



Hidden Pages for QX Advanced Configuration

Abstract: This document describes the frequently used hidden web pages available on QX units. It is intended to help administrators to identify and resolve the issues caused by inappropriate configuration of hidden settings.

Table of Contents:

1	Introduction	4
2	List of Hidden Pages	4
3	Accessing Hidden Pages	5
3.1	Advanced	5
3.2	Callcontrol	5
3.3	Codecsconfig	7
3.4	Extmanagement	8
3.5	Fxocfg	11
3.6	Fxoconfig	13
3.7	Fxodisconnect	15
3.8	Generalconfig	16
3.9	Lantable	18
3.10	Linesconfig	19
3.11	Netcapture	20
3.12	Networkphycfg	21
3.13	Trunkecho	21
3.14	Tos	22

Document Revision History

Revision	Date	Description	Valid for SW	Valid for models
1.0	19-Feb-15	Initial Version	6.0.x and higher	All QX products

1 Introduction

Along with the regular web management pages available for the administrator, the Epygi QXs contain a number of hidden pages that allow advanced configurations in the system. The default system settings will work for most applications but they can be changed if required. Some hidden pages also provide access to troubleshooting tools to collect additional information.

This document describes the frequently used hidden pages available on the Epygi QX systems, provides explanations of the most commonly used options and parameters to help administrators identify and resolve issues that cannot be resolved through the QXs documentations.

Please Note: The hidden pages contain low-level system settings and are mainly intended for Epygi's technical support engineers and advanced distributors for fine-tuning the system in some specific configuration scenarios. Wrong or inappropriate configuration of these settings may result in undesired system behavior. It is strongly recommended to leave the default settings if the meanings are not fully understood.

2 List of Hidden Pages

The list of frequently used hidden pages with their brief descriptions is presented in the Table 1.

	Pages	Description
1.	advanced	Allows switching off the software watchdog and changing the default http/https server ports.
2.	callcontrol	Allows making auto-generated call on the IP PBX
3.	codecsconfig	Allows configuring voice codec settings.
4.	extmenegement	Allows configuring certain advanced settings for extensions and attendants.
5.	fxocfg	Allows adjusting FXO general configurations.
6.	fxoconfig	Allows configuring FXO lines' settings.
7.	fxodisconnect	Allows selecting one of the five basic scenarios for call disconnection.
8.	generalconfig	Allows configuring the system general settings.
9.	lantable	Allows packet traffic prioritization based on IP address.
10.	linesconfig	Allows configuring FXS lines.
11.	netcapture	Allows making packet captures from a specified network interface.
12.	networkphycfg	Allows configuring speed and mode parameters for LAN and WAN interfaces
13.	trunkecho	Allows configuring echo settings of ISDN, E1/T1 trunks.

	Pages	Description
14.	tos	Allows specifying the value for the Type of Service setting (ToS) for SIP and RTP packets.

Table 1: List of hidden pages

3 Accessing Hidden Pages

To access a hidden page, log into the system as an administrator and type the CGI name into the address field of the browser:

`http://QX_hostname_or_IP/hidden_page`

For example: <http://192.168.74.12/foxcfg.cgi>

3.1 Advanced

The **Advanced System Setup** page allows switching off the software watchdog and specifying custom listening ports for HTTP and HTTPS.

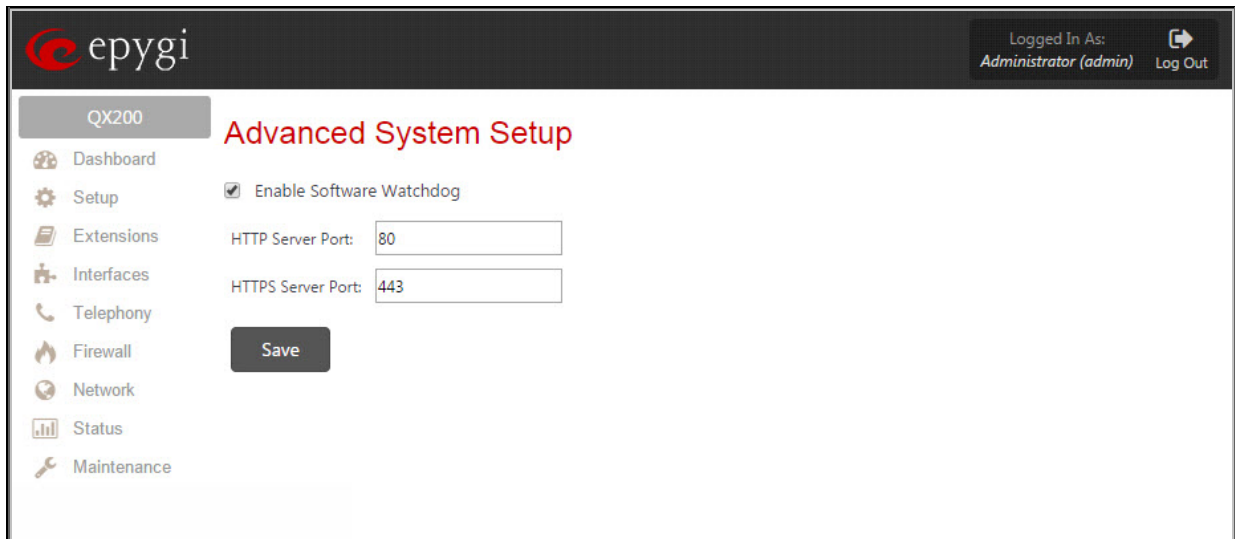


Figure 1: Advanced System Setup page

- Enables Software Watchdog:** The QX software watchdog is an application that checks the activity of sub-systems that are vital for the proper operation of the whole system and restarts the voice system or reboots the whole system if a sub-system has failed. The activity checking is executed every five minutes. The following sub-systems are monitored by the watchdog: *voice, vpd, ipsecd, stats, tftpd, dhclient, dhcpd, named, ntpd, ntp, httpd*.
 - HTTP Server Port:** Listening port for HTTP.
 - HTTPS Server Port:** Listening port for HTTPS.

3.2 Callcontrol

The **callcontrol.cgi** page allows activating a call from a WEB, connecting two parties together via auto-generated call (see Figure 2). This feature is useful for remote diagnostic purposes.

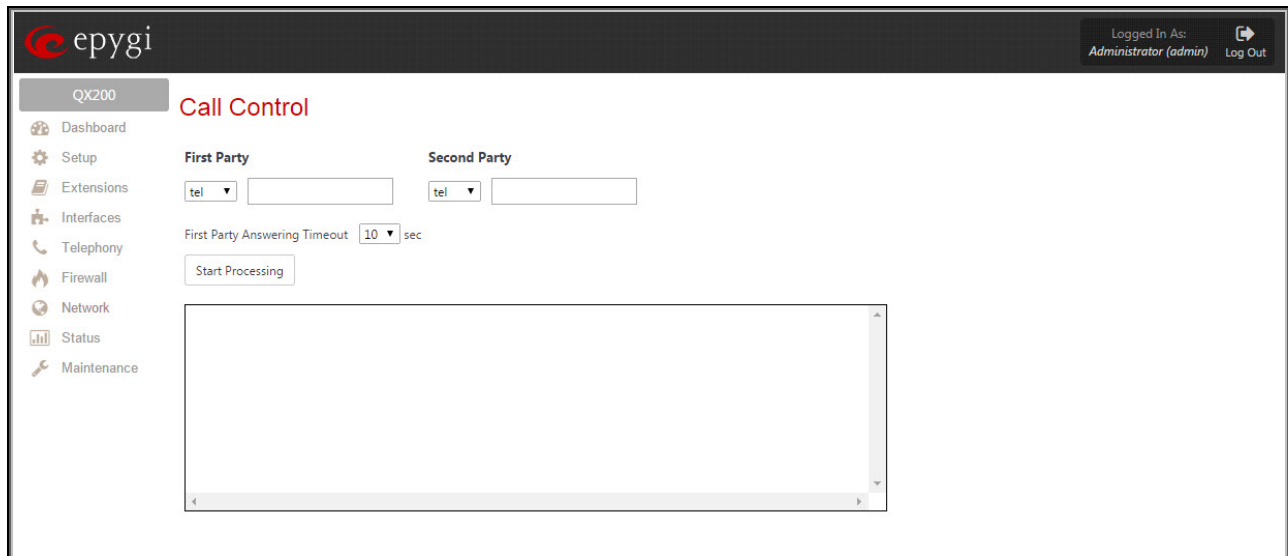


Figure 2: Call Control page

- **First Party** text field requires the address for the first party of the auto-generated call.
- **Second Party** text field requires the address for the second party of the auto-generated call.
- **First Party Answering Timeout** drop down list requires the time period during which the first party has to answer the call in order to transfer it to the second party.
 - **Start Processing** button is used to initiate the auto-generated call.

