





Configuring QX IP PBXs with Adiptel SIP Trunks

Abstract: This guide describes the configuration of QX IP PBXs to use the SIP trunk service from Adiptel.



Document Revision History

Revision	Date	Description	Valid for FW	Valid for Models
1.0	29-Aug-17	Initial Release	6.1.50 and higher	QX IP PBXs



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1 Introduction

This document describes the configuration of Epygi QX IP PBXs (herein QXs) to use the VoIP SIP trunking service from Adiptel – the industry leader among business VoIP providers in Spain. The QX is capable of making IP-PSTN calls via Adiptel SIP trunks. This solution allows QX users to make cost saving calls to the global PSTN network.

Note:

- The described configuration is generic for all Epygi QX IP PBX models, such as the QX20, QX50, QX200, QX500, QX2000 and QXISDN4+.
- Security issues and calling rates are beyond the scope of this document. See the listed documents in <u>References</u> section to get more information on the security related issues.

2 Scenario

Provider: Adiptel

- Offers outbound and inbound calls.
- Allows parallel outbound calls to be made from one account.
- Allows parallel calls to be received on one account.

Customer:

• The customer will make long distance cost saving PSTN calls through the Adiptel SIP trunks.

2.1 Requirements and Preparations

- QX is connected to the network and all network settings are properly configured.
- One or more IP phones in Epygi supported phones list are autoconfigured with QX.
- Always use the **latest** available QX **firmware** to achieve maximum compatibility for the QX's telephony features and settings.

2.2 Account Information from Adiptel

Adiptel will provide the customer with the following data (all listed data below are just samples):

- Username (authorization username/userID) 4997414497
- Password **********
- SIP server clientes.adiptel.com
- Signaling port for SIP server 5060
- Telephone number(s) (DID allocated to the customer) 9476XXXXX



3 Configuration

The sections below describe the configuration steps required on the QX to allow the users to

- Make outgoing calls through the Adiptel SIP trunks.
- Receive incoming calls from the Adiptel SIP trunks
- Send and receive faxes through the Adiptel SIP trunks.

We will use the QX's **VoIP Carrier Wizard** designed to simplify the configuration of QX with different VoIP SIP providers. The wizard allows collecting the account information from provider and generating the needed configuration for each specific VoIP SIP provider on QX. Just after finishing the wizard, the QX local PBX extensions will be able to place calls to the PSTN using the provider's SIP trunks, as well as receive calls from the provider's VoIP SIP trunks.

3.1 Making Outgoing Calls through the Adiptel

Create automatically a new extension on the QX and configure if with the Adiptel SIP trunks as follows:

- 1. Go to the **Telephony**→**VoIP Carrier Wizard**, pass through the wizard by inserting the below listed parameters:
 - > Select Manual from the VoIP Carrier list.
 - Description optional (e.g. Adiptel)
 - Click Next.

	QX50	Resumen Provvedor VoIP Enrutamiento de llamadas Grabación de llamadas NAT Transversal RTP SIP	Schedules Avanzado
69 0	Panel de Control Configuración	Asistente de Configuración del Proveedor de VoIP	Hostname: epygiArmInteropQX50 Halp •
<i>₽</i>	Extensiones Interfaces		
C A	Telefonia Eirewall	← Anterior → Próximo	
0	Red	Selecione proveedor de VoIP	
1	Estatus Mantenimiento	Proveedor de VolP: Manual	
		Pration	
		← Anterior → Próximo	

Figure 1: Select VoIP Carrier section

- 2. Insert the following parameters in the VoIP Carrier Settings section (Figure 2):
 - > Account Name the username provided by the Adiptel (4997414497 for this example)

 - SIP server clientes.adiptel.com
 - SIP Server Port 5060
 - Use RTP Proxy selected
 - Click Next.



