

## **Spectrum Enterprise SIP Trunking Service AastraLink Pro 160 Firmware 1.2.2 build 1005 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 AstraLink Pro 160 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 AstraLink Pro 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	AastraLink Pro 160
Firmware	1.2.2 build 1005

### TWC Network Equipment:

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

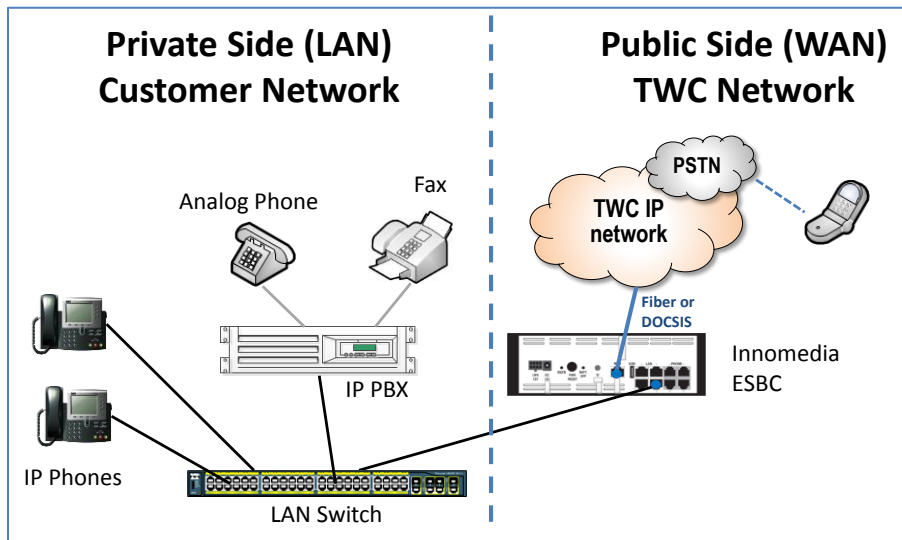
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The Time Warner Cable Business Class (TWCBC) SIP Trunks product is an IP-based, voice only trunk that uses Session Initiation Protocol (SIP) to connect an IP PBX to the PSTN. The IP PBX uses SIP to exchange signaling information with the service provider 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	IP: xxx.xxx.xxx.xxx	
To setup LAN configuration for signaling of voice traffic to the ESG	Subnet: 255.255.xxx.xxx	
		TWC could provide IP address

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

## AastraLink Pro 160 Configuration

The instructions provided in this section are intended to help configure the AastraLink Pro 160 to connect to the ESG. They are not intended for advanced functionality setups. It is further assumed that the customer has knowledge of the AastraLink Pro 160 operations.

1. Connect Aastra IP phones to the same LAN as the AastraLink Pro 160 PBX, the IP Phones should automatically find the AastraLink Pro 160.
2. Login to the AastraLink Pro 160 PBX.

## Configure Extensions with Direct Inbound Dialing Numbers (DID)

1. Navigate to **Users -> User List**
2. **User List:** Choose the Operator Account (default account 200). (It is assumed you have configured your device with at least one Extension already)

The screenshot shows the AastraLink Pro 160 web interface. The top navigation bar includes 'My Phone', 'Users', 'Configuration', and 'Maintenance'. Under 'Users', there are sub-tabs for 'User List', 'Groups', 'Default SoftKeys', and 'Default SoftKey Permissions'. The main content area displays a table with the following data:

<input type="checkbox"/>	Extension Name	Account Type	Account Flags	IP Address	MAC Address	SIP DID Number	Firmware Version
<input type="checkbox"/>	200 Dave Test	Administrator	Local, Offline, Operator	172.16.0.100	00:08:5D:20:F7:B0	2404983515	2.5.2.2041-SIP
<input type="checkbox"/>	201 Jim Jones	User	Local	172.16.0.112	00:08:5D:21:07:38	2404983516	2.5.2.2041-SIP

Below the table, it says 'Displaying 1-2 of 2'. At the bottom of the interface, there are buttons for 'Delete', 'Add Phones', 'Upload User List', 'Download User List', and 'Reboot Phones'. The footer contains the copyright notice: 'Copyright © 2006-2010 Aastra Telecom. All Rights Reserved.' and the status 'Status: Ready'.

