Release Note QX50/QX200 6.1.25 Edition 1

1 Introduction

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

QX50/QX200 firmware 6.1.25 Date: May 4, 2016

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

Date: May 4, 2016

2 Requirements

2.1 Hardware Requirements

The firmware (FW) can be used on all QX200 and QX50 models.
The model name is written on the back plate of the unit and the model number is on the bottom label.

2.2 Firmware Requirements

Attention: The firmware upgrade can be made from 6.0.2 or later firmware. If the QX50/QX200 is running on a FW version lower than 6.0.2 then 6.0.2 needs to be installed from scratch. For details on the installation procedure, see Upgrading Instructions section.

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 6.1.25. All the phones in this list can be automatically configured to work with QX50/QX200 FW 6.1.25.

Please Note:

- QX50/QX200 FW 6.1.25 supports also the Plug-and-Play (PnP) option for most IP phones. The configuration options for each specific IP phone is described in detail in **Configuring Epygi Supported IP Phones** document.
- Any known issues and limitations regarding the usage of the QX50/QX200 FW 6.1.25 telephony services and features for each IP phone described in details in the **QX IP PBX Features on Epygi Supported IP Phones** document.

Both mentioned documents are available at www.epygi.com and can be found under the Support Portal.

Vendor	Model	SW/FW Version	PnP Support
Akuvox	SP-R53P	53.0.1.23	Yes
Alcatel	IP2015	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	Yes
AudioCodes	320HD	1.6.0_build_37	Yes
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

Vendor	Model	SW/FW Version	PnP Support
Fanvil	X3/X3P	1.3.511.1821	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
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	GXP1610	1.0.2.27	No
Grandstream	GXP1620/GXP1625	1.0.2.27	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	1.0.5.23	Yes
Grandstream	GXP2140	1.0.5.23	Yes
Grandstream	GXP2160	1.0.5.23	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)	6863	4.0.0.92-SIP	Yes
Mitel (Aastra)	6865	4.0.0.92-SIP	Yes
Mitel (Aastra)	6867	4.0.0.92-SIP	Yes
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9480	3.3.1.4305-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

Vendor	Model	SW/FW Version	PnP Support
Polycom	SoundStation IP 6000	3.3.5.0247	Yes
Polycom	VVX 1500	3.3.5.0247	Yes
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	1.0.3.97	No
snom	300	8.4.35	Yes
snom	320	8.4.35	Yes
snom	360	8.4.35	Yes
snom	370	8.4.35	Yes
snom	710	8.7.3.25.9	Yes
snom	720	8.7.3.25.9	Yes
snom	760	8.7.3.25.9	Yes
snom	821	8.4.35	Yes
snom	870	8.4.35	Yes
snom	D715/715	8.7.5.17	Yes
snom	D725	8.7.5.17	Yes
snom	m9	9.4.7	Yes
snom	MeetingPoint	8.4.35	Yes
snom	M700	03.24.0007	Yes
Spectralink	KIRK Wireless Server 300	PCS14C_	No
Spectralink	KIRK Wireless Server 6000	PCS14C_	No
Yealink	CP860	37.72.0.10	Yes
Yealink	SIP-T19P	31.72.0.1	Yes
Yealink	SIP-T19P E2	53.80.0.70	No
Yealink	SIP-T20P	9.72.0.1	Yes
Yealink	SIP-T21P	34.72.0.1	Yes
Yealink	SIP-T21P E2	52.80.0.70	No
Yealink	SIP-T22P	7.72.0.1	Yes
Yealink	SIP-T23G(P)	44.80.0.70	No
Yealink	SIP-T26P	6.72.0.1	Yes
Yealink	SIP-T27P	45.80.0.70	No
Yealink	SIP-T28P	2.72.0.1	Yes
Yealink	SIP-T29G	46.80.0.70	No
Yealink	SIP-T32G	32.70.0.130	Yes
Yealink	SIP-T38G	38.70.0.125	Yes
Yealink	SIP-T41P	36.72.0.1	Yes
Yealink	SIP-T42G	29.72.0.1	Yes
Yealink	SIP-T46G	28.72.0.1	Yes
Yealink	SIP-T48G	35.72.0.34	Yes
Yealink	VP-530	23.70.0.40	Yes
Yealink	W52P	25.30.0.20	Yes

2.4 Interaction with Other Epygi Software Releases

To achieve maximum compatibility with QX50/QX200 FW 6.1.25, use the latest SW and FW versions.

