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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 :
  - o Port 5060 (UDP/TCP) for SIP communications
  - o Port 6000-6255 (UDP) for RTP/RTCP communications.

## 3. Epygi PBX Architecture Examples

### Network Scenario - On-Premises Deployment



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## 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

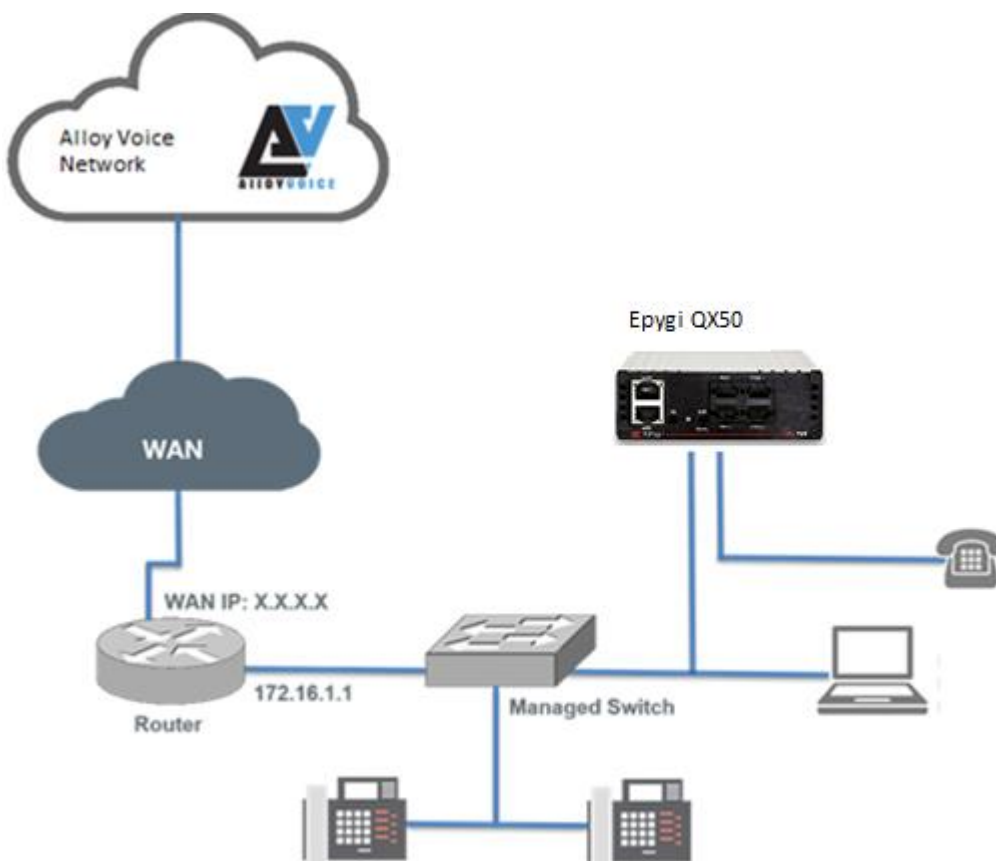
If the transmission medium is Ethernet, the IP packet is encapsulated in an Ethernet Frame which adds an 18-byte header, for a total of 218 bytes per frame \* 50 packets per second \* 8 bits per byte.

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.



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## 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.



Welcome to Epygi QX50  
Please log in to get access.

 Log In

Username / Extension:  Password:

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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.

The screenshot shows the Epygi QX50 Management Dashboard. The top navigation bar includes the Epygi logo, the date and time (Tue, 12-Jun-2018 12:47 AEST), a 'Go To Extension' dropdown, a 'Viewing Events' indicator, and the user 'Administrator (admin)' with a 'Log Out' button. The main content area is titled 'Epygi QX50 Management' and features a sidebar with navigation options: Dashboard, Setup, Extensions, Interfaces, Telephony, Firewall, Network, Status, and Maintenance. The 'Active Calls' section is currently empty, displaying a table with columns for 'Call Start Time', 'Call Duration', 'Calling Phone', and 'Called Phone', and a message 'No items in list.' To the right, two informational boxes are visible: 'Firmware Version: 6.2.6/Release' and 'Users currently logged in: - admin from 192.168.58.88, expires 13:07'. At the bottom left, it indicates 'Internet connection status: static IP'. A copyright notice at the bottom center reads '© 2005-2018 Epygi Technologies LTD. All Rights Reserved (en\_US)'.



