Release Note QX2000 6.1.50 Edition 1

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1 Introduction

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

QX50/QX200 firmware 6.1.50 Date: June 14, 2017

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

Date: June 16, 2017

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2 Requirements

2.1 Hardware Requirements

The firmware (FW) can be used on QX2000 model only.

2.2 Firmware Requirements

Attention: The firmware upgrade to **6.1.50** 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 SW (FW) versions that are tested and recommended for use with QX2000 FW 6.1.50. All the phones in this list can be automatically configured to work with QX2000 FW 6.1.50.

Note:

- QX2000 FW 6.1.50 supports also the Plug-and-Play (PnP) option for most IP phones. The configuration options for each specific IP phone are described in detail in the Configuring Epygi Supported IP Phones guide.
- Any known issues and limitations regarding the usage of the QX2000 FW 6.1.50 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 Support
Akuvox	SP-R53P	53.0.1.23	Yes
Alcatel	IP2015 (IP15)	1.0.7A-0	No
Alcatel	Temporis IP100	1.0.6A-0	No
Alcatel	Temporis IP150	1.0.6A-0	No
Alcatel	Temporis IP200	13.60.0.89	Yes
Alcatel	Temporis IP300	1.0.7B-0	No
Alcatel	Temporis IP600	14.60.0.89	Yes
Alcatel	Temporis IP700G	1.0.7A-0	No
Alcatel	Temporis IP800	15.60.0.89	Yes
AudioCodes	310HD	1.6.0_build_37	No
AudioCodes	320HD	1.6.0_build_37	No
Cisco	SPA303	7.4.9c	Yes
Cisco	SPA501G	7.4.9c	Yes
Cisco	SPA509G	7.4.9c	Yes
Cisco	SPA525G2	7.4.9c	Yes
Fanvil	C58/C58P	2.3.233.129	No
Fanvil	C62/C62P	2.3.235.128	No
Fanvil	C400	11.20.12.2.B	No
Fanvil	C600	11.20.12.2.B	No
Fanvil	F52/F52P	2.3.123.78	No
Fanvil	X3/X3P	1.3.511.1821	Yes
Fanvil	X4/X4G	2.0.2.2830	Yes
Fanvil	X5/X5G	1.3.511.1821	Yes
Grandstream	GXP1100	1.0.8.6	Yes
Grandstream	GXP1105	1.0.8.6	Yes
Grandstream	GXP1160	1.0.8.6	Yes

Vendor	Model	SW/FW Version	PnP Support
Grandstream	GXP1165	1.0.8.6	Yes
Grandstream	GXP1400	1.0.8.6	Yes
Grandstream	GXP1405	1.0.8.6	Yes
Grandstream	GXP1450	1.0.8.6	Yes
Grandstream	GXP1615/1610	1.0.4.55	Yes
Grandstream	GXP1625/1620	1.0.4.55	Yes
Grandstream	GXP1628	1.0.4.55	Yes
Grandstream	GXP1630	1.0.4.55	Yes
Grandstream	GXP1760	1.0.0.48	No
Grandstream	GXP1782/1780	1.0.0.48	No
Grandstream	GXP2100	1.0.8.6	Yes
Grandstream	GXP2110	1.0.8.6	Yes
Grandstream	GXP2120	1.0.8.6	Yes
Grandstream	GXP2124	1.0.8.6	Yes
Grandstream	GXP2130	1.0.7.99	Yes
Grandstream	GXP2135	1.0.7.99	Yes
Grandstream	GXP2140	1.0.7.99	Yes
Grandstream	GXP2160	1.0.7.99	Yes
Grandstream	GXP2170	1.0.7.99	Yes
Grandstream	GXP2200	1.0.3.27	Yes
Grandstream	GXV3140	1.0.7.80	Yes
Grandstream	GXV3175	1.0.3.76	Yes
Grandstream	GXV3240	1.0.3.62	Yes
Grandstream	GXV3275	1.0.3.62	Yes
Mitel (Aastra)	6730	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6731	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6735	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6737	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6739	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6753	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6755	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6757	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9480	3.3.1.4305-SIP	Yes
Mitel	6863	4.2.0.2023-SIP	Yes
Mitel	6865	4.2.0.2023-SIP	Yes
Mitel	6867	4.2.0.2023-SIP	Yes
Mitel	6869	4.2.0.2023-SIP	Yes
Panasonic	KX-TGP550T04	12.17	No
Panasonic	KX-UT123	01.061	No
Panasonic	KX-UT123NE	01.221	No
Panasonic	KX-UT136	01.061	No
Polycom	SoundPoint IP 330SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 331SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 335SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 450SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 550SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 650SIP*	3.3.5.0247	Yes
Polycom	SoundPoint IP 670SIP*	3.3.5.0247	Yes
Polycom	SoundStation IP 5000*	3.3.5.0247	Yes
Polycom	SoundStation IP 6000*	3.3.5.0247	Yes
Polycom	VVX 1500*	3.3.5.0247	Yes

