

## Spectrum Enterprise SIP Trunking Service

### Allworx 6x Firmware 7.3.9.5

### IP PBX Configuration Guide

#### About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

#### About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Time Warner Cable Business Class. All references to Time Warner Cable Business Class, TWCBC or TWC should be read as Spectrum Enterprise.

**Thank you,**

**Spectrum Enterprise**

## Document Purpose and Target Audience

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This document will serve as a reference guide to configure the Allworx 6x IP PBX to interoperate with Time Warner Cable (TWC) SIP Trunk Service.

**This guide is not intended to be a replacement of the PBX manufacturer's user or configuration guide. It is intended to provide additional guidance on configuring the PBX in preparation to receive voice service from the SIP Trunk. It provides detailed instructions and best practices for a successful installation with TWC SIP Trunks.**

There are many options for establishing and maintaining service using the Allworx 6x series. This guide focuses on the minimum configurations essential for successful interoperability with Time Warner Cable Business Class SIP Trunks.

This configuration guide is based on:

### Customer Premise Equipment:

Model	Allworx 6x
Firmware	7.3.9.5

### TWC Network Equipment:

ESG	InnoMedia ESBC 9378-4B
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## SIP Trunk Components

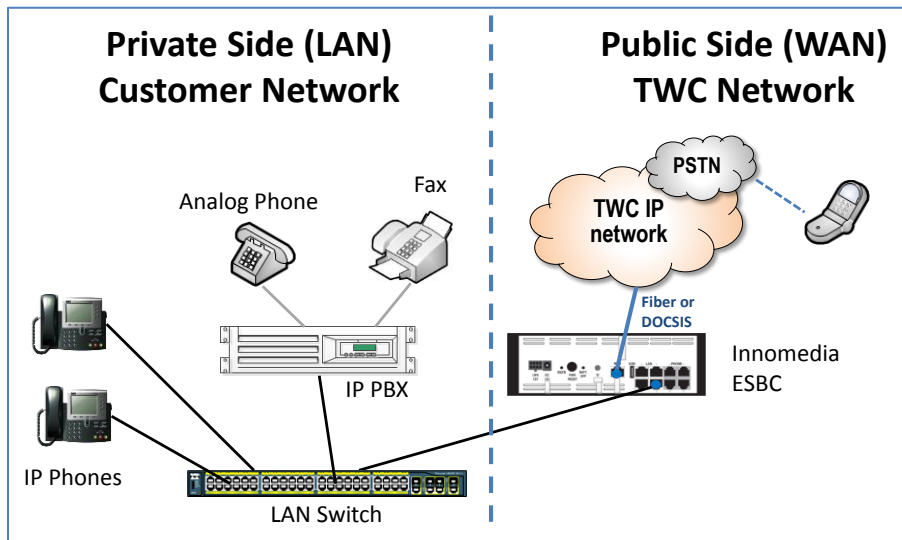
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The Time Warner Cable Business Class (TWCBC) SIP Trunks product is an IP-based, voice only trunk that uses Session Initiation Protocol (SIP) to connect an IP PBX to the PSTN. The IP PBX uses SIP to exchange signaling information with the service provider and to deliver and receive voice in IP packets.

The IP PBX is connected to the TWC Enterprise SIP Gateway (ESG), which provides network access for voice traffic. The customer is responsible for the LAN infrastructure and configuration, including the physical connection to the LAN port 2 on the ESG.

The ESG is the demarcation point to the TWC network. The ESG is connected to a dedicated router for SIP Trunks delivered over a fiber connection or to a cable modem when delivered over a DOCSIS connection.

SIP Trunk components located on the customer premise, including connections to the TWC network, are illustrated below.



All TWC SIP Trunk calls are routed over Time Warner Cable's IP network and are not routed over the public internet.

## Getting Started

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You will need to have the TWC “**SIP Trunk Questionnaire**” and “**Business Class (BC) SIP Trunks: Customer Cut Sheet**” in order to configure your IP PBX for TWC Business Class SIP Trunk service.

Confirm that your **IP PBX model number and software versions** recorded on the **Customer Cut Sheet** match those associated with your current equipment. If they do not, be sure to alert your TWC sales engineer or TWC project manager as this can impact how TWC designs your service configuration.

