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Interoperability Guide

Integrating ADTRAN SBC Gateways with the Allworx 6X IP PBX

This interoperability guide provides instructions for integrating the Allworx 6x Internet Protocol private branch exchange (IP PBX) with an ADTRAN Session Border Controller (SBC) gateway to provide a Session Initiation Protocol (SIP) connection to a service provider. This guide provides an overview and instructions for the integration as well as a list of equipment used for testing the integration, the features supported by the integration, and the verified functionality of the integration.

This guide consists of the following sections:

Application Overview

ADTRAN SBC products, such as the Total Access 908e, are used by service providers to provide SIP trunking to customer IP PBX systems. They provide SBC 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 SBC terminates the SIP trunk from the service provider and operates as a SIP back-to-back user agent (B2BUA). A second SIP trunk connects the ADTRAN SBC to the IP PBX. *Figure 1* illustrates the use of the ADTRAN SBC in a typical network deployment.

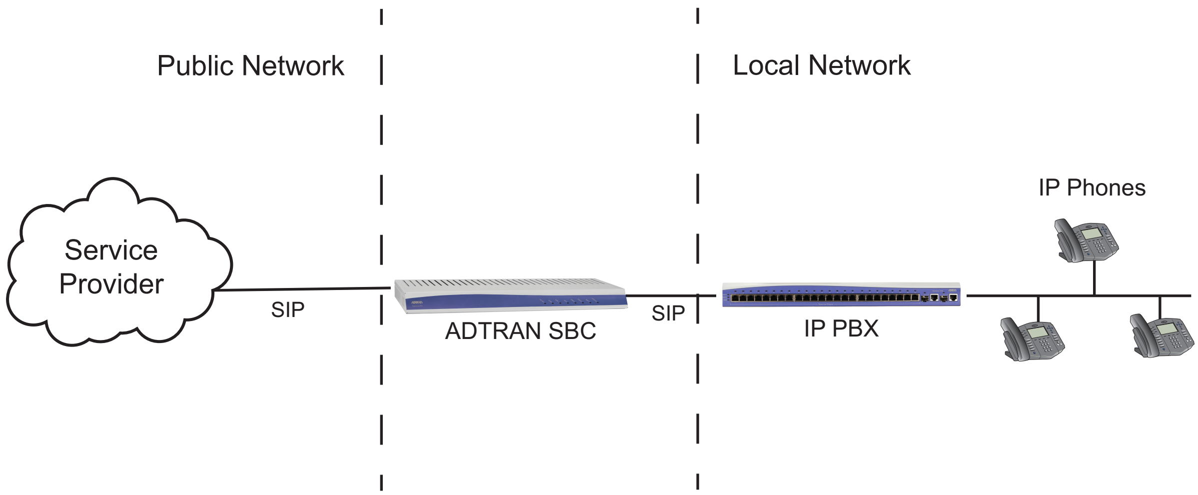


Figure 1. ADTRAN SBC in the Network

Interoperability

This interoperability guide provides interoperability support for SIP trunking between an ADTRAN SBC and the Allworx 6x IP PBX. The configuration is a typical SIP trunking application where the ADTRAN gateway Ethernet interface provides the Ethernet wide are network (WAN) connection to the service provider's network. A second Ethernet interface connects to the Allworx 6x IP PBX. Two SIP trunks are configured on the ADTRAN SBC: one to the service provider and another to the Allworx 6x. The ADTRAN SBC operates as a SIP B2BUA, and outbound and inbound calls to the public switched telephone network (PSTN) are routed through the ADTRAN SBC. The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN SBC and the Allworx 6x.

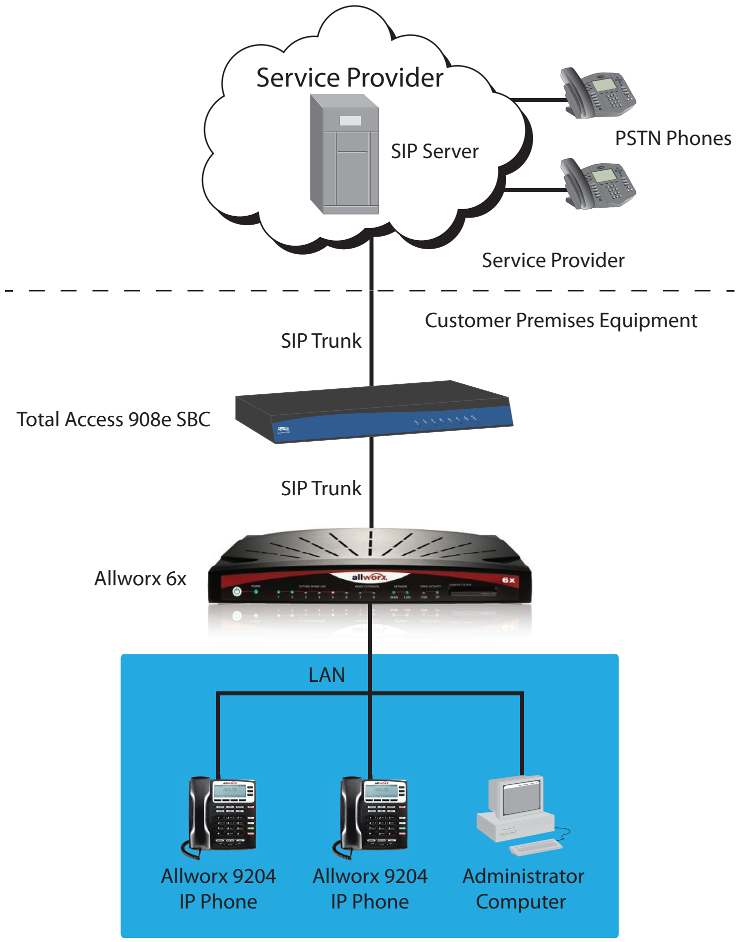


Figure 2. Network Topology Used for Interoperability Verification

Hardware and Software Requirements and Limitations

The Allworx 6x IP PBX is interoperable with ADTRAN products with the SBC feature pack. A list of ADTRAN products that support SBC feature pack is 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 software/firmware versions used during verification testing:

Table 1. Hardware Components Tested

Provider	Hardware Component	Version
ADTRAN	Total Access 908e with SBC	SBC AOS R10.9.0
Allworx	6x Server IP PBX	7.5.8.5
Allworx	9204 IP Phone	Software: 2.5.8.3 DSP: 204C

Supported Features and Exceptions

The following sections provide information on the feature verification performed and issues discovered during interoperability verification. The features listed in the *Supported Features* section below are the features verified to work with the testing equipment. These are the only features you can expect to function with the configuration provided in this guide.

Supported Features

The focus of the interoperability verification for this solution was SIP trunking. The following features were tested and are supported by the integration:

- Allworx 6x SIP trunking with ADTRAN Total Access 908e SBC
- Basic calling (internal and external SIP trunking to PSTN)
- Calling party number presentation
- Anonymous calling
- Call transfers (blind, attended, and unattended)
- Call forwarding
- Call hold and resume
- Call pickup
- Call waiting
- Do not disturb (DND) operations
- Call park
- Three-way conference calling
- Auto attendant operations
- RFC 2833 dual-tone multi-frequency (DTMF) operation
- CODEC transcoding (G.729, G.711)

Exceptions

The following minor issues were discovered during interoperability verification:

- For outbound calls to an external PSTN number from a local IP phone, when the caller places the call on hold, the external user does not hear Music on Hold. This is not the case for inbound calls. When an external PSTN user originates the call to a local user and the local user places the call on hold, the external PSTN user hears Music on Hold. This occurs with the Allworx configured to provide internal Music on Hold.

