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.**

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).

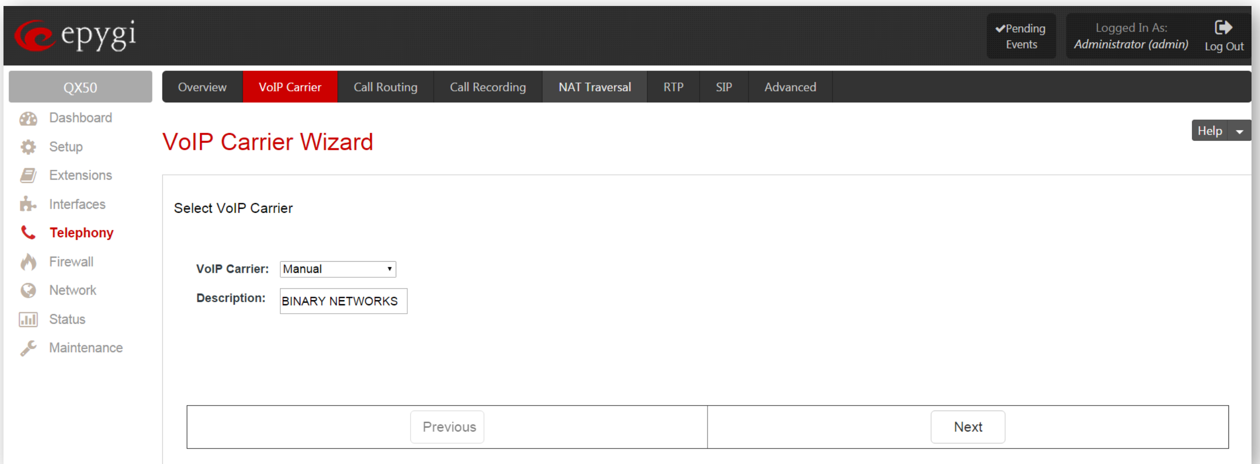

Figure 1: VoIP Carrier Wizard- Page 1

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).

