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

This document describes the configuration on Egypti 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 solutions 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 INFORMTAION 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).

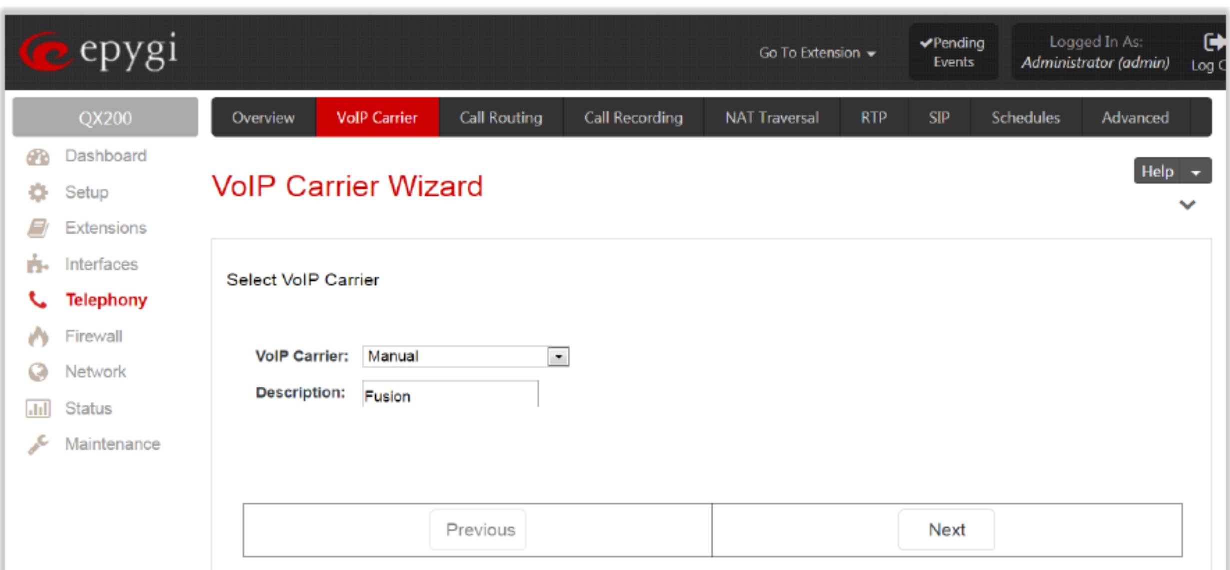


Figure 1: VoIP Carrier Wizard – page 1

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).