- The Allworx 6x IP PBX is not configurable for inband DTMF. However, RFC 2833 out-of-band DTMF operates as expected.

Configuring the ADTRAN SBC

To configure the ADTRAN SBC for integration with the Allworx 6x, follow these steps:

- *Step 1: Access the CLI on page 5*
- *Step 2: Configure the WAN Interface to the Service Provider on page 6*
- *Step 3: Configure the LAN Interface to the Allworx 6x on page 7*
- *Step 4: Configure a Static Default Route to the Service Provider's Gateway on page 8*
- *Step 5: Configure a SIP Trunk to the Service Provider on page 8*
- *Step 6: Configure a SIP Trunk To the Allworx 6x on page 9*
- *Step 7: Configure a Trunk Group for the Service Provider SIP Trunk on page 10*
- *Step 8: Configure a Trunk Group to the Allworx 6x on page 12*
- *Step 9: Enable Media Anchoring and Symmetric Filtering on page 14*
- *Step 10: Disable Prefer Double reINVITEs on page 14*
- *Step 11: Configure the Forward and Transfer Modes on page 15*

Step 1: Access the CLI

To access the command line interface (CLI) on the ADTRAN SBC, 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 Internet Protocol (IP) address (**10.10.10.1**), use the configured IP address.*

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

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

4. Enter Enable mode by entering **enable** at the prompt as follows:

   >enable

5. Enter your Enable mode password at the prompt.

> *The default Enable mode password is* ***password****. If your product no longer has the default Enable password, contact your system administrator for the appropriate password.*

6. Enter the unit's Global Configuration mode as follows:

   #configure terminal
   (config)#

Step 2: Configure the WAN Interface to the Service Provider

To configure a WAN interface on the ADTRAN SBC to the service provider, follow these steps:

> *The configuration in this section uses an Ethernet interface. Other interfaces can be used for the WAN interface.*

1. From the Global Configuration mode, use the **interface** <*interface type*> <*slot/port*> command to enter the Ethernet Interface Configuration mode for the specified interface. The <*slot/port*> variable specifies a slot and port for the interface being configured.

Syntax	Description
<interface type>	Specifies the interface type (e.g., Ethernet, Gigabit Ethernet, etc.). Type **interface range ?** for a complete list of valid interfaces.
<slot/port>	Specifies the slot and port of the interface.

   The following example enters the Ethernet Interface Configuration mode for Ethernet **0/1**:

   (config)#**interface ethernet 0/1**
   (config-eth 0/1)#

2. Use the **ip address** command to define the IPv4 address of the interface.

   (config-eth 0/1)#**ip address** *<ipv4 address> <subnet mask>*

Syntax	Description
ipv4 address	Specifies a valid IPv4 address. IPv4 addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).
subnet mask	Specifies the subnet mask that corresponds to a range of IPv4 addresses (network) or a specific host. Subnet masks can be expressed in dotted decimal notation (for example, **255.255.255.0**) or as a prefix length (for example, **/24**).

   The following example configures the interface with an IPv4 address of **198.51.100.1 /30**:

   (config-eth 0/1)#**ip address 198.51.100.1 /30**

3. Use the **media-gateway ip primary** command to specify that the interface's primary IPv4 address (configured in the previous step) should be used for SIP and Realtime Transport Protocol (RTP) traffic.

   (config-eth 0/1)#**media-gateway ip primary**

4. Use the **no shutdown** command to activate the interface.

   (config-eth 0/1)#**no shutdown**

Step 3: Configure the LAN Interface to the Allworx 6x

To configure a LAN interface on the ADTRAN SBC to the Allworx 6x, follow these steps:

1. From the Global Configuration mode, use the **interface** *<interface type> <slot/port>* command to enter the Ethernet Interface Configuration mode for the specified interface. The *<slot/port>* variable specifies a slot and port for the interface being configured.

Syntax	Description
<interface type>	Specifies the interface type (e.g., Ethernet, Gigabit Ethernet, etc.). Type **interface range ?** for a complete list of valid interfaces.
<slot/port>	Specifies the slot and port of the interface.

   The following example enters the Ethernet Interface Configuration mode for Ethernet interface **0/2**:

   (config)#**interface ethernet 0/2**
   (config-eth 0/2)#

2. Use the **ip address** command to define the IPv4 address of the interface.

   (config-eth 0/2)#**ip address** *<ipv4 address> <subnet mask>*

Syntax	Description
ipv4 address	Specifies a valid IPv4 address. IPv4 addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).
subnet mask	Specifies the subnet mask that corresponds to a range of IPv4 addresses (network) or a specific host. Subnet masks can be expressed in dotted decimal notation (for example, **255.255.255.0**) or as a prefix length (for example, **/24**).

   The following example configures the interface with an IPv4 address of **192.168.1.254 /24**:

   (config-eth 0/2)#**ip address 192.168.1.254 /24**

3. Use the **media-gateway ip primary** command to specify that the interface's primary IPv4 address (configured in the previous step) should be used for Realtime Transport Protocol (RTP) traffic.

   (config-eth 0/2)#**media-gateway ip primary**

4. Use the **no shutdown** command to activate the interface.

   (config-eth 0/2)#**no shutdown**

Step 4: Configure a Static Default Route to the Service Provider's Gateway

The default route is used as the destination for packets for which no more specific route is present. To set a static default route to the next hop gateway, use the **ip route** command from the Global Configuration mode:

(config)#**ip route 0.0.0.0 0.0.0.0** *<ip address>*

Syntax	Description
<ip address>	Specifies the IPv4 network address to add to the route table. IPv4 addresses should be expressed in dotted decimal notation. **The address used should be the IP address of the service provider's gateway.**

The following example creates a default route to **198.51.100.2**:

(config)#**ip route 0.0.0.0 0.0.0.0 198.51.100.2**

Step 5: Configure a SIP Trunk to the Service Provider

To create and configure a SIP trunk on the ADTRAN SBC to the service provider, follow these general steps:

> *You should consult with your service provider for specific requirements for configuring the SIP trunk.*

1. From the Global Configuration mode, use the **voice trunk** *<Txx>* **type sip** command to create a SIP trunk and enter the Voice SIP Trunk Configuration mode.

   (config)#**voice trunk** *<Txx>* **type sip**.

Syntax	Description
<Txx>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**)

   The following example creates SIP trunk **T01** and enters the Voice SIP Trunk Configuration mode:

   (config)#**voice trunk T01 type sip**
   (config-T01)#

2. From the Voice SIP Trunk Configuration mode, use the **sip-server primary** command to specify the fully qualified domain name (FQDN) or IP address of the SIP server to which the trunk will send SIP messages. The value used should be the address of the service provider's SIP server.

