Manual-II: Administration Guide for QX IP PBXs

This manual is effective for QX IP PBXs: QX20, QX50, QX200, QX500, QX2000, QX3000 and QXISDN4+.

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Emergency 911 Calls

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If the product proves to be defective during this warranty period, please contact the establishment where the unit was purchased. The Integrator will provide guidance on how to return the unit in accordance with its established procedures. Epygi will provide the Return Merchandise Authorization Number to your retailer.

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Extended Warranty

Extended Warranty Option

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Epygi Technologies, LLC.

2233 Lee Road Suite 201 Winter Park, Florida 32789

Administrative Council for Terminal Attachments (ACTA) Customer Information

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. Located on the equipment is a label that contains, among other information, the ACTA registration number and ringer equivalence number (REN). If requested, this information must be provided to the telephone company.

The REN is used to determine the quantity of devices which may be connected to the telephone line. Excessive REN's on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of the REN's should not exceed five (5.0). To be certain of the number of devices that may be connected to the line, as determined by the total REN's contact the telephone company to determine the maximum REN for the calling area.

This equipment cannot be used on the telephone company-provided coin service. Connection to Party Line Service is subject to State Tariffs.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right the file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make the necessary modifications in order to maintain uninterrupted service.

If trouble is experienced with this equipment, please contact EPYGI TECHNOLOGIES, LLC.

If the trouble is causing harm to the telephone network, the telephone company may request you to remove the equipment from the network until the problem is resolved.

Electrical Safety Advisory

To reduce the risk of damaging power surges, we recommend you install an AC surge arrestor in the AC outlet from which the QX or Quadro is powered.

Industry Canada Statement

This product meets the applicable Industry Canada technical specifications.

Safety Information

Before using the QX or Quadro, please review and ensure the following safety instructions are adhered to:

- To prevent fire or shock hazard, do not expose your QX or Quadro to rain or moisture.
- To avoid electrical shock, do not open the QX or Quadro. Refer servicing to qualified personnel only.
- Never install wiring during a lightning storm.
- Never install telephone jacks in wet locations unless the jack is specified for wet locations.
- Never touch non-insulated telephone wire or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying cable or telephone lines.
- Avoid using your QX or Quadro during an electrical storm.
- Do not use your QX, Quadro or telephone to report a gas leak in the vicinity of the leak.
- An electrical outlet should be as close as possible to the unit and easily accessible.

Emergency Services

The use of VoIP telephony is made available through IP networks such as the Internet and is dependent upon a constant source of electricity, network availability and proper operation of the equipment. If a power outage, network disruption or equipment failure occurs, the VoIP telephony service could be disabled. User understands that in any of those events the QX or Quadro may not be able to support 911 emergency services, and further, such services may only be available via the user's regular telephone line or mobile lines that are not connected to the QX or Quadro. User further acknowledges that any interruption in the supply or delivery of electricity, network availability or equipment failure is beyond Epygi's control and Epygi shall have no responsibility for losses arising from such interruption.

Music on Hold Copyright

The default Music on Hold on the QX and Quadro is a 22 second fragment from Chopin's Nocturne Op.9 #2 performed by Marina Vardanyan and kindly provided to Epygi Technologies, LLC. The recording is royalty free.

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Document Edition History

Revision	Date	Description	Valid for Models	Valid for FW
1.0	24-Mar-17	Initial Release	QX IP PBXs	6.1.45 and higher
1.1	16-Jun-17	Added a new licensable feature - Calling Cost Control. Updated.	QX IP PBXs	6.1.50 and higher
1.2	11-Dec-17	Updated for the new QX3000.	QX20, QX50, QX200, QX500, QX2000 QX3000 and QXISDN4+	6.2.1 and higher
1.3	31-May-18	Updated	QX20, QX50, QX200, QX500, QX2000 QX3000 and QXISDN4+	6.2.11 and higher

Table of Contents

1 About Administration Guide

Administration guide is intended for administrators as an aid to configure and operate QX IP PBX (herein QX). The functionality and configuration of QX with reference to other guides, manuals and complementary resources are described in this guide.

Many screen illustrations can be found in this guide. Since QX offers a wide variety of features and functionality, the example screenshots shown may not appear exactly the same for a specific QX as they appear in this manual. The example screenshots are for illustrative and explanatory purposes and should not be construed to represent your unique setup and scenarios.

2 Conventions Used in this Guide

Following conventions are used in this guide:

- Add button is used to create and add new entry.
- Edit button is used to modify the selected entry(s).
- Delete button is used to remove the selected entry(s).
- Save button is used to apply the changes.
- Start button is used to start a service, connection, etc.
- Stop button is used to start a service, connection, etc.
- Enable/Disable button is used to enable/disable the selected entry(s).
- Move Up and Move Down buttons are used to sort the entries in the specific table in the order they need to be accessed.
- Generate Password button is used to generate a system defined strong password.
- Show Hot Desking Settings and Hide Hot Desking Settings links are used to show/hide the Hot Desking settings respectively.
- Hide extensions attached to disabled IP lines and Show all extensions buttons are used to hide extensions which are attached to disabled IP lines or show all created extensions respectively.
- Call Type lists the available call types:
  - ➢ PBX – local calls to QX extensions.
  - ➢ SIP – calls via SIP.
  - ➢ PSTN – calls to a legacy telephone network (N/A for QX20, QX500, QX2000 and QX3000).
  - ➢ Auto – calls to a destination resolved by the Call Routing Table.
- Address (Redirect Address, Calling Address or Call to) field is used to define the destination address the call will be addressed to. The address strictly depends on the call type. Thus, define an extension number for the PBX calls, SIP address for the SIP calls, phone number for the PSTN calls, and, finally, define a routing pattern for the Auto type calls.
- Description field is used to enter any optional information about the entry.
- Wildcard supported notification is used to mention that wildcards are allowed for the field. Go to the Allowed Characters and Wildcards section to see the complete list of the supported characters and wildcards.
- The following options are available on QX to select the way custom voice message will be provided:
  - ➢ RTP Channel is used to stream messages through RTP Channels.
  - ➢ Audio Line In is used to stream messages through Audio Line In. This option is not available on QXISDN4+, QX2000 and QX3000.
  - ➢ File is used to upload/record custom messages.
    - ♦ Click Choose File to open a file chooser window to upload the file.

- Click **Record from Extension** to record a message directly on the phone.
- Once the message has been uploaded/recorded the following links will appear. The **Download … message** link used to download the uploaded/recorded message. The **Remove … message** link used to remove the uploaded/recorded message or restore the default one.

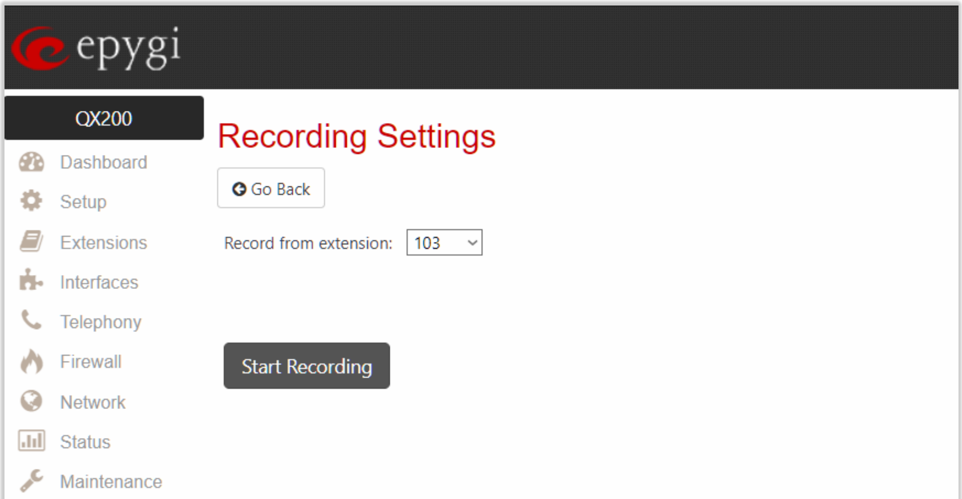


Figure 1: Recording Settings page

The **Recording Settings** page is used to initiate a custom voice message recording for the current extension directly from an IP phone. The **Record from extension** drop-down box lists all phone extensions that are available for recording.

Record a message as follows:

1. Select the extension from the **Record from extension** list.
2. Click **Start Recording**. The phone for the selected extension will start ringing.
3. Answer the call and follow the audio prompts to record a message.
4. Once the message has been recorded the following buttons will appear:

- **Download Recording** is used to download the recorded message.
- **Restore Default Recording** is used to remove the recorded message and restore the default one.

Note:

- The uploaded file should be either in (***.wav**) or (***.mp3**) format.
- The maximum duration of uploaded file is limited to 5 minutes.
- The maximum size of uploaded file is limited to 7.5 MB.

3 QX Graphical Interface

The following top menus and links are available when logged in as an administrator:

- Dashboard
- Setup
- Extensions
- Interfaces
- Telephony
- Firewall
- Network
- Status
- Maintenance
- Go To Extension allows quick access to User Settings for the selected extension.
- Pending Events allows quick access to system events and event settings.
- Language is available in case a custom Language Pack is installed and is used to enable custom language for GUI or revert back to default English.
- Date/Time displays the current time of device.
- Hostname displays the hostname of device.
- Renew WAN IP Address is available in case a WAN IP address for QX is assigned dynamically via DHCP.

4 Dashboard

If you are logged in as an administrator (users: admin or localadmin), you will see the number of calls currently active on the QX. The Active Calls table includes information about the calling/called parties, call start time and duration.

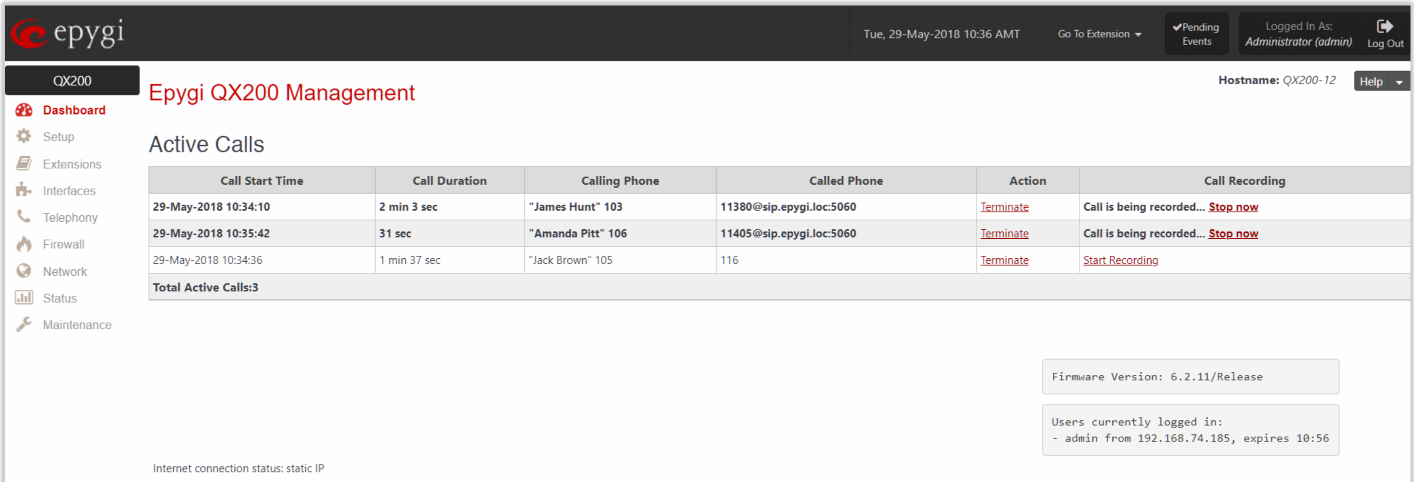


Figure 2: Dashboard menu

- Terminate link is used to terminate the active call.
- Start Recording link is used to manually start the recording of the call. Once the call recording starts, the link changes to Stop now and is used to manually stop the recording.
- The list of users currently logged into the system appears in the lower right corner of the page. The IP address of the user, the time until the next automatic logout and the current version of the firmware are displayed as well. The idle session timeout is set to 10 minutes. If no action is performed within 10 minutes, the user will automatically get logged out.

5 Setup Menu

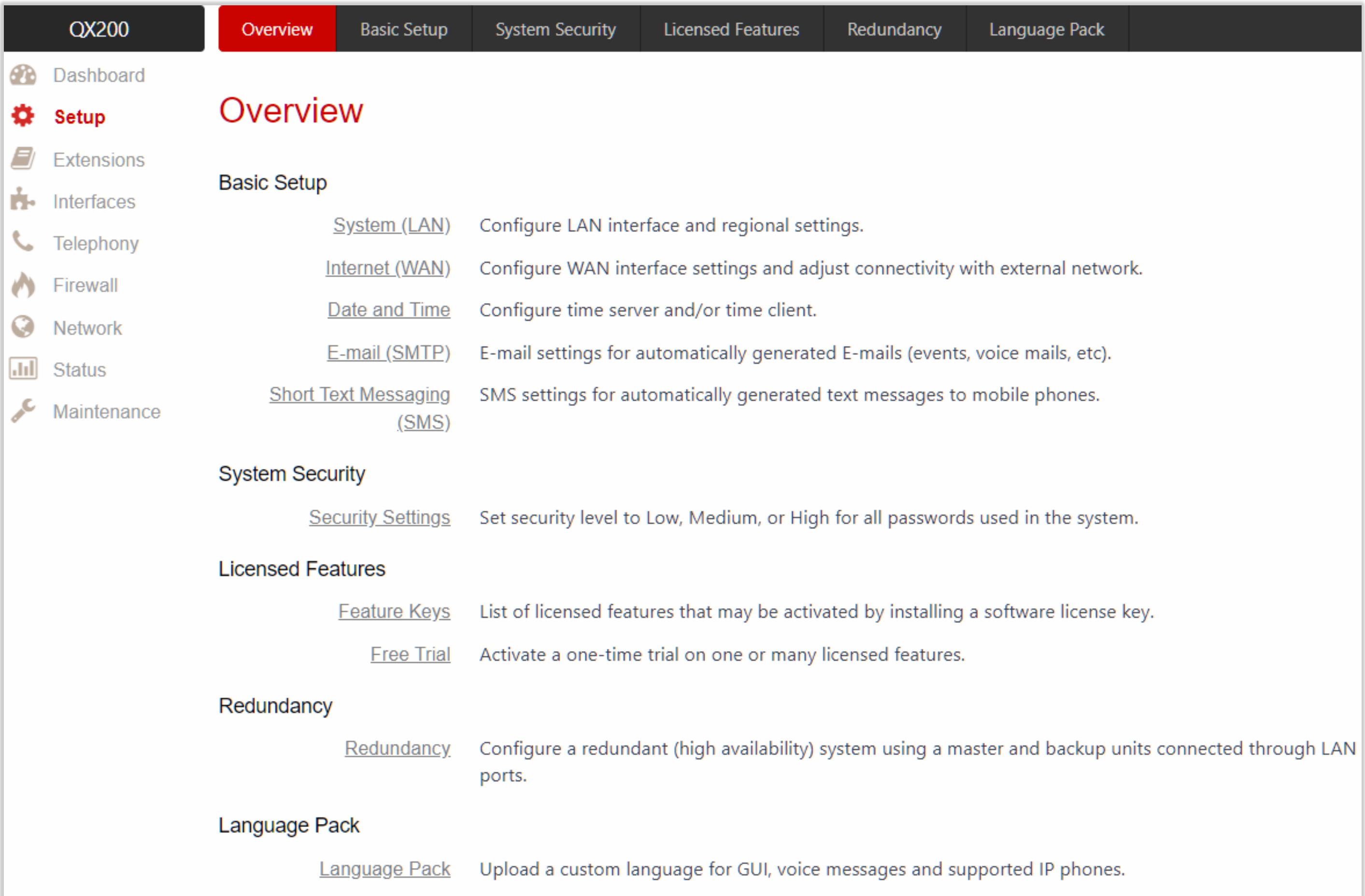


Figure 3: Setup Menu overview

5.1 Basic Setup

5.1.1 System (LAN)

You can login the QX WEB GUI through LAN interface using the default IP address:

- For QX20, QX50, QX200, QX500 and QXISDN4+ – 172.30.0.1
- For QX2000 and QX3000 – 192.168.0.200

Go to **Setup→Basic Setup→System (LAN)** to adjust the network parameters for the LAN interface. The **System Configuration Wizard** consists of the following sections:

- System Configuration
- DHCP Settings for the LAN Interface
- Regional Settings and Preferences
- Emergency Codes and PSTN Access Code Settings
- Call Alert Settings
- Summary

System Configuration

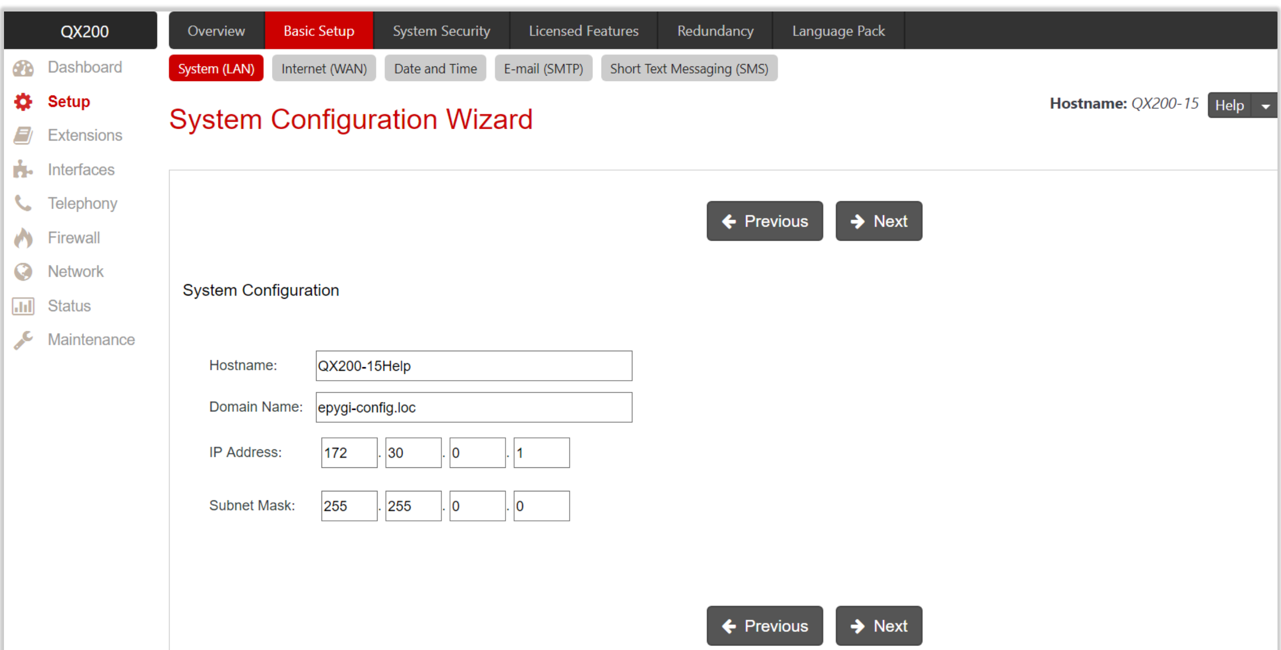


Figure 4: System Configuration section

The **System Configuration** section is used to configure the following settings (options):

- Hostname
- Domain Name
- IP Address
- Subnet Mask

DHCP Settings for the LAN Interface

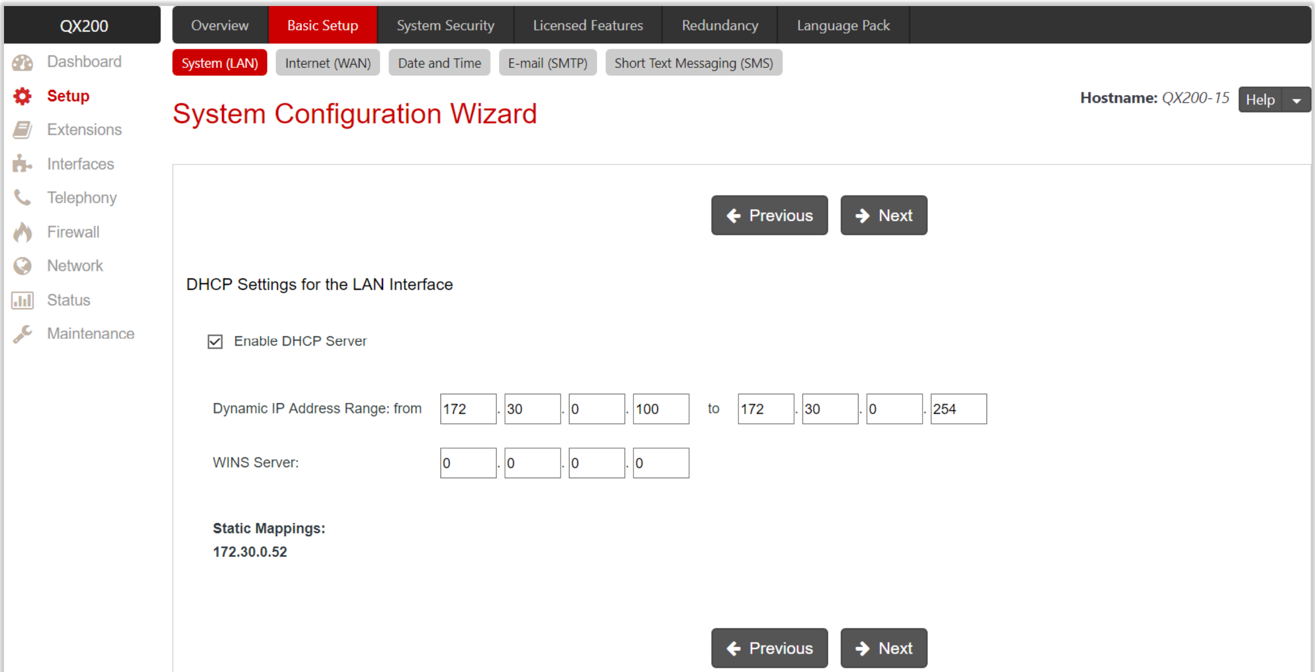


Figure 5: DHCP Settings for the LAN Interface section

The **DHCP Settings for the LAN Interface** section is used to configure the following settings (options):

- **Enable DHCP Server** is used to activate DHCP server on QX.
- **Dynamic IP Address Range** is used to set the IP address pool.
- **WINS Server** is used to set the IP address for the WINS server.

Regional Settings and Preferences

Regional Settings are important for the functionality of QX voice subsystem.

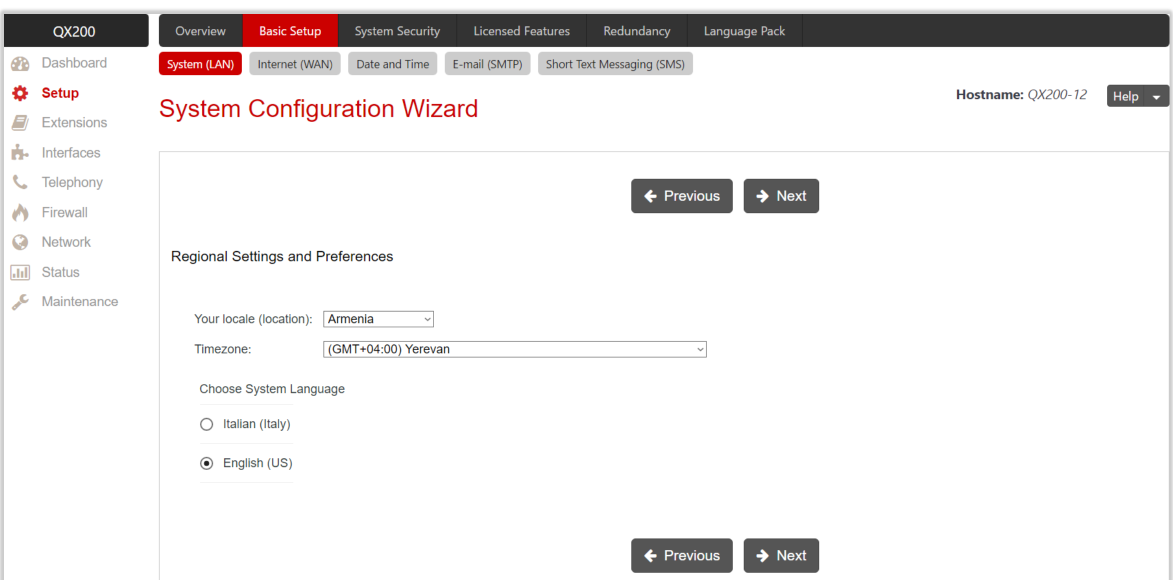


Figure 6: Regional Settings and Preferences section

The **Regional Settings and Preferences** section is used to configure the following settings (options):

- **Your Locale (location)** is used to select the location and time zone of QX.
- **Timezone** is used to select the proper time zone so QX can display the correct time. TIP: QX supports **Daylight Savings (DST)** correction if it is available for the selected time zone.
- **Choose System Language** is used to select the language for system voice messages: **custom** or **default English**. TIP: This selection is available when a custom **Language Pack** has been uploaded.

Emergency Codes and PSTN Access Code Settings

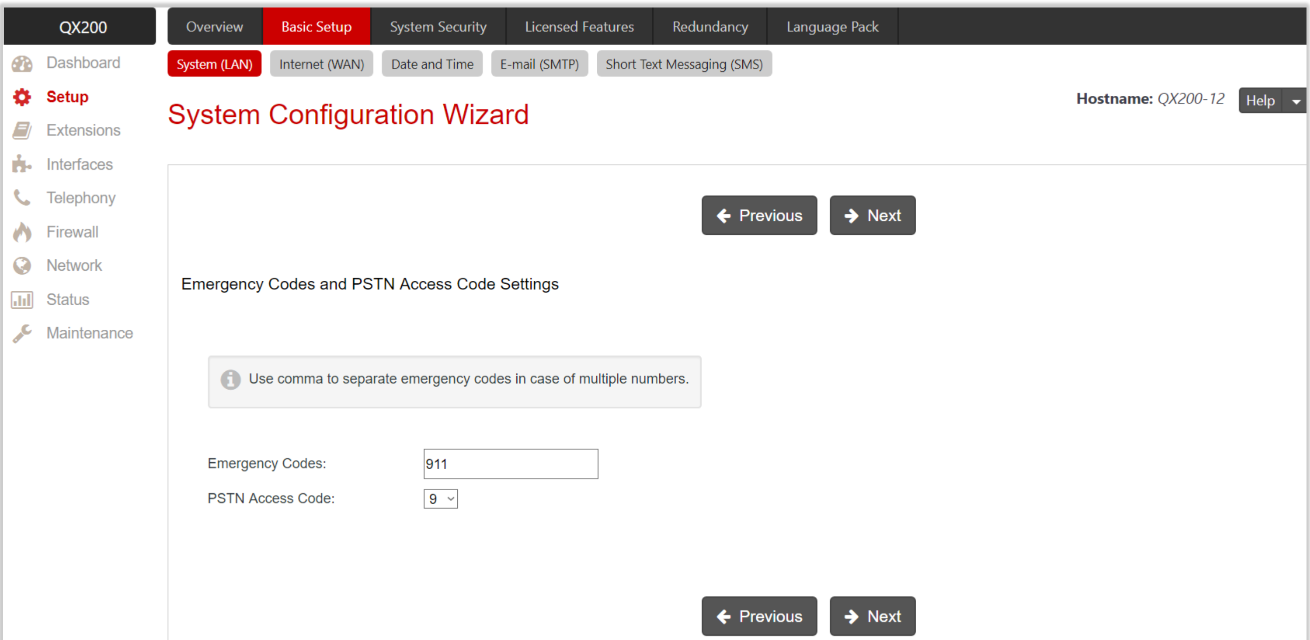


Figure 7: Emergency Codes and PSTN Access Code Settings section

The **Emergency Codes and PSTN Access Code Settings** section is used to configure the following settings (options):

- **Emergency Codes** is used to set PSTN number(s) of emergency service(s). For each emergency code, a routing pattern will be generated in the **Call Routing Table**, allowing faster and easier calls to emergency services. TIP: Use commas to separate emergency codes in case of multiple numbers.
- **PSTN Access Code** is used to select prefix code for accessing PSTN line through routing table.

Call Alert Settings

This section is used to activate the call alert service and configure settings. The Call Alert service is used to notify the designated personnel about emergency calls, as well as calls through the certain call routing rules. The following information will be included in the notification: the routing pattern, the user extension who placed the call, the dialed number and the call Date/Time.

The following settings (options) are available:

- Generate System Event is used to generate and display event notification in the System Events.
- Send Notification via E-mail is used to send a notification to a specified address via e-mail.
- Send Notification via SMS is used to send a notification to a specified number via SMS.
- Leave Voice Message is used to leave a voice mail on defined extension(s) with a voice message.

Note: Use commas to separate email addresses, mobile numbers and user extensions in case of multiple entries.

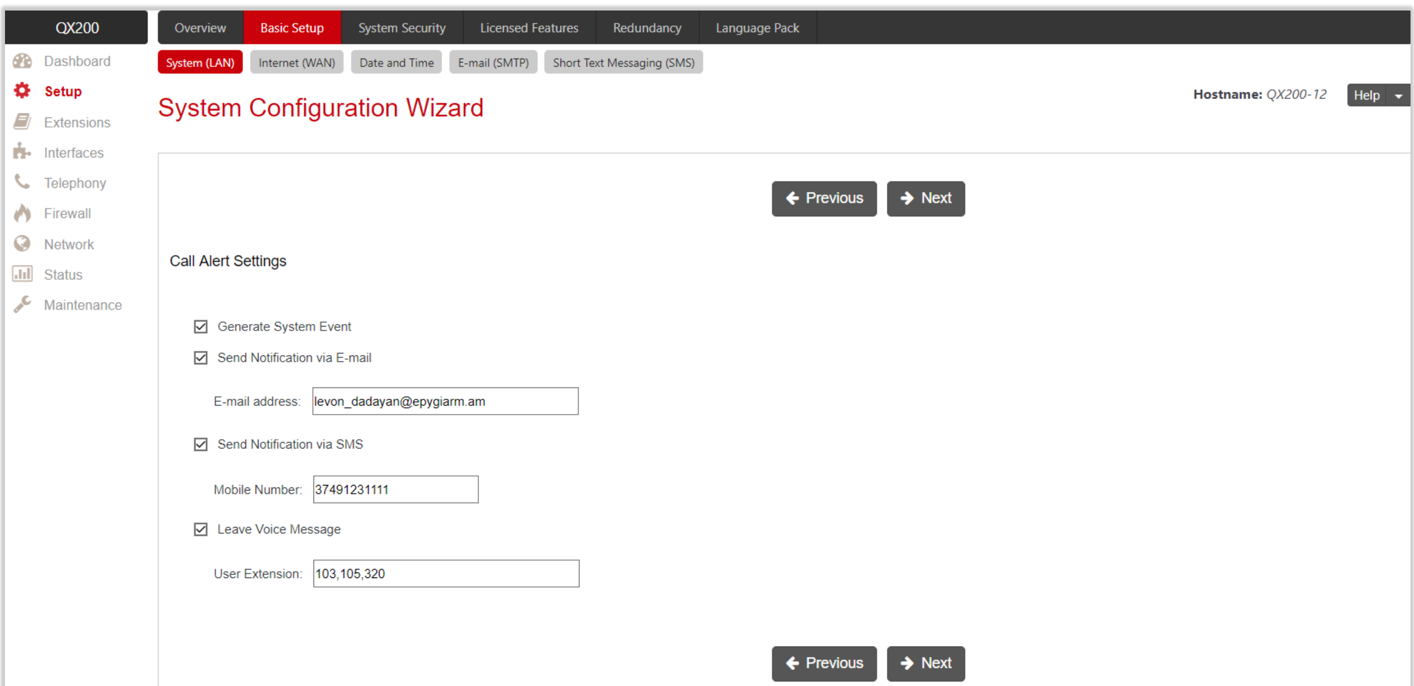


Figure 8: Call Alert Settings section

Summary

This section displays all configured settings (options) before applying them.

Note:

- Finish the wizard and click "OK" to apply the changes made in all sections of the wizard. You must confirm the settings within 20 minutes. Otherwise the device will return back to the previous configuration and reboot.
- It is strongly recommended to leave the factory default settings unchanged if their meanings are not fully clear to you.

5.1.2 Internet (WAN)

Go to the **Setup→Basic Setup→Internet (WAN)** to configure or adjust the network parameters for QX WAN interface. **Internet Configuration Wizard** consists of the following sections:

- Uplink Configuration
- WAN Interface Configuration
- DNS Settings
- Summary

Uplink Configuration

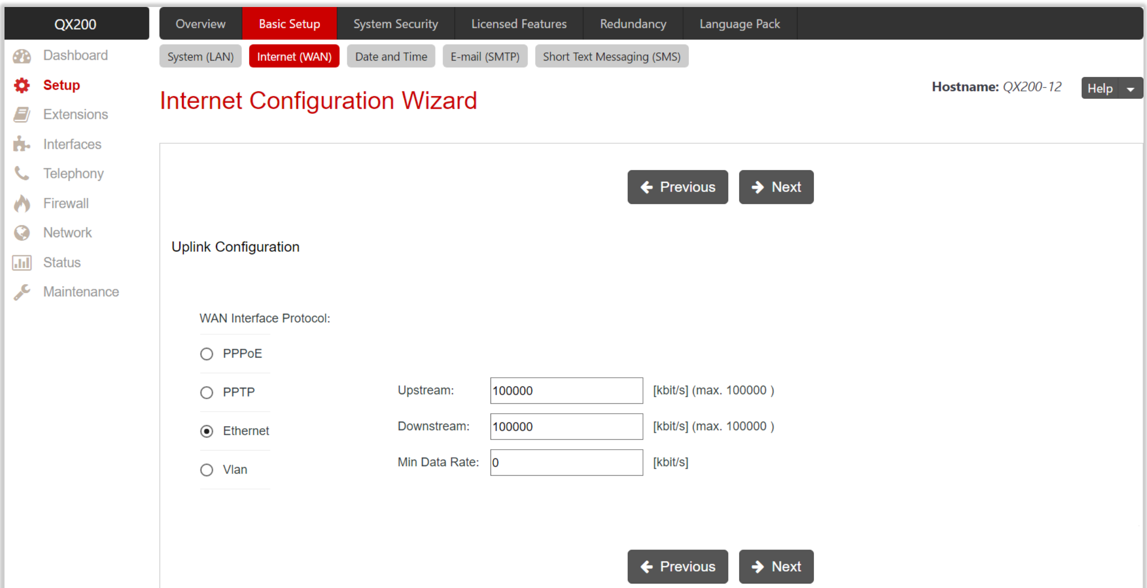


Figure 9: Uplink Configuration section

The **Uplink Configuration** section is used to configure the following settings (options):

- **WAN Interface Protocol** is used to select the protocol for the WAN interface. Based on this selection the wizard configuration sections may differ. The following protocols are available:
  - PPPoE
  - PPTP
  - Ethernet
  - VLAN (**TIP:** This option becomes available only when VLAN is configured on QX.)
- **WAN Interface Bandwidth** is used to specify the upstream and downstream speeds in Kbit/s, helping to assure the quality of IP calls. IP call loses the voice quality if there is no available bandwidth. When the limits of a bandwidth capacity are approaching, another IP call will be rejected.
- **Min Data Rate** is used to set the amount of upstream bandwidth that ought to remain for data traffic even if voice applications use the entire available upstream bandwidth. The value selected here needs to be smaller than the upstream bandwidth.

PPPoE

- **Keep Connection Alive** is used to keep the connection alive by sending control packets for the link state verification.
- **Authentication Settings** is used to set the authentication parameters (**Username** and **Password**) to register on the ISP server.
- **Dial Behavior** is used to select the **Dial Behavior** type.
  - **Dial manually** – if selected, a button will be displayed on the **Dashboard** to switch the connection on/off.
  - **Always connected** – if selected, the connection will always stay active and connected.
- **IP Address Assignment** is used to select the IP address assignment type:
  - **Obtain an IP Address automatically** – if selected, QX will get the IP address from local network or ISP automatically.
  - **Use the following IP Address** is used to set the IP address manually.

PPTP

- **Obtain an IP Address automatically** – if selected, QX will get the IP address from local network or ISP automatically.
- **Use the following IP Address** is used to set the IP address manually.

Click **Next** to continue the configuration of **PPP/ PPTP** settings:

- **PPTP Server** is used to set the IP address of PPTP server.
- **Encryption** is used to select the encryption for the traffic over PPTP interface.
- **Keep Connection Alive** is used to keep the connection alive by sending control packets for the link state verification.
- **Authentication Settings** is used to set the authentication parameters (Username and Password) to register on the ISP server.
- **Dial Behavior** is used to select the **Dial Behavior** type:
  - **Dial manually** – if selected, a button will be displayed on the **Dashboard** to switch the connection on/off.
  - **Always connected** – if selected, the connection will always stay active and connected.
- **IP Address Assignment** is used to select the IP address assignment type:
  - **Obtain an IP Address automatically** – if selected, QX will get the IP address from local network or ISP automatically.
  - **Use the following IP Address** is used to set the IP address manually.

Ethernet

- **Obtain an IP Address automatically** – if selected, QX will get the IP address from local network or ISP automatically.
- **Use the following IP Address** is used to set the IP address manually.

VLAN

- **VLAN ID** is used to select **VLAN ID** from the configured VLAN list.

Click **Next** to continue the configuration of the **VLAN IP Configuration** settings:

- **Obtain an IP Address automatically** – if selected, QX will get the IP address from local network or ISP automatically.
- **Use the following IP Address** is used to set the IP address manually.

WAN Interface Configuration

The **WAN Interface Configuration** section is used to set the MAC address of QX. This might be necessary if the ISP requires a specified MAC address (e.g. for authentication).

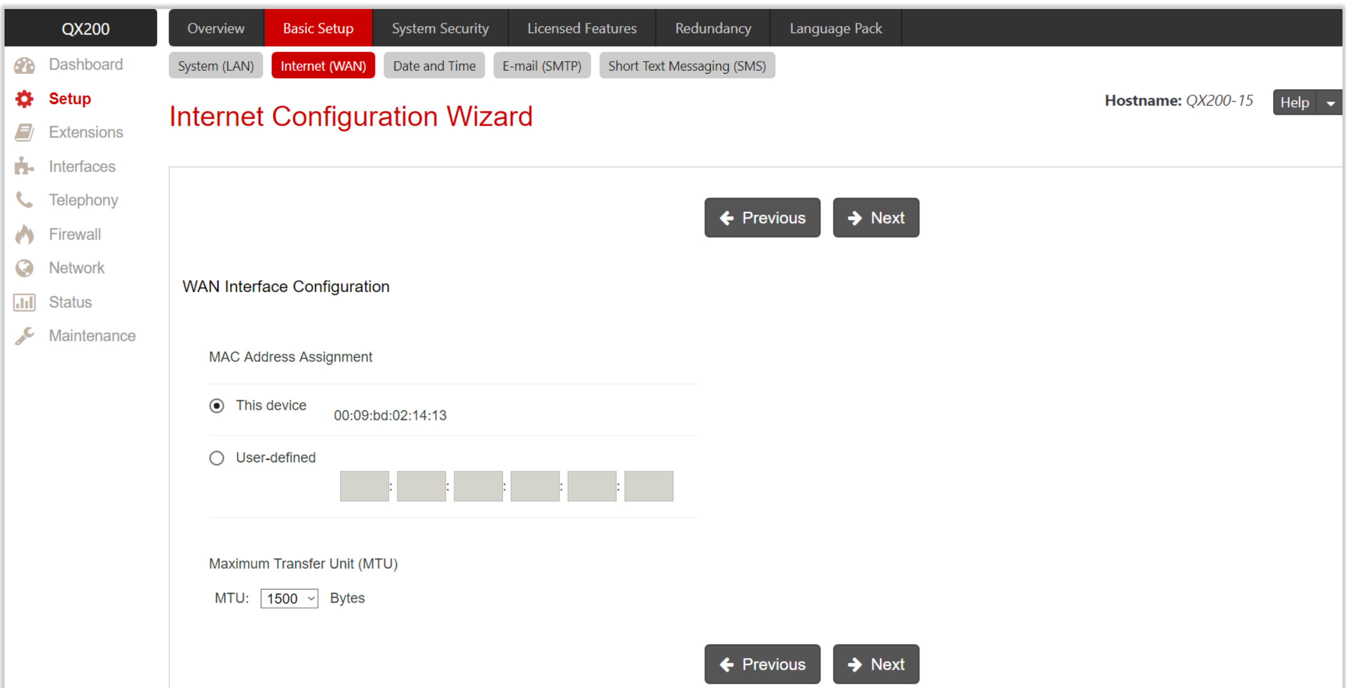


Figure 10: WAN Interface Configuration section

The following settings (options) are available:

- **This device** is used to select the default MAC address of the WAN interface.
- **User-defined** is used to set the MAC Address manually.
- **MTU** is used to select the maximum size of packet that can be sent in a packet or frame-based network such as the Internet. QX supports packet fragmentation. TIP: The default MTU size is **1500** bytes for Ethernet protocol and **1400** bytes for PPPoE.

DNS Settings

The **DNS Settings** section is used to configure the following settings (options):

- **Obtain DNS Server Address automatically** – if selected, QX will get the IP address of DNS server from local network or ISP automatically.
- **Use the following DNS Server Address** is used to manually assign a name server as follows:
  - ➢ **Preferred DNS** is used to set the IP address of name server.
  - ➢ **Alternate DNS** is used to set the IP address of the secondary name server that will be used if the main name server cannot be accessed.

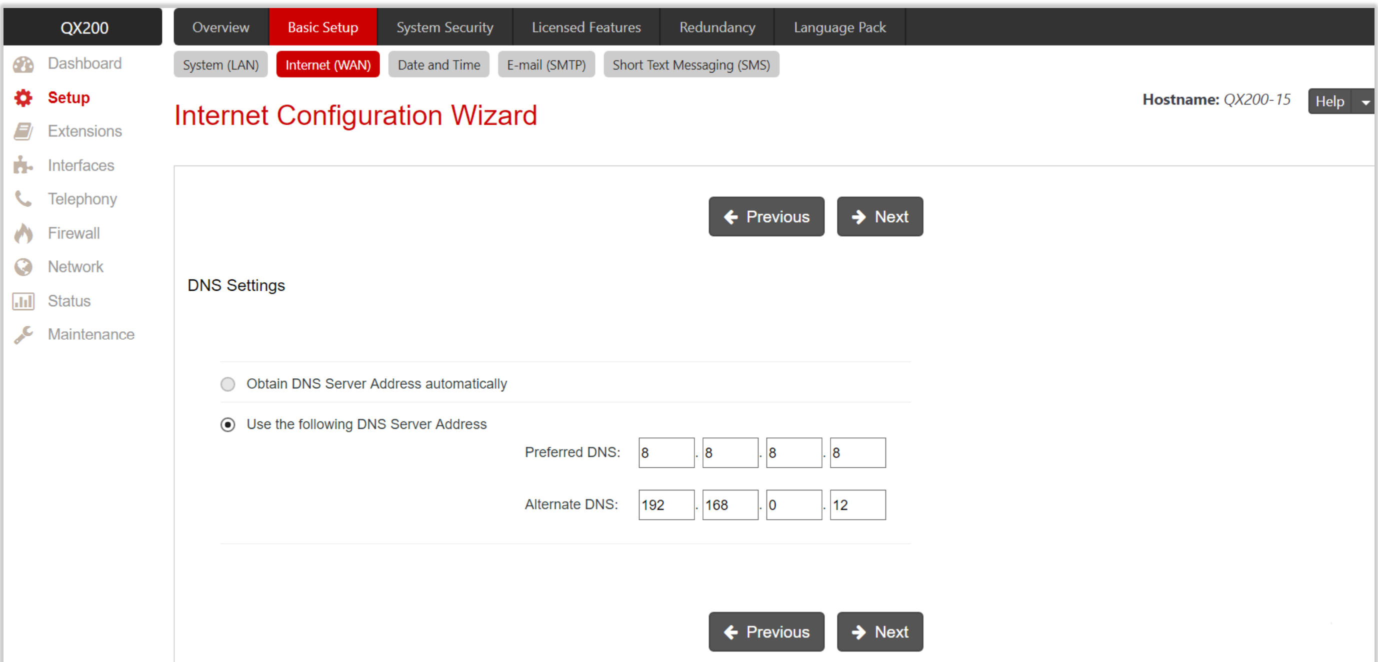


Figure 11: DNS Settings section

Summary

This section displays all configured settings (options) before applying them.

Note:

- Finish the wizard and click "**OK**" to apply the changes made in all sections of the wizard. You must confirm the settings within **20** minutes. Otherwise the device will return back to the previous configuration and reboot.
- It is strongly recommended to leave the factory default settings unchanged if their meanings are not fully clear to you.
- **Internet Configuration Wizard** is renamed to **Uplink Configuration Wizard** on QX2000 and QX3000.

5.1.3 Date and Time

QX Date and Time settings can be updated through international time servers.

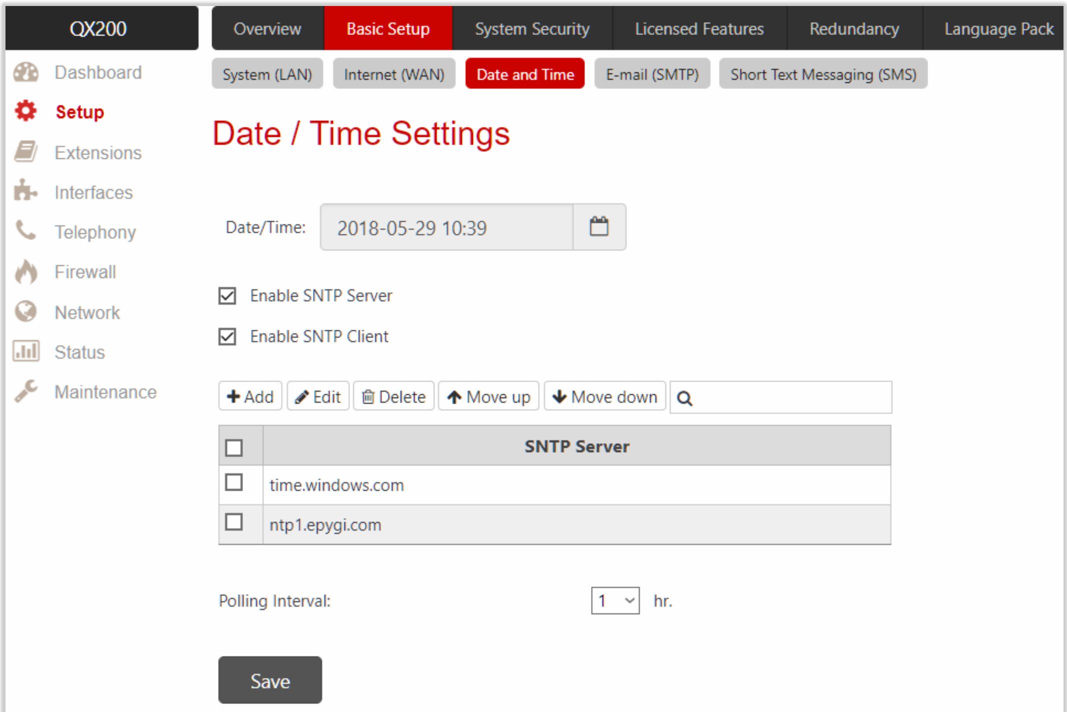


Figure 12: Date / Time Settings page

The following settings (options) are available:

- Date/Time shows the current system time.
- Enable SNTP Server is used to activate SNTP server on QX.
- Enable SNTP Client is used to activate SNTP client on QX. If not selected, the current system time can be configured manually.
- Polling Interval is used to select the time interval for periodical synchronization between the timeserver and QX.

The SNTP Server table lists all defined SNTP servers. To add a new SNTP Server:

1. Click Add to define new server parameters. The following options are available:
   - Manual is used to set the Full Qualified Domain Name (FQDN) or IP address of the SNTP server.
   - Predefined is used to select the FQDN of the SNTP server from the drop-down list.
2. Click Save to add the new SNTP server to the SNTP Servers table.
3. Click Move Up or Move Down to sort NTP servers in the order they need to be accessed. TIP: If the NTP server in the first position of the SNTP Server table does not answer, NTP server in the next position will be attempted to reach.

5.1.4 E-mail (SMTP)

The SMTP service allows QX to automatically generate and send alert and notification e-mails as specified in the Event Settings. The E-mail Settings page is used to configure SMTP settings (parameters):

- Enable SMTP Service activates the SMTP service.
- SMTP Host is used to set the IP address or hostname of the SMTP server.
- E-mail Sender Address is used to set the e-mail address that is either registered on the selected SMTP server or has permission to use SMTP server for e-mail transmissions.
- E-mail Recipient Address – an active address to send e-mails to.
- E-mail Recipient Address (CC) – an active address to deliver e-mails' carbon copy (CC) to.
- The server requires a secure connection (TLS) is used to select if the specified SMTP server requires secure connection using TLS. If the specified SMTP server allows to use both secure and unsecure connections, then this selection forces to establish the secure connection.
- Enable SMTP Authentication is used to select if the specified SMTP server requires authentication. Then enter the Username and Password configured on the SMTP server.

Below is the sample of e-mail settings on the QX, (smtp.gmail.com is used as a SMTP server).

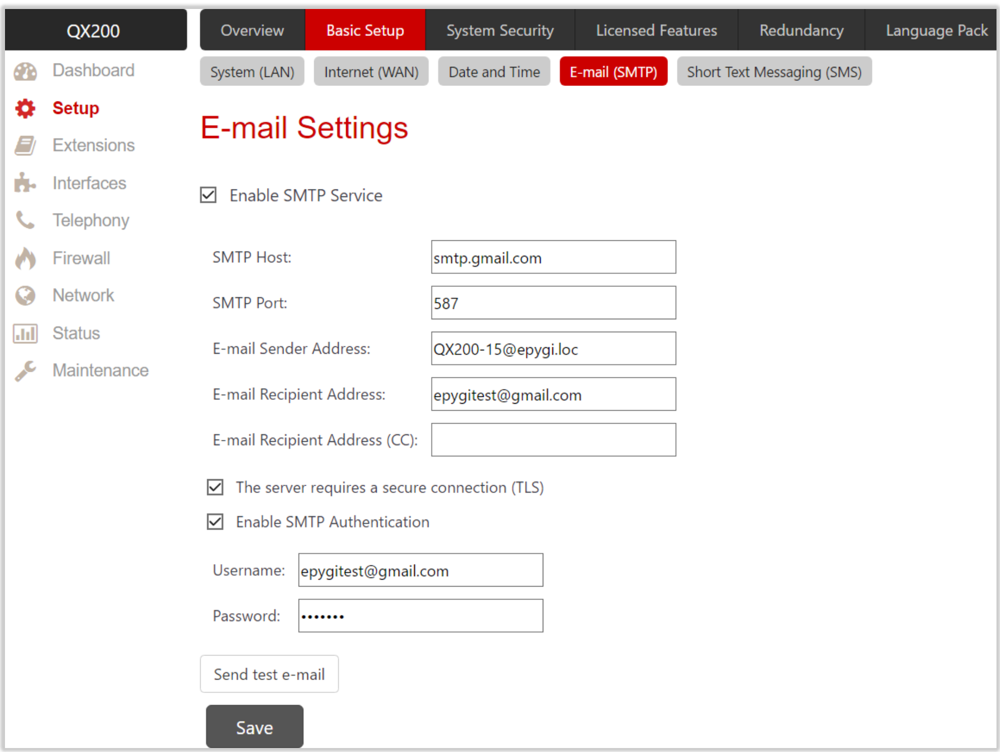


Figure 13: E-mail Settings page

Once configured, click Send test e-mail to send a test e-mail to the defined e-mail address to verify the settings.

5.1.5 Short Text Messaging (SMS)

The **SMS** service allows QX to automatically generate and send alert and notification events via SMS. The **SMS Settings** page is used to configure SMS settings (parameters):

- **Enable SMS Service** is used to activate SMS service on QX.
- **Username** and **Password** is used to set the authentication parameters configured on the SMS server.
- **SMS Sender Address** is used to set the sender's address.
- **SMS Recipient Address** is used to set the recipient's address. TIP: Use a space, semicolon or a comma to separate mobile numbers in case of multiple recipients.

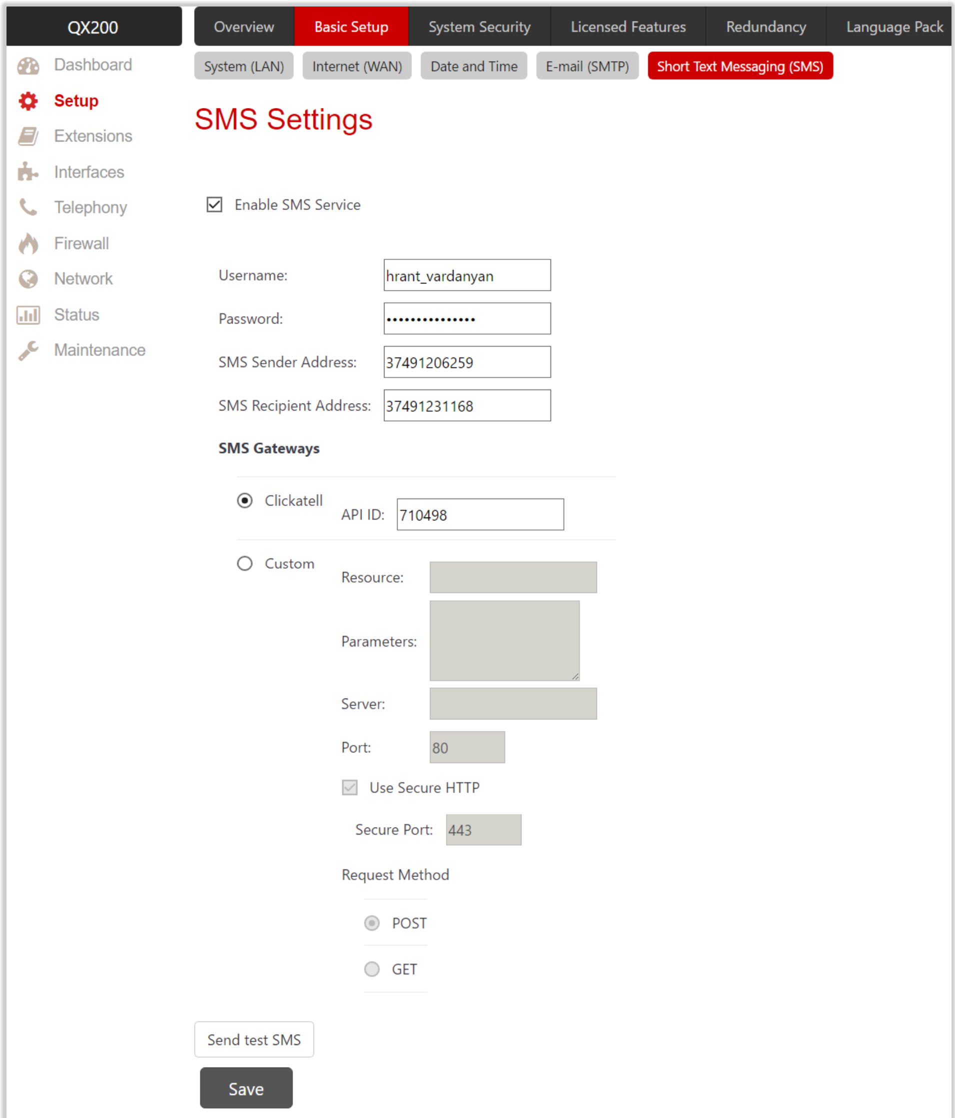


Figure 14: SMS Settings page

You may either use predefined SMS gateway (Clickatell) or define a custom one.

- **Clickatell** – if selected, then set the **Clickatell** specific parameter provided by the server in the activated **API ID** field. This parameter must be identical on both sides.
- **Custom** – if selected, then set the gateway parameters as follows:
  - **Resource** is used to set the HTTP resource name on the SMS gateway.
  - **Parameters** is used to set parameters to be submitted to the resource address. The value of this field represents a string with tokens (separated by percent (%) symbols) inside. Each token indicates a value of the certain field on this page. The value depends on the SMS gateway requirements. The tokens are the strings that have the following dependencies from the field in this page:
    - **%username%** indicates the username set in the **Username** field.
    - **%password%** indicates the password set in the **Password** field.
    - **%to%** indicates the password set in the **SMS Recipient Address** field.
    - **%from%** indicates the password set in the **SMS Sender Address** field.
    - **%text%** indicates the SMS text generated by QX (voice mail notification, event notification, etc.).

For example: user=%username%&password=%password%&to=%to%&from=%from%&text=%text%

- ➢ **Server** is used to set the IP address or hostname of the SMS gateway.
- ➢ **Port** is used to set the port number of the SMS gateway.
- ➢ **Use Secure HTTP** to access the SMS server via HTTPS. Then set the port number for HTTPS traffic in the activated **Secure Port** field.

- Select one of the HTTP request methods (**POST** or **GET**) through the **Request Method** options. The QX uses one of these methods to access the SMS gateway.

Once configured, click **Send test SMS** to send a test SMS to the defined mobile number to verify the settings.

5.2 System Security

System Security Management is used to manage the global security of QX.

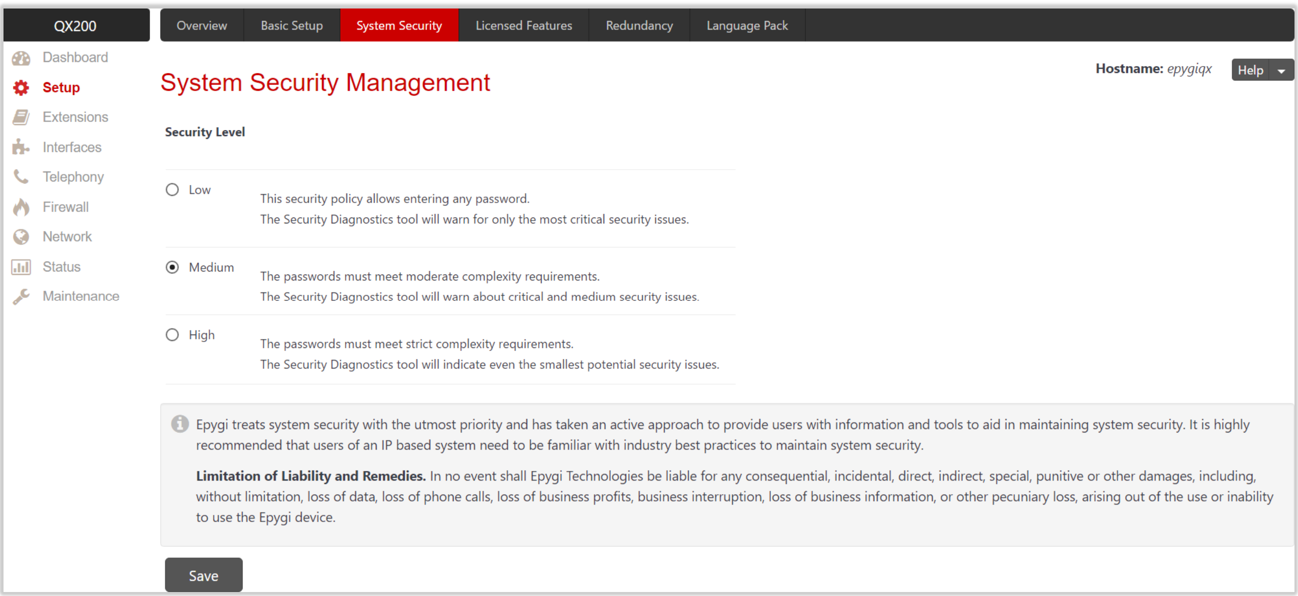


Figure 15: System Security Management page

The security levels are the following:

- **Low** – there are no specific restrictions on the strength of the saved password. The **Security Report** will generate warning for critical routing rules (PSTN and IP-PSTN) and if the Firewall and IDS services are disabled.
- **Medium** – the minimum strength of passwords must be "**moderate**". The **Security Report** will generate warnings for all unsecured routing rules, IP line and extension passwords, Firewall level (if it is set below **Medium**), disabled IDS, default administrator passwords.
- **High** – the minimum strength of passwords must be "**strong**". The **Security Report** will generate warnings for the IP line and extension passwords, disabled IDS, all unsecured routing rules, Firewall level (if it is set below **High**), default administrator passwords etc.

5.3 Licensed Features

5.3.1 Feature Keys

The **Feature Keys** page is used to show available and activated licensable feature keys on QX. Two types of licensable feature keys are available on QX:

- **Permanent keys** is used to activate licensable features on QX permanently, without time limitation. **Permanent keys** is available for all licensable features, except for two: **Epygi ACD Console** and **Epygi Hotel Console**.
- **Time limited keys** is used to activate or extend the operation for already activated licensable feature temporarily, for the specified period. The feature will no longer be functional after the period expiration date. The **Time limited keys** is available for all licensable features.

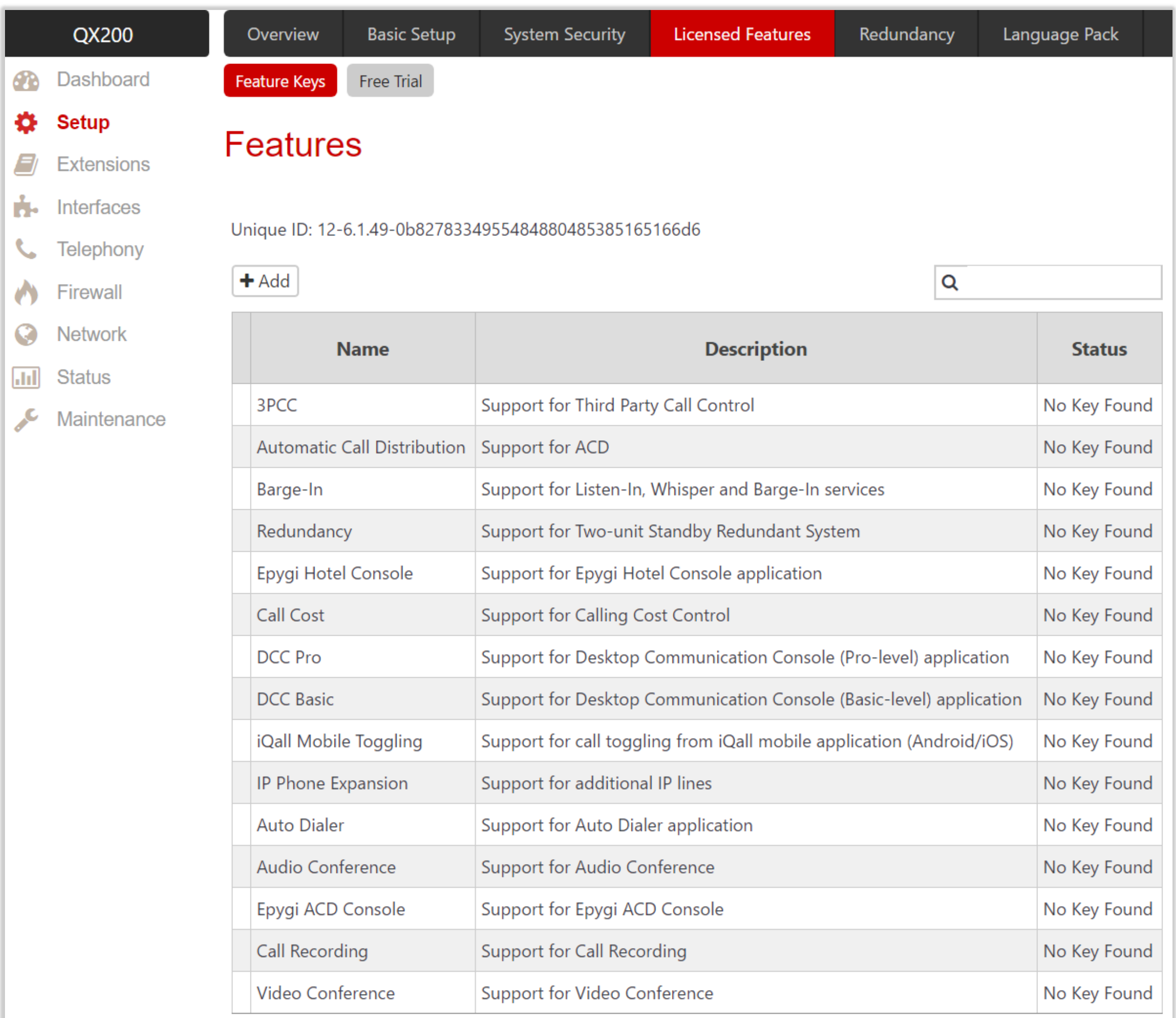


Figure 16: Features page

The following licensable features are available on QX:

- **Debug** enables SSH connection towards the QX for debugging purposes.
- **3PCC** activates **Third Party Call Control** feature on QX. This feature allows the call controlling applications running on PC to remotely initiate and handle calls on QX and to subscribe for certain event notifications from QX.

- Automatic Call Distribution activates the ACD feature which provides contact center solution for queuing and automatic distribution of the calls between contact center agents.
- Barge-In activates the Barge-In feature on QX. This feature allows PBX users to participate in the third-party calls while remaining imperceptible.
- Redundancy activates the Redundancy feature on QX.
- Epygi Hotel Console activates EHC application support for QX.
- Call Cost allows to limit and control the cost of calls through the routing rules on QX.
- PMSLINK Connection is used to enable the interface for connecting to PMSLINK middleware from char and integrate QX with PMS used in hotels.
- DCC Pro activates Desktop Communication Console Pro-level application support for QX.
- DCC Basic activates Desktop Communication Console Basic-level application support for QX.
- iQall Mobile Toggling allows users to alternate between their mobile (iPhone/Android) running iQall application and their desk phone without the call being disconnected.
- IP Phone Expansion activates additional IP lines (IP phone support) on QX.
- Auto Dialer activates Auto Dialer application support for QX.
- Audio Conference activates the Conference feature allowing the system to act as a standalone conference server.
- Epygi ACD Console activates Web monitoring support for ACD processes on QX.
- Call Recording activates the Call Recording feature which is used to record PBX, SIP or PSTN calls on QX and save recordings into the local recording box or upload to the remote server.
- Video Conference activates the Video Conference feature.

To receive a Feature Key, register the QX device and send a corresponding request to Epygi Technical Support. This request must include the Unique ID that is displayed in the Features page above the features list.

Enter a Feature Key as follows:

1. Click Add.
2. Enter the key in the Feature Key field.
3. Click Save. The status of the selected feature will turn to "Reboot needed".
4. Reboot QX to complete the installation. The status of feature will turn to "Activated".

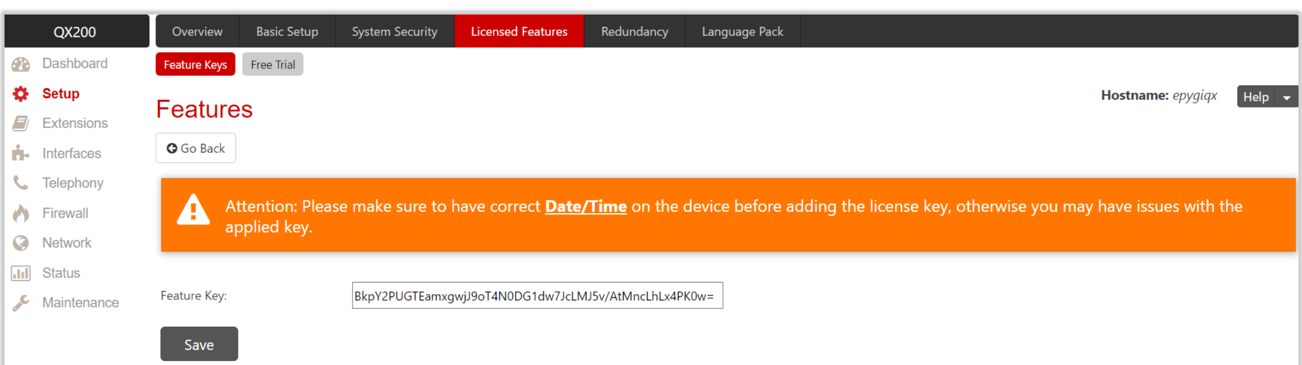


Figure 17: Adding a feature key

Note:

- Make sure to have the correct Date/Time on device before adding the license key, otherwise you may have issues with the applied key.
- When using **Call Recording** and/or **ACD** features on the QX50/QX200 it is advisable to use an SD memory card to expand the system memory. Currently, the recommended SD card's largest capacity is **32** GB.
- When using **Call Recording** and/or **ACD** features on the QX20/QX500 it is advisable to use a micro SD memory card to expand the system memory. Currently, the largest capacity of the recommended micro SD cards is **64** GB.

For more information on **Licensable Features**, refer to the Licensable Features on QX IP PBXs guide.

5.3.2 Free Trial

The **Trial Features Activation** page lists all QX features that can be activated for a trial period.

Expiration Date/Time is used to specify the trial period. Upon expiring the specified period, QX will reboot and trial feature(s) will be deactivated. TIP: The trial option can be activated on QX only once. You cannot activate the trial for the same or any other feature again after the first activation.

To activate **Trial** feature:

1. Tick the **Activate** checkbox next to the feature.
2. Specify the needed count under the **Count** column (depending on the selected feature).
3. Click **Save**. The QX will reboot and activate the selected trial feature(s).

5.4 Redundancy

The **Redundancy** feature is used to increase QX availability by using the second QX as a backup unit. This requires two units running the same firmware version and connected to each other through Ethernet or LAN ports, depending on the device model. The purpose of redundancy is to ensure uninterrupted functionality of QX. **Redundancy Settings** must be configured on both units. One of the units is configured as a master, the second one as a backup.

For more information on how to configure and use **Redundancy** feature, refer to the Redundancy Feature on QX IP PBXs guide.

5.5 Language Pack

All Epygi supported **Language Packs** (LPs) will change system voice messages to custom language. Some of LPs will change the QX WEB GUI and also the GUI interface on most of supported IP phones. For more information on **Language Packs**, refer to the Language Packs Overview for Epygi QX Line guide.

To upload a **Language Pack**:

1. Click **Choose File** to browse and select the LP file.
2. Click **Save** to start uploading the language pack. Clicking **Save** will stop some vital processes on QX, therefore it is required to manually reboot the device even if you have cancelled the LP update procedure.
3. Click **Yes** to proceed the upload. QX will be rebooted automatically.
4. Uploaded LP will appear in the **Current language pack** field. After successful upload, you will be able to:
   - Change the language of WEB GUI session from **Login** page or from **Main Menu**.

- Switch the system voice messages to the custom language and change the GUI interface of some supported IP phones. TIP: Choose the language from the Regional Settings and Preferences section to change the system voice messages and GUI language for the IP phones. The IP phones will be automatically rebooted to change language.

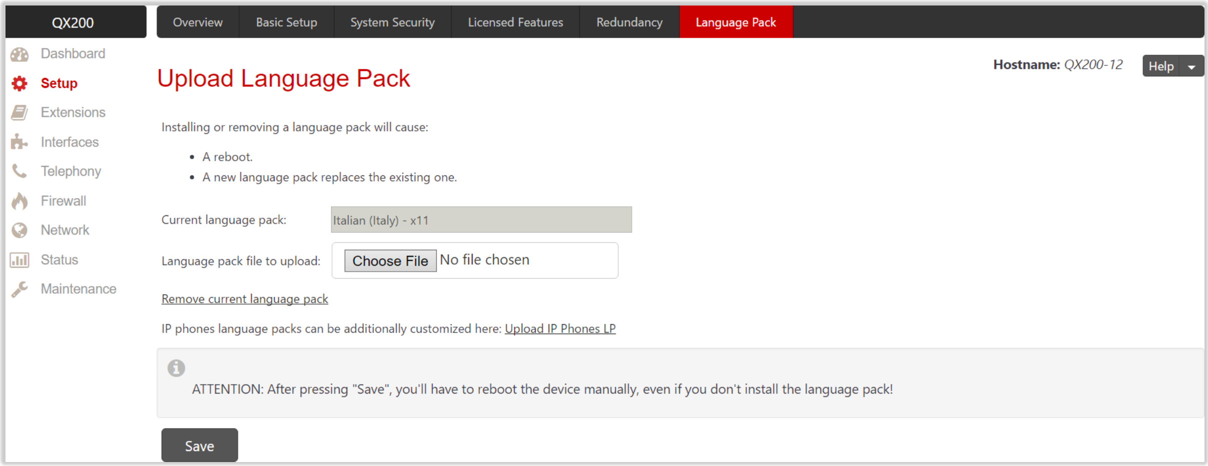


Figure 18: Language Pack page

To upload a custom LP on the IP phone(s):

1. Click the **Upload IP Phones LP** link.
2. Click the hyperlinked IP phone vendor.
3. Click **Choose File** to browse and select the LP file.
4. Click **Yes** to proceed the upload. Then reboot the IP phone to activate the new LP. TIP: Clicking **Save** will stop some vital processes on the IP phone, therefore reboot your phone manually even if you have cancelled the LP update procedure.

Note: Only one custom LP can be uploaded at a time. Thus, the new LP will remove the existing one and reboot the QX. Once QX is rebooted, the connected IP phones will reboot then.

6 Extensions Menu

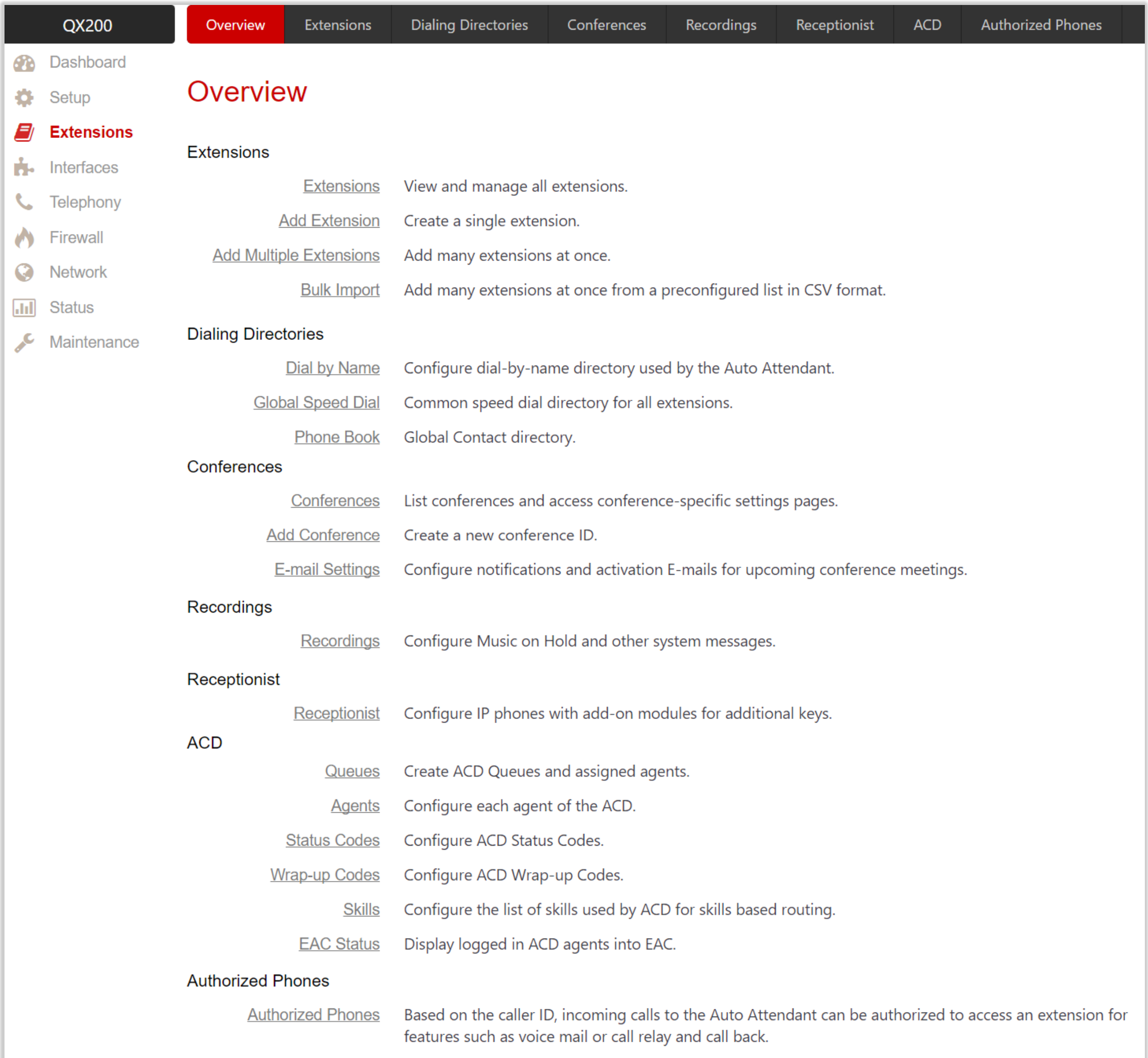


Figure 19: Extensions Menu overview

6.1 Extensions

6.1.1 Extensions

Navigating to the Extensions Management page for the first time after the QX initial start or configuration restore you will be prompted to choose the extensions length applicable to all QX default extensions.

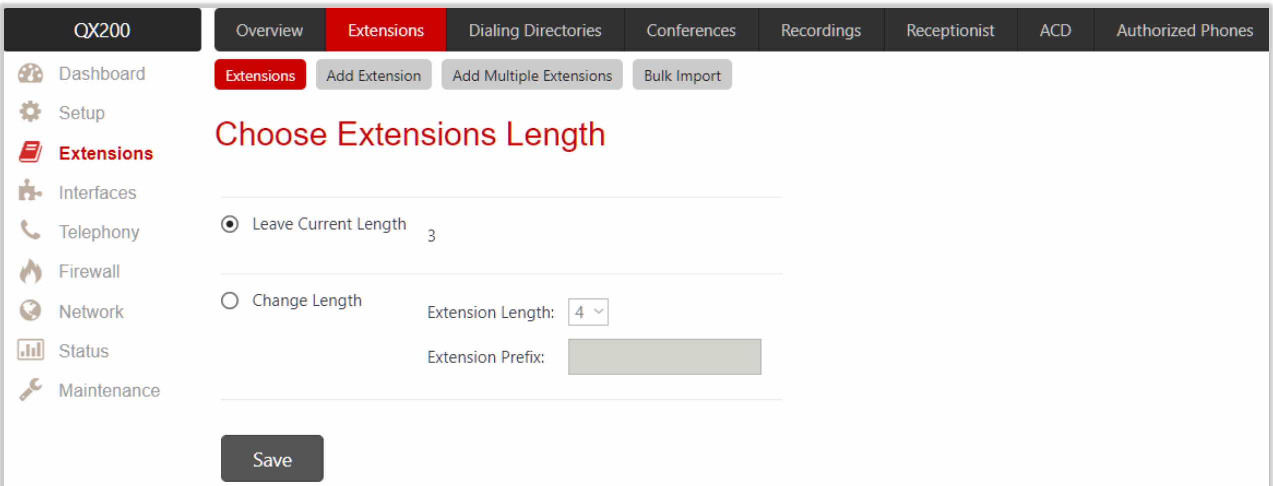


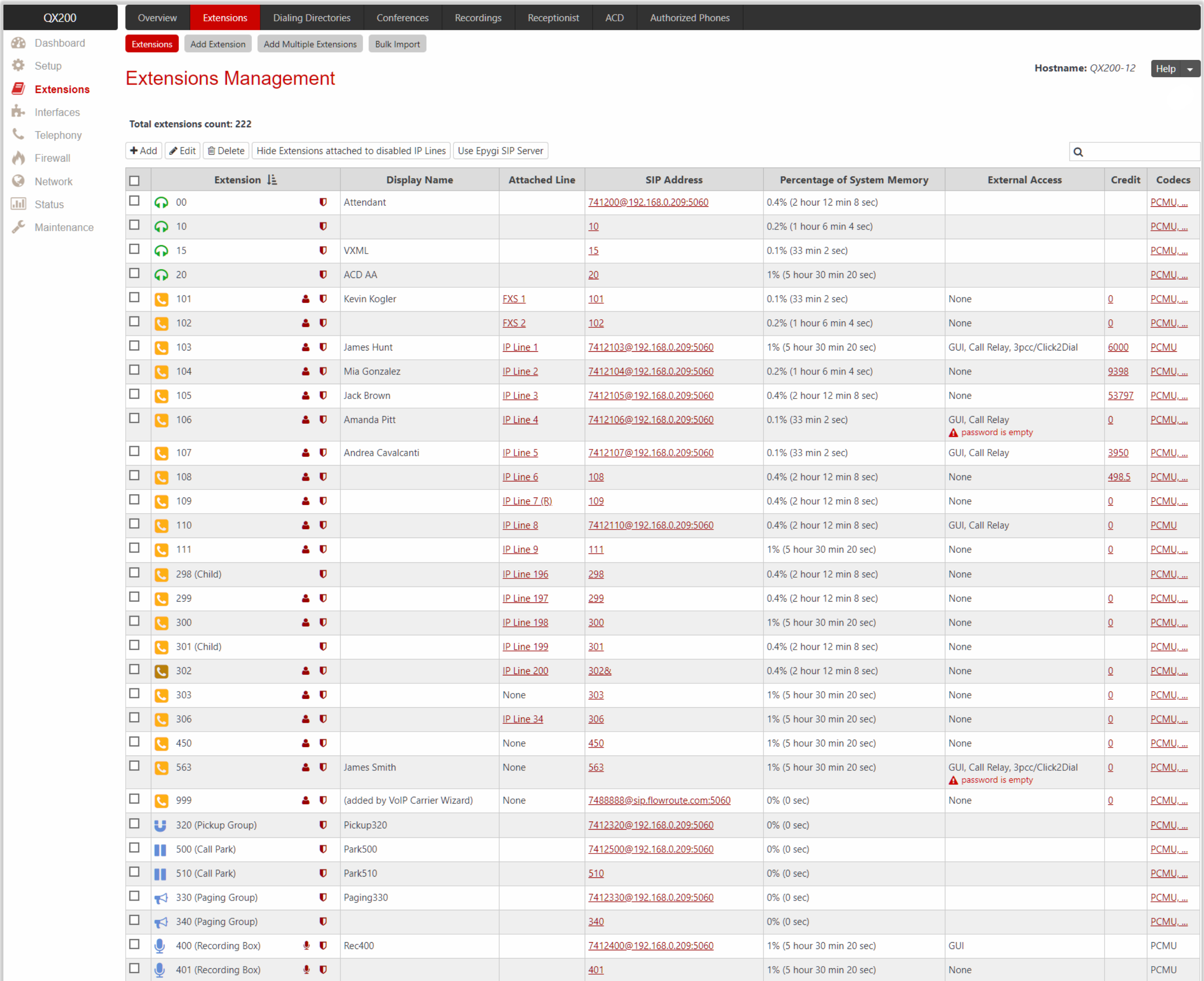
Figure 20: Choose Extensions Length page

The following options are available:

- Leave Current Length – by default, extension length is 3 on QX20, QX50, QX200, QX500, QXISDN4+ and is 4 on the QX2000 and QX3000. In front of this selection, the actual length of configured extensions is displayed.
- Change Length changes the length of extensions as follows:
  - Extension Length is used to select the length of extensions. It will be applied for all existing extensions on QX.
  - Extension Prefix is used to set the prefix the existing extensions as well as the newly created extensions should start with. The prefix cannot start with 0 or 9.

Attention:

- In case of saving the settings on the Choose Extensions Length page, all existing extensions will lose the custom voice messages and voice mails in the mailbox. The device will be rebooted. The Choose Extensions Length page will not appear again unless the default configuration settings are not restored on QX.
- QX20 is limited to 100, QX50/QXISDN4+ to 200, QX200 to 400, QX500 to 800, QX2000 to 2400 and QX3000 to 3400 extensions in total.



Extension	Display Name	Attached Line	SIP Address	Percentage of System Memory	External Access	Credit	Codecs
00	Attendant		741200@192.168.0.209:5060	0.4% (2 hour 12 min 8 sec)			PCMU, ...
10			10	0.2% (1 hour 6 min 4 sec)			PCMU, ...
15	VXML		15	0.1% (33 min 2 sec)			PCMU, ...
20	ACD AA		20	1% (5 hour 30 min 20 sec)			PCMU, ...
101	Kevin Kogler	FXS 1	101	0.1% (33 min 2 sec)	None	0	PCMU, ...
102		FXS 2	102	0.2% (1 hour 6 min 4 sec)	None	0	PCMU, ...
103	James Hunt	IP Line 1	7412103@192.168.0.209:5060	1% (5 hour 30 min 20 sec)	GUI, Call Relay, 3pcc/Click2Dial	6000	PCMU
104	Mia Gonzalez	IP Line 2	7412104@192.168.0.209:5060	0.2% (1 hour 6 min 4 sec)	None	9398	PCMU, ...
105	Jack Brown	IP Line 3	7412105@192.168.0.209:5060	0.4% (2 hour 12 min 8 sec)	None	53797	PCMU, ...
106	Amanda Pitt	IP Line 4	7412106@192.168.0.209:5060	0.1% (33 min 2 sec)	GUI, Call Relay password is empty	0	PCMU, ...
107	Andrea Cavalcanti	IP Line 5	7412107@192.168.0.209:5060	0.1% (33 min 2 sec)	GUI, Call Relay	3950	PCMU, ...
108		IP Line 6	108	0.4% (2 hour 12 min 8 sec)	None	498.5	PCMU, ...
109		IP Line 7 (R)	109	0.4% (2 hour 12 min 8 sec)	None	0	PCMU, ...
110		IP Line 8	7412110@192.168.0.209:5060	0.4% (2 hour 12 min 8 sec)	GUI, Call Relay	0	PCMU
111		IP Line 9	111	1% (5 hour 30 min 20 sec)	None	0	PCMU, ...
298 (Child)		IP Line 196	298	0.4% (2 hour 12 min 8 sec)	None		PCMU, ...
299		IP Line 197	299	0.4% (2 hour 12 min 8 sec)	None	0	PCMU, ...
300		IP Line 198	300	1% (5 hour 30 min 20 sec)	None	0	PCMU, ...
301 (Child)		IP Line 199	301	0.4% (2 hour 12 min 8 sec)	None		PCMU, ...
302		IP Line 200	302&	0.4% (2 hour 12 min 8 sec)	None	0	PCMU, ...
303		None	303	1% (5 hour 30 min 20 sec)	None	0	PCMU, ...
306		IP Line 34	306	1% (5 hour 30 min 20 sec)	None	0	PCMU, ...
450		None	450	1% (5 hour 30 min 20 sec)	None	0	PCMU, ...
563	James Smith	None	563	1% (5 hour 30 min 20 sec)	GUI, Call Relay, 3pcc/Click2Dial password is empty	0	PCMU, ...
999	(added by VoIP Carrier Wizard)	None	7488888@sip.flowroute.com:5060	0% (0 sec)	None	0	PCMU, ...
320 (Pickup Group)	Pickup320		7412320@192.168.0.209:5060	0% (0 sec)			PCMU, ...
500 (Call Park)	Park500		7412500@192.168.0.209:5060	0% (0 sec)			PCMU, ...
510 (Call Park)	Park510		510	0% (0 sec)			PCMU, ...
330 (Paging Group)	Paging330		7412330@192.168.0.209:5060	0% (0 sec)			PCMU, ...
340 (Paging Group)			340	0% (0 sec)			PCMU, ...
400 (Recording Box)	Rec400		7412400@192.168.0.209:5060	1% (5 hour 30 min 20 sec)	GUI		PCMU
401 (Recording Box)			401	1% (5 hour 30 min 20 sec)	None		PCMU

Figure 21: Extensions Management page

The **Extensions Management** table consists of the following components:

- **Extension** lists the numbers for extensions on QX. These numbers are used for calling extensions internally.
- **Display Name** is an optional name given to extension mainly to identify the extension owner at the called side.
- **Attached Line** indicates the IP or FXS line the extension is attached to. TIP: If the **Remote Extension** service is enabled on the extension, **R** will be shown. **None** is shown when no FXS or IP line is attached to the extension.
- **SIP Address** shows the full SIP address of the extension, (i.e., **username@sipserver:port**) when **Registration on SIP Server** is enabled, otherwise the SIP address will be displayed in the following format: "**username, Proxy: sipserver:port**". If no **username** is defined, the extension number will be displayed instead.
- **Percentage of System Memory** indicates the part of total memory allocated to extension and shows the duration available for the voice mails and custom messages of the extension. The available time duration depends on the selected Voice Mail Recording Codec.

- **External Access** indicates whether the **Allow Call Relay**, **Allow GUI Login** or **Allow 3pcc/Click2Dial Access** options are enabled on the extension.
- **Credit** indicates the available credit amount of the extension.
- **Codecs** shows activated **Codecs** on the extension. Click the **Codecs** link to access and modify the codecs on the extension.

6.1.2 Add Extension

To add a new **Extension**:

1. Click **Add Extension**.
   - Enter the extension number.
   - Select the **extension type**. The following types are available: **Auto Attendant, User Extension, Pickup Group, Call Park, Paging Group** and **Recording Box.**
2. Click **Save** to add the new extension to the **Extension Management** table.

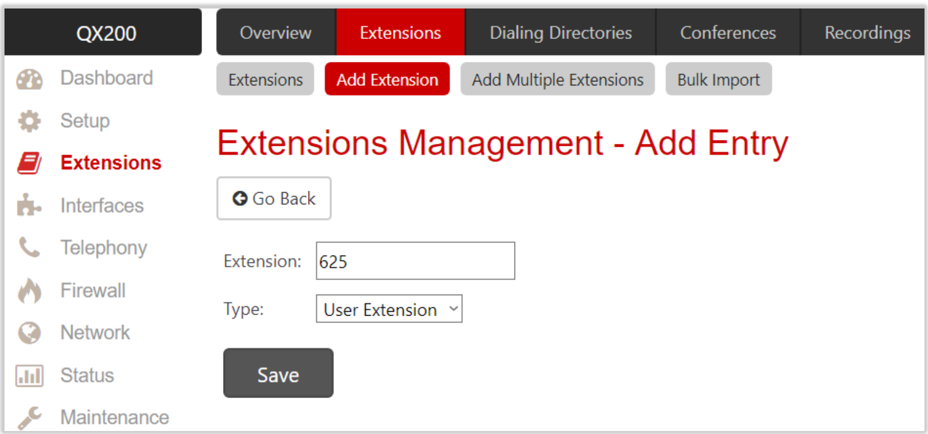


Figure 22: Extensions Management – Add Entry page

Two types of user extensions, **active** and **inactive**, can be created on QX.

- **Active extensions** are those that are attached to the line (IP or FXS), can place and receive calls and use available telephony services.
- **Inactive extensions** are those that are not attached to the line (IP or FXS), cannot place and receive calls and use only part of telephony services.

Note:

- **Manually** adjust the routing rules for calling extensions with custom length since the call routing rule(s) for calling PBX extensions will not be adjusted automatically.
- A maximum extension length is **20** digits.
- The **Recording Box** extension type becomes available if the **Call Recording** feature is activated on QX.