	QX50	Resumen Proveedor VolP	Enrutamiento de llamadas	Grabación de llamadas NAT	Iransversal RIP SI	P Schedules	Avenzado	
-	Panal de Control Configuración Extensiones	Asistente de Cor	nfiguración del Pro	oveedor de VoIP			Hostname: epygiArmInteropQX50	Help +
ń-	Interfaces							
0	Telefonia			← Anterior	Próximo			
1	Frewall							
0	Red	Ajustes de Carrier VOIP						
lately	Estatus							
F	Mantenimiento	Parámetros Comunes C	arrier VolP	Parámetros Avanzado	s Carrier VolP			
		Authentication by IP	Address	Use el proxy RTP				
			Lose Lucies	Autenticación de				
		Nombre de Cuenta:	4397414497	Nombre de Usuario:				
		Contraseña		Enviar mensajes o	ie keep-alive al proxy			
		Confirme contraseña:		Tiempo finalizado:	60	seg		
		Servidor SIP:	clientes.adiptel.com	Proxy Exterior				
		Puerto de Servidor SIP:	5060	Dirección de Host				
				Puerto:				
				Servidor Secundario	de SIP			
				Dirección de Host.		1		
				Puerto:				
				Proxy Exterior para se	arvidor secundario de SIP			
				Dirección de Host		1		
				Puerto:				
				(2)				
				+ Anterior	Próximo			

Figure 2: VoIP Carrier Settings section

- 3. Configure the following parameters in the VoIP Carrier Access Code section: (Figure 3):
 - Access Code 1 (for this example)
- **Emergency Code** leave the default value or put your emergency call number for your area.
- Route Incoming Calls to 00 (the QX default Auto Attendant). Routing all incoming calls to the Auto Attendant is the most frequently used scenario. Using other QX extension as a call receiver is also applicable.
- Failover to PSTN Enable the Failover to PSTN service if it is desirable to allow calls failover through the QX's on-board FXO/ISDN lines. This option is available for QX50, QX200 and QXISDN4+ models.
- Click **Next**.



	QX50	Resumen	Proveedor VolP	Enrutamiento de llamadas	Grabación de llamadas	NAT Transversal		Schedules	Avanzado	
€3 ¢ ₽	Panel de Control Configuración Extensiones	Asisten	te de Conf	guración del P	roveedor de Vo	IP			Hostname: epygiArminteropQ	W50 Нир •
in C	Interfaces Telefonia				← Ante	rior 🔷 Próximo	,			
0 0	Rod Estatus	Código de	Acceso a Carrier \	/oIP						
£	Mantenimiento	Código	de Acceso:	Por Prefijo: Por Patron:						
		Código	de Emergencia: ¹	911						
		E Falk	over a la PSTN	<u>.</u>						
					🗲 Ante	rior 🔶 Próximo				

Figure 3: VoIP Carrier Access Code section

4. Confirm the entered settings on the last section of VoIP Carrier Wizard and click Finish.

	QX50	Resumen Proveedor VolP	Essutamiento de llamadas	Grabación de Barnadas	NAT Transversal	RTP SIP	Schedules	Avaruado	
ß	Panel de Control							Mostname anunideministerer(1)/50	
0	Configuración	Asistente de Config	guración del Pro	oveedor de Vol	P			Hose and the second sec	-
10	Extensiones								
h -	Interfaces								
6	Telefonia			← Anter	ior 👂 Terminar	7			
n	Firewall								
0	Rad	Sumario de Carrier VolP							
Ghill	Estatus								
1	Mantenimiento	Proveedor de VoIP:	Manual						
1		Descripción:	Adiptel						
		Parámetros Comunes Carrier	VolP						
		Nombre de Cuenta:	4997414497						
		Servidor SIP:	clientes.adiptei.co	m					
		Puerto de Servidor SIP:	5060						
		Parámetros Avanzados Carri	er VolP						
		Use el proxy RTP:	SI						
		Autenticación de Nombre de	Usuario:						
		Enviar mensajes de keep-aliv	e al proxy No						
		Código de Acceso a Carrier \	/oIP						
		Código de Acceso de PSTN:							
		Código de Emergencia:	911						
		Enrute llamada entrante a	00						
		Failover a la PSTN:	No						
				6 Anton	or ISI Terminas				
				A 100 million					

