



Interoperability Guide

Integrating an ADTRAN eSBC with a Panasonic KX-TDE100

This interoperability guide provides instructions for integrating an ADTRAN enterprise session border controller (eSBC) and the Panasonic KX-TDE100 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 Panasonic KX-TDE100 products.

- *Overview on page 2*
- *Interoperability on page 2*
- *Hardware and Software Requirements and Limitations on page 3*
- *Configuring the ADTRAN eSBC on page 4*
- *ADTRAN eSBC Sample Configuration on page 9*
- *Configuring the Panasonic KX-TDE100 on page 10*
- *Additional Resources on page 31*

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, and 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.

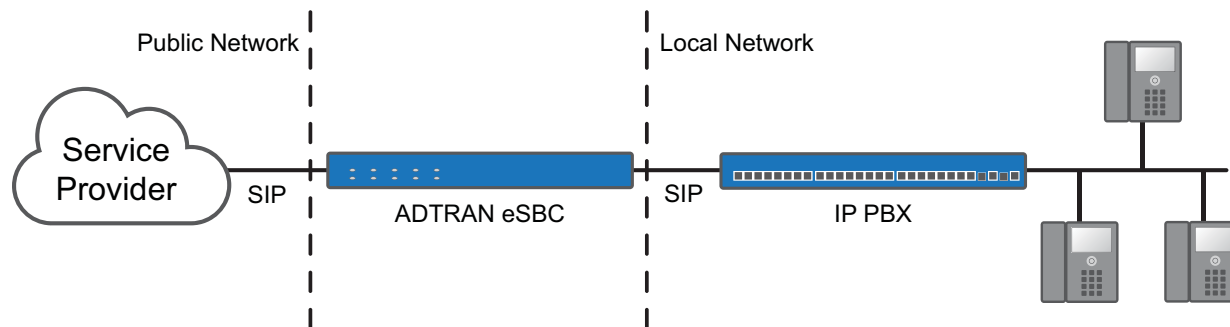


Figure 1. ADTRAN eSBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN eSBC and the Panasonic KX-TDE100. 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 Panasonic KX-TDE100 LAN interface 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 Panasonic KX-TDE100. 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 Panasonic KX-TDE100 supports various phone types (including digital, H.323, and SIP IP phones). The phones register locally to the Panasonic KX-TDE100. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN eSBC gateway.

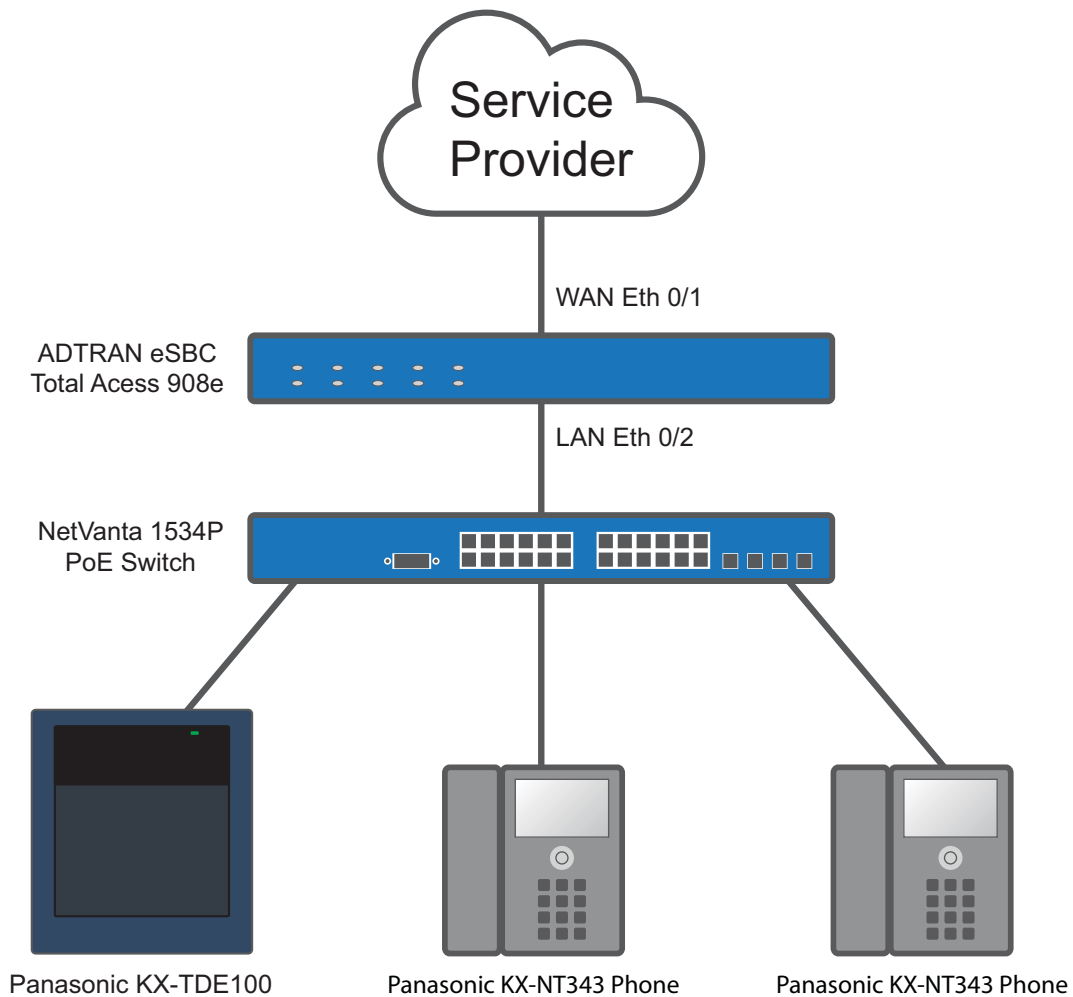


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with the Panasonic KX-TDE100 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 IP Business Gateway with eSBC	R10.9.4
Panasonic KX-TDE100	N/A
Panasonic IPCMPR	5.0002
Panasonic KX-NT343 Phone	See IPCMPR

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 Panasonic KX-TDE100, 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 Panasonic KX-TDE100 on page 8*
- *Step 9: Configure the Double reINVITE Preference on page 8*
- *Step 10: Configure SIP Privacy (Optional) on page 9*
- *ADTRAN eSBC Sample Configuration 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.



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**. 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 Panasonic KX-TDE100 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.



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 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 Panasonic KX-TDE100 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 Panasonic KX-TDE100 LAN IP address.

In addition, the Panasonic KX-TDE100 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 Panasonic KX-TDE100

After configuring a trunk group for the service provider, create a trunk group for the Panasonic KX-TDE100 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 Panasonic KX-TDE100 and a service provider.



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.



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 Panasonic KX-TDE100

The KX-TDE100 is configured using the Panasonic Unified Maintenance Console software installed on a PC. Refer to the Panasonic documentation listed in [Additional Resources on page 31](#) 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.

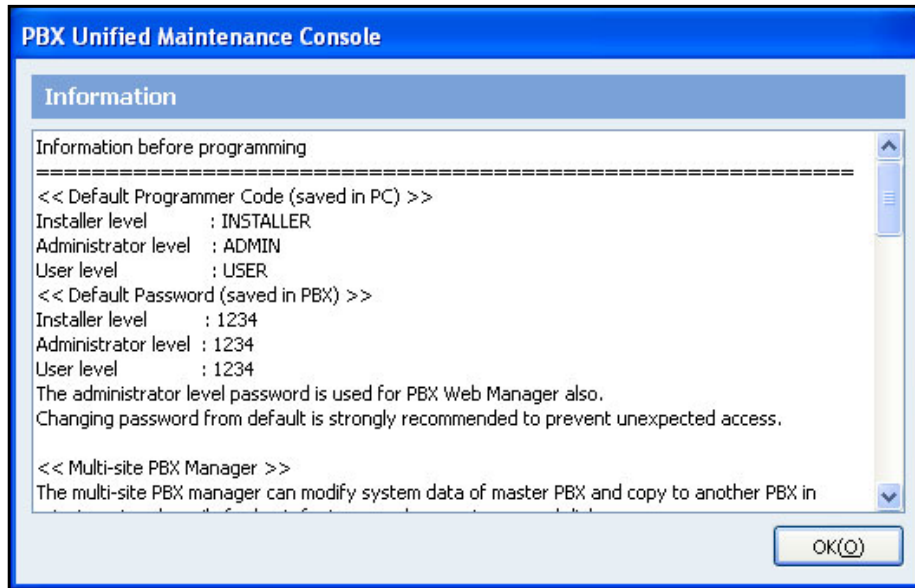
To configure the Panasonic IP PBX, follow these steps:

- *Step 1: Connect to the KX-TDE100 with the Unified Maintenance Console on page 12*
- *Step 2: Configure the KX-TDE100's Default Gateway on page 14*
- *Step 3: Verify or Change the SIP Client Port Number on page 16*
- *Step 4: Configure SIP Trunk IP Address on page 20*
- *Step 5: Associate DIDs to IP Phone Extensions on page 23*
- *Step 6: Configure Public Caller ID for IP Phones on page 26*
- *Step 7: Configure Outbound Calling on the SIP Trunk on page 27*
- *Step 8: Verify and Add Dial Plans on page 29*
- *Step 9: Bring the Card In Service on page 30*

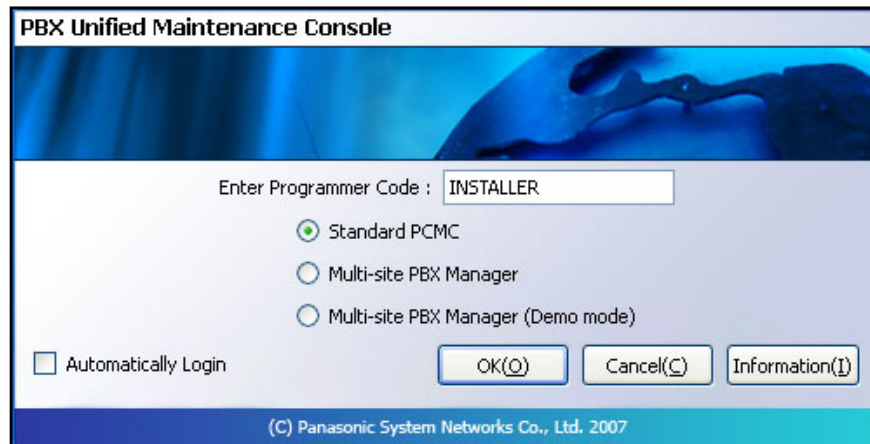
Step 1: Connect to the KX-TDE100 with the Unified Maintenance Console

Use the following steps to connect the Panasonic KX-TDE100 with the Unified Maintenance Console:

1. From the PC with the Panasonic configuration software, open the Unified Connection Manager. Select **OK** to continue.