Vendor	Model	SW/FW Version	PnP Support
Polycom	VVX 300/310	4.1.7.1210	Yes
Polycom	VVX 400/410	4.1.7.1210	No
Polycom	VVX 500	4.1.7.1210	No
Polycom	VVX 600	4.1.7.1210	Yes
QOSIP	Q7104/Q7204	1.0.3.98	No
snom	300	8.4.35	Yes
snom	320	8.4.35	Yes
snom	360	8.4.35	Yes
snom	370	8.7.5.35	Yes
snom	720	8.7.5.35	Yes
snom	760	8.7.5.35	Yes
snom	821	8.7.5.35	Yes
snom	870	8.7.5.35	Yes
snom	D345	8.9.3.35	Yes
snom	D375	8.9.3.35	Yes
snom	D710/710	8.7.5.35	Yes
snom	D715/715	8.7.5.35	Yes
snom	D725	8.7.5.35	Yes
snom	D765	8.7.5.35	Yes
snom	m9	9.4.7	Yes
snom	MeetingPoint	8.7.5.35	Yes
snom	M700 (M85/M65/M25)	03.24.0007	Yes
Spectralink	KIRK Wireless Server 300	PCS14C_	No
Spectralink	KIRK Wireless Server 6000	PCS14C_	No
VTech	ErisStation VCS754	1.1.4.0-0	No
VTech	ErisTerminal VSP600 (VSP601)	1.1.4.1-0	No
VTech	ErisTerminal VSP715	1.1.4.0-0	No
VTech	ErisTerminal VSP725	1.1.4.0-0	No
VTech	ErisTerminal VSP726	2.0.3.2-0	Yes
VTech	ErisTerminal VSP735	1.1.4.0-0	No
VTech	ErisTerminal VSP736	2.0.3.2-0	Yes
Yealink	CP860	37.80.0.30	Yes
Yealink	SIP-T19P	31.72.0.1	Yes
Yealink	SIP-T19P E2	53.81.0.25	Yes
Yealink	SIP-T20P	9.72.0.1	Yes
Yealink	SIP-T21P	34.72.0.1	Yes
Yealink	SIP-T21P E2	52.81.0.25	Yes
Yealink	SIP-T22P	7.72.0.1	Yes
Yealink	SIP-T23G(P)	44.81.0.25	Yes
Yealink	SIP-T26P	6.72.0.1	Yes
Yealink	SIP-T27G	69.81.0.25	Yes
Yealink	SIP-T27P	45.81.0.25	Yes
Yealink	SIP-T28P	2.72.0.1	Yes
Yealink	SIP-T29G	46.81.0.25	Yes
Yealink	SIP-T32G	32.70.0.130	Yes
Yealink	SIP-T38G	38.70.0.125	Yes
Yealink	SIP-T40P	54.81.0.25	Yes
Yealink	SIP-T41P	36.81.0.25	Yes
Yealink	SIP-T41S	66.81.0.25	Yes
Yealink	SIP-T42G	29.81.0.25	Yes
Yealink	SIP-T42S	66.81.0.25	Yes
Yealink	SIP-T46G	28.81.0.25	Yes

Vendor	Model	SW/FW Version	PnP Support
Yealink	SIP-T46S	66.81.0.25	Yes
Yealink	SIP-T48G	35.81.0.25	Yes
Yealink	SIP-T48S	66.81.0.25	Yes
Yealink	SIP VP-T49G	51.80.0.100	Yes
Yealink	VP-530	23.70.0.40	Yes
Yealink	W52P	25.30.0.20	Yes

Note: In the model's list the Polycom phones with (*) sign are also presented as Polycom-xx-Pre-3.3.0 due to backward incompatibility of UC Software 3.1.1 configuration. It is recommended to use Pre-3.3.0 models with Application SIP software 3.2.2.0477.

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 6.1.50:

- QXISDN4, QXE1T1 or QXFXO4 gateways used in the shared mode should have FW 6.1.17 or higher.
- QXFXS24 should have FW 6.1.40 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 1.0.4 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 Tool version 1.0.17 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 FW 6.1.50 with a 3PCC or Click2Dial application, the 3pcc/Click2Dial Access Allowed checkbox should be enabled for each extension(s) 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.