6.1.3 Add Multiple Extensions

The **Extensions Management – Add Multiple Extensions** page is used to create multiple extensions at once.

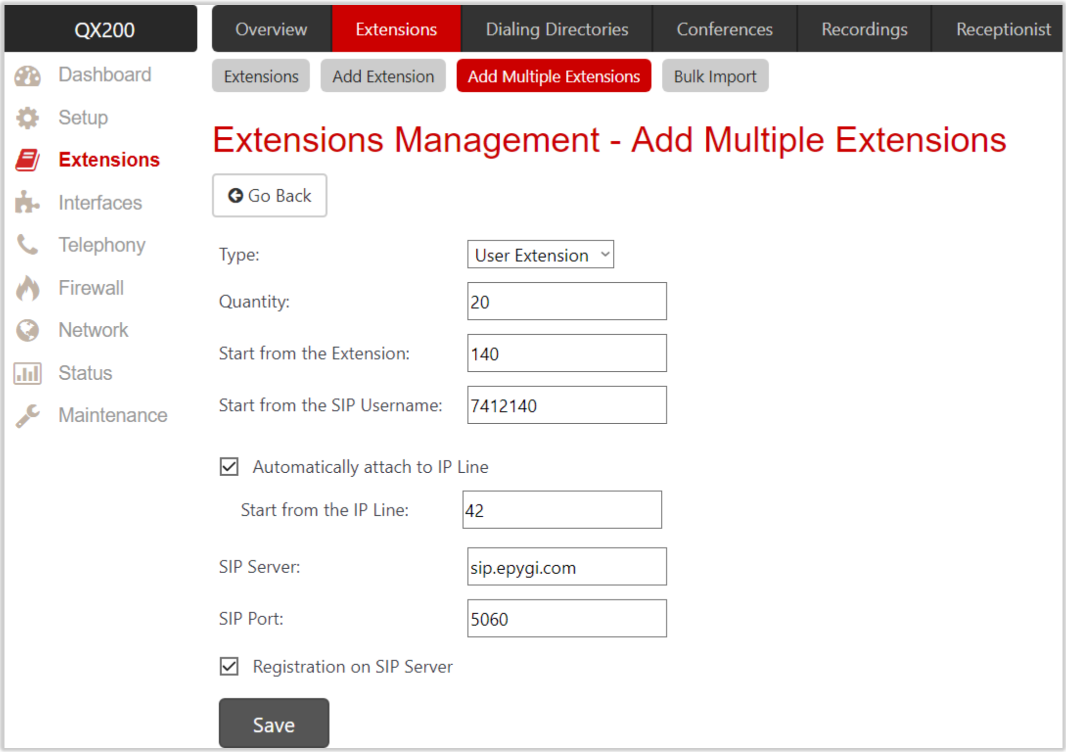


Figure 23: Extensions Management – Add Multiple Extensions page

To add multiple **Extensions**:

1. Select the extension type. The following types are available: **Auto Attendant, User Extension, Pickup Group, Call Park, Paging Group** and **Recording Box.**
2. Enter the amount of extensions.
3. Enter the **number** for the first extension. Based on the **Quantity**, next extensions will have subsequent numbers.
4. Enter the **SIP Username** of the first extension. Based on the **Quantity**, next extensions will have subsequent SIP usernames.
5. Tick the **Automatically attach to IP Line** checkbox to attach user extensions to IP lines.
6. Enter the **number** of the first IP line to be attached.
7. Enter the **SIP Server** and **SIP Port**. If the latter is not specified, QX will access the SIP server via the default **5060** port.
8. Tick the **Registration on SIP Server** checkbox to enable registration of the extensions on the SIP server.
9. Click **Save** to add the new extensions to the **Extension Management** table.

Note:

- **Manually** adjust the routing rules for calling extensions with custom length since the call routing rule(s) for PBX extensions will not be adjusted automatically.
- The **Recording Box** extension type becomes available if the **Call Recording** feature is activated on QX.
- A maximum extension length is **20** digits.
- A maximum SIP Username length is **32** characters. The SIP Username can consist of lowercase and uppercase alphabetic characters, digits and symbols.

6.1.4 Edit Extension

You can modify both **admin** and **user** settings of the extension.

- To modify extension **admin** settings, click the **Admin Settings** icon or tick the checkbox next to the extension and click **Edit**. Remember to save changes before moving between configuration sections.
- To modify extension **user** settings, click the **User Settings** icon.

You can modify **admin** settings of two or more extensions at once by ticking checkboxes next to extensions and clicking **Edit**. When editing multiple extensions, fields that cannot be edited for multiple records have **Multiple** values. When editing user extension and auto attendant together, only common fields will be shown. Additionally, tick the **Select to modify fields** checkbox to submit changes of the corresponding settings (options), otherwise the changes won't be applied.

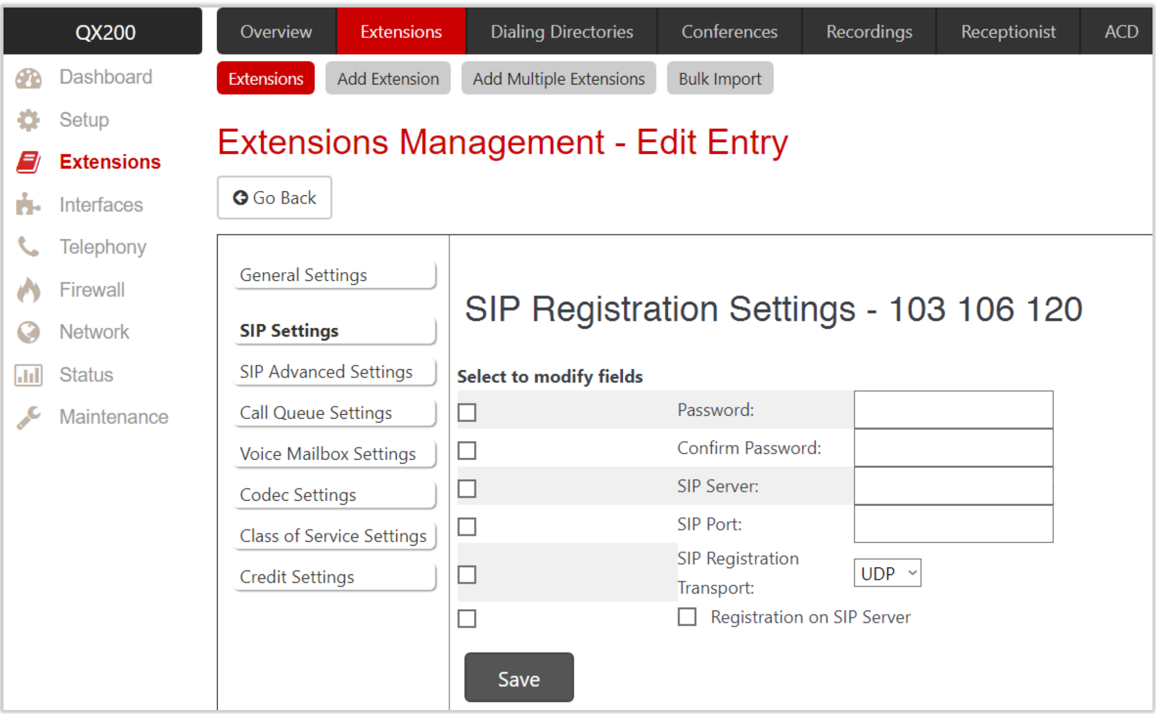


Figure 24: Extensions Management – Edit Entry page (for multiple edit operation)

6.1.5 User Extension

The following sections are available for configuration:

- General Settings
- SIP Settings
- SIP Advanced Settings
- Remote Settings
- Call Queue Settings
- Voice Mailbox Settings
- Class of Service Settings
- Credit Settings
- Licensing

General Settings

This section is used to uniquely identify the extension through parameters described below:

- Display Name is the caller ID that will be displayed on the callee's phone.
- Password is used to assign a password to the extension. TIP: This password will be used for GUI Login, for Call Relay and remote access to voice mailbox.
- Attached Line lists all free lines an extension can be attached to. Extension should be attached to a line (either IP or FXS) to be able to make and receive calls. If there is no line attached to an extension, then it is called Virtual Extension (VE). VE can't place/receive calls but is allowed to use a limited number of QX telephony services, such as call forwarding service or voice mail service to store and manage the messages from callers. Any VE can easily become a real extension after attaching a line and vice versa. By default, all extensions on QX have lines attached already. Extensions cannot be detached from the line if the Remote Extension service is enabled on. To detach the extension from the line, disable the Remote Extension service on the extension first.
- Use Kickback enables the Kickback service on the extension for the blind transfer scenario. When an extension blindly transfers the call to other extension and if there is no answer from the called extension, the call will automatically get back to the extension who initiated the transfer instead of getting into the destination's voice mailbox or being disconnected.
- Allow Call Relay enables the extension to be used to access the Call Relay service from auto attendant. It is recommended to set a proper and non-blank password when enabling this service in order to protect it from an unauthorized access.
- Allow GUI Login activates WEB GUI access (by extension number and password) for the extension.
- Allow 3pcc/Click2Dial Access enables the current extension to be used with applications based on the QX interface and QX Click2Dial application.
- Show on Public Directory – if selected, automatically includes the extension display name and number in the Phone Book (Directory) and Extension Directory.

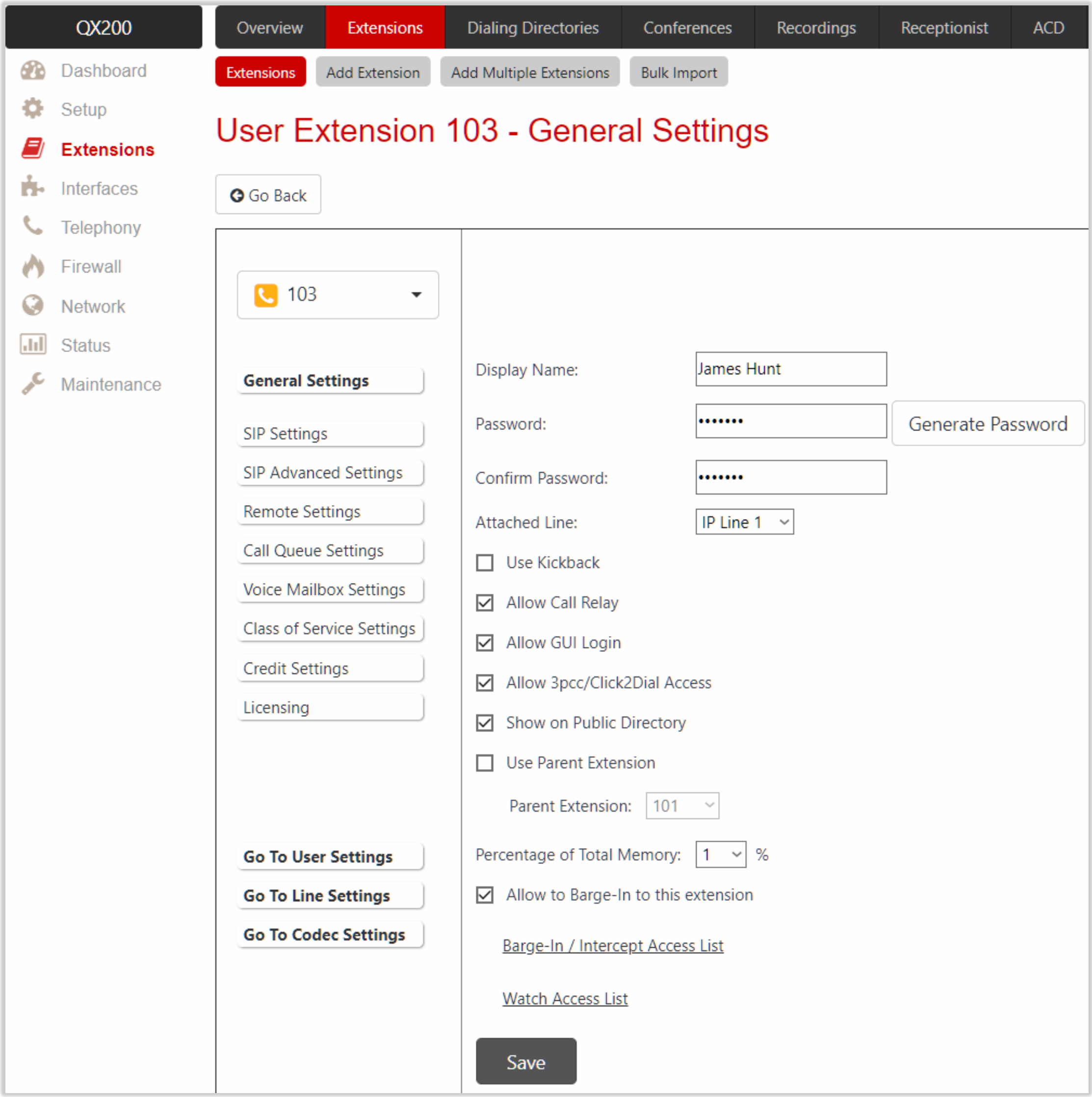


Figure 25: User Extension – General Settings section

- Use Parent Extension allows the current extension to be configured as a Child for the Parent, selectable from the Parent Extension drop-down list. When done, the Use Parent Extension checkbox will disappear for the Parent and the Child Extension List link will appear instead. For more information, refer to the Parent-Child Configuration.
  - ➢ Child Extension List leads to the Child Extension of Parent Extension page, where you can see the list of extensions defined as Child for the Parent extensions. The General Settings section of the Child extension has the following components:
    - ♦ Use Parent Extension – if not selected, interrupts the Use Parent Extension service on the Child extension.
    - ♦ Parent Extension is used to select Parent extension for the Child extension.
- Allow Concurrent Calls to Parent-Child Group allows to choose between the following options available for handling inbound call to Parent-Child group:
  - ➢ If selected, incoming calls continue ringing on available phones when one of the phones in Parent-Child group is busy or rejects the call.
  - ➢ If not selected, incoming calls will follow busy state rules (Busy Call Forwarding, Call Queue, VMS, etc.) depending on what is configured, if any of the phones in the Parent-Child group is busy. If all extensions in the Parent-Child group are free and are ringing, and any of them presses Reject button

(or somehow else declines the incoming call), then the **entire group** will be considered as busy. Therefore, incoming call will follow busy state rules depending on what is configured. Note: If the **Call Waiting Service** is enabled on the **Parent** extension, then extensions of **Parent-Child** group will receive the second call.

- **Percentage of Total Memory** is used to allocate memory for voice mails and custom messages of the extension.
- **Enable Ringing Simulation** – if selected, extra ring tones will be played to the caller before the voice mail service gets activated (available on **Virtual Extensions** only), otherwise the voice mail service will be activated immediately. The ring tones will be played during the timeout specified in the **Ringing Simulation Timeout**.
- **Intercept Access List** leads to the **Intercept Access List of User Extension** page to define extension(s) allowed to intercept calls.
- **Allow to Barge-In to this extension** enables **Barge-In** on the extension. The **Barge-In / Intercept Access List** leads to the **Call Barge-In / Intercept Access List** page to define extensions allowed to barge-in to the current extension calls or intercept calls.
- **Watch Access List** leads to the **Watch Access List** page to define the extensions allowed to watch calls.

Barge-In / Intercept Access List of User Extension

This page is used to define a list of extensions that are capable to **Barge-In / Intercept** the extension calls and defines the appropriate permissions.

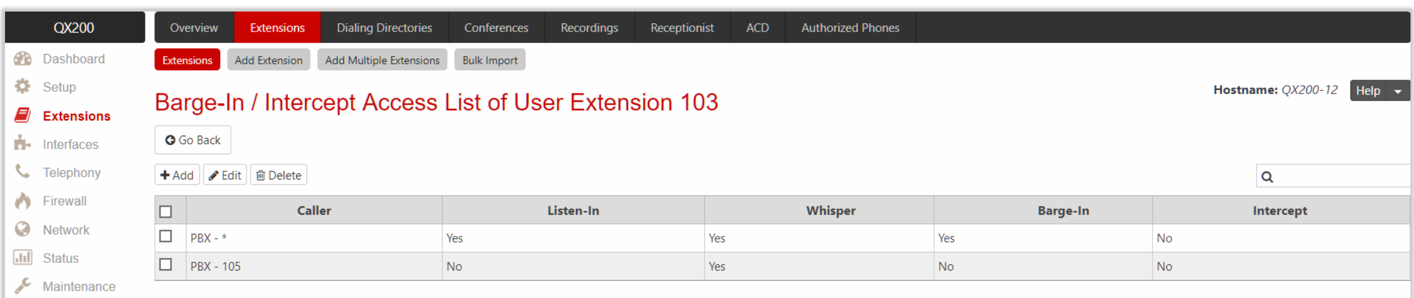


Figure 26: Call Barge-In/Intercept Access List

To add a new extension:

1. Click **Add**.
   - ➢ Enter the extension number(s) allowed to **Barge-In / Intercept** the current extension calls.
   - ➢ Select **Barge-In**, **Intercept** options, to allow the selected action only. The following options are available: **Listen-In**, **Whisper**, **Barge-In** and **Intercept**.
2. Click **Save** to add the new entry to the **Barge-In / Intercept Access List** table.

Note: The **Barge-In / Call Intercept** calls neither will be displayed in the **Active Calls** table on the **Dashboard** nor will be registered in the **Call History** table.

Watch Access List of User Extension

This page is used to define a list of extensions that are able to watch the current extension calls and defines the appropriate permissions.

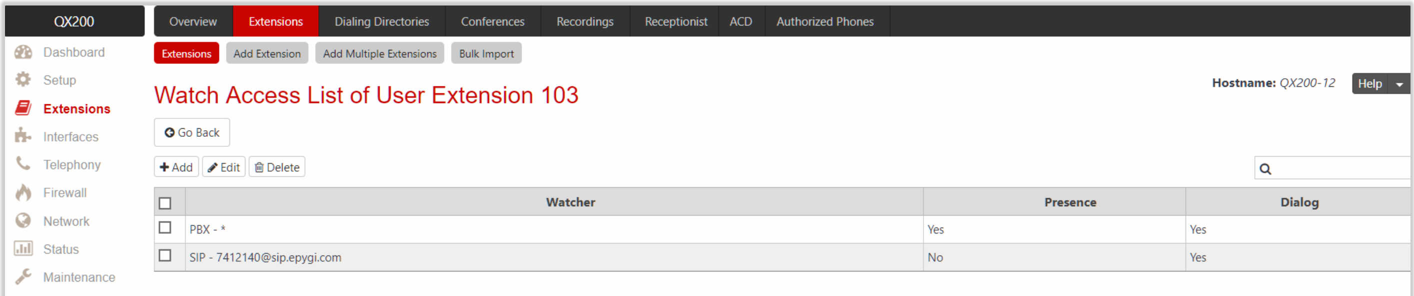


Figure 27: Watch Access List

To add a new extension:

1. Click **Add**.
   - ➢ Enter the extension number(s).
   - ➢ Select the **Allow Presence Subscriptions** and **Allow Dialog Subscriptions** options to allow subscriptions to the current extension.
2. Click **Save** to add the new entry to the **Watch Access List** table.

SIP Settings

This section describes how to register QX extension on a SIP server to receive external SIP calls.

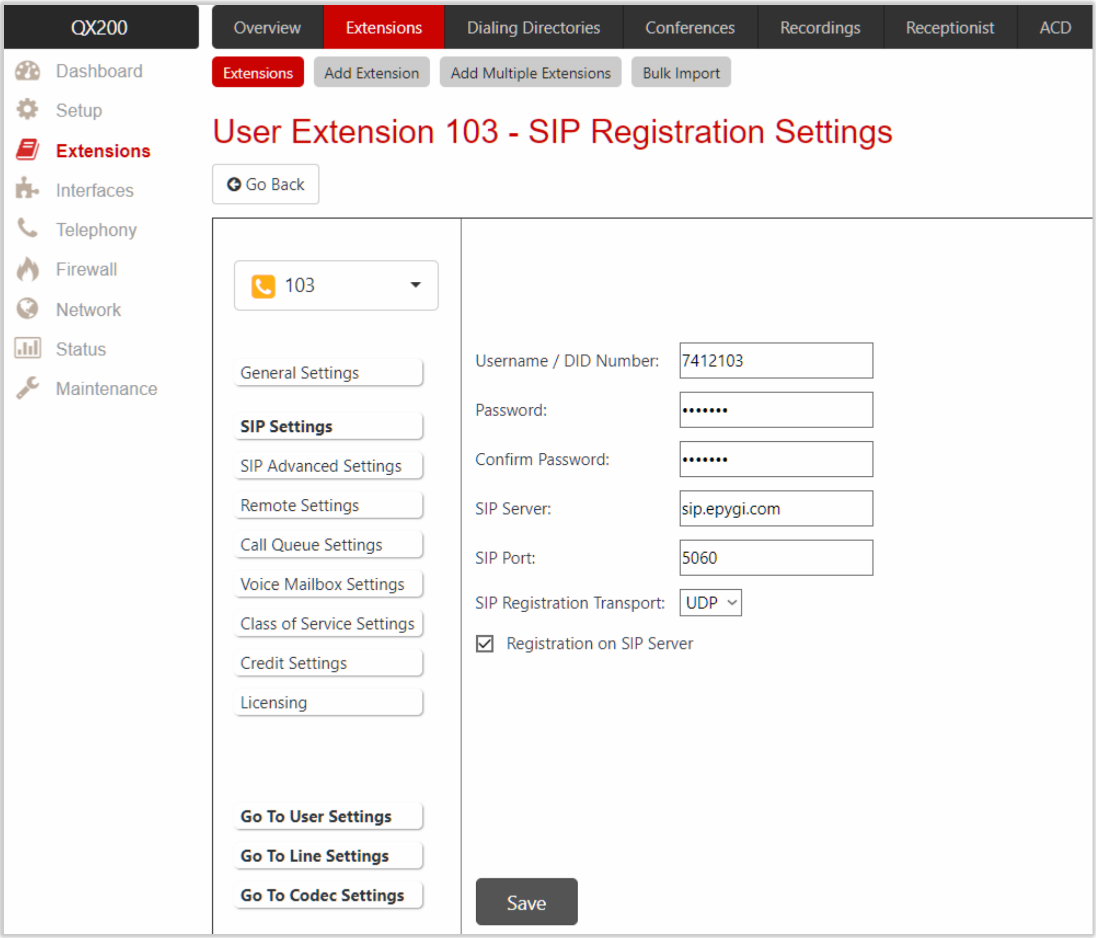


Figure 28: SIP Settings section

The following settings (options) are available:

- Username / DID Number is the registration username or the DID number on the external server. TIP: The maximum SIP Username length is 32 characters. The SIP Username can consist of lowercase and uppercase alphabetic characters, digits and symbols.
- Password is the registration password on the SIP server.
- SIP Server is the address of the SIP server. It can be either an IP address (e.g. 198.51.100.1) or a hostname (e.g. sip.epygi.com). TIP: The maximum SIP server length is 32 characters. The SIP server can consist of lowercase and uppercase alphabetic characters, digits and symbols.
- SIP Port is the port number used to connect to the SIP server. TIP: If the SIP port is not specified, QX will access the SIP server through the default 5060.
- SIP Registration Transport is used to select SIP Transport (UDP, TCP and TLS) for the registration. TIP: If the QX is located behind a NAT router, the TCP ports (for TCP and TLS) should be manually configured from NAT Traversal – SIP Parameters page and opened on the NAT router accordingly.
- Registration on SIP Server is used to register extension on the SIP server.

How it works: Upon receiving a SIP Invite message from an external server, QX will look to match the called number in the Username / DID Number field. If the ITSP does not require each DID to uniquely register on the SIP server, then only enter the DID number in the Username / DID Number field and leave other fields blank.

SIP Advanced Settings

This section describes how to configure advanced and specific SIP settings for QX extension.

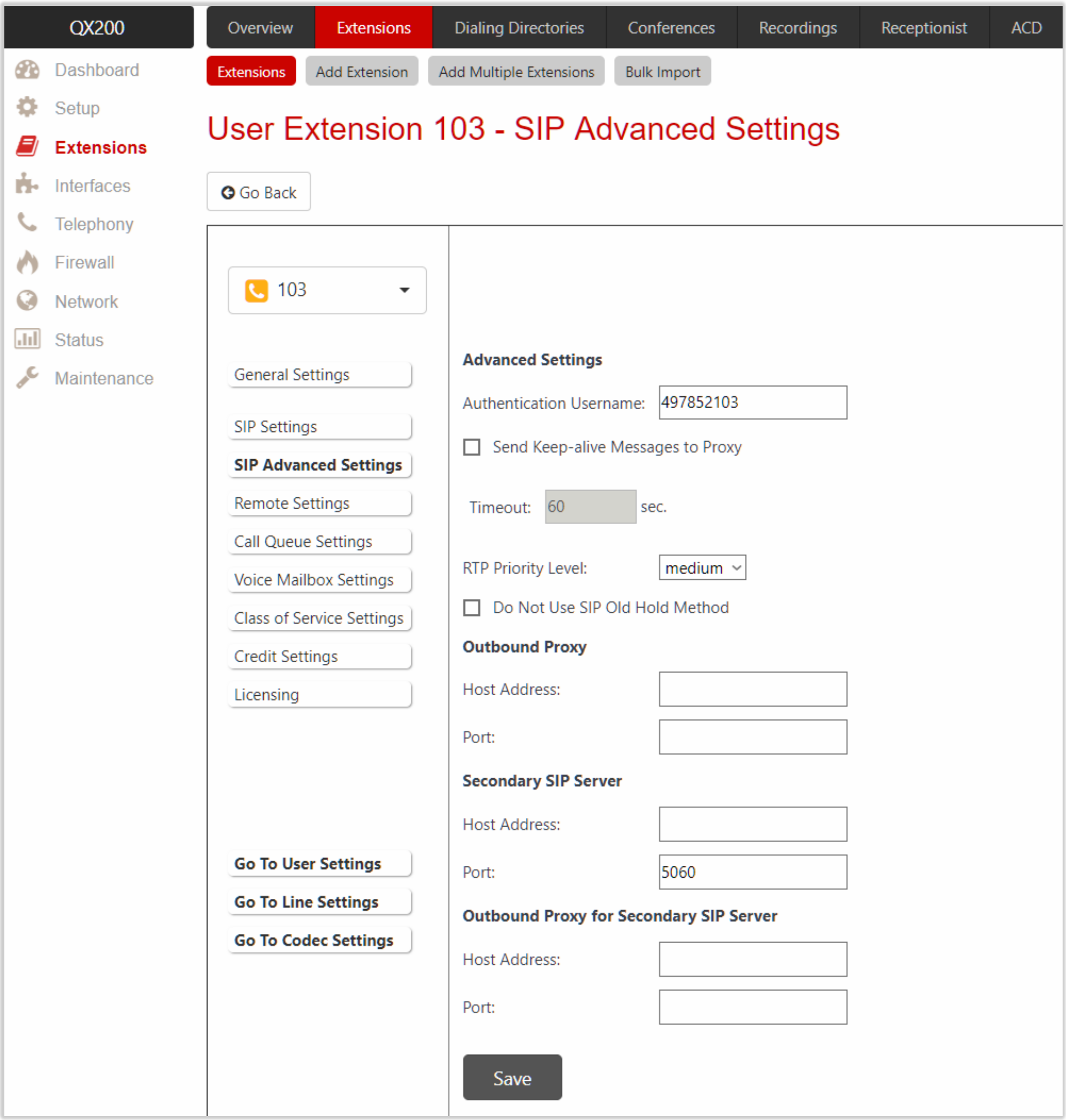


Figure 29: SIP Advanced Settings section

The following settings (options) are available:

- Authentication Username is used to set an identification parameter. It should be provided by ITSP and can be requested for some SIP servers only. For others, the field should be left blank.
- Send Keep-alive Messages to Proxy enables the SIP registration server accessibility to the verification mechanism.
- Timeout is used to set the timeout between two attempts for the SIP registration server accessibility verification. If no reply is received from the primary SIP server within this timeout, the secondary SIP server will be contacted. When the primary SIP server recovers, SIP packets will resume being sent to it.
- RTP Priority Level is used to select the level of priority (low, medium or high) of the RTP packets sent from the extension. RTP packets with higher priority will be sent first in case of heavy traffic.
- Do Not Use SIP Old Hold Method – if selected, a new recommended method of call hold in SIP (the call hold request is indicated with the "a=sendonly" media attribute, rather than with the IP address of

0.0.0.0) will be used. This option should be selected if the remote party does not recognize hold requests initiated from the QX.

- **Outbound Proxy** is the SIP server where all SIP requests and SIP messages are transferred to. Some SIP servers use an outbound proxy to escape NAT restrictions. If an outbound proxy is specified for an extension then all SIP calls originating from that extension will go through that outbound proxy, i.e., all requests will be sent to that outbound proxy.
- **Secondary SIP Server** is used to act as an alternative SIP server when the primary SIP server becomes inaccessible. If the connection with the primary SIP server fails, QX will automatically start sending SIP messages to the secondary SIP server. It will switch back to the primary SIP server as soon as the connection is reestablished.
- **Host Address** and **Port** are used to set the host address and SIP port of the **Outbound Proxy**, **Secondary SIP Server** and the **Outbound Proxy for the Secondary SIP Server** respectively. These settings are provided by ITSP and are used by QX to reach the SIP servers.

Remote Settings

This section describes how to configure **Remote Extension** (RE) service for QX extension. This is an advanced telephony service that allows users to connect phone to QX remotely. The user needs to register an IP phone or softphone on the QX by defining the QX global IP address and an appropriate Username/Password. The registered phone can fully act as a phone connected locally to QX, i.e. you can use all QX telephony services, place and receive calls, access voice mails, etc. TIP: The RE service can be enabled only for extensions attached to the line (IP or FXS). The following settings (options) are available:

- **Enable Remote Extension** is used to activate service on QX.
- **Username** and **Password** are used to set the identification parameters used by the remote phone to register it on QX. TIP: The **Username** and **Password** must match on both QX and IP phone for successful registration.
- **Line Appearance** is used to set a number of simultaneous calls supported by the remote phone.
- **Enable RTP Proxy** – if selected, the incoming and outgoing RTP streams to/from the remote IP phone will be routed through QX, otherwise RTP packets will move directly between peers.
- **Fallback to local extension when not registered** – if selected, the incoming calls to the local extension will be forwarded to the remote IP phone only if it is

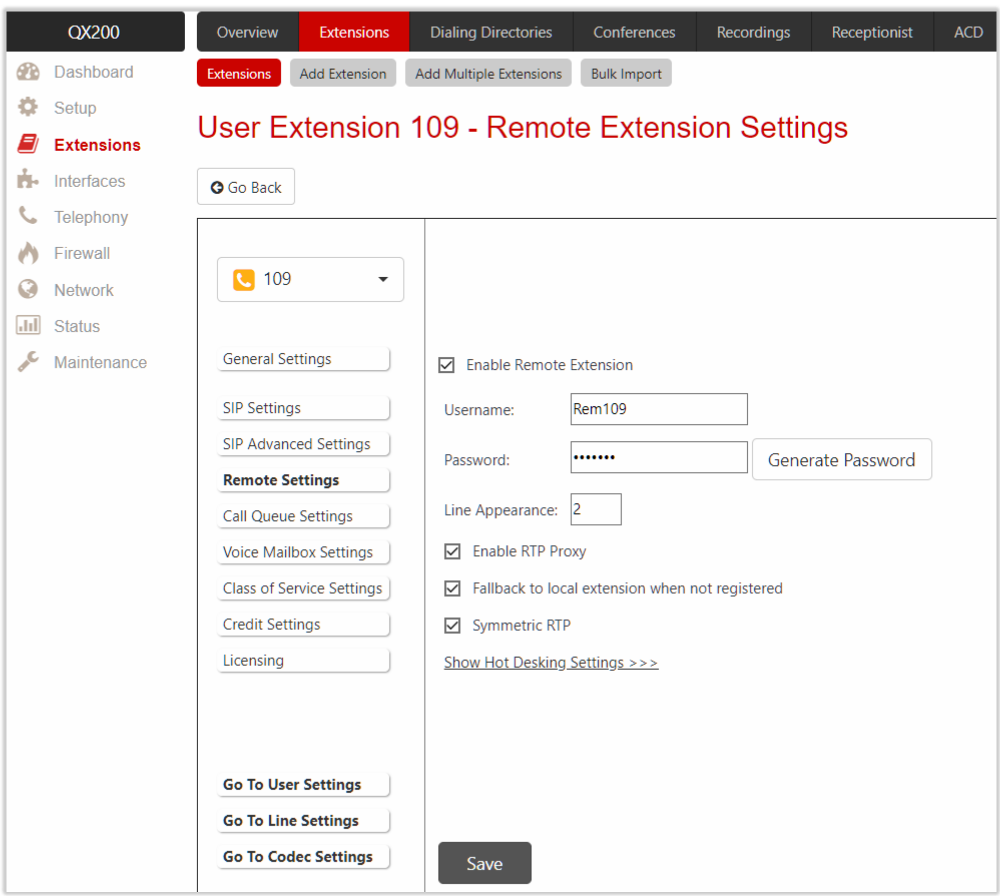


Figure 30: Remote Settings section

registered. Otherwise, when the remote IP phone is unregistered, incoming calls will be routed to the local extension it is attached to.

- Symmetric RTP must be selected when RE is located behind the NAT router.
- Enable Hot Desking is used to activate Hot Desking service on RE. Note: The Hot Desking section is the same as for IP line.

For more information on how to configure and use Remote Extension service, refer to the Remote Extension Configuration on QX IP PBXs guide.

Call Queue Settings

This section describes how to configure the Call Queue service on QX extension allowing multiple incoming calls to wait in the queue and be answered in the order they have been received. This service can be used in the Receptionist as well. The following settings (options) are available:

- Enable Call Queue is used to activate service on QX.
  - Call Queue Size is used to set the length of call queue. This is the maximum number of calls that will be accepted into the queue and kept on hold while the extension is on a call. If the queue is filled up then the next incoming call will be forwarded to the extension Voice Mail (if enabled). Otherwise the call will be disconnected.
  - Max Calls Presented to Extension is used to set the maximum number of active calls on the line. So, if the maximum call number is set to 1 and the extension is in call then an incoming call will go to the call queue. If the maximum call number is set to 2 and the extension is in call then the alert for the next incoming will be played in the background (if the Call Waiting service is enabled on the extension) and the extension will put the first call on hold to answer the second one or they can be joined for a call conference.
- Enable No Answer Redirect – if activated and configured, callers will be redirected to the specified address after some time waiting in the queue. The Prompt Repetition is used to set the number of prompts to be played before redirection.
- ZeroOut Redirection – if activated and configured, callers dialing 0 during queue welcome message or recurring prompt will be redirected to the specified address.
  - Voice Mail redirects the call to the extension Voice Mail.
  - Call Type, Calling Address (identical for both Call Redirection and ZeroOut Redirection) is used to set the destination address the call will be redirected to. The address strictly depends on the call type.
- Call Queue Welcome Message is used to play a message (default or custom) once when reaching the extension Call Queue.
- Call Queue Prompt is used to play a queue prompt after Call Queue Welcome Message.

Note: The Call Forwarding if Busy and Voice Mail services will function once the call queue will be filled up. Thus, these services will affect those calls that are left out of the queue.

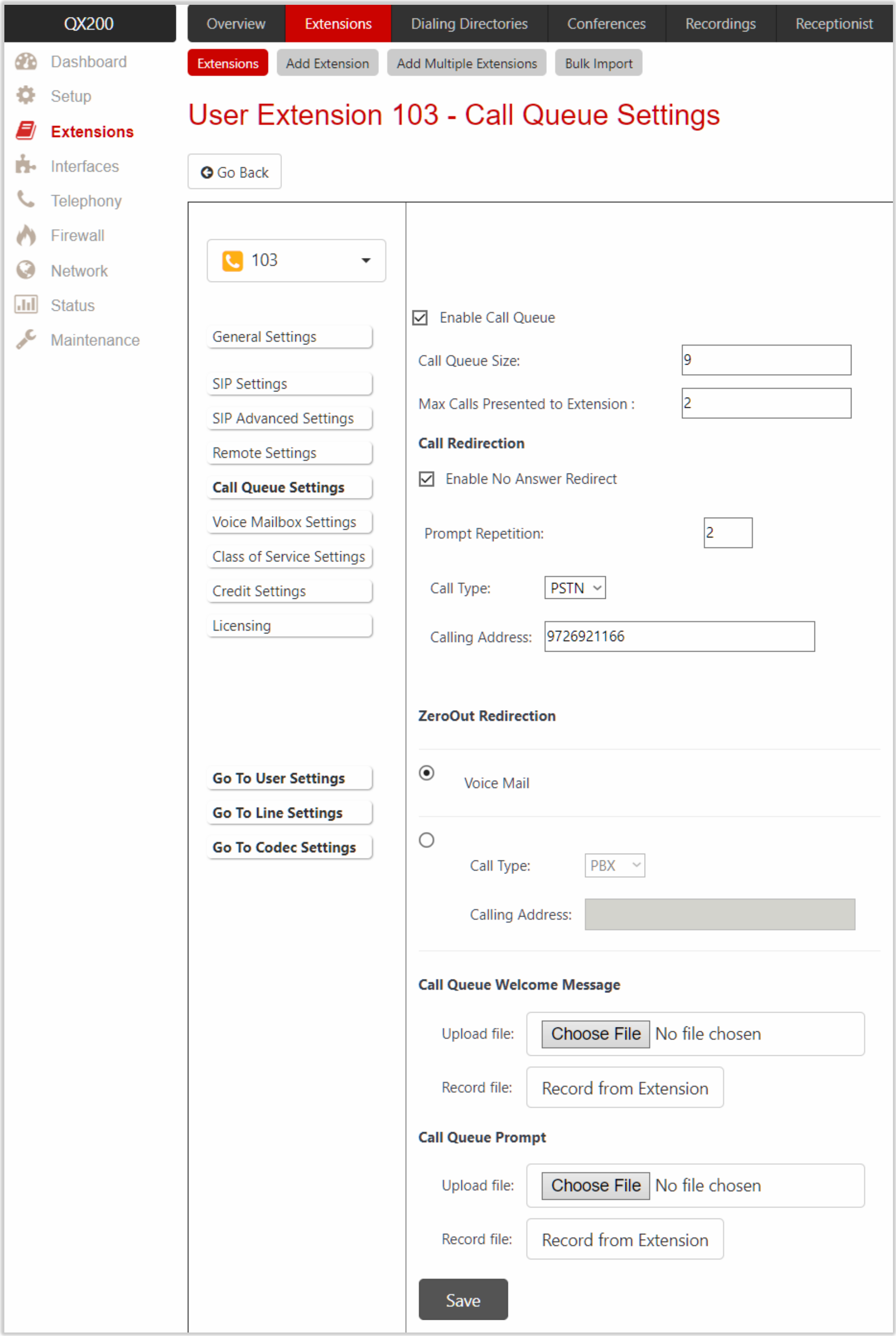


Figure 31: Call Queue Settings section

Voice Mailbox Settings

This section describes how to configure **Voice Mailbox Settings** on user extension. By default, the **Voice Mail** service is active for all user extensions and a certain percentage of memory space is assigned.

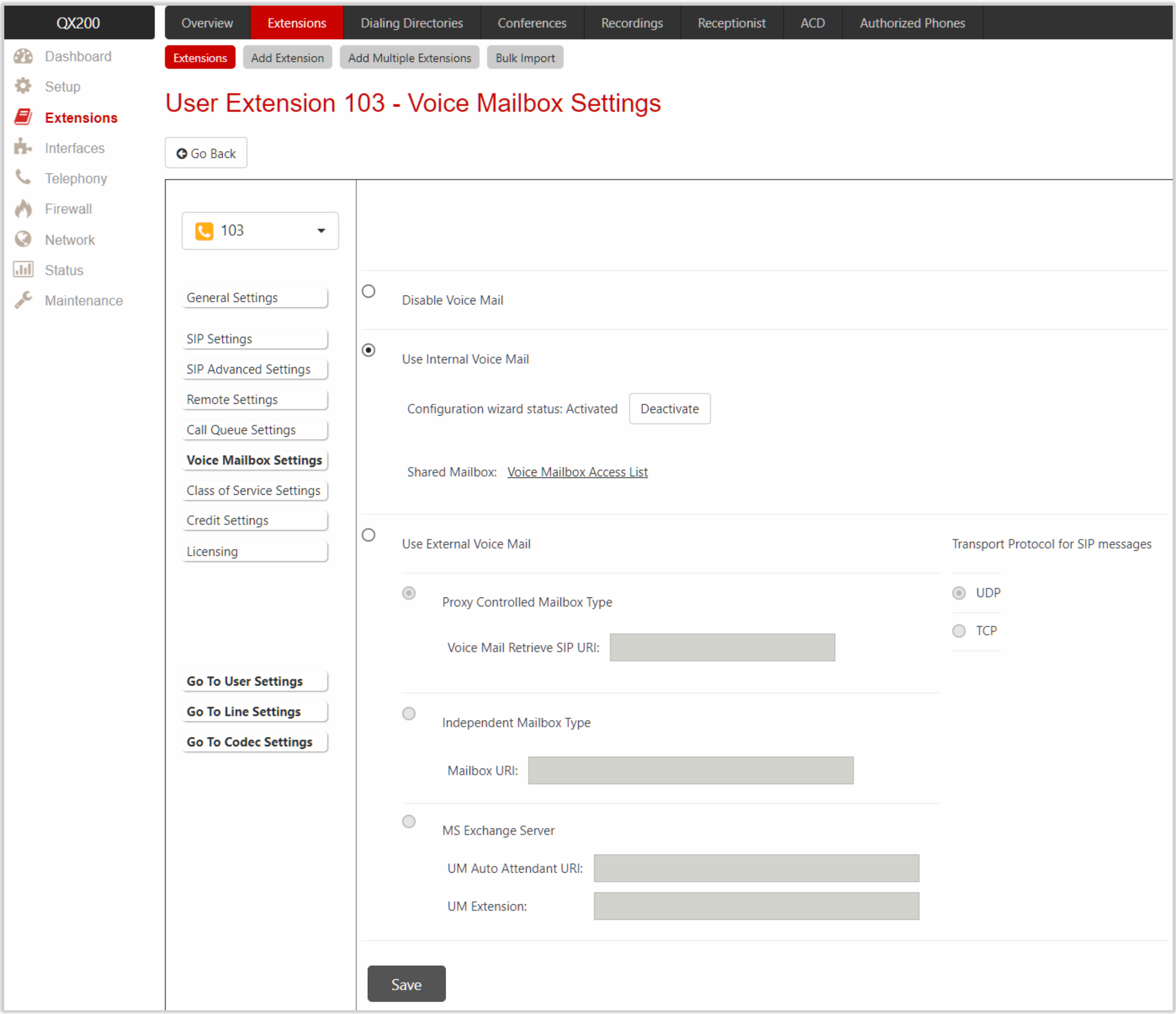


Figure 32: Voice Mailbox Settings section

The following settings (options) are available:

- **Disable Voice Mail** is used to disable the **Voice Mail** service denying caller to leave a voice message. User will still be able to access his **Voice Mailbox** and manage the existing messages as well as setup the personal settings (password, voice mail greeting and so on) from the handset.
- **Use Internal Voice Mail** is used to enable the **Voice Mail** service and set QX internal storage as a location for voice messages.
  - **Voice Mail Configuration Wizard** – if activated, prompts user to configure personal settings while entering the **Voice Mailbox** first time. Click **Deactivate** to stop the **Voice Mail Configuration Wizard**.
  - **Shared Mailbox** is used to setup **Shared Voice Mailbox** service. The **Voice Mailbox Access List** link leads to the **Voice Mailbox Access List of User Extension** page to define a list of extensions that are capable to access **Voice Mailbox** without password authentication.
- **Use External Voice Mail** is used to enable the **Voice Mail** service and set external storage as a location for the voice messages.

- Proxy Controlled Mailbox Type keeps the recorded voice messages on the SIP Proxy Server. When user accesses mailbox by dialing *0, the call will be redirected to the Voice Mailbox on the proxy server. It is recommended to select the Proxy Controlled Mailbox Type option if the Voice Mail Server is combined with the SIP Proxy Server.
- Proxy Controlled Mailbox Type redirects the recorded voice messages to the preconfigured Voice Mail Server. When user accesses the mailbox by dialing *0, the call will be redirected to the remote Voice Mail Server. It is recommended to select the Independent Mailbox Type option if the Voice Mail Server acts as a standalone location for the voice mails. TIP: It is required to set the SIP URI of Voice Mail Server where voice mails of the current extension will be collected for both options described above.
- Transport Protocol for SIP messages is used to select the transport protocol (UDP or TCP) for the transmission of SIP messages.
- MS Exchange Server keeps recorded voice messages into one universal inbox.
  - UM Auto Attendant URI is used to set the SIP URI of MS Exchange Server. When user accesses mailbox by dialing *0, the call will be redirected to the Voice Mailbox on MS Exchange Server.
  - UM Extension is used to enter the extension number that Unified Messaging will use when voice messages are submitted to user Voice Mailbox.

Note:

- For more information on how to configure and use MS Exchange Server, refer to the Configuring MS Exchange Server as External VM Server for QX IP PBX guide.
- Some internal Voice Mail services will become unavailable while choosing the Use External Voice Mail option. Instead, services of external Voice Mail service will become available to user. Consult with the external Voice Mail service administrator before enabling this option.

Class of Service Settings

This section describes how to assign the defined classes to extensions. The **Class of Service** specifies **user** or **conference** extensions that can use specific call routing rules to make a call. Extension not assigned to a certain class of service can't use a routing rule with **Class of Service** enabled.

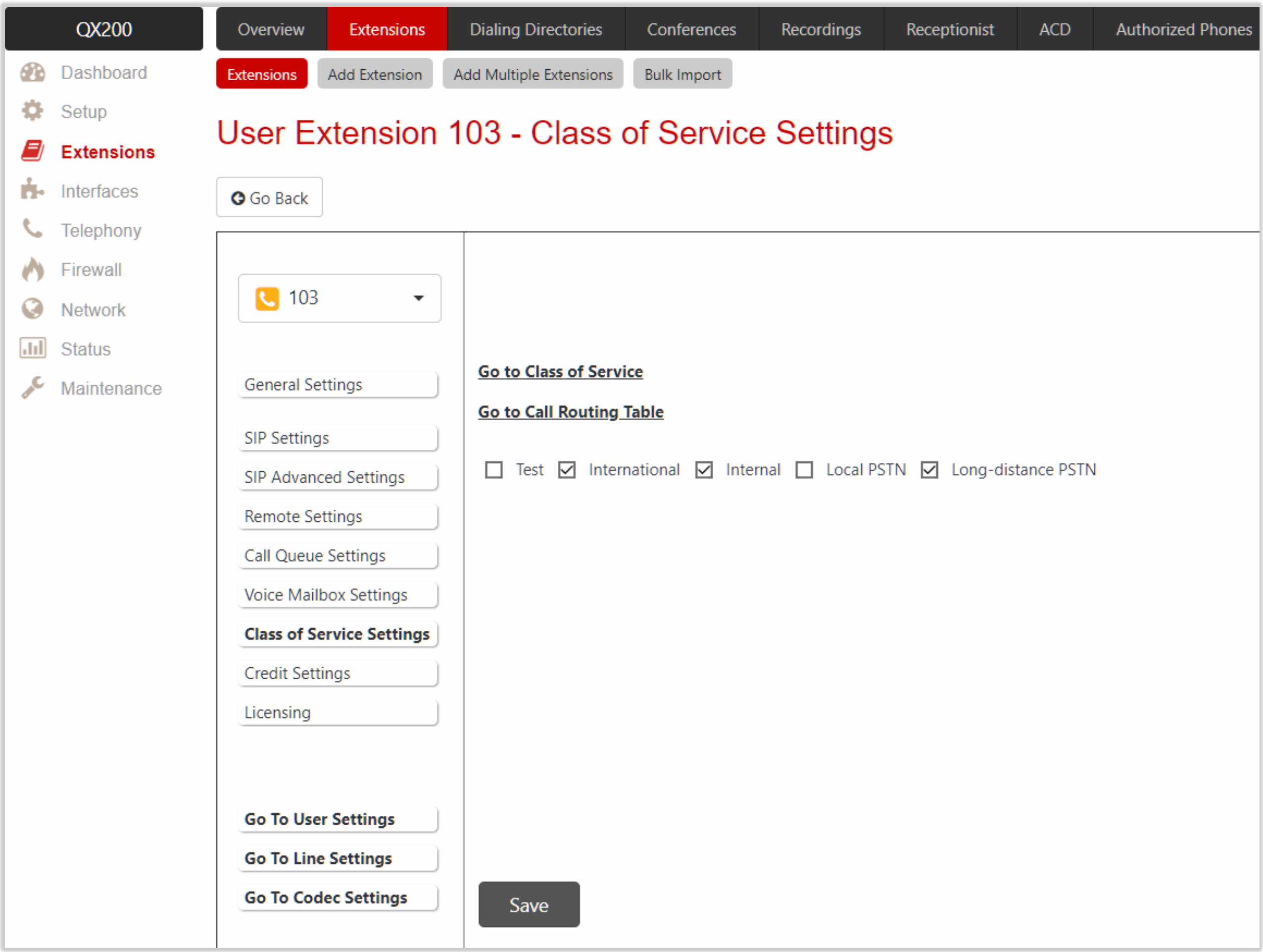


Figure 33: Class of Service Settings section

Note: **User** and **Conference** extensions can be attached to several **Class of Services** at the same time.

Credit Settings

The **Calling Cost Control** service allows to assign and manage credits to each extension for making calls. The assigned credit would be used and controlled when making a call through specific ("**payable**") call routing rules. Extensions not having credit can't use the routing rules with Calling Rate Settings enabled. **Credit Settings** is used to set the credit amount for the extension.

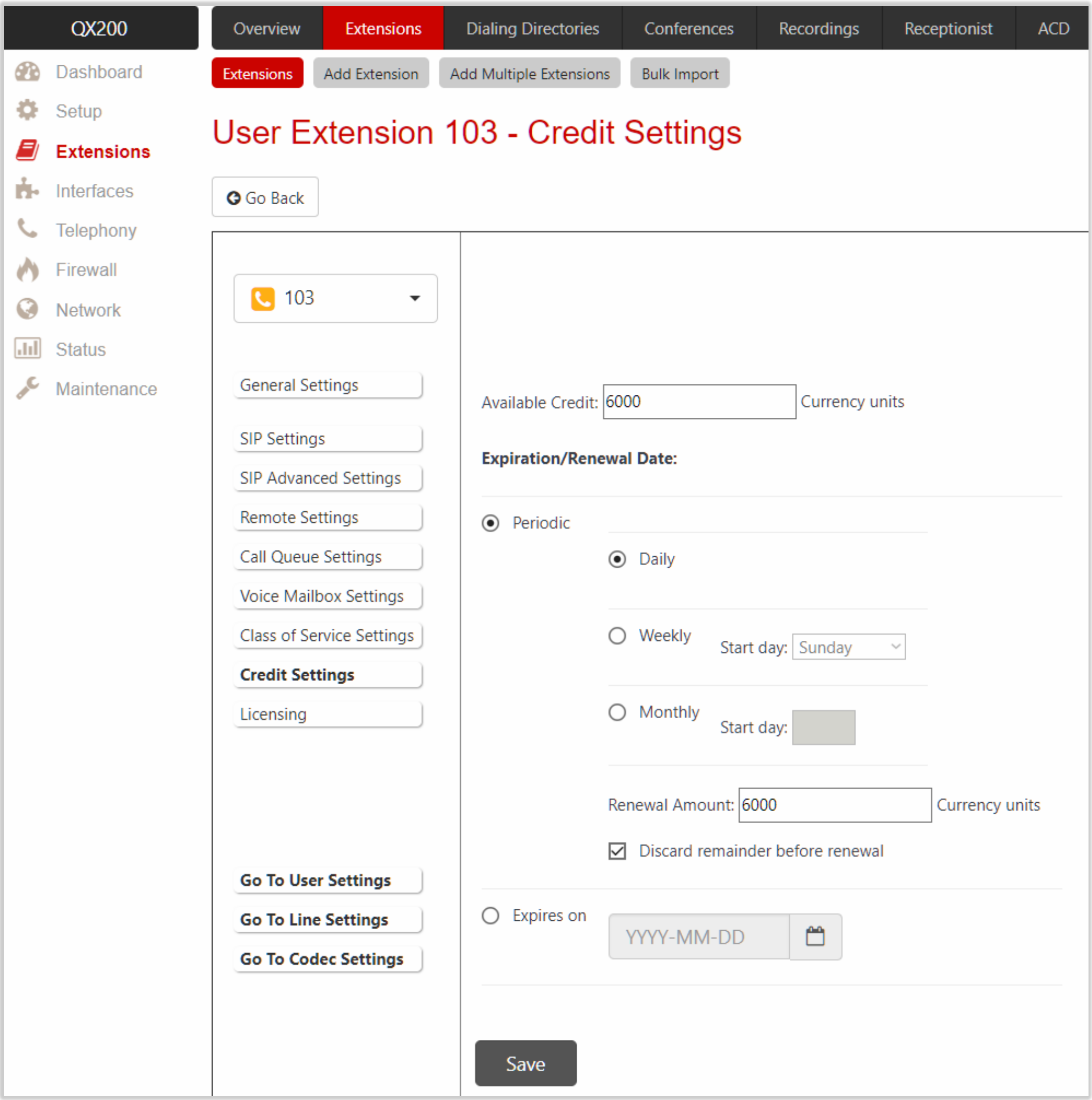


Figure 34: Credit Settings section

The following settings (options) are available:

- **Available Credit** is used to set the credit that can be used by extension. Once the **Available Credit** expires, the call will be disconnected without a prior notice. Placing a new call through routing rule(s) with **Call Rate Settings** option enabled is not possible until the **Available Credit** is updated (either manually or automatically by the renewal date and amount).
- **Periodic** is used to select one of the **Renewal Date** options:
  - **Daily** – the defined **Available Credit** will be renewed every day.
  - **Weekly** – the defined **Available Credit** will be renewed every week on a specified weekday.
  - **Monthly** – the defined **Available Credit** will be renewed every month on a specified day.
  - **Renewal Amount** is used to set the renewal amount to be added to **Available Credit** when the expiration date of **Available Credit** is reached. Leave the field blank, if you don't need to renew **Available Credit**.

- ➢ **Discard remainder before renewal** is used to discard the remainder of **Available Credit** before renewal and set **Renewal Amount** as a new **Available Credit**.
- **Expires on** is used to manually set the expiration date for **Available Credit**. After **Expiration Date**, extension will not be able to make a new call through call routing rule(s) with the **Call Rate Settings** option enabled.

Licensing

The **Licensing** section becomes available only if the corresponding licenses are activated.

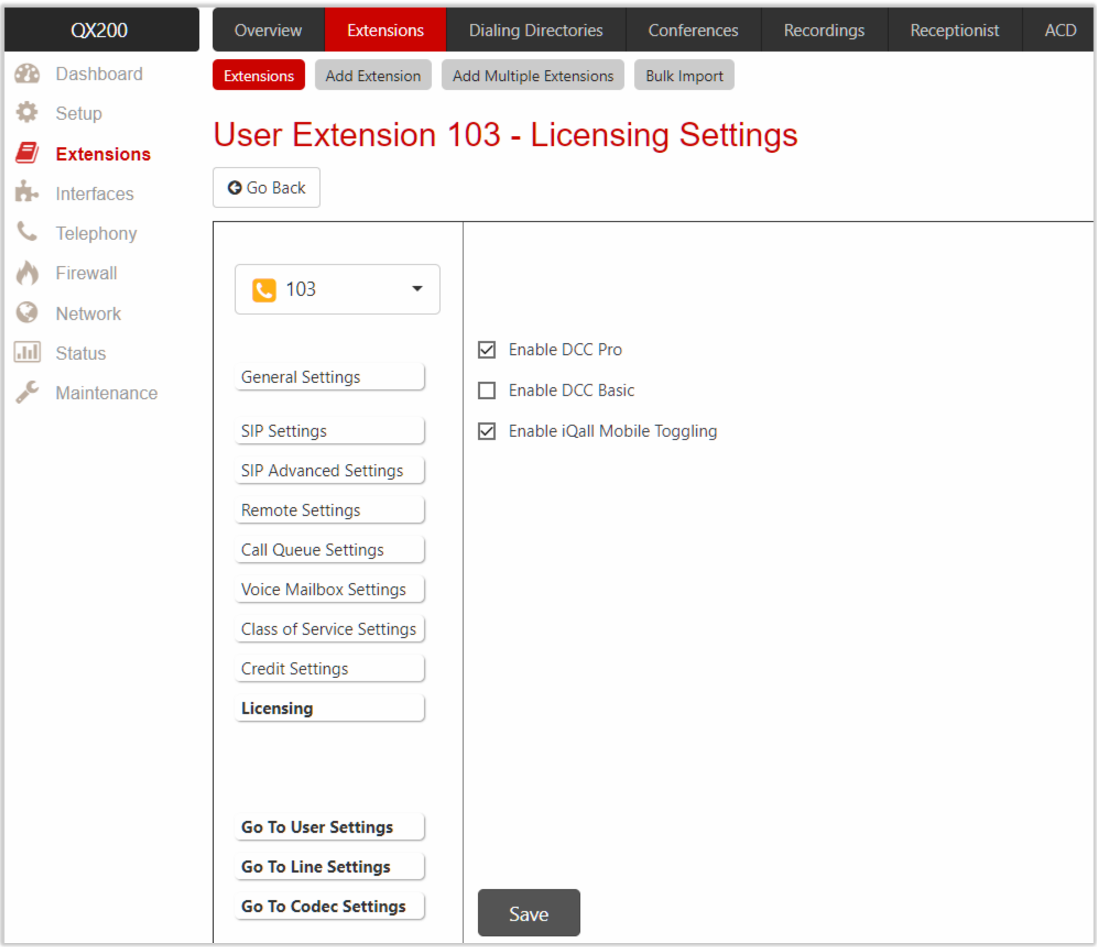


Figure 35: Licensing section

The following settings (options) are available:

- **Enable DCC Pro** allows to set the extension to be used by the **DCC Pro** application. **TIP:** DCC Pro/Basic licenses can't be activated simultaneously for the same extension.
- **Enable DCC Basic** allows to set the extension to be used by the **DCC Basic** application.
- **Enable iQall Mobile Toggling** allows to allocate the **iQall Mobile Toggling** license to the extension.

For more information on how to configure and use these features, refer to the Licensable Features on QX IP PBXs guide.

Parent-Child Configuration

The **Parent-Child** configuration allows to assign a number of extensions (phones) to the certain **Parent** extension as a **Child**. Phone(s) configured as a **Child** will make outbound calls on behalf of an extension configured as a **Parent**. **Child** extension(s) will ring simultaneously in case of inbound call to the **Parent**.

The Parent-Child configuration can be used in specific cases, to create the appearance that many phones are connected to the same extension. This feature can be used, for example, with Epygi Hotel Console (EHC) feature for hotel rooms having many phones or with other applications where many phones are linked to the same extension.

In case of outbound calls, Child extensions are not visible for called destinations. When placing an outbound call from the Child phone the Caller ID and the name of Parent extension would appear at the destination.

In case of inbound calls to the Parent, all phones configured as a Child will ring simultaneously with the Parent. The Parent or any of Child phones can answer the call.

Note:

- Child extension(s) will lose the SIP registration, the configured Basic and Caller ID Services.
- Child extension(s) will not be able to receive incoming calls directly and will ring only when the Parent extension is dialed.

6.1.6 Pickup Group

The Call Pickup service allows to pick up calls ringing on a certain group of extensions by dialing Pickup Group extension number.

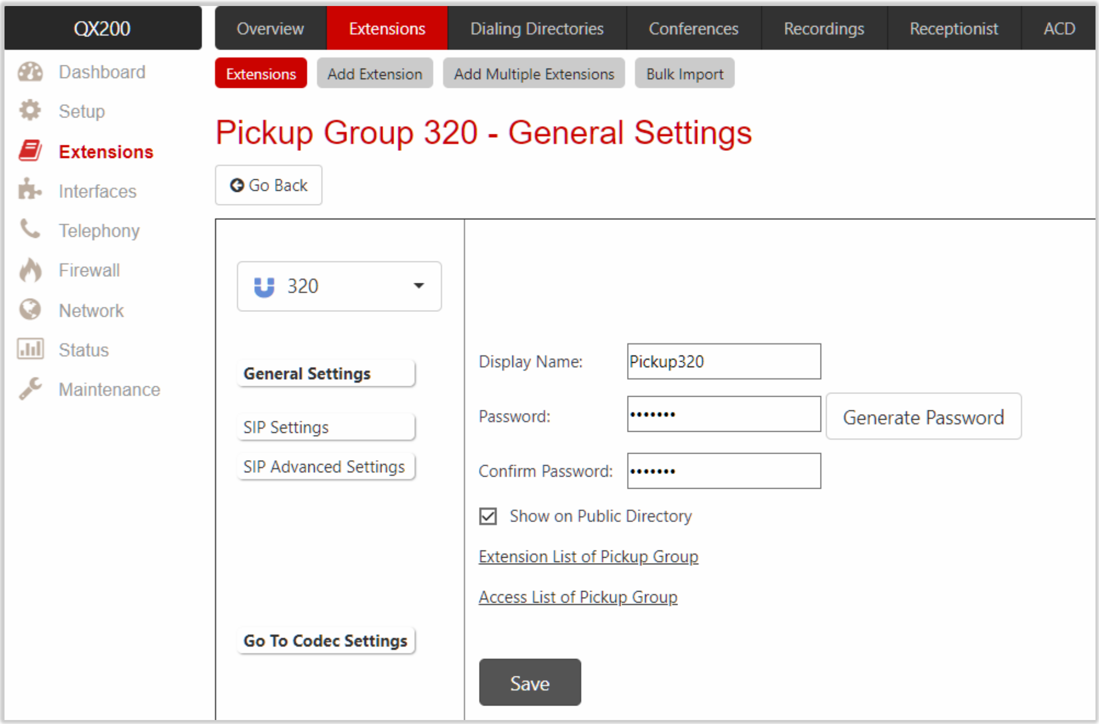


Figure 36: Pickup Group – General Settings section

To configure Pickup Group extension:

1. Click the Extension List of Pickup Group link.
2. Select the extension(s) and click Enable. Calls to these extensions can be picked up.
3. Go Back and click the Access List of Pickup Group link.
4. Click Add and enter the extension(s) allowed or denied permission to pick up the ringing calls.
5. Click Save to add the new entry to the Access List of Pickup Group table.

How it works: When call is ringing on another phone, you can pick up that call on your own phone by dialing the number of the Pickup Group extension.

Note:

- The General Settings, SIP Settings, SIP Advanced Settings and Go To Codec Settings sections are the same as for user extension.
- When a caller not listed in the Access List calls the Pickup Group extension, password authorization (the password of the Pickup Group extension) will be required to allow the call pickup.
- If a user dials the Pickup Group extension when several extensions of the Pickup Group are ringing, the first (oldest in time) call will be picked up.

6.1.7 Call Park

The Call Park service allows to park a call, (the call will be automatically placed on hold) then retrieve the parked call from another phone by dialing the Call Park extension number.

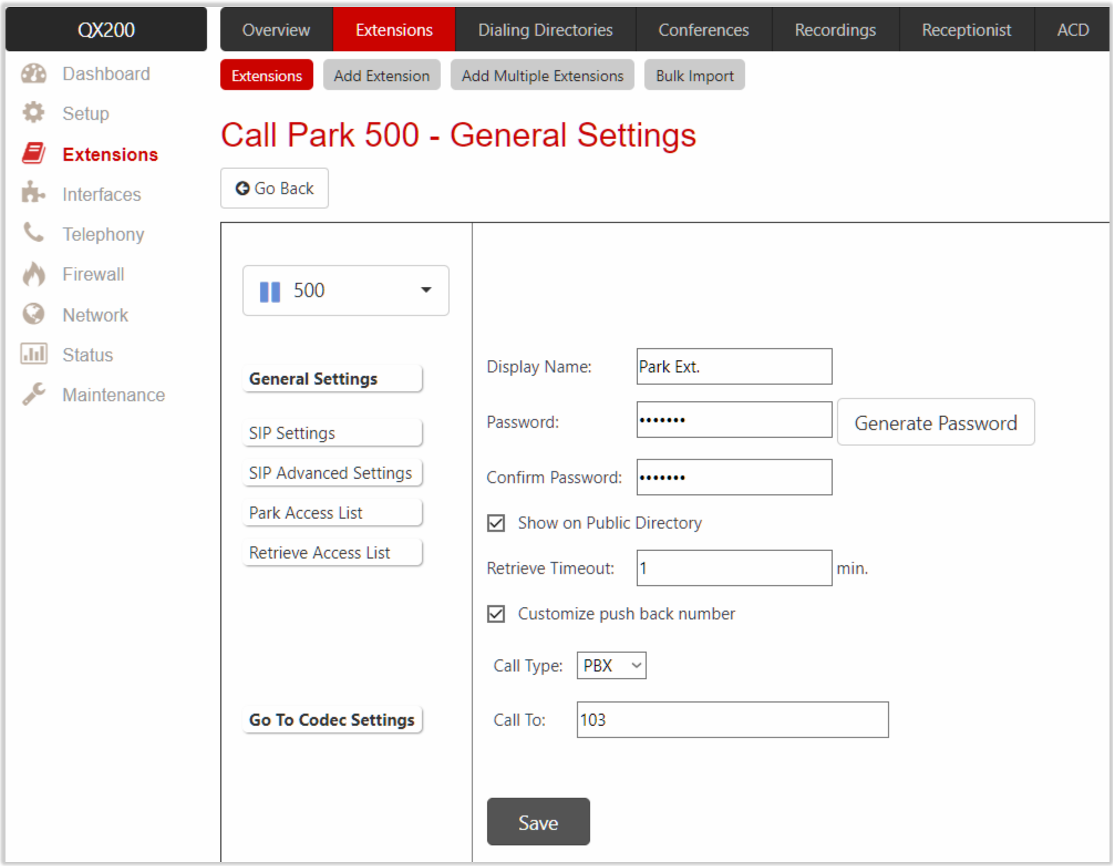


Figure 37: Call Park – General Settings section

The following settings (options) are available:

- Retrieve Timeout is used to set the timeout during which the call will stay in Call Park, i.e. the parked user will remain on-hold.
- Customize push back number – if selected, after the call park retrieve timeout expires, the hold music will stop playing to the parked party (user) and a new call will be placed towards the push back number configured in the Customize push back number field. TIP: If the Customize push back number option is not selected, then after the call park retrieve timeout expires, the call will be forwarded back to the extension which parked the call.

To configure Call Park extension:

1. Click the Park Access List link, then click Add.
2. Enter the extension number(s). These extensions will be able to park calls on the Call Park extension.
3. Click Save to add the new entry to the Park Access List table.
4. Click the Retrieve Access List link, then click Add.
5. Enter the extension number(s) allowed to retrieve the parked calls from the current Call Park extension.
6. Click Save to add the new entry to the Retrieve Access List table.

How it works: To park a call, put the active call on hold and either dial *5 or the Call Park extension number. The call will be parked. To retrieve the call, dial the Call Park extension number.

Note:

- The General Settings, SIP Settings, SIP Advanced Settings and Go To Codec Settings sections are the same as for user extension.
- Any extension missing from the **Park Access List** won't be able to park a call to the current call park extension.
- When a caller not listed in the **Retrieve Access List**, calls the **Call Park** extension, password authorization (the password of the **Call Park** extension) will be required to allow retrieving the parked call.
- For more information on how to park/retrieve calls on Epygi supported IP phones, refer to the QX IP PBX Features on Epygi Supported IP Phones guide.

6.1.8 Paging Group

The **Call Paging** service is used to page a group of extensions (phones) by forcing extensions to go off-hook and opening a one-way communication. The service is particularly used for announcements addressed to a group of extensions. This service allows to page multiple extensions by dialing the **Paging Group** extension.

To configure **Paging Group** extension:

1. Click the **Extension List of Paging Group** link.
2. Select the extension(s) and click **Enable** to page these extensions.
3. Go **Back** and click the **Access List of Paging Group** link.
4. Click **Add** and enter the extension(s). These extensions will be allowed/denied to dial **Paging Group** extension.
5. Click **Save** to add the new entry to the **Access List of Paging Group** table.

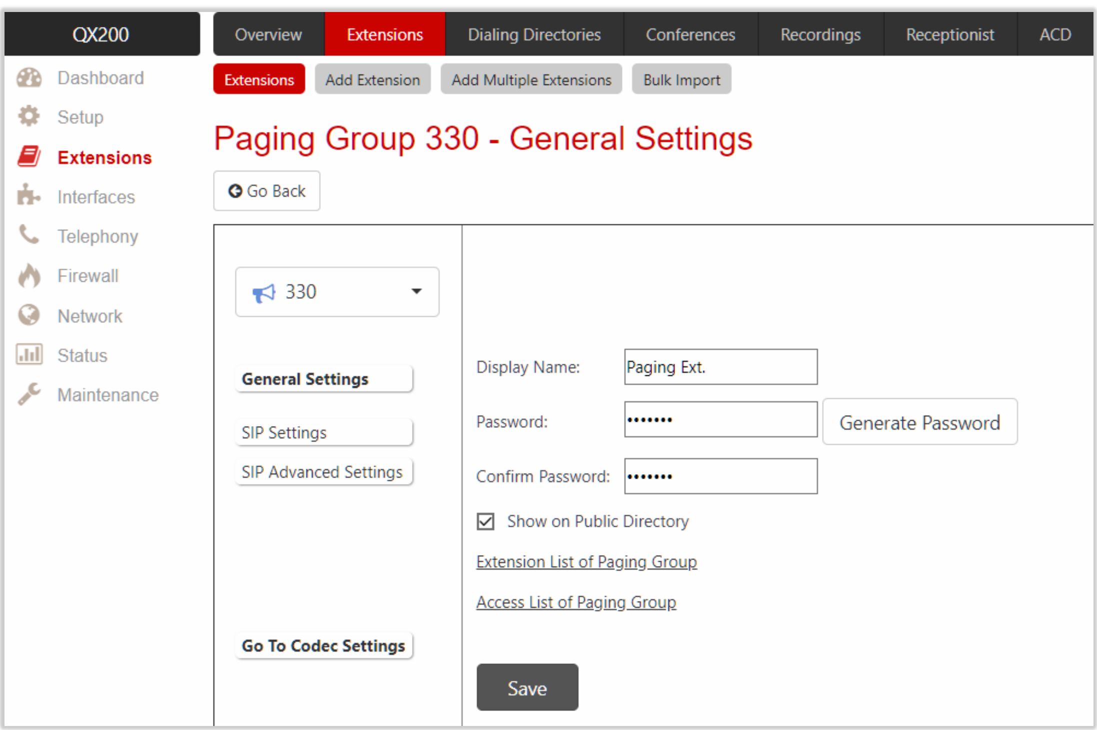


Figure 38: Paging Group – General Settings section

How it works: When calling to the **Paging Group** extension, the call will be forwarded to the extensions listed in the **Paging Group** table. The phones of the called extensions will automatically go off-hook (the phone speaker automatically becomes activated) and the caller will be able to make announcement. Since the paging call opens one-way communication, the called extensions will not be able to give an answer to the caller. To terminate the paging call, caller should simply hang up.

Note:

- The SIP Settings, SIP Advanced Settings and Go To Codec Settings sections are the same as for user extension.
- When a caller (not listed in the **Access List of Paging Group** table) calls the **Paging Group** extension, password authorization (the password of the **Paging Group** extension) will be required to start the call paging.
- **Paging** will not work if the called phone is in call.
- Make sure the called phones support automatic off-hook. For more information how to use **Paging** service on Epygi supported IP phones, refer to the QX IP PBX Features on Epygi Supported IP Phones guide.

6.1.9 Recording Box

Recorded calls on QX can either be stored locally in the **Recording Box** or transferred to the remote server. Recording Box is used to store the recorded calls locally. Users can access the Recording Box either from WEB GUI or from handset by calling the corresponding Recording Box extension. In both cases, the user can play and delete the recordings.

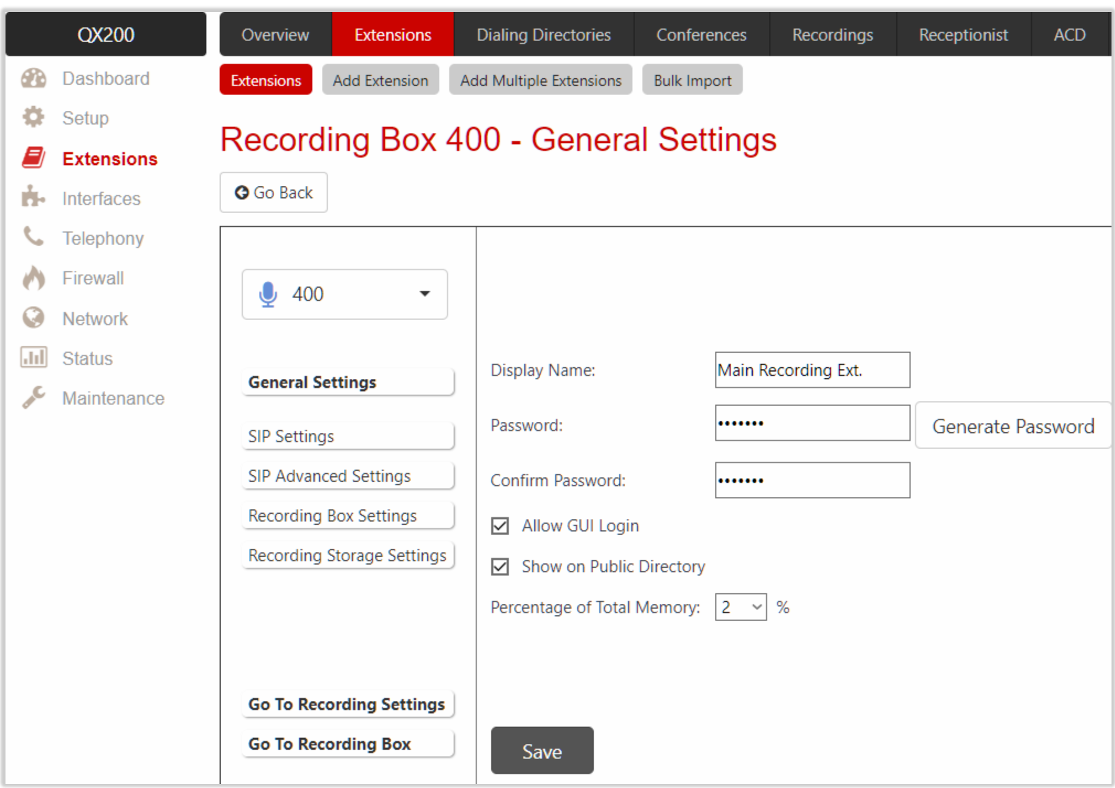


Figure 39: General Settings section

Recording Box Settings

This section describes how to configure specific settings of Recording Box. The following settings (options) are available:

- Ask Password on Local Access protects local access to the Recording Box. If selected, the Recording Box password will be required to access the Recording Box locally.
- Ask Password on Remote Access protects remote access to the Recording Box. If selected, the Recording Box password will be required to access the Recording Box remotely.
- Play Welcome Message enables the welcome message that is played when accessing the Recording Box.
- Maximum Recordings refers to the maximum number of recordings allowed to be stored in the Recording Box. If this number is reached, some of call recordings should be deleted from the Recording Box, to free up space for new recordings.
- Single Recording Duration refers to the maximum recording duration for a single call. When the recording duration expires, recording will be stopped while the call will stay active.
- Forward/Rewind Duration is used to select the timeout in seconds to shift the recording playback from the handset.
- Play announcement when starting recording is used to play an announcement before starting the recording. The call recording will start without notification if this option is disabled.

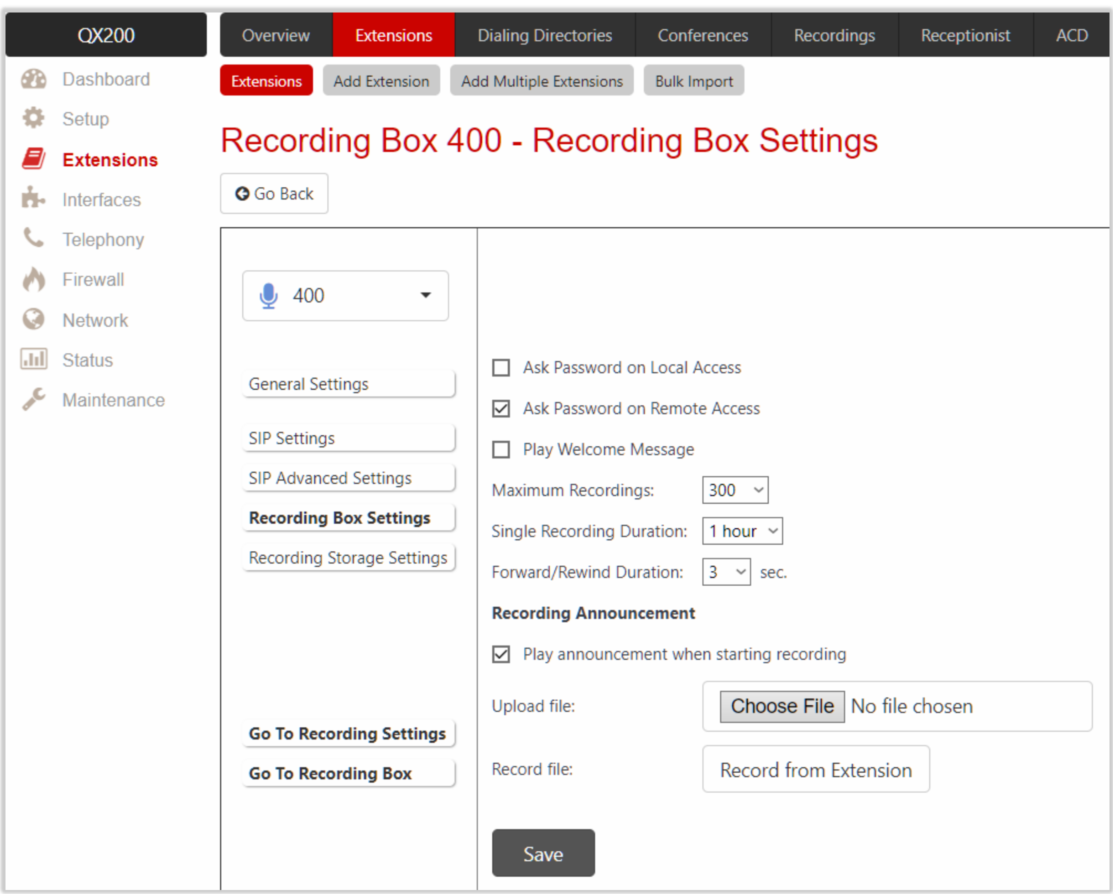


Figure 40: Recording Box Settings section

Recording Storage Settings

This section describes how to configure the recording storage settings. The following settings (options) are available:

- Recording Storage Modes offers the following recording storage options:
  - FTP sends recordings directly to FTP server.
  - Simple Local keeps recordings locally. When local space is full or maximum recording count is reached, stop recording calls and generate an event.
  - Cyclic Local keeps recordings locally. When local space is full or a maximum recording count is reached, delete oldest recordings to free up some space.
  - Mixed keeps recordings to FTP server and keeps a copy in local storage. When local space is full or a maximum recording count is reached, delete oldest recordings to free up some space.
- FTP Settings is used to set the FTP server parameters:
  - Server Name is used to set the IP address or hostname.
  - Server Port is used to set the port number.
  - Use SFTP enables SSH FTP (SFTP) support, which allows using secure FTP connection.
  - Username and Password are used to set the authentication parameters.
  - Directory on Server is used to set the location on the server where the recordings will be stored.
  - File Naming Scheme is used to set the naming scheme of the files to be uploaded to the FTP server. This scheme helps to distinguish files among others and to avoid possible overwriting of the files. This field may contain any distinctive text and also offers a list of variables:
    - call_guid – unique GUID of the call
    - recording_id – unique recording ID of the call
    - caller_dispname – caller's display name
    - caller_username – caller's username
    - caller_fullname – caller's full name in the username@host[:port] format
    - callee_dispname – called user's display name
    - callee_username – called user's username
    - callee_fullname – called user's full name in the username@host[:port] format
    - duration – duration of the call
    - time_hour – hour when the call recording started
    - time_min – minute when the call recording started
    - time_sec – second when the call recording started
    - date_year – year when the call recording started
    - date_month – month when the call recording started
    - date_day – day when the call recording started
    - extension – recording box extension
    - hostname – QX hostname
    - recording_id – unique recording ID of the call

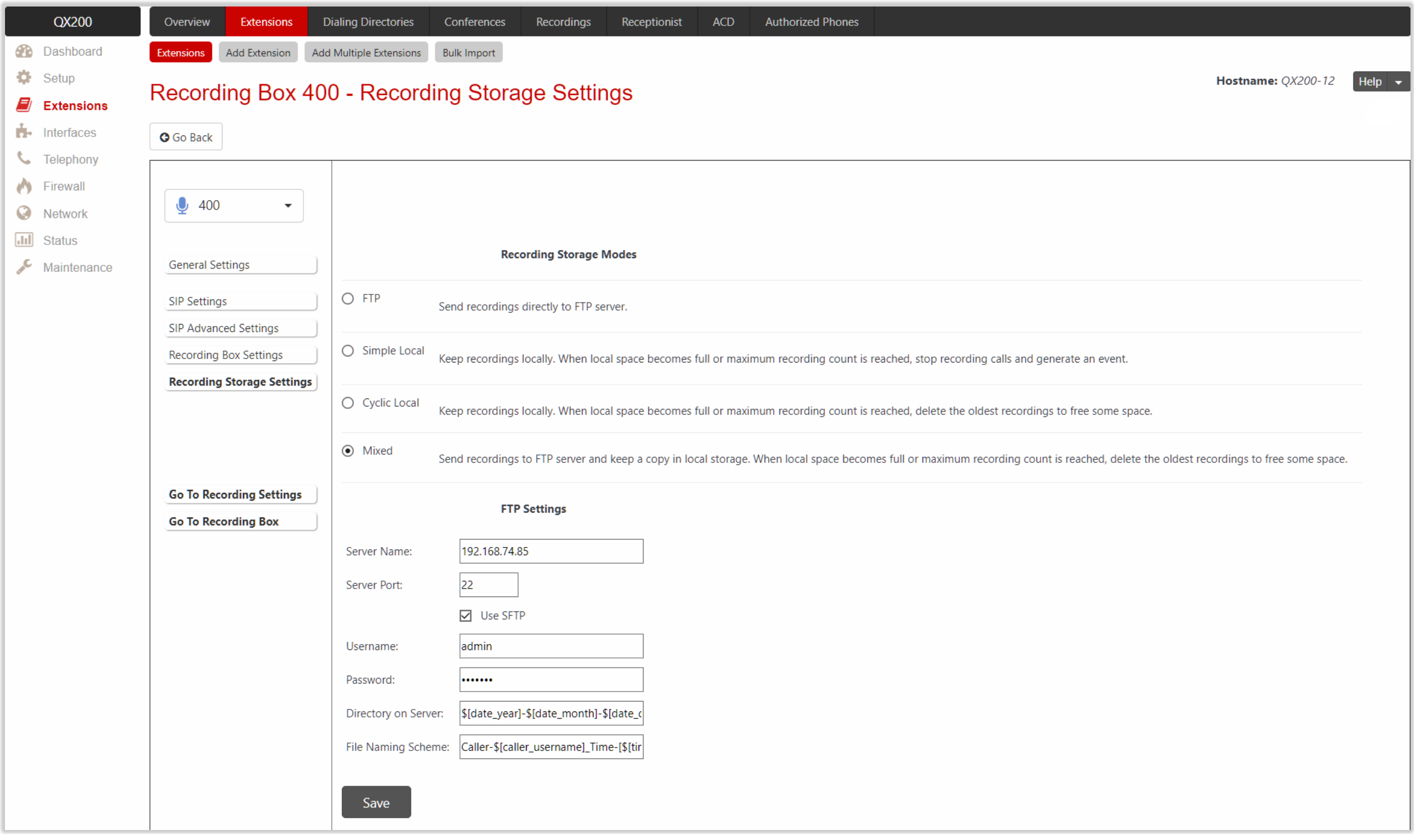


Figure 41: Recording Storage Settings section

Any combination of listed variables can be used in the **File Naming Scheme** field.

Example for a file naming scheme: MyQX-$[caller_dispname]-$[duration]-$[time_hour]-$[time_min] business.

If Andrew is the caller, call recording started at 14:10 and lasted 15 seconds then the files stored on the FTP server for this Recording Box will have the following name: MyQX-Andrew-15 sec-14-10-business.wav

Recording Box

Users can access the Recording Box either from WEB GUI or from handset by calling the Recording Box extension. In both cases, the user can play and delete the recorded calls in the Recording Box.

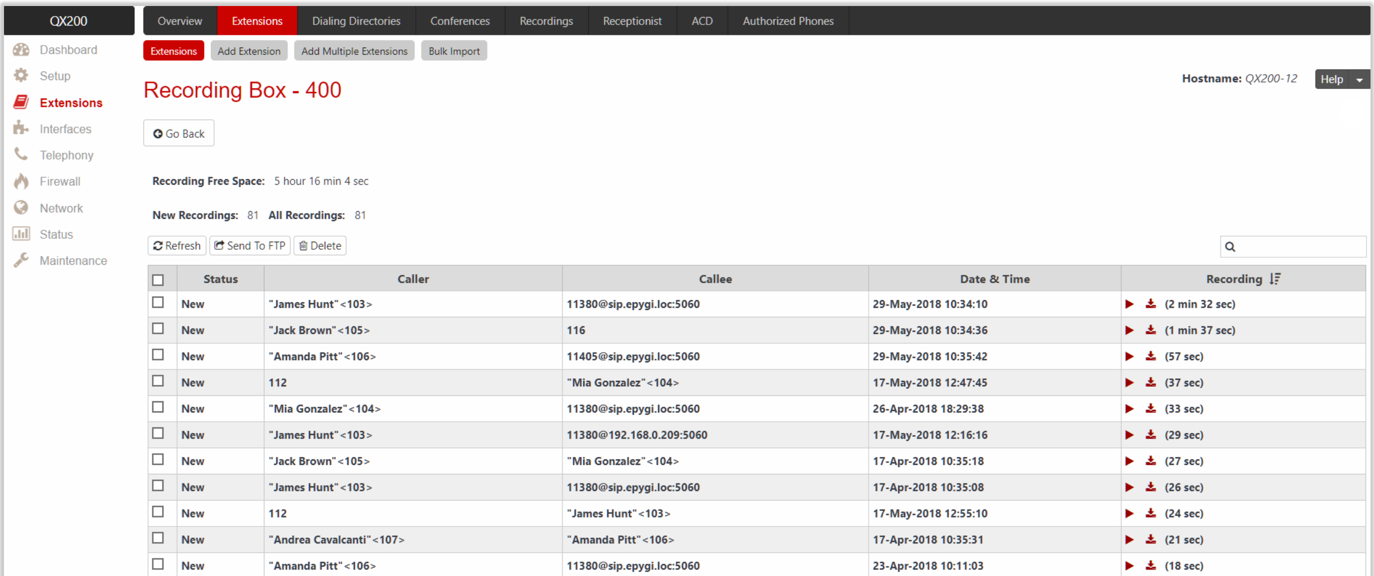


Figure 42: Recording Box page

Note:

- The General Settings, SIP Settings, SIP Advanced Settings and Go To Codec Settings sections are the same as for user extension.
- When using **Call Recording** on the QX50/QX200 it is advisable to use an SD memory card to expand the system memory.
- When using **Call Recording** on the QX20/QX500 it is advisable to use a micro SD memory card to expand the system memory.

6.1.10 Auto Attendant

Auto Attendant is an IVR system that replaces a receptionist and allows to distribute calls to QX extensions or services through audio prompts.

Note: The SIP Settings, SIP Advanced Settings and Go To Codec Settings sections are the same as for user extension.

General Settings

This section describes how to configure general settings of auto attendant.

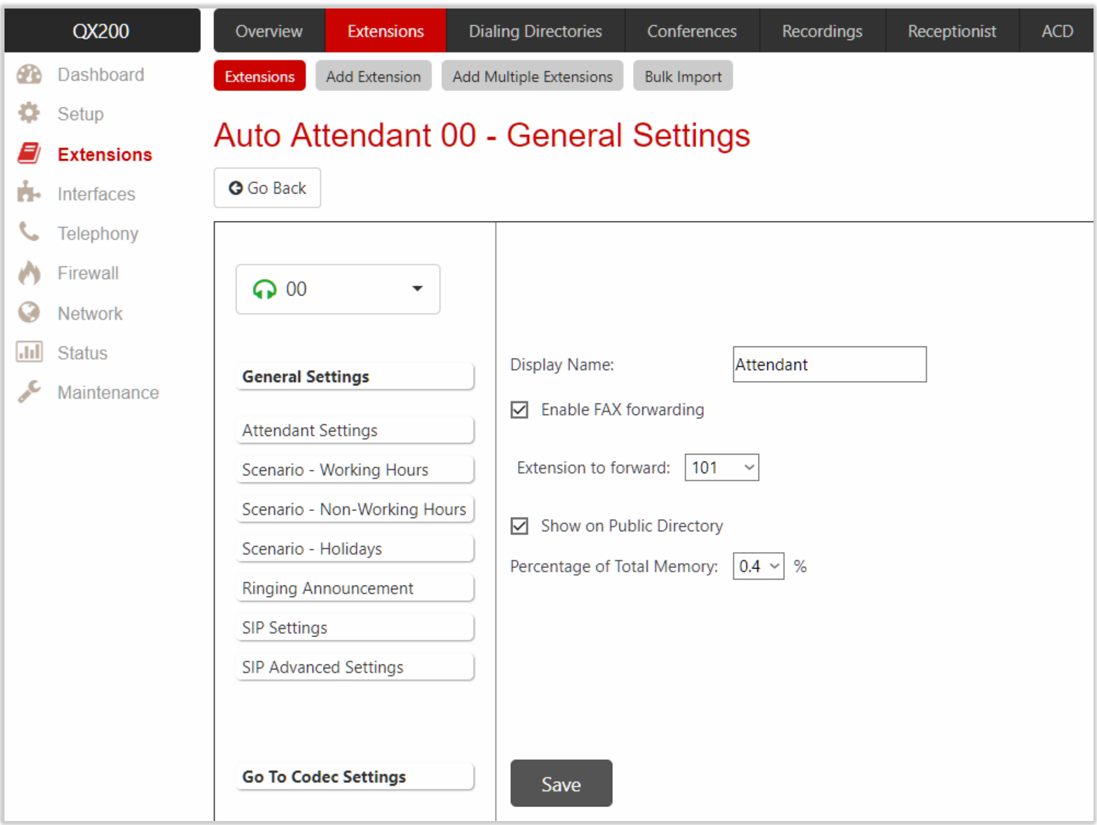


Figure 43: Auto Attendant – General Settings section

The following settings (options) are available:

- Display Name is the caller ID that will be displayed on the phone when making a call to/from auto attendant (e.g. when using Callback service).
- Enable FAX forwarding – if selected, the system forwards the FAX messages to the selected extension in case incoming calls are routed to the auto attendant and FAX tone is detected.
- Extension to forward is used to select the extension where the incoming FAX addressed to the auto attendant will be forwarded. The list contains only those extensions that have FAX support enabled. The FAX support can be enabled from the Extension Codecs page. TIP: FAX forwarding is applicable only for incoming calls from PSTN and SIP.
- Show on Public Directory – if selected, automatically includes the extension display name and number in the Phone Book (Directory) and Extension Directory.
- Percentage of Total Memory is used to allocate memory for custom messages.

