Configuring Epygi QX IP PBXs with Flowroute SIP Trunks

Abstract: This document describes the configuration of the Epygi QX IP PBXs to use the SIP trunking service from Flowroute.

Document Revision History

Revision	Date	Description	Valid for FW	Valid for Models
1.0	26-Sep-16	Initial Release	6.1.x and higher	QX IP PBXs

Table of Contents

1 Introduction

This document describes how to configure the Epygi QX IP PBXs (herein IP PBXs) to use the VoIP SIP trunking service from Flowroute. The IP PBX is capable of making IP-PSTN calls via Flowroute 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 Epygi QX IP PBX models, such as the QX50/QX200/QX2000/QXISDN4+.
- 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: Flowroute 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 Flowroute 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 Flowroute

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

- Username – 49135259
- Password – ************
- SIP Registrar – sip.flowroute.com
- Signaling port for SIP Registrar – 5060
- Telephone number(s) (DIDs allocated to the customer) – 12012994794, 12015354360

3 Configuration

The **Flowroute** allows both IP-based authentication and SIP registration when configuring VoIP PBX systems. The QX VoIP Carrier Wizard supports both methods; however, the SIP registration is used in the configuration below.

The two sections below describe the configurations required on the IP PBX to allow the users to

- Make outgoing calls through the **Flowroute** SIP trunks.
- Receive incoming calls from the **Flowroute** SIP trunks.

3.1 Making Outgoing Calls through Flowroute

To create a new extension on the IP PBX and configure it with the provided account automatically follow the steps below:

1. Go to the **Telephony→VoIP Carrier Wizard** page, pass through the wizard by inserting the below listed parameters:
   - Select **Manual** for VoIP Carrier
   - **Description** – optional
   - Press **Next** (Figure 1).

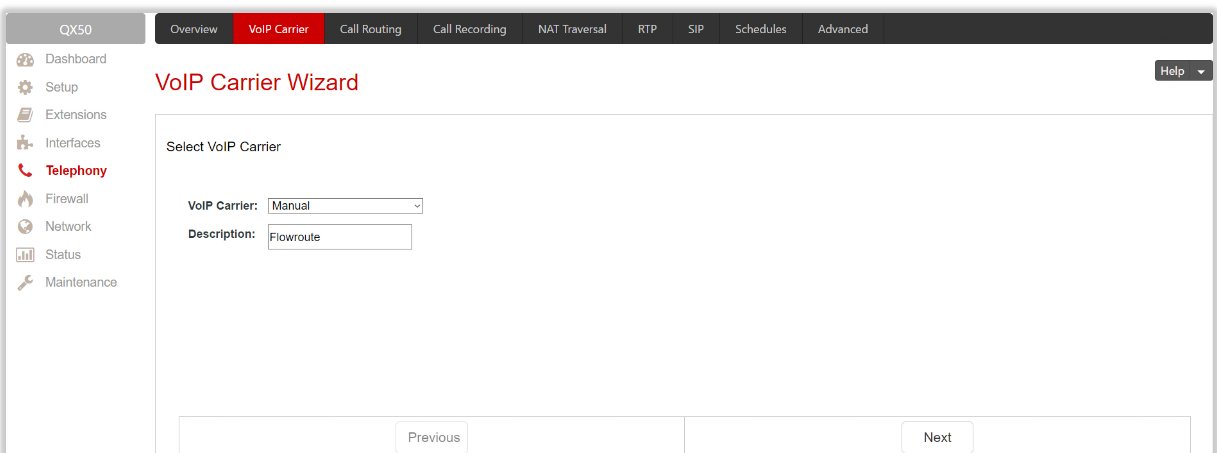

Figure 1: VoIP Carrier Wizard – page 1

2. Insert the following parameters in the next opened page:
   - **Account Name** – the Authentication Username provided by the **Flowroute** (e.g. **49135259**)
   - **Password** – ************
   - **SIP Registrar** – sip.flowroute.com
   - **SIP Server Port** – 5060
   - Enable **Use RTP Proxy** service and press **Next** (Figure 2).

