TECHNICAL NOTE

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HOW TO CONFIGURE ALLOYVOICE SIP TRUNKS ON EPYGI QX PBX SERIES

1. Introduction

This Technical note will go through information on how to setup AlloyVoice on Epygi QX PBX series, as well as general information on the SIP Protocol.

SIP trunks are a VoIP service that can be provided from an ITSP (Internet Telephony Service Provider) to extend telephony features beyond IP-PBX local area. SIP trunks can carry voice calls, video calls, multimedia conferences, and other SIP-based, real-time communications services.

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2. Requirements

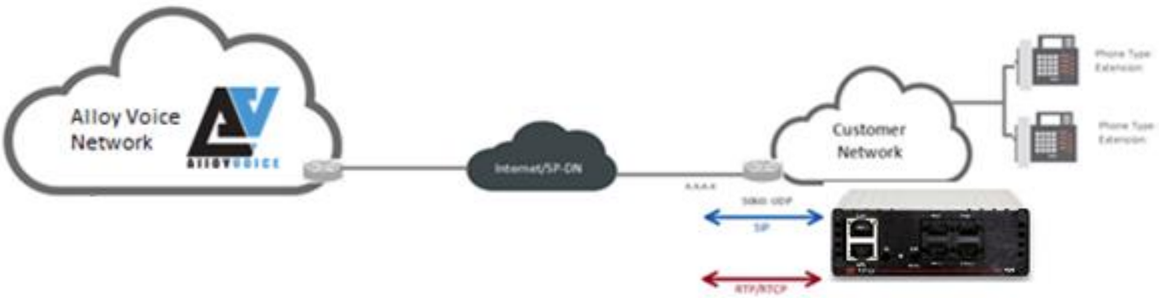
Before configuring the Epygi QX PBX, the following minimum requirements should be considered.

Prerequisites

- Internet/Network Services
- It is also highly recommended that you have a static IP address. If your external IP changes intermittently, inbound calls will fail.
- A firewall/router/NAT device that supports static port mapping.
- Open the following ports to allow Epygi QX PBX to communicate with the Alloy Voice SIP Trunk :
    - Port 5060 (UDP/TCP) for SIP communications
    - Port 6000-6255 (UDP) for RTP/RTCP communications.

3. Epygi PBX Architecture Examples

Network Scenario - On-Premises Deployment



3.1 SIP ALG

To maximize your chances of success, make sure you choose a device that does not implement a SIP Helper or SIP ALG (Application Layer Gateway), or choose a device on which SIP ALG can be disabled.

3.2 Bandwidth Requirements

VoIP is in real time, so it does place a demand on your Internet connection.
Example: Call Using G711 Codec
Each RTP packet contains 20ms of audio (typical).
Each 20ms of audio requires 160 bytes.
Each second of audio will require 50 packets, each containing an audio payload of 160 bytes.

Before being transmitted over the network the IP packet will contain:
160 bytes for the audio payload
12 bytes for the RTP header
8 bytes for the UDP header
20 bytes for the IP header
= for a total of 200 bytes per packet

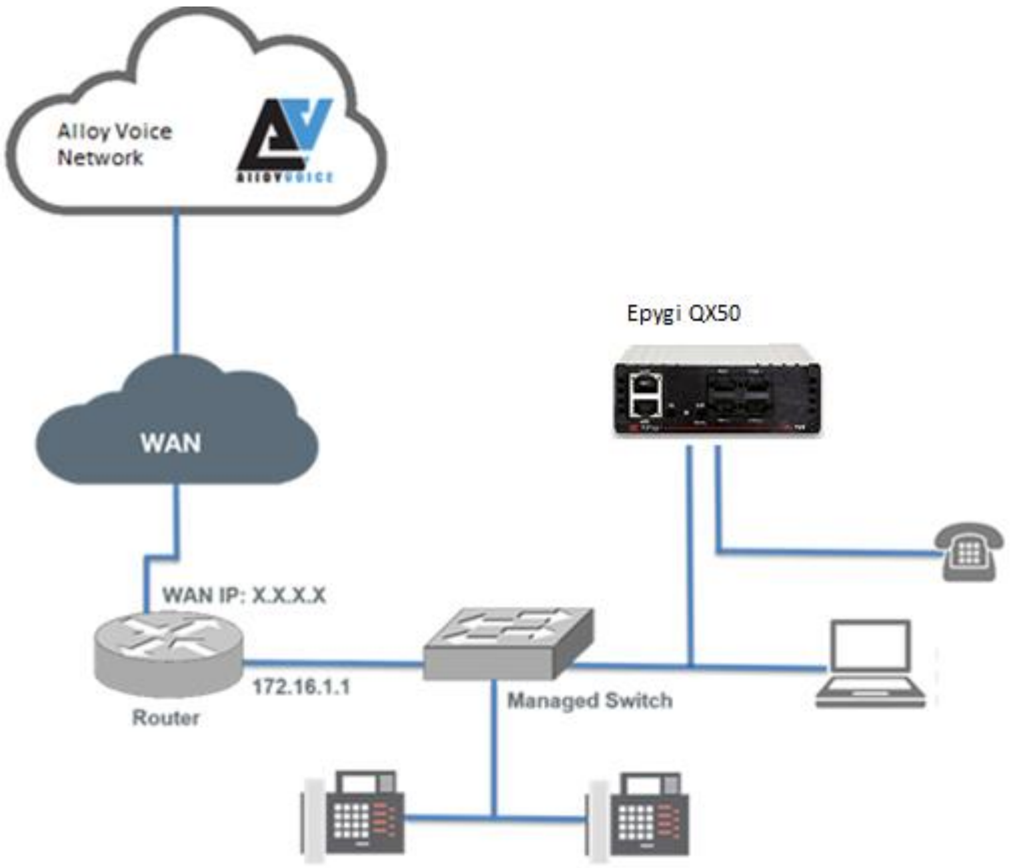
If the transmission medium is Ethernet, the IP packet is encapsulated in an Ethernet Frame which adds an 18-byte header, for a total of 218 bytes per frame * 50 packets per second * 8 bits per byte.