The options of the **call type** drop down list with descriptions and examples for both first and second parties are the following:

Call Type	Definition	Example
Tel	Used for the calls via the call routing table	Type 820225 to call to the 20225 number via call routing
Sip	Used when called destination reached through a SIP server	Type 20225@sip.epygi.com to call to the extension 20225 registered on sip.epygi.com server
PSTN	Used when called destination dialed to PSTN	Type 0123456789 to call to the 0123456789 PSTN number via FXO, ISDN or E1/T1
PBX	Used for local calls to a PBX extensions	Type 103 to call to the extension 103

Table 2: Options of the Call Type drop down list

How it works:

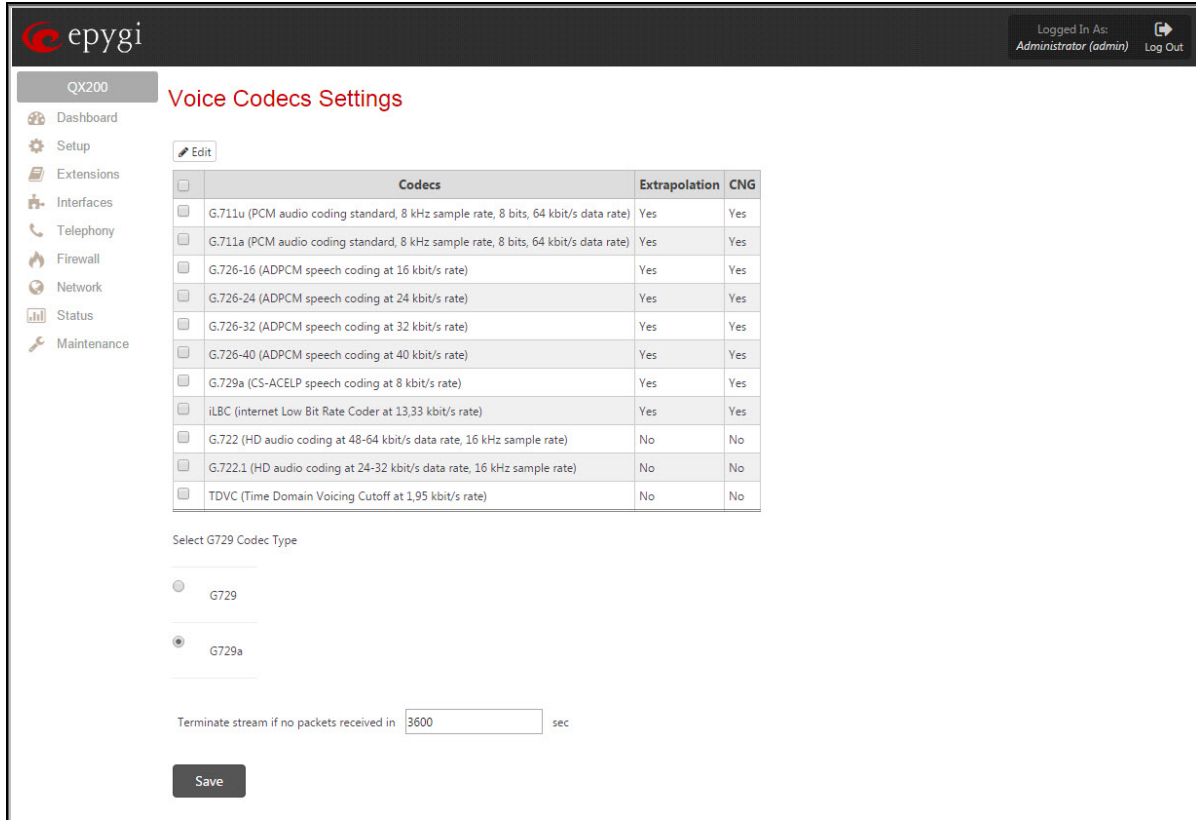
The user enters the addresses for the first and second parties and presses the **Start Processing** button.

The call setup takes the following steps: QX makes a call to the “First Party” (QX extension or a remote destination). As soon as the extension answers the call, QX plays a message, then transfers the call to the second party (remote user) and connects the call between QX extension and remote user.

In case if the first party does not answer within the specified timeout, QX shows an error in the progress text box and stops processing the call.

3.3 Codecsconfig

The **Voice Codecs Settings** page enables/disables the CNG and Extrapolation options for voice codecs and configuring the below listed settings.



The screenshot shows the 'Voice Codecs Settings' page in the epygi QX200 interface. The page includes a sidebar with navigation options like Dashboard, Setup, Extensions, Interfaces, Telephony, Firewall, Network, Status, and Maintenance. The main content area features a table of codecs and their settings.

<input type="checkbox"/>	Codecs	Extrapolation	CNG
<input type="checkbox"/>	G.711u (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)	Yes	Yes
<input type="checkbox"/>	G.711a (PCM audio coding standard, 8 kHz sample rate, 8 bits, 64 kbit/s data rate)	Yes	Yes
<input type="checkbox"/>	G.726-16 (ADPCM speech coding at 16 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	G.726-24 (ADPCM speech coding at 24 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	G.726-32 (ADPCM speech coding at 32 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	G.726-40 (ADPCM speech coding at 40 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	G.729a (CS-ACELP speech coding at 8 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	iLBC (internet Low Bit Rate Coder at 13.33 kbit/s rate)	Yes	Yes
<input type="checkbox"/>	G.722 (HD audio coding at 48-64 kbit/s data rate, 16 kHz sample rate)	No	No
<input type="checkbox"/>	G.722.1 (HD audio coding at 24-32 kbit/s data rate, 16 kHz sample rate)	No	No
<input type="checkbox"/>	TDVC (Time Domain Voicing Cutoff at 1.95 kbit/s rate)	No	No

Below the table, there is a section for 'Select G729 Codec Type' with two radio buttons: 'G729' and 'G729a' (which is selected). Below this is a text input field labeled 'Terminate stream if no packets received in' with the value '3600' and a unit 'sec'. A 'Save' button is located at the bottom of the page.

Figure 3: Voice Codecs Settings page

- **Select G729 Codec Type:** allows choosing between the G729 codec types.
- **Terminate stream if no packets received in:** Time (in seconds) after which the connection with the remote endpoint will be closed if no packet is received.
- **CNG (Comfort Noise Generation):** A technique used in codecs to generate an artificial noise if no voice packet or a CN packet is received from the remote party. With typical PSTN networks people are accustomed to hearing a very low level of background noise on a call. Without Comfort Noise packets being locally generated by the QX or far-end there would be complete silence on the call when a person is not speaking and the other party would think the call was disconnected.
- **Extrapolation:** A technique used in codecs to recover a lost or delayed RTP packet to provide a seamless voice experience to the user.
- To disable the **CNG** or **Extrapolation** option for a codec, select the codec and click on **Edit** (see Figure 4). Clear the checkbox for the appropriate option and click on **Save**.

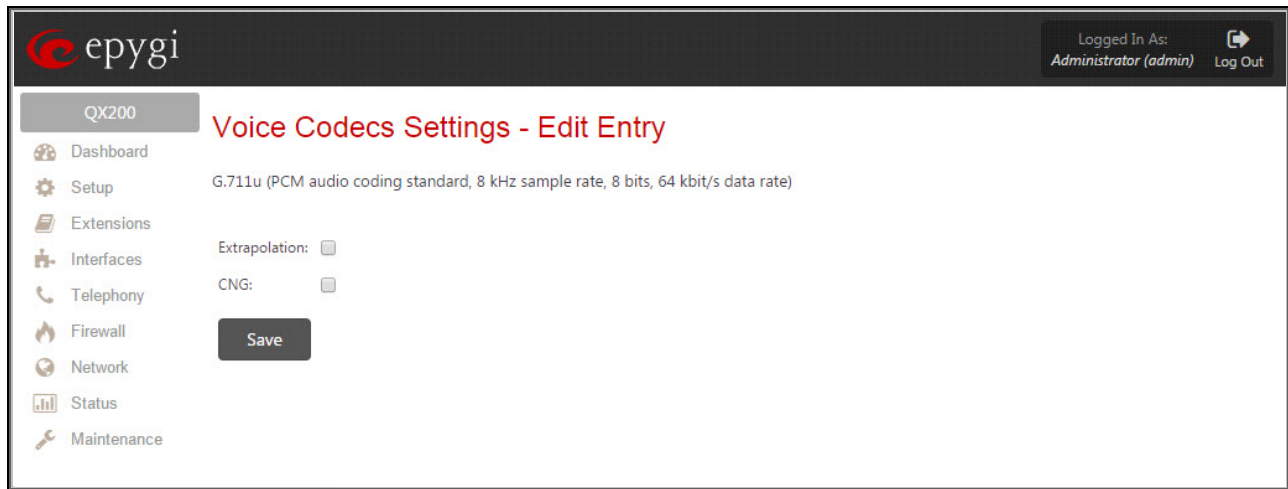


Figure 4: Voice Codec Settings - Edit Entry page

3.4 Extmanagement

The **Extensions Management** page allows configuring certain advanced settings for extensions and attendants.