Figure 4: VoIP Carrier Wizard – Summary section

Now the provided account is configured with the QX. The extension (e.g. 999) with provided credentials (username, password) will be created automatically in the **Extensions Management** (Figure 5). The appropriate routing rules with **1*** and **{911}** patterns will be automatically added on the **Call Routing Table** (Figure 6).



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n Panel de Cantral	Dates	Agrega Enteriodo	Agingario	nühiptes Estensiones Importación máxica						
Configuration Extensiones	Ges	stión de Extens	Hartname: spygiArminiero	hi0)(20	Help =					
- interfactors	Conte	ro total de extensiones: 52/1								
A Frend	+ 46	adir / Cambiar Billiorrar	Q							
Q Rall		Extensión		Nombre del Cliente	Lines conectada	Dirección de SIP	Porciento de Memoria del sistema	Accesso externo	Credit	CODECS
F Martenenionia		Q 00	U	Attendant		20236@stalepygi.com/5060	5% (5 hour 43 min 54 sec)			POVU.
		Q 10	0			10	1% (1 hour B min 47 sec)			2000
		Q 20	U	Scheekulee texting		713497420, Prosycatoreputa.com/5080	2% (2 hour 17 min 34 sec)			ROMD.
		CB 101	4.0		881	2404988091. Prospassiop Libroadworks.net 5060	5% (5 hour 43 min 54 sec)	Call Relay, 3pec/Click2Dad		PENAL.
		102	A U		EKS.2	7134974102, Provesto apage corr.5060	0.7% (5 min 53 sec)	Ninguno		PCMU.
		100	4.0	Maria	Litten de IV-1	2404050095. Proscasiop Litradeoxis.ret5060	0.1% (8 min 53 sec)	GLIL Call Relay, 3ptc/Okt/20al		PCM2
		104	4 0	Andrea	Lines do P.2	7427104, Prove 192,168,0,209,5060	0.4% (27 min 31 sec)	Nikguno		PCMU.
		105	4.0		Lines de P.J	12012504704	0.4% (27 mm 37 sec)	Ninguno		BOMM
		105	4 0		Unes de 17.4	T134914106, Provented epygl.com 5061	0.4% (27 min 31 sec)	Ninguno		PCMU
		107	4 U		Lites de IP 5	1134974107, Programenya com5060	0.4% (27 min 31 sec)	Ninguno		PCMU
		108	4 0		uites de 17.6	7134974100. Proxy siz epypi.com/5060	1% (1 hour Bimin 47 sec)	Ninguno		ECM2
		C 301	4 0	EAX	Ninguno	20	5% (5 hour 43 min 54 led)	Nirguno		PGMD
		C 17	4 0	shared malbox	Ninguno	12. 12.	5% (5 hour 43 min 54 tec)	Ninguno		PCMU.
		38	4 0		Ninguno	29	1% (1 hour 8 min 47 sec)	Nieguno		PEMU
		500	4 0	Dial & Announce	Ninguno	7069278142	2% (2 hour 17 min 34 sec)	Ninguno		BCMU.
		C 600	A 0	VE for Hot Desking	Ninguno	7134974600, Provinsio epogi com 5060	1% (1 hour 8 min 47 sec)	Ninguno		G726-24
		700.	A 0	VE for Hot Desking	Ninguno	7134974700, Program apyg.com 5060	1% (1 hour 8 min 47 sec)	Nirguno		BCMU.
		C 19	4 0	maiketing	Ningunia	19	1%-(1 hour 5 min 47 sec)	Ninguno		PCMU
		1 99	4.0	Adiptel Jagregado por el Asistente de Catter Vol73	Ninguno	40074144071@clientes.adiptel.com/5080	0% (0 sec)	Ninguno		PENU.
		36 (Srupo de torne de li	ertadai) O			25	0%-(0,sec)			ECML_
		35 (Estacionar Llamada)	U			25	0% (0 sec)			RCMU.
		456 Estacionar Llamada	U U			456	0% (0 sec)			BONN
		📢 367 (Grupo Paging)	Ø			7624995042	9% (0.sec)			RONU
		16 (Equipo de Gratevio)				28	1% (1 hour Erren 47 sec)	Ninguno		PCMU

