

Epygi QXs with Fusion
June 2, 2016



Document Revision History

Version	Date	Reason for Change
1.0	October 31, 2013	Initial Draft
2.0	June 2, 2016	Revised

CONTENTS:

introduCtion.....	4
Scenario	4
Requirements and preparations.....	4
Account informtaion from Fusion.....	4
CONFIGURATION	5
making outgoing calls through fusion.....	5
receiving inbound calls from fusion.....	10
ADditional notes	13
sending music on hold to remote parties.....	14
References	15

INTRODUCTION

This document describes the configuration on Egyptian QX IP PBXs and QX GQs (herein QXs) to use the VoIP SIP Trunking service from Fusion. The QXs are capable of making calls with Fusion SIP Trunks. This solution allows QX users to make cost savings calls to the global PSTN.

Please note: The described configuration is general for all QX models, such as the QX50/QX200/QX2000/QXISDN4+ and QXE1T1/QXFX04/QXISDN4.

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

SCENARIO

Provider: Fusion SIP Trunks

- 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 be making long distance cost saving PSTN calls through the Fusion SIP Trunks.

REQUIREMENTS AND PREPARATIONS

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

ACCOUNT INFORMATION FROM FUSION

Fusion will provide the customer with the following data (all data used in the configuration below are just samples):

- Username
- Password
- SIP server
- Signaling port for SIP server
- Telephone number(s) (DIDs allocated to the customer)

CONFIGURATION

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

- Make outgoing calls through the Fusion SIP Trunks.
- Receive incoming calls from the Fusion SIP Trunks.

MAKING OUTGOING CALLS THROUGH FUSION

Follow the steps to automatically create a new extension on the QX and configure it with the provided account:

1. Go to Telephony>VoIP Carrier Wizard page, pass through the wizard by inserting the parameters:
 - a. Select Manual for VoIP Carrier
 - b. Description - Optional
 - c. Press Next (figure 1).

The screenshot shows the Epygi web interface. The top navigation bar includes the Epygi logo, a 'Go To Extension' dropdown, a 'Pending Events' indicator, and a 'Logged In As: Administrator (admin)' session bar. The main navigation menu on the left lists various system components, with 'Telephony' highlighted. The central content area is titled 'VoIP Carrier Wizard' and contains a form with the following fields:

- Select VoIP Carrier:** A dropdown menu with 'Manual' selected.
- Description:** A text input field containing 'Fusion'.

At the bottom of the form, there are two buttons: 'Previous' and 'Next'.

Figure 1: VoIP Carrier Wizard – page 1

Insert the following parameter in the next opened page:

- a. Account Name – the account name provided by Fusion (in this example – 2164164495)
- b. Password - *****
- c. SIP Server – Sip2.thevoicemanager.com
- d. SIP Server Port - 5060
- e. Enable Use RTP Proxy service and press Next (figure 2).

The screenshot displays the 'VoIP Carrier Wizard' configuration page in the epygi interface. The page is titled 'VoIP Carrier Wizard' and is part of the 'VoIP Carrier' section. It is divided into two main columns of settings:

- VoIP Carrier Common Settings:**
 - Authentication by IP Address
 - Account Name: 2164164495
 - Password: [Redacted]
 - Confirm Password: [Redacted]
 - SIP Server: Sip2.thevoicemanager.cc
 - SIP Server Port: 5060
- VoIP Carrier Advanced Settings:**
 - Use RTP Proxy
 - Authentication User Name: [Redacted]
 - Send Keep-alive Messages to Proxy
 - Timeout: 60 sec
 - Outbound Proxy:**
 - Host Address: [Redacted]
 - Port: [Redacted]
 - Secondary SIP Server:**
 - Host Address: [Redacted]
 - Port: [Redacted]
 - Outbound Proxy for Secondary SIP Server:**
 - Host Address: [Redacted]
 - Port: [Redacted]