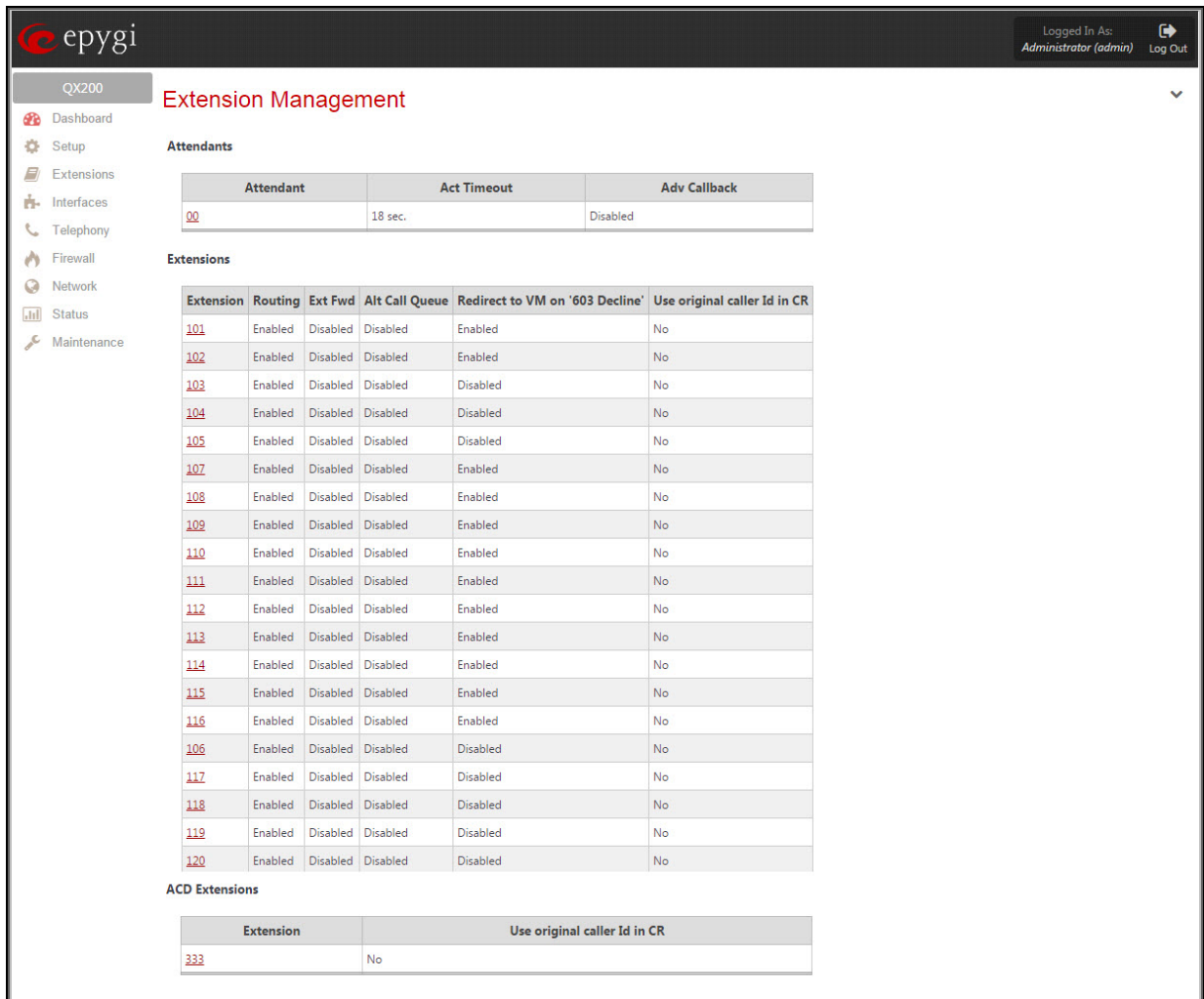


Figure 5: Extension Management page

The below listed setting can be edited through this hidden page for attendants.



Figure 6: Extension Management page for attendant 00

- **Callback Activation Timeout:** The time range within which the trusted users may interrupt the call to auto attendant in order to initiate the callback (18 sec is the default).
 - **Enable Advanced Callback Capability (experimental):** If enabled, all incoming SIP calls received to the selected attendant will be required to pass authentication on the RADIUS server the QX is configured with. If authentication is successful, the system will disconnect the SIP call and call back the caller via ITSP using a corresponding call routing rule. The scenario that this setting uses assumes that incoming SIP calls are received from another QX (typically a QX gateway), which prior to making a SIP call, substitutes the originator's SIP username by the caller's PSTN number. If authentication fails, the system will disconnect the call without calling back. If disabled, incoming calls to the attendant will be handled in the usual way according to the system configuration.

Please Note: This setting is available in QX IP PBXs and is used in a scenario that allows initiating a long-distance PSTN call from a remote QX IP PBX by a local PSTN call to a QX Gateway.

The below listed settings can be edited through the **Extensions Management** hidden page.

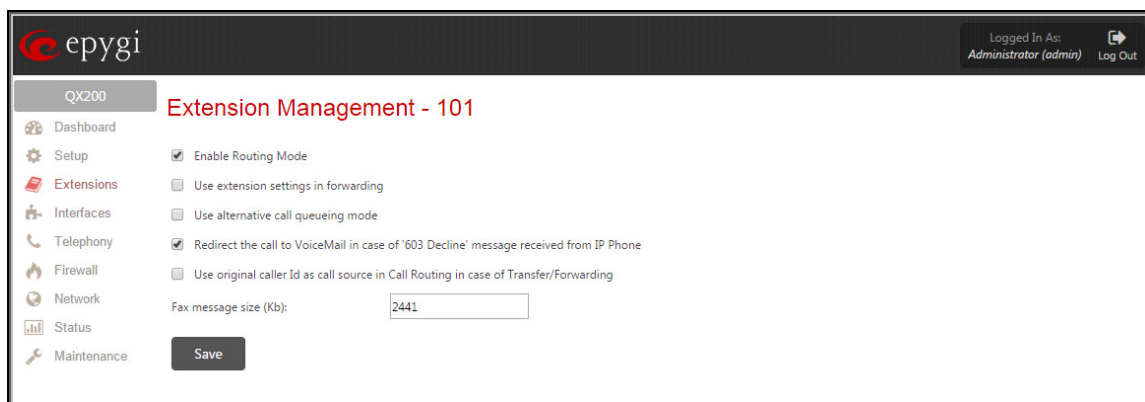


Figure 7: Extension Management page for extension 101

- **Enable Routing Mode:** Indicates whether calls initiated from the selected extension should be made using the rules in the **Call Routing** table. If disabled, the extension will only be able to make PBX calls to extensions, PSTN calls and SIP calls through a server. If enabled, calls initiated from the extension will be routed to the call routing table where the best matching pattern will be used to place the call.

Please Note: This option should not be disabled so that the extension can use all the patterns in the call routing table.

- **Use extension settings in forwarding:** Indicates whether CallerIDs of SIP calls received on the selected extension and forwarded to another SIP destination will be modified to contain the SIP username of the extension that has forwarded the call. If enabled, the callerIDs of forwarded SIP calls will contain the call originator's display name and the SIP URL of the extension that has forwarded the call. If disabled, the originator's callerID will be sent to the forwarded destination without any modification.
- **Use alternative call queuing mode:** If this checkbox is enabled, the call queue message will be played simultaneously to all callers in the queue. If disabled, the call queue message will be played to callers at regular intervals.

Please Note: Do not enable this checkbox, it will break incoming calls' handling of the extension.