This equates to 87,200 bits per second or 87.2 kbps (RTP Call Only), to include message overhead, add +5%.

4. Connecting to the Epygi QX Series PBX.

Please follow the below steps to connect the Epygi QX Series.

- Connect the WAN Port of the Epygi QX Series Phone System to the Local Area Network.
    - *Note: For Initial Set-up be sure to utilise the WAN port to your network to receive an IP Address via DHCP. Alternately you can connect to the LAN port assign a static IP address of 172.30.0.50 on your Network card. The IP address of the LAN port on a default QX PBX is 172.30.0.1*
- Power-on the Epygi QX PBX.
- Connect the computer to the same local area network to which the Epygi PBX WAN port is connected to, alternately the LAN side if you would prefer to connect to the PBX via this method.



5. Logging into the Web Management Console

The Epygi PBX provides a web console to monitor and configure the system parameters.

Please follow the following steps to login to the Epygi QX PBX.

- Open a web browser such as Google Chrome
- Enter in the IP Address of the Epygi QX, EG 192.168.1.250. https://192.168.11.250
- Enter in the username and password. If you have not logged in or changed the password before this will be;
  Username: admin
  Password: 19
  Note*: Please update this password ASAP as 19 is not a secure password.*

After successfully logging in, you will be presented with the Epygi QX Management Dashboard which will show you the current active calls, firmware version and who is logged into the management console.

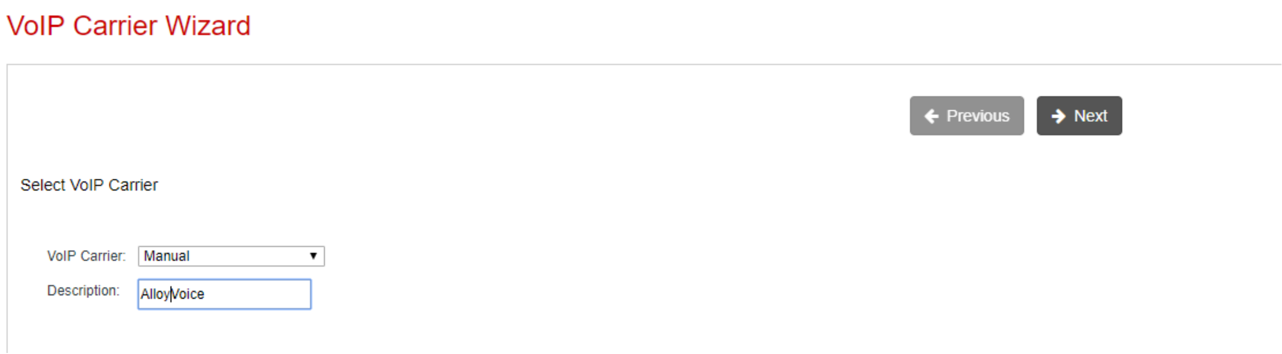
6. Configure SIP Trunk

In this section we discuss the configuration of the Alloy Voice SIP VoIP Trunk through the Epygi web console.

6.1. Add New SIP Trunks

To add a new SIP trunk on the Epygi PBX, follow the steps below.

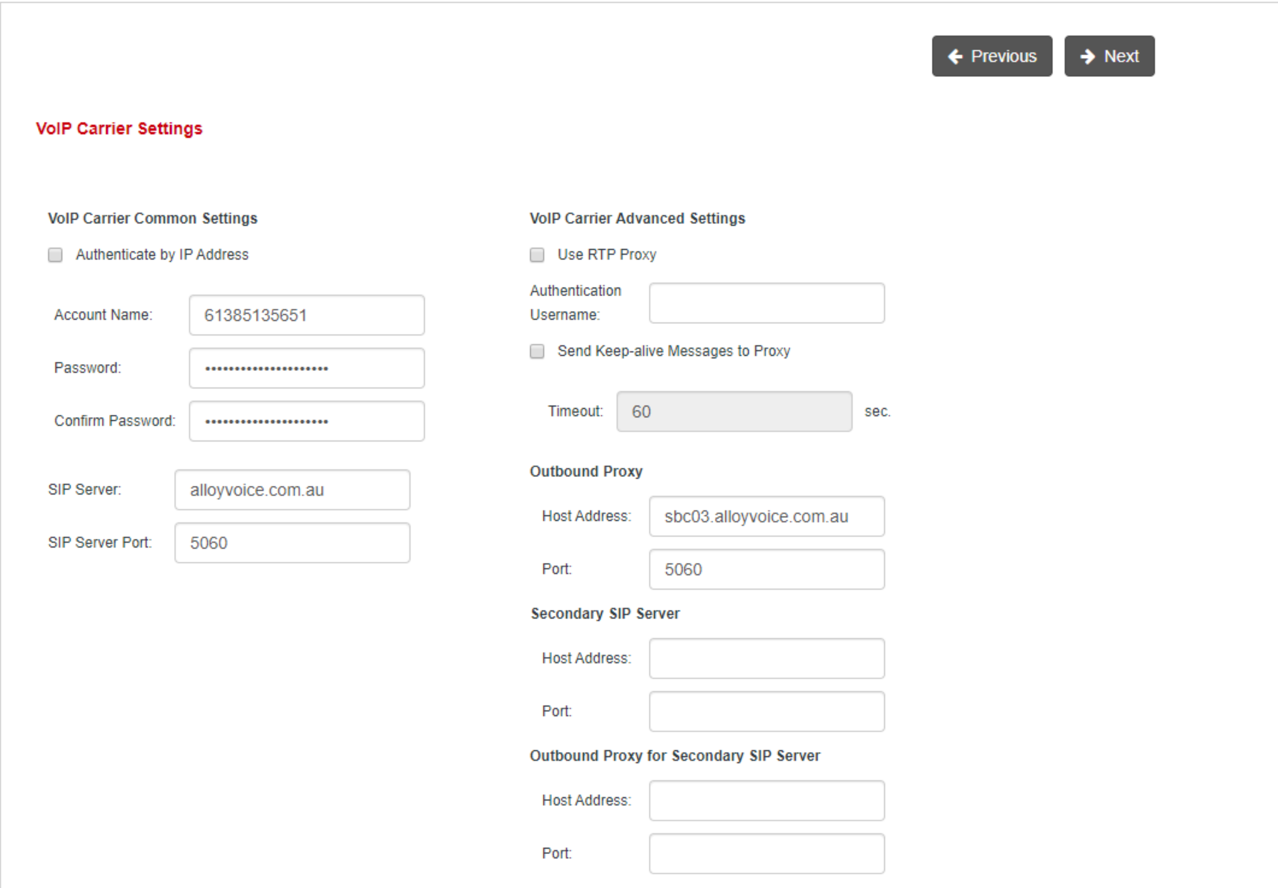
- Select **Telephony** from the left hand side menu
- Select the **VoIP Carrier Wizard** tab
- For VoIP Carrier Select Manual
- Under **Description** Add AlloyVoice SIP Trunk
- Select **Next**