Figure 5: Extensions Management page



C)(SO	Re	sum	m Prov	veedor VolP Enrutan	iento de llamadas Grabación	de llamadas NAT Transversal	RTP SIP	Schedules	Awanzado						
Panel de Control	Tabl	a de P	luteo de llam	Enrutamiento de l	amadas Tabla local de AAA Tu	Inel SIP Clase de Servicio									
 Configuración Extensiones 	Tal	bla	de R	uteo de llama	das					Hos	tname: g	iygiArm/nt	evapQXS9 Help •		
🖆 Interfaces	Ve	r Vist	a Detallada	Nostrar todos	los registros										
Frewall	ØÅ	Activado Desactivado + Añadir Zamibiar Duplicado Borrar + Mover Hacia Aniba + Mover Hacia Abajo Mover Hacia													
Red Estatus Mantenimiente		ID	Estado	Puerto del número o destino	e Destination Number Modification	Ajustes de Llamada	Razón de faila(s)	Autenticación Local	Patrón de Número de Origen/ Modificación de Identificador de Tamada	Tipo de Origen	UES / URP	Metrico	Descripción		
		1	Activado	911		FXO Puerto(s): Cualquier puerto	Cualquiera	No	*	РВХ		10	Emergency Call		
		2	Activado	92*	NDS: 1	FXO Puerto(c): Cualquier puerto	Cualquiera	No		PEX		10	Make PSTN Call		
		3	Activado	8*	NDS: 1	SIP sip.epygi.com/5060, RNSC: No	Ninguno	No			URP: No	10	Make SIP call		
		4	Activado	(11,777,7777,77777) 📿		PBX	Cualquiera	No			URP: SI	10	Call to Extensions		
		7	Activado	55777	NDS: 2	PIDI-Intercom	Ninguno	No				10			
		8	Activado	44777	NDS 2	P8X-Voicemail	Ninguno	No	•	PEX		10			
		9	Activado	7° 📿	NDS: 1	SIP 192.168.0.209:5060, RNSC: No	Ninguno	No	8	PBX	URP: SI	10			
		12	Activado	1* 🖉	NDS 1	IP-PSTN clientes.adiptel.com/5060, RNSC: No	Ninguno	No		PEX	UES: 999 URP: SI	10	Acliptei		
		13	Activado	(911)		IP-PSTN clientes.acliptel.com:5060, RNSC: No	Cualquiera	No		PBX	UES: 999 URP: ST	10	Adiptel : Emergency Call		

Figure 6: Call Routing Table page

How this works: The system will route all outbound calls matching the pattern **1*** to the Adiptel SIP trunks. Adiptel, in its turn, will route all inbound calls to the DID 9476XXXXX number to the QX Auto Attendant (00).

3.2 Receiving Inbound Calls from Adiptel

To receive incoming calls from the Adiptel SIP trunks, the required configuration is already created through the **VoIP Carrier Wizard**, so now all incoming calls to the DID number 9476XXXXX will go to the extension 00, which is the QX's default Auto Attendant.



4 Additional Notes

4.1 Sending Music on Hold to Remote Parties

Each extension of the QX 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 QX, 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.
- 2. Enable the "Send Hold Music to remote IP party" checkbox and click Save.

