

**Spectrum Enterprise SIP Trunking Service
SIPfoundry sipXecs
Firmware 4.0-1-051823
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

This document will serve as a reference guide to configure the SIPfoundry sipXecs 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 SIPfoundry sipXecs 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	SIPfoundry sipXecs
Firmware	4.0-1-051823

TWC Network Equipment:

ESG	InnoMedia ESBC 9378-4B
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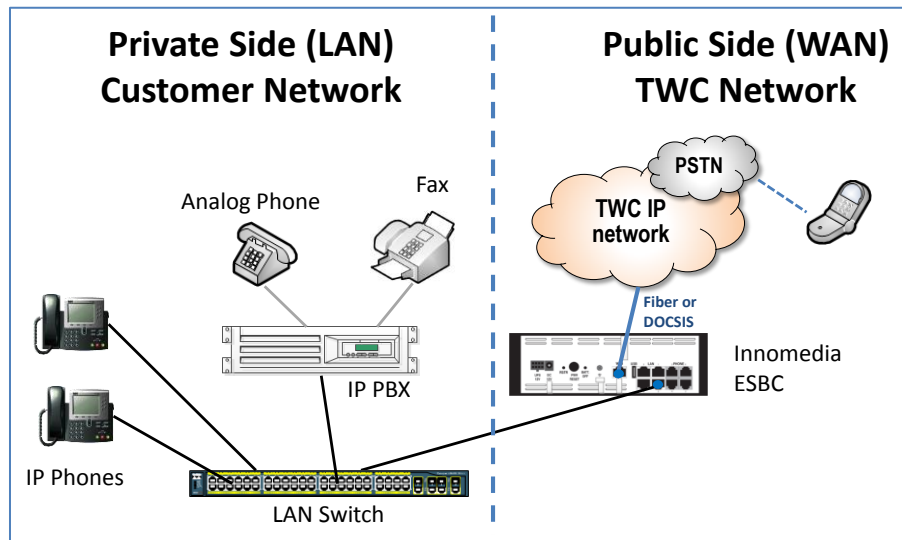
SIP Trunk Components

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

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.

sipXecs Configurations

The instructions provided in this section are intended to help configure the sipXecs to connect to the TWC ESG. They are not intended for advanced functionality setups. It is further assumed that there is already knowledge of sipXecs operations

Once logged into the sipXecs GUI as an Administrator, follow these steps to configure SIP Trunk Service.

Navigate your web browser to the IP address of your sipXecs SIP Server. Log in using the proper credentials.

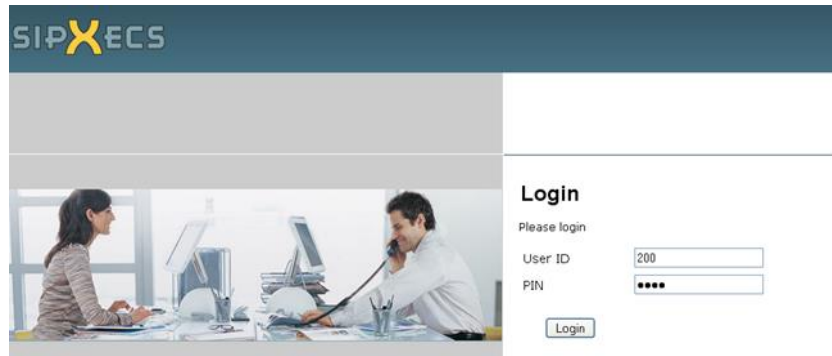


Figure 1 Login

Adding Extensions

1. Navigate to **Users**
2. Select **Add New User** to add your extension.
3. Configure **extension**.
 1. **Alias**: this should be set according the User ID of ESG SIP UA which is assigned to this extension.
4. Click **OK** or **Apply** button.



Figure 2 Add an Extension

Devices	Features	System	Diagnostics
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User: 200

[Hide Advanced Settings](#)

Existing Groups:
administrators

New Groups: You can create new groups simply by adding the new group name to the Groups form value.

Select *Phones* to add this user to one or more phones.

User ID:
The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DNIS portion) as an alias.

Last name:

First name:

Active greeting: Voice mail prompt callers will hear before leaving a message.

E-mail address:
Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.

Attach voicemail:
If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.

Additional E-mail address:
Used for sending voicemail message notification to the additional e-mail address.

