



Release Notes for QX2000 6.2.47, Edition 1

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1 Introduction

This Release Notes describes hardware and firmware requirements to use with the
QX2000 firmware 6.2.47 Date: November 20, 2019

Additional enhancements, bug fixes and known issues incorporated in this firmware will be listed as known.

Date: November 29, 2019

2 Requirements

2.1 Hardware Requirements

- The firmware (FW) can be used on QX2000 model only.
- The model name is shown on the front panel of the unit.

2.2 Firmware Requirements

Attention: The firmware upgrade to 6.2.47 can **ONLY** be done from 6.0.2 and higher versions.

2.3 Supported IP Phones

Listed below are the Epygi Supported IP phones with the corresponding software (firmware) versions that are tested and recommended for use with QX2000 FW 6.2.47.

Note:

- The **Auto Configuration** and **PnP** services are described in detail in the [Configuring Epygi Supported IP Phones with QX IP PBXs](#) guide.
- Any known issues and limitations regarding the usage of the QX2000 FW 6.2.47 telephony services and features for each IP phone described in detail in the [QX IP PBX Features on Epygi Supported IP Phones](#) guide.

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Akuvox	R15(P)	15.0.5.235	Yes	Yes	Yes
Akuvox	SP-R53(P)	53.0.6.115	Yes	Yes	Yes
Alcatel	IP2015 (IP15)	1.0.7A-0	No	No	Yes
Alcatel	Temporis IP100	1.0.6A-0	No	No	Yes
Alcatel	Temporis IP150	1.0.6A-0	No	No	Yes
Alcatel	Temporis IP200	13.60.0.89	Yes	Yes	Yes
Alcatel	Temporis IP300	1.0.7B-0	No	No	Yes
Alcatel	Temporis IP600	14.60.0.89	Yes	Yes	Yes
Alcatel	Temporis IP700G	1.0.7A-0	No	No	Yes
Alcatel	Temporis IP800	15.60.0.89	Yes	Yes	Yes
AudioCodes	310HD	1.6.0_build_37	No	No	Yes
AudioCodes	320HD	1.6.0_build_37	No	No	Yes
Cisco	CP-6851	11-1-1	No	No	Yes
Cisco	CP-7861	11-1-1	No	No	Yes
Cisco	CP-8851	11-1-1	No	No	Yes
Cisco	SPA303	7.4.9c	No	Yes	Yes
Cisco	SPA501G	7.4.9c	No	Yes	Yes
Cisco	SPA509G	7.4.9c	No	Yes	Yes
Cisco	SPA525G2	7.4.9c	No	Yes	Yes
Fanvil	C58/C58P	2.3.233.129	Yes	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Fanvil	C58/C58P	2.3.233.129	No	No	Yes
Fanvil	C62/C62P	2.3.235.128	No	No	Yes
Fanvil	C400	14.0.0.3.r1	No	No	Yes
Fanvil	C600	14.0.0.3.r1	No	No	Yes
Fanvil	F52/F52P	2.3.123.78	Yes	Yes	Yes
Fanvil	H2/H2S	2.8.0.6251	Yes	Yes	Yes
Fanvil	H3	2.8.0.6251	Yes	Yes	Yes
Fanvil	H5	2.8.0.6251	Yes	Yes	Yes
Fanvil	X3/X3P	1.4.0.2016	Yes	Yes	Yes
Fanvil	X3S/X3G	2.8.0.6251	Yes	Yes	Yes
Fanvil	X4/X4G/X4S	2.8.0.6251	Yes	Yes	Yes
Fanvil	X5/X5G	1.4.0.2016	Yes	Yes	Yes
Fanvil	X5S	1.8.0	Yes	Yes	Yes
Fanvil	X6	1.8.0	Yes	Yes	Yes
Gigaset	Maxwell 3 PRO	82.2.22.7	Yes	Yes	Yes
Grandstream	GXP1100	1.0.8.6	No	Yes	Yes
Grandstream	GXP1105	1.0.8.6	No	Yes	Yes
Grandstream	GXP1160	1.0.8.6	No	Yes	Yes
Grandstream	GXP1165	1.0.8.6	No	Yes	Yes
Grandstream	GXP1400	1.0.8.6	No	Yes	Yes
Grandstream	GXP1405	1.0.8.6	No	Yes	Yes
Grandstream	GXP1450	1.0.8.6	No	Yes	Yes
Grandstream	GXP1615/1610	1.0.4.55	No	Yes	Yes
Grandstream	GXP1625/1620	1.0.4.55	No	Yes	Yes
Grandstream	GXP1628	1.0.4.55	No	Yes	Yes
Grandstream	GXP1630	1.0.4.55	No	Yes	Yes
Grandstream	GXP1760	1.0.0.48	No	No	Yes
Grandstream	GXP1782/1780	1.0.0.48	No	No	Yes
Grandstream	GXP2100	1.0.8.6	No	Yes	Yes
Grandstream	GXP2110	1.0.8.6	No	Yes	Yes
Grandstream	GXP2120	1.0.8.6	No	Yes	Yes
Grandstream	GXP2124	1.0.8.6	No	Yes	Yes
Grandstream	GXP2130	1.0.7.99	No	Yes	Yes
Grandstream	GXP2135	1.0.7.99	No	Yes	Yes
Grandstream	GXP2140	1.0.7.99	No	Yes	Yes
Grandstream	GXP2160	1.0.7.99	No	Yes	Yes
Grandstream	GXP2170	1.0.7.99	No	Yes	Yes
Grandstream	GXP2200	1.0.3.27	No	Yes	Yes
Grandstream	GXV3140	1.0.7.80	No	Yes	Yes
Grandstream	GXV3175	1.0.3.76	No	Yes	Yes
Grandstream	GXV3240	1.0.3.62	No	Yes	Yes
Grandstream	GXV3275	1.0.3.62	No	Yes	Yes
Htek	UC902	2.0.4.4.41	Yes	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Htek	UC903	2.0.4.4.41	Yes	Yes	Yes
Htek	UC912G	2.0.4.4.41	Yes	Yes	Yes
Htek	UC912P	2.0.4.4.41	Yes	Yes	Yes
Htek	UC923	2.0.4.4.41	Yes	Yes	Yes
Htek	UC924	2.0.4.4.41	Yes	Yes	Yes
Htek	UC924E	2.0.4.4.41	Yes	Yes	Yes
Htek	UC926	2.0.4.4.41	Yes	Yes	Yes
Htek	UC926E	2.0.4.4.41	Yes	Yes	Yes
iServ	8660 (8430/8630/8830)	03.55.0025	No	No	Yes
Mitel (Aastra)	6730	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6731	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6735	3.3.1.8140-SIP	Yes	Yes	Yes