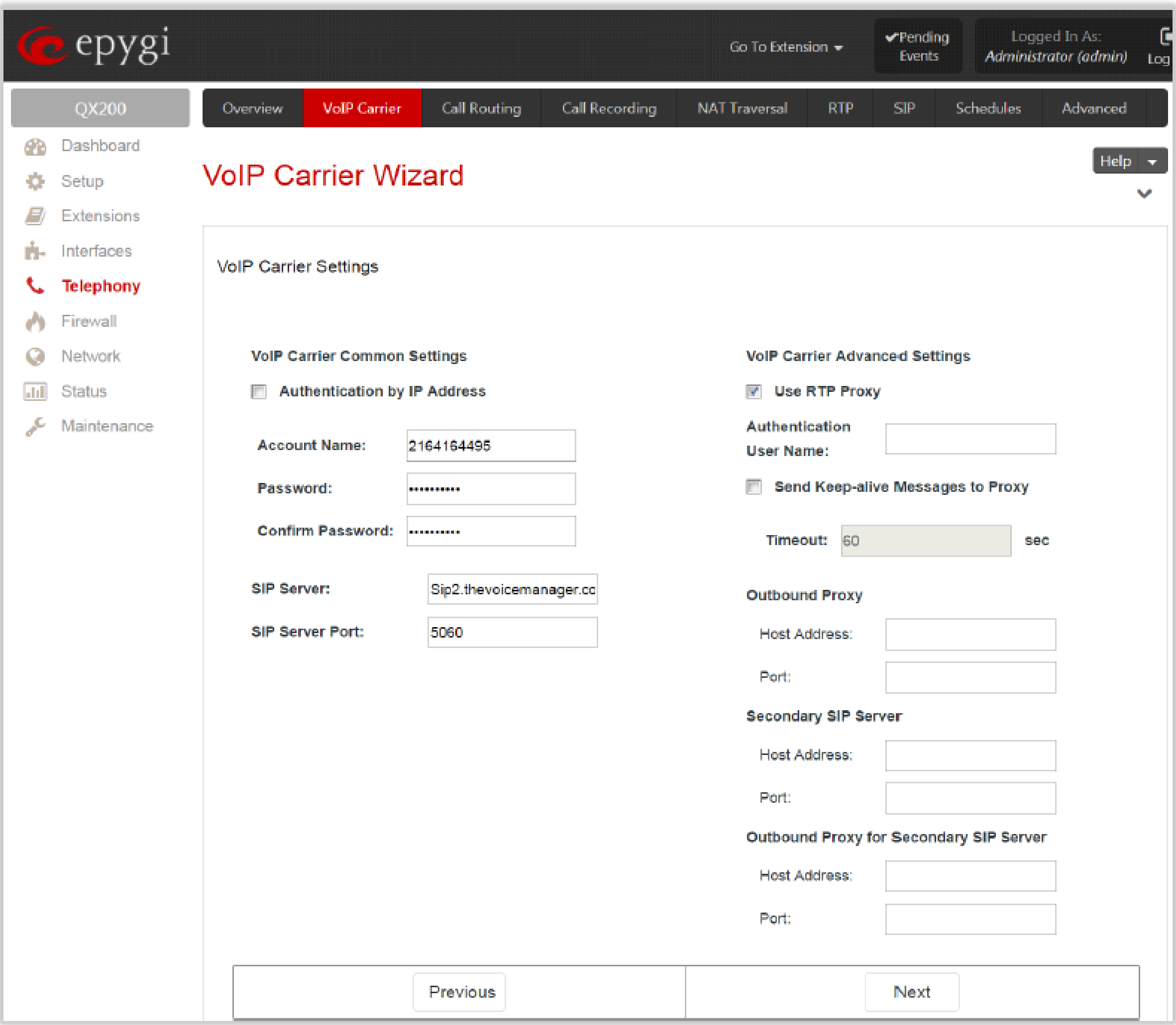


Figure 2: VoIP Carrier Wizard – page 2

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.

a. Access Code - 1
b. Emergency Code – leave the default value or put in your country emergency call
c. Route inkling 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).

