Spectrum Enterprise SIP Trunking Service
AllWorx 6X R7.5.19.1
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 Charter or Charter Business. All references to Charter should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise

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AllWorx 6X R7.5.19.1 Configuration Guide

1 Introduction

The document describes how to configure the AllWorx 6x Release 7.5.19.1 IP PBX to interoperate within the Charter network. It does not provide any information how to provision, configure or use the features of the IP PBX. Please refer to the documentation provided with the IP PBX or contact the vendor.

2 Configuration

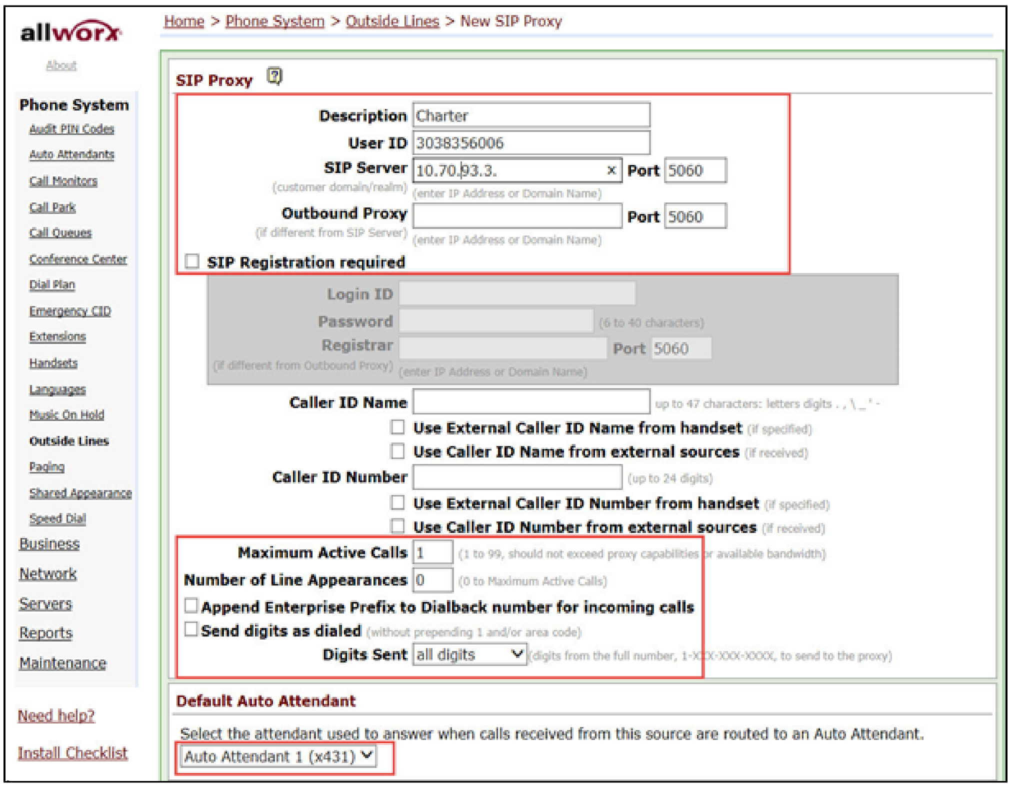
2.1 Creating SIP Trunk Profile

To create a sip trunk profile, follow the step-by-step procedure.

