



Configuring Epygi QX IP PBXs with BINARY NETWORKS

Abstract: This document describes the configuration of the Epygi QX IP PBXs to use the IP-PSTN service from **BINARY NETWORKS**.

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Document Revision History

Revision	Date	Description	Valid for SW	Valid for Models
1.0	01-Oct-2015	Initial release	6.1.x and higher	QX IP PBXs





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1 Introduction

This document describes the configuration of Epygi QX IP PBXs (herein IP PBX) to use the SIP trunking service from **BINARY NETWORKS**. The IP PBX is capable of making IP-PSTN calls via **BINARY NETWORKS** SIP trunks. This solution allows IP PBX users to make cost saving calls to the global PSTN.

Please Note: The described configuration is generic for all QX IP PBX models.

Please Note: Security issues and calling rates are beyond the scope of this document. See the listed documents in References section to get more information on the security related issues.

2 Scenario

Provider: BINARY NETWORKS

- offers outbound and inbound calls.
- allows parallel outbound calls to be made from one account.
- allows parallel calls to be received to one account.

Customer:

• will be able to make long distance PSTN calls through **BINARY NETWORKS** SIP trunks.

2.1 Requirements and Preparations

- The IP PBX is connected to the network and all network settings are properly configured.
- The IP PBX is running software version 6.1.x or higher.

2.2 Account Information from BINARY NETWORKS

BINARY NETWORKS will provide the customer with the following data:

- Username 61390208487 (sample)
- Password **********
- SIP Server sip.binaryvoice.com.au
- Signaling port for SIP server 5060
- Telephone Number(s) a single DID or groups of DIDs of 5, 10, 20 or 100.

Please Note: The above listed values have demonstrational purposes.

3 Configuration

Sections below describe the configuration required on the IP PBX allowing to:

- Make and receive long distance IP-PSTN calls through BINARY NETWORKS SIP trunks.
- Send and receive faxes through BINARY NETWORKS SIP trunks.



3.1 Configuration for Making Outgoing Calls through BINARY NETWORKS

Configure the provided account on the IP PBX as follows to make outgoing calls.

- 1. Go to the **Telephony**->**VoIP Carrier Wizard** menu, pass through the wizard by inserting the below listed parameters to create a new extension and configure it with the **BINARY NETWORKS** SIP trunks:
 - Select Manual for VoIP Carrier
 - Description BINARY NETWORKS (optional)
 - Press Next (Figure 1).

e chàg	1	Events Administrator (admin) Log
	Overview VolP Carrier Call Routing Call Recording NAT Traversal RTP SIP Advanced	
Dashboard		Held
setup	VoIP Carrier Wizard	net
Extensions		
Interfaces	Select VoIP Carrier	
Telephony		
Firewall	VolP Carrier: Manual	
Network	Description: BINARY NETWORKS	
I Status		
Maintenance		
	Previous	Next

Figure 1: VoIP Carrier Wizard- Page 1

- 2. Insert the following parameters in the next opened page:
 - Account Name the provided account (in this case it coincidences with one of the DID numbers)
 - Password and Confirm Password fields
 - SIP Server sip.binaryvoice.com.au
 - SIP Server Port 5060
 - > Enable **Use RTP Proxy** service and press **Next** (Figure 2).



6	epygi						✓Pending Events	g Logged In As: Administrator (admin)	[∳ Log Out
	QX50	Overview VoIP Carrie	er Call Routing	Call Recording NAT	Traversal RTP	SIP Advanced			
80 \$2	Dashboard Setup Extensions	VoIP Carrier W	Vizard						Help 👻
ite C	Interfaces Telephony	VoIP Carrier Settings							
() ()	Firewall Network	VoIP Carrier Commo	on Settings		VolP Carrier Advan	ced Settings			
dil	Status	Account Name:	61390208487		Use RTP Proxy				
₽ ^C	Maintenance	Password:			Authentication User Name:				
		Confirm Password:	•••••		Send Keep-aliv	e Messages to Proxy			
		SIP Server:	sip.binaryvoice.com.au		Timeout: 60	Sec			
		SIP Server Port:	5060						
					Outbound Proxy				
					Host Address:				
					Port:				
					Host Address				
					Port:				
					Outbound Proxy for	r Secondary SIP Server			
					Host Address:				
					Port:				
			Pre	evious			Next		

