InnoMedia ESBC SIP Trunking Configuration Guide for Allworx 6x SIP PBX

The purpose of this application note is to describe the steps needed to configure the Allworx 6x SIP PBX for proper operation in a SIP Trunking Application with the InnoMedia ESBC.

Related Documents

Name of Document	Location
ESBC Administrator's Guide	http://www.innomedia.com/support_manuals.shtml
Allworx	http://www.allworx.com/phone_systems/6x.aspx

www.innomedia.com

Document version: 1.5 (Oct, 2012)

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1. Introduction

This application note provides configuration steps for Service Providers to deliver SIP Trunking Services to enterprise customers deploying Allworx 6x SIP PBXs using InnoMedia's Enterprise SIP Gateway (ESG) ESBC9x80-4B. The ESBC is owned and managed by the service provider, and performs necessary SIP normalization, NAT and firewall traversal, security operations, performance reporting, and remote diagnosis.

Delivering Business Voice Services to Enterprises

Figure 1 illustrates how business voice services can be delivered to enterprise customers having legacy TDM PBXs, IPPBXs, or through hosted voice services.

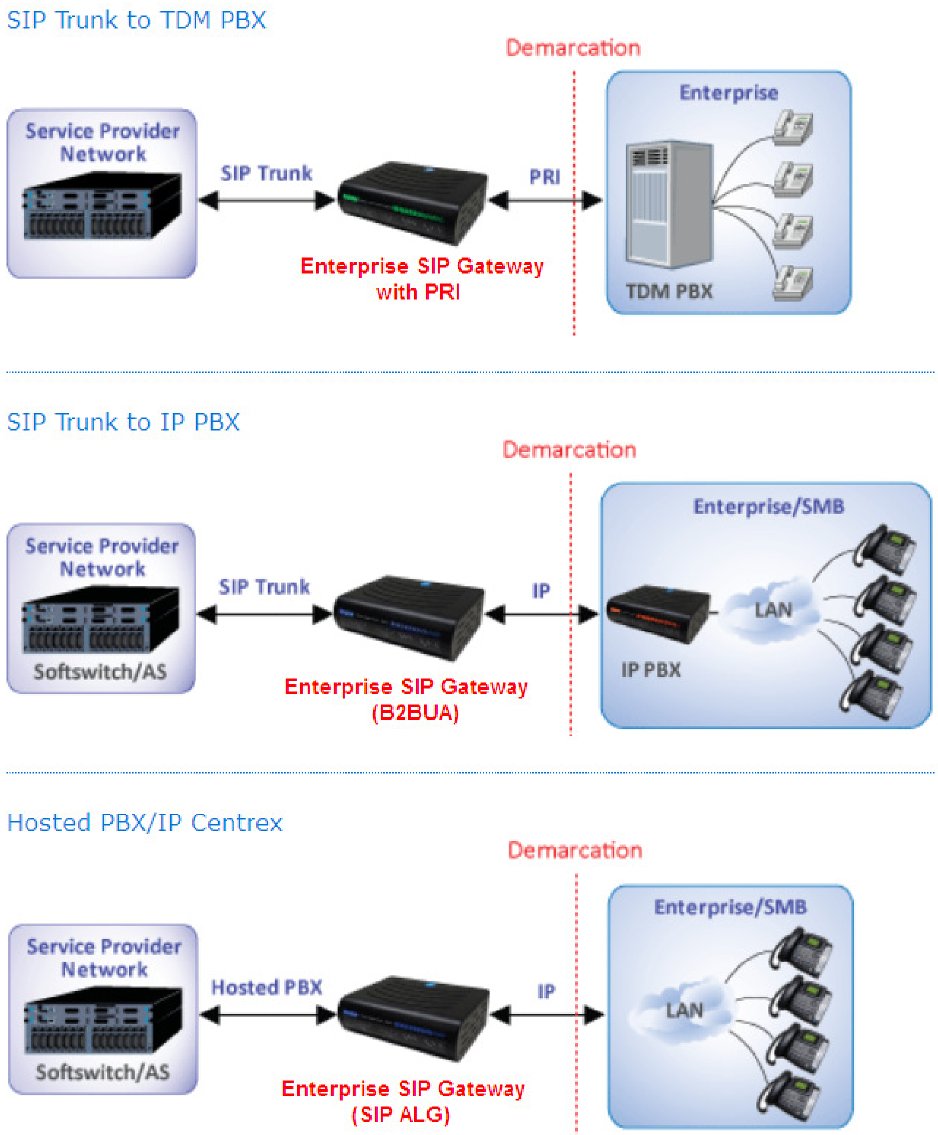


Figure 1. Reference Network Architecture

Getting Started

The following pieces of information are required.

- SIP Proxy server IP address or FQDN (and/or SIP Domain information)
- SIP Trunk service user account(s), inward dialing (DID) numbers, and/or a main pilot number.
- SIP authentication credential with the service provider network.

2. Interoperability Compliance Testing

The SIP Trunking ESG – PBX interoperability lab setup is shown in Figure 2 where an Allworx 6x is connected to an ESBC which in turn is registered to a softswitch/SIP Server. To verify ESBC - PBX interoperability, the following features and functionality were exercised during the interoperability compliance tests.

Features Supported

- Inbound calls to Allworx IP phones, and FAX machines connected to Allworx 6x FXS ports.
- Outbound calls from Allworx IP Phones and Fax machines connected to Allworx 6x FXS ports.
- Various Call features:
    - Direct Inward Dialing
    - Blind and Consulted Call Transfer
    - Call Forwarding
    - Caller ID/Caller Name presentation and Caller ID restriction
    - Three way conference call
    - User features such as hold, resume
- DTMF transmission using in-band and RFC2833 tones
- Codec: G.711 mu-law
- FAX: G.711 pass-through

Not Supported Features and Limitations

- T.38 FAX relay
- GIN Registration

Equipment and Software Validated of this Application Note

The lab network for the SIP Trunk reference configuration is illustrated below.

The lab network consists of the following components:

ESBC

Model	: InnoMedia ESBC 9380-4B
Firmware Version	: esbc-9x-2-0-12-36

PBX

Model	: Allworx 6x
Firmware version	: V7.3.9.5

PBX LAN Phone

Model	: Allworx 9212Phones on the local LAN

Firmware version : 2.4.3.4 DSP 2026

WAN Phone

Model : InnoMedia 6308-SL2

Firmware version

Model : InnoMedia 7308-SLY65P

Firmware

SIP Server

Broadsoft Broadworks

Version : R18

LAB Configuration Example

Table 1 – IP Address

Component	IOT LAB Value	Your Value
InnoMedia ESBC		
LAN IP Address	172.16.251.1	
LAN Subnet Mask	255.255.0.0	
WAN IP Address	DHCP	
Allworx 6		
WAN IP Address	172.16.251.5	
Voice VLAN IP Address	192.168.2.1	
Data VLAN IP Address		

DID Account Number	Account Attribute	Extension
2418884820	Pilot Number	4820
2418884826	DID	4826
2418884827	DID	4827

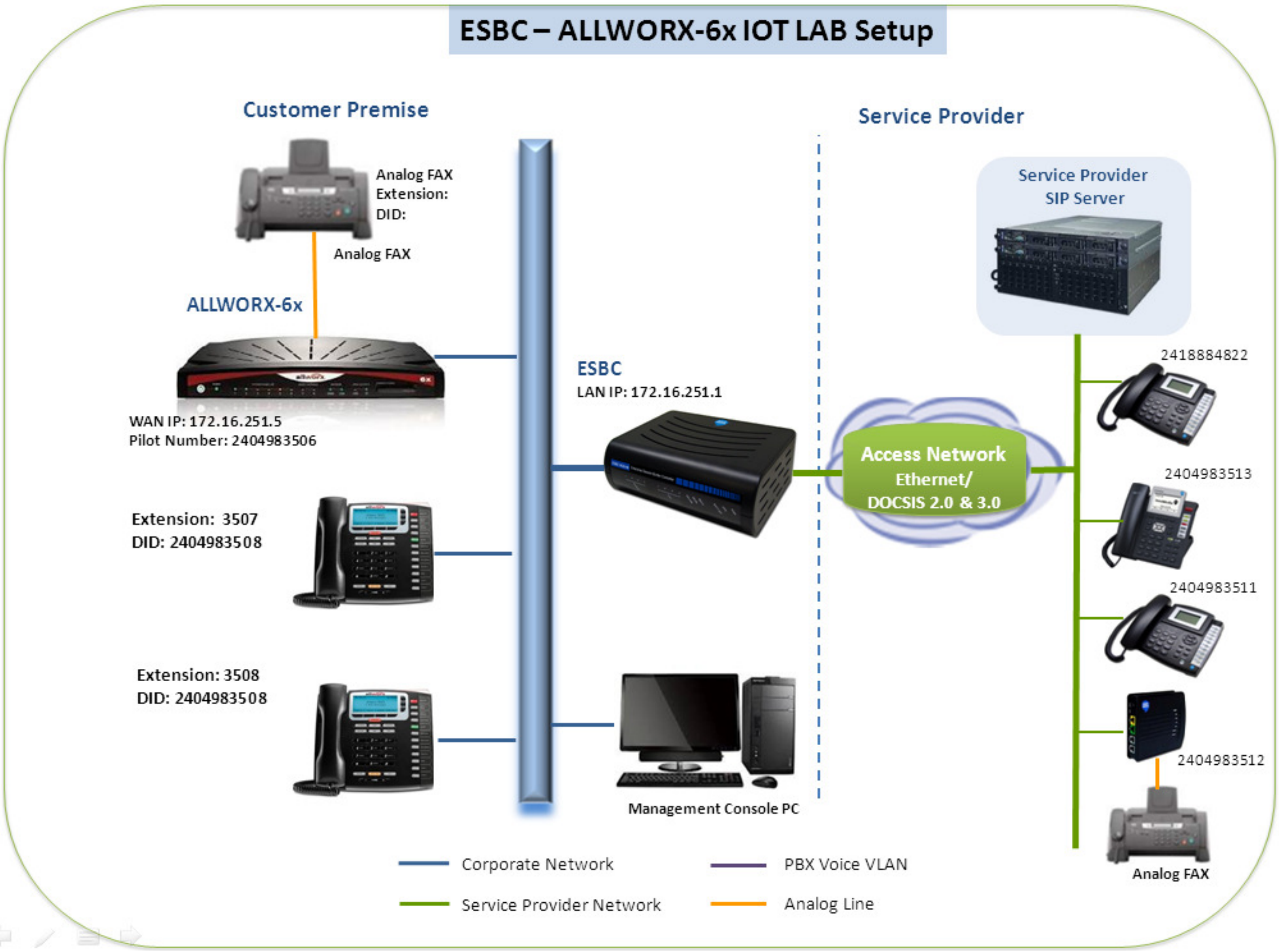


Figure 2. SIP Trunk LAB Reference Network

The lab network consists of the following components:

- Device Under IOT Test:
  - PBX: Allworx 6x PBX
  - Enterprise SIP Gateway: InnoMedia ESBC9380-4B
- Two Allworx 9212 phones on the local LAN registered to the Allworx 6x and one fax machine connected to an FXS port of Allworx 6x.
- Broadsoft Broadworks R18 as Service Provider Core Network Softswitch
- Three IP Phones and an ATA registered to Broadworks R18. A fax machine is connected to the ATA for fax testing.
- DHCP Server and SIP trunk termination.