- QXISDN4, QXE1T1 or QXFXO4 gateways used in the shared mode should have FW 6.1.10 or higher.
- QXFXS24 should have FW 6.1.10 or higher for PnP configuration.
- Auto Dialer SW 1.0.11 or higher should be used.
- Desktop Communication Console (DCC) SW 1.17 or higher should be used.
- iQall (IOS application) version 1.1.0 and iQall (Android application) version 1.0.3 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 1.0.12 or higher should be used.
- QX-Quadro Configuration Console (QCC) SW 2.2 or higher should be used.
- ActiveX Control SW 5.7.0 or higher should be used.
- To use QX50/QX200 FW 6.1.25 with a 3PCC or Click2Dial application the 3pcc/Click2Dial Access Allowed checkbox should be enabled for each extension(s) using this feature.

Important Note: QX50/QX200 FW 6.1.2 and higher is not compatible with ACD Service and Statistics Monitoring and Reporting (SMR) system. The replacement for SMR is the Epygi ACD Console (EAC) web application. EAC requires a software license key.

3 New Features History

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
6.1.25	Added support for the **SIP Registration Transport** UDP/TCP/TLS options in the Extension's SIP Registration.
	Added a new **Schedules** menu to define schedule names and apply them to specific entries in the Call Routing Table. The **Schedules** menu allows various times to be defined for each day of the week. It also allows **Holidays** and **Special Days** to be applied to each Schedule Name.
	Added auto configuration support for the new QOSIP Q7104 IP phone.
	Added 3-way conference support for Mitel (Aastra) 6863.
	Added **Push Route to LAN side** option for OpenVPN server configuration.
	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.
6.1.24	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.
6.1.23	Added support (with a license key) for connecting QX50/QX200 to **PMSLINK** middleware from char (software solution partner). This allows integrating the QX50/QX200 with the PMS systems used in hotels.
	Added PnP and auto configuration support for the Grandstream GXV3240 and GXV3275 IP phones.
	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.

Release	New Features
	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.
6.1.10	Added auto configuration support for the new Yealink SIP-T19P E2, T21P E2, T23G(P) and T27P IP phones.
	Added **OpenVPN** support, which allows secure point-to-point or site-to-site connections in routed or bridged configurations between the QX50/QX200 and other devices and remote access facilities.
	Added support for the new **Epygi Hotel Console (EHC)** application (with a license key).
	Added new **Alarm** feature. This feature allows to be schedule alarms for each extension. The configured alarm will be automatically activated at the scheduled time.
	Added support for **Barge-In**, **Whisper** and **Listen-In** features from **EAC** web GUI.
	Added **Reporting** feature in the **EAC** application that allows reports to be generated and downloaded in a **CSV** format related to the **ACD** statistics and **CDR** records.
	Added support for direct outbound/inbound calls from/to agents along with the ACD call statistics and CDR records for these calls.
	Added a new **ACD Service** in the **Firewall→Management Access** to allow remote access to the **EAC** application from the WAN.
	Added a new **EHC Service** in the **Firewall→Management Access** to allow remote access to the **EHC** application from the WAN.
6.1.5	Added auto configuration support for the new Alcatel Temporis IP100, IP150, IP300, IP700G IP phones and Alcatel IP2015 DECT phone.
	Added PnP and auto configuration support for the new snom D715/715 and D725 IP phones.
6.1.2	Added support for the new **Epygi ACD Console** web application (with a license key), which allows call center agents, supervisors and administrators to monitor the queues and agents, to view call statistics, to chat and to update the agents' status, etc. **EAC** stores and formats the data and produces real-time information and statistical reports on ACD activities.
	Added **Phone Book** support, which allows external contacts to be easily added into the IP phones' Directory.
	Added a new **Recording tool**, which allows system voice messages to be directly recorded from an IP phone.
	Added auto configuration support for the new Fanvil X3/X3P, X5/ X5G, C400 and C600 IP phones.
	Added PnP and auto configuration support for the new snom M700 base station.
	Added **Play Ringback Tone** option in Find Me/Follow Me (FM/FM), which allows a ringback to be played during the **Welcome Message Delay** and **Audio Wait** period.
	Added a new configuration option in the IP Phone templates to allow the **Record** button functionality for snom phones to be enabled/disabled.
	Added the **Live Dialpad** option for Yealink phones in IP Phone templates.