**Example from Customer Cut Sheet for Cisco UC 560:**

SERVICE INFORMATION	
PRODUCT	Business Class SIP TRUNK
IP-PBX MAKE	Cisco
IP-PBX MODEL	UC560
IP-PBX SOFTWARE VERSION	15.1(4)

While configuring your IP PBX for BC SIP Trunk service, you will need to know your Lead Telephone Number and the IP address of your IP PBX.

The **Lead Number** is confirmed on the **Customer Cut Sheet** as seen below:

Trunk Groups				
TWC TRUNK Group ID	DID Range	Lead Number	Inbound Call Blocking	Outbound Call Blocking

The **IP Address** of the IP PBX was recorded on the **SIP Trunk Questionnaire**, Section 5. Signaling and Media as shown below:

5- Signaling and Media		
IP Address for PBX or SBC To setup LAN configuration for signaling of voice traffic to the ESG	IP: xxx.xxx.xxx.xxx	TWC could provide IP address
	Subnet: 255.255.xxx.xxx	

This document is intended as an aid to help configure a customer’s IP PBX for interoperability with TWCBC SIP Trunk Service.

## Allworx 6x Configurations

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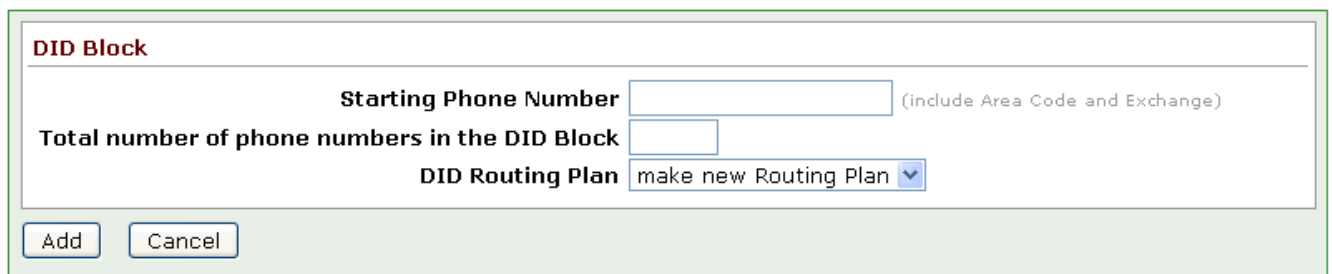
The instructions provided in this section are intended to configure the Allworx 6x IP-PBX to connect to the ESG. It is not intended for advanced functionality setups.

## Direct Inward Dialing Blocks

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1. Enter your Starting Phone Number in your Dialing Block (Lead Telephone Number)
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.

[Home](#) > [Phone System](#) > [Outside Lines](#) > [New DID Block](#)



**DID Block**

Starting Phone Number  (include Area Code and Exchange)

Total number of phone numbers in the DID Block

DID Routing Plan

Figure 1

## Direct Inward Dial Routing Plans

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- Navigate to Home > Phone System > Outside lines > DID Routing Plan
- 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

**Routing Plan Information** [modify](#)

<b>Description</b>	Routing Plan 1
<b>Default Extension</b>	0 - Operator
<b>Default DNIS Name</b>	
<b>DID Blocks using this plan</b>	(241) 888-4820 / 1 numbers

**Phone Number to Extension Mapping**

Phone Number	Extension	DNIS Name	Action
(241) 888-4820	4820 - Main Number 4820	2418884820	<a href="#">Modify</a>

TIP  
Phone Numbers that do not appear in the table above use the Default Extension for this DID Block as defined above.

To remove a phone number from the table, select Modify, then change the number to use the default extension.

Figure 2

- 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”

**Add Phone Number(s) to Extension Mapping**

**Phone Number(s)**

**Extension**

**DNIS Name**  (up to 47 characters: letters digits . , \ \_ ' -)

Figure 3

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

## SIP Trunk Setting

Gateway	Action
Allworx Test 4820 User ID: 2900 Gateway IP Address: 172.16.251.1:5060	<a href="#">Modify</a> <a href="#">Delete</a>

Figure 4

1. Navigate to Phone System > Outside Lines > SIP Gateway
2. Click “add new SIP Gateway”
3. Enter a 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.