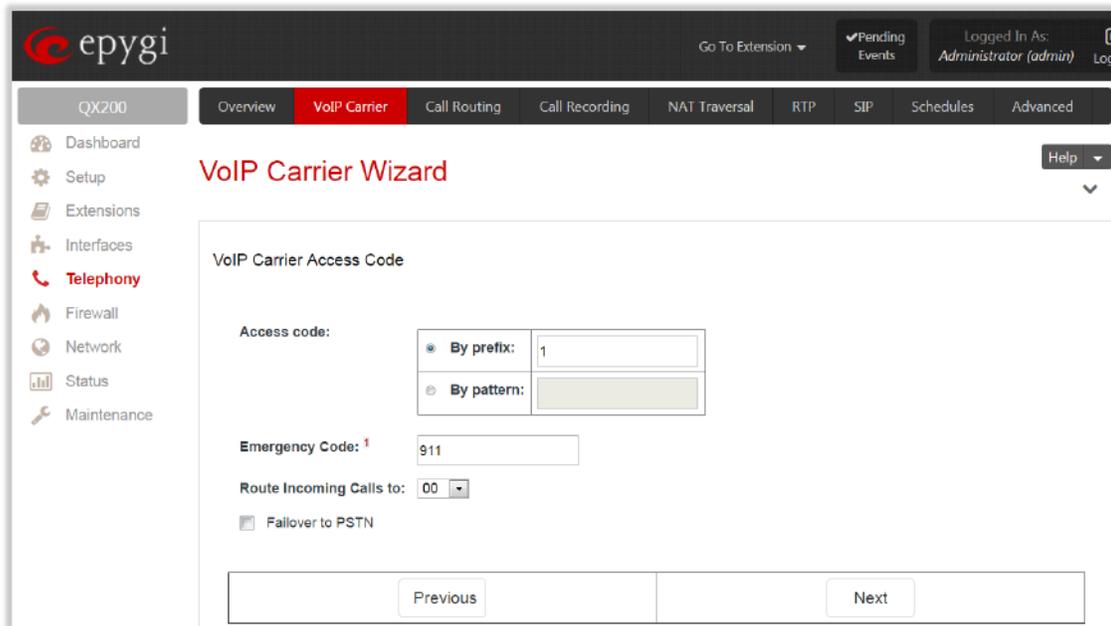
At the bottom of the form, there are 'Previous' and 'Next' navigation buttons.

Figure 2: VoIP Carrier Wizard – page 2

On the third page of the VoIP Carrier Wizard, define the Access Code (let's say 1) which will be used in the Call Routing Table for making outgoing calls, and the QX extension (let's say it is the Auto Attendant -00) which will receive all incoming calls from the Fusion SIP Trunks. Routing all incoming calls to the Auto Attendant is the most frequently used scenario. Defining another extension as the call receiver is also applicable.

- Access Code - 1
- Emergency Code – leave the default value or put in your country emergency call
- Route incoming calls to -00.

Enable the Failover to PSTN service it is desirable to allow calls to failover through QX onboard FXO lines (if available on your model) and press Next (figure 3).



The screenshot shows the epygi VoIP Carrier Wizard configuration page. The interface includes a top navigation bar with the epygi logo, a 'Go To Extension' dropdown, a 'Pending Events' indicator, and a 'Logged In As: Administrator (admin)' session. Below this is a secondary navigation bar with tabs for 'Overview', 'VoIP Carrier' (which is highlighted in red), 'Call Routing', 'Call Recording', 'NAT Traversal', 'RTP', 'SIP', 'Schedules', and 'Advanced'. A left sidebar contains a menu with icons for 'Dashboard', 'Setup', 'Extensions', 'Interfaces', 'Telephony' (highlighted in red), 'Firewall', 'Network', 'Status', and 'Maintenance'. The main content area is titled 'VoIP Carrier Wizard' and contains a 'VoIP Carrier Access Code' section. This section includes: 'Access code:' with two radio button options, 'By prefix:' (selected) with a text input field containing '1', and 'By pattern:' with an empty text input field; 'Emergency Code: 1' with a text input field containing '911'; 'Route Incoming Calls to:' with a dropdown menu showing '00'; and a checkbox labeled 'Failover to PSTN' which is currently unchecked. At the bottom of the form are two buttons: 'Previous' and 'Next'.

