softswitch server. In this application, the ADTRAN eSBC gateway operates as a SIP back-to-back user agent (B2BUA). One Ethernet interface on the ADTRAN eSBC provides the wide area network (WAN) connection to the service provider network and terminates the service provider SIP trunk. A second Ethernet interface connects the customer's local area network (LAN) and provides a SIP trunk connection to the IP PBX for VoIP applications. *Figure 1* illustrates the use of the ADTRAN eSBC in a typical network deployment.

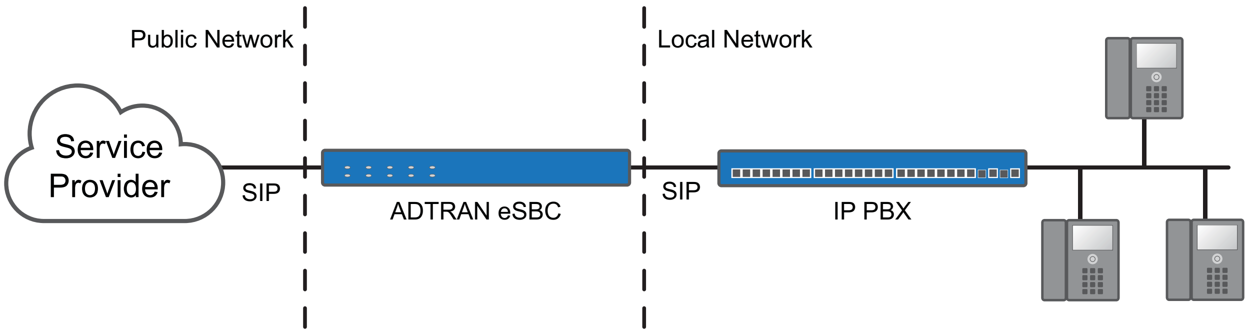


Figure 1. ADTRAN eSBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN eSBC and the NEC SV8100. The configuration is a typical SIP trunking application, in which the ADTRAN eSBC gateway Ethernet interface provides the Ethernet WAN connection to the service provider network. It should be noted that the WAN connection is not limited to Ethernet. A second Ethernet interface connects to the customer LAN. The NEC SV8100 connects to the customer LAN. Two SIP trunks are configured on the ADTRAN eSBC gateway: one to the service provider SIP network and the second to the NEC SV8100. The ADTRAN eSBC gateway operates as a SIP B2BUA, and outbound and inbound calls to the public switched telephone network (PSTN) are routed through the ADTRAN eSBC.

The NEC SV8100 supports various phone types, including digital, H.323, and SIP IP phones. The phones register locally to the NEC SV8100. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN eSBC gateway.