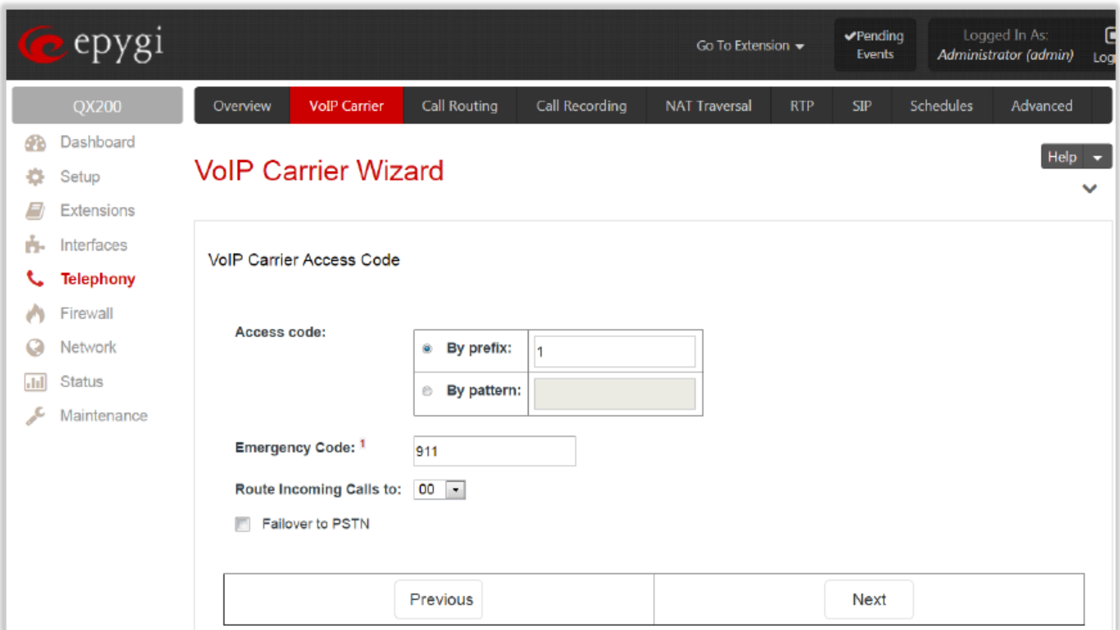


Figure 3: VoIP Carrier Wizard – Page 3

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