2. At the prompt, enter the programmer code **INSTALLER** and select **OK**.



3. Select **Connect** from the PBX Unified Maintenance Console.



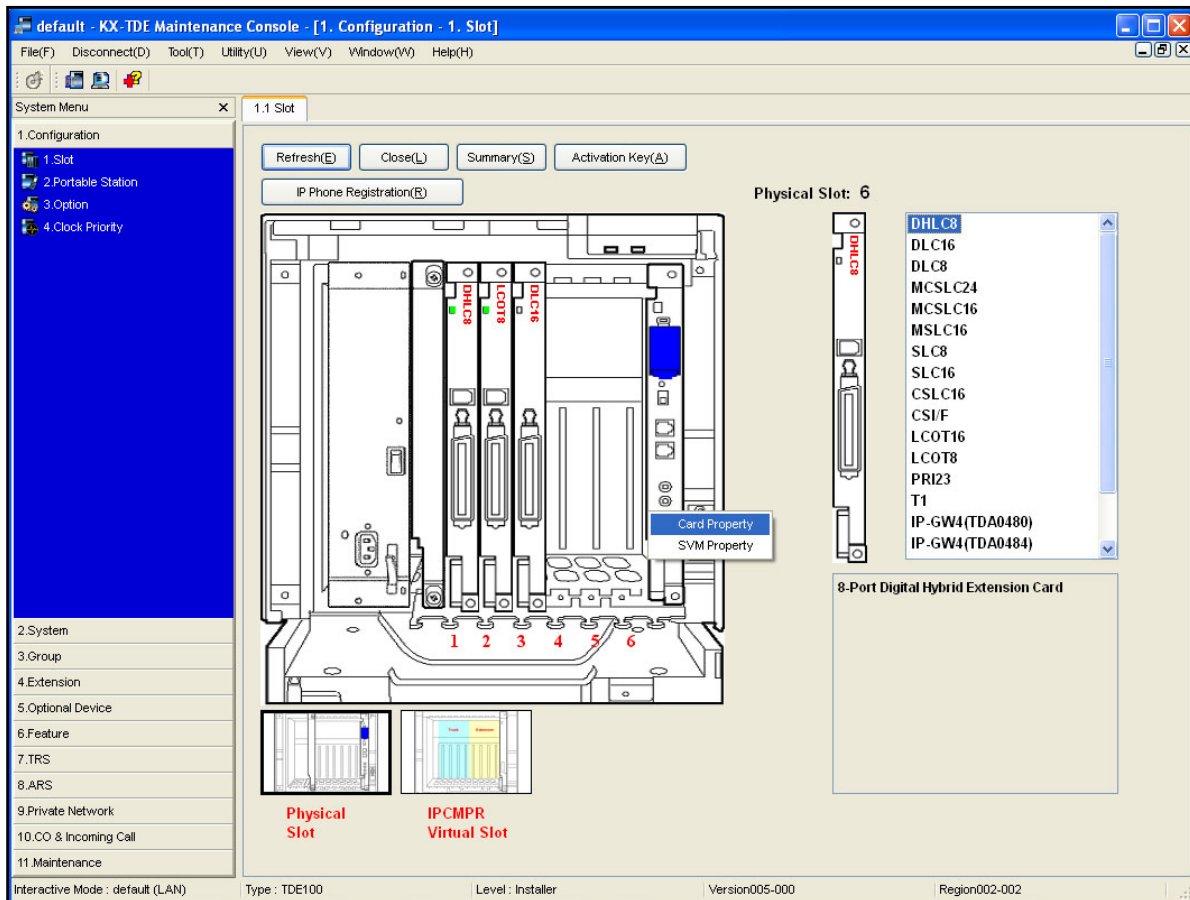
4. Select the LAN tab and enter the KX-TDE100's IP Convergence Main Processing Card (IPCMR) LAN IP address and port number. Select **Connect**.

The screenshot shows the "Connect" dialog box within the PBX Unified Maintenance Console. The "Profile File(P)" dropdown is set to "default". Under "Connection Property", the "PBX Model" is set to "KX-TDE100/200". The "LAN" radio button is selected. Below this, there are tabs for "LAN", "Modem", "RS-232C", and "USB". The "LAN" tab is active, showing an "IP Address" field with "192.168.1.2" and a "Search(S)" button. The "Port" field contains "35300". There is an "Enter Password" field with four dots. A "Save Password" checkbox is checked. A note at the bottom says "*) Please change the password frequently." At the bottom right, there are "Connect(O)" and "Cancel(C)" buttons.

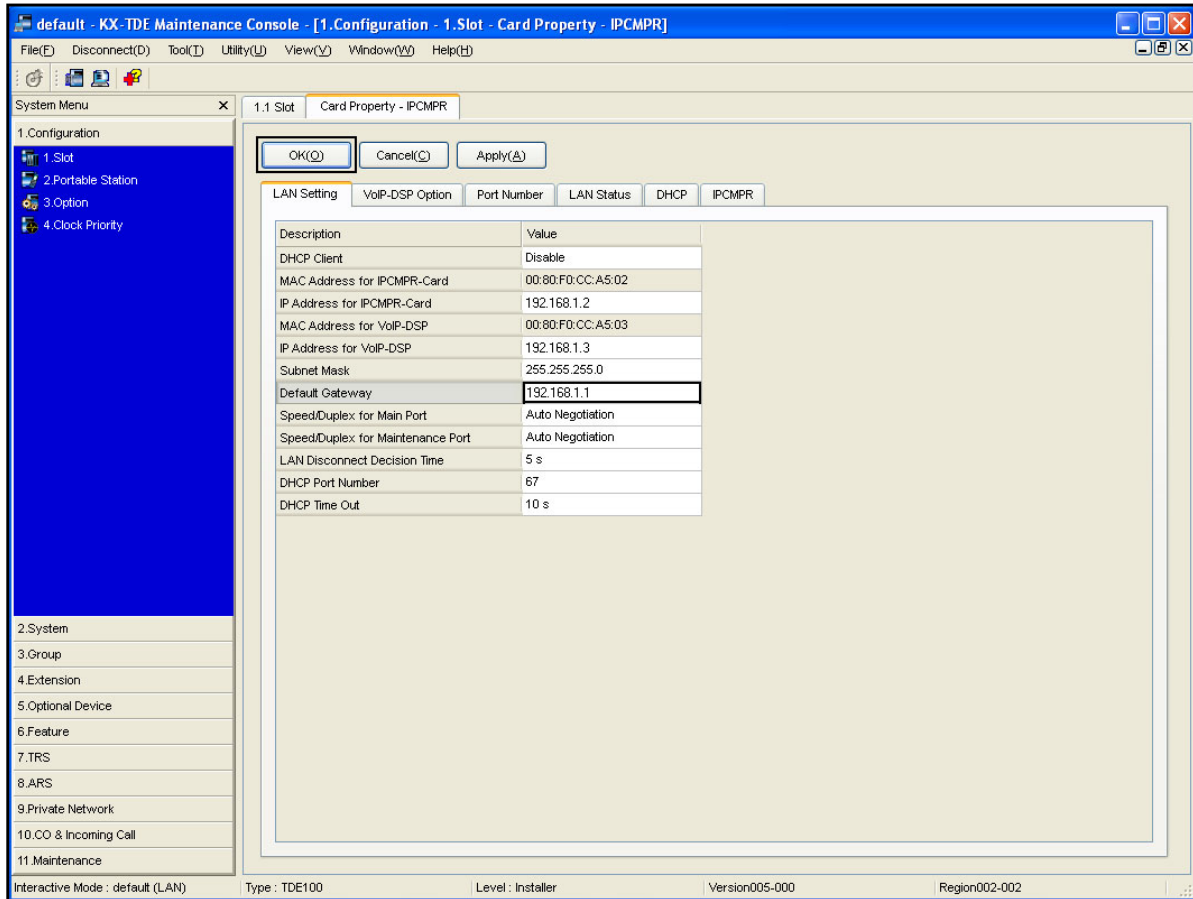
Step 2: Configure the KX-TDE100's Default Gateway

Use the following steps to configure the KX-TDE100's default gateway:

1. In the **System Menu** on the left, select **1.Configuration > 1.Slot**. The chassis view will display. Mouse over the IPCMPR card and select **Card Property**.