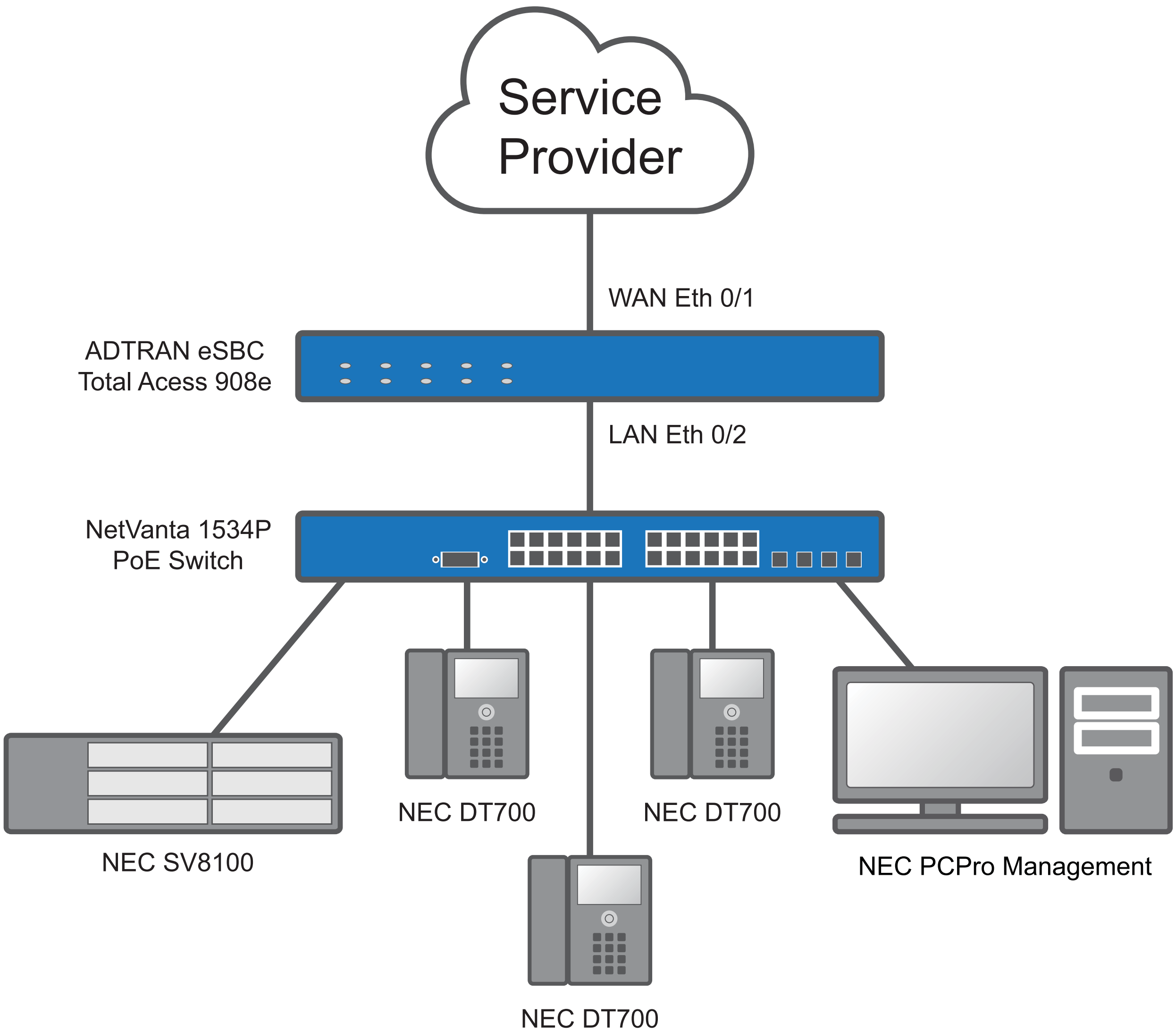


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the NEC SV8100 PBX is available on ADTRAN products with the eSBC 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 with eSBC	R10.9.6
NEC SV8100 PBX	06.00
NEC PCPro Management Software	1.03.0n.pipk
NEC DT700 IP Phone, ITL-6DE-1	5.0.3.0
NEC DT700 IP Phone, ITL-12D-1	5.0.2.0
NEC DT700 IP Phone, ITL-320C-2	5.0.2.0

Configuring the ADTRAN eSBC

The following sections describe the key configuration settings required for this solution. These settings are implemented using the ADTRAN Operating System (AOS) command line interface (CLI).

To configure the ADTRAN eSBC for interoperability with the NEC SV8100 PBX, follow these steps:

- *Step 1: Access the eSBC CLI on page 4*
- *Step 2: Configure the Basic Network Settings on page 5*
- *Step 3: Configure Global Voice Modes for Local Handling on page 6*
- *Step 4: Enable Media Anchoring on page 6*
- *Step 5: Configure the Service Provider SIP Trunk on page 6*
- *Step 6: Configure the PBX SIP Trunk on page 6*
- *Step 7: Configure a Trunk Group for the Service Provider on page 7*
- *Step 8: Configure a Trunk Group for the NEC SV8100 on page 8*
- *Step 9: Configure the Double reINVITE Preference on page 8*
- *Step 10: Configure SIP Privacy (Optional) on page 9*

Step 1: Access the eSBC CLI

The AOS unit can be managed using the console port, Hypertext Transfer Protocol (HTTP), HTTP Secure (HTTPS), Telnet, and Secure Shell (SSH). Most of the initial configuration is performed through the console port or Telnet session. Accessing the AOS unit is described in this step.

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****. The default enable password is* ***password****. If your product no longer has the default user name and passwords, contact your system administrator for the appropriate user name and passwords.*

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: Configure the Basic Network Settings

Basic network configuration includes setting up two Ethernet interfaces, one for the Ethernet LAN interface to the NEC SV8100, and the second for the Ethernet WAN interface to the service provider. 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 IPv4 address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic.

> NOTE *When configuring the basic network settings, use the IP address information supplied by the service provider.*

Enter the commands from the Ethernet interface configuration mode as follows:

For the WAN:
(config)#**interface ethernet 0/1**
(config-eth 0/1)#**description WAN**
(config-eth 0/1)#**ip address 203.0.113.2 255.255.255.252**
(config-eth 0/1)#**media-gateway ip primary**
(config-eth 0/1)#**no shutdown**

For the LAN:
(config)#**interface ethernet 0/2**
(config-eth 0/2)#**description LAN**
(config-eth 0/2)#**ip address 192.168.1.1 255.255.255.0**
(config-eth 0/2)#**media-gateway ip primary**
(config-eth 0/2)#**no shutdown**

Step 3: Configure Global Voice Modes for Local Handling

Configure the ADTRAN eSBC 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: Enable Media Anchoring

Media anchoring is an eSBC feature that forces RTP traffic through the ADTRAN eSBC 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. The RTP symmetric filter is enabled using the **ip rtp symmetric-filter** command.

Enter the commands as follows:

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

Step 5: Configure 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 eSBC. Check with your service provider for any specific requirements beyond those listed in this document. Your service provider will provide you with the IP addresses or fully qualified domain name (FQDN) and possibly the port numbers for their SIP server. They may also provide a backup or secondary SIP server.

The **voice trunk** <*Txx*> **type sip** command is used to define a new SIP trunk and activate the Voice Trunk Configuration mode for the trunk. From the Voice Trunk Configuration mode, you can provide a descriptive name for the trunk and define the IP address (or host name) of the SIP server. The **description** <*text*> command is used to label the trunk. The **sip-server primary** <*ip address* | *hostname*> command is used to define the host name or IP address of the primary server to which the trunk sends SIP messages.

Enter the commands as follows:

(config)#**voice trunk T01 type sip**
(config-T01)#**description PROVIDER**
(config-T01)#**sip-server primary sip.example.com**

Step 6: Configure the PBX SIP Trunk

The second voice trunk that must be configured is the SIP trunk to the NEC SV8100 from the ADTRAN eSBC. The **voice trunk** <*Txx*> **type sip** command is used 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 IP address (or host name) of the PBX. The **description** <*text*> command is used to label the trunk. The **sip-server-primary** <*ip address* | *hostname*> command is used to set the server address to the NEC SV8100 LAN IP address.

In addition, the NEC SV8100 must control call transfers. This is accomplished using the **transfer-mode-network** command in the trunk's configuration. The **grammar from host local** command is used to specify that the IP address of the interface is used in the SIP From header for outbound messages.