Attendant Settings

This section describes how to apply schedules and manage the scenario(s) for the scheduled periods. The following settings (options) are available:

- **Enable Schedule** is used to select and apply preconfigured **schedule.** The applied schedule allows to configure different scenarios for scheduled periods (working hours, non-working hours and holidays).
- **Attendant Scenario** is used to select the scenario. The following scenarios are available:
  - Standard scenario is available and active on the **00** auto attendant and for newly created auto attendants by default.
  - VXML scenario allows to upload a custom scenario file in **VXML** format.
  - Custom scenario allows to configure the custom scenario with the embedded scenario builder.
  - ACD scenario allows to activate a special scenario for **ACD** agents.

Note: Enable the **Schedule** option and apply a schedule to the auto attendant, to be able to select a scenario for each period.

- **Authorized Phones** leads to the Authorized Phones page. The trusted user (external SIP or PSTN caller) will be able to use QX services after calling the auto attendant, as if a user extension. If the **Callback** service is activated the trusted user will get a call back from auto attendant.

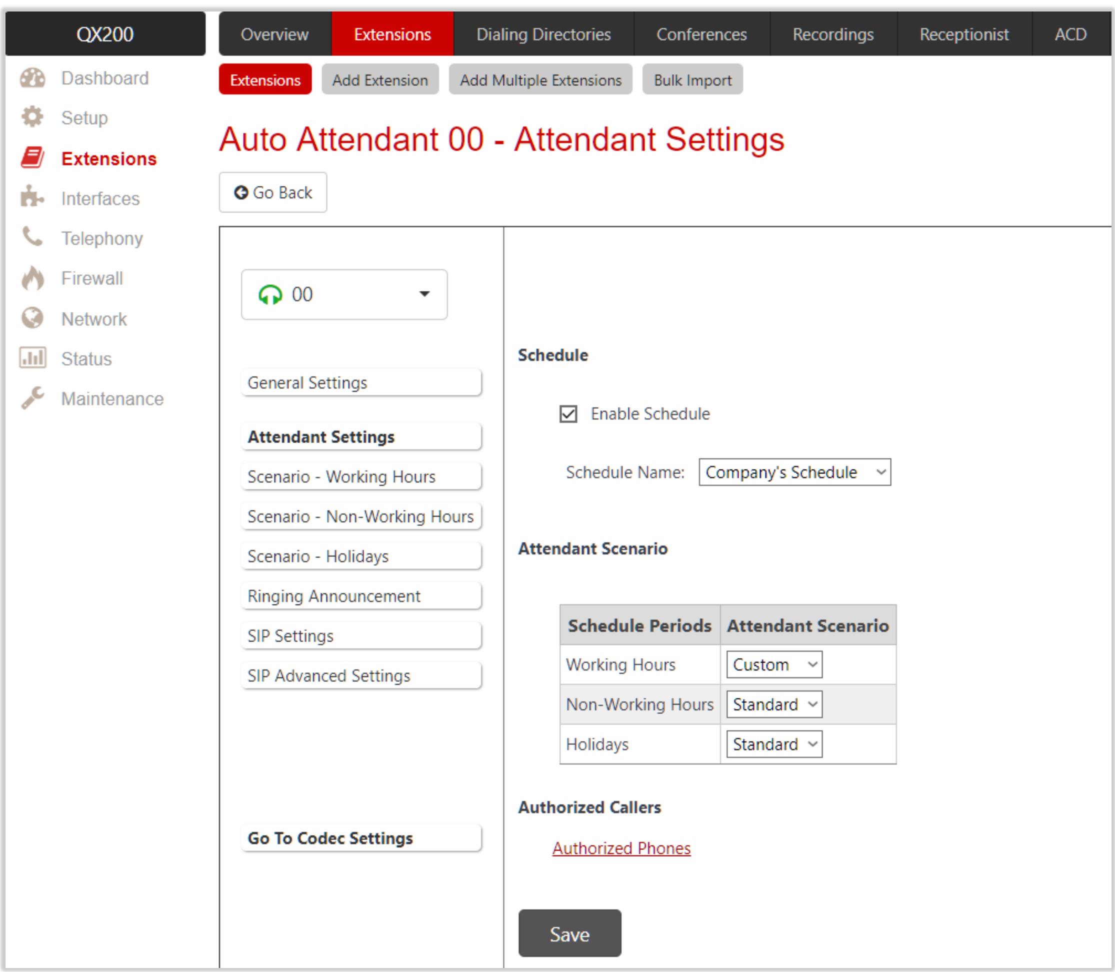


Figure 44: Attendant Settings section

Attendant Scenario

This section is used to configure the selected scenario.

Standard scenario

The following settings (options) will be available by selecting the Standard scenario:

- Pass Dialed Digits through Call Routing – if selected, sends the dialed numbers to the Call Routing Table.
- Enable No Input Redirect – if activated and configured, callers will be redirected to the specified address in case if no action by caller on the recurring prompt. The Prompt Repetition is used to set the number of prompts to be played before redirection.
- Enable ZeroOut Redirect – if activated and configured, callers dialing 0 during welcome message or recurring prompt, will be redirected to the specified address. Note: The routing patterns in the Call Routing Table starting with digit 0 will not work for incoming calls to auto attendant if both the ZeroOut and Pass Dialed Digits through Call Routing options are enabled. The ZeroOut option has a higher priority. If enabled, the system will redirect calls to the specified destination. As a result, calls prefixed with 0 will never reach call routing.
- Welcome Message allows to enable and customize auto attendant welcome message.
- Recurring Prompt allows to customize the auto attendant recurring prompt (played after the welcome message and then periodically repeated while being in the auto attendant).

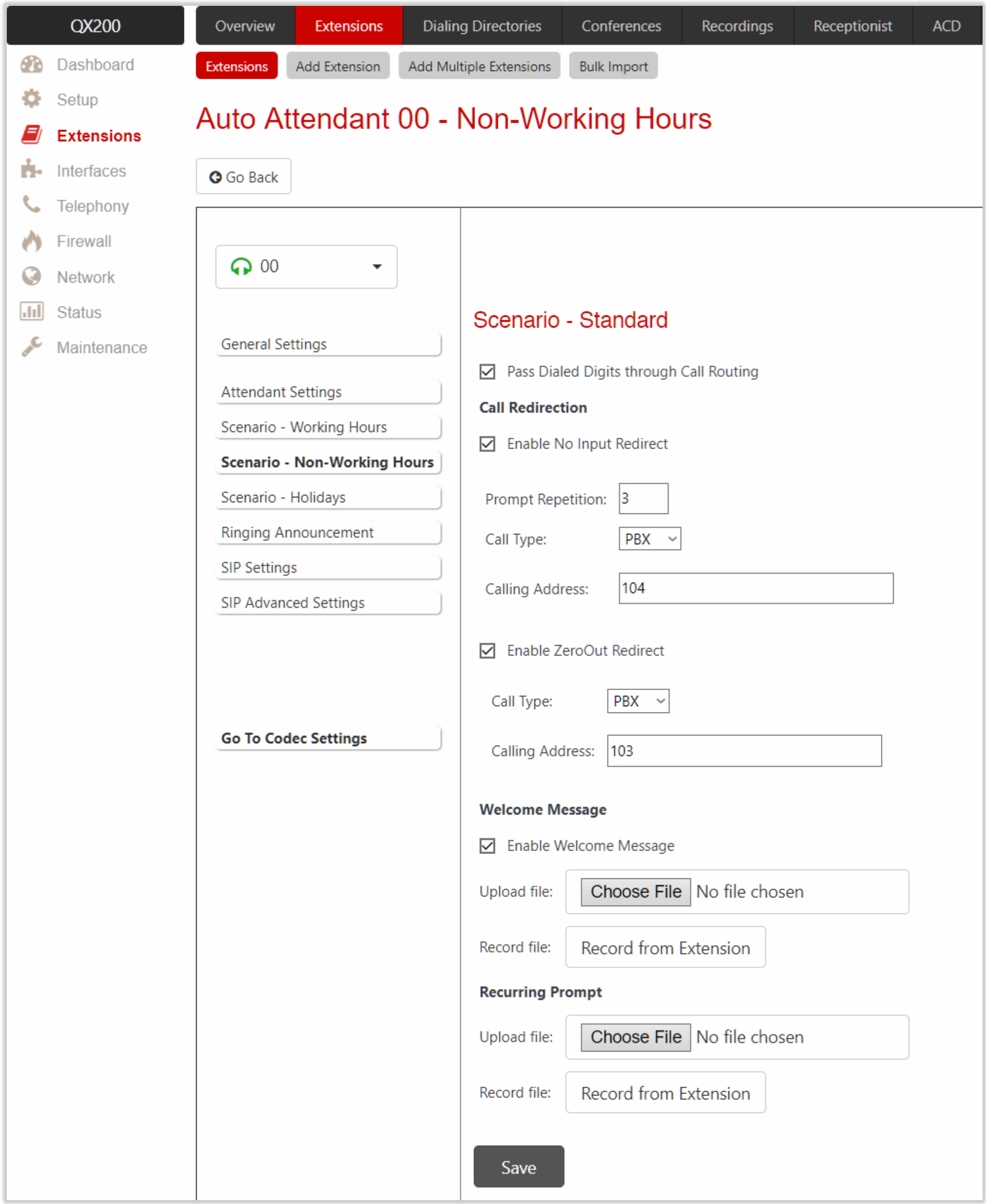


Figure 45: Auto Attendant – Standard scenario

VXML scenario

The **VXML** scenario allows to upload a custom scenario file and voice messages.

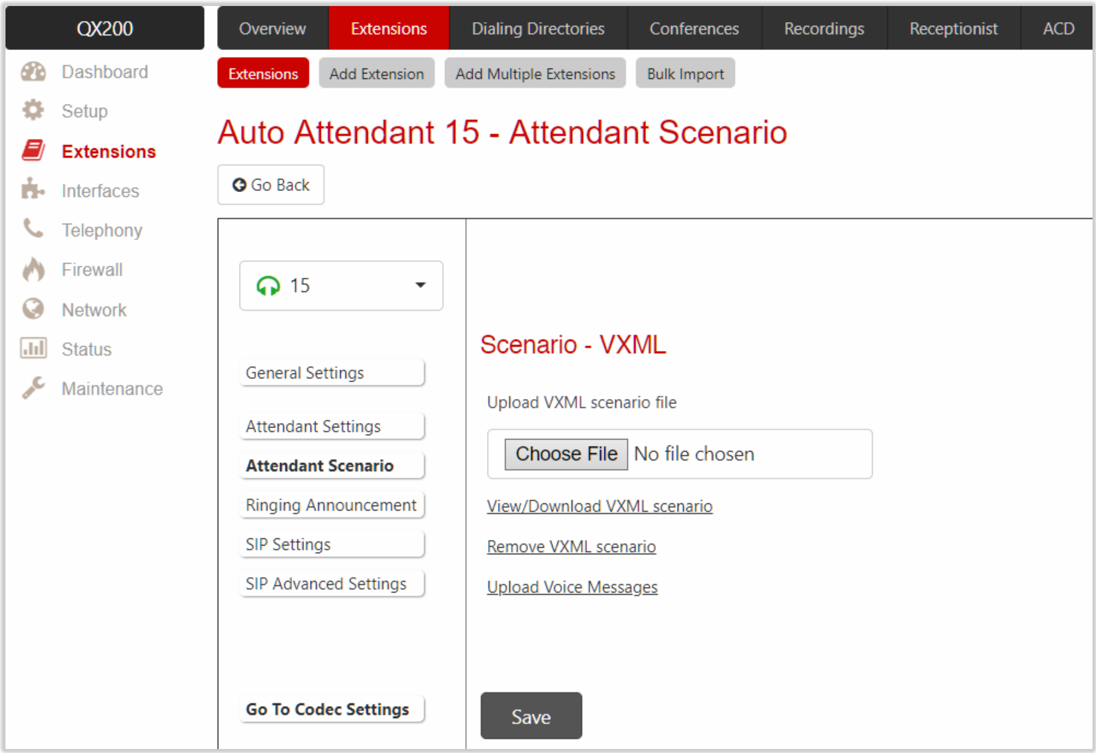


Figure 46: Auto Attendant – VXML scenario

To upload **VXML** scenario and voice messages:

1. Click **Choose File** to browse and select the **VXML** scenario file. **TIP:** The uploaded file needs to be in EpygiXML format and is restricted to **20** KB file size.
2. Click the **Upload Voice Messages** link.
3. Click **Choose File** to browse and upload the voice messages. **TIP:** To upload all voice messages at once, create an archive file of the (***.tar.gz**) type containing all the necessary files and upload it.

Custom scenario

The **Custom** scenario allows to use the embedded scenario builder. The following components are available:

- **Create scenario** leads to the **Auto Attendant – Main Menu** page to create a new scenario.
  TIP: The **Create scenario** link will be renamed into the **Edit scenario** after creating a scenario.
- **Import/Export scenario** leads to the **Auto Attendant – Import/Export Scenario** page to import/export the scenario file.
- **Remove scenario** is used to remove the current scenario.
- **View/Download VXML scenario** is used to view and download the scenario script in **VXML** format.

Click the **Edit scenario** link to modify the custom scenario through scenario builder. Two main sections are available: **Main Menu** and **Submenus**. All incoming calls to auto attendant will be placed to the **Main Menu** first. The **Submenus** are the supplementary menus which can be called from other menus. There are no limitations on the depth and nesting levels of menus.

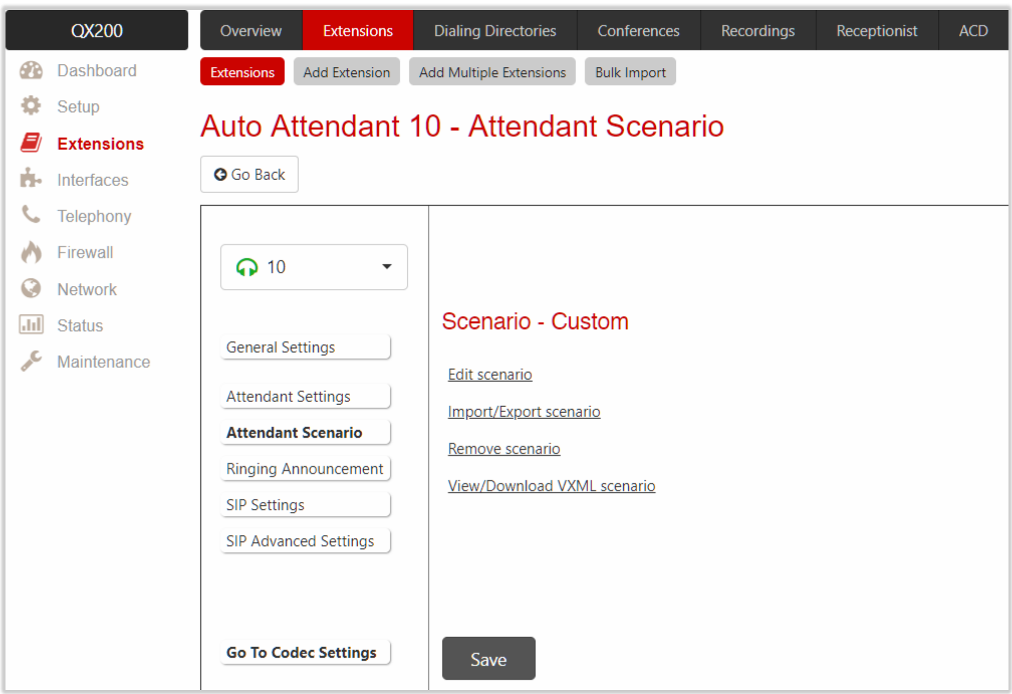


Figure 47: Auto Attendant – Custom scenario

Main Menu

Main Menu consists of the following sections:

- **Welcome Message** is used to play a welcome message (**default** or **custom**) once when entering the **Main Menu**. TIP: If the **Welcome Message** is not specified, then the welcome message for **Standard** scenario will be played.
  - **Enable Welcome Message** is used to activate the welcome message (**default** or **custom**).
  - **Delay after message** is used to set the break between the welcome message and recurring prompt.

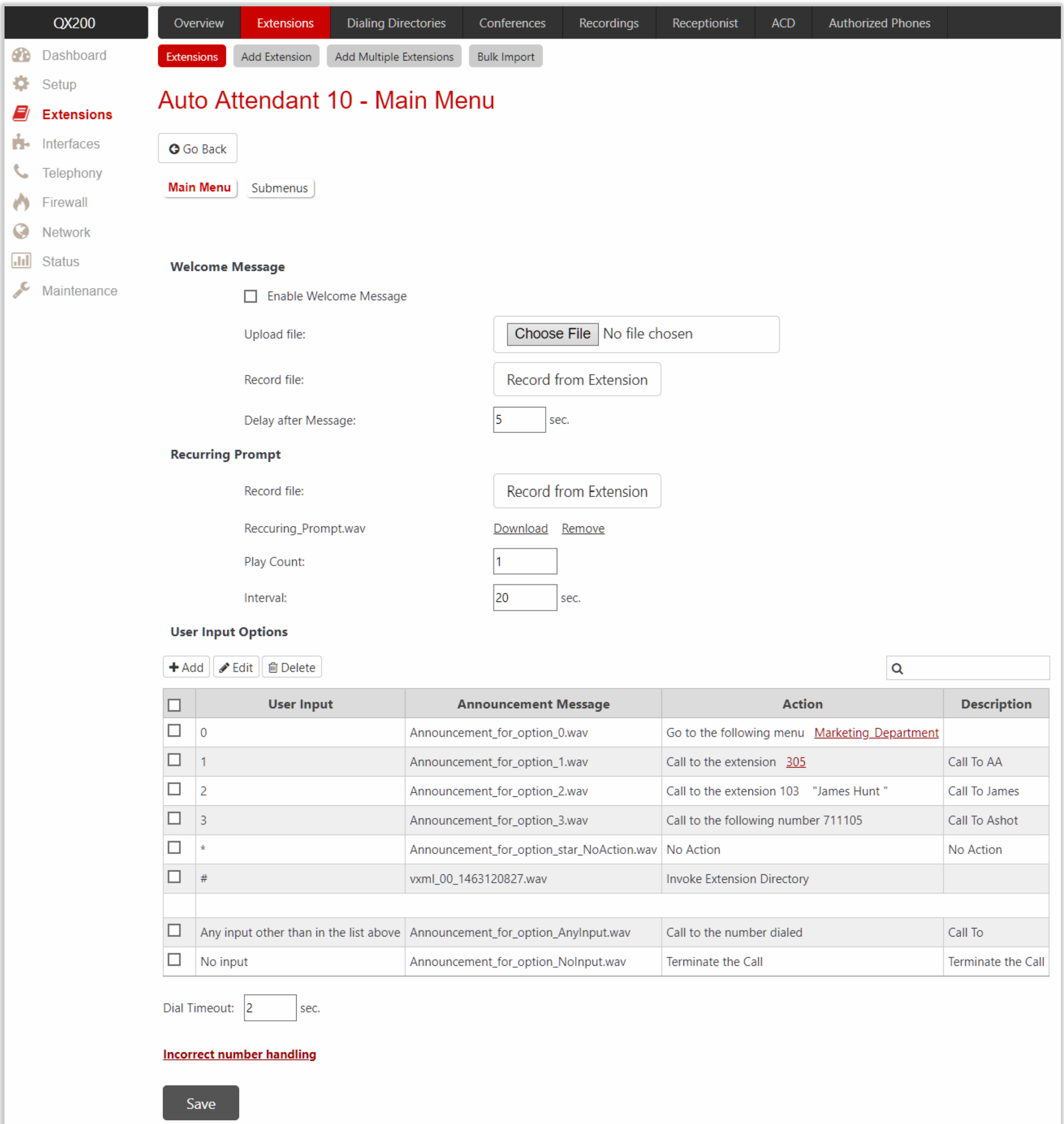


Figure 48: Create scenario – Main Menu page

- **Recurring Prompt** is used to play a recurring prompt (**default** or **custom**) after the **Welcome Message**. TIP: If the **Recurring Prompt** is not specified, then the recurring prompt for **Standard** scenario will be played.
  - **Play Count** is used to set the repetition count of the recurring prompt.
  - **Interval** is used to set the silence duration between consecutively played recurring prompts.
- **User Input Options** table consists of the following components:

- Option is used to select the user input and configure it with some announcement and action to be taken.
- Any input other than preconfigured in the list configures the action taken when the caller makes a selection other than options listed in the User Input table. If it is configured to No Action, then the timer for No input will reset and it will be counting the No input time again.
- No input configures the action taken when the caller doesn't enter anything during the certain period. The No input timeout is composed of [Welcome Message duration] + Delay after message + [Recurring Prompt duration] * Play Count + Play Count * Interval. If there is no input during that time, the action specified for No input will take effect.

- Dial Timeout is used set the timeout after user has completed the dialing before the call processing starts. The timer will start after the last digit or symbol is entered. Press the # sign to process the call immediately.
- Incorrect number handling leads to the Option Incorrect Number Handling page to configure the action taken when the user has selected a destination that resulted in a failed call, such as an invalid extension number.

Note:

- The Incorrect number handling will be activated only if either an attempt was made to call to a non-existing extension or to a number not matching with any Destination Number Pattern in the Call Routing Table.
- The Incorrect number handling will be activated only if the call comes to the auto attendant from SIP, PSTN or IP-PSTN side.

Input Option Configuration

- Add leads to the Add Option page to configure previously unspecified inputs.
- Edit leads to the selected Option page to modify the actions of Input option.

The following components are available:

- Option is used to select the user input and configure it with some announcement and action to be taken (applicable only for User Input). The following input options are available:
  - Digits (from 0 to 9)
  - Signs (* and #)
- Announcement is used to upload/record an announcement message for the selected User Input option. As soon as the caller presses the preconfigured digit, the message will be played and only then the action configured for that User Input option will be activated.
- Action is used to configure the action which will be taken after the Announcement message.
  - No Action – the system continues playing the Recurring Prompt.
  - Go to the following menu leads to the selected submenu. The drop-down list allows to select a previously created submenu or create a new one by choosing the Create New Submenu option.
  - Call to the following extension is used to call to the selected extension.
  - Call to the following number is used to call to the specified destination via the Call Routing Table.
  - Call to the number dialed is used to send the user inputs to the Call Routing Table (available only for the Any input other than in the list above option).
  - Invoke Extensions Directory is used to connect the caller to the Extension Directory.
  - Terminate the call is used to disconnect the call.

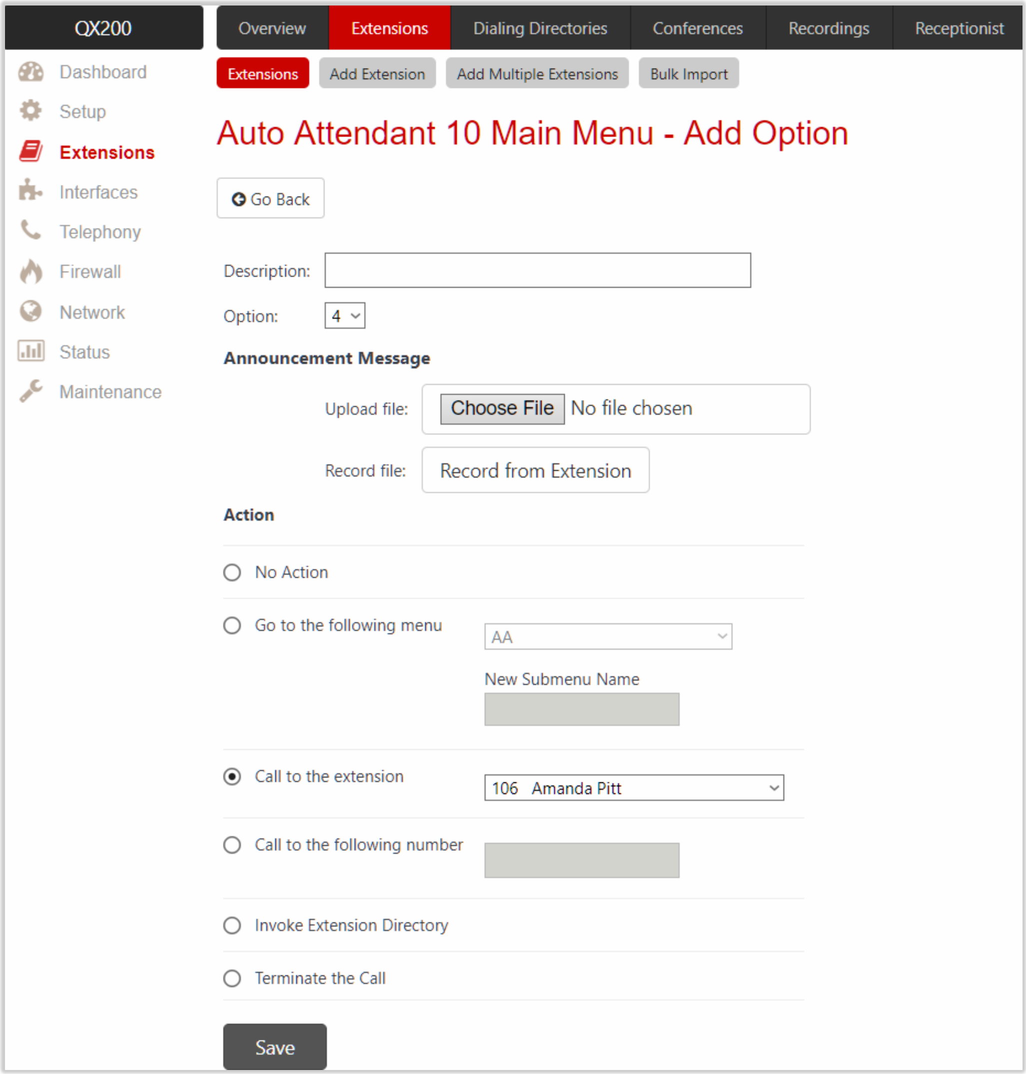


Figure 49: Main Menu – Add Option page

Submenus

Submenu is a supplementary menu accessible from the Main Menu. Submenu allows to configure multilevel scenarios. Submenu consists of the same sections and configuration options as Main Menu.

ACD scenario

ACD scenario allows to use a special scenario for ACD agents. This scenario allows ACD agents to change/update their status by dialing to auto attendant and following voice prompts.

Note: This selection is only available if the ACD feature is activated.

Ringing Announcement

The Ringing Announcement section is used to play an optional custom message to callers instead of ring-back tones when making calls through the auto attendant. Select the Enable Ringing Announcement option to activate service on the auto attendant.

Note: Ringing Announcement is played to SIP-to-Extension and PSTN-to-Extension calls only. It can also be played to SIP-Attendant-SIP and PSTN-Attendant-SIP calls if they are made by a call routing rule with the Use RTP Proxy option enabled.

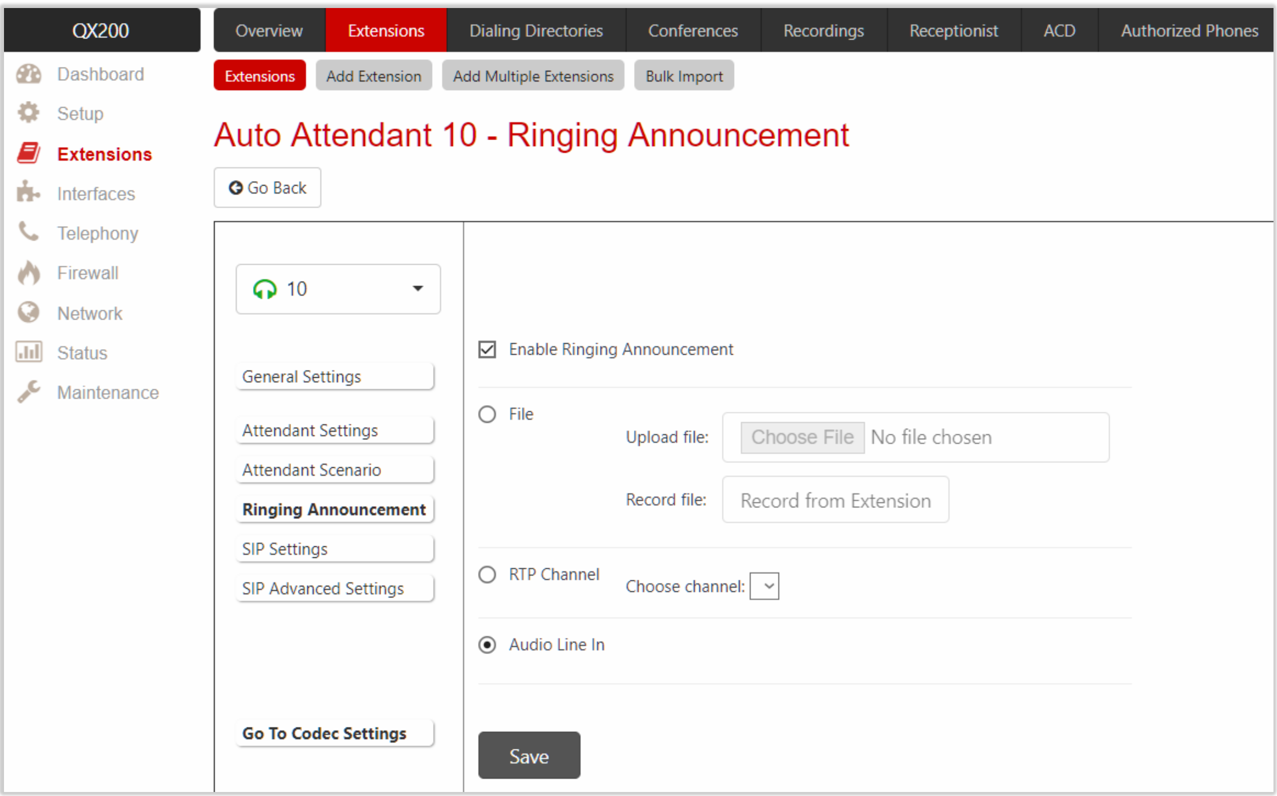


Figure 50: Auto Attendant – Ringing Announcement section

6.1.11 Extension Codecs

The **Codecs** table lists regular audio and video codecs available for extensions. Checkboxes in the table are used to **enable** or **disable** the selected codec, to **move** it **up** or **down** in the table.

The order of the **enabled codecs** is important. Codec has a higher priority over those located below in the table. **Move Up** moves the selected codec one level up, increasing the codec priority and the **Move Down** moves the selected codec one level down, decreasing its priority.

The codec at the top of the table is used as a **preferred** one. When establishing a call, the system will try this codec first. If the called party does not support the preferred codec, the following codecs will be tried out strictly in the order given in the **Codecs** table. The **Make preferred** button moves the selected codec to the top of the table, setting its priority to the highest. Clicking the **Make preferred** when a disabled codec is selected will first enable that codec, then move it to the top.

Disabled codec(s) will never be used for the call setup. At least one codec must be enabled in the table; otherwise communication will be impossible.

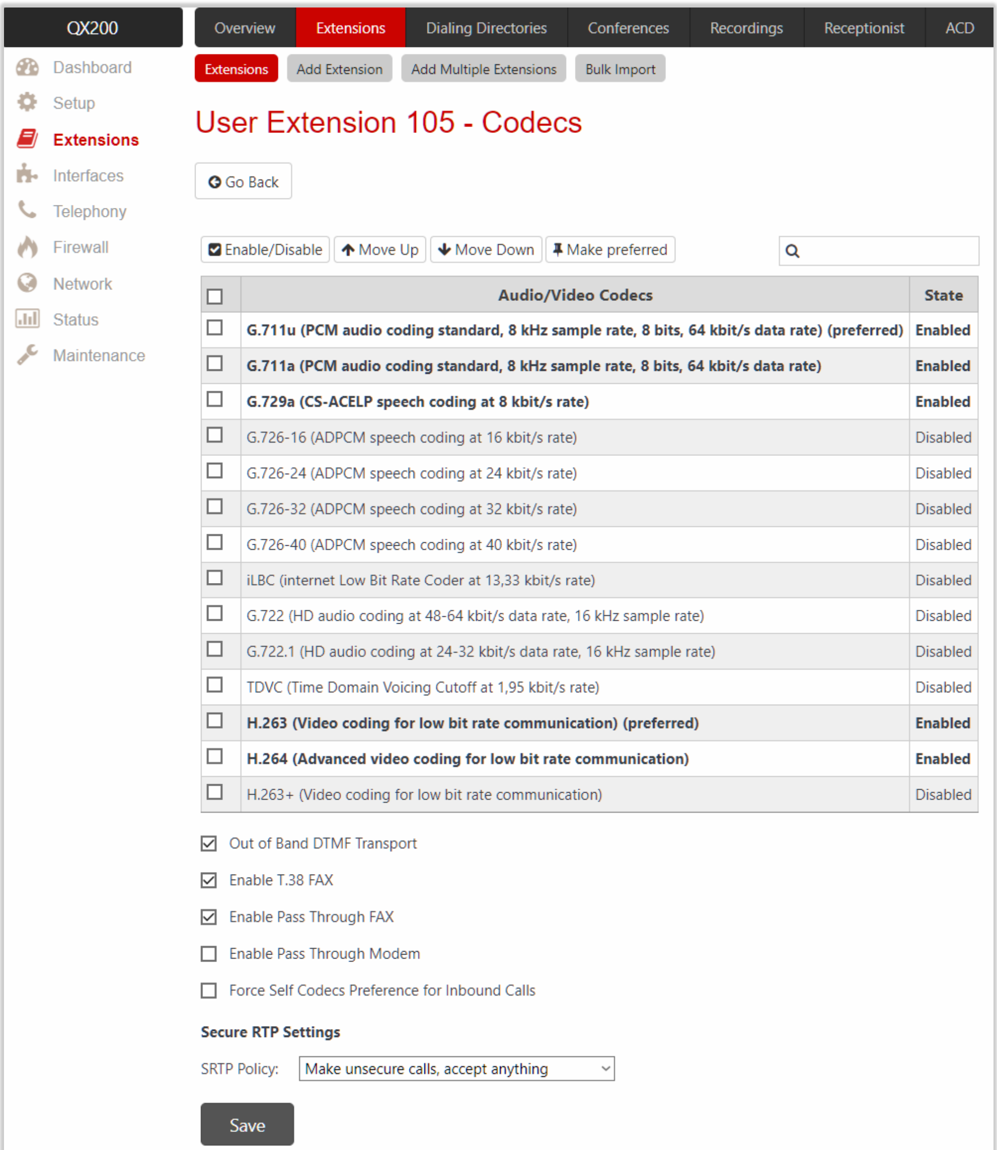


Figure 51: Extension Codecs list

The following settings (options) are available:

- Out of Band DTMF Transport enables the DTMF code transmission in parallel with the voice stream. Destination that received the DTMF code will play it locally if it supports the feature too. This helps to avoid DTMFs loss in case of heavy traffic.
- Enable T.38 FAX enables the T.38 codec support for the FAX transmission from/to the FAX machine/modem in case if that FAX machine/modem is connected to FXS line attached to target extension. It also enables the T.38 codec support of FAX transmission for incoming unified FAX messages (fax to mailbox) and remote IP devices connected to Epygi device via routing rules which using the target extension user settings.
- Enable Pass Through FAX enables the G.711 codec support for the FAX transmission from/to the FAX machine/modem in case if that FAX machine/modem is connected to FXS line attached to target extension. It also enables the G.711 codec support for incoming unified FAX messages (fax to mailbox) and IP devices connected to the attached IP line. TIP: If both of the above options are enabled, the T.38 FAX will be used as a preferred codec for FAX transmission. If it is not supported by the peer, the G.711 codec will be used instead. For virtual extensions, the incoming FAX can only be stored in the extension Voice Mailbox. To allow FAX message to be stored in the voice mailbox, the user should not answer the incoming calls, so that they are forwarded to the Voice Mailbox. If the T.38 FAX and Pass Through FAX options are disabled, no FAX transmission to the peer Voice Mailbox will be possible.
- Enable Pass Through Modem enables the modem tone detection and the G.711 codec support for the data transmission from/to the modem attached to the line. During data transmission, the Silence Suppression and Echo Cancellation are automatically disabled on the line. TIP: If the user extension or auto attendant is intended to accept modem connections, disable the T.38 FAX option to allow the system to identify the modem tones correctly, otherwise the modem connection may fail.
- Force Self Codecs Preference for Inbound Calls enables the usage of your own preferred codecs (if available on both peers).
- Secure RTP Settings are used to configure secure VoIP communication on QX. The following options are available:
  - Make and accept only secure calls – only secure calls will be generated and accepted.
  - Make and accept only unsecure calls – only unsecure calls will be generated and accepted.
  - Try to establish secure calls, accept anything – first the system will try to establish a secure call, but will fall back to unsecure call if the party doesn't accept secure calls. Both secure and unsecure incoming calls will be accepted, as requested by the remote party, with the preference given to establishing a secure call.
  - Make unsecure calls, accept anything – the system will establish unsecure outgoing calls, but both secure and unsecure incoming calls will be accepted as requested by the remote party.

6.1.12 Bulk Import

Extension Template Management and Bulk User Extensions Importer tools are used to create and update multiple user extensions on QX.

The Extension Template Management tool is for configuring common settings for extensions (e.g. SIP Server, SIP Port, etc.), while the Bulk User Extensions Importer tool is for configuring specific settings (Display Name, Extension Password, etc).

For more information on how to configure and use Bulk Import service, refer to the Extensions Bulk Import on QX IP PBXs guide.

6.2 Dialing Directories

QX provides different services allowing PBX extensions and external callers to dial the desired destinations in a more simplified way. These services are known as Dialing Directories:

- Dial by Name allows dialing the desired extension by simply spelling the extension's User name on the phone keypad.
- Global Speed Dial allows dialing the desired destination by using a preconfigured speed dial code (shortcut number).
- Phone Book allows to dial the desired contact by using the contact's name from the Local Directory on the phone.

For more information on how to configure and use Dialing Directories, refer to the Dialing Directories on QX IP PBXs guide.

6.3 Conferences

Epygi conferencing is composed of the following two licensable features:

- Audio conference feature activated by installing the Audio Conference license key.
- Video conference feature activated by installing the Video Conference license key.

For more information on how to configure and use Audio-Video conferences, refer to the Audio-Video Conferencing on QX IP PBXs guide.

6.4 Recordings

Universal Extension Recordings is used to define voice messages universal for all user extensions on QX. The defined messages become applicable by default to all extensions, unless no custom messages have been uploaded or recorded.

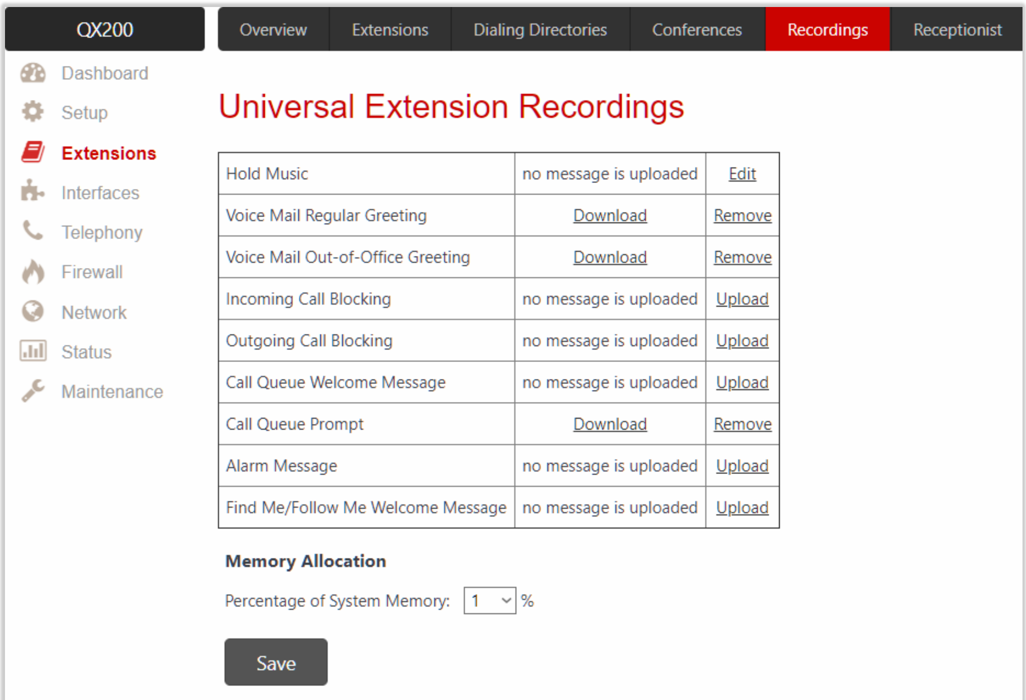


Figure 52: Universal Extension Recordings page

The following messages are available:

- Hold Music – played to the user on hold.
- Voice Mail Regular Greeting – played when a caller reaches to the Voice Mail of extension.
- Voice Mail Out-of-Office Greeting – played when a caller reaches to the Voice Mail of extension if the Out-of-office greeting is enabled.
- Incoming Call Blocking – played when a blocked user calls the extension.
- Outgoing Call Blocking – played when the extension dials a blocked destination number.
- Call Queue Welcome Message – played when a caller joins the extension call queue.
- Call Queue Prompt – played when a caller is being held in the queue.
- Alarm Message – played to the user after answering the call.
- Find Me/Follow Me Welcome Message – played when a user calls the extension with enabled FM/FM service.

To change the message:

1. Set Percentage of System Memory and click Save to allocate memory for Universal Extension Recordings.
2. Click the Upload link next to the message (or Edit in case of Hold Music).
3. Click Choose File to open the file chooser window and browser for the file.
4. Click Save to upload the file.

6.5 Receptionist

QX **Receptionist** service offers a variety of services to manage multiple calls: answer calls, keep the calls in the queue and forward to users.

For more information on how to configure and use the **Receptionist** service, refer to the Receptionist Service on QX IP PBXs guide.

6.6 ACD

Automatic Call Distribution (ACD) feature is a complete solution for today's call centers. ACD is designed to receive and queue high-volume inbound calls, then distribute queued calls to the available agents in a call center.

Epygi ACD Console (EAC) is a web application designed to support call center agents monitoring ACD activity and performance on the QX. EAC stores and formats the data and produce real-time information and statistical reports on ACD activity.

- In order to activate ACD feature the **Automatic Call Distribution** license key should be installed.
- In order to activate EAC feature the **Epygi ACD Console** license key should be installed.

For more information on how to configure and use **ACD** and **EAC**, refer to the ACD and EAC - User Guide.

6.7 Authorized Phones

Authorized Phones is used to create a list of trusted external users allowing them to access QX auto attendant services without authentication.

To add a new entry:

1. Click **Add**. The **Authorized Phones – Add Entry** page will be opened.
2. Tick the **Enable** checkbox to activate service for the created entry.
3. Enter the caller SIP address or PSTN number.
4. Select the **Login Extension**. When calling the auto attendant, a trusted user will automatically be logged in as the selected extension (the extension number and password will be automatically submitted by the system and the trusted user will directly access to auto attendant services).

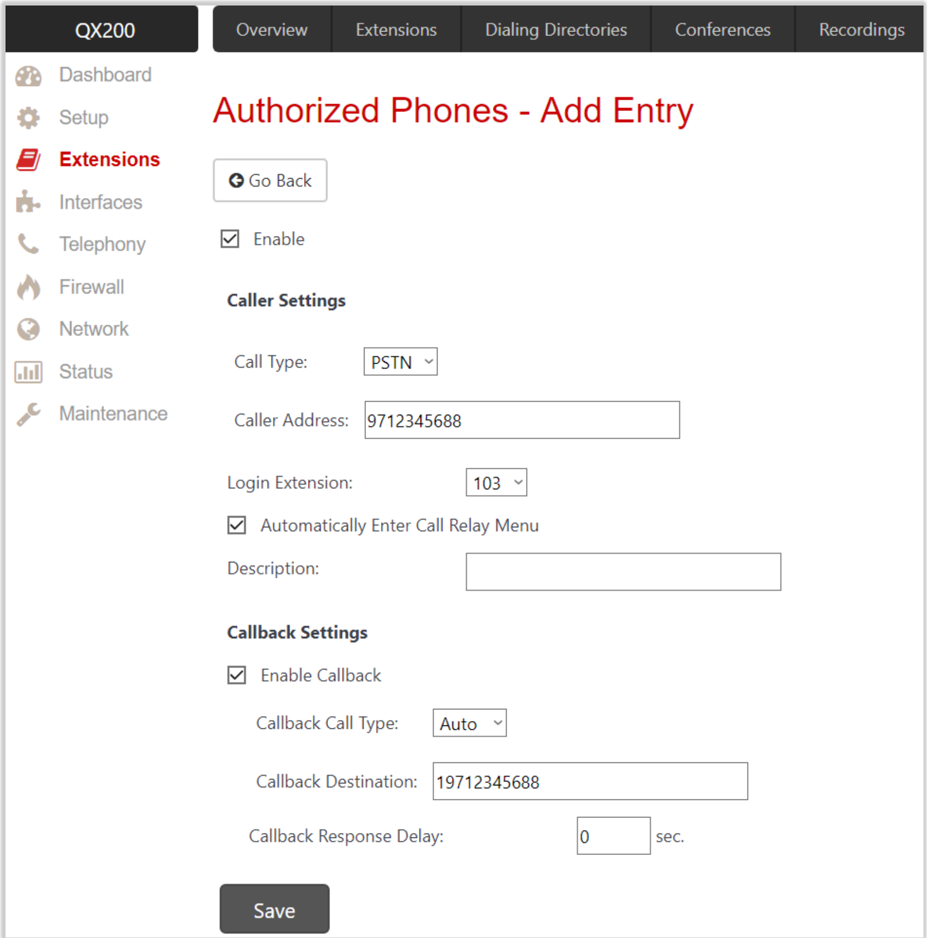


Figure 53: Authorized Phones – Add Entry page

5. Select the **Automatically Enter Call Relay Menu** option. If selected, allows direct access for trusted user to **Call Relay** menu. Otherwise a trusted caller will be only directed to the auto attendant, but still will be able to reach to the **Call Relay** services (by dialing *2) without authentication.
6. Configure **Callback Settings** (optional).

- Tick **Enable Callback** checkbox to allow a specified caller to use the **Callback** service.
- Specify the **Callback Destination**. TIP: If the **Callback Destination** is left blank, the trusted caller address will be implied as a **Callback** destination.
- Define **Callback Response Delay** before the **Callback** will be activated.

How it works: The trusted user will be able to use QX services after calling the auto attendant, as if a user extension. If the **Callback** service is activated the trusted user will get a call back from auto attendant.

Note:

- **Authorized Phones** will only work when the trusted caller connects to the **auto attendant** running the **Standard** scenario.
- For more information on how to configure and use **Callback** service, refer to the Callback Service on QX IP PBXs guide.

7 Interfaces Menu

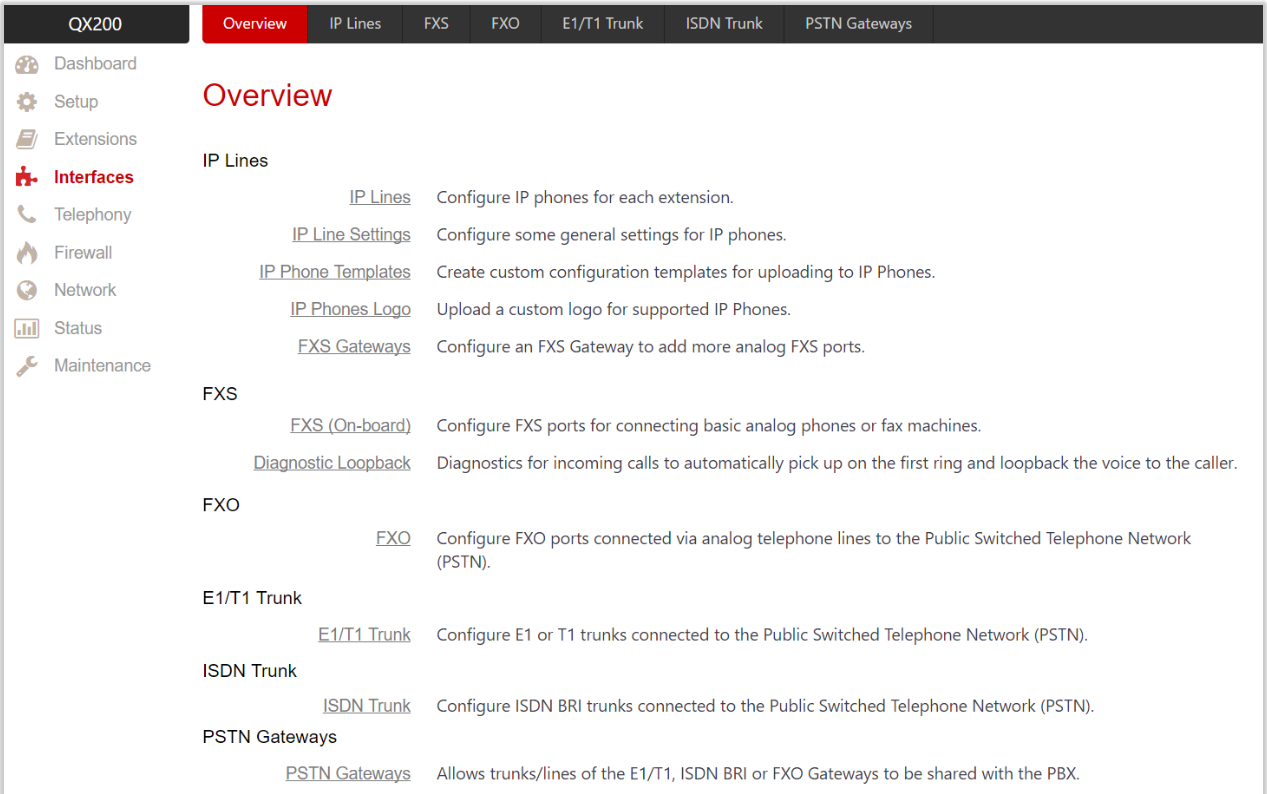


Figure 54: Interfaces Menu overview

7.1 IP Lines

The **IP Lines** table lists all IP lines available on QX with specific details for each. The following buttons and parameters are available:

- **Reboot** is used to reboot selected IP phone(s).
- **Deactivate** is used to change the status for selected group(s) of IP lines to **free** (inactive).
- **Show disabled IP lines/Hide disabled IP lines** is used to show or hide the IP lines not activated with a feature key.
- **Enable/Disable OpenVPN** is used to provide configuration file for selected group(s) of IP lines through OpenVPN.

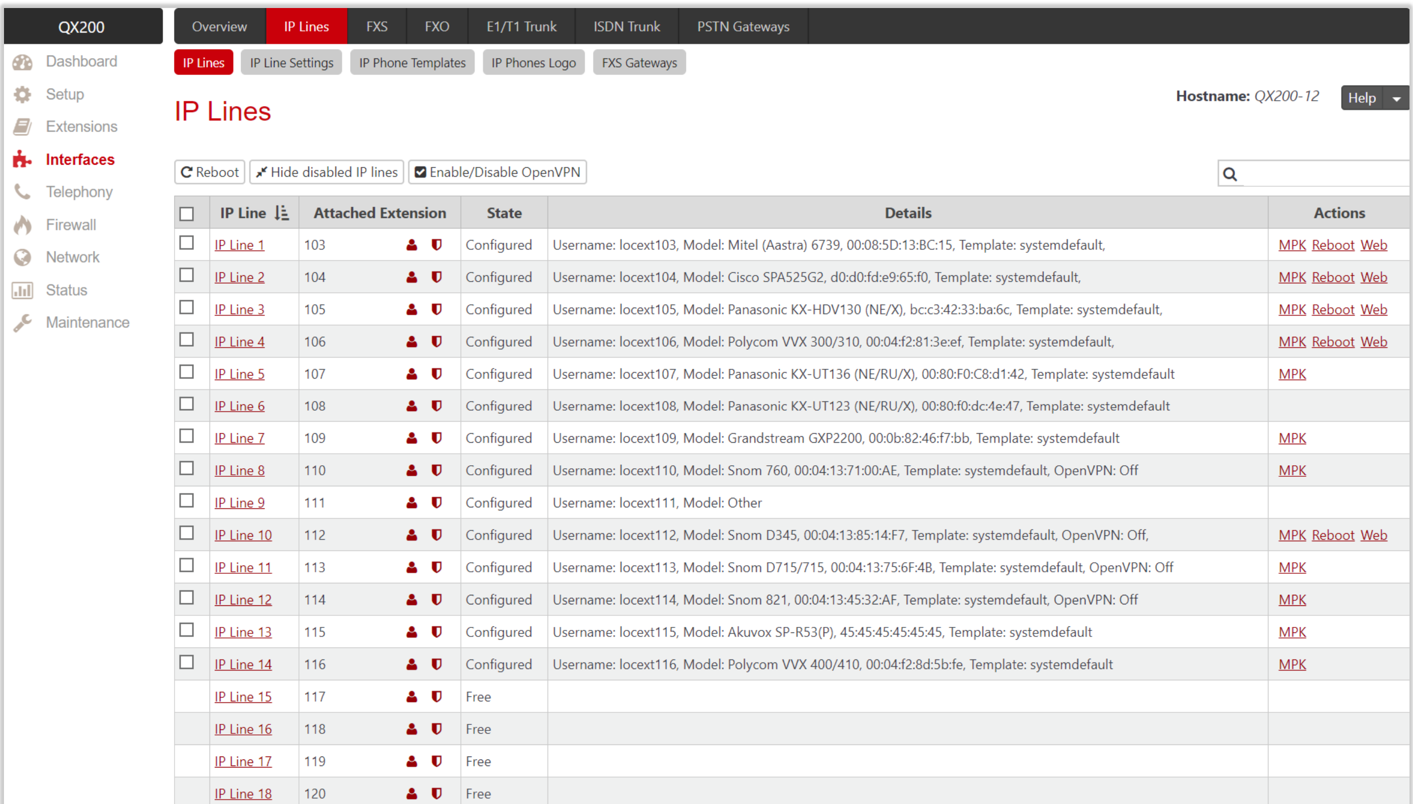


Figure 55: IP Lines page

- **IP Line** shows all IP lines available on QX. Click on **IP Line** to go the IP Line Settings - IP Line page.
- **Attached Extension** shows the user extension attached to the IP line. TIP: **None** is displayed if there is no extension attached to that line.
  - ➢ Click the **Admin Settings** icon to go to the extension **admin** settings.
  - ➢ Click the **User Settings** icon to go to the extension **user** settings.
- **State** shows whether the IP line is **Disabled**, **Configured** or **Free**.
- **Details** shows the settings for the IP phone configured on the corresponding line, such as the phone model, MAC address, attached IP phone template and the authorization credentials.
- **Actions** – the following actions are available to manage the IP phone:
  - ➢ **MPK** leads to the Multi-functional Programmable Keys page of the phone.
  - ➢ **Reboot** is used to reboot the IP phone.
  - ➢ **Restart** is used to restart FXS Gateway (QXFXS24 or QuadroM FXS26) attached to the line.
  - ➢ **Web** leads to the WEB GUI of the IP phone. TIP: This link only works from the LAN side of the QX, i.e. when the WEB GUI is accessed from a PC located in the LAN of QX. If you wish to access the WEB

GUI of the IP phone through WAN, an appropriate Incoming Traffic/Port Forwarding filtering rule should be added on QX.

IP Line Settings – IP Line

The **IP Line Settings – IP Line #** page is used to configure IP phone with QX. The following settings (options) are available:

- **Inactive** – if selected, changes the IP line state from **Configured** to **Free.**
- **IP Phone** – if selected, activates the IP line to configure with the IP phone as follows:
  - **Phone Model** is used to select the IP phone model. Select **Other** if the phone model is not listed or the phone should be configured as a Remote Extension.
  - **MAC Address** is used to set the MAC Address of the phone.
  - **Line Appearance** is used to set the number for intended simultaneous calls (the total number of active and held calls).
  - **Username** and **Password** are used to set the authentication parameters to register the IP phone on QX. By default, QX assigns unique username and password to each IP line. You may either keep these values or specify new ones. TIP: The **Username** and **Password** should match on both QX and IP phone for the successful registration.
  - **Transport** is used to select the transport protocol for SIP messages – **UDP**, **TCP** or **TLS**. For TLS, you may activate the TLS Certificates update mechanism from an IP phone to obtain the latest certificate generated by the QX.
  - **Use Template** is used to select a preconfigured custom template for the IP phone. When the **Use default** option is selected, the template selected on the IP Line Settings page will be used.
  - **Use Session Timer** enables the SIP session timer for the IP line. This option allows both user agents and proxies to check and determine if the SIP session is still active.
  - **Symmetric RTP** must be selected when the IP phone attached to the IP line is located behind the NAT router.
  - **Use OpenVPN Settings** is used to select this option to auto configure phone using the OpenVPN settings. The OpenVPN service for auto configuration is available on majority of **Epygi Supported IP phones.**
  - **OpenVPN client configuration** is used to select and download OpenVPN client configuration file for the IP phone attached to the IP line. TIP: This option is **NOT** used to apply the OpenVPN configuration on the phone.

Hot Desking

If QX has limited number of IP phones connected, but much more users wishing to make and receive calls through QX, some of the connected phones can be announced as "public". "Public" phones have no static owners; they are just connected to the IP lines. Each user that accesses the "public" phone should login using personal credentials (extension number and password).

The Hot Desking section is used to enable and configure the Hot Desking service on the IP line. The following settings (options) are available:

- Enable Hot Desking is used to activate service on the IP line.
- Hot Desking Automatic Logout is used to configure the Hot Desking service expiration on the current extension. The following options are available:
  - Never – the service will never expire and the extension will remain logged into the "public" phone.
  - After – the extension will automatically get logged out from the "public" phone after a specified period of time.
  - At – the extension will automatically get logged out from the "public" phone at the specified moment (hour and minute).

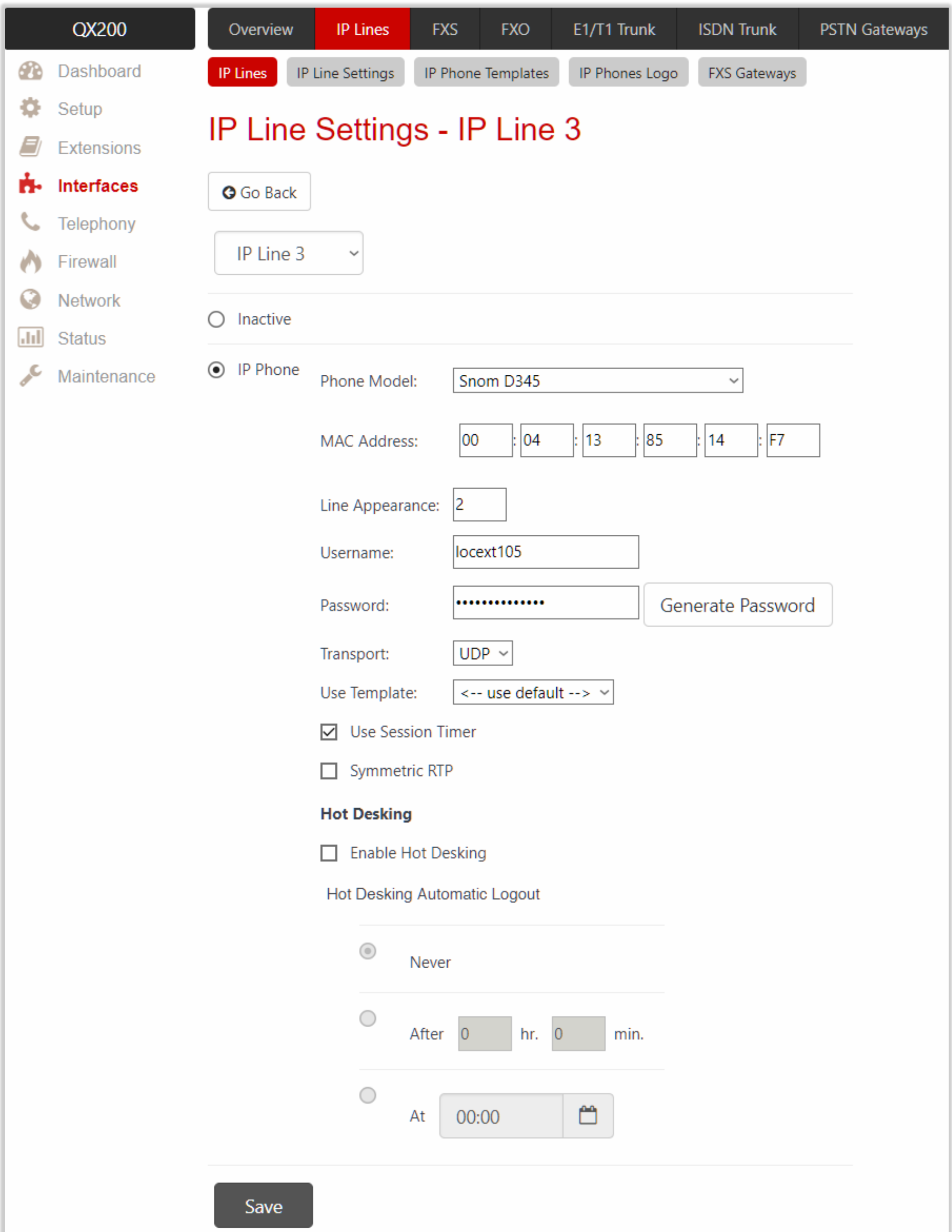


Figure 56: IP Line Settings – Edit page

For more information on how to configure and use **Hot Desking** service, refer to the Hot Desking Service on QX IP PBXs guide.

Multi-functional Programmable Keys

The **Multi-functional Programmable Keys** page is used to assign functions to the programmable keys of the IP phone. The design of this page depends on the IP phone model. However, regardless of the IP phone model, this page contains a number of programmable keys and a **Function** list assigned to each of them.

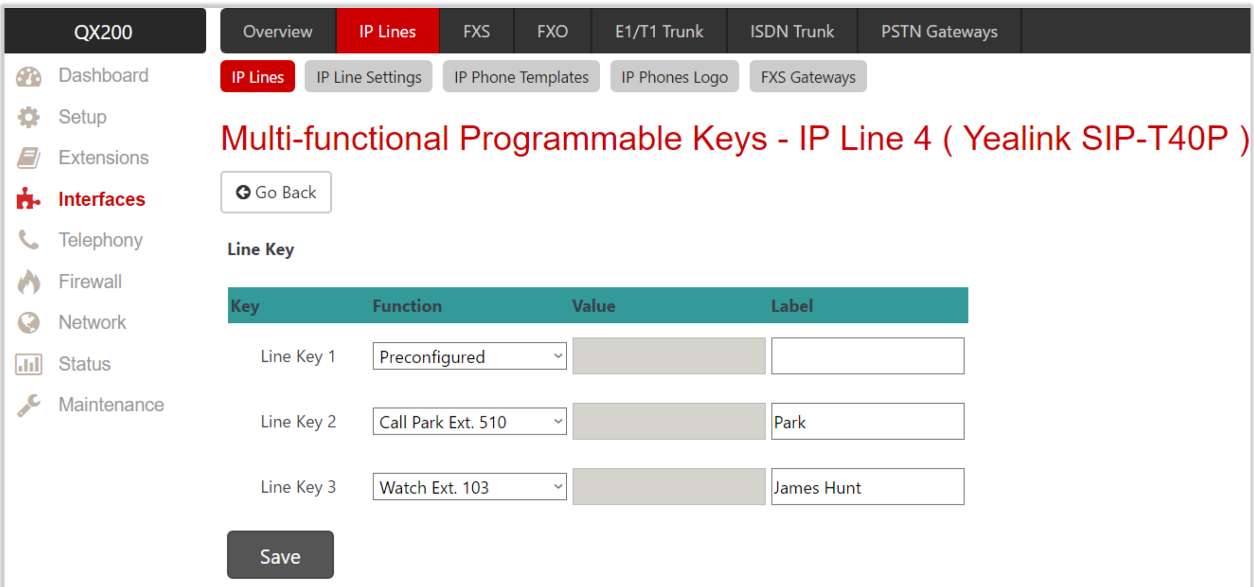


Figure 57: Programmable Keys Configuration page

The following options are available in the **Function** list:

- **Preconfigured** will not change anything for the key functionality. In fact, it will keep the previously configured function for that key.
- **None** eliminates any functionality for the key. In fact, it disables the key.
- **IP Line** allows to assign a key to the corresponding IP line. Press the key to get dial tone. The key will flash when a call is ringing to that line. The key illuminates green when the IP line is busy with another call. TIP: Based on the phone model, the status of the BLF key and the status of the IP Line will vary.
- **Watch Ext. #** allows to watch the extension and intercept calls addressed to that extension.
- **Call Park Ext. #** allows to watch the parked calls on the corresponding **Call Park** extension and retrieve the parked calls.
- **Shared Vmail Ext. #** allows to watch and access to the Shared Voice Mailbox.
- **Schedule #** allows to watch and update the state for a specific schedule.
- **Vmail** allows to access to the voice mailbox of the extension.
- **DND** allows to activate/deactivate the **Do Not Disturb** service on the extension.
- **CallFwd** allows to configure/toggle (activate/deactivate) **Unconditional Call Forwarding** on the extension.
- **AutoReDl** automatically redials the last dialed number.
- **CallBack** calls back to the last caller.
- **LineInfo** plays information about the IP line.
- **CallBlk** blocks the last caller.
- **Record** allows to start the call recording (in case if the **Manual** mode for call recording is configured in the Call Recording Settings).

- **ACD Login/Logout** allows to login/logout the corresponding **ACD agent** to/from all queues he/she is involved in.
- **LDAP** allows to retrieve contacts from 3-rd party LDAP server.
- **URL** is basically HTTP GET Requests (often XML over HTTP) that allow the phone to interact with web server applications. It can be used to retrieve various data from the web server.

Note: The system will ask a conformation to remotely reboot the IP phone to save changes. It is recommended to reboot the IP phone after configuration changes on this page to make the new configuration effective on the IP phone.

Supported IP Phones

Below is the list of IP phones that are officially supported by Epygi and can be configured with QX using **Plug and Play** (PnP) or **Auto Configuration** services.

Vendor	Model	SW/FW Version	PnP Support
Akuvox	R15(P)	15.0.5.235	Yes
Akuvox	SP-R53(P)	53.0.6.115	Yes
Alcatel	IP2015 (IP15)	1.0.7A-0	No
Alcatel	Temporis IP100	1.0.6A-0	No
Alcatel	Temporis IP150	1.0.6A-0	No
Alcatel	Temporis IP200	13.60.0.89	Yes
Alcatel	Temporis IP300	1.0.7B-0	No
Alcatel	Temporis IP600	14.60.0.89	Yes
Alcatel	Temporis IP700G	1.0.7A-0	No
Alcatel	Temporis IP800	15.60.0.89	Yes
AudioCodes	310HD	1.6.0_build_37	No
AudioCodes	320HD	1.6.0_build_37	No
Cisco	SPA303	7.4.9c	Yes
Cisco	SPA501G	7.4.9c	Yes
Cisco	SPA509G	7.4.9c	Yes
Cisco	SPA525G2	7.4.9c	Yes
Fanvil	C58/C58P	2.3.233.129	No
Fanvil	C62/C62P	2.3.235.128	No
Fanvil	C400	11.20.12.2.B	No
Fanvil	C600	11.20.12.2.B	No
Fanvil	F52/F52P	2.3.123.78	No
Fanvil	H2/H2S	2.0.2.2776	Yes
Fanvil	H3	2.0.2.2770	Yes
Fanvil	H5	2.0.2.2770	Yes
Fanvil	X3/X3P	1.3.511.1821	Yes
Fanvil	X3S/X3G	2.0.3.3049	Yes
Fanvil	X4/X4G/X4S	2.0.2.2830	Yes
Fanvil	X5/X5G	1.3.511.1821	Yes
Fanvil	X5S	R0.7.0.1	Yes
Fanvil	X6	R0.5.3	Yes
Grandstream	GXP1100	1.0.8.6	Yes
Grandstream	GXP1105	1.0.8.6	Yes

Vendor	Model	SW/FW Version	PnP Support
Grandstream	GXP1160	1.0.8.6	Yes
Grandstream	GXP1165	1.0.8.6	Yes
Grandstream	GXP1400	1.0.8.6	Yes
Grandstream	GXP1405	1.0.8.6	Yes
Grandstream	GXP1450	1.0.8.6	Yes
Grandstream	GXP1615/1610	1.0.4.55	Yes
Grandstream	GXP1625/1620	1.0.4.55	Yes
Grandstream	GXP1628	1.0.4.55	Yes
Grandstream	GXP1630	1.0.4.55	Yes
Grandstream	GXP1760	1.0.0.48	No
Grandstream	GXP1782/1780	1.0.0.48	No
Grandstream	GXP2100	1.0.8.6	Yes
Grandstream	GXP2110	1.0.8.6	Yes
Grandstream	GXP2120	1.0.8.6	Yes
Grandstream	GXP2124	1.0.8.6	Yes
Grandstream	GXP2130	1.0.7.99	Yes
Grandstream	GXP2135	1.0.7.99	Yes
Grandstream	GXP2140	1.0.7.99	Yes
Grandstream	GXP2160	1.0.7.99	Yes
Grandstream	GXP2170	1.0.7.99	Yes
Grandstream	GXP2200	1.0.3.27	Yes
Grandstream	GXV3140	1.0.7.80	Yes
Grandstream	GXV3175	1.0.3.76	Yes
Grandstream	GXV3240	1.0.3.62	Yes
Grandstream	GXV3275	1.0.3.62	Yes
Htek	UC902	2.0.4.4.33	No
Htek	UC903	2.0.4.4.33	No
Htek	UC912G	2.0.4.4.33	No
Htek	UC912P	2.0.4.4.33	No
Htek	UC923	2.0.4.4.33	No
Htek	UC924	2.0.4.4.33	No
Htek	UC924E	2.0.4.4.33	No
Htek	UC926	2.0.4.4.33	No
Htek	UC926E	2.0.4.4.33	No
Mitel (Aastra)	6730	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6731	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6735	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6737	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6739	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6753	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6755	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6757	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9480	3.3.1.4305-SIP	Yes
Mitel	6863	4.2.0.2023-SIP	Yes

Vendor	Model	SW/FW Version	PnP Support
Mitel	6865	4.2.0.2023-SIP	Yes
Mitel	6867	4.2.0.2023-SIP	Yes
Mitel	6869	4.2.0.2023-SIP	Yes
Panasonic	KX-HDV130	03.004	Yes
Panasonic	KX-HDV130NE, KX-HDV130X	06.101	Yes
Panasonic	KX-HDV230	03.004	Yes
Panasonic	KX-HDV230NE, KX-HDV230X	06.101	Yes
Panasonic	KX-TGP550T04	12.17	No
Panasonic	KX-UT123 (NE/RU/X)	01.302	No
Panasonic	KX-UT136 (NE/RU/X)	01.302	No
Polycom	SoundPoint IP 330	3.3.5.0247	Yes
Polycom	SoundPoint IP 331	4.0.13.1445	Yes
Polycom	SoundPoint IP 335	4.0.13.1445	Yes
Polycom	SoundPoint IP 450	4.0.13.1445	Yes
Polycom	SoundPoint IP 550	4.0.13.1445	Yes
Polycom	SoundPoint IP 650	4.0.13.1445	Yes
Polycom	SoundPoint IP 670	4.0.13.1445	Yes
Polycom	SoundStation IP 5000	4.0.13.1445	Yes
Polycom	SoundStation IP 6000	4.0.13.1445	Yes
Polycom	VVX 300/310	5.7.0.11768	Yes
Polycom	VVX 301/311	5.7.0.11768	No
Polycom	VVX 400/410	5.7.0.11768	No
Polycom	VVX 401/411	5.7.0.11768	No
Polycom	VVX 500	5.7.0.11768	No
Polycom	VVX 600	5.7.0.11768	Yes
Polycom	VVX 1500	5.7.0.11768	Yes
QOSIP	Q7104/Q7204	1.0.3.98	No
snom	300	8.4.35	Yes
snom	320	8.4.35	Yes
snom	360	8.4.35	Yes
snom	370	8.7.5.35	Yes
snom	720	8.9.3.60	Yes
snom	760	8.9.3.60	Yes
snom	821	8.7.5.35	Yes
snom	870	8.7.5.35	Yes
snom	D345	8.9.3.60	Yes
snom	D375	8.9.3.60	Yes
snom	D710/710	8.9.3.60	Yes
snom	D715/715	8.9.3.60	Yes
snom	D725	8.9.3.60	Yes
snom	D745	8.9.3.60	Yes
snom	D765	8.9.3.60	Yes
snom	m9	9.4.7	Yes
snom	MeetingPoint	8.7.5.35	Yes
snom	M700 (M85/M65/M25)	03.24.0007	Yes

Vendor	Model	SW/FW Version	PnP Support
Spectralink	KIRK Wireless Server 300	PCS14C_	No
Spectralink	KIRK Wireless Server 6000	PCS14C_	No
VTech	ErisStation VCS754	1.1.4.0-0	No
VTech	ErisTerminal VSP600 (VSP601)	1.1.4.1-0	No
VTech	ErisTerminal VSP715	1.1.4.0-0	No
VTech	ErisTerminal VSP725	1.1.4.0-0	No
VTech	ErisTerminal VSP726	2.0.3.2-0	Yes
VTech	ErisTerminal VSP735	1.1.4.0-0	No
VTech	ErisTerminal VSP736	2.0.3.2-0	Yes
Yealink	CP860	37.81.0.10	Yes
Yealink	CP920	78.81.0.15	Yes
Yealink	CP960	73.80.0.25	Yes
Yealink	SIP-T19P	31.72.0.1	Yes
Yealink	SIP-T19P E2	53.81.0.25	Yes
Yealink	SIP-T20P	9.72.0.1	Yes
Yealink	SIP-T21P	34.72.0.1	Yes
Yealink	SIP-T21P E2	52.81.0.25	Yes
Yealink	SIP-T22P	7.72.0.1	Yes
Yealink	SIP-T23G(P)	44.81.0.25	Yes
Yealink	SIP-T26P	6.72.0.1	Yes
Yealink	SIP-T27G	69.81.0.25	Yes
Yealink	SIP-T27P	45.81.0.25	Yes
Yealink	SIP-T28P	2.72.0.1	Yes
Yealink	SIP-T29G	46.81.0.25	Yes
Yealink	SIP-T32G	32.70.0.130	Yes
Yealink	SIP-T38G	38.70.0.125	Yes
Yealink	SIP-T40G	76.81.0.110	Yes
Yealink	SIP-T40P	54.81.0.110	Yes
Yealink	SIP-T41P	36.81.0.25	Yes
Yealink	SIP-T41S	66.81.0.25	Yes
Yealink	SIP-T42G	29.81.0.25	Yes
Yealink	SIP-T42S	66.81.0.25	Yes
Yealink	SIP-T46G	28.81.0.25	Yes
Yealink	SIP-T46S	66.81.0.25	Yes
Yealink	SIP-T48G	35.81.0.25	Yes
Yealink	SIP-T48S	66.81.0.25	Yes
Yealink	SIP VP-T49G	51.80.0.100	Yes
Yealink	SIP-T52S	70.81.0.10	Yes
Yealink	SIP-T54S	70.81.0.10	Yes
Yealink	SIP-T56A	58.80.0.25	Yes
Yealink	SIP-T58A/V	58.80.0.25	Yes
Yealink	VP-530	23.70.0.40	Yes
Yealink	W52P	25.30.0.20	Yes

Table 1: Supported IP Phones

7.1.2 IP Line Settings

IP Line Settings is used to control the basic settings for configuring IP phones.

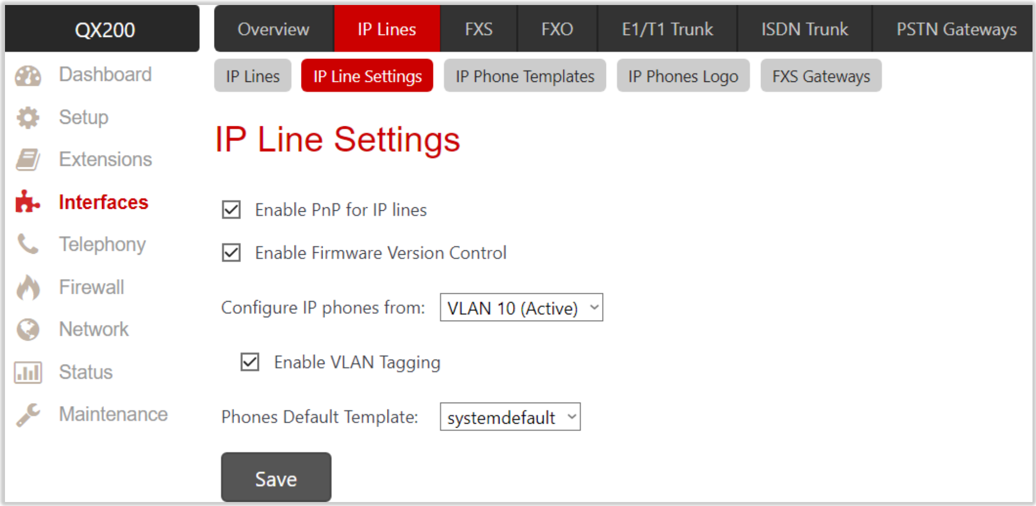


Figure 58: IP Line Settings page

The following settings (options) are available:

- Enable PnP for IP lines activates the PnP option on QX. The PnP allows Epygi supported IP phones to be automatically configured without any manual intervention in the QX and phone settings. If selected, connect the phone to the QX and factory reset the phone. After a clean boot-up of the phone, QX will detect the phone settings, automatically generate the specific configuration file of the phone and upload it. The phone will be then configured on the first Free IP line.
- Enable Firmware Version Control is used to control and manage the firmware version running on the IP phone. This service will allow to replace the firmware running on the phone (upgrade or downgrade) with the recommended one. Note: Currently the Firmware Version Control service is applicable for Mitel, Mitel (Aastra), snom and Yealink phones.
- Configure IP phones from is used to select the network interface on QX, where the IP phones should be connected to. Besides LAN and WAN (LAN1 for QX2000/QX3000), this list also includes the VLAN interface if available.
  - Enable VLAN Tagging is used to set the VLAN ID and priority for IP phones. TIP: The provided IP address will always be from VLAN network. This option is enabled by default.
- Phones Default Template is used to select the IP phone template that will be used as default for IP lines.

7.1.3 IP Phone Templates

The **Manage IP Phone Templates** page is used to create custom templates for IP phones. The templates contain a set of configuration settings that are applied to the IP phone once it is registered on QX. With the custom templates, the most popular configuration settings may be adjusted accordingly. The saved custom templates can be then configured from the IP Line Settings – IP Line # page to be used on a particular IP phone.

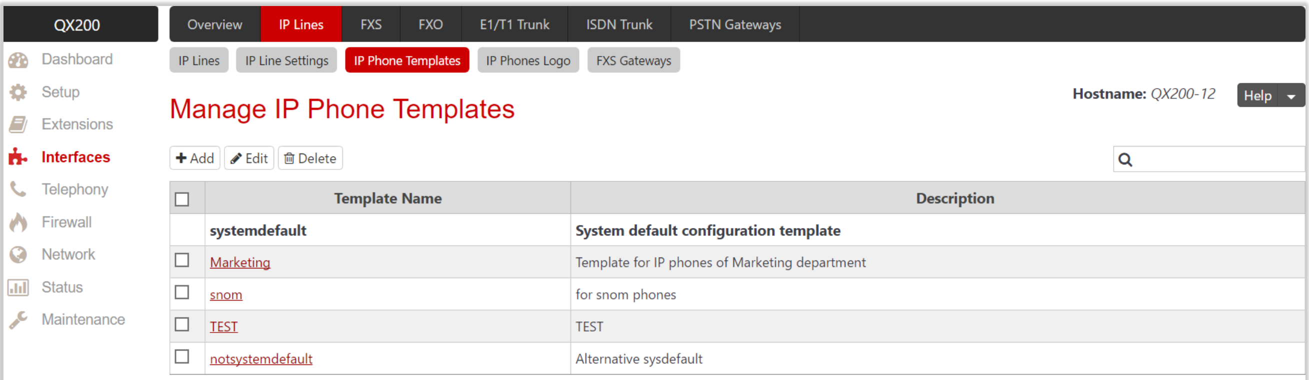


Figure 59: Manage IP Phone Templates page

To create a new **IP phone template**:

1. Click **Add** to define a template name.
   - **Template Name** is used to set a template name.
   - Enter a **Description**, if needed.
2. Click **Save** to add a new template to the **Manage IP Phone Templates** table.
3. Click on the **Template Name** link to adjust the advanced settings for different IP phone vendors and assigned functions to the programmable keys for each phone model. You are allowed to manage the settings for a group of IP phones at once.
4. Click **Save** to apply changes.

7.1.4 IP Phones Logo

IP Phones Logo is used to upload a custom logo for the IP phone. The uploaded custom logo will be visible on the display of the IP phone.

To upload a custom logo:

1. Click the Choose File button and browse for a logo file.
2. Tick the Enable Logo checkbox.
3. Click Save to apply changes.

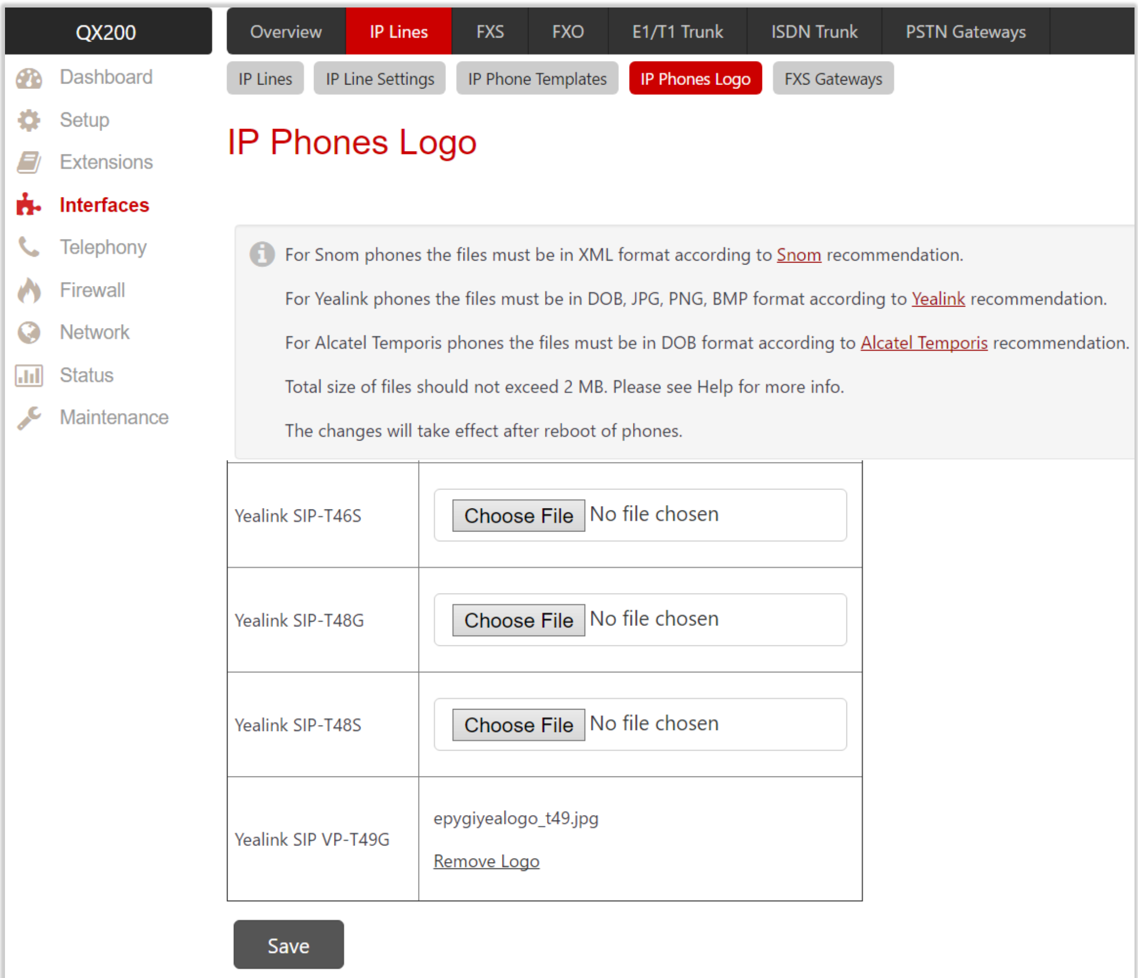


Figure 60: IP Phone Logo page

7.1.5 FXS Gateways

FXS Gateway Management is used to automatically configure QXFXS24 with QX IP PBX. QXFXS24 is an analog VoIP Gateway connects analog phones to a VoIP network. The device can be used with QX IP PBX to emulate FXS ports. The **FXS Gateway Management** table lists all configured FXS gateways.

Click **Add** to run **FXS Gateway Configuration Wizard** and configure FXS gateway with QX. The wizard consists of the following sections:

- FXS Gateway Model
- Line Mapping – Add Entry
- Summary

FXS Gateway Model

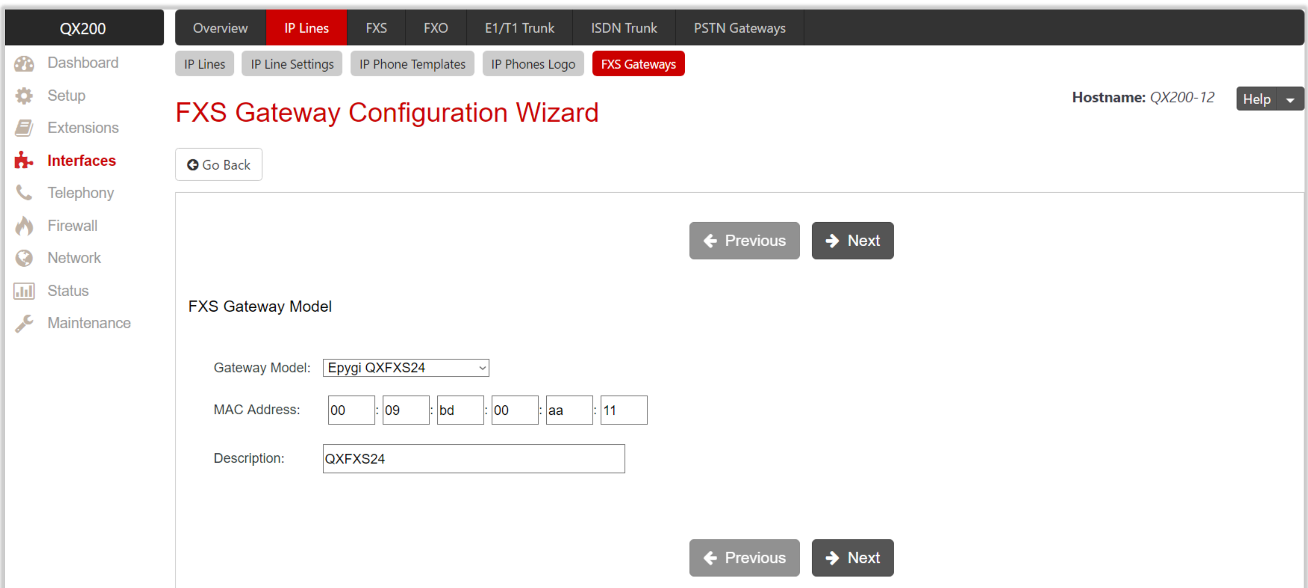


Figure 61: FXS Gateway Model section

The following settings (options) are available:

- **Gateway Model** is used to select the gateway model from the list.
- **MAC Address** is used to set the MAC Address of the gateway.

Line Mapping – Add Entry

This section is used to assign each FXS line to an IP line. The system will automatically assign the provided FXS lines to the first available IP lines on QX. **Line Mapping** can be manually adjusted. FXS lines can be assigned only to free (inactive) IP lines on QX. If there aren't any free IP lines, you should first free (deactivate) the IP line.

- **Use OpenVPN Settings** – if selected the configuration file will be provided through OpenVPN.

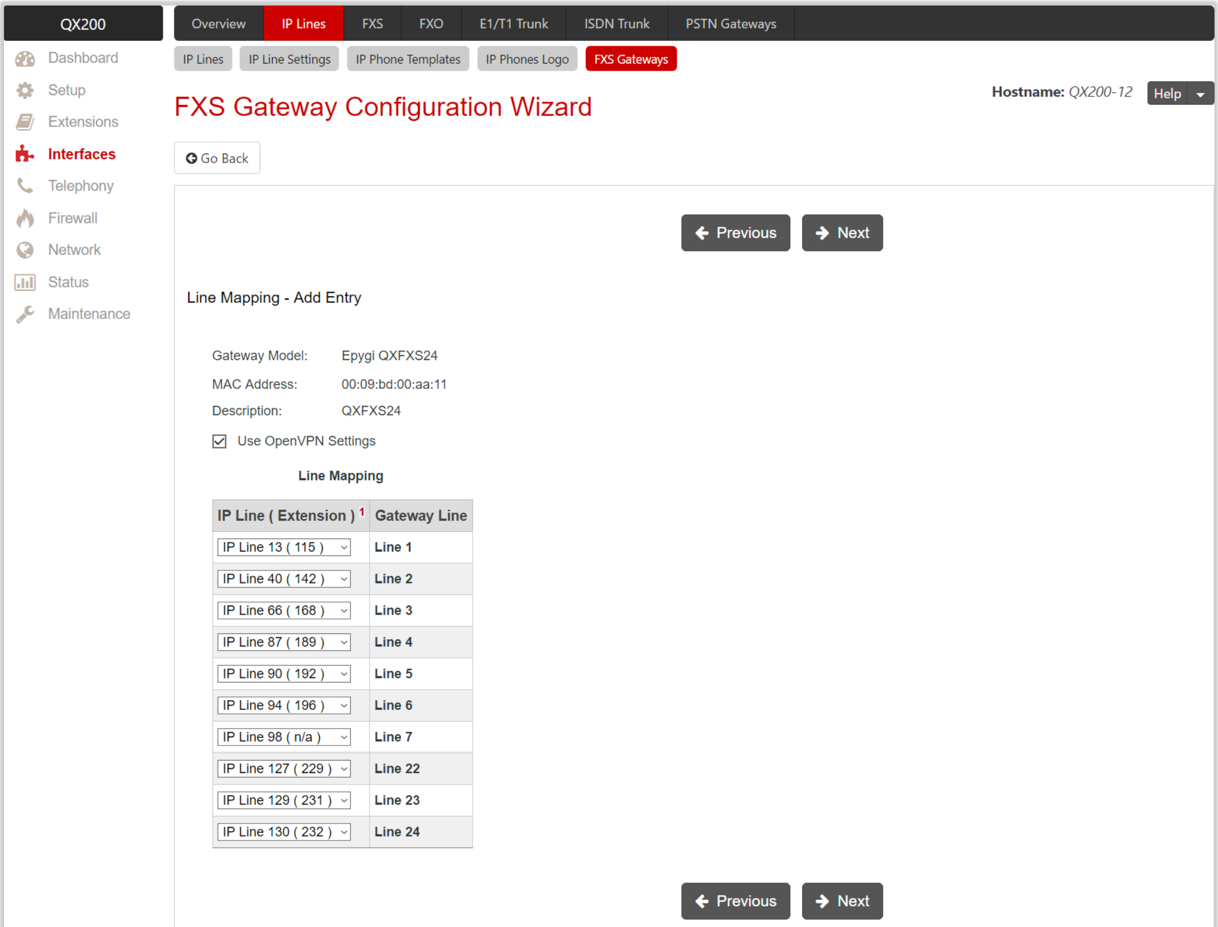


Figure 62: Line Mapping section

Summary

This section displays all configured settings (options) before applying them.