Figure 2: VoIP Carrier Wizard- Page 2

- 3. On the third page of the VoIP Carrier Wizard define the Access Code (let's say 0) which will be used in the Call Routing Table for making outgoing calls to BINARY NETWORK, and the IP PBX extension which will receive all incoming calls from BINARY NETWORKS SIP trunks. Routing all incoming calls to the Auto Attendant (00) is the most frequently used scenario. Defining another extension as the call receiver also applicable.
 - Access Code 0
 - **Emergency Code** leave the default value or put your country emergency call, let's say 000.
 - > Route Incoming Calls to -00.

Enable the **Failover to PSTN** service if it is desirable to allow calls failover through the IP PBX onboard FXO lines and press the **Next** button (Figure 3).



🕜 epygi		✓Pending Events Logged In As: Administrator (admin) Log Out
QX50	Overview VolP Carrier Call Routing Call Recording NAT Traversal RTP SIP Advanced	
 Dashboard Setup Extensions 	VoIP Carrier Wizard	Help 🗸
 Interfaces Telephony 	VoIP Carrier Access Code	
Firewall Network Status	Access code: By prefix: By prefix: By pattern: By pat	
🔏 Maintenance	Emergency Code: 1 000	
	Fallover to PSTN	
	Previous	Next

Figure 3: VoIP Carrier Wizard- Page 3

4. Confirm entered settings on the last page of VoIP Carrier Wizard page and press Finish (Figure 4).

	epygi						✓Pending Events	Logged In As: Administrator (admin)	C Log Out
	QX50	Overview VoIP Carrier Call Re	outing Call Recording	NAT Travers	al RTP	SIP	Advanced		
€3 ↓ ↓ ↓ ↓	Dashboard Setup Extensions Interfaces	VolP Carrier Wizard							Help 🔻
د (۸) (2)	Telephony Firewall Network	VolP Carrier:	Manual						
.11 	Status Maintenance	Description: VolP Carrier Common Settings Account Name:	BINARY NETWORKS 61390208487						
		SIP Server: SIP Server Port: VoIP Carrier Advanced Settings	sip.binaryvoice.com.au 5060						
		Use RTP Proxy: Authentication User Name: Send Keep-alive Messages to Pro	Yes xy: No						
		VoIP Carrier Access Code Access Code: Emergency Code:	0* 000						
		Route Incoming Calls to: Failover to PSTN:	00 No						
		P	revious				Finish		

Figure 4: VoIP Carrier Wizard- Page 4



As a result of above listed steps, the provided account is configured on the automatically created extension 989 and the routing rule **0*** is automatically added to the **Call Routing** table. This allows making outbound calls through **BINARY NETWORK** SIP trunks using 0 prefix when dialing, as well as receiving inbound calls to the extension 989 (Figure 5).

6	epygi											Logged In As: Administrator (admin)	€ Log Out
	QX50	0	verview	Extensions	Dialing Directories	Conferences	Recordings	Receptionist	ACD	Authorized Phones			
2	Dashboard	Exte	ensions	Add Extension	Add Multiple Extensions	Bulk Import							
•	Setup	Fv	ton	sions Man	agement								Help 👻
	Extensions	L^	len.		agement								
÷.	Interfaces	erfaces Total extensions count: 72											
6	Telephony	Total extensions count: 72 ★ Add Edit: Deleta Hida extensions attached to disabled IR line: O Lise Enviri SID server.											
0	Firewall						Attached				Porcontago of		
	Network			Extension	Display Na	ame	Line		SIP A	Address	System Memory	External Access	Codecs
	Status		* (<u>)0</u>	Attendant			20236@sip.e	pygi.com:50	60	5% (0 sec)		<u>PCMU,</u>
a	Walliteriarice		* 1	<u>.0</u>				713497410@	sip.epygi.co	m:5060	1% (0 sec)		<u>PCMU,</u>
			* 2	20				713497420@	sip.epygi.co	m:5060	2% (0 sec)		<u>PCMU,</u>
			<u> </u>	.01			FXS 1	101, Proxy:si	p.epygi.loc:5	060	5% (0 sec)	None	<u>PCMU,</u>
			Q <u>9</u>	1 <u>89</u>	BINARY NETWORKS (adde Wizard)	ed by VoIP Carrier	None	61390208487	7@sip.binary	voice.com.au:5060	0% (0 sec)	None	<u>PCMU,</u>
			U	36 (Pickup Group)				36			0% (0 sec)		<u>PCMU,</u>
				35 (Call Park)				35			0% (0 sec)		<u>PCMU,</u>
				456 (Call Park)				456			0% (0 sec)		<u>PCMU,</u>
				444 (Paging Group)				444			0% (0 sec)		<u>PCMU,</u>
			9 3	8 (Recording Box)				38			5% (0 sec)		PCMU