[Home](#) > [Phone System](#) > [Outside Lines](#) > [New SIP Gateway](#)

**SIP Gateway** ?

Description

Number of Line Appearances  (0 to 99, should not exceed number of CO lines attached to gateway)

**SIP Registration** ?

Gateway uses SIP Registration

Login ID

Password  (maximum 40 characters)


Gateway uses static IP Address

IP Address

SIP Port  5060

Figure 5

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

**SIP Gateway** 

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**Description**

**Caller ID Name**  (up to 47 characters: letters digits . , \ \_ ' -)

**Use External Caller ID Name from handset** (if specified)

**Use Caller ID Name from external sources** (if received)


**Caller ID Number**  (up to 24 digits)

**Use External Caller ID Number from handset** (if specified)

**Use Caller ID Number from external sources** (if received)

**Number of Line Appearances**  (0 to 99, should not exceed number of CO lines attached to gateway)

---


**SIP Registration** 

**Gateway uses SIP Registration**

(maximum 40 characters)

**Gateway uses static IP Address**

Figure 6

**Advanced Settings** 

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**Pad DTMF RTP Packets**

**Enable Early Media** (allow audio from 183 Session Progress responses)

**Supports SIP REFER** (when calls from this gateway are transferred back to this gateway)

**Supports SIP Redirect** (when call requests from this gateway are routed back to the gateway)


**Use E.164 format for phone numbers**

**Offer '100rel' support** (RFC 3262 - PRACK)

**Obtain DID/DNIS number from**

**Use**  **in Request URI of outbound calls**

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**Features** 

**Prefix String**  (digits/characters sent by the Allworx to gateway before sending number dialed)

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**Default Auto Attendant**

Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.

Figure 7



**Call Route**

Calls received from this SIP Gateway go to:

Extension

Auto Attendant

Voicemail for user

Routed using DID Block(s):

[check all](#)   [uncheck all](#)

(241) 888-4820 / 1 Numbers / Routing Plan 1

(241) 888-4826 / 3 Numbers / Routing Plan 2

**Figure 8**

8. Enter your Desired Caller ID Name – something like your company name for example.
9. Enter your Caller ID Number – Lead Telephone 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

**SIP Proxies** [add new SIP Proxy](#)

Proxy	Action
ESBC 2418884826 User ID: 2418884826 Proxy Address: 172.16.251.1:5060 (expires: Aug 14, 2012 04:21 pm)	<a href="#">Modify</a> <a href="#">Delete</a> <input type="button" value="Register Now"/>
SIP Phone User ID: 2418884820 (expires: Aug 14, 2012 04:20 pm)	<a href="#">Modify</a> <a href="#">Delete</a> <input type="button" value="Register Now"/>

**Figure 9. Allworx 6X SIP Trunk Settings**

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

**SIP Proxy** 

**Description**

**User ID**

**SIP Server**  **Port**   
(customer domain/realm) (enter IP Address or Domain Name)

**Outbound Proxy**  **Port**   
(if different from SIP Server) (enter IP Address or Domain Name)

**SIP Registration required**

**Login ID**

**Password**  (maximum 40 characters)

**Registrar**  **Port**   
(if different from Outbound Proxy) (enter IP Address or Domain Name)

**Caller ID Name**  (up to 47 characters: letters digits , \ \_ ' -')

**Use External Caller ID Name from handset** (if specified)

**Use Caller ID Name from external sources** (if received)

**Caller ID Number**  (up to 24 digits)

**Use External Caller ID Number from handset** (if specified)

**Use Caller ID Name from external sources** (if received)

**Caller ID Number**  (up to 24 digits)

**Use External Caller ID Number from handset** (if specified)

**Use Caller ID Number from external sources** (if received)

**Maximum Active Calls**  (1 to 99, should not exceed proxy capabilities or available bandwidth)

**Number of Line Appearances**  (0 to Maximum Active Calls)

**Append Enterprise Prefix to Dialback number for incoming calls**

**Send digits as dialed** (without prepending 1 and/or area code)

**Digits Sent**  (digits from the full number, 1-XXX-XXX-XXXX, to send to the proxy)

Figure 10

**Default Auto Attendant**

Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.