QX50	Correo de Voz Historial de LLan	adas Información del I	'BX Discado Rá	pido Cuenta	Servicios básicos	Servicios de Identificador de Llamadas
📞 Extensión:999	General Música en Espera No Mo	estar Alarm Activate Di	Línea Caliente			
← Regresar	Servicios Básicos -	Ajustes de Mú	sica en Es	pera		
	Extensión: 999					
	Ajustes Generales	Enviar Música en Espe	ra a clientes IP remot	os		
	Parámetros de Música en Espera	Escuchar Música en Espera	Own_Music ~			
	Parámetro de No Molestar	 Archivo 				
	Alarm Settings	A	chivos Cargados:	Choose File N	o file chosen	
	D&A Schedule					
	Parámetros de Línea Caliente	G	abar:	Extensión de Gra	bación	
		O Canal RTP Set	eccionar Canal Basic	~		
		O Entrada de Audio				
		Guardar				

Figure 7: Basic Services – Hold Music Settings page

If the QX 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:

- 1. Go to the "<u>http://xxx.xxx.xxx/generalconfig.cgi</u>" hidden page (Figure 8).
- 2. On this page, select the "Force Hold Music" checkbox and click Save.



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QX50	Configuración General										
Configurac	ón Max Number of Records in DB cache	32	recs	Accept stray SIP requests							
h Interfaces	DNS cache MAX size	32	recs	Change SIP Error Code to Busy Here							
📞 Telefonía	DNS cache cleanup timeout	6	Horas	Ignore To header in incoming SIP INVITE requests							
Firewall	Flash timeout	2	seq	Add SIP Diversion header on forwarding							
🚱 Red	Call presence actilization timeout	10] ~~2	Use Rport							
Lil Estatus	Call progress notification timeout	10] seg	Use External Call Control Forwarding							
🔑 Mantenimie	nto SIP DNS SRV Failover Timeout	16	seg	Force Hold Music							
	IP line registration timeout maximum	3600	seg	Do Not Send External RF-INVITE							
	IP line registration timeout minimum	120	seg	Do Not Send REFER							
	Play user friendly voice messages instead of tones	Play user friendly voice messages instead of tones default ~									
	IP phones settings	P phones settings									
	SIP registration timeout	3600	seg	Allow Multiple Parallel Calls on an IP Line							
	SIP subscription timeout	3600	seg								
	SIP session refresh timeout	600	seg								
	SIP failed registration retry timeout	30	seg								
	Clean IP Phone VLAN settings if no VLAN on PBX (reboot required)	2	-								
	SIP TLS										
	SSL server method	SSLv23 ~									
	SSL client method	SSLv23 v									
	Templates for Caller ID 1										
	IP call	%a	(%a%d%u%h)								
	PBX call	%a	(%a%d%u)								
	PSTN call	%a	(%a%d%u)								
	Presencia										
	Subscription limitation (reboot required)	1000]								
	Do not use "partial update" method in BLF notifications										
	Directorio Telefónico										
	Max number of contacts:	1000]								
	Enable VM silence disconnect										
	Disconnect timeout 60 v	Disconnect timeout 60 ~									
	check meet aneout										
	VM Session timeout 6000 seg	VM Session timeout 6000 seg									
	Guardar										

Figure 8: General Configuration hidden page



4.2 Sending and Receiving Faxes through the Adiptel

To send a FAX connect the FAX machine to one of FXS ports on IP PBX and enable **T.38 FAX** and **Enable Pass Through FAX** options in the codecs' list for the corresponding FXS extension (extension 102, FXS-2 in this example).

For receiving FAX from the **Adiptel** SIP trunks you can use an already created configuration through the VoIP Carrier Wizard. After the additional configuration steps described below you will receive FAX on the FAX machine attached to the FXS-2, extension 102:

- 1. Choose the Extensions → Extensions Management page.
- 2. On the Extensions Management page, click the Codecs link of the extension 102.
- 3. On the Extension Codecs page select the Enable T.38 FAX and Enable Pass Through FAX checkboxes.