Mitel (Aastra)	6737	3.3.1.8140-SIP	Yes	Yes	Yes
Mitel (Aastra)	6739	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6753	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6755	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6757	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	9480	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel	6863	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6865	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6867	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6869	4.2.0.2023-SIP	Yes	Yes	Yes
Panasonic	KX-HDV130	03.004	Yes	Yes	Yes
Panasonic	KX-HDV130NE, KX-HDV130X	06.101	Yes	Yes	Yes
Panasonic	KX-HDV230	03.004	Yes	Yes	Yes
Panasonic	KX-HDV230NE, KX-HDV230X	06.101	Yes	Yes	Yes
Panasonic	KX-TGP550T04	12.17	No	No	Yes
Panasonic	KX-UT123 (NE/RU/X)	01.302	No	No	Yes
Panasonic	KX-UT136 (NE/RU/X)	01.302	No	No	Yes
Polycom	SoundPoint IP 330	3.3.5.0247	No	Yes	Yes
Polycom	SoundPoint IP 331	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 335	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 450	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 550	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 650	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 670	4.0.13.1445	No	Yes	Yes
Polycom	SoundStation IP 5000	4.0.13.1445	No	Yes	Yes
Polycom	SoundStation IP 6000	4.0.13.1445	No	Yes	Yes
Polycom	VVX 300/310	5.7.0.11768	No	Yes	Yes
Polycom	VVX 301/311	5.7.0.11768	No	No	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Polycom	VX 400/410	5.7.0.11768	No	No	Yes
Polycom	VX 401/411	5.7.0.11768	No	No	Yes
Polycom	VX 500	5.7.0.11768	No	No	Yes
Polycom	VX 600	5.7.0.11768	No	Yes	Yes
Polycom	VX 1500	5.7.0.11768	No	Yes	Yes
QOSIP	Q7104/Q7204	1.0.3.98	No	No	Yes
snom	300	8.4.35	Yes	Yes	Yes
snom	320	8.4.35	Yes	Yes	Yes
snom	360	8.4.35	Yes	Yes	Yes
snom	370	8.7.5.35	Yes	Yes	Yes
snom	720	8.9.3.60	Yes	Yes	Yes
snom	760	8.9.3.60	Yes	Yes	Yes
snom	821	8.7.5.35	Yes	Yes	Yes
snom	870	8.7.5.35	Yes	Yes	Yes
snom	D120	10.1.10.1	Yes	Yes	Yes
snom	D345	8.9.3.60	Yes	Yes	Yes
snom	D375	8.9.3.60	Yes	Yes	Yes
snom	D710/710	8.9.3.60	Yes	Yes	Yes
snom	D712	8.9.3.60	Yes	Yes	Yes
snom	D715/715	8.9.3.60	Yes	Yes	Yes
snom	D725	8.9.3.60	Yes	Yes	Yes
snom	D745	8.9.3.60	Yes	Yes	Yes
snom	D765	8.9.3.60	Yes	Yes	Yes
snom	D785	10.1.20.0	Yes	Yes	Yes
snom	M700 (M85/M65/M25)	03.24.0007	Yes	Yes	Yes
snom	m9	9.4.7	Yes	Yes	Yes
snom	MeetingPoint	8.7.5.35	Yes	Yes	Yes
Spectralink	KIRK Wireless Server 300	PCS14C_	No	No	Yes
Spectralink	KIRK Wireless Server 6000	PCS14C_	No	No	Yes
VTech	ErisStation VCS754	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP600 (VSP601)	1.1.4.1-0	No	No	Yes
VTech	ErisTerminal VSP715	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP725	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP726	2.0.3.2-0	Yes	Yes	Yes
VTech	ErisTerminal VSP735	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP736	2.0.3.2-0	Yes	Yes	Yes
Yealink	CP860	37.81.0.10	Yes	Yes	Yes
Yealink	CP920	78.84.0.15	Yes	Yes	Yes
Yealink	CP960	73.83.0.30	Yes	Yes	Yes
Yealink	SIP-T19P	31.72.0.1	Yes	Yes	Yes
Yealink	SIP-T19P E2	53.83.0.50	Yes	Yes	Yes
Yealink	SIP-T20P	9.72.0.1	Yes	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Yealink	SIP-T21P	34.72.0.1	Yes	Yes	Yes
Yealink	SIP-T21P E2	52.83.0.50	Yes	Yes	Yes
Yealink	SIP-T22P	7.72.0.1	Yes	Yes	Yes
Yealink	SIP-T23G(P)	44.83.0.50	Yes	Yes	Yes
Yealink	SIP-T26P	6.72.0.1	Yes	Yes	Yes
Yealink	SIP-T27G	69.83.0.50	Yes	Yes	Yes
Yealink	SIP-T27P	45.83.0.50	Yes	Yes	Yes
Yealink	SIP-T28P	2.72.0.1	Yes	Yes	Yes
Yealink	SIP-T29G	46.83.0.50	Yes	Yes	Yes
Yealink	SIP-T32G	32.70.0.130	Yes	Yes	Yes
Yealink	SIP-T38G	38.70.0.125	Yes	Yes	Yes
Yealink	SIP-T40G	76.83.0.50	Yes	Yes	Yes
Yealink	SIP-T40P	54.83.0.50	Yes	Yes	Yes
Yealink	SIP-T41P	36.83.0.50	Yes	Yes	Yes
Yealink	SIP-T41S	66.83.0.50	Yes	Yes	Yes
Yealink	SIP-T42G	29.83.0.50	Yes	Yes	Yes
Yealink	SIP-T42S	66.83.0.50	Yes	Yes	Yes
Yealink	SIP-T46G	28.83.0.50	Yes	Yes	Yes
Yealink	SIP-T46S	66.83.0.50	Yes	Yes	Yes
Yealink	SIP-T48G	35.83.0.50	Yes	Yes	Yes
Yealink	SIP-T48S	66.83.0.50	Yes	Yes	Yes
Yealink	SIP VP-T49G	51.80.0.100	Yes	Yes	Yes
Yealink	SIP-T52S	70.83.0.50	Yes	Yes	Yes
Yealink	SIP-T54S	70.83.0.50	Yes	Yes	Yes
Yealink	SIP-T56A	58.83.0.15	Yes	Yes	Yes
Yealink	SIP-T58A/V	58.83.0.15	Yes	Yes	Yes
Yealink	VP-530	23.70.0.40	Yes	Yes	Yes
Yealink	W52P	25.30.0.20	Yes	Yes	Yes