Under VoIP Carrier Settings;

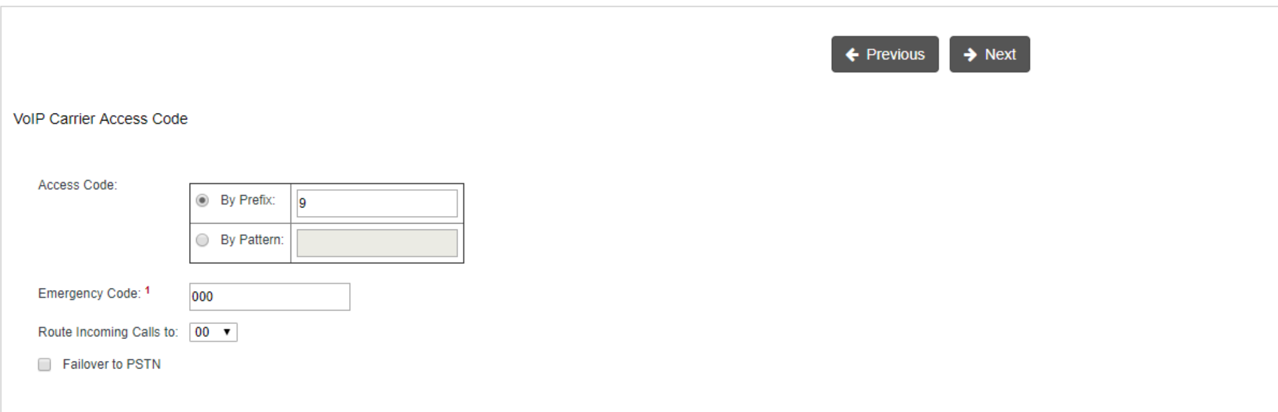
- **Account Name:** AlloyVoice SIP Trunk
- **Account Password:** Your AlloyVoice SIP Trunk Password
- **Confirm Password:** Your AlloyVoice SIP Trunk Password
- **SIP Server:** alloyvoice.com.au
- **Sip Server port:** 5060
- **Outbound Proxy:** sbc03.alloyvoice.com.au
- **Port:** 5060

Select **Next**



- **Access Code:** Select By Prefix and enter in the prefix you wish to use. EG 9.
  Note: you can remove this prefix later if you do not want to have a prefix for outgoing calls
- **Emergency code:** Enter in 000
- Select **Next**

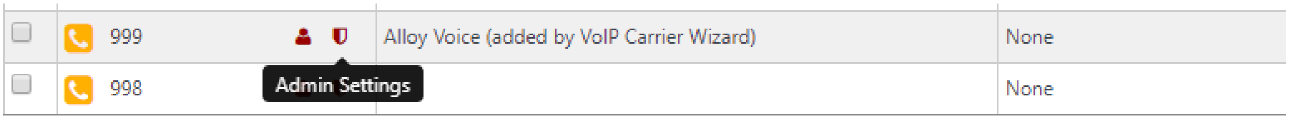
VoIP Carrier Wizard



- Select **Finish** to save the VoIP Trunk

To Make any changes to the SIP trunk follow the below steps;

- Click the **Extensions** menu from the left hand side
- Select the **Extensions** Tab at the top of the screen
- Scroll down to extension 999 and click the **Admin Settings** icon