- **Redirect the call to VoiceMail in case of '603 Decline' message received from IP Phone:** When the checkbox is enabled, and IP phone replies '603 Decline' for incoming call (user presses 'Reject' button), the call is being redirected to the Voice Mail of that extension. If the checkbox is disabled, then the call is being rejected in that scenario. Note: not all IP phones send '603 Decline' upon pressing of 'Reject' – this is implementation specific feature.
- **Use original caller Id as call source in Call Routing in case of Transfer/Forwarding:** When the checkbox is enabled, the CM uses original caller's Id for passing to the Call Routing upon resolving of call destination (call source filtering works on original caller Id). If the checkbox is disabled (default), in such scenarios caller Id of transferor/forwarder extension is being used. This feature was added for some very specific setups, when filtering in the CR should be done based on caller's Id.
- **Fax message size (kb):** Size of system RAM to be allocated for temporary storage of an incoming fax. Once the fax receipt is complete, it will be stored in the system memory assigned to the extension as a tiff file. If an incoming fax requires larger storage space, the system will reject part of the fax that does not fit into the specified space.

To edit any of the advanced settings using the **Extensions Management** page, click on the link associated with the corresponding extension or attendant.

3.5 Fxocfg

The **Fxo Configuration** page allows configuring the FXO port settings to match the characteristics of PSTN lines.

The below settings are found on the **Fxo Configuration** page (see Figure 8):

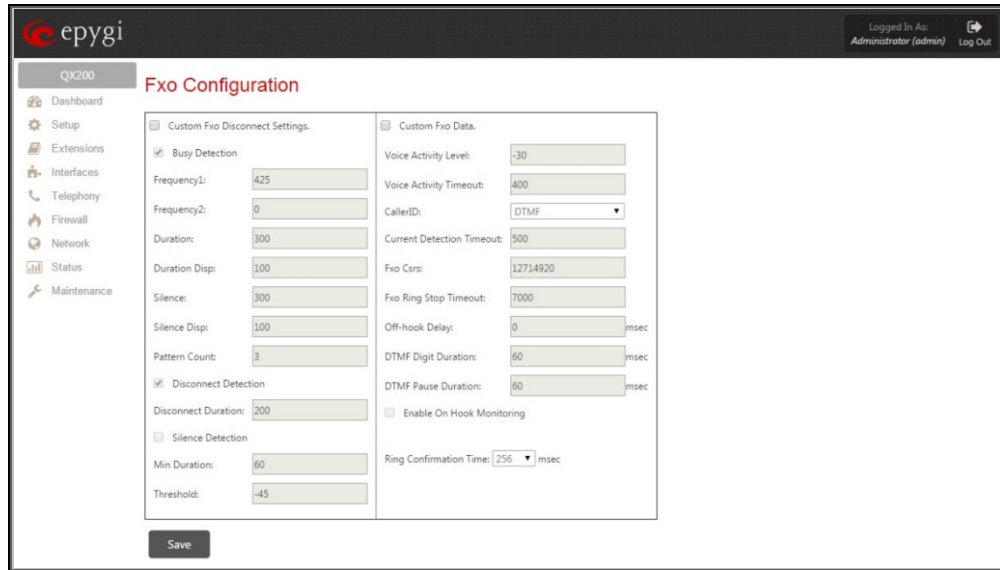


Figure 8: FXO Configuration page

Custom Fxo Disconnect Settings: Enables manual configuration of disconnect-call detection settings. Central offices (COs) typically use any or a mixture of the following three methods to notify call disconnection:

- **Disconnect notification by sending busy/disconnect tones.** The CO sends specific tones to notify that the call party has hung up. To configure the QX to detect disconnect tones and close the call upon detection, enable the **Busy Detection** checkbox and specify the following settings:
 - **Frequency 1:** Frequency of the disconnect tone used by CO. If the CO sends dual tone signals, enter the high frequency of the tone into the **Frequency 1** text field and the low frequency into the **Frequency 2** text field. In case of single tone signals, set the **Frequency 2** to **0**.
 - **Duration:** Tone duration in ms.
 - **Duration Disp:** Amount of time (ms) that the tone signal may vary from its assigned value. It should be chosen reasonably large taking into account the value of **Duration**. By default 100 ms is recommended.
 - **Silence and Silence Disp:** Silence duration and the amount (in ms) that it may vary respectively.
 - **Pattern Count:** Number of tone(s)-silence patterns the system should identify as a disconnect notification. Leave the factory default value if the exact number is unavailable. If the QX detects signals with above-specified settings, it will close the call.
- **Disconnect notification by power interruption.** The CO interrupts the power on the line to notify that the call party has hung up. To configure the QX to detect power interruption, enable the **Disconnect Detection** checkbox. Specify the **Disconnect Duration** setting, which is the minimum duration (ms) of power interruption to be identified as disconnect notification. If the QX does not detect the power for the time specified by the **Disconnect Duration** setting, it will close the call.

- **Disconnect notification by putting silence on the line.** The CO puts silence on the line to notify that the call party has hung up. To configure the QX to close the call upon detecting silence, enable the **Silence Detection** checkbox and specify the following settings:
- **Min Duration:** Minimum duration of silence the system should identify as a disconnect notification.
- **Threshold:** Highest audio level that the system should identify as silence.
- If the system detects audio lower than the specified threshold for more than the specified duration, it will close the call.

The **Custom Fxo Data** checkbox enables the manual configuration of different FXO related settings.

- **Voice Activity Level:** Lowest audio level (dial tone) the system should detect on an FXO line to know that the line is working. The activity detection takes place each time the system takes the FXO line off hook to make an outgoing call.
- **Voice Activity Timeout:** Maximum length of time that the system will wait for the voice activity on an FXO line. If the system does not detect the required voice activity level for the specified length of time, it assumes that that the PSTN line is not working (or no PSTN line is attached to the FXO port) and plays the notification message, “Check your external phone line.”

Please Note: The **Voice Activity Level** and **Voice Activity Timeout** settings take effect only if the **Wait for dialtone** checkbox in the `fxoconfig.cgi` is enabled for the corresponding FXO line (see section [3.6](#)).

- **Caller ID:** Different standards of Caller ID transmissions. The standard appropriate to the local telephony company should be chosen from this list.

Please Note: The selected standard takes effect only if the **Caller ID** setting for the corresponding FXO line is set to **Default** in the `fxoconfig.cgi`. Otherwise, the standard selected in `fxoconfig.cgi` will be used.

- **Current Detection Timeout:** Maximum length of time that the system will wait for the current on an FXO line to know that the line is working. On call origination, if the system does not detect the current within this timeout, it assumes that the PSTN line is not working (or no PSTN line is attached to the FXO port) and plays the notification message, “Check your external phone line.”

Please Note: The **Current Detection Timeout** setting takes effect only if the **Current Detection** checkbox in the `fxoconfig.cgi` is enabled for the FXO line (see section [3.6](#)).

- **Fxo Csrs:** Country-specific regional setting that should correspond to the line settings of the local telephony company. The value for this setting is set according to the regional settings selected in the System Configuration wizard. Please do not modify this setting unless instructed to do so by Epygi’s technical support engineers.
- **Fxo Ring Stop Timeout:** Period of time after which the system will validate ringing termination if no new ring pulse is received from CO. The value for this setting should be slightly greater than the time between two successive ringing signals sent by CO. Note that too small value for this setting will cause the system to drop the call before the next ringing signal arrives. Too large of a value will cause the system to ring the phone long after the remote party has hung up, i.e. if the remote party hangs up before this timeout expires, the local user will hear busy tones when answering the ringing phone.
- **Off-hook Delay:** Delay time (in milliseconds) for incoming calls on FXO line to avoid the ring tone playing with some providers.
- **Enable On Hook Monitoring:** Enables on-hook monitoring of Caller ID signals. If selected, the system will continuously monitor FXO lines to detect Caller ID signals that are not preceded by

a notification signal. If this checkbox is not selected, the system will detect only Caller IDs that are transmitted after a notification signal.

Ring Confirmation Time: Minimum length of the ringing signal (in ms) required for the system to validate an incoming call. Ringing signals shorter than the specified time will be ignored by the system.

All settings found on this page are automatically adjusted to the regional settings set in the **System Configuration Wizard**. It is strongly recommended to leave the settings with their default values unless the characteristics of a specific PSTN line differ from those predefined for the region.

Please Note: For the instructions on how to determine the correct values for FXO settings appropriate to your CO, please see the troubleshooting guide **FXO Diagnostics**, which is available in the Support Portal of the www.epygi.com.

3.6 Fxoconfig

The **FXO Settings** page allows configuring each FXO line (see Figure 9). As opposed to the `fxocfg.cgi` responsible for all FXO lines' common settings, the `fxoconfig.cgi` allows distinct configuration of each FXO line.

FXO Lines	Enabled	Allowed Call Type	Route Incoming Call to
FXO.1	No	N/A	N/A
FXO.2	Yes	Both incoming and outgoing calls	00
FXO.3	Yes	Both incoming and outgoing calls	Routing
FXO.4	Yes	Both incoming and outgoing calls	00

Figure 9: FXO Settings page

- To modify the settings of an FXO line, click on the link associated with the line on the **FXO Settings** page. This will open the **FXO Settings** page for the selected line and enable to configure the below listed settings.