How this rule works: The system will route all dialed digits matching the **0** prefix through **BINARY NETWORKS** SIP trunks to PSTN.

3.2 Configuration for Receiving Inbound Calls from BINARY NETWORKS

There are a couple of ways to allow incoming calls to be received from **BINARY NETWORKS** SIP trunks.

1. One is already done in 3.1. For receiving incoming calls from **BINARY NETWORKS** SIP trunks, the required configuration is already created through the **VoIP Carrier Wizard**, so now all incoming calls to the DID number 61390208487 will go to the extension 989 and then automatically forwarded to the extension 00, which is the system Auto Attendant (Figure 6).



🕐 epygi		Logged In As: Administrator (admin) Log Out
QX50 Voice Mail Call History PBX Info Your Extension Return Caller ID Based Servic Extension: 989	rmation Speed Calling Account Basic Services	Caller ID Services
Hiding Caller Information Incoming Call Blocking Outgoing Call Blocking Distinctive Ringing Call Hunting Many Extension Ringing Unconditional Call Forwarding Busy Call Forwarding No Answer Call Forwarding Unregistered/Inaccessible Call Forwarding Find Me / Follow Me Emergency Interrupt Intercom Voicemail Profile	 Enable Service Enable/Disable + Add / Edit Delete State Enabled Send Notification Via SMS Mobile Number Send E-mail E-mail Address Toggle from Handset 	Forward to PBX-00

Figure 6: Unconditional Call Forwarding page for extension 989

2. The second option can be used when a range of DIDs is allocated and provided to the customer. In this case it is possible to receive inbound calls directly to the same or different extensions. See below an example with a range of 100 DID numbers (61390208400 – 61390208499) routed to the range of extensions 100 - 199 accordingly. For this purpose create the appropriate call routing rule.

Go to the **Telephony**→**Call Routing** menu and press the **Add** button. The **Call Routing Wizard** appears. Fill the following fields (Figure 7):

- Destination Number Pattern 613902084{00-99} (for our example)
- Number of Discarded Symbols 9 (the first 9 digits in the incoming DID number will to be discarded)
- Prefix 1 (will be added at the beginning of the remaining two digits)
- Destination Type PBX
- Disable Filter on Source/Modify Caller ID service (to allow SIP callers to use this routing rule) and press the Next button.
- Proceed to the end of the Call Routing Wizard by leaving other settings unchanged and finish the wizard.



n epygi		✓Pending Logged In As: Events Administrator (admin) Log Out							
QX50	Overview VoIP Carrier Call Routing Call Recording NAT Traversal RTP SIP Advanced								
Dashboard	Call Routing Table Call Routing Local AAA Table SIP Tunnel Class of Service								
SetupExtensions	Call Routing Wizard	Help 🗸							
interfaces	G Go Back								
Firewall Network	Routing Call Type - Add Entry								
🖬 Status 🖋 Maintenance	Enable Record								
	Destination Number Pattern: 613902084{00-99} (wildcard supported)	Enabler Key:							
	Number of Discarded Symbols. 9	Disabler Key:							
	Prefix: 1	Require Authorization for							
	Suffix:	Enabling/Disabling							
	Destination Type: PBX •								
	Metric: 10								
	Description:								
	Filter on Source / Modify Caller ID								
	Set Date/Time Period(s)								
	Set Overall Calling Time Limit								
	Set Tracing / Debug Options on This Rule								
	Previous	Next							

Figure 7: Call Routing Wizard page

How the rule works: An inbound call matching one of the DIDs within the 613902084{00-99} range will be forwarded to the extension 1{00-99}.