2.4 Interaction with Other Epygi Software Releases

Use the latest SW and FW versions for other Epygi products to achieve maximum compatibility with QX2000 FW 6.2.47:

- QXE1T1, QXFXO4 and QXISDN4 gateways used in the **Share** mode should have FW 6.2.35 or higher.
- QXFXS24 should have FW 6.2.35 or higher for PnP configuration.
- Auto Dialer SW 1.0.11 or higher should be used.
- Desktop Communication Console (DCC) SW 1.18 or higher should be used.
- iQall (IOS application) version 1.1.0 and iQall (Android application) version 1.0.5 or higher should be used.
- Epygi Hotel Console (EHC) SW 1.0.7 or higher should be used.
- Epygi Media Streamer (EMS) SW 2.4 or higher should be used.
- HotCall Add-In SW 2.5 or higher should be used.
- HotKeyCall SW 1.14 or higher should be used.
- Bulk User Extensions Importer version 1.6 or higher should be used.
- QX-Quadro Configuration Console (QCC) SW 2.3 or higher should be used.
- CallControl Pack SW 5.8.0 or higher should be used.
- To use QX2000 with a 3PCC or Click2Dial application, the **Allow 3pcc/Click2Dial Access** option should be enabled for each extension using this feature.

3 New Features

The table below indicates a high-level list of new features that have been added beginning with the most recent QX2000 FW release.

Release	New Features
6.2.47	
6.2.46	
6.2.41	<p>The Phone Book service has been enhanced with a new capability to create and manage the contact list for the phone's directory. In addition to the existing methods of 1) adding the QX selected extensions to the contact list automatically in the extensions management, and 2) composing the list of external users as a CSV file and afterwards importing that file into the QX, a new method is added to manage the list of contacts by directly adding, editing and deleting contacts using the QX WEB GUI.</p>
	<p>Added support for voice Transcoding when the call is passing through the QX. An incoming SIP call can be transcoded before being sent to another SIP destination. The Transcoding function is applicable for all Epygi supported voice codecs.</p>
	<p>Added support for the new Low Credit Notification feature. Based on a predefined threshold, the system will check and play the balance available for the extension before establishing the call. Two options of playing the balance are available - either by available credit amount or by available calling duration.</p>
	<p>Added support to provide notifications when the predefined limit for a single call duration is reached. Notifications are provided via system events, via email and SMS.</p>
	<p>Enhancements in Hot Desking feature. Added two new options for Hot Desking:</p> <ul style="list-style-type: none"> • Play Balance After Login • Hot Desking Automatic Logout after each phone call
	<p>Added support for a new feature and new configuration options in the ACD queue settings:</p> <ul style="list-style-type: none"> • Playing Caller's Position in Queue – this option allows to inform the caller of his/her position in the queue. The queue administrator has additional control over this feature by specifying the exact number of calls in the queue after which the position in queue message will be played. • Playing Estimated Wait Time – this option allows to announce to the caller the approximate wait time in the queue, so that the caller can then make a decision to continue waiting in the queue or to take other actions, such as a dial zero option, call back later or to receive a call back, mentioned below. • ACD Call Back – when this option is activated the caller is given a choice to terminate the call and receive a call back if they choose not to wait in the queue.
	<p>Added custom logo support for the Htek IP phones: UC923, UC924, UC924E, UC926 and UC926E.</p>
	<p>Two new Voice Mail Indication options have been added to the IP Phone Templates for the Yealink IP phones: T19P E2, T21P E2, T23G(P), T27G(P), T29G, T40G(P), T41S(P), T42S(P), T46S(P), T48S(P), T52S, T54S, T56A, T58V/A. The options are Voice Mail Tone and Voice Mail Power Light Flash and they are enabled by default.</p> <ul style="list-style-type: none"> • Voice Mail Tone - the user will hear a specific beep during the call (just once) if a new VM is received in the mailbox • Voice Mail Power Light Flash – the VM power light will flashing when there is a new VM in the mailbox
<p>Support for Two-Factor Authentication (2FA).</p>	
<p>Added PnP and auto configuration support for the new snom D120, D712 and D785 IP phones.</p>	