<table>
<tr><th>Release</th><th>New Features</th></tr>
<tr><td rowspan="12">6.1.50</td><td>Added PnP and auto configuration support for the new Grandstream GXP1615, GXP1628, GXP1630, GXP2135 and GXP2170 IP phones.</td></tr>
<tr><td>Added auto configuration support for the new Grandstream GXP1760 and GXP1782/1780, IP phones.</td></tr>
<tr><td>Added PnP support for the Grandstream GXP1610 and GXP1625/1620 IP phones.</td></tr>
<tr><td>Added PnP and auto configuration support for the new Mitel 6869 IP phone.</td></tr>
<tr><td>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.
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.</td></tr>
<tr><td>Configuration Management enhancements
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.</td></tr>
<tr><td>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.</td></tr>
<tr><td>Added the SSH FTP (SFTP) support, which allows to send the configuration backup files to an FTP server using the secure FTP connection.</td></tr>
<tr><td>Added a new "Archive Now" option on the Call History – Archiving Settings page, allowing to archive immediately the available data.</td></tr>
<tr><td>Added new Reporting types in EAC: CDRs by Agent, by Queue, by Date and CDRs by Queue, by Agent, by Date.</td></tr>
<tr><td>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.</td></tr>
<tr><td>The Client Code Identification option can be activated and used by other billing systems as well as it is done for RADIUS server.</td></tr>
<tr><td rowspan="3">6.1.45</td><td>Added PnP and auto configuration support for new Yealink SIP-T27G, SIP-T41S, SIP-T42S, SIP-T46S and SIP-T48S IP phones.</td></tr>
<tr><td>Added new Media Streamer service allowing to upload and stream uploaded audio files to the RTP destinations from the QX2000.</td></tr>
<tr><td>Added "Firmware Version Control" support for Yealink phones to upgrade or downgrade the phones to the Epygi recommended firmware version.</td></tr>
</table>

Release	New Features
	Call Recording feature enhancements: • Improved wildcard character support for the **Call Recording** rules. • Added support to move the call recording entry up/down by keeping the checkbox selected.
	Added support allowing to **Restrict Simultaneous Calls** for "**SIP**" call types.
	GUI improvements and enhancements in the **Extensions Management** page.
	GUI enhancements on the **Menu** bar: • Added device's current Date/Time. • Added device's hostname.
	General improvements and enhancements in the **SIP TLS certificate.**
6.1.41	Added the new **Priority** option in the Redundancy settings, which allows the particular device to always run as a **Master**.
6.1.40	Added PnP and auto configuration support for the new **snom D345** IP phone.
	Added PnP and auto configuration support for the new **Vtech ErisTerminal VSP726** and **ErisTerminal VSP736** IP phones.
	Added auto configuration support for the new **VTech** phones: **ErisTerminal VSP715, ErisTerminal VSP725, ErisTerminal VSP735** IP phones, **ErisStation VCS754** conference phone and **ErisTerminal VSP 600 (VSP601)** DECT phone.
	Uploading audio files for customizing any of the system or extension audio messages on the QX2000 has been simplified: • Apart from the files in the (***.wav**) format, the system can now accept (***.mp3**) files for uploading as custom messages. • The (***.wav**) and (***.mp3**) files can now be uploaded directly to the system without the need to convert to the proper telephony format. The uploaded files will be automatically converted to the QX supported wav format: (**CCITT u-law, 8 kHz, 16-bit, Mono**).
	Dial & Announce service enhancements: • Added support to allow configuring the repetition count for the **Announcement Message.** • Added support to allow configuring the silence duration between consecutively played **Announcement Messages.**
	Add the **Backup Device GUI Access Port** option on **Setup→Redundancy** page. This option allows to access **Backup** device WEB GUI by entering the IP address of **Master** device followed by **Backup Device GUI Access Port.**
	GUI improvements and enhancements in the **EAC** application.
6.1.35	Added support for the new **Day/Night Switching** service for the Scheduling feature. This service allows the state of the schedules to be manually controlled by either using the MPK keys on the phones or the new feature code ***86**, available also remotely on the QX Auto Attendant. Schedules are assigned Schedule IDs. The **Schedule IDs are assigned** to MPK keys in the **IP Lines→Programmable Keys Configuration** page. **Attention:** After FW update to 6.1.50 the **Schedules** comes up with blank **Schedule IDs**. Thus, the **Day/Night Switching** service won't work until the schedule(s) are updated with a **Schedule ID**. Therefore, after FW update select schedules one by one in the **Telephony→Schedules** page, edit and update with **Schedule IDs**.
	Added PnP and auto configuration support for the new **Yealink SIP-T40P** IP phone.
	Added PnP and auto configuration support for the new **Fanvil X4/X4G** IP phone.
	Added support allowing the QX2000 to act as an **OpenVPN** client.