4 Additional Notes

4.1 Sending Music on Hold to Remote Parties

Each extension of the IP PBX can be configured to send its own hold music to remote parties on hold (PSTN, IP, or IP-PSTN destinations). While sending the extensions' music on hold (MOH) to PSTN parties does not require any configuration on the IP PBX, certain configuration is needed when the remote party is an IP or IP-PSTN destination. The following steps describe how to configure an extension to send its own MOH to remote IP parties:

- 1. Open the **Basic Services**→**Hold Music Settings** page (Figure 8).
- 2. Enable the Send Hold Music to remote IP party checkbox and click Save.



🤕 epygi			Language 👻	Logged In As: Extension (103)	C Log Out
QX200	Voice Mail Call Hist General Hold Music Basic Service Extension: 10	ary PBX Information Speed Calling Account Basic Services Caller ID Services Do Not Disturb Hot Line s - Hold Music Settings 3			Help 👻
	General Settings Hold Music Settings Do Not Disturb Settings Hot Line Settings	 Send Hold Music to remote IP party Listen Hold Music: Own_Music • File Restore default Hold Music file Upload file: Choose File Record file: Record file: Record file: Record file: Choose Channel: QX200 • Audio Line In 			
		Save			

Figure 8: Basic Services - Hold Music Settings page

If the IP PBX is configured with an ITSP that does not support remote MOH (the ITSP closes the received audio stream when receiving a SIP re-INVITE message with the c=IN IP4 0.0.0.0, a=send only media attributes), please follow these steps to complete the configuration:

- 3. Type "generalconfig.cgi" in the address field of the browser to open the General Configuration page (Figure 9).
- 4. On this page, select the Force Hold Music checkbox and click Save.



Call progress r Call progress r SIP DNS SRV F IP line registrat IP line registrat IP gains r IP phones set SIP subscriptio SIP subscriptio SIP scall PSX call PSTN call Presence 2 Subscription li Phone Book	notification timeout allover Timeout tion timeout maximum distribute timeout minimum distribute timeout n timeout • caller ID ¹	2 10 10 16 2600 120 6600 3600 660 %a %a 1000	sec sec sec sec sec sec sec (%a%d%u%h) (%a%d%u)	 Add SIP Diversion header on forwarding Use Rport Enable IP Loop Force Hold Music Do Not Send External RE-INVITE Do Not Send REFER Callback through Routing Enable Call Recording of Early Media Allow Multiple Parallel Calls on an IP Line Do not use "partial update" method in BLF notifications 	
Max number o Enable VM Disconnect til VM Session t	f contacts: silence disconnect meout 5 • imeout 60	1000 sec			



4.2 Sending and Receiving Faxes through BINARY NETWORKS SIP Server

Connect the fax machine to one of the FXS extensions attached to FXS 1/FXS 2 line and enable the **Enable T.38 FAX** and **Enable Pass Through FAX** options for the selected extension to send a fax.

For receiving faxes on the fax machine the IP PBX supports the following configuration options:

- 1. Incoming calls are routed directly to one of the FXS extension with the FAX machine attached. A DID number is dedicated for that extension in this case;
- Incoming calls are routed to the Auto Attendant (00) with FAX forwarding enabled to the appropriate FXS extension (FAX extension) that has the fax machine attached. Pressing START from the sending fax machine while listening to the Auto Attendant greeting message will forward the call to the predefined FAX extension.

For receiving faxes from **BINARY NETWORK** SIP Server with the second option you can use the configuration already created during the **VoIP Carrier Wizard**. After the configuration steps described below you will receive faxes on the fax machine attached to the extension 102:

 Go to the Auto Attendant (00)→General Settings menu and enable FAX Forwarding to the extension 102 (Figure 10).