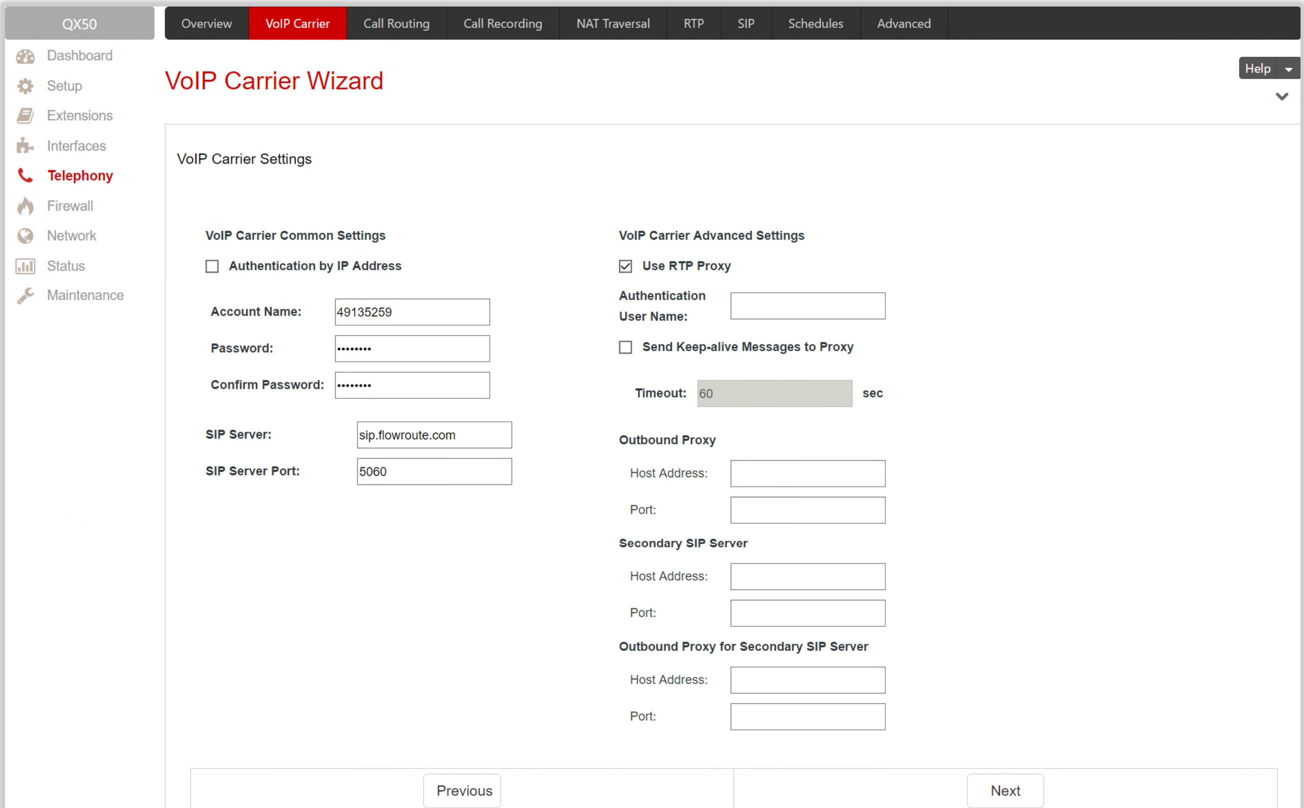

Figure 2: VoIP Carrier Wizard – page 2

3. On the third page of the VoIP Carrier Wizard define the Access Code (e.g. 1) which will be used in the IP PBX Call Routing Table for making outgoing calls through the Flowroute, and the IP PBX extension which will receive all incoming calls from the Flowroute SIP trunks (e.g. extension 00; it is the IP PBX's default Auto Attendant). 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 your country emergency call
   - Route Incoming Calls to – 00.
4. Enable the Failover to PSTN service if it is desirable to allow calls failover through IP PBXs onboard FXO lines or trunks (*N/A* on the QX2000) and press Next (Figure 3).

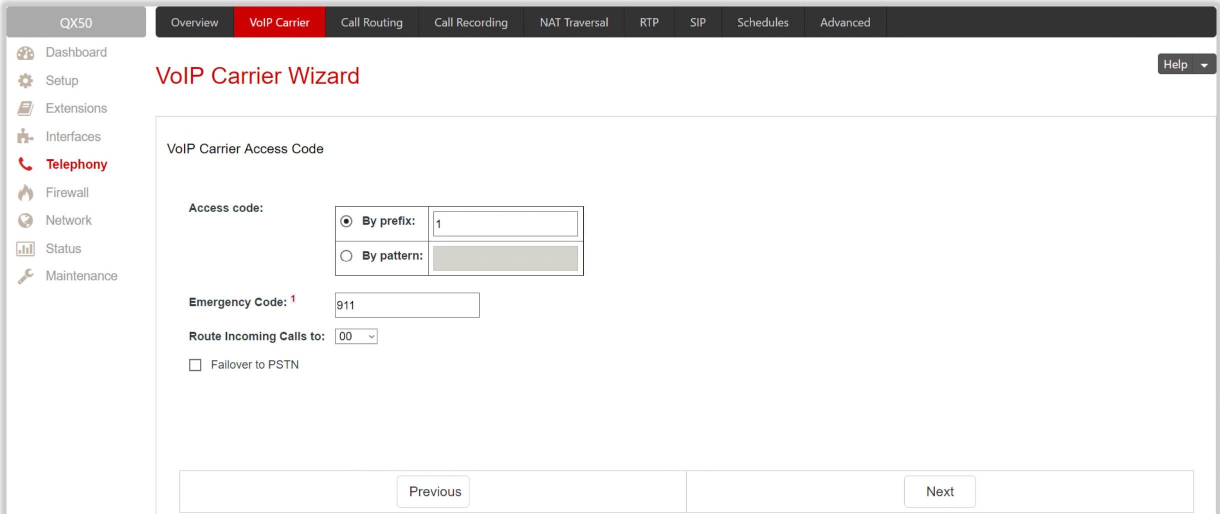

Figure 3: VoIP Carrier Wizard – Page 3

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

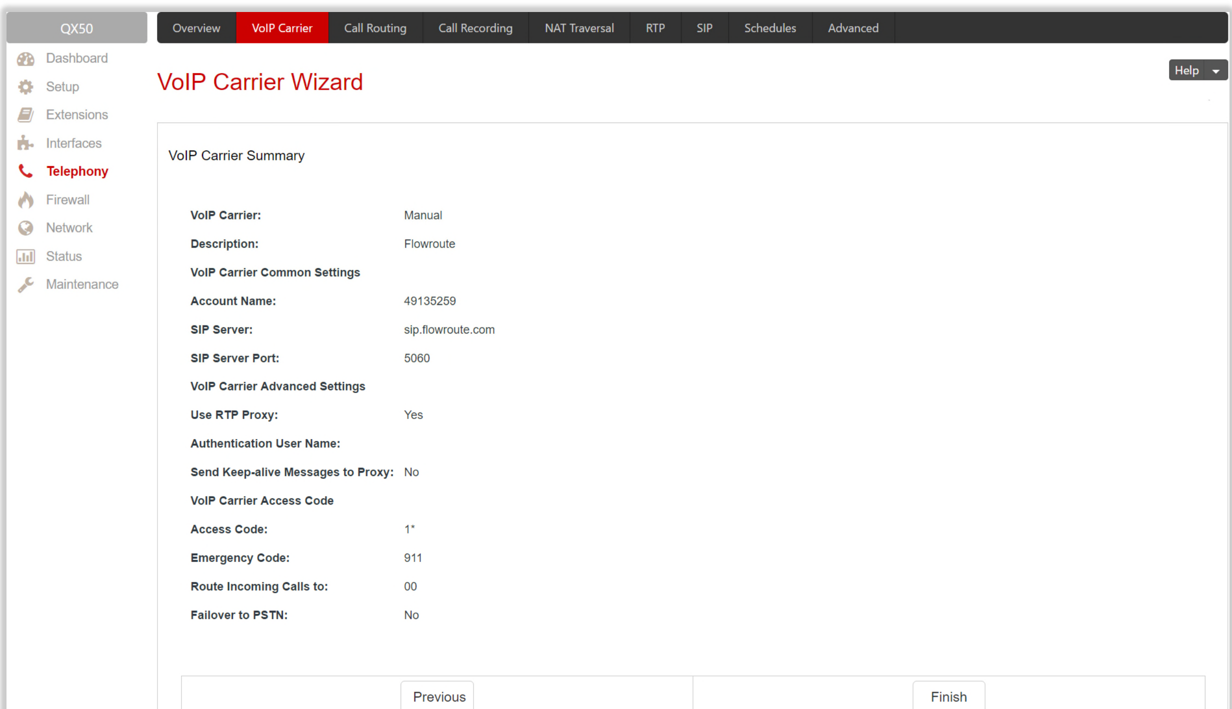

Figure 4: VoIP Carrier Wizard – Summary page

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

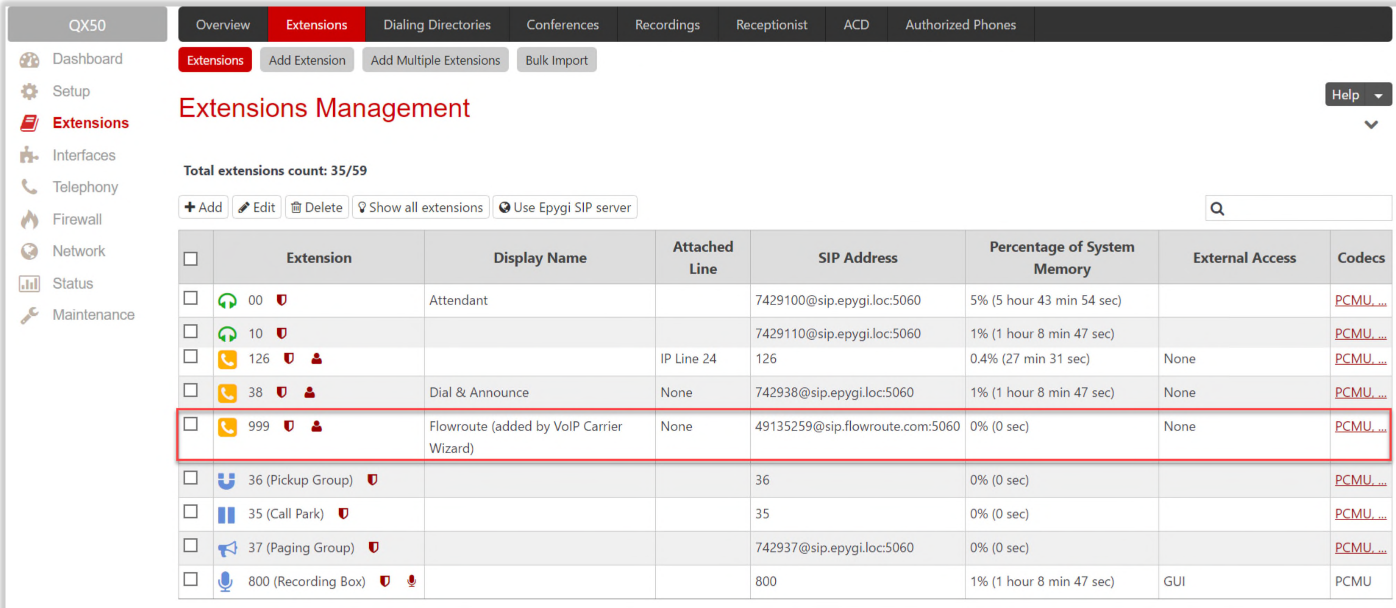

Figure 5: Extensions Management page

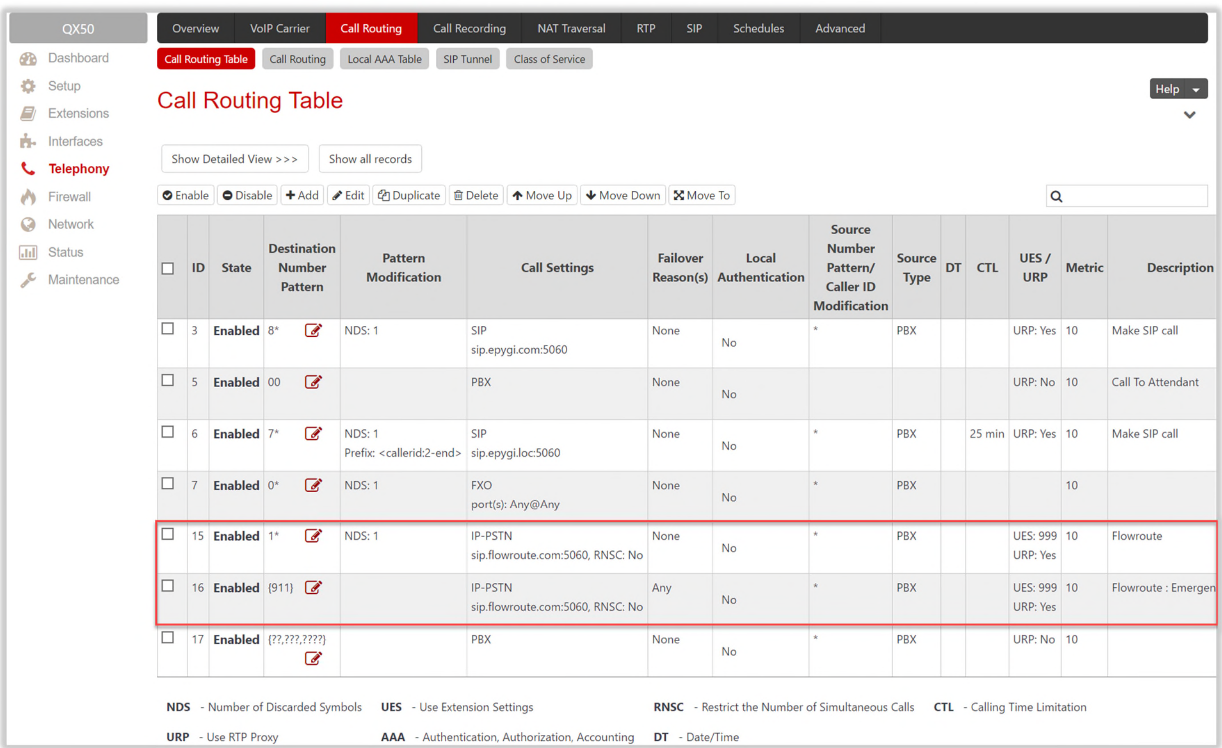

Figure 6: Call Routing Table page

How this rule works: The IP PBX will route all outbound calls matching the prefix 1* through the **Flowroute** SIP trunks.

1.1 Modifying Caller ID for Outbound Calls

The **Flowroute** allows the outbound Caller ID to be modified. In the outgoing SIP Invite message **Flowroute** needs to see a known number in the P-Preferred field using the globalized E.164 format (+1 and 10 digits) in order for outbound calls to be allowed, recognized and called back. Epygi does support this and it is doable in the call routing entry for outgoing calls as follows:

1. Go to **Telephony→Call Routing** page, select the outgoing Call Routing entry *1 and press the **Edit** button. The **Call Routing Wizard** appears.
   - Enable the **Filter on Source/Modify Caller ID** option and press **Next**.

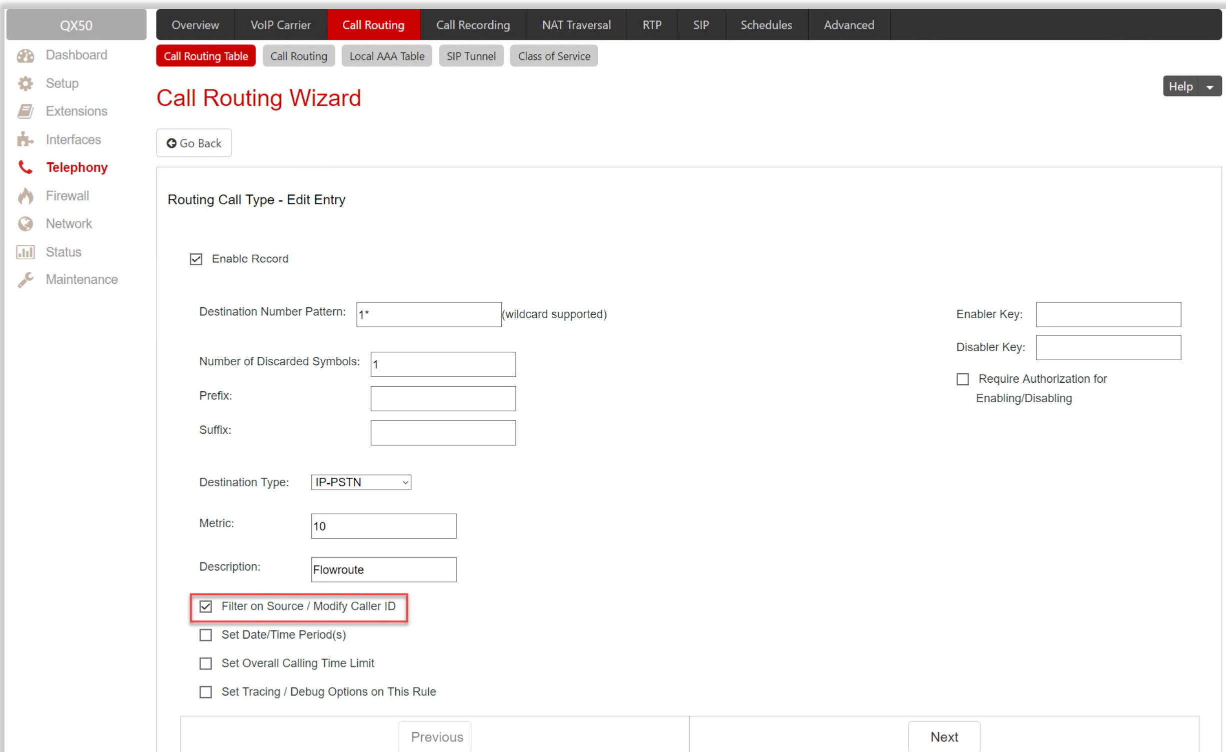

Figure 7: Call Routing Wizard page

2. On the second page deselect the **Use Extension Setting** field, fill-in the **Username** and **Password** fields with information from **Flowroute** and press Next (Figure 8).

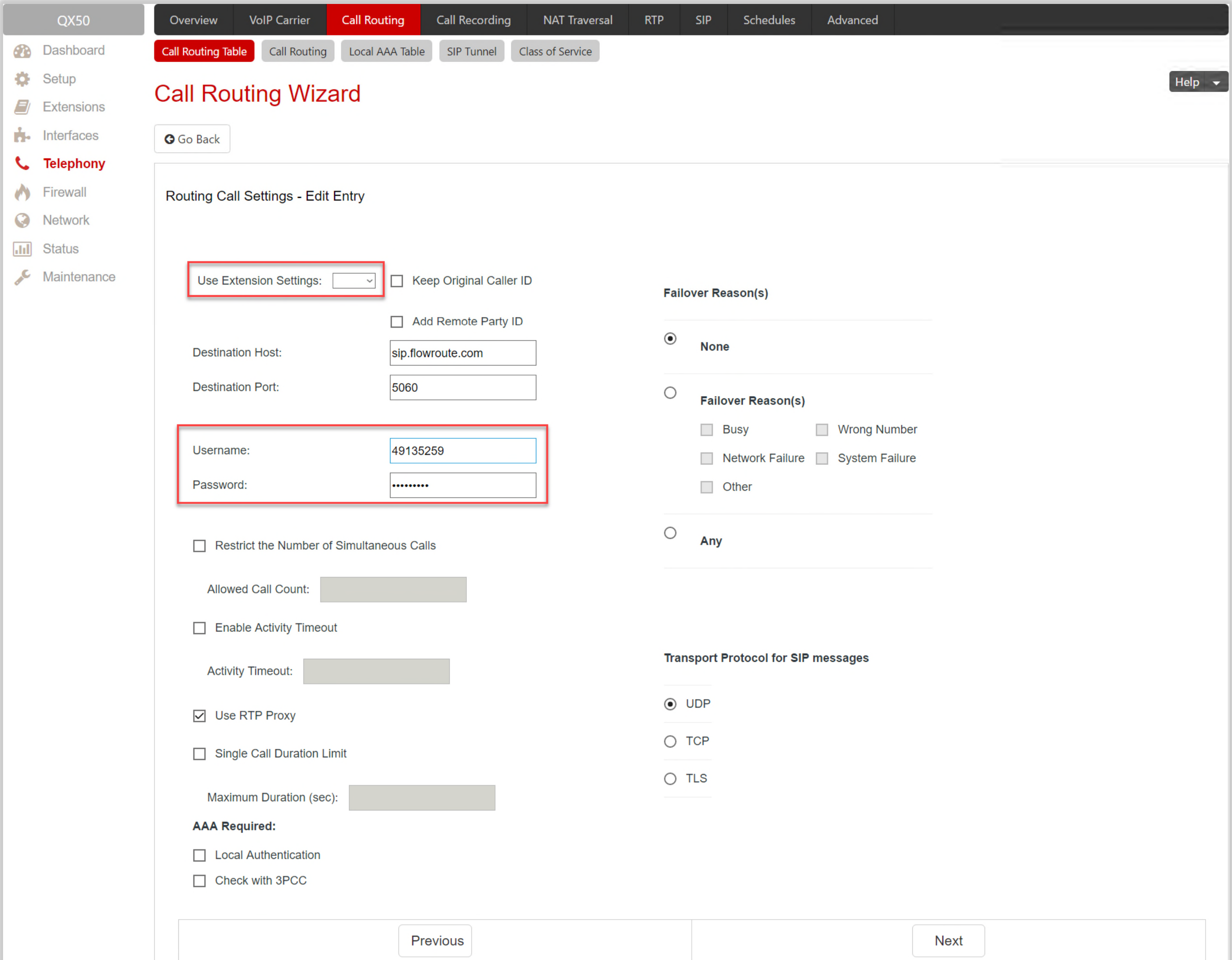

Figure 8: Call Routing Wizard page

3. On the next page, enter the following parameters:
   - Source Number Pattern – *
   - Source Type – PBX
   - Number of Discarded Symbols – 99;
   - Prefix – 12012994794 (new Caller ID number)

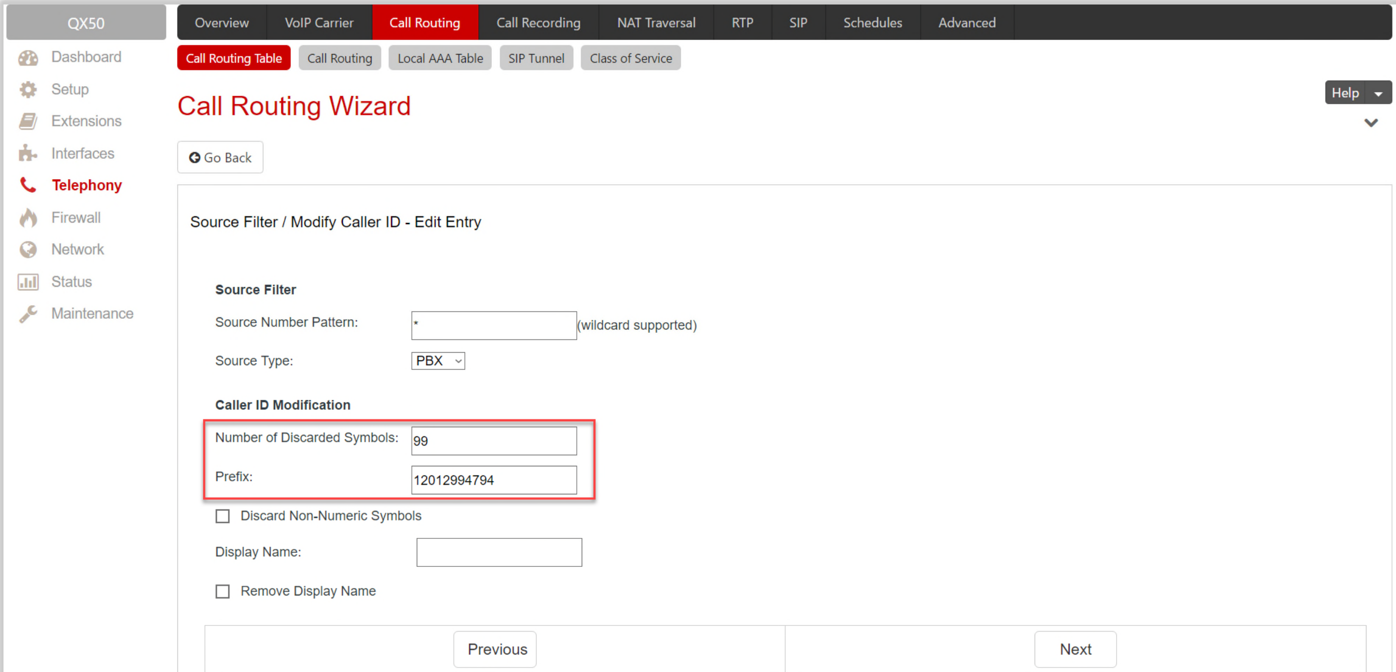

Figure 9: Call Routing Wizard page

4. Press **Next** and finish the wizard.

How this rule works: Outgoing calls will now have the new Caller ID as the DID **12012994794**.

3.2 Receiving Inbound Calls from Flowroute

Apply the provided DID number in the SIP Settings for the auto attendant extension **00,** in the **SIP Settings-Username/DID number** section, and save (Figure 10).

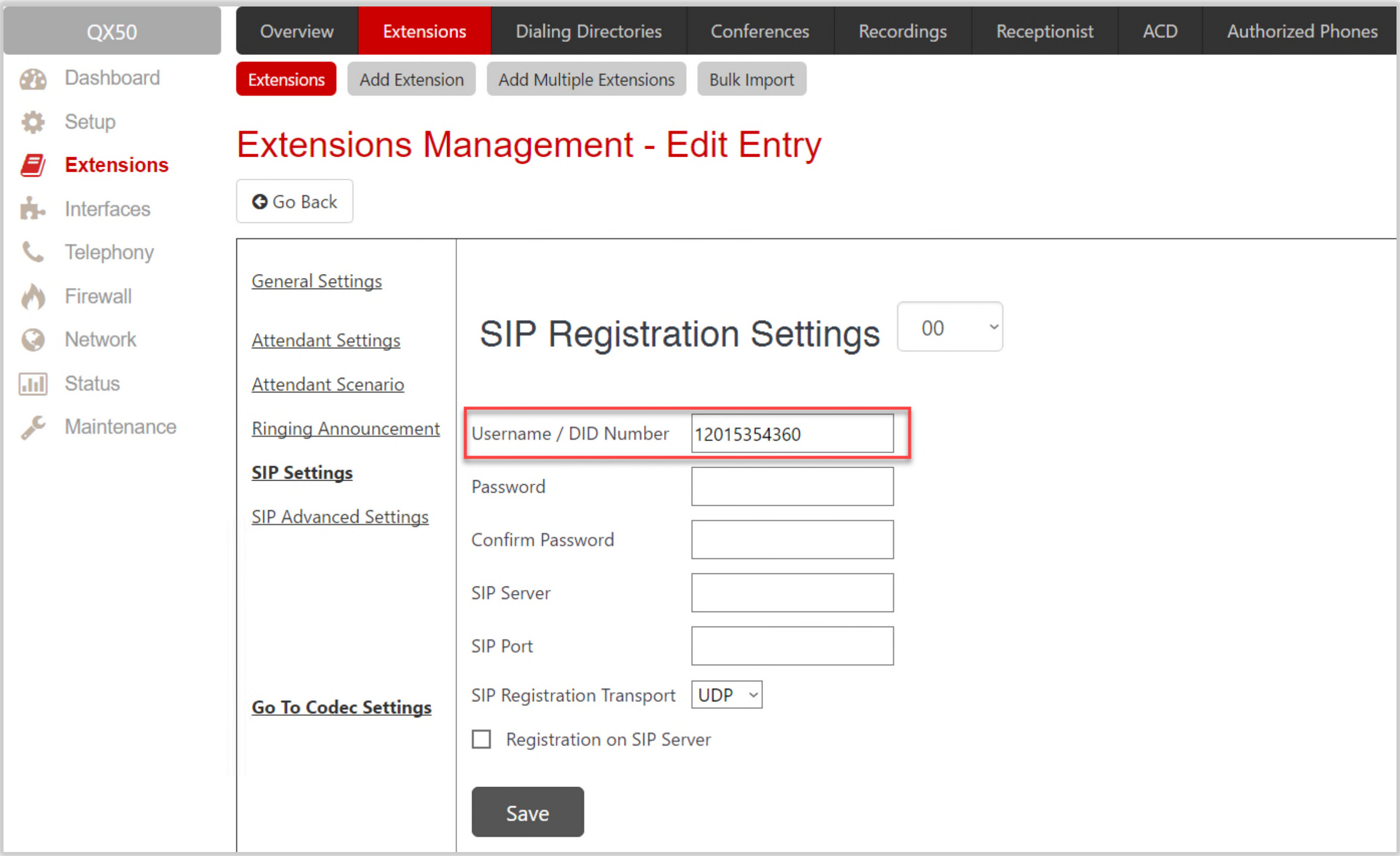

Figure 10: Extension 00 – SIP Settings page

How this configuration works: The system will route all inbound calls from the **Flowroute** SIP trunks matching the pattern **12015354360** to the auto attendant **00** on IP PBX.

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 11).
2. Enable the Send Hold Music to remote IP party checkbox and press Save.

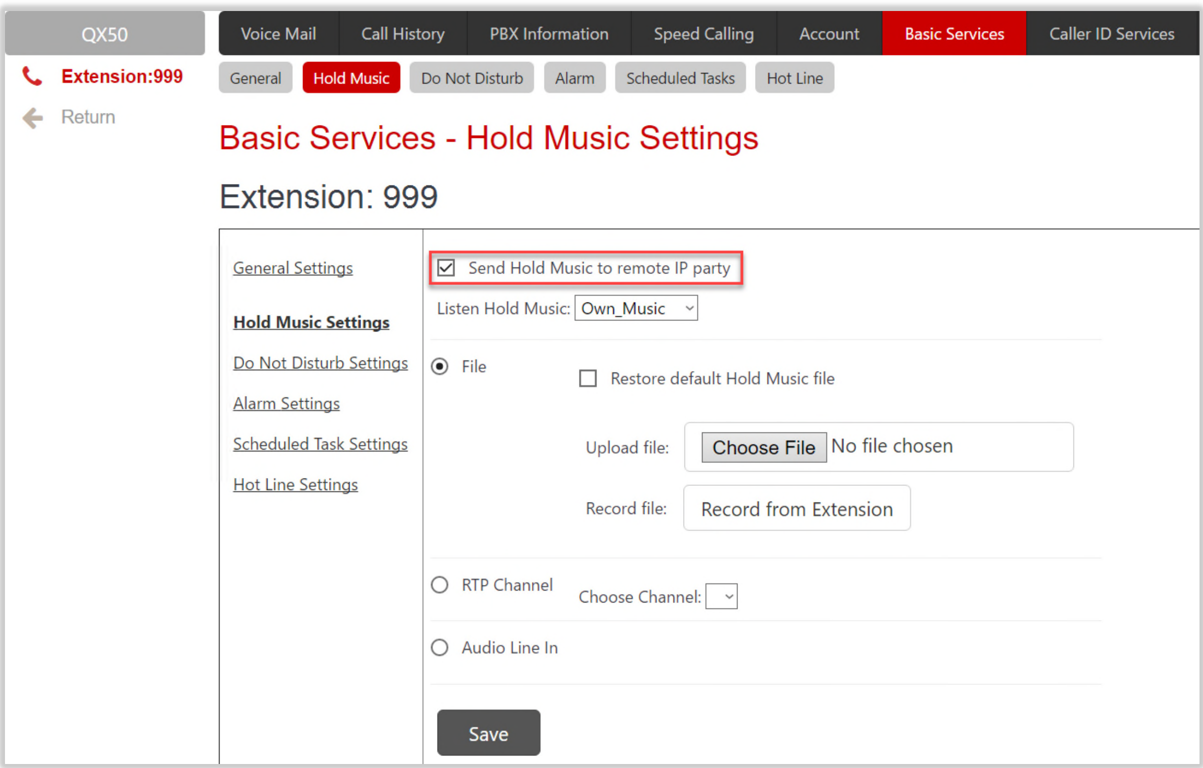

Figure 11: Basic Services – Hold Music Settings page

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:

1. Go to the "http://xxx.xxx.xxx.xxx/generalconfig.cgi" hidden page (Figure 12).
2. On this page, select the **Force Hold Music** checkbox and press **Save**.

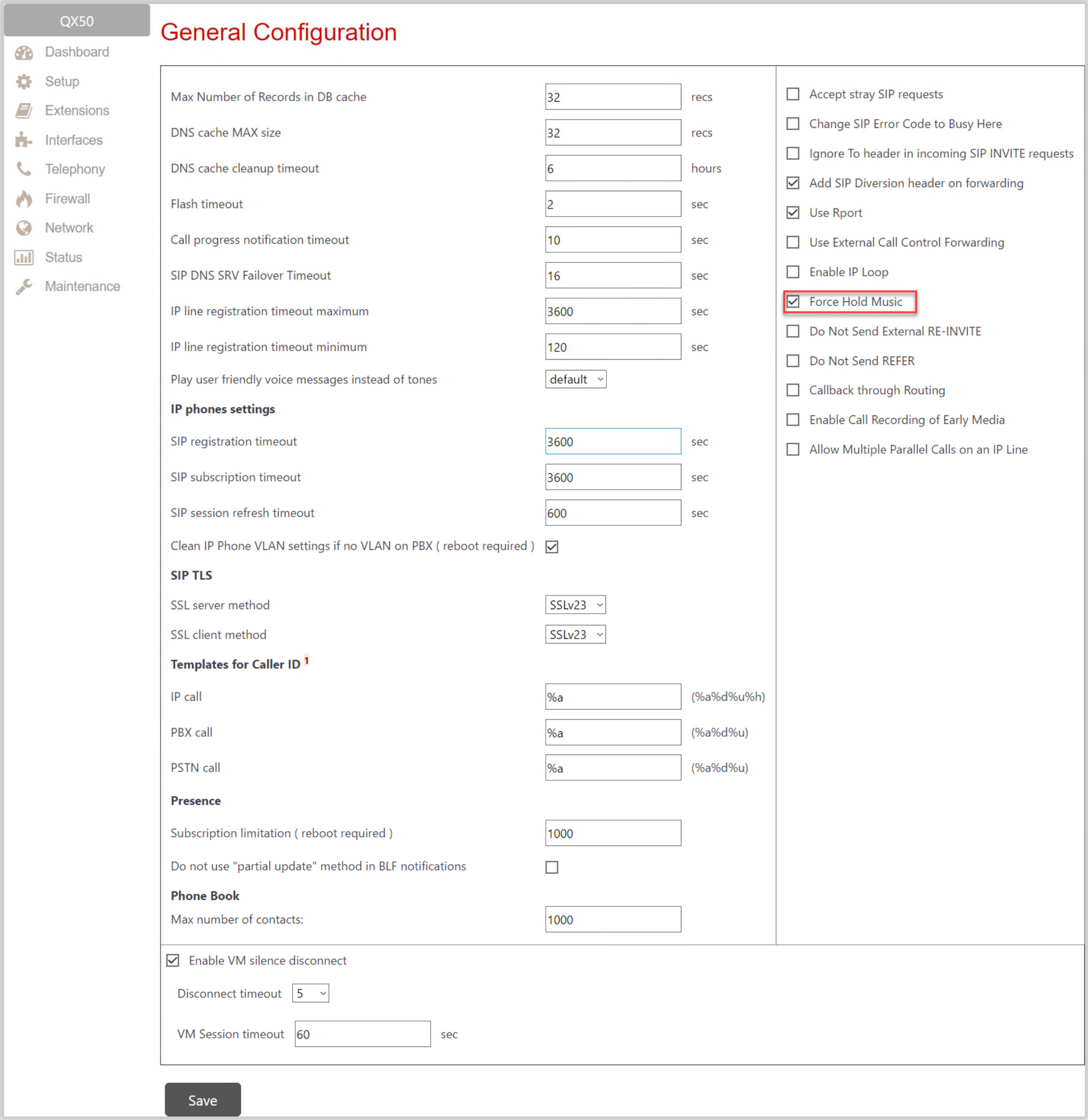

Figure 12: General Configuration page

4.2 Sending and Receiving Faxes through the Flowroute

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 **Flowroute** 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 on 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 (Figure 13).

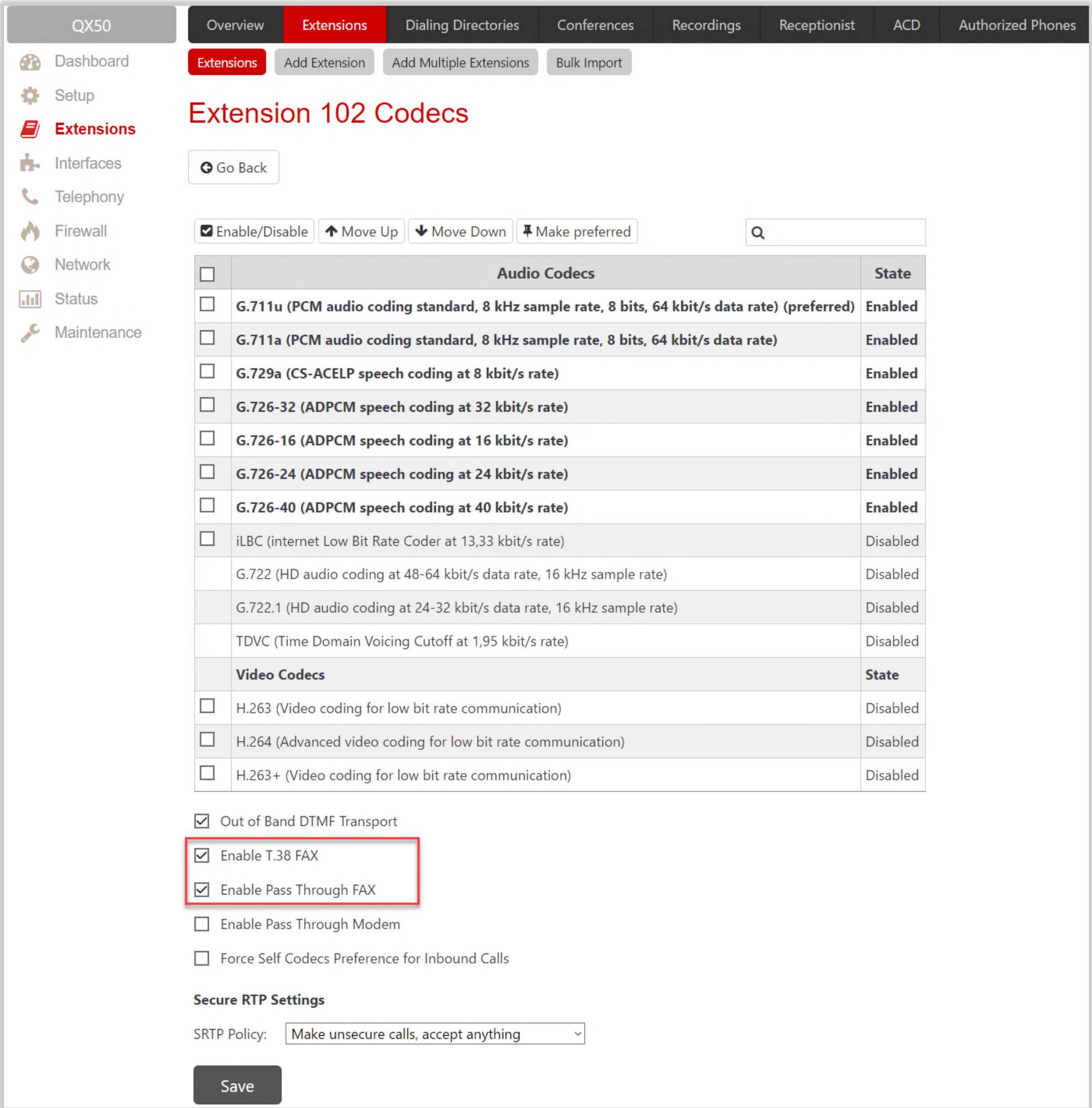

Figure 13: Codecs page for extension 102

These are the configuration options for receiving FAX on the IP PBX:

- 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 IP PBX 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.

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

Refer to the below listed recourses to get more details about the configurations described in this guide:

- Manual I – Installation Guide
- 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

Find the above listed documents on Epygi Support Portal.