Release Notes for QX50/QX200 6.2.45 Edition 1

1 Introduction

This Release Notes describes hardware and firmware requirements to use with the

QX50/QX200 firmware 6.2.45 Date: June 10, 2019

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

Date: June 21, 2019

2 Requirements

2.1 Hardware Requirements

- The firmware (FW) can be used on QX200 and QX50 models only.
- The model name is written on the front plate of the unit; the unit serial number is on the bottom label.

2.2 Firmware Requirements

Attention: The firmware upgrade to 6.2.45 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 QX50/QX200 FW 6.2.45.

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 QX50/QX200 FW 6.2.45 telephony services and features for each IP phone are 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
Alcatel	Lucent - 8058S	1.51.05	No	No	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	SPA508G	7.5.2	No	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Cisco	SPA509G	7.4.9c	No	Yes	Yes
Cisco	SPA525G2	7.4.9c	No	Yes	Yes
Fanvil	C58/C58P	2.3.233.129	No	No	Yes
Fanvil	C62/C62P	2.5.787.97	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	Auto
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

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Grandstream	GXV3275	1.0.3.62	No	Yes	Yes
Htek	UC902	2.0.4.4.41	Yes	Yes	Yes
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

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Polycom	VVX 300/310	5.7.0.11768	No	Yes	Yes
Polycom	VVX 301/311	5.7.0.11768	No	No	Yes
Polycom	VVX 400/410	5.7.0.11768	No	No	Yes
Polycom	VVX 401/411	5.7.0.11768	No	No	Yes
Polycom	VVX 500	5.7.0.11768	No	No	Yes
Polycom	VVX 600	5.7.0.11768	No	Yes	Yes
Polycom	VVX 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	Auto
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

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Yealink	SIP-T19P E2	53.83.0.50	Yes	Yes	Yes
Yealink	SIP-T20P	9.72.0.1	Yes	Yes	Yes
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-T53/53W	95.84.0.30	Yes	Yes	Yes
Yealink	SIP-T54S	70.83.0.50	Yes	Yes	Auto
Yealink	SIP-T54W	96.84.0.30	Yes	Yes	Yes
Yealink	SIP-T56A	58.83.0.15	Yes	Yes	Yes
Yealink	SIP-T57W	97.84.0.30	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 **QX50/QX200 FW 6.2.45**:

- **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 2.5 or higher should be used.
- **CallControl Pack** SW 5.8.0 or higher should be used.
- To use QX50/QX200 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 QX50/QX200 FW release.

Release	New Features
6.2.45	Added support for FW update, configuration backup upload & restore from ecMON.
	Added PnP and autoconfiguration support for the Cisco 508G/509G phone.
	Added PnP and autoconfiguration support for the Yealink T53W, T54W and T57W IP phones.
	Added autoconfiguration support for Alcatel-Lucent lCTouch-Phone 8058S.
	Added iQall advanced features support for IOS: • voice mail push notifications • missed calls push notifications
	Added the new "Send Push notification" option under the caller ID based services → Find me / Follow me settings for extensions. Enabling this option on the extension configured for iQall allows to receive Find me/Follow me notifications on the mobile phone.
	Added the new "Send Push notification" option under the caller ID based services → call forwarding settings for the extensions. Enabling this option on the extension configured for iQall allows call forwarding notifications on the mobile phone.
6.2.40	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.
	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.
	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.
	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.
	Enhancements in Hot Desking feature. Added two new options: • Play Balance After Login • Hot Desking Automatic Logout after each phone call
	Added support for a new feature and new configuration options in the ACD queue settings: • **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.