Release	New Features
	Added **OpenVPN** support for the following IP phones: • snom 370, 720, 760, 821, 870, D375, D710/710, D715/715, D725, D765 IP phones and snom MeetingPoint Conference phone. Note: OpenVPN support will only work for the phone after applying **small patch firmware** (http://wiki.snom.com/VPN_Support). • Grandstream GXV3240 and GXV3275 Video phones. • QOSIP Q7104/Q7204 IP phone.
	Added **Use OpenVPN Settings** option for **IP Lines Settings**. Select this option when you want to configure IP Phone using OpenVPN settings. Note: This option is available only for some phones.
	Added **Symmetric RTP** support for IP Lines. Select this option when the IP phone attached to the IP line is behind the NAT router.
	ACD & **EAC** enhancements: • Added support to display information about the agents logged into EAC on **Extensions→ACD→EAC Status** page. • Added support to terminate the EAC session for the logged in agent(s) from QX GUI. • Added support to display "**Wait Time in Queue**" for each call in EAC. • Added support to select "**Duration Format**" from **EAC→Settings** page for EAC reports.
	Dial & Announce service enhancements: • Added support to allow the **Dial & Announce** service to be automatically activated on a schedule. • Added a new option to interrupt the active call and play an announcement to the users in the dial & announce list. This option is available for **QX extensions** only. • Added **Auto Answer** option to page the extension's IP phone by forcing the phone to go off-hook and play the announcement. This option is available for **QX extensions** only.
	GUI enhancements for **Voice Mail Settings** and **Profiles for Voice Mail Settings** pages.
6.1.30	Added PnP support for the **Yealink SIP-T19P E2**, **T21P E2**, **T23G(P)**, **T27P** and **T29G** IP phones. Note: PnP option will only work for the phones running **x.80.0.130** or higher firmware.
	Added **EXP20** expansion module support for the **Yealink SIP-T27P** and **T29G** IP phones.
6.1.29	Added PnP and auto configuration support for the new **Yealink SIP VP-T49G** video phone.
6.1.28	
6.1.27	Added PnP and auto configuration support for the new **snom D375, D710** and **D765** IP phones.
	Added **snom D7** expansion module support for **snom 720, 760, D715/715, D725** and **D765** IP phones.
	The **Schedules** feature is enhanced to be applicable except of the call routing to the auto attendant scenarios as well.
	GUI enhancements for IP Lines page: • Added support to reboot multiple number of IP Phones at once. • Added support to allow quicker edits when moving between IP Lines. • Added support to allow quicker access to the attached extension's **Admin** and **User** settings.
	The GUI is enhanced to allow quicker access to the extension's **Admin** and **User** settings.
6.1.25	Added support for the **SIP Registration Transport** UDP/TCP/TLS options in the Extension's SIP Registration.