Enabled	Pattern Name	Ring Duration	Pause Duration	Route Pattern
<input type="checkbox"/>	duet	Min 150 -250 Max	Min 350 -450 Max	
<input type="checkbox"/>	trio	Min 150 -250 Max	Min 350 -450 Max	

Figure 10: FXO Settings page for FXO 2

Enable FXO: Enables the corresponding FXO line.

- **Wait for dialtone:** Enables checking the availability of the connected PSTN line by detecting voice activity (dial tone) on the line. If enabled, the system will proceed with placing an outgoing PSTN call only if the required voice activity level is detected on the line. The voice activity and the timeout to wait for it are set in the `fxocfg.cgi`. If disabled, the system will send DTMFs regardless of whether or not the PSTN line is working.
- **DSP Echo Cancellation:** Enables the DSP echo canceller for the FXO line. The echo canceller uses adaptive linear filtering for linear processing of the echo that can be followed by non-linear processing depending on the residual signal level.
 - **Echo Tail:** Maximum anticipated length of echo signals. The setting value is expressed in samples with the sampling rate of 8000 samples/sec (8 samples = 1 ms) and is used in the adaptive liner filter.
 - **Echo NLP Diff:** Minimum value for the **Echo NLP Diff** setting, which is the difference in echo level before and after linear processing. If the difference in echo level is greater than the specified value, the echo signals will be further processed by the non-linear processor to remove the residual echo signal.
 - **Echo NLP Shift:** Number of times to shift right the residual signal to decrease the echo signal level. This is equivalent to multiplying the specified number by 6 dB and deducting it from the residual signal strength. Note that this setting is used only if the difference in echo level is greater than the value specified by the **Echo NLP Diff** setting.

Please Note: For the instructions on how to determine the correct values for echo parameters appropriate to your QX setup, please see the troubleshooting guide **Echo Cancellation on QX**, which is available in the Support Portal of the www.epygi.com.

- **Current Detection:** Enables checking the availability of the connected PSTN line by detecting current on the line. If enabled, the system will proceed with placing an outgoing PSTN call only if the required current is detected on the line. The timeout to wait for the current is set in the `fxocfg.cgi`. If disabled, the system will send DTMFs regardless of whether or not the PSTN line is working.
- **Enable Parallel Handset Detection:** This setting is used when a PSTN line is shared between an FXO port and a parallel-connected phone(s). The checkbox allows the system to detect whether a parallel-connected phone is using the line and prevent placing an outgoing call if the line is in use. If enabled, the system will place an outgoing PSTN call only if the line is not in use. If disabled, the system will place an outgoing PSTN call if the level of the current on the line is sufficient for making a call.
- **Enable Pulse Dialing:** Enables pulse dialing on the line. If enabled, the dialed number will be sent to the PSTN using pulse signaling. If disabled, tone signaling will be used.
- **Wait for Next Ring Before Accept:** Clicking this checkbox enables the FXO agent to go offhook only when port is ringing. It intends to minimize the risk of going offhook when the remote side already dropped the call.

Please Note: Regardless of the dialing method selected for the FXO line, numbers from devices connected to QX FXS ports (phones, modems or other terminal equipment) should be dialed using DTMF dialing.

- **Caller ID:** This drop-down list allows selecting the standard of Caller ID transmission used on the PSTN line. The *default* option for this setting configures the system to use the standard selected in the `fxocfg.cgi`.

- Route Patterns:** This table is used to route incoming FXO calls whose ringing signals match a certain pattern to a specified call routing record. (The duet pattern is *ring-short pause-ring-long pause*; the trio pattern is *ring-short pause-ring-short pause-ring-long pause*). The capability of routing calls to different destinations based on the ringing pattern can be used, for example, to separate fax calls from ordinary phone calls.

3.7 Fxodisconnect

The **Fxo Disconnect Scenarios** page allows selecting one of the five basic scenarios for the disconnect-call notification methods. Each scenario is a set of prescribed values for busy tone parameters and duration of power interruption used by COs in specific countries (see Table 3).

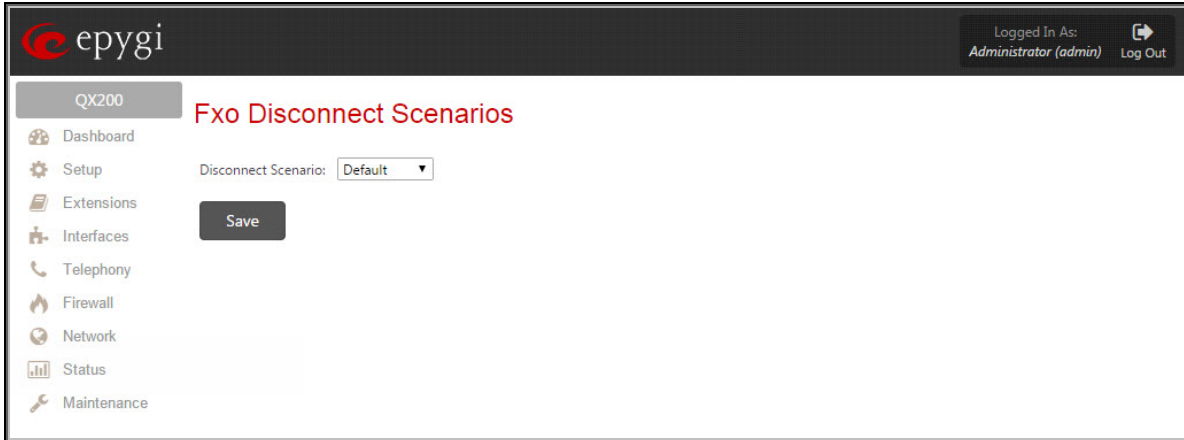


Figure 11: Fxo Disconnect Scenarios page

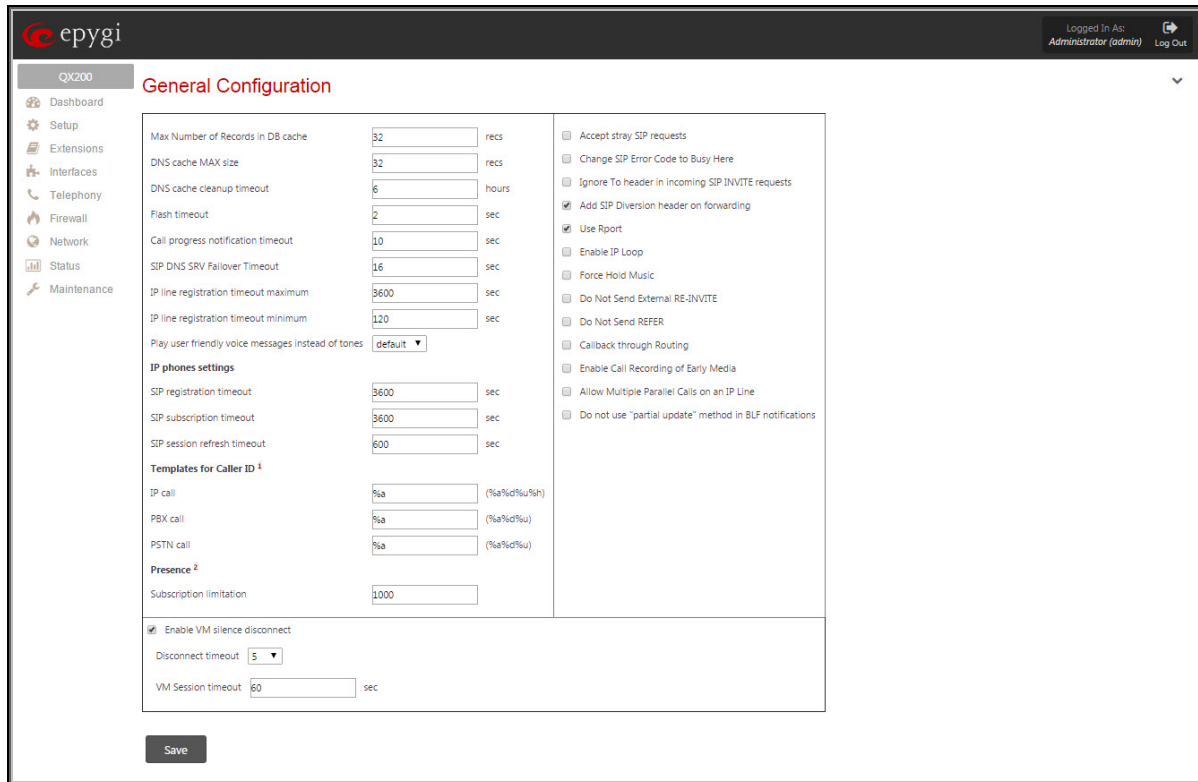
The **Disconnect Scenario** drop-down list contains the five basic scenarios for call disconnection. The **Default** option corresponds to the scenario selected based on the system regional settings. Selecting a different disconnect scenario is useful if the QX does not recognize the disconnect signals sent from the CO resulting in calls not releasing.