Release	New Features
	Added custom logo support for the Htek IP phones: UC923, UC924, UC924E, UC926 and UC926E.
	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. • **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
6.2.35	Support for Two-Factor Authentication (2FA).
	Added PnP and auto configuration support for the new **snom D120**, **D712** and **D785** IP phones.
	Added PnP and auto configuration support for the new **Gigaset Maxwell 3 PRO** IP phone.
	Added support for the OPUS codec.
	Added support for having remote login access to QX devices through the cloud based ecMON application. This will allow remote access even to QX units behind a firewall and having a private IP.
	New **Automatic Outbound Calling** feature to automate the outbound dialing and contact with customers. The contacts are called from a web-based application and this 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.
	Paging service enhancements. A new **Store and Forward Call** option is added in the settings for paging groups and provides three enhancements: • 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.
	In addition to **iQall mobile toggling** new advanced licensable features have been added to the iQall application, such as: • voice mail push notifications • missed calls push notifications A special license key is not needed for these new functions. The license key for **iQall Advanced Features** will cover all three.
	Added Receptionist support for Cisco CP-8851 with expansion modules 8800 and 8851/61.
	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.
	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.
	Added IP Phones Logo support for Cisco CP-8851 and Polycom VVX1500 IP phones.
	Added autoconfiguration support using OpenVPN service for the Fanvil phones X5S and X6.
6.2.23	
6.2.22	Added auto configuration support for the new **Cisco CP-6851**, **CP-7861** and **CP-8851** IP phones.
	Added PnP support for **Htek UC902, UC903, UC912G, UC912P, UC923, UC924, UC924E, UC926** and **UC926E** IP phones.

Release	New Features
	Added auto configuration support for the new **iServ 8660 (8430/8630/8830)** Dect phone.
	Added support for the **Twilio** service as a new **SMS Gateway**. Use **Twilio** API for sending notifications via SMS.
	Added **P-Asserted Identity** (PAI) support for Polycom IP phones. The **PAI** option is configurable from **IP Phone Templates**. It is enabled by default.
	Added support for the **Use Session Progress** option instead of **Ringing** for the external calls.
	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.
	Enhancements in **EAC WEB application**: • 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 6.2.45. • The **Dashboard** menu has been completely modified. For FW version 6.2.45 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 LAN and VLAN interfaces.
	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.
	Added **Htek UC46 (LCD)** expansion module support for **Htek UC924, UC924E, UC926** and **UC926E** IP phones.
	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).
	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.
	Added a new **Allow Concurrent Calls to Parent-Child Group** option, allowing to control handling of calls 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.
	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.

Release	New Features
	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 Response**s (6xx messages).
	Added possibility to access QX50/QX200 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 **VVX 301/311** and **VVX 401/411** IP phones.
	Added **Phone Book** service support for Polycom phones.
	Added **Watching – Call Interception** support for Fanvil phones.
	Added support for **SNMP v3**.
	Added support for **TLSv1.1** and **TLSv1.2**.
	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. **Note:** • 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.
	Added support in **EAC** to show the Agent Status Statistics, Queue Statistics in hourly/minutely basis, as well as to generate Reports for selected **Hours** and **Minutes**.
	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).
	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.
	Added a new **Billed Extension** column in the **Call History** pages to provide information about the extensions that are charged for the calls.
	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.
6.2.1	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.
	Added PnP and auto configuration support for the new **Fanvil H2/H2S, H3, H5, X3S/X3G, X5S** and **X6** IP phones.
	Added PnP and auto configuration support for the new **snom D745** and **Akuvox R15(P)** IP phones.
	Added auto configuration support for the new **Htek UC924** and **UC926** IP phones.
	Added PnP and auto configuration support for the new **Panasonic KX-HDV130** and **KX-HDV230** IP phones.
	Added a checking mechanism for emergency codes: • Prevents an emergency code from being added if there is already an extension with the same number. • Prevents an extension from being created if there already exists an emergency code with the same number.
	Added a new **Call Completion Fee** option in the **Calling Cost Control** allowing to calculate call cost per the number of completed calls.

Release	New Features
	Added new **Auto Reload Queue Statistics** option in EAC settings allowing to automatically reload (refresh) Queue Summary pages.
	Added a new option for **Agent status by Queue, by Date (Summary)** report in EAC which allows to show the status duration in percentage.

4 Changed Features

The table below provides a high-level list of changed features that have been changed beginning with the most recent QX50/QX200 FW release.