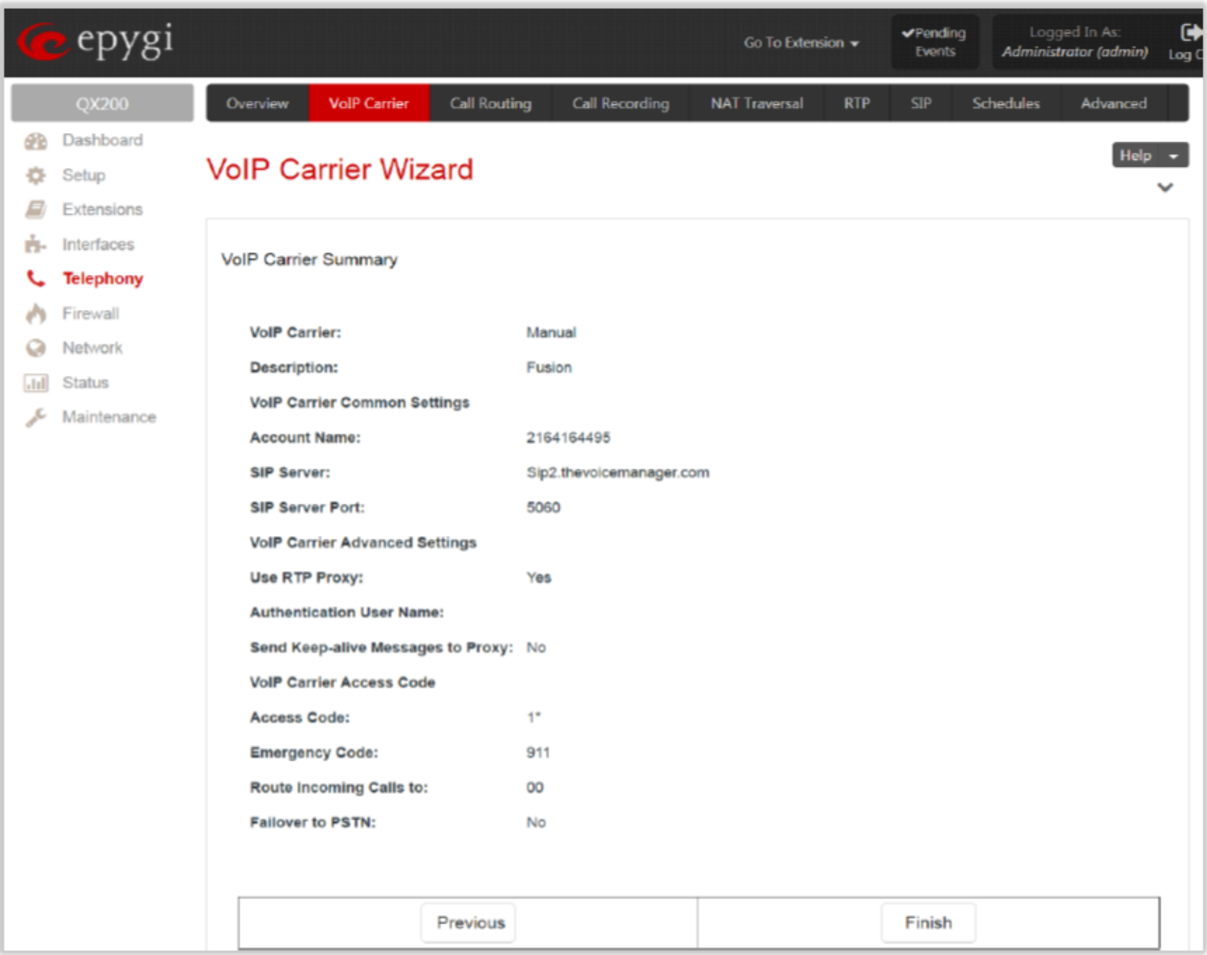


Figure 4: VoIP Carrier Wizard – Summary page

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).