Release	New Features
	Added **Caller ID Source** parameter for Yealink phones in IP Phone templates.
	Added **BLF LED Mode** states to be optioned for Yealink phones (except Yealink SIP-T19P) in IP Phone templates.
	Added an additional option to verify the status for trial and time limited features keys.
6.0.13	Added PnP and auto configuration support for the new Aastra 6863i, 6865i, 6867i IP Phones.
	Added PnP support for Cisco SPA IP phones: 303, 501, 509 and 525.
	Added programmable keys configuration support for Cisco SPA IP phones: 303, 501, 509 and 525.
6.0.9	
6.0.8	
6.0.2	

4 Changed Features History

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
6.1.25	**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 60 message/sec.
	The maximum length of **Connection Name** field for **PPTP** and **L2TP** has been increased up to **64** symbols. Support for the following symbols "@", "-", ".", "_" is added.
	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, but these calls will not be counted as ACD calls and will not be displayed in EAC.
	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**.
6.1.24	
6.1.23	Added support to download Call Detail Records in CSV format for Successful, Missed and Unsuccessful Outgoing calls.
	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.
	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.
	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) system will clear/remove all VLAN settings configured on the LAN interface of IP phones. • When this option is disabled 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.
	ACD Agents can now receive calls from another queue(s) when busy on a call.
	The **Blueface Ireland, Blueface Italy and Blueface UK** carriers have been removed from the VoIP Carrier Wizard list.
6.1.17	Added support allowing to enable/disable entries in the **Authorized Phones**.
6.1.15	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.

Release	Changed Features
	The recommended FW versions for Grandstream GXP2130, 2140, 2160 IP phones have been changed from 1.0.3.9 to 1.0.5.23.
	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.
	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.
	Added SoTel/VoIPLINK as a new carrier to the VoIP Carrier Wizard list.
	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.
	The user can make calls with clicktodial.cgi either using admin credentials or his own (username/password).
	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).
	ACD agent will not receive a second call from the same queue if he/she is already in the call from the same queue.
	ACD Agents Status Records archiving is removed.
6.1.10	The default option for Configure IP phones from changed from WAN to LAN in the IP Line Settings.
	The maximum count for IP Phones trial feature has been changed from 99 to 176 for QX200 and to 32 for QX50.
	Added support in Extensions Multiple Editing for the following fields: Call Barge-In / Intercept Access List from General Settings page.
	The behavior for the VoIP Carrier Wizard is changed. The new Authentication by IP Address checkbox allows bypassing the Account Name and Password information and filling the SIP Server and SIP Server Port information only for the carriers not requiring account authentication.
	Added BINARY NETWORKS as a new carrier to the VoIP Carrier Wizard list.
	Added Blueface Ireland, Blueface Italy and Blueface UK as new carriers to the VoIP Carrier Wizard list.
	Added IP Directions as a new carrier to the VoIP Carrier Wizard list.
	Added MyNetFone as a new carrier to the VoIP Carrier Wizard list.
	Added ThinkTel Communications as a new carrier to the VoIP Carrier Wizard list.
	Kebu.it carrier has been removed from the VoIP Carrier Wizard list.
	Consultative call transfer action has been removed from EAC→Dashboard page.
	The old password will not be required when change the Phone Access Password for Administrator.
	The LAN IP address of the backup configuration displayed in the shell window, while restoring previously backed up configuration file.
6.1.5	Archive by call records count maximum is decreased to 3000 in the Call History→Archiving Settings for system stability purposes.
	In addition to expiration period the expiration date/time has been displayed for time limited license keys in Licensed features table.
	Added support in Extensions Multiple Editing for the following fields: Password and Confirm password from General Settings page and Authentication User Name from SIP Advanced settings.
	No Answer Call Forwarding (NACF) enhancement for FM/FM. Unanswered FM/FM call will be forwarded to NACF destination's Voice Mailbox.