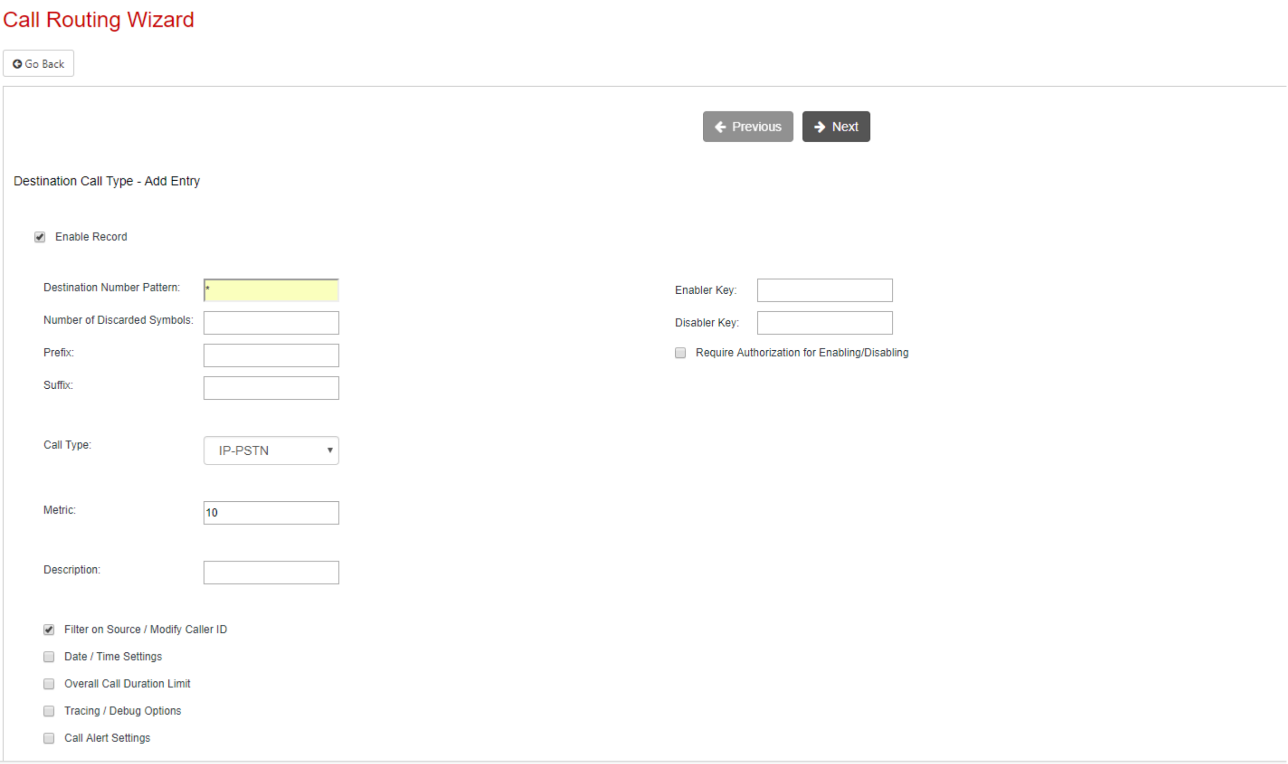
7. Configure Outbound Routes

In this section we discuss the configuration of enabling outgoing calls for extensions on the Epygi QX PBX to route through the AlloyVoice SIP Trunk.

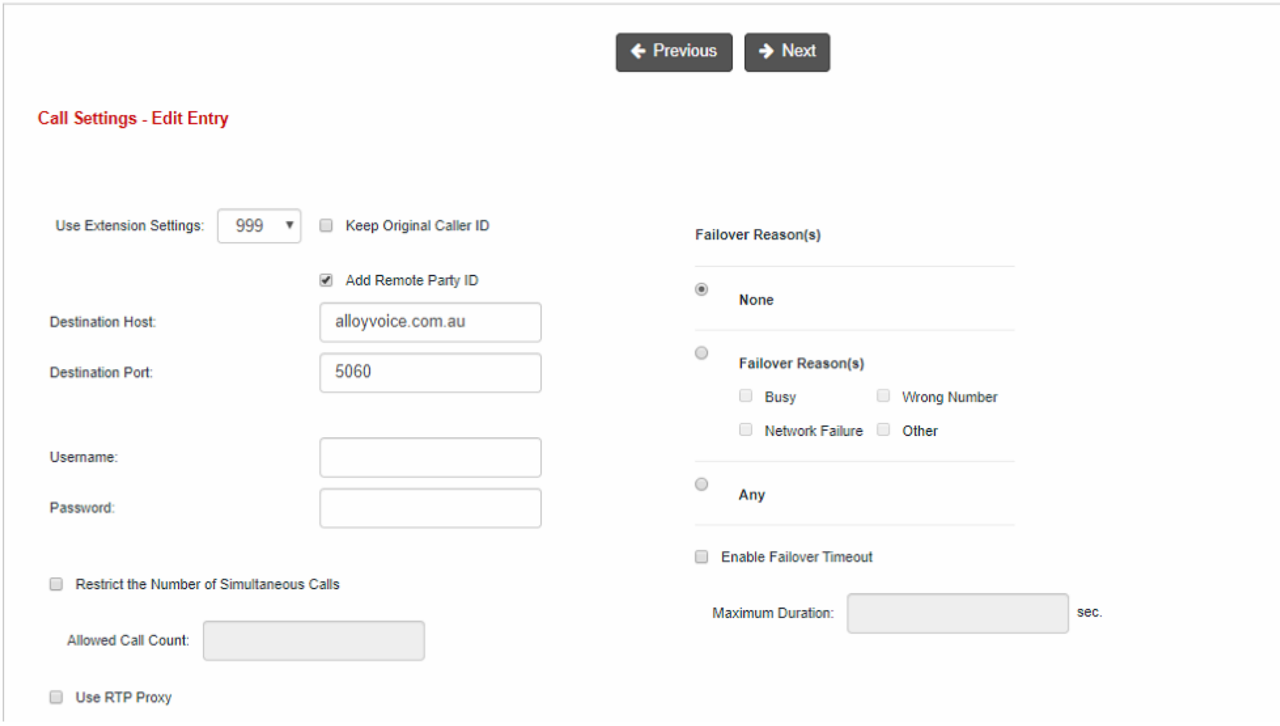
- Select **Telephony** from the left hand side menu
- Select the **Call Routing Table** under the call routing heading
- Click the **+Add** Button