Attach voicemail:
If checked, the voicemail message will be attached to the notification email sent to the additional e-mail address.

PIN:

Confirm PIN:
The PIN is a password used to log in to voicemail or to the user portal. Numeric PINs are recommended, since only numbers can be dialed.

SIP password:
The SIP password is used by the user's phone to register with the SIP proxy. For phones managed by this system, the SIP password entered here will be configured automatically on the phone. For unmanaged phones, the SIP password is needed when manually configuring lines on the phone. The security of this password is very important and that is why a secure password is auto-generated.

Groups:

List all groups for this user. If a group does not exist, it will be created. When entering multiple groups, separate them with spaces.

Aliases:
Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with spaces.

Figure 3 Extension Identification

Adding SBC's

1. Navigate to Devices > SBCs.

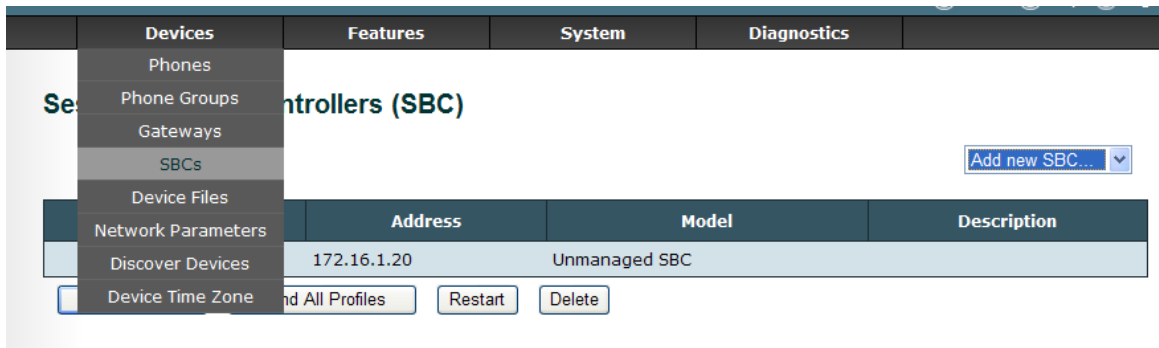


Figure 6 SBCs link

2. Choose **Unmanaged SBC option** from **Add New SBC** drop down list.
3. Setting SBC

Session Border Controller (SBC)

Name

Address

Port (Default: 5060)

Description

Figure 4 Configuring SBC

1. **Address:** this should be set as LAN ip address of ESBC.
4. Click **OK** or **Apply** button.