Note: FXS gateway (mapped IP lines) will be added in the IP Lines table after successful configuration. The corresponding routing rules will be added to the Call Routing Table of FXS gateway.

7.2 FXS

7.2.1 FXS (On-board)

The FXS Line page is used to configure on-board FXS lines, define the Caller ID Type, configure Remote Party Disconnect Indication and select the Ringer Type on each of them.

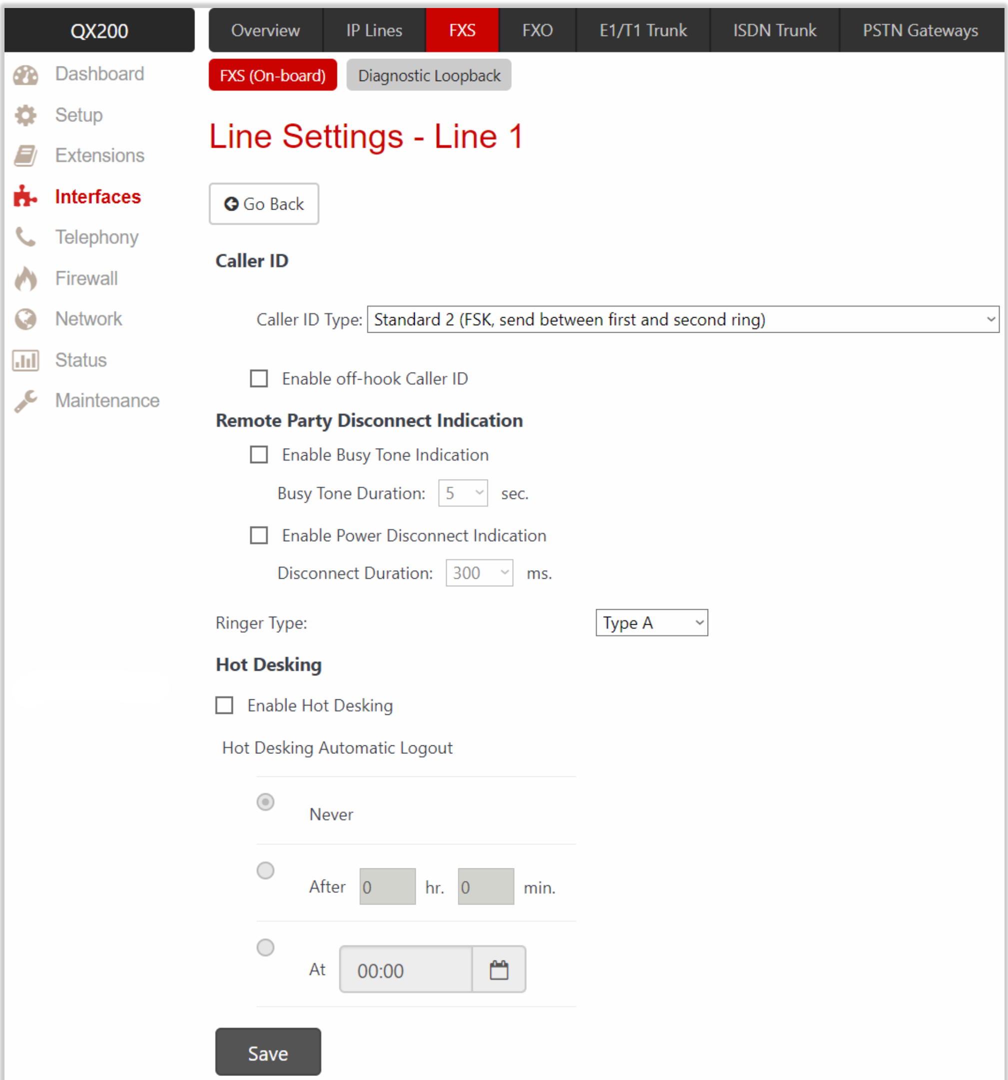


Figure 63: FXS Line Settings page

Click the hyperlinked FXS # to open the Line Settings page to configure specific settings for the selected line. The following settings (options) are available:

- Caller ID Type is used to send the calling party's information to the phone attached to the selected line:
  - No Caller ID
  - FSK, send prior to the first ring
  - FSK, send between the first and second ring
  - FSK, send both prior to a ring and between the first and second ring
  - DTMF, send prior to the first ring
  - DTMF, send between the first and the second ring
  - Combined, send both DTMF prior to the first ring and FSK between the first and the second rings.

Note: The caller ID detection method is different for various types of phones and can be found in the phone manual.

- Enable off-hook Caller ID is used to enable Caller ID transmission to the phone in the off-hook state attached to a certain line. Service is applicable to the phones supporting the Call Waiting Caller ID feature.
- Remote Party Disconnect Indication parameters are used to configure the private PBX attached to the QX FXS port.
  - Enable Busy Tone Indication is used to enable a busy tone transmission to the FXS port when the remote party being called is disconnected. Busy Tone Duration is used to select the period when a busy tone is transmitted to the FXS port.
  - Enable Power Disconnect Indication is used to enable the power cycling on the FXS line when the remote party being called is disconnected. Power Disconnect is applied after the busy tone transmission on the FXS line. Disconnect Duration is used to select the period when the FXS line power will be down.
- Ringer Type is used to select the frequency of ringer supported by the phone attached to the line. The supported ringer type can be found on the bottom of the phone, in the Ren:x.xN value where N is the ringer type supported by the phone. For example, if N=A, the Type A ringer type should be selected, if N=B, the Type B&Z ringer type should be selected.

The Hot Desking section is used to enable and configure the Hot Desking service on the FXS line. Note: The Hot Desking section is the same as for IP line.

Information on the Caller ID system

Caller ID service is used to identify the caller (when establishing a call or sending a voice mail) and notify the called party about the identity of the caller. The Caller ID service is available only for phones with a display to show that information. Two types of Caller ID notifications are available on QX: FSK and DTMF.

FSK Standard

The FSK standard supports caller ID indication either with the phone handset on-hook or if the called party is already busy with another call. For internal calls, caller ID notification in FSK can show up to two lines of identifiable parameters on the called phone's display. The first line shows the caller's extension number. The second line shows the caller's nickname (if indicated in the configuration). For external IP calls, caller ID notification in FSK can also show up to two lines of identifiable parameters on the called phone's display. The first line shows the caller's user name. The second line shows the caller's nickname (if indicated in configuration). If the nickname is not available and there is a display name, provided by the caller party, the second line will display it, otherwise the URL, in the format: username@host will be displayed. For calls from the PSTN network, the entire caller ID message will be shown.

DTMF Standard

The DTMF standard supports caller ID indication only if the phone handset is on-hook (phone is free and ready to accept calls). This standard also has caller ID notification conditions but they are non-configurable. Caller ID notification in DTMF can show only one line of identifiable parameters on the called phone's display. For internal calls, it is the caller's extension number. For external IP calls, it is the caller's user name. For calls from the PSTN network, caller ID will only display the caller's phone number.

Note: DTMF supports only parameters consisting of digits. If any letter symbol has been used in the external caller user name, DTMF will not display caller ID.

7.2.2 Diagnostic Loopback

The **FXS Lines Loopback Settings** page is used to configure the lines for voice loopback diagnostics. When loopback is enabled on the line, incoming call to the line will be automatically picked up after the first ring and voice towards the line will automatically be sent back to the caller (the caller will hear his voice).

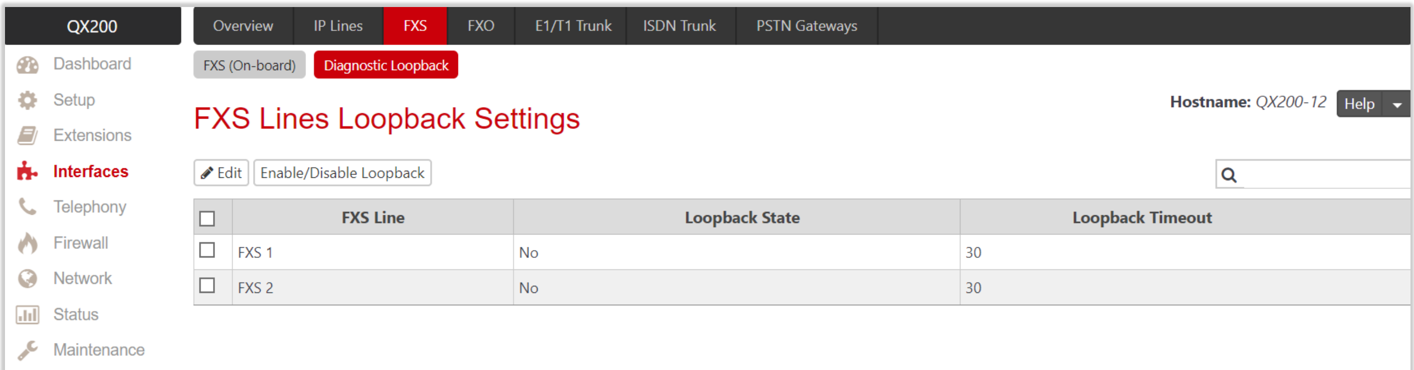


Figure 64: FXS Lines Loopback Settings page

- **Edit** leads to **FXS Lines Loopback Settings – Edit Entry** page to configure the **Loopback Timeout** for the selected FXS line(s).
  - **Loopback Timeout** is used to put a limit on the voice loopback diagnostics duration, i.e. the caller will be disconnected from the QX when the **Loopback Timeout** expires.
- **Enable/Disable Loopback** is used to enable/disable the service on the selected FXS line(s).

7.3 FXO

The **FXO Settings** page is used to configure on-board FXO Lines to make PSTN calls through the on-board FXO ports.

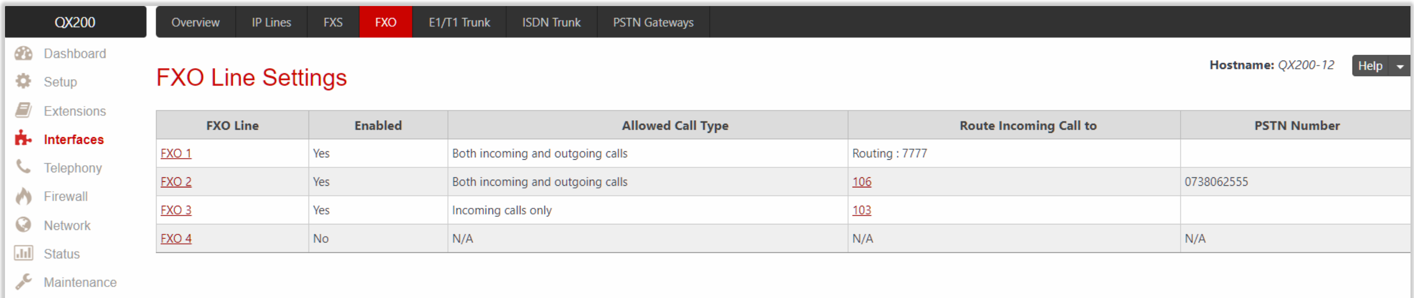


Figure 65: FXO Settings page

Click the hyperlinked **FXO #** to open the **FXO Settings – FXO #** page to configure specific settings of the selected line. The following settings (options) are available:

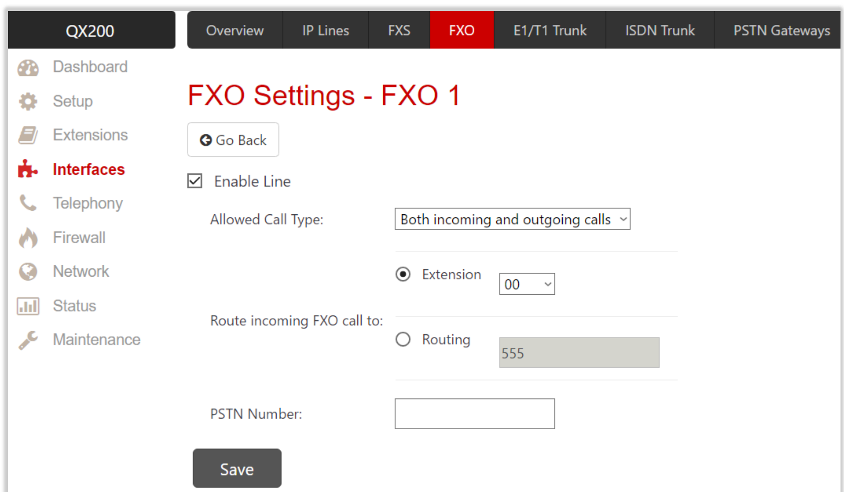


Figure 66: FXO Line Settings page

- **Enable Line** is used to activate FXO line.
- **Allowed Call Type** is used to select the allowed call directions for the FXO line. The following options are available:
  - ➢ **Both incoming and outgoing calls** will be allowed through the selected FXO line.
  - ➢ **Incoming calls only** will be allowed through the selected FXO line.
  - ➢ **Outgoing calls only** will be allowed through the selected FXO line.
- **Route incoming FXO Call to** is used to define the destination where the incoming calls will be forwarded to.
  - ➢ **Extension** is used to forward the calls to either PBX user extension or auto attendant extension. The calls will be forwarded to **Voice Mailbox** if an inactive extension is chosen.
  - ➢ **Routing** is used to forward the calls to the destination defined through the **Call Routing Table.** Enter the routing pattern that will be used for forwarding purposes.
- **PSTN Number** is used to enter any descriptive information, if needed.

For more information on how to configure and use **FXO** lines, refer to the Manual-II: Administration Guide for QX Gateways.

7.4 ISDN Trunk

The Integrated Services Digital Network (ISDN) involves the digitization of the telephone network, which permits voice, data, text, graphics, music, video, and other source material to be transmitted over existing telephone wires. The ISDN Basic Rate Interface (BRI) service offers two B channels (voice transfer) and one D channel (signaling data transfer). The BRI B-channel service operates at 64 kbit/s and is meant to carry user data. The BRI D-channel service operates at 16 kbit/s and is meant to carry control and signaling information, although it can support user data transmission under certain circumstances.

For more information on how to configure and use ISDN trunks, refer to the Manual-II: Administration Guide for QX Gateways.

The ISDN service allows QXISDN4+ act in the following modes:

- network – if connected to a private PBX
- user – if connected to the ISDN trunk from the Central Office (CO). The QXISDN4+ supports the Multiple Subscriber Number (MSN) service, i.e., thus it can be subscribed to multiple numbers from the CO allowing to place two simultaneous calls at a time.

The ISDN Trunk Settings page is used to configure the ISDN trunks and their signaling parameters.

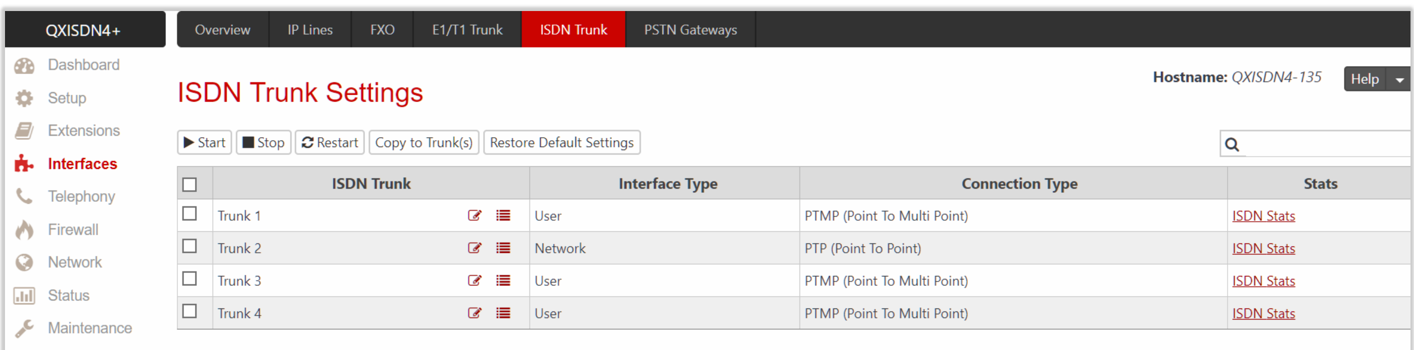


Figure 67: ISDN Trunk Settings page

The following buttons are available:

- Start and Stop are used to start/shutdown the selected ISDN trunk(s). When an ISDN trunk is in a shutdown state, ISDN calls cannot be placed or received.
- Restart is used to bring channel(s) to the initial idle state on both sides, any active traffic on the channel(s) will be terminated.
- Copy to Trunk(s) is used to copy the settings of the selected trunk to another trunk(s).
- Restore Default Settings is used to restore the default settings of the selected ISDN trunk(s).
- Click the Incoming Interdigit Service icon to configure dial plan for incoming ISDN calls from CO/PBX to the QX.
- Click the Modify ISDN Trunk icon to run the ISDN wizard to configure the ISDN trunk settings.

7.4.1 ISDN Wizard

Click the **Modify ISDN Trunk** icon to run the ISDN wizard to configure the selected trunk settings. In general, the wizard consists of the following sections:

- ISDN Settings
- MSN Settings
- Routing Settings
- ISDN Low Level Settings
- L2 & L3 Settings
- Summary of ISDN Settings

ISDN Settings

This section is used to select the interface type and the connection type of the selected trunk.

- **Trunk** shows the selected trunk number.
- **Interface Type** allows to select between the **User** and the **Network** options. If the ISDN trunk is connected to the CO, then the **User** option should be selected. If the trunk is connected to legacy PBX, then the **Network** option should be selected.
- **Connection Type** allows to select between the PTP and PTMP connection types.
  - **PTP** (Point to Point) – in case of connection to the CO, **User** interface type is selected. TIP: No other ISDN device should be connected to ISDN trunk. In case of connection to the legacy PBX, **Network** interface type is selected. TIP: No other ISDN devices should be connected to ISDN trunk. In both cases, with this selection, QX sets the TEI to **manual** mode, assigning the default value (0). TIP: If needed, that value can be changed from the **Advanced Settings** section.

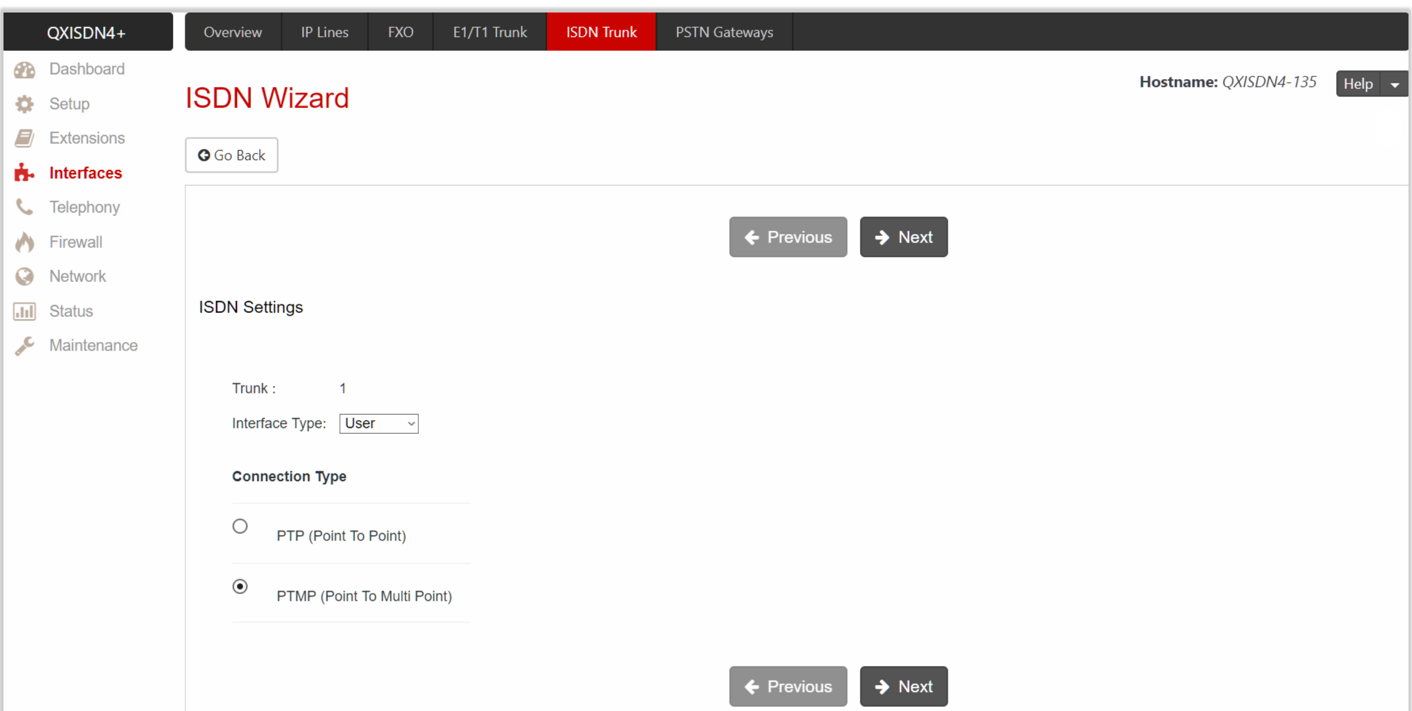


Figure 68: ISDN Settings section

- PTMP (Point to Multi Point) – in case of connection to the CO, User interface type is selected. In case of connection to the legacy PBX, Network interface type is selected. In both cases, with this selection, QX sets the TEI to automatic mode.

MSN Settings

This section is used to turn on the MSN configuration. The section becomes available only in case of User interface type. It is recommended to enable the MSN when there are multiple ISDN devices connected to the same ISDN bus.

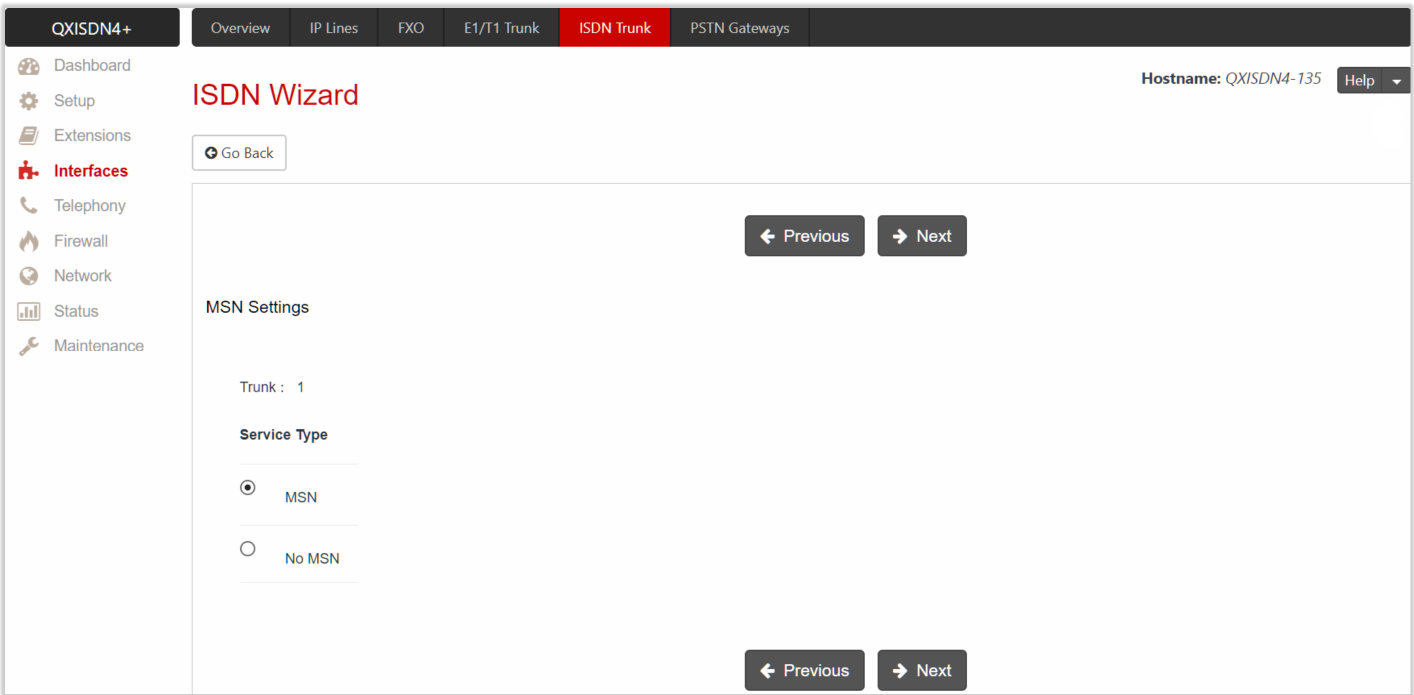


Figure 69: MSN Settings section

Routing Settings

The content of this section depends on the interface type and service type selected from the previous sections of the wizard.

- Trunk displays the selected trunk number.
- Routing Settings – if MSN service is enabled, this section is used to assign MSN numbers to the certain destinations on QX.
  - The fields in the MSN Number column require the MSN numbers allocated to QX. At least one MSN number should be defined.
  - Route Incoming Call to is used to define the destination where the incoming calls addressed to the certain MSN number will be forwarded to. The following options are available:
    - The calls can be forwarded to either user extension or auto attendant.
    - Routing with inbound destination number is used to forward the calls to the destination defined through Call Routing Table.
- Routing Settings – if MSN service is disabled or the selected interface type is Network, this section has only one Route Incoming Call to option.
- Use Default outgoing Caller ID is used to overwrite the source Caller ID with the one specified in this field.
- Advanced Settings – tick this checkbox if you want to adjust L2 and L3 Settings of the trunk manually in the next section.

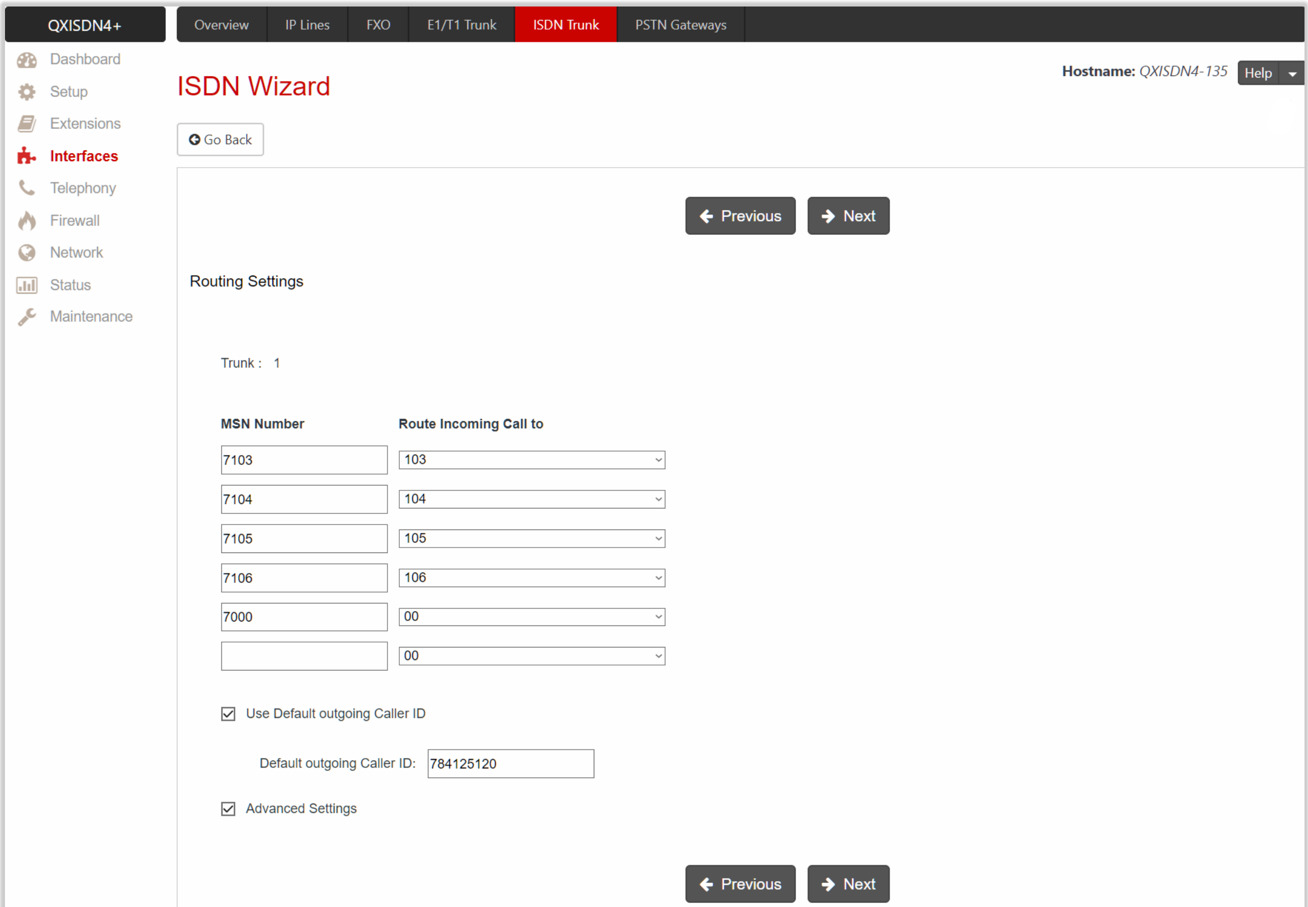


Figure 70: Routing Settings section

ISDN Low Level Settings

This section is used to enable Power Source option. The section becomes available only in case of Network interface type.

- Trunk displays the selected trunk number.
- Power Source – if selected, the QX will supply power for the connected ISDN phones, otherwise ISDN phones should have their own power supplies. TIP: Power Source option should be always disabled when a legacy PBX or Telecom is connected to the QX.

L2 & L3 Settings

This section is used for the advanced configuration of L2 and L3 settings. The section becomes available only if the Advanced Settings checkbox is ticked on the previous section. The following options are available:

- Trunk displays the selected trunk number.
- Excessive Ack. Delay T200 is used to configure the period between the transmitted signaling packet and the acknowledgement received.
- Idle Timer T203 is used to configure the period for the ISDN client idle timeout.
- T302 Timer indicates the time frame, system will wait for digits to be dialed after which the system initiates the call.
- T309 Timer is used to configure call steadiness during link disconnection. If the value in this field is 0, the T309 timer will be disabled.

- T310 Timer is used to configure the outgoing call steadiness when CALL PROCEEDING is already received from the destination but call confirmation (ALERT, CONNECT, DISC or PROGRESS) has not arrived yet.
- Alert Guard Timeout is used to set the value for the Alert Guard Timer between CALL PROC and ALERT messages. Alert Guard Timer is used when QX is connected to an old legacy PBX. Recommended values are:
  - fast connection (0ms)
  - normal (150ms), default
  - slow ISDN-PBX (350ms)
  - very slow ISDN-PBX (500ms)
- Coding Type is used to select between a-law and mu-law coding types.
- Keep ISDN Layer 1 UP is used to force ISDN layer 1 connection to always stay active.
- Passive Mode is used to leave the ISDN Layer1 connection in the Slave mode. If selected, Layer1 remains idle when calls are not available, otherwise QX keeps its Layer1 always active.
  - Enable TEI Remove Procedure – if selected, the trunk will lose the assigned TEI when entering into passive mode on the Layer 2.
  - Permanent TEI Value – if selected, the trunk will keep the assigned TEI when entering into passive mode on the Layer 2 or when QX detected ISDN link DOWN signal from carrier.

Note: These options are available only for PTMP connection type. If PTP connection type is selected, these two options are replaced with a TEI Address option which requires the channel number for connection establishment between the CO and the ISDN client.

- Switch Type – this configuration parameter depends on the Service Provider when acting in the User mode and the legacy PBX capabilities when acting in the Network mode.
- Channel Selection is used to select between the Preferred and Exclusive B channel selection methods. For Preferred channel selection, the CO answers to the call request by the first available timeslot. With the Exclusive channel selection, the CO should feedback only by the timeslot asked in the call request.
- Bearer Establishment Procedure allows to select the session initiation method on B channels. The transmission path completion prior to receipt of a call acceptance indication can be selected:
  - on channel negotiation at the destination interface
  - on progress indication with in-band information
  - on call acceptance

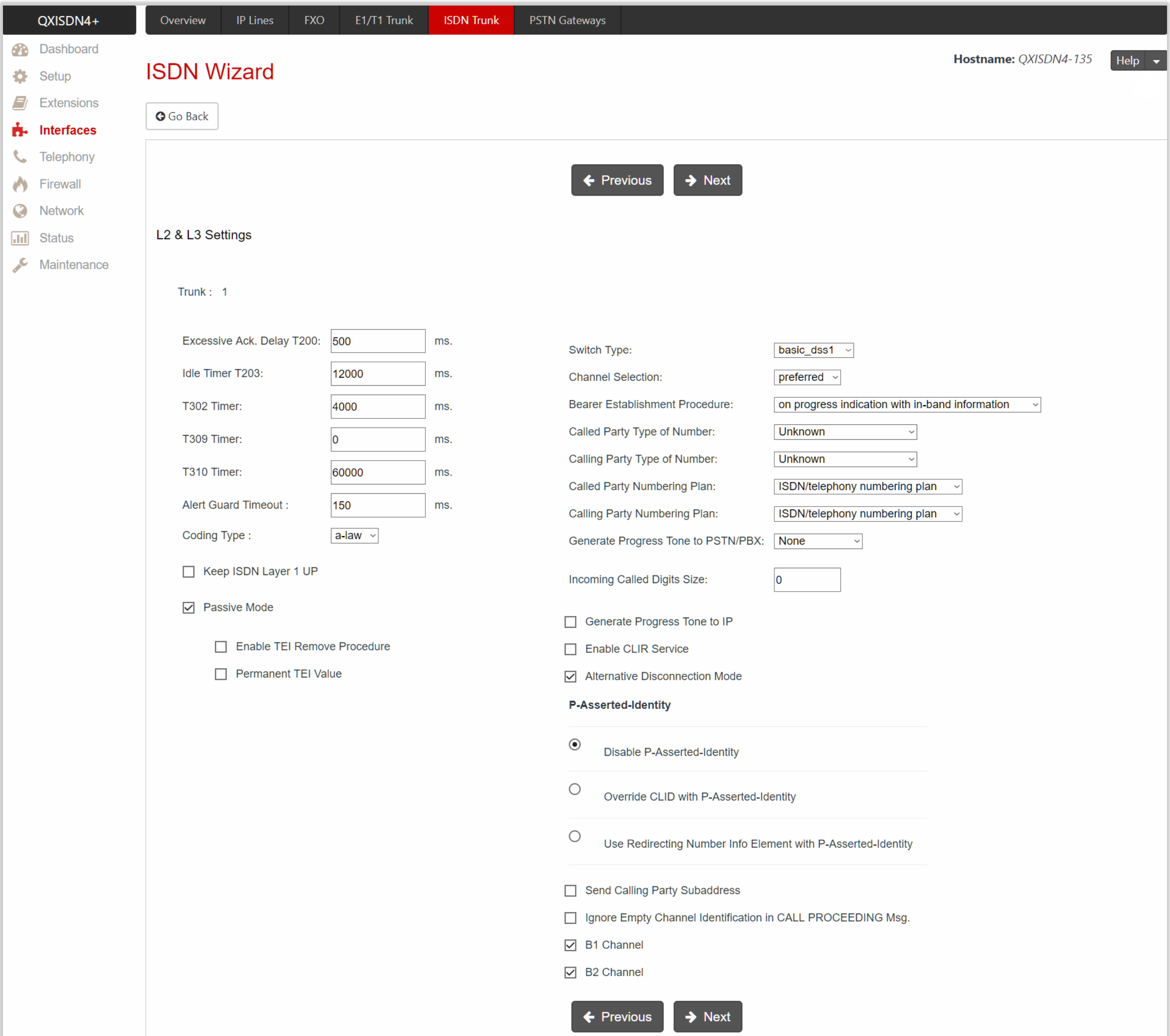


Figure 71: ISDN Low Level Settings section

- Called Party Type of Number allows to select the type identifying the sub address of the called party.
- Calling Party Type of Number allows to select the type identifying the origin of a call.
- Called Party Numbering Plan and Calling Party Numbering Plan are used to select numbering plans.
- Generate Progress Tone to PSTN/PBX contains the options for sending progress (ring-back) tone to callers from the PSTN/PBX. The following options are available in the list:
  - None is used to configure the system to send ALERT messages without the Progress Indicator Information Element.
  - Unconditional is used to configure the system to send ALERT/PROGRESS messages with the Progress Indicator Information Element. With this option, the system will send its own progress tone.
  - Conditional is used to configure the system to send ALERT/PROGRESS messages with Progress Indicator Information Element. With this option, the system will send its own progress tone only if there is no early media (180/183 with SDP) from the called party.
- Incoming Called Digits Size indicates the number of received digits required to establish a call. When this field has 0 value, system uses either the timeout defined in the T302 field or the Sending Complete

Information element messages to establish a call. Independent on the value in this field, Sending Complete Information element and # always cause the call establishment.

- Generate Progress tone on IP – if selected, the progress tone to IP (SIP) will be generated.
- Enable CLIR Service – if selected, Calling Line Identification Restriction (CLIR) service will be activated and this will display the incoming caller ID only in case if Presentation Indication is allowed on the remote side. Otherwise, if CLIR service is disabled, caller ID will be unconditionally displayed.
- Alternative Disconnection Mode – if not selected, QX will disconnect the call as soon as disconnect message has been received from the peer. Otherwise, the QX user may hear a busy tone when the peer has been disconnected.
- P-Asserted-Identity is used to configure P-Asserted-Identity for the calls from SIP to ISDN and vice-versa.
  - Disable P-Asserted-Identity is used to disable the P-Asserted-Identity for both incoming and outgoing calls.
  - Override CLID with P-Asserted-Identity enables the SIP P-Asserted-Identity support. For the calls from SIP to ISDN if the Invite SIP message contains a P-Asserted-Identity or a P-Preferred-Identity or a Remote-Party-ID, then the Caller ID on ISDN is sent with the original Caller ID. The latter comes from the identity field. SIP user agent should check for the existence of the P-Asserted-Identity, then the P-Preferred-Identity, then the Remote-Party-ID to fill out the identity field. For the calls from ISDN to SIP with restricted Caller ID, the SIP Invite message contains P-Asserted-Identity field with the value from the Caller ID on ISDN. The "SIP From" field contains anonymous.
  - Use Redirecting Number Info Element with P-Asserted-Identity radio button selection enables full support of the SIP P-Asserted-Identity. For the calls from SIP to ISDN, if the SIP Invite message contains a P-Asserted-Identity or a P-Preferred-Identity or a Remote-Party-ID, then the Caller ID on ISDN contains the number from the user name field and the Redirecting Number IE contains the original number from the identity field. SIP user agent should check for the existence of the P-Asserted-Identity, then the P-Preferred-Identity, then the Remote-Party-ID to fill the identity field. For the calls from ISDN to SIP with Caller ID, the SIP Invite message contains P-Asserted-Identity field with the original number value from the Redirecting Number IE on ISDN. The "SIP From" field contains the value from the user name.
- Send Calling Party Subaddress – if selected, QX will send the extension number as sub address and the value defined in the Default outgoing Caller ID field as caller ID on the outgoing call. Otherwise no sub address information will be sent and the caller ID will be defined according to the selection of the Use Default Outgoing Caller ID checkbox. Caller ID information, along with the Subaddress, can be displayed on the phone display depending on the phone and PBX settings and capabilities.
- Ignore Empty Channel Identification in CALL PROCEEDING Msg. – if selected, QX will ignore the empty ISDN L3 Channel Identification information element in CALL PROCEEDING message and will not response with STATUS message, otherwise QX will respond with STATUS message on empty Channel Identification information element.
- B1 Channel and B2 Channel enable/disable timeslots for voice transfer. Disabling the timeslot will prevent both incoming and outgoing calls.

Summary of ISDN Settings

This section displays all configured settings (options) before applying them.

7.4.2 ISDN Stats

The ISDN Trunk Status page shows information on the link state, transfer and error statistics. The following sections are available:

General Information

This section contains the following components:

- Active Calls shows the currently active calls.
- Outgoing Calls shows the total amount of outgoing calls (historical data).
- Incoming Calls shows the total amount of incoming calls (historical data).
- Last Time Cleared shows the date and time when the ISDN Stats has been manually cleared last time. TIP: Click the Clear Statistics button, to reset the statistics counters.

Layer 1 - Trunk Settings and Link Status

This section contains the following components:

- Link shows the ISDN link state: up or down.
- Frame Synchronization shows the signal synchronization state in the trunk: Yes or No.

Layer 1 - HDLC Statistics

This section contains the following components:

- HDLC Receive shows the number of packets received in HDLC format.
- HDLC CRC Error shows the number of packets received with CRC errors.
- HDLC Packet Abort shows the number of aborted packets received.
- HDLC Transmit shows the number of packets transmitted in HDLC format.
- HDLC Octet Count shows the number of error packets received in HDLC format.

Layer 2 Settings

This section contains the following components:

- TEI Value shows the actual TEI assigned value.
- L2 State shows the state of BRI L2.

Layer 2 - Transfer Statistics

This section contains the following components for received and transmitted packets:

- Information Frame shows the signaling packets for call initiation and termination.
- Receive Ready shows the control packets when the ISDN link is up.
- Receive Not Ready shows the control packets when unable to accept calls.
- SABME shows the packets when establishing connection.
- Disconnected Mode shows the packets when connection is being terminated.
- Disconnect shows the packets when connection is terminated.
- Unnumbered Acknowledgement shows the packets when accepting connection, call establishment and termination.

- Framer shows the packets as a report of an error condition.
- TEI Request shows the packets containing TEI to initiate subscription of the device at the network.
- Unnumbered Information Frame shows the broadcast signaling packets received for call initiation and termination.
- Exchange Identification shows the received packets containing connection management settings.

Layer 2 - Error Statistics

This section contains the following components:

- Incorrect Length shows the packets with incorrect length.
- Bad Supervisory Frame shows the packets with incorrect supervisory header.
- Bad Unnumbered Information Frame shows the packets with incorrect unnumbered information frame header.
- Bad Frame Type shows the packets with bad frame type.
- Bad Unnumbered Frame shows the packets with incorrect unnumbered acknowledgement frame header.
- Foreign TEI Value shows the packets with bad or foreign TEI value.

7.5 E1/T1 Trunk

QXs don't have on-board E1/T1 ports. Connect QXE1T1 gateway(s) to use the shared E1/T1 trunks.

For more information on how to configure and use E1/T1 trunks, refer to the Manual-II: Administration Guide for QX Gateways.

7.6 PSTN Gateways

The PSTN lines (FXO, E1/T1 or ISDN) of the QX gateway(s) can be shared with QX IP PBXs.

The **Shared PSTN Gateways** page is used to create accounts for the **slave** QX gateway(s) to connect it to the **master** QX IP PBX for **PSTN line sharing** (FXO lines, E1/T1 and/or ISDN trunks).

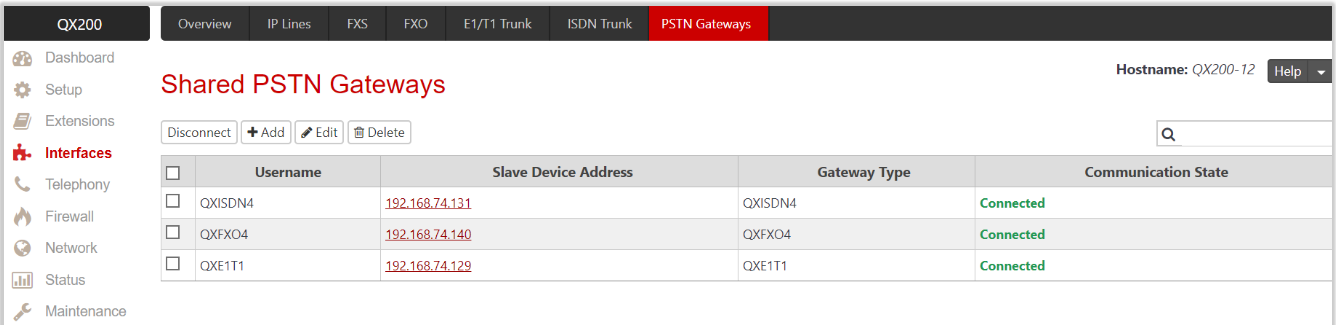


Figure 72: Sharedl PSTN Gateways page

To connect QX gateway to QX IP PBX and share the PSTN lines of the gateway:

1. Click **Add** and enter the following information:
   - **Username** and **Password** are used to set the authentication parameters. TIP: The **Username** and **Password** should match on both master and slave for the successful **PSTN line sharing**.
   - Click **Save** to add the new entry to the **Shared PSTN Gateways** table.
2. The QX will start listening connection requests from slave device.
3. Make corresponding configurations on QX gateway to establish **master-slave** connection. Once the **slave-master** connection is successfully established, appropriate routing rules will be created on the **Call Routing Table** for both devices (slave and master) to support **PSTN line sharing**.
4. Click **Disconnect** to disconnect the slave device from the QX. Note: The slave device will not be reconnected automatically. You need to manually reconnect the slave device to QX from slave's WEB GUI.

For more information on how to configure and use **QX gateway(s)** with **QX IP PBX** in **Share mode**, refer to the Configuring QX Gateways with QX IP PBXs in Sharing Mode guide.

8 Telephony Menu

Figure 73: Telephony Menu overview

8.1 VoIP Carrier

QX supports the SIP trunking service from VoIP providers. This solution allows QX users to make cost saving calls to the global PSTN. VoIP Carrier Wizard simplifies the QX configuration with the SIP trunking services from VoIP providers.

Moreover, for many of industry leaders in VoIP business, the pre-configured templates are included in the QX configuration, allowing one-touch setup for SIP trunking services from these providers. For each SIP trunking service, the wizard automatically creates a specific IP-PSTN type call routing rule in the Call Routing Table. Additionally, a Virtual Extension is automatically generated in the Extensions Management table and registered on the SIP server of VoIP provider. The settings of that extension will be used to make calls towards the configured SIP trunks.

Commonly, just after finishing the wizard, QX users will be able to place calls to the PSTN using the carrier SIP trunk, as well as receive calls. Only in some rare cases some extra configuration should be done.

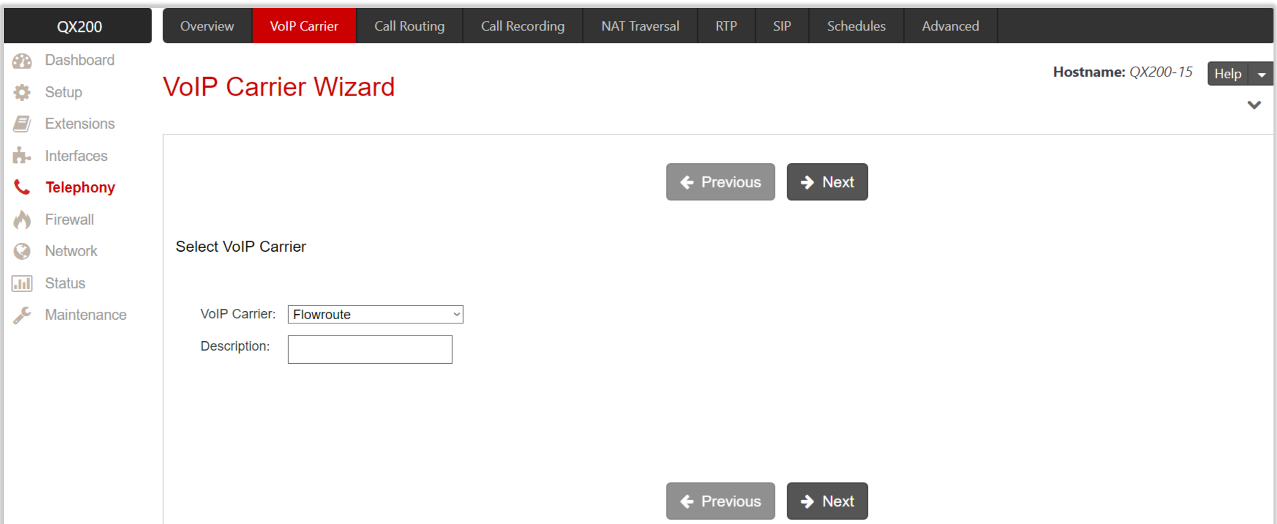


Figure 74: Select VoIP Carrier section

The VoIP Carrier Wizard consists of the following sections:

- Select VoIP Carrier
- VoIP Carrier Settings
- VoIP Carrier Access Code
- Summary

Select VoIP Carrier

This section is used to select a carrier from the VoIP Carrier list. Once the carrier is found and selected, the carrier SIP Server and SIP Port will automatically appear on the next section of the wizard. The Manual option selection allows to configure the VoIP Carrier Settings manually from scratch.

VoIP Carrier Settings

This section is used to define and configure the account from provider. The following settings (options) are available:

- Authentication by IP Address – if selected, deactivates the Account Name and Password fields, thus allowing to skip the account settings. This option is intended for the VoIP carriers requiring IP address authentication instead of account authentication and will be available if Manual option has been selected in the previous section.
- Account Name is used to set the username for authentication on the carrier SIP server.
- Password is used to set the password for authentication on the carrier SIP server and confirm it in the Confirm Password field.
- SIP Server is used to set the IP address or hostname.
- SIP Server Port is used to set the SIP server port.
- Use RTP Proxy – if selected, the RTP (audio) streams between external users will be routed through the QX, otherwise RTP packets will move directly between peers. This option is applicable only when a route is used for calls towards a configured carrier from a peer located outside the QX.
- Authentication Username is used to set an identification parameter to reach the SIP server. It should be provided by the SIP trunking service and may be requested only for certain SIP servers. Commonly, this field should be left blank.

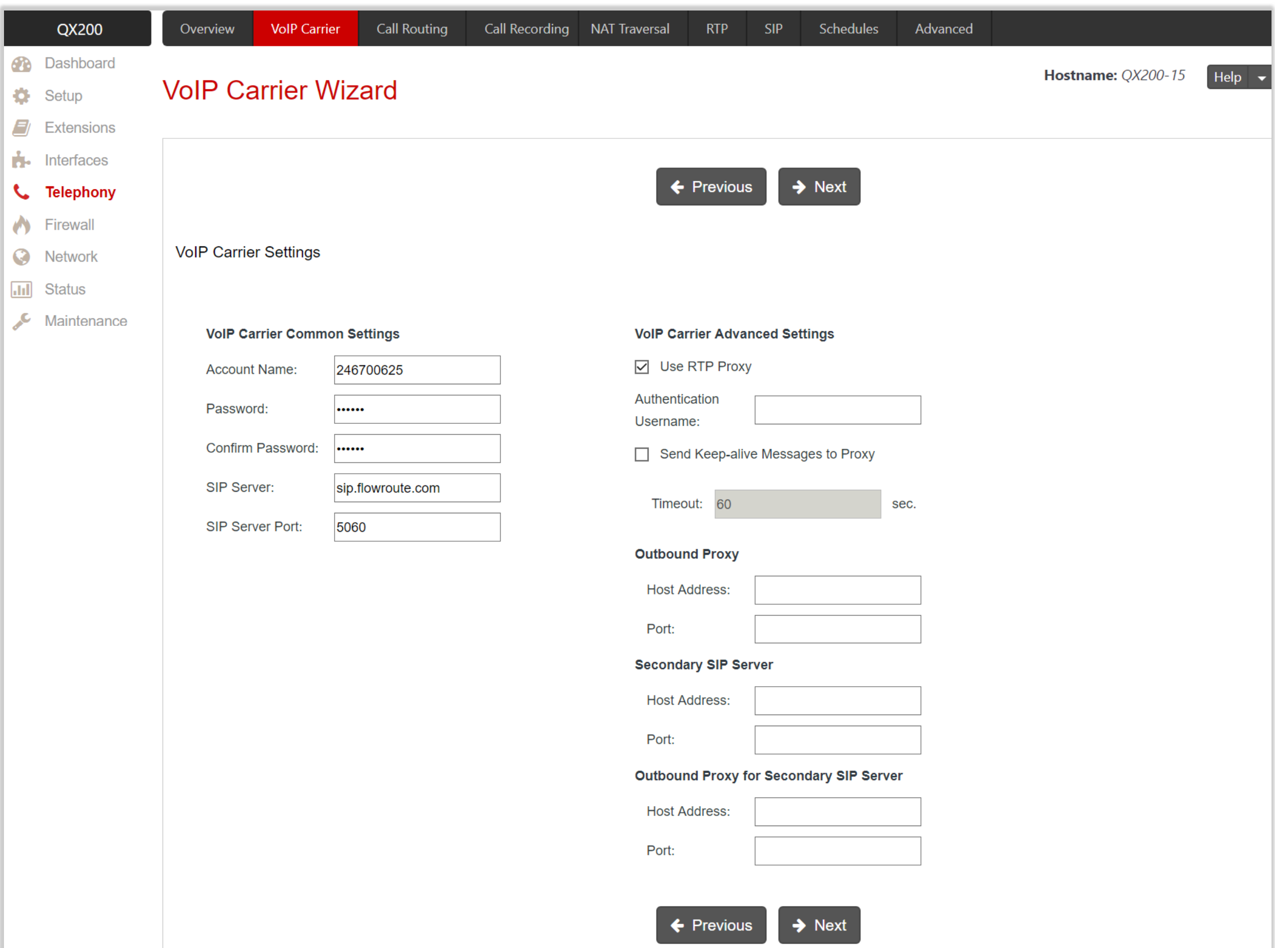


Figure 75: VoIP Carrier Settings section

- Send Keep-alive Messages to Proxy enables the SIP registration server accessibility to the verification mechanism. Timeout is used to set the timeout between two attempts of SIP registration server accessibility verification. If a response is not received from the primary SIP server within this timeout, the secondary SIP server will be contacted. When the primary SIP server recovers, SIP packets will continue to be sent to the server.
- Define the Outbound Proxy, Secondary SIP Server and Outbound Proxy for Secondary SIP Server by entering the Host Address and Port for each of them respectively. These settings are provided by the SIP Trunking service and are used by the QX to reach to the selected SIP servers.

VoIP Carrier Access Code

This section is used to define the routing rules for outbound/inbound calls through the SIP trunks.

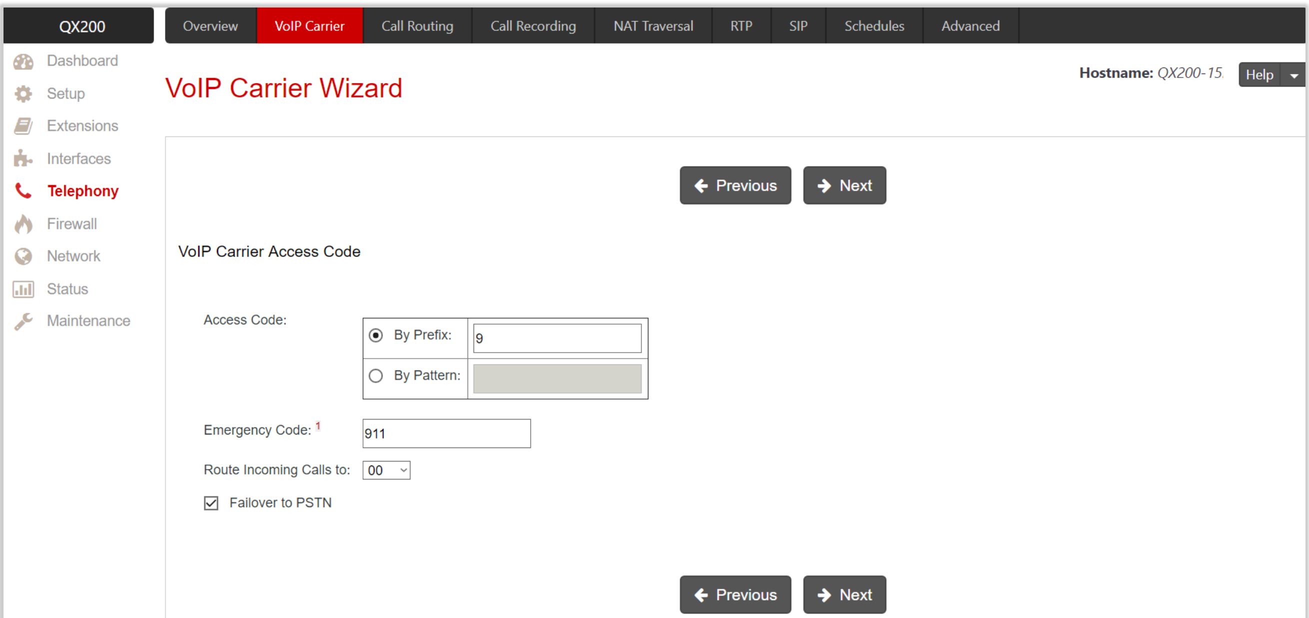


Figure 76: VoIP Carrier Access Code section

The following settings (options) are available:

- Access Code is used to define the routing rule for outbound calls.
  - By Prefix is used to specify the numeric prefix that should be dialed to route call through the SIP trunks. The system will route all digits matching this prefix to the SIP trunks.
  - By Pattern is used to specify the pattern that should be dialed to route call through the SIP trunks. If an outbound call has a destination number that matches the specified pattern, it will be completed according to the current rule.
- Emergency Code is used to set the emergency code supported by the specified VoIP provider. By default, this field is filled with the information defined in the System Configuration Wizard. It also allows to set the carrier specific emergency codes. In case your system has both local PSTN emergency codes and IP-PSTN codes configured, when dialing the certain emergency code, QX will first try to reach the local PSTN allocated emergency, and if failed will dial the IP-PSTN emergency.
  TIP: If the defined VoIP service is 911 compliant then you have to bind this account with the geographical address of your device. If the provider is not 911 compliant, then the public safety agency will not be able to determine the address automatically.
- Route Incoming Calls to is used to select an extension (user extension or auto attendant) on QX where the incoming calls from the configured carrier should be routed to. The unconditional call forwarding is

configured and activated automatically. This setup will be used to forward incoming calls from the VoIP carrier to the selected extension.

- Failover to PSTN – if selected, an additional entry will be added to the Call Routing Table to route calls to the PSTN network through the QX on-board PSTN lines in case if the carrier SIP trunks are not available.

Summary

This section displays all configured settings (options) before applying them.

8.2 Call Routing

8.2.1 Call Routing Table

All calls from/to QX are being processed according to call routing rules that specify the destinations based on the dialed number. When dialing a number, QX matches that number against the Destination Number Pattern among the available call routing records. If the dialed number matches a pattern, then the record with respective pattern will be used to set up the call.

Call Routing Table allows to create and manage call routing rules for different type of calls and destinations. Call Routing Table lists the settings of all call routing rules (records) either generated manually or added automatically with one of the system wizards of QX: Call Routing Wizard, System Configuration Wizard or VoIP Carrier Wizard.

Note: Based on the Emergency Codes and PSTN Access Codes Settings, the automatically added records in the Call Routing Table will be marked in bold and placed in the first position of the table. Additionally, they cannot be modified and deleted from the Call Routing Table. To remove these rules, pass through the System Configuration Wizard and remove them from the Emergency Codes and PSTN Access Code Settings section.

All calls from QX extensions, as well as some calls from external sources, are being routed in QX according to call routing rules that specify the destination based on the dialed number. When user dials a number, QX matches the dialed number against the destination number patterns in the call routing rules.

For more information on how to configure and use Call Routing Rules, refer to the Call Routing on QX IP PBXs guide.

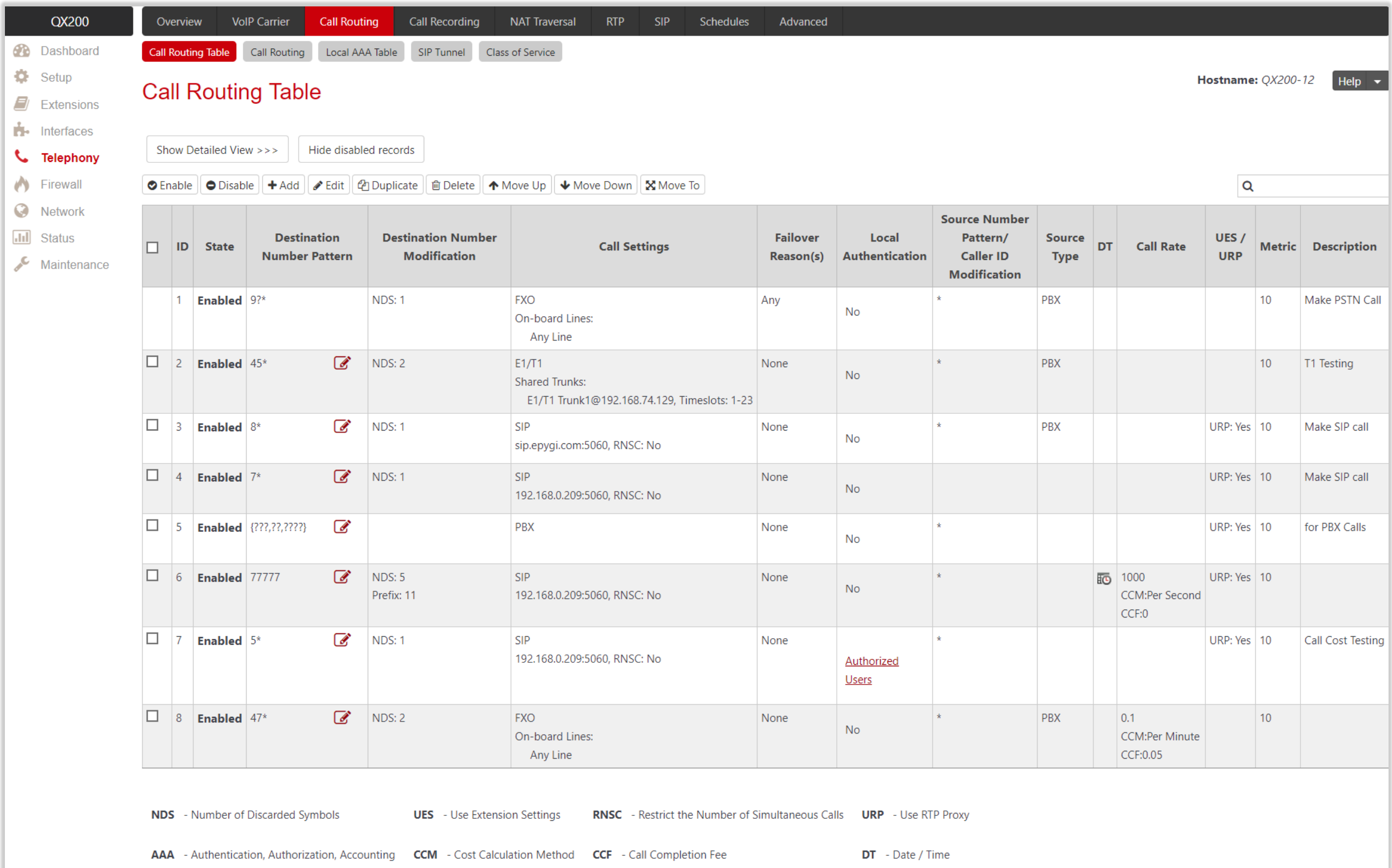


Figure 77: Call Routing Table (brief view) page

Click **Add** to run **Call Routing Wizard** and configure a new call routing rule. In general, the wizard consists of the following sections:

- Destination Call Type
- Call Settings
- Filter on Source / Modify Caller ID
- Date / Time Settings
- Overall Call Duration Limit
- Calling Rate Settings
- Tracing / Debug Options
- Call Alert Settings
- Summary

Destination Call Type

This section contains the following components:

- **Enable Record** is used to enable the call routing rule.
- **Destination Number Pattern** is used to specify a template for filtering out the calls that can be routed via respective call routing rule. If destination number of the call matches with a specified pattern, then the call can be completed via respective call routing rule.

- Number of Discarded Symbols is used to specify the number of digits/characters/symbols that should be removed from the beginning of the destination number after matching it against Destination Number Pattern. Leave the field blank, if nothing needs to be discarded.
- Prefix is used to specify the digits/characters/symbols that will be added in front of the destination number after discarding the digits/characters/symbols as described above. Except for single characters or character strings, the following tags can be used for this field:
  - <callerid:range> allows to use the caller ID or its part as a prefix. For example: <callerid:1-3> indicates that the first 3 digits of the caller ID will be considered as a prefix, <callerid:3-end> indicates that the caller ID from its 3rd digit and up to the end will be assigned to prefix.
  - <dialednum:range> allows to use the dialed number or its part as a prefix. For example: <dialnum:1-3> indicates that the first 3 digits of the dialed number will be as assigned to the prefix, <dialnum:1-end> indicates that the dialed number from its 3rd digit and up to the end will be assigned to prefix.
  - aaa,,,bbb allows two-stage dialing. The aaa and bbb are the numbers to call; bbb can also be a series of digits to inject; a comma indicates a delay of one second. For example: 11,,,11018 will call to 11, wait until the call is established, wait for three seconds and then dial/inject 11018. The two-stage dialing is available for FXO, ISDN, and E1/T1call types.
- Suffix is used to specify the digits/characters/symbols that will be added to destination number from the end after discarding the digits/characters/symbols and adding the prefix as described above.
- Call Type is used to select the call destination type. The following call types are available:
  - PBX – local call to QX extension.
  - PBX-Voicemail – call directly to the user extension Voice Mailbox.
  - PBX-Intercom – call to user extension with request to activate the Intercom service.
  - SIP – calls through a SIP server.
  - SIP_Tunnel – calls through an established SIP Tunnel.
  - IP-PSTN – calls through the IP-PSTN provider to the global PSTN network.
  - RTSP – connection to RTSP server. The Number of Discarded Symbols, Prefix and Suffix fields are not available for the RTSP call type.
  - FXO – calls to the PSTN network either through on-board FXO lines or available shared FXO lines.
  - ISDN – calls to the PSTN network either through on-board ISDN trunks or available shared ISDN trunks.
  - E1/T1 – calls to the PSTN network through available shared E1/T1trunk(s).
- Metric is used to set a rating for the routing pattern in a range from 0 to 20. If no value is set, 10 will be used as the default. If two route entries match the dialed string, the routing pattern with the lower metric will be chosen.
- Description is used to enter description, if needed.
- Enabler Key and Disabler Key are digital codes which should be dialed from handset or auto attendant to enable or disable the routing rule. You can set the same Enabler/Disabler key for multiple routing rules (the same key may be used as enabler for one call routing rule, and as a disabler for another one). This will allow to manage several call routing rules with a single key.
  - Require Authorization for Enabling/Disabling – if selected, enter Phone Access Password after the Enabler/Disabler key to pass authorization. TIP: If the password has been entered incorrectly for 3 times, no status changes will be applied to any of the call routing rule(s), even to those which have no authorization enabled.

The following options give additional configuration possibilities:

- Filter on Source / Modify Caller ID puts a limit on the routing pattern availability for selected caller(s) or allows to modify the caller ID. This option is checked off by default.
- Date / Time Settings allows to set a validity period for the routing pattern by setting date/time rules manually or simply assigning a working schedule.

- **Overall Call Duration Limit** allows to control and limit the total calls duration for the routing pattern.
- **Calling Rate Settings** is used to configure calling rate settings.
- **Tracing / Debug Options** allows to enable generating event notifications on the result of using the call routing rule.
- **Call Alert Settings** allows to notify the designated personnel about the emergency calls, as well as calls through the certain call routing rules.

Call Settings

The content of this section strictly depends on the Call Type selected on the previous section.

Call Type – PBX

- **Use RTP Proxy** – if selected, RTP (audio) streams between the peers will be routed through QX. This is applicable when peers are located in different subnets. If not selected, the RTP streams will move directly between peers.
- **Local Authentication** – if selected, the caller(s) will need to pass authentication to make PBX calls.
- **Client Code Identification** – if selected, the **Code Identification** service will be activated. After dialing the destination phone number, the caller may optionally enter * and then the **Identity Code**. The **Identity Code** is an arbitrary digit string entered by the user to identify a specific call or call group. The **Identity Code** is sent with **Call Detail Reports** (CDRs) and might be used by a billing program for grouping the calls having the same code.
- **Check with 3PCC** is used to request a **3-rd party call control** (3PCC) approval before placing a call through the routing rule. If the checkbox is selected and the routing rule is used to place a call, QX sends a request to the 3PCC application to accept or reject the specific call. The call will be placed if the request is accepted, otherwise it will be skipped. In case of no feedback from the call controlling application, the call will be accepted after a timeout defined in the configuration of the 3PCC application.
- **Failover Reason(s)** – the system will use next matching routing pattern(s) to establish the call if the call setup fails due to the failover reasons presented below:
  - **None** – the system will not use next matching routing pattern(s) regardless of the failover.
  - **Busy** – the system will use next matching routing pattern(s) if the dialed destination is busy.
  - **Wrong Number** – the system will use next matching routing pattern(s) if the dialed number is wrong.
  - **Any** – the system will use next matching routing pattern(s) regardless of the failover reason.

Note: The above-mentioned configuration settings (options) are available and applicable for other call types.

Call Type – PBX-Voicemail

Voice Mail Profile is used to define the custom **Voice Mail Profile** name to activate custom **Voice Mail Settings** on the extension. If the extension does not have a profile specified here or the specified profile name is incorrect, the default profile will be used.

Note: Other settings (options) are the same as for Call Type – PBX.

Call Type – PBX Intercom

Play audible signal before Intercom activation – if selected, the audible signal will be played once **Intercom** service is activated.

Note: Other settings (options) are the same as for Call Type – PBX.

Call Type – SIP

- Use Extension Settings is used to select the extension (user extension or auto attendant) the call will be placed from. SIP settings of the selected extension will be used as caller information. If nothing is selected from the list, the original caller information will be kept.
- Keep Original Caller ID – if selected, the called destination will receive the original caller information.
- Add Remote Party ID – if selected, the Remote Party ID parameter will be added in the outgoing Invite message.
- Destination Host is the IP address or hostname of the destination (for a direct call) or SIP server (for calls through the SIP server). TIP: This field is renamed to Modified Destination Host if the configured Destination Number Pattern contains "@" symbol.
- Destination Port is the port number of the destination or the SIP server. TIP: This field is renamed to Modified Destination Port if the configured Destination Number Pattern contains "@" symbol.
- Username and Password are used to set the authentication parameters for the SIP server if needed.
- Restrict the Number of Simultaneous Calls is used to restrict the number of simultaneous calls to the SIP server with the same username. Allowed Call Count is used to set the number of simultaneous calls.
- Use RTP Proxy – if selected, RTP (audio) streams between the peers will be routed through QX. This is applicable when peers are located in different subnets. If not selected, the RTP streams will move directly between peers.
- Single Call Duration Limit is used to limit the duration of the call placed through the routing rule. Maximum Duration is used to set the maximum duration of the call. The call will be disconnected without prior notice if the maximum duration is reached.
- Local Authentication – if selected, the caller(s) will need to pass authentication to make SIP calls.
- Client Code Identification – if selected, the Code Identification service will be activated. After dialing the destination phone number, the caller may optionally enter * and then the Identity Code. The Identity Code is an arbitrary digit string entered by the user to identify a specific call or call group. The Identity Code is sent with CDRs and might be used by a billing program for grouping the calls having the same code.
- Check with 3PCC is used to request a 3PCC approval before placing a call through the routing rule. If the checkbox is selected and the routing rule is used to place a call, QX sends a request to the 3PCC application to accept or reject a specific call. The call will be placed if the request is accepted, otherwise it will be skipped. In case of no feedback from the call controlling application, the call will be accepted after a timeout defined in the configuration of the 3PCC application.
- Failover Reason(s) – the system will use next matching routing pattern(s) to establish a call if the call setup fails due to the failover reasons presented below:
  - None – the system will not use next matching routing pattern(s) regardless of the failover.
  - Busy – the system will use next matching routing pattern(s) if the dialed destination is busy.
  - Wrong Number – the system will use next matching routing pattern(s) if the dialed number is wrong.
  - Network Failure – the system will use next matching routing pattern(s) in case of system overload, network failure or timeout expiration.
  - Other – the system will use next matching routing pattern(s) in case of Server Failure Responses (5xx messages) and Global Failure Responses (6xx messages).
  - Any – stands for all failure reasons mentioned in the Failover Reason(s) group.
- Enable Failover Timeout is used to set the period after which the call can be considered as failed (SIP response message isn't received). The Failover Timeout is used to set the timeout duration (in the range from 1 to 180 seconds). The call will be established through next matching routing pattern(s) after the timeout expires if the failover reason is enabled for the call routing rule.

- SIP Privacy is used to select the security level of the SIP route by means of hiding or replacing (depending on the configuration of the SIP server) the key headers of the SIP messages used to establish the call.
  - Default Privacy – if selected, no QX specific SIP privacy will be applied, and all privacy will be relied on the configuration of the SIP server.
  - Disable Privacy – if selected, SIP call security will be disabled, all headers of the SIP message will be transparently visible to the destination.
  - Enable Privacy – if selected, QX specific SIP privacy will be applied. Selection enables a group of checkboxes to choose the key headers to be fully or partly hidden or replaced. The Require Privacy option is used to restrict the delivery of the SIP message if either of the selected headers cannot be hidden (or replaced, depending on the configuration of the SIP server) before being sent to destination.
- Transport Protocol for SIP messages is used to select the transport protocol (UDP, TCP or TLS) for transmitting the SIP messages.

Call Type – SIP Tunnel

SIP Tunnel is used to select the SIP tunnel to route the calls through tunnel to the remote QX device (QX IP PBXs and QX Gateways).

Note: Other settings (options) are the same as for Call Type – SIP.

Call Type – IP-PSTN

Note: Settings (options) are the same as for Call Type – SIP.

Call Type – RTSP

- RTSP URI is used to define the RTSP server URI for receiving stream(s). Audio and video streams are available depending on RTSP server configuration.
- Username and Password are used to set the authentication parameters for RTSP server if needed.

Note: Other settings (options) are the same as for Call Type – SIP.

Call Type – FXO

- FXO Lines to Use – a group of radio buttons allowing to select whether a specific or any available FXO line will be used to route the call. The following options are available:
  - None – selection means no on-board (local) FXO lines will be used to route the call.
  - Any Line – the call will be established through the first available on-board FXO line.
  - Specific Line – the call will be established only through the selected on-board FXO line.

If QXFXO4 gateway is connected to QX in share mode, the following options will be available:

  - Any Available Line – the call will be established through the first available on-board FXO line, then through shared FXO lines.
  - Any Line@ – the call will be established through the first available shared FXO line.
  - Specific Line@ – the call will be established only through a selected shared FXO line.
- FXO Lines Load Balancing is used to enable load balancing mechanism on FXO lines.
  - None – the system will not apply load balancing mechanism and the call will be routed through the first available FXO line (among the selected ones).
  - Round Robin – the system will apply load balancing mechanism according to internally gained statistics of most used FXO lines, the call will be routed to the less used and currently available FXO line (among the selected ones).

- Single Call Duration Limit is used to limit the duration of the call placed through the routing rule. Maximum Duration is used to set the maximum duration of the call. The call will be disconnected without prior notice if the maximum duration is reached.
- Local Authentication – if selected, caller(s) will need to pass authentication to make FXO calls.
- Client Code Identification – if selected, the Code Identification service will be activated. After dialing the destination phone number, the caller may optionally enter * and then the Identity Code. The Identity Code is an arbitrary digit string entered by the user to identify a specific call or call group. The Identity Code is sent with CDRs and might be used by a billing program for grouping the calls having the same code.
- Check with 3PCC is used to request a 3PCC approval before placing a call through the routing rule. If the checkbox is selected and the routing rule is used to place a call, QX sends a request to the 3PCC application to accept or reject a specific call. The call will be placed if the request is accepted, otherwise it will be skipped. In case of no feedback from the call controlling application, the call will be accepted after a timeout defined in the configuration of the 3PCC application.
- Failover Reason(s) – the system will use next matching routing pattern(s) to establish a call if the call setup fails due to the failover reasons presented below:
  - None – the system will not use next matching routing pattern(s) regardless of the failover.
  - Cannot Establish Connection – the system will use next matching routing pattern(s) if the connection cannot be established.
  - Any – the system will use next matching routing pattern(s) regardless the failover reason.

Call Type – ISDN

- Keep Original Caller ID – if selected, the called party will receive the original caller information (mobile number, PSTN/SIP number, etc.) instead of extension information when call(s) are forwarded.
- ISDN Trunks to Use is used to select a specific or any available trunk to route the call(s). The following options are available only for QXISDN4+:
  - Any Trunk(User) – calls will be established through any ISDN trunk running in User mode.
  - Any Trunk(Network) – calls will be established through any ISDN trunk running in Network mode.
  - ISDN Trunk# – calls will be established through the selected ISDN trunk.

If QXISDN4 gateway is connected to the QX in share mode, the following options will be available:

  - Any Trunk(User)@Any – calls will be established through the first available on-board ISDN trunk running in User mode, then through shared ISDN trunks (running in User mode).
  - Any Trunk(Network)@Any – calls will be established through the first available on-board ISDN trunk running in Network mode, then through shared ISDN trunks (running in Network mode).
  - ISDN Trunk@ – calls will be established through the selected shared ISDN trunk.
  - Any Trunk(User)@ – calls will be established through the first available shared ISDN trunk running in User mode.
  - Any Trunk(Network)@ – calls will be established through the first available shared ISDN trunk running in Network mode.
- Collect Call is used when the calling party wants to place a call at the called party's expense. This service is applicable only if the Collect Call service is enabled on both calling and called party.
- Single Call Duration Limit is used to limit the duration of the call placed through the routing rule. Maximum Duration is used to set the maximum duration of the call. The call will be disconnected without prior notice if the maximum duration is reached.
- Local Authentication – if selected, the caller(s) will need to pass authentication to make ISDN calls.
- Client Code Identification – if selected, the Code Identification service will be activated. After dialing the destination phone number, the caller may optionally enter * and then the Identity Code. The Identity

Code is an arbitrary digit string entered by the user to identify a specific call or call group. The **Identity Code** is sent with CDRs and might be used by a billing program for grouping the calls having the same code.

- **Check with 3PCC** is used to request a 3PCC approval before placing a call through the routing rule. If the checkbox is selected and the routing rule is used to place a call, QX sends a request to the 3PCC application to accept or reject a specific call. The call will be placed if the request is accepted, otherwise it will be skipped. In case of no feedback from the call controlling application, the call will be accepted after a timeout defined in the configuration of the 3PCC application.
- **Failover Reason(s)** – the system will use next matching routing pattern(s) to establish a call if the call setup fails due to the failover reasons presented below:
  - **None** – the system will not use next matching routing pattern(s) regardless of the failover.
  - **Cannot Establish Connection** – the system will use next matching routing pattern(s) if the connection cannot be established.
  - **Any** – the system will use next matching routing pattern(s) regardless the failover reason.

Note: Additional wizard section will be available for ISDN call type to configure trunk timeslots.

- **Select Timeslots** is used to select timeslot(s) which will be used for placing ISDN calls.

Call Type – E1/T1

QX IP PBXs don't have on-board E1/T1 trunks. The E1T1 call type becomes available when QXE1T1 gateway is connected to QX in share mode.

- **Keep Original Caller ID** – if selected, the called party will receive the original caller information (mobile number, PSTN/SIP number, etc.) instead of extension information when the call(s) are forwarded.
- **E1/T1 Trunks to Use** is used to select a specific E1/T1 trunk to route the call(s). The following option is available:
  - **E1/T1 Trunk1@** – the calls will be established through the selected E1/T1 trunk.
- **Collect Call** is used when the calling party wants to place a call at the called party's expense. This service is applicable only if the **Collect Call** service is enabled on both calling and called party.
- **Single Call Duration Limit** is used to limit the duration of the call placed through the routing rule. **Maximum Duration** is used to set the maximum duration of the call. The call will be disconnected without prior notice if the maximum duration is reached.
- **Local Authentication** – if selected, the caller(s) will need to pass authentication to make E1/T1 call.
- **Client Code Identification** – if selected, the **Code Identification** service will be activated. After dialing the destination phone number, the caller may optionally enter * and then the **Identity Code**. The **Identity Code** is an arbitrary digit string entered by the user to identify a specific call or call group. The **Identity Code** is sent with CDRs and might be used by a billing program for grouping the calls having the same code.
- **Check with 3PCC** is used to request a 3PCC approval before placing a call through the routing rule. If the checkbox is selected and the routing rule is used to place a call, QX sends a request to the 3PCC application to accept or reject a specific call. The call will be placed if the request is accepted, otherwise it will be skipped. In case of no feedback from the call controlling application, the call will be accepted after a timeout defined in the configuration of the 3PCC application.
- **Failover Reason(s)** – the system will use next matching routing pattern(s) to establish a call if the call setup fails due to the failover reasons presented below:
  - **None** – the system will not use next matching routing pattern(s) regardless of the failover.
  - **Cannot Establish Connection** – the system will use next matching pattern(s) if the connection cannot be established.
  - **Any** – the system will use next matching routing pattern(s) regardless of the failover reason.

Note: Additional wizard section will be available for E1/T1 call type to configure trunk timeslots.

- Select Timeslots is used to select timeslot(s) which will be used for placing E1/T1 calls.
  - Up to 30 timeslots will be available for placing E1 calls regardless of the signaling type of trunk.
  - Up to 23 timeslots will be available for placing T1 calls if the trunk signaling type is CCS.
  - Up to 24 timeslots will be available for placing T1 calls if the trunk signaling type is CAS.

Radius Authentication and Authorization

The RADIUS Authentication and Authorization options are available for the routing pattern regardless of Destination Call Type, if the RADIUS Client is enabled.

- RADIUS Authentication and Authorization is used to make the caller pass authentication through the RADIUS server to make calls.
- RADIUS Accounting is used to send the CDRs of the calls made through the call routing rule to the RADIUS server.

Filter on Source / Modify Caller ID

The following settings (options) are available:

- **Source Filter** is used to limit the routing pattern availability for selected caller(s).
  - **Source Number Pattern** is used to enter the caller address the routing pattern will be available for.
  - **Source Type** is used to select the caller source type. The following options are available:
    - **Any** – any caller will be able to make calls regardless of caller source type.
    - **PBX** – only PBX extensions will be able to make calls. TIP: The **Class of Service** section will become available to select **Class of Service(s).**
    - **SIP** – only inbound SIP callers will be able to make calls. TIP: The **Source Filter** section will become available to configure **Source Host** address (IP address or hostname).
    - **SIP_Tunnel** – only inbound callers from the selected SIP_Tunnel will be able to make calls. TIP: **Source Filter** section will become available to select **Inbound SIP Tunnel.**
    - **FXO** – only inbound FXO callers will be able to make calls. TIP: The **Source Filter** section will become available to select **Port ID** for FXO call type.
    - **ISDN** – only inbound ISDN caller(s) will be able to make calls. TIP: The **Source Filter** section will become available to select **Port ID** for ISDN call type.
    - **E1/T1** – only inbound E1/T1 caller(s) will be able to make calls. TIP: The **Source Filter** section will become available to select **Port ID** for E1/T1 call type.

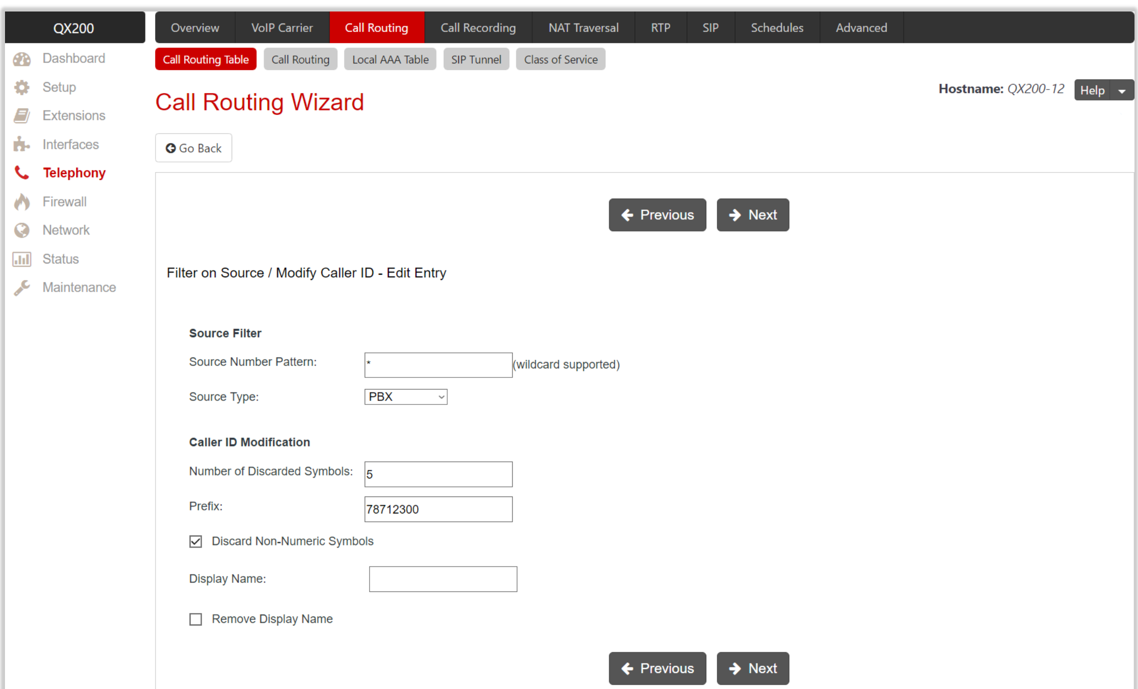


Figure 78: Filter on Source / Modify Caller ID section

- **Caller ID Modification** is used to modify the **Caller ID** before sending them to remote party.
  - **Number of Discarded Symbols** is used to specify the number of digits/characters/symbols that should be discarded from the beginning of the **Source Number Pattern.** Leave the field blank if there is no need to discard the digits.
  - **Prefix** is used to specify the digits/characters/symbols that will be placed in front of the **Source Number Pattern.**
  - **Discard Non-Numeric Symbols** is used to discard any non-numeric symbols/characters from the **Source Number Pattern.**
  - **Display Name** is used to replace an original **Caller ID** with the custom display name.
  - **Remove Display Name** is used to remove **Caller ID.**

Date / Time Settings

This section is used to set a validity period(s) for the routing pattern.

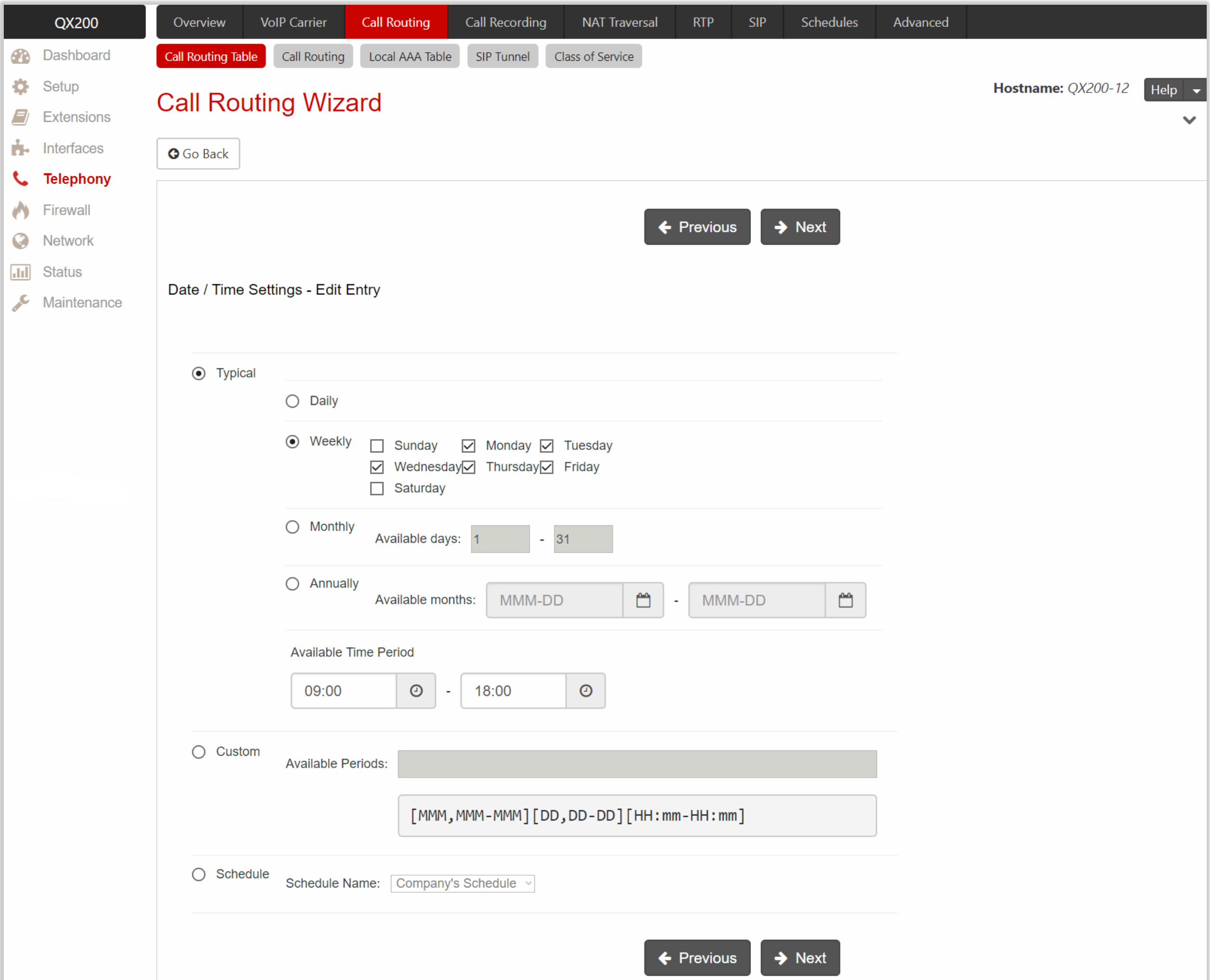


Figure 79: Date / Time Settings section

The following settings (options) are available:

- Typical is used to select one of the validity periods:
  - ➢ Daily – the routing pattern will be available for each day.
  - ➢ Weekly – the routing pattern will be available for the selected weekday(s).
  - ➢ Monthly – the routing pattern will be available for the selected day(s) in each month.
  - ➢ Annually – the routing pattern will be available for the selected day(s) and month(s) for each year.
  - ➢ Available Time Period is used to set the validation time range for the routing pattern.
- Custom is used to manually set the validity period(s). TIP: The entered values need to be in the following format: [MMM,MMM-MMM][DD,DD-DD][HH:mm-HH:mm]
- Schedule is used to apply one of the configured schedules to the routing pattern. Select the desired schedule from the Schedule Name drop-down list.