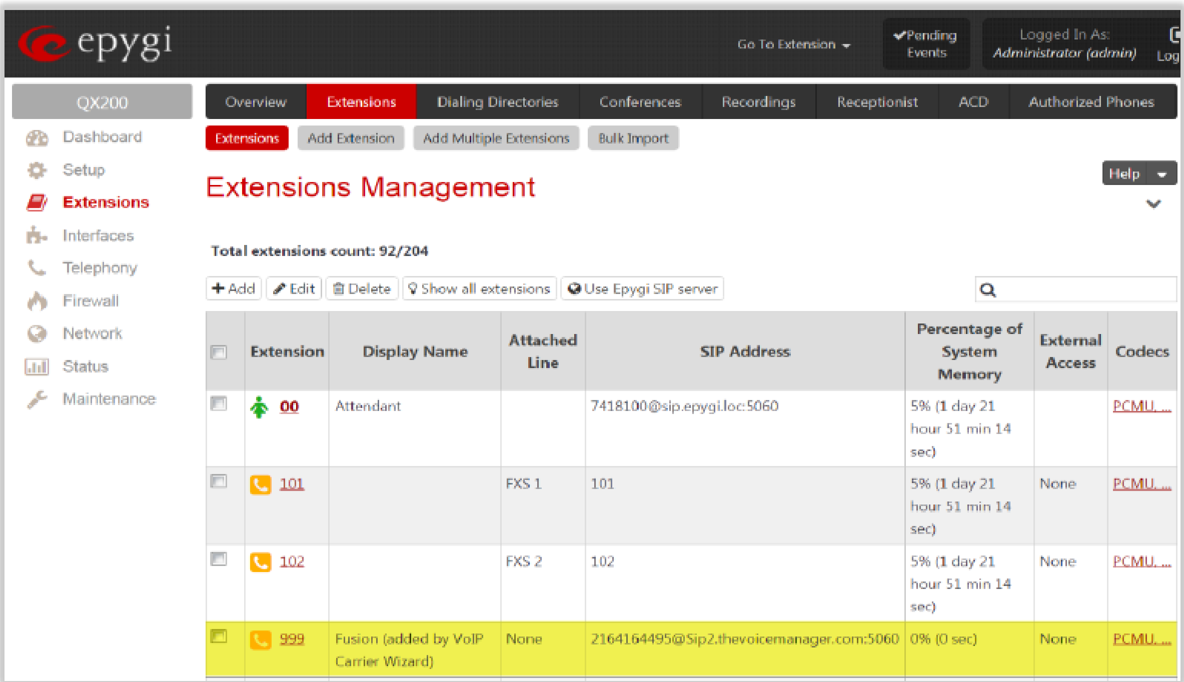


Figure 5: Extensions Management page

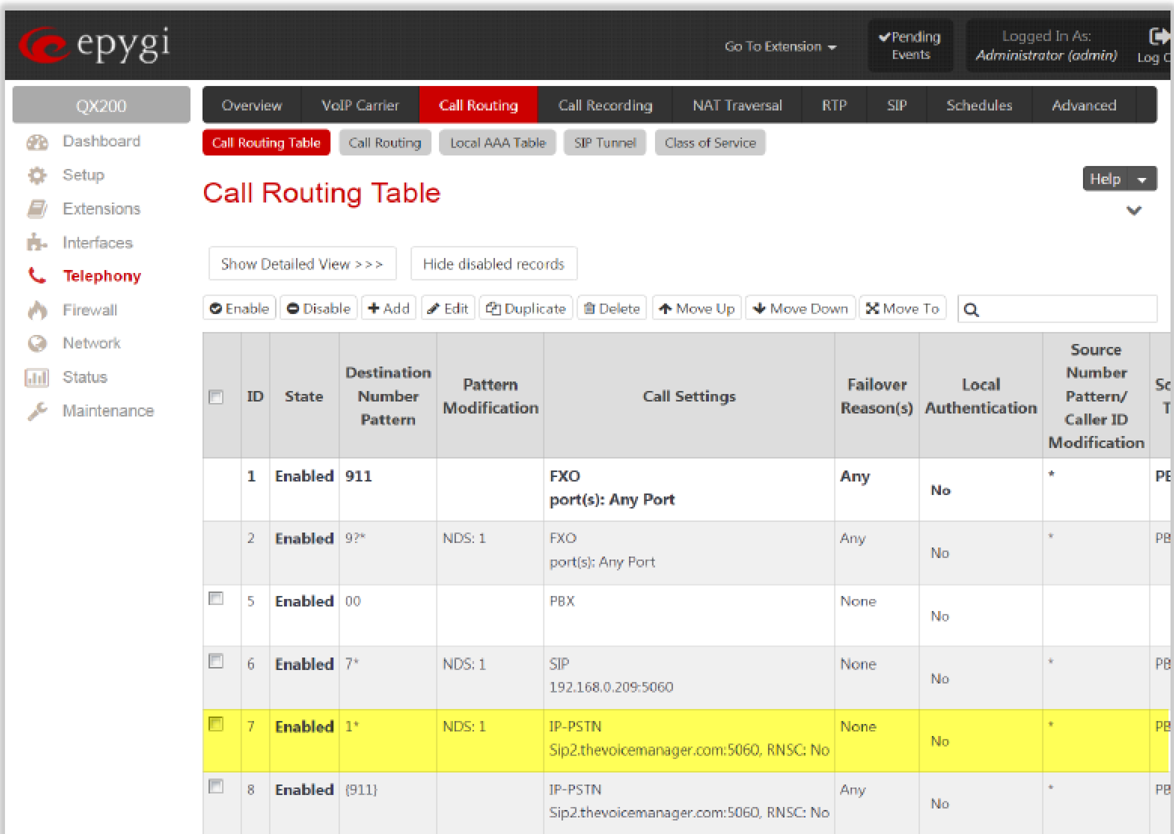


Figure 6: Call Routing Table page

<u>How this rule works</u>: The system will route all outbound calls mathcin 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 create though 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).