Release	New Features
	Added the new **Schedules** feature to define schedules and apply them to the call routing rules. The scheduling allows different scenarios for scheduled periods such as working hours, non-working hours and holidays.
	Added auto configuration support for the new QOSIP Q7104 IP phone.
	Added 3-way conference support for Mitel 6863.
	The **Universal Extensions Recording** list is updated with the **Find Me/Follow Me Welcome Message,** allowing the message to be configured for all extensions at once.
	The GUI is enhanced to allow quicker edits when moving between extensions. The update is added in the **Extension General Settings** page and also for the **Extension User Settings**, such as the Caller ID based services, voice mailbox, etc.
	Added support to configure the **Forward/Rewind duration** for Recording Box Extension.
	Added a new **Dial & Announce** service in the list of Caller ID based services. This service allows simultaneously calling to the predefined list for up to 32 destinations and play the announcement audio message when the destinations answer the call.
	Added support (with a license key) for connecting QX2000 to **PMSLINK** middleware from char (software solution partner). This allows integrating the QX2000 with the PMS systems used in hotels.
	Added PnP and auto configuration support for the Grandstream GXV3240 and GXV3275 IP phones.
	Added a new **Dial & Announce** service that allows to configure simultaneously calling to the predefined list of destinations with the option of playing audio messages on the incoming call from the certain caller.
	Added the new **Reports Scheduling** feature for the **EAC** application that allows the reports to be automatically generated, then stored on an FTP server and/or delivered by e-mail.
	Added a new feature to allow **Call Recordings** from the **EAC** application to be downloaded and played.
	Added the **SSH FTP** (**SFTP**) support, which allows to send the call recordings and call history archive files to an FTP server using the secure FTP connection.
	Added **Ignore Push Routes** option for OpenVPN client configuration. If disabled, the client side will accept push route commands from the server side, which allows an OpenVPN client to reach the QX's LAN side.
	Added support to upload a custom logo for the IP phones: Yealink SIP-T19, T19 E2, T21, T21 E2, T23G(P), T27P, T29G, T41P, T42G, T46G, T48G and CP860 conference phone.
	Added a **Collect Call** option for shared ISDN and E1/T1 trunks in the call routing wizard.
	Added a new **Search** option in the QX **Online Help**.
6.1.17	
6.1.15	Added auto configuration support for the new Grandstream GXP1610 and GXP1620/ GXP1625 IP phones.
	Added auto configuration support for the new Yealink SIP-T29G IP phone.
	Added PnP and auto configuration support for the new Yealink CP860 Conference phone.
	Added PnP support for the Fanvil X3/X3P and X5/X5G IP phones.
	Added **RTSP** support, which allows live media streaming from RTSP server to the video phones.
	Added **OpenVPN** support for Yealink phones.
	Added the **Disable DND Button** option for Grandstream GXP14xx series, GXP2110, GXP2120 phones in IP Phone templates.
	Added the **Blink message LED on ringing** option for Grandstream GXP110x, GXP116x, GXP140x series and GXP1610 phones in IP Phone templates.

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
6.1.50	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.
	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.
	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.
	The HTML5 **Date/Time picker** is implemented for Date/Time selection.
	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**
	Added option allowing to display Media Streamer's allocated and used memory space on the **Status→System Status→Memory** page.
	Added new option allowing to select and change **Schedule State** from WEB GUI.
	The **Network Capture** page has been moved to **Maintenance→Diagnostics→Network Capture** page.
	GUI Enhancements for **Call Routing Table**.
	GUI Enhancements on the **Setup→Licensed Features** page.
	GUI Enhancements for **IP Phone Templates**.
6.1.45	The recommended FW versions have been changed for some **Yealink** phones. For **CP860** from 37.80.0.10 to 37.80.0.30, for **T19 E2** from 53.80.0.130 to 53.81.0.25, for **T21 E2** from 52.80.0.130 to 52.81.0.25, for **T23G** from 44.80.0.130 to 44.81.0.25, for **T27P** from 45.80.0.130 to 45.81.0.25, for **T29G** from 46.80.0.130 to 46.81.0.25, for **T40P** from 54.80.0.130 to 54.81.0.25, for **T41P** from 36.80.0.130 to 36.81.0.25, for **T42G** from 29.80.0.130 to 29.81.0.25, for **T46G** from 28.80.0.130 to 28.81.0.25 and for **T48G** from 35.80.0.130 to 35.81.0.25.
6.1.41	Added the "**Electronic Hookswitch**" option for Polycom IP phones into the **IP Phone Templates**. When you use a headset that supports electronic hookswitch (EHS), you can place, answer, and end calls by using controls on your headset.
6.1.40	Changes in behavior for the **Parent-Child** extensions configuration: • If any of the extensions in the **Parent/Child** group are busy, then entire group will be considered busy, therefore the incoming call will follow the busy state rules (busy forwarding, call queue, VMS, etc.) 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. • If all extensions in the **Parent/Child** group are free (not busy) and are ringing, and any of them presses the "**Reject**" button (or somehow declines the incoming call), then the entire group will be considered as busy. Therefore, incoming calls will follow busy state rules (busy forwarding, queue, VMS, etc.) depending on what is configured.