Enter the commands as follows:

(config)#**voice trunk T11 type sip**
(config-T11)#**description PBX**
(config-T11)#**sip-server primary 192.168.1.2**
(config-T11)#**transfer-mode network**
(config-T11)#**grammar from host local**

Step 7: Configure 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 that particular trunk. The *<Txx>* parameter specifies the trunk identity where *xx* is the trunk ID number.

Use the **accept** *<pattern>* 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 *<pattern>* 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 NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 1-NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 011-$ cost 0**
(config-PROVIDER)#**accept 411 cost 0**
(config-PROVIDER)#**accept 611 cost 0**
(config-PROVIDER)#**accept 911 cost 0**

Step 8: Configure a Trunk Group for the NEC SV8100

After configuring a trunk group for the service provider, create a trunk group for the NEC SV8100 trunk account using the **voice grouped-trunk** <*name*> command. Add an existing trunk to the trunk group using the **trunk** <*Txx*> command. The outbound allowed calls are defined using the **accept** <*pattern*> command and are assigned a cost using the **cost** parameter, as described in *Step 7: Configure a Trunk Group for the Service Provider on page 7*.

Enter the commands as follows:

(config)#**voice grouped-trunk PBX**
(config-PBX)#**trunk T11**
(config-PBX)#**accept $ cost 0**

Step 9: Configure the Double reINVITE Preference

The **sip prefer double-reinvite** command is used from the Global Configuration mode to specify that a double reINVITE is preferred globally for all calls in the system. 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** form of the **sip prefer double-reinvite** command from the Global Configuration mode.

Enter the command as follows:

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

Step 10: Configure SIP Privacy (Optional)

The ADTRAN eSBC supports SIP user privacy using the Privacy and P-Asserted-Identity (PAI) SIP headers. The **sip privacy** command is used from the Global Configuration mode to enable SIP privacy support. The **trust-domain** command is used from the Voice SIP Trunk Configuration mode to connect the trunk to a trusted domain and enable PAI support, adding security measures for user's identity and privacy.

Enter the commands as follows:

(config)#**sip privacy**
(config)#**voice trunk T01 type sip**
(config-T01)#**trust-domain**
(config)#**voice trunk T11 type sip**
(config-T11)#**trust-domain**

ADTRAN eSBC Sample Configuration

The following example configuration demonstrates a typical installation of an ADTRAN eSBC configured as the SIP trunking gateway between a NEC SV8100 and a service provider.

> 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. This configuration should not be copied without first making the necessary adjustments to ensure it will function properly in your network.*

> NOTE *For details on each configuration option, refer to the AOS Command Reference Guide and other IP business gateway/eSBC configuration guides on the ADTRAN support forums at https://supportforums.adtran.com.*

```
!
interface eth 0/1
  description WAN
  ip address 203.0.113.2 255.255.255.252
  media-gateway ip primary
  no shutdown
!
interface eth 0/2
  description LAN
  ip address 192.168.1.1 255.255.255.0
  media-gateway ip primary
  no shutdown
!
ip route 0.0.0.0 0.0.0.0 203.0.113.1
!
sip
sip udp 5060
!
```

```
voice feature-mode network
voice transfer-mode local
voice forward-mode local
!
voice trunk T01 type sip
  description PROVIDER
  sip-server primary sip.example.com
  trust-domain
!
voice trunk T11 type sip
  description PBX
  sip-server primary 192.168.1.2
  trust-domain
  grammar from host local
  transfer-mode network
!
voice grouped-trunk PROVIDER
  trunk T01
  accept NXX-NXX-XXXX cost 0
  accept 1-NXX-NXX-XXXX cost 0
  accept 011-$ cost 0
  accept 411 cost 0
  accept 611 cost 0
  accept 911 cost 0
!
voice grouped-trunk PBX
  trunk T11
  accept $ cost 0
!
sip privacy
!
no sip prefer double-reinvite
!
ip rtp symmetric-filter
ip rtp media-anchoring
!
```

Configuring the NEC SV8100

The NEC SV8100 PBX is configured using NEC's PCPro Manager Software that runs on a Microsoft Windows PC. The SV8100 also supports web access. However, ADTRAN recommends using the client software for better performance. Refer to the NEC documentation for detailed instructions on configuring additional features and capabilities. The following sections describe the minimum configuration required for SIP trunking interoperability with the ADTRAN eSBC.

To configure the NEC SV8100 PBX, follow these steps:

- *Step 1: Installing NEC SV8100 PBX and NEC SV8100 PCPro on page 11*
- *Step 2: Logging in to NEC SV8100 PCPro on page 11*
- *Step 3: Establishing Communications with the NEC SV8100 Chassis on page 12*
- *Step 4: Configuring a SIP Trunk Group on page 14*
- *Step 5: Configuring the SV8100 System Data Settings on page 14*

Step 1: Installing NEC SV8100 PBX and NEC SV8100 PCPro

Install the SV8100 PBX as shown in the Network Topology diagram in *Figure 2 on page 3*. Refer to the *Univerge SV8100 System Hardware Manual* available from NEC for installation details. The LAN1 Ethernet interface connects to the LAN switch (NetVanta 1534P).

NEC SV8100 PBX configuration is performed using the NEC SV8100 PCPro software installed on a Windows PC. Refer to the *Univerge SV8100 PC Programming Manual* available from NEC for installation details.