Figure 3: VoIP Carrier Wizard – Page 3

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

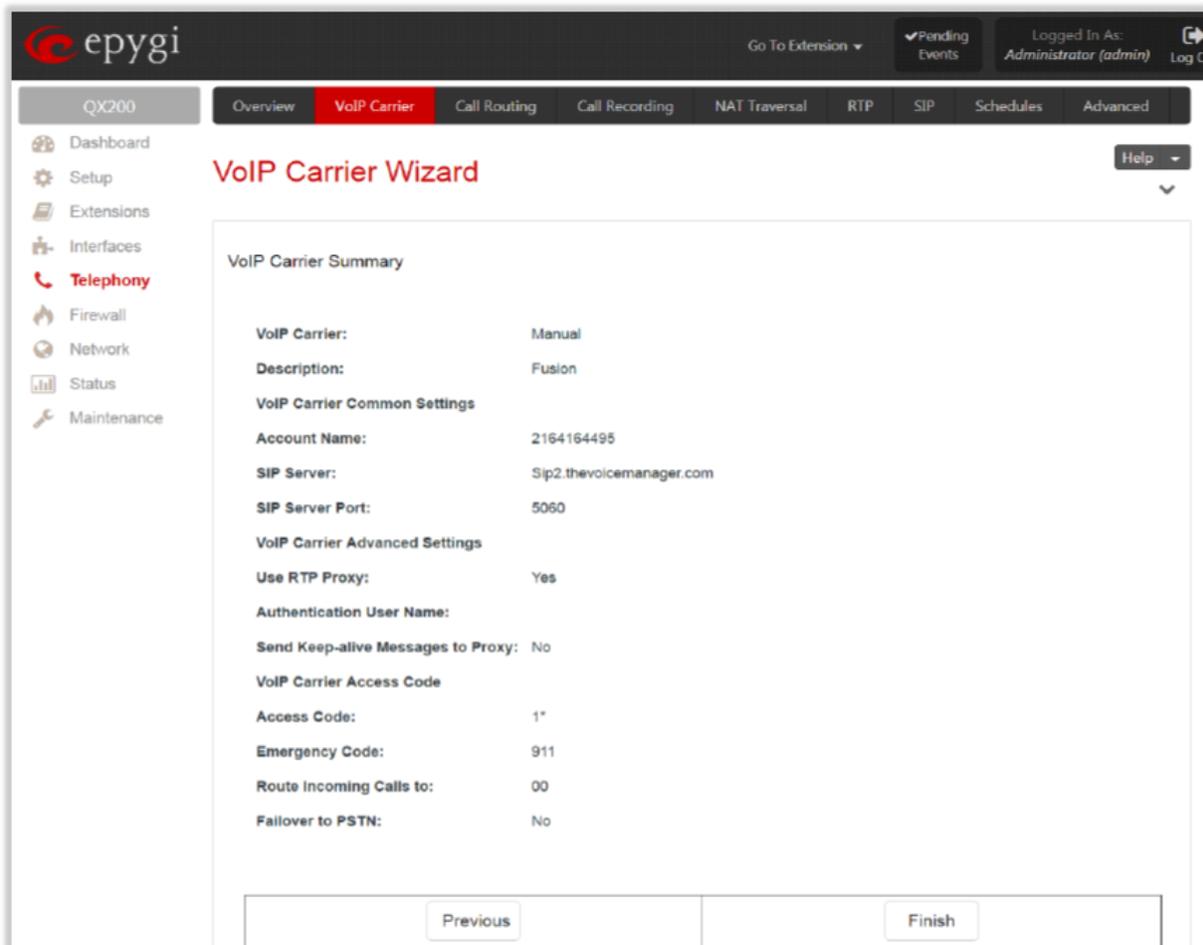


Figure 4: VoIP Carrier Wizard – Summary page

Now the provided account is configured with the automatically created extension 999. This allows making outgoing calls through Fusion SIP trunks using the settings for extension 999 (figure 5). The appropriate Call Routing rule with 1* pattern is also automatically added to the Call Routing table – (see figure 6).