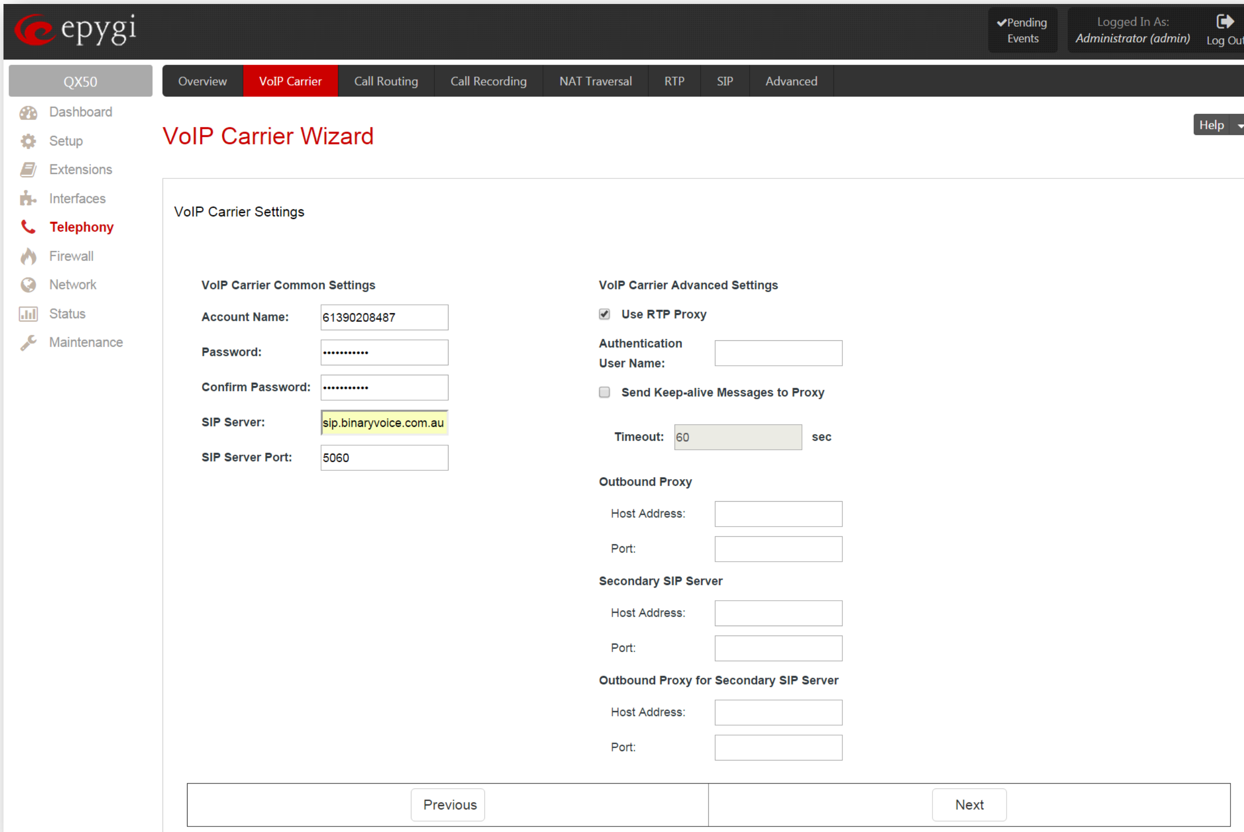

Figure 2: VoIP Carrier Wizard- Page 2

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).