Overall Call Duration Limit

This section is used to limit and control the total duration of calls through the routing pattern.

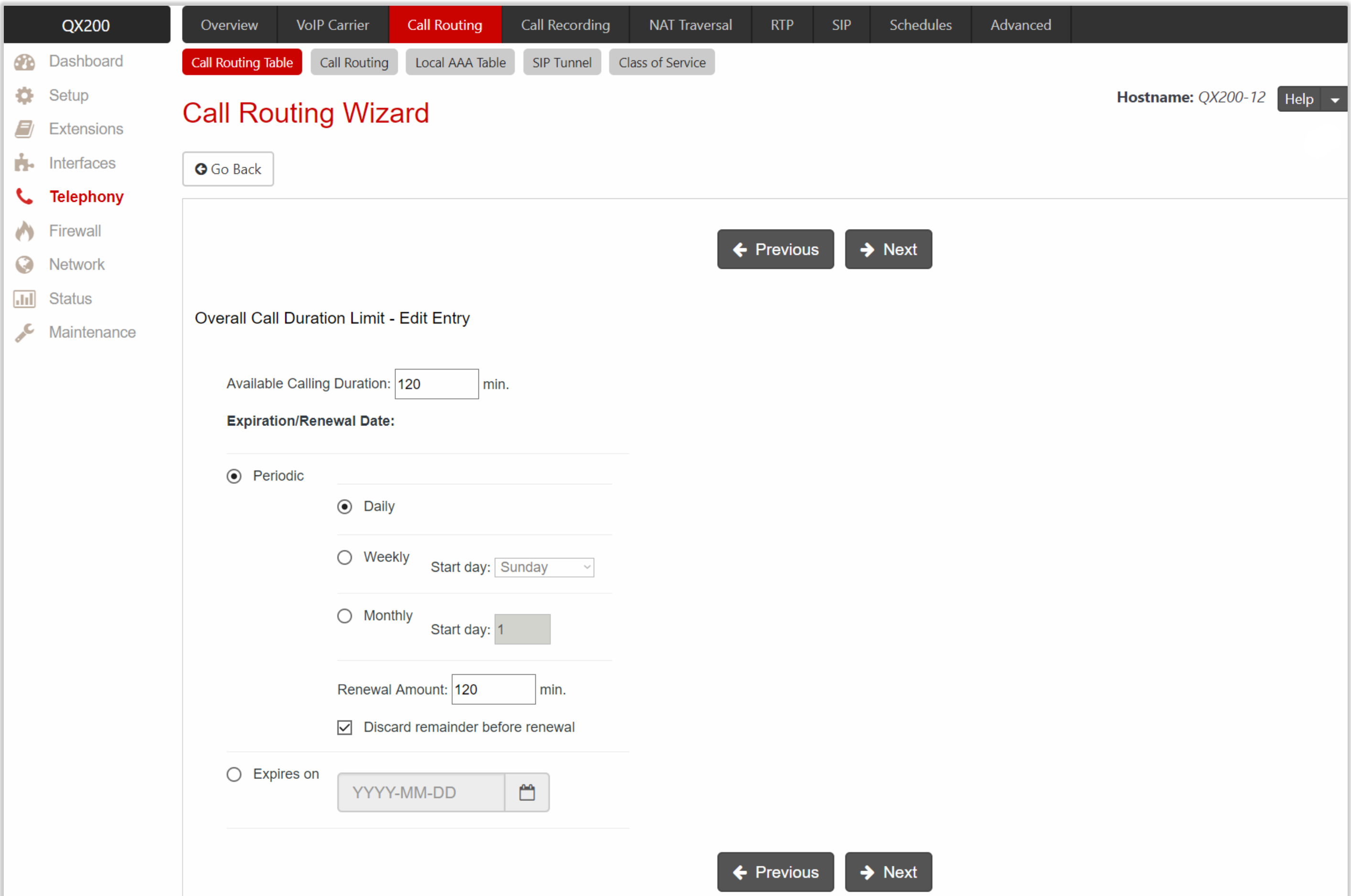


Figure 80: Overall Call Duration Limit section

The following settings (options) are available:

- Available Calling Duration is used to set the total duration for the calls through the selected call routing rule. Once Available Calling Duration expires, the current call will be disconnected without prior notice. Placing new calls through this rule is not possible until Available Calling Duration is not updated either manually or automatically by Renewal Date and Amount.
- Periodic is used to select one of the Renewal Date options:
  - Daily – the defined Available Calling Duration will be renewed every day.
  - Weekly – the defined Available Calling Duration will be renewed every week on a specified weekday.
  - Monthly – the defined Available Calling Duration will be renewed every month on a specified day.
  - Renewal Amount is used to set the renewal amount to be added to the available calling duration when the expiration date of Available Calling Duration is reached. Leave the field blank, if you don't need to renew Available Calling Duration.
  - Discard remainder before renewal is used to discard the remainder of Available Calling Duration before renewal and set Renewal Amount as new Available Calling Duration.
- Expires on is used to set the expiration date for Available Calling Duration. After Expiration Date, the call routing rule becomes unavailable and it is impossible to place a new call until this field is updated.

Calling Rate Settings

This section is used to configure calling rate settings. The following settings (options) are available:

- Call Completion Fee is used to set the cost of a single call, regardless of the call duration. The actual cost of the call depends on the cost calculation method.
- Rate per Minute is used to set the cost of one minute of call. The actual cost of the call depends on the cost calculation method.
- Cost Calculation Method is used to select one of the options:
  - ➢ Per Second – if this method is selected, the call cost is calculated as:

    Call Cost = Call Completion Fee + CDIS/60 x CRPM

  - ➢ Per Minute – if this method is selected, the call cost is calculated as:

    Call Cost = Call Completion Fee + Roundup(CDIS/60) x CRPM.

TIP: If CRPM is equal to 0, flat fee is charged for the call.

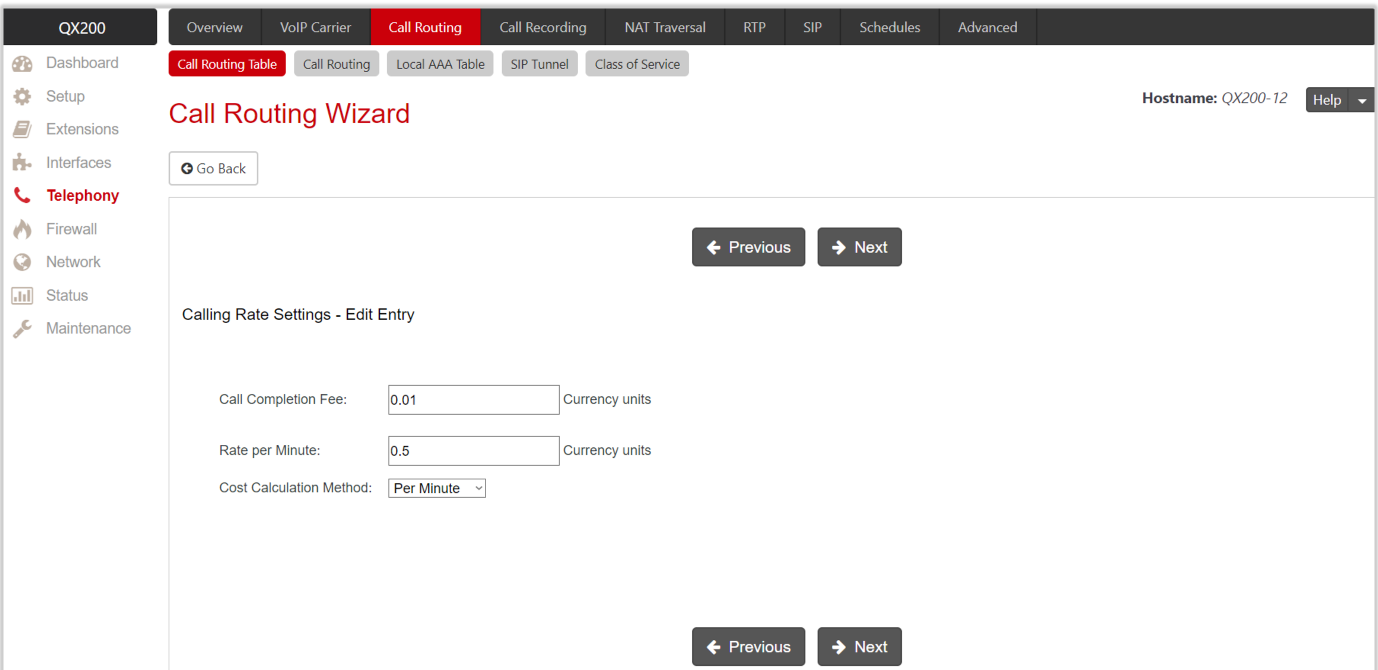


Figure 81: Calling Rate Settings section

The calling credit assigned to extension will be charged in the following scenarios:

- Extension places a call to the destination through a payable routing rule directly.
- Extension transfers incoming call (blind or consultative) to destination through a payable routing rule.
- Extension forwards the call to destination through a payable routing rule automatically.
- Trusted caller uses the Call Relay service on auto attendant to make call to destination passing authentication by the extension credentials (number and the password).

Note:

- The Calling Rate Settings section becomes available once Call Cost feature is activated.
- The Calling Rate Settings is not applicable for PBX, PBX-Voicemail and PBX-Intercom call types.
- The Single Call Duration Limit option cannot be used with Calling Rate Settings.

Tracing / Debug Options

These options are used to generate event notifications on a certain execution result for the call routing rule.

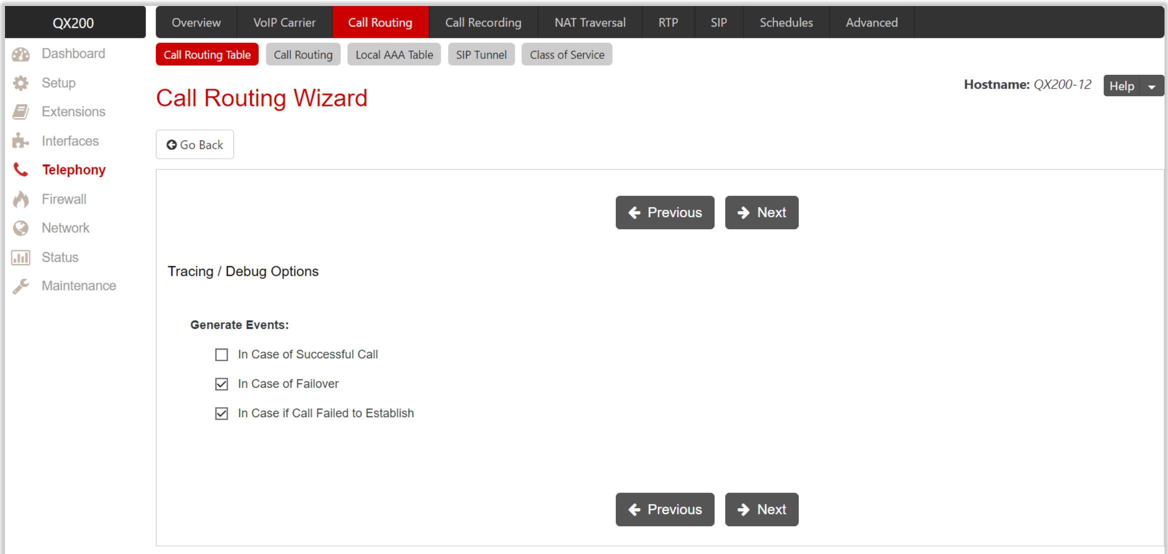


Figure 82: Tracing / Debug Options section

The events will be generated and displayed in the System Events for the following cases:

- In Case of Successful Call
- In Case of Failover
- In Case if Call Failed to Establish

Call Alert Settings

This section is used to activate the call alert service and configure settings. The Call Alert service is used to notify the designated personnel about the emergency calls, as well as calls through certain call routing rules. The following information will be included in the notification: the routing pattern, the extension who placed the call, the dialed number and the call Date/Time.

The following settings (options) are available:

- Generate System Event – this option is used to generate and display event notification in the System Events.
- Send Notification via E-mail – this option is used to send notification to a specified address via e-mail.
- Send Notification via SMS – this option is used to send notification to a specified number via SMS.
- Leave Voice Message – this option is used to leave voice mail on the defined extension(s) with a voice message.

Note: Use commas to separate email addresses, mobile numbers and user extensions in case of multiple entries.

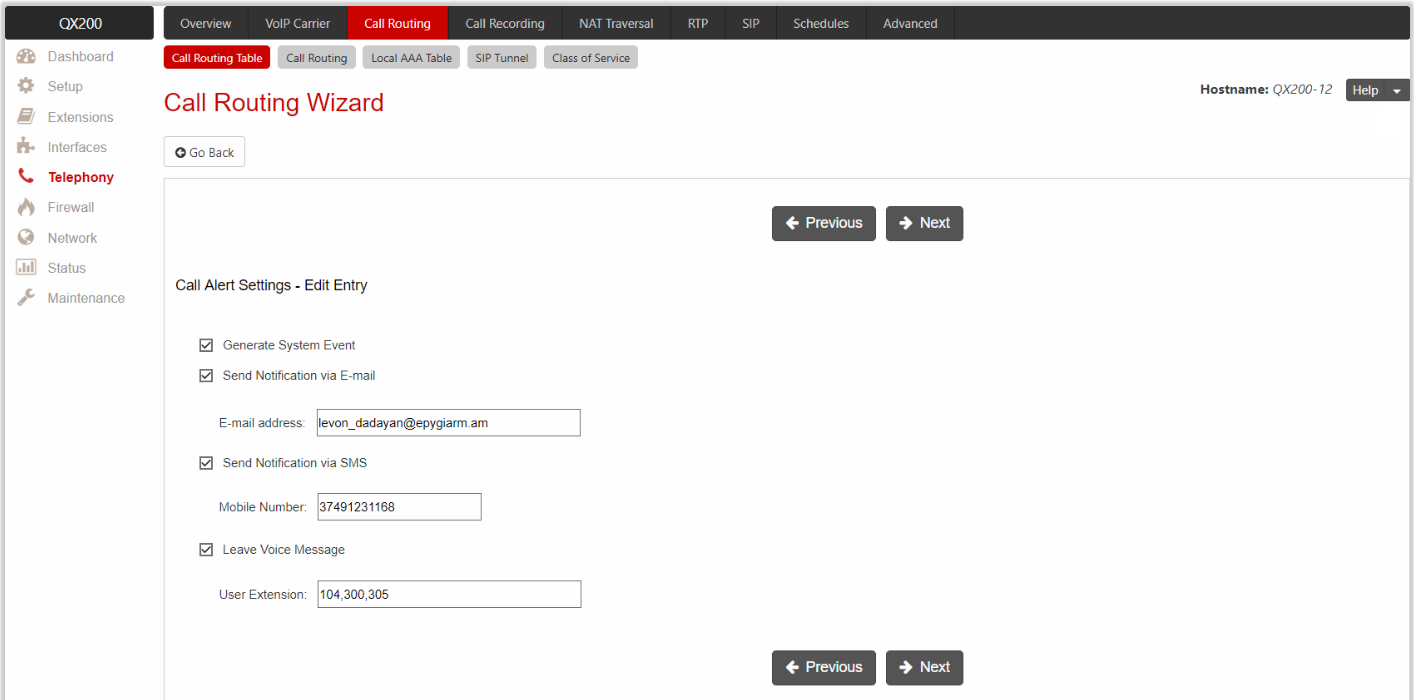


Figure 83: Call Alert Settings section

Summary

This section displays all configured settings (options) before applying them.

8.2.2 Call Routing

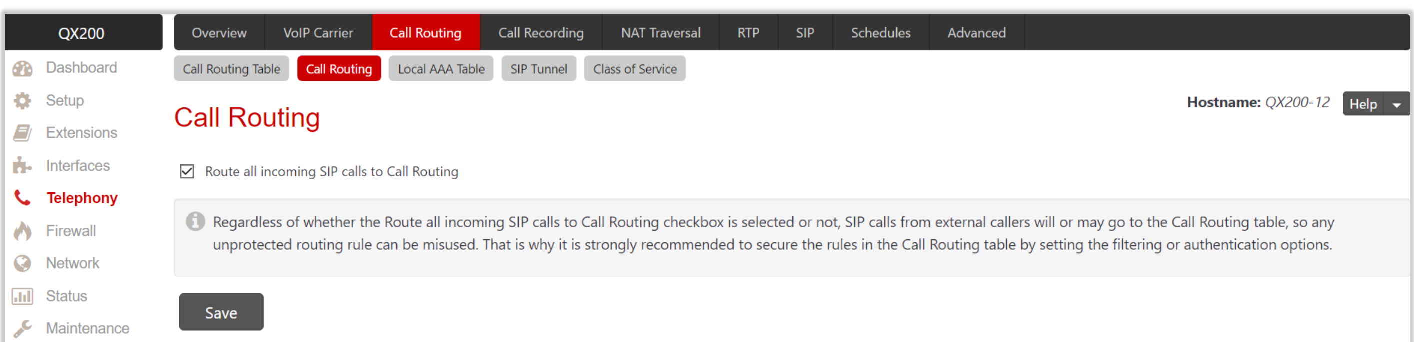


Figure 84: Call Routing page

Route all incoming SIP calls to Call Routing – if selected the system will directly look for a matching call routing rule in the Call Routing Table and ignore the possible matches with the SIP address of extension (Username or DID Number). Otherwise the system will first try to match the SIP address of extension. If it matches, the incoming call will ring on the extension, otherwise the system will look for a matching call routing rule in the Call Routing Table.

Note: Regardless of whether Route all incoming SIP calls to Call Routing is selected or not, SIP calls from external callers will or may go to the Call Routing Table, so any unprotected rule can be misused. That is why it is strongly recommended to secure the rules in the Call Routing Table by setting the filtering or authentication options.

8.2.3 Local AAA Table

The Call Routing – Local AAA Table page is used to configure and manage the local authentication database.

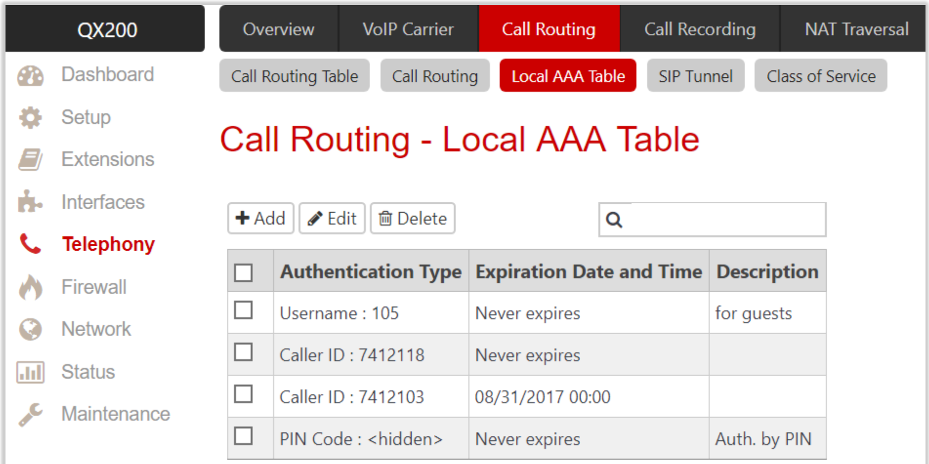


Figure 85: Call Routing - Local AAA Table page

To add a new AAA entry:

1. Click Add and configure the following information:
2. Select one of the Authentication methods.
   - Authentication by Caller ID is used to set the authentication based on the caller phone number or SIP address (Username or DID Number). The system will automatically pass the authentication.
   - Authentication by Login is used to set the authentication based on the Username and Password provided by the user upon login.
   - Authentication by PIN is used to set the authentication based on the PIN Code provided by the user upon login.
3. Configure the Expiration Date and Time, if needed.
   - Expires on is used to set the expiration date for the configured AAA entry.
4. Enter any Description, if needed.
5. Click Save to add the new AAA entry to the Call Routing – Local AAA table.

Authorized Users

If the **Local Authentication** option is enabled on the call routing rule, then the caller should use configured **AAA** entries to pass authentication in order to make calls. The caller will automatically pass authentication if the caller's phone number or SIP address (Username or DID Number) is enabled in the **Authorized Users** table. Otherwise the caller will be asked to login (enter username and password) or enter the **PIN Code**.

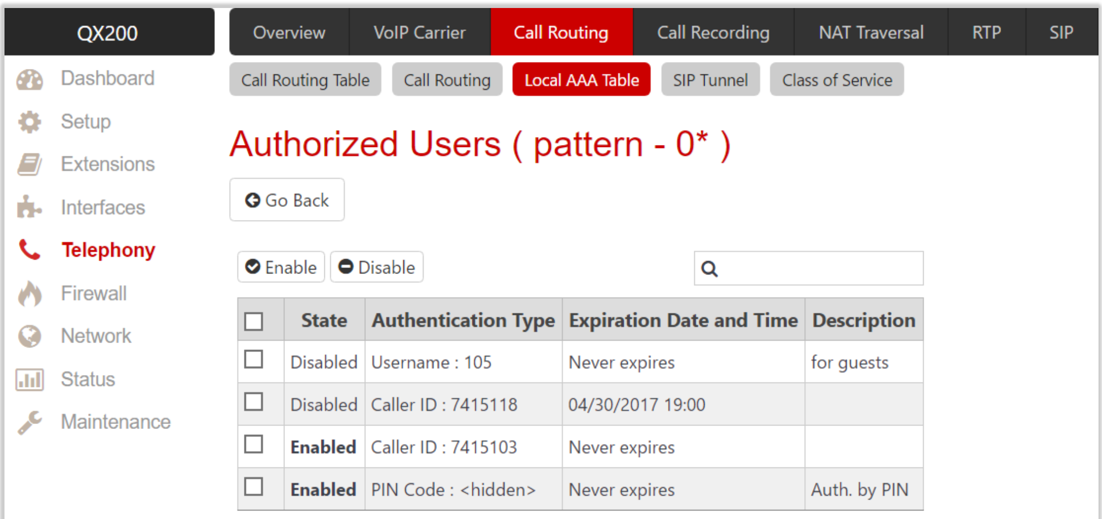


Figure 86: Authorized Users

Note: **Authentication by Login** cannot be combined with **Authentication by PIN** on the same call routing rule.

8.2.4 SIP Tunnel

The **SIP Tunneling** feature provides means for building network on QX IP PBXs. This network is based on many "**slave**" QXs in satellite offices and one or more "**master**" QXs in the main office(s) with SIP tunnels configured between "**slave**" and "**master**" devices. One possible scenario for using SIP Tunneling is routing SIP calls through the remote QX device. Another scenario is building a redundant distributed PBX system based on many slave QXs in satellite offices and two or more master QXs in the main office.

For more information on how to configure and use **SIP Tunnels**, refer to the SIP Tunneling Feature on QX IP PBXs guide.

8.2.5 Class of Service

QX Class of Service (CoS) is used to define the permissions that extensions (User or Conference) will have when using certain call routing rules to make a call.
The CoS provides the ability to set restrictions on the call routing rules for each extension, thus allowing to permit or deny the extensions to use call routing rules

For example, the following restrictions can be applied for extensions:

- Only Internal – internal calls to other extensions on the QX are only allowed from this extension(s). Calls to SIP and PSTN are not allowed.
- Only Local PSTN – calls to the local PSTN are only allowed from this extension(s).
- Long-distance IP-PSTN only – long distance IP-PSTN calls are only allowed from this extension(s).
- International only – international calls are only allowed from this extension(s).

The above defined abstract restrictions, they should be implemented as service classes on the call routing patterns in the Call Routing Table. For example, to implement the long-distance service class, select all call routing rule on the QX that can be used for making long distance PSTN calls and assign them to the Long-distance IP-PSTN only CoS.

Pay close attention to the configuration of call routing rules on QX. To avoid from ambiguities, create a call routing rule for each CoS.

Configure a CoS as follows:

1. Assign the specified CoS(s) to a certain call routing rule(s).
2. Assign the specified CoS(s) to the extension (User or Conference).

Note:

- CoS is applicable only for call routing rules with PBX source type
- If there is no CoS assigned to the call routing rule, that rule will be generally available for any extension (User or Conference) whether it is attached to a CoS or not.
- If the Enable Class of Service option is disabled, call routing rules that are assigned to a certain CoS(s) will be available for any extension (User or Conference), if there are no any other filtering limitations.

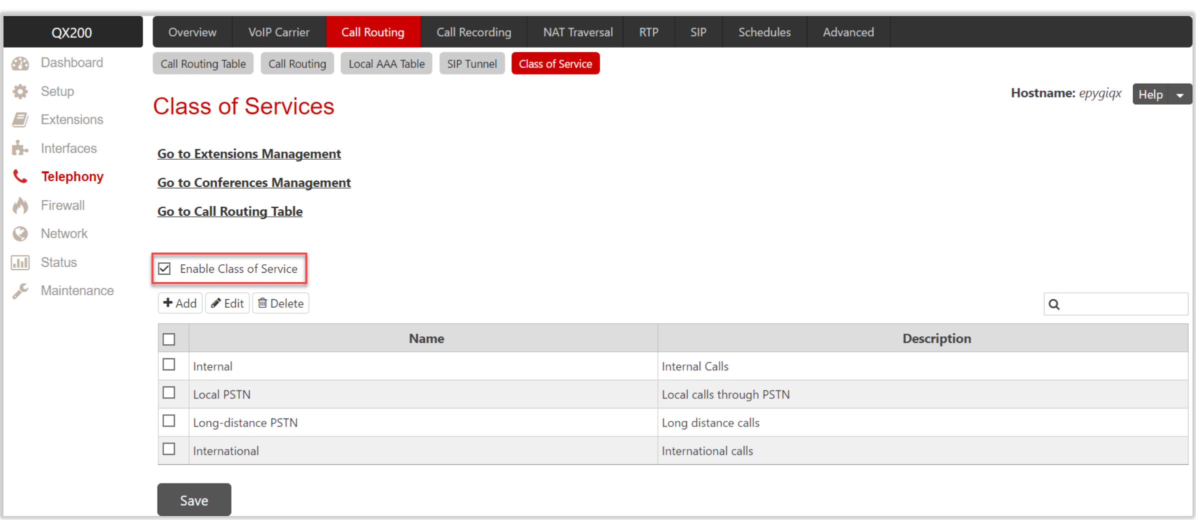


Figure 87: Class of Services page

To create a new Class of Service and activate CoS functionality:

1. Click the Add button.
   - Enter a Name for the CoS.
   - Enter a Description, if needed.
2. Click Save to add the newly created CoS to the Class of Services table.
3. Tick the Enable Class of Service option to activate the Class of Service functionality on the QX.

8.2.6 Best Matching Algorithm

All calls from QX extensions, as well as some calls from external sources, are being routed in QX according to call routing rules (records) that specify the destination based on the dialed number. When a user dials a number, the QX matches the dialed number against the destination number patterns in call routing records.

- If the dialed number matches only with a single pattern, then the rule with respective pattern will be used to set up the call.
- If multiple patterns have been found to match the dialed number, the system uses the Best Matching Algorithm to prioritize the matching patterns.

Once the patterns are prioritized, the rule with the highest priority will be used as a preferred one for call setup. The order of the call routing rules is important. The rule has a higher priority over those located below in the table. Move Up moves the selected rule one level up, increasing the rule priority and the Move Down moves the selected rule one level down, decreasing the rule priority.

Note: The subsequent prioritized pattern will be used only if the destination specified by a pattern with higher priority is unreachable and the Failover(s) is configured.

To prioritize the matching patterns, the following criteria are sequentially applied to matching patterns. The criteria are ordered by their priorities: Each consecutive criterion is calculated only for the patterns that take the same value for the preceding criterion: that is, Criterion 3 is calculated only for patterns that take the same value for Criterion 1 and Criterion 2.

Criteria list

- Criterion 1 shows the available asterisks (*) in a pattern. The patterns without (*) have a higher priority.
- Criterion 2 is the total number of matching digits/symbols inside and outside the braces/brackets. The more matching digits a pattern contains, the higher its priority.
- Criterion 3 is the number of matching digits/symbols outside the braces/brackets. The more matching digits outside braces/brackets a pattern contains, the higher its priority. TIP: This criterion is only used if several patterns take an equal but non-zero value for Criterion 2.
- Criterion 4 is the total number of question marks (?) inside and outside the braces/brackets. The more question marks a pattern contains, the higher its priority.
- Criterion 5 is the number of question marks (?) outside braces/brackets. The more question marks outside braces/brackets a pattern contains, the higher its priority. TIP: This criterion is only used if several patterns take an equal but non-zero value for Criterion 4.
- Criterion 6 is the number of square brackets ([]). The more brackets a pattern contains, the higher its priority.
- Criterion 7 is the number of braces ({}). The more braces a pattern contains, the higher its priority.
- Criterion 8 is the number of asterisks (*). The fewer asterisks a pattern contains, the higher its priority.
- Criterion 9 is the value of the metric. The lower the metric of a pattern is, the higher its priority.
- Criterion 10 is the position in the routing table. The higher the position of a pattern in the routing table is, the higher its priority.

For example: User dials 1231, the following matching patterns are found in the Call Routing Table.

Pattern Position	Routing Pattern
1	*1*
2	123*
3	{11–15}3*
4	??1
5	[1–3]*
6	{100–150, asd, *\?}1
7	1[1–3]3[0–8]
8	123?
9	*2*1
10	*

Table 2: Example – The list of Patterns

Step 1: The list is sorted and the patterns with asterisks (*) are pushed back to the end of the list, due to lower priority (Criterion 1).

Position after Step1	Routing Pattern
1	?2?1
2	{100–150, asd, *\?}1
3	1[1–3]3[0–8]
4	123?
5	*1*
6	123*
7	{11–15}3*
8	[1–3]*
9	*2*1
10	*

Table 3: Example – The list of Patterns after the Step 1

Step 2: The list is sorted and the patterns with fewer number of matching digits inside and outside the braces/brackets are pushed back to the end of the list, due to lower priority (Criterion 2). The patterns that contain the same number of matching digits are grouped into sub-lists.

Position after Step2	Routing Pattern	Matching Digits
1	1[1–3]3[0–8]	4
2	{100–150, asd, *\?}1	4
3	123?	3
4	{11–15}3*	3
5	123*	3
6	?2?1	2
7	*2*1	2
8	[1–3]*	1
9	*1*	1
10	*	0

Table 4: Example – The list of Patterns after the Step 2

Step 3: Each consecutive criterion is calculated only for the patterns that take the same value for the preceding criterion: that is Criterion 3 is calculated only for patterns that take the same value for Criterion 1 and Criterion 2.

The list is sorted and the patterns with the fewer number of matching digits outside the braces/brackets are pushed back to the end of the list, due to lower priority (Criterion 3).

Position after Step2	Routing Pattern	Matching Digits
1	1[1–3]3[0–8]	2
2	{100–150, asd, *\?}1	1

Table 5: Example – The list of the Patterns after Step 3

The Best Matching Algorithm will stop after executing Step 3 and the dialed number 1231 will pass through 1[1-3]3[0-8] routing pattern.

8.2.7 Allowed Characters and Wildcards

Below is the complete list of the characters and wildcards supported in the QX. Not all characters and wildcards are supported for all options and settings. Thus, depending on the meaning of the option some limitations can be applied.

Characters

- Numbers – 0…9
- Letters – A…Z, a…z
- Special symbols – =; +; -; $; / ; ~ ; _ ; – ; . ; & ; ( ) ; ' ; ! ; * ; ? ; {} ; [ ]

Note:

- The symbols (*, ?, -, ! and ,) should be prefixed with a slash (\) symbol if they are used as ordinary characters; otherwise the system will interpret them as wildcards.
- The symbols !; { }; [ ]; – and , are used to define a range of characters and cannot be used as ordinary characters.

Wildcards

- * – any number of any characters
- ? – any single character
- {} – a character or a string from the specified set of characters and strings
- [] – a character from the specified set of characters and strings
- Note: You can use the wildcard ? within the braces, but not *.

The following control symbols are used to specify a set:

- Use a comma (,) to separate the elements of a set. For example: The pattern is: 9{1,3,11,a}. Numbers matching the pattern will be: 91, 93, 911, 9a. Note: No spaces are allowed within braces.
- Use a minus sign (-) to specify a range of characters. Each successive element of the range is obtained by increasing the previous element (the element code) by one. For example: The pattern is: 2{11-15,a-d}5 Numbers matching the pattern will be: 2115, 2125, 2135, 2145, 2155, 2a5, 2b5, 2c5, 2d5.
- Use an exclamation point (!) to exclude a character or a string from a set. For example: The pattern is: 2{11-15,a-d,!14,!c}5. Numbers matching the pattern will be: 2115, 2125, 2135, 2145, 2155, 2a5, 2b5, 2d5. Note: The exclamation point (!) cannot be used to exclude a range of symbols.
- Use a slash (\) before control symbols (*, ?, -, ! and ,) to use them as an ordinary character. For example: The pattern is: 1*[1–3]. Numbers matching the pattern will be: 1*1, 1*2, 1*3

- Use an at sign (@) to indicate full SIP address (for example: 20233@sip.epygi.com). This pattern is mainly used to call back users registered on the SIP server different from the one where the called party is registered. Note: Patterns containing @ symbol will not be parsed among those that do not have @ symbol in the Call Routing Table. When calling from local extensions (the calling number for PBX extension is sip_number@ip_address_of_QX, e.g. 20233@192.168.35.25), only the sip number part of the pattern will be parsed among other entries with @ symbol in the Call Routing Table.

Allowed SIP Addresses

Calls over IP are implemented based on Session Initiating Protocol (SIP) on QX. When making a call to a destination that is somewhere on the Internet, a SIP address must be provided.

SIP address needs to be entered in one of the following formats:

- "display name" <username@ipaddress:port>
- "display name" <username@ipaddress>
- username@ipaddress:port
- username@ipaddress
- username

The display name and port number are optional parameters in the SIP address. If a port is not specified, 5060 will be set up as the default one. The range of valid ports is between 1024 and 65536. The SIP address may contain wildcards. The following combinations can be used:

- *@ipaddress – any user from the specified SIP server
- username@* – a specified user from any SIP server
- *@* – any user from any SIP server

8.3 Call Recording Settings

The Call Recording feature allows to record all inbound and outbound calls, including calls that pass through the QX, keep the recordings locally or send them to the FTP server.

For more information on how to configure and use Call Recording feature, refer to the Call Recording Feature on QX IP PBXs guide.

8.4 NAT Traversal

NAT Traversal is divided into separate pages that are used to configure the General NAT Traversal Settings, SIP, RTP and STUN parameters for NAT as well as configuring Exceptions List for NAT Traversal.

8.4.1 General

The NAT Traversal Settings page is used to select the NAT Traversal mode which will be used for the SIP traffic. The following modes are available:

- Automatic – if selected, the system will analyze the QX WAN IP address. If the address is in the IP range specified for the private networks (according to RFC), the SIP traffic (any incoming and outgoing SIP messages from/to QX) will be routed through the NAT router. Otherwise no SIP traffic will be routed through the NAT router.
- Force – if selected, all SIP traffic will be routed through the NAT router.
- Disable – if selected, no SIP traffic will be routed through the NAT router.

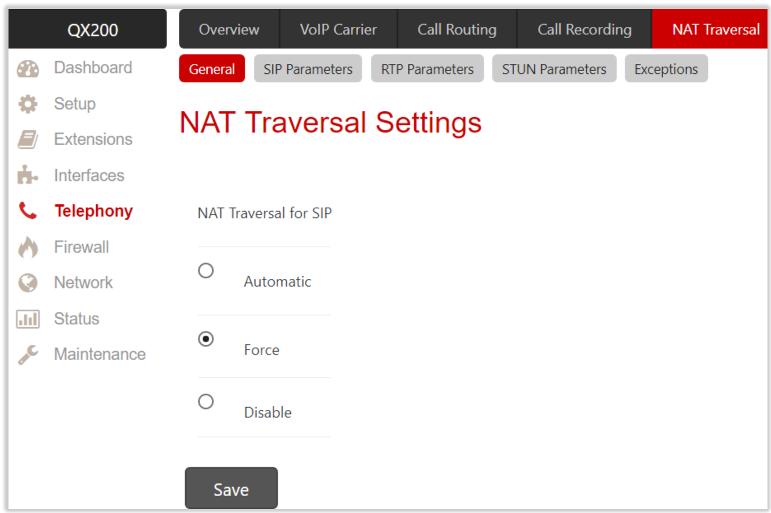


Figure 88: NAT Traversal Settings page

8.4.2 SIP Parameters

The **NAT Traversal – SIP Parameters** page is used to configure the NAT specific settings for SIP and offers two independent groups of settings.

The **UDP Parameters** section allows to select the type of connection over NAT as follows:

- **Use STUN** is used to automatically discover the mapped settings for the SIP UDP traffic over NAT. STUN settings are configured on the **STUN Parameters** page.
- **Use Manual NAT Traversal** is used to manually set the mapped settings for the SIP UDP traffic over NAT:
  - **Mapped Host** is used to set the IP address of the mapped host for SIP UDP traffic over NAT.
  - **Mapped Port** is used to set the port number on the mapped host for the SIP UDP traffic over NAT.

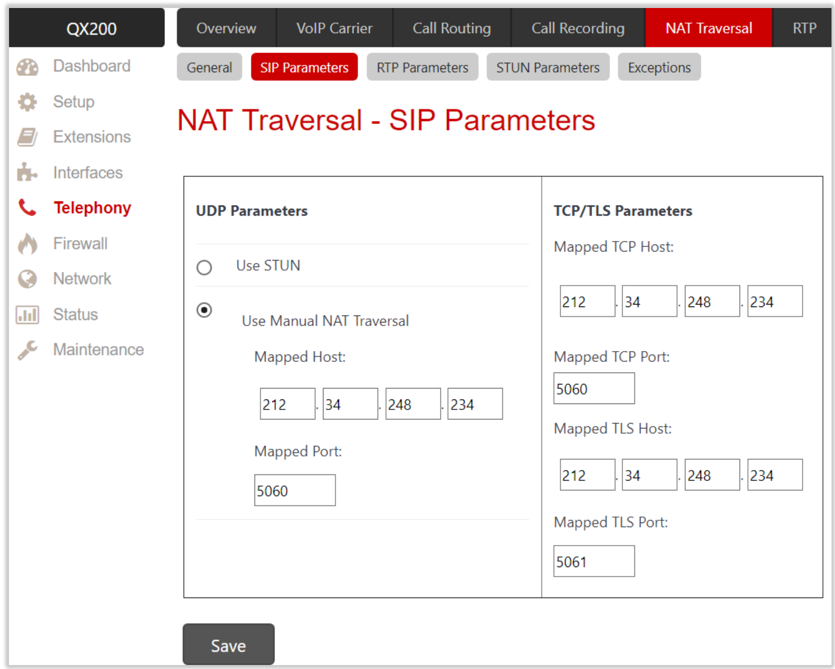


Figure 89: NAT Traversal – SIP Parameters page

The **TCP/TLS Parameters** section allows to set TCP/TLS ports for the connection over NAT as follows:

- **Mapped TCP Host** is used to set the IP address of the mapped host for SIP TCP traffic over NAT.
- **Mapped TCP Port** is used to set the port number on the mapped host for the SIP TCP traffic over NAT.
- **Mapped TLS Host** is used to set the IP address of the mapped host for SIP TLS traffic over NAT.
- **Mapped TLS Port** is used to set the port number on the mapped host for the SIP TLS traffic over NAT.

8.4.3 RTP Parameters

The **NAT Traversal – RTP Parameters** page is used to choose between the STUN and Manual NAT traversal connection for the RTP traffic and set the RTP/RTCP ports for the connection over NAT.

- **Use STUN** is used to automatically discover the mapped settings for the RTP UDP traffic over NAT. STUN settings are configured on the **STUN Parameters** page.
- **Use Manual NAT Traversal** is used to manually define the RTP/RTCP port ranges for the RTP traffic over NAT:
  - **Mapped Host** is used to set the mapped host IP address for RTP traffic over NAT.
  - **Min** and **Max** are used to set the port numbers on the mapped host for RTP and RTSP traffic.

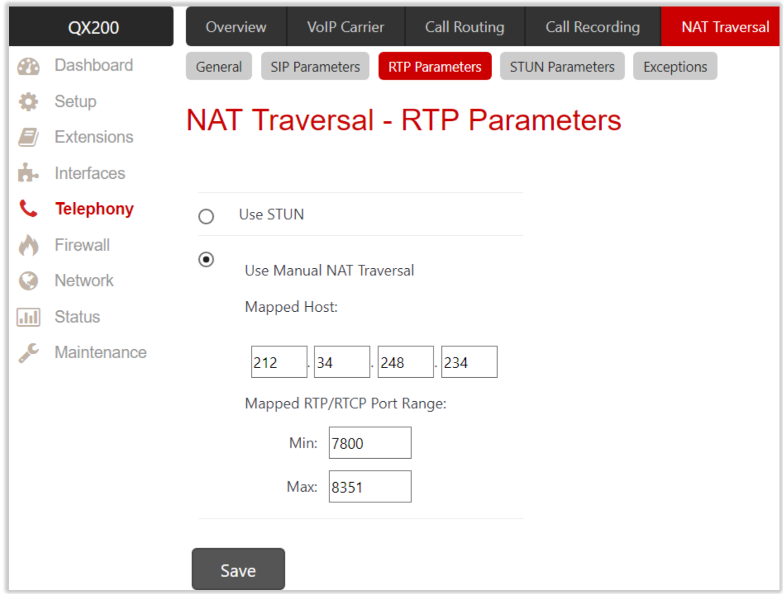


Figure 90: NAT Traversal – RTP Parameters page

8.4.4 STUN Parameters

The **NAT Traversal – STUN Parameters** page is used to enable automatic NAT configuration through the STUN server and is used to configure the STUN client on QX as follows:

- **Primary STUN Server** is used to set the STUN server hostname or IP address.
- **Primary STUN Port** is used to set the STUN server port number.
- **Secondary STUN Server** and **Secondary STUN Port** are used to set the respective parameters of the secondary STUN server.
- **Polling Interval** is used to select the possible time intervals between referrals to the STUN server.
- **Keep-alive Interval** is used to set the time interval for keeping NAT mapping alive.
- **NAT IP checking Interval** is used to set the interval between the NAT IP checking attempts (used to distinguish the possible NAT IP address changes and to perform registration on the new host).

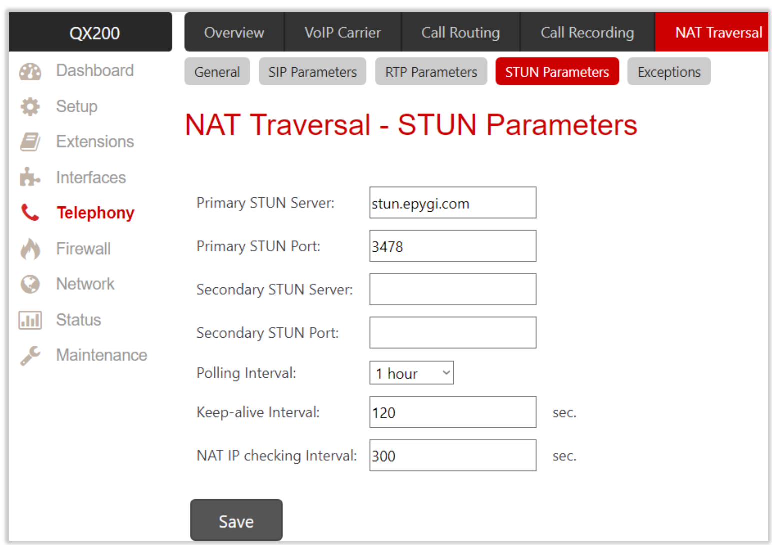


Figure 92: NAT Traversal – STUN Parameters page

8.4.5 Exceptions

The NAT Traversal Exceptions page displays all possible IP ranges that are not included in the NAT process. IP addresses that are not listed in the NAT Traversal Exceptions are accessed over NAT. For example, if a QX user needs to make SIP calls within the local network as well as outside that network, all local IP addresses are required to be excluded from NAT traversal settings by being listed in this table.

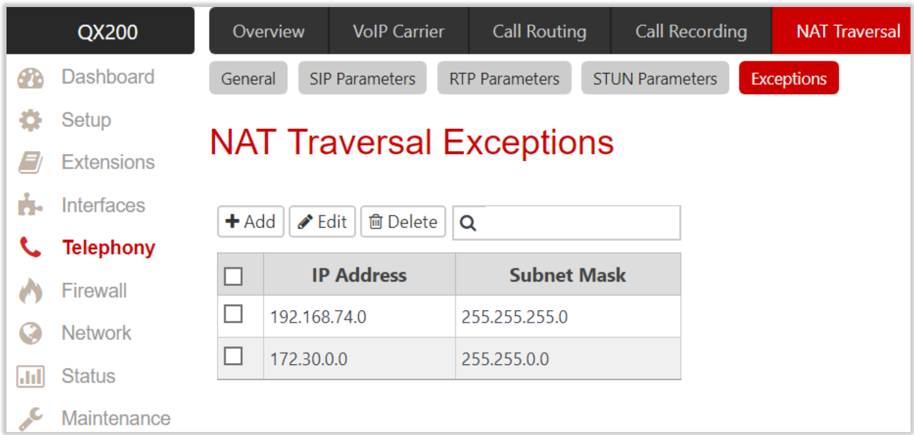


Figure 93: NAT Traversal Exceptions page

To add a new exception:

1. Click Add and enter the following information:
   - Enter the IP Address.
   - Enter the Subnet Mask. TIP: Enter 255.255.255.255 as a Subnet Mask to add only the IP address in exception list.
2. Click Save to add the new exception entry to the NAT Traversal Exceptions table.

8.5 RTP

The **RTP Settings** page is used to configure the packet size and silence suppression for each voice codec. The **Codec Properties** table lists all codecs with the packetization interval and silence suppression associated to each.

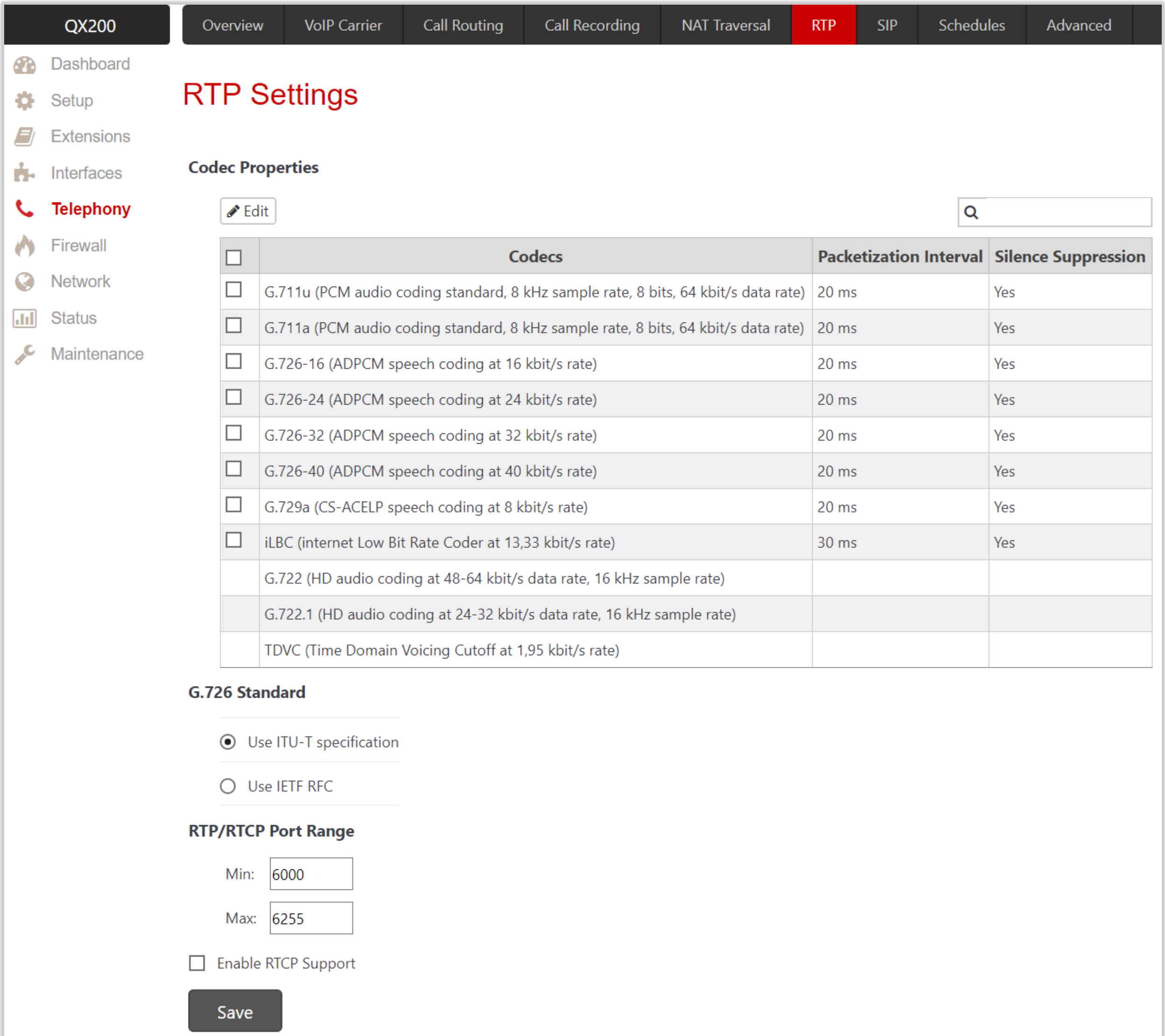


Figure 94: RTP Settings page

To modify the **Codec** parameters:

1. Tick the checkbox next to the codec and click **Edit**. The following settings (options) are available:
   - **Packetization Interval** is used to set the time interval between two RTP packets of the same stream. If this interval is increased, the overhead will decrease, but the voice quality may deteriorate as a result. If the interval is decreased, the network load will increase and the delay will reduce.
   - **Enable Silence Suppression** is used to stop RTP packet transmission in case of no voice activity. This option helps to avoid extra traffic if the RTP stream contains no voice activity. It is activated after two seconds of silence and restarts immediately in case of any audio.
2. Click **Save** to apply changes.

The following settings (options) are available:

- G.726 Standard is used to select between packaging method of the G.726 code words into octets.
  - Use ITU_T specification – if selected, the ITU I.366.2 (AAL2 type 2 service specific convergence sublayer for narrow-band services) type packaging of code words is used, where packing code words into octets start from the most significant rather than the least significant positions in the octet.
  - Use IETF RFC – if selected, the IETF RFC (RTP Profile for Audio and Video Conferences with Minimal Control) type packaging of code words is used, where packing code words start from the least significant positions in the octet.
- RTP/RTCP Port Range is used to set the port numbers for RTP/RTCP traffic.
- Enable RTCP Support is used to enable Real Time Control Protocol (RTCP) support and allows the RTCP packets transmission. RTCP is used for monitoring the RTP streams and changing RTP characteristics depending on network conditions.

8.6 SIP

8.6.1 SIP Settings

The SIP Settings page allows to configure SIP ports (UDP, TCP and TLS), the DNS Server for SIP and the SIP timers scheme. The following settings (options) are available:

- UDP Port indicates the SIP UDP receive port. By default, 5060 is selected and used.
- TCP Port indicates the SIP TCP receive port. By default, 5060 is selected and used. QX will not use TCP protocol as a transport for SIP messages if the TCP port is left blank.
- TLS Port indicates the SIP TLS receive port. By default, 5061 is selected and used.
- Realm is used to set the messaging level information to be included in SIP messages sent by QX. This information might be used by remote side for authentication purposes.
- Enable Session Timer is used to enable advanced mechanisms for connection activity checking. This option allows both user agents and proxies to determine if the SIP session is still active.
- DNS Server for SIP allows to choose between regular DNS servers configured in the DNS page and specific DNS servers for SIP traffic.
  - Default is used to apply regular DNS servers for SIP traffic.
  - Specific is used to enable SIP specific DNS servers. For this selection, both primary and secondary SIP DNS servers should be defined in the SIP DNS 1 and SIP DNS 2 fields.
- SIP Timers is used to set the timeouts of the SIP messages retransmission.
  - RFC 3261 is used to apply standard SIP timers described in the corresponding specification.
  - High Availability is used to apply SIP timers to shorten the call establishment, registration confirmation and registration failure procedures. This selection provides more firmness to the SIP connection but increases the network traffic on QX.
  - Custom is used to manually define the Registration Timeout, Registration Failure Timeout, Transaction Duration and Session Refresh Timeout timers.

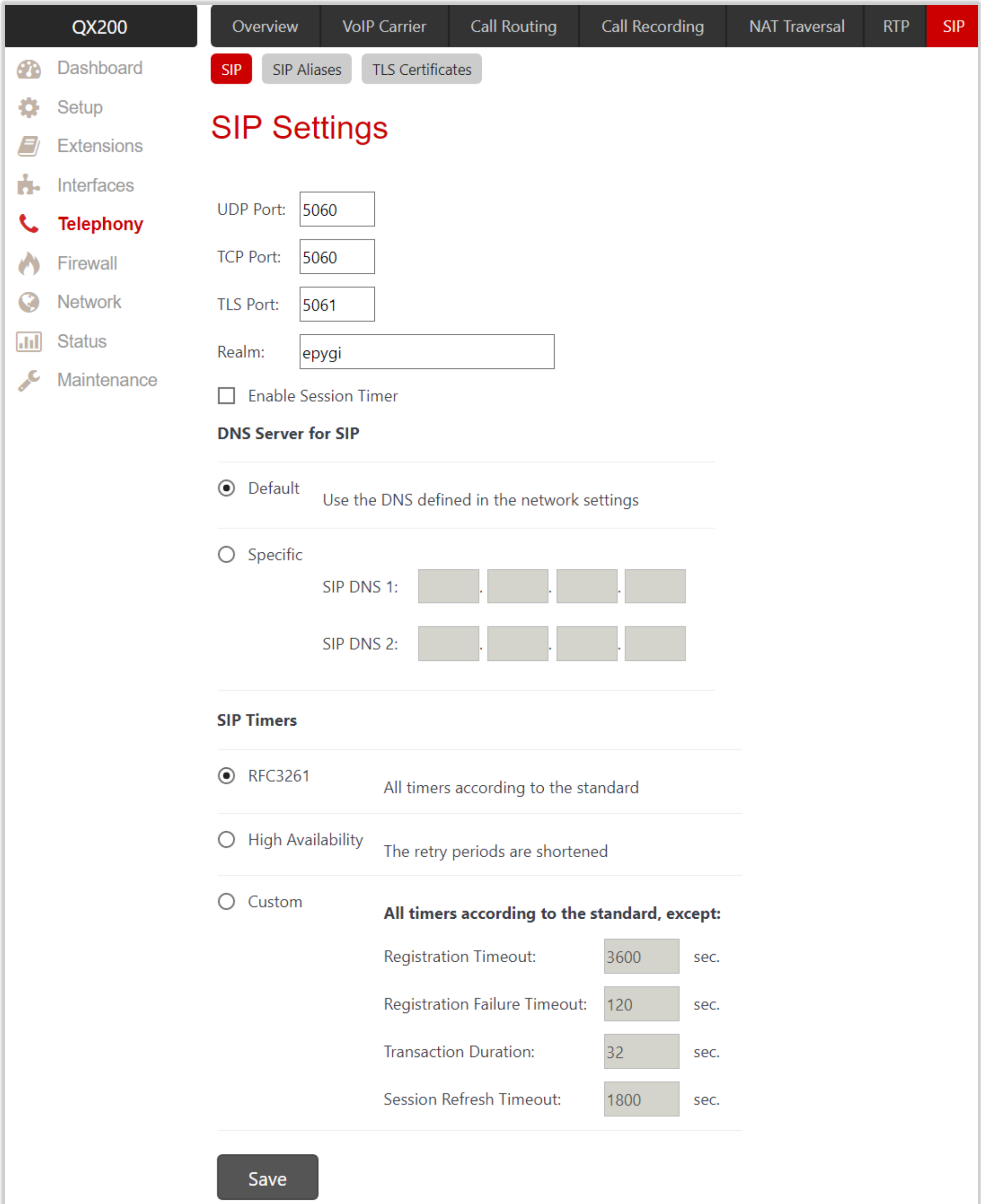


Figure 95: SIP Settings page

8.6.2 SIP Aliases

The Host Aliases for SIP page is used to add the hostname(s) registered on remote DNS server to the Host Aliases for SIP list. This list will be used to identify SIP packets received from remote servers where QX is registered with different names.

8.6.3 TLS Certificates

The **Generate and Install New CA Root Certificate** page is used to define, generate and install a new CA root certificate for SIP TLS traffic. All fields in this page require specific information on root certificate.

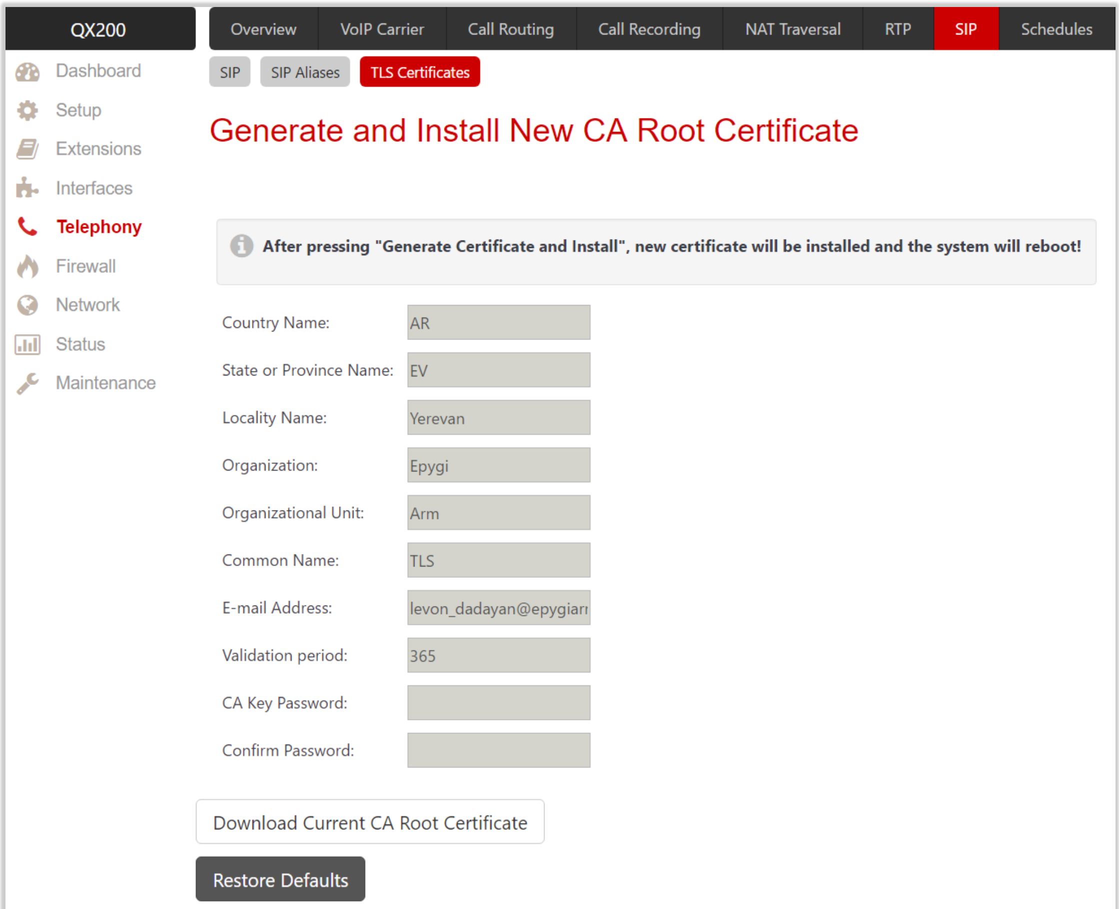


Figure 96: Generate and Install New CA Root Certificate page

- **Generate Certificate and Install** generates a new CA root certificate based on the defined data and installs it on QX. QX will reboot automatically once the new certificate is installed. You may download the actual copy of the certificate from SIP Settings page.
- **Download Current CA Root Certificate** is used to download the actual CA root certificate in (*.crt) file format.

To ensure a secure TLS connection with QX, both sides should have the same certificate installed. If the end user is an IP phone, you may activate the TLS certificate update mechanism from it to obtain the latest certificate generated by QX. If the end user is a server or other device, you may download the certificate from QX and apply it manually on the remote side.

8.7 Schedules

The **Schedules** feature is designed for creating flexible weekly working schedule(s). The preconfigured schedules then can be applied to the **Call Routing** and **auto attendant**. The **Day/Night Switching** service allows to control and change the state of schedules manually by using the phone handset instead of going into the GUI.

For more information on how to configure and use **Schedules**, refer to the Scheduling Feature on QX IP PBXs guide.

8.8 Advanced

8.8.1 Voice Mail

The Voice Mail Common Settings page is mainly used to select the codec for the Voice Mail recording.

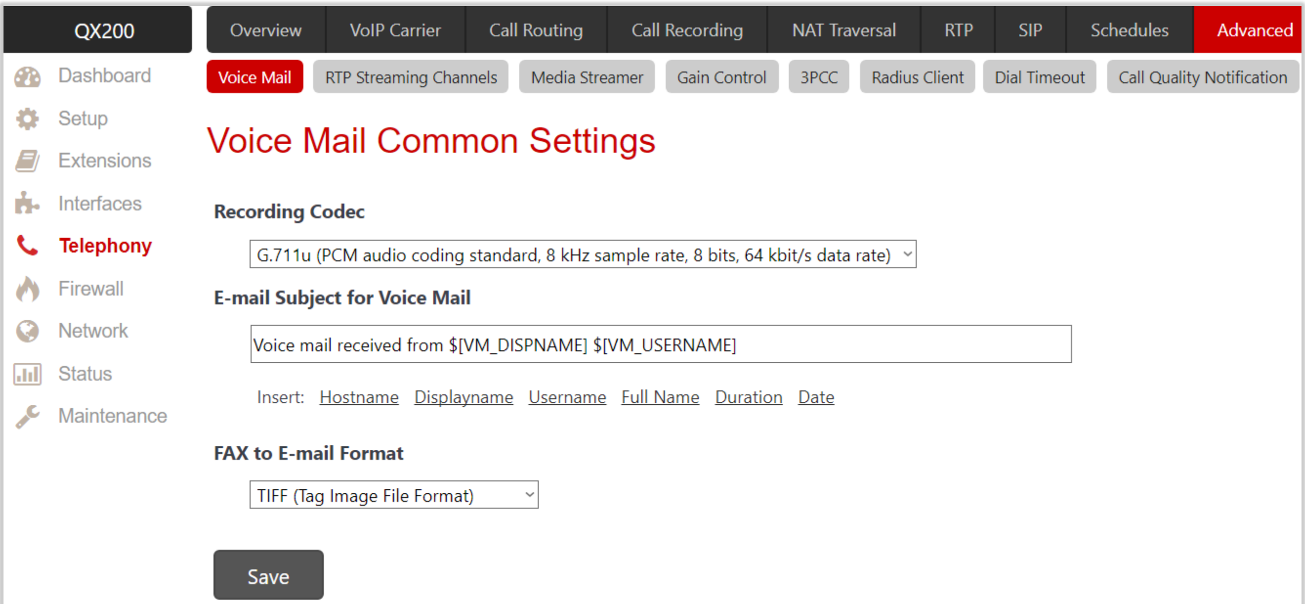


Figure 97: Voice Mail Common Settings page

The following settings (options) are available:

- Recording Codec is used to select the codec for voice mail recording. Changing the Voice Mail recording codec will directly affect the allocated memory size for extensions.
- E-mail Subject for Voice Mail is used to define a flexible subject for all e-mails sent from the QX and carrying the voice mails.

Besides using a static text in the subject line, you may use the predefined tags to combine the needed subject:

- Hostname is the hostname of QX.
- Displayname is the caller's display name. This value is not displayed for PSTN callers.
- Username is the caller's SIP username. For PBX callers, this is the caller's extension number and for PSTN callers, this is the caller's PSTN number.
- Full Name is the caller's full SIP address (SIP username and the SIP server). For PBX callers, this is the caller's extension number and for PSTN callers, this is the caller's PSTN number.
- Duration is the voice mail duration.
- Date is the date the voice mail has been received.

To enter the predefined tag to the subject line, you should simply click on the corresponding tag. The following format should be maintained to create a flexible subject:

Example: Voice mail received from $[VM_DISPNAME] $[VM_DATE].

In this example, all email subjects will contain a static text "Voice mail received from" which is followed by the display name of the caller and the date the voice mail has been received.

- FAX to E-mail Format is used to set the format of the FAX document received in the extension's voice mailbox and send as an attachment to the e-mail (in case if Send new voice messages via e-mail option is enabled for the extension). The (*.tiff) or (*.pdf) formats may be selected here.

8.8.2 RTP Streaming Channels

The RTP Streaming Channels page is used to define the channels for the RTP streaming. These channels may be then used when configuring RTP channel streaming for Music on Hold (MoH), auto attendant ringing announcement and for other custom messages.

For more information on how to configure and use RTP Streaming Channels, refer to the Customizing Voice Messages on QX IP PBXs guide.

8.8.3 Media Streamer

The Media Streamer service expand the audio streaming capacity of QX. Audio files uploaded on QX can be streamed out to specified destinations, thus allowing to play music on hold or other messages to callers while they are placed on hold. Audio files can be played either to remote destinations in network, or to extensions on the same QX.

The Media Streamer page allows to add and manage playlists for media streamer, start and stop audio streaming with playlists. The configured playlists can be used to stream audio to the extensions through the RTP Streaming Channels.

For more information on how to configure and use Media Streamer, refer to the Customizing Voice Messages on QX IP PBXs guide.

8.8.4 Gain Control

The **Gain Control** settings are used to set the **Transmit** and **Receive** gains. The **Gain Control** page consists of **Transmit Gain** and **Receive Gain** drop-down lists for each line that contains allowed gain values, which can be set up for every line.

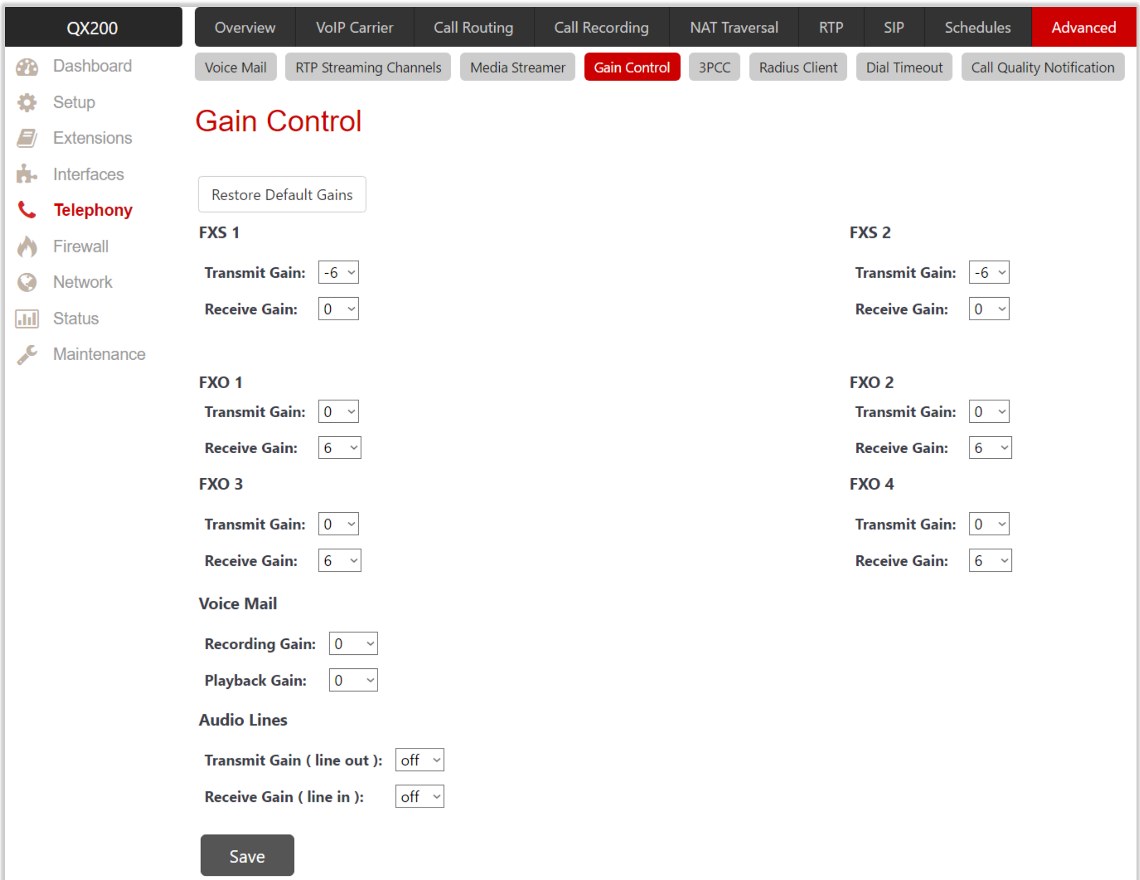


Figure 98: Gain Control page

- For **FXS** lines:
  - **Transmit Gain** is used to set the phone speaker volume on the call.
  - **Receive Gain** is used to set the volume of the phone microphone on the call.
- For **FXO** lines:
  - **Transmit Gain** is used to set the level of voice transmitted from QX to the FXO network.
  - **Receive Gain** is used to set the volume of voice received by QX from the FXO network.
- For **ISDN** trunks:
  - **Transmit Gain** is used to set the level of voice transmitted from QX to the ISDN network.
  - **Receive Gain** is used to set the volume of voice received by QX from the ISDN network.
- For Voice Mail:
  - **Recording Gain** is used to set the volume of the phone microphone upon playing voice mails or system messages.
  - **Playback Gain** is used to set the phone speaker volume upon playing voice mails or system messages.
- For Audio Lines:
  - **Transmit Gain (line out)** is used to set the level of voice transmitted from QX to the **Audio Line Out** port.
  - **Receive Gain (line in)** is used to set the volume of voice received by QX from the **Audio Line In** port.
- **Restore Default Gains** is used to restore the default values.

8.8.5 3PCC

The **3PCC Settings** page is used to adjust the **3rd party call controlling** (3PCC) settings. 3PCC service allows call controlling applications to remotely initiate and handle calls on QX and subscribe for certain event notifications from QX.

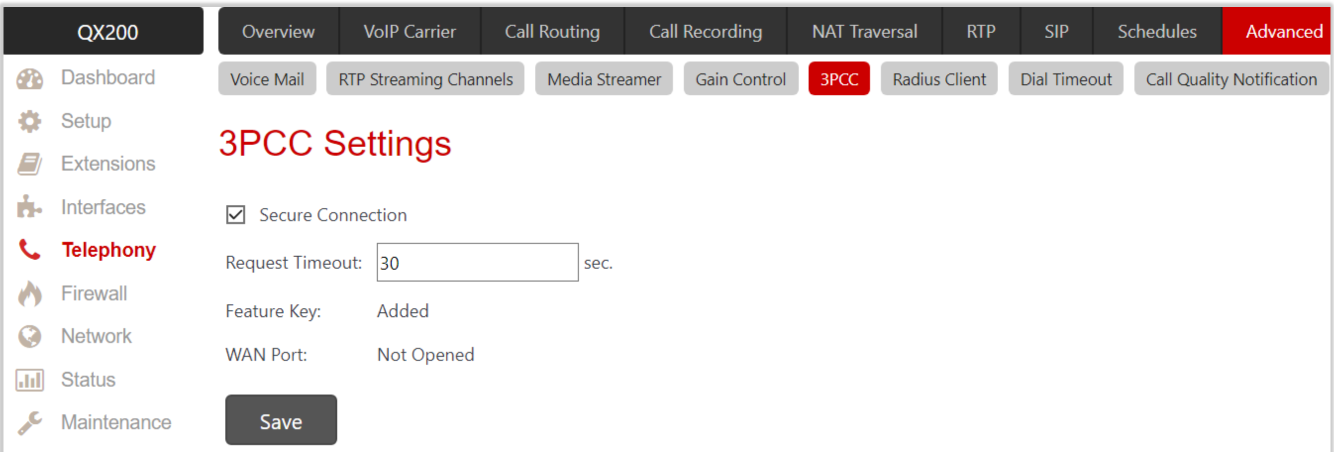


Figure 99: 3PCC Settings page

The following settings (options) are available:

- **Secure Connection** – if selected, a secure encrypted connection will be used between the call controlling application and QX. The **Secure Connection** must be set up in the same way on both sides for successful connection.
- **Request Timeout** is used to set the timeout during which QX should receive a response to the request from the call control application. If no response is received during this timeout, QX will perform a request according to default action. Let's say the call control application is configured to handle incoming calls on QX. Once incoming call is received, QX will try to transfer the call to the call control application. If the call control application does not respond within the mentioned timeout, QX will answer the call or perform an action configured for missed incoming calls. This setting depends on the network conditions therefore consult with your network administrator before changing the default value.
- **Feature Key** indicates whether the feature key for the **3PCC support** is installed on the system or not. The system will not accept connections from 3PCC applications if no key is found. The **3PCC support** is a licensable feature and can be activated from the **Licensed Features** page.
- **WAN Port** indicates whether there is a filtering rule specified for the Call Control Access or not. If a 3PCC application connects to QX from the WAN interface, a filtering rule should be created on the **Call Control Access** page to allow remote access. Creating a filtering rule is not required if the firewall is not setup on QX. The field shows **Opened** if there is at least one enabled filtering rule for the **Call Control Access**.

8.8.6 RADIUS Client

Remote Authentication Dial in User Service (RADIUS) is a networking protocol that provides centralized **Authentication**, **Authorization**, and **Accounting** (AAA) management for users who connect and use a network service. The RADIUS server provides the option for a caller from/through QX to make calls after passing authentication.

When a RADIUS client is enabled on QX, and according to the configuration of **AAA Required** option, the RADIUS server will be used to authenticate user and/or to account for the call. This can be accomplished by automatic detection of the caller's number or a customized login prompt where the caller is expected to enter a username and password.

Transactions between the RADIUS client and server are authenticated through the use of a shared Secret Key, which is never sent over network. In addition, user passwords are encrypted when sent between the client and server. If no response from the RADIUS server is returned once the Receive Timeout expires, the authentication request will be resent. The client can also forward requests to the secondary server if the primary server is down or unreachable. The secondary server can be used after a number of failed attempts to the primary server.

Once the RADIUS server receives the request from client, it determines the client's validity. If the client is valid, the RADIUS server addresses the user database to find the user whose name matches the request. The user entry in the database contains a list of parameters (username, password, etc.) that must be met to give access to the user. If all conditions are met, the user gets access to the QX.

The following settings (options) are available:

- Enable RADIUS Client is used to activate service on QX. TIP: This service cannot be disabled if the RADIUS Authentication and Authorization or RADIUS Accounting options are enabled at least on one call routing rule.

Registration Settings

- Primary Server is used to set the IP address of the primary RADIUS server.
- Secondary Server is used to set the IP address of the secondary RADIUS server.
- NAT Station IP is used to set the IP address of the NAT station. If no IP address is set, QX IP address will be sent to the RADIUS server.
- Secret Key is used to specify the secret key.
- Retry Count is used to select the number of unsuccessful requests before canceling the authentication on RADIUS server.
- Receive Timeout is used to select the timeout between two attempts to authenticate.
- Encoding Type is used to select the encoding type (PAP or CHAP) that should be unique on both the client and the server sides for the establishment of a successful connection. Encoding type should also be requested from RADIUS server.
- Authorization Port is used to set the port number on the RADIUS server where QX is to send the authentication requests.

- **Accounting Port** is used to set the port number on the RADIUS server where QX is to send the accounting messages.

Authentication Settings

- **Enable common login for all users in time of by phone authentication** is used to activate phone authentication service This checkbox enables **Username** and **Password** fields to set common authentication parameters.
- **Authentication on Destination RADIUS Server** is used to set the authentication parameters. Leave these fields blank if you want to use the original authentication parameters.

Accounting Settings

- **Username** is used to set the identification parameter for accounting services only. The source username will be used if no username is specified
- **Send Accounting messages** is used to select sending **Both Start and Stop** or **Only Stop** accounting message.

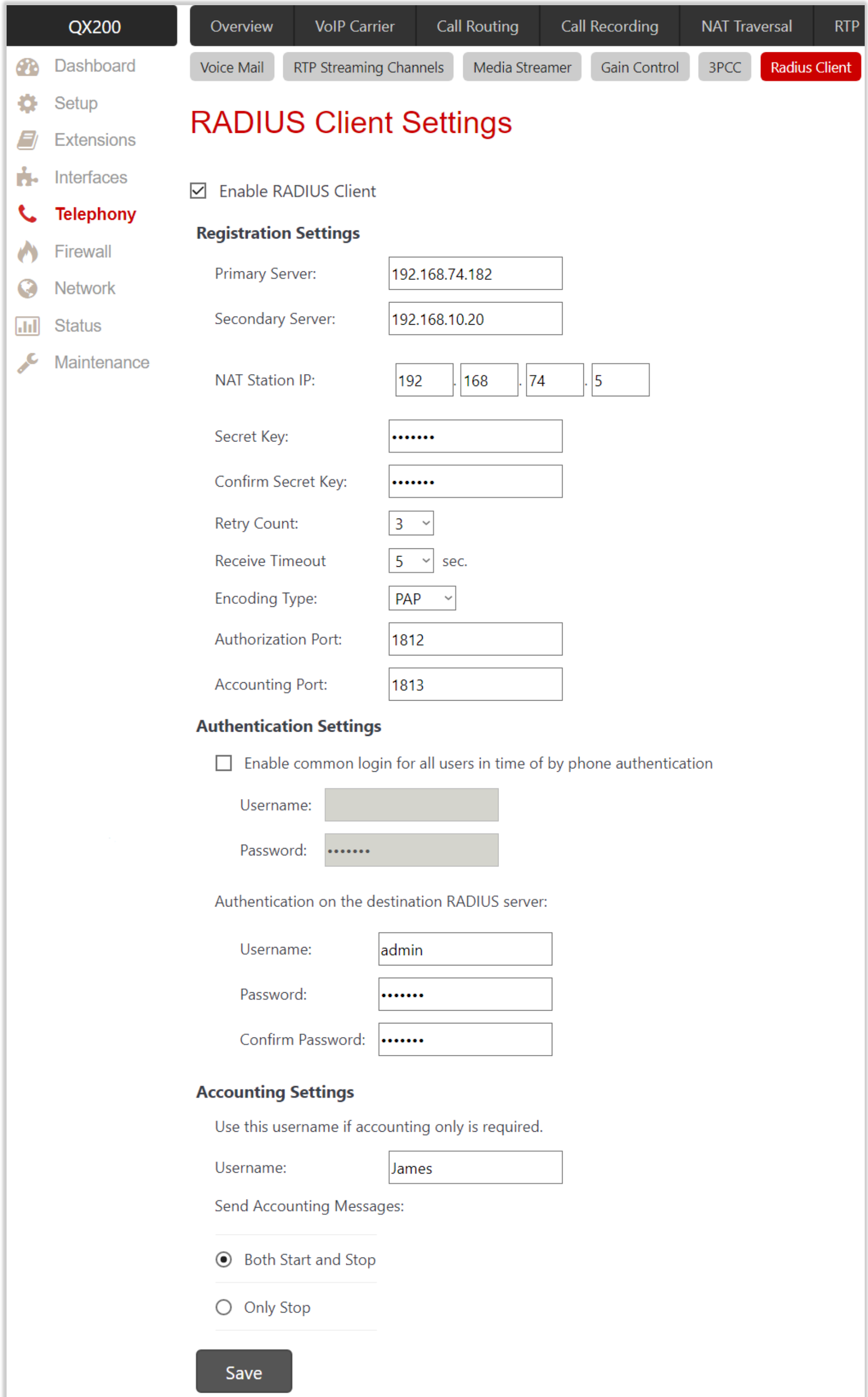


Figure 100: Radius Client Settings page

8.8.7 Dial Timeout

The **Dial Timeout Settings** page is used to adjust the timeout setting when dialing on the phone. The **Routing Dial Timeout** option is used to set timeout after the last dialed digit that the system identifies as a completion of dialing. If the user does not press any key within the specified timeout, the system assumes that the dialing is completed and starts processing the dialed number. This option is also applicable to all supported IP phones. The modified value will take effect after rebooting IP phone.

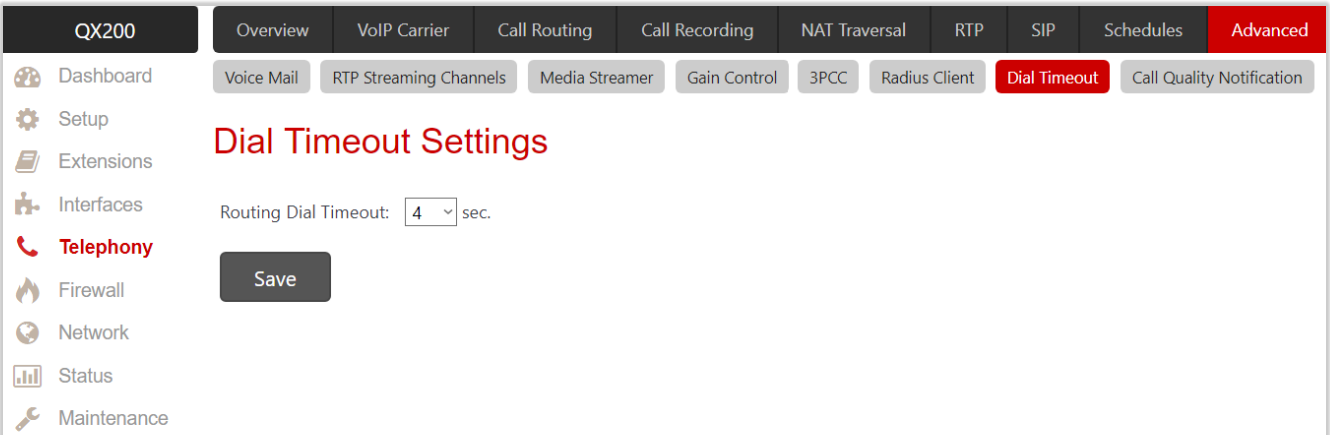


Figure 101: Dial Timeout Settings page

8.8.8 Call Quality Notification

The **Configure Call Quality Event Notification** page is used to configure the policy for event notification when the call quality is lower than the selected level.

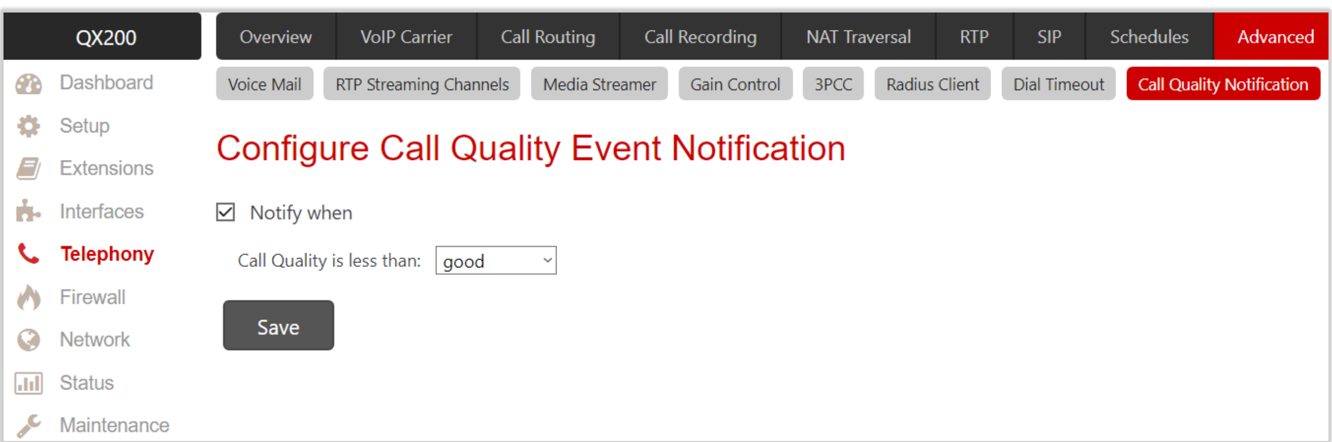


Figure 102: Configure Call Quality Event Notification page

To activate **Call Quality Event Notification** service:

1. Tick the **Notify when** option to enable the call quality monitoring mechanism.
2. Select the **call quality level** below which the notification will be generated and displayed in **System Events**.
3. Click **Save** to apply changes.

9 Firewall Menu

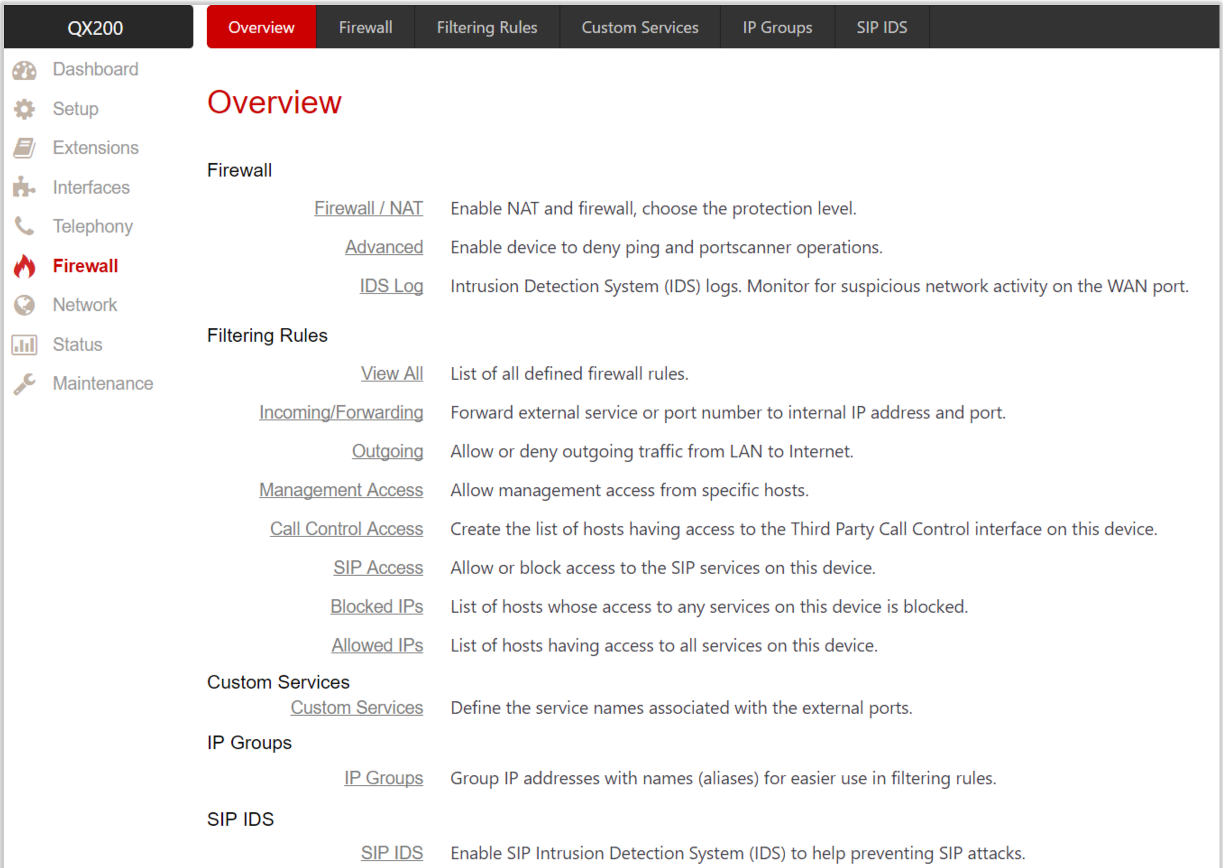


Figure 103: Fireval Menu overview

9.1 Firewall

The **Firewall Configuration** page allows setting up the **Firewall** and activating **Network Address Translation** (NAT) and **Intrusion Detection System** (IDS) services on QX.

Firewall is a security service configurable through various criteria. It has three level of security policies: low, medium and high. QX Firewall allows or blocks traffic based on the policies, services and/or IP addresses. Filtering rules will take effect only if the Firewall has been enabled and are independent from the selected firewall security level. Additional service-based rules can be added as well.

NAT is used to connect the QX LAN devices (IP phones, PCs, etc.) to Internet using QX WAN IP address. NAT also forwards incoming packets from the WAN to the QX LAN devices.

IDS is a type of firewall. The latter deletes dangerous packets or packets containing intrusion attacks, also generates a log file containing information about the dropped packets and senders responsible for those packets. The log can be viewed on the IDS Log page. Users can be notified about the generated logs through an email, flashing LED display notification, etc.

9.1.1 Firewall and NAT

The **Firewall Configuration** page is used to configure the following settings (options):

- **Enable IDS** is used to enable service on QX.
- **Enable NAT** is used to enable service on QX.
- **Enable Firewall** is used to enable service on QX. To activate Firewall, the firewall security level should be selected. The **Firewall Security** levels are the following:
  - **Low Security** – everything that is not explicitly forbidden will be allowed. This security level doesn't block anything by default. It is recommended if the device is already located behind another firewall or if every filter has been configured correctly.
  - **Medium Security** – traffic originating from the LAN side may pass and traffic from the WAN side will be blocked by default. This is the recommended security level.
  - **High Security** – everything that is not explicitly allowed will be blocked, including traffic from the LAN side.