3. Allworx 6x Configurations

The instructions provided in this section are intended to help Field Install Technicians configure the Allworx 6x to connect to the InnoMedia ESBC9X80-4B. It is not intended for advanced functionality setups. It is further assumed that the Field Install Technicians already have knowledge of the Allworx 6x.

Direct Inward Dialing Blocks –

1. Enter your Starting Phone Number in your Dialing Block
2. Enter Total number of Phone Numbers in the DID Block
3. Choose the DID Routing Plan to use for this Block of numbers, or if none, then “make new Routing Plan
4. Click Update for changes to take effect and a new Routing Plan to be created.

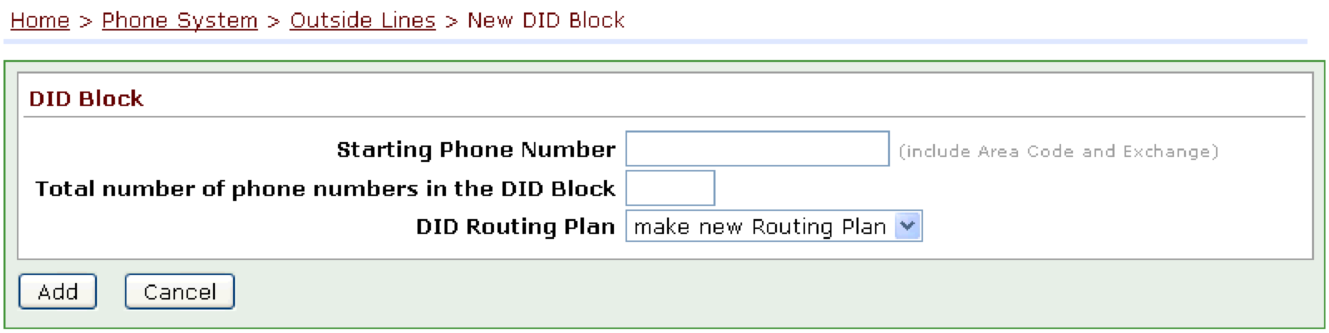


Figure 3

Direct Inward Dial Routing Plans –

1. Navigate to Home > Phone System > Outside lines > DID Routing Plan
2. Add/Modify the Routing Plan Information to send numbers not mapped to an Extension to be sent to the Operator/ or desired extension of your choice

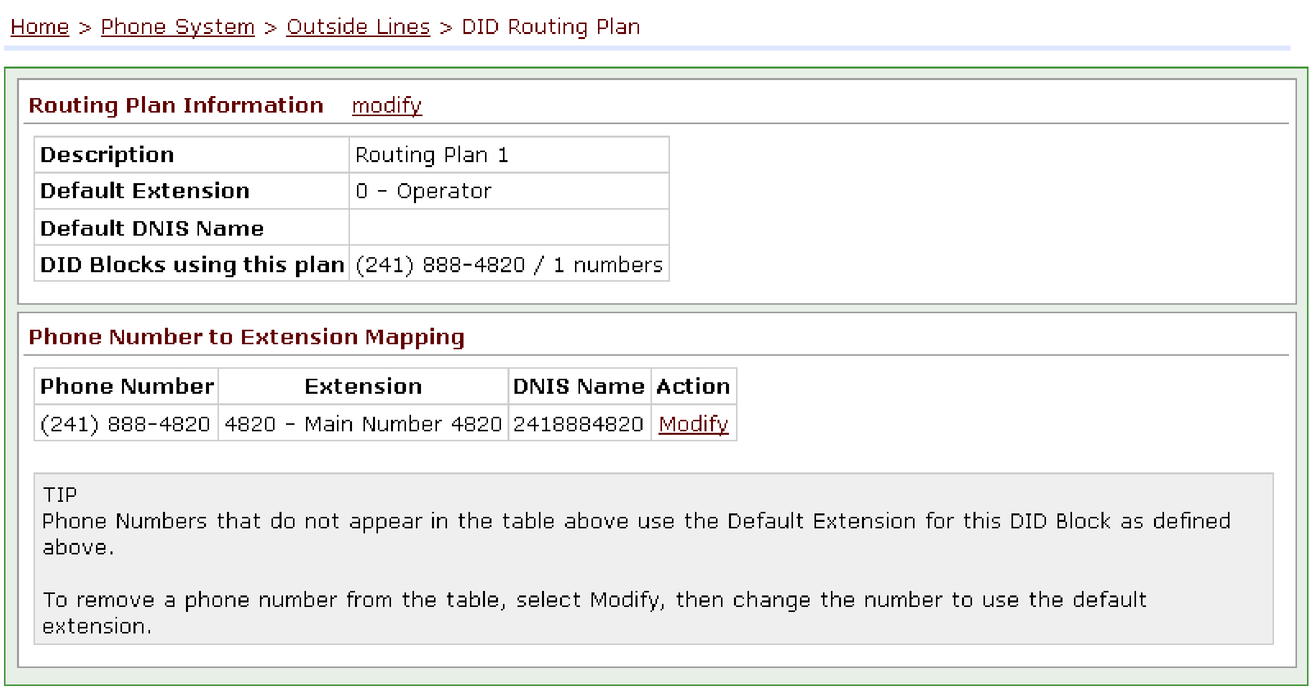


Figure 4

3. To send a DID to a specific Extension, click “add number to table” in the Phone Number to Extension Mapping box. If all DID’s have been assigned an extension, you will no longer have the option to click on “add number to table”

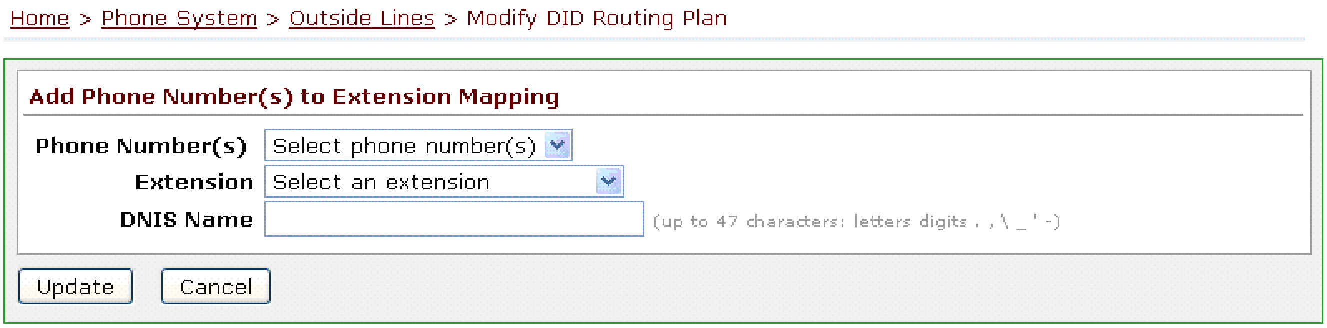


Figure 5

4. Select phone number
5. Select the extension you want the phone number to be sent to.
6. Enter the Number or Name you want to associate to this number.

SIP Trunk Setting

Figure 6

1. Navigate to Phone System > Outside Lines > SIP Gateway
2. Click “add new SIP Gateway”
3. Enter something for the Description
4. Enter a value for the “Number of Line Appearances” - If you have a total of 5 SIP lines, then you should not enter a number higher than 5.

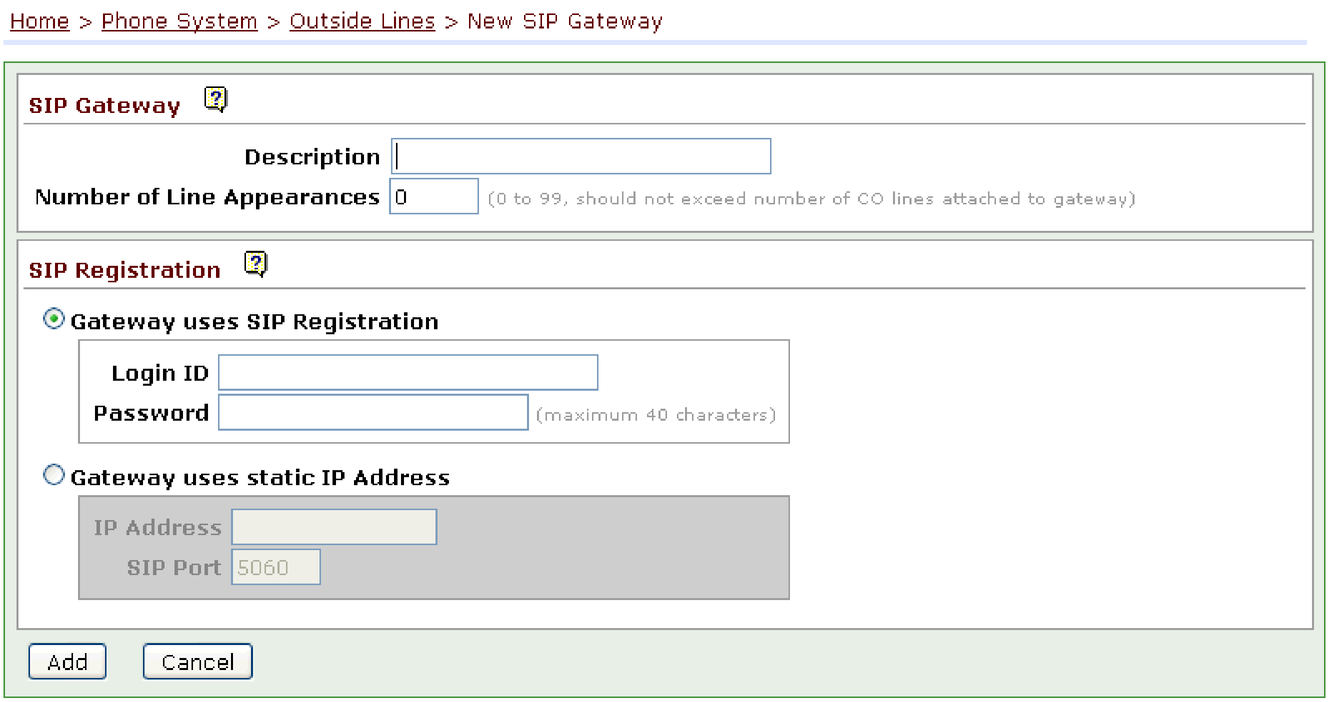


Figure 7

5. Choose “Gateway uses static IP Address” , and then enter the IP Address of the ESBC LAN port
6. Click “Add” to create the new Gateway
7. Now Modify the gateway