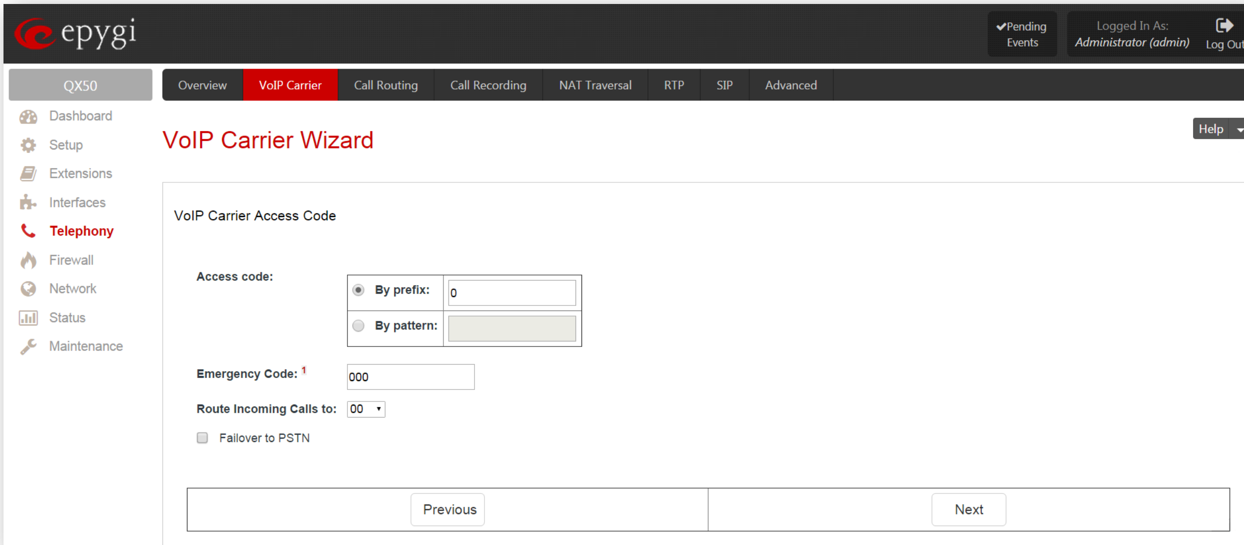

Figure 3: VoIP Carrier Wizard- Page 3

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

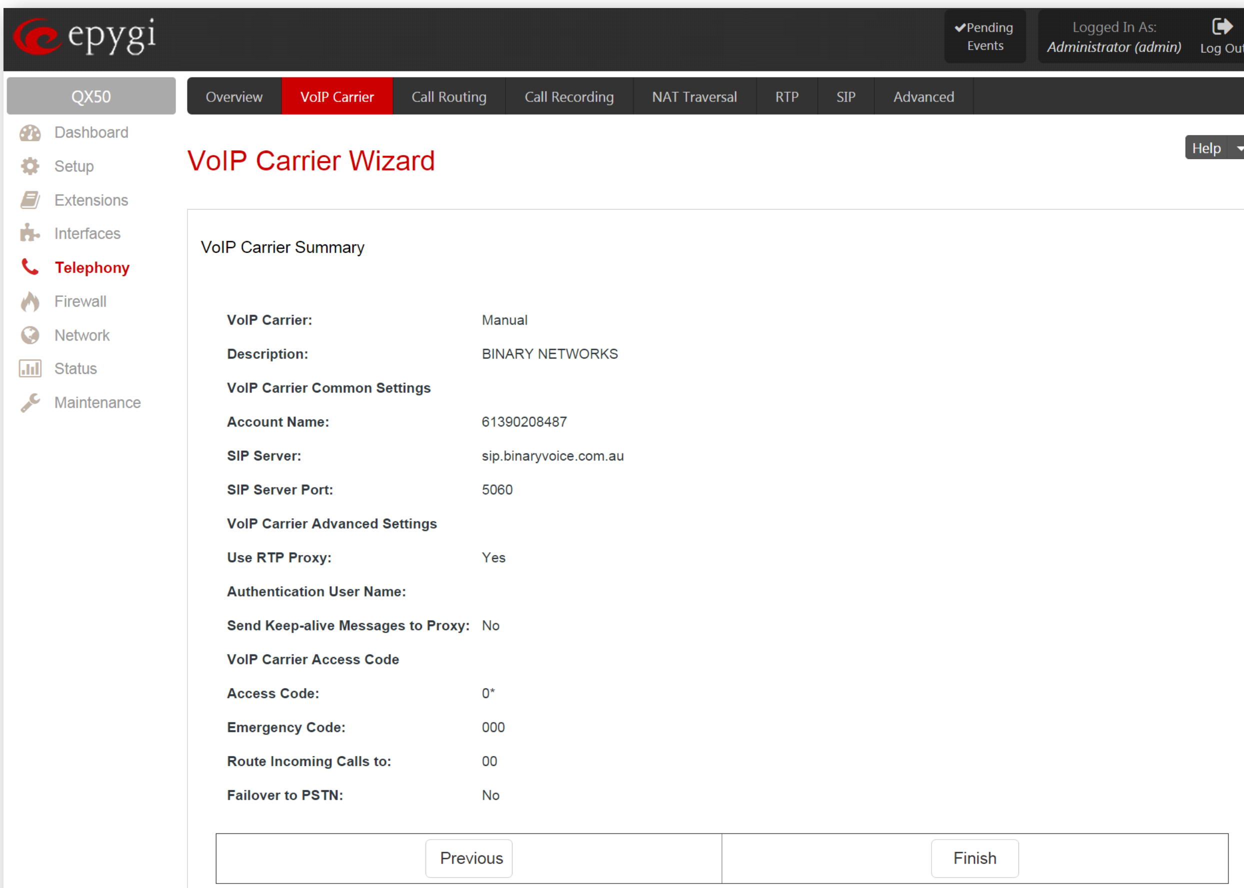

Figure 4: VoIP Carrier Wizard- Page 4

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).