Auto Attendant 1 (x\*31) ▼

**Advanced Settings** ?

Pad DTMF RTP Packets

Enable Early Media (allow audio from 183 Session Progress responses)

Supports Symmetric Response Routing (RFC 3581 - include "rport" in requests)

Use SIP Diversion for deflected calls (draft-levy-sip-diverison-08.txt)

Supports SIP REFER (when calls from this proxy are transferred back to this proxy)

Supports SIP Redirect (when call requests from this proxy are routed back to the proxy)

Use E.164 format for phone numbers

Offer '100rel' support (RFC 3262 - PRACK)

Obtain DID/DNIS number from SIP To: header field ▼

Use dialed number ▼ in Request URI of outbound calls

**Features** ?

Prefix String  (digits/characters sent by the Allworx to proxy before sending number dialed)

**Call Route** ?

Proxy is an "Enterprise Server" (calls received from this proxy follow the server's internal dial plan)

**Calls received from this SIP Proxy go to:**

Extension

Auto Attendant

Voicemail for user

Routed using DID Block(s):

(241) 888-4820 / 1 Numbers / Routing Plan 1

(241) 888-4826 / 3 Numbers / Routing Plan 2

Figure 11

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 ESG LAN port
7. Check “SIP Registration required” if not using Static Registration on the ESG PBX Profile Settings.
8. If using SIP Registration, complete the Login ID (ESBC User ID), password (ESG 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	<a href="#">Modify</a>
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	<a href="#">Modify</a>
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 12

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

[Internal Extension Length](#)
[Internal Dial Plan](#)
[External Dialing Rules](#)
[Dialing Privileges Groups](#)
[Service Groups](#)

Allworx phones must be rebooted after changes to the Internal Extension Length, Internal Dial Plan, or External Dialing Rules.

**Dialing Rules**

The Allworx uses the table below to determine how numbers in your region are dialed and which Service Group is used to complete the call. Enter your **Home** Area Code and any area codes that do not require dialing 1 before the area code. If some exchanges inside an area code require dialing 1 while others do not, you need only to enter the area code/exchanges that require dialing 1. You may also enter any area codes or area code/exchanges for which you require a specific Service Group to be used to complete the call.

Area Code	Exchange	Dial Method	Service Group
<a href="#">add new row</a>			
<input type="text" value="X 240"/>	<input type="text"/>	Area Code dialed	All CO Lines & SIP Gateways
<b>Home</b> <input type="text" value="241"/>		Area Code dialed	All CO Lines & SIP Gateways
all others		1 + Area Code dialed	All CO Lines & SIP Gateways

**NOTE**  
 If the **Home** Area Code has been set, seven digit phone numbers (nnn-nnnn) will be routed using the Service Group selected for the **Home** Area Code. If the **Home** Area Code has *not* been set, seven digit numbers will be routed using the "All Trunk Devices" Service Group.

Figure 13

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 14.** Allworx 6X SIP Trunk Line Dialing Rule Settings

## Add Users and 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
<a href="#">4820</a>	4820, Main Number (Allworx4820)  ! <a href="#">System User (Default)</a>	In Office	(local)	<a href="#">Modify</a> <a href="#">Delete</a>
<a href="#">4199</a>	Administrator, System (admin)	In Office	(local)	<a href="#">Modify</a>
<a href="#">4827</a>	Port 7, FXS (FXS)  ! <a href="#">Copy of System User (Default)</a>	In Office	(local)	<a href="#">Modify</a> <a href="#">Delete</a>
<a href="#">4828</a>	Port 8, FXS (Port8)  ! <a href="#">Copy of System User (Default)</a>	In Office	(local)	<a href="#">Modify</a> <a href="#">Delete</a>

**User Templates**

Name	Action
System User (Default)	<a href="#">View</a> <a href="#">Copy</a>
Copy of System User (Default)	<a href="#">View</a> <a href="#">Copy</a>