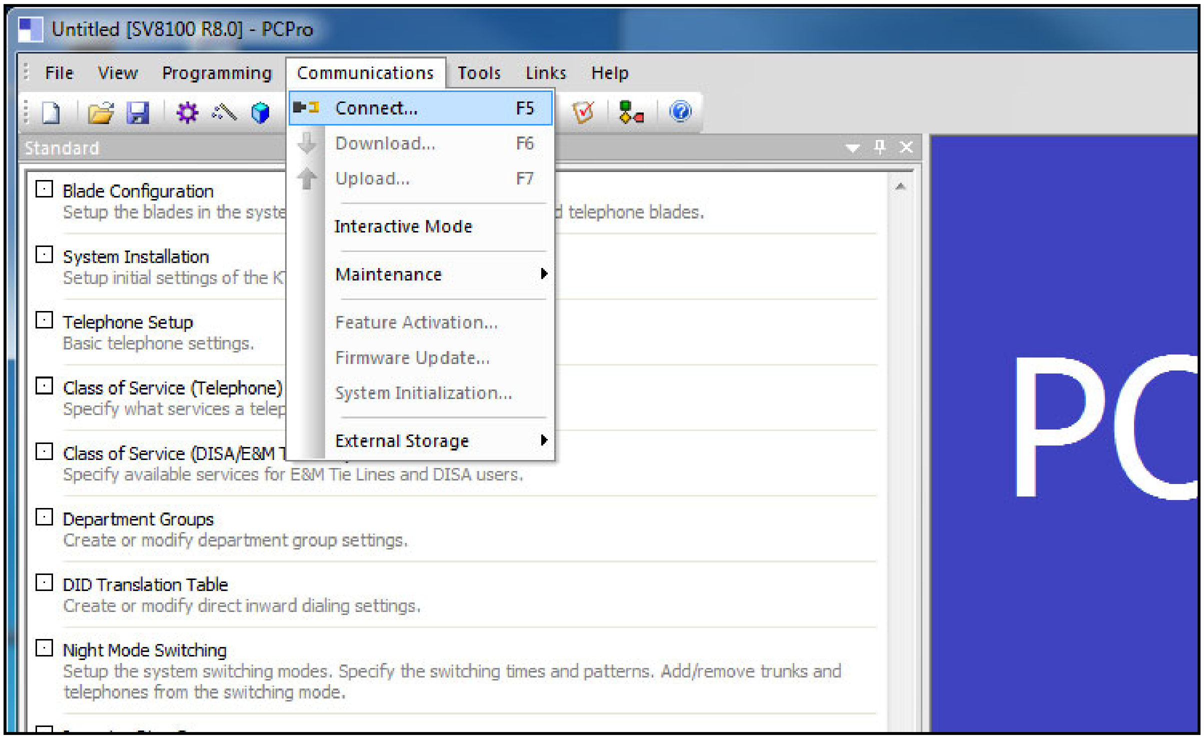
Step 2: Logging in to NEC SV8100 PCPro

Log in to PCPro with the **Installer Mode** default PCPro account using the user name **tech** and password **12345678**. Then, select **OK**. The **NEC SV8100 PCPro** main menu appears.

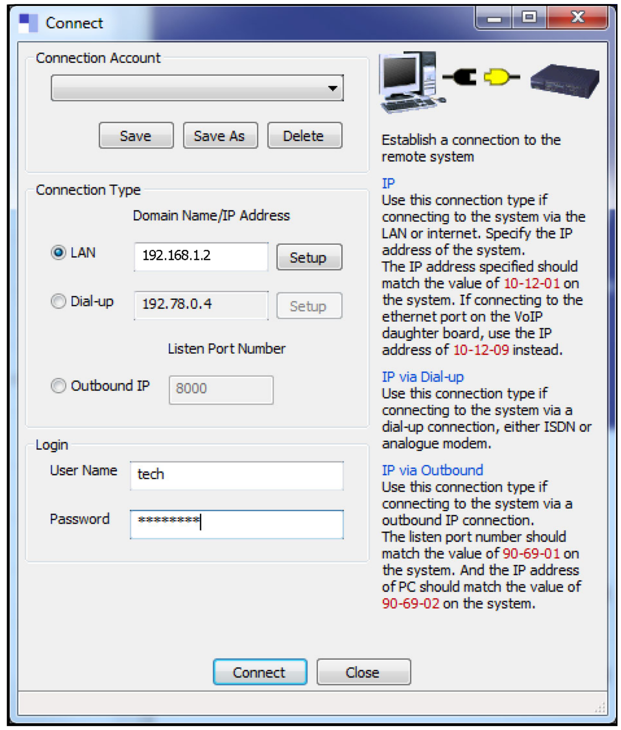
Step 3: Establishing Communications with the NEC SV8100 Chassis

To establish communication with the NEC SV8100 chassis, follow these steps:

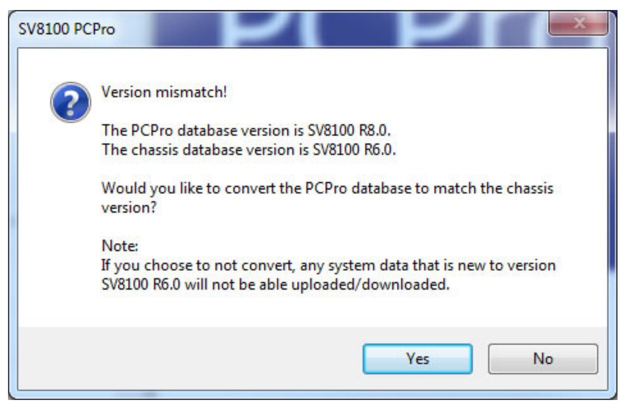
1. In the tool bar of the NEC SV8100 PCPro software, select **Communications** > **Connect**. The **Connect** menu will appear.