Release	New Features
	<p>Added PnP and auto configuration support for the new Gigaset Maxwell 3 PRO IP phone.</p> <p>Added support for the OPUS codec.</p> <p>Added support for having remote access to QX devices through the cloud based ecMON application, even if the QXs are behind a firewall and having a private IP. Two options of remote access are supported: login to the QX GUI and Proxy to IP phones connected to QX.</p> <p>New Automatic Outbound Calling feature to automate the outbound dialing and contact with customers. The contacts are called from a web-based application and is integrated with ACD. The application allows to control and monitor automatic outbound calling sessions on the QX and then placing the answered calls in a high priority queue to be answered by an agent. Implemented with a license key.</p> <p>Paging service enhancements. A new Store and Forward Call option is added in the settings for paging groups and provides three enhancements:</p> <ul style="list-style-type: none"> • A Page is recorded but it is not sent to the paging group until the caller hangs up or presses the # key. • Multiple users can initiate a page simultaneously since each page will now be queued and it will be announced in the order that it was recorded. • This will also prevent acoustic feedback when a person currently does a live page. <p>Additional features have been added to the iCall application, such as:</p> <ul style="list-style-type: none"> • voice mail push notifications • missed calls push notifications <p>A special license key is not needed for these new functions. A license key is only needed for the iCall mobile toggling option.</p> <p>Added Receptionist support for Cisco CP-8851 with expansion modules 8800 and 8851/61.</p> <p>Added support for sending random outbound caller ID. An outbound caller ID will be randomly chosen from the list of caller IDs defined in the Call Routing entry for the outbound call.</p> <p>Added P-Asserted Identity (PAI) support for Cisco, Htek, Grandstream and QOSIP IP phones. The PAI option is configurable from the IP Phone Templates. It is enabled by default.</p> <p>Added IP Phones Logo support for Cisco CP-8851 and Polycom WX1500 IP phones.</p> <p>Added autoconfiguration support using OpenVPN service for the Fanvil phones X5S and X6.</p>
6.2.23	<p>Added auto configuration support for the new Cisco CP-6851, CP-7861 and CP-8851 IP phones.</p> <p>Added PnP support for Htek UC902, UC903, UC912G, UC912P, UC923, UC924, UC924E, UC926 and UC926E IP phones.</p> <p>Added auto configuration support for the new iServ 8660 (8430/8630/8830) Dect phone.</p> <p>Added support for the Twilio service as a new SMS Gateway. Use Twilio API for sending notifications via SMS.</p> <p>Added P-Asserted Identity (PAI) support for Polycom IP phones. The PAI option is configurable from IP Phone Templates. It is enabled by default.</p> <p>Added support for the Use Session Progress option instead of Ringing for the external calls.</p> <p>Added a new Threshold to enter Master Mode option on the Redundancy Settings page. This is an advanced configuration option and is related to Backup mode. This option allows to specify the time interval to wait before moving to Master mode. Default values of Threshold to enter Master Mode is 1. Attention: Please contact Epygi TSS before changing the value for this option.</p>

Release	New Features
	Enhancements in EAC WEB application: <ul style="list-style-type: none"> • Added new menu Calls, which is used to show the active calls addressed to the logged in agent. This Calls menu has the same functionality as the Dashboard menu in the QX FW versions bellow 6.2.40. • The Dashboard menu has been completely modified. For FW version 6.2.40 it shows various statistics and data for the selected Queue(s). This menu is available only when you are logged in as ACD Admin. • Added a new Agent Tracing report, which details agent activities regarding the status changes and calls for the selected agent(s), queue(s) and timeframe(s). • Added new values (Every 2, 3 and 4 minutes) for the Auto Reload Queue Statistics option. • The Settings menu has been improved and updated.
6.2.18	Added support allowing to configure MTU size on LAN1 and VLAN interfaces.
6.2.18	Added P-Asserted Identity (PAI) support for Mitel (Aastra) IP phones. The PAI option is configurable from IP Phone Templates . It is enabled by default.
6.2.11	Added auto configuration support for the new Htek UC902, UC903, UC912G, UC912P, UC923, UC924E and UC926E IP phones.
6.2.11	Added Htek UC46 (LCD) expansion module support for Htek UC924, UC924E, UC926 and UC926E IP phones.
6.2.11	In the Call Alert Settings added a new Leave a Voice Message option, allowing to leave the actual message as a voice mail, available for playback on the defined extension(s).
6.2.11	Added a new Advertisement Interval option in Redundancy Settings page which allows to specify the time interval between the advertisement packets that are being sent to the Backup device.
6.2.11	Added a new Allow Concurrent Calls to Parent-Child Group option, allowing to control handling of calls to Parent-Child group: <ul style="list-style-type: none"> • 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.
6.2.6	Added a new Reset All function in the Multi-functional Programmable Keys (MPKs) for IP phones, allowing to clean the MPKs configuration quickly for the selected IP line, IP Phone Template and Receptionist. Note: This option is not programmed to remove already configured MPKs from the IP Phone.
6.2.6	Added new failover reason – Other . The system will use next matching routing pattern(s) in case of Server Failure Responses (5xx messages) and Global Failure Responses (6xx messages).
6.2.6	Added possibility to access QX2000 WEB GUI using HTTP. Enter the following line http://xxx.xxx.xxx.xxx/unsecure in the address bar of the browser to access WEB GUI, where xxx.xxx.xxx.xxx is the IP address or hostname of the QX.
6.2.6	
6.2.5	Added auto configuration support for the new Polycom VX 301/311 and VX 401/411 IP phones.
6.2.5	Added Phone Book service support for Polycom phones.