	QX50	Rest	imen	Extensiones	Dialing Directories	Conferencias	Grabaciones	Operadora /	CD Teléfonos A	utorizados		
æ	Panel de Control	Extens	iones	Agregar Extensión	Agregar múltiples Exte	ensiones Importa	ción másiva					
۰	Configuración	Evt	onoi	én 102 Cé	(diago							
	Extensiones	EXU	ensi		bulgos							
÷.	Interfaces	O At	ras									
6	Telefonía											
0	Firewall	🖬 Ac	tivar/De	sactivar 🛧 Mover	Hacia Arriba 🔶 Mov	er Hacia Abajo 📮	Ponerlo como prefe	erido Q				
0	Red				Co	decs de Audio			Estado			
.til	Estatus		G.711u	ı (estándar de codi	ficación de audio PCN	l, muestra de 8 kH	z, 8 bits, velocidad	d 64 kbit/s) (Preferi	do) Activado			
10	Mantenimiento		G.711a	(estándar de codi	ficación de audio PCM	l, muestra de 8 kH	z, 8 bits, velocidad	d 64 kbit/s)	Activado			
			G.729a	(codificación de v	oz CS-ACELP a 8 kbit/	s)			Activado			
			G.726-1	16 (codificación de	voz ADPCM a 16 kbit/s)				Deactivado			
			G.726-2	24 (codificación de	voz ADPCM a 24 kbit/s)				Deactivado			
			G.726-3	32 (codificación de	voz ADPCM a 32 kbit/s)				Deactivado			
			G.726-4	40 (codificación de	voz ADPCM a 40 kbit/s)				Deactivado			
			ilbc (C	odificación internet	Low Bit Rate Coder a 1	3,33 kbit/s)			Deactivado			
			G.722 (HD codificación de	audio en 48-64 kbit / s	de velocidad de da	tos, frecuencia de n	nuestreo 16 kHz)	Deactivado			
			G.722.1	(HD codificación d	e audio en 24-32 kbit /	s de velocidad de d	latos, frecuencia de	muestreo 16 kHz)	Deactivado			
			TDVC (Tasa de Dominio de	Tiempo de corte de vo	z a 1,95 kbit/s)			Deactivado			
			Codecs	s de Video					Estado			
			H.263 (codificación de vide	eo para comunicaciones	de baja velocidad)			Deactivado			
			H.264 (codificación de vide	eo avanzada para comu	nicaciones de baja v	velocidad)		Deactivado			
			H.263+	(codificación de vie	deo para comunicacione	es de baja velocidad	i)		Deactivado			
		⊠ Tr	ansporte	e de DTMF fuera de	banda							
		₽н	abilitar r	modo FAX T.38	٦							
		₽н	abilitar F	AX Pass through								
		Пн	abilitar t	ransmisión transpa	rente de Mödem							
		Forzar la preferencia de los codificadores para llamadas entrantes										
	Configuraciones de RTP Seguro											
		Política SRTP: Realizar llamadas no seguras, aceptar cualquier llamada v										
		Gu	ardar									

Figure 9: Codecs page for extension 102



These are the configuration options for receiving FAX on the QX:

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

The QX also allows receiving FAX messages as a TIFF file into the extension's voice mailbox if there is no FAX machine attached to the extension. In this case, the following should be configured on that extension:

- The voice mail service should be enabled (default).
- Enough memory space should be allocated to the selected extension for storing incoming faxes.
- The **No answer timeout** should be set to its min value in the extension settings.
- The Enable T.38 FAX and Enable Pass Through FAX options for that extension should be enabled as well.

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

5 References

Refer to the below listed recourses to get more details about the configuration settings used in this guide:

- Manual-II: Administration Guide for QX IP PBXs
- Manual-III: User Guide for QX IP PBXs
- User Rights Management on QX IP PBXs
- Preventing Unauthorized Calls on QX IP PBXs

Find the above listed documents on Epygi Support Portal.

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