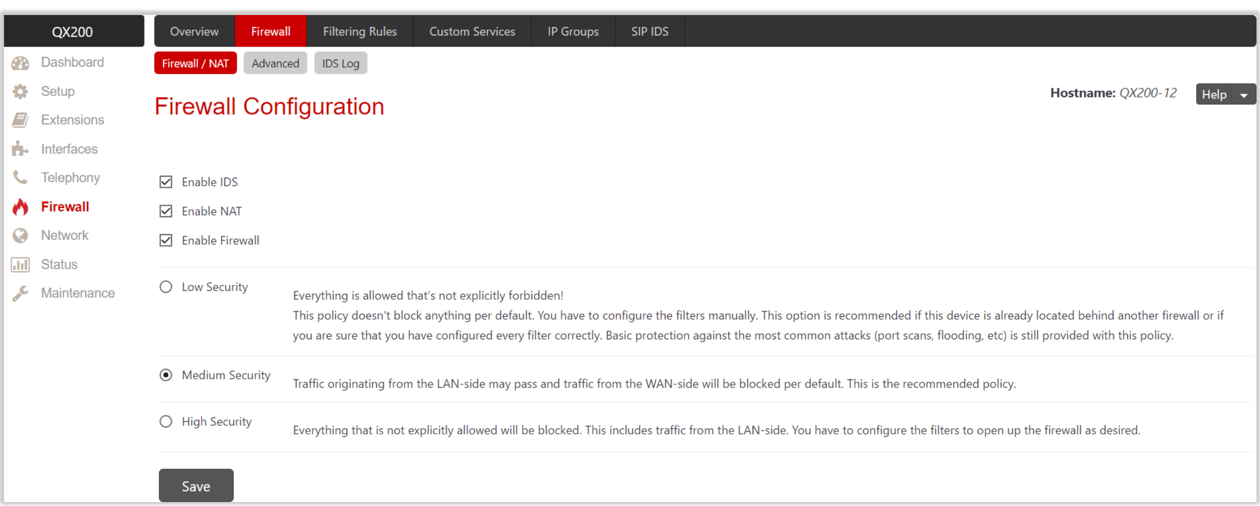


Figure 104: Firewall Configuration page

9.1.2 Advanced Firewall Settings

Advanced Firewall Settings is used to activate Ping Stealth and Fool Portscanner services to enhance system security. These services will be activated when Firewall is enabled on QX.

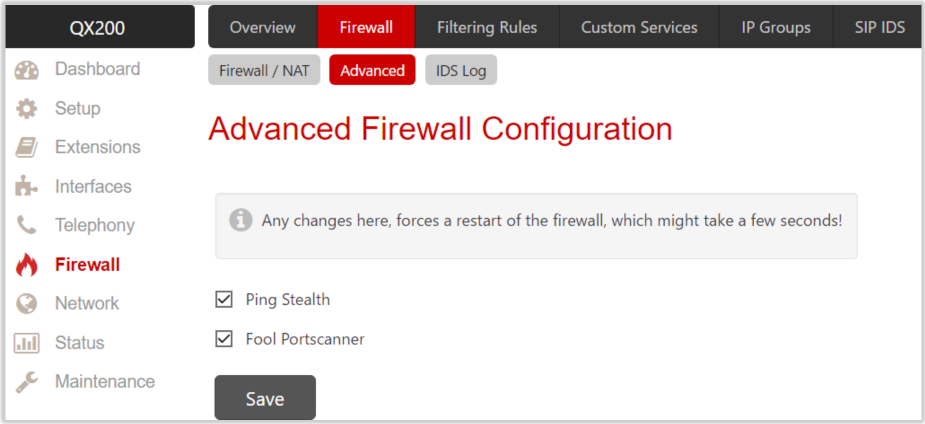


Figure 105: Advanced Firewall Settings page

9.1.3 IDS Log

The IDS Logs page contains information on dropped packets and the senders responsible for those packets. The system discards dangerous packets or packets including intrusion attacks. It generates a table with the IDS log report. Administrator can be notified about newly logged entries in various ways (e-mail, display notification, etc.) depending on the settings in the System Events page. IDS logs will be reported as soon as IDS is enabled from the Firewall Configuration page. The IDS Logs table shows the IDS entries and descriptions referring to them.

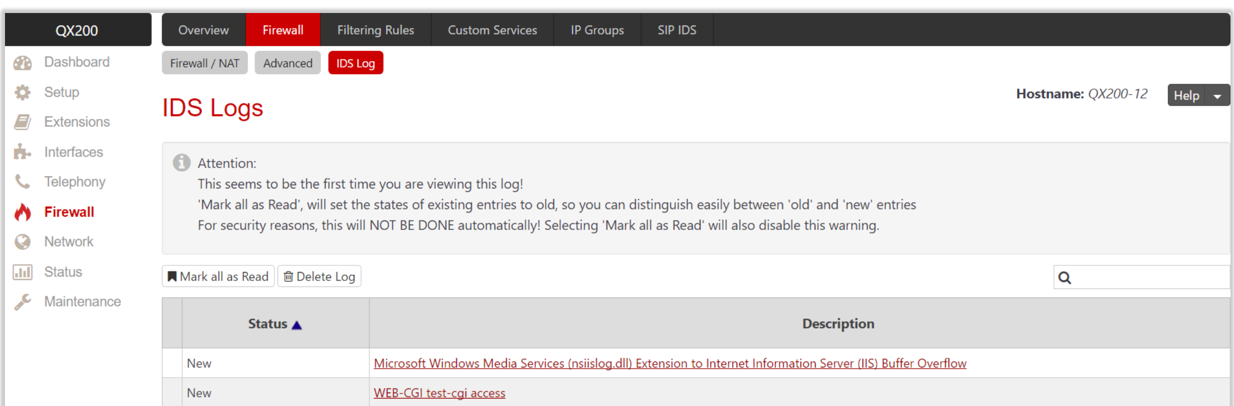


Figure 106: IDS Logs page

Click on the desired entry to see its detailed log in the IDS Detailed Logs table. The IDS Logs table shows the detailed log: additional information about the access protocol, IP address and port number as well as date and time of the event.

9.2 Filtering Rules

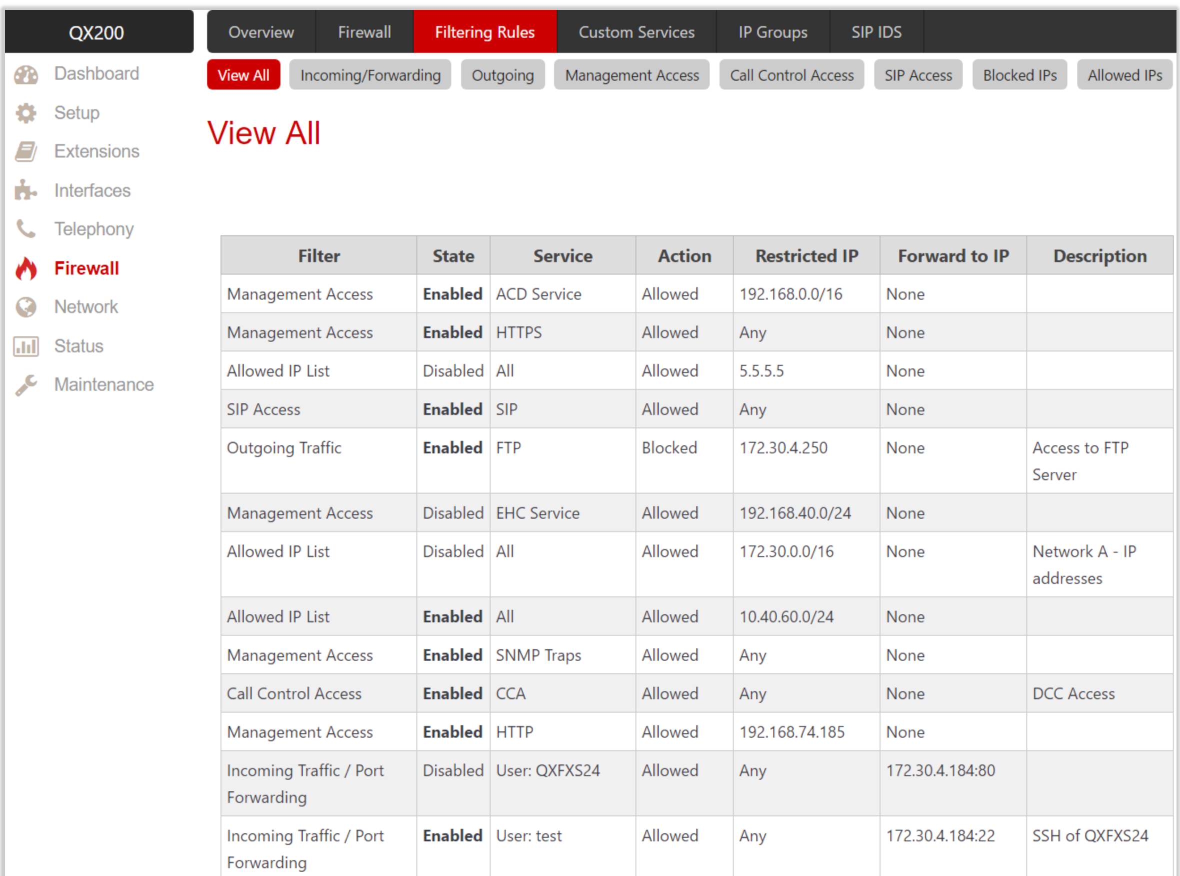
The **Filtering Rules** pages are used to configure the filters for incoming and outgoing traffic. It is allowed to create only one rule per service to prevent inaccurate configuration. You may use IP groups to include several IP addresses for any rule. Since the filtering rules specify the operation mode of the firewall, they only take effect if the firewall has been enabled (also NAT is enabled to use the **Port Forwarding** function in the Incoming Traffic/Port Forwarding filtering rules). The filtering rules are independent from the security level, so they will work regardless of the type of selected security level.

Note:

- Applying firewall rules will prevent the establishment of new connections that violate the rules. Applying rules does not kill existing connections that violate the rule.
- The newly created blocking filtering rules will take effect immediately only if the IP address(es) is added into the Blocked IPs.

9.2.1 View All Filtering Rules

The **View All** table presents all configured filters, specified by their **State** (enabled or disabled), selected **Service**, type of **Action** (allowed or blocked), **Restricted IP** addresses and **Destination** of port forwarding.



Filter	State	Service	Action	Restricted IP	Forward to IP	Description
Management Access	**Enabled**	ACD Service	Allowed	192.168.0.0/16	None	
Management Access	**Enabled**	HTTPS	Allowed	Any	None	
Allowed IP List	Disabled	All	Allowed	5.5.5.5	None	
SIP Access	**Enabled**	SIP	Allowed	Any	None	
Outgoing Traffic	**Enabled**	FTP	Blocked	172.30.4.250	None	Access to FTP Server
Management Access	Disabled	EHC Service	Allowed	192.168.40.0/24	None	
Allowed IP List	Disabled	All	Allowed	172.30.0.0/16	None	Network A - IP addresses
Allowed IP List	**Enabled**	All	Allowed	10.40.60.0/24	None	
Management Access	**Enabled**	SNMP Traps	Allowed	Any	None	
Call Control Access	**Enabled**	CCA	Allowed	Any	None	DCC Access
Management Access	**Enabled**	HTTP	Allowed	192.168.74.185	None	
Incoming Traffic / Port Forwarding	Disabled	User: QXFXS24	Allowed	Any	172.30.4.184:80	
Incoming Traffic / Port Forwarding	**Enabled**	User: test	Allowed	Any	172.30.4.184:22	SSH of QXFXS24

Figure 107: Filtering Rules – View All page

9.2.2 Incoming Traffic/Port Forwarding

The Incoming Traffic/Port Forwarding rules are used to allow or deny incoming traffic to reach to QX LAN. Enable the NAT service on QX to allow Port Forwarding in the Incoming/Forwarding filtering rules.

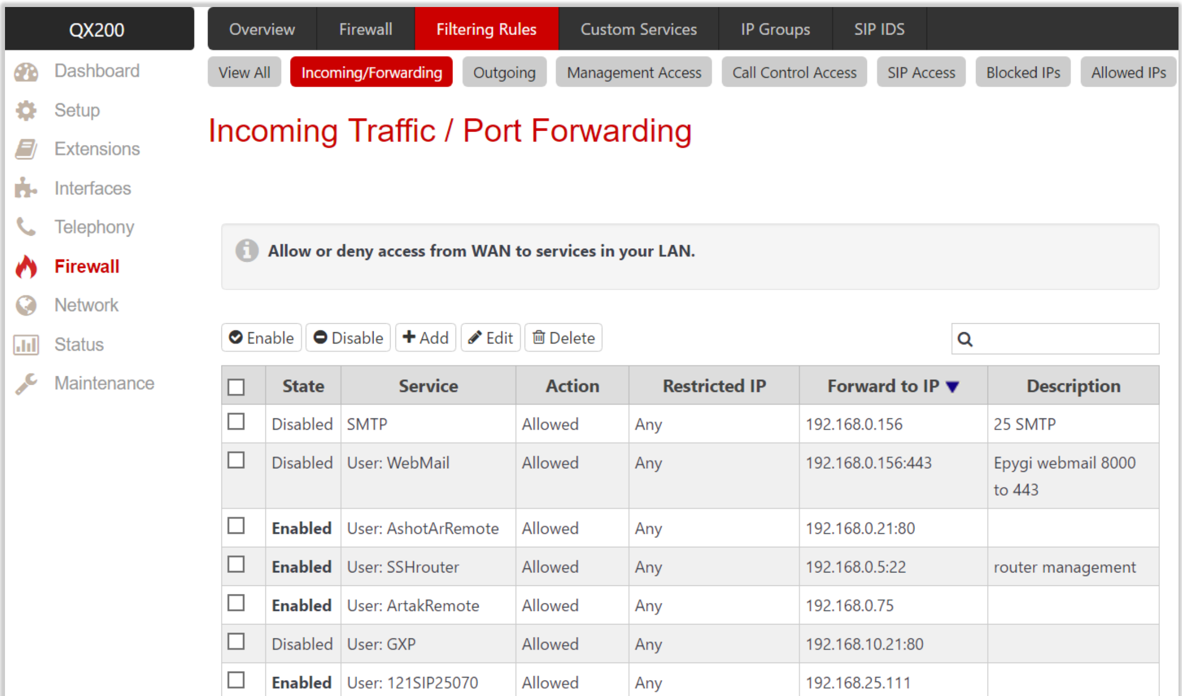


Figure 108: Filtering Rules – Incoming Traffic / Port Forwarding page

9.2.3 Outgoing Traffic

The Outgoing Traffic rules are used to allow or deny access to the external services for QX LAN users.

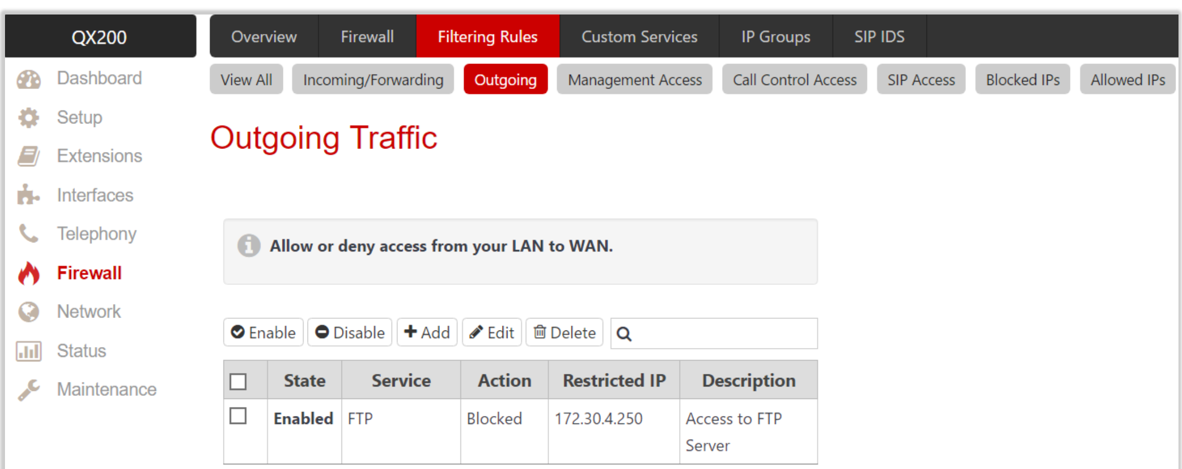


Figure 109: Filtering Rules – Outgoing Traffic page

9.2.4 Management Access

The Management Access rules are used to allow or deny WEB GUI access to QX.

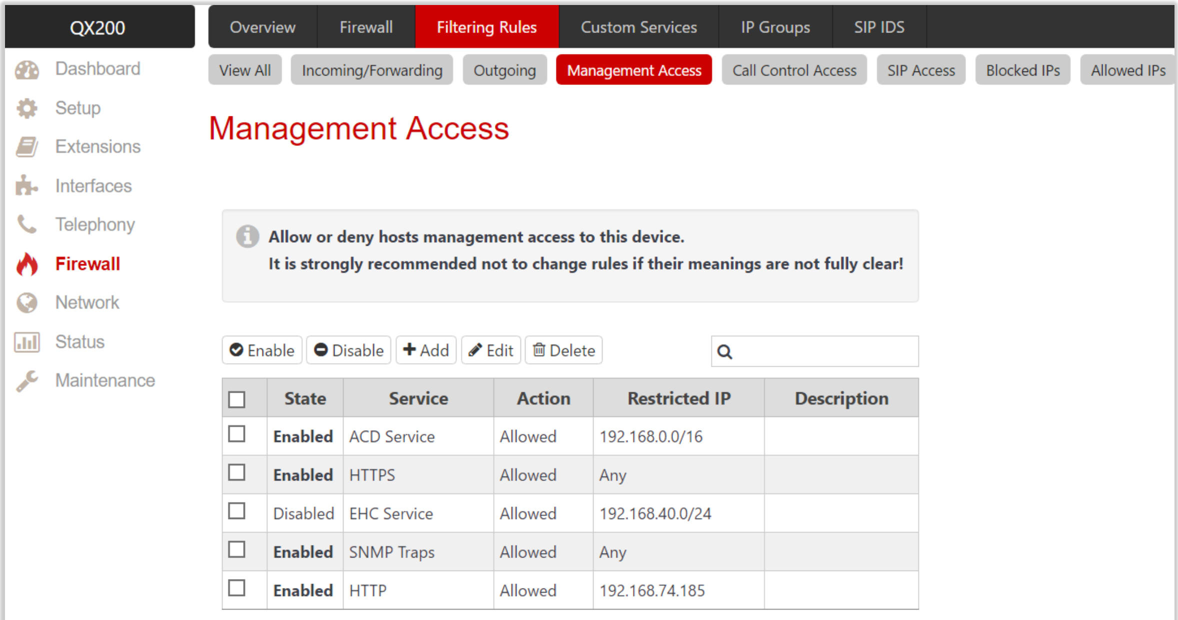


Figure 110: Filtering Rules – Management Access page

9.2.5 Call Control Access

The Call Control Access rules are used to allow or deny devices to access QX Call Control interface. It can be used to enable access from the call controlling applications (DCC, HotCall Add-In, etc.) to QX.

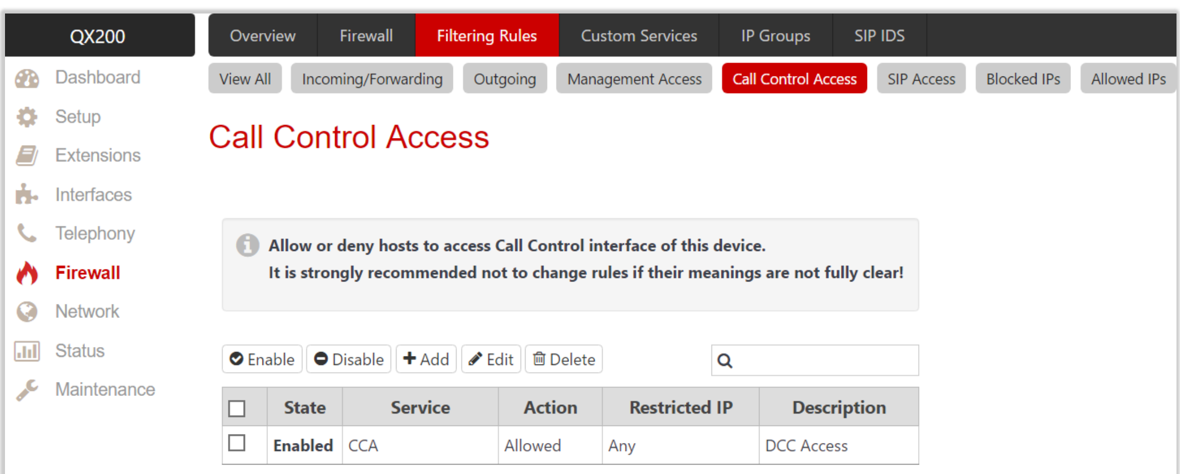


Figure 111: Filtering Rules – Call Control Access page

9.2.6 SIP Access

The **SIP Access** rules are used to allow or deny SIP traffic to QX from SIP servers and other SIP devices. It can be used to allow incoming/outgoing SIP calls from IP phones and SIP servers.

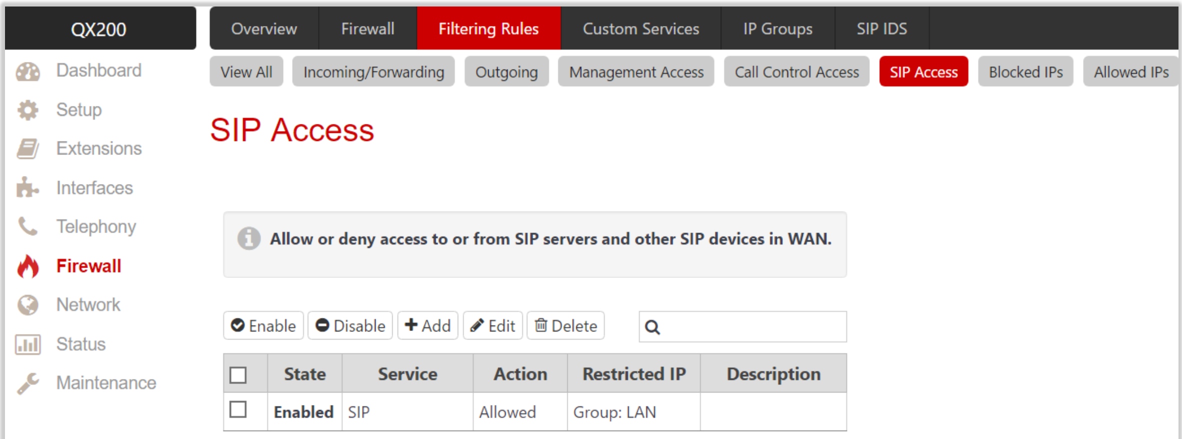


Figure 112: Filtering Rules – SIP Access page

9.2.7 Blocked IPs

The **Blocked IP List** rules are used to deny access for special devices. Traffic to or from these devices will be blocked in any case, no matter what services are configured in other pages. The **Blocked IP List** has a higher priority over the **Allowed IP List**: if the same host is listed in both tables, it will be blocked.

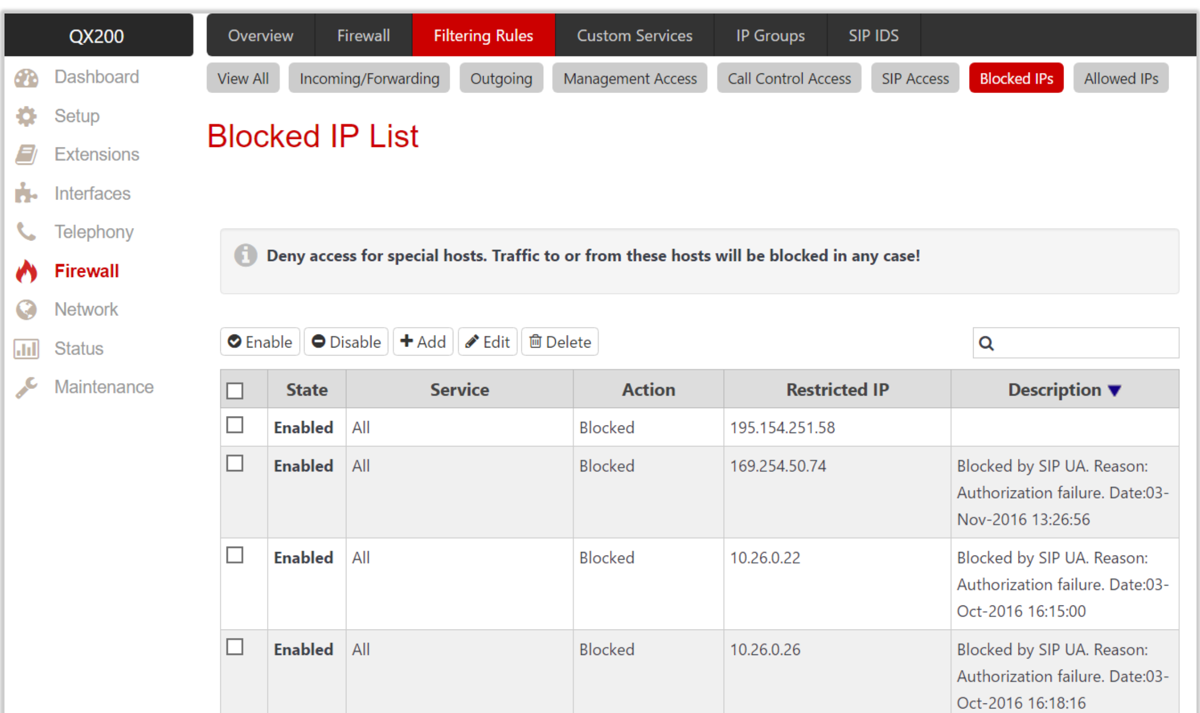


Figure 113: Filtering Rules – Blocked IP List page

9.2.8 Allowed IPs

The **Allowed IP List** rules are used to allow trusted devices to reach your network and vice versa. TIP: If the IP address of the device also appears in the **Blocked IP List**, then the traffic will be blocked as the **Blocked IP List** has a higher priority.

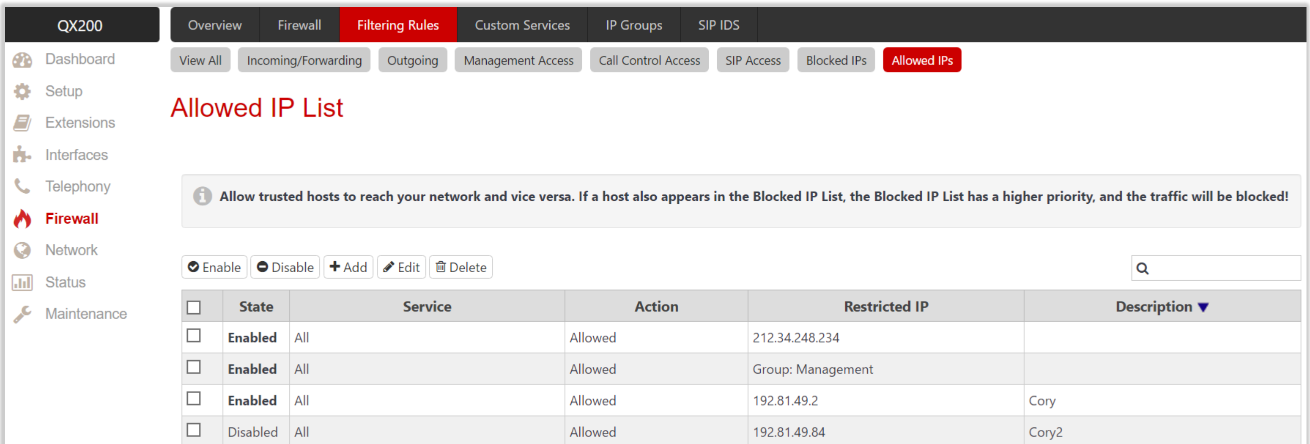


Figure 114: Filtering Rules – Allowed IP List page

To add a Filtering Rule

1. Go to the **Filtering Rules** (Incoming Traffic/Port Forwarding, Outgoing Traffic, Management Access, Call Control Access, SIP Access, Blocked IP List or Allowed IP List) page to add a rule.
2. Click **Add** on the corresponding filtering rule page.
   - Select the **Service** to configure a rule for it.
   - Select an **Action** to setup the rule.
   - Enter the destination **IP address** in the **Forward to IP** where traffic should be transferred to if it comes from the restricted host (**Incoming Traffic/Port Forwarding** rule).
   - Enter a **port number** in the **Port Translation** field which will stand instead of the original port number when incoming packet is being forwarded (**Incoming Traffic/Port Forwarding** rule).
   - Choose the **restriction type** by selecting **Any**, **Single IP**, **IP/Mask** or **Single URL** and enter the required information in the text fields or select a **Group**.
   - Enter a **Description**, if needed.
3. Click **Save** to create a rule with the given parameters. The newly created filtering rule will be shown in the corresponding **Filtering Rule** table and in the **View All** page.
4. Click **Enable** to activate the newly created filtering rule from the corresponding table.

9.3 Custom Services

The **Service Pool Configuration** page is used to create new services with the appropriate settings (protocol type and port range). New services can be used to add a restriction or allowance upon creating a new filtering rule.

To add a new **service**:

1. Click **Add**.
   - Enter a **Service Name**.
   - Select a **Protocol** type.
   - Set the **Port Range**.
2. Click **Save** to add the new service to the **Service Pool Configuration** table.

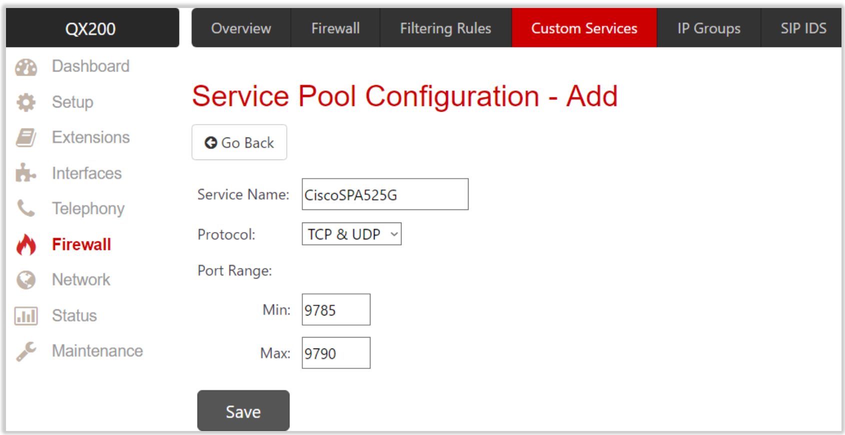


Figure 115: Service Pool Configuration – Add Service page

9.4 IP Groups

The **IP Pool Configuration** page is used to add groups of IP addresses that have the same restriction criteria. When adding a new filtering rule, a group can be used instead of several IP addresses. TIP: Changing a group name will also change the references to this group, including filtering rules and member relations to the other groups. Deleting a group will also delete any reference to the corresponding group, including filtering rules and member relations to the other groups.

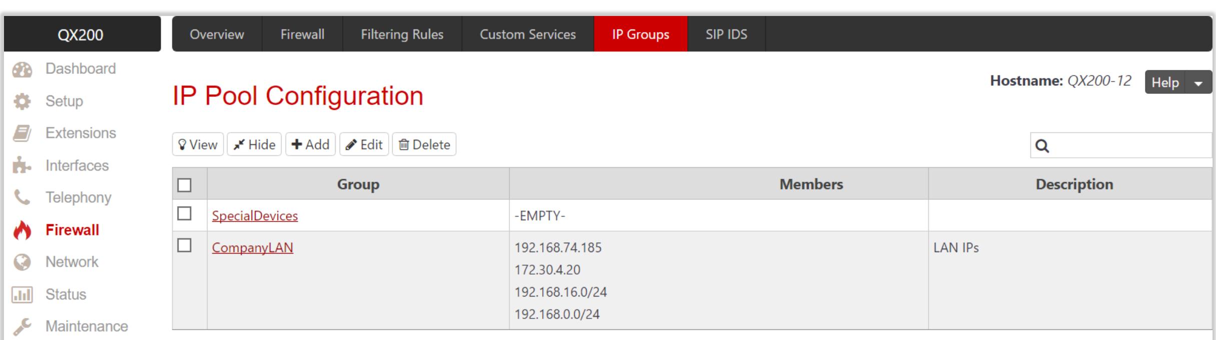


Figure 116: IP Pool Configuration page

Click **Group** name link to display the **IP Pool Group Configuration** page with the **Members** list for the current group.

To add a new **Group** with **Members**:

1. Click Add on the IP Pool Configuration page.
2. Enter a **Group Name** and fill out the **Group Description**, if needed.
3. Click **Save** to add the group. The newly added group will be shown on the **IP Pool Configuration** table.
4. Open the **IP Pool Group Configuration** page by clicking on the group name.
5. Click **Add** on the **IP Pool Group Configuration** page. A page opens where new members can be added to the group.
   - Choose the member addition type by selecting **IP Address, IP Subnet** and enter the required information in the text fields or select **A user-defined Group**.
   - Enter a **Member description**, if needed.
6. Click **Save** to add the new member to the **Current Group** table.

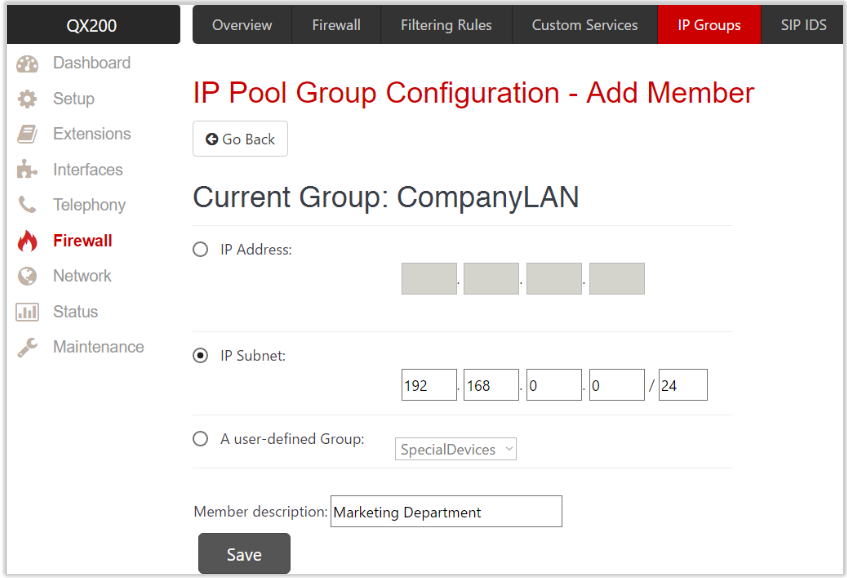


Figure 117: IP Pool Group Configuration – Add Member

9.5 SIP IDS

The **SIP IDS Settings** page includes the following components:

- **Enable SIP IDS** is used to enable service on QX.
- **Add the IP address into the Blocked IP List in Firewall** – if selected, the system will block the SIP attacker IP address by adding it to the **Blocked IP List**. This action will take effect if **Firewall** is enabled on QX.
- **Discard SIP messages from IP address for** – if selected, the system will ignore the SIP messages from attacker IP address for the specified time period after attack detection (default period is 32 seconds).
- **SIP IDS Exceptions** link leads to the **Exceptions for SIP IDS** page where you can specify the trusted IP address(es) that shouldn't be blocked.

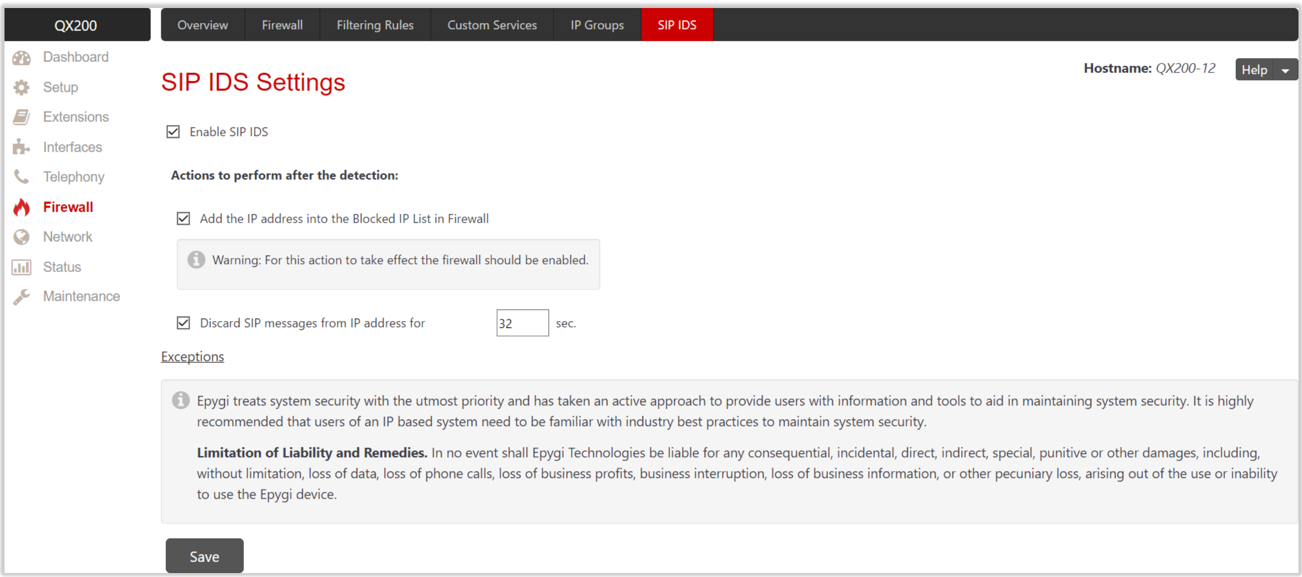


Figure 118: SIP IDS Settings page

To add a new **exception**:

1. Click the **SIP IDS Exceptions** link.
2. Click **Add** and enter the following information:
   - Enter the **IP Address**.
   - Enter the **Mask**. TIP: Enter **32** as a Mask to add only the IP address in exception list.
3. Click **Save** to add the new exception entry to the **SIP IDS Exceptions** table.

The Bad IP detection logic

The **Bad IP detection logic** is the following:

- 2 failures of SIP authorization/authentication from the same IP during **250** milliseconds.
- 2 messages causing **Non-self-Request-URI** from the same IP during **250** milliseconds.
- If there are **10** failures in a row during any period of time from the same IP, then the IP will be blocked.

Note: Any successful registration attempt from that IP will reset the counter. For example, if IP=xxx.xxx.xxx.xxx failed to register **9** times and then successfully registered on the **10**th attempt, then it resets the counter to **0**. Next time the same IP can make another **9** unsuccessful attempts before being blocked.

10 Network Menu

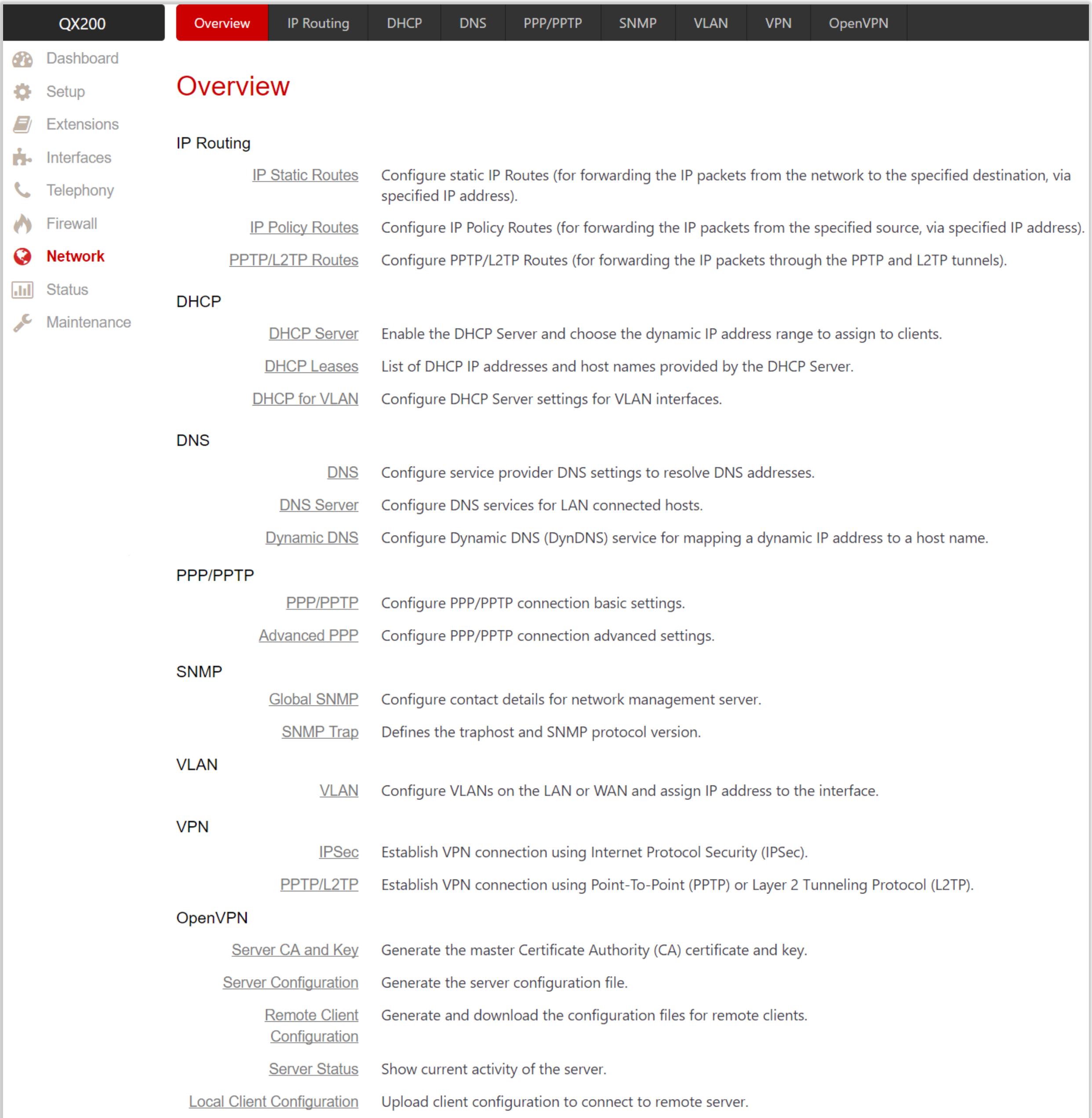


Figure 119: Network Menu overview

10.1 Second LAN Interface

The **Second LAN Interface Settings** page is used to activate second interface on the QX2000 and QX3000.

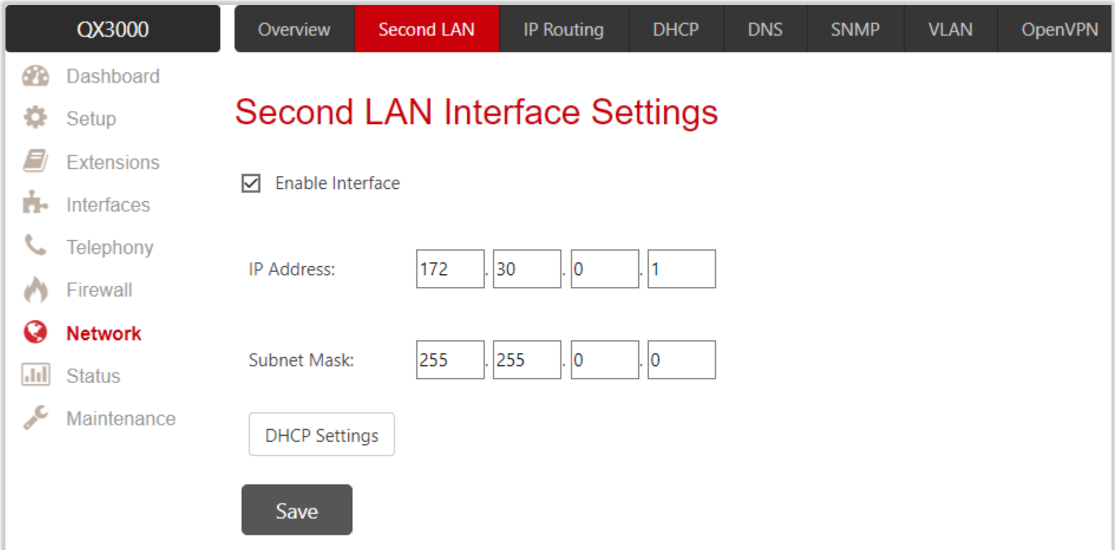


Figure 120: Second LAN Interface Settings page

This following settings (options) are available:

- **Enable Interface** is used to activate the second interface.
- **IP Address** is used to set the LAN IP address.
- **Subnet Mask** is used to set the subnet mask.
- **DHCP Settings** button is used to configure DHCP Settings on the second interface.

Note: The network for the second LAN interface shouldn't be overlapped with the main LAN interface.

10.2 IP Routing

QX **IP Routing** service allows to route IP packets from one destination to another (or to a specified router) through QX or QX VPN. The **IP Routing** is used to make IP Static, IP Policy and PPTP/L2TP routes for IP packets routing.

10.2.1 IP Static Routes

IP Static Routes are used to forward IP packets from the network (the QX is connected) to a specified destination.

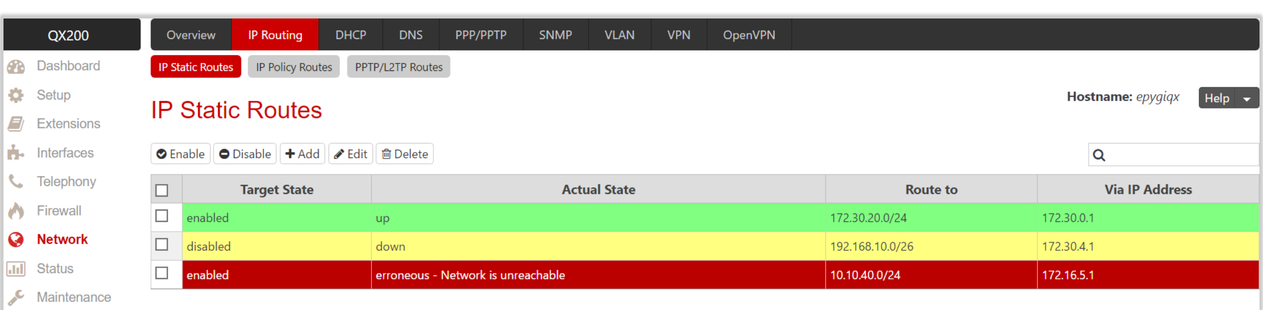


Figure 121: IP Static Routes page

To add a new IP Static Route:

1. Click Add and enter the following information:
   - Route to is used to set the IP address and subnet mask of the destination the IP packet will be routed to.
   - Via IP Address is used to set the IP address of the router that will forward the IP packet to the specified destination.
2. Click Save to add the new route to the IP Static Routes table.
3. Click Enable to activate the newly created route.

Note: The rule with the longest subnet (smallest IP range) will take effect when having two or more IP Static routing rules with the coinciding subnets.

10.2.2 IP Policy Routes

IP Policy Routes allows to forward IP packets to a specified router depending on the source IP address as well as set the priority for the current routing rule.

To add a new IP Policy Route:

1. Click Add and enter the following information:
   - Priority is used to set the priority of the routing rule. Enter any numeric value from the 1-252 range. The lower the number, the sooner the routing rule will take effect (higher priority).
   - From is used to set the packet source IP address and subnet mask of the specified destination to match with the rule.
   - Via IP Address is used to set the IP address of the subsequent router to forward the IP packet to.
2. Click Save to add the new route to the IP Policy Routes table.
3. Click Enable to activate the newly created route.
4. Click Raise Priority or Lower Priority to increase/decrease the priority of the selected policy route by one.

10.2.3 PPTP/L2TP Routes

PPTP/L2TP Routes allows to forward IP packets through the PPTP and L2TP tunnels of QX. VPN routes cannot be generated if PPTP/L2TP connections do not exist on QX.

To add a new PPTP/L2TP Route:

1. Click Add and enter the following information:
   - Route via is used to select the available PPTP or L2TP connection from the drop-down list. A connection selected from this list will be used to route the IP packet from the QX LAN to the peer behind the PPTP/L2TP tunnel.
   - Route to is used to set the IP address range of the possible peers behind the PPTP/L2TP tunnel the IP packets should be routed to.
2. Click Save to add the new route to the PPTP/L2TP Routes table.
3. Click Enable to activate the newly created route.

10.3 DHCP

10.3.1 DHCP Server

The **DHCP Settings for the LAN Interface** page is used to enable DHCP server and configure network parameters for DHCP server.

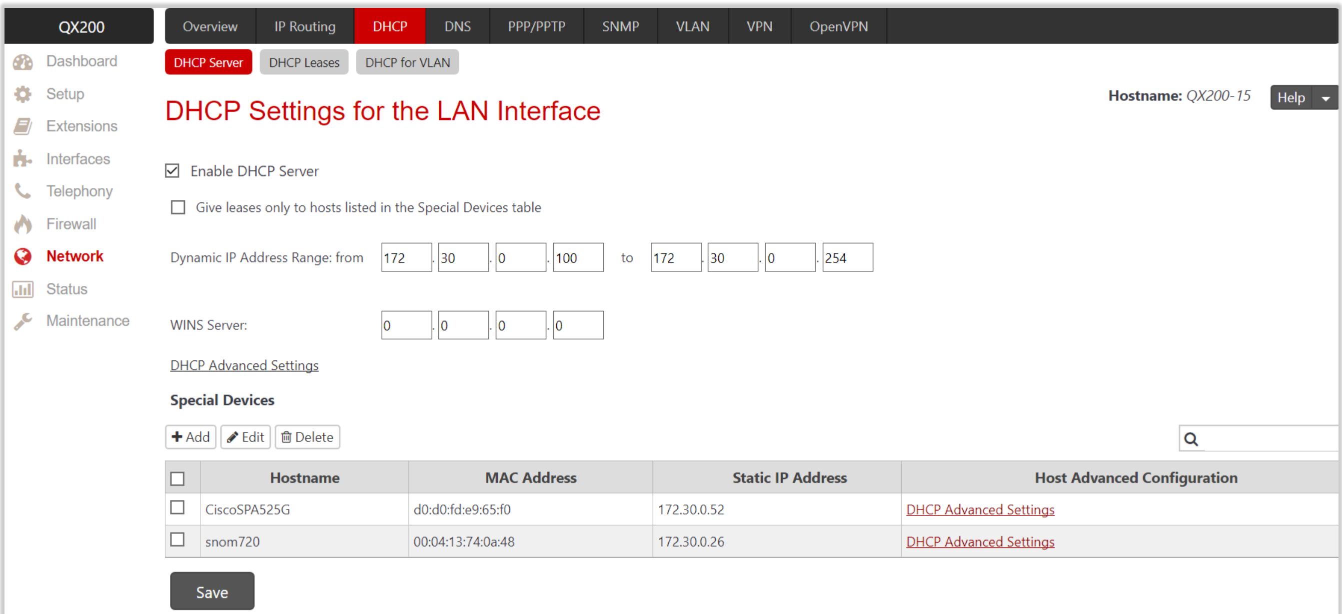


Figure 122: DHCP Settings page for the LAN interface page

The following settings (options) are available:

- **Enable DHCP Server** is used to activate DHCP service on LAN interface. If selected, the QX will be able to assign dynamic IP addresses to devices in QX LAN.
- **Give leases only to hosts listed in the Special Devices table** – if selected then DHCP services will be provided only to devices listed in the **Special Devices** table.
- **Dynamic IP Address Range (from to)** is used to set the range of IP addresses that will be assigned to devices (IP phones, PCs, etc.).
- **WINS Server** is used to set an IP address of WINS server.
- **DHCP Advanced Settings** leads to the DHCP Advanced Settings page to configure the advanced options for DHCP server.
- **Special Devices** allows to set a static IP address binding on the MAC address of the device. When this table is configured, the devices (with defined hostname and MAC address) will always get the same IP address from DHCP server. Devices not listed in this table will get dynamic IP addresses.

To add a new **Host**:

1. Click **Add** and enter the following information:
   - **Hostname**
   - **MAC Address**
   - **Static IP Address** is used to set a fixed IP address. TIP: If you leave this field blank, the device will get the first available IP address from the range defined in the **DHCP Settings** page.
2. Click **Save** to add the new host to the **Special Devices** table.

DHCP Advanced Settings

The **DHCP Advanced Settings** page is used to add new advanced options of the DHCP sever and modify the existing ones. The **DHCP Advanced Settings** table lists DHCP server default options. All options will be sent to the DHCP clients.

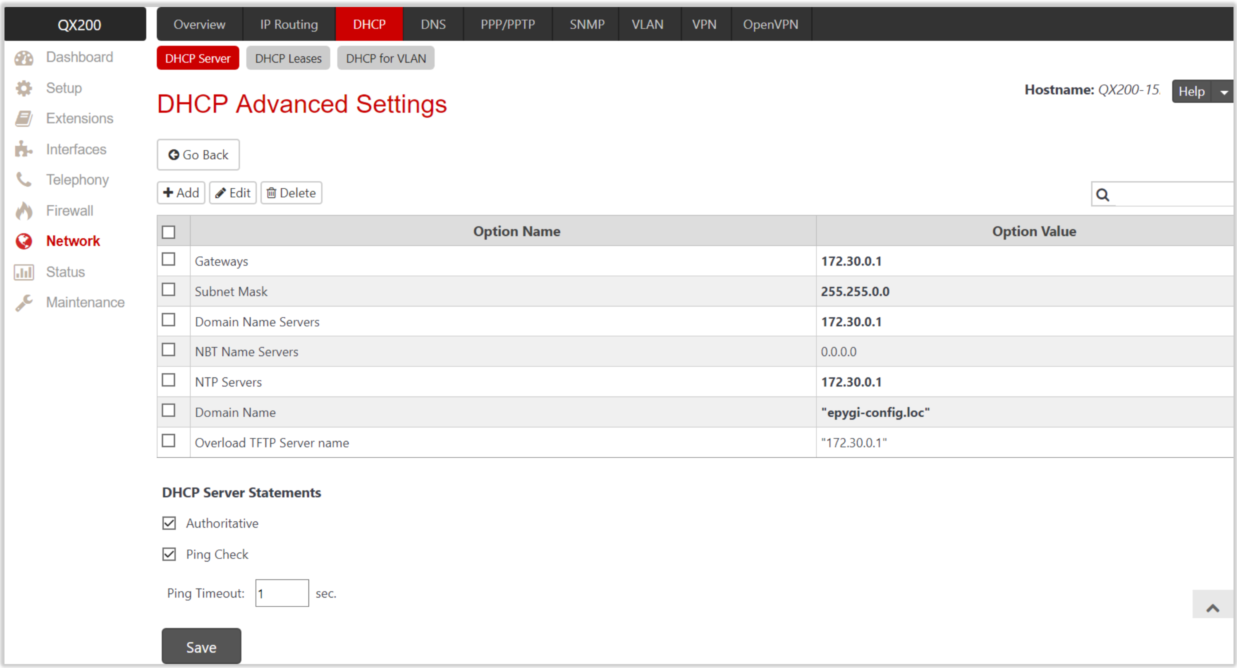


Figure 123: DHCP Advanced Settings page

To add a new **DHCP option**:

1. Click **Add** and enter the following information:
   - ➢ Select one of the predefined DHCP server options or define a custom one.
   - ➢ **Predefined Options** is used to select one of the predefined DHCP server options.
     - ♦ **Option Name** is used to select DHCP server option.
     - ♦ **Option Value** is used to set the value for the selected option. **Type** and **format** of the value depends on the option selected from the **Option Name** list.
   - ➢ **Custom Options** is used to define a new DHCP server option. The following parameters must be entered for a new option:
     - ♦ **Option Code** is used to set a code for the option. It may have values in a range from **0** to **255**.
     - ♦ **Option Type** is used to select the type of the option value. It may be an IP address, a Boolean or integer value, etc.
     - ♦ **Option Value** is used to set the value of the option. This value depends on the selected **Option Type**.
2. Click **Save** to add a new DHCP option to the **DHCP Advanced Settings** table.

Note:

- Use commas to separate values in case of multiple entries.
- The changes made through the **System Configuration Wizard** regarding the DHCP server options will not immediately reflect on the **DHCP Advanced Settings** if option parameters of DHCP sever are modified, so you will have to reconfigure changes in the **DHCP Advanced Settings** manually. The settings will be changed automatically if the parameters in DHCP server options are in "**bold**". In this case, the **DHCP Advanced Settings** will be changed automatically if you make changes through the **System Configuration Wizard**.

The following **DHCP Server Statements** are available:

- **Authoritative** is used to enable authoritative mode on DHCP server. TIP: If several DHCP servers are used on the network and QX has to provide network parameters to IP phones only, then disable this option.
- **Ping Check** – if selected, verifies the availability of an IP address on the network before providing it to a client. QX will first ping the IP address retrieved from the IP pool and wait for a reply. If no reply is received within a timeout specified in the **Ping Timeout**, the retrieved IP address will be provided to the client. Otherwise, a new IP address will be retrieved from the IP pool and the procedure will be repeated. If not selected, QX will provide an IP address immediately when requested.

10.3.2 DHCP Leases

The **DHCP Leases** table shows the list of clients that obtain a lease for an IP address from DHCP server. Before the lease expires, DHCP server will renew the lease for the client or the client will obtain a new lease. By default, the **DHCP address lease time** is 7 days.

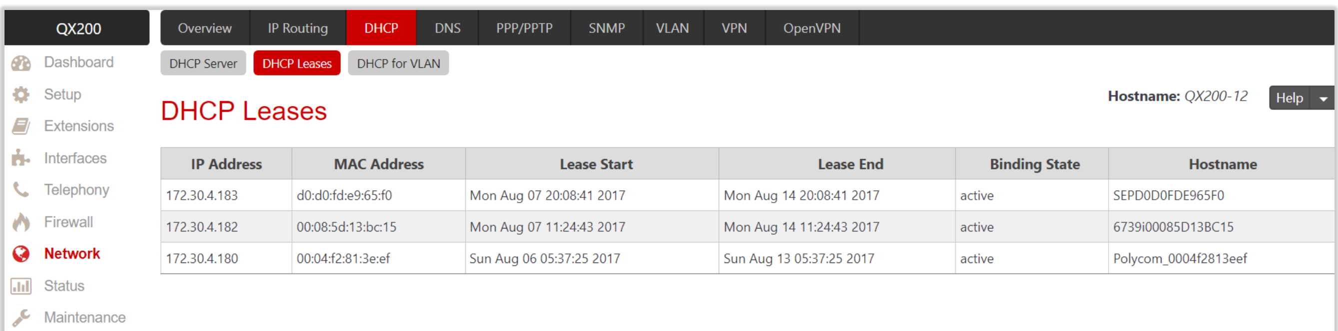


Figure 124: DHCP Leases page for LAN interface

10.3.3 DHCP for VLAN

The **DHCP Settings for the VLAN Interface** page is used to enable DHCP server and configure network parameters for DHCP server.

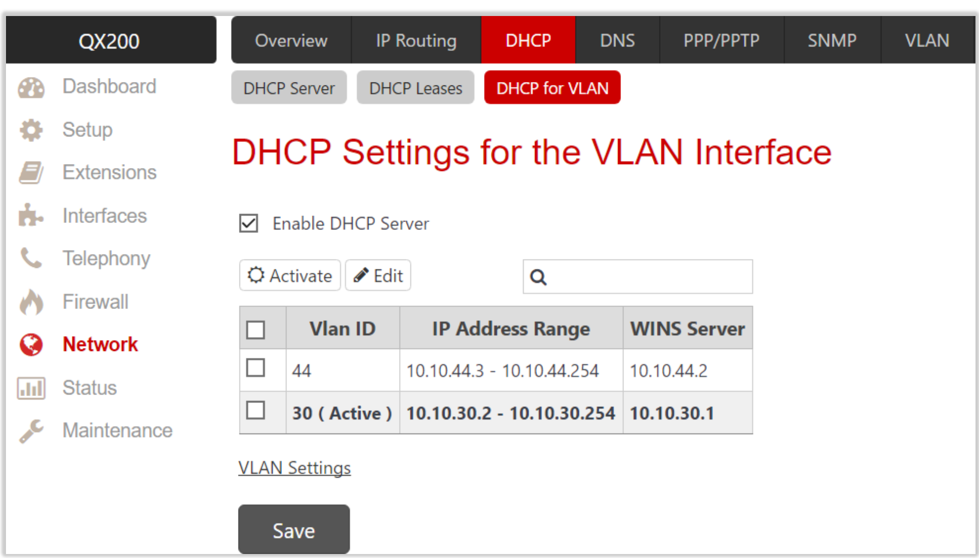


Figure 125: DHCP Settings for the VLAN Interface

The **DHCP Settings for the VLAN Interface** table lists all **enabled** VLAN interfaces created in the **VLAN Settings** page. The following settings (options) are available:

- **Enable DHCP Server** is used to activate the DHCP service. If selected, QX will be able to assign dynamic IP addresses to devices (e.g. IP phones, PCs, etc.).
- **Activate** is used to activate the DHCP server for the selected VLAN interface. TIP: The DHCP server can be activated only for one VLAN interface at once.
- **Edit** is used to modify the selected VLAN interface. This page contains all the same settings and options as the DHCP Server page.
- **Go to VLAN Settings** leads to the VLAN Settings page to create a new VLAN or modify existing ones.

10.4 DNS

10.4.1 DNS

The **DNS Settings** page allows to set up name server(s) for QX.

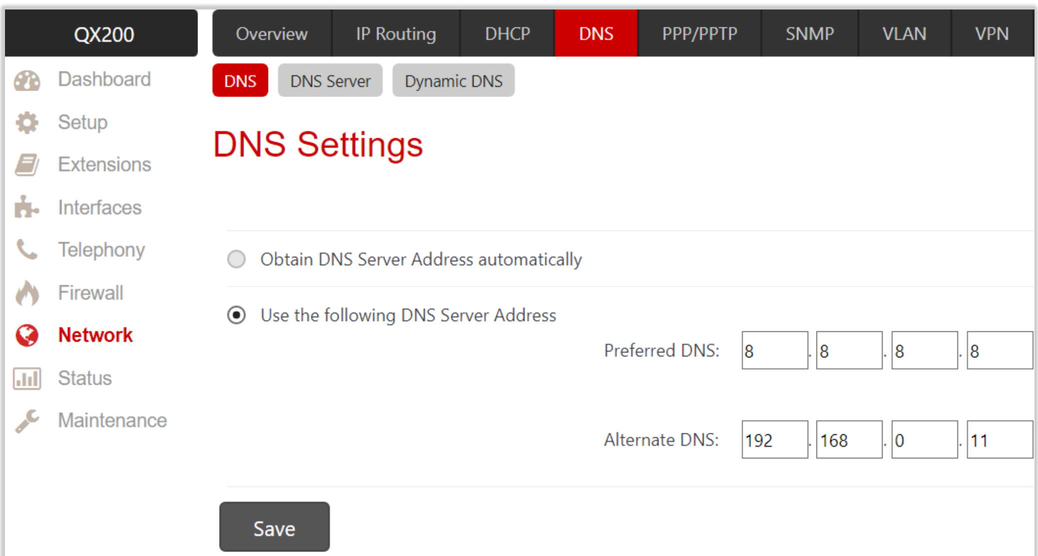


Figure 126: DNS Settings page

The following settings (options) are available:

- **Obtain DNS Server Address automatically** – if selected, QX will get the IP address of DNS server from local network or ISP automatically.
- **Use the following DNS Server Address** is used to manually assign a name server as follows:
  - **Preferred DNS** is used to set the IP address of name server.
  - **Alternate DNS** is used to set the IP address of the secondary name server that will be used if the main name server cannot be accessed.

10.4.2 DNS Server

The DNS Server returns the correct IP address to the requested domain name, so that any device located in the LAN side can be accessed by its hostname or alternative alias name. The DNS Server Settings page is used to configure DNS server settings on the QX and define a list of aliases for the devices.

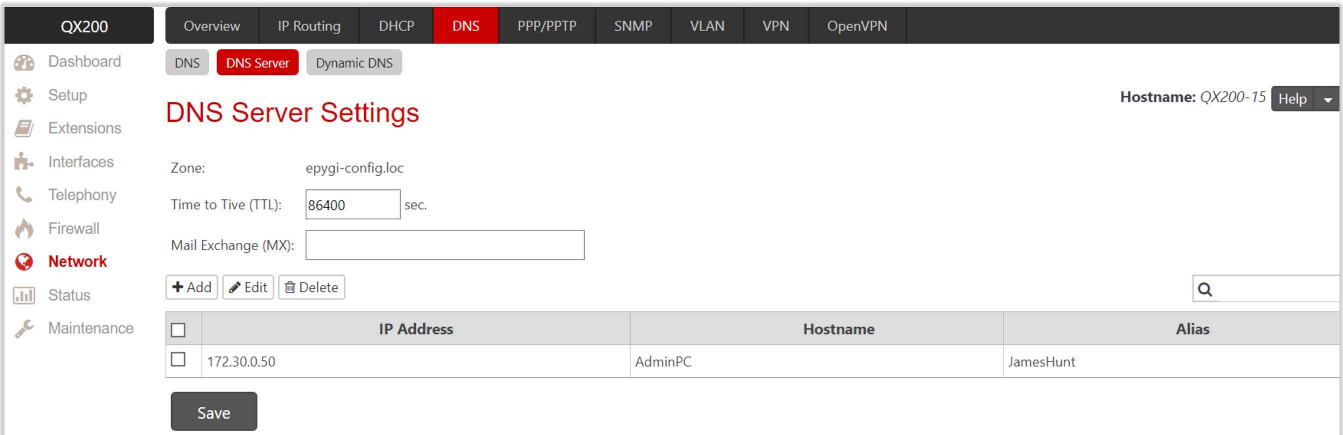


Figure 127: DNS Server Settings page

The following settings (options) are available:

- Zone displays the QX domain name as it is configured in the System Configuration Wizard.
- Time to Live (TTL) – indicates the time (in seconds) during which the DNS server will keep the resolved names in its cache. During this time, the same address will be resolved from the cache of the DNS server. Once the timeout expires, the requested address will be resolved newly.
- Mail Exchange (MX) – indicates the mail server's hostname. When resolving the email address, the reference will go to the mail server defined in this field, before being sent out to external network. The value in this field will be used in the MX record in the DNS server on the QX.

The DNS Server Settings table lists aliases for each host (device).

To add a new Host:

1. Click Add and enter the following information:
   - IP Address
   - Hostname
   - Alias is used to enter up to 5 alias names by which the device will be resolved.
2. Click Save to add the new host to the DNS Server Settings table.

10.4.3 Dynamic DNS

Dynamic DNS (DynDNS) associates your address with a consistent domain name without the need to buy a pricey static IP. Dynamic DNS can help by assigning a custom domain name to your IP address that will update automatically as your IP continues to change.

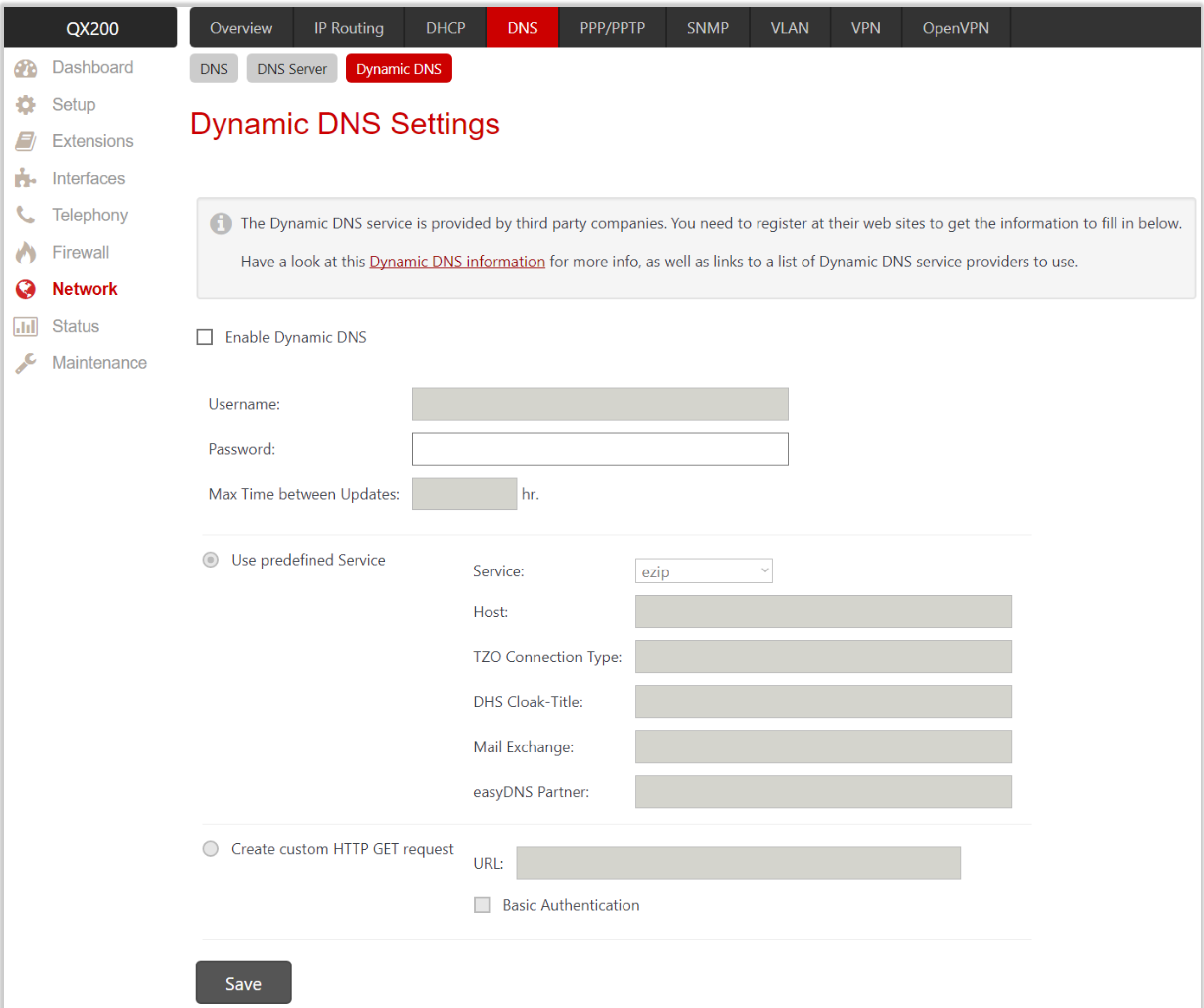


Figure 128: Dynamic DNS Settings page

The following settings (options) are available:

- Enable Dynamic DNS is used to activate service on QX. TIP: To activate the DynDNS service on ecQX, first, choose a DynDNS provider and register at the provider's website.
- Username and Password are used to set the authentication parameters specified during registration at the DynDNS provider.
- Max Time between updates is used to set the interval between two updates. The values entered in these fields should be greater than 24. Normally, whenever you set up a connection to the Internet, the DynDNS is updated at least once in the period indicated in this field.
- Use predefined Service enables the manual configuration of the DynDNS service.
  - Service is used to select the provider to be subscribed to.
  - Host is used to set the name of the host on the Internet.
  - TZO Connection Type is used to set the special parameter required by the TZO provider.
  - DHS Cloak-Title is used to set the special parameter required by the DHS provider.

- **Mail Exchange** is used to set the address of the e-mail server the DynDNS service provider will relay e-mails to. If this service is used, ensure that port forwarding is configured for SMTP to the internal e-mail server.
- **easyDNS Partner** is used to enter a special parameter required by the DynDNS provider easyDNS.

- **Create Custom HTTP GET Request** is used to switch to the custom settings of the DynDNS service. Normally, the DynDNS provider uses HTTP get requests to map dynamic IP addresses to host names. If the **HTTP receive request** is known to you, click **Create Custom HTTP GET Request** and enter the appropriate value into the URL field.
  - **URL** is used to set the complete request to be sent to the DynDNS server. The request modifies the name server database so that the hostname will be resolved to the new IP address.
  - **Basic Authentication** enables the encoding of the username and password entered in the text fields above, and then uses the **Basic Authentication** method to notify the provider about the user's authentication settings. Most of the DynDNS providers require an authentication for security. Authentication parameters can be provided in the URL field to be used for the HTTP GET request. Select **Basic Authentication** if no authentication parameters are provided.

10.5 PPP/PPTP

10.5.1 PPP/PPTP

The **PPP/PPTP Settings** page is used to establish a connection over the DSL link, or any other type of uplink, to ISP. A connection is needed to set up and make or receive calls through PPP over Ethernet. The connection may be configured for manual setup or always up. Once a connection has been established between the QX and the provider, QX users will be able to make and receive calls at any time. The following settings (options) are available:

- **PPTP Server** is used to set the IP address of the PPTP server.
- **Encryption** is used to select the encryption for the traffic over the PPTP interface.
- **Keep Connection Alive** is used to keep the connection alive by sending control packets to verify the link state.
- **Authentication Settings** is used to set the authentication parameters (Username and Password) to register on the ISP server.
  - **Dial manually** – if selected, a button will be displayed on the **Dashboard** to switch the connection on/off.
  - **Always connected** – if selected, the connection will always stay active and connected.
- **IP Address Assignment** is used to select the IP address assignment type:
  - **Obtain an IP Address automatically** – if selected, QX will get the IP address

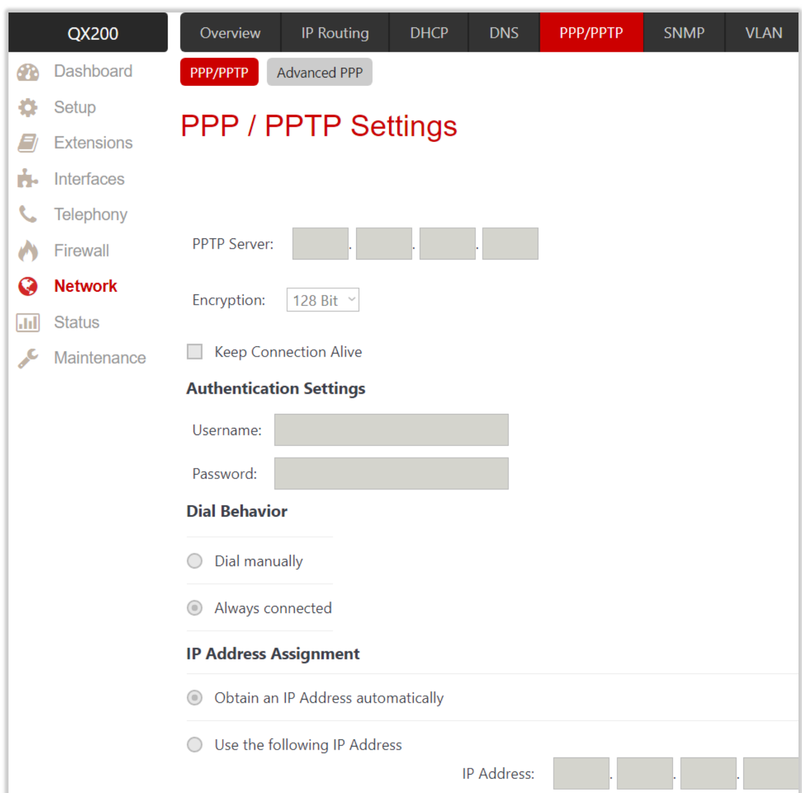


Figure 129: PPP/PPTP Settings page

from local network or ISP automatically.

➢ **Use the following IP Address** is used to set the IP address manually.

10.5.2 Advanced PPP

The **Advanced PPP Settings** page is used to enable certain settings (options) of the negotiation process during connection establishment. These settings are available only if QX has a PPPoE WAN interface.

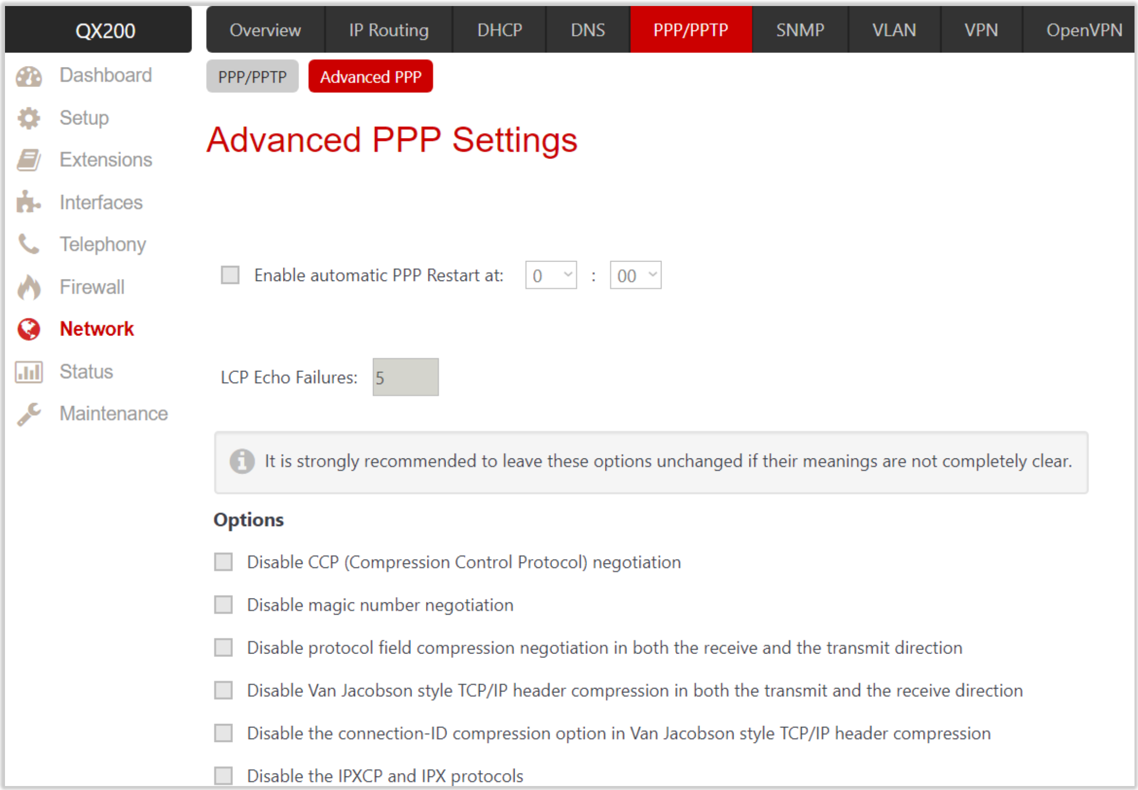


Figure 130: Advanced PPP Settings page

The following settings (options) are available:

- **Enable automatic PPP Restart** is used to select the time when the PPP connection will automatically be restarted.
- **LCP Echo Failures** shows the number of the LCP echo failure packets received before the PPP connection will be considered as dead and will be restarted.
- **Disable CCP (Compression Control Protocol) negotiation** is used to select if the peer system is not working properly. For example, if it is not accepting requests from the PPPD (Point-to-Point Daemon) for CCP negotiation.
- **Disable magic number negotiation** is used to select if the peer system is not working properly. If selected, PPP cannot detect a looped-back line.
- **Disable protocol field compression negotiation in both the receive and the transmit direction** – if selected, no protocol field compression will take place.
- **Disable Van Jacobson style TCP/IP header compression in both the transmit and the receive direction** – if selected, no negotiation of TCP/IP header compression will take place and the header will always be sent uncompressed.
- **Disable the connection-ID compression option in Van Jacobson style TCP/IP header compression** – if selected, PPPD will not compress the connection-ID byte from Van Jacobson and will not ask the peer to do so.

- **Disable the IPXCP and IPX protocols** is used to select if the peer is not working properly and cannot handle requests from PPPD for IPXCP negotiation.

Note: It is strongly recommended to leave these switches unchanged if their meanings are not completely clear.

10.6 SNMP

Simple Network Management Protocol (SNMP) is an application layer protocol that facilitates the exchange of management information between network devices and is used by network administrators to manage network performance, find and solve network problems, and plan for network growth. The SNMP agent is running to allow administrators to remotely manage QX network and device configuration.

For more information on how to configure and use **SNMP**, refer to the Configuring SNMP Agent on QX IP PBXs guide.

10.7 VLAN

The **VLAN Settings** page is used to create new VLAN interface(s) or modify the existing ones. The **VLAN Settings** table lists all existing virtual interfaces on QX.

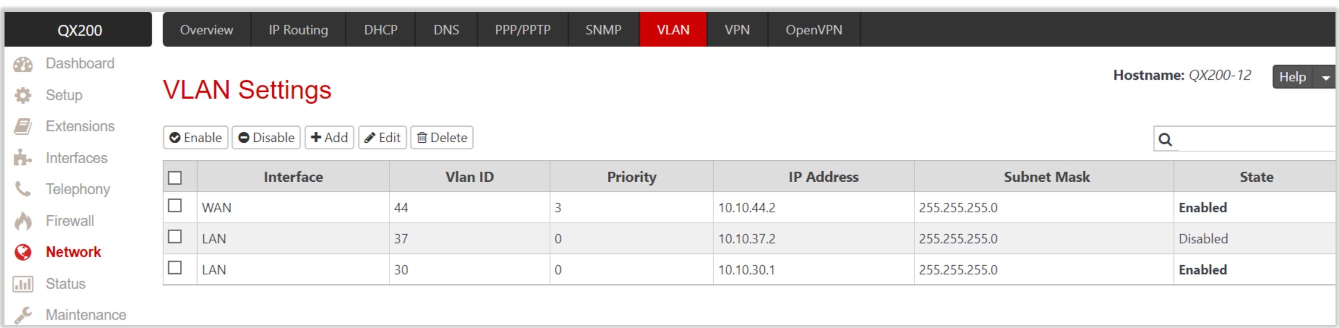


Figure 131: VLAN Settings page

To configure a new **VLAN interface**:

1. Click **Add** and enter the following information:
   - **Enable** is used to activate virtual interface after creating it.
   - **Interface Type** is used to select whether the virtual interface will be created on LAN or WAN interface.
   - **VLAN ID** is used to set the virtual network ID from the range of **0** to **4094**.
   - **Priority** is used to select the priority of packets in the corresponding interface. Packets with the lower priority (0) will be delivered first.
   - **IP Address** is used to set the IP address.
   - **Subnet Mask** is used to set the subnet mask.
2. Click **Save** to add the new interface to the **VLAN Settings** table.

10.8 VPN

Virtual Private Network (VPN) is used to connect two local networks (intranets) securely over the Internet. The VPN routers manage authentication between servers and clients and handle data encryption for the connection. In general, VPN connection is similar to Internet connection, both of them are based on IP detection.

VPN gateway must authenticate the IP addresses of the partner VPN gateway(s). Each time a specific VPN is to be established, the same IP addresses are usually expected. This will not create problems if both VPN partners have fixed WAN IP addresses. In some cases, you may use dynamically allocated IP addresses. Devices that use a dynamic IP address as part of a VPN, are turned into **Road Warriors** once they are enabled. Every VPN needs at least one VPN gateway with a fixed IP address.

The endpoints of a VPN must have different WAN IP addresses, and if they are connected to LAN, the LAN must have different IP addresses. As all QX devices have the same default IP addresses on delivery, at least one of them must be reconfigured in order to set a new IP address.

QX supports the following VPN connections: **IPSec**, **PPTP** and **L2TP**. Note: It is strongly recommended not to run different types of VPN tunnels between the same endpoints simultaneously.

10.8.1 IPSec

Internet Protocol security (IPSec) is a framework of open standards which aim to ensure private, secure communications over IP networks through the use of cryptographic security services. IPSec supports network-level data integrity, data confidentiality, data origin authentication, and replay protection. As IPSec is integrated at the Internet layer (layer 3), it provides security for almost all protocols in the TCP/IP suite, and as IPSec is applied transparently to applications, there is no need to configure separate security for each application that uses TCP/IP.

Connection

The **Connection** sub-page is used to create a new IPSec connection or manage the existing ones.

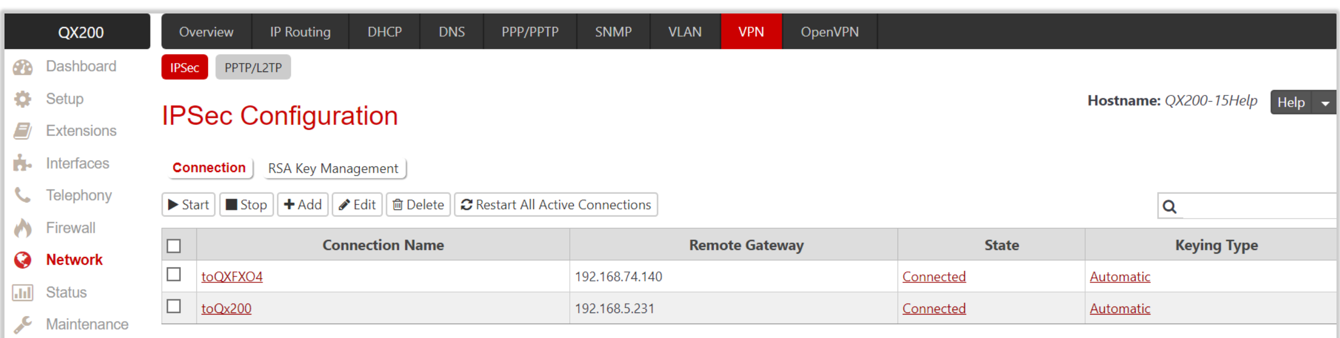


Figure 132: IPSec Configuration – Connection Settings page

Click **Add** to run **IPSec Configuration Wizard** and configure a new connection. The wizard consists of the following sections:

- New IPSec Connection
- IPSec Keying Properties
- Automatic Keying
- IPSec Connection Properties
- Summary

New IPSec Connection

- **Connection Name** is used to enter the name of a new IPSec connection.
- **Peer Type** is used to select the remote machine type for the connection. If the required type of machine is not listed, choose **Other**.

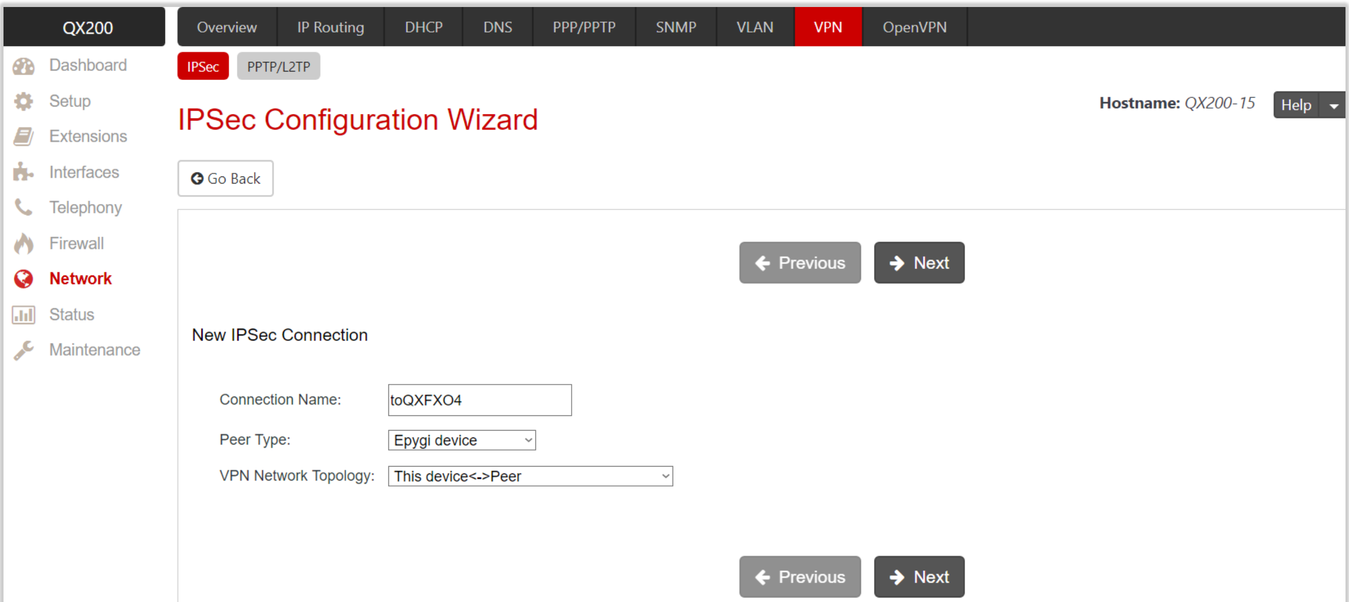


Figure 133: New IPSec Connection section

- **VPN Network Topology** is used to select the location of the peers participating to the VPN connection. The following options are available:
  - **This device<>Peer** – direct connection between QX and peer.
  - **This device<>[Internet]<>Peer** – connection between QX and peer over Internet.
  - **This device<>NAT<>[Internet]<>Peer** – connection between QX and peer over Internet through QX provider's NAT.
  - **This device<>[Internet]<>NAT<>Peer** – connection between QX and peer over Internet through peer provider's NAT.

IPSec Keying Properties

The **Internet Key Exchange** (IKE) and **Encapsulated Security Payload** (ESP) parameters are used to define the security of your IPSec tunnel.

The IKE parameters group is used to set up **Security Association** (SA) in the IPSec protocol suite.

- **Encryption** is used to select encryption standard. The following standards are available:
  - **Triple DES** uses three different keys on a single data block to achieve a higher security than is available from a single DES pass (block cipher algorithm with 64-bit blocks and a 56-bit key).
  - **AES (128 bit)** cryptography scheme is a symmetric block cipher, which encrypts and decrypts 128-bit blocks of data.
  - **AES (192 bit)** cryptography scheme is a symmetric block cipher, which encrypts and decrypts 192-bit blocks of data.
  - **AES (256 bit)** cryptography scheme is a symmetric block cipher, which encrypts and decrypts 256-bit blocks of data.

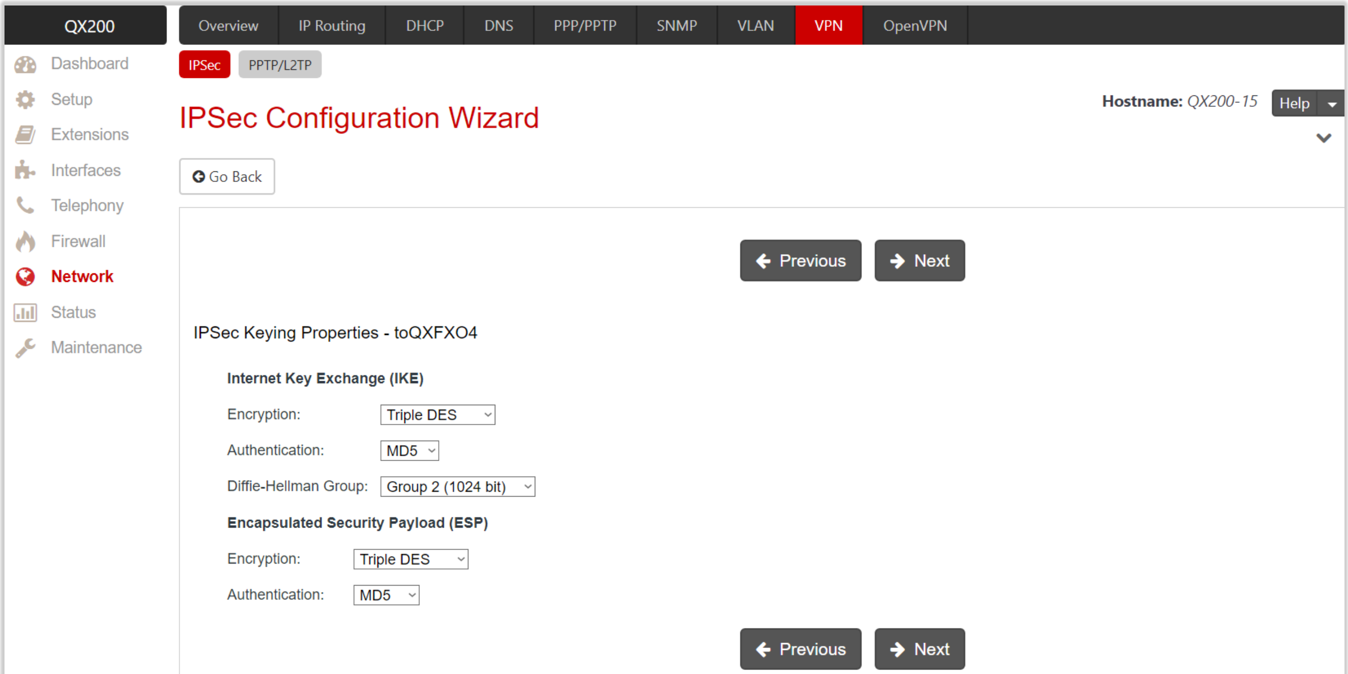


Figure 134: IPSec Keying Properties section

- Authentication is used to select authentication type:
  - SHA (Secure Hash Algorithm) is a strong digest algorithm proposed by the US NIST (National Institute of Standards and Technology) agency as a standard digest algorithm and is used in the Digital Signature standard, FIPS number 186 from NIST. SHA is an improved variant of MD4 producing a 160-bit hash. SHA and MD5 are the message digest algorithms available in IPSec.
  - MD5 (Message Digest) is a hash algorithm that makes a checksum over the messages. The checksum is sent with the data and enables the receiver to notice whether the data has been altered.
- Diffie-Hellman Group is used to determine the length of the base prime numbers used during the key exchange process. The cryptographic strength of any key derived depends, in part, on the strength of the Diffie-Hellman group, which is based upon the prime numbers. The higher is the group bit rate, the better is encryption. If mismatched groups are specified on each peer, negotiation fails.