Under the Call Routing Wizard enter in the following

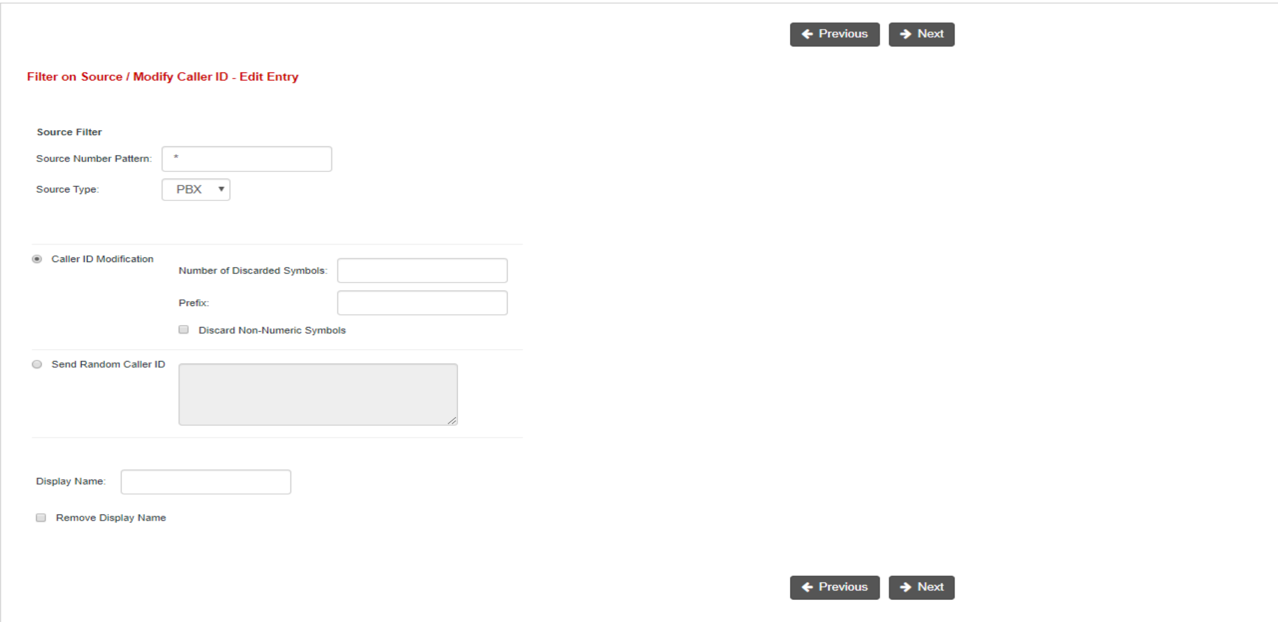
- **Destination Number Pattern**: Enter an asterisk (this means any call dialled) you can change this if you only want to route certain calls out via this Rule. EG Landline, mobile, international.
- **Call Type:** Select IP-PSTN from the drop down menu
- **Click** Next



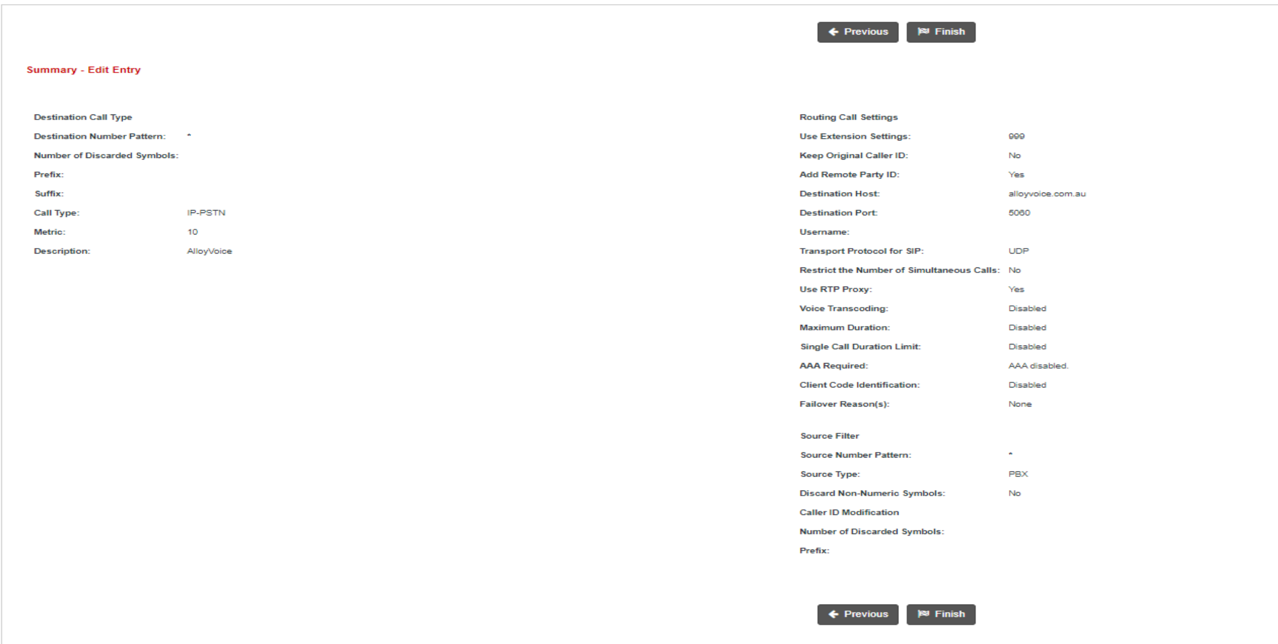
- **Destination Host:** alloyvoice.com.au
- **Destination Port: 5060**
- **Keep Original Caller ID:** If selected, the called destination will receive the original caller's information. (use this option if you want to use the DID of the extension as the outbound caller ID)
- **Add Remote Party ID: Yes**
- Click **Next**



- **Source Number Pattern:** Enter an asterisk. If you would like to restrict this to certain extensions, enter the extension numbers in here.
- **Source Type**: PBX
- Caller ID Modification
  - **Number of Discarded Symbols:** optional – If you would like to add a caller ID on outbound calls with this route, enter in 99 here.
  - **Prefix:** Enter the number you would like to present on outbound calls. Note: if you enter in a number that is not on your account the call route will fail and outbound calls will not work.
- **Click** Next



