



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.....	4
2	Scenario	4
2.1	Requirements and Preparations	4
2.2	Account Information from Flowroute	4
3	Configuration.....	5
3.1	Making Outgoing Calls through Flowroute.....	5
1.1	Modifying Caller ID for Outbound Calls.....	9
3.2	Receiving Inbound Calls from Flowroute	12
4	Additional Notes	13
4.1	Sending Music on Hold to Remote Parties.....	13
4.2	Sending and Receiving Faxes through the Flowroute	14
5	References.....	16

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).

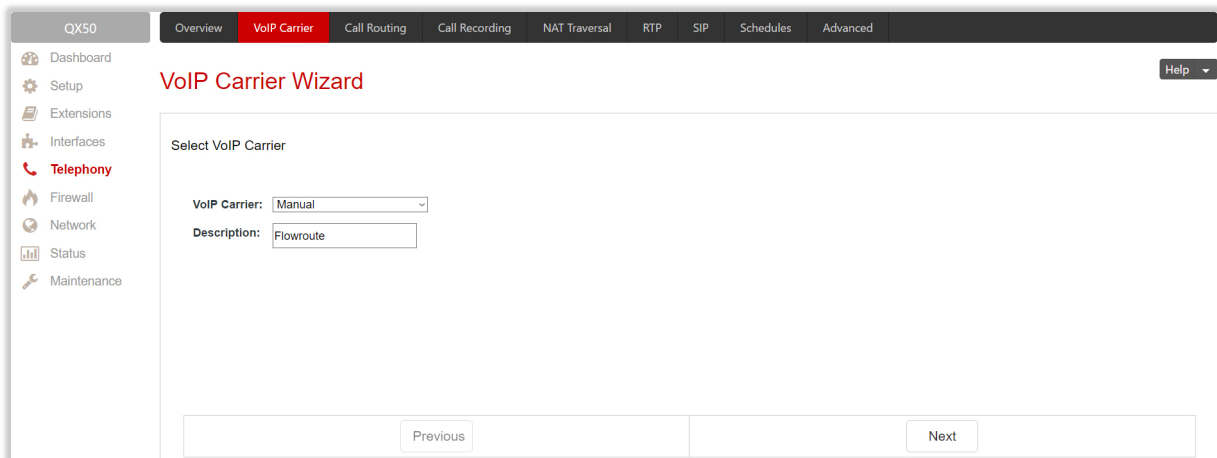


Figure 1: VoIP Carrier Wizard – page 1

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).