   (config-T01)#**sip-server primary** *<FQDN or IP address>* **[tcp** *<port>* **| udp** *<port>***]**

Syntax	Description
<FQDN or IP address>	Specifies the FQDN or IP address of the SIP server. IP addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).

Syntax	Description
tcp	Optional. Sets the Transmission Control Protocol (TCP) port of the outbound proxy server.
udp	Optional. Sets the User Datagram Protocol (UDP) of the outbound proxy server.
<port>	Optional. Specifies the TCP or UDP port number. Range is **0** to **65535**.

The following example sets the SIP server IP address to **203.0.113.1**:

(config-T01)#**sip-server 203.0.113.1**

3. Optional. If required by your service provider, use the **dial-sting source to** command to specify the SIP To header field as the dial string source.

(config-T01)#**dial-string source to**

4. Optional. If your service provider requires SIP registration, use the **register** command to define the SIP name for registration and the authorization name(s) and password(s).

(config-T01)#**register [***<name>* **| range** *<begin> <end>***] auth-name [***<username>* **| range** *<begin> <end>***] password [***<password>* **| range** *<begin> <end>***]**

Syntax	Description
register	Configures the SIP user to register.
<name>	Specifies the name of the SIP user to register.
range *<begin> <end>*	Specifies the beginning and ending of the range of SIP users to register.
auth-name	Optional. Configures a user name for authentication.
<username>	Specifies a single user name for authentication.
range *<begin> <end>*	Specifies the beginning and ending of the range of user names for authentication.
password	Optional. Configures a password for authentication.
<password>	Specifies a single password for authentication.
range *<begin> <end>*	Specifies the beginning and ending of the range of passwords for authentication.

The following example specifies the authorization name and password range for a group of SIP users:

(config-T01)#**register range 2565553000 2565553999 auth-name range Adtran3000 Adtran3999 password range adtn3000 adtn3999**

Step 6: Configure a SIP Trunk To the Allworx 6x

To create and configure a SIP trunk on the ADTRAN SBC to the Allworx 6x, follow these steps:

1. From the Global Configuration mode, use the **voice trunk** *<Txx>* **type sip** command to create a SIP trunk and enter the Voice SIP Trunk Configuration mode.

   (config)#**voice trunk** *<Txx>* **type sip**.

Syntax	Description
<Txx>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**)

   The following example creates SIP trunk **T02** and enters the Voice SIP Trunk Configuration mode:

   (config)#**voice trunk T02 type sip**
   (config-T02)#

2. From the Voice SIP Trunk Configuration mode, use the **sip-server primary** command to specify the FQDN or IP address of the SIP server to which the trunk will send call-related SIP messages. Record the IP address used, as it will be used later when configuring the Allworx 6x WAN IP address.

   (config-T02)#**sip-server primary** *<FQDN or IP address>* **[tcp [***<port>***] | udp [***<port>***]]**

Syntax	Description
<FQDN or IP address>	Specifies the FQDN or IP address of the SIP server. IP addresses should be expressed in dotted decimal notation (for example, **10.10.10.1**).
tcp	Optional. Sets the TCP port of the outbound proxy server.
udp	Optional. Sets the UDP port of the outbound proxy server.
<port>	Optional. Specifies the TCP or UDP port number. Range is **0** to **65535**.

   The following example sets the SIP server IP address to **192.168.1.1**:

   (config-T02)#**sip-server 192.168.1.1**

Step 7: Configure a Trunk Group for the Service Provider SIP Trunk

Trunk groups combine one or more trunk accounts and assign outbound call characteristics to the group to determine what types of numbers can and cannot be dialed on the trunk. Additionally, a cost can be assigned to each number template in the trunk group; in cases where a call is accepted by several trunks; the call will be routed to the trunk with the lowest cost. To create and configure a trunk group for the SIP trunk to the service provider, follow these steps:

1. From the Global Configuration mode, use the **voice grouped-trunk** *<name>* command to create a trunk group to the service provider and enter the Voice Trunk Group Configuration mode.

   (config)#**voice grouped-trunk ITSP**

Syntax	Description
<name>	Specifies the name of the trunk.

The following example creates a trunk group named **ITSP** and enters the Voice Trunk Group Configuration mode:

(config)#**voice grouped-trunk ITSP**
(config-ITSP)#

2. From the Voice Trunk Group Configuration mode, use the **trunk** <*Txx*> command to add the SIP trunk configured to the service provider to the trunk group.

(config-ITSP)#**trunk** <*Txx*>

Syntax	Description
<*Txx*>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**).

The following example adds trunk **T01** to the trunk group.

(config-ITSP)#**trunk T01**

3. Use the **accept** command to specify the numbers that can be dialed on the SIP trunk.

(config-ITSP)#**accept** <*template*> **cost** <*value*>

Syntax	Description
<*template*>	Specifies the patterns users can dial on the trunk. You can enter a complete phone number or wildcards can be used to help define accepted numbers.
cost <*value*>	Specifies the cost value for the trunk. This option is used if a call is accepted by several trunks. The call will be routed to the trunk with the lowest cost value. The valid range is **0** to **499**.

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

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$; XXXX$.

The following example allows users on the trunk to dial 10-digit numbers, 11-digit numbers, international numbers, 411, 611, and 911:

(config-ITSP)#**accept NXX-NXX-XXXX cost 0**
(config-ITSP)#**accept 1-NXX-NXX-XXXX cost 0**
(config-ITSP)#**accept 011-$ cost 0**
(config-ITSP)#**accept 411 cost 0**
(config-ITSP)#**accept 611 cost 0**
(config-ITSP)#**accept 911 cost 0**

4. Use the **reject** *<template>* command to specify the numbers that cannot be dialed on the trunk.

(config-ITSP)#**reject** *<template>*

Syntax	Description
<template>	Specifies the patterns users cannot dial on the trunk. You can enter a complete phone number or wildcards can be used to help define rejected numbers.

The following example blocks calls to any 900 number on the trunk group:

(config-ITSP)#**reject 900-NXX-XXXX**
(config-ITSP)#**reject 1-900-NXX-XXXX**

Step 8: Configure a Trunk Group to the Allworx 6x

To create and configure a trunk group for the SIP trunk to the Allworx 6x, follow these steps:

1. From the Global Configuration mode, use the **voice grouped-trunk** *<name>* command to create a trunk group to the Allworx 6x and enter the Voice Trunk Group Configuration mode.

(config)#**voice grouped-trunk Allworx**

Syntax	Description
<name>	Specifies the name of the trunk.

The following example creates a trunk group named **Allworx** and enters the Voice Trunk Group Configuration mode:

(config)#**voice grouped-trunk Allworx**
(config-Allworx)#

2. From the Voice Trunk Group Configuration mode, use the **trunk** <*Txx*> command to add the SIP trunk configured to the Allworx 6x to the trunk group.

(config-Allworx)#**trunk** <*Txx*>

Syntax	Description
<*Txx*>	Specifies the trunk identity in the format Txx, where xx is the trunk ID number between **01** and **99** (for example, **T01**).

The following example adds trunk **T02** to the trunk group.