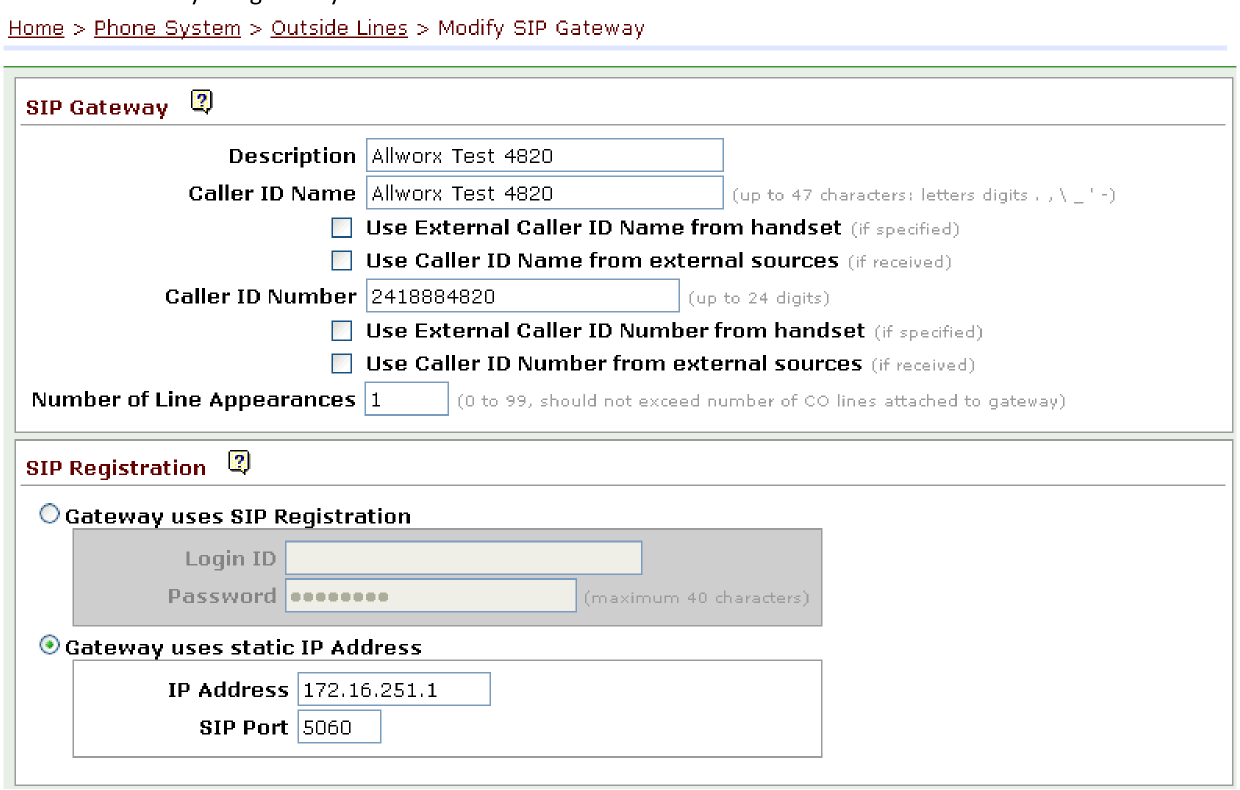


Figure 8

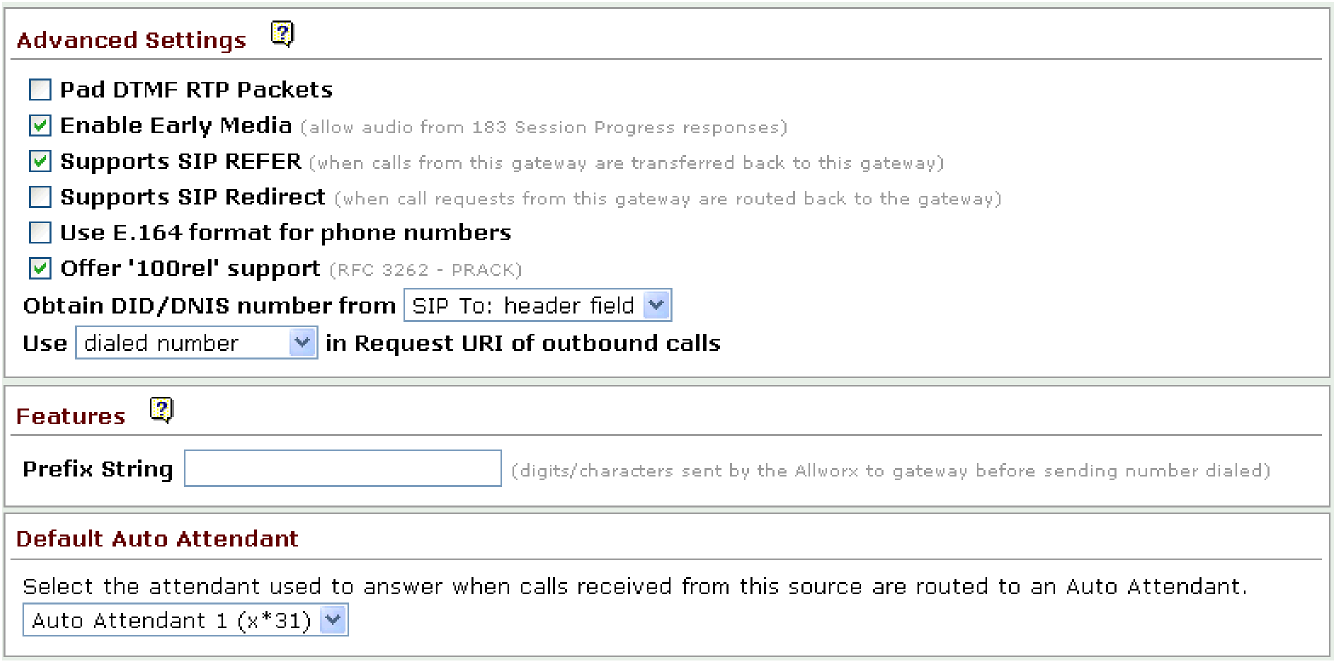


Figure 9

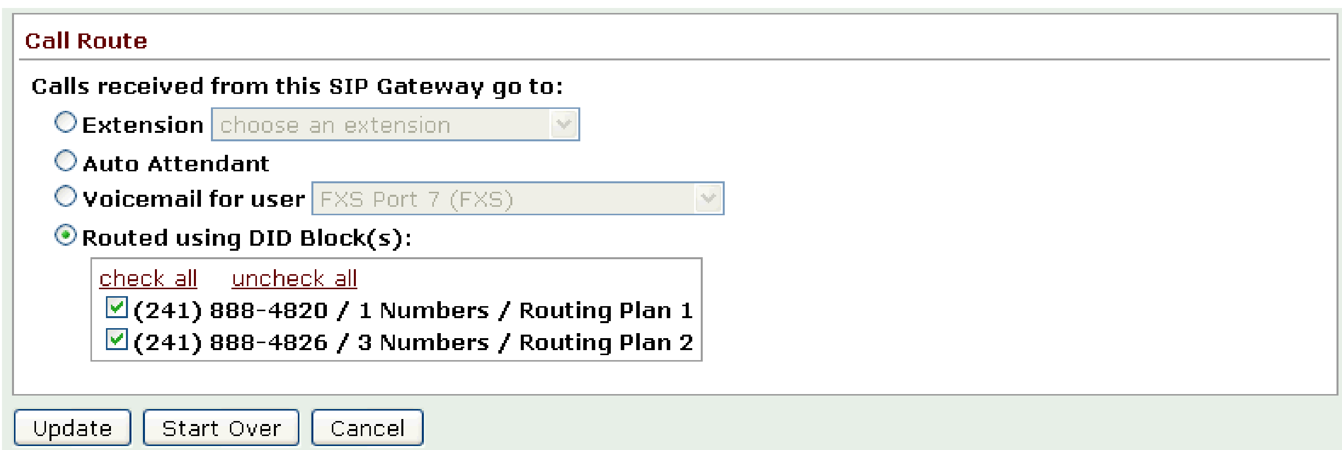


Figure 10

8. Enter your Desired Caller ID Name – something like your company name for example.
9. Enter your Caller ID Number – MAIN Pilot Number.
10. Ensure “Gateway uses Static IP Address” is clicked and the correct IP is entered
11. Setup the Call Route – Choose “Routed using DID Block(s)” - choose the Routing plan(s) you want it to use.