The ESP parameters group is used to provide authenticity, integrity and confidentiality protection of packets. The same IKE encryption and authentication parameters are used.

Automatic Keying

The **Automatic Keying** section is used to specify a **Shared Secret** password or RSA public key to secure the IPSec connection.

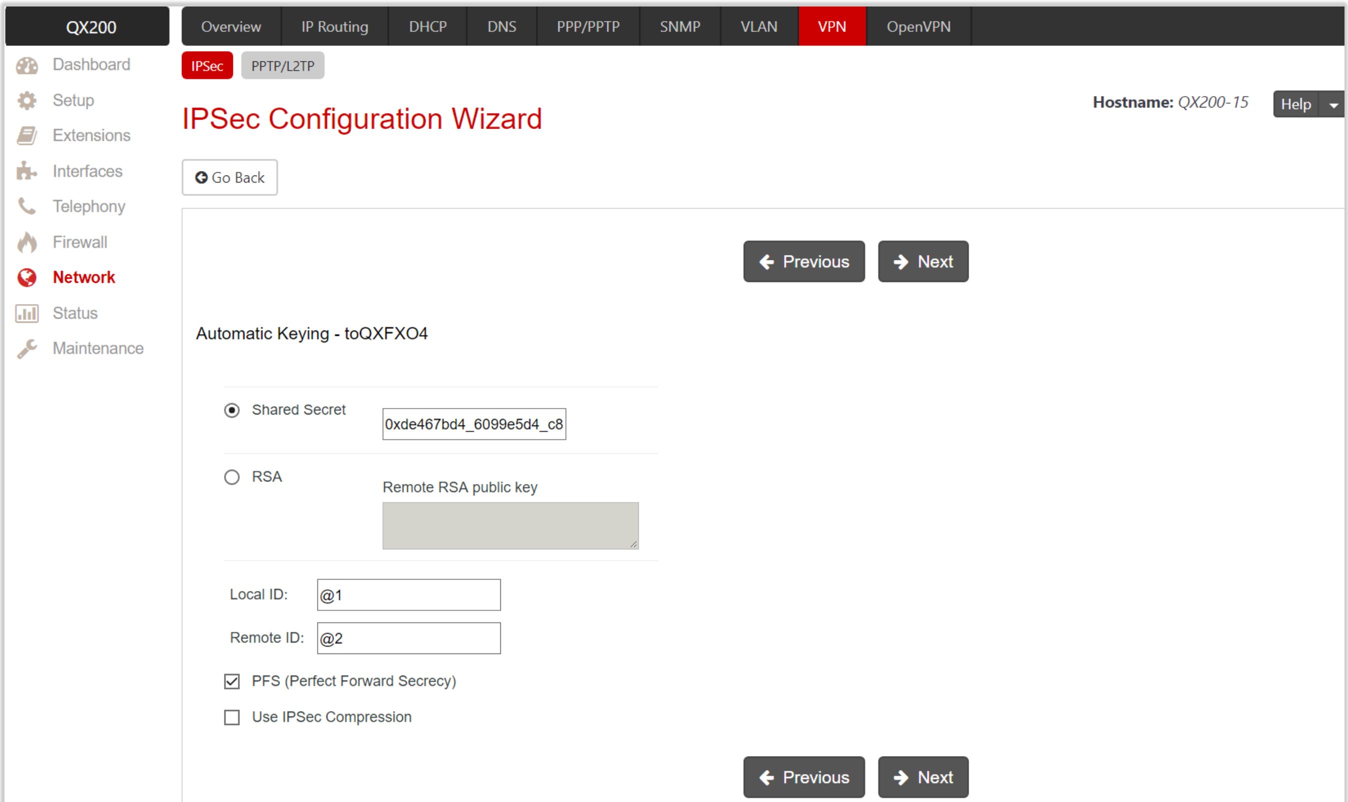


Figure 135: Automatic Keying section

- **Shared Secret** is a type of password that both of the IPSec connection partners must know. The authentication will be done with this shared secret. All encryption functions below will remain concealed.
- **RSA** is used to define the public RSA key of your IPSec connection partner.
- **Local ID** is used to set the QX **Fully Qualified Domain Name** (FQDN) that is resolved to an IP address, or any @-ed string that is used in the same way.
- **Remote ID** is used to set the remote endpoint FQDN that is resolved to an IP address, or any @-ed string that is used in the same way.

The **Local ID** and **Remote ID** fields can have the values in one of the formats presented below:

- ♦ **IP address** – for example: 10.1.19.32.
- ♦ **Hostname** – for example: vpn.epygi.com. This form requires additional resources to resolve the hostname, therefore it is not recommended to use this format.
- ♦ **@FQDN** – for example: @vpn.epygi.com. This form is considered as a string and is not being resolved. It is recommended to use this form for most applications.
- ♦ **user@FQDN** – for example: test1@vpn.epygi.com. This form is also considered as a string and is not being resolved. It has no advantage over the previous form.

- **PFS** (**Perfect Forward Secrecy**) is a procedure of a system key exchange, which uses a long-term key and generates short-term keys as is required. Thus, an attacker who acquires the long-term key can neither read previous messages that they may have captured nor read future ones.
- **Use IPSec Compression** enables IPSec data compression. This option is displayed only if the IPSec-VPN partner supports it.

Note:

- It is not recommended to start multiple road warrior connections with the Shared Secret option selected. In order to start multiple road warriors simultaneously, it is recommended to use RSA option with Local ID and Remote ID fields configured.
- QX will prevent to start a connection with Shared Secret option selected if there is already a connection with RSA option started, and vice-versa.
- The Local ID and Remote ID values are mandatory for the RSA selection and are optional for Shared Secret. However, it is recommended to define the Local ID and Remote ID values for multiple road-warrior connections.

IPSec Connection Properties

The Dynamic IP/Road Warrior and Static IP/ Remote Gateway radio buttons are used to select whether the remote QX (or another VPN gateway device) is connected to the Internet with a dynamic IP address and is acting as a Road Warrior or is connected to the Internet with a fixed IP address and is acting as a VPN Gateway. The following settings (options) is used to configure IPSec connection:

- Dynamic IP/RoadWarrior – if selected, the Remote Gateway IP Address field will automatically generate the value "any", to allow access regardless of IP address.
- Static IP/Remote Gateway is used to set the IP address or hostname of the remote QX (or another VPN gateway device) in the Remote Gateway field.
- This device<>Remote Gateway allows to access from the local QX to the remote VPN gateway (local subnet and remote subnet are not included). This includes management access. The checkbox is disabled if the This device<>NAT<>[Internet]<>Peer or This device<>[Internet]<>NAT<>Peer option is selected from the VPN Network Topology drop-down list on the New IPSec Connection section.
- Local Subnet<>Remote Gateway allows to access from all stations connected to the local network to the remote VPN gateway device (local QX and remote subnet are not included). The checkbox is disabled when the This device<>[Internet]<>NAT<>Peer option is selected from the VPN Network Topology drop-down list on the New IPSec Connection section.
- This device<>Remote Subnet allows to access from the local QX to all stations of the remote LAN (local subnet and remote VPN gateway devices are not included). The checkbox is disabled when the This device<>NAT<>[Internet]<>Peer option is selected from the VPN Network Topology drop-down list on the New IPSec Connection section.
- Local Subnet<>Remote Subnet allows to access from all stations of the local network to all stations of the remote LAN (VPN gateway devices are not included). In this case, the local and remote subnet IP addresses and subnet masks have to be entered in the Local Subnet IP and Remote Subnet IP fields.
- Stop connection if not successful allows to stop the IPSec connection attempts if the partner remains unreachable after the timeout period. If not selected, the system will continue to try to reach the IPSec connection partner.

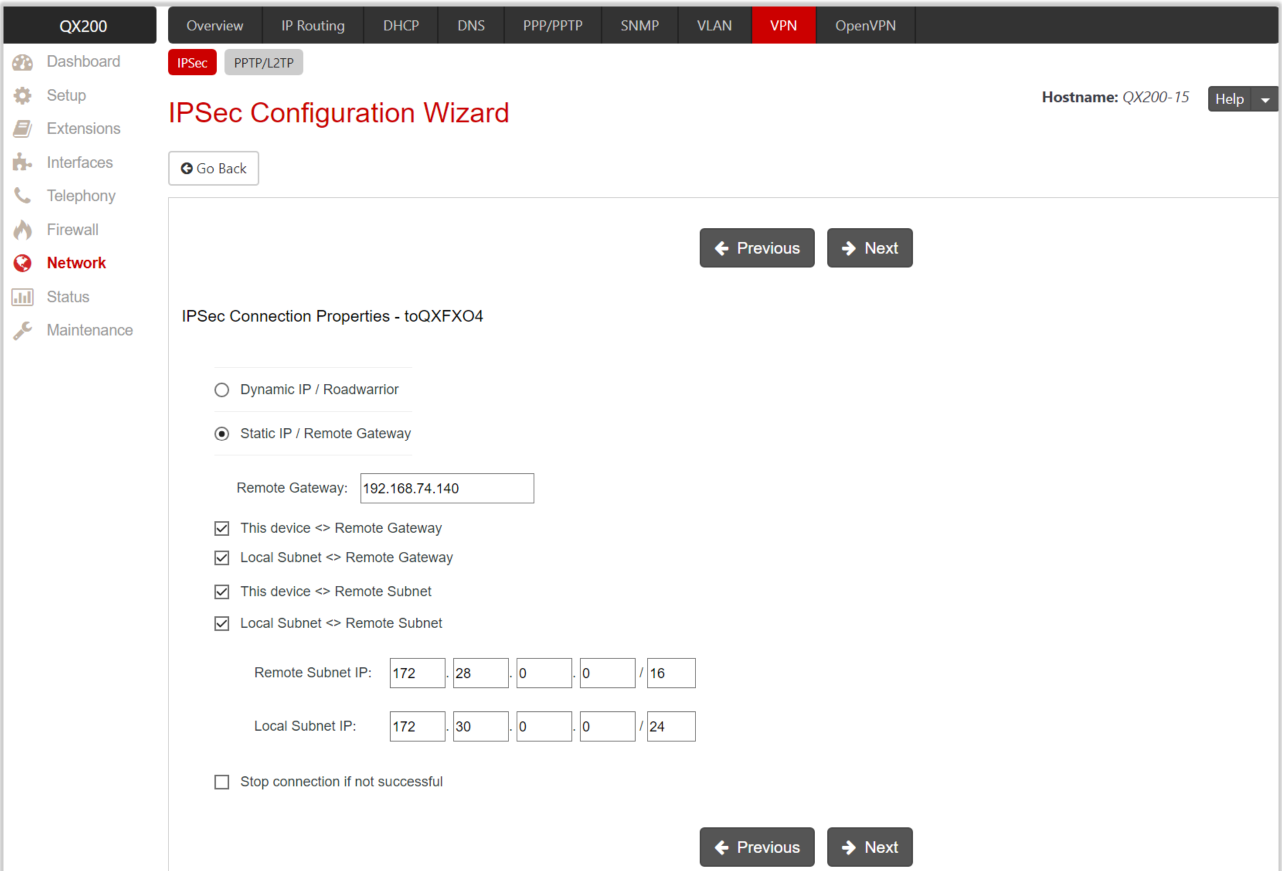


Figure 136: IPSec Connection Properties section

Note:

- It is not recommended to simultaneously start a static and a dynamic connection configured to use the same secret key. A dynamic connection may capture the static connection peer and vice versa, depending on which connection is established first.
- The **Static IP/ Remote Gateway** selection is not applicable if the QX is positioned behind NAT, since the IP address of the remote gateway is not reachable directly in this case.

Summary

This section displays all configured settings (options) before applying them.

RSA Key Management

The RSA Key Management sub-page is used to generate a new RSA public key. This page also displays the current public RSA key and allows to send it to the IPSec connection partner.

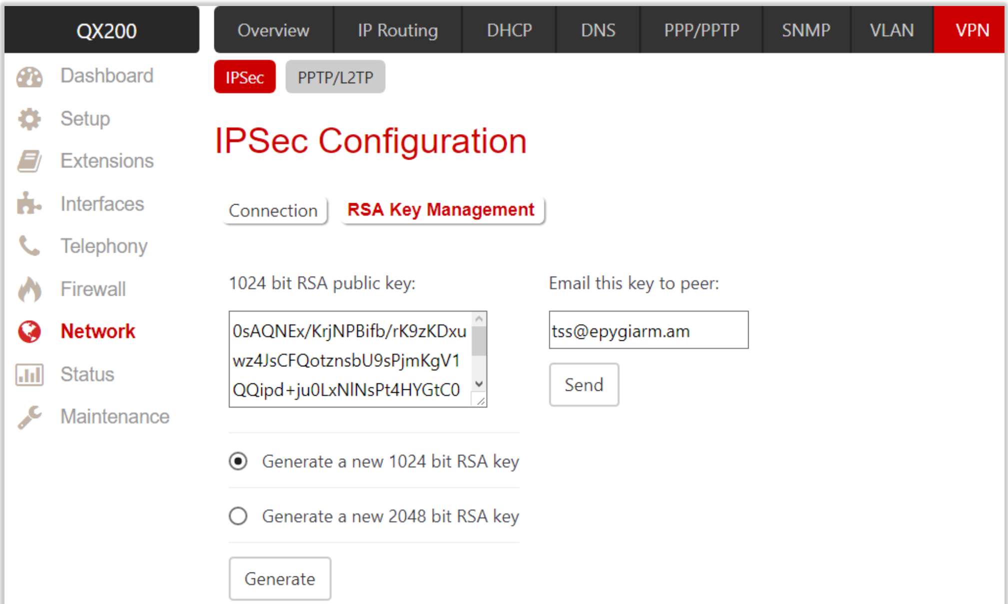


Figure 137: RSA Key Management page

To generate a new RSA key:

1. Select one of two available RSA key lengths (1024 or 2048).
2. Click Generate to generate the key.
3. Enter the e-mail address and click Send to send the generated key to the partner via e-mail.

Note: A pair of keys will always be generated, a public one and a private one. The previously generated pair of keys will become invalid as well as all existing IPSec connections that use RSA keying.

10.8.2 PPTP/L2TP

Point-to-Point Tunneling Protocol (PPTP) is a set of communication rules that govern the secure implementation of VPNs, which allows organizations to extend their own private networks over the public Internet via "tunnels". PPTP enables the creation of a secure route of data transfer from a remote client to a server in a private enterprise network through the creation of a VPN over TCP/IP-based networks, such as the Internet. It allows remote users to securely access corporate networks over the Internet, as if the client is physically present in the corporate network.

Layer 2 Tunneling Protocol (L2TP) is a networking protocol used by ISPs to enable VPN operations. L2TP is similar to the Data Link Layer Protocol in the OSI reference model, but it is actually a session layer protocol.

For PPTP and L2TP connections, two parties are required: Client and Server. The client is responsible for establishing connection. The server is waiting for clients; it is not able to initiate the connection itself. Servers define the range of IP addresses that are assigned to the Server and Clients participating in the connection. Each side is specified by the Hostname and Password. The client should know the server name and password (QX server has no password) and the server should set the client's hostname and password. The client and server settings have to match on both sides for successful establishment of connection.

Note:

- L2TP tunnels have no data encryption mechanism.
- Only one client can be connected to the server in the same network.
- After creating a PPTP server connection, PPTP connections between devices placed on the QX LAN and external devices will no longer be possible. The PPTP pass-through service for incoming and outgoing traffic will be automatically disallowed once a PPTP server connection is created.

Connections

The **Connections** sub-page is used to create a new PPTP or L2TP connection or manage the existing ones.

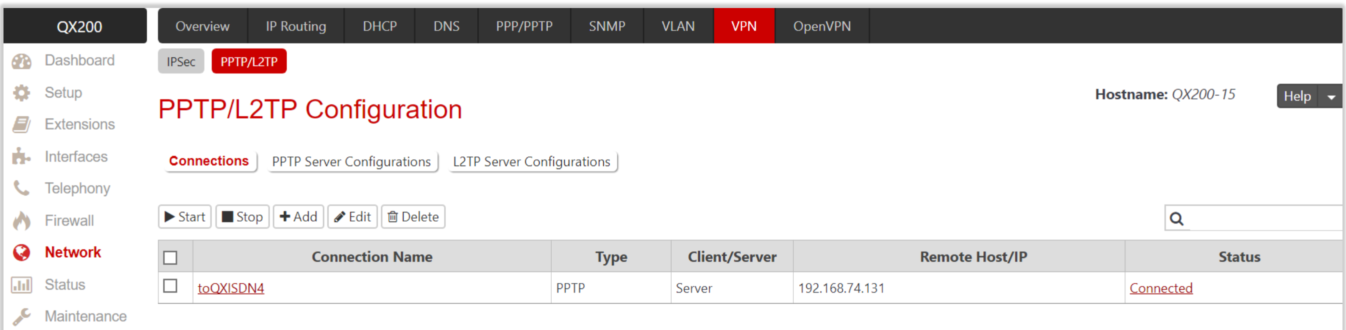


Figure 138: PPTP/L2TP Configuration – Connections page

Click **Add** to run **PPTP/L2TP Connection Wizard** and configure a new connection. The wizard consists of the following sections:

- New PPTP/L2TP Connection
- PPTP Connection Properties
- Summary

New PPTP/L2TP Connection

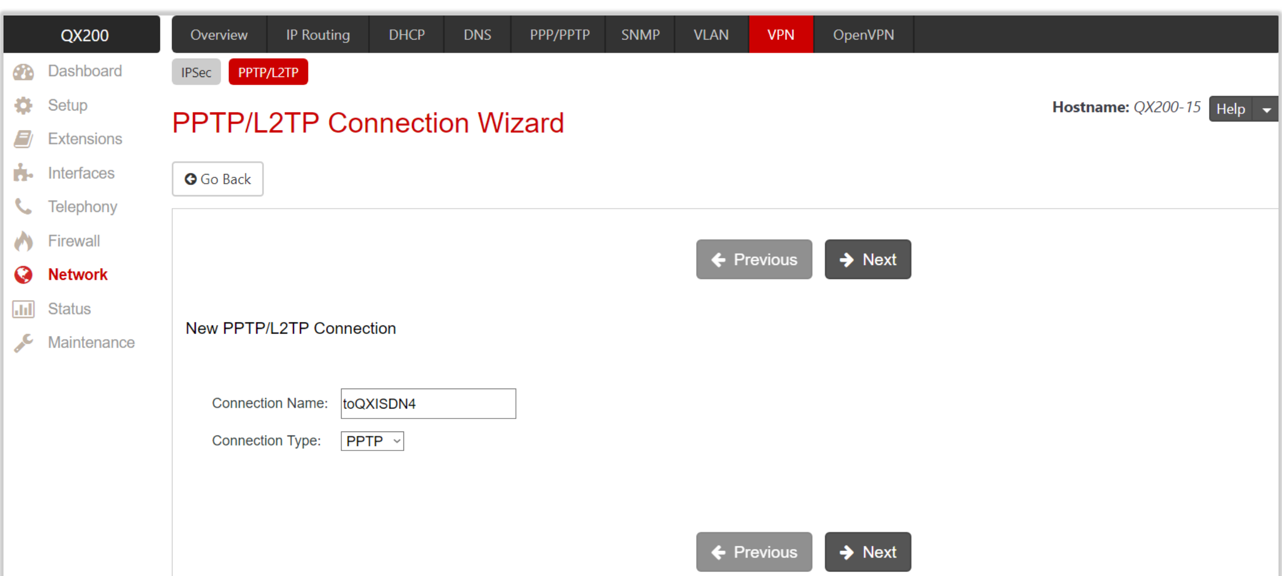


Figure 139: New PPTP/L2TP Connection section

- Connection Name is used to enter the connection name.
- Connection Type is used to select the type of connection (PPTP or L2TP).

PPTP Connection Properties

- Peer Name is used to enter the connection peer name. TIP: The Peer Name must be written with Latin characters. When creating a connection with a Windows server, ensure that a user with QX hostname and Dial-in access exists on the server. When creating a connection with a Windows client, ensure that the specified Peer Name matches the Dial-in connection username.
- Password is used to enter the password.
- Server/Client is used to select whether the new connection will be a server or client. The following information should be configured when the Client option is selected:
  - PPTP Server (if the PPTP connection type is selected) is used to set the IP address or hostname of the PPTP server.
  - L2TP Server (if the L2TP connection type is selected) is used to set the IP address of the L2TP server.
- Authentication is used to select the authentication protocol. This section is available only if the PPTP connection type is selected on the previous section. The MSCHAPv2 selection enables the Encryption drop-down list where the encryption method can be selected. TIP: These authentication settings should be identically configured on both peers for the successful connection.

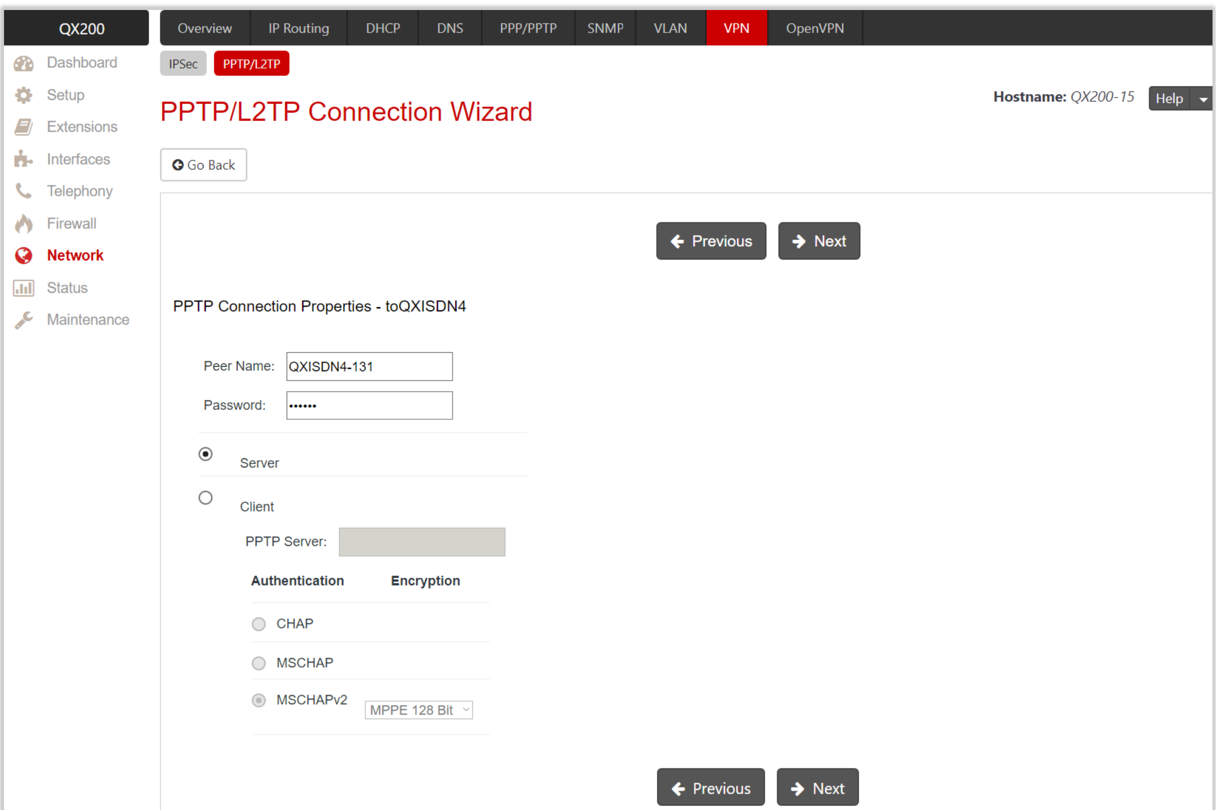


Figure 140: PPTP/L2TP Connection Properties section

Summary

This section displays all configured settings (options) before applying them.

PPTP Server Configurations

The **PPTP Server Configuration** sub-page is used to configure the PPTP server settings.

- **Subnet** is used to set the IP address range for the PPTP server and clients within the PPTP tunnel. TIP: The first address specified in the **PPTP Subnet** will be assigned to the PPTP server, others will be assigned to the clients. The PPTP server subnet must be different from the L2TP server subnet.
- **Authentication** is used to select the authentication protocol through which the client will communicate with the server. TIP: The **MSCHAPv2** selection enables **Encryption** drop-down list where the encryption method can be selected.

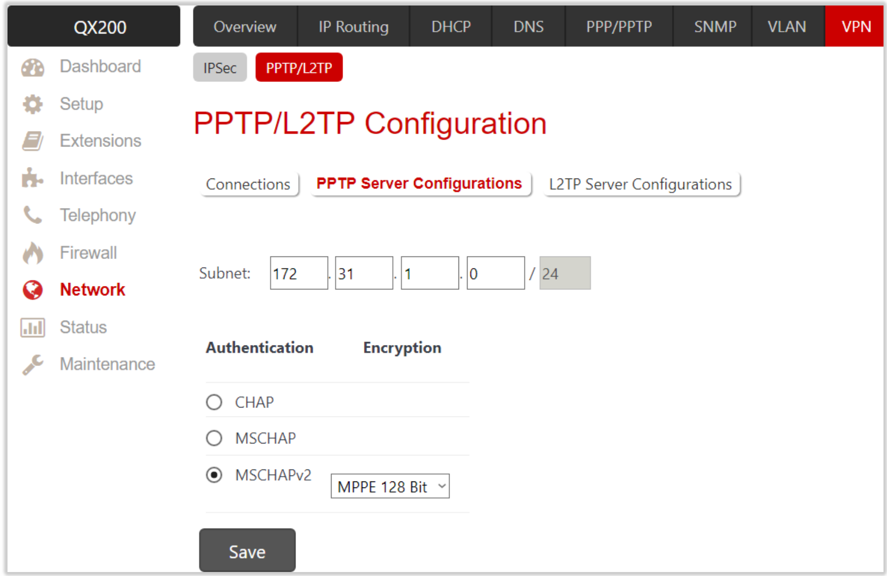


Figure 141: PPTP Server Configuration page

L2TP Server Configuration

The **L2TP Server Configuration** sub-page is used to configure the L2TP server settings. The **Subnet** is used to set the IP address range for the L2TP server and clients within the L2TP tunnel. TIP: The first address specified in the **L2TP Subnet** will be assigned to the L2TP server, others will be assigned to the clients. The L2TP server subnet must be different from the PPTP server subnet.

10.9 OpenVPN

OpenVPN allows secure point-to-point or site-to-site connections in routed or bridged configurations between the QX and other devices and remote access facilities. **OpenVPN** supports bidirectional authentication based on certificates, meaning that the client must authenticate the server certificate and the server must authenticate the client certificate before mutual trust is established.

Both server and client will authenticate each other first by verifying if the presented certificate was signed by the certificate authority (CA), then by checking the information in the now-authenticated certificate header, such as the certificate common name or certificate type (client or server).

For information on how to configure and use **OpenVPN** service, refer to the OpenVPN Service on QX IP PBXs and Auto Configuration of Epygi Supported IP Phones using OpenVPN guides.

11 Status Menu

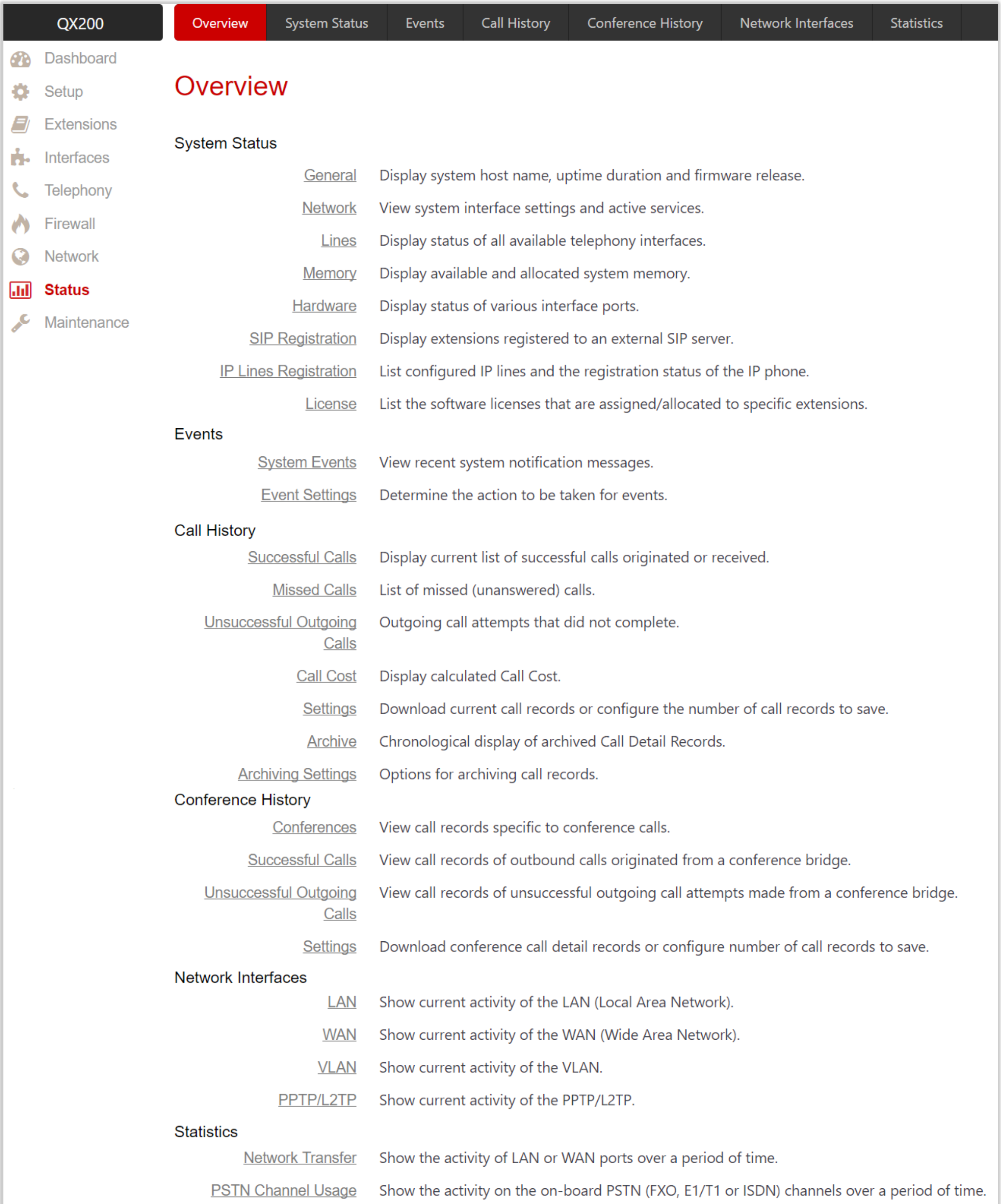


Figure 142: Staus Menu overview

11.1 System Status

11.1.1 General

The **General Information** page provides the following information:

- **Uptime Duration** shows the time period the QX is running since last reboot.
- **Device Hostname** shows the device hostname of QX.
- **Firmware Version** shows the version of the QX firmware and the file system.
- **Language Pack** – this information is presented only when a custom language pack is uploaded and indicates the version of language pack.

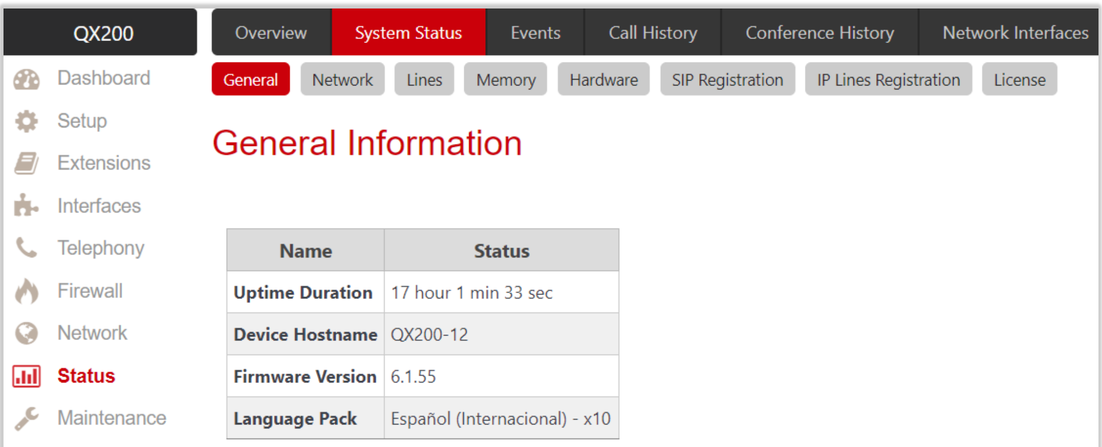


Figure 143: Status – General Information page

11.1.2 Network

The **Network Status** page provides information on available network interfaces and services on the QX.

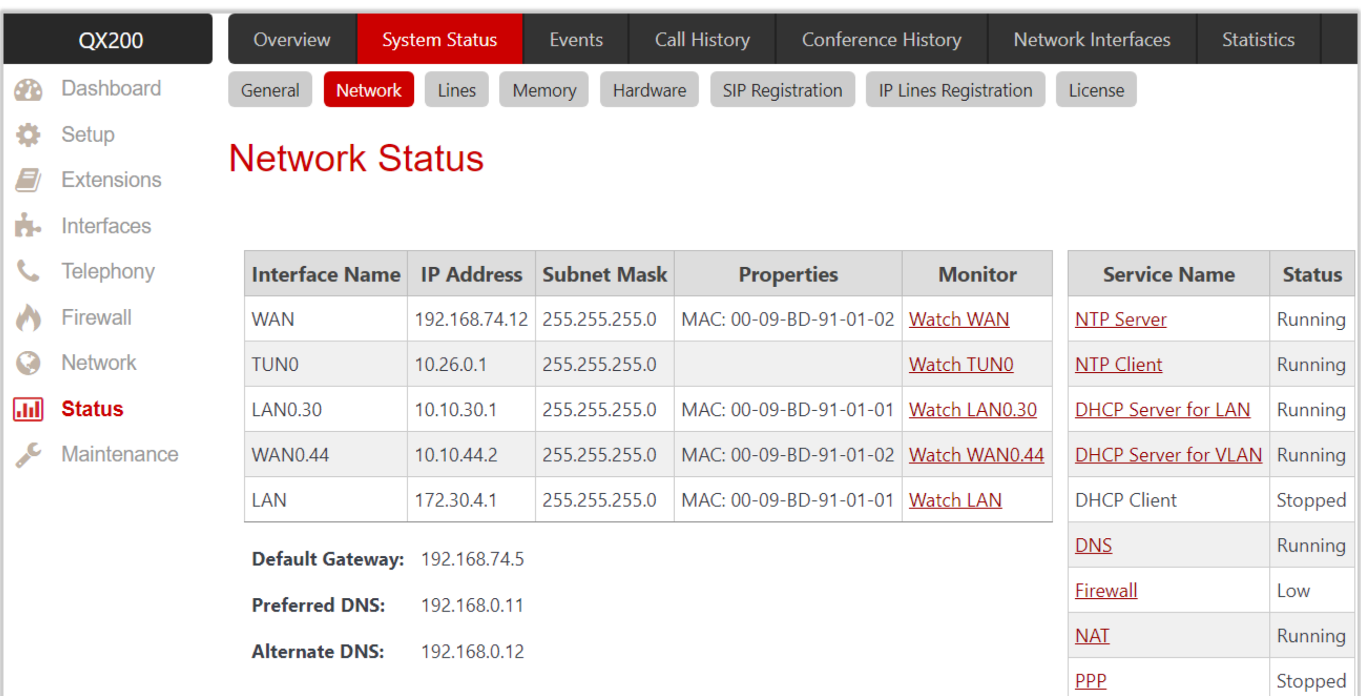
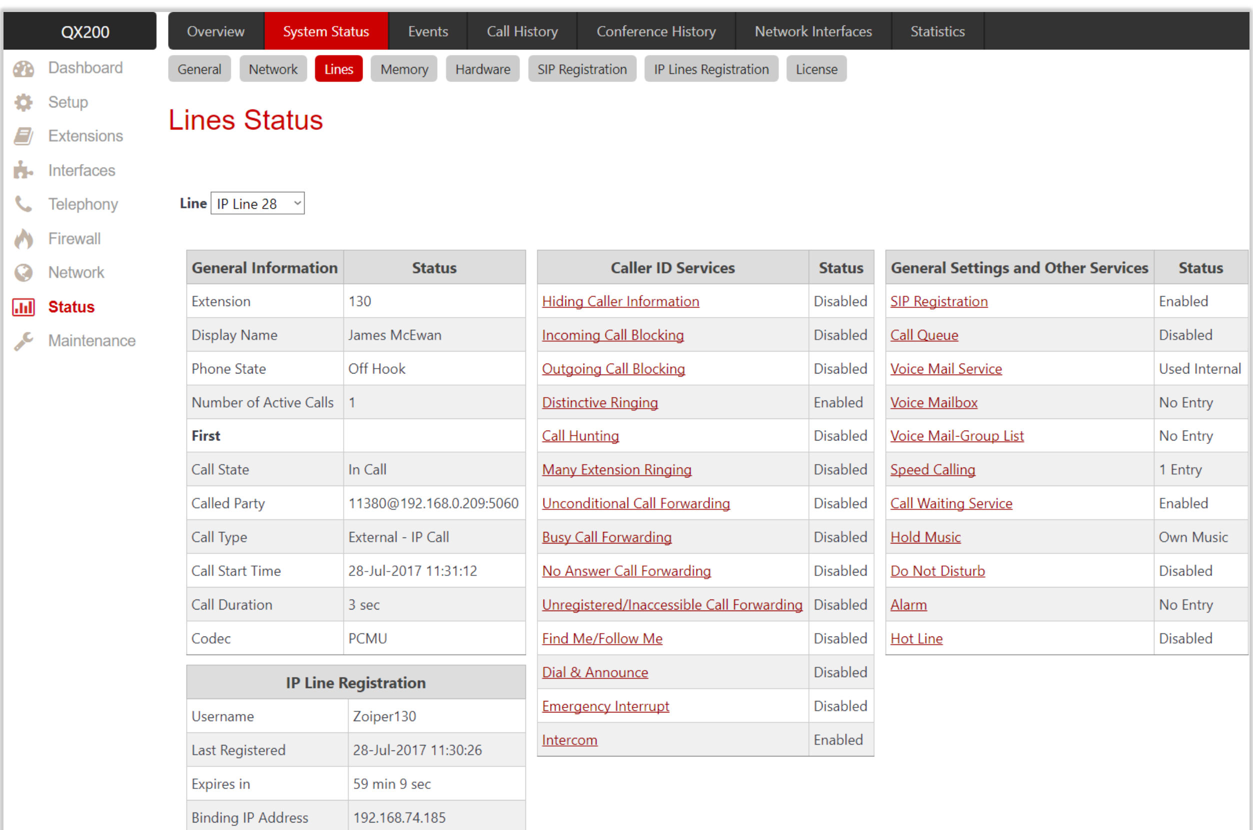


Figure 144: Status – Network Status page

11.1.3 Lines

The **Lines Status** page displays the current status and general information for the selected **Line** or **Trunk**.

- **FXS Line** contains the following tables:
  - **General Information** shows the attached extension number, display name, the phone state and the number of active calls.
  - **Caller ID Services** shows the status for the **Caller ID Services** (enabled or disabled) on the attached extension.
  - **General Settings and Other Services** shows the settings and services configured on the attached extension.
- **IP Line** contains the following tables:
  - **General Information** shows the number of attached extension, display name, the phone state and the number of active calls.
  - **IP Line Registration** shows the IP line registration status.
  - **Caller ID Services** –shows the status for the **Caller ID Services** (enabled or disabled) on the attached extension.
  - **General Settings and Other Services** shows the settings and services configured on the attached extension.
- **FXO Line** shows the **Allowed Call Type**, the destination for **Incoming Calls** (user extension, auto attendant or Call Routing Table) and the **state** of the line (Free or Busy).
- **ISDN Trunk** shows the status of **B1** and **B2 channels** and the **state of the trunk** (Free or Busy). The table includes a group of static and dynamic parameters. The static parameters are always displayed. The dynamic parameters appear only whenever an event takes place on the channel.



General Information	Status
Extension	130
Display Name	James McEwan
Phone State	Off Hook
Number of Active Calls	1
First	
Call State	In Call
Called Party	11380@192.168.0.209:5060
Call Type	External - IP Call
Call Start Time	28-Jul-2017 11:31:12
Call Duration	3 sec
Codec	PCMU

IP Line Registration	
Username	Zoiper130
Last Registered	28-Jul-2017 11:30:26
Expires in	59 min 9 sec
Binding IP Address	192.168.74.185

Caller ID Services	Status
Hiding Caller Information	Disabled
Incoming Call Blocking	Disabled
Outgoing Call Blocking	Disabled
Distinctive Ringing	Enabled
Call Hunting	Disabled
Many Extension Ringing	Disabled
Unconditional Call Forwarding	Disabled
Busy Call Forwarding	Disabled
No Answer Call Forwarding	Disabled
Unregistered/Inaccessible Call Forwarding	Disabled
Find Me/Follow Me	Disabled
Dial & Announce	Disabled
Emergency Interrupt	Disabled
Intercom	Enabled

General Settings and Other Services	Status
SIP Registration	Enabled
Call Queue	Disabled
Voice Mail Service	Used Internal
Voice Mailbox	No Entry
Voice Mail-Group List	No Entry
Speed Calling	1 Entry
Call Waiting Service	Enabled
Hold Music	Own Music
Do Not Disturb	Disabled
Alarm	No Entry
Hot Line	Disabled

Figure 145: Status – Lines Status page

11.1.4 Memory

The **Memory Status** pages show information on available memory size and memory allocation among the applications and services on the QX. The **Memory Size** is expressed in time units calculated using a specific codec. The following sub-pages are available:

- **General Information** shows the memory size and current memory allocation (usage) between the system messages, voice mails, recorded calls and recorded conferences. The **Databases** table shows the memory size used by different QX services.
- **User Extension** shows the memory size available and currently allocated (used) to voice mails and recorded/uploaded system voice messages for each **user extension. Universal Extension Recordings** shows the space used to define the system default voice messages common for all user extensions.
- **Attendant** shows the memory size available and currently allocated (used) to recorded/uploaded system voice messages for each **auto attendant.**
- **Recorded Box** shows the memory size available and currently allocated (used) to recorded calls, recorded/uploaded system messages for each specific **Recording Box**. Only **G711 codec** is used to record calls.
- **ACD Queue** shows the memory size available and allocated (used) to recorded/uploaded system messages for each specific **ACD Queue.**
- **Conference** shows the memory size available and allocated (used) to currently recorded conferences and recorded/uploaded system voice messages for each **Conference.**

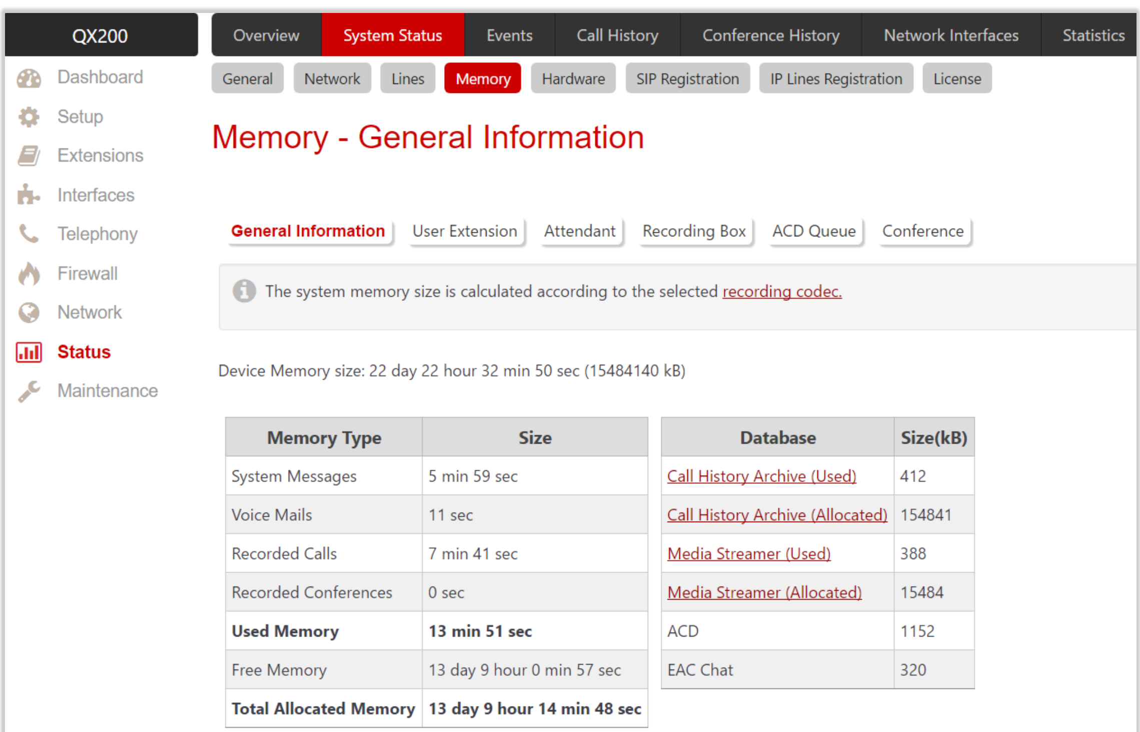


Figure 146: Status – Memory Status page

For more information on **Memory Status**, refer to the Memory Management on QX IP PBXs guide.

11.1.5 Hardware

The **Hardware Status** table shows the list of network interfaces, on-board and external devices and parts currently available on the QX with their parameters and statuses.

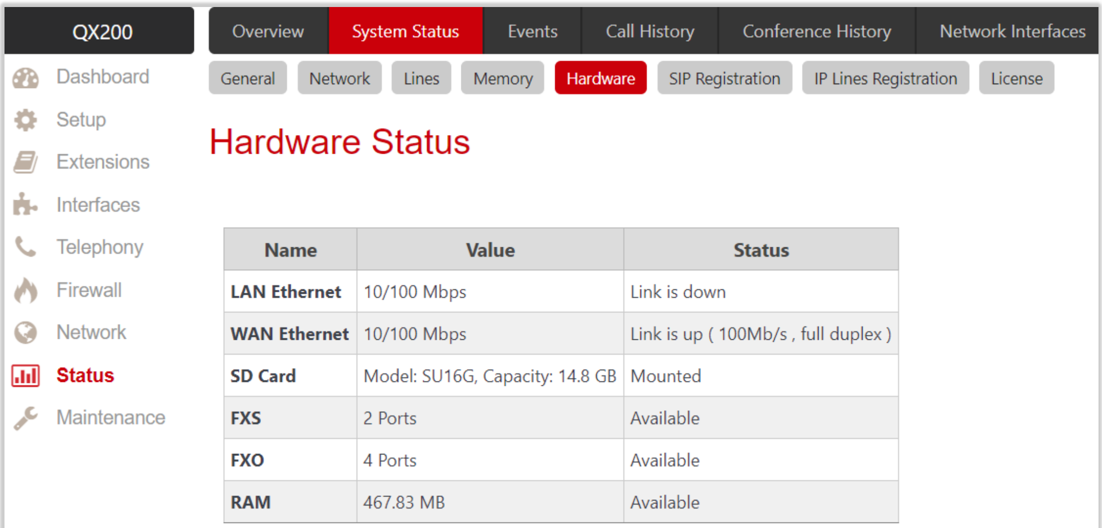


Figure 147: Status – Hardware Status page

11.1.6 SIP Registration

The **SIP Registration Status** page displays information about the registration of QX extensions on SIP server(s). Information about the configured **SIP Tunnels** between Epygi devices is displayed here as well.

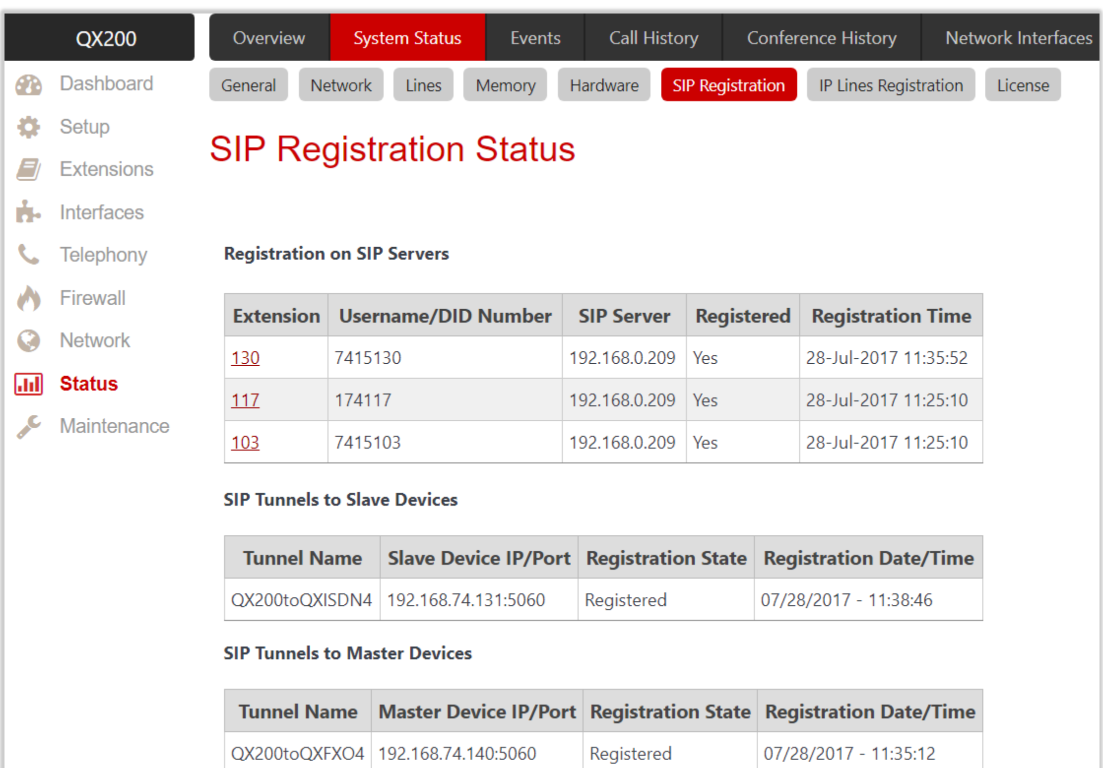


Figure 148: Status – SIP Registration Status page

11.1.7 IP Lines Registration

The **IP Lines Registration Status** page provides information on IP lines registration and subscriptions used on the QX. The **IP Lines Registration table** lists the IP lines and remote extensions registered on the QX. The **Subscriptions Count** table shows the used and maximum allowed subscriptions on the QX. The subscriptions are events originated by QX services or IP phones. The following information is available:

- **Dialog (BLF) event** – IP phone's **Busy Lamp Field** (BLF) subscriptions, used for watching the extensions, as well as showing the states for other telephony services on the phone.
- **Message Waiting Indication (MWI) event** – IP phone's MWI subscriptions, used for voice mailbox status indication on the phone.
- **Presence event** and **Other events** are used by the QX internal services.

Note:

- When the allowed number of subscriptions is reached, new subscriptions are no longer possible. In order to avoid losing subscriptions, make sure the number of subscription is kept reasonably below the maximum allowed number.
- The number of **Maximum allowed subscriptions** can be changed from the **generalconfig.cgi** hidden page. Reboot the QX to apply changes.

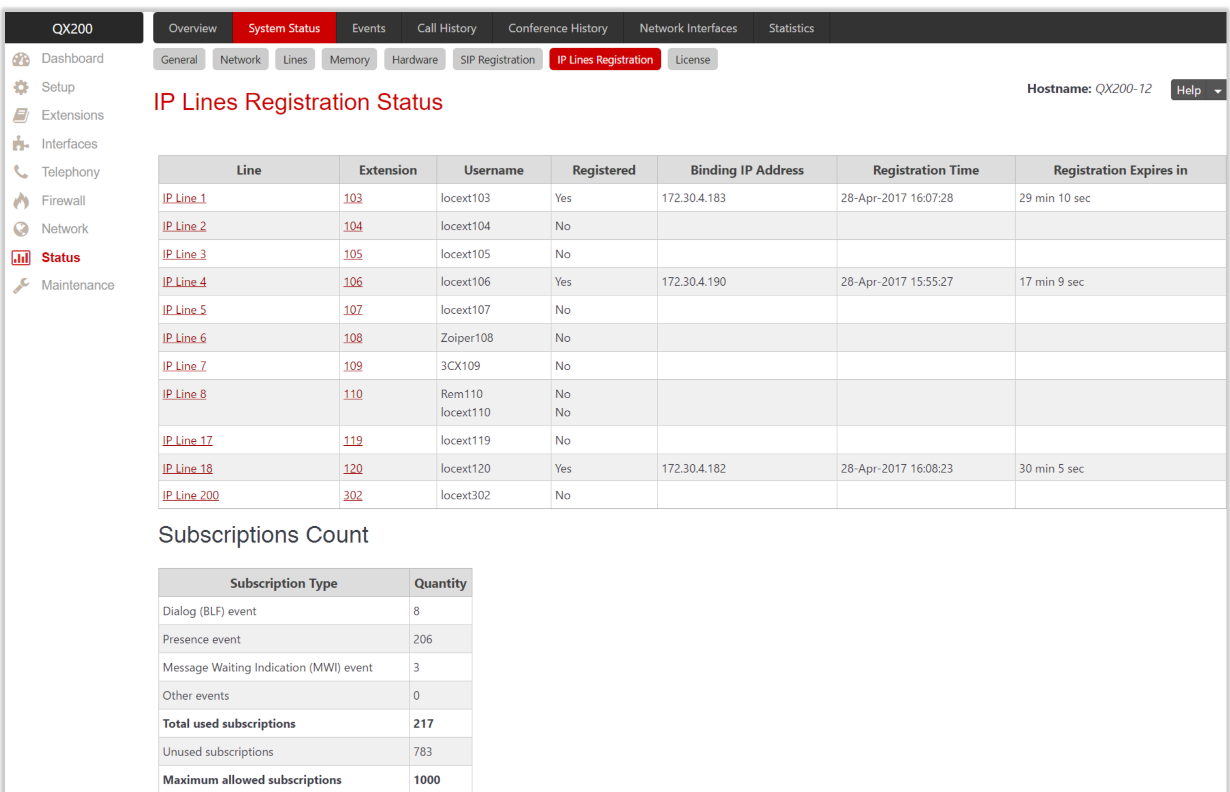


Figure 149: Status – IP Lines Registration Status page

11.1.8 License

The **License Status** page provides information about the following licensable features on the QX.

- DCC Basic
- DCC Pro
- iQall Mobile Toggling

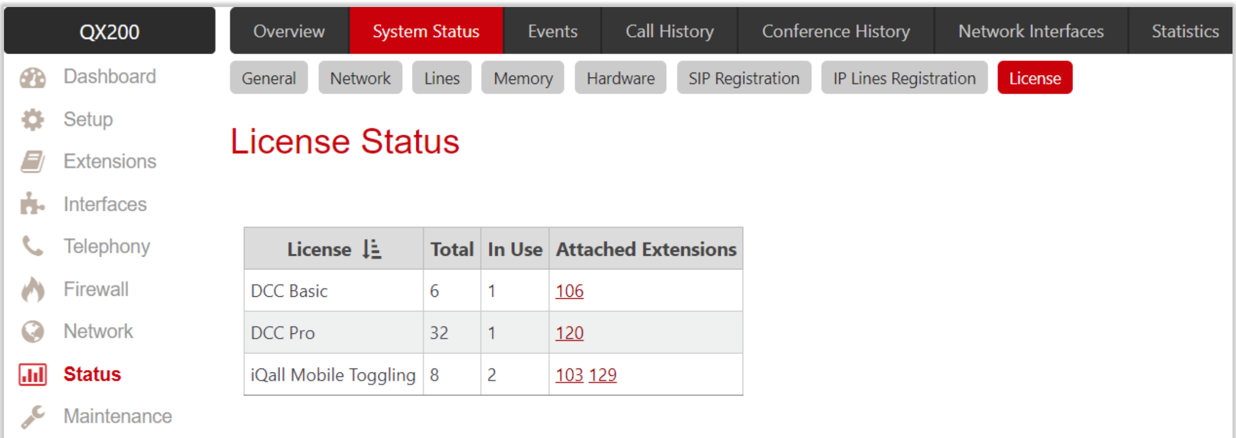


Figure 150: Status – License Status page

11.2 Events

11.2.1 System Events

The **System Events** page lists information about system events that have occurred on the QX. When a new event takes place, a record is added to the **System Event** table.

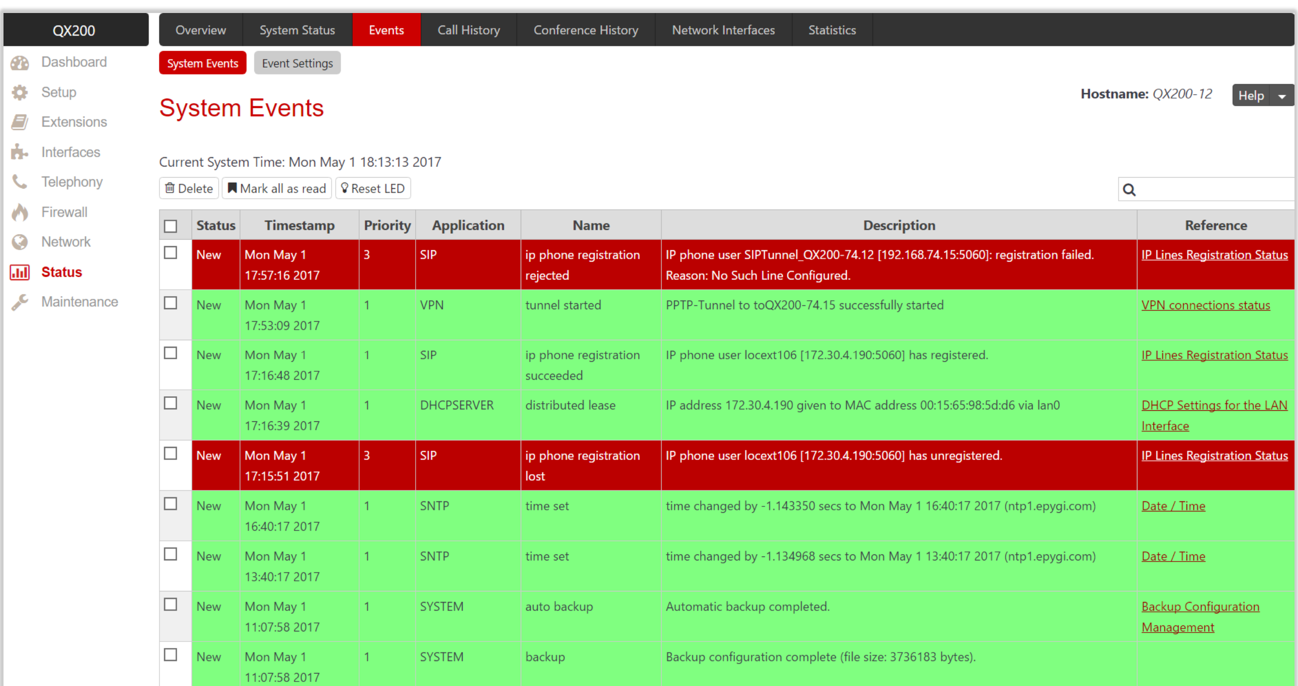


Figure 151: System Events list

The **System Events** table is the list of new and read system events. Events are marked by different colors depending on the nature of the event: **success** (priority 1, color green), **low importance failure** (priority 2, color yellow), **critical failure** (priority 3, color red). This table shows the **status** of the event (new or read) as well as the name of the application the event refers to, event description, and the date when the event occurred. TIP: Once the administrator marks all new events as "**read**", the **Pending Events** link will disappear from **Top Menu** bar.

11.2.2 Event Settings

The **Event Settings** page lists all available events on QX and allows to notify admins/users in case of any event.

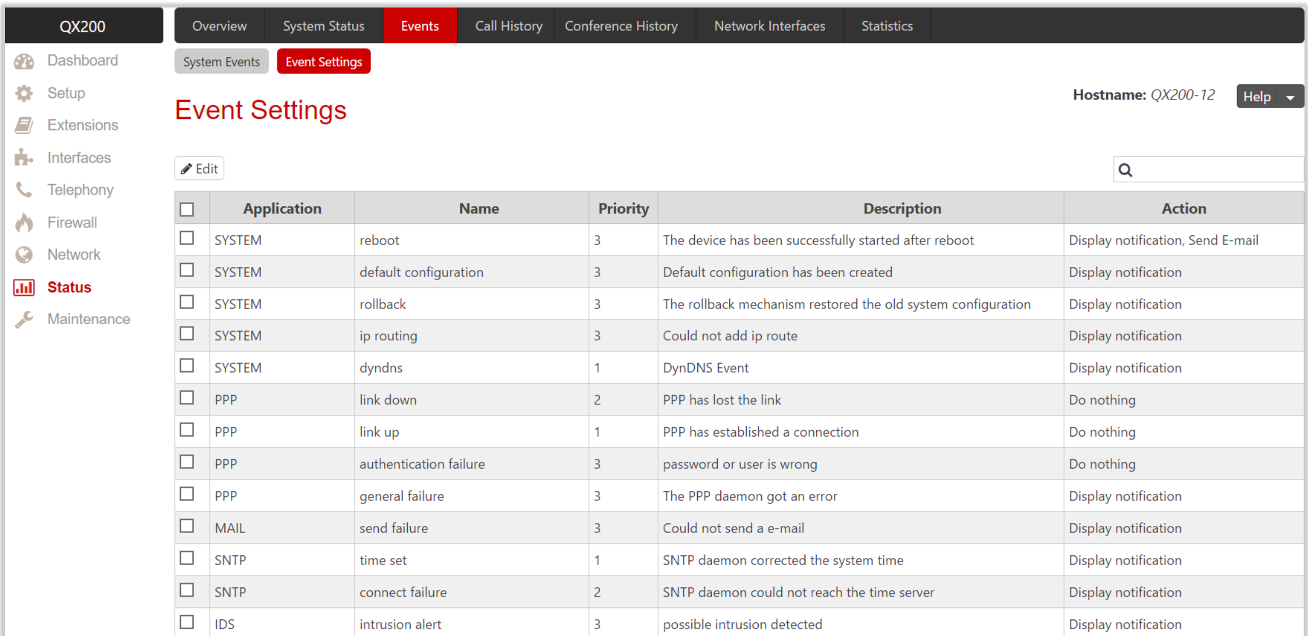


Figure 152: Event Settings page

By default, the notification will be displayed in the **System Events** page. You can modify and select other notification methods (actions) as well.

To change the **Notification** option for the event:

1. Tick the **checkbox** next to the event and click **Edit**. Multiple selection is supported.
2. Tick the **checkbox** next to the available **Action**. The following actions are available:
   - **Display Notification** displays notification in the **System Events** page.
   - **Flash LED** – LED flashes every second. For some events, the LED will start flashing after a delay.
   - **Send Mail** –e-mail will be sent to the e-mail address(es) specified in the E-mail Settings page.
   - **Send SNMP Trap** – the trap will be sent to the traphost(s) listed in the SNMP Trap Receiver Settings table.
   - **Send SMS** –SMS will be sent to the mobile number specified in the SMS Settings page.
   - **Rest Request** – the notification will be sent to the **Monitoring** server(s) specified in the **https://xxx.xxx.xxx.xxx/ecmon** hidden page.
3. Click **Save** to apply changes.

Note:

- Actions that are not allowed for the selected event (e.g. mail notification if the PPP link is down or the mail server is configured improperly) are hidden. When editing multiple events, Actions that are not appropriate for one of the selected events will be hidden.
- In case of the IDS intrusion alert, only the first intrusion in each 10-minute period will raise an event.
- If QX cannot receive an IP address (from the DHCP or PPP servers) or cannot register an extension on the SIP server or cannot reach an NTP server, it raises only one event for the entire period the action has failed but will continue to try.

11.3 Call History

Call History allows to track and report the Call Detail Records (CDRs) for calls originated and terminated on QX, as well as for calls passed through QX.

11.3.1 Successful Calls, Missed Calls and Unsuccessful Outgoing Calls

The Successful Calls, Missed Calls and Unsuccessful Outgoing Calls pages list successful, missed and unsuccessful outgoing calls and their parameters. The following components are available:

- Filter allows to search for call records based on at least one of the following criteria: Call Start Time, Call Duration, Call Cost, Caller and Called parties.
- Clear Filter is used to remove the filter.
- The Download and Download in CSV format buttons are used to download the displayed CDRs for each page (Successful, Missed and Unsuccessful Outgoing) in (*.log) and (*.csv) formats respectively.

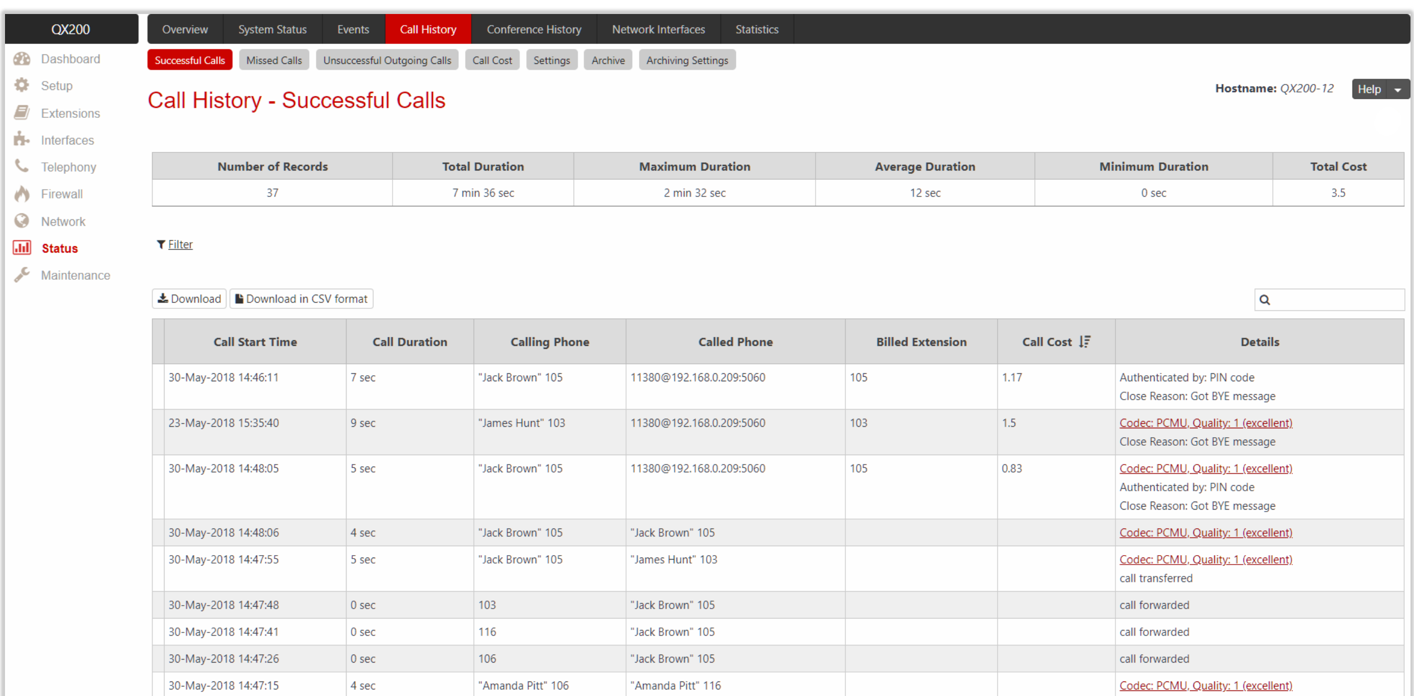


Figure 153: Call History – Successful Calls page

CDRs listed in the Call History – Successful Calls table are characterized by the following parameters:

- Call Start Time
- Call Duration

- Calling Phone
- Called Phone
- **Billed Extension** shows the extension which is charged for the call (if available).
- **Call Cost** shows the calculated call cost (if available).
- **Details** provides the following additional information:
  - **Details** shows information on the call quality, audio codec and the call close reason. The call close reason appears to provide more information about the call termination, such as a network problem, call termination by one of the parties, **Voice Mail Service** activation, etc. The **Codec** link leads to the RTP Statistics page where the **RTP parameters** of the call are shown.
  - **Authenticated By** shows the authentication parameters (e.g. **login** or **PIN code**) used to pass the authentication when making a call.
  - Information about **FAX statistics** for the calls that have a FAX transmission handled. The **FAX** link leads to the FAX Statistics page where the **FAX parameters** of the call are shown.

11.3.2 Call Cost

The **Call Cost** page shows the summarized information regarding the chargeable calls. The following components are available:

- **Filter** allows to search for call records based on at least one of the following criteria: **Timeframe**, **Duration**, **Call Cost** and **Billed Extension**.
- **Clear Filter** is used to remove the filter.
- The **Download** and **Download in CSV** format buttons are used to download the displayed CDRs in the (*.log) and (*.csv) formats respectively.

The **Call Cost** table is characterized by the following parameters:

- **Billed Extension** shows the extension which is charged for the call.
- **Duration** shows the total duration of all chargeable calls for the extension.
- **Cost** shows the total cost of all chargeable calls for the extension.

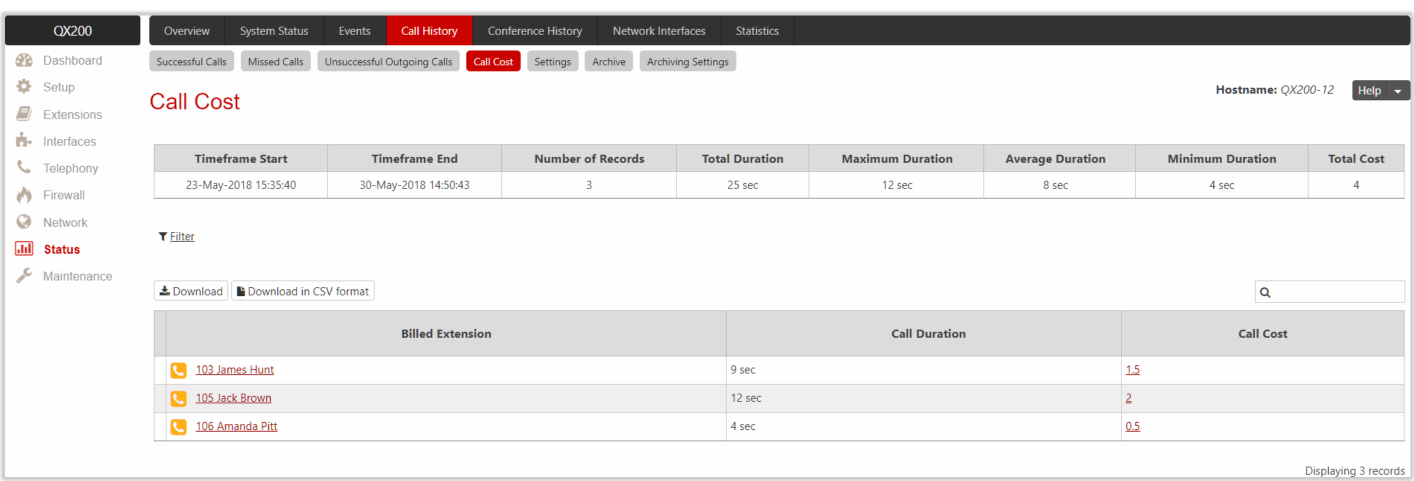


Figure 154: Call Cost page

11.3.3 Settings

The **Call History – Settings** page is used to configure specific parameters for displaying **Call History**.
The following settings (options) are available:

- **Enable Call Reporting** is used to activate service and allows to select the maximum number of CDR entries to be displayed in the **Call History** tables respectively.
  - **Maximum Number of Successful/Missed/Unsuccessful Call Records** is used to select the maximum number of CDR entries to be displayed in the **Call History** tables. When the number of CDRs exceeds the defined numbers, the oldest entries will be automatically deleted. To keep the **Call History** safe, configure and use the Archiving Settings service.
- **CDR Parameters** section provides the full list for CDR parameters on QX. You can select the specific parameters to be excluded from the downloaded/archived CDR files to make them more compact, thus more readable. For detailed information about **CDR parameters**, refer to the Call Detail Records on QX IP PBXs guide.

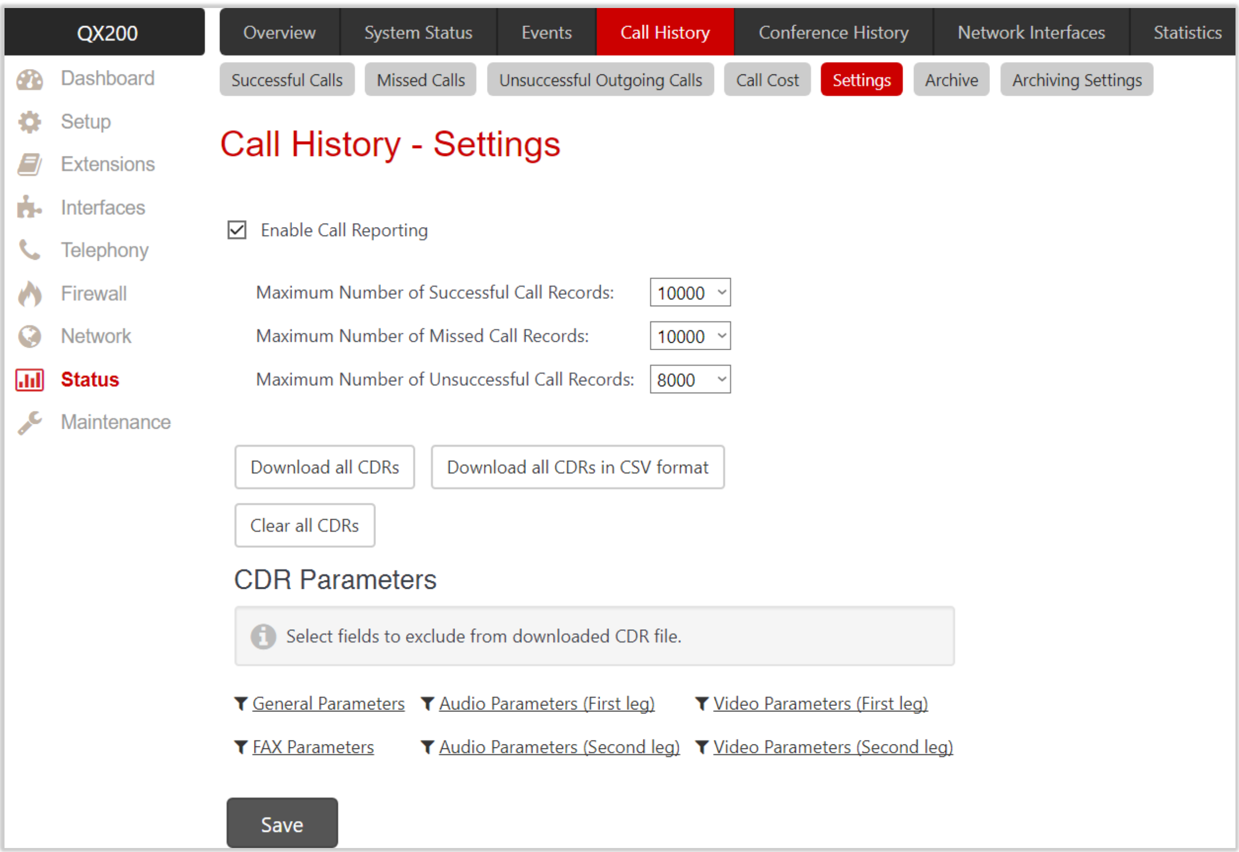


Figure 155: Call History – Settings page

11.3.4 Archive

The **Call History – Archive** page shows the archived CDR files and allows the user to download them either in (*.log) and (*.csv) format.

The following functions are available on this page:

- **Filter** allows to search for specific archived CDR records in the **Archive** table by the record's full name or some part of the name.
- **Delete** is used to remove the selected record(s) from the **Archive.**
- **Clear all Records** is used to remove all archived files.

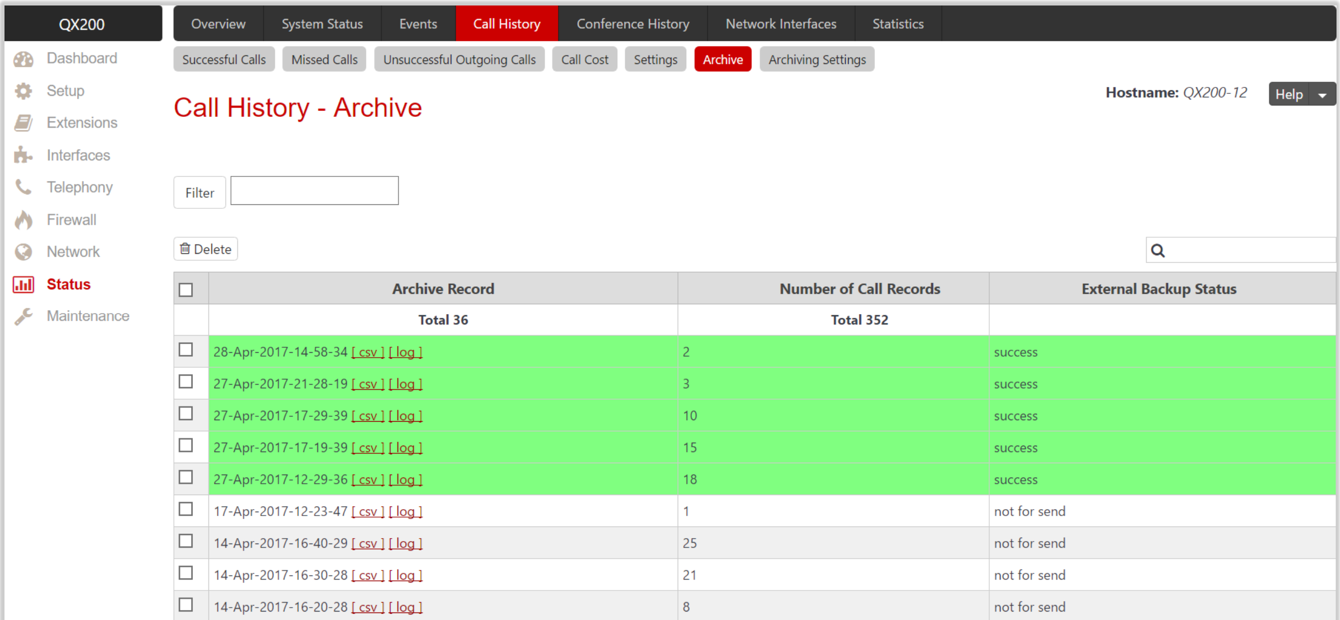


Figure 156: Call History – Archive page

CDRs listed in the Call History – Archive table are characterized by the following specifications:

- Archive Records shows the archived record (file) name which is actually the archiving date and time. Click the hyperlinked [csv] or [log] to download the archived file.
- Number of Call Records shows the number of call records in the archived file.
- External Backup Status shows the status of the archived file backup. The following statuses are available:
  - Success shows that the archived file has been successfully backed up.
  - Failed shows that the archived file failed to be backed up. The Try to send now link will appear next to this status allowing to repeat the backup process.

11.3.5 Archiving Settings

The Call History Archiving service is used to configure the automatic archiving of Call History. The following settings (options) are available:

- Percentage of Total Memory allocated for Archive is used to allocate memory for archiving.
- Enable Call History Archiving is used to activate service on QX.
- File Format is used to select the format of archived file as (*.log) and (*csv).

Archiving Mode

This section is used to select the archiving mode. The following modes are available:

- Archive by Record Count – if selected, the file will be archived as soon as the number of records specified in the drop-down list is collected.
- Archive by Time Interval – if selected, the file will be archived as soon as the timeframe specified in the drop-down list is elapsed from the last archiving. If no CDRs were produced during that timeframe, archived file for that period will not be generated.

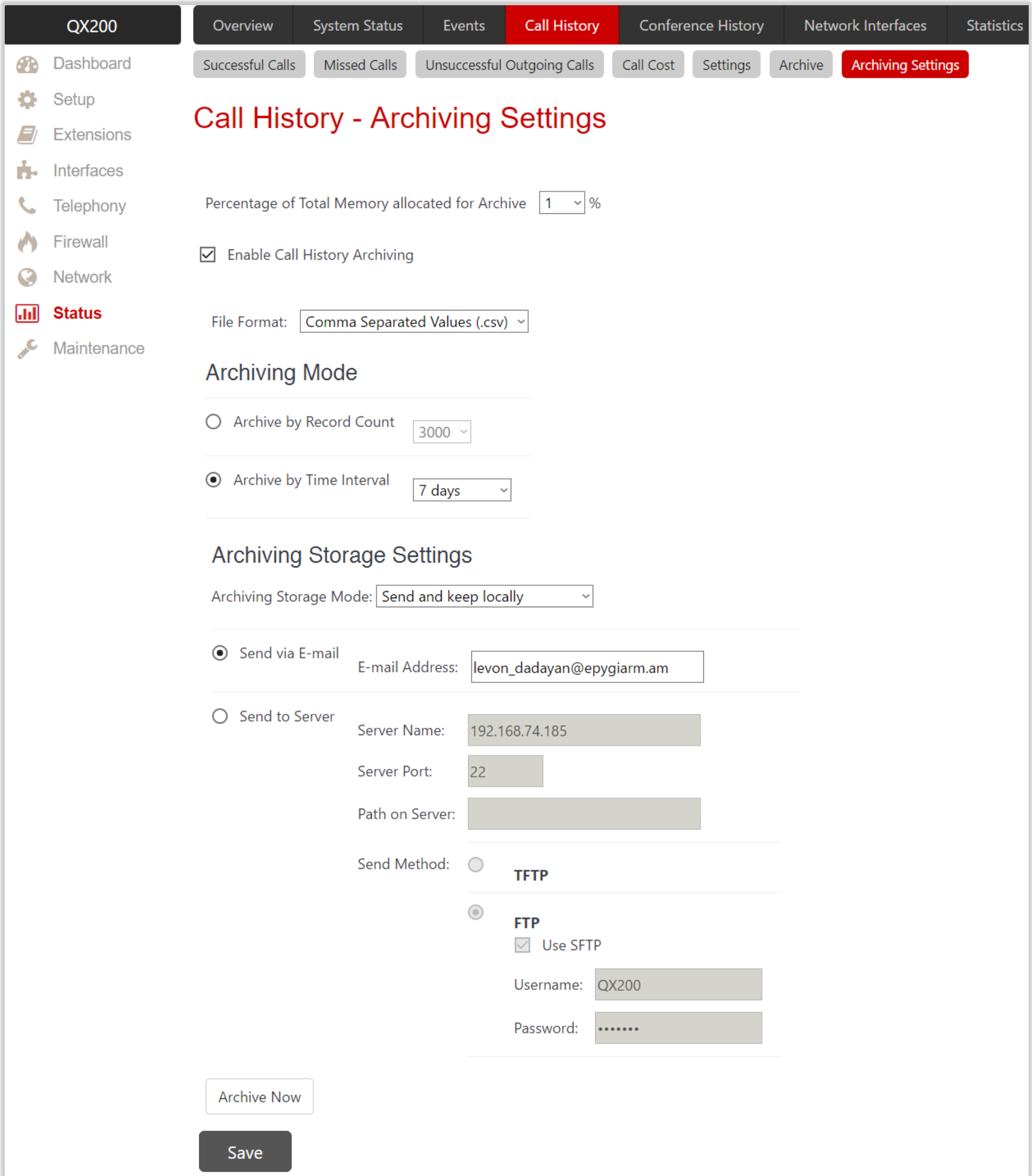


Figure 157: Call History – Archiving Settings page

Archiving Storage Settings

This section is used to select archiving storage and configure the backup settings.

- **Archiving Storage Mode** is used to select one of the following archiving modes:
  - **Do not send** – if selected, the CDRs will be archived and kept locally only.
  - **Send and keep locally** – if selected, the CDRs will be sent to the server and kept locally.
  - **Send and delete from archive** – if selected, the CDRs will be sent to the server and removed from the archive.
- The following options are available for storing archived CDRs:
  - **Send via E-mail** allows to send the archived files via e-mail. The destination e-mail address has to be entered in the **E-mail Address** field.
  - **Send to Server** allows to send the archived files to the external server. This selection enables the following fields to be filled:

- Server Name is used to set the IP address or hostname of the server.
- Server Port is used to set the port of the server.
- Path on Server is used to enter the path on the server.
- Send Method – the server type: TFTP or FTP. Specify the Username and Password in case of the FTP. If these fields are left blank, anonymous authentication will be used. TIP: Select the Use SFTP option to enable SFTP support.

- Archive Now is used to archive CDRs immediately.

11.3.6 RTP Statistics

The RTP Statistics page provides detailed information about the established call. When QX serves as an RTP proxy, this page displays two groups (legs) of RTP statistics. Normally, one leg describes the RTP stream from caller to the QX and the other leg describes the RTP stream from QX to the destination. The following parameters are available:

- Quality indicates the call quality, which depends on RTP statistics. Below is the legend for Call Quality:
  - excellent – RX Lost Packets < 1% and RX Jitter < 20
  - good – RX Lost Packets < 5% and RX Jitter < 80
  - satisfactory – RX Lost Packets < 10% and RX Jitter < 150
  - bad – RX Lost Packets < 20% and RX Jitter < 200
  - very bad – RX Lost Packets > 20% or RX Jitter > 200
- Local and Remote indicate the two peers the RTP stream is transmitted in between. The table below describes the characteristics of RTP stream between these peers.
  - Rx/Tx Codec is the codec for received and transmitted RTP stream respectively.
  - Rx/Tx Packets is the number of RTP packets received and transmitted respectively.
  - Rx/Tx Packet Size is the size of RTP packets (payload) received and transmitted respectively.
  - Rx Lost Packets is the number of lost RTP packets for received stream.
  - Rx Jitter – is an estimate of the statistical variance of the RTP data packet inter-arrival time, measured in timestamp units.

The inter-arrival jitter is defined to be the mean deviation (smoothed absolute value) of the difference D in packet spacing at the receiver compared to the sender for a pair of packets. If Si is the RTP timestamp from packet i, and Ri is the time of arrival in RTP timestamp units for packet i, then for two packets i and j, D may be expressed as:

D(i,j) = (Rj - Ri) - (Sj - Si) = (Rj - Sj) - (Ri - Si) J(i) = J(i-1) + (|D(i-1,i)| - J(i-1))/16, where J(i) is Rx Jitter for packet i.

For more details about Jitter calculations, refer to the RFC1889.

- Rx Maximum Delay is the maximum variance (absolute value) of actual arrival time of the RTP data packet compared to estimated arrival time, measured in milliseconds. If Si is the RTP timestamp from packet i, and Ri is the time of arrival in RTP timestamp units for packet i, then variance for packet i may be expressed as following:

V(i) = |(Ri - R1) - (Si - S1)| = |(Ri - Si) - (R1 - S1)| Rx Maximum Delay = max V(i) / 8

- RX Delay Increase Count indicates the number of times the delay in jitter buffer is increased during the call.
- RX Delay Decrease Count indicates the number of times the delay in jitter buffer is decreased during the call.
- Configure Call Quality Event Notification leads to the Call Quality Notification page to configure call quality control notifications.

- **Configure System Events** leads to the **Event Settings** page to configure the methods of notification for each system event.

RTP Statistics is logged only when at least one of the call endpoints is located on the QX. For example, it will not be logged when:

- Calls from or addressed to the IP lines or remote extension.
- Calls from an external user are routed to another external user through call routing rules.

In the first case, RTP statistics will be logged if remote extension or IP line user is calling locally to the user extension or auto attendant.

11.3.7 FAX Statistics

The **FAX statistics** page provides information on received and transmitted packets, lost, bad and duplicated packets. These statistics only refer to **T.38 FAX** transmission. FAX statistics are not available for the FAX transmitted with other protocols.

11.4 Conference History

Conference History allows to track and report the details of conference calls that have been activated on QX.

For more information on **Audio-Video conferencing**, refer to the Audio-Video Conferencing on QX IP PBXs guide.

11.5 Network Interfaces

The **Interface Statistics** pages display statistics (e.g. the number of received and transmitted packets, errors, etc.) for each interface.

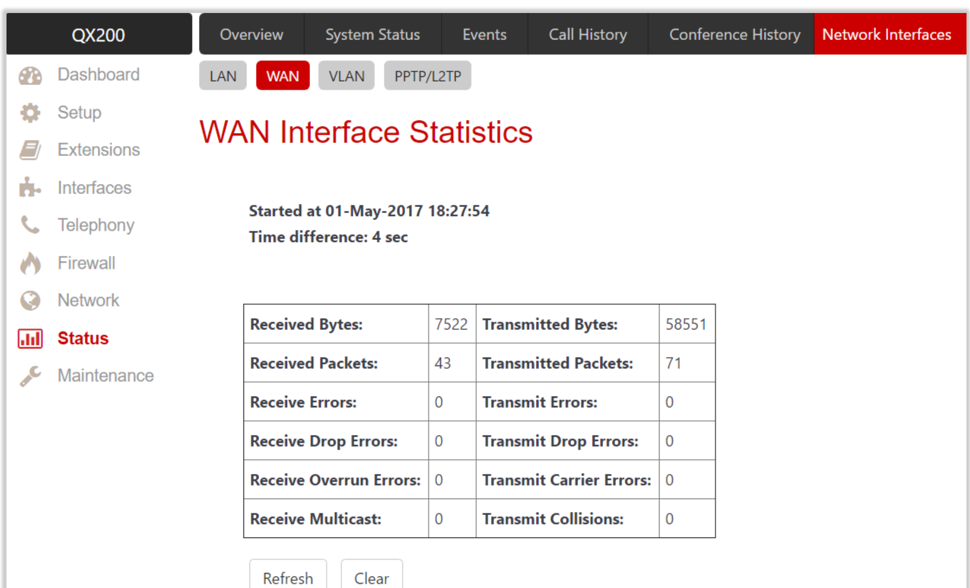


Figure 158: LAN Interface Statistics page

11.6 Statistics

11.6.1 Network Transfer

The **Transfer Statistics** page is used to generate charts with the transmit/receive values (criteria), interface type and time period. Select the desired criteria and click **Show** to generate the **Transfer Statistics** chart and the table showing the transfer statistics values (if enabled). The letters **M** (millions) and **K** (thousands) used in the legend of the displayed chart show the total number of specified criteria.