Figure 1 Extension/User List

3. Enter the SIP DID Number that this extension will use – Since this is the Operator Extension I would recommend to use the Pilot Number/Main number

The screenshot displays the AastraLink PRO web interface. At the top, the user is logged in as 'Dave Test (x200)' with links for 'About' and 'Logout'. The main navigation bar includes 'My Phone', 'Users', 'Configuration', and 'Maintenance'. Under 'Users', there are sub-links for 'User List', 'Groups', 'Default SoftKeys', and 'Default SoftKey Permissions'. The 'Users' section is active, showing tabs for 'General', 'SoftKey Permissions', 'SoftKeys', and 'Top SoftKeys'. The 'General' tab is selected, displaying a configuration form for user 'Dave Test'. The form includes various settings such as 'User Enabled', 'Voicemail Enabled', 'Rings before Voicemail', 'Secondary Extension', 'SLA Enabled', 'Private Extension', 'Extension', 'First Name', 'Last Name', 'Password', 'Bypass PIN', 'Email', 'Account Type', 'Operator', 'Outgoing Line', 'Phone Type', 'Phone Firmware Version', 'IP Address', 'MAC Address', and 'SIP DID Number'. The 'SIP DID Number' field, containing the value '2404983515', is circled in red. At the bottom of the form are 'Save' and 'Cancel' buttons.

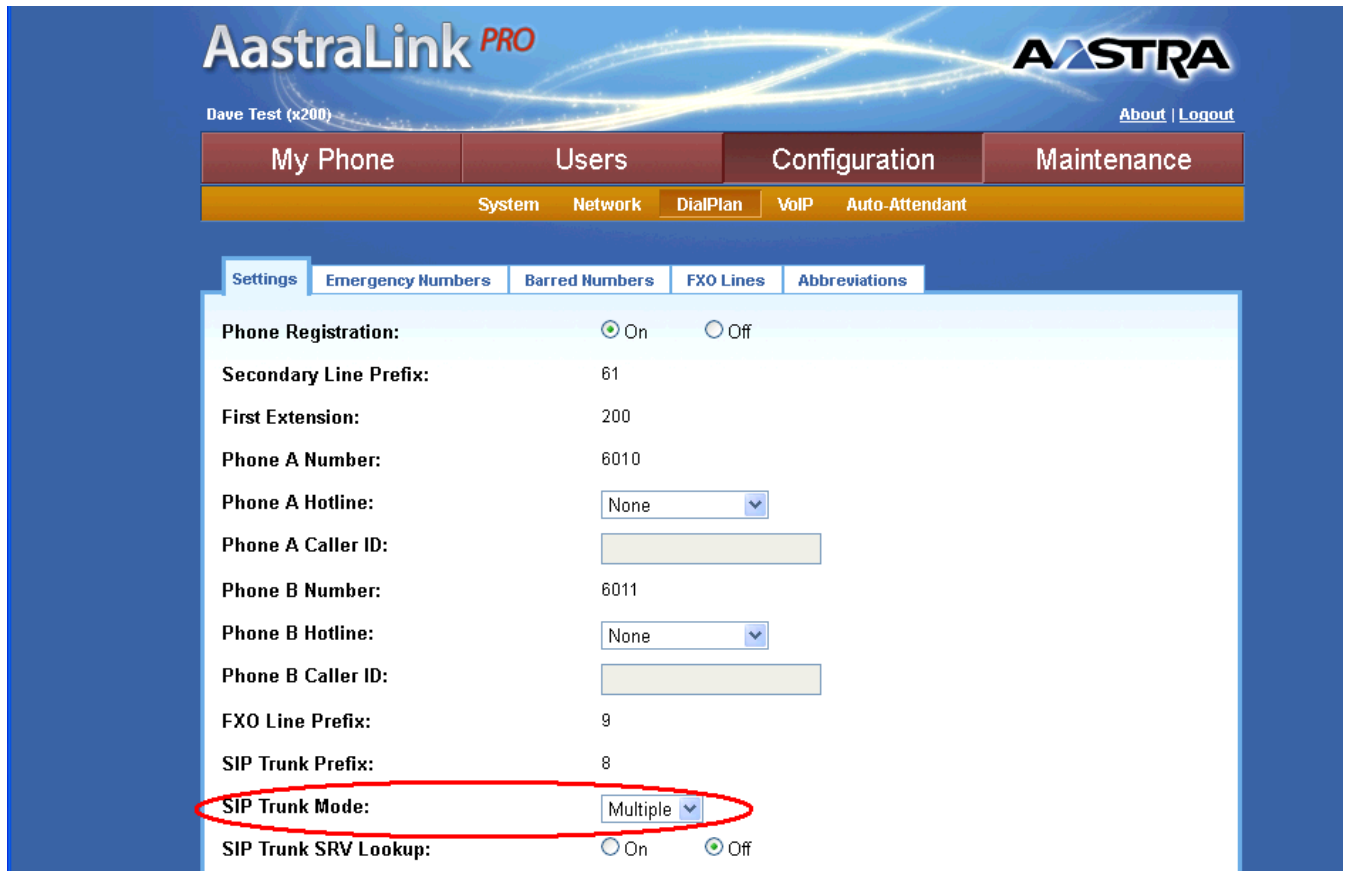
User Enabled:	<input checked="" type="checkbox"/>
Voicemail Enabled:	<input type="checkbox"/>
Rings before Voicemail:	Unlimited
Secondary Extension:	Disabled
SLA Enabled:	<input checked="" type="checkbox"/>
Private Extension:	<input type="checkbox"/>
Extension:	200
First Name:	Dave
Last Name:	Test
Password:	•••
Bypass PIN:	<input type="checkbox"/>
Email:	
Account Type:	Administrator
Operator:	<input checked="" type="checkbox"/>
Outgoing Line:	Any
Phone Type:	Aastra6757i
Phone Firmware Version:	2.5.2.2041-SIP
IP Address:	172.16.0.100
MAC Address:	00:08:5D:20:F7:80
SIP DID Number:	2404983515