Release	New Features
	<p>Added Watching – Call Interception support for Fanvil phones.</p> <p>Added support for SNMP v3.</p> <p>Added support for TLSv1.1 and TLSv1.2.</p> <p>Security enhancements: Users will be redirected to HTTPS for the QX Login and Logout pages. This will allow to encrypt traffic between user's device (PC, smartphone, etc.) and the QX.</p> <p>Note:</p> <ul style="list-style-type: none"> • Check and reconfigure Port Forwarding settings on the router, if the QX is located behind router to make sure that there is also Port Forwarding for HTTPS. • If you have already configured Port Forwardings to access the devices located on the QX LAN side, then check the entered address link to be with HTTP (instead of HTTPS) or reconfigure the Port Forwarding to HTTPS. <p>Added support in EAC to show the Agent Status Statistics, Queue Statistics in hourly/minutes basis, as well as to generate Reports for selected Hours and Minutes.</p> <p>Added a new Deactivate button on the IP Lines page allowing to change the status for selected group(s) of IP lines to inactive (free).</p> <p>Added a new Use Epygi SIP Server button on the Conference Management and ACD Queues pages to allow quick SIP registration of Conference extensions and ACD queues on Epygi SIP Server.</p> <p>Added a new Billed Extension column in the Call History pages to provide information about the extensions that are charged for the calls.</p> <p>Added support to provide QX users with e-mail, sms and event notifications in case of calls (emergency calls, etc.) completed through the respective call routing rules.</p>
6.2.1	<p>Added PnP and auto configuration support for the new Yealink CP920, CP960, SIP-T40G, SIP-T52S, SIP-T54S, SIP-T56A and SIP-T58A/V conference, audio and video phones.</p> <p>Added PnP and auto configuration support for the new Fanvil H2/H2S, H3, H5, X3S/X3G, X5S and X6 IP phones.</p> <p>Added PnP and auto configuration support for the new snom D745 and Akuvox R15(P) IP phones.</p> <p>Added auto configuration support for the new Htek UC924 and UC926 IP phones.</p> <p>Added PnP and auto configuration support for the new Panasonic KX-HDV130 and KX-HDV230 IP phones.</p> <p>Added a new Call Completion Fee option in the Calling Cost Control allowing to calculate call cost per the number of completed calls.</p> <p>Added new Auto Reload Queue Statistics option in EAC settings allowing to automatically reload (refresh) Queue Summary pages.</p> <p>Added a new option for Agent status by Queue, by Date (Summary) report in EAC which allows to show the status duration in percentage.</p>
6.1.50	<p>Added PnP and auto configuration support for the new Grandstream GXP1615, GXP1628, GXP1630, GXP2135 and GXP2170 IP phones.</p> <p>Added auto configuration support for the new Grandstream GXP1760 and GXP1782/1780, IP phones.</p> <p>Added PnP support for the Grandstream GXP1610 and GXP1625/1620 IP phones.</p> <p>Added PnP and auto configuration support for the new Mitel 6869 IP phone.</p>

Release	New Features
	<p>Added support for the new Calling Cost Control licensable feature. This feature allows to limit and control the cost of calls through the routing rules. The following changes are done concerning mainly the Extensions Settings and the Call Routing.</p> <ul style="list-style-type: none"> • You can assign a credit amount for each specific extension for making calls through the "payable" routing rules. • It allows to configure and use "payable" call routing rules to be used only by extensions with a calling credit assigned. • The overall calling costs for "payable" routing rules are calculated and reported in the call history.
	<p>Configuration Management enhancements</p> <ul style="list-style-type: none"> • Added a new option to allow the EAC data to be backed up and saved along with the system configuration and voice data. The EAC data includes the EAC Chat database, Agents' Status and Call Statistics. • Added a new service to restore the system configuration and voice data together with the EAC data. Note: The current EAC data with system configuration will be overwritten after configuration restore.
	<p>Added a new Click to Dial & Announce feature allowing the Dial & Announce service to be activated on the QX extensions by using the 3PCC Request URI method from a WEB browser.</p>
	<p>Added the SSH FTP (SFTP) support, which allows to send the configuration backup files to an FTP server using the secure FTP connection.</p>
	<p>Added a new Archive Now option on the Call History – Archiving Settings page, allowing to archive immediately the available data.</p>
	<p>Added new Reporting types in EAC: CDRs by Agent, by Queue, by Date and CDRs by Queue, by Agent, by Date.</p>
	<p>Added the new Enable VLAN Tagging option. This option is used to enable/disable setting the VLAN ID and priority for IP phones. Note: The provided IP address will always be from the VLAN network.</p>
	<p>The Client Code Identification option can be activated and used by other billing systems as well as it is done for RADIUS server.</p>

4 Changed Features

The table below provides a high-level list of changed features that have been changed beginning with the most recent QX2000 FW release.