Settings	Scenario A	Scenario B	Scenario C	Scenario D	Scenario E
Frequency 1	425	450	425	400	480
Frequency 2	0	400	0	0	625
Duration	300	300	250	500	300
Duration Disp	100	100	100	100	100
Silence	300	300	250	500	300
Silence Disp	100	100	100	100	100
Pattern	3	3	3	3	3
Disconnect Duration	200	200	200	200	200

Table 3: Types of FXO disconnect scenarios

3.8 Generalconfig

The **General Configuration** page allows configuring the system’s general settings.



The screenshot shows the 'General Configuration' page in the epygi web interface. The page is titled 'General Configuration' and has a 'Save' button at the bottom. The settings are organized into several sections:

- Cache and Cleanup:** Max Number of Records in DB cache (32), DNS cache MAX size (32), DNS cache cleanup timeout (6 hours), Flash timeout (2 sec).
- Call Progress:** Call progress notification timeout (10 sec).
- SIP DNS SRV Failover:** SIP DNS SRV Failover Timeout (16 sec).
- IP Line Registration:** IP line registration timeout maximum (3600 sec), IP line registration timeout minimum (120 sec).
- Play user friendly voice messages instead of tones:** default (dropdown).
- IP phones settings:** SIP registration timeout (3600 sec), SIP subscription timeout (3600 sec), SIP session refresh timeout (600 sec).
- Templates for Caller ID:** IP call (%a), PBX call (%a), PSTN call (%a).
- Presence:** Subscription limitation (1000).
- VM Silence Disconnect:** Enable VM silence disconnect (checked), Disconnect timeout (5), VM Session timeout (60 sec).
- Other Settings:** Checkboxes for 'Accept stray SIP requests', 'Change SIP Error Code to Busy Here', 'Ignore To header in incoming SIP INVITE requests', 'Add SIP Diversion header on forwarding', 'Use Rport', 'Enable IP Loop', 'Force Hold Music', 'Do Not Send External RE-INVITE', 'Do Not Send REFER', 'Callback through Routing', 'Enable Call Recording of Early Media', 'Allow Multiple Parallel Calls on an IP Line', and 'Do not use "partial update" method in BLF notifications'.

Figure 12: General Configuration page

- **Max Number of Records in DB cache:** Maximum number of records in the external SIP callers’ cache. The setting was used with a feature available in older software versions.
- **DNS cache MAX size:** Maximum number of DNS records (hostnames to IP address resolutions) stored for Call Manager’s reference.
- **DNS cache cleanup timeout:** Time after which the system will refresh all records stored in the DNS cache.
- **Flash timeout:** Time that the system will wait for the input of a next digit(s) to handle the keys as a key combination (for example ‘Flash’+‘3’). If no digit is pressed within the specified timeout, the system will assume that the Flash key has been followed by the key 0. Digits pressed after the timeout expiration will be ignored.
- **Call progress notification timeout:** Period of time after which the system will play the message “Trying to connect, please wait” if no response has been received from the remote endpoint when establishing a SIP call.
- **SIP DNS SRV Failover Timeout:** The expiration timeout after which the system will try to reach the next resolved SIP DNS server if available.
- **IP line registration timeout:** Registration timeout to be included in configuration files of supported IP phones generated by the QX. A registration timeout is the time after which an IP phone will try to re-register on the QX.
- **IP line registration timeout minimum:** It is the minimum registration timeout which is supported by the QX. If some device tries to register on the QX with the less registration timeout, QX will reject that registration with “Too Brief” SIP error message.
- **Play user friendly voice messages instead of tones:** Response type (system voice messages or busy tones) to receive from the system if it encounters a problem while executing user-

initiated operations (e.g., the user tries to dial out when all FXO lines are in use). If the *default* or *always* options are selected, the system will play a voice message to notify the user that a problem occurred. When the *never* option is selected, the system will send busy tones.

- **Templates for CallerID:** The text boxes allow specifying which component(s) of the received caller ID the system should send to the called extension's phone. The following options for forming a caller ID pattern are available:

%a (automatic) – send the caller ID as provided by the calling party's service provider.

%d (display name) – send the calling party's display name.

%u (username) – send the calling party's SIP or PSTN number.

%h (hostname, available for SIP calls only) – send the hostname of the calling party's SIP URL.

Please Note: These templates for CallerID are effective for FXS lines only. The **IP call**, **PBX call**, **PSTN call** text boxes are used to assign different caller ID patterns to different call types. For example, to configure the system to send the display name and host name for incoming SIP calls, enter `%d%h` into the **IP call** text box.

- **Presence:** This section allows to specify the count of subscriptions which can be supported by the QX.
- **Subscription limitation:** This text box allows to specify the number of subscriptions which can be supported by the QX. Increasing this value can bring instability of QX.
- **Accept stray SIP requests:** Indicates whether the system should accept SIP requests without checking the presence of the destination URI in the sip user agent.
- **Change SIP Error Code to Busy Here:** Indicates whether the system should substitute SIP error code 480 (Temporary Unavailable) with code 486 (Busy Here). As responses to SIP error 480 may be implemented differently depending on the SIP device, error code substitution is useful to ensure identical responses from different SIP devices.
- **Ignore To header in incoming SIP INVITE requests:** When this feature is enabled and in the incoming SIP "INVITE" message "To" header and "Request Line" contains different information, QX will ignore information provided in the "To" header of SIP "INVITE" message and will use the information which is provided in the "Request Line" of the same SIP "INVITE" message.
- **Use Rport:** Indicates whether the system should add an empty **Rport** to SIP packets. The **Rport** parameter allows the sender (the QX) to request proxy servers to send the responses to a particular address and port (the originating address and port) if it supports the feature.
- **Enable IP Loop:** Indicates whether the system should make a SIP call when extension-to-extension calls are made using extensions' SIP usernames. If disabled, the system will make a PBX call in the mentioned scenario. This checkbox is designed for testing purposes only. Please do not enable this checkbox unless instructed to do so by Epygi's technical support engineers.
- **Force Hold Music:** Forces QX's music on hold (MOH) to be sent to the remote party when the QX is behind an IP/PSTN provider that does not properly support remote MOH. If the checkbox is disabled, the QX will send an SIP re-INVITE (HOLD) message with the `c=IN IP4 0.0.0.0, a=sendonly` media attributes to notify the server that the local party has placed the remote party on hold. In this case, some servers completely close the received audio stream resulting in the remote party not hearing the QX's MOH. If the checkbox is enabled, the QX will send an SIP re-INVITE message with active media attributes, leaving local IP (`c=`) address and media stream direction (`a=`) unchanged, forcing the server to send QX's MOH to the remote party.

- **Do Not Send External RE-INVITE:** When this feature is enabled QX will not send external “RE-INVITE” message to the remote party. This feature can be used for solving Hold/Un-Hold, Transfer and similar problems.
- **Do Not Send REFER:** When this feature is enabled QX will not send external “REFER” message to the remote party. This feature can be used when the remote party doesn’t support SIP “REFER” message.
- **Callback through Routing:** This feature allows to make a Call Back (*2 feature code) through the Call Routing table. This feature is relevant to use, when you want the system depending on the caller ID to decide how to make the Call Back. If you do not have proper routing for call back destination, the Call Back feature will not work.
- **Enable Call Recording of Early Media:** If the extension is making an external call to SIP or E1 destination and the sip server replies with **Session Progress** or the **Generate progress tone to IP** checkbox is enabled on the E1 settings, the corresponding call recording will be divided into two different recordings in the recording box. One of the recordings will present the ring-back tones (Session Progress part) and the second contains the audio of the call after the call establishment. The **Enable Call Recording of Early Media** is disabled by default.
- **Allow Multiple Parallel Calls on an IP Line:** This feature allows to receive a second incoming call from an IP phone without holding the first call. The **Allow Multiple Parallel Calls on an IP Line** checkbox is available only for QX1000
- **Do not use "partial update" method in BLF notifications** checkbox is available only for QuadroM32x.
- **Enable VM silence disconnect:** This feature allows to stop the Voice Mail recording when QX receives silence from remote party and close that call.
- **Disconnect timeout:** This drop-down list allows to specify the silence timeout, after which Voice Mail recording will be automatically stopped and that call will be closed.