The screenshot shows the Epygi web interface for Extensions Management. The page title is "Extensions Management" and it indicates a total of 92/204 extensions. A table lists several extensions, including an Attendant extension and two FXS extensions (101 and 102). The table columns are: Extension, Display Name, Attached Line, SIP Address, Percentage of System Memory, External Access, and Codecs.

Extension	Display Name	Attached Line	SIP Address	Percentage of System Memory	External Access	Codecs
00	Attendant		7418100@sip.epygi.loc:5060	5% (1 day 21 hour 51 min 14 sec)		PCMU...
101		FXS 1	101	5% (1 day 21 hour 51 min 14 sec)	None	PCMU...
102		FXS 2	102	5% (1 day 21 hour 51 min 14 sec)	None	PCMU...
999	Fusion (added by VoIP Carrier Wizard)	None	2164164495@Sip2.thevoicemanager.com:5060	0% (0 sec)	None	PCMU...

Figure 5: Extensions Management page

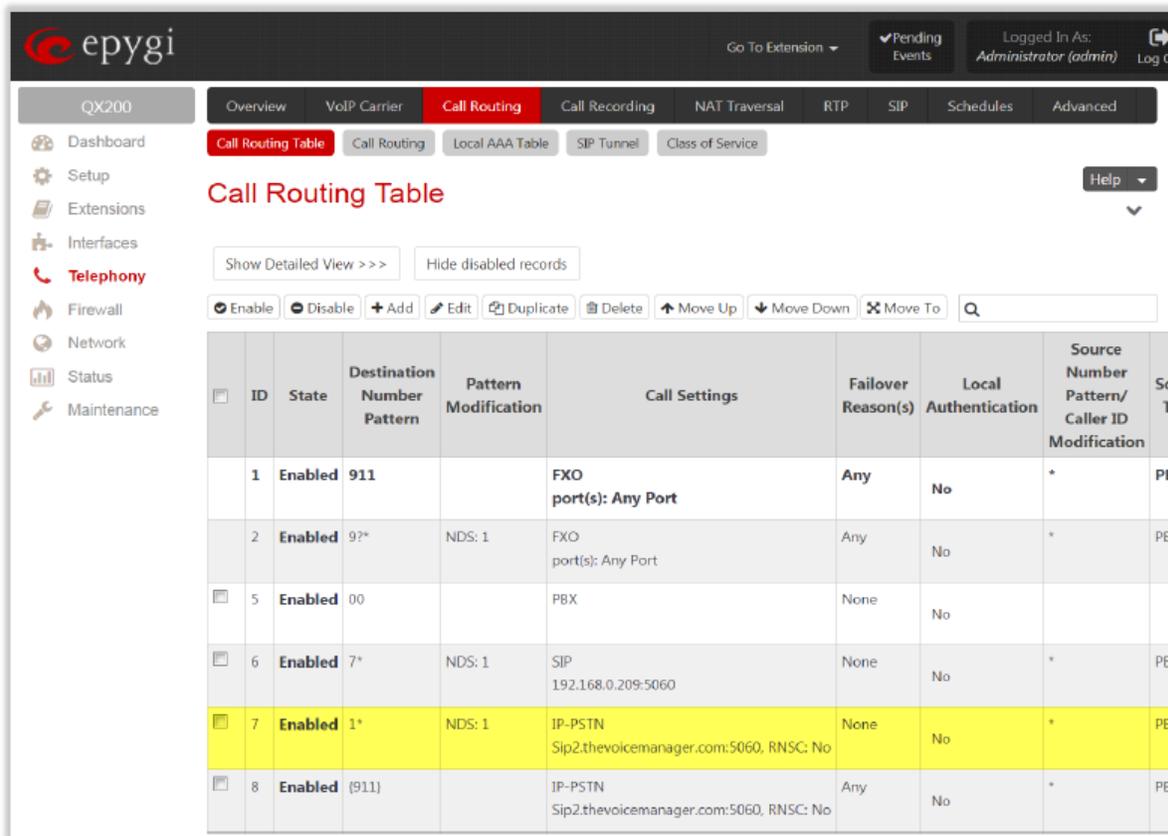


Figure 6: Call Routing Table page

How this rule works: The system will route all outbound calls matching the prefix 1 through Fusion SIP Trunks.