Release	Changed Features
	Added the **Transfer on Conference Hang Up** option for Grandstream IP phones into the **IP Phone Templates**. This allows the conference initiator to be disconnected from the conference while keeping the other two parties in the call.
	Added **MC-Link** as a new carrier to the VoIP Carrier Wizard list.
	Added **Flowroute** as a new carrier to the VoIP Carrier Wizard list.
6.1.35	The recommended FW versions have been changed for some **Yealink** phones. For **C860** from 37.72.0.10 to 37.80.0.10, for **T41P** from 36.72.0.1 to 36.80.0.130, for **T42G** from 29.72.0.1 to 29.80.0.130, for **T46G** from 28.72.0.1 to 28.80.0.130 and for **T48G** from 35.72.0.34 to 35.80.0.130.
	The max number of **Line appearance** has been increased for some **Yealink** phones. For **T29G** from **10** to **16**, for **T41P** from **6** to **10**, for **T42G** from **6** to **12**, for **T46G**, **T48G** and **T49G** from **10** to **16**.
	The recommended FW versions have been changed for some **snom** phones. For **MeetingPoint**, **370**, **821** and **870** from 8.4.35 to 8.7.5.35.
	QOSIP Q7104 IP phone has been renamed to **QOSIP Q7104/Q7204**.
	The recommended FW version has been changed for **QOSIP Q7104/Q7204** from 1.0.3.97 to 1.0.3.98.
	The default **Digitmap** for Polycom phones has been changed. • Added support allowing user to dial out automatically after **Digitmap Timeout**, without pressing # sign or **Send** softkey. • Added support allowing to accelerate dial out (don't wait for **Digitmap Timeout**) by pressing # sign.
	Programmable key support (DSS Keys) has been increased up to **37** keys for **Fanvil X5/X5G**.
	Hot Desking feature enhanced regarding the voice notifications when login/logout on the public phones: • Added voice prompt asking user to login before using the phone. • Added voice prompt notifying user about login extensions in use. • Added voice prompt informing the user about the successful logging out.
	Added support to allow/deny access to the **Diagnostics** and **Reboot** pages for QX localadmin.
	The **Status→System Status→Memory** page is redesigned and modernized.
	The **Subscriptions Count** section (**Status→System Status→IP Line Registration** page) is enhanced to display correct and informative data for used/allowed subscriptions.
6.1.30	The recommended FW versions have been changed for some **Yealink** phones. For **T19P E2** from 53.80.0.70 to 53.80.0.130, for **T21P E2** from 52.80.0.70 to 52.80.0.130, for **T23G(P)** from 44.80.0.70 to 44.80.0.130, for **T27P** from 45.80.0.70 to 45.80.0.130 and for **T29G** from 46.80.0.70 to 46.80.0.130.
	Added the option to enable/disable the **Welcome Message** in the menus for Auto Attendant **Custom** scenario.
	Changes in Auto Attendant behavior for the **Custom** scenario: • If the **Welcome Message** is not specified, then the Welcome Message for **Standard** scenario will be played. • If the **Recurring Prompt** is not specified, then the Recurring Prompt for **Standard** scenario will be played.
	Enhancement in Auto Attendant for **ACD** scenario: No need to enter **ACD Queue ID** for changing **Agent** status.

Release	Changed Features
	The **EXP38** expansion module support has been discontinued for the **Yealink SIP-T27P** and **T29G** IP phones.
6.1.29	
6.1.28	
6.1.27	The ACD Call Detail Records archiving is removed. The old records will not be archived and removed from EAC, instead they will be moved gently to another database within QX. Thus, allowing to increase the stability and the speed of ACD and EAC systems.
	Enhancements in **ACD Scheduling Reports**: The system will try to send scheduled reports to FTP server or e-mail address periodically in case of failures: 1. 4 attempts within 20 minutes 2. 4 attempts within 4 hours 3. 18 attempts within 72 hours After each period, an event will be generated in **Status→System Events** about the failure. If the scheduled reports failed to send within 72 hours, they would be deleted.
	Calls addressed to ACD Agent cannot be intercepted by another agent or extension.
	The recommended FW versions have been changed for some **snom** phones. For **710**, **720** and **760** from 8.7.3.25.9 to 8.7.5.35, for **D715/715** and **D725** 8.7.5.17 to 8.7.5.35.
	The maximum count of recordings has been changed from **1000** to **10000**.
	Added support to configure **Voice Mail Profiles** when logged in as an **Extension**.
	Added **MO=1** parameter in the **SMS Settings**.
	Added support for the following symbols "<", ">" in the password field for **E-mail Settings**.
	Added support to download Extension's Call Detail Records for Successful, Missed and Unsuccessful Outgoing calls, when logged into the system using extension's credentials.
	Added support to exclude/include different **CDR parameters** in generated CDR reports for the Call and Conference History.
	Added support to display **SRTP** parameters in the Call and Conference History.
	Added **recording_id** as a new **File Naming Scheme** in Recording Storage Settings page.
	Added **Fusion** as a new carrier to the VoIP Carrier Wizard list.
6.1.25	**SIP IDS enhancement:** added a special rule in QX firewall configuration to drop the messages, to exclude the load on the system, in case of huge amount of invite messages from the sender's IP addresses. This rule is applied automatically only for the SIP messages (new established UDP and TCP ports), and limits their number according the criteria: max average match rate as 600 message/sec.
	Added support in **Extensions Multiple Editing** for the following fields: **Ringing Simulation** and **Ringing Simulation Timeout** from General Settings page.
	Added an option for each Queue to set the **Wrap-up** timeout for all Agents of that queue in common.
	ACD Agents will not receive calls from other Queues within their Wrap-up timeout. With the exception of Direct Inbound Calls, those will change the Agent status from **Wrap-up** to **Busy**.
	ACD Agent can make and receive direct calls, when his status is set to **Offline**, **Away** or any **User-defined** state.
	Added an option for each Queue to set the ACD Agents status to **Away** if the Agent(s) receives a call and doesn't answer within the **Agent Ring timeout**.
	ACD Agents can now receive calls from another queue(s) when busy on a call.
	Added support to download **Call Detail Records** in the (***.csv**) format for Successful, Missed and Unsuccessful Outgoing calls.