- On the next screen select **Finish**



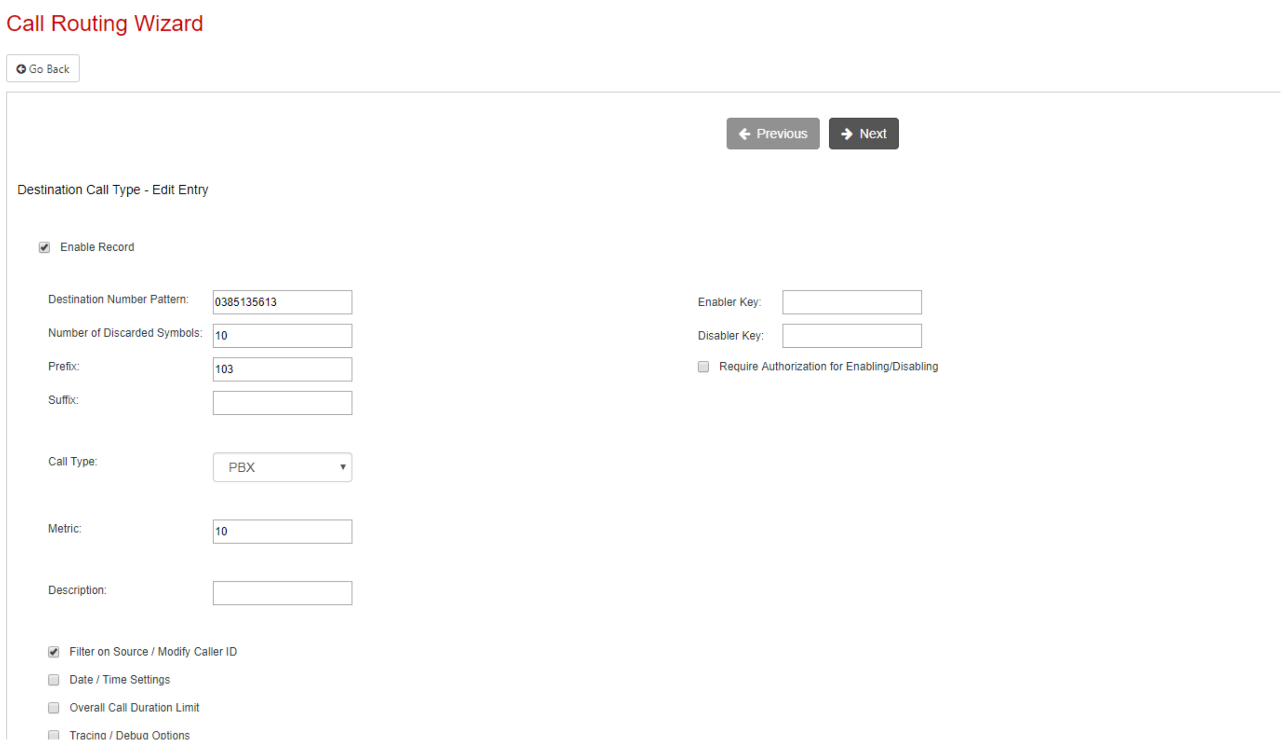
8. Configure Inbound Routes

In this section we discuss the configuration of enabling Inbound calls for extensions on the Epygi QX PBX to route through the AlloyVoice SIP Trunk.
The inbound route we are creating will be for the main number 0385135613 to route to extension 103.

- Select **Telephony** from the left hand side menu
- Select the **Call Routing Table** under the call routing heading
- Click the **+Add** Button

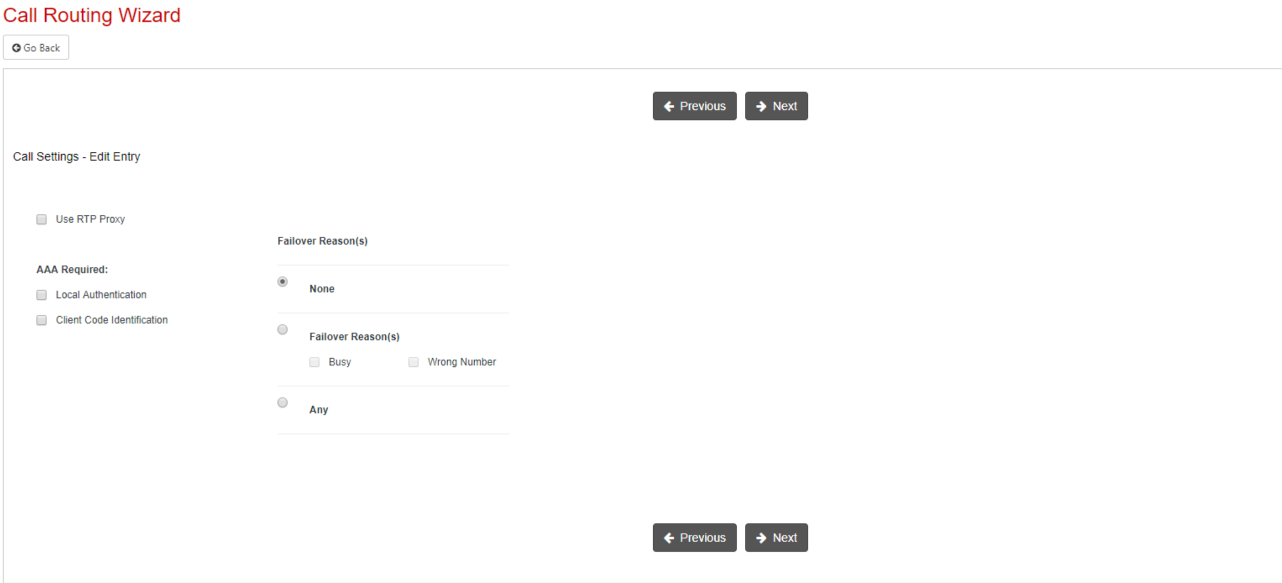
Under the Call Routing Wizard enter in the following

- **Destination Number Pattern**: Enter in the Inbound number that you wish to assign to this rule. In the below example we are using 0385135613.
- **Number of Discarded Symbols**: Enter in 10
- **Prefix**: Enter in the extension you wish to route the call to. EG 103
- **Call Type:** Select PBX from the drop down menu
- Tick **Filter on Source / Modify Caller ID**
- **Click** Next