Adding Gateways

1. Navigate to **Devices > Gateways**.

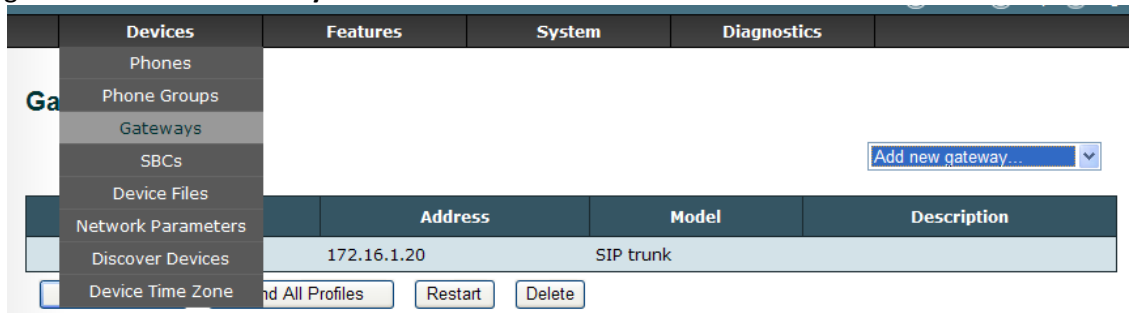


Figure 5 Gateways link

2. Choose **SIP Trunk** from **Add new gateway** drop down list.
3. Setting gateway.

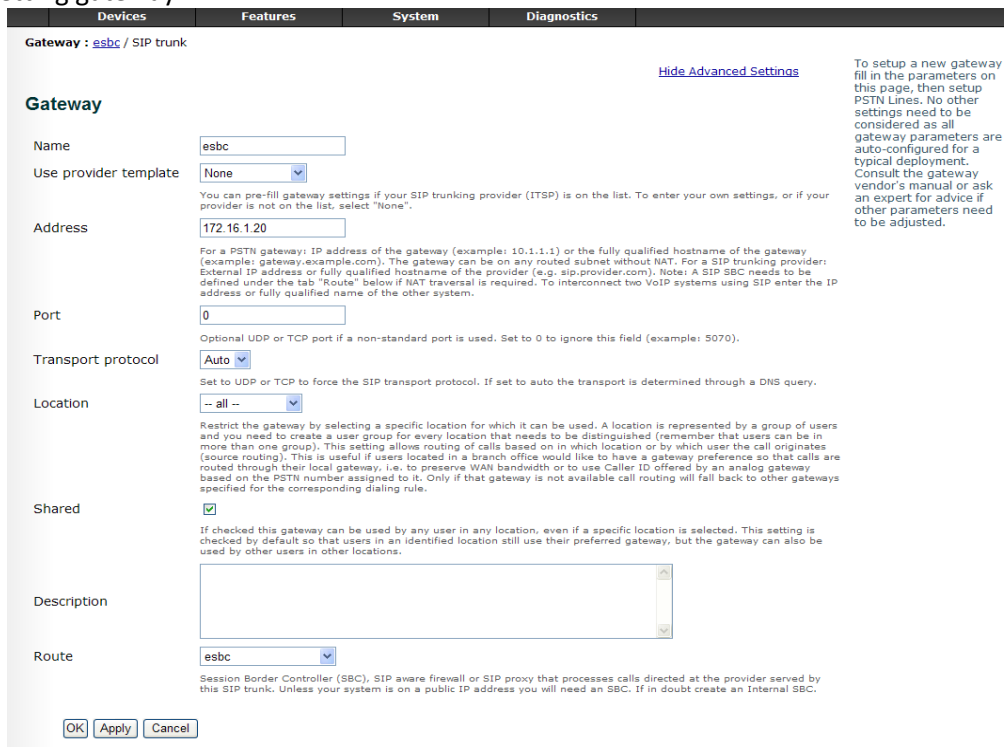


Figure 6 Setting gateway

1. **Address:** this should be set as LAN IP address of ESG.
2. **Port:** this should be set as 5060 except that ESG has changed the port.
3. **Route:** this should be set as the SBC you add before.
4. Click **OK** or **Apply** button.

Adding Dial Plans

1. Navigate to **System >Dial Plans**.
2. Choose **Custom** from **Add New Rule** drop down list.
3. Setting rule

Dial Rule

Enabled

Name

Description

Dial Number

Prefix and [Delete](#)

Prefix and [Add](#) [Delete](#)

Required Permissions

900 Dialing

Attendant Directory

International Dialing

Local Dialing

Long Distance Dialing

Mobile Dialing

Record System Prompts

Toll Free

Voice Mail

Resulting Call

Dial and append

Schedule

Gateways

	Name	Address	Model	Description
<input type="checkbox"/>	esbc	172.16.1.20	SIP trunk	

Prefixes
Prefixes can be integers (e.g. 2), ranges (e.g. 1-5), or lists (e.g. 2,3,4)

Figure 7 Setting Inbound DID Number

4. Check **Enabled**.
5. Enter **Name**.
6. Choose dialed number pattern, and click **Add**.
7. From drop down list **More actions**, choose the gateway you add before.
4. Click **OK** or **Apply** button.