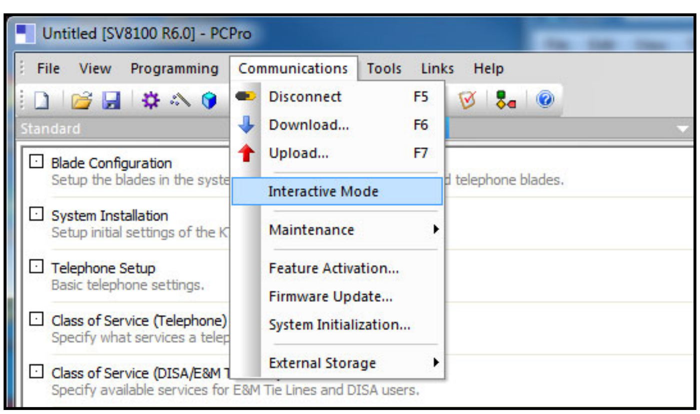
2. In the **Connect** menu, perform the following:
   a. Select the **LAN** radio button.
   b. Enter the LAN IP address of the NEC SV8100 (**192.168.1.2**) in the **LAN** field.
   c. Enter the **Installer Mode** user name (**tech**) in the **User Name** field.
   d. Enter the **Installer Mode** password (**12345678**) in the **Password** field.
   e. Select **Connect**. Communication should be established with the SV8100.



3. If the PCPro database version is different from the SV8100 database version, a version mismatch message will appear. Select **Yes** on the message to update the SV8100 database. The PCPro main menu will update with information from the SV8100 chassis.



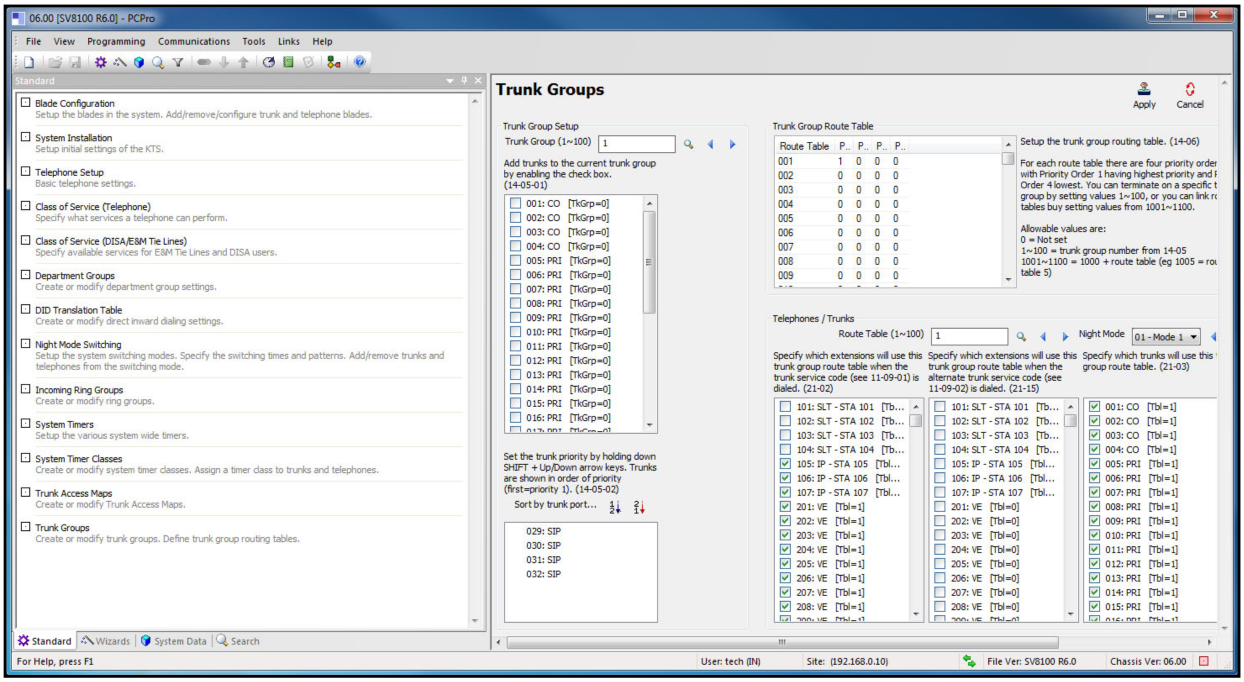
4. From the tool bar in the PCPro main menu, select **Communications > Interactive Mode**. Communication should now be established with the SV8100 chassis for configuration.



Step 4: Configuring a SIP Trunk Group

Once you have established communication with the NEC SV8100, create a SIP trunk group with four unassigned trunks. The trunks used for the trunk group should not be the trunks already in use for the central office (CO) or the Primary Rate Interface (PRI) (i.e., trunks **1** through **28**).

Trunk groups can be configured in either the **Standard** view or **System Data** view. To access the trunk group configuration from the **Standard** view, select the **Standard** tab in the bottom left corner of the PCPro main menu, then select **Trunk Groups** from the **Standard** pane. To access the trunk group configuration from the **System Data** view, select the **System Data** tab in the bottom left corner of the PCPro main menu, then select **14-XX: Trunk Setup** > **14-05: Trunk Groups** from the **System Data** pane. Refer to the *Univerge SV8100 PC Programming Manual* available from NEC for configuration details.



Step 5: Configuring the SV8100 System Data Settings

Use the following sections to configure various SV8100 settings using the **System Data** view.

Allocating the SIP Trunk Ports (10-03: IPL Configuration)

To allocate physical trunk ports to logical trunk ports and assign the trunks a trunk type, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup** > **10-03: IPL Configuration** from the **System Data** pane. The **10-03: IPL Configuration** menu will appear.
2. In the **10-03: IPL Configuration** menu, allocate physical ports for each SIP trunk configured on the SV8100 (i.e., **29, 30, 31, 32**):

a. For **Physical Port 001**, enter **29** in the **Trunk Logical Port** field, and select **SIP** using the provided **Trunk Type** drop-down menu.

b. For **Physical Port 002**, enter **30** in the **Trunk Logical Port** field, and select **SIP** using the provided **Trunk Type** drop-down menu.

c. For **Physical Port 003**, enter **31** in the **Trunk Logical Port** field, and select **SIP** using the provided **Trunk Type** drop-down menu.

d. For **Physical Port 004**, enter **32** in the **Trunk Logical Port** field, and select **SIP** using the provided **Trunk Type** drop-down menu.