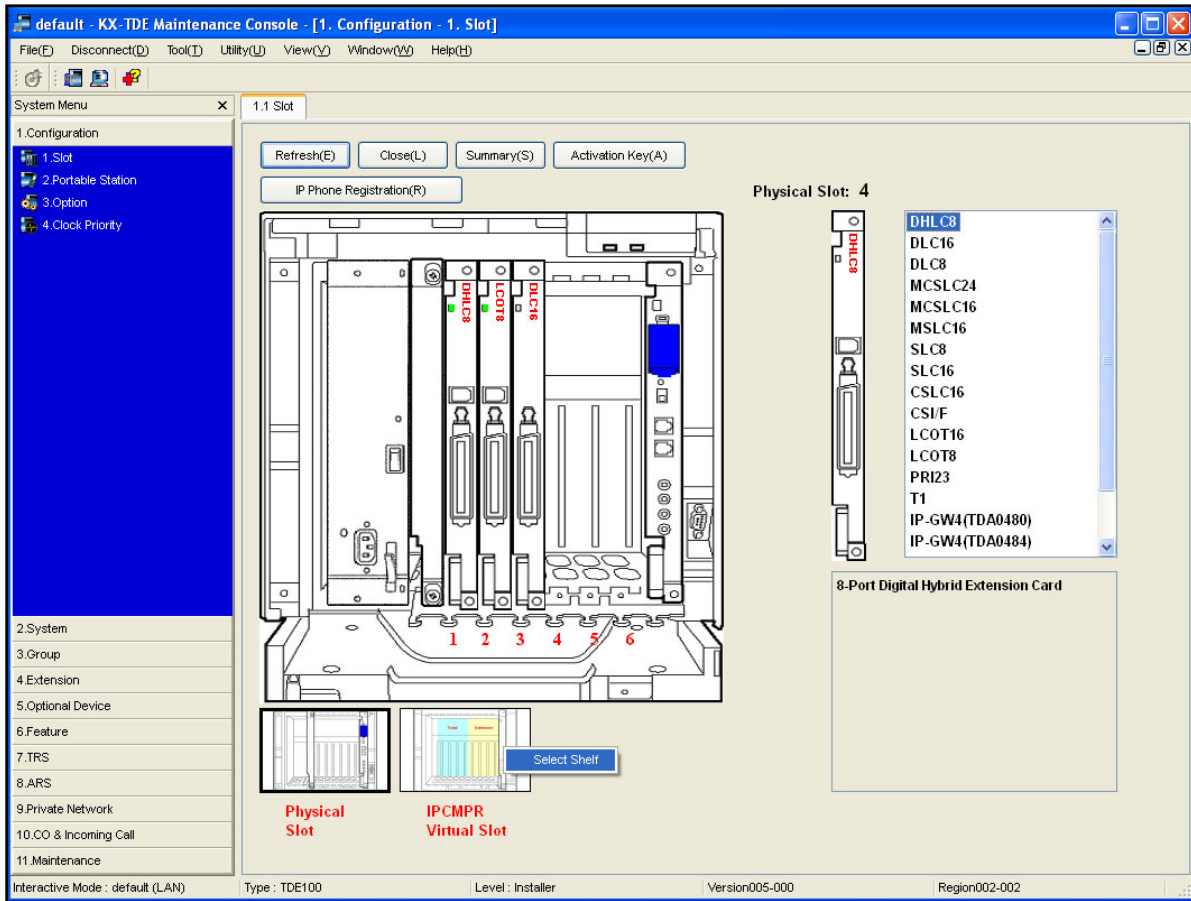
- From the **LAN Setting** tab, highlight the **Default Gateway** option, and enter the LAN IP address of the ADTRAN eSBC. Select **OK**.



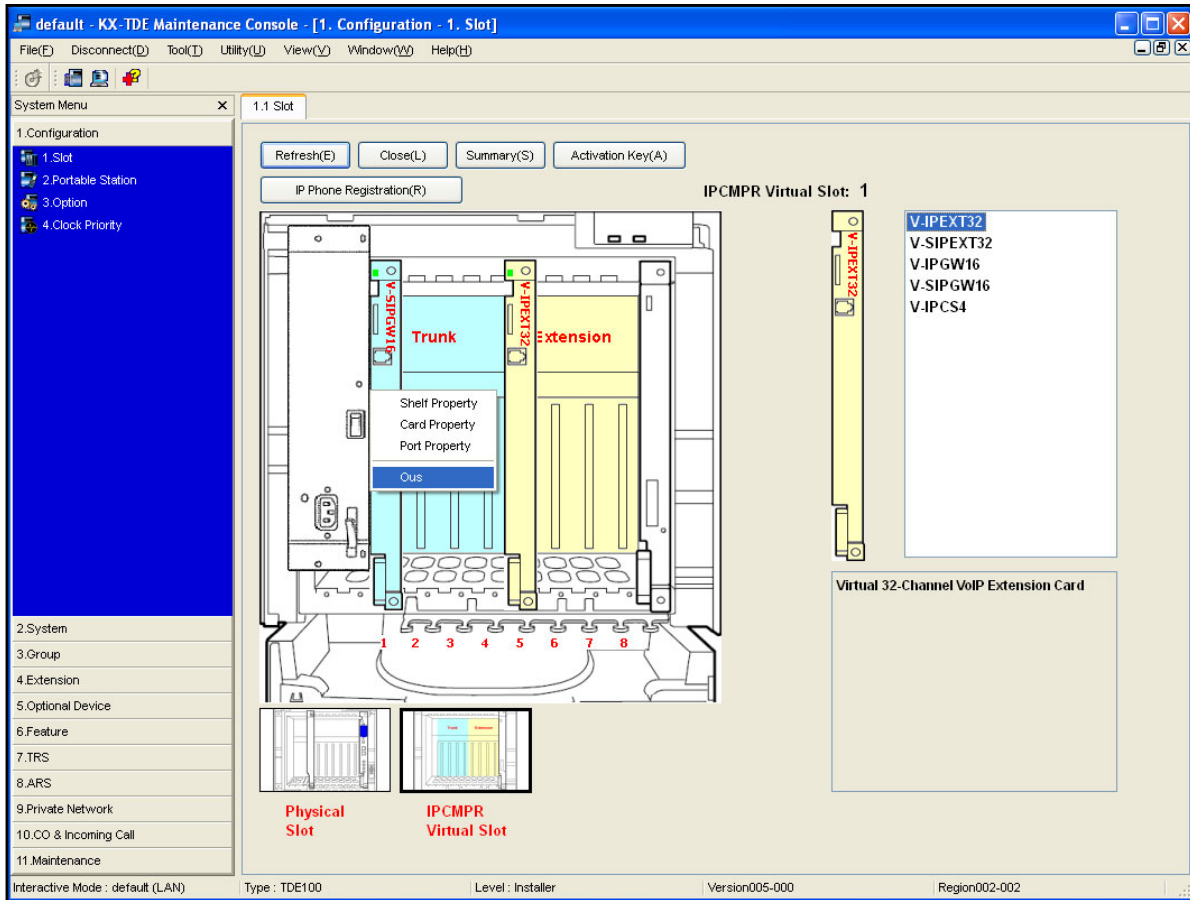
Step 3: Verify or Change the SIP Client Port Number

Use the following steps to verify or change the SIP client port number:

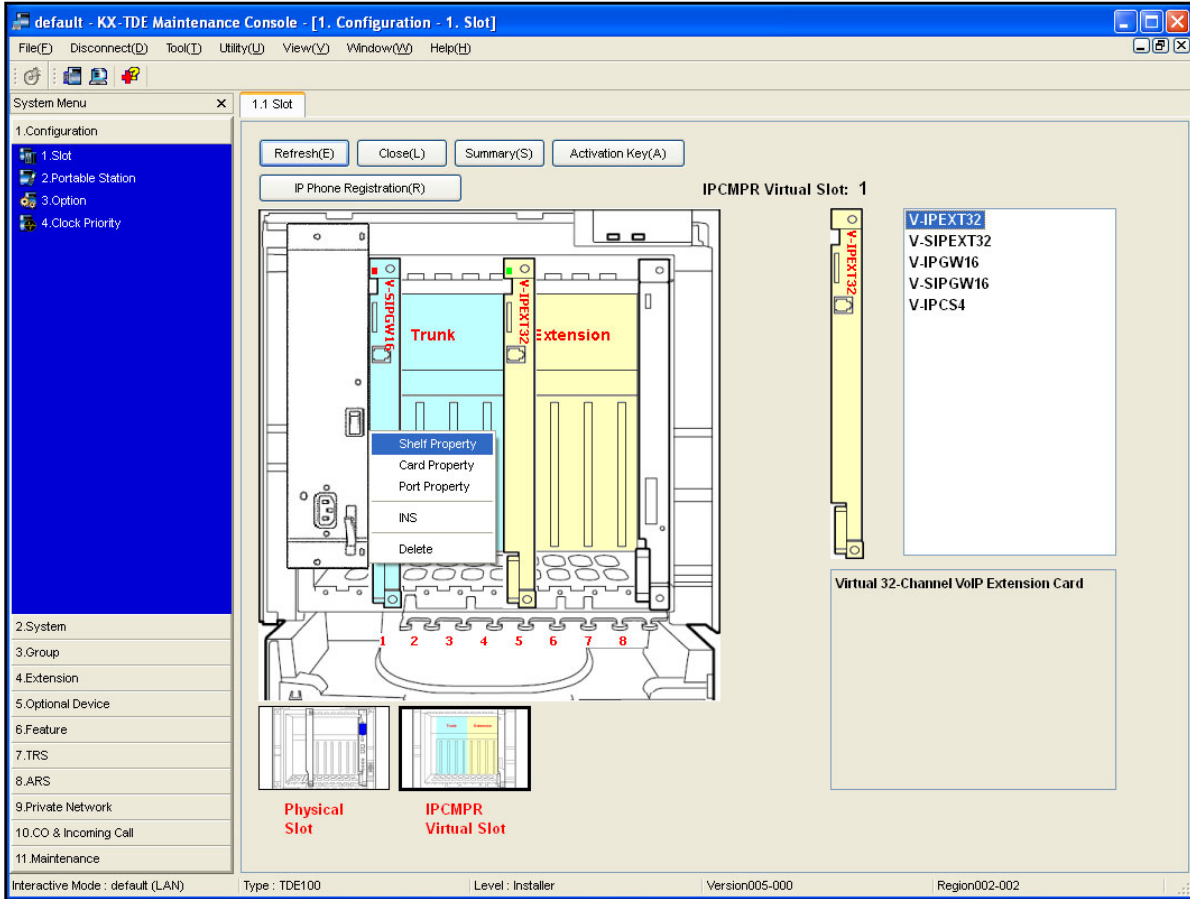
1. From the **Chassis** view, mouse over the **IPCMPR Virtual Slot** picture at the bottom. Select **Select Shelf**.