(config-Allworx)#**trunk T02**

3. Use the **accept** command to specify the numbers that can be dialed on the SIP trunk. At a minimum, the DID numbers of the Allworx 6x auto attendant and phones should be added.

(config-Allworx)#**accept** <*template*> **cost** <*value*>

Syntax	Description
<*template*>	Specifies the patterns users can dial on the trunk. You can enter a complete phone number or wildcards can be used to help define accepted numbers.
cost <*value*>	Specifies the cost value for the trunk. This option is used if a call is accepted by several trunks. The call will be routed to the trunk with the lowest cost value. The valid range is **0** to **499**.

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

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$; XXXX$.

The following example allows inbound calls to phone numbers in the range **256-555-3000** to **256-555-3999**:

(config-Allworx)#**accept 256-555-3XXX**

Step 9: Enable Media Anchoring and the RTP Symmetric Filter

Media anchoring forces all RTP traffic to pass through through the ADTRAN SBC. The RTP symmetric filter works in conjunction with media anchoring to filter non-symmetric RTP packets.

1. To enable media anchoring, use the following command from the Global Configuration mode:

(config)#**ip rtp media-anchoring [qos dscp** <*value*> **| session timeout** <*value*>**]**

Syntax	Description
qos dscp <*value*>	Optional. Specifies the differentiated services code point (DSCP) value for anchored RTP traffic. Range is **0** to **63**.
session timeout <*value*>	Optional. Specifies the timeout period, in seconds, of an anchoring association after the associated RTP packet flow ends. Range is **32** to **900** seconds. This value should not be changed without the guidance from ADTRAN technical support.

The following example enables media anchoring on the ADTRAN SBC:

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

2. To enable the RTP symmetric filter, use the **ip rtp symmetric-filter** command from the Global Configuration mode.

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

Step 10: Disable Prefer Double reINVITEs

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 when it is not necessary, double reINVITEs should be disabled using the following command from the Global Configuration mode:

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

Step 11: Configure the Forward and Transfer Modes

To ensure that call forwards and transfers are handled locally by the ADTRAN SBC, the forward and transfer modes should be set to local.

1. Use the **voice forward-mode local** command from the Global Configuration mode to set the forward mode to local.

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

2. Use the **voice transfer-mode local** command from the Global Configuration mode to set the transfer mode to local.

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

Sample Configuration

The following is the configuration used during interoperability testing. It is a sample configuration only and should be modified to suit your specific integration.

```
!
interface eth 0/1
  description ITSP
  ip address  198.51.100.1 /30
  media-gateway ip primary
  no shutdown
!
interface eth 0/2
  description ITSP
  ip address  192.168.1.254 /24
  media-gateway ip primary
  no shutdown
!
ip route 0.0.0.0 0.0.0.0 198.51.100.2
!
voice transfer-mode local
voice forward-mode local
!
voice trunk T01 type sip
  description "ITSP SIP Server"
  sip-server primary 203.0.113.1
  dial-string source to
  register 2565553000
!
voice trunk T02 type sip
  description "Allworx PBX"
  sip-server primary 192.168.1.1
!
voice grouped-trunk ITSP
  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 T02
  accept 256-555-3XXX cost 0
!
no sip prefer double-reinvite
!
ip rtp media-anchoring
ip rtp symetric-filter
!
```

Configuring the Allworx 6x

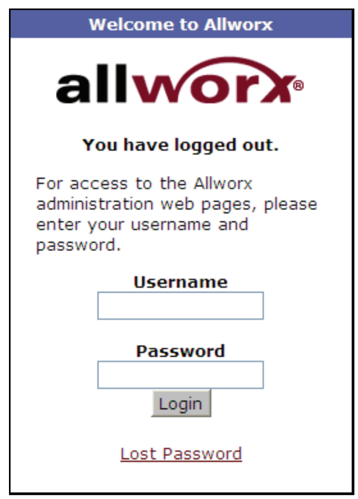
> *NOTE: The following sections provide general instructions for configuring the Allworx 6x for integration with an ADTRAN SBC. Refer to the Allworx Server Administrator's Guide for detailed instructions on how to configure the Allworx 6x.*

To configure the Allworx 6x for integration with the ADTRAN SBC, follow these steps:

- *Step 1: Log in to the Allworx 6x*
- *Step 2: Configure the Network Settings on page 18*
- *Step 3: Create a Default Route to the ADTRAN SBC on page 19*
- *Step 4: Create a DID Block for Telephone Numbers on page 19*
- *Step 5: Create a Call Routing Plan for the DID Block on page 20*
- *Step 6: Create a SIP Trunk to the ADTRAN SBC on page 21*
- *Step 7: Add Phones to the Allworx 6x on page 23*
- *Step 8: Create Users and Extensions on page 25*

Step 1: Log in to the Allworx 6x

The Allworx 6x System Administration is accessed through a web browser. To log in to the Allworx 6x, follow these steps:



1. Open a new web page in your Internet browser.
2. To log in to the administrator GUI, enter the following in the Internet brower's address field:

   http://<*Allworx IP Address*>:8080

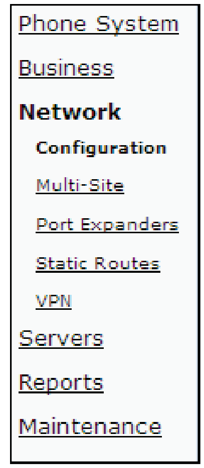
   The <*Allworx IP Address*> is the LAN IP address of the Allworx 6x. The default LAN IP address for the Allworx 6x is **192.168.2.254**. The following example accesses the administrator GUI login for an Allworx 6x with a default LAN IP address:

   http://192.168.2.254:8080
3. Enter the username for the Allworx 6x in the **Username** field, and enter the password in the **Password** field. The default username is **admin**, and the default password is **admin**. Then, select **Login**.

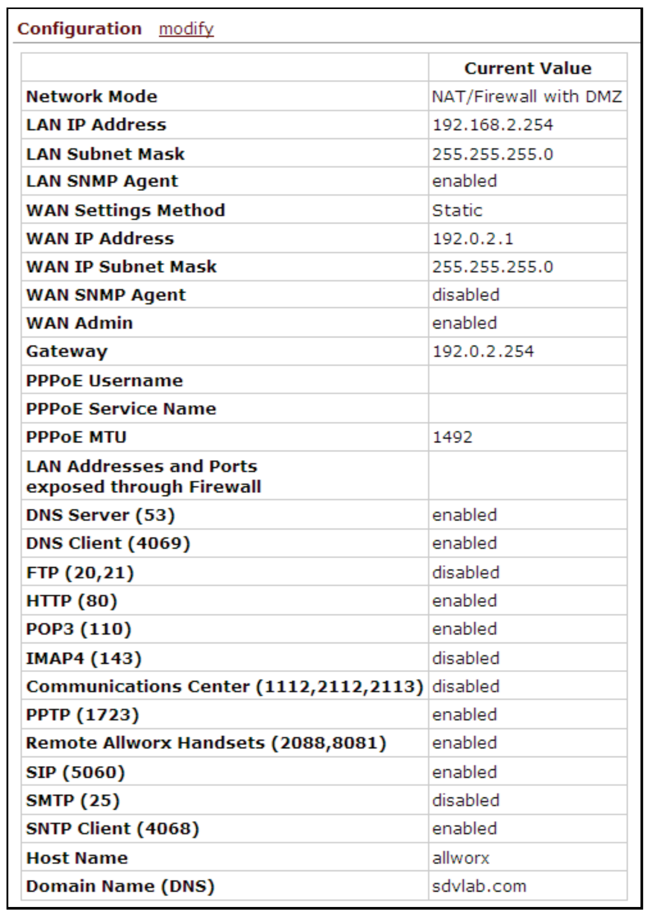
Step 2: Configure the Network Settings

To configure the network settings on the Allworx 6x required to integrate with the ADTRAN SBC, follow these steps:

1. Use the menu bar on the left side of the browser to navigate to **Network** > **Configuration**. The **Configuration** menu appears.