3. Select **Apply** to apply the settings.

Configuring the SIP Trunk Service (10-12: CD-CP00 Network Setup)

To configure the SIP trunk service, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-12: CD-CP00 Network Setup** from the **System Data** pane. The **10-12: CD-CP00 Network Setup** menu will appear.
2. In the **10-12: CD-CP00 Network Setup** menu, perform the following:
   a. Enter **0.0.0.0** in the **01 - IP Address** field.
   b. Enter the LAN IP address of the ADTRAN eSBC (i.e., **192.168.1.1**) in the **03 - Default Gateway** field.
   c. Enter the LAN IP address of the NEC SV8100 (i.e., **192.168.1.2**) in the **09 - IPL IP Address** field.
   d. Select the subnet mask (i.e., **255.255.255.0**) using the **10 - IPL Subnet Mask** drop-down menu
3. Select **Apply** to apply the settings.

Enabling the DHCP Server mode (10-13: DHCP Server Setup)

The NEC SV8100 provides an optional Dynamic Host Configuration Protocol (DHCP) server. NEC IP phones can be setup to use either DHCP or static IP addressing. To enable DHCP server mode, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-13: DHCP Server Setup** from the **System Data** pane. The **10-13: DHCP Server Setup** menu will appear.
2. In the **10-13: DHCP Server Setup** menu, check the **01 - DHCP Server Mode** check box.
3. Select **Apply** to apply the settings.

Specifying the DHCP IP Address Range (10-14: Managed Network Setup)

To specify the range of IP addresses which the SV8100 DHCP server leases to a client, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-14: Managed Network Setup** from the **System Data** pane. The **10-14: Managed Network Setup** menu will appear.

2. In the **10-14: Managed Network Setup** menu, perform the following:
   a. In the **01 - Min** field, enter the minimum IP address for the range of IP addresses leased by the DHCP server.
   b. In the **02- Max** field, enter the maximum IP address for the range of IP addresses leased by the DHCP server.
3. Select **Apply** to apply the settings.

Configuring DHCP Option Information (10-16: Option Information Setup)

To specify the option given from the DHCP server to each client, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-16: Option Information Setup** from the **System Data** pane. The **10-16: Option Information Setup** menu will appear.
2. In the **10-16: Option Information Setup** menu, perform the following:
   a. In the **16 - SIP Server IP Address** field, enter the IP address of the SV8100 VoIP Interface (i.e., **192.168.1.2**)
   b. In the **27 - SIP Server Receiver Port** field, enter **5080**.
3. Select **Apply** to apply the settings.

Configuring the VoIP DSP Operating Mode (10-19: IPL DSP Resource Selection)

To allocate digital signal processor (DSP) resources for each SIP trunk to ensure their availability, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-19: IPL DSP Resource Selection** from the **System Data** pane. The **10-19: IPL DSP Resource Selection** menu will appear.
2. In the **10-19: IPL DSP Resource Selection** menu, allocate DSP resources for each SIP trunk configured on the SV8100 (i.e., **29, 30, 31, 32**). For **DSP Resource 001** through **004**, use the provided drop-down menu to select **Used for IP trunks** or **Commonly used for both IP extensions and trunks**.
3. Select **Apply** to apply the settings.

Configuring the IP System Interconnection (10-23: IP System Interconnection Setup)

To configure the SIP server address and trunk access code used for the SV8100 PBX, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-23: IP System Interconnection Setup** from the **System Data** pane. The **10-23: IP System Interconnection Setup** menu will appear.
2. In the **10-23: IP System Interconnection Setup** menu, perform the following perform the following for **Sys No. 0001**:
   a. Check the **System Interconnection** check box.
   b. Enter the LAN IP Address of the ADTRAN eSBC (i.e., **192.168.1.1**) in the **IP Address** field.
   c. Enter the desired trunk access code (e.g., **9**) in the **Dial Number** field.
3. Select **Apply** to apply the settings.

Configuring the SIP System Information (10-28: SIP System Information Setup)

To configure the SIP system information, such as domain name, host name, and transport protocol, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-28: SIP System Information Setup** from the **System Data** pane. The **10-28: SIP System Information Setup** menu will appear.
2. In the **10-28: SIP System Information Setup** menu, perform the following:
   a. Enter the LAN IP address of the ADTRAN eSBC (i.e., **192.168.1.1**) in the **01 - Domain Name** field.
   b. Enter the LAN IP address of the ADTRAN eSBC (i.e., **192.168.1.1**) in the **02 - Host Name** field.
   c. Select **UDP** using the **03- Transport Protocol** drop-down menu.
   d. Enter the main business direct inward dialing (DID) number in the **04 - User ID** field.
   e. Select **IP Address** using the **05 - Domain Assignment** drop-down menu.
   f. Uncheck the **06 - IP Trunk Port Binding** check box.
3. Select **Apply** to apply the settings.

Configuring the SIP Server Information (10-29: SIP Server Information Setup)

To configure the SIP server information, such as the outbound default proxy and default proxy IP address, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-29: SIP Server Information Setup** from the **System Data** pane. The **10-29: SIP Server Information Setup** menu will appear.
2. In the **10-29: SIP Server Information Setup** menu, perform the following perform the following:
   a. Check the **01 - Outbound Default Proxy** check box.
   b. Enter the LAN IP address of the ADTRAN eSBC (i.e., **192.168.1.1**) in the **03 - Default Proxy IP Address** field.
   c. Enter **5060** in the **04 - Default Proxy Port** field.
   d. Select **Default** using the **14 - SIP Carrier Choice** drop-down menu
3. Select **Apply** to apply the settings.