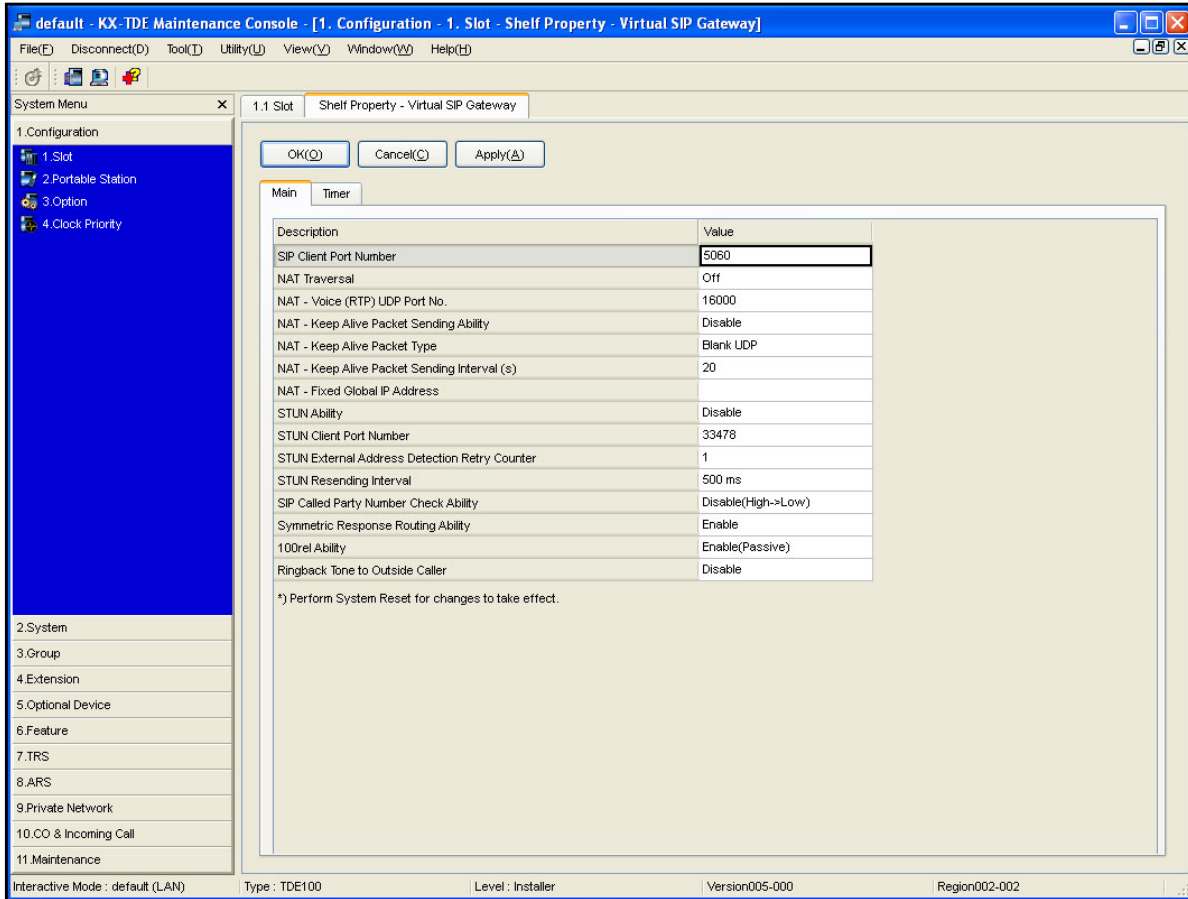
2. Mouse over the **V-SIPGW16** card, and select **OUS** to take the card out of service. Select **Yes** when prompted if you want to take the card out of service.



3. Mouse over the **V-SIPGW16** card, and select **Shelf Property**.



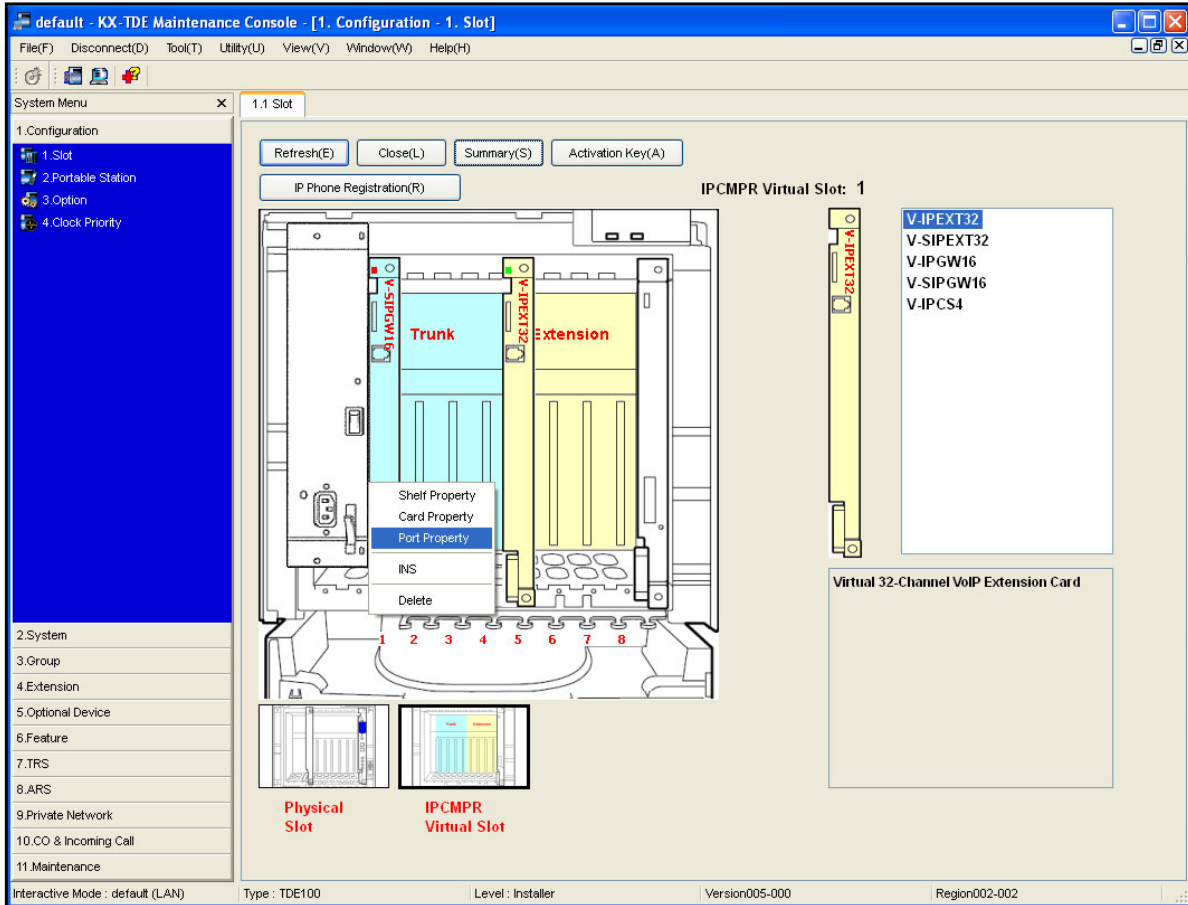
4. From the **Main** tab, verify or change the **SIP Client Port Number** to match the port configured under the ADTRAN eSBC Voice Trunk configuration. This port must match the port configured on the ADTRAN eSBC's trunk configuration connected to the PBX. **NAT Traversal** should be **Off**. Select **OK** when complete.



Step 4: Configure SIP Trunk IP Address

Use the following steps to configure the SIP trunk IP address:

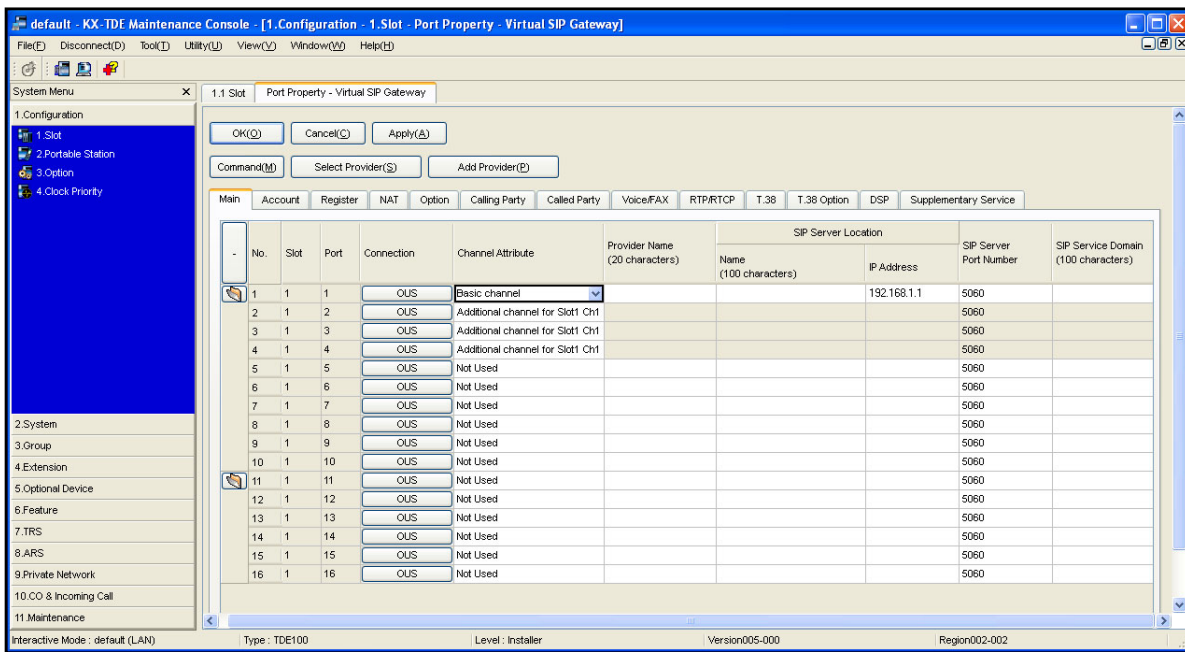
1. Mouse over the **V-SIPGW16** card, and select **Port Property**.