SIP Proxy

1. For each DID, you will need to create a SIP Proxy
2. Navigate to Phone System > Outside Lines > SIP Proxies

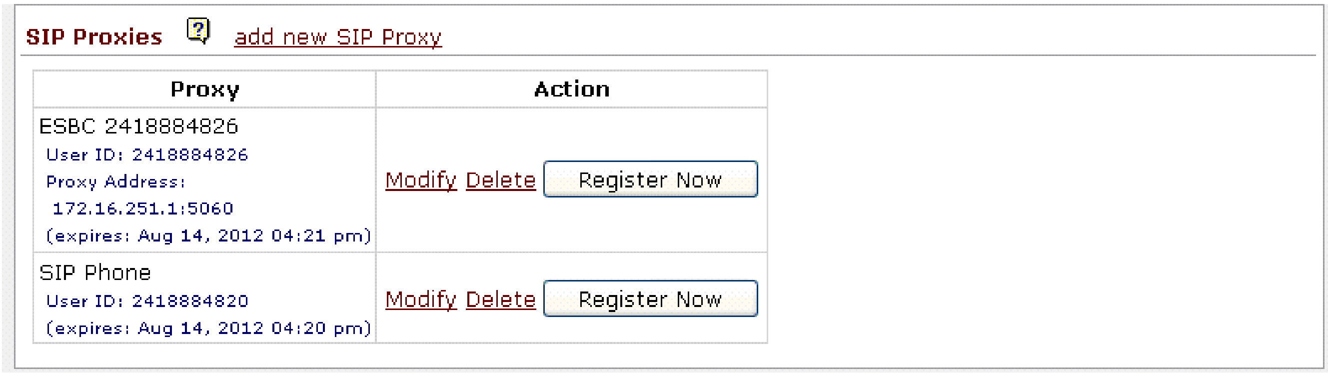


Figure 11. Allworx 6X SIP Trunk Settings

3. Click “add new SIP Proxy” to add a new proxy for each DID number

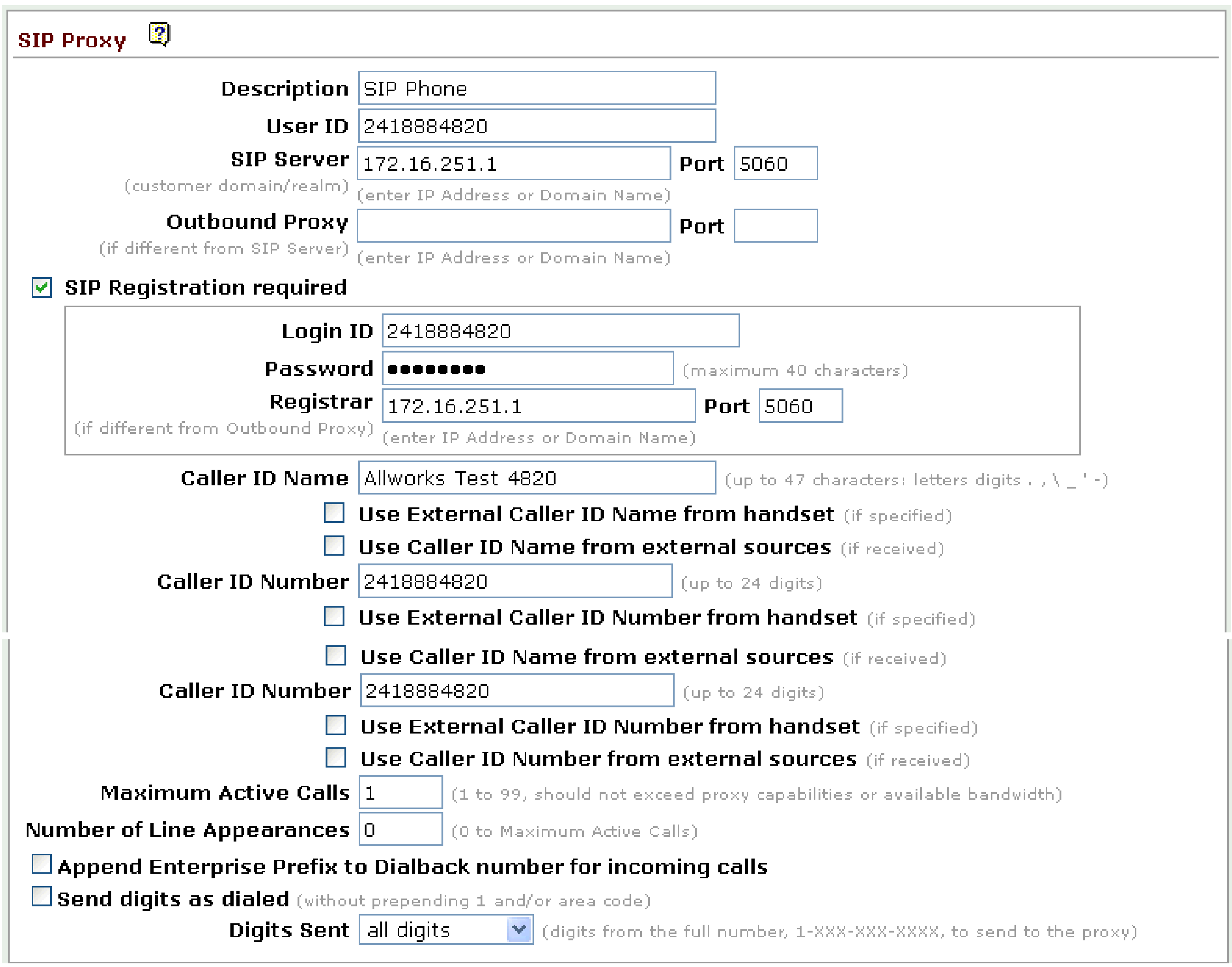


Figure 12

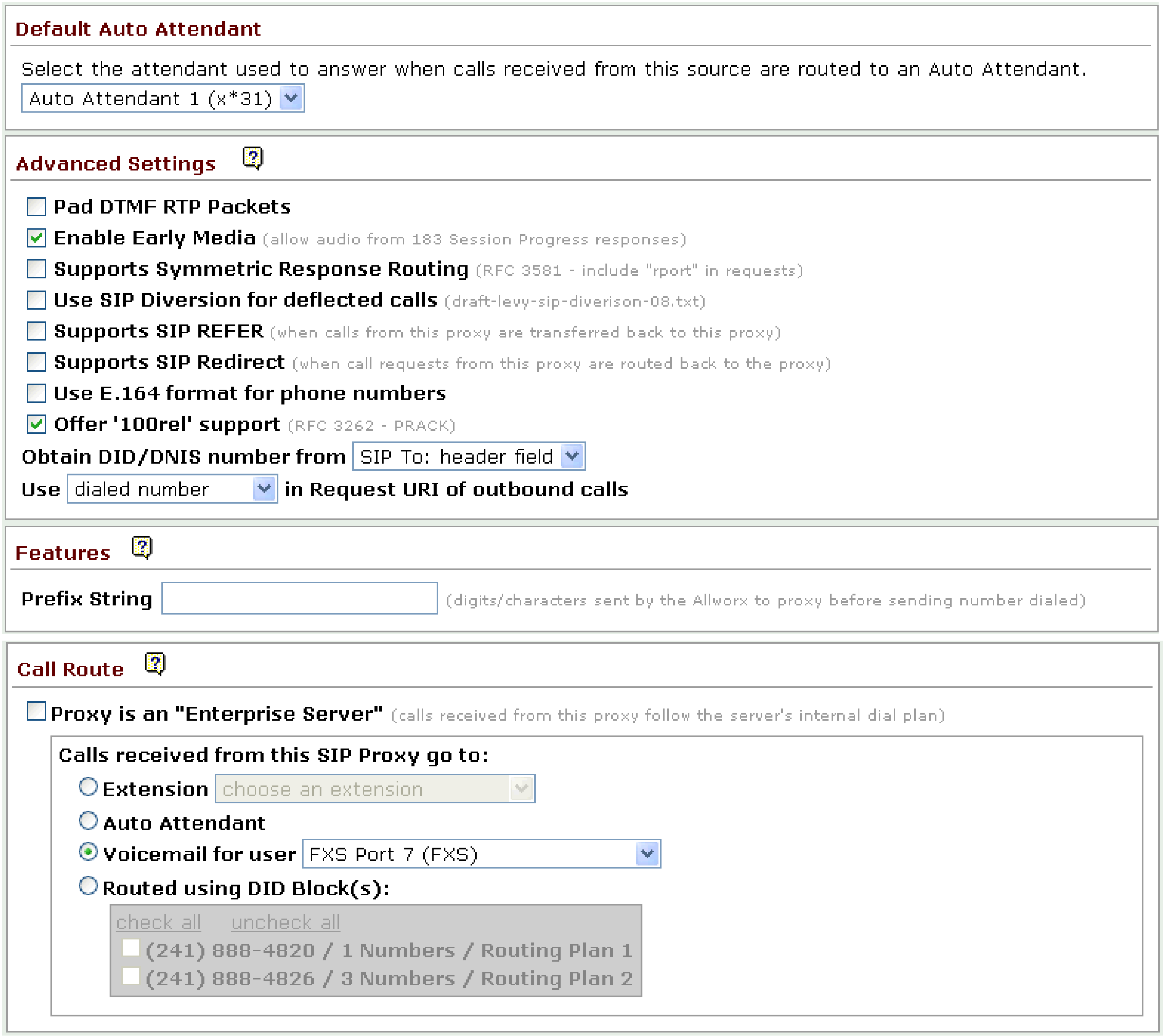


Figure 13

4. Enter a Description for this SIP Proxy
5. Enter the UserID – the Phone Number in most cases
6. Enter the SIP Server – IP address of the ESBC LAN port
7. Check “SIP Registration required” if not using Static Registration on the ESBC PBX Profile Settings.
8. If using SIP Registration, complete the Login ID (ESBC User ID), password (ESBC User Password), and Registrar (ESBC Lan IP) fields.
9. Enter a Caller ID Name and Caller ID Number
10. Enter the desired Call Route – Route using DID Block(s) or the specific Extension

Configuring Outbound Routing

1. Ensure you are using North American Numbering Plan Administration (NANPA)
2. Modify the External Dialing rules for dialing out from your Area Code so the correct Service Group is used, correct dialing string is passed on.

External Dialing Rules

North American Numbering Plan Administration (NANPA)	enabled	Modify

Area Code	Exchange	Number Dialed	Service Group	Action
240		9+240-xxx-nnnn 9+1+240-xxx-nnnn	All CO Lines & SIP Gateways	Modify
Home 241		9+241-xxx-nnnn 9+1+241-xxx-nnnn	All CO Lines & SIP Gateways	
all others		9+1+aaa-xxx-nnnn	All CO Lines & SIP Gateways	

aaa - area code xxx - exchange nnnn - number

Type	Number Dialed	Service Group	Action
Emergency	9+911	see Dialing Privileges Group for source of call	Modify
Phone Services (211,311,411,511,611,711,811)	9+n11	All Trunk Devices	
Operator	9+0	All Trunk Devices	
Long Distance Services	9+1010...	All CO Lines & SIP Gateways	
International Calls	9+011...	All Trunk Devices	
Public SIP Directory	1+nnnn (4 digits)	All SIP Proxies	
PIN Code	78+nnnnn (5 digits)	All CO Lines	
Outside Line Seizure	9#	All Trunk Devices	

Emergency Call Email Notifications are not enabled. Modify

Figure 14

3. Enter your Home Area Code and choose the service group to dial out from.

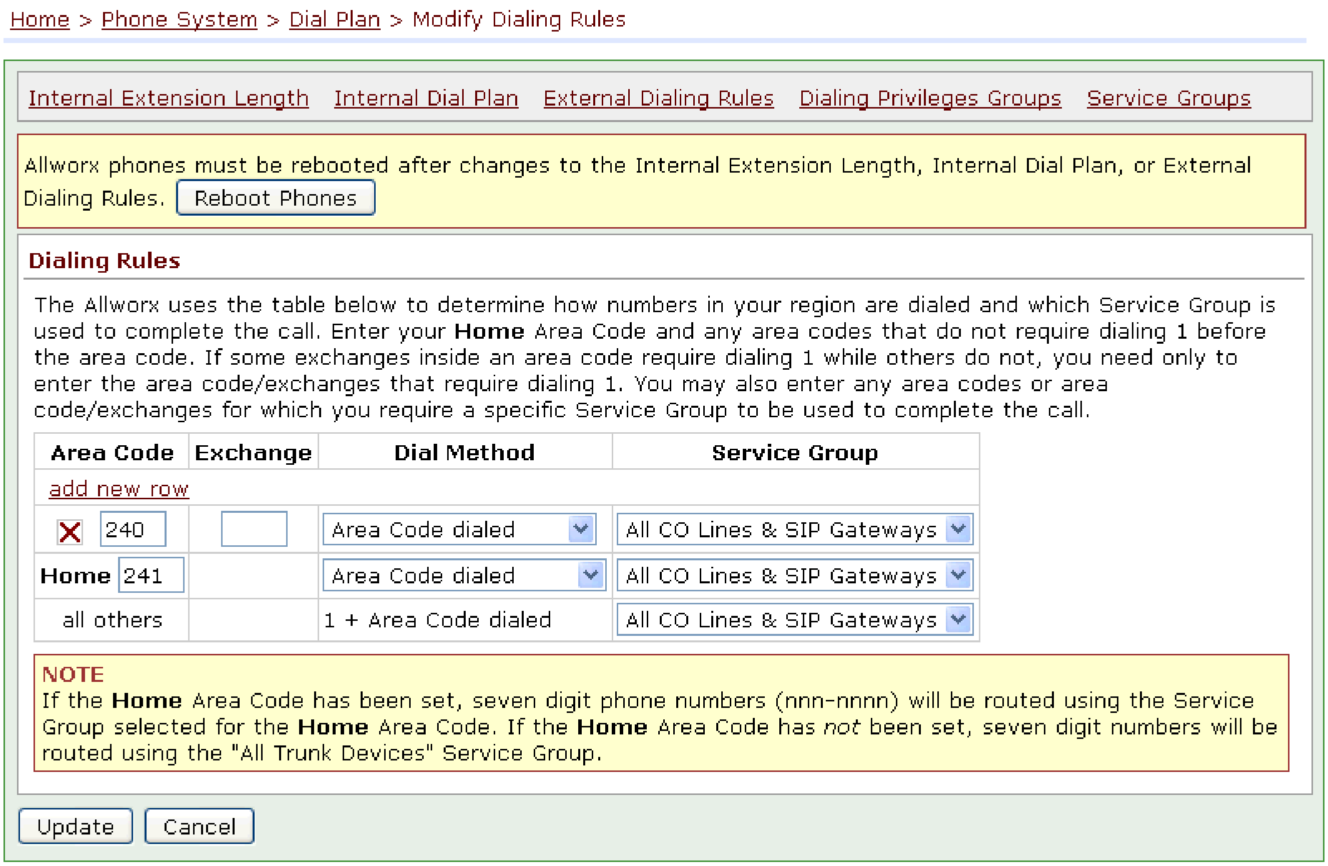


Figure 15

4. Ensure you have an Internal Dial Plan with an External Call Access number, normally this is a 9 + External Number. If not modify the Internal Dial Plan so you have one.