Release	Changed Features
6.2.45	The status of the "Force Hold Music" option in the **generalconfig.cgi** is changed to "enabled" by default.
	The behavior has been changed for the estimated serving time calculation in ACD queue: • The calculation is based on the ACD call statistics during the time period defined by user in the QX GUI. • For this purpose, a new field under the ACD Queue settings is added and named as "Average serving time calculation period".
	The extension number "the call was blindly transferred to" is shown together with caller name after the call was "kicked back" to the receptionist.
	The distinctive ring pattern has been changed on Yealink phones running the FW V.83. The fix configures the phone in such a way that the phone plays its own ring tones instead of the Bellcore ones.
	The method of the configuration restore from the ecMON has been changed. The network related configuration files on "configuration restore" from ecMON do not changed.
	The SIP proxy server has been changed in the **VoIP Carrier Wizard** from the **link.voipvoice.it** to **user.voipvoice.it** for the ITSP **voipvoice.it**.
6.2.40	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 on 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.
6.2.35	The recommended FW version has been changed for most of the supported Fanvil phones: • For C62/C62P from 2.3.235.128 to 2.5.787.97 • 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/X5G 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	

Release	Changed Features
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: • 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.
	The configured **SRTP policy** of PBX extension will be provided to the Yealink IP phones during configuration.
	The configured **DTMF parameters** of PBX extension will be provided to the IP phones during configuration.
6.2.11	The maximum length of **API ID** field for **Clickatell** SMS Gateway has been increased up to **128** symbols.
	The recommended FW version has been changed for some of the supported **Htek** phones. For **UC924** and **UC926** it is increased from 2.0.4.2.24 to 2.0.4.4.33.
	GUI Enhancements for the following pages: • Admin Settings of the extensions (user, auto attendant, etc.) • User Settings of the extensions • Call History • Conference History
	Depending on the IP phone model, the **Use Session Timer** option will be enabled for the configured IP line.
	The number of concurrent G729 calls has been increased for QX50 from **4** to **10** and for QX200 from **6** to **10**.
	The default **Line Appearance** has been increased from **2** to **5** for each IP line.
6.2.6	**Network** and **Broadcast** IP addresses will not be included into **Usable Host IP Range**. These IPs will be reserved for network purposes.
6.2.5	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.
	The recommended FW version has been changed for **Yealink CP860** from 37.80.0.30 to 37.81.0.10.
	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 **VVX 300/310, VVX 400/410, VVX 500** and **VVX 600** from 4.1.7.1210 to 5.7.0.11768, for **VVX 1500** from 3.3.5.0247 to 5.7.0.11768.
	The first programmable key on **Polycom** phones is reserved for the phone account.

Release	Changed Features
	The **Call Quality Warning** in the System Events has been modernized to show the information about the callee, caller and call date/time.
	GUI enhancements for the **Call History** and **Conference History** pages.
	Redundancy feature has been redesigned to allow sending VRRP packets to unicast IP address (virtual IP) of the slave device. **Note:** Having VRRP packets sent via unicast IP address will allow the Redundancy feature to work in a scenario when the master and backup devices are located in different places, meaning that there is a router between the master and the backup.
6.2.1	The **PSTN Gateways Line Sharing** mechanism has been changed and updated, bringing more stability, and improving the connection between PBXs and Gateways. **Important Note:** Please update the firmware version to **6.2.1** both on QX50/QX200 and QX Gateway(s) to be able to successfully connect the devices and share the lines.
	Added option allowing to share and synchronize the configured Incoming Interdigit Service settings with QXISDN4 and QXE1T1 gateways when connected with QX50/QX200 in the shared mode.
	The allowed duration of recorded voice mail sent as an 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.
	The timezone database has been updated on QX IP PBXs: • The current local time has been corrected for Israel, Venezuela, Sri Lanka, Apia, Samoa and Fiji. • Added new timezone Nukualofa, Tonga (GMT+14).
	New **Date/Time** pickers have been implemented for all applicable GUI pages, allowing to select or define the date/time options easier and conveniently.
	Enhancements for the **Call History – Call Cost** page: • Added new filtering options supporting multi-criteria search for payable call records. • Added support to download the displayed CDRs in the (*.log) and (*.csv) formats respectively.
	The recommended FW version has been changed for **Yealink SIP-T40P** from 54.81.0.25 to 54.81.0.110.
	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.
	Panasonic KX-UT123 and **KX-UT123NE** IP phones have been merged and renamed as **KX-UT123 (NE/RU/X)**.
	Panasonic KX-UT136 IP phone has been renamed as **KX-UT136 (NE/RU/X)**.
	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.
	Akuvox SP-R53P/SP-R53 IP phone has been renamed as **Akuvox SP-R53(P)**.
	The recommended FW version has been changed for the **Akuvox SP-R53(P)** IP phone from 53.0.1.23 to 53.0.6.115.
	The **Emergency number** configured on the QX IP PBX will be added in the provisioning file for **snom** IP phones.
	Added **nexogy**, **ClarityTel** and **Adiptel** as the new carriers on the VoIP Carrier Wizard list.
	Added a new option allowing to select Conference extensions from the **Unconditional**, **Busy**, **No Answer** and **Unregistered** Call Forwarding lists.
	The default TLS port number (5061) will be selected for SIP.