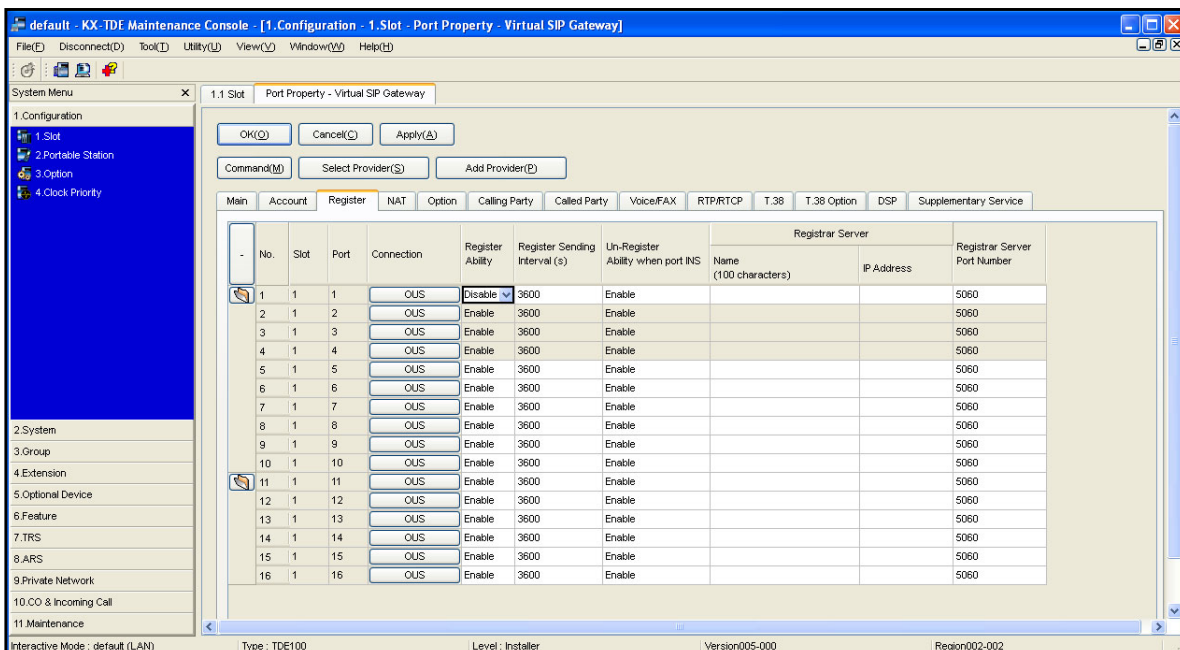
- From the **Main** tab, find the first row (Slot 1 Port 1). Change the **Channel Attribute** to **Basic channel**. In the **SIP Server Location/IP Address** column, enter the ADTRAN eSBC's LAN IP address, and verify port **5060** is set for the **SIP Server Port Number**.

For additional SIP trunk channels, change the **Channel Attribute** to **Additional channel for Slot1 Ch1**. These channels will share the configuration from slot 1 channel 1. The number of channels available is dependent on the number licensed on the PBX.

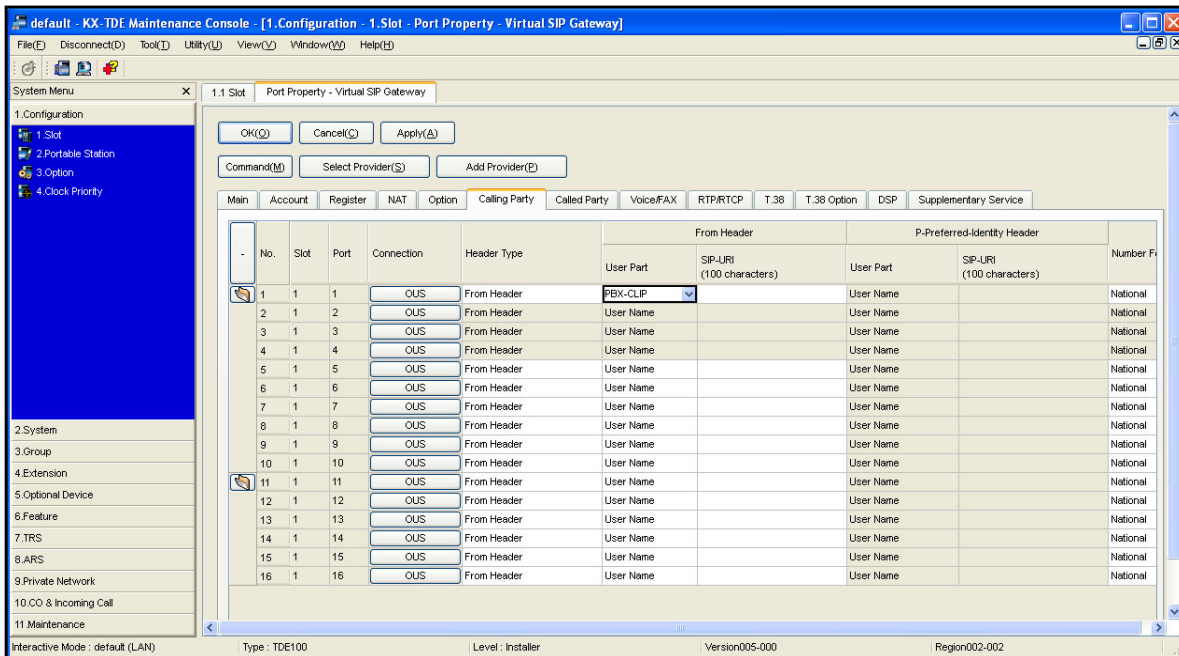
Select **OK** when complete.



- From the **Register** tab, find the first row (Slot 1 Port 1) and **Disable** the **Register Ability**. In this configuration the PBX will not be registering extensions. Select **OK** when complete.



- From the **Calling Party** tab, change the **User Part** to **PBX-CLIP**. Select **OK** when complete.



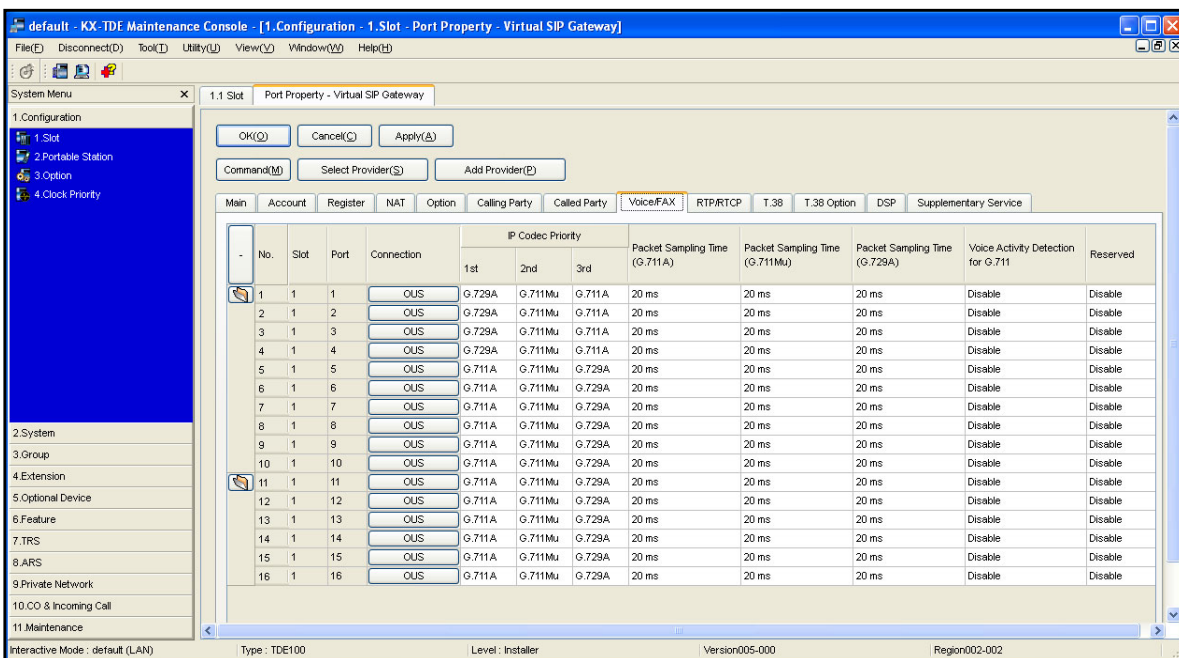
- From the **Voice/FAX** tab, change the **IP Codec Priority** of the SIP trunk as follows:

1st = G.729A

2nd = G.711mu

3rd = G.711A

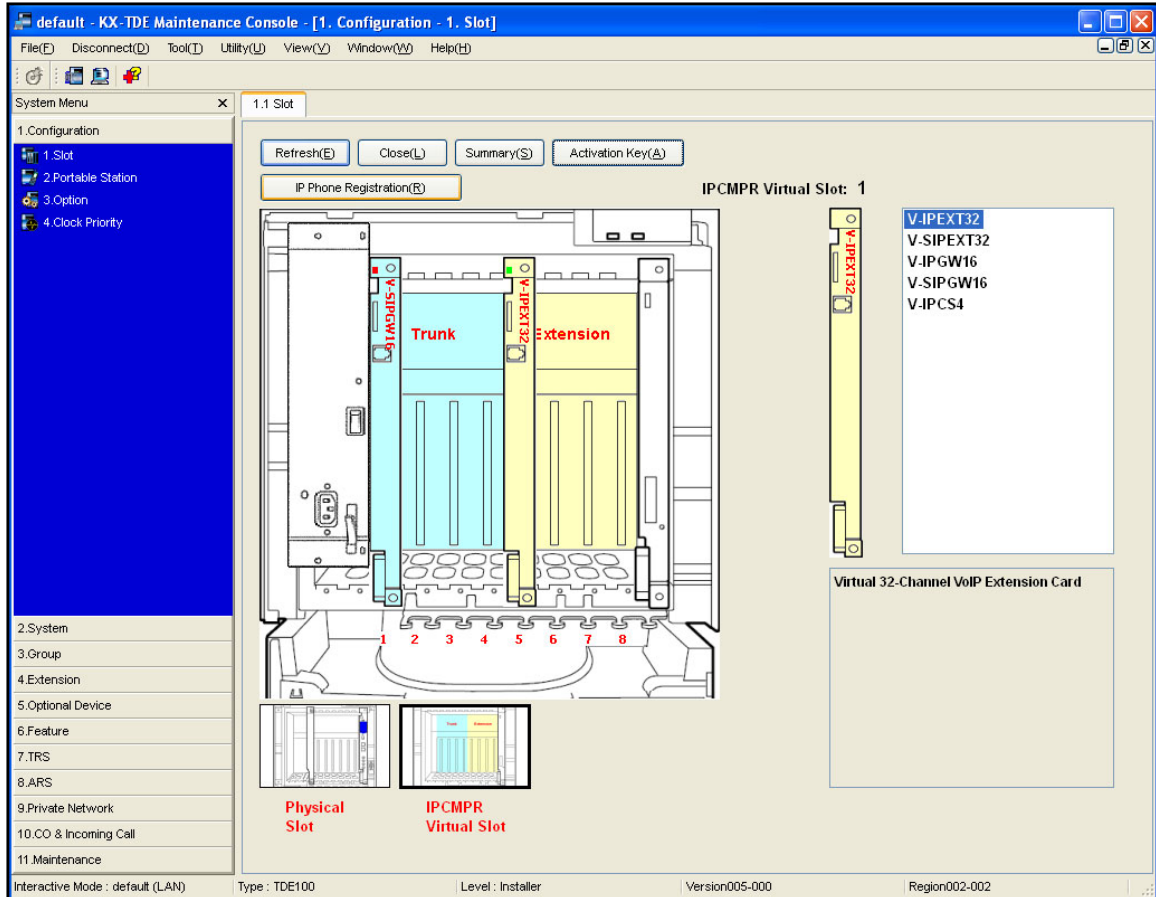
Ensure all **Packet Sampling Time** remains **20ms**. Select **OK** when complete.



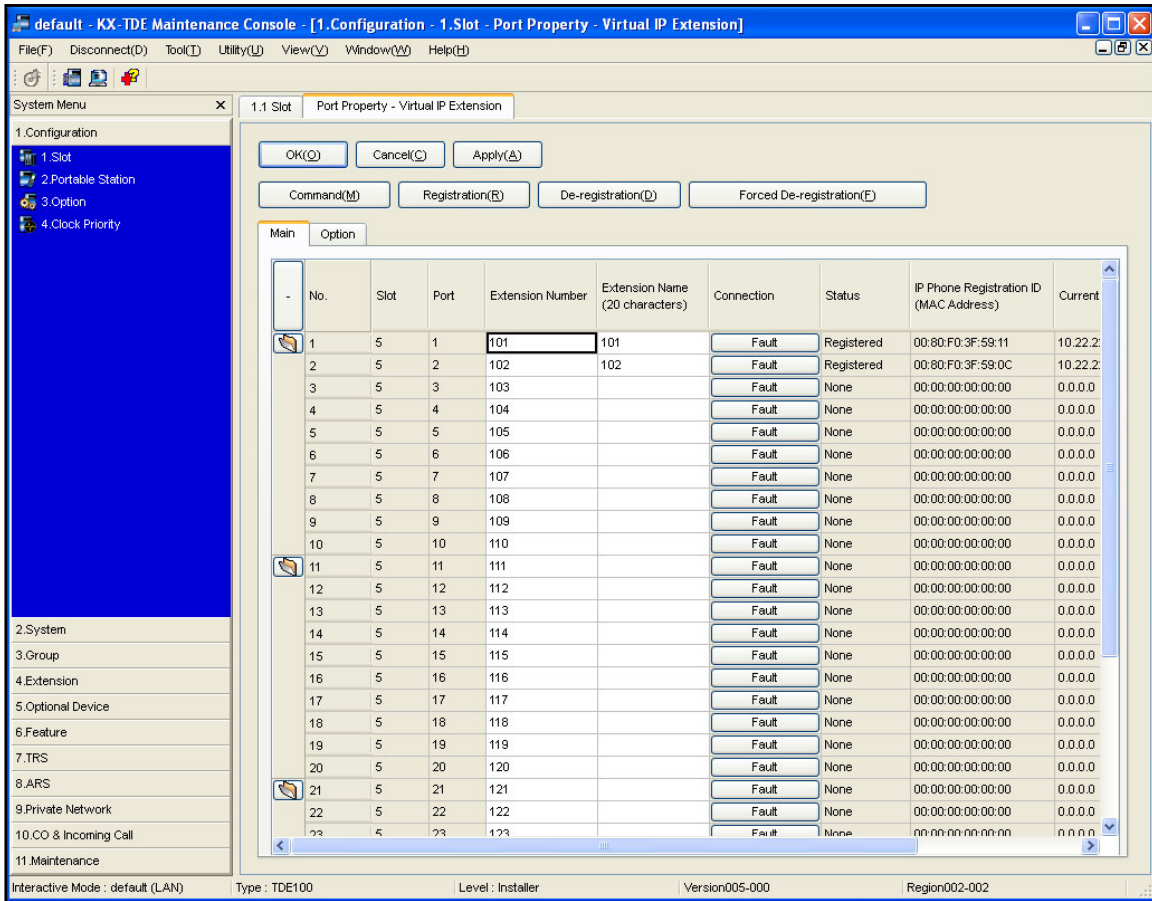
Step 5: Associate DIDs to IP Phone Extensions

Use the following steps to associate direct inward dialing (DID) numbers with IP phone extensions.

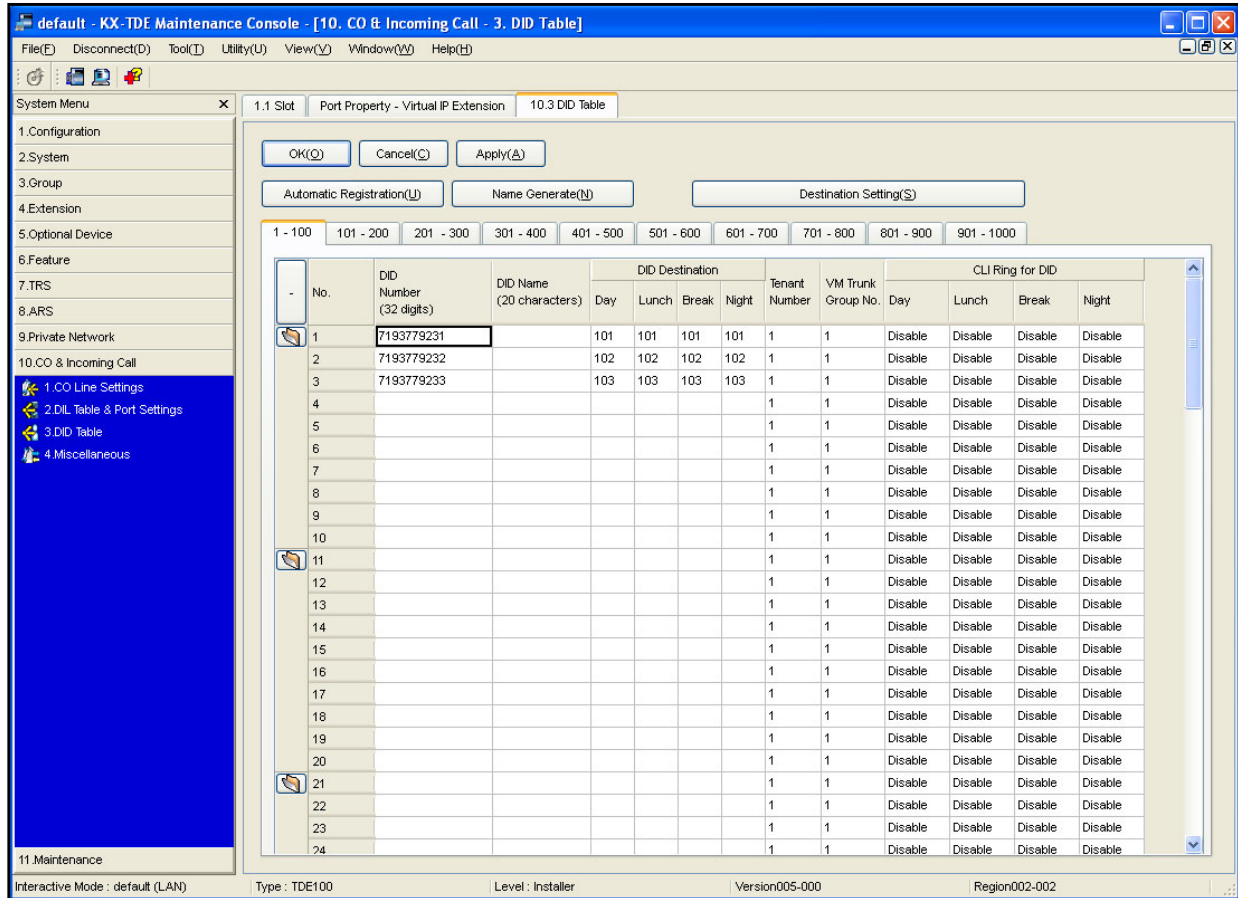
1. From the chassis view, select the **IP Phone Registration(R)** button.