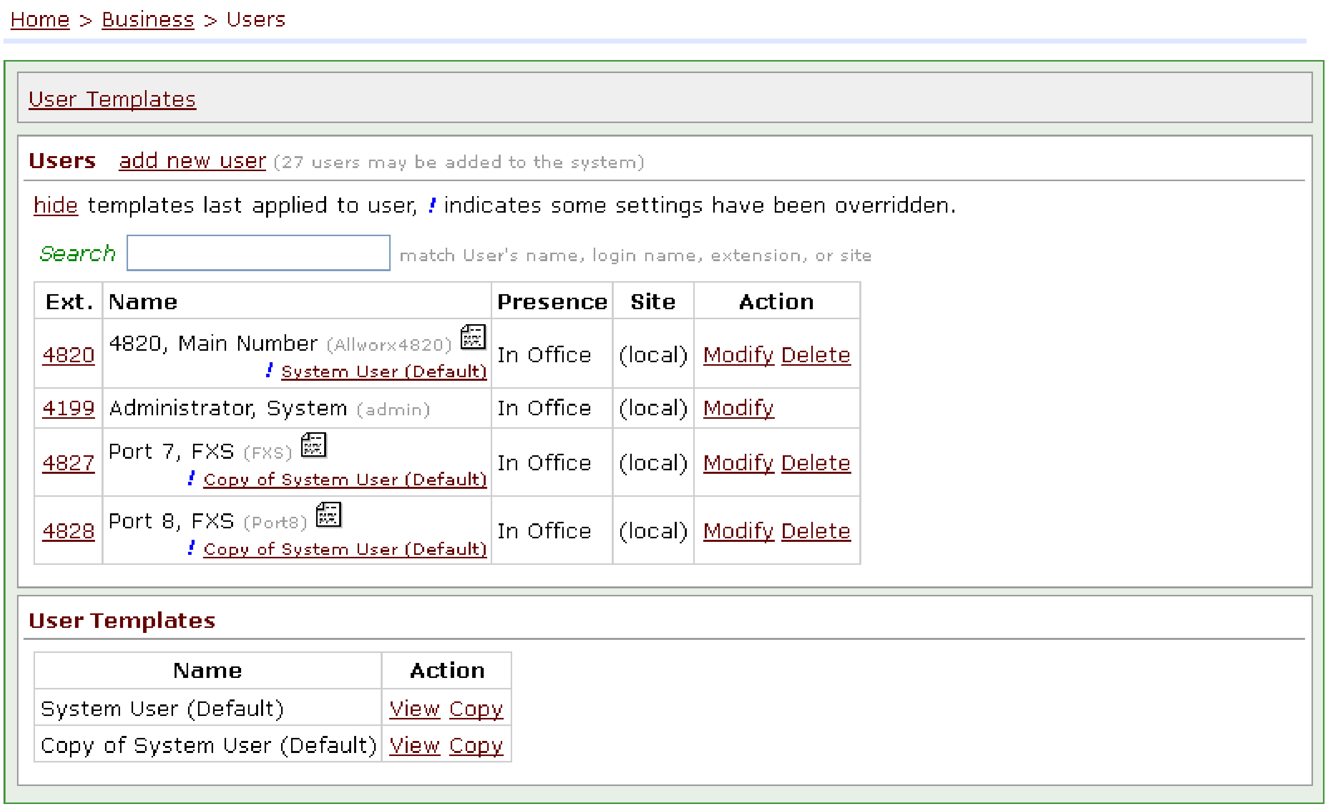
Internal Dial Plan modify view the Phone Functions Reference Card

Plan	
4xxx 5xxx	User and System Extensions
0	Operator
9 + external number	External Call access (follows External Dialing Rules below)
1 + enterprise number	Enterprise calling
2nnn	Internal station access (reserved for system)
350-399 34nnn	Speed dial numbers
6 + user extension	Message Center
700 call park 701-709 call retrieve 7xxxx call pickup 78 + pin code	Call Functions (park/pickup/audit pin code)
8 + user extension	Leave a voicemail for extension
*03 door relay *08 conference center *2n do not disturb *3n auto attendants *4nn call queues *950-*999 call retrieve *5xxxx call forwarding *6n paging	PBX Functions

Figure 16. Allworx 6X SIP Trunk Line Dialing Rule Settings

Add Users & Extensions

1. Navigate to Home > Business > Users



Home > Business > Users

User Templates

Users add new user (27 users may be added to the system)

hide templates last applied to user, ! indicates some settings have been overridden.

Search match User's name, login name, extension, or site

Ext.	Name	Presence	Site	Action
4820	4820, Main Number (Allworx4820) ! System User (Default)	In Office	(local)	Modify Delete
4199	Administrator, System (admin)	In Office	(local)	Modify
4827	Port 7, FXS (FXS) ! Copy of System User (Default)	In Office	(local)	Modify Delete
4828	Port 8, FXS (Port8) ! Copy of System User (Default)	In Office	(local)	Modify Delete

User Templates

Name	Action
System User (Default)	View Copy
Copy of System User (Default)	View Copy

Figure 17 Add Users and Extensions to Allworx

2. Click “add new user”

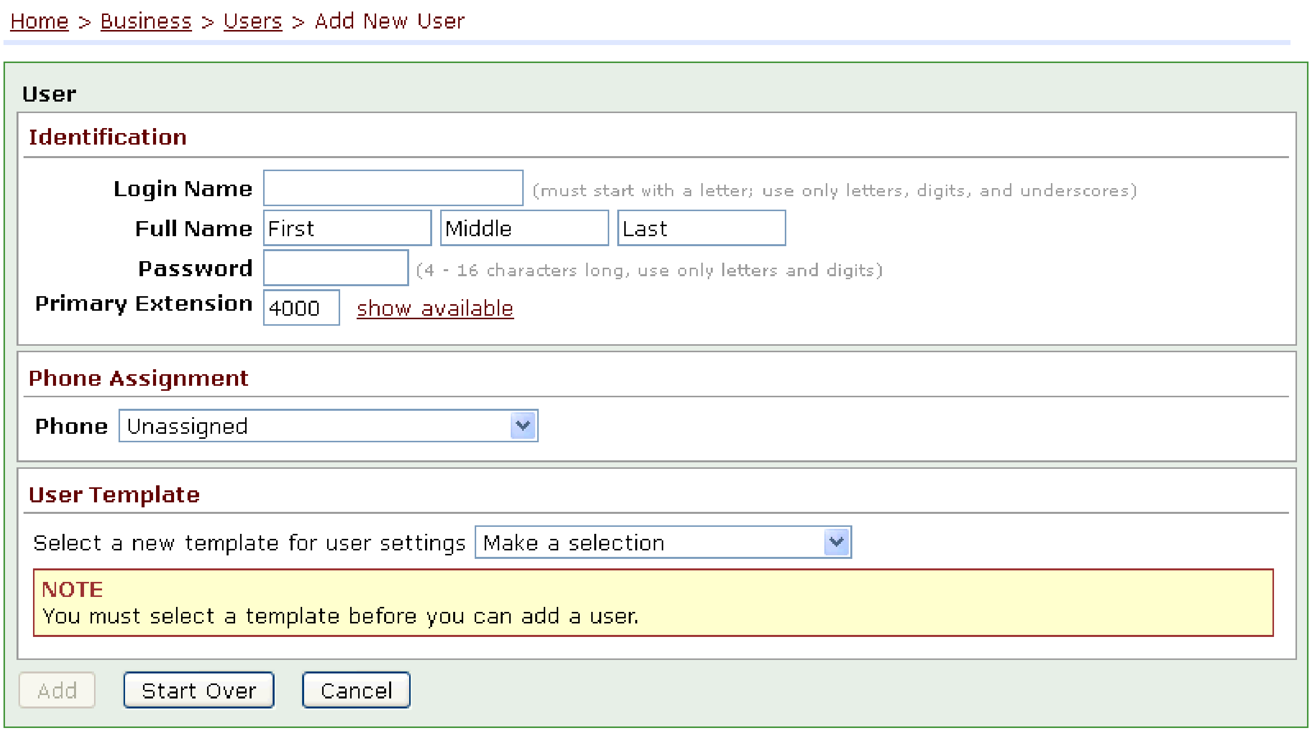


Figure 18

3. Enter Login Name
4. Enter First and Last name
5. enter a minimum 4 character password
6. Enter a Primary Extension – I would use the last 4 digits of the DID number if possible, so it is easy to follow.
7. If you have plugged in a SIP Phone, and it has been recognized, you can assign it to this user, choose from the pull down menu.
8. Choose a User Template, the default user should be fine.
9. Click "add" at bottom of screen. Note that when you choose the Template, you will get more options, and Defaults should be fine, unless you want to make changes.

FXS/Analog Handset and SIP Handsets

FXS Handset

1. Navigate to Home > Phone System > Handsets

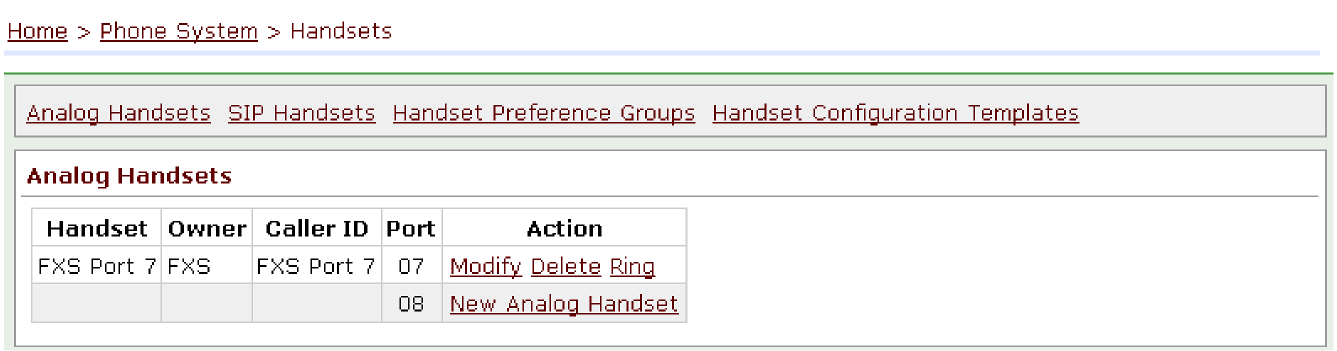


Figure 19

1. Click on "New Analog Handset"

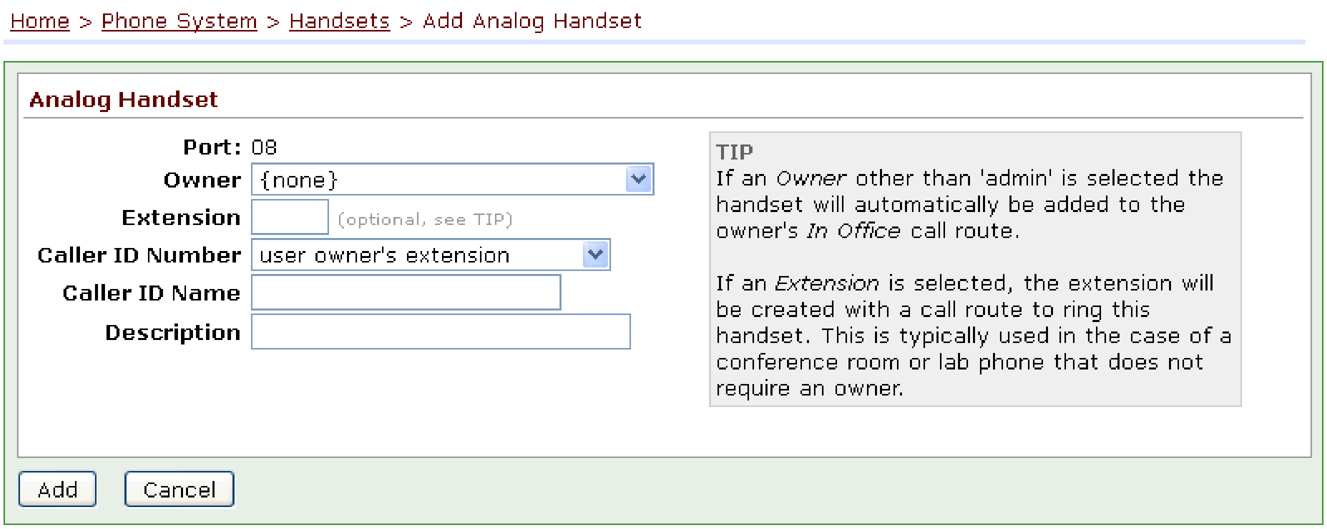


Figure 20

2. Choose an Owner from the pull down menu from one of the users you created earlier.
3. Choose a Caller ID Number from the pull down menu for the same owner. This should populate the Caller ID Name and Description as well.
4. Click "Add" to assign this information to your FXS port