Figure 2 Extension/User Configuration

4. Continue this for each extension you wish to have Direct Inbound Dialing for

## Add SIP Trunk

1. Navigate to **Configuration -> DialPlan -> Settings**



The screenshot displays the AstraLink PRO web interface. At the top, the user is logged in as 'Dave Test (x200)'. The navigation menu includes 'My Phone', 'Users', 'Configuration', and 'Maintenance'. Under 'Configuration', there are sub-menus for 'System', 'Network', 'DialPlan', 'VoIP', and 'Auto-Attendant'. The 'DialPlan' sub-menu is selected, and the 'Settings' tab is active. The settings are as follows:

Phone Registration:	<input checked="" type="radio"/> On <input type="radio"/> Off
Secondary Line Prefix:	61
First Extension:	200
Phone A Number:	6010
Phone A Hotline:	None
Phone A Caller ID:	
Phone B Number:	6011
Phone B Hotline:	None
Phone B Caller ID:	
FXO Line Prefix:	9
SIP Trunk Prefix:	8
<b>SIP Trunk Mode:</b>	<b>Multiple</b>
SIP Trunk SRV Lookup:	<input type="radio"/> On <input checked="" type="radio"/> Off

Figure 3 DialPlan Settings

- Configure for Multiple SIP Accounts
  - i. Set the SIP Trunk Mode to be Multiple – this allows up to 10 DID's to be configured on the AstraLink Pro 160

## Configure SIP Accounts

1. Navigate to **Configuration -> VoIP -> SIP Trunking**
2. Configure SIP accounts.
3. Choose the first line **8 + 0 + (number)** so you can configure your first account.

The screenshot shows the AastraLink PRO web interface. At the top, there is a header with the AastraLink PRO logo and the Aastra logo. Below the header, there is a navigation menu with four main categories: My Phone, Users, Configuration, and Maintenance. Under Configuration, there are sub-categories: System, Network, DialPlan, VoIP, and Auto-Attendant. The VoIP category is selected. Under VoIP, there are sub-categories: SIP Trunking, SIP DIDs, AastraLink Trunks, and Mobility Base Units. The SIP Trunking category is selected. The main content area displays a table of SIP Trunking lines. The table has the following columns: Dial Prefix, Local, Keep Alive, Anonymous, Username, Registrar Server, Registrar Port, and Status. The first three lines are registered, while the remaining seven are unregistered.

<input type="checkbox"/>	Dial Prefix	Local	Keep Alive	Anonymous	Username	Registrar Server	Registrar Port	Status
<input type="checkbox"/>	8 + 0 + (number)	off	off	off	2404983515	172.16.253.1	5060	Registered
<input type="checkbox"/>	8 + 1 + (number)	off	off	off	2404983516	172.16.253.1	5060	Registered
<input type="checkbox"/>	8 + 2 + (number)	off	off	off	2404983517	172.16.253.1	5060	Registered
<input type="checkbox"/>	8 + 3 + (number)							
<input type="checkbox"/>	8 + 4 + (number)							
<input type="checkbox"/>	8 + 5 + (number)							
<input type="checkbox"/>	8 + 6 + (number)							
<input type="checkbox"/>	8 + 7 + (number)							
<input type="checkbox"/>	8 + 8 + (number)							
<input type="checkbox"/>	8 + 9 + (number)							