RECEIVING INBOUND CALLS FROM FUSION

There are a couple of ways to receive incoming calls from Fusion SIP Trunks.

1. One is already done in 3.1. For receiving incoming calls from the Fusion SIP Trunks, the required configuration is already created through the VoIP Carrier Wizard, so not all incoming calls to the DID number 2164164495 will go to the extension 00, which is the System Auto Attendant (figure 7).

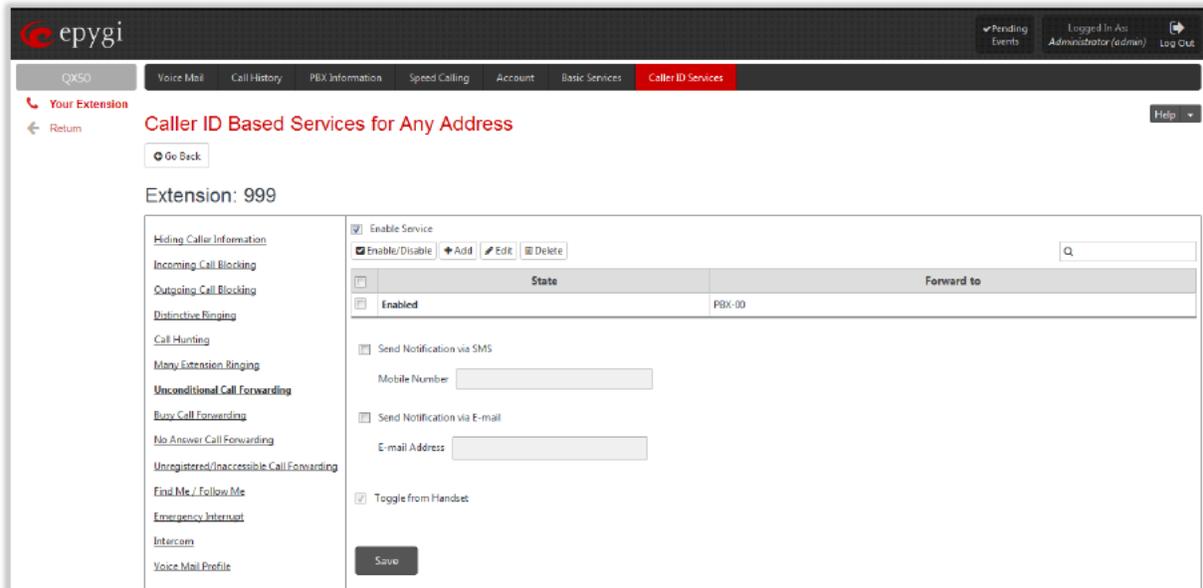
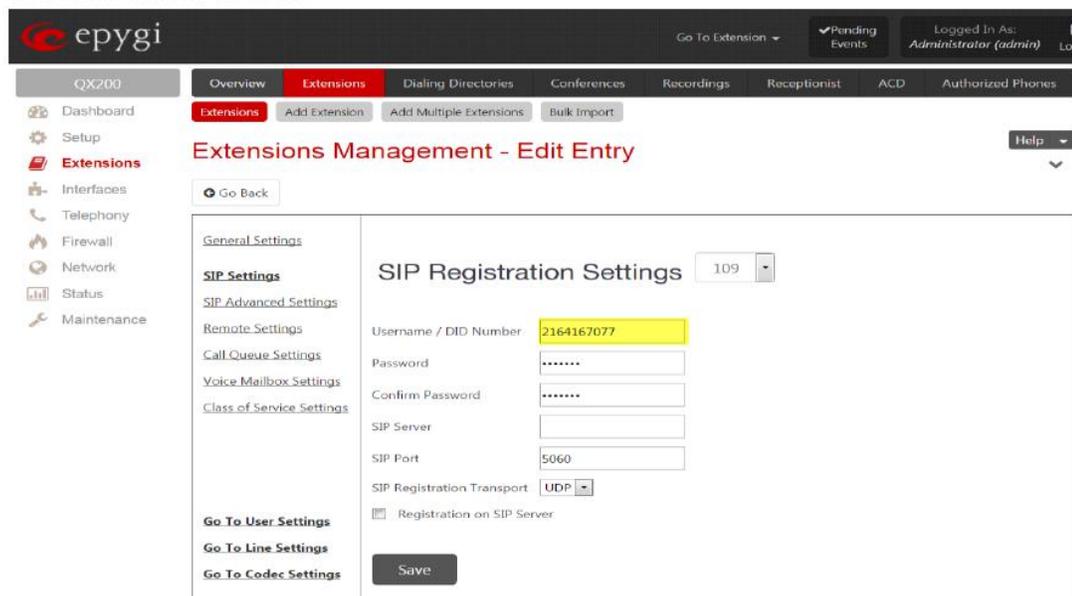