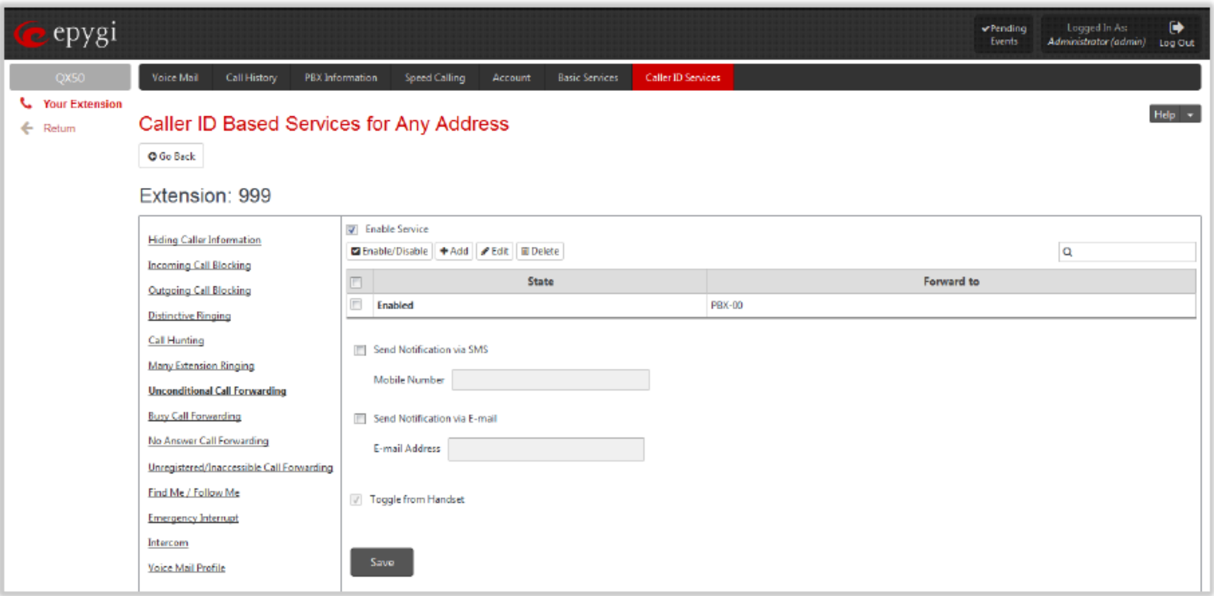
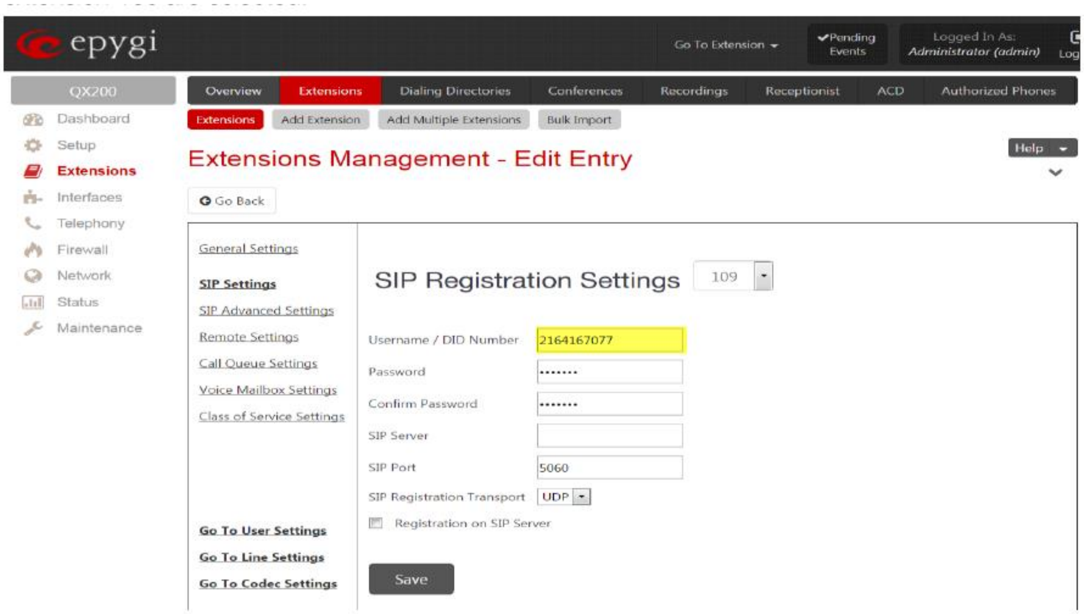


Figure 7: Caller ID Based Services - Unconditional Call Forwarding page

2. 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.