2. Select the **Registration(R)** button.

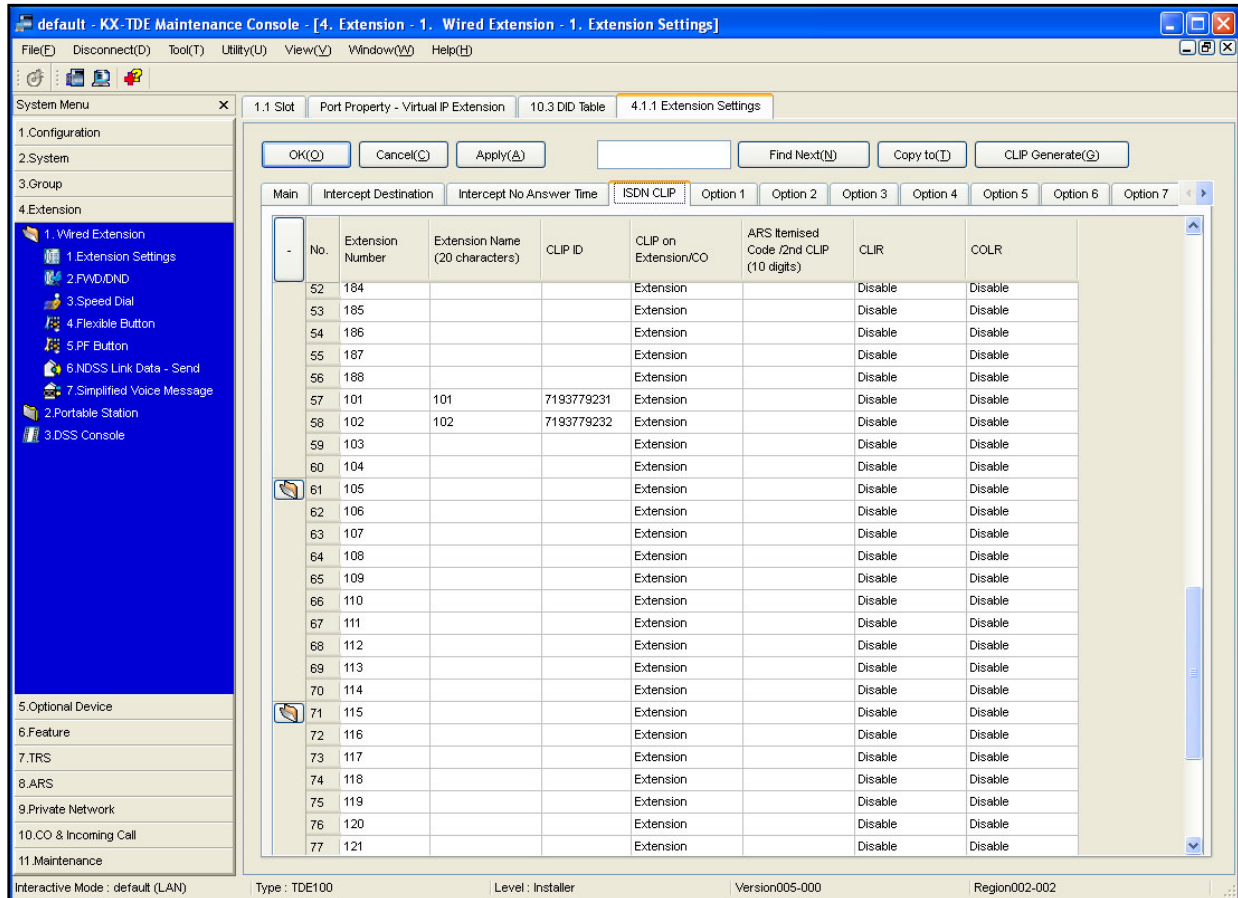


- From the **System Menu** on the left, select **10.CO & Incoming Call > 3.DID Table**. For each phone extension, enter an associated DID. The DID is the public telephone number provided by the service provider. Enter the DID number under the **DID Number** column. Then, enter the appropriate extension under the **DID Destination** columns (day/lunch/break/night) for that row. Select **OK** when complete.



Step 6: Configure Public Caller ID for IP Phones

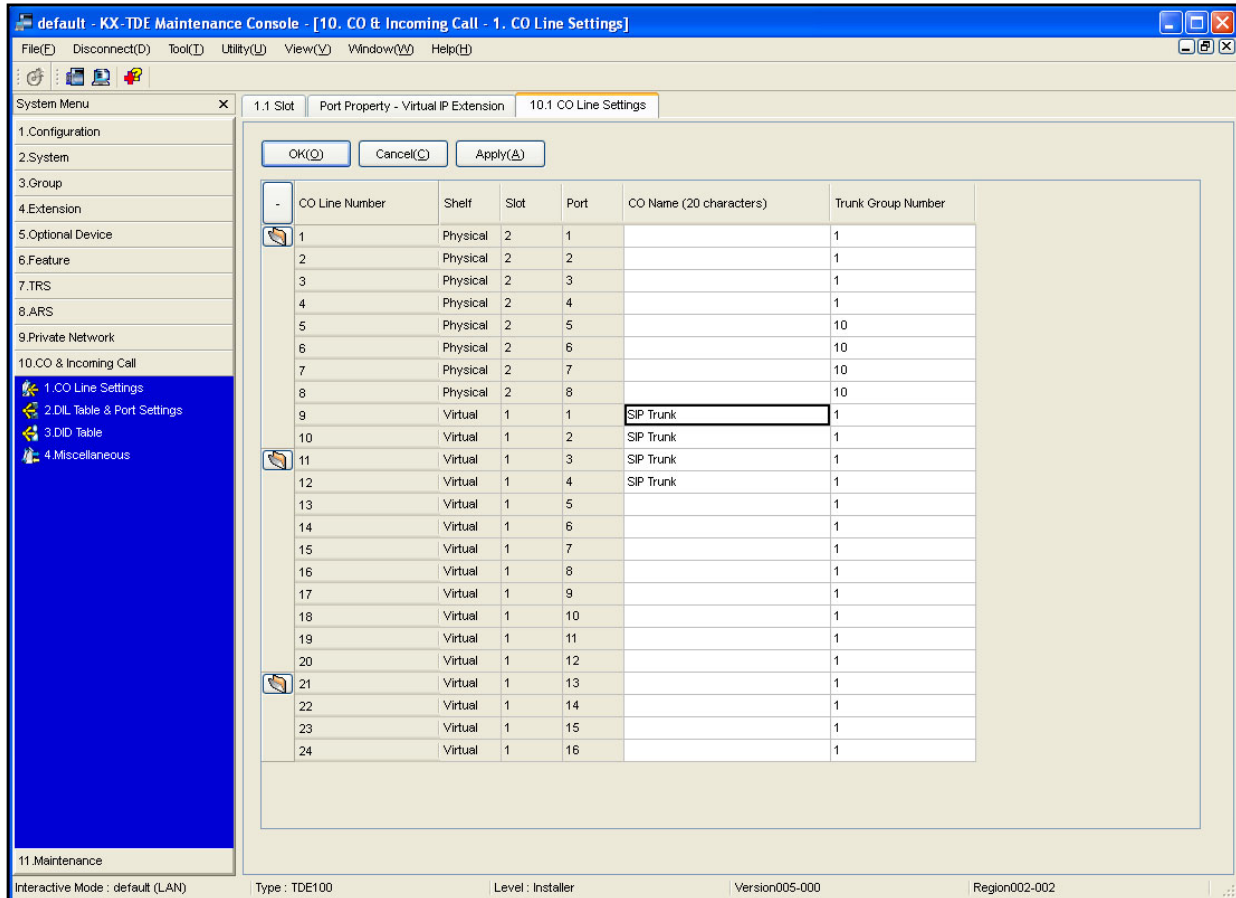
From the **System Menu** on the left, select **4.Extension > 1.Wired Extension > 1.Extension Settings**. Locate the **Extension Number** for each IP phone. Enter an **Extension Name** (this example uses the extension number) and the associated public DID under the **CLIP ID**. This will be used as the external caller ID for the extension. Select **OK** when complete.



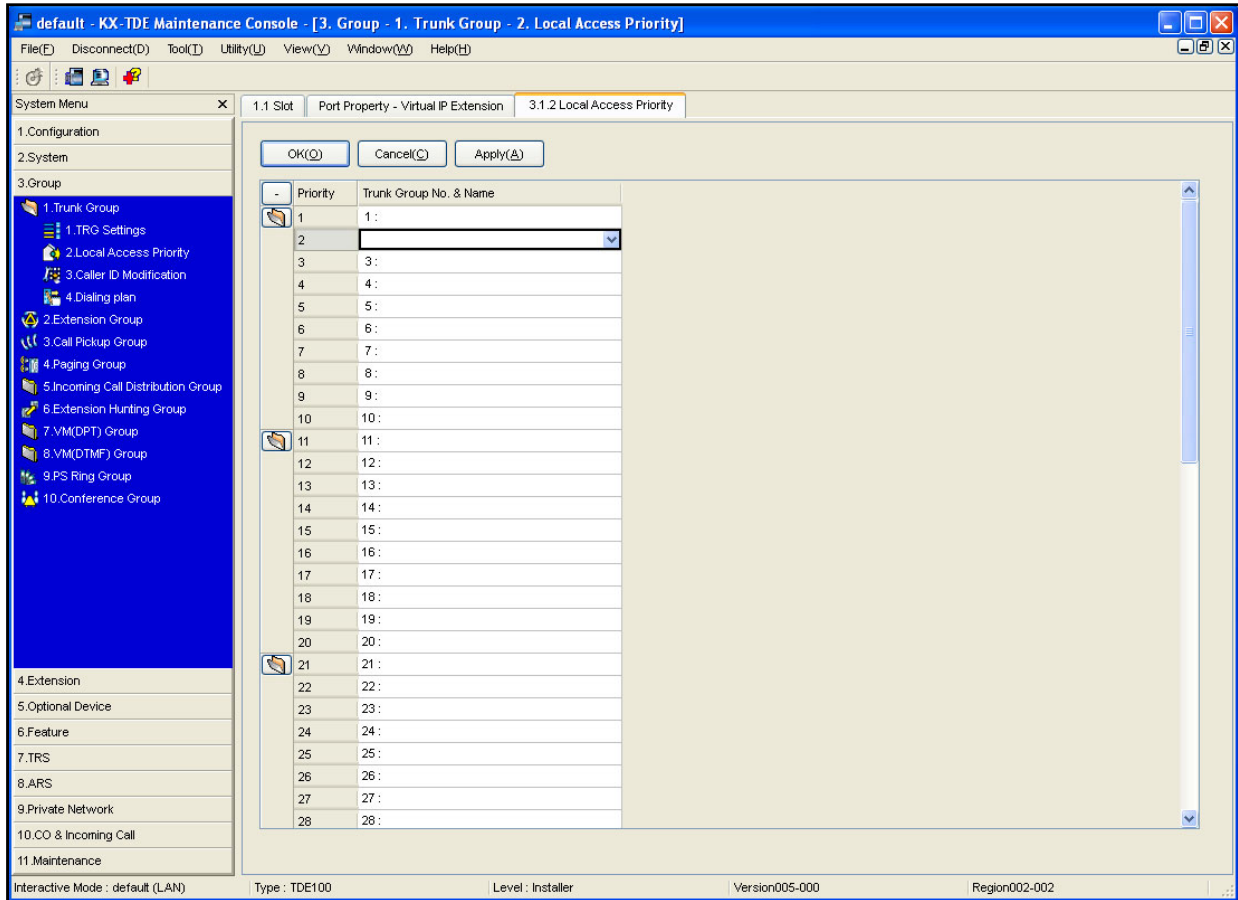
Step 7: Configure Outbound Calling on the SIP Trunk

Use the following steps to configure outbound calling on the SIP trunk.