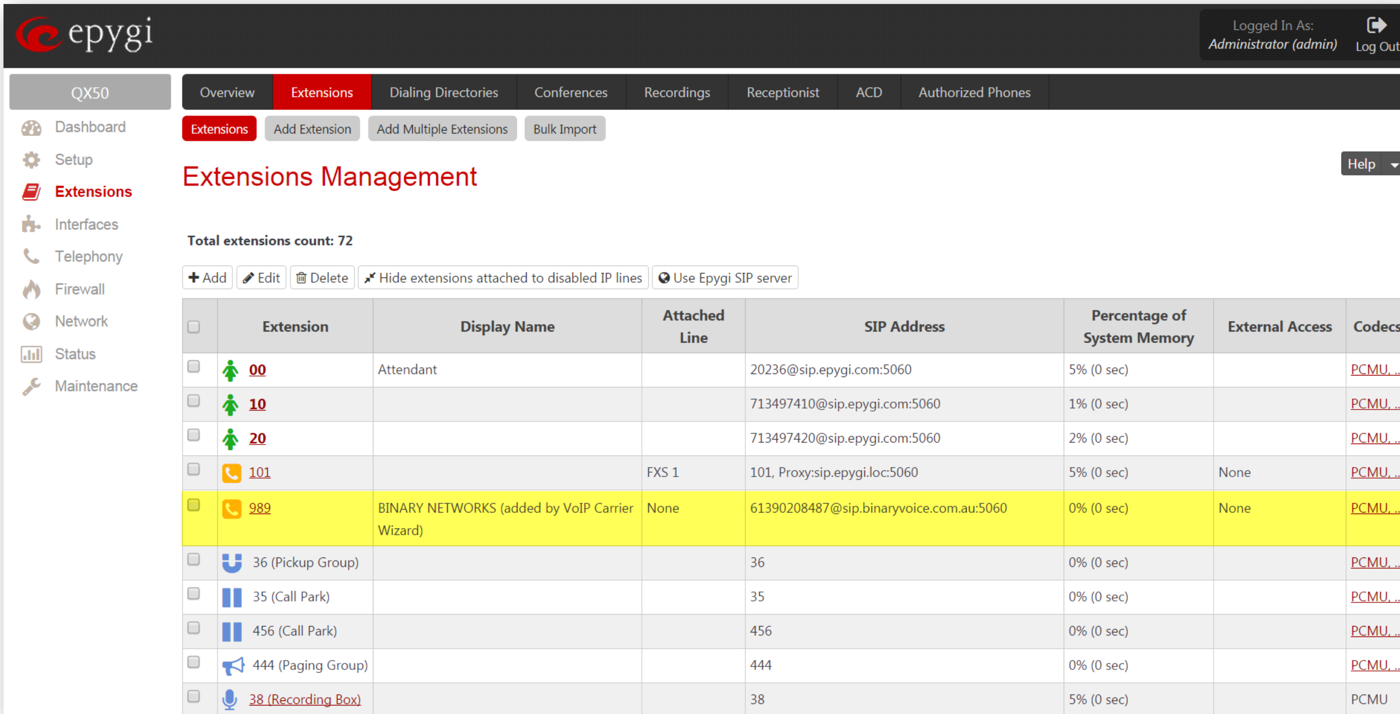

Figure 5: Call Routing Table page

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).

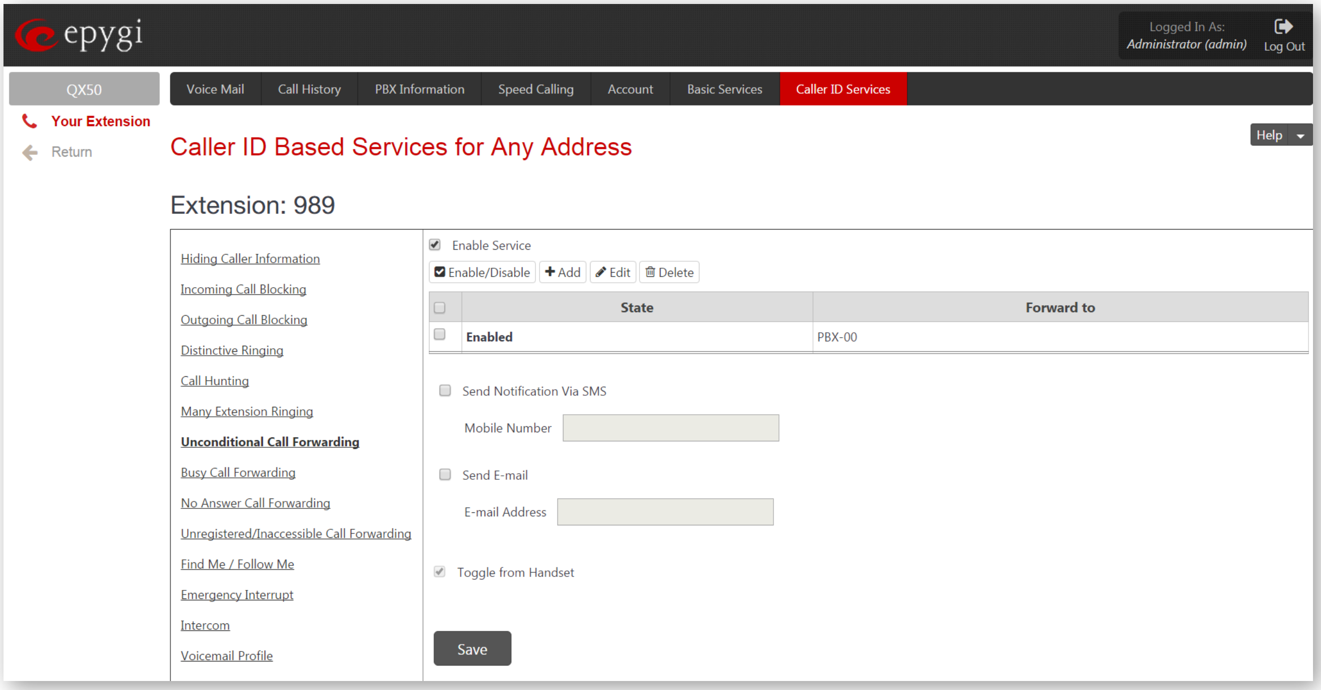

Figure 6: Unconditional Call Forwarding page for extension 989

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.

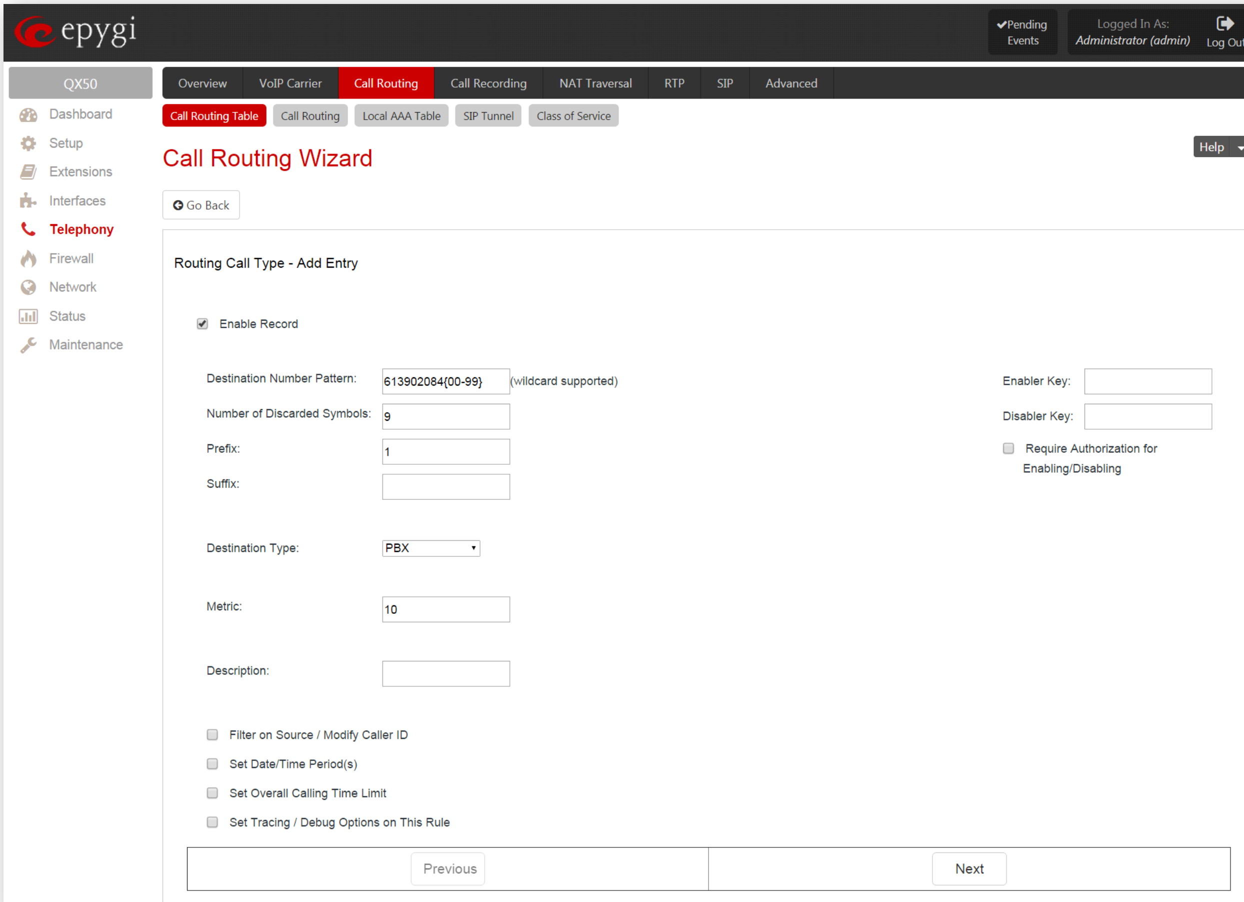

Figure 7: Call Routing Wizard page

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**.

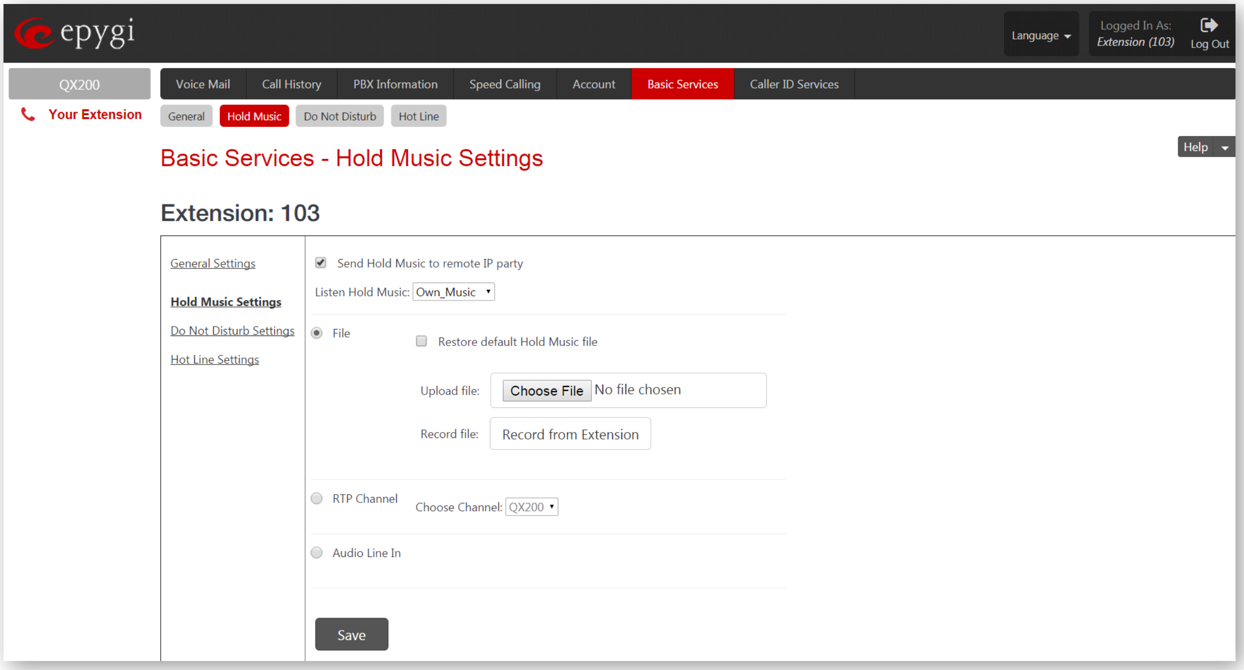

Figure 8: Basic Services – Hold Music Settings page

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.**

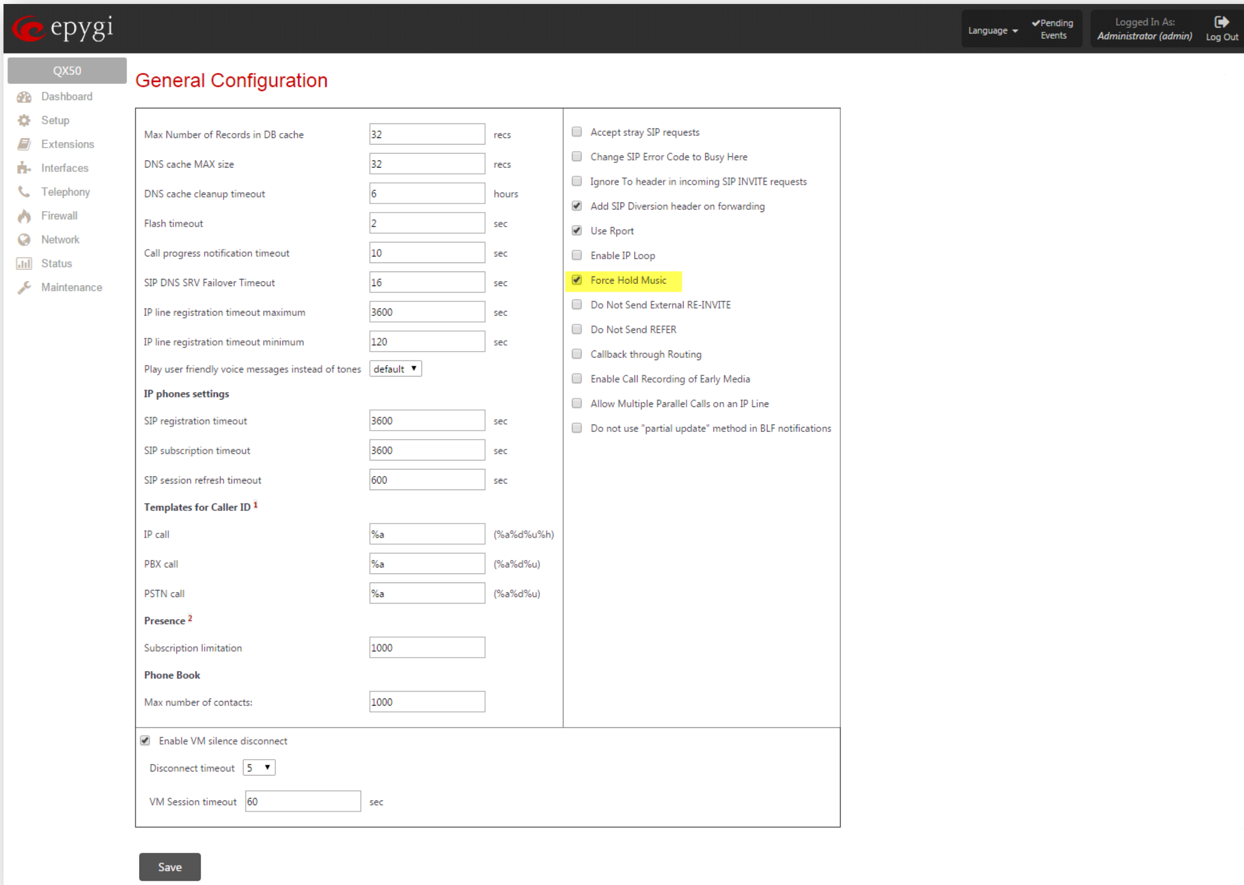

Figure 9: General Configuration page

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;
2. 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:

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

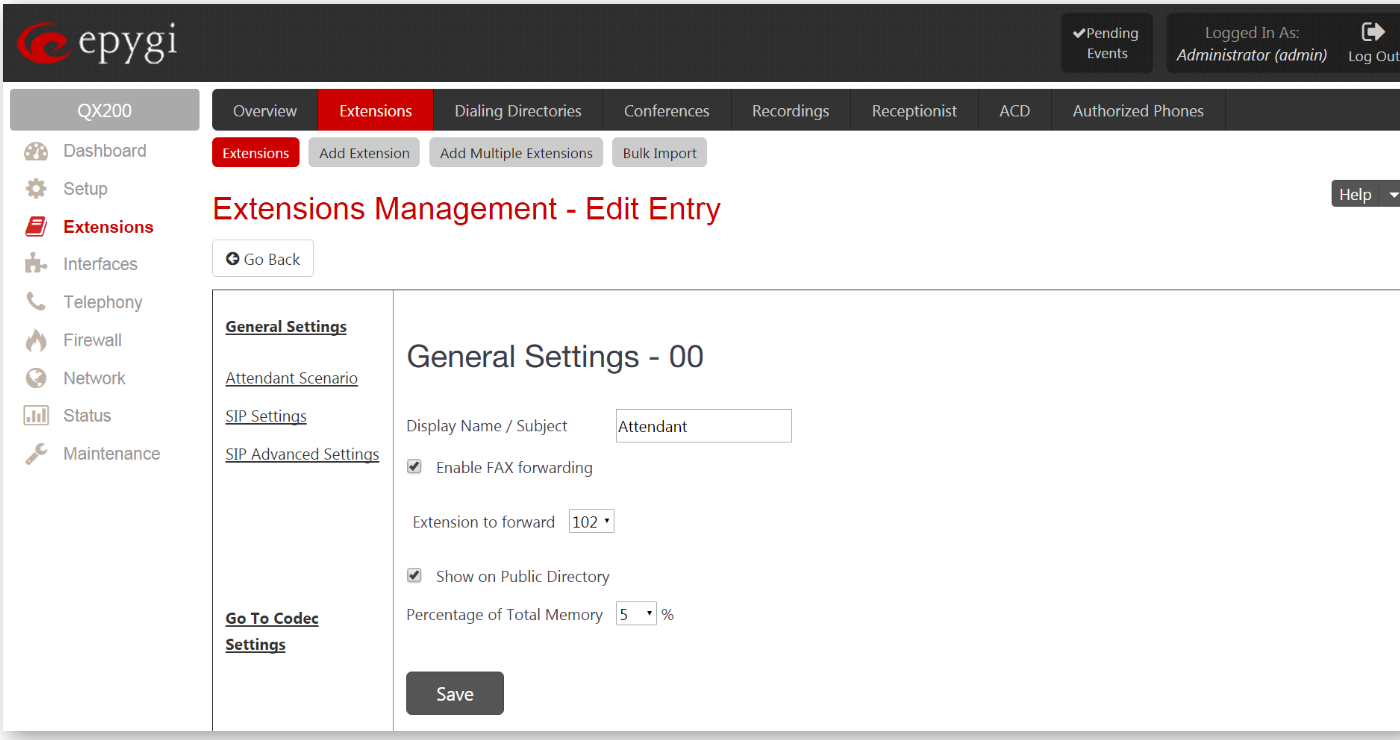

Figure 10: Auto Atendant General Settings page

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).

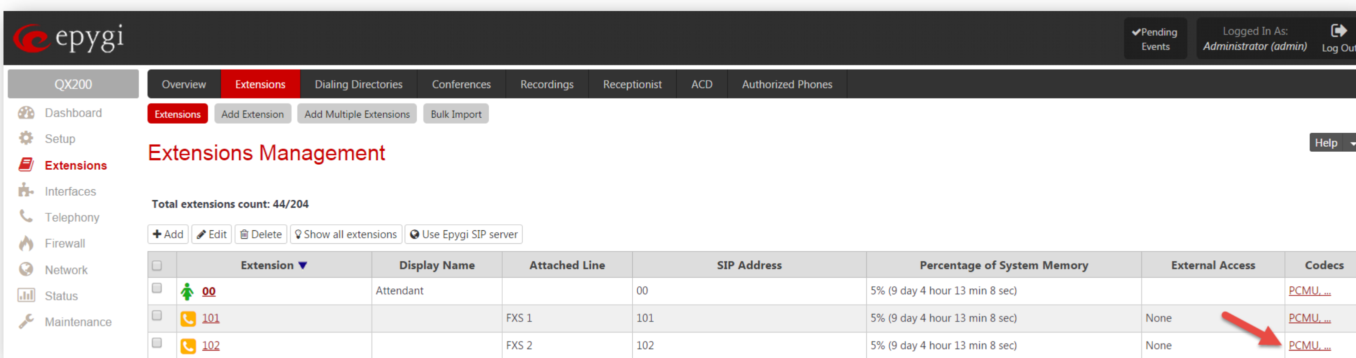

Figure 11: Extension Management page

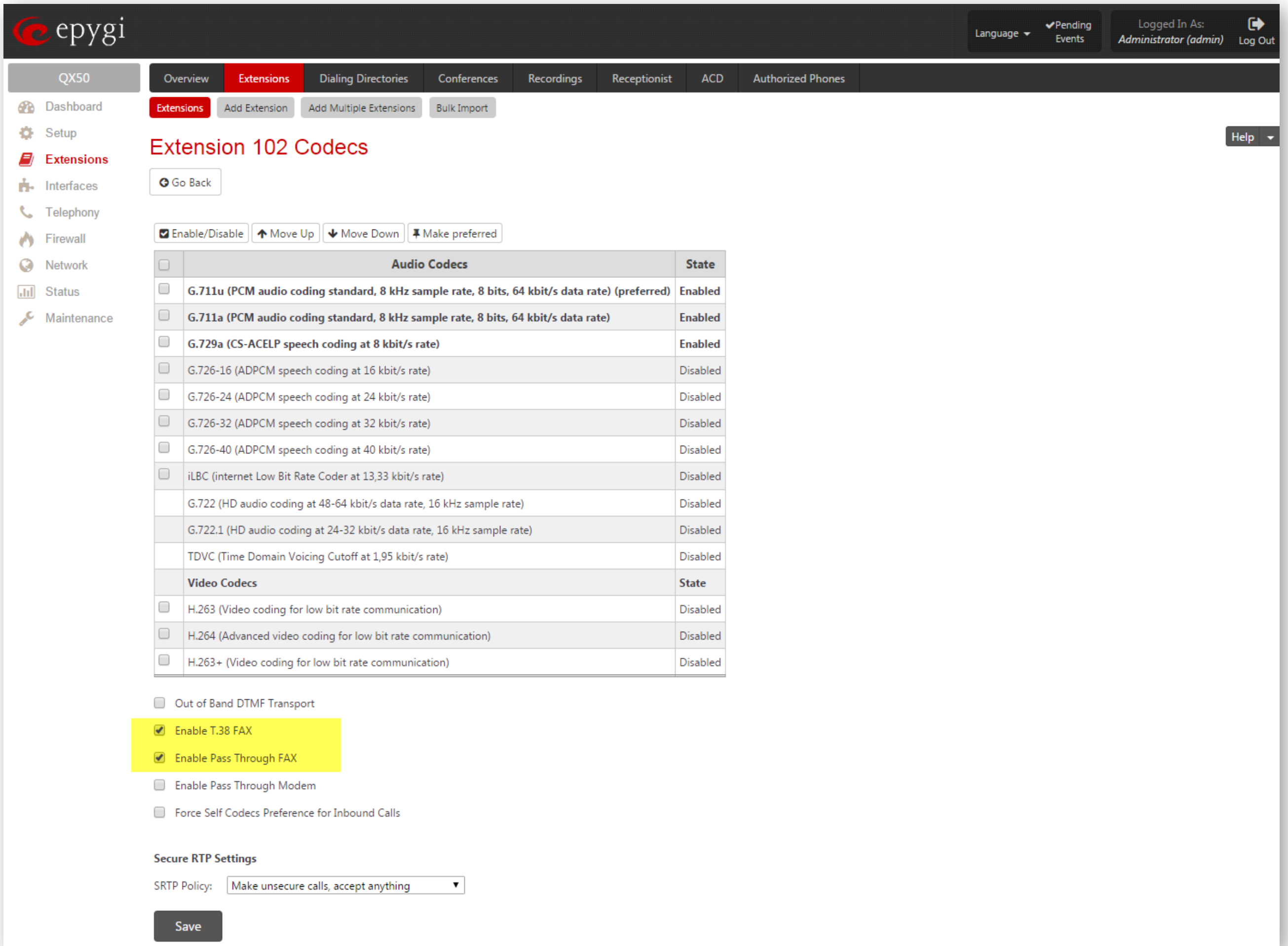

Figure 12: Codecs page for extension 102

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 www.epygi.com:

- 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