Release	Changed Features
6.2.47	
6.2.46	
6.2.41	The recommended FW version has been changed for most of the supported Yealink phones.
	Added a new setting in the paging group GUI, allowing to send the page by simply hanging up, without waiting for #.
	The list of recording codecs in the voice mail common settings is updated. All codecs except of the G.711U/A and G.729 are removed in the list.
	The Single Call Duration Limit option has been implemented for PBX type calls in Call Routing Table as well.
	The value for TTL (Time to Live) setting is changed to 64.
	The Telemetry page is renamed to ecMON on the QX WEB GUI.
	The recommended FW version has been changed for most of the supported Fanvil phones: <ul style="list-style-type: none"> • For C400/C600 from 11.20.12.2.B to 14.0.0.3.r1 • For H2/H2S from 2.0.2.2776 to 2.8.0.6251 • For H3/H5 from 2.0.2.2770 to 2.8.0.6251 • For X3/X3P from 1.3.511.1821 to 1.4.0.2016 • For X3S/X3G from 2.0.3.3049 to 2.8.0.6251 • For X4/X4G/X4S from 2.0.2.2830 to 2.8.0.6251 • For X5/X5GT from 1.3.511.1821 to 1.4.0.2016 • For X5S from R0.7.0.1 to 1.8.0 • For X6 from R0.5.3 to 1.8.0
	The Stun Polling Interval parameters have been adjusted. Added two extra parameters for the Polling Interval option: 10 and 20 minutes. The default value is still set to 1 hour.
	The VLAN interface is added in the Network Capture settings.
	The LLDP parameter is not overwritten for Yealink IP phones after getting provisioned by the QX.
6.2.23	
6.2.22	Major Security Enhancements.
	The recommended FW version has been changed for Htek UC902, UC903, UC912G, UC912P, UC923, UC924, UC924E, UC926 and UC926E phones from 2.0.4.4.33 to 2.0.4.4.41.
	GUI Enhancements for the following pages: <ul style="list-style-type: none"> • IP Lines, IP Lines Settings • SMS Settings • Extensions Management • ACD Management • NAT Traversal • VoIP Carrier Wizard
	The system Music on Hold (MOH) file will be used as default Background Music for ACD Queue custom prompts.
6.2.18	Major Security Enhancements.
	The default MTU size for VLAN interfaces has been decreased from 1500 to 1432 bytes.

Release	Changed Features
	<p>The configured SRTP policy of PBX extension will be provided to the Yealink IP phones during configuration.</p> <p>The configured DTMF parameters of PBX extension will be provided to the IP phones during configuration.</p>
6.2.11	<p>The maximum length of API ID field for Clickatell SMS Gateway has been increased up to 128 symbols.</p> <p>The recommended FW version has been changed for some of supported Htek phones. For UC924 and UC926 from 2.0.4.2.24 to 2.0.4.4.33.</p> <p>GUI Enhancements for the following pages:</p> <ul style="list-style-type: none"> • Admin Settings of the extensions (user, auto attendant, etc.) • User Settings of the extensions • Call History • Conference History <p>Depending on the IP phone model, the Use Session Timer option will be enabled for the configured IP line.</p> <p>The default Line Appearance has been increased from 2 to 5 for each IP line.</p>
6.2.6	<p>Network and Broadcast IP addresses will not be included into Usable Host IP Range. These IPs will be reserved for network purposes.</p>
6.2.5	<p>The Second LAN Interface Settings have been moved from the hidden page to the Network→Second LAN page.</p> <p>The function of Mixed mode for Recording Storage Settings has been updated to keep the call recordings safe in case of FTP failure. Now this mode allows to send recordings to FTP server immediately together with keeping a copy in the local storage.</p> <p>The recommended FW version has been changed for Yealink CP860 from 37.80.0.30 to 37.81.0.10.</p> <p>The recommended FW version has been changed for some of the Polycom phones. For Polycom SoundPoint IP 331, IP 335, IP 450, IP 550, IP 650, IP 670 from 3.3.5.0247 to 4.0.13.1445, for SoundStation IP 5000 and IP 6000 from 3.3.5.0247 to 4.0.13.1445, for VWX 300/310, VWX 400/410, VWX 500 and VWX 600 from 4.1.7.1210 to 5.7.0.11768, for VWX 1500 from 3.3.5.0247 to 5.7.0.11768.</p> <p>The first programmable key on Polycom phones is reserved for the phone account.</p> <p>The Call Quality Warning in the System Events has been modernized to show information about the callee, caller and call date/time.</p> <p>GUI enhancements for Call History and Conference History pages.</p> <p>Redundancy feature has been redesigned to allow sending VRRP packets to unicast IP address (virtual IP) of slave device. Note: Having VRRP packets sent via unicast IP address will allow Redundancy feature to work in scenario when the master and backup devices are located in different places, meaning that there is router between master and backup.</p>
6.2.1	<p>The PSTN Gateways Line Sharing mechanism has been changed and updated, bringing more stability, improving the connection between PBXs and Gateways. Important Note: Please update the firmware version to 6.2.1 both on QX2000 and QX Gateway(s) to be able successfully connect the devices and share the lines.</p> <p>Added option allowing to share and synchronize the configured Incoming Interdigit Service settings with QXISDN4 and QXE1T1 gateways when connected with QX2000 in shared mode.</p> <p>The allowed duration of recorded voice mail sent as attachment via e-mail has been increased from 3 to 5 minutes, when G729a codec is used for recording voice mails. Note: If G711u codec is used for recording, the attached voice mail will not be truncated before being sent via e-mail.</p>