1. From the **System Menu** on the left, select **10.CO & Incoming Call > 1.CO Line Settings**. Locate the SIP trunks. These will be **Virtual** under the **Shelf** column and correlate to the same **Slot** and **Port** numbers as configured in *Step 4: Configure SIP Trunk IP Address on page 20*. Enter a SIP trunk description in the **CO Name** field (if desired), and verify the **Trunk Group Number** is the same on all configured SIP trunks. Select **OK** when complete.

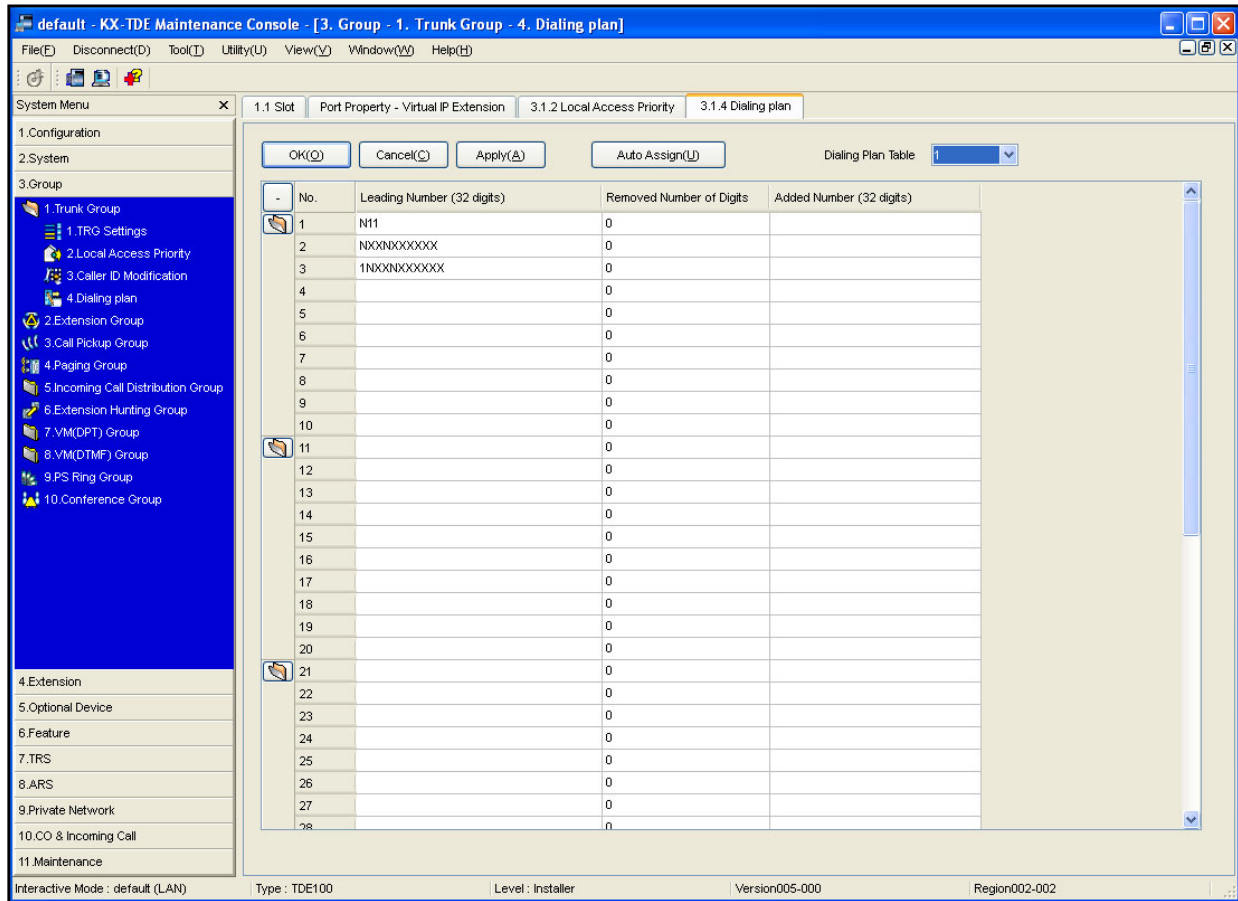


- From the **System Menu** on the left, select **3.Group > 1.Trunk Group > 2.Local Access Priority**. Next to the **Priority 1** entry in the **Trunk Group No. & Name** column, select the **Trunk Group Number** identified in the previous step. Select a blank **Trunk Group Number** for **Priority 2**. Select **OK** when complete.



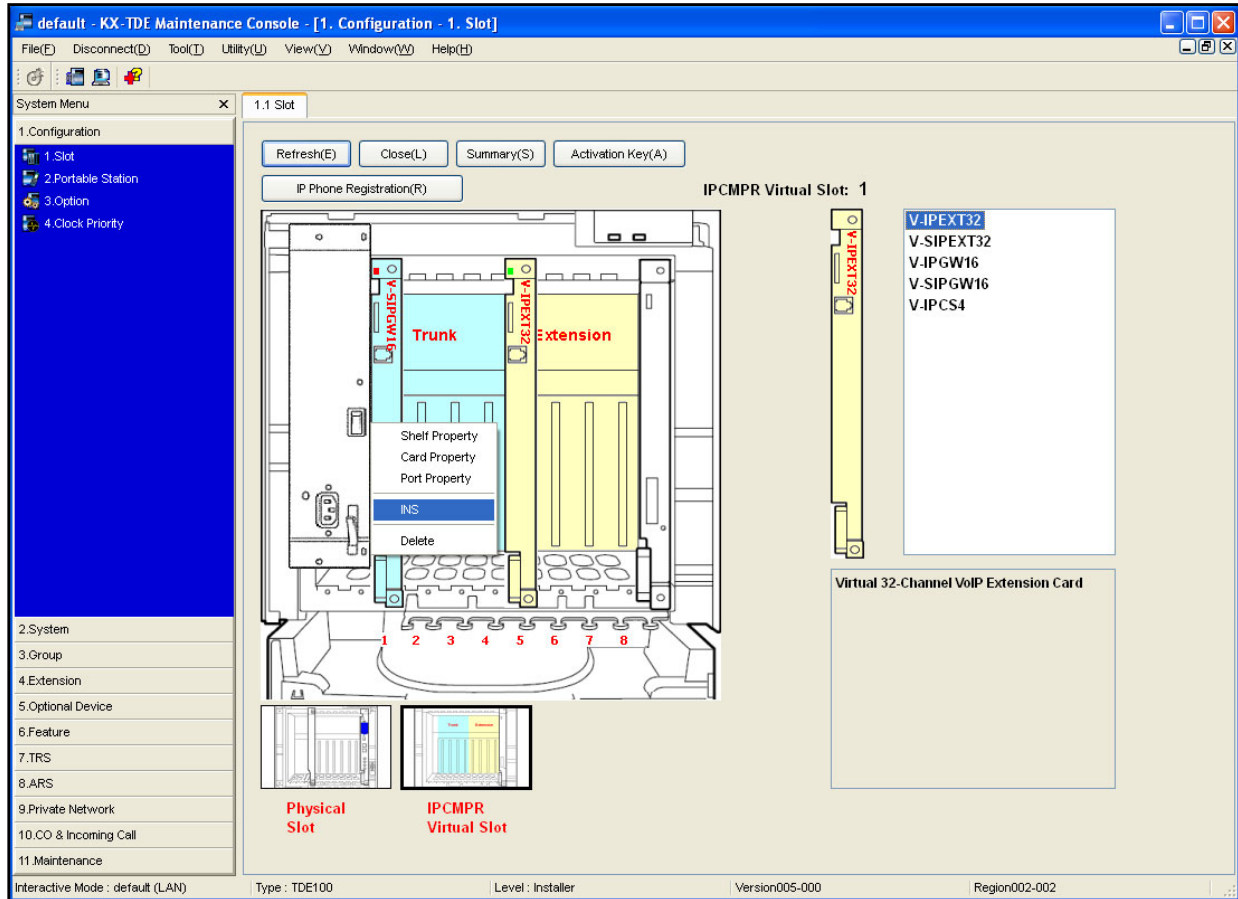
Step 8: Verify and Add Dial Plans

From the **System Menu** on the left, select **3.Group > 1.Trunk Group > 4.Dialing plan**. Verify all necessary call dial plans are present or add additional as needed. Select **OK** when complete.



Step 9: Bring the Card In Service

From the **System Menu** on the left, select **1.Configuration > 1.Slot**. Mouse over the **V-SIPGW16** card and select **INS**. This will bring the card in service, and the basic configuration of the Panasonic KX-TDE100 is complete.



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	<i>AOS Command Reference Guide</i>
ANI and DNIS Substitution	<i>Enhanced ANI/DNIS Substitution in AOS Voice Products</i>
eSBC Product Overview	<i>Session Border Controllers in AOS</i>
Media Anchoring	<i>Configuring Media Anchoring in AOS</i>

For additional information on configuring Panasonic products, refer to the following guide and manuals available from Panasonic:

- KX-TDE100 Feature Guide
- KX-TDE100 User Manual
- KX-TDE100 Operating Manual