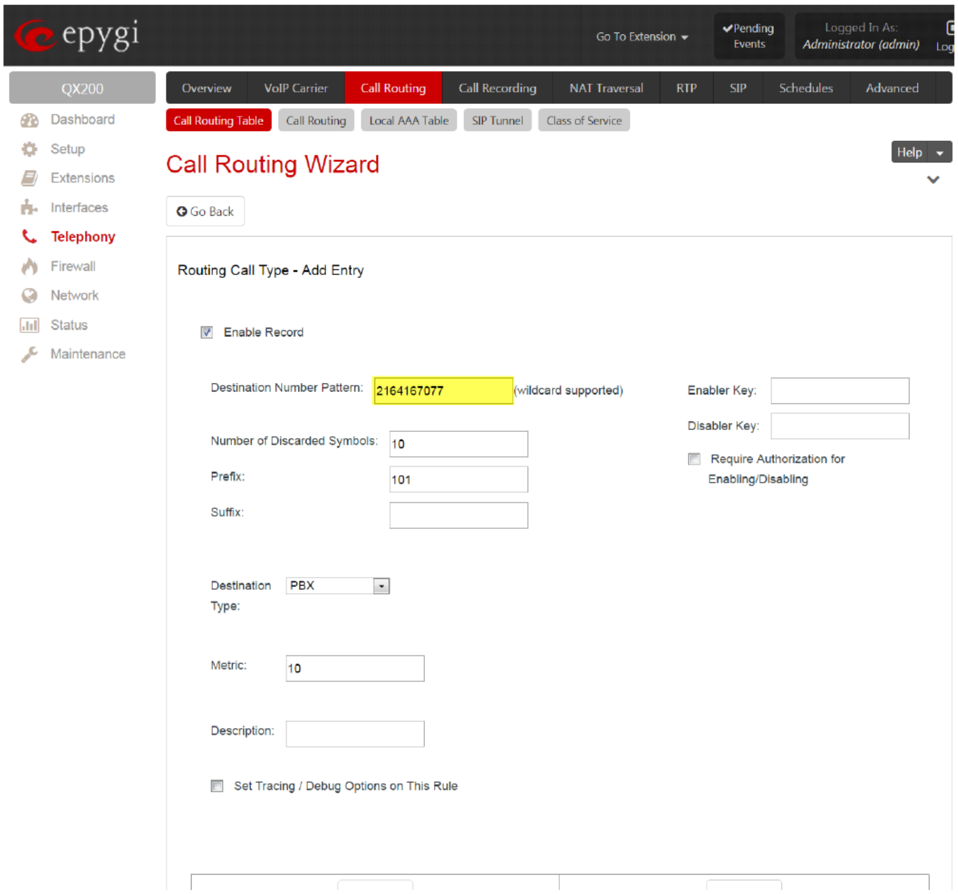


Figure 9: Call Routing Wizard page

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.