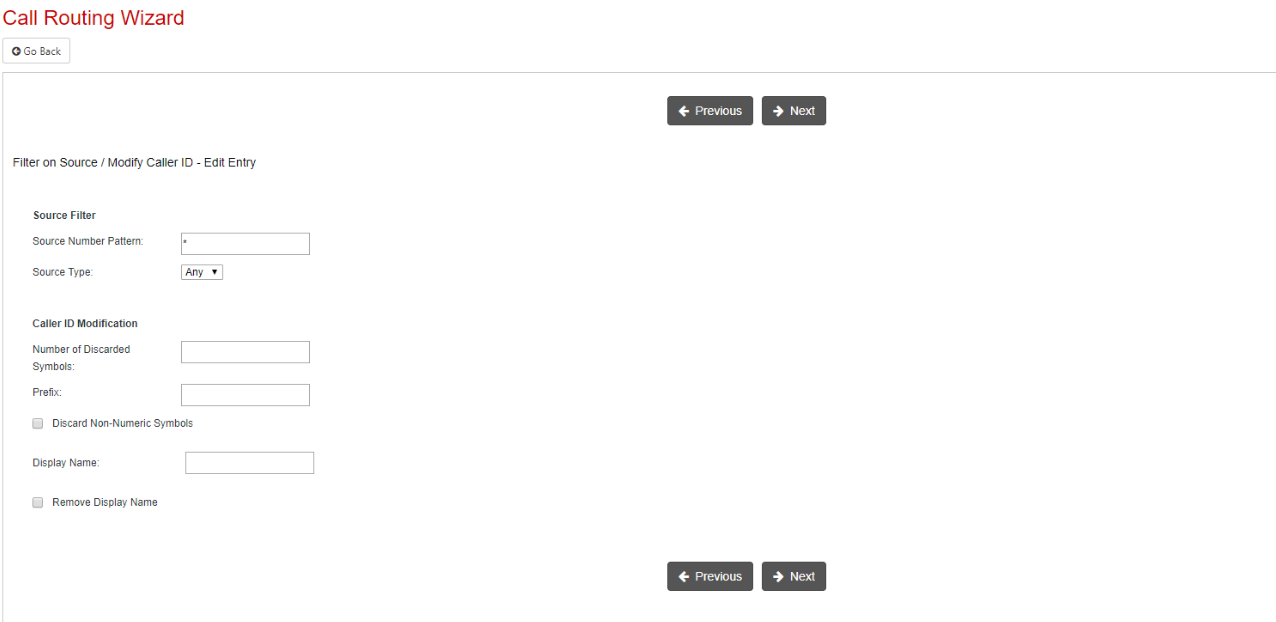
For Call Settings – Edit Entry

Leave this page as default and click **Next**



For Filter on Source / Modify Caller ID

- **Source Number Pattern**: Enter an asterisk
- Click **Next**



- Select **Finish**

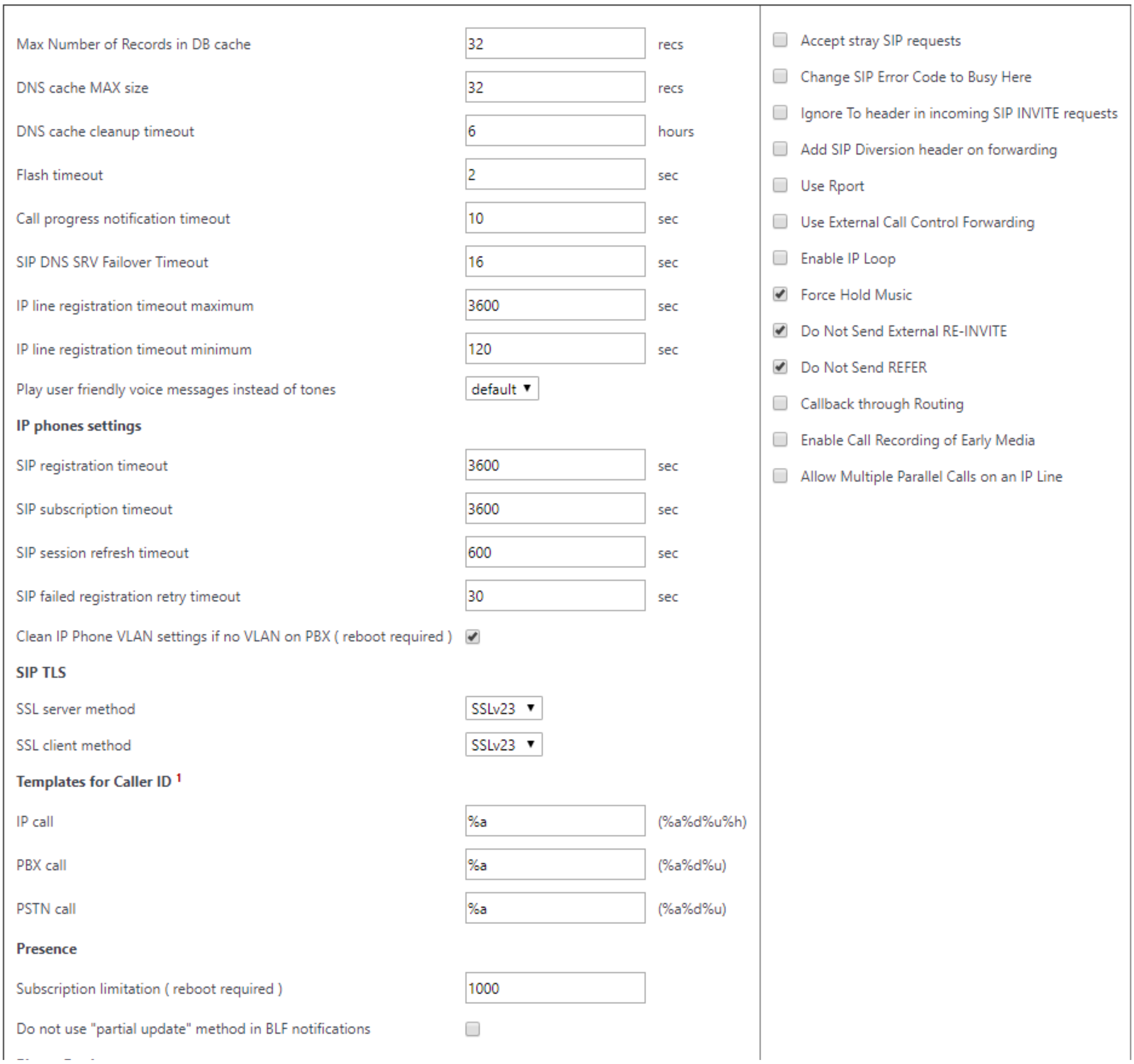
9. Epygi QX SIP changes

Before testing the SIP trunk you need to make some modifications to the Epygi's SIP parameters or there will be issues with transfers and hold music. Please follow the below steps to make the changes.

- The General Configuration Page is not found through the menus and must be navigated to via modifying the web address. If your Epygi PBX is on 192.168.1.200 for example, you need to enter in the following; https://192.168.1.200/generaconfig.cgi
- Uncheck use **Rport**
- Uncheck **Add SIP Diversion header on forwarding**
- Check **Force Hold Music**
- Check **Do Not Send External RE-INVITE**
- Check **Do Not Send REFER**
- Select **Save**
- **Reboot** the PBX after making these changes.

A Screenshot of how the general config page should look is on the next page.

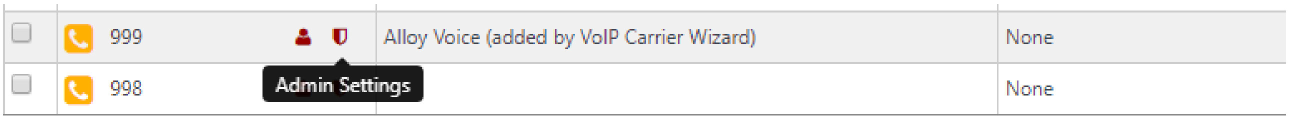
General Configuration



10 . Selecting the correct codec

AlloyVoice Only supports G711a voice codec, so this must be set correctly on the SIP trunk that has been recently created.
To make these changes please follow the steps below

- Click the **Extensions** menu from the left hand side
- Select the **Extensions** Tab at the top of the screen
- Scroll down to extension 999 and click the **Admin Settings** icon



- Select **Go To Codec Settings**
- Disable each codec except the G.711a by clicking the checkbox and **Enable/Disable**
- Enable **Out of Band DTMF Transport**
- Select **Save**

Extension 999 Codecs

Go Back

Enable/Disable | Move Up | Move Down | Make preferred

	Audio Codecs	State
☐	**G.711a (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)**	**Enabled**
☐	G.711u (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)	Disabled
☐	G.726-16 (ADPCM speech coding at 16 kbit/s rate)	Disabled
☐	G.726-24 (ADPCM speech coding at 24 kbit/s rate)	Disabled