2. On the **Configuration** menu, select the **Modify** link to modify the network configuration for the Allworx 6x.
3. In the **Modify** menu, complete the following:
   a. Set the Network Mode to **NAT/Firewall with DMZ.**
   b. If desired, change the **LAN IP Address** from the default **192.168.2.254**.
   c. Configure the **WAN IP Address** with the IP address for the primary SIP server configured on the ADTRAN SBC SIP trunk to the Allworx 6x. This is the IP address configured in *Step 6: Configure a SIP Trunk To the Allworx 6x on page 9*
   d. Configure the **Gateway** IP address with the LAN IP address of the ADTRAN SBC. This is the IP address configured in *Step 3: Configure the LAN Interface to the Allworx 6x on page 7*.
   e. Configure the firewall ports, DHCP Server, and NTP Server as necessary for your configuration.



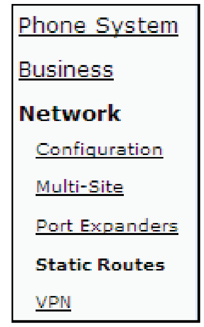
Configuration modify

	Current Value
Network Mode	NAT/Firewall with DMZ
LAN IP Address	192.168.2.254
LAN Subnet Mask	255.255.255.0
LAN SNMP Agent	enabled
WAN Settings Method	Static
WAN IP Address	192.0.2.1
WAN IP Subnet Mask	255.255.255.0
WAN SNMP Agent	disabled
WAN Admin	enabled
Gateway	192.0.2.254
PPPoE Username	
PPPoE Service Name	
PPPoE MTU	1492
LAN Addresses and Ports exposed through Firewall	
DNS Server (53)	enabled
DNS Client (4069)	enabled
FTP (20,21)	disabled
HTTP (80)	enabled
POP3 (110)	enabled
IMAP4 (143)	disabled
Communications Center (1112,2112,2113)	disabled
PPTP (1723)	enabled
Remote Allworx Handsets (2088,8081)	enabled
SIP (5060)	enabled
SMTP (25)	disabled
SNTP Client (4068)	enabled
Host Name	allworx
Domain Name (DNS)	sdvlab.com

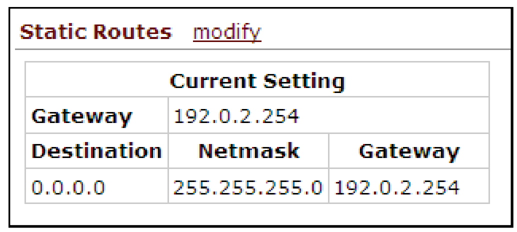
Step 3: Create a Default Route to the ADTRAN SBC

Static routes create a fixed routing path from the Allworx 6x to a specific network destination. To create a static route to the ADTRAN SBC, follow these steps:

1. Use the menu bar on the left side of the browser to navigate to **Network > Static Routes**. The **Static Routes** menu will appear.



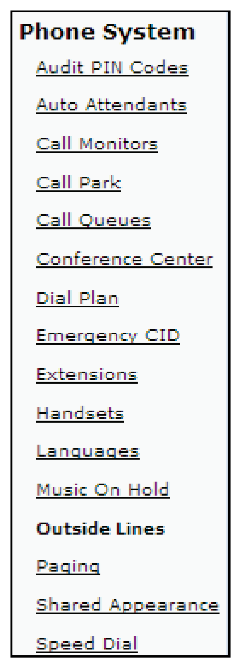
2. In the **Static Routes** menu, select the **modify** link to configure a static route.
3. In the **Destination** field, enter 0.0.0.0.
4. In the **Gateway** field, enter the LAN IP address of the ADTRAN SBC.
5. Select **Update** to add the static route.



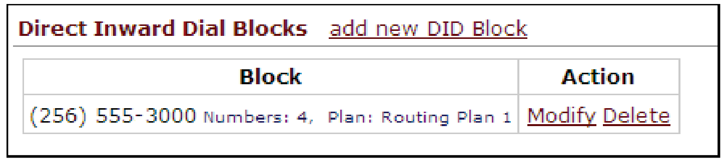
Step 4: Create a DID Block for Telephone Numbers

DID telephone numbers assigned by the service provider allow extensions on a PBX to be reached externally by a standard telephone number. On the Allworx 6x, DID numbers are created in blocks. To create a DID block, follow these steps:

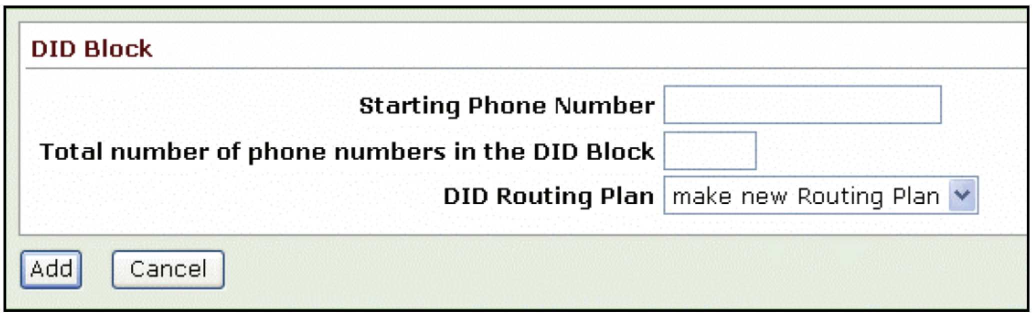
1. Use the menu bar on the left side of the browser to navigate to **Phone System > Outside Lines**. The **Outside Lines** menu will appear.



2. In the **Outside Lines** menu, scroll down to the **Direct Inward Dial Blocks** section, and select the **add new DID Block** link. The **DID Block** menu will appear.



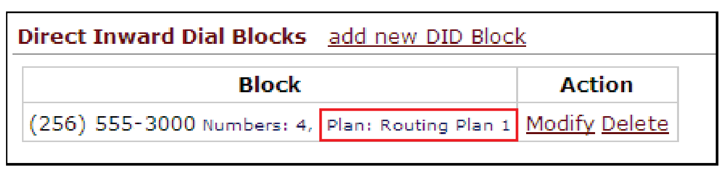
3. In the **DID Block** menu, enter the phone number that will begin the DID block in the **Starting Phone Number** field. Use the provided field to enter the **Total number of phone numbers in the DID Block**. Use the **DID Routing Plan** drop-down menu to select **make new Routing Plan**. Then, select **Add**.