Configuring Gateway

1. Navigate to **Devices > Gateways**.
2. Click the gateway name you add before.

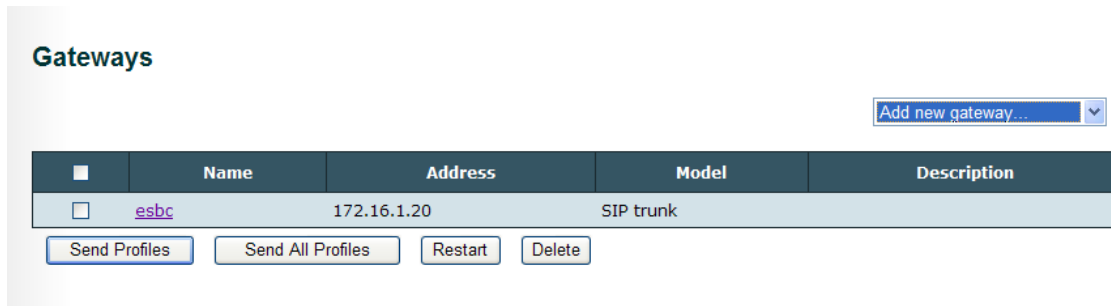


Figure 8 Gateway List

3. Navigate to **Caller ID**.
4. Click **Show Advanced Settings**.
5. Setting Caller ID.



Figure 9 Setting Caller ID

8. Check **Specify Caller ID**.
9. Enter **Caller ID**
The format of caller ID is <User ID of SIP UA>@<LAN IP address of ESBC>
6. Click **OK** or **Apply** button.
7. Navigate to **Dial Plan**.
8. From drop down list **More actions**, choose the dial plan you add before.

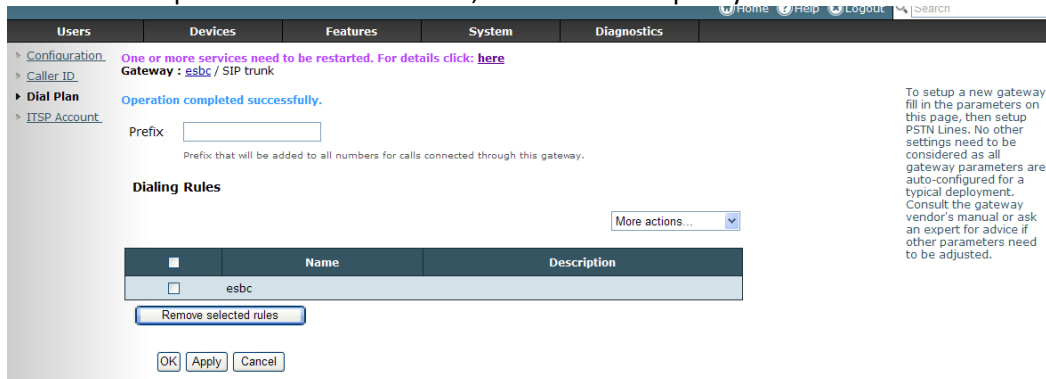


Figure 10 Choose Dial Plan

9. Click **OK** or **Apply** button.

Apply Configuration Changes

You will now be prompted to “One or more services need to be restarted”.

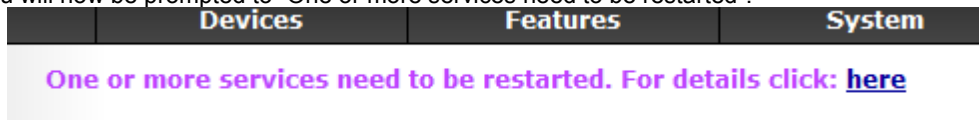


Figure 11 Prompt to restart service

1. Click **here** link
2. Select all servers. Click **Restart**.

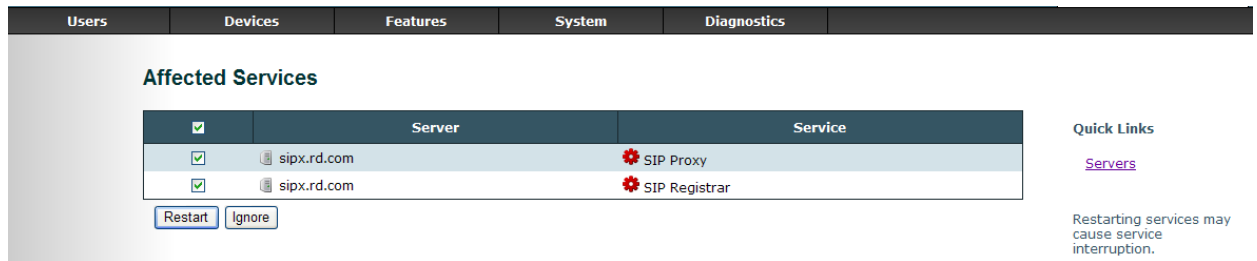


Figure 12 Restart Services

3. After restarting the services. You may now add your SIP endpoint settings (“User Extension”, “secret” and SIP server IP address) to your SIP device and make incoming and outgoing calls.

Appendix

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

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