☐	G.726-32 (ADPCM speech coding at 32 kbit/s rate)	Disabled
☐	G.726-40 (ADPCM speech coding at 40 kbit/s rate)	Disabled
☐	G.729a (CS-ACELP speech coding at 8 kbit/s rate)	Disabled
☐	iLBC (internet Low Bit Rate Coder at 13,33 kbit/s rate)	Disabled
☐	G.722 (HD audio coding at 48-64 kbit/s data rate, 16 kHz sample rate)	Disabled
☐	G.722.1 (HD audio coding at 24-32 kbit/s data rate, 16 kHz sample rate)	Disabled
☐	TDVC (Time Domain Voicing Cutoff at 1,95 kbit/s rate)	Disabled
	Video Codecs	**State**
☐	H.263 (Video coding for low bit rate communication)	Disabled
☐	H.264 (Advanced video coding for low bit rate communication)	Disabled
☐	H.263+ (Video coding for low bit rate communication)	Disabled

- [x] Out of Band DTMF Transport
- [ ] Enable T.38 FAX
- [ ] Enable Pass Through FAX
- [ ] Enable Pass Through Modem
- [ ] Force Self Codecs Preference for Inbound Calls

10. SIP Trunk status

To check if the SIP trunk is registered, follow the steps below

- Click **Status** from the left hand side menu
- Select **SIP Registration**

Under Registration of SIP Servers this will show the status of each SIP trunk connected to the Epygi QX PBX. Under Registered this will either be Yes or No.

SIP Registration Status

Registration on SIP Servers

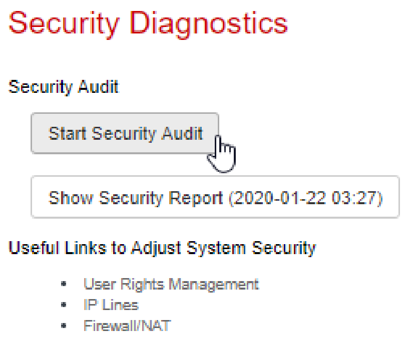
Extension	Username/DID Number	SIP Server	Registered	Registration Time
999	0385135613	alloyvoice.com.au	Yes	13-Jun-2018 10:21:21

11. Security Audit

Security Audit is a security reporting system, which generates the warnings regarding the Epygi PBX's weaknesses relative to the selected Security Level.

Follow the steps below to perform a security audit

- Click **Maintenance** from the left hand side menu
- Go to **Diagnostics → Security Diagnostics**
- Select **Start Security Audit**



- Select **Show Security Report** to see the report

Security Scanning Report

Security level is set to Medium Date: 22-Jan-2020 10:29
Firewall and IDS
No security flaw is found
IP Lines passwords
No security flaw is found
Call Routing
No security flaw is found
Extensions
No security flaw is found
Users
No security flaw is found