## 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**

#### VoIP Carrier Wizard

A screenshot of the 'VoIP Carrier Wizard' configuration screen. At the top right, there are two buttons: 'Previous' with a left arrow and 'Next' with a right arrow. Below these, the text 'Select VoIP Carrier' is displayed. Underneath, there is a 'VoIP Carrier:' label followed by a dropdown menu showing 'Manual'. Below that is a 'Description:' label followed by a text input field containing 'AlloyVoice'.

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**



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## VoIP Carrier Settings

### VoIP Carrier Common Settings

Authenticate by IP Address

Account Name:

Password:

Confirm Password:

SIP Server:

SIP Server Port:

### VoIP Carrier Advanced Settings

Use RTP Proxy

Authentication Username:

Send Keep-alive Messages to Proxy

Timeout:  sec.

### Outbound Proxy

Host Address:

Port:

### Secondary SIP Server

Host Address:

Port:

### Outbound Proxy for Secondary SIP Server

Host Address:

Port:

- **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

← Previous    → Next

### VoIP Carrier Access Code

Access Code:  
 By Prefix:   
 By Pattern:

Emergency Code: <sup>1</sup>

Route Incoming Calls to:  ▼

Failover to PSTN

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

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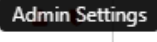
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To Make any changes to the SIP trunk follow the below steps;

- 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

<input type="checkbox"/>	 999	 	Alloy Voice (added by VoIP Carrier Wizard)	None
<input type="checkbox"/>	 998			None



## 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**

### Call Routing Wizard

Go Back

← Previous    Next →

Destination Call Type - Add Entry

Enable Record

Destination Number Pattern:

Number of Discarded Symbols:

Prefix:

Suffix:

Call Type:

Metric:

Description:

Filter on Source / Modify Caller ID

Date / Time Settings

Overall Call Duration Limit

Tracing / Debug Options

Call Alert Settings

Enabler Key:

Disabler Key:

Require Authorization for Enabling/Disabling

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- **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**

[← Previous](#) [Next →](#)

**Call Settings - Edit Entry**

Use Extension Settings:   Keep Original Caller ID

Add Remote Party ID

Destination Host:

Destination Port:

Username:

Password:

Restrict the Number of Simultaneous Calls

Allowed Call Count:

Use RTP Proxy

**Failover Reason(s)**

None

Failover Reason(s)

Busy  Wrong Number

Network Failure  Other

Any

Enable Failover Timeout

Maximum Duration:  sec.

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- **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
  - o **Number of Discarded Symbols:** optional – If you would like to add a caller ID on outbound calls with this route, enter in 99 here.
  - o **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**

← Previous    → Next

**Filter on Source / Modify Caller ID - Edit Entry**

Source Filter

Source Number Pattern:

Source Type:

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Caller ID Modification

Number of Discarded Symbols:

Prefix:

Discard Non-Numeric Symbols

Send Random Caller ID

---

Display Name:

Remove Display Name

← Previous    → Next

- On the next screen select **Finish**

← Previous    🏠 Finish

**Summary - Edit Entry**

<b>Destination Call Type</b>		<b>Routing Call Settings</b>	
Destination Number Pattern:	*	Use Extension Settings:	000
Number of Discarded Symbols:		Keep Original Caller ID:	No
Prefix:		Add Remote Party ID:	Yes
Suffix:		Destination Host:	alloyvoice.com.au
Call Type:	IP-PSTN	Destination Port:	5050
Metric:	10	Username:	
Description:	AlloyVoice	Transport Protocol for SIP:	UDP
		Restrict the Number of Simultaneous Calls:	No
		Use RTP Proxy:	Yes
		Voice Transcoding:	Disabled
		Maximum Duration:	Disabled
		Single Call Duration Limit:	Disabled
		AAA Required:	AAA disabled.
		Client Code Identification:	Disabled
		Failover Reason(s):	None
		Source Filter	
		Source Number Pattern:	*
		Source Type:	PBX
		Discard Non-Numeric Symbols:	No
		Caller ID Modification	
		Number of Discarded Symbols:	
		Prefix:	

← Previous    🏠 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**

### Call Routing Wizard

[Go Back](#)

[← Previous](#) [Next →](#)

Destination Call Type - Edit Entry

Enable Record

Destination Number Pattern:

Number of Discarded Symbols:

Prefix:

Suffix:

Call Type:

Metric:

Description:

Filter on Source / Modify Caller ID

Date / Time Settings

Overall Call Duration Limit

Tracing / Debug Options

Enabler Key:

Disabler Key:

Require Authorization for Enabling/Disabling

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## For Call Settings – Edit Entry

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

### Call Routing Wizard

[Go Back](#)

[← Previous](#) [→ Next](#)

Call Settings - Edit Entry

Use RTP Proxy

AAA Required:

Local Authentication

Client Code Identification

Failover Reason(s)

None

Failover Reason(s)

Busy  Wrong Number

Any

[← Previous](#) [→ Next](#)

## For Filter on Source / Modify Caller ID

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

### Call Routing Wizard

[Go Back](#)

[← Previous](#) [→ Next](#)

Filter on Source / Modify Caller ID - Edit Entry

Source Filter

Source Number Pattern:

Source Type:

Caller ID Modification

Number of Discarded Symbols:

Prefix:

Discard Non-Numeric Symbols

Display Name:

Remove Display Name

[← Previous](#) [→ 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.



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## General Configuration

Max Number of Records in DB cache	<input type="text" value="32"/>	recs	<input type="checkbox"/> Accept stray SIP requests
DNS cache MAX size	<input type="text" value="32"/>	recs	<input type="checkbox"/> Change SIP Error Code to Busy Here
DNS cache cleanup timeout	<input type="text" value="6"/>	hours	<input type="checkbox"/> Ignore To header in incoming SIP INVITE requests
Flash timeout	<input type="text" value="2"/>	sec	<input type="checkbox"/> Add SIP Diversion header on forwarding
Call progress notification timeout	<input type="text" value="10"/>	sec	<input type="checkbox"/> Use Rport
SIP DNS SRV Failover Timeout	<input type="text" value="16"/>	sec	<input type="checkbox"/> Use External Call Control Forwarding
IP line registration timeout maximum	<input type="text" value="3600"/>	sec	<input type="checkbox"/> Enable IP Loop
IP line registration timeout minimum	<input type="text" value="120"/>	sec	<input checked="" type="checkbox"/> Force Hold Music
Play user friendly voice messages instead of tones	<input type="text" value="default"/>		<input checked="" type="checkbox"/> Do Not Send External RE-INVITE
<b>IP phones settings</b>			<input checked="" type="checkbox"/> Do Not Send REFER
SIP registration timeout	<input type="text" value="3600"/>	sec	<input type="checkbox"/> Callback through Routing
SIP subscription timeout	<input type="text" value="3600"/>	sec	<input type="checkbox"/> Enable Call Recording of Early Media
SIP session refresh timeout	<input type="text" value="600"/>	sec	<input type="checkbox"/> Allow Multiple Parallel Calls on an IP Line
SIP failed registration retry timeout	<input type="text" value="30"/>	sec	
Clean IP Phone VLAN settings if no VLAN on PBX ( reboot required )	<input checked="" type="checkbox"/>		
<b>SIP TLS</b>			
SSL server method	<input type="text" value="SSLv23"/>		
SSL client method	<input type="text" value="SSLv23"/>		
<b>Templates for Caller ID <sup>1</sup></b>			
IP call	<input type="text" value="%a"/>	(%a%d%u%h)	
PBX call	<input type="text" value="%a"/>	(%a%d%u)	
PSTN call	<input type="text" value="%a"/>	(%a%d%u)	
<b>Presence</b>			
Subscription limitation ( reboot required )	<input type="text" value="1000"/>		
Do not use "partial update" method in BLF notifications	<input type="checkbox"/>		

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## 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

<input type="checkbox"/>	999		Alloy Voice (added by VoIP Carrier Wizard)	None
<input type="checkbox"/>	998	<b>Admin Settings</b>		None

- 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**

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## Extension 999 Codecs

[Go Back](#)

Enable/Disable    Move Up    Move Down    Make preferred  

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

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

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## 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
<a href="#">999</a>	0385135613	alloyvoice.com.au	Yes	13-Jun-2018 10:21:21



## 11. Security Audit

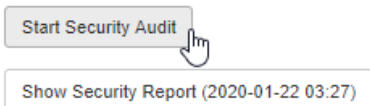
Security Audit is a security reporting system, which generates the warnings regarding the Epygi PBX's weaknesses relative to the selected Security Level.

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**

### Security Diagnostics

Security Audit



Useful Links to Adjust System Security

- User Rights Management
- IP Lines
- Firewall/NAT

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

### Security Scanning Report

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