Figure 7: Caller ID Based Services - Unconditional Call Forwarding page

- The next step is setting up the provided DID number(s) as the User Name / DID Number under the SIP settings for the selected extension(s). (Figure 8). For this example, the DID number 2164167077 and extension 109 are selected.



3. One more option can be used when a range of DIDs is allocated and provided to the customer. To use this option, you need to create call routing rules to route incoming calls to different extensions for each provided DID number.

In the example below, a call routing rule is created to ring extension 101 in case incoming call from the Fusion's SIP server match the DID number 2164167077.

To add an appropriate rule to the Call Routing, go to Telephony>Call Routing pages and press Add. The Call routing Wizard appears. Insert the following settings (figure 9).

- Destination Number Pattern – 2164167077
- Number of Discarded Symbols – 10 (all digits in this DID number need to be discarded)
- Prefix – the extension number as a prefix the call should be routed to (in our example – 101)
- Destination Type – PBX
- Disable Filter on Source/modify Caller ID service and press Next.
- Proceed to the end of the Call Routing Wizard by leaving other settings unchanged and finish the wizard.

The screenshot shows the Epygi web interface for configuring call routing. The main heading is "Call Routing Wizard". Below it, there's a "Go Back" button and a "Routing Call Type - Add Entry" form. The form includes the following fields and options:

- Enable Record
- Destination Number Pattern: (wildcard supported)
- Number of Discarded Symbols:
- Prefix:
- Suffix:
- Destination Type:
- Metric:
- Description:
- Require Authorization for Enabling/Disabling
- Set Tracing / Debug Options on This Rule

Figure 9: Call Routing Wizard page

As a result of this configuration the QX will receive incoming calls from the Fusion SIP Trunks to the DID number 2164167077 directly to the extension 101.

In conclusion, to create separate rules for each DID number it is required to put the appropriate DID number in the Destination Number Pattern field and discard 10 digits.

How the rule works: Inbound call matching the pattern '2164167077' will be forwarded to the extension 101.

ADDITIONAL NOTES

SENDING MUSIC ON HOLD TO REMOTE PARTIES

Each extension of the QX IP PBX can be configured to send its own hold music to the 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 (figure 10).
2. Enable the Send Hold Music to remote IP party checkbox and press Save.