Configuring IP Trunk Availability (10-40: IP Trunk Availability)

IP trunk availability is configured in the **10-40: IP Trunk Availability** menu. To configure IP trunk availability, follow these steps:

1. Select the **System Data** tab, then select **10-XX: Trunk Setup > 10-40: IP Trunk Availability** from the **System Data** pane. The **10-40: IP Trunk Availability** menu will appear.
2. In the **10-40: IP Trunk Availability,** menu perform the following:
   a. Check the **01 - IP Trunk Availability** check box.
   b. Select the number of supported SIP trunks (i.e., 4) using the **02 - IP Trunk Port Count** drop-down menu.
3. Select **Apply** to apply the settings.

Configuring the System Number Plan (11-01: System Numbering)

The numbering plan defines the first and second digits dialed and specifies the digits a user must dial to call other extensions and access features. To configure the numbering plan, follow these steps:

1. Select the **System Data** tab, then select **11-XX: System Numbering Plan > 11-01: System Numbering** from the **System Data** pane. The **11-01: System Numbering** menu will appear.
2. In the **11-01: System Numbering** menu, perform the following:
    a. For the **1x** dial digit, enter **3** in the **Dial Digit Length** field, and select **Extension** from the **Type** drop-down menu.
    b. For the **9x** dial digit, enter **1** in the **Dial Digit Length** field, and select **Trunk** from the **Type** drop-down menu.
3. Select **Apply** to apply the settings.

Configuring the Trunk Access Code (11-09: Trunk Access Code)

The trunk access code is the code extension users dial to access automatic route selection. To configure the trunk access code, follow these steps:

1. Select the **System Data** tab, then select **11-XX: System Numbering Plan > 11-09: Trunk Access Code** from the **System Data** pane. The **11-09: Trunk Access Code** menu will appear.
2. In the **11-09: Trunk Access Code** menu, enter the desired trunk access code in the **01 - Trunk Access Code** field.
3. Select **Apply** to apply the settings.

Configuring Night Mode Switching (12-01: Night Mode Switching Setup)

The night mode provides for scheduling different call behaviors for different time periods. Night mode switching should be set for manual operation to keep the call routing fixed. To enable manual night mode switching, follow these steps:

1. Select the **System Data** tab, then select **12-XX: Night Mode Service > 12-01: Night Mode Switching Setup** from the **System Data** pane. The **12-01: Night Mode Switching Setup** menu will appear.
2. In the **12-01: Night Mode Switching Setup** menu, check the **01-Manual Night Mode Switching** check box.
3. Select **Apply** to apply the settings.

Configuring Trunk Night Mode Groups (12-06: Night Mode Group Assignment for Trunks)

The configured SIP trunks (i.e., **29**, **30**, **31**, and **32**) should be assigned to **Night Mode Group 1**. Trunk mode groups are configure in the **12-06: Night Mode Group Assignment for Trunks** menu.

Configuring Trunk Groups and Trunk Priority (14-05: Trunk Groups)

Assign the configured SIP trunks (i.e., **29**, **30**, **31**, and **32**) to a trunk group, and assign the SIP trunks the highest priority. To configure trunk priority and trunk groups, follow these steps:

1. Select the **System Data** tab, then select **14-XX: Trunk Setup > 14-05: Trunk Groups** from the **System Data** pane. The **14-05: Trunk Groups** menu will appear.
2. In the **14-05: Trunk Groups** menu, perform the following:
   a. For each SIP trunk configured on the system, enter the same number in the provided **Trunk Group** field to assign the SIP trunks to the same trunk group.
   b. For each SIP trunk configured on the system, enter **1** in the **Priority** field. Specifying **1** assigns the trunks the highest priority.
3. Select **Apply** to apply the settings.

Configuring SIP Trunk Routing (14-06: Trunk Group Routing)

Trunks are selected based on priority (optionally, Round-Robin). The SIP trunks configured on the system should be given the highest priority. To give the SIP trunks the highest routing priority, the SIP trunk group should be assigned to the first entry of the highest priority in the route table. To configure trunk group routing for the SIP trunk group, follow these steps:

1. Select the **System Data** tab, then select **14-XX: Trunk Setup > 14-06: Trunk Group Routing** from the **System Data** pane. The **14-06: Trunk Group Routing** menu will appear.
2. In the **14-06: Trunk Group Routing** menu, enter the SIP trunk group number configured in *Configuring Trunk Groups and Trunk Priority (14-05: Trunk Groups) on page 19* in the first field of the route table. This is the field in **Route Table 001** with **Priority Order 1**.
3. Select **Apply** to apply the settings.

Configuring Trunk Access (14-07: Trunk Access Map Setup)

By default, all trunks are given full incoming, outgoing, and hold access. To verify that the SIP trunks (i.e., **29**, **30**, **31**, and **32**) are configured for full access, follow these steps:

1. Select the **System Data** tab, then select **14-XX: Trunk Setup > 14-07: Trunk Access Map Setup** from the **System Data** pane. The **14-07: Trunk Access Map Setup** menu will appear.
2. In the **14-07: Trunk Access Map Setup**, verify the SIP trunks are configured with the **OTG/INC/Hold** option.
3. Select **Apply** to apply the settings.