Release	Changed Features
6.1.2	The ACD system has been completely redesigned for QX50/QX200 FW 6.1.2. The SMR system isn't compatible with QX50/QX200 FW 6.1.2 and higher firmware. The replacement for SMR is the new Epygi ACD Console web application. EAC requires a software license key. Please Note: In general, most of the ACD configuration settings will remain while updating to FW 6.1.2 and higher. The ACD settings should be reviewed before and after the update as new fields will be added in FW 6.1.2.
	If an ACD Agent rejects a call by pressing the Reject button on the phone, then that call will not ring the agent's phone again within the Queue Ring timeout duration.
	When an ACD Agent receives a call and doesn't answer within the Agent Ring timeout, the agent state will change to Away. Either the Agent or the Supervisor will need to change the agent's status back to Online.
	Added Ring Duration option for each FM/FM destination. The Ring Duration is used to select the ringing timeout of the destination.
	Loadlogo.cgi hidden page (for updating company details) has been renamed/ changed to uploadlogo.cgi
	Aastra IP phones (6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i, 6863i, 6865i, 6867i, 9143i, 9480i) have been renamed to Mitel (Aastra) 6730, 6731, 6735, 6737, 6739, 6753, 6755, 6757, 6863, 6865, 6867, 9143 and 9480.
	Polycom KIRK Wireless Server 300 and KIRK Wireless Server 6000 stations have been renamed to Spectralink KIRK Wireless Server 300 and Spectralink KIRK Wireless Server 6000 accordingly. Please Note: QX50/QX200 doesn't support the KWS's redundancy feature for these stations.
	The recommended FW versions for Grandstream GXP14xx, 2100, 2110, 2120, 2124 have been changed from 1.0.4.9 to 1.0.6.7.
	The recommended FW versions for snom 710, 720, 760 have been changed from 8.7.3.15 to 8.7.3.25.9.
	The recommended FW versions have been changed for Mitel (Aastra) IP phones. For 6139 from 3.2.2.2088-SIP to 3.3.1.4305-SIP, for 6735, 6737 from 3.2.2.7137-SIP to 3.3.1.8140-SIP, for 6730, 6731, 6753, 6755, 6757, 9143, 9480 from 2.6.0.2019-SIP to 3.3.1.4305-SIP.
	The recommended FW versions for Aastra 6757iCT and 9480iCT have been changed from 2.6.0.2019-SIP to 3.3.1.2256-SIP.
6.0.13	The max number of Line appearance for Cisco SPA525G2 is changed from 2 to10.
	New parameters have been added to Aastra IP Phones templates.
	The recommended FW versions for Grandstream GXP2130, GXP2140 and GXP2160 have been changed from 1.0.2.9 to 1.0.3.9.
	Changed the behavior for Polycom phone display in case of incoming calls to watched extensions. Show or not the call appearance/caller's ID on incoming calls to watched extensions is configurable now in the IP phones templates.
6.0.9	License key support for Redundancy feature.
6.0.8	Removed support for Polycom IP phones using the older 3.3.x.x firmware. This phone will need to be updated to the recommended FW.
6.0.