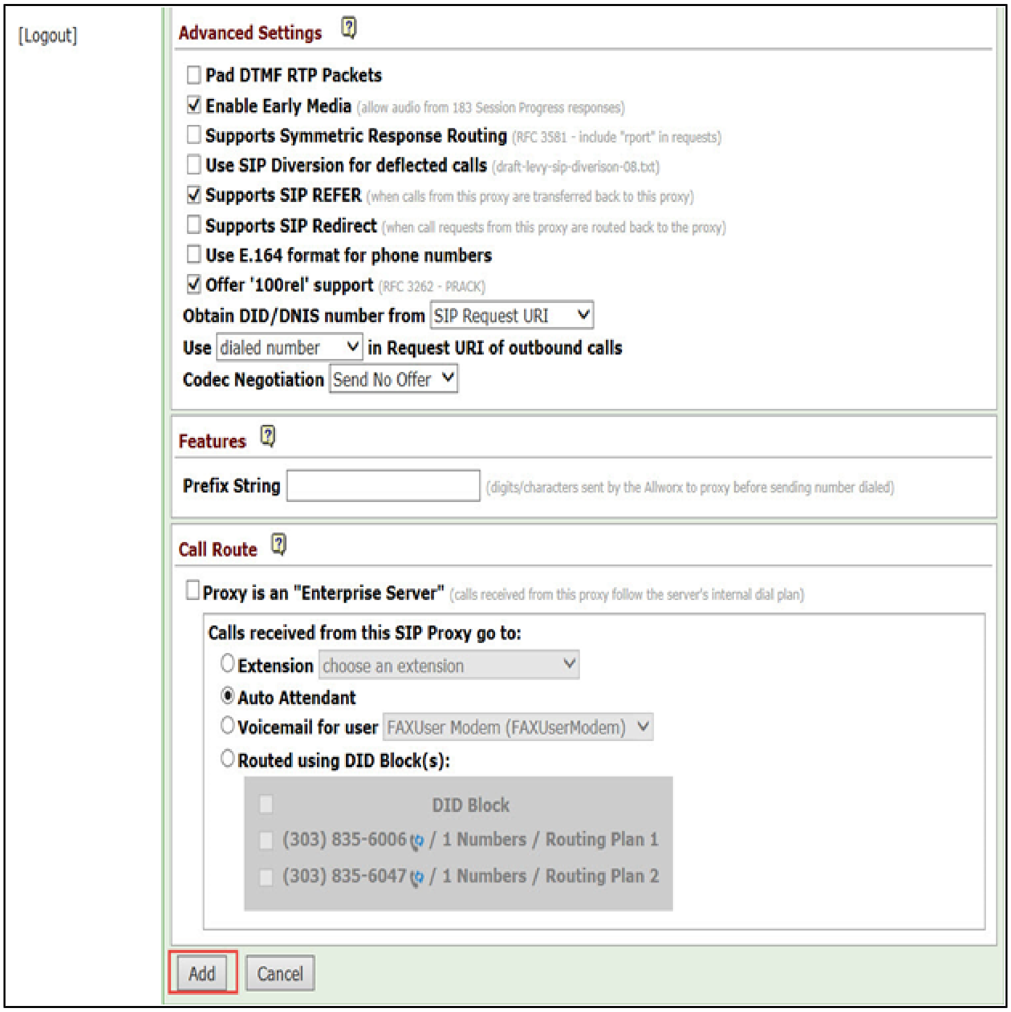
Step	Action	Result
1	Navigate to **Home > Phone System > Outside Lines > New SIP Proxy**	SIP Proxy screen opens
2	Description Enter **Charter**	
3	UserID Enter **3038356006**	
4	SIP Server Enter the **IP LAN address of the eSBC**	Use the actual (e-SBC) LAN IP for the network.
5	Port Enter **5060**	
6	Outbound Proxy Confirm the text box is blank (no data)	
7	SIP Registration required Check box is **unchecked**	
8	Maximum Active Calls Enter **10**	
9	Number of Line Appearances Enter **10**	
10	Send digits as dialed Check box is **unchecked**	

11	Digits Sent Select **all digits**	
12	Default Auto Attendant Select the appropriate **response**	For this example 1 (x432) is used
13	Click the **Add** button	
14	Procedure completed	

Add SIP Profile



Add SIP Profile

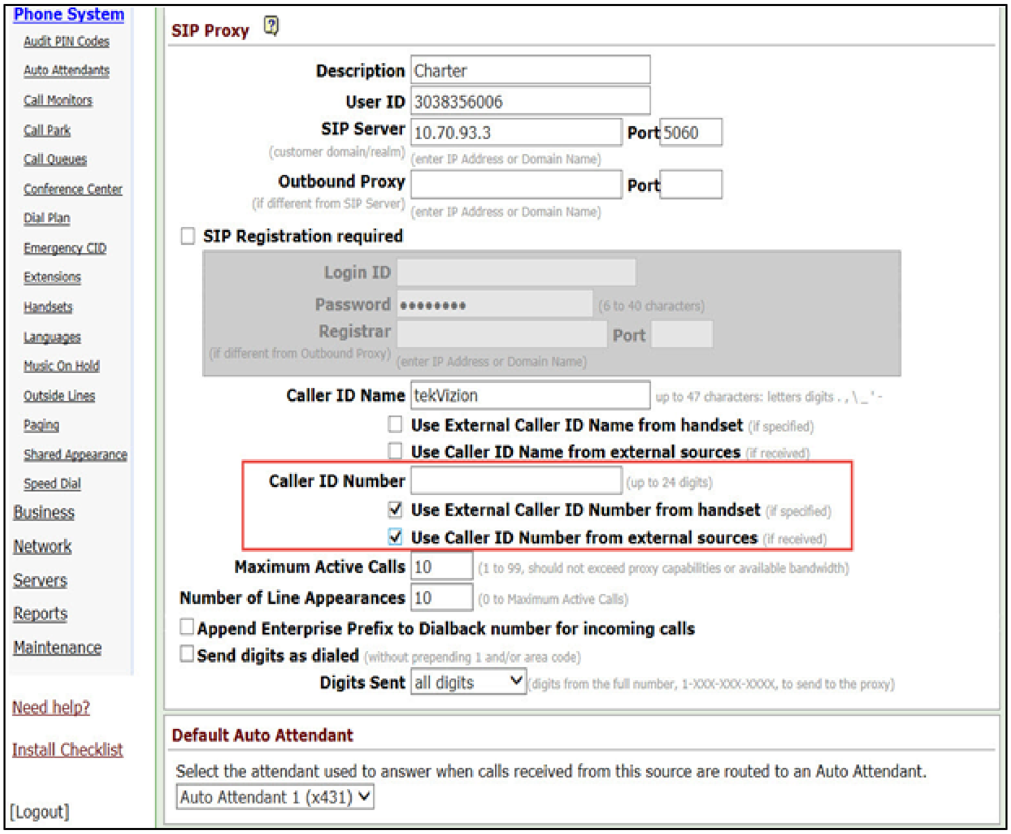


2.2 Modifying SIP Trunk

To modify a sip trunk, follow the step-by-step procedure.