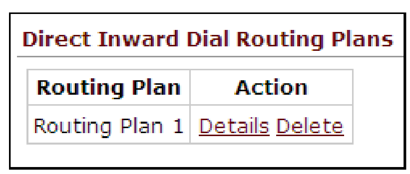
Step 5: Create a Call Routing Plan for the DID Block

On the Allworx 6x, call routing plans map a DID number to an extension number on the Allworx server. Additionally, a Dialed Number Identification Service (DNIS) name can be assigned for each DID number. The DNIS name is displayed on a recipient's phone during a call between two Allworx users. To create a call routing plan for the DID block created in *Step 4: Create a DID Block for Telephone Numbers on page 19*, follow these steps:

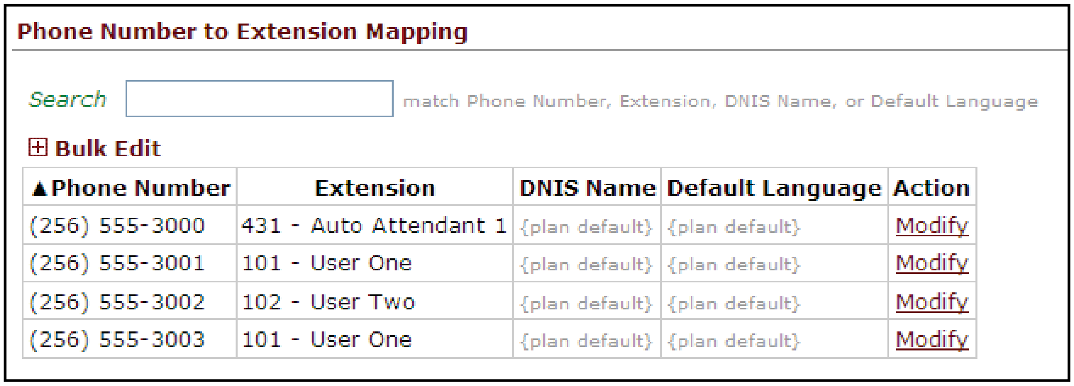
1. Use the menu bar on the left side of the browser to navigate to **Phone System > Outside Lines**. The **Outside Lines** menu will appear.
2. Scroll down to the **Direct Inward Dial Blocks** section to find the name of the routing plan associated with the DID block you created in *Step 4: Create a DID Block for Telephone Numbers on page 19*.



3. In the **Outside Lines** menu, scroll down to the **Direct Inward Dial Routing Plans** section, and select the **Details** link in the **Action** column next to the routing plan associated with the DID block. The **DID Routing Plan** menu will appear.



4. In the **DID Routing Plan** menu, use the **Modify** links in the **Phone Number to Extension Mapping** section to configure the **Extension**, **DNIS Name**, and **Default Language** assigned to each DID phone number.



Phone Number to Extension Mapping

Search match Phone Number, Extension, DNIS Name, or Default Language

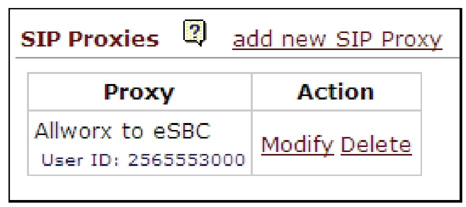
⊞ Bulk Edit

▲Phone Number	Extension	DNIS Name	Default Language	Action
(256) 555-3000	431 - Auto Attendant 1	{plan default}	{plan default}	Modify
(256) 555-3001	101 - User One	{plan default}	{plan default}	Modify
(256) 555-3002	102 - User Two	{plan default}	{plan default}	Modify
(256) 555-3003	101 - User One	{plan default}	{plan default}	Modify

Step 6: Create a SIP Trunk to the ADTRAN SBC

To create a SIP trunk to the ADTRAN SBC, follow these steps:

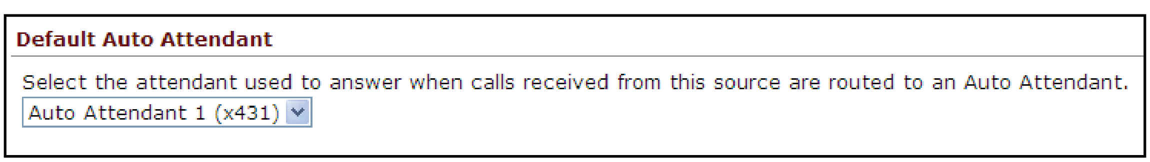
1. Use the menu bar on the left side of the browser to navigate to **Phone System > Outside Lines**. The **Outside Lines** menu will appear.
2. In the **Outside Lines** menu, scroll down to the **SIP Proxies** section, and select the **add new SIP Proxy** link. The **SIP Proxy** menu will appear.