SIP Handset

1. Navigate to Home > Phone System > Handsets

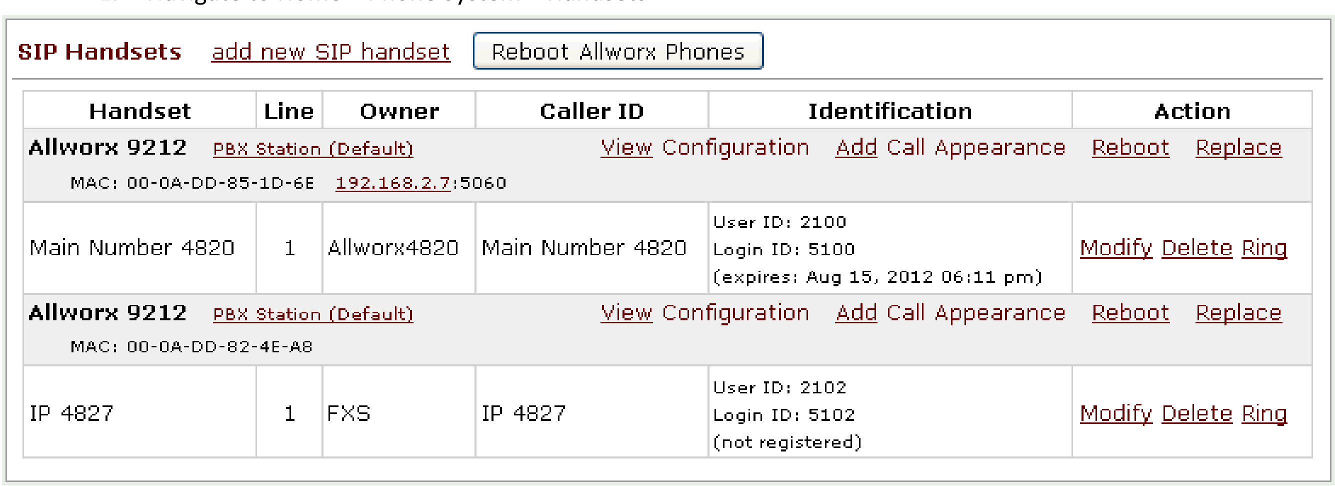


Figure 21

2. Click "add new SIP handset"

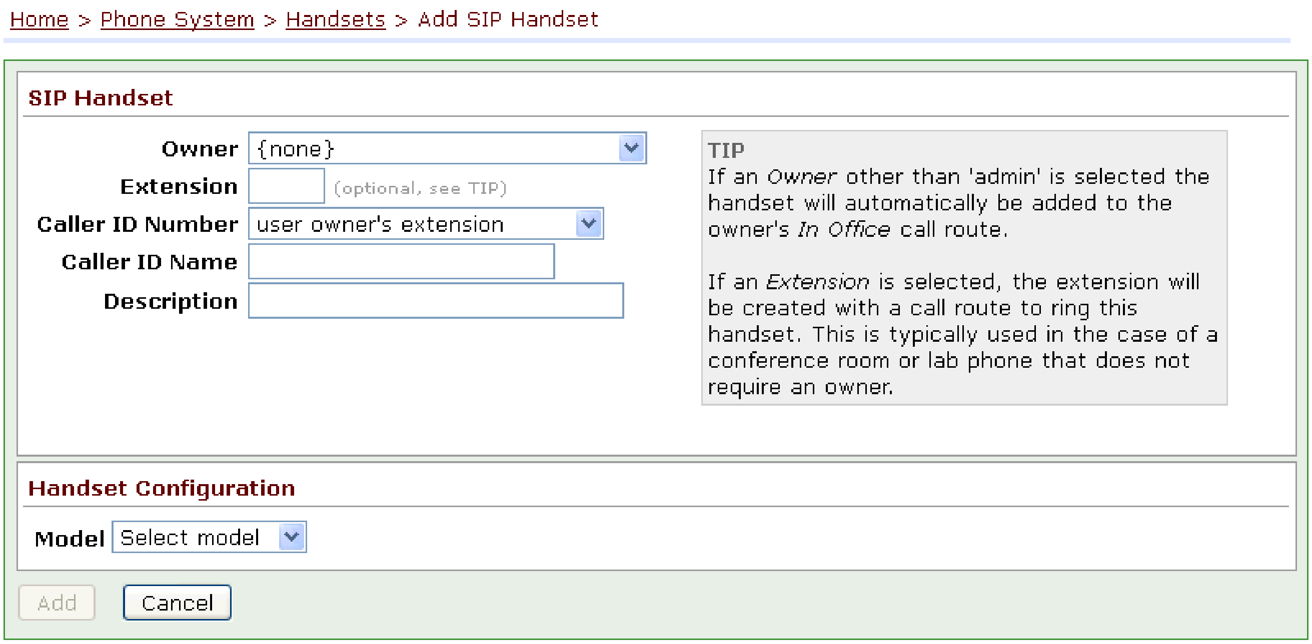


Figure 22

3. Choose the Owner from the Owner's pull down list
4. Choose Caller ID Number from the pull down list of the Owner you choose. _This will cause the Caller ID Name and Description to be filled in.
5. Choose the Model of your SIP Phone. When you choose the model, it will cause 3 additional items to be filled out

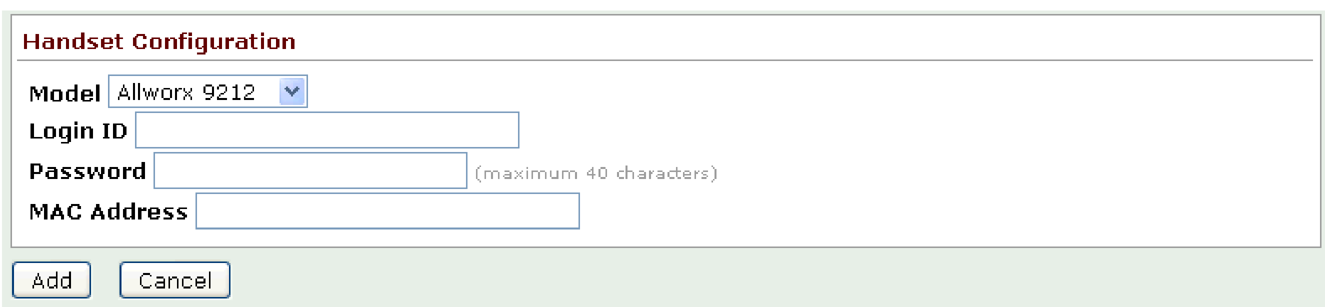


Figure 23

6. Enter a Login ID for this phone
7. Enter a Password for the phone
8. Enter the MAC Address of this phone.
9. Click "Add" to add this handset.

Allworx 6X Configuration File

Export creates an external copy of a configuration backup such that it can be imported later into this or another Allworx 6X device.

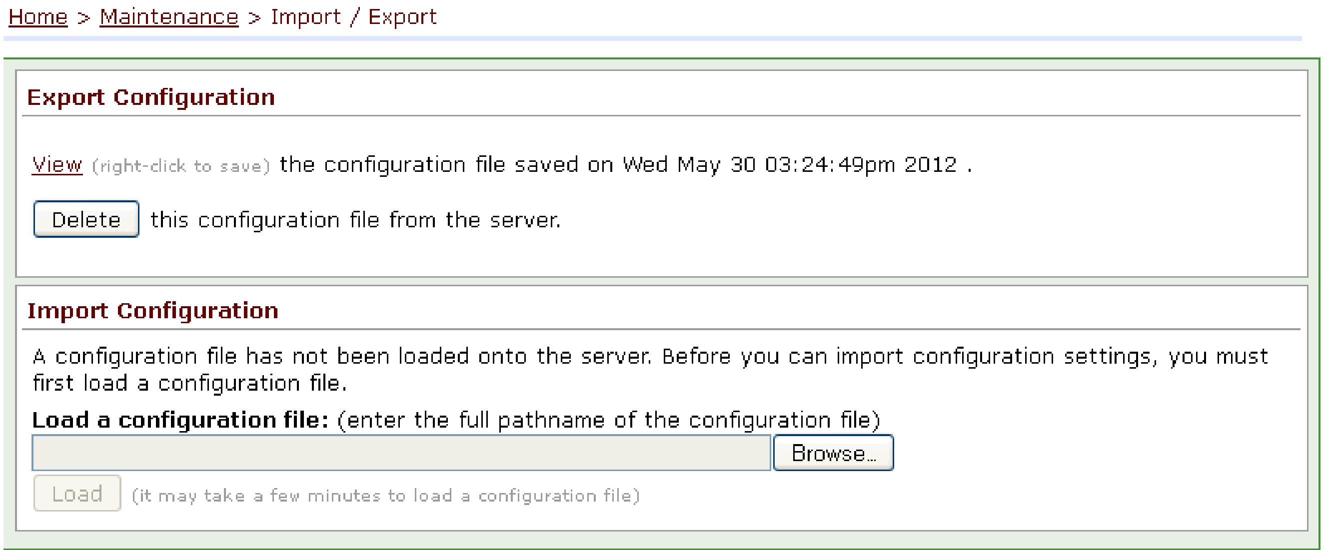

Home > Maintenance > Import / Export
Export Configuration
View (right-click to save) the configuration file saved on Wed May 30 03:24:49pm 2012 .
Delete this configuration file from the server.
Import Configuration
A configuration file has not been loaded onto the server. Before you can import configuration settings, you must first load a configuration file.
Load a configuration file: (enter the full pathname of the configuration file)
Browse...
Load (it may take a few minutes to load a configuration file)

Figure 24

4. ESBC Configurations

Please refer to ESBC Administrative Guide for detailed information.
http://www.innomedia.com/support_manuals.shtml

Configuring SIP Trunk Information

1. Navigate to **Telephony** > **SIP TRUNKS** > **Trunk Settings**
2. Enter the SIP Service information provided by your service provider.