5 Fixed Issues

Issues fixed since version 6.2.40

T: Title
D: Description

20514	T:	**The FM/FM is calling to the preconfigured No Answer call forwarding destination regardless if the forwarding is enabled or not**
	D:	
20498	T:	**An issue regarding the phone book update from QX GUI**
	D:	The new contacts added to the phone book from QX GUI was not shown on the IP phones if the phones were rebooted manually
20491	T:	**The call type wouldn't save in the reporting rule of ACD/EAC**
	D:	Create a scheduled reporting rule for CDRs by Queue, Call Type by Date. Select any call type for the report rule and save. After reopening the rule for edit, the call type was blank.
20490	T:	**The ACD/EAC scheduled report for the abandoned calls was always empty when the file was sent via email**
	D:	
20484	T:	**Showed the DID (name) for the specific ACD queue instead of "system" when ACD callback**
	D:	
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, was not updated correctly after powering off/on the phone. If you turned off the phone and turned it on again, the MPK on the phone would always show the default logout state.

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

20471	T:	**An issue concerning the Hot Desking Automatic Logout → "After the call" option**
	D:	The Hot Desking extension won't logout automatically after making a call if you just select the "After the Call" option and press Save. You have to logout/login and additionally, use 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.
20361	T:	**In some cases, call to Htek phone cannot be established when using the OPUS codec**
	D:	In some scenarios call to Htek phone with OPUS is being dropped due to the media parameters incompatibility, in spite of the fact OPUS is configured on both parties correctly. Example, when calling to Htek from an FXS phone. No such problem when calling from Htek to FXS phone. The source of the problem is the method how Htek is processing the name for dynamic codecs, specifically for OPUS. The Htek doesn't perform case-insensitive checking for the codec's name and it brings about this problem. If the codec's name matches with lowercase "opus", all is OK, otherwise the phone tries to recognize the codec by the payload type. The phone wants the value for the payload type to be either internally defined 120 or match with one what is configured on the phone. On the other hand, the value of payload type for opposite party is not obligated to match with the value for Htek. That is why sometimes Htek works with "OPUS" OK, sometimes not.
	C:	Call is being dropped due to the media parameters incompatibility, in spite of the fact that OPUS is configured as the preferred codec for both sides.
	Fix:	Should be fixed by Htek in future release.
20359	T:	**In case of IP line to FXS calls with OPUS the codec is always incorrectly shown as "N/A" in Call History details**
	D:	
	C:	No consequences.
	Fix:	Will be fixed in future release.
20074	T:	**Fanvil IP Phones have issues 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 Fanvil 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 restoring a config backup downloaded from the QX with custom language Pack (LP) is installed
	D:	Scenario that cause this problem: • 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 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.
	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 QX 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.
18577	T:	The voice traffic is not encrypted when using IPSec connection between two QX devices
	D:	
	C:	
	Fix:	Will be fixed in the next release.
18549	T:	Could not dial out (*1) or use any other moderator feature while welcome message has been playing
	D:	Could not dial out (***1**) or use other moderator features while welcome message has been playing. You should listen to the whole welcome message 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 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.

7   General Hints

7.1   Technical Advisory

- 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.
- Always power down the QX50/QX200 before inserting/removing SD memory card.
- Some system information (Call History, Conference History and Pending Events) may be lost when QX50/QX200 is powered down. You may maximum lose the portion of the above-mentioned system information, which occurred during last hour before the QX is powered down. It's recommended to enable **Call History – Archiving** to minimize the loss of Call History.

7.2   Firmware Update

It is recommended to execute the update by downloading the firmware first to a PC located in the LAN side of the QX50/QX200 and perform the firmware update from the LAN side. This is to ensure that the Internet connection will not affect the upgrade process.

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.

7.3   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 **DSP Capture** duration is limited to **160** seconds.
- 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 **300** voice mails for each extension.