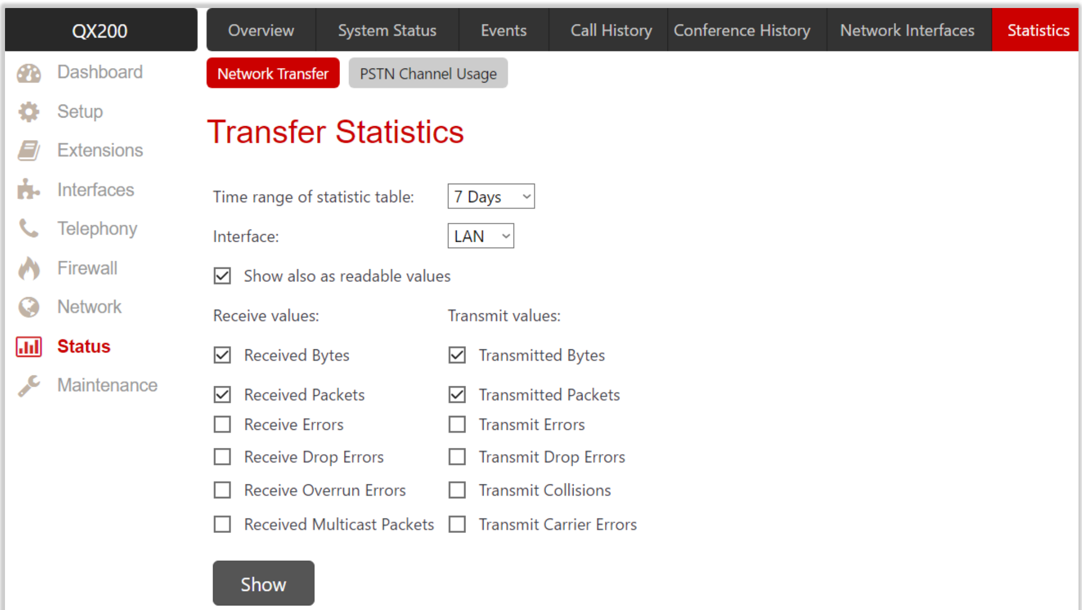


Figure 159: Transfer Statistics page

11.6.2 PSTN Channel Usage

The **PSTN Channel Usage** page is used to generate charts with the incoming/outgoing calls and maximum active calls (criteria) and time period for the selected on-board lines or trunks.

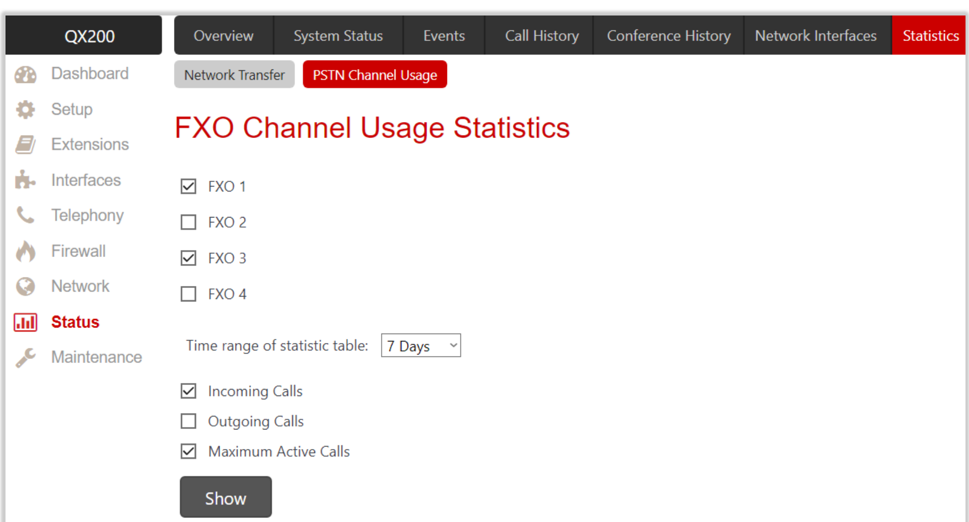


Figure 160: FXO Channel Usage Statistics page

Select the desired criteria and click **Show** to generate the **Channel Usage Statistics** chart.

12 Maintenance Menu

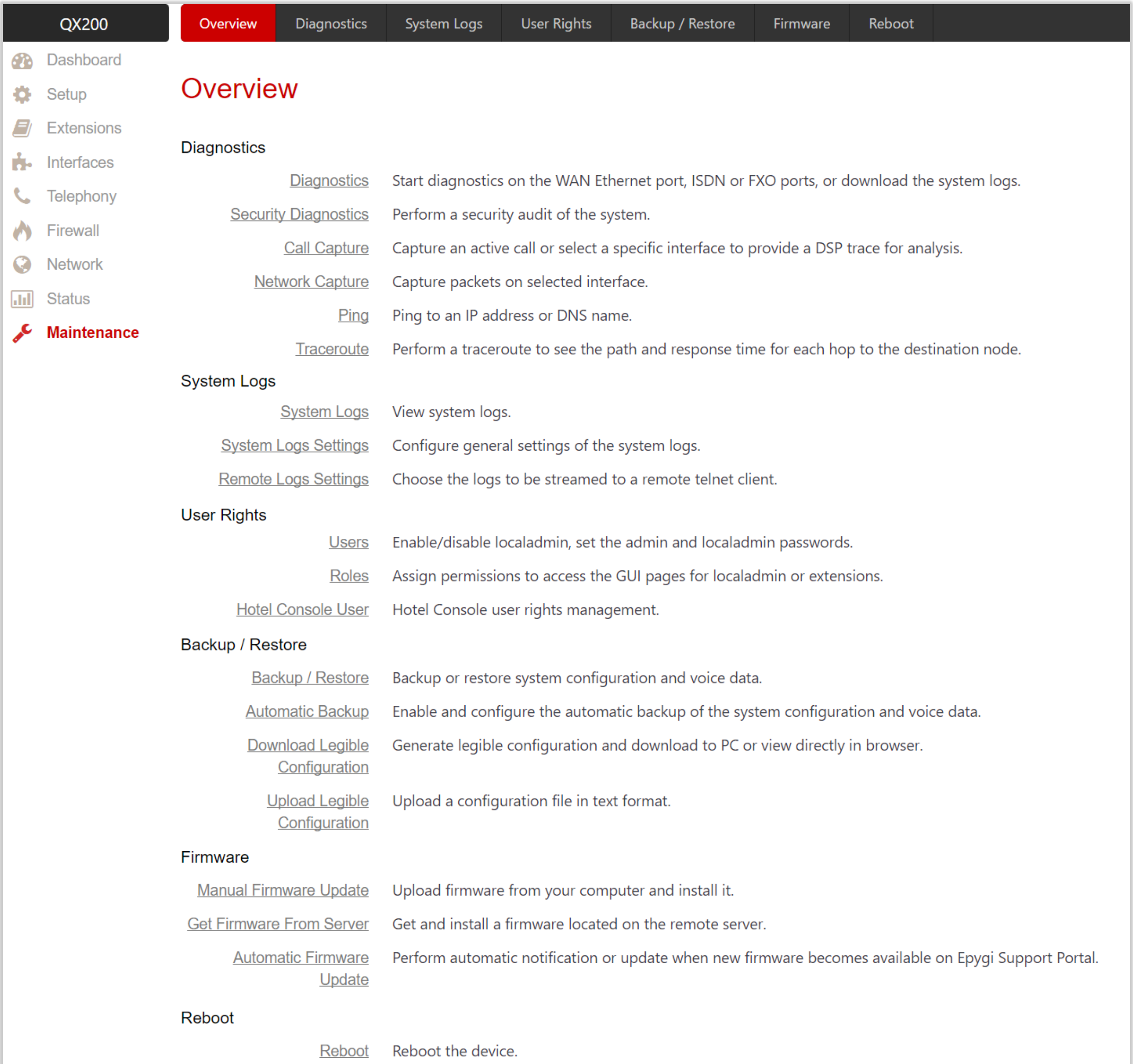


Figure 161: Maintenance Menu overview

12.1 Diagnostics

The **Diagnostics** page allows to run network and on-board lines (trunks) diagnostics to verify QX connectivity and collect system logs for diagnostic purposes.

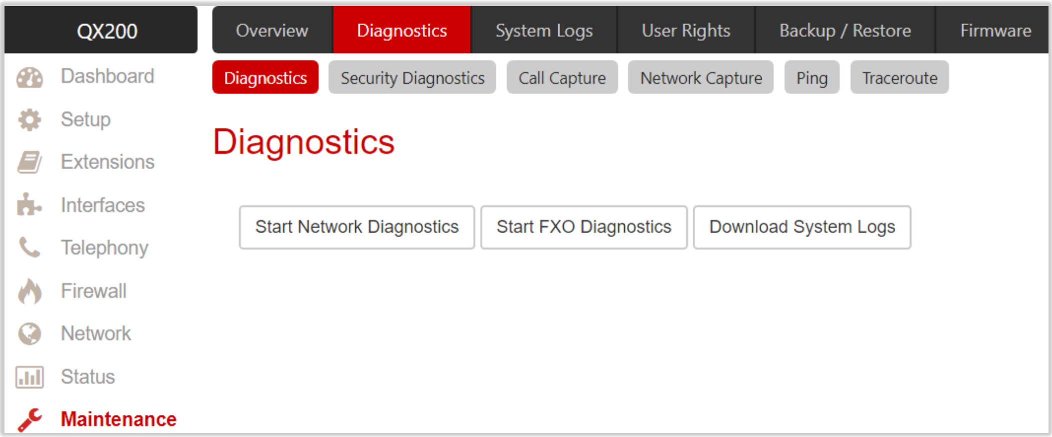


Figure 162: Diagnostics page

- **Start Network Diagnostics** is used to run network diagnostics, i.e., to check the WAN link and network parameters such as IP configuration, Default Gateway, primary and secondary DNS servers' accessibility.
- **Start FXO Diagnostics** is used to run FXO diagnostics to determine the optimal value for the **Country Specific Regional Setting** (CSRS) appropriate to your PSTN provider. Once the FXO diagnostics is complete, the recommended value should be manually set on the **http://xxx.xxx.xxx.xxx/fxocfg** hidden page. Setting this value may resolve echo or poor audio quality issues on FXO lines.
- **Start ISDN Diagnostics** is used to run ISDN diagnostics to initiate ISDN BRI low level diagnostics. With these tests, the ISDN physical link is checked and the **Frame Synchronization** is verified.
- **Download System Logs** is used to download all logs in (*.tar) file format. These logs can then be used by Epygi Technical Support to determine the issues that have occurred on QX.

12.1.1 Security Diagnostics

The **Security Diagnostics** page allows to run security audit and get security reports.

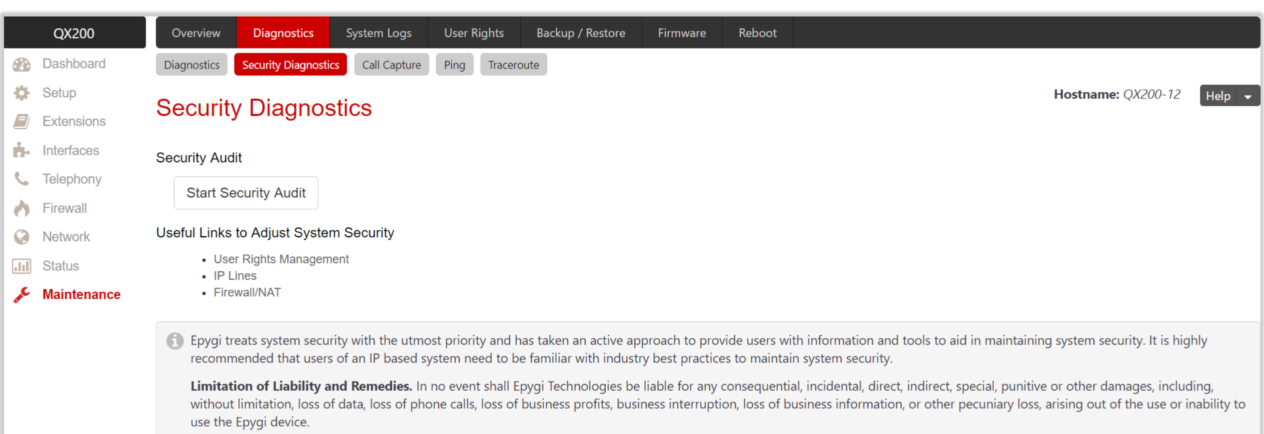


Figure 163: Security Diagnostics page

- **Start Security Audit** is used to run security audit. QX **Security Audit** is a security reporting system, which generates the warnings regarding QX weaknesses for the selected Security Level. The warnings may vary depending on the selected Security Level. Security Audit will detect configuration issues related to security in Firewall, IDS, IP line passwords, Call Routing and extension settings. Show Security Report allows to display the last security audit report.

- The following useful links are available to adjust the system security:
  - User Rights Management
  - IP Lines
  - Firewall/NAT

12.1.2 Call Capture

Call Capture is used to capture the calls to/from on-board interfaces. You can capture calls on the following interfaces (depending on the QX model): FXS, FXO or ISDN. This page consists of two sub-pages:

- Active Calls lists all active calls on the QX for a certain moment.
- Interfaces lists all available interfaces on the QX.

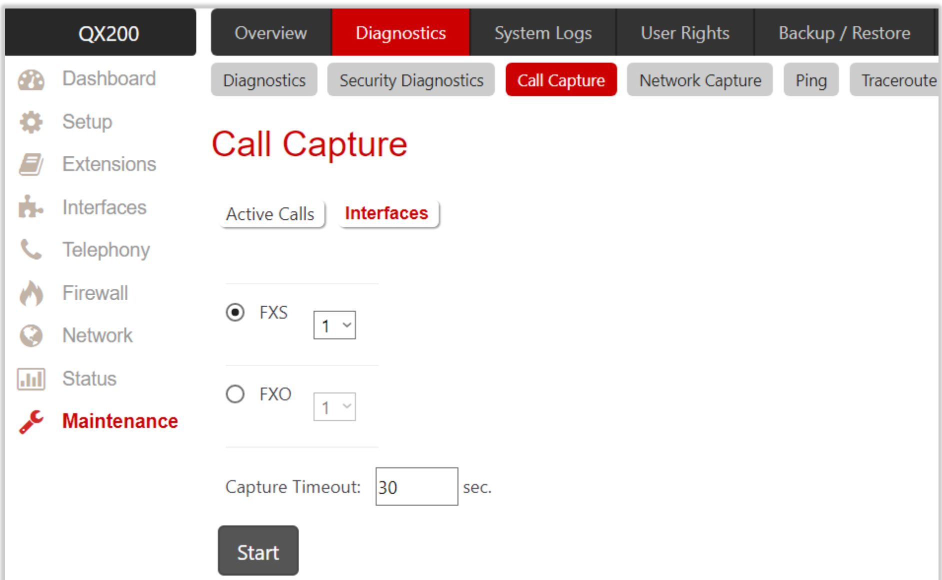


Figure 164: Call Capture - Interfaces subpage

To start Call Capture:

1. Tick the checkbox next to the call, which should be captured from Active Calls sub-page or select the available interface from Interfaces sub-page.
2. Configure the Capture Timeout, during which the call will be captured. TIP: The call capture will automatically be stopped, when the capture timeout expires.
3. Click Start to start a call capture.
4. Click Stop to stop a call capture and download the captured file.

The captured call will be downloaded in (*.tar) format. It contains two streams (receive and transmit) of the captured call. These streams can be then played with an audio player application.

Note: The Call Capture duration is limited to 160 seconds.

12.1.3 Network Capture

Network Capture is used to capture packets for the selected network interface. The following options are available:

- Capture on Interface is used to select the interface to capture packets. The Local Loopback Interface option is used to capture the traffic within the unit.
- Stop after receiving count packet is used to enter the number of packets to be captured.
- Restrict to Host is used to enter a specific IP address packets should be captured for.

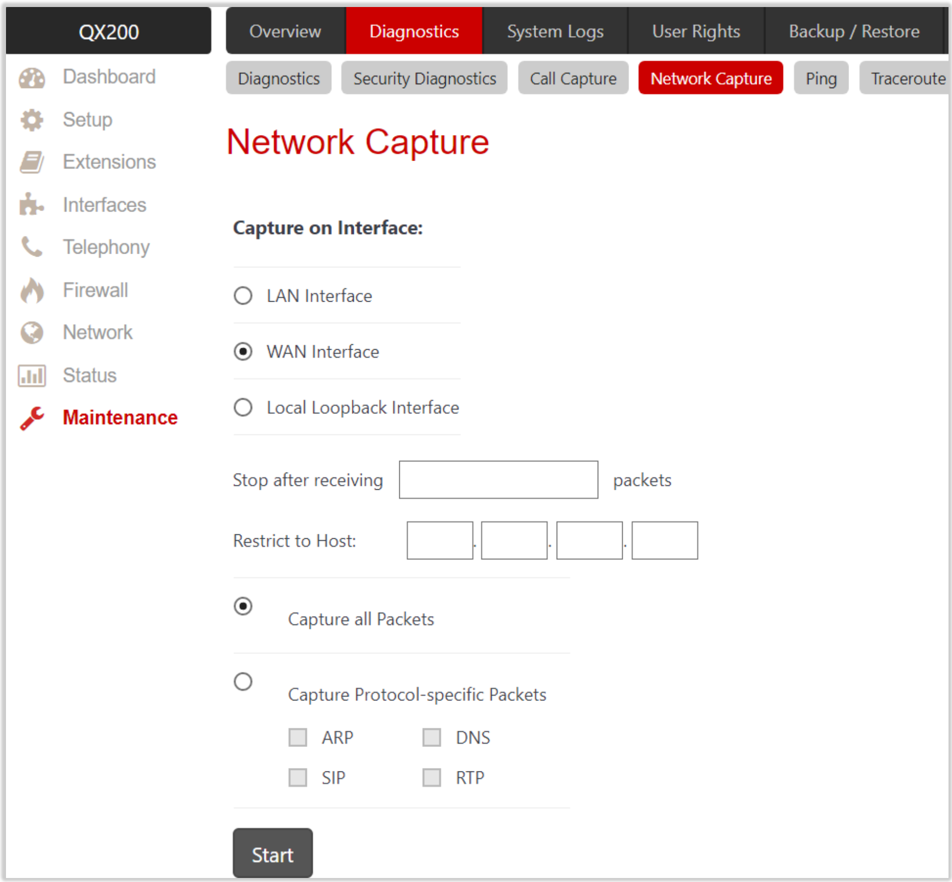


Figure 165: Network Capture page

- Capture all Packets allows capturing all packets on the selected interface.
- Capture Protocol-specific Packets is used to restrict capturing specific packets only (ARP, SIP, DNS, and RTP).

To start Network Capture:

1. Select the Interface.
2. Configure restriction parameters, if needed.
3. Select packets to capture: all or specific ones.
4. Click Start to start a network capture.
5. Click Stop to stop a network capture and download the captured file.

Note: Network Capture size is limited to 24 MB. This will limit the duration of captured file.

12.1.4 Ping

Ping is used diagnostically to ensure that a destination (e.g. host computer) the user is trying to reach is operating. Ping works by sending an **Internet Control Message Protocol** (ICMP) Echo Request to a specified interface on the network and waiting for a reply. Ping can be used for troubleshooting to test connectivity and determine response time.

To ping a target:

1. Enter the destination IP address or hostname in the **Ping Target** field.
2. Click **Start Ping**.
3. The results of the ping will be displayed in the **Ping Output** window.

12.1.5 Traceroute

Traceroute is a utility that records the route (the specific gateway at each hop) through the Internet between your device and a specified destination. It also calculates and displays the amount of time each hop took.

To traceroute a target:

1. Enter the destination IP address or hostname in the **Traceroute Target** field.
2. Tick the **Use ICMP** checkbox to send an ICMP request to the ping destination (MS Windows standard), otherwise a UDP request will be sent (Linux standard).
3. Click **Start Traceroute**.
4. The results of the ping will be displayed in the **Traceroute Output** window.

Note: No **Traceroute** is possible if the **Firewall level** is set to "**High**". For the purpose of tracerouting, several IP packets are sent out. UDP is used to send packets and ICMP is used to receive information about the routers. In their headers, the **TTL** value increases from **1** to **30**. When the first IP packet is received by the first router, its IP address will be returned in its acknowledgement.

12.2 System Logs

The **System Logs** page shows the logs on QX. System logs are useful to determine any kind of problems on the QX.

You can collect **user logs** from handset. Dial *82 to collect the logs. The collected logs will be a part of the **System Logs** when you download them next time. This could be used to collect the logs at the exact moment when a problem occurs.

12.2.1 System Logs Settings

The System Logs Settings page is used to adjust system logging settings. The following settings (options) are available:

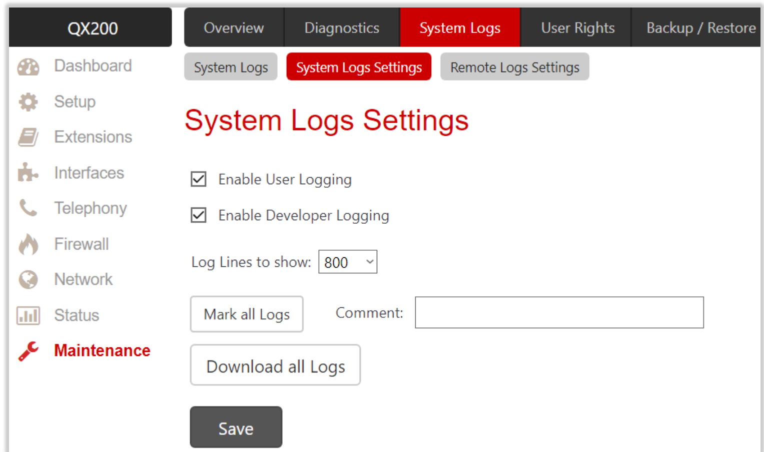


Figure 166: System Logs Settings page

- Enable User Logging – this logging contains brief information about events on QX.
- Enable Developer Logging – this logging contains detailed information about events on QX.
- Log Lines to Show is used to select the maximum number of log lines to display on the System Logs page.
- Mark all Logs is used to set a line marker in the logs.
- Comment is used to describe the problem captured in the following logs.
- Download all Logs is used to download all logs in (*.tar) file format. These logs can then be used by Epygi Technical Support to determine the issues that has occurred on QX.

12.2.2 Remote Logs Settings

The Remote Logs Settings page is used to adjust the system logging settings for collecting the logs remotely. These logs can then be used by Epygi Technical Support to determine the issues that has occurred on QX.

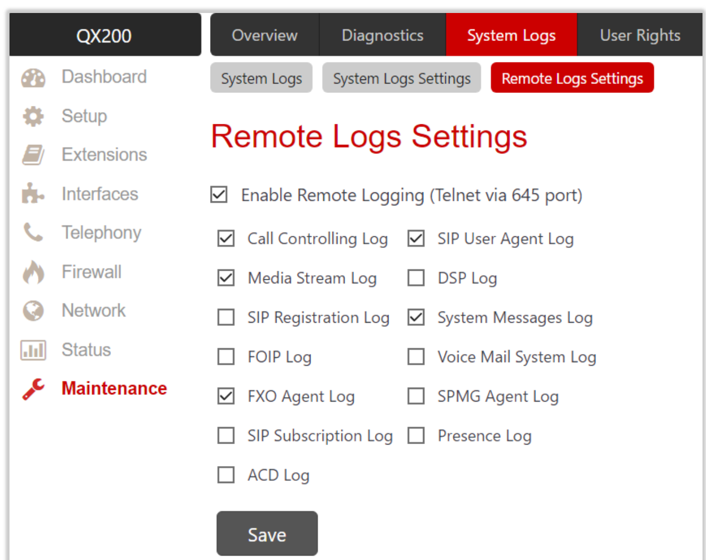


Figure 167: Remote Logs Settings page

12.2.3 Logs Archive

The **System Logs Archive** page shows the archived logs table with time period by **Date**. Clicking on the corresponding date will open the archived system logs table on an hourly basis. **Hour** shows the initiation time of the system logs. It can be used to collect the logs at the exact moment when the issue has started.

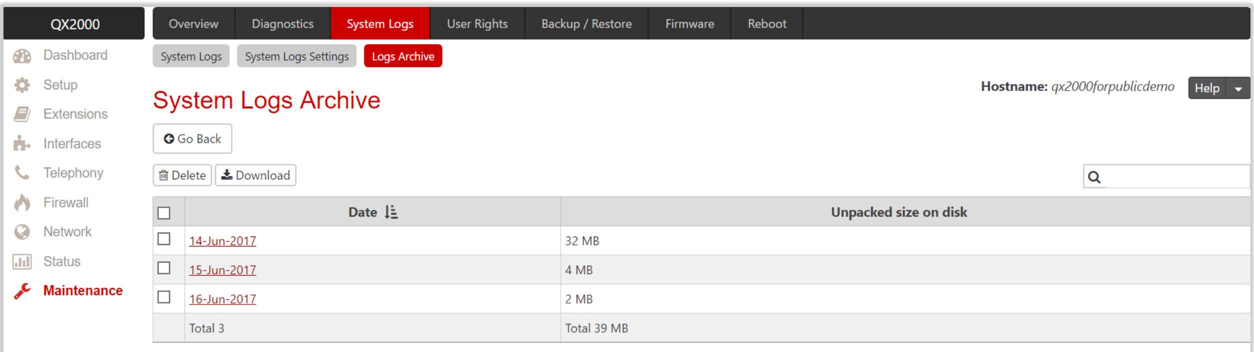


Figure 168: System Logs Archive page

12.3 User Rights

The **User Rights** service is used to configure permissions/restrictions on the GUI access for various users.

12.3.1 Users

The **Users** page contains a table where the **Administrator** and **Local Administrator** accounts are listed. This page allows to modify the passwords of **Administrator** and **Local Administrator** accounts. Two levels of QX GUI administration are available:

- **admin** – this is the **Administrator** account. The latter has access to all WEB GUI pages. The administrator is responsible for granting access to all other user groups. By default, as well as after factory reset of QX, the **admin password** is set to **19**.
- **localadmin** – this is a common **sub-administrator** account. **Local Administrator** has permission to access and adjust each WEB GUI page. By default, as well as after factory reset of QX, the **localadmin password** is set to **19**. The **localadmin** account is disabled by default.

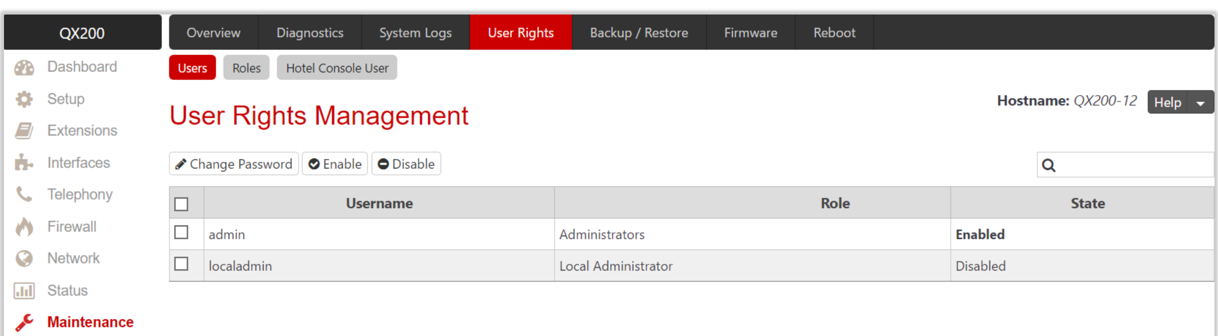


Figure 169: User Rights Management – Users page

To change the **GUI Access Password**:

1. Tick the checkbox next to the **admin** or **localadmin** entry in the table and click **Change Password**.
2. The **Change Password** page appears for the selected user. Select **GUI Access Password** tab.
   - Enter the old password (by default – 19)
   - Enter a new password and then re-enter it to confirm.
3. Click **Save** to change the password.

The **Phone Access Password** is used for authentication purposes (when connecting to 3PCC application using **admin** account) as well as for Administrator Login (*75).

To change the **Phone Access Password**:

1. Tick the checkbox next to the **admin** entry in the table and click **Change Password**.
2. The **Change Password** page appears for selected user. Select **Phone Access Password** tab.
   - Enter a new password and then re-enter it to confirm.
3. Click **Save** to change the password.

Note:

- The **GUI Access Password** can consist of lowercase and uppercase alphabetic characters, digits and symbols. A maximum password length is **20** characters.
- The **Phone Access Password** can consist of only digits. A maximum password length is **20** characters.
- In order to keep passwords safe, make sure you write it down in a safe place and don't share it with others.

12.3.2 Roles

The **Roles** page contains a table where the user roles are listed. This page allows to set access permissions to the GUI pages for each role in the table.

- **Local Administrator** – this role can have permissions to adjust each GUI page.
- **Extension** – this role refers to all user extensions created on QX. Permissions for each GUI page can be adjusted.

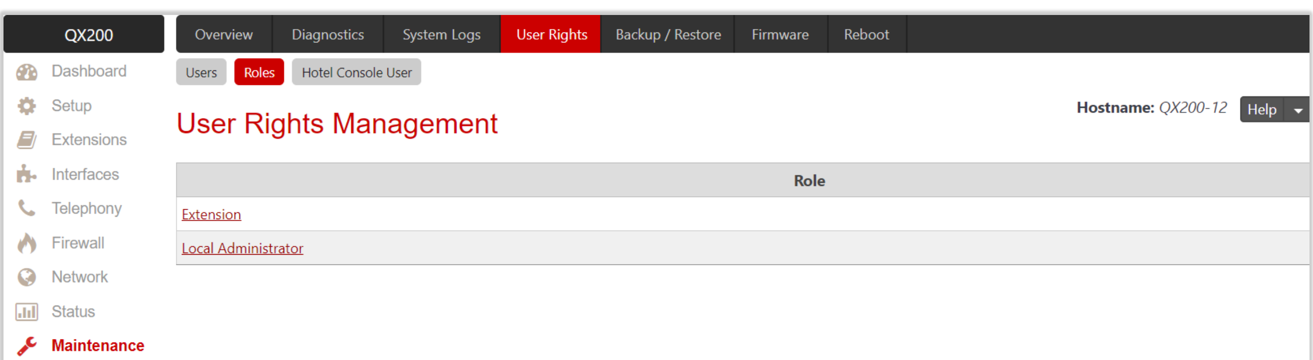


Figure 170: User Rights Management – Roles page

To manage the permissions for the selected role:

1. Click the hyperlinked role (**Extension** or **Local Administrator**). The **Access Rights** page will be opened.
2. Tick the checkbox(es) next to **CGI Name**.
3. Click the **Grant Access** or **Deny Access** to grant/deny access for the selected page(s).

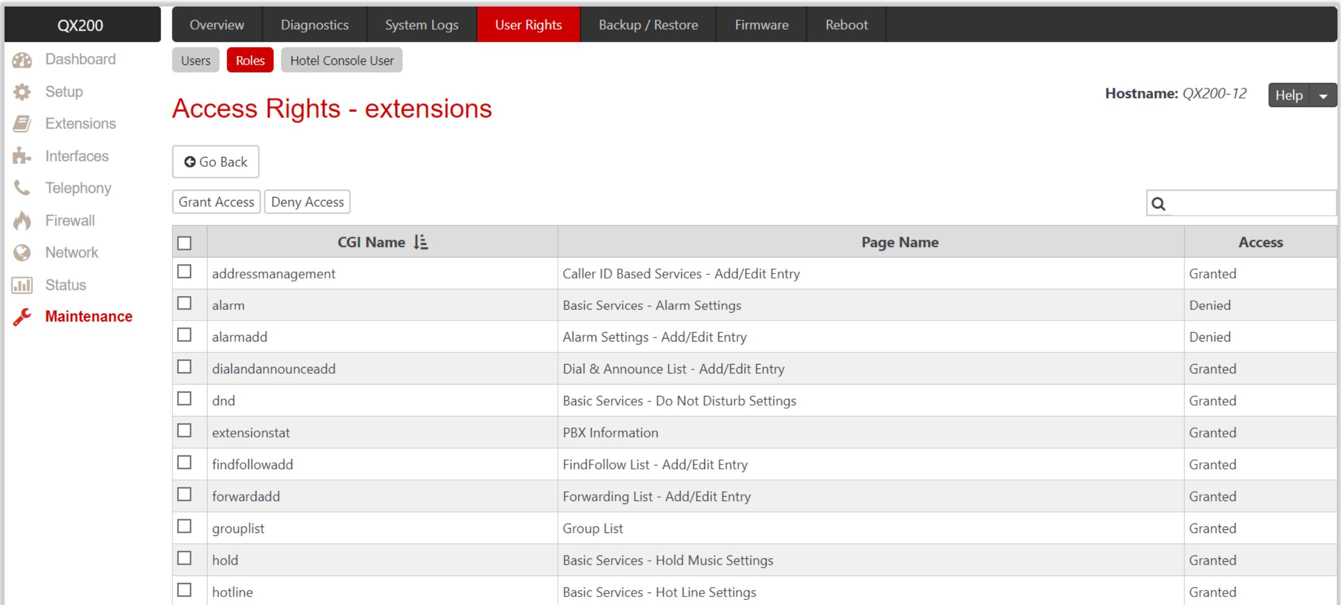


Figure 171: Access Rights – extensions page

12.3.3 Hotel Console User

The **Hotel Console User Rights Management** page is used for managing the users allowed to connect to the QX from the **Epygi Hotel Console** (EHC) application.

For more information on how to configure and use **EHC** application with QX, refer to the Epygi Hotel Console (EHC) - User Guide.

12.4 Backup / Restore

12.4.1 Backup / Restore

Configuration Management includes features that allow to back up and save the current configuration of QX, restore the configuration from backups created earlier, as well as to restore the system default configuration.

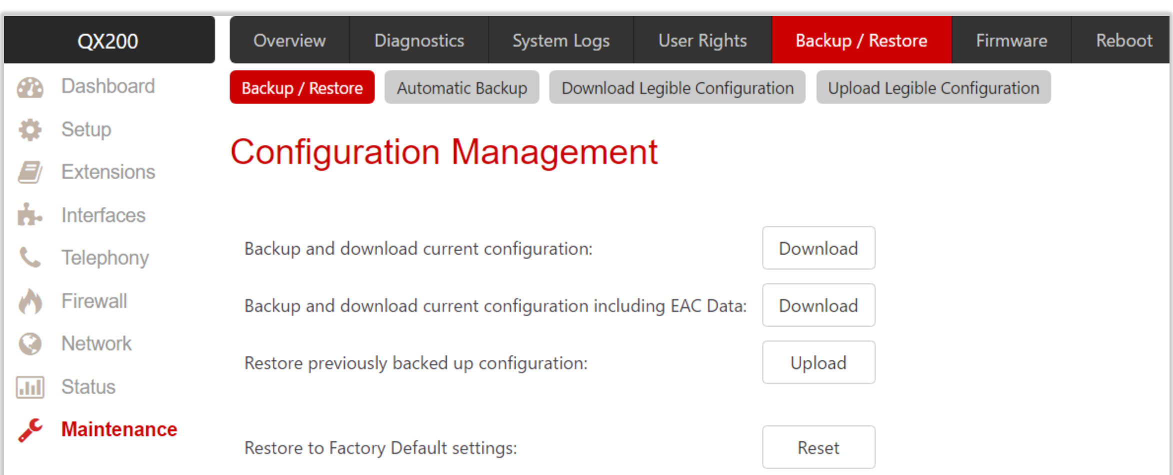


Figure 172: Configuration Management page

The following settings (options) are available:

- Backup and download current Configuration – this option is used to create a backup file with all current configuration settings and system voice messages (default and customized). Click the Download button to back up and download the current configuration. The file will be saved in the (*.bin) format. The backup filename will have the following format: config_[Hostname]_[Firmware Version]_[Date/Time].bin
- Backup and download current configuration including EAC data – this option is also used to create a backup file with all current configuration settings and system voice messages (default and customized). Compared to the previous option the current configuration includes the EAC data, covering the EAC Chat database, Agents Status and Call Statistics. Click the Download button to back up and download the current configuration. The file will be saved in the (*.bin) format. The backup filename will have the following format: config_[Hostname]_[Firmware Version]_[Date/Time].bin

Note: Voice Mails and Call Recordings are not backed up and included in the configuration file.

- Restore previously backed up configuration is used to restore earlier created backup file and replace the current configuration settings and system voice messages.

1. Click the Upload button.
2. Click Choose File to open the file chooser window and browse for the file.
3. Click Save to start configuration restore.

Note: QX doesn't allow to restore the earlier created backup in case it is running a firmware version lower than the version at the time of configuration backup.

- Restore to Factory Default settings is used to reset all configuration settings and restores factory default settings of device.

1. Click the Reset button.
2. Click Yes to proceed the factory reset procedure.

Note: Unlike the factory reset done by pressing the Reset pin on QX manually, this option will keep the following data:

- The device registration with Epygi Technical Support.
- The installed license keys.

12.4.2 Automatic Backup

The Backup Configuration Management service allows to activate and configure the automatic backup of the current configuration and system voice messages (default and customized).

The following settings (options) are available:

- Enable Automatic Backup is used to activate service on QX.
- Include EAC Data is used to include the EAC data, covering the EAC Chat database, Agents Status and Call Statistics in the backup file.
- Send via Email allows sending the backup file via e-mail. The destination e-mail address has to be entered in the E-mail Address field.
- Send to Server allows sending the backup file to an external server. This selection enables the following fields to be filled:
  - Server Name is used to set the IP address or the hostname of the server.
  - Server Port is used to set the port of the server.
  - Path on Server is used to set the path on the server.

- Send Method – the server type: TFTP or FTP. Specify the Username and Password in case of the FTP. If these fields are left empty, anonymous authentication will be used. TIP: Select the Use SFTP option to enable SFTP support.

- Backup Interval Selection is used to schedule the automatic backup.
- Backup Now is used to back up the configuration immediately.

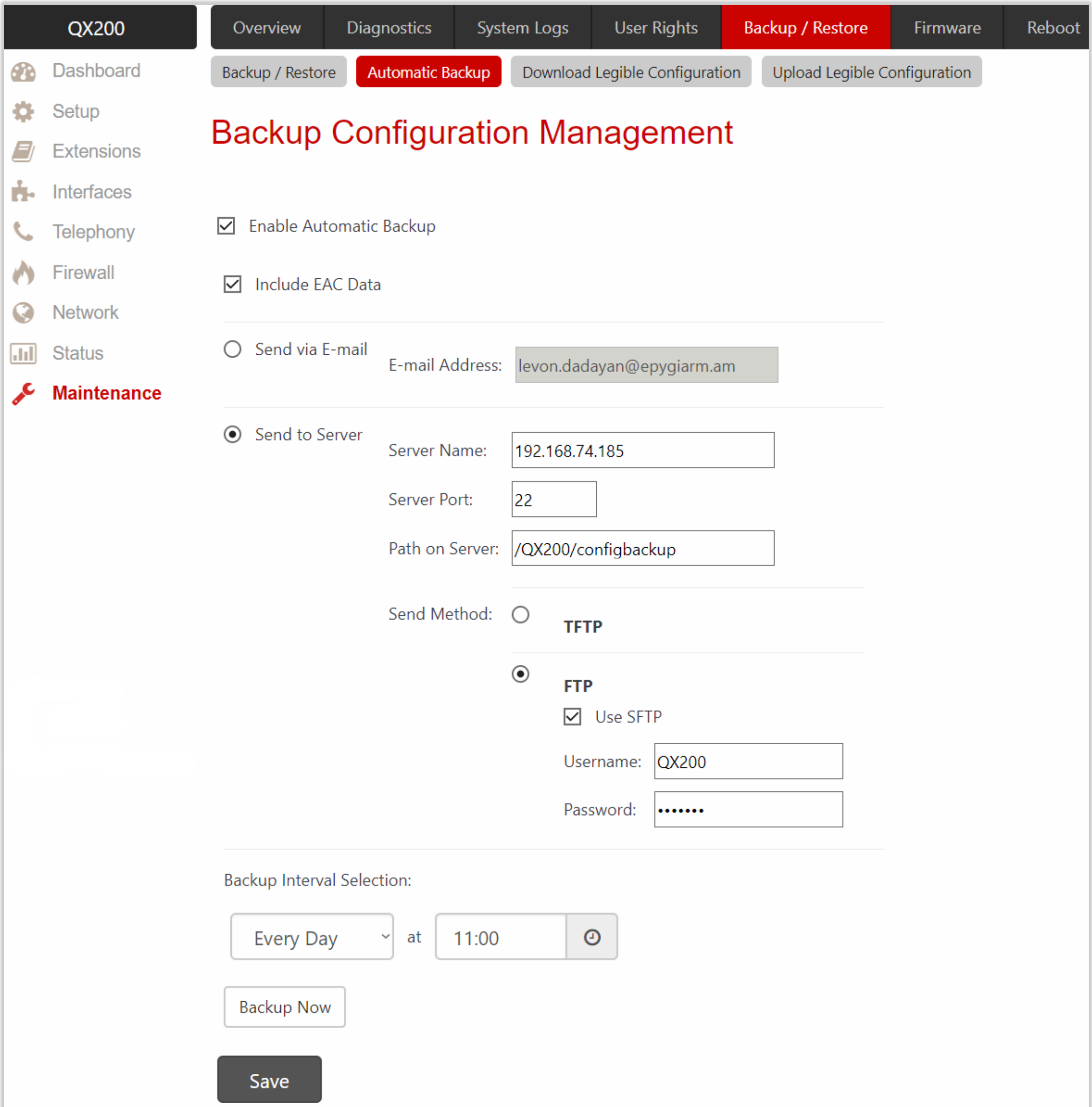


Figure 173: Automatic Backup page

12.4.3 Download Legible Configuration

The **Legible Configuration** service allows to generate a piece of QX configuration, download it to review and make necessary changes, then upload back to update the configuration. The downloaded **Legible Configuration File** (LCF) contain QX configuration parameters in (*.txt) file format. LCF can be edited with any text editor and uploaded back to save the changes on the same or another QX system.

For more information on how to configure and use **Legible Configuration** service, refer to the Legible Configuration on QX IP PBXs guide.

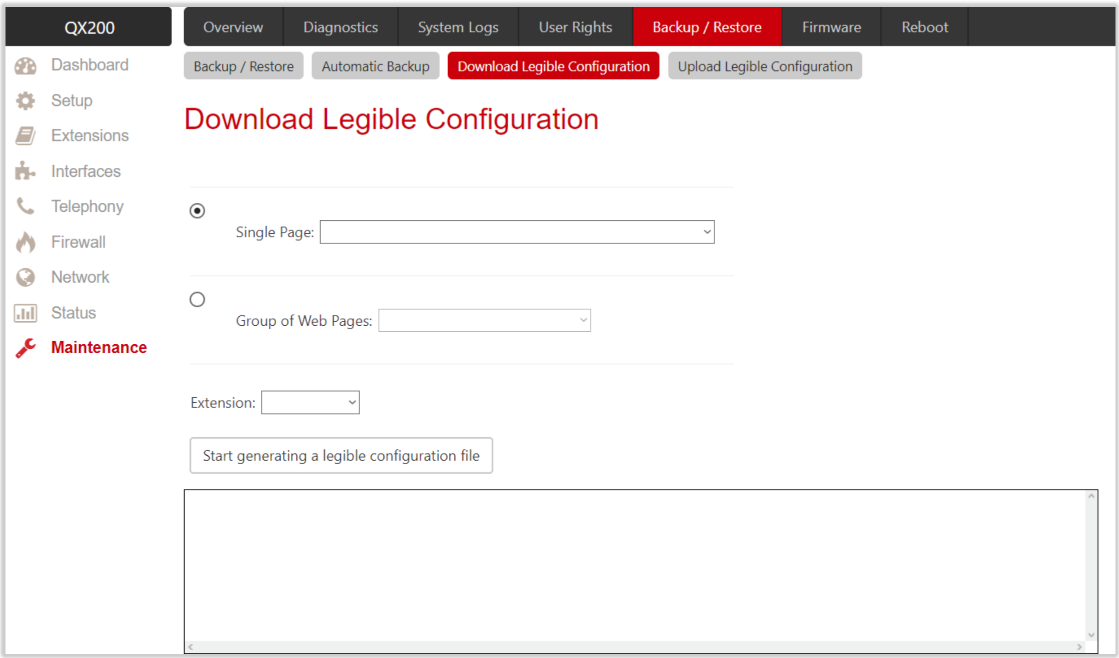


Figure 174: Download Legible Configuration page

The following radio buttons are used to select between a specific CGI or a group of CGIs:

- **Single Page** is used to select a certain page from the list of WEB GUI pages the legible configuration can be manually managed for. For example: selecting **RTP Settings** will generate a legible configuration file with parameters present on the RTP Settings page.
- **Group of Web Pages** is used to choose among the four predefined groups: **Internet Connection Settings**, **LAN Configuration Settings**, **Telephony General Settings** and **Extension Settings**. Each of these groups refer to all pages characterized by the selected criteria, e.g. **Internet Connection Settings** group contains all parameters on the pages related to the networking and **WAN** configuration.
- **Extension** is used to select the settings in the generated legible configuration file to one specific extension. For example: each of the extensions on QX has its own SIP settings or Codecs. To download the settings for a particular extension only, you need to choose the corresponding extension from the list. The drop-down may also have a blank selection. In that case, the LCF will contain the parameter of all available extensions on QX (if the selected parameter applies to the extension and not to the overall system, like RTP settings).

The following functional buttons are available:

- **Start generate a legible configuration file** is used to start parsing the configuration structure of the device for the defined parameters. The progress will be displayed in the window.
- **Cancel generation process** is used to stop the generation procedure. This button appears once the configuration generation procedure has been started.

- **Download generated configuration!** is used to download the generated file in the (*.txt) format. This button appears when the legible configuration generation is finished. Necessary changes can be made in the downloaded configuration file and then uploaded back to the system.
- **View generated configuration!** is used to view the generated file directly in the browser. This button appears once the legible configuration generation is finished.
- **Restart generation!** is used to cancel the generated configuration file and start over. This button appears once the legible configuration generation is finished.

12.4.4 Upload Legible Configuration

The **Upload Legible Configuration** page is used to upload a configuration file in (***.txt**) file format.

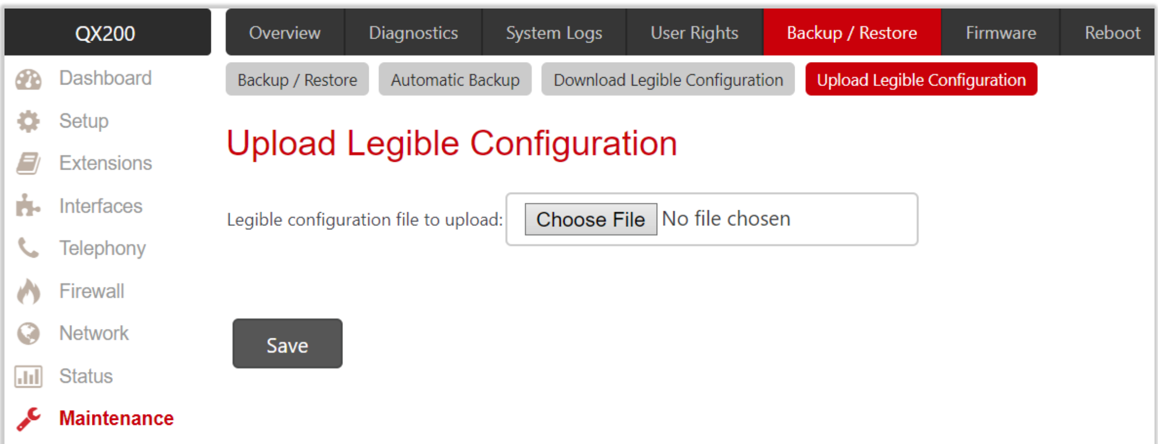


Figure 175: Upload Legible Configuration page

Checking the Validity of a LCF

Before applying the changes specified in the LCF, QX checks the validity of the uploaded LCF. First, the QX compares the FW version indicated in the LCF with the currently running one on the QX. If they match, the QX will proceed checking the correctness of the specified settings similarly as it does when the user presses **Save** to submit the changes. At any point, the QX detects a mistake (version mismatch, wrong value for a setting, a wrong syntax). It will generate an error and delete the LCF without applying any change. If no mistakes are found in the edited LCF, the QX will start to sequentially apply the changes.

12.5 Firmware

The **Firmware** section is used to update the firmware of QXs. The following options are available for updating the current firmware:

- Upload and update firmware manually.
- Download and update firmware manually.
- Download and update firmware automatically

For more information on how to update the QX **firmware**, refer to the Firmware Update Service on Epygi QX Line guide.

Attention:

- It is recommended to back up the configuration for **emergency purposes** prior to upgrading the firmware. You can do that by clicking the **Download Configuration** link in the **Manual Firmware Update** page. The current configuration will remain once the firmware has been updated. Moreover, voice mails, call recordings, all custom messages and call history will be saved during the upgrade.
- Firmware installation will take about 5 minutes. During that time, QXs will be in non-operational condition, neither telephony nor Internet access is possible.
- You will not be automatically redirected to the Login page. To access QX WEB GUI, connect to QX again and login.
- QX will factory reset and the system configuration will be lost while downgrading the firmware.
- After the firmware update, all IP phones attached to the QX will be restarted.

12.5.1 Manual Firmware Update

The **Manual Firmware Update** page is used to upload and update the QX firmware manually.

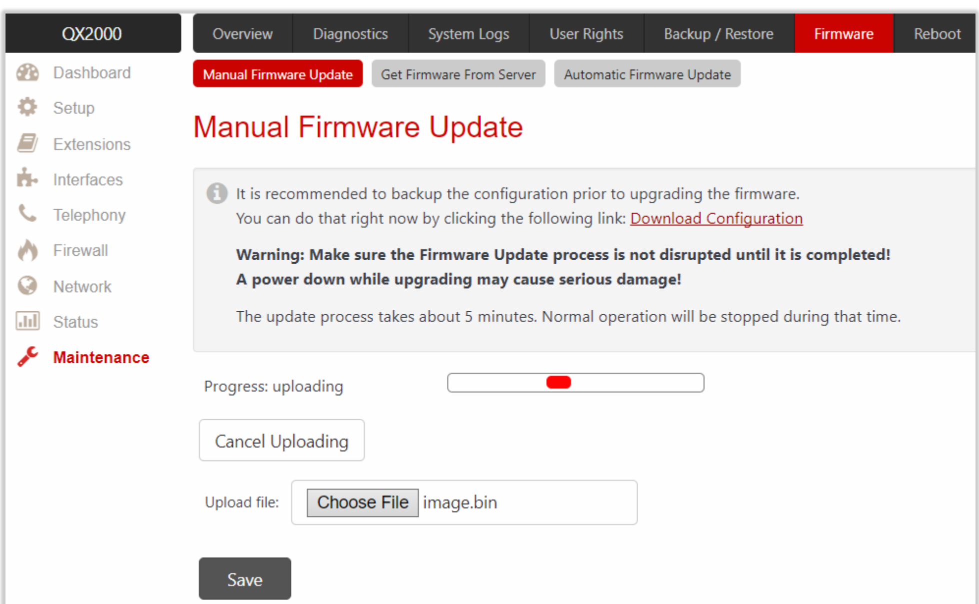


Figure 176: Manual Firmware Update page

To perform **Manual Firmware Update**:

1. Go to the **Maintenance→Firmware→Manual Firmware Update** page.
2. Click the **Download Configuration** link to back up the current configuration (recommended).
3. Click the **Choose File** button to browse for **image.bin** file.

4. Click **Save** to start uploading the file. The following information will be displayed once the firmware has been uploaded.
   - **Firmware check** shows the status of the uploaded firmware. **Invalid** status means that the uploaded firmware is not compatible with the QX hardware version.
   - **Current Firmware Version/New Firmware Version** shows the current/new firmware versions accordingly.
5. Click **Yes** to proceed the update or click **Discard this firmware** to close the message without updating the device.

Note: The update process takes about 5 minutes. Normal operation will be stopped during that time.

12.5.2 Get Firmware From Server

The **Manual Firmware Update from Server** page is used to manually download and update the QX firmware from the FTP server.

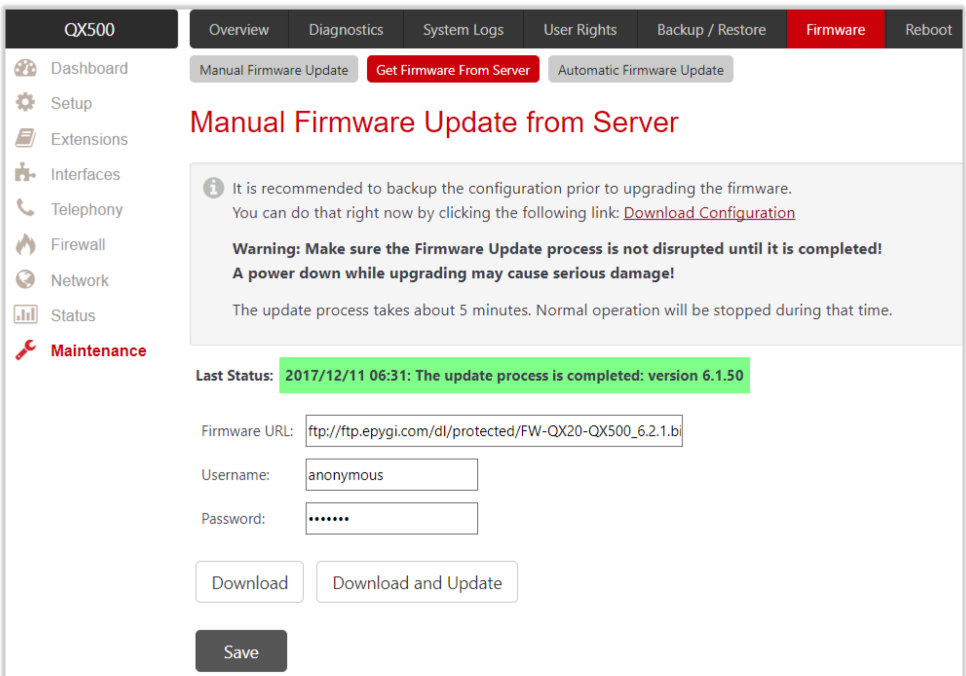


Figure 177: Manual Firmware Update from Server page

To perform **Manual Download and Firmware Update**:

1. Go to the **Maintenance→Firmware→Manual Firmware Update** page.
2. Click the **Download Configuration** link to back up the current configuration (recommended).
3. Set the **Firmware URL** to get the new firmware located in the FTP server.
4. Set the **Username** and **Password** to pass the FTP server authentication (if needed).
5. Click **Save** to apply changes before starting downloading and updating the firmware.
6. Click **Download and Update** to automatically download and update the firmware or click **Download** to start downloading firmware from FTP server.
   - **Firmware check** shows the status of the uploaded firmware. **Invalid** status means the firmware is not compatible with the QX hardware version.
   - **Current Firmware Version/New Firmware Version** shows the current/new firmware versions accordingly.
7. Click **Update** to proceed the update or click **Discard** to close the warning message without updating the device.

Note: The update process takes about 5 minutes. Normal operation will be stopped during that time.

12.5.3 Automatic Firmware Update

The **Automatic Firmware Update** page is used to enable and configure the automatic firmware update settings on QX. When this service is enabled, on the scheduled time the QX will automatically check if a new firmware is available on the server. Then, based on the preconfigured settings, will notify user or update the firmware immediately.

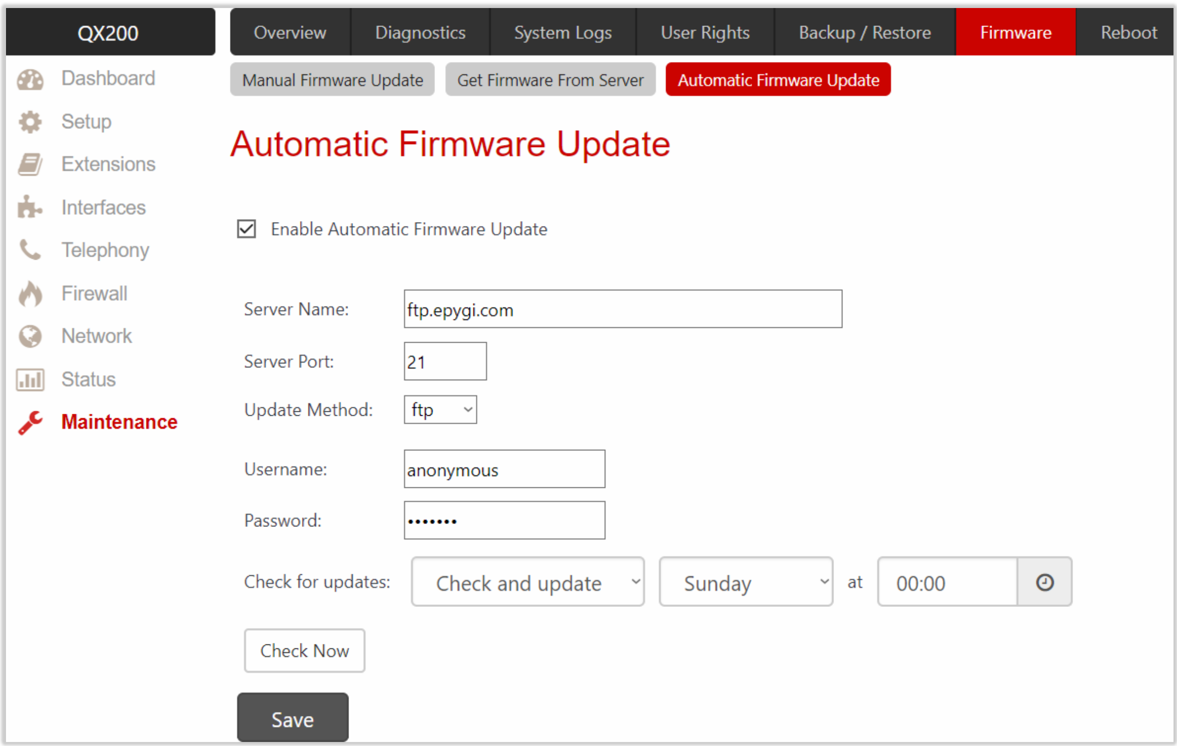


Figure 178: Automatic Firmware Update page

Note: The server configuration can be done manually. The recommended and simplest method is to use the Epygi public FTP server.

Check for updates based on one of the following options:

- Select the **Check and notify** option if you want QX to check for a new firmware in the server at the scheduled time and notify.
- Select **Check and update** option if you want QX to check for a new firmware, automatically download and install it on a scheduled time.
- Click **Check Now** to manually initiate the action selected from the **Check for updates** drop-down list.

To perform the automatic firmware update from Epygi FTP server:

1. Tick the **Enable Automatic Firmware Update** option.
2. Leave the **Server Name**, **Server Port**, **Update Method**, **Username** and **Password** text fields to their default values (ftp.epygi.com, 21, ftp and anonymous respectively, use blank for password) to use Epygi's public ftp server.
3. Select the **Check and update** option from the **Check for updates** drop-down list.
4. Configure the **Date/Time** settings.
5. Click **Save**.

The system will check for a new firmware at a scheduled time. If there is a new firmware available, QX will download and update it automatically.

12.6 Reboot

The **Yes, Reboot Device** button is used to reboot the QX. TIP: The WEB GUI session with the QX will be terminated, i.e., after successful reboot you need to log in again.

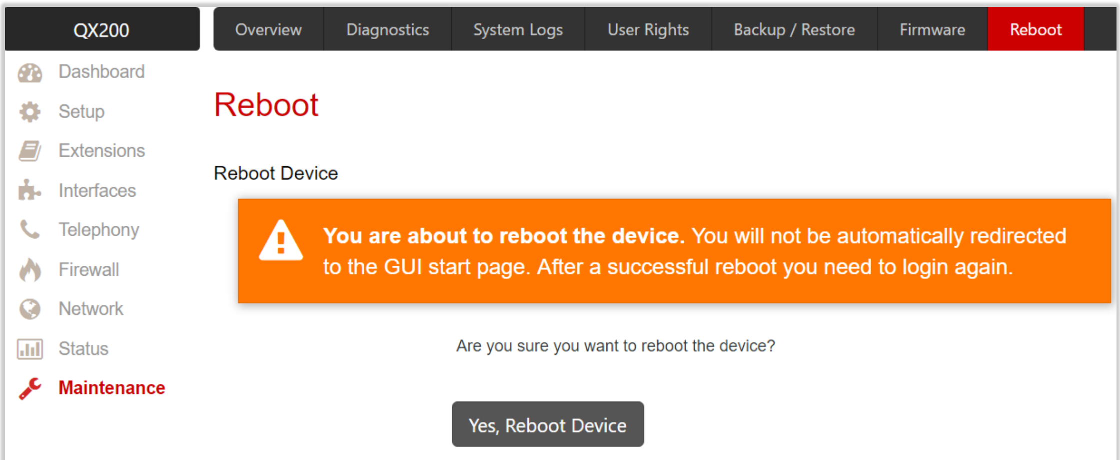


Figure 179: Reboot Device page

12.7 Registration Form

The **Register Your Device in Technical Support Center** page appears when configuring the unregistered QX, and it has been created for customer support purposes. The page allows you to register on Epygi Technical Support.

Figure 180: Device Registration page

The following registration options are available:

- **Register now** leads to the **Epygi Technical Support System Registration** page and requires customer information to submit into QX registration form.
- **Remind me later** hides the registration notification until the next login.
- **Don't remind me again** hides the registration notification forever.

13 Appendices

13.1 Administrator Login

The **Administrator Login** is used to review and modify the auto attendant greeting and recurring prompt as well as the universal extension messages. Phone Access Password will be required for login.

1. Dial *75 to log in.
2. Enter the **Phone Access Password**.
3. Follow the voice prompts to review and change system messages.
4. Dial *0 or hang up to logout.

System will notify about the messages that can be reviewed and modified.

Administrator Login menu						
1 Review Attendant Greeting	2 Review Attendant Recurring Prompt	3 Review Universal Extension Messages				
Enter the Attendant Number (in case of multiple AAs)	Enter the Attendant Number (in case of multiple AAs)	1 Voice Mail Greeting message	3 Incoming Call Blocking message	4 Outgoing Call Blocking message	6 Out of Office message	7 Find Me/ Follow Me message
1 Listen to the current greeting	1 Listen to the current prompt	1 Listen to the current message	1 Listen to the current message	1 Listen to the current message	1 Listen to the current message	1 Listen to the current message
2 Record a new greeting	2 Record a new prompt	2 Record a new message	2 Record a new message	2 Record a new message	2 Record a new message	2 Record a new message
3 Restore system default greeting	3 Restore system default prompt	3 Restore system default message	3 Restore system default message	3 Restore system default message	3 Restore system default message	3 Restore system default message
# Stop recording or playback	# Stop recording or playback	# Stop recording or playback	# Stop recording or playback	# Stop recording or playback	# Stop recording or playback	# Stop recording or playback

Table 6: Administrator Login menu

13.2 Needed Bandwidth for IP Calls

The bandwidth required for an IP call depends on the used codec. The codec specifications are listed in the tables below.

Codecs	Packet Size (in msec)					
	10	20	30	40	50	60
G.711u/G.711a	105	84	76	74	71	67
G.726-16	58	37	30	27	25	22
G.726-24	66	45	38	34	32	30
G.726-32	74	53	45	42	40	37
G.726-40	82	61	53	50	48	45
G.729a	50	29	22	19	17	15
iLBC	–	–	27	–	–	20
G.722	105	84	76	74	71	67
G.722.1	74	53	45	42	40	37

Table 7: Required Bandwidth for Standard Packets

Codecs	Packet Size (in msec)					
	10	20	30	40	50	60
G.711u/G.711a	114	89	81	76	74	72
G.726-16	66	41	33	28	26	24
G.726-24	74	49	41	36	34	32
G.726-32	82	57	49	44	42	40
G.726-40	90	65	57	52	50	48
G.729a	58	33	26	20	18	16
iLBC	–	–	31	–	–	22
G.722	114	89	81	76	74	72
G.722.1	82	57	49	44	42	40

Table 8: Required Bandwidth for Encrypted Packets when using a SRTP

Codecs	Packet Size (in msec)					
	10	20	30	40	50	60
G.711u/G.711a	148	105	90	85	80	74
G.726-16	95	59	43	38	34	29
G.726-24	108	65	52	45	41	37
G.726-32	118	74	60	53	48	45
G.726-40	124	81	66	61	56	52
G.729a	92	49	35	30	26	22
iLBC	–	–	41	–	–	26
G.722	148	105	90	85	80	74
G.722.1	118	74	60	53	48	45

Table 9: Required Bandwidth for Encrypted Packets when using a VPN

13.3 System Default Values

The values are described ONLY for features, services and options which are enabled and preconfigured by default. The following abbreviations are used:

- n/a – not applicable for the selected QX model
- E – the setting (option) is enabled
- D – the setting (option) is disabled
- SA – the service or setting (option) is activated/preconfigured
- SNA – the service or setting (option) is NOT activated/preconfigured
- SDM – system default message

13.3.1 System Settings

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
User Rights Management								
Username		admin	admin	admin	admin	admin	admin	admin
GUI Password		19	19	19	19	19	19	19
Phone Access Password		19	19	19	19	19	19	19
admin (role - Administrators)		E	E	E	E	E	E	E
localadmin (role – Local Administrator)		D	D	D	D	D	D	D
Access Rights – user extension	Access for all available pages	granted	granted	granted	granted	granted	granted	granted
Access Rights – localadmin	Access for all available pages	granted	granted	granted	granted	granted	granted	granted
System Configuration Wizard								
LAN IP Address		172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	192.168.0.200	192.168.0.200
LAN Subnet Mask		255.255.0.0	255.255.0.0	255.255.0.0	255.255.0.0	255.255.0.0	255.255.240.0	255.255.240.0
Hostname		epygiqx	epygiqx	epygiqx	epygiqx	epygiqx	epygiqx	epygiqx
Domain Name		epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc
DHCP Server	DHCP Settings for the LAN Interface	SNA	SNA	SNA	SNA	SNA	SNA	SNA
Your locale (location)		US	US	US	US	US	US	US
Timezone		(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)	(GMT-06:00) Central Time (US&Canada)
Emergency Codes		n/a	911	911	911	n/a	n/a	n/a
PSTN Access Code		n/a	9	9	9	n/a	n/a	n/a

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
System Configuration Wizard								
Generate System Event		n/a	SNA	SNA	SNA	SNA	n/a	n/a
Send Notification via E-mail		n/a	SNA	SNA	SNA	SNA	n/a	n/a
Send Notification via SMS		n/a	SNA	SNA	SNA	SNA	n/a	n/a
Leave Voice Message		n/a	SNA	SNA	SNA	SNA	n/a	n/a
Internet Configuration Wizard								
WAN Interface Protocol		Ethernet	Ethernet	Ethernet	Ethernet	Ethernet	Ethernet	Ethernet
Upstream [kbit/s]		1.000.000	100.000	100.000	100.000	100.000	1.000.000	1.000.000
Downstream [kbit/s]		1.000.000	100.000	100.000	100.000	100.000	1.000.000	1.000.000
Min Date Rate [kbit/s]		0	0	0	0	0	0	0
IP configuration of the WAN interface		Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	n/a	n/a
MAC Address Assignment		This device	This device	This device	This device	This device	n/a	n/a
Maximum Transfer Unit (MTU)		1500	1500	1500	1500	1500	1500	1500
DNS Settings		Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	Obtain an IP Address automatically	Undefined	Undefined
Date / Time Settings								
SNTP Server		E	E	E	E	E	E	E
SNTP Client		E	E	E	E	E	E	E
SNTP Server		ntp1.epygi.com	ntp1.epygi.com	ntp1.epygi.com	ntp1.epygi.com	ntp1.epygi.com	ntp1.epygi.com	ntp1.epygi.com
Polling Interval		6 hr.	6 hr.	6 hr.	6 hr.	6 hr.	6 hr.	6 hr.
System Security Management								
Security Level		Medium	Medium	Medium	Medium	Medium	Medium	Medium
Extensions Management								
Extension Length		3	3	3	3	3	4	4
Extensions attached to FXS lines		n/a	(101-102) to (1-2)	n/a	(101-102) to (1-2)	n/a	n/a	n/a
Extensions attached to IP lines		(101-112) to (1-12)	(103-118) to (1-16)	(101-116) to (1-16)	(103-126) to (1-24)	(101-200) to (1-100)	(101-1200) to (1-200)	(101-1200) to (1-200)
Percentage of System Memory for Extensions attached to FXS lines		n/a	5%	n/a	5%	n/a	n/a	n/a
Percentage of System Memory for Extensions attached to IP lines		0.1%	0.4%	0.4%	0.4%	0.1%	0.04%	0.02%

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Auto Attendant 00 – General Settings								
Display Name		Attendant	Attendant	Attendant	Attendant	Attendant	Attendant	Attendant
Enable FAX forwarding		D	D	D	D	D	D	D
Show on Public Directory		E	E	E	E	E	E	E
Percentage of System Memory		5%	5%	5%	5%	5%	0.08%	0.08%
Auto Attendant 00 – Attendant Settings								
Schedule		D	D	D	D	D	D	D
Attendant Scenario		Standard	Standard	Standard	Standard	Standard	Standard	Standard
Auto Attendant 00 – Attendant Scenario								
Pass Dialed Digits through Call Routing		D	D	D	D	D	D	D
Call Redirection		SNA	SNA	SNA	SNA	SNA	SNA	SNA
ZeroOut Redirection		SNA	SNA	SNA	SNA	SNA	SNA	SNA
Welcome Message		E	E	E	E	E	E	E
Welcome Message and Recurring Prompt	The system default messages are used.	SDM	SDM	SDM	SDM	SDM	SDM	SDM
Auto Attendant 00 – SIP Registration Settings								
Username / DID Number		00	00	00	00	00	00	00
Password		left blank	left blank	left blank	left blank	left blank	left blank	left blank
SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank
SIP Port		5060	5060	5060	5060	5060	5060	5060
SIP Registration Transport		UDP	UDP	UDP	UDP	UDP	UDP	UDP
Registration on SIP Server		D	D	D	D	D	D	D
Auto Attendant 00 – SIP Advanced Settings								
Authentication Username		None	None	None	None	None	None	None
Send Keep-alive Messages to Proxy		D	D	D	D	D	D	D
RTP Priority Level		Medium	Medium	Medium	Medium	Medium	Medium	Medium
Do Not use SIP Old Hold Method		D	D	D	D	D	D	D
Outbound Proxy		left blank	left blank	left blank	left blank	left blank	left blank	left blank
Secondary SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank
Outbound Proxy for Secondary SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Auto Attendant 00 – Codecs								
G711u, G711a and G729		E	E	E	E	E	E	E
Preferred Codec		G711u	G711u	G711u	G711u	G711u	G711u	G711u
G726-16, G726-24, G726-32, G726-40, iLBC, G.722, G.722.1, TDVC		D	D	D	D	D	D	D
H.263, H.263+ and H.264		D	D	D	D	D	D	D
Out of Band DTMF Transport		E	E	E	E	E	E	E
T.38 FAX		E	E	E	E	E	E	E
Pass Through FAX		E	E	E	E	E	E	E
Pass Through Modem		D	D	D	D	D	D	D
Force Self Codecs Preference for Inbound Calls		D	D	D	D	D	D	D
SRTP Policy		Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything
User Extension – General Settings								
Display Name		None	None	None	None	None	None	None
Password		left blank	left blank	left blank	left blank	left blank	left blank	left blank
Use Kickback		D	D	D	D	D	D	D
Allow Call Relay		D	D	D	D	D	D	D
Allow GUI Login Allowed		D	D	D	D	D	D	D
Allow 3pcc/Click2Dial Access		D	D	D	D	D	D	D
Show on Public Directory		D	D	D	D	D	D	D
Use Parent Extension		D	D	D	D	D	D	D
User Extension – SIP Registration Settings								
Username / DID Number		Same as the extension number	Same as the extension number	Same as the extension number	Same as the extension number	Same as the extension number	Same as the extension number	Same as the extension number
Password		left blank	left blank	left blank	left blank	left blank	left blank	left blank
SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank
SIP Port		5060	5060	5060	5060	5060	5060	5060
SIP Registration Transport		UDP	UDP	UDP	UDP	UDP	UDP	UDP
Registration on SIP Server		D	D	D	D	D	D	D

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
User Extension – SIP Advanced Settings								
Authentication Username		None	None	None	None	None	None	None
Send Keep-alive Messages to Proxy		D	D	D	D	D	D	D
RTP Priority Level		Medium	Medium	Medium	Medium	Medium	Medium	Medium
Do Not use SIP Old Hold Method		D	D	D	D	D	D	D
Outbound Proxy		left blank	left blank	left blank	left blank	left blank	left blank	left blank
Secondary SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank
Outbound Proxy for Secondary SIP Server		left blank	left blank	left blank	left blank	left blank	left blank	left blank
User Extension – Voice Mailbox Settings								
Voice Mailbox type		Use Internal Voice Mail	Use Internal Voice Mail	Use Internal Voice Mail	Use Internal Voice Mail	Use Internal Voice Mail	Use Internal Voice Mail	Use Internal Voice Mail
Configuration wizard status		Activated	Activated	Activated	Activated	Activated	Activated	Activated
User Extension – Codecs								
G711u, G711a and G729		E	E	E	E	E	E	E
Preferred Codec		G711u	G711u	G711u	G711u	G711u	G711u	G711u
G726-16, G726-24, G726-32, G726-40, iLBC, G.722, G.722.1, TDVC		D	D	D	D	D	D	D
H.263, H.263+ and H.264		D	D	D	D	D	D	D
Out of Band DTMF Transport		E	E	E	E	E	E	E
T.38 FAX		E	E	E	E	E	E	E
Pass Through FAX		E	E	E	E	E	E	E
Pass Through Modem		D	D	D	D	D	D	D
Force Self Codecs Preference for Inbound Calls		D	D	D	D	D	D	D
SRTP Policy		Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything	Make unsecure calls, accept anything
Universal Extension Recordings								
System Messages	The system default messages are used.	SDM	SDM	SDM	SDM	SDM	SDM	SDM
Percentage of System Memory	Memory allocation	0.1%	1%	1%	1%	0.1%	0.08%	0.08%

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
IP Lines								
IP lines attached to extensions		(1-12) to (101-112)	(1-16) to (103-118)	(1-16) to (101-116)	(1-24) to (103-126)	(1-100) to (101-200)	(1-200) to (101-1200)	(1-200) to (101-1200)
IP line State		Free	Free	Free	Free	Free	Free	Free
IP Line Settings								
PnP for IP lines		E	E	E	E	E	E	E
Firmware Version Control		E	E	E	E	E	E	E
Configure IP phones from		LAN	LAN	LAN	LAN	LAN	LAN	LAN
Phones Default Template		systemdefault	systemdefault	systemdefault	systemdefault	systemdefault	systemdefault	systemdefault
FXS Lines								
FXS lines attached to extensions		n/a	(1-2) to (101-102)	n/a	(1-2) to (101-102)	n/a	n/a	n/a
Caller ID Type		n/a	Standard 2	n/a	Standard 2	n/a	n/a	n/a
Ringer Type		n/a	Type A	n/a	Type A	n/a	n/a	n/a
FXO Line Settings								
FXO lines		n/a	2	n/a	4	n/a	n/a	n/a
Enable Line		n/a	E	n/a	E	n/a	n/a	n/a
Allowed Call Type		n/a	Both incoming and outgoing calls	n/a	Both incoming and outgoing calls	n/a	n/a	n/a
Route incoming FXO call to		n/a	00	n/a	00	n/a	n/a	n/a
ISDN Trunk Settings								
ISDN trunks		n/a	n/a	4	n/a	n/a	n/a	n/a
Interface Type		n/a	n/a	User	n/a	n/a	n/a	n/a
Connection Type		n/a	n/a	PTMP	n/a	n/a	n/a	n/a
Service Type		n/a	n/a	No MSN	n/a	n/a	n/a	n/a
Route incoming call to		n/a	n/a	00	n/a	n/a	n/a	n/a
Call Routing Table								
Call Routing Rule 1	Destination Number Pattern to call 00 Auto Attendant	00	00	00	00	00	00	00
Call Routing Rule 2	Destination Number Pattern to call PBX extensions	???	???	???	???	???	????	????
Call Routing Rule 3	Destination Number Pattern to call SIP (sip.epygi.com)	8*	8*	8*	8*	8*	8*	8*

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
NAT Traversal								
NAT Traversal for SIP		Automatic	Automatic	Automatic	Automatic	Automatic	Automatic	Automatic
NAT Traversal - SIP Parameters		Use STUN	Use STUN	Use STUN	Use STUN	Use STUN	Use STUN	Use STUN
NAT Traversal - RTP Parameters		Use STUN	Use STUN	Use STUN	Use STUN	Use STUN	Use STUN	Use STUN
NAT Traversal – STUN Parameters								
Primary STUN Server		stun.epygi.com	stun.epygi.com	stun.epygi.com	stun.epygi.com	stun.epygi.com	stun.epygi.com	stun.epygi.com
Primary STUN Port		3478	3478	3478	3478	3478	3478	3478
Polling Interval		1 hour	1 hour	1 hour	1 hour	1 hour	1 hour	1 hour
Keep-alive Interval		120 sec.	120 sec.	120 sec.	120 sec.	120 sec.	120 sec.	120 sec.
NAT IP checking Interval		300 sec.	300 sec.	300 sec.	300 sec.	300 sec.	300 sec.	300 sec.
RTP Settings								
Packetization Interval for G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a		20	20	20	20	20	20	20
Packetization Interval for iLBC		30	30	30	30	30	30	30
Silence Suppression for G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, iLBC		Yes	Yes	Yes	Yes	Yes	Yes	Yes
G726 Standard		Use ITU-T specification	Use ITU-T specification	Use ITU-T specification	Use ITU-T specification	Use ITU-T specification	Use ITU-T specification	Use ITU-T specification
RTP/RTCP Port Range		6000-6509	6000-6255	6000-6255	6000-6255	6000-6509	6000-7799	6000-8399
Enable RTCP Support		D	D	D	D	D	D	D
SIP Settings								
UDP Port		5060	5060	5060	5060	5060	5060	5060
TCP Port		5060	5060	5060	5060	5060	5060	5060
TLS Port		5061	5061	5061	5061	5061	5061	5061
Realm		epygi	epygi	epygi	epygi	epygi	epygi	epygi
Session Timer		D	D	D	D	D	D	D
DNS Server for SIP		Default	Default	Default	Default	Default	Default	Default
SIP Timers		RFC3261	RFC3261	RFC3261	RFC3261	RFC3261	RFC3261	RFC3261

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Schedules								
Work Hours	for Company's Schedule	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)	09:00-13:00, 14:00-18:00 (from Monday to Friday)
Observe Holidays	for Company's Schedule	D	D	D	D	D	D	D
State	for Company's Schedule	Running on Schedule	Running on Schedule	Running on Schedule	Running on Schedule	Running on Schedule	Running on Schedule	Running on Schedule
Voice Mail Common Settings								
Recording Codec		G711u	G711u	G711u	G711u	G711u	G711u	G711u
E-mail Subject for Voice Mail		Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]	Voice mail received from $[VM_DISPNAME] $[VM_USERNAME]
FAX to E-mail Format		TIFF	TIFF	TIFF	TIFF	TIFF	TIFF	TIFF
Gain Control								
FXS 1 (Transmit Gain/Receive Gain)		-6/0	-6/0	-6/0	-6/0	-6/0	-6/0	-6/0
FXS 2 (Transmit Gain/Receive Gain)		-6/0	-6/0	-6/0	-6/0	-6/0	-6/0	-6/0
FXO 1 (Transmit Gain/Receive Gain)		0/6	0/6	0/6	0/6	0/6	0/6	0/6
FXO 2 (Transmit Gain/Receive Gain)		0/6	0/6	0/6	0/6	0/6	0/6	0/6
FXO 3 (Transmit Gain/Receive Gain)		0/6	0/6	0/6	0/6	0/6	0/6	0/6
FXO 4 (Transmit Gain/Receive Gain)		0/6	0/6	0/6	0/6	0/6	0/6	0/6
ISDN 1 (Transmit Gain/Receive Gain)		0/0	0/0	0/0	0/0	0/0	0/0	0/0
ISDN 2 (Transmit Gain/Receive Gain)		0/0	0/0	0/0	0/0	0/0	0/0	0/0
ISDN 3 (Transmit Gain/Receive Gain)		0/0	0/0	0/0	0/0	0/0	0/0	0/0
ISDN 4 (Transmit Gain/Receive Gain)		0/0	0/0	0/0	0/0	0/0	0/0	0/0
Audio Line Out (Transmit Gain)		Off	Off	Off	Off	Off	Off	Off
Audio Line In (Receive Gain)		Off	Off	Off	Off	Off	Off	Off
Voice Mail (Recording Gain)		0	0	0	0	0	0	0
Voice Mail (Playback Gain)		0	0	0	0	0	0	0
Dial Timeout Settings								
Routing Dial Timeout		4 sec.	4 sec.	4 sec.	4 sec.	4 sec.	4 sec.	4 sec.

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Firewall Configuration								
IDS		E	E	E	E	E	n/a	n/a
NAT		E	E	E	E	E	n/a	n/a
Firewall		D	D	D	D	D	D	D
Firewall Security Level		SNA	SNA	SNA	SNA	SNA	SNA	SNA
Advanced Firewall Configuration								
Ping Stealth		E	E	E	E	E	E	E
Fool Portscanner		D	D	D	D	D	n/a	n/a
Filtering Rules								
Management Access	HTTPS service allowed for Any IP	E	E	E	E	E	E	E
SIP Access	SIP service allowed for Any IP	E	E	E	E	E	E	E
SIP IDS Settings								
SIP IDS		E	E	E	E	E	E	E
Add the IP address into the Blocked IP List in Firewall		E	E	E	E	E	E	E
Discard SIP messages from IP address for		32 sec.	32 sec.	32 sec.	32 sec.	32 sec.	32 sec.	32 sec.
Second LAN Interface Settings								
Second LAN Interface		n/a	n/a	n/a	n/a	n/a	E	E
IP Address		n/a	n/a	n/a	n/a	n/a	172.30.0.1	172.30.0.1
Subnet Mask		n/a	n/a	n/a	n/a	n/a	255.255.0.0	255.255.0.0
DHCP Server for the Second LAN interface		n/a	n/a	n/a	n/a	n/a	D	D
DHCP Advanced Settings – DHCP Options								
Gateways		172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	192.168.0.200	192.168.0.200
Subnet Mask		255.255.0.0	255.255.0.0	255.255.0.0	255.255.0.0	255.255.0.0	255.255.254.0	255.255.254.0
Domain Name Servers		172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	192.168.0.200	192.168.0.200
NBT Name Servers		0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0	0.0.0.0
NTP Servers		172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	192.168.0.200	192.168.0.200
Domain Name		epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc
Overload TFTP Server Name		172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	172.30.0.1	192.168.0.200	192.168.0.200

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
DHCP Advanced Settings – DHCP Server Statements								
Authoritative		D	D	D	D	D	D	D
Ping Check		E	E	E	E	E	E	E
Ping Timeout		1 sec.	1 sec.	1 sec.	1 sec.	1 sec.	1 sec.	1 sec.
DNS Server Settings								
Zone		epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc	epygi-config.loc
Time to Live (TTL)		86400 sec.	86400 sec.	86400 sec.	86400 sec.	86400 sec.	86400 sec.	86400 sec.
Mail Exchange (MX)		Undefined	Undefined	Undefined	Undefined	Undefined	Undefined	Undefined
PPTP Server Configuration								
Subnet		172.31.1.0/24	172.31.1.0/24	172.31.1.0/24	172.31.1.0/24	172.31.1.0/24	n/a	n/a
Authentication		MSCHAPv2	MSCHAPv2	MSCHAPv2	MSCHAPv2	MSCHAPv2	n/a	n/a
Encryption		MPPE 128-bit	MPPE 128-bit	MPPE 128-bit	MPPE 128-bit	MPPE 128-bit	n/a	n/a
L2TP Server Configuration								
Subnet		172.31.2.0/24	172.31.2.0/24	172.31.2.0/24	172.31.2.0/24	172.31.2.0/24	n/a	n/a
Event Settings								
All available system events		Display notification	Display notification	Display notification	Display notification	Display notification	Display notification	Display notification
PPP	System events concerning PPP application	Do nothing	Do nothing	Do nothing	Do nothing	Do nothing	n/a	n/a
Call History – Settings								
Call Reporting		E	E	E	E	E	E	E
Maximum Number of Successful Call Records		100	100	100	100	100	100	100
Maximum Number of Missed Call Records		100	100	100	100	100	100	100
Maximum Number of Unsuccessful Call Records		100	100	100	100	100	100	100
CDR Parameters	CDR Parameters exclusion from CDR file	D	D	D	D	D	D	D
Call History – Archiving Settings								
Percentage of Total Memory allocated for Archive		0%	0%	0%	0%	0%	0%	0%
Call History Archiving		D	D	D	D	D	D	D

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
System Logs Settings								
User Logging		E	E	E	E	E	E	E
Developer Logging		E	E	E	E	E	E	E
Log Lines to show		1000	1000	1000	1000	1000	1000	1000
Automatic Firmware Update								
Automatic Firmware Update		E	E	E	E	E	D	D
Server Name		ftp.epygi.com	ftp.epygi.com	ftp.epygi.com	ftp.epygi.com	ftp.epygi.com	ftp.epygi.com	ftp.epygi.com
Server Port		21	21	21	21	21	21	21
Update Method		ftp	ftp	ftp	ftp	ftp	ftp	ftp
Username		anonymous	anonymous	anonymous	anonymous	anonymous	anonymous	anonymous
Password		left blank	left blank.	left blank.	left blank.	left blank.	left blank.	left blank.
Check for updates		Check and notify every day at 0:00	Check and notify every day at 0:00	Check and notify every day at 0:00	Check and notify every day at 0:00	Check and notify every day at 0:00	Check and notify every day at 0:00	Check and notify every day at 0:00

13.3.2 User Extension Settings

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Voice Mail Settings – General Settings								
Maximum Voice Mail Duration		5 min.	5 min.	5 min.	5 min.	5 min.	5 min.	5 min.
Forward/Rewind Duration		3 sec.	3 sec.	3 sec.	3 sec.	3 sec.	3 sec.	3 sec.
Ask password before granting local access to Voice Mailbox		D	D	D	D	D	D	D
Ask password before granting remote access to Voice Mailbox		E	E	E	E	E	E	E
Play welcome message		D	D	D	D	D	D	D
Play Voice Mail help		E	E	E	E	E	E	E
Automatically play Voice Mail		E	E	E	E	E	E	E
Play Voice Mails count information message		D	D	D	D	D	D	D
Play date/time information message		E	E	E	E	E	E	E
Play beep at the end of message		E	E	E	E	E	E	E
Silent Voice Mail recording		D	D	D	D	D	D	D
Voice Mail Greeting Message		SDM	SDM	SDM	SDM	SDM	SDM	SDM
Voice Mail Settings – VM Notifications								
Send new Voice Mail notifications via E-mail		D	D	D	D	D	D	D
Send new Voice Mail notifications via SMS		D	D	D	D	D	D	D
Send new Voice Mail notifications via phone call		D	D	D	D	D	D	D
Voice Mail Notification Message		SDM	SDM	SDM	SDM	SDM	SDM	SDM
Voice Mail Settings – VM Indication								
Lamp indication		E	E	E	E	E	E	E
Tone indication		n/a	E	n/a	E	n/a	n/a	n/a
Ringing indication		n/a	D	n/a	D	n/a	n/a	n/a
Voice Mail Settings – VM Redirection								
Zero Out Redirect	Calls will be redirected to 00 Auto Attendant.	E	E	E	E	E	E	E
FAX Redirection		D	D	D	D	D	D	D
Automatic Fax Receiving Mode		D	D	D	D	D	D	D

Service / Option / Parameter	Description	QX20	QX50	QXISDN4+	QX200	QX500	QX2000	QX3000
Account Settings								
Display Name		None	None	None	None	None	None	None
User Password Protection	Both for incoming and outgoing calls	D	D	D	D	D	D	D
Remote Extension service		D	D	D	D	D	D	D
User's name for Dial by Name Directory		Undefined	Undefined	Undefined	Undefined	Undefined	Undefined	Undefined
Basic Services – General Settings								
No Answer Timeout		20 sec.	20 sec.	20 sec.	20 sec.	20 sec.	20 sec.	20 sec.
Call Waiting service		E	E	E	E	E	E	E
Redial Interval		10 sec.	10 sec.	10 sec.	10 sec.	10 sec.	10 sec.	10 sec.
Redial Period		15 min.	15 min.	15 min.	15 min.	15 min.	15 min.	15 min.
Basic Services – Hold Music Settings								
Send Hold Music to Remote IP Party		D	D	D	D	D	D	D
Listen Hold Music		Own_Music	Own_Music	Own_Music	Own_Music	Own_Music	Own_Music	Own_Music
Hold Music		SDM	SDM	SDM	SDM	SDM	SDM	SDM
Basic Services – Do Not Disturb Settings								
Actual Status		SNA	SNA	SNA	SNA	SNA	SNA	SNA
Expires after		30 min	30 min	30 min	30 min	30 min	30 min	30 min
Send Message to Caller		E	E	E	E	E	E	E

14 References

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

- Manual-I: Installation Guide for QX IP PBXs
- Manual-III: User Guide for QX IP PBXs
- System Capacity of QX IP PBXs
- QX IP PBX Features on Epygi Supported IP Phones
- Licensable Features on QX IP PBXs
- Language Packs Overview for Epygi QX Line
- Audio-Video Conferencing on QX IP PBXs
- Receptionist Service on QX IP PBXs
- QX IP PBX Remote Extension Configuration
- Extensions Bulk Import on QX IP PBXs
- Auto Configuration of Epygi Supported IP Phones using OpenVPN
- Call Detail Records on QX IP PBXs
- Firmware Update Service on Epygi QX Line
- DCC – User Guide

Find the above listed documents on Epygi Support Portal.

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