Release	Changed Features
	<p>The timezone database has been updated on QX IP PBXs:</p> <ul style="list-style-type: none"> • The current local time has been corrected for Israel, Venezuela, Shri Lanka, Apia, Samoa and Fiji. • Added new timezone Nukualofa, Tonga (GMT+14). <p>New Date/Time pickers have been implemented for all applicable GUI pages, allowing to select or define the date/time options easier and conveniently.</p> <p>Enhancements for the Call History – Call Cost page:</p> <ul style="list-style-type: none"> • Added new filtering options supporting multicriteria searching for payable call records. • Added support to download the displayed CDRs in the (*.log) and (*.csv) formats respectively. <p>The recommended FW version has been changed for Yealink SIP-T40P from 54.81.0.25 to 54.81.0.110.</p> <p>The recommended FW version has been changed for some of snom phones. For snom 720, 760, D710/710, D715/715, D725, D765 from 8.7.5.35 to 8.9.3.60 and for D345, D375 from 8.9.3.35 to 8.9.3.60.</p> <p>Panasonic KX-UT123 and KX-UT123NE IP phones have been merged and renamed to KX-UT123 (NE/RU/X).</p> <p>Panasonic KX-UT136 IP phone has been renamed to KX-UT136 (NE/RU/X).</p> <p>The recommended FW version has been changed for some Panasonic phones. For KX-UT123 (NE/RU/X) and KX-UT136 (NE/RU/X) from 01.221 to 01.302.</p> <p>Akuvox SP-R53P/SP-R53 IP phone has been renamed to Akuvox SP-R53(P).</p> <p>The recommended FW version has been changed for Akuvox SP-R53(P) IP phone from 53.0.1.23 to 53.0.6.115.</p> <p>Added nexogy, ClarityTel and Adiptel as the new carriers to the VoIP Carrier Wizard list.</p> <p>Added a new option allowing to select Conference extensions from the Unconditional, Busy, No Answer and Unregistered Call Forwarding lists.</p> <p>The default TLS port number (5061) will be selected for SIP.</p>
6.1.50	<p>The recommended FW versions have been changed for some Grandstream IP phones. For GXP1610 and GXP1625/1620 from 1.0.2.27 to 1.0.4.55, for GXP2130, GXP2140 and GXP2160 from 1.0.5.23 to 1.0.7.99.</p> <p>The recommended FW versions have been changed for Mitel IP phones. For 6863, 6865 and 6867 from 4.0.0.92-SIP to 4.2.0.2023-SIP.</p> <p>The maximum number of Watched Extensions for DCC Pro has been increased: for QX20 from 30 to 32, for QX50 and QXISDN4+ from 30 to 50, for QX200 from 100 to 200, for QX500 and QX2000 from 100 to 300.</p> <p>The HTML5 Date/Time picker is implemented for Date/Time selection.</p> <p>The backup configuration filename format has been updated and will include the installed firmware version of the QX: config_[Hostname]_[Firmware Version]_[Date/Time].bin</p> <p>Added option allowing to display Media Streamer's allocated and used memory space on the Status→System Status→Memory page.</p> <p>Added new option allowing to select and change Schedule State from WEB GUI.</p> <p>The Network Capture page has been moved to Maintenance→Diagnostics→Network Capture page.</p> <p>GUI Enhancements for Call Routing Table.</p> <p>GUI Enhancements on the Setup→Licensed Features page.</p> <p>GUI Enhancements for IP Phone Templates.</p>

5 Fixed Issues

Issues fixed since version 6.2.46:

T: Title
D: Description

20640	T:	Scheduled reports for Agent status by Queue by Date not calculating correctly
	D:	Scheduled reports for Agent status are not calculating for a 24-hour period regardless of time preset.

6 Known Issues

T: Title
D: Description
C: Consequences
Fix: How to avoid the situation, or what to do in case the situation has occurred

20481	T:	An issue concerning the IP phone Multi-functional Programmable Key (MPK) for ACD Login/Logout
	D:	The state of the MPK on IP phone, configured for ACD login/logout, is not updated correctly after power off/on the phone. If you turn off the phone and turn on again, the MPK on the phone will always show the default logout state.
	C:	No consequences.
	Fix:	Will be fixed in future release.
20471	T:	An issue concerning the Hot Desking Automatic Logout -> "After the call" option
	D:	The Hot Desking extension wouldn't logout automatically after making a call if just select the "After the Call" option and press Save. You have to logout/login additionally using the feature code *78 to make it work.
	C:	No consequences.
	Fix:	Will be fixed in future release.
20390	T:	The notification message about the low credit for extension is not playing when using redialing method for calling (*1)
	D:	
	C:	No consequences.
	Fix:	Don't use the redialing method if you want to get low credit notification. Will be fixed in future release.
20074	T:	Fanvil IP Phones have issue with firmware downgrade in general. Fanvil Phones stop working when downgrading the firmware, even if you downgrade to Epygi recommended version
	D:	
	C:	
	Fix:	Don't downgrade the firmware on Fanvil IP Phones. Will be fixed by in next version.
19805	T:	The BLF indication (for programmable keys) on snom phones is switched off after the subscription timeout expires, regardless of the actual state of the BLF event
	D:	The issue appears on snom 3xx and 8xx series running 8.7.5.35 firmware version.