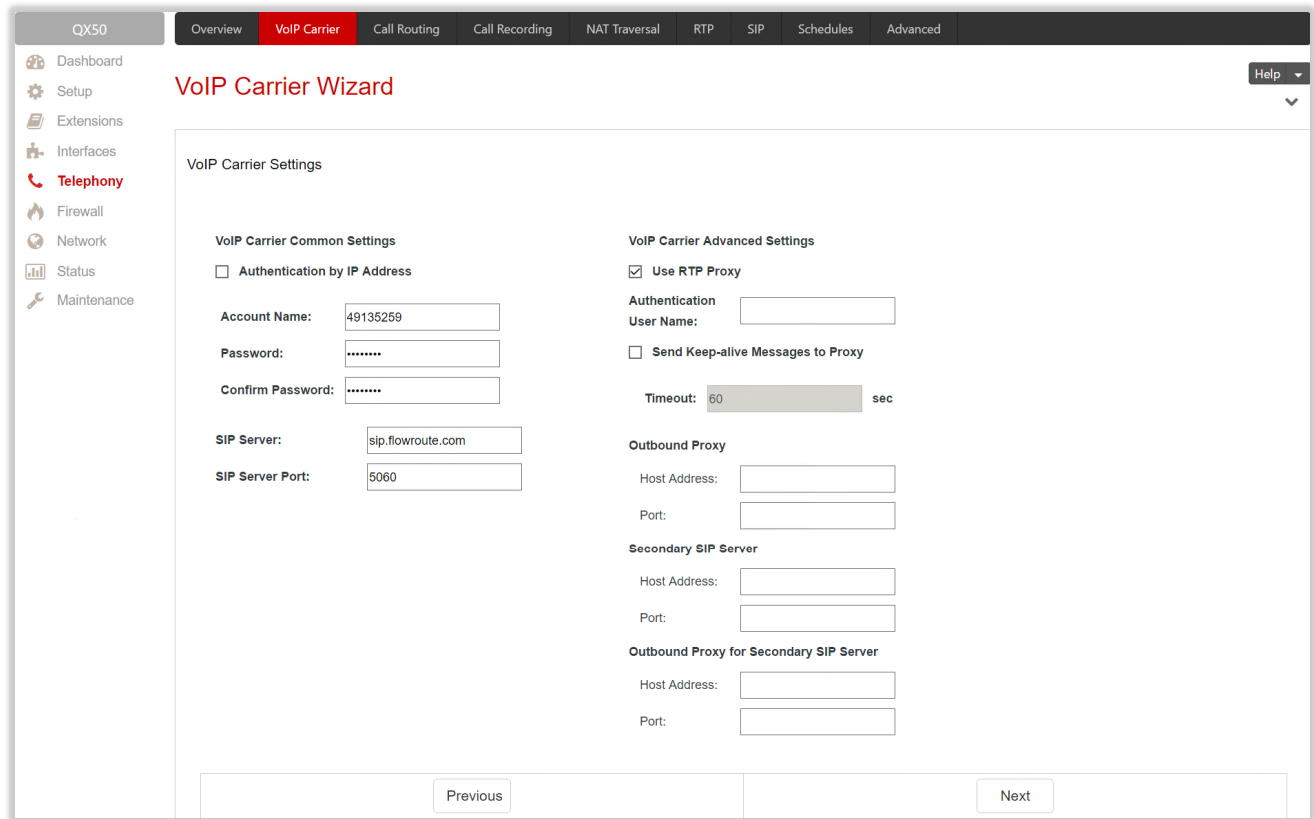


Figure 2: VoIP Carrier Wizard – page 2

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).