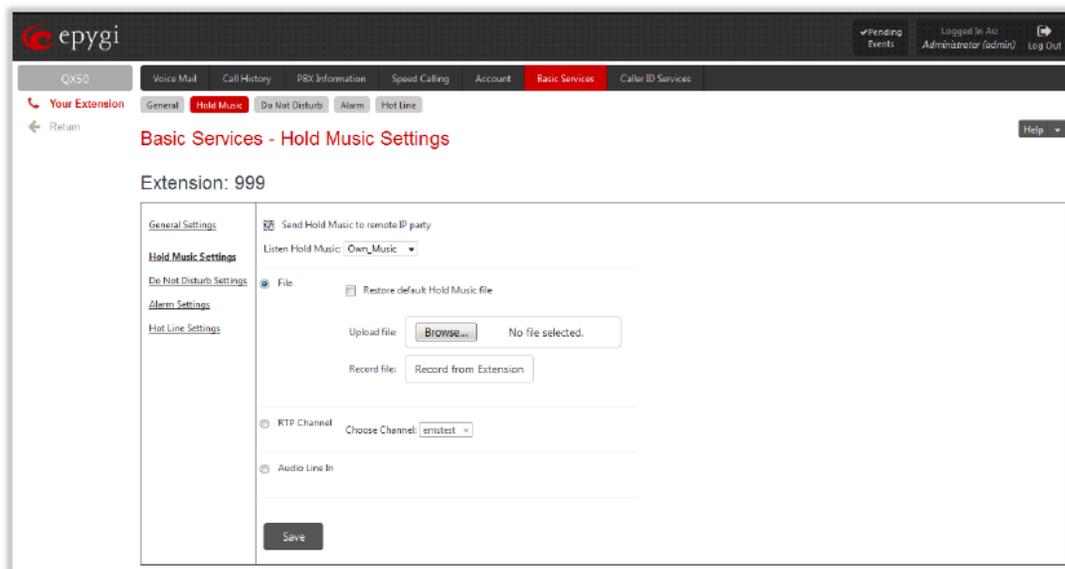


Figure 10: Basic Services – Hold Music Settings page

If the QX IP PBX is configured with an ITSP that does not support remote MOH (the ITSP closes the received audio stream while 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. Type “generalconfig.cgi” in the address field of the browser to open the General Configuration page (figure 11)
2. On this page, select the Force Hold Music checkbox and press Save.

The screenshot displays the 'General Configuration' page for the Epygi QX50 system. The interface includes a sidebar with navigation options like Dashboard, Setup, Extensions, Interfaces, Telephony, Firewall, Network, Status, and Maintenance. The main content area is divided into several sections:

- Database Settings:** Max Number of Records in DB cache (32), DNS cache MAX size (32), DNS cache cleanup timeout (5), Flush timeout (2), Call progress notification timeout (30), SIP DNS SRV Failover Timeout (36), IP line registration timeout maximum (3600), and IP line registration timeout minimum (320).
- Play user friendly voice messages instead of tones:** Set to 'default'.
- IP phones settings:** SIP registration timeout (3600), SIP subscription timeout (3600), and SIP session refresh timeout (600).
- Templates for Caller ID:** IP call (70), PBX call (70), and PSTN call (70).
- Presence:** Subscription limitation (1000).
- Phone Book:** Max number of contacts (1000).
- VM Settings:** Enable VM silence disconnect (checked), Disconnect timeout (5), and VM Session timeout (60).
- Advanced SIP Options:** A list of checkboxes including 'Accept stray SIP requests', 'Change SIP Error Code to Busy Here', 'Ignore To header in incoming SIP INVITE requests', 'Add SIP Diversion header on forwarding', 'Use Roport', 'Enable IP Loop', 'force Hold Music' (highlighted), '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', and 'Do not use "partial update" method in BLF notifications'.

A 'Save' button is located at the bottom left of the configuration area.

Figure 11: General Configuration Page

REFERENCES

The following documents can be helpful for further configuration of the QX IP PBX. The can be downloaded from Epygi's WEB Portal at www.epygi.com

- QX Manual I – Installation guide
- QX Manual II – Administrator's guide
- Preventing unauthorized call from the Epygi QX IP PBX
- Web Access control and privileges on the Epygi QX IP PBX