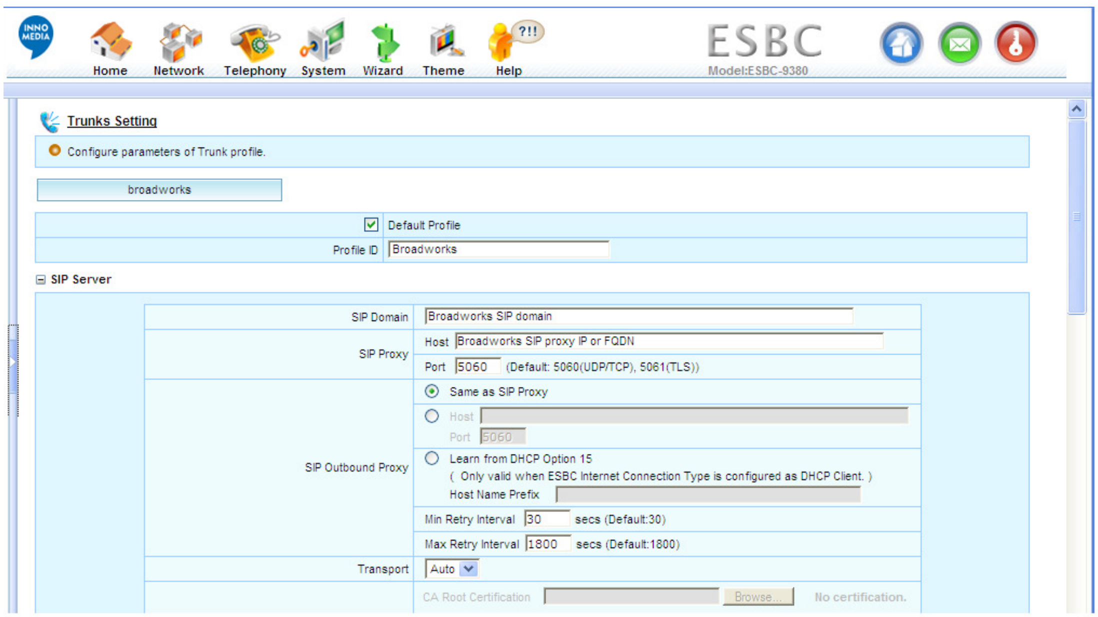


Figure 25 ESBC SIP Trunk – Trunk Settings

Adding and Configuring SIP Trunk and Account(s) on ESBC

ESBC supports two operational modes to access SIP Trunk telephony service provided by service provider:

- Registration mode
- Static mode

Configuring SIP Account(s) to ESBC with Registration Mode

1. Navigate to **Telephony** > **SIP ACCOUNTS** > **SIP UA Setting**
2. Click **Batch Config** button to **add/modify/delete** accounts in bulk mode
3. Click **Add** tag to add SIP account(s) to ESBC

It is possible to provision ESBC with only one pilot number (or main public identity). In such case, default route should be configured as well. ESBC connects to service provider network through either implicit registration mode or static operational mode. Default route does not work for individual DID registration mode.

Marks of different SIP UA types

Main Public Identity

Default Route

Only one pilot number is provisioned on the ESBC with implicit registration operation in the IOT example. The pilot number is also configured as the default route on the ESBC.

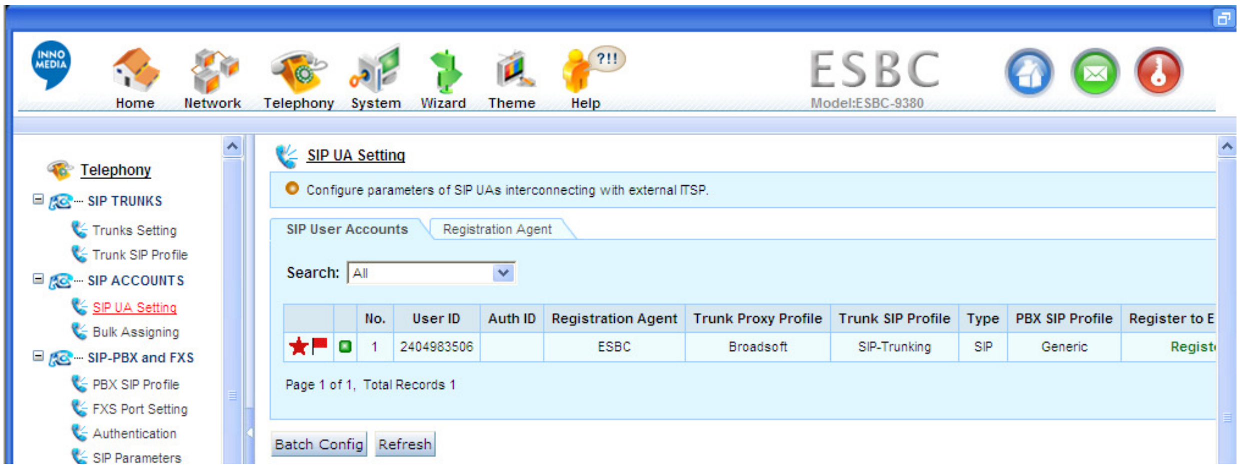


Figure 26 ESBC SIP Account Settings

Configuring Registration Agent

1. Click **Registration Agent** tab, as shown in Figure 27.
2. Add a Registration Agent (RA) by giving its name and associate a trunk line to this RA.

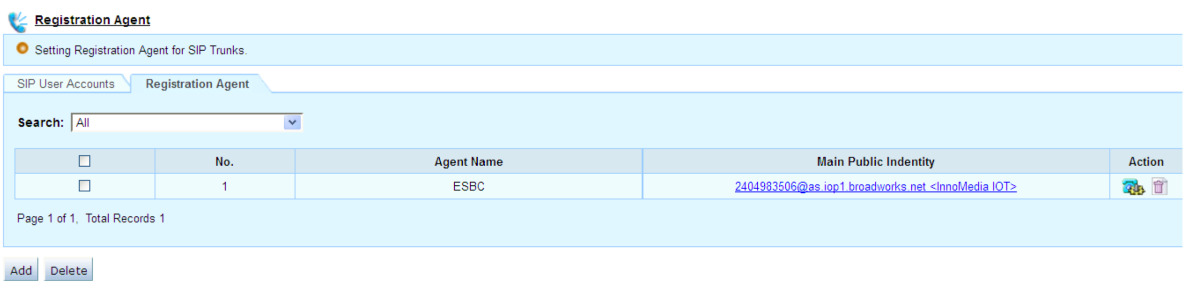
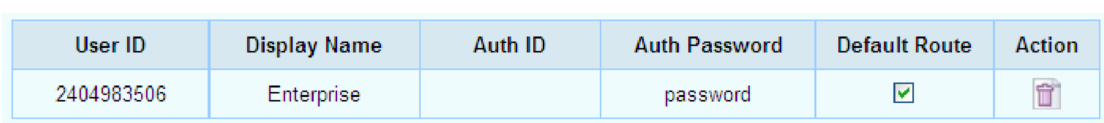


Figure 27 Configuring Registration Agent

Configuring Default Route

Enable Default Route can be configured by either selecting default when creating new UA accounts as in Figure 28 , or by editing an existing UA account as in Figure 29.**Error! Reference source not found.**



User ID	Display Name	Auth ID	Auth Password	Default Route	Action
2404983506	Enterprise		password	☑	

Figure 28 Configuring Default Route-1

Figure 29 Configuring Default Route-2

Configuring SIP Account(s) to ESBC using Static Mode

By default, the ESBC Trunk SIP Profile is configured with registration mode to access service provider network. When it is required to be configured as static mode, follow the steps.

1. Navigate to **Telephony > SIP TRUNKS > Trunk SIP Profile**
2. Add a new Trunk SIP Profile.
3. Check the box of **Static Registration.**

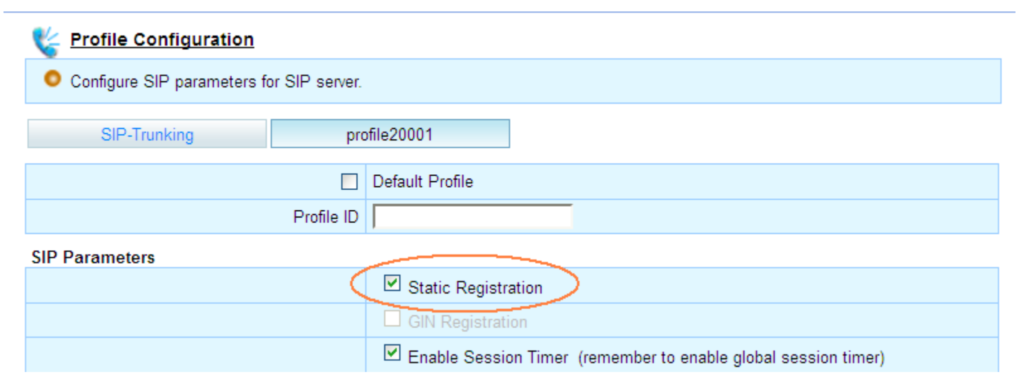


Figure 30 Configuring static operation mode

Configuring Call Transfer/Forward Methods on ESBC Interfacing with Service Provider's Core Network

Call Transfer

Different service providers may expect different Call Transfer/Forward methods when interfacing with their core networks. The ESBC supports two methods to interface with core network. (1) SIP REFER & (2) re-INVITE, the default configuration.

1. Navigate to **Telephony > SIP TRUNKS > Trunk SIP Profile.**

2. Choose the Target **Profile ID** (or create a new Profile), and configure this profile by clicking <**Action>.** Scroll down and configure this Feature item: **Process Call Transfer and Call Forward Locally**.
   - REFER method: Uncheck
   - Re-INVITE method: Check

Call Forward

Configuration for Call Forward is coupled with Call Transfer on ESBC. The configuration is the same as Call Transfer, as shown in Figure 31.

- When **REFER** is selected for Call Transfer, then the '**302 moved temporarily**' SIP method is used for Call Forwarding.
- When **re-INVITE** is selected for Call Transfer, then **re-INVITE** is used for Call Forwarding.