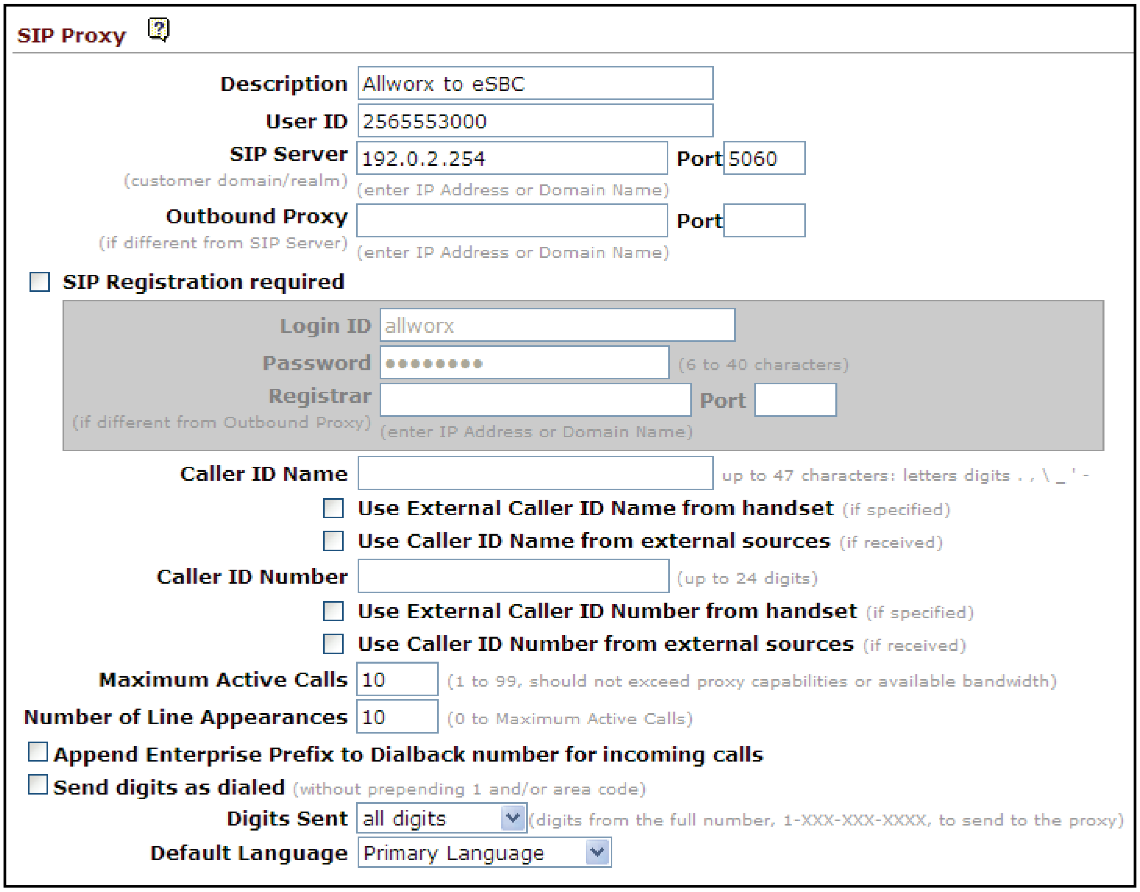
SIP Proxies add new SIP Proxy

Proxy	Action
Allworx to eSBC User ID: 2565553000	Modify Delete

3. In the **SIP Proxy** section of the **SIP Proxy** menu, complete the following:
   a. In the **Description** field, enter a description for the SIP proxy (for example, **Allworx to eSBC**).
   b. In the **SIP Server** field, enter the LAN IP address of the ADTRAN SBC configured in *Step 3: Configure the LAN Interface to the Allworx 6x on page 7*.
   c. In the **Port** field next to the **SIP Server** field, enter **5060**.
   d. Ensure the **Outbound Proxy** field is blank.
   e. Ensure that **SIP Registration required** is not checked.
   f. Use the **Digits Sent** drop-down menu to select **all digits**.
4. In the **Default Auto Attendant** section, use the drop-down menu to select the Allworx auto attendant that should be used when inbound calls are routed to an auto attendant.



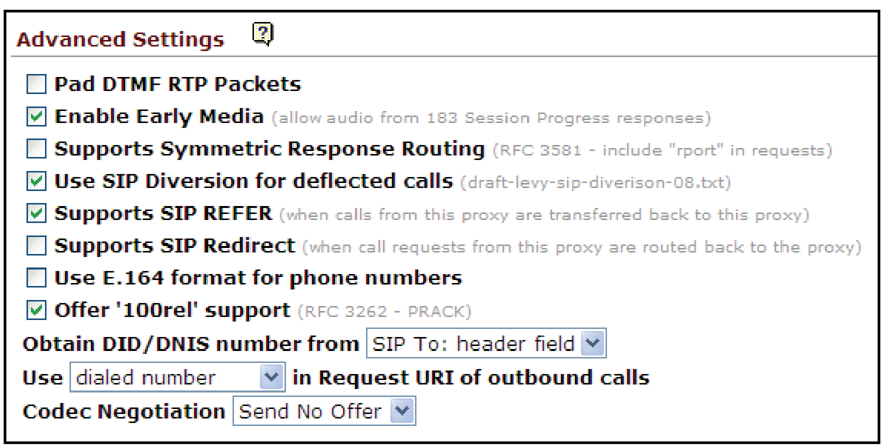
5. In the **Advanced Settings** section, complete the following:
   a. Ensure that **Enable Early Media** is enabled.
   b. Ensure that **Use SIP Diversion for deflected calls** is enabled.
   c. Ensure that **Supports SIP REFER** is enabled.



d. Ensure that **Offer '100rel' support** is enabled.

e. Use the **Obtain DID/DNIS number from** drop-down menu to select **SIP To: header field**.

f. Use the drop-down menu to select that the **dialed number** is used in the **Request URI of outbound calls**.



6. In the **Call Route** section, select the **Routed using DID block** radio button and select the DID block configured in *Step 4: Create a DID Block for Telephone Numbers on page 19*.

7. Select **Add** to create the SIP proxy.

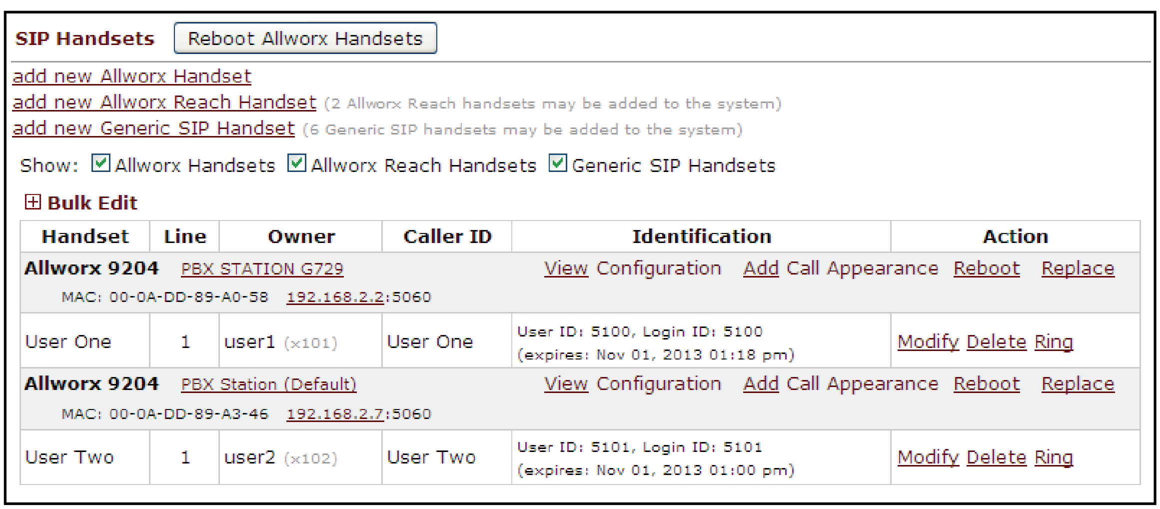
Step 7: Add Phones to the Allworx 6x

Allworx brand phones can be installed on the Allworx 6x using plug-and-play installation. If the Allworx 6x is connected to the network, Allworx brand phones will automatically register with the server when they restart. Allworx brand phones and phones from other manufacturers can also be manually configured.

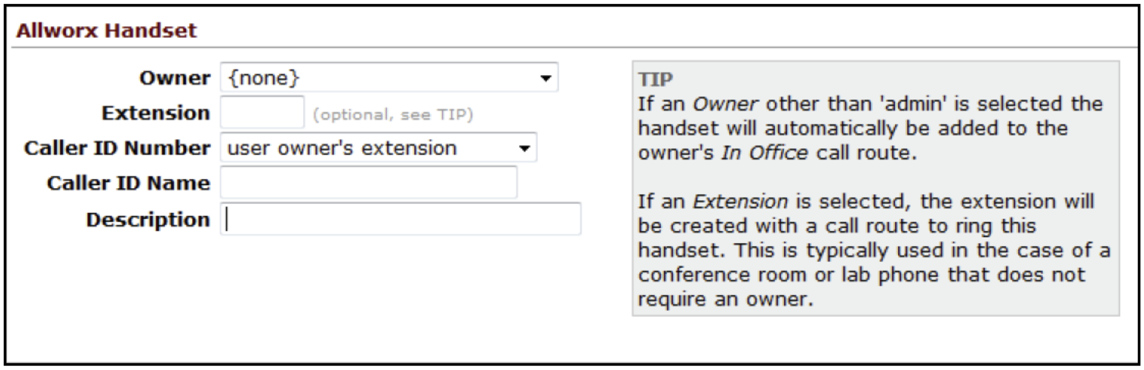
Manually Adding an Allworx Brand Phone

To manually configure Allworx brand phones, follow these steps:

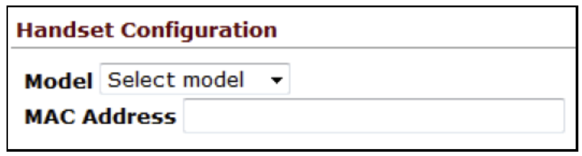
1. Use the menu bar on the left side of the browser to navigate to **Phone System > Handsets**. The **Handsets** menu will appear.
2. In the **SIP Handsets** section of the **Handsets** menu, select the **add new Allworx Handset** link. The **SIP Handset** menu will appear.