3.9 Lantable

By default, all traffic from the LAN to WAN interfaces has ordinary priority. The **LAN Prioritization Table** allows changing the precedence of the traffic created by a particular IP device residing on the LAN from ordinary to high. Prioritizing the outbound traffic of a device can be useful, for example, in network configurations where one QX resides on the LAN of another QX. In this case the QX residing on the LAN should have higher traffic priority on the other QX to ensure better call quality.

To change the priority of a device from ordinary to high, add its IP address to the **LAN Prioritization** table (see Figure 13). Note that IP phones should not be added to the table as they are added implicitly once registered on the QX. To add an entry to the table, click the **Add** button and specify the IP address of the device (see Figure 14).

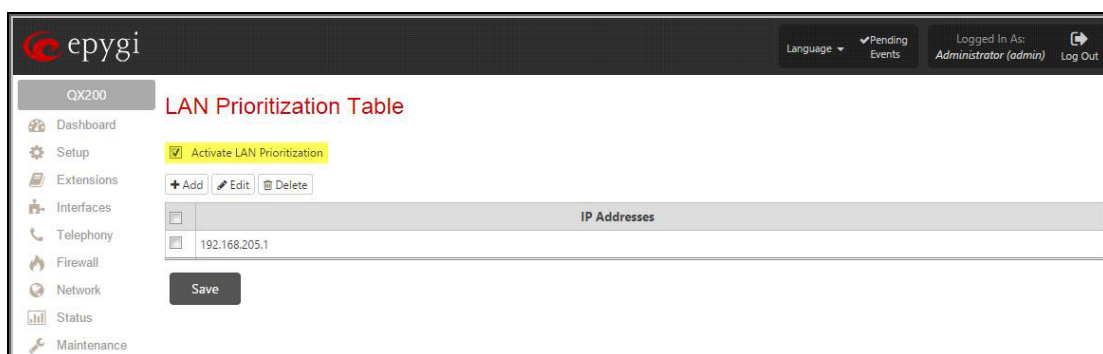


Figure 13: LAN Prioritization Table page



Figure 14: LAN Prioritization Table - Add Entry page

3.10 Linesconfig

The **Lines Configuration** page allows configuring FXS lines. Each FXS line has an associated tab on the page that should be clicked to access the line settings.

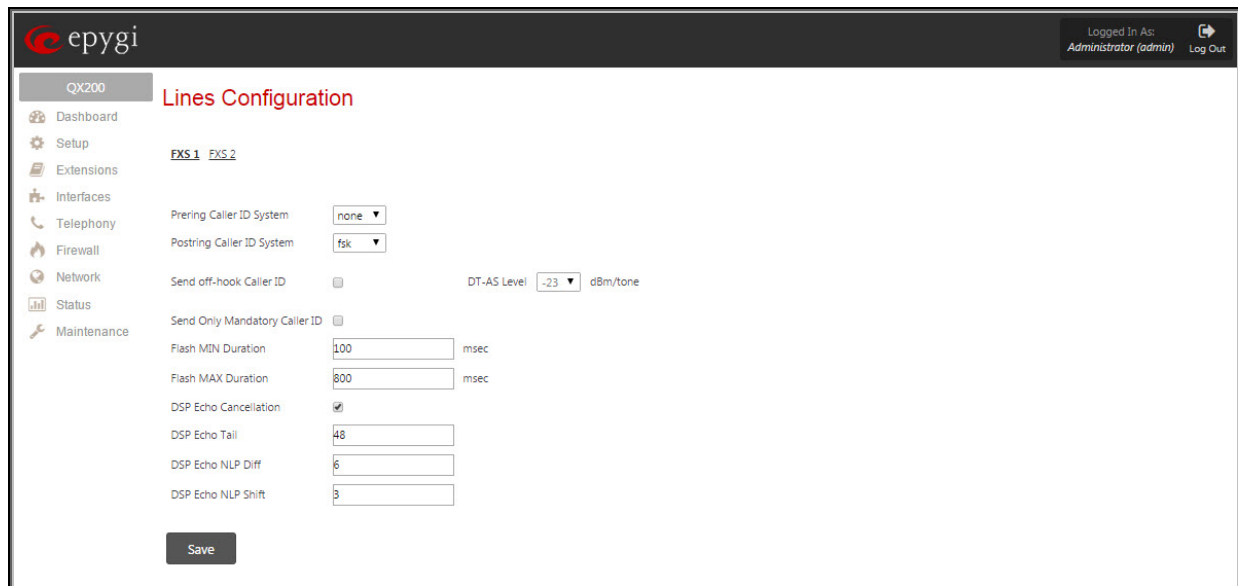


Figure 15: Lines Configuration page

The following settings can be configured for a selected FXS line:

- **Prering Caller ID System:** The standard of caller ID transmissions to be used on the line when sending the caller ID prior the first ring.
- **Postring Caller ID System:** The standard of caller ID transmissions to be used on the line when sending the caller ID between the first and second rings.
- **Send off-hook Caller ID:** Enables caller ID transmission to the FXS line when the line is in a call (provided that the call waiting feature is enabled on the extension).
- **DT-AS Level:** The level of alert signals preceding off-hook caller ID signals.
- **Send Only Mandatory Caller ID:** If selected, only certain fields of caller ID information will be sent to the called party when using the FSK system. These fields include *caller ID*, *caller name* and *time* as defined by the ETSI standard. Selecting this checkbox is useful if the called party's phone does not recognize all FSK parameters and doesn't show Caller ID on incoming calls.
- **Flash MIN Duration** and **Flash MAX Duration:** The Flash button causes the line to go on-hook for a short time then return back to the off-hook state again. The **Flash MIN Duration** and

Flash MAX Duration specify the minimum and maximum durations of keeping the line on-hook for the system to assume the Flash button is pressed (i.e. if the line goes on hook for longer than the specified **Flash MIN Duration** period of time, but shorter than the specified **Flash MAX Duration**, then the system will assume that the Flash button has been pressed). On-hook intervals shorter than the **Flash MIN Duration** will be ignored by the system. On-hook intervals longer than the **Flash MAX Duration** will be interpreted as hanging up the handset.

- **DSP Echo Cancellation:** This checkbox enables the DSP echo canceller for the FXS line. For the meaning of the **DSP Echo Tail**, **DSP Echo NLP Diff**, and **DSP Echo NLP Shift**, (see section [3.6](#)).

Please Note: The DSP echo canceller provides a better suppression of echo compared with the Codec echo canceller.

3.11 Netcapture

The **Network Capture** page allows network packets captures from a specified network interface.

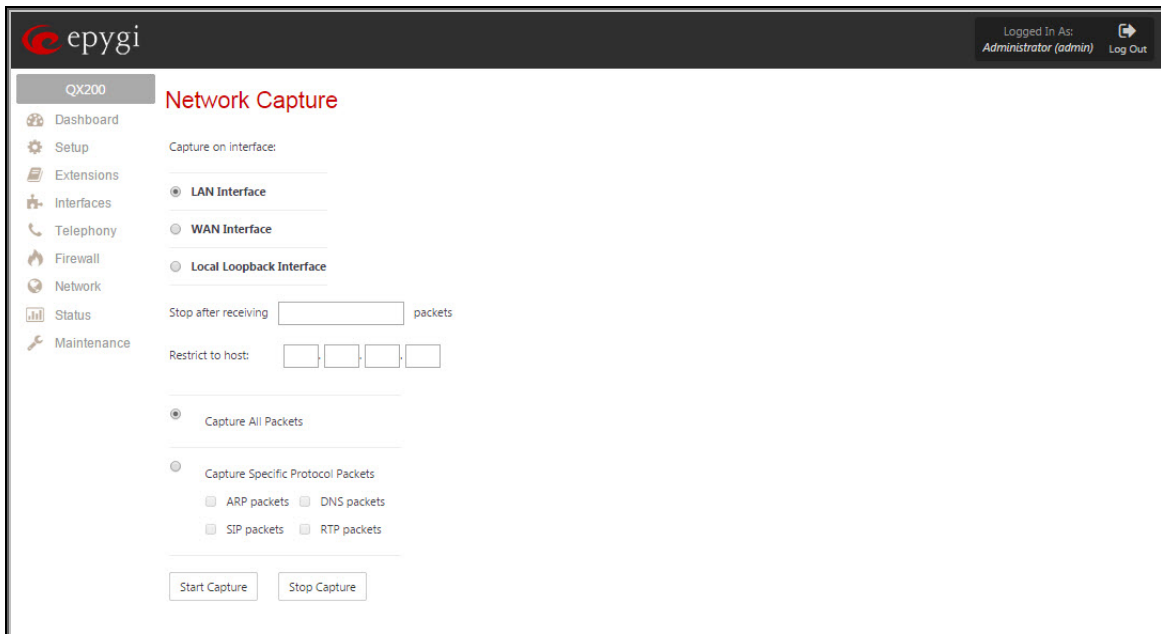


Figure 16: Network Capture page

- **Capture on interface:** Network interface from where the capture should be made. The **Local Loopback Interface** option is used to capture the traffic within the unit.
- **Stop after receiving count packet:** Number of packets to be captured.
- **Restrict to host:** Packets can be captured from the specified IP address.
- **Capture All Packets** allows capturing all the packets on the interface.
- **Capture Specific Protocol Packets** enables restricting the capture to specific packets only (ARP, SIP, DNS, and RTP).