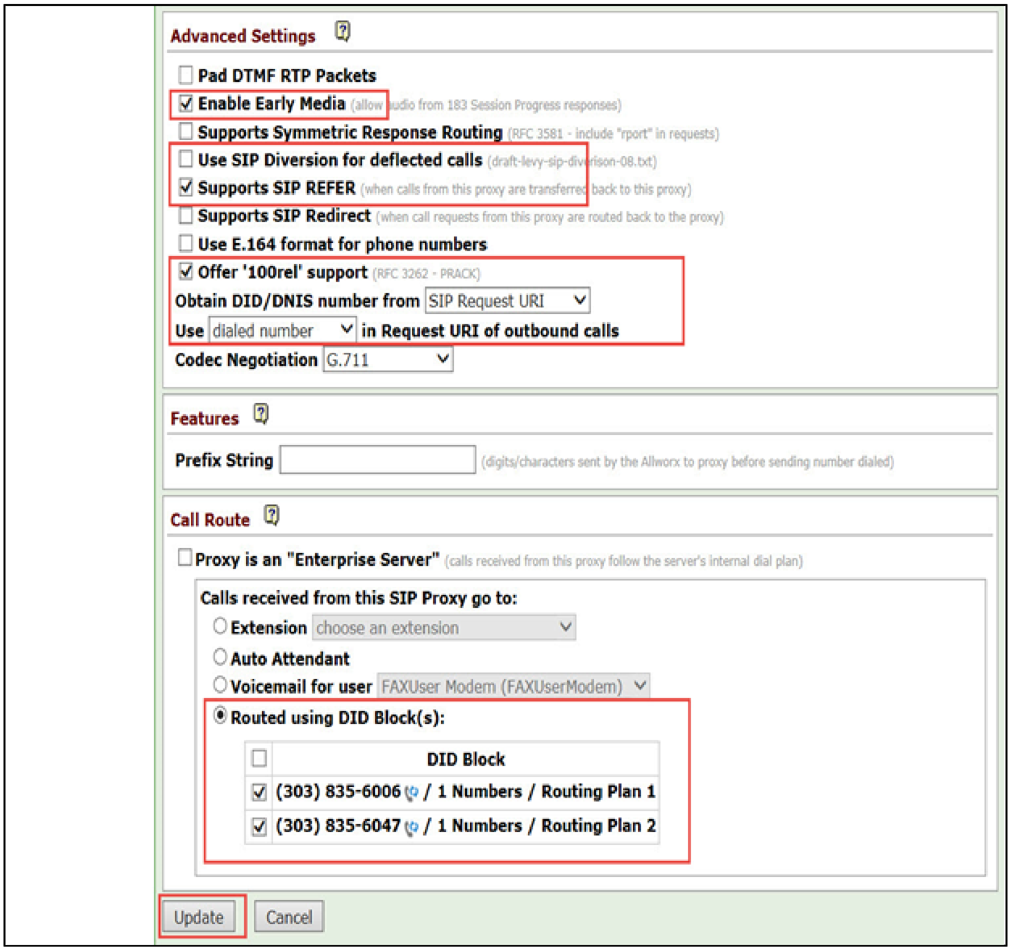
Step	Action	Result
1	Navigate to **Home > Phone System > Outside Lines**	
2	At SIP Proxies Action section Click on **Modify**	
3	In the SIP Proxy section under Caller ID Number: • Check **Use External Caller ID Number from handset** • Check **Use Caller ID Number from external sources**	
4	Go to the next table	

SIP Trunk Modifications SIP Proxy



Step	Action	Result
5	In the Advance Settings section, confirm the following is **checked**: • Enable Early Media • Use SIP Diversion for deflected calls • Supports SIP REFER • Offer '100rel' support • Obtain DID/DNIS number from 'SIP to: header field' • Use 'dialed number' in Request URI of outbound calls	
6	In the Call Route section, confirm the following is **checked**: • Routed using DID Blocks	**Note:** If the DID range is not created you may have to come back to this window to check the DID group.
7	Click **Update** button	
8	Procedure completed	

Trunk Modifications Advanced Settings


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-diversion-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 Request URI
Use dialed number in Request URI of outbound calls
Codec Negotiation G.711
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 choose an extension
Auto Attendant
Voicemail for user FAXUser Modem (FAXUserModem)
Routed using DID Block(s):
DID Block
(303) 835-6006 / 1 Numbers / Routing Plan 1
(303) 835-6047 / 1 Numbers / Routing Plan 2
Update
Cancel

TOP