<table>
<tr><th>Release</th><th>Changed Features</th></tr>
<tr><td rowspan="4"></td><td>The recommended FW versions have been changed. For Grandstream GXP2200 IP phone from 1.0.3.25 to 1.0.3.27, for GXV3140 from 1.0.7.3 to 1.0.7.80 and for GXV3175 from 1.0.3.22 to 1.0.3.76.</td></tr>
<tr><td>The voice mail forwarding procedure on the handset is simplified. Now you can skip the accompanying message recording when forwarding a voice mail on the handset. Just press # twice quickly when prompted to record the accompanying message.</td></tr>
<tr><td>Added the new Clean IP Phone VLAN settings if no VLAN on PBX option in the generalconfig.cgi hidden page. This option allows to clear or leave unchanged VLAN settings manually configured on the LAN interface of IP phones.
When this option is enabled (default), the system will clear/remove all VLAN settings configured on the LAN interface of IP phones.
When this option is disabled, the system will leave unchanged all VLAN settings configured on the LAN interface of IP phones.
Note: The system doesn’t touch the PC port configuration of the phones.</td></tr>
<tr><td>The Blueface Ireland, Blueface Italy and Blueface UK carriers have been removed from the VoIP Carrier Wizard list.</td></tr>
<tr><td>6.1.17</td><td>Added support allowing to enable/disable entries in the Authorized Phones.</td></tr>
<tr><td rowspan="10">6.1.15</td><td>The recommended FW versions for Yealink SIP-T19 E2, T21P E2, T23G(P) and T27P IP phones have been changed from xx.80.0.60 to xx.80.0.70.</td></tr>
<tr><td>The recommended FW versions for Grandstream GXP2130, 2140, 2160 IP phones have been changed from 1.0.3.9 to 1.0.5.23.</td></tr>
<tr><td>The recommended FW versions for Grandstream GXP1100, 1105, 1160, 1165, 1400, 1405, 1450, 2100, 2110, 2120, 2124 IP phones have been changed from 1.0.6.7 to 1.0.8.6.</td></tr>
<tr><td>The recommended FW versions for Fanvil X3/X3P and X5/X5G IP phones have been changed from (1.3.221.1531 and 1.3.115.1425) to 1.3.511.1821 accordingly.</td></tr>
<tr><td>Added SoTel/VoIPLINK as a new carrier to the VoIP Carrier Wizard list.</td></tr>
<tr><td>If the IP phones are configured from VLAN side, the corresponding VLAN cannot be deleted or modified unless the interface for IP phones configuration is changed.</td></tr>
<tr><td>The user can make calls with clicktodial.cgi either using admin credentials or his own (username/password).</td></tr>
<tr><td>ACD agent's status will not be changed to Away if he/she is busy with another call and doesn't answer the second call (ACD call).</td></tr>
<tr><td>ACD agent will not receive a second call from the same queue if he/she is already in the call from the same queue.</td></tr>
<tr><td>ACD Agents Status Records archiving is removed.</td></tr>
</table>

5 Fixed Issues

Issues fixed since version 6.1.45:

T: Title
D: Description

19931	T:	The selected Tracing / Debug options aren't disabled, when you check off the "Tracing / Debug Options" from the Destination Call Type section
	D:	
19922	T:	Address update (modification) of the "Caller ID based Services" for new entries doesn't work properly
	D:	
19921	T:	The expiration/renewal isn't calculated correctly for the "Overall Call Duration Limit" service
	D:	
19917	T:	The codec information isn't shown correctly in the Call History – RTP Statistics page for calls with G726 codec
	D:	
19916	T:	Disabling FM/FM service in Caller ID based services for additionally added address, disables whole 3PCC (FM/FM, Dial&Announce) services of that extension
	D:	
19910	T:	Unable to add working time intervals for Sundays in Schedule configuration
	D:	
19902	T:	The "Firmware version control" service doesn't work on Mitel (Aastra) 6757, 9480 and Aastra 6757iCT, 480iCT IP phones
	D:	
19893	T:	The Date/Time is not displayed correctly on the Menu bar, when you change the timezone or manually set the Date/Time on the device
	D:	
19872	T:	Unable to change ACD Agent's status when the agent makes call to its own or another ACD Queue
	D:	
19804	T:	Auto attendant hangs and stops responding when pressing a digit during the announcement message playing in custom scenario
	D:	
18863	T:	The "Symmetric RTP" option doesn't work after retrieving the parked call, which was parked from the same remote phone
	D:	
18397	T:	When you change the Date/Time on the QX, the WEB GUI session will be automatically terminated and you will be logged out.
	D:	
17993	T:	Voice Mail notifications are sent even after disabling the corresponding "Send new Voice Mail notifications …" option
	D:	

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

19947	T:	Unable to park calls using "Call Park" MPK on Yealink SIP-T40P IP phone
	D:	
	C:	
	Fix:	Workaround: Use direct call park. Press Tran softkey during an active call. The call will be put on hold. Dial Park Extension number and press # to park the call. Will be fixed in future release.
19894	T:	Automatic "Daylight Saving Time" doesn't work on Fanvil IP phones
	D:	
	C:	
	Fix:	Workaround: Create an IP Phone Template for Fanvil phones, select the "manual" option for "Daylight Saving Time". Attach this template to the IP lines for Fanvil phones. Will be fixed in future release by Fanvil.
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, 7xx, D7xx, 8xx series and MeetingPoint running 8.7.5.35 firmware version.
	C:	
	Fix:	Workaround: The issue is solved in snom 8.7.5.44 beta firmware.
19804	T:	Auto attendant hangs and stops responding when pressing a digit during the announcement message playing in custom scenario
	D:	
	C:	
	Fix:	Workaround: Disconnect the call and dial again. Will be fixed in future release.
19537	T:	ACD call recordings cannot be played from EAC when using the Mozilla Firefox browser
	D:	The Mozilla Firefox browser doesn't have native support for (*.wav) audio format.
	C:	When you click Play instead of playing the recording, it will be downloaded.
	Fix:	Workaround: Install corresponding add-ons or use other browsers (Chrome, Opera, Microsoft Edge).
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 QX2000 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.

19329	T:	Outgoing calls through default PSTN routing rule cannot be established in a specific scenario
	D:	Scenario: QX E1T1 connected with QX2000 in share mode. After adding the PSTN access code from System Configuration Wizard on QX2000, the default (9?* or 0?*) routing rule isn't added in QX E1T1's Call Routing Table.
	C:	Outgoing calls through the default (9?* or 0?*) routing rule cannot be established.
	Fix:	Workaround: Reboot QX E1T1 to resolve this issue. 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.
18419	T:	Cannot establish call if you change signaling type for time slots using CAS Signaling Wizard
	D:	
	C:	
	Fix:	Workaround: Need to stop/start E1 trunk to make a call. Will be fixed in future release.
18397	T:	After changing the Time/Date Settings manually, it takes you to the QX2000 login page
	D:	
	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.
16683	T:	Find Me / Follow Me does not work for incoming Secure RTP call
	D:	Though the call came with SRTP option the FM/FM is making unsecure calls.
	C:	As a result, the call is not established.
	Fix:	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.
16533	T:	A problem with incoming Secure RTP call in a specific scenario
	D:	When incoming Secure RTP call is connecting to the destination via Call Routing table, QX2000 always tries to connect it as an unsecure call and the call is being dropped due to the media parameters incompatibility.
	C:	
	Fix:	Will be fixed in future release.
15942	T:	It is not possible to pick up (via pickup group) the call to extension with FM/FM enabled
	D:	
	C:	
	Fix:	Will be fixed in the next releases.

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 6.0.1 or for Emergency Recovery of a system. This procedure will result in a system that is at factory default. 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, voice mails, 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 VM sent by email is limited to three minutes.
- The number of VMs in the mailbox for an extension is limited to 500.
- Use Session Timer in IP Line Settings is deselected by default.