Please Note: After the capture is initiated, the system will create a file and log all the packets meeting the specified criteria into it. Specifying a large number for this setting or leaving it unspecified and making a long capture may consume system resources and cause a system reboot.

3.12 Networkphycfg

The **Network PHY Config** page allows configuring speed and mode parameters for the LAN and WAN interfaces.



Figure 17: Network PHY Config page

The following options are available for configuration of the speed and mode parameters:

- 10BT-HD
- 10BT-FD
- 100BTX-HD
- 100BTX-FD

Please Note: By default these parameters are set in **auto** mode.

3.13 Trunkecho

The **ISDN Trunk Echo** page allows configuring echo settings of ISDN and E1/T1 trunks.

Trunks	Interface Type	Echo Delay	Echo Tail	NLP Difference	NLP Shift
Trunk 1	User	80	128	6	3
Trunk 2	Network	80	128	6	3
Trunk 3	Network	80	128	6	3
Trunk 4	Network	80	128	6	3

Figure 18: ISDN Trunk Echo page

- **Echo Delay:** Delay between the original voice and the corresponding echo signal. For the meaning of the **DSP Echo Tail**, **DSP Echo NLP Diff**, and **DSP Echo NLP Shift** (see section [3.6](#)).
- **Advanced Echo Settings:** Indicates whether the system should automatically switch off the DSP echo canceller if no echo is detected on the call, i.e. if the call path does not contain any echo source, such as FXO or FXS interfaces.
 - If the **Disable** option is selected, the echo canceller will be on regardless of whether or not echo occurs on the call.
 - If the **Enable** option is selected, the system will automatically switch off the canceller if the transmitted signal level is higher than the specified threshold (**Transmit Energy**) and the

received signal level is lower than the specified threshold (**Receive Energy**) for the specified length of time (**Process Saturation**) and these conditions hold true for the specified number of times (**Pattern Count**). Note that the higher the **Transmit Energy** level and the lower the **Receive Energy** level, the stricter the conditions for switching the canceller off.

To modify the settings of an ISDN (or E1/T1) trunk, click on the link associated with the trunk. This will open the **Trunk – Echo** page for the corresponding trunk (see):

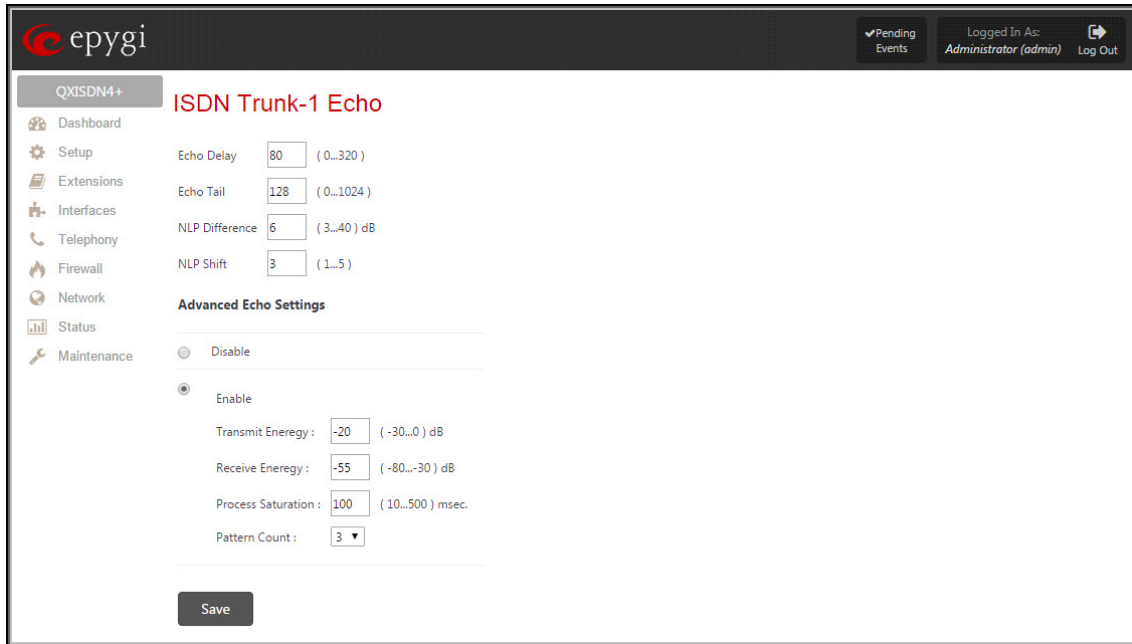


Figure 19: Configuring Echo Settings of ISDN Trunk-1

3.14 Tos

The **TOS Settings** page allows specifying the value for the **Type of Service** setting (ToS) for SIP and RTP packets (see Figure 20). The ToS setting is used in IP packets to notify routers how to handle the packet (precedence, delay, reliability, etc.).



Figure 20: TOS Settings page

- **low, medium, high:** Values for the ToS setting for low, medium and high priority RTP packets respectively. The priority level of RTP is chosen from the Users→Extension Management→Edit Entry page for the selected extension (see Figure 21).
- **SIP tos:** The ToS value for SIP packets.

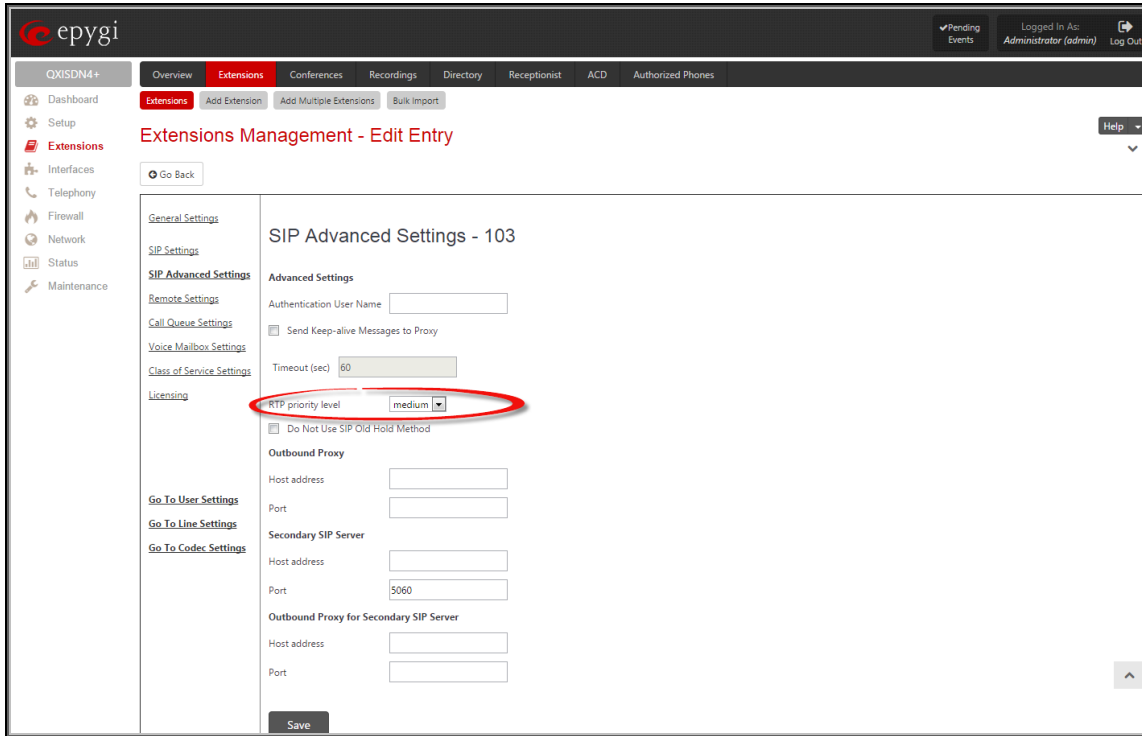


Figure 21: Extensions Management - Edit Entry page

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