3. In the **Allworx Handset** section of the **SIP Handset** menu, enter a description for the phone in the **Description** field.



4. In the **Handset Configuration** section, complete the following:
   a. Use the **Model** drop-down menu to select an Allworx phone model.
   b. Enter the phone's media access control (MAC) address in the **MAC Address** field.

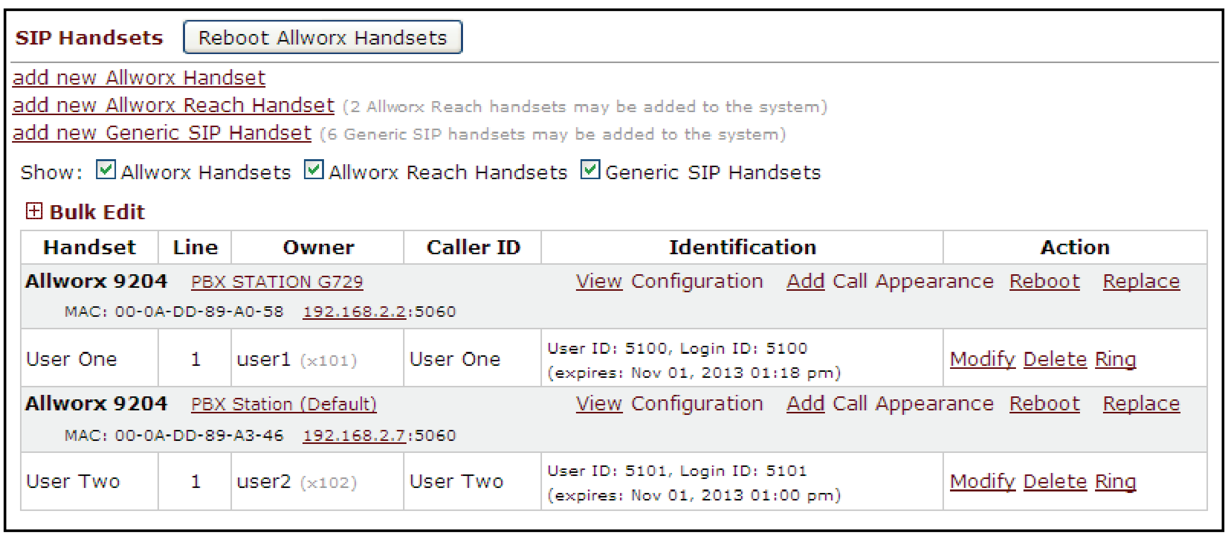


5. Select the **Add** button to add the phone.

Manually Adding a Third-Party SIP Phone

To manually configure third-party SIP phones, follow these steps:

1. Use the menu bar on the left side of the browser to navigate to **Phone System > Handsets**. The **Handsets** menu will appear.
2. In the **SIP Handsets** section of the **Handsets** menu, select the **add new Generic SIP Handset** link. The **SIP Handset** menu will appear.



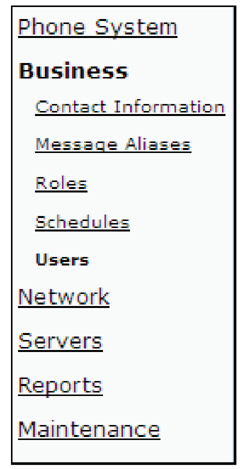
3. In the **SIP Handset** section of the **SIP Handset** menu, enter a description for the phone in the **Description** field.

4. In the **Handset Configuration** section, complete the following:
   a. Enter the number of lines available on the phone in the **Number of Lines** field.
   b. Enter a **Login ID** and **Password** in the provided fields for authenticating the phone with the server.
5. Select the **Add** button to add the phone.

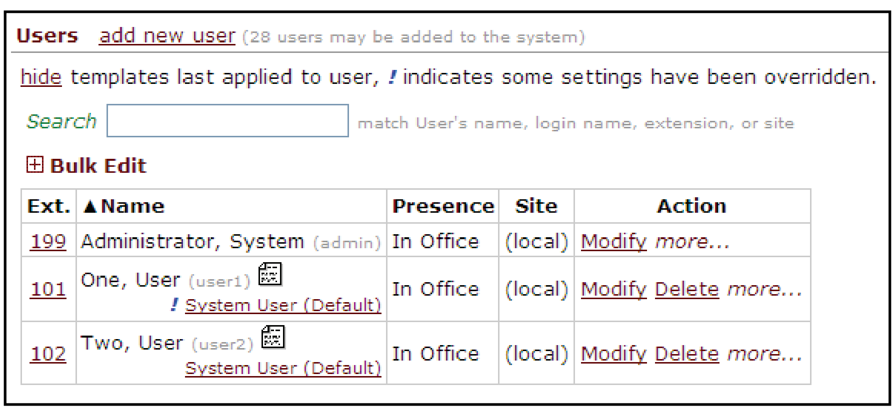
Step 8: Create Users and Extensions

Allworx system users are created and assigned to phones and extensions in the **Users** menu. To create a new user, follow these steps:

1. Use the menu bar on the left side of the browser to navigate to **Business > Users**. The **Users** menu appears.



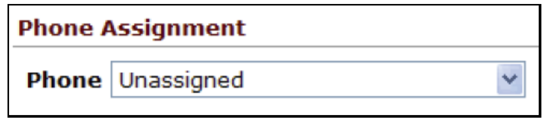
2. In the **Users** menu, select the **add new user** link. The **Add New User** menu will appear.



3. In the **Identification** section of the **Add New User** menu, complete the following:
   a. Enter the login name for the user in the **Login Name** field. This is the username used by the user to log in to the **My Allworx Manager**.
   b. Enter the first, middle, and last names (in that order) of the user in the **Full Name** fields.
   c. Enter a password for the user in the **Password** field. This is the password used by the user to log in to the **My Allworx Manager**.

d. Enter an extension for the user in the **Primary Extension** field. This extension is assigned to the user's phone.

4. In the **Phone Assignment** section, use the **Phone** drop-down menu to assign the user a phone configured on the Allworx 6x.



5. In the **User Template** section, use the drop-down menu to assign a user template to the user.