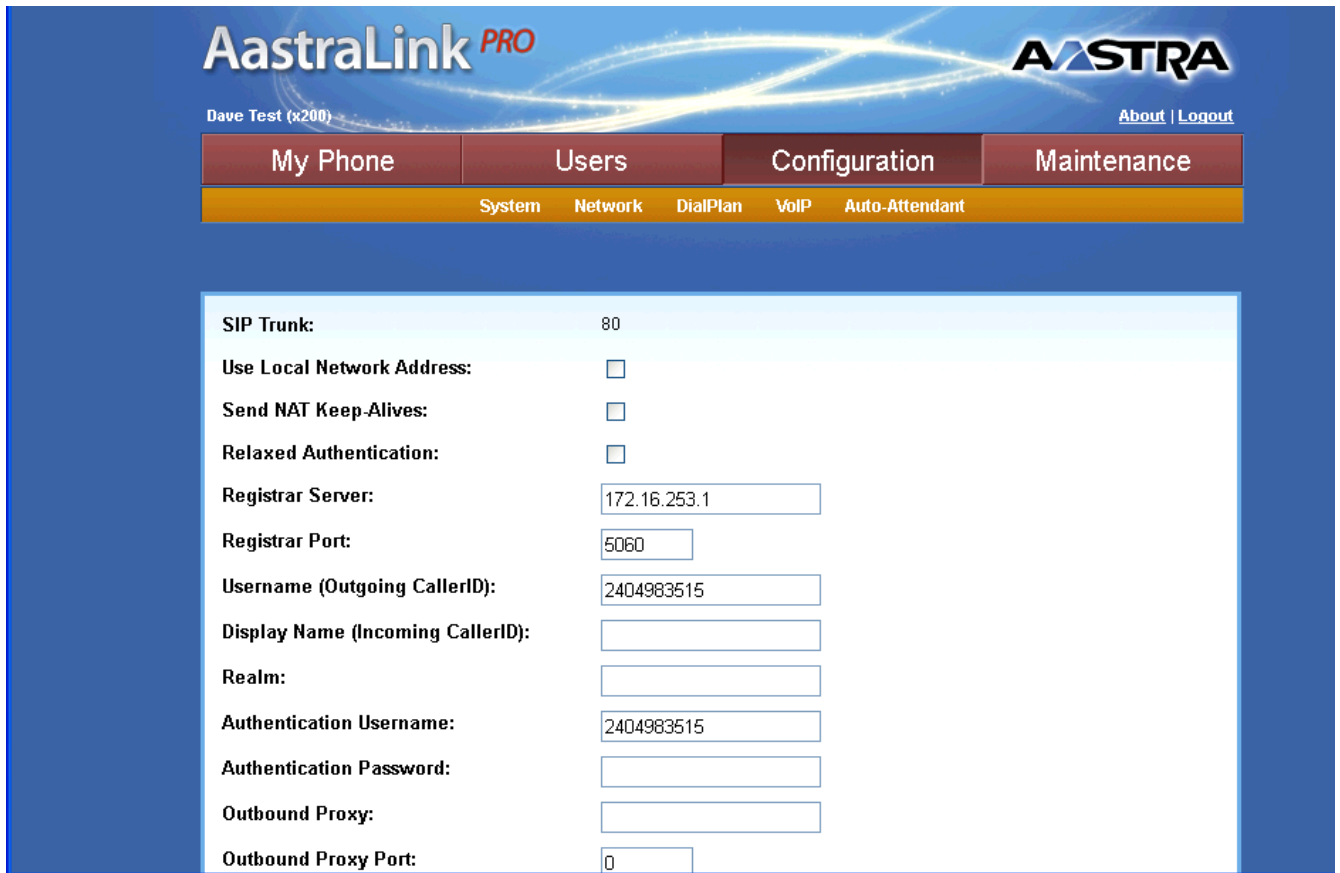
Displaying 1-10 of 10

Delete Selected SIP Trunks

Figure 4 Sip Trunk Line Status



4. Enter the Registrar Server – This will be the ESG LAN IP – For our example here it is 172.16.253.1
5. Enter Username (Outgoing Caller ID) – This is your DID number
6. Enter Authentication Name and if required Password fields – This is the SIP Account Info that the service provider supplied for your accounts.
7. Save the settings



**AastraLink PRO** **Aastra**

Dave Test (x200) [About](#) | [Logout](#)

**My Phone**   **Users**   **Configuration**   **Maintenance**

System   Network   DialPlan   VoIP   Auto-Attendant

<b>SIP Trunk:</b>	80
<b>Use Local Network Address:</b>	<input type="checkbox"/>
<b>Send NAT Keep-Alives:</b>	<input type="checkbox"/>
<b>Relaxed Authentication:</b>	<input type="checkbox"/>
<b>Registrar Server:</b>	172.16.253.1
<b>Registrar Port:</b>	5060
<b>Username (Outgoing CallerID):</b>	2404983515
<b>Display Name (Incoming CallerID):</b>	
<b>Realm:</b>	
<b>Authentication Username:</b>	2404983515
<b>Authentication Password:</b>	
<b>Outbound Proxy:</b>	
<b>Outbound Proxy Port:</b>	0

**Figure 5 Sip Trunk Line Configuration**

## 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.