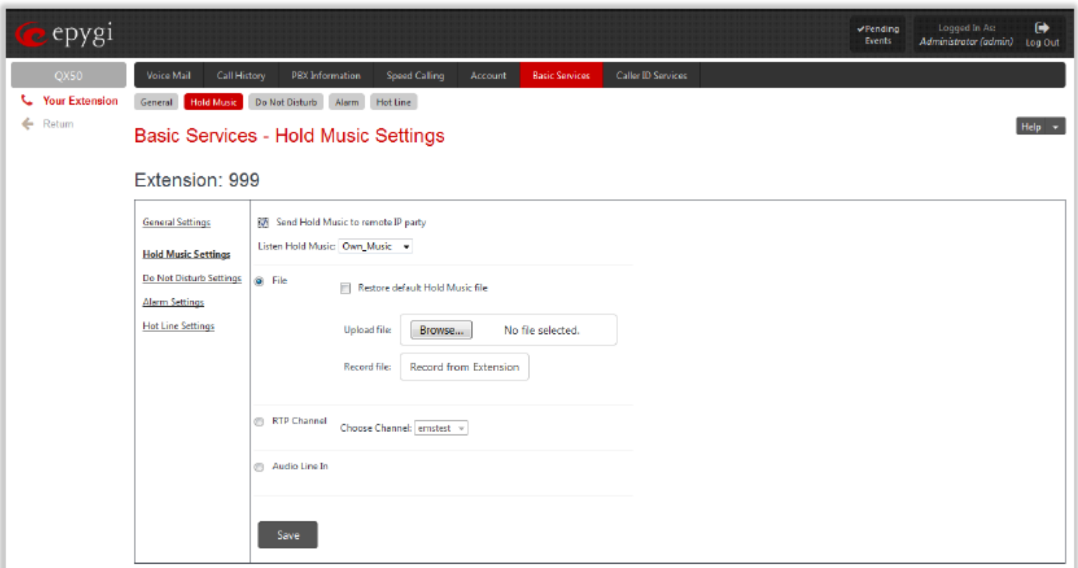


Figure 10: Basic Services – Hold Music Settings page

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.

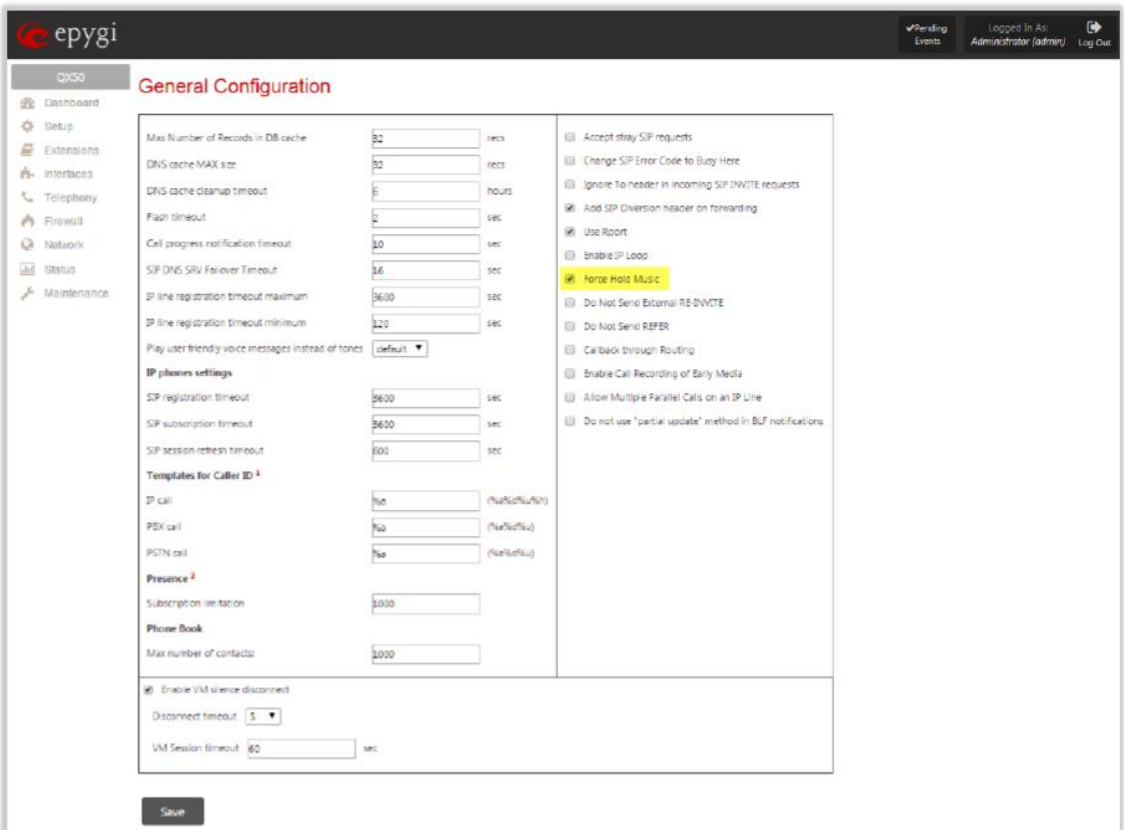


Figure 11: General Configuration Page

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