	C:	
	Fix:	Workaround: The issue is solved in snom 8.7.5.44 beta firmware.
19725	T:	A problem when restore a config backup downloaded from the QX with custom language Pack (LP) installed
	D:	<p>Scenario that cause this problem:</p> <ul style="list-style-type: none"> • Backup and download a config file on the QX with a custom LP installed • Remove the custom LP • Upload and restore back that config file to the same QX <p>At this point none of the system voice messages is playing. The reason why this is happening is the following: the system is looking for the audio files specific for that LP.</p>
	C:	No consequences.
	Fix:	Install the same LP on the QX, after remove that LP in the upload language pack page. Will be fixed in future release.
19463	T:	3-way conference doesn't work on Grandstream GXP1100 and GXP1105 IP phones in a specific scenario
	D:	3-way call conference cannot be established on Grandstream GXP1100 or GXP1105 phones when they receive a call.
	C:	
	Fix:	Workaround: Login into WEB GUI of the phone and assign 3-way conference key as a MPK. Use this key to initiate 3-way call conference when the phone is already in the active call. Will be fixed in future release.
19446	T:	After changing QX2000 LAN IP configuration, the phones configured from LAN side lose registration
	D:	After changing QX LAN IP configuration (changing the network part of the IP address) the system doesn't reboot phones automatically.
	C:	IP phones lost registration.
	Fix:	Workaround: Reboot phones manually. Will be fixed in future release.
18839	T:	It's not possible to park a call twice to the same call park extension by using programmable key on Yealink T32G and T38G
	D:	Upon successful call park/pick up the second attempt to park the call, using the park ext. programmable key fails. The problem is happening only if you park the call to the same park extension (by pressing Call Park key).
	C:	
	Fix:	Workaround: Park the call to different call park extension.
18549	T:	Could not dial out (*1) or use any other moderator feature while welcome message file has been playing
	D:	Could not dial out (*1) or use other moderator features while welcome message file has been playing. You should listen to the whole welcome message file first, after that use moderator features. It is recommended to keep the welcome message to a short duration.
	C:	
	Fix:	Will be fixed in future release.
18548	T:	Part of conference recording is lost after recording pause/resume
	D:	When pausing the conference recording and then resuming it again, the final recording contains only the part after resuming.
	C:	
	Fix:	Will be fixed in future release.
17404	T:	Calls which are done using Call Relay (*2) on the auto attendant are not shown in Call History

	D:	Only the call to attendant is shown in the call history. The call leg after call relay is missing in the call history in case if the external caller is terminating the call first.
	C:	
	Fix:	Workaround: Use feature code *1 instead of *2 for call relay. Will be fixed in future release.
16635	T:	Shared Mailbox watching does not work when using "Allow access to Shared Mailbox for enabled extensions" option in Many Extension Ringing configuration
	D:	Extension has Many Extension Ringing enabled with a few extensions configured for Shared Mailbox.
	C:	However, in the IP Line settings, the Shared VMail Ext. xxx option is not listed in the drop-down list on IP Lines→MPK page.
	Fix:	Workaround: Use the Shared Mailbox: Edit Voice Mailbox Access List link in the Voice Mailbox Settings for extension. Will be fixed in future release.
16635	T:	Shared Mailbox watching does not work when using "Allow access to Shared Mailbox for enabled extensions" option in Many Extension Ringing configuration
	D:	Extension has Many Extension Ringing enabled with a few extensions configured for Shared Mailbox.
	C:	However, in the IP Line settings, the Shared VMail Ext. xxx option is not listed in the drop-down list on IP Lines→MPK page.
	Fix:	Workaround: Use the Shared Mailbox: Edit Voice Mailbox Access List link in the Voice Mailbox Settings for extension. Will be fixed in future release.

7 General Hints

7.1 Firmware Installation and Update

The steps below describe shortly the QX2000 manual installation procedure used to install the firmware from scratch. This would be used to install version 6.1.45 or for **Emergency Recovery** of a system. This procedure will result in a system that is at factory defaults. Please refer to [QX1000/2000 System Recovery Procedure](#) document for more details.

1. Turn on the PC.
2. Insert CD/DVD disk including installation program to the DVD ROM.
3. Restart (reset) the PC.
4. Installation will start automatically after PC reboot. After the successful installation, the PC will automatically shut down (this may take from 10-15 minutes). The beep sound will indicate that the installation successfully completed.
5. Turn on the PC and quickly remove the installation CD/DVD disk from the DVD ROM.

Attention: It is recommended to back up the configuration for **emergency purposes** prior to upgrading the firmware. You can do that from **Maintenance**→**Backup/Restore**→**Backup and download current Configuration** page. The current configuration will remain after the firmware update. Moreover, the locally saved voice mails and call recordings, all custom messages and call history will be saved during the upgrade.

To perform the 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.
5. Click **Yes** to proceed the firmware upgrade.

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

7.2 Limitations and Restrictions

- The **Network Capture** size is limited to **24** MB. This will put a limitation on the duration of captured file.
- The **Call Capture** duration is limited to **160** seconds.
- The capture duration is limited to **160** seconds in **DSP Capture** hidden page.
- In case if **Voice Mail Recording Codec** is other than **PCMU**, the maximum length of voice message sent by email is limited to **5** minutes.
- The **Voice Mailbox** size is limited to **500** voice mails for each extension.