6	epygi							✓Pending Events	Logged In As: Administrator (admin)	C Log Out
	QX200	Overview Exten	sions Dialing Directories	Conferences	Recordings	Receptionist	ACD	Authorized	Phones	
	Dashboard	Extensions Add Exte	Add Multiple Extensions	Bulk Import						
•	Setup	Extensions	Management - F	dit Entry						Help 👻
	Extensions									
÷.	Interfaces	🕒 Go Back								
6	Telephony	General Settings								
ø	Firewall	General Settings	General Settin	nas - 00						
Q	Network	Attendant Scenario		.90 00						
.11	Status	SIP Settings	Display Name / Subject	Attendant						
on C	Maintenance	SIP Advanced Settings	Enable FAX forwarding							
		Extension to forward 102								
			Show on Public Director	у						
		Go To Codec	Percentage of Total Memory	5 • %						
		<u>Settings</u>	Save							

Figure 10: Auto Atendant General Settings page

- 2. Go to the Extensions > Extensions Management menu (Figure 11).
- 3. Click on the Codecs link of the extension 102.
- 4. Select the Enable T.38 FAX and Enable Pass Through FAX checkboxes and press Save (Figure 12).

🜈 epygi											✓Pending Events	Logged In As Administrator (ad	min) Log Out
QX200	Overvi	ew Extensions	Dialing Director	es Conferences	Recordings	Receptionist	ACD	Authorized Phones					
Dashboard	Extensio	Add Extension	Add Multiple Exten	ions Bulk Import									
🔅 Setup	Exto	nsions Mar	agomont										Help 👻
Extensions	LAIC	1510115 11141	lagement										
nterfaces	Total ex	tensions count: 44/2	04										
📞 Telephony	Add		0 Show all outansia	a Lico Equai SID co	D IOT								
Firewall	TAUU	Fuir B Delete	show all extension	s See Epygr StP se	iver								
Network		Extension	•	Display Name	Attached Line	e	SIF	Address	Percentage	of System Memory	Exte	ernal Access	Codecs
JII Status	• 🛉	<u>00</u>	At	endant		00			5% (9 day 4 hour 13 mi	n 8 sec)			<u>PCMU</u>
🔎 Maintenance		101			FXS 1	101			5% (9 day 4 hour 13 mi	n 8 sec)	None		PCMU,
		<u>102</u>			FXS 2	102			5% (9 day 4 hour 13 mi	n 8 sec)	None		<u>PCMU,</u>

Figure 11: Extension Management page



epygi		Language → Pending Logged In As: Events Administrator (admin)										
QX50	Overview Extensions Dialing Directories Conferences Recordings Rece	rtionist ACD Authorized Phones										
Dashboard	Extensions Add Extension Add Multiple Extensions Bulk Import											
Setup	Extension 102 Codece	[
Extensions	Extension 102 Codecs											
Interfaces	O Go Back											
Telephony												
Firewall	Canable/Disable A Move Up A Move Down A Make preferred											
Network	Audio Codecs	State										
Status	G.711u (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate) (pre	erred) Enabled										
Maintenance	G.711a (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)	Enabled										
	G.729a (CS-ACELP speech coding at 8 kbit/s rate)	Enabled										
	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										
	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										
	Out of Band DTMF Transport											
	✓ Enable T.38 FAX											
	🖉 Enable Pass Through FAX											
	Enable Pass Through Modem											
	Force Self Codecs Preference for Inbound Calls											
	Secure RTP Settings											
	SRTP Policy: Make unsecure calls, accept anything											
	Save											

Figure 12: Codecs page for extension 102

Please Note: The IP PBX allows also receiving FAX messages as a **TIF** file into the extension's voice mailbox if there is no FAX machine attached to that extension. In this case the following should be configured on that extension:

- Enable Voice Mail Service.
- Allocate enough memory space for storing incoming faxes.
- Set the **No answer timeout** to its minimum value.
- Enable Enable T.38 FAX and Enable Pass Through FAX services for that extension as well.

Please Note: In all scenarios the Enable T.38 FAX and Enable Pass Through FAX checkboxes should be selected for the FAX extension.



5 References

The following documents can be helpful for further configuration of the QX IP PBX. They can be downloaded from Epygi's WEB portal at <u>www.epygi.com</u>:

- QX IP PBX Manual I Installation Guide
- QX IP PBX Manual II Administrator's Guide
- Preventing Unauthorized Calls on the Epygi QX IP PBX
- Web Access Control and Privileges on the Epygi QX IP PBX
- Receiving Unified Fax Messages on the Epygi IP PBX

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