Figure 15 Add Users and Extensions to Allworx

2. Click “add new user”

### User

#### Identification

**Login Name**  (must start with a letter; use only letters, digits, and underscores)

**Full Name**  First  Middle  Last

**Password**  (4 - 16 characters long, use only letters and digits)

**Primary Extension**  4000 [show available](#)

#### Phone Assignment

**Phone**  Unassigned

#### User Template

Select a new template for user settings  Make a selection

**NOTE**  
You must select a template before you can add a user.

Figure 16

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

[Home](#) > [Phone System](#) > [Handsets](#)

<a href="#">Analog Handsets</a> <a href="#">SIP Handsets</a> <a href="#">Handset Preference Groups</a> <a href="#">Handset Configuration Templates</a>				
<b>Analog Handsets</b>				
Handset	Owner	Caller ID	Port	Action
FXS Port 7	FXS	FXS Port 7	07	<a href="#">Modify</a> <a href="#">Delete</a> <a href="#">Ring</a>
			08	<a href="#">New Analog Handset</a>

Figure 17

1. Click on “New Analog Handset”

[Home](#) > [Phone System](#) > [Handsets](#) > [Add Analog Handset](#)

<b>Analog Handset</b>	
Port: 08	
Owner	{none} <input type="button" value="v"/>
Extension	<input type="text"/> (optional, see TIP)
Caller ID Number	user owner's extension <input type="button" value="v"/>
Caller ID Name	<input type="text"/>
Description	<input type="text"/>
<input type="button" value="Add"/> <input type="button" value="Cancel"/>	

**TIP**  
If an *Owner* other than 'admin' is selected the handset will automatically be added to the owner's *In Office* call route.

If an *Extension* is selected, the extension will be created with a call route to ring this handset. This is typically used in the case of a conference room or lab phone that does not require an owner.

Figure 18

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

**SIP Handsets** [add new SIP handset](#) Reboot Allworx Phones

Handset	Line	Owner	Caller ID	Identification	Action
<b>Allworx 9212</b> <a href="#">PBX Station (Default)</a> <a href="#">View Configuration</a> <a href="#">Add Call Appearance</a> <a href="#">Reboot</a> <a href="#">Replace</a>					
MAC: 00-0A-DD-85-1D-6E <a href="#">192.168.2.7</a> :5060					
Main Number 4820	1	Allworx4820	Main Number 4820	User ID: 2100 Login ID: 5100 (expires: Aug 15, 2012 06:11 pm)	<a href="#">Modify</a> <a href="#">Delete</a> <a href="#">Ring</a>
<b>Allworx 9212</b> <a href="#">PBX Station (Default)</a> <a href="#">View Configuration</a> <a href="#">Add Call Appearance</a> <a href="#">Reboot</a> <a href="#">Replace</a>					
MAC: 00-0A-DD-82-4E-A8					
IP 4827	1	FXS	IP 4827	User ID: 2102 Login ID: 5102 (not registered)	<a href="#">Modify</a> <a href="#">Delete</a> <a href="#">Ring</a>

Figure 19

2. Click “add new SIP handset”

[Home](#) > [Phone System](#) > [Handsets](#) > [Add SIP Handset](#)

**SIP Handset**

Owner: {none}

Extension:  (optional, see TIP)

Caller ID Number: user owner's extension

Caller ID Name:

Description:

**TIP**

If an *Owner* other than 'admin' is selected the handset will automatically be added to the owner's *In Office* call route.

If an *Extension* is selected, the extension will be created with a call route to ring this handset. This is typically used in the case of a conference room or lab phone that does not require an owner.

**Handset Configuration**

Model:

Figure 20

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

**Handset Configuration**

Model

Login ID

Password  (maximum 40 characters)

MAC Address

**Figure 21**

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 Configuration File

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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 .

this configuration file from the server.

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**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)

(it may take a few minutes to load a configuration file)

**Figure 22**

## Appendix

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### TWC Turn-up Testing Procedure

To ensure proper service between the IP PBX and the TWC network, test calls from the IP PBX will be made. Typically, the following call types will be used (call testing varies depending on service configuration)

1. Outbound/Inbound call to a local number
2. Outbound/Inbound call to a long distance number
3. Calls to 411 and 611
4. Outbound calls to a blocked number to verify call blocking settings
5. Other calls based on customer request , e.g. FAX testing using T.38 or calls to an auto-attendant to verify DTMF

### Questions

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If you have questions, please contact your Time Warner Cable Business Class Account Executive.