Configuring CAP Keys for Phones (15-07: Function Keys)

Call appearance (CAP) keys are required to answer incoming DID calls on phones. At least two CAP keys should be defined for each phone on the system. A CAP key for each external trunk can be defined. To configure CAP keys, follow these steps:

1. Select the **System Data** tab, then select **15-XX: Extension Setup > 15-07: Function Keys** from the **System Data** pane. The **15-07: Function Keys** menu will appear.
2. In the **15-07: Function Keys** menu, use the **Function** drop-down menu to assign ***08 - CAP key** to two or more **Function Keys**.
3. Select **Apply** to apply the settings.

Enabling Caller ID Display (20-09: Class of Service Options (Incoming Call Service))

To enable caller ID display for all classes of server, follow these steps for every **Class of Service** available (1-15):

1. Select the **System Data** tab, then select **20-XX: System Options > 20-09: Class of Service Options (Incoming Call Service)** from the **System Data** pane. The **20-09: Class of Service Options (Incoming Call Service)** menu will appear.
2. In the **20-09: Class of Service Options (Incoming Call Service)** menu, check the **02 - Caller ID Display** check box.
3. Select **Apply** to apply the settings.

Configuring SIP Calling Party Numbers (21-19: IP Trunk (SIP) Calling Party Number Setup for Extensions)

To specify the SIP calling party number for each extension, follow these steps:

1. Select the **System Data** tab, then select **21-XX: Outgoing Call Setup > 21-19: IP Trunk (SIP) Calling Party Number Setup for Extension** from the **System Data** pane. The **21-19: IP Trunk (SIP) Calling Party Number Setup for Extension** menu will appear.
2. In the **21-19: IP Trunk (SIP) Calling Party Number Setup for Extension** menu, enter the caller ID information in the **Calling Party Number** field for each assigned extension.
3. Select **Apply** to apply the settings.

Specifying the Incoming Trunk Type (22-02: Incoming Call Trunk Setup)

To assign the incoming trunk type for each SIP trunk, follow these steps:

1. Select the **System Data** tab, then select **22-XX: Incoming Call Service > 22-02: Incoming Call Trunk Setup** from the **System Data** pane. The **22-02: Incoming Call Trunk Setup** menu will appear.
2. In the **22-02: Incoming Call Trunk Setup** menu, select **DID** for the **Incoming Type** for all the service modes for each SIP trunk (identified by its **Trunk Logical Port** number).
3. Select **Apply** to apply the settings.

Configuring Basic DID Options (22-09: DID Basic Setup)

To configure the basic settings for DID calls, follow these steps:

1. Select the **System Data** tab, then select **22-XX: Incoming Call Service > 22-09: DID Basic Setup** from the **System Data** pane. The **22-09: DID Basic Setup** menu will appear.
2. In the **22-09: DID Basic Setup** menu, perform the following:
   a. In the **01 - Dial-in Receive Digits** field enter **4**.
   b. Select **Transfer** using the **02 - Received Vacant Number Operation** drop-down menu.
   c. Select **DID Translation Table** using the **03 - Sub-addressing Mode** drop-down menu.
3. Select **Apply** to apply the settings.

Configuring the DID Translation Table (22-11: DID Translation Table)

To configure the DID translation table, use these steps to add a DID translation entry for each extension using DID:

1. Select the **System Data** tab, then select **22-XX: Incoming Call Service > 22-11: DID Translation Table** from the **System Data** pane. The **22-11: DID Translation Table** menu will appear.
2. In the **22-11: DID Translation Table** menu, perform the following:
   a. In the **01 - Received Number** field enter the last four digits of the called DID number (e.g., **4444** for DID number **256-555-4444**).
   b. In the **02 - Target Number** field enter the target extension for the DID number.
   c. In the **03 - Dial-in Name** field, enter the name that is assigned to the DID digits.
3. Select **Apply** to apply the settings.

Configuring the SIP Trunk CODEC (84-13: SIP Trunk Codec Information Basic Setup)

To configure the SIP trunk CODEC and dual-tone multi-frequency (DTMF) settings, follow these steps:

1. Select the **System Data** tab, then select **84-XX: VoIP Hardware Setup > 84-13: SIP Trunk Codec Information Basic Setup** from the **System Data** pane. The **84-13: SIP Trunk Codec Information Basic Setup** menu will appear.
2. In the **84-13: SIP Trunk Codec Information Basic Setup** menu, perform the following:
   a. Select **u-law** using the **03 - G.711 Type** drop-down menu.
   b. Select **20ms** using the **07 - G.729 Maximum Audio Frame Size** drop-down menu.

   > NOTE: *If outbound call failure is encountered, specify a maximum audio frame size of **20ms** for **all** CODEC types.*

   c. Select **RFC2833** using the **32 - DTMF Relay Mode** drop-down menu.
3. Select **Apply** to apply the settings.

Configuring the SIP Trunk Settings (84-14: SIP Trunk Basic Setup)

To configure the SIP trunk basic setup, follow these steps:

1. Select the **System Data** tab, then select **84-XX: VoIP Hardware Setup** > **84-14: SIP Trunk Basic Setup** from the **System Data** pane. The **84-14: SIP Trunk Basic Setup** menu will appear.
2. In the **84-14: SIP Trunk Basic Setup** menu, perform the following:
   a. Select **Request URI** using the **09 - Called Party Info** drop-down menu.
   b. Select **SIP-URL** using the **10 - URL Type** drop-down menu.
   c. Select **SIP UA Domain** using the **11 - URL/TO Header Information** drop-down menu.
3. Select **Apply** to apply the settings.

Additional Resources

There are additional resources available to aid in configuring your ADTRAN eSBC unit. Many of the topics discussed in this guide are complex and require additional understanding, such as using the CLI, eSBC 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*
eSBC Product Overview	*Session Border Controllers in AOS*
Media Anchoring	*Configuring Media Anchoring in AOS*