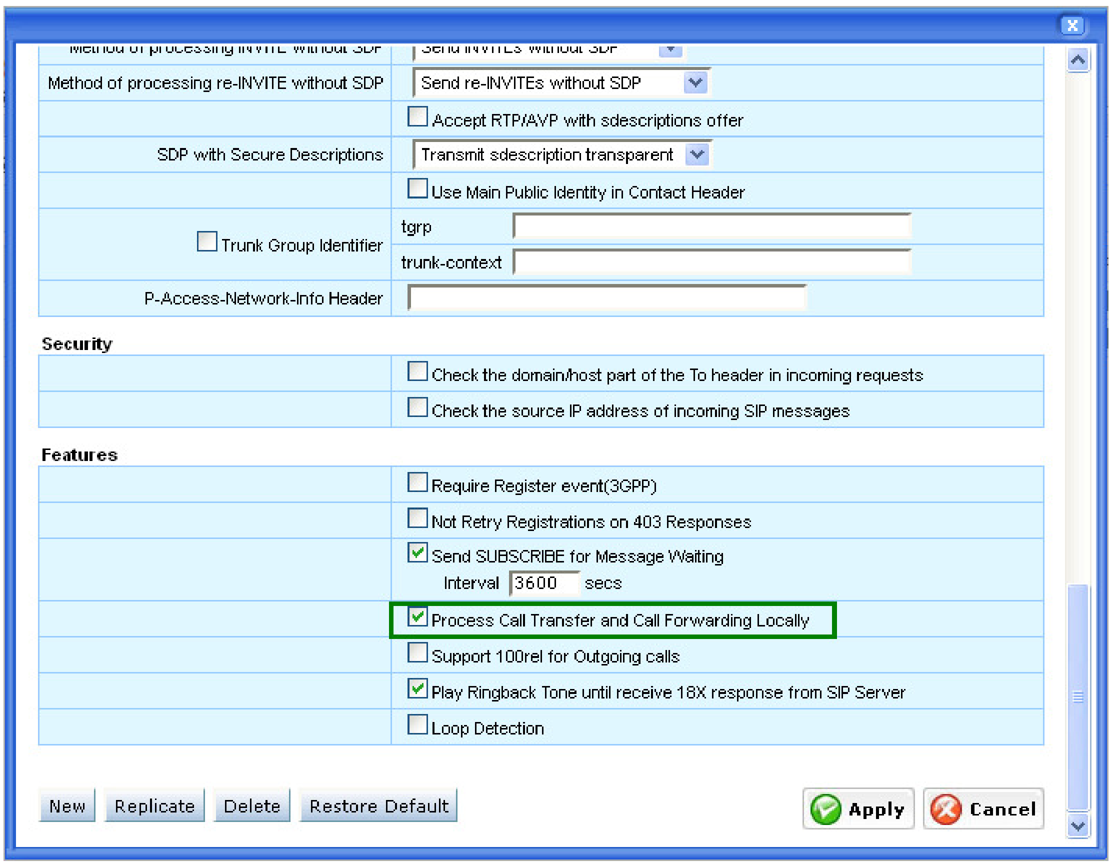


Figure 31 Call Transfer and Call Forward method configuration

Configuring Caller ID Display for Outbound Calls

ESBC provides the flexibility of configuring Caller ID display for outbound calls, by defining the "From" header of SIP messages.

The steps and options are as follows.

1. Navigate to **Telephony>SIP TRUNKS>Trunk SIP Profile**
2. Choose the target **Profile ID**, and click the **<Action>** button to configure this setting, as displayed in Figure 32**Error! Reference source not found.**.
   - Display "Main Public Identity": pilot number (and/or default route)

- Display “Alternate Identity”: individual DID numbers configured on ESBC
- Display “the Original Caller”: the caller IDs of PBX users. ESBC would take the “FROM” header as it is sent from PBX.

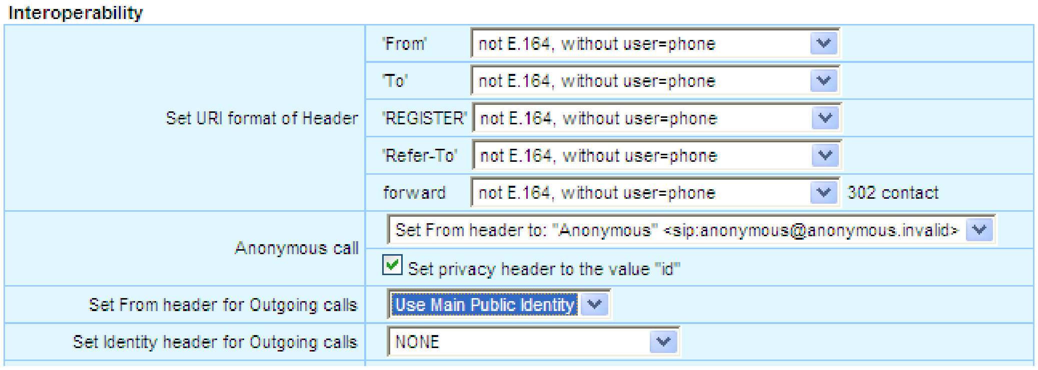


Figure 32. Configuring Caller IDs for Outbound Calls

5. Appendix

Trouble Shooting with Call Transfer/Call Forward Failure from SIP-PBX—SIP PBX Adding Invalid Domain in REFER-To Header

Some SIP-PBX's may adopt non-standard methods of handling call transfer/call forward requests which cause transfer (or forward) failure when the transferee UA is external to the ESBC.

When performing Call Transfer using the SIP REFER method, the REFER method indicates that the recipient (identified by the Request-URI) should contact a third party using the contact information provided in the request. However, some SIP-PBX while handling calls originating from other networks may have an invalid domain name in the REFER-To header and hence making the call may be routed to an unexpected destination.

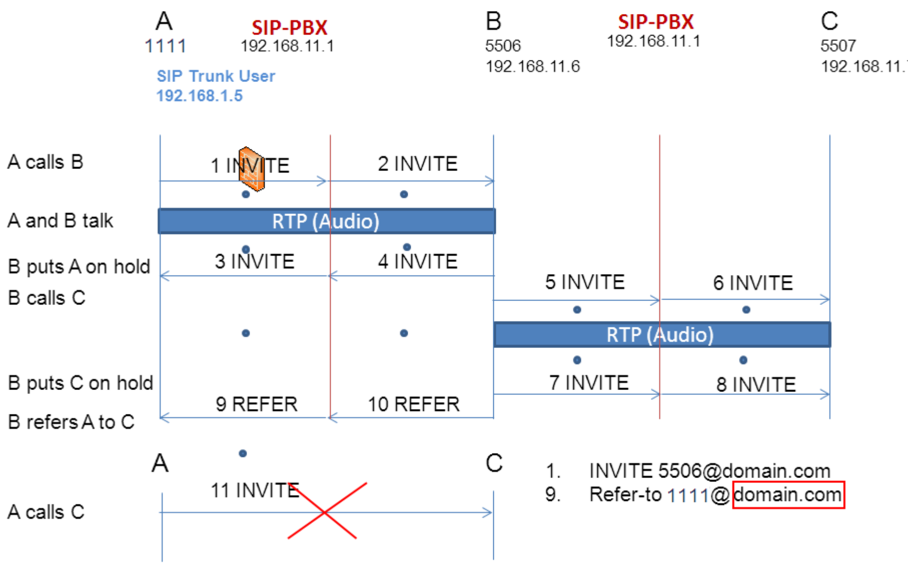


Figure 33 SIP-PBX sends local IP in contact information in REFER messages

ESBC Resolving the Network Contact Information of REFER-To Header

ESBC ensures the original caller receives a REFER request containing a Refer-To URI that is reachable. ESBC has PBX profile settings for SIP-PBX.

1. Navigate to **Telephony > SIP-PBX and FXS > PBX SIP Profile**

2. Choose the target **SIP-PBX profile** (or create a new profile),

3. Check the item "**Ignore Domain in Refer-To**" header.

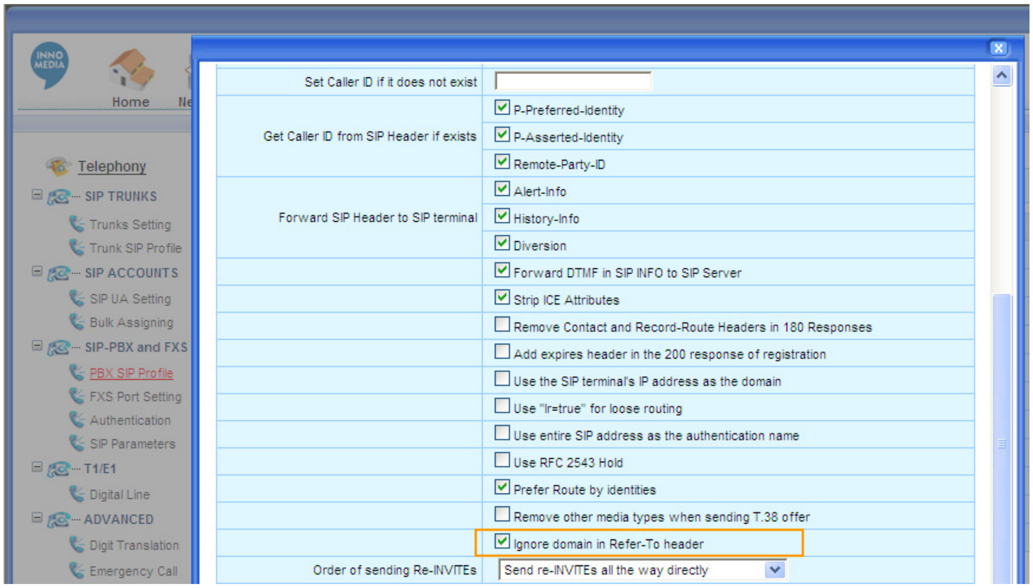


Figure 34. Ignore domain in Refer-To header to solve the unexpected network contact information

Trouble Shooting with Call Transfer/Call Forward Failure from SIP-PBX—SIP PBX Sending Local Call-ID in SIP REFER Messages

Some SIP-PBX's may handle Call Transfer in an unexpected way: The REFER message sent from the SIP-PBX carries a locally generated Call-ID, i.e., Call-ID: Y instead of conforming to Call-ID: X1 as in the first call, as illustrated in Figure **35Error! Reference source not found.**. Hence, it cause a failure to associate this Call-ID:Y to any of previously established call. In this case, ESBC handles this local "REFER" (transfer) with a new INVITE message to caller. **Please refer to Figure 31, this feature "Process Call Transfer and Forwarding Locally" should be checked.**

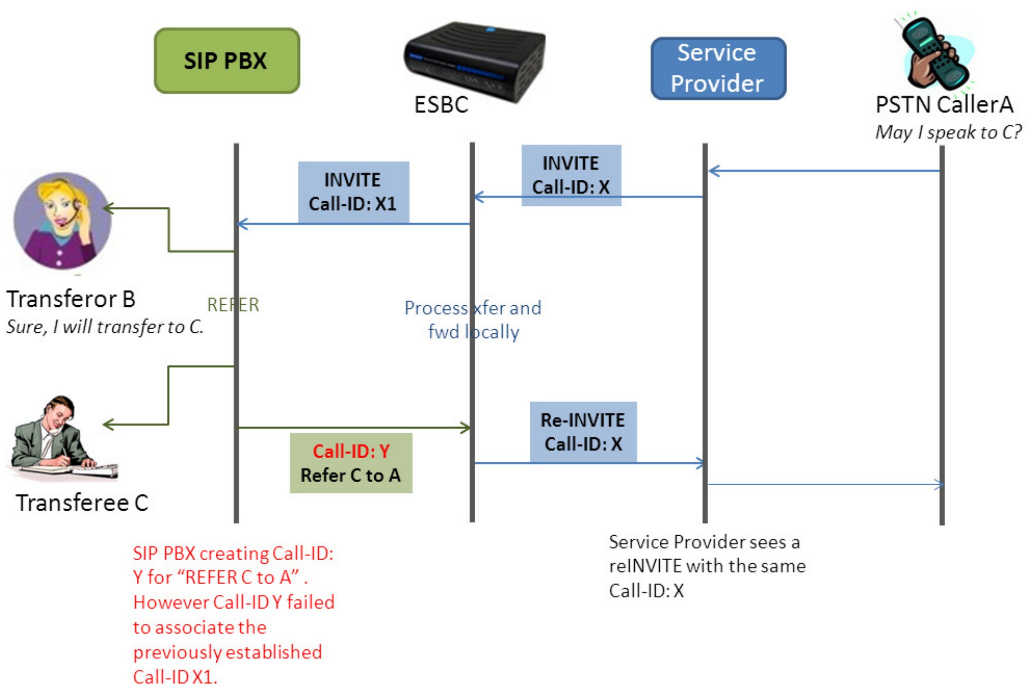


Figure 35 The SIP PBX Transfer Problem with REFER Message