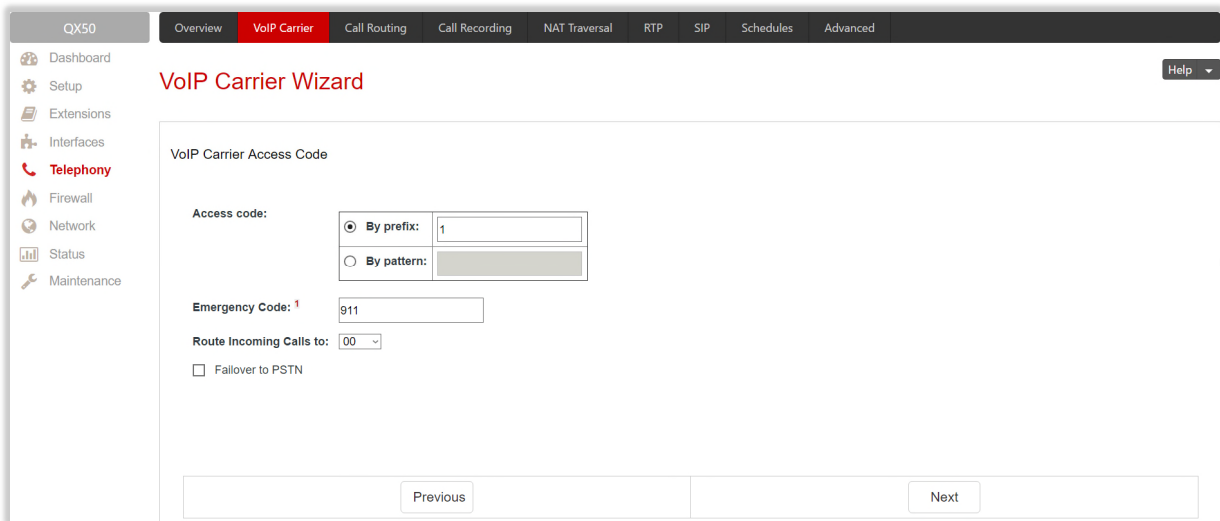


Figure 3: VoIP Carrier Wizard – Page 3

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

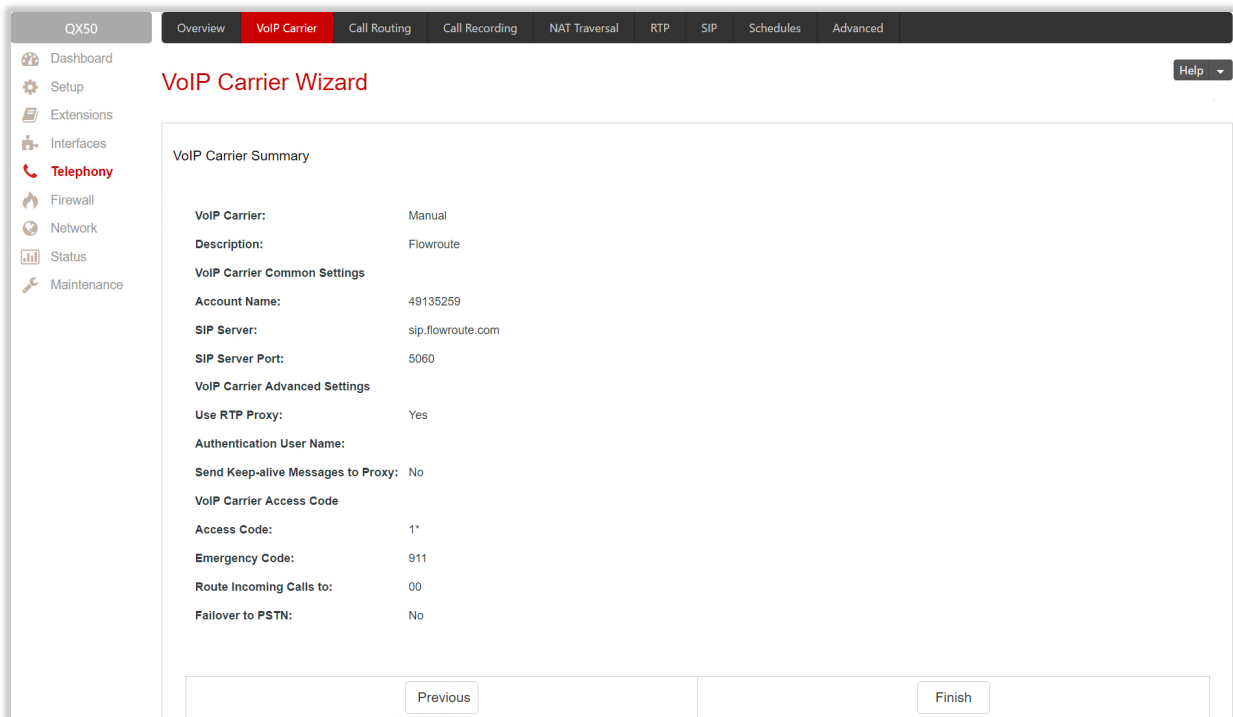


Figure 4: VoIP Carrier Wizard – Summary page

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).

Extensions Management

Total extensions count: 35/59

Extension	Display Name	Attached Line	SIP Address	Percentage of System Memory	External Access	Codes
00	Attendant		7429100@sip.epygi.loc:5060	5% (5 hour 43 min 54 sec)		PCMU...
10			7429110@sip.epygi.loc:5060	1% (1 hour 8 min 47 sec)		PCMU...
126		IP Line 24	126	0.4% (27 min 31 sec)	None	PCMU...
38	Dial & Announce	None	742938@sip.epygi.loc:5060	1% (1 hour 8 min 47 sec)	None	PCMU...
999	Flowroute (added by VoIP Carrier Wizard)	None	49135259@sip.flowroute.com:5060	0% (0 sec)	None	PCMU...
36 (Pickup Group)			36	0% (0 sec)		PCMU...
35 (Call Park)			35	0% (0 sec)		PCMU...
37 (Paging Group)			742937@sip.epygi.loc:5060	0% (0 sec)		PCMU...
800 (Recording Box)			800	1% (1 hour 8 min 47 sec)	GUI	PCMU

Figure 5: Extensions Management page

Call Routing Table

ID	State	Destination Number Pattern	Pattern Modification	Call Settings	Failover Reason(s)	Local Authentication	Source Number Pattern/ Caller ID Modification	Source Type	DT	CTL	UES / URP	Metric	Description
3	Enabled	8*	NDS: 1	SIP sip.epygi.com:5060	None	No	*	PBX			URP: Yes	10	Make SIP call
5	Enabled	00		PBX	None	No					URP: No	10	Call To Attendant
6	Enabled	7*	NDS: 1 Prefix: <callerid:2-end>	SIP sip.epygi.loc:5060	None	No	*	PBX	25 min		URP: Yes	10	Make SIP call
7	Enabled	0*	NDS: 1	FXO port(s): Any@Any	None	No	*	PBX			URP: No	10	
15	Enabled	1*	NDS: 1	IP-PSTN sip.flowroute.com:5060, RNSC: No	None	No	*	PBX			UES: 999 URP: Yes	10	Flowroute
16	Enabled	(911)		IP-PSTN sip.flowroute.com:5060, RNSC: No	Any	No	*	PBX			UES: 999 URP: Yes	10	Flowroute : Emergen
17	Enabled	??,???,????		PBX	None	No	*	PBX			URP: No	10	

NDS - Number of Discarded Symbols UES - Use Extension Settings RNSC - Restrict the Number of Simultaneous Calls CTL - Calling Time Limitation
 URP - Use RTP Proxy AAA - Authentication, Authorization, Accounting DT - Date/Time

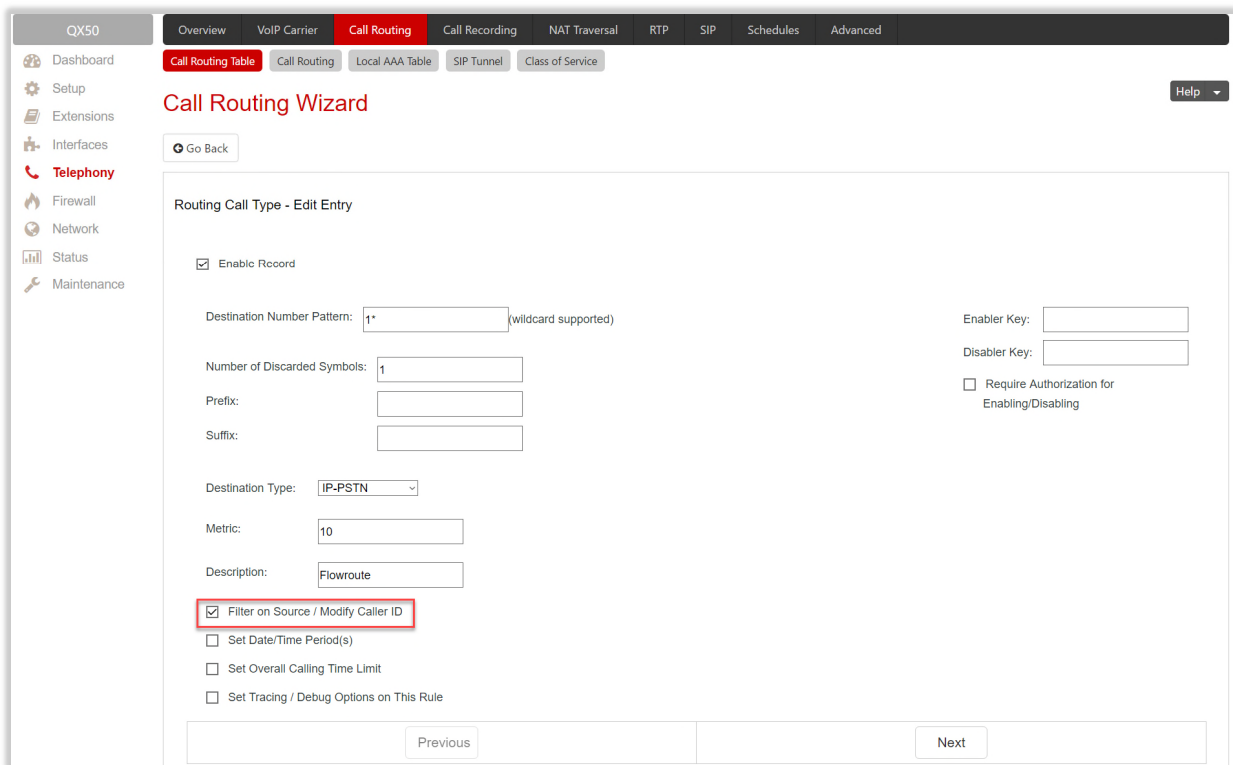
Figure 6: Call Routing Table page

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**.



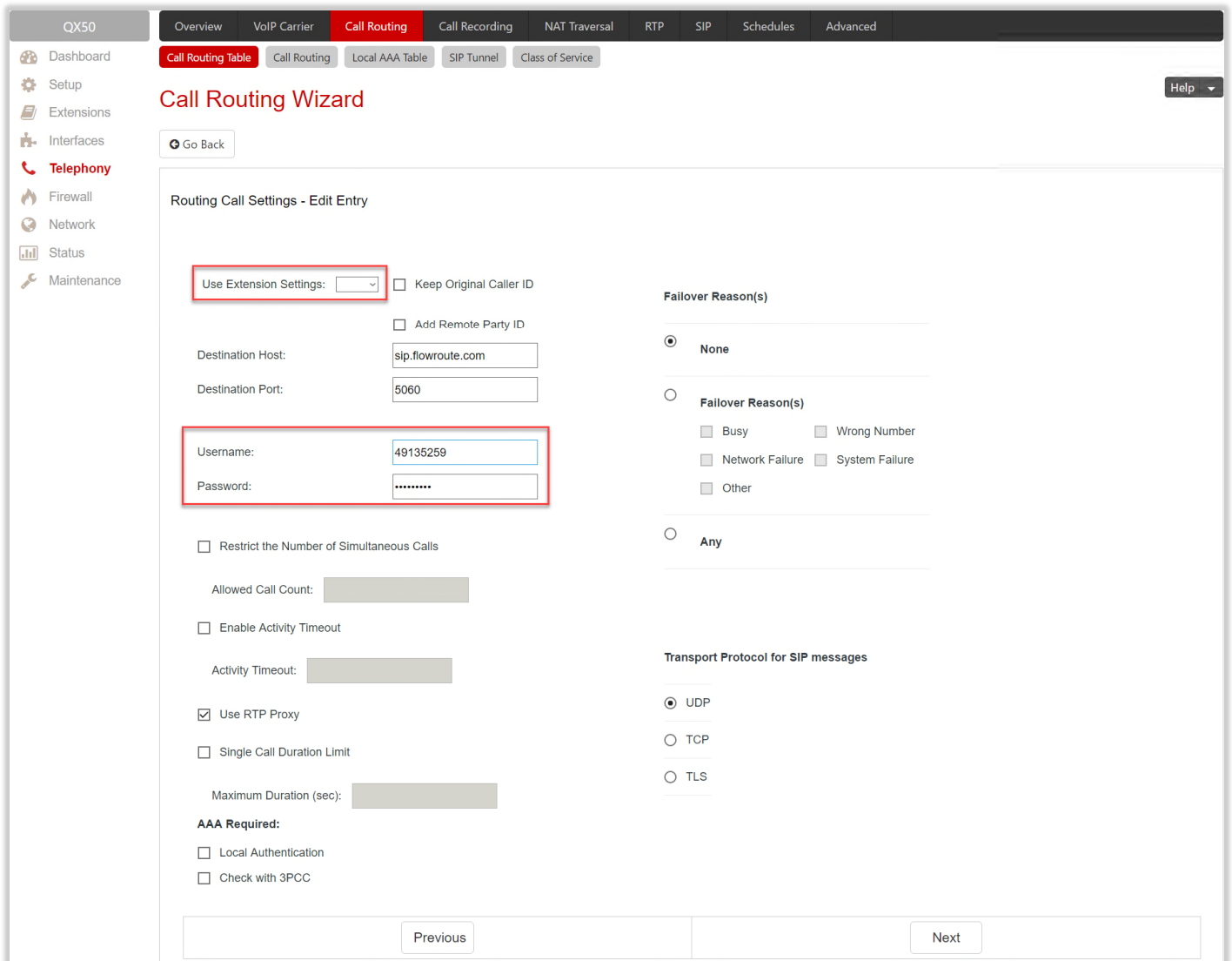
The screenshot shows the 'Call Routing Wizard' interface. The main content area is titled 'Routing Call Type - Edit Entry'. It contains several form fields and checkboxes:

- Enable Record
- Destination Number Pattern: (wildcard supported)
- Number of Discarded Symbols:
- Prefix:
- Suffix:
- Destination Type:
- Metric:
- Description:
- Filter on Source / Modify Caller ID (highlighted with a red box)
- Set Date/Time Period(s)
- Set Overall Calling Time Limit
- Set Tracing / Debug Options on This Rule
- Enabler Key:
- Disabler Key:
- Require Authorization for Enabling/Disabling

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

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).



QX50 Overview VoIP Carrier **Call Routing** Call Recording NAT Traversal RTP SIP Schedules Advanced

Call Routing Table Call Routing Local AAA Table SIP Tunnel Class of Service

Dashboard Setup Extensions Interfaces **Telephony** Firewall Network Status Maintenance

Call Routing Wizard

Go Back

Routing Call Settings - Edit Entry

Use Extension Settings: Keep Original Caller ID

Add Remote Party ID

Destination Host:

Destination Port:

Username:

Password:

Restrict the Number of Simultaneous Calls

Allowed Call Count:

Enable Activity Timeout

Activity Timeout:

Use RTP Proxy

Single Call Duration Limit

Maximum Duration (sec):

AAA Required:

Local Authentication

Check with 3PCC

Failover Reason(s)

None

Failover Reason(s)

Busy Wrong Number

Network Failure System Failure

Other

Any

Transport Protocol for SIP messages

UDP

TCP

TLS

Previous Next

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)

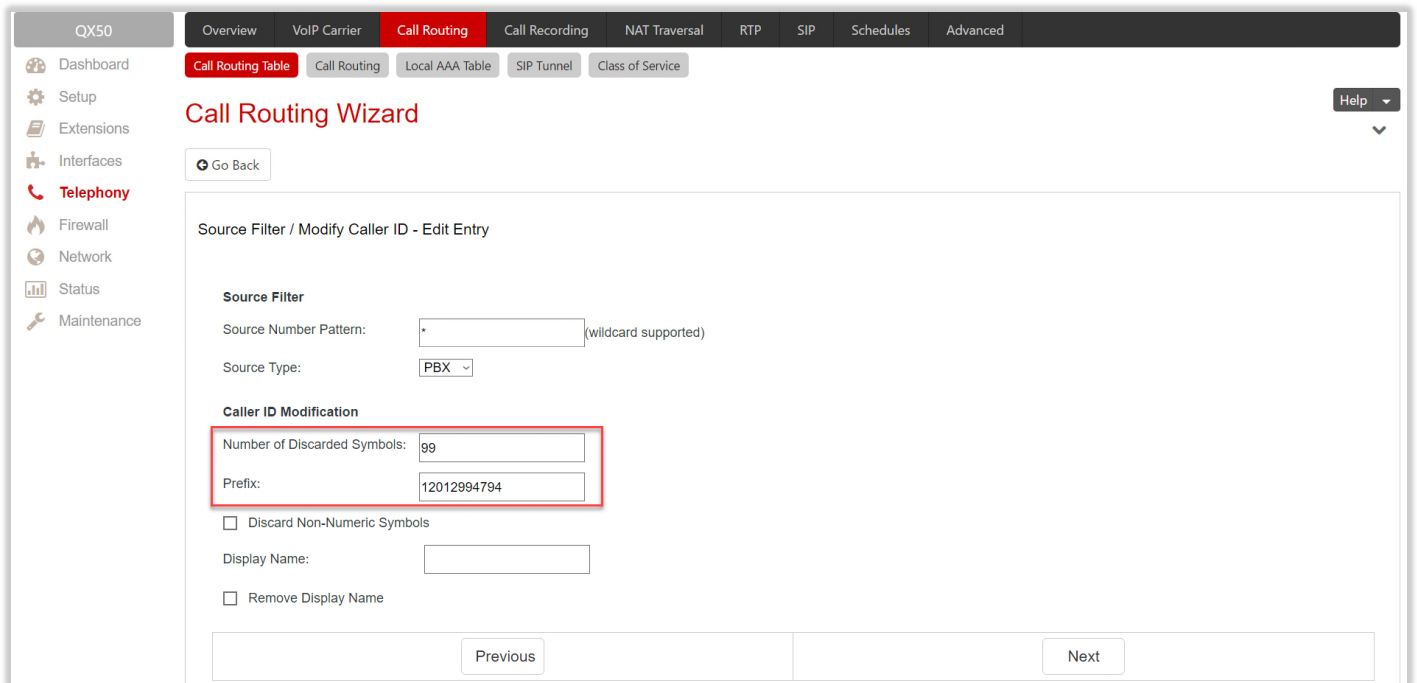


Figure 9: Call Routing Wizard page

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).

The screenshot shows the 'Extensions Management - Edit Entry' page for extension 00. The left sidebar contains navigation options: Dashboard, Setup, Extensions (selected), Interfaces, Telephony, Firewall, Network, Status, and Maintenance. The top navigation bar includes Overview, Extensions, Dialing Directories, Conferences, Recordings, Receptionist, ACD, and Authorized Phones. The main content area is titled 'SIP Registration Settings' for extension 00. It includes a 'Go Back' button and several sections: General Settings, Attendant Settings, Attendant Scenario, Ringing Announcement, SIP Settings (selected), SIP Advanced Settings, and Go To Codec Settings. The SIP Settings section contains the following fields: Username / DID Number (12015354360), Password, Confirm Password, SIP Server, SIP Port, SIP Registration Transport (UDP), and a checkbox for Registration on SIP Server. A Save button is located at the bottom of the form.

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.

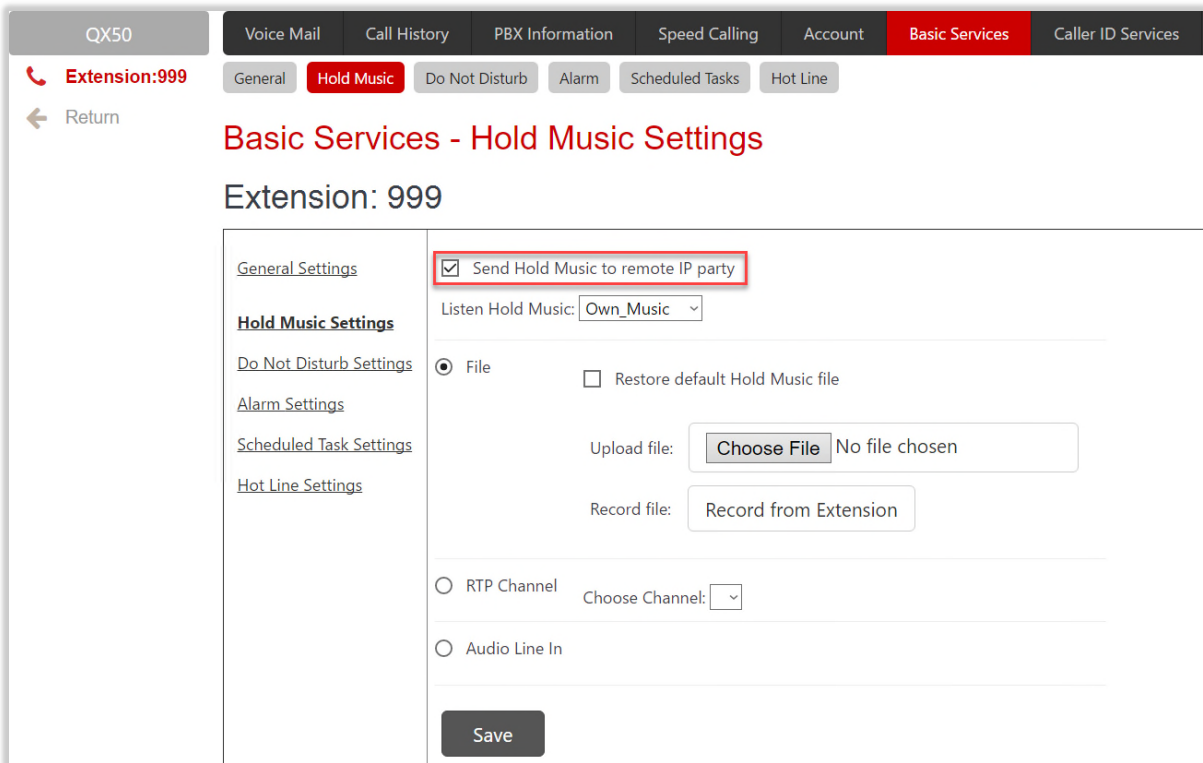


Figure 11: Basic Services – Hold Music Settings page

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**.

QX50
General Configuration

- Dashboard
- Setup
- Extensions
- Interfaces
- Telephony
- Firewall
- Network
- Status
- Maintenance

Max Number of Records in DB cache	<input type="text" value="32"/>	recs	
DNS cache MAX size	<input type="text" value="32"/>	recs	
DNS cache cleanup timeout	<input type="text" value="6"/>	hours	
Flash timeout	<input type="text" value="2"/>	sec	
Call progress notification timeout	<input type="text" value="10"/>	sec	
SIP DNS SRV Failover Timeout	<input type="text" value="16"/>	sec	
IP line registration timeout maximum	<input type="text" value="3600"/>	sec	
IP line registration timeout minimum	<input type="text" value="120"/>	sec	
Play user friendly voice messages instead of tones	<input type="text" value="default"/>		
IP phones settings			
SIP registration timeout	<input type="text" value="3600"/>	sec	
SIP subscription timeout	<input type="text" value="3600"/>	sec	
SIP session refresh timeout	<input type="text" value="600"/>	sec	
Clean IP Phone VLAN settings if no VLAN on PBX (reboot required)	<input checked="" type="checkbox"/>		
SIP TLS			
SSL server method	<input type="text" value="SSLv23"/>		
SSL client method	<input type="text" value="SSLv23"/>		
Templates for Caller ID ¹			
IP call	<input type="text" value="%a"/>	(%a%d%u%h)	
PBX call	<input type="text" value="%a"/>	(%a%d%u)	
PSTN call	<input type="text" value="%a"/>	(%a%d%u)	
Presence			
Subscription limitation (reboot required)	<input type="text" value="1000"/>		
Do not use "partial update" method in BLF notifications	<input type="checkbox"/>		
Phone Book			
Max number of contacts:	<input type="text" value="1000"/>		
<input checked="" type="checkbox"/> Enable VM silence disconnect			
Disconnect timeout	<input type="text" value="5"/>		
VM Session timeout	<input type="text" value="60"/>	sec	

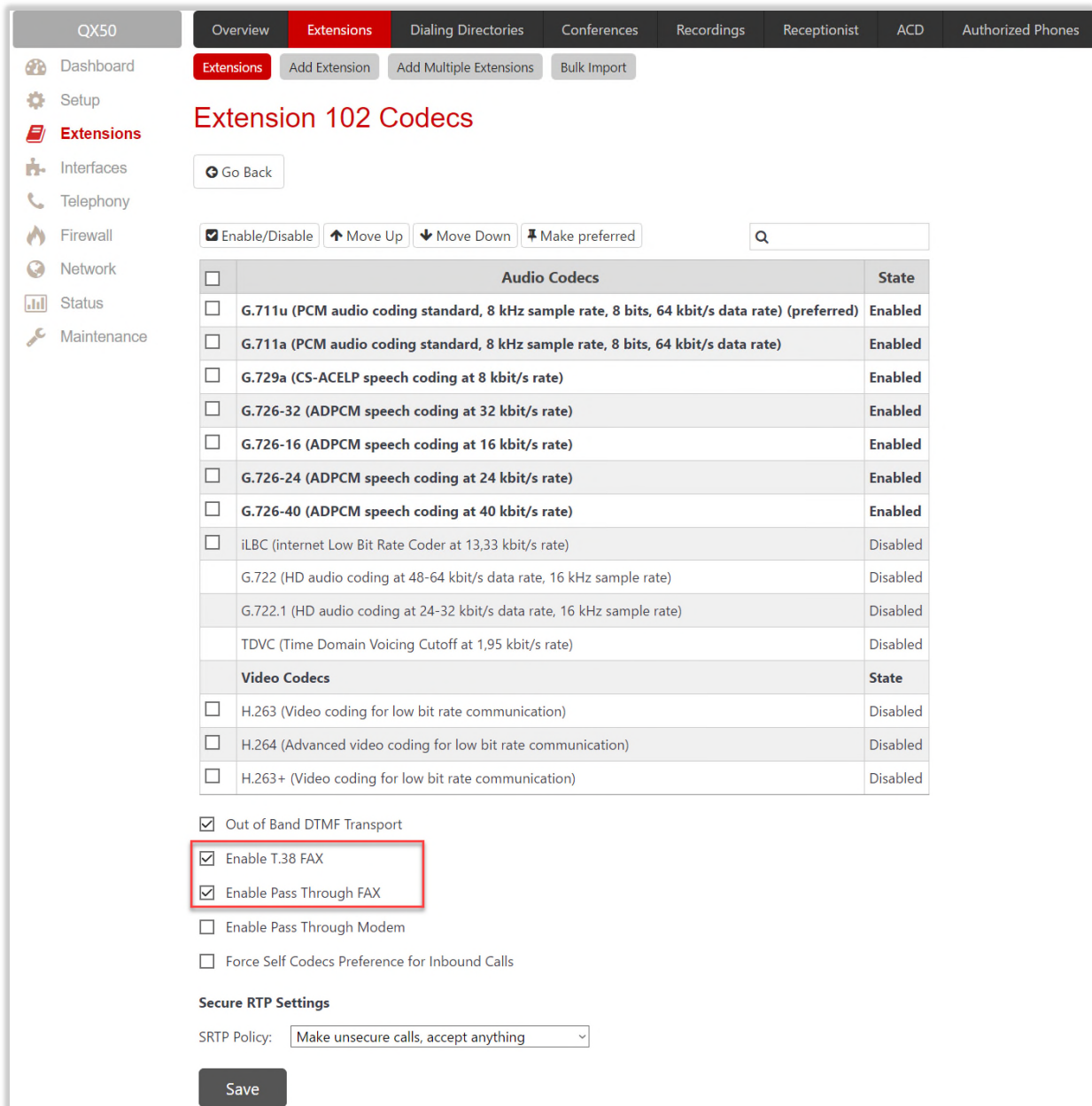
Figure 12: General Configuration page

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).



Extension 102 Codecs

Enable/Disable
 Move Up
 Move Down
 Make preferred

<input type="checkbox"/>	Audio Codecs	State
<input type="checkbox"/>	G.711u (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate) (preferred)	Enabled
<input type="checkbox"/>	G.711a (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)	Enabled
<input type="checkbox"/>	G.729a (CS-ACELP speech coding at 8 kbit/s rate)	Enabled
<input type="checkbox"/>	G.726-32 (ADPCM speech coding at 32 kbit/s rate)	Enabled
<input type="checkbox"/>	G.726-16 (ADPCM speech coding at 16 kbit/s rate)	Enabled
<input type="checkbox"/>	G.726-24 (ADPCM speech coding at 24 kbit/s rate)	Enabled
<input type="checkbox"/>	G.726-40 (ADPCM speech coding at 40 kbit/s rate)	Enabled
<input type="checkbox"/>	iLBC (internet Low Bit Rate Coder at 13.33 kbit/s rate)	Disabled
<input type="checkbox"/>	G.722 (HD audio coding at 48-64 kbit/s data rate, 16 kHz sample rate)	Disabled
<input type="checkbox"/>	G.722.1 (HD audio coding at 24-32 kbit/s data rate, 16 kHz sample rate)	Disabled
<input type="checkbox"/>	TDVC (Time Domain Voicing Cutoff at 1,95 kbit/s rate)	Disabled
<input type="checkbox"/>	Video Codecs	State
<input type="checkbox"/>	H.263 (Video coding for low bit rate communication)	Disabled
<input type="checkbox"/>	H.264 (Advanced video coding for low bit rate communication)	Disabled
<input type="checkbox"/>	H.263+ (Video coding for low bit rate communication)	Disabled

Out of Band DTMF Transport
 Enable T.38 FAX
 Enable Pass Through FAX
 Enable Pass Through Modem
 Force Self Codecs Preference for Inbound Calls

Secure RTP Settings

SRTP Policy:

Save

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](#).

THIS DOCUMENT IS PROVIDED TO YOU FOR INFORMATIONAL PURPOSES ONLY. The information furnished in this document, believed by Epygi Technologies to be accurate as of the date of publication, is subject to change without notice. Epygi Technologies assumes no responsibility for any errors or omissions in this document and shall have no obligation to you as a result of having made this document available to you or based upon the information it contains.

Epygi is a registered trademark of Epygi Technologies, Ltd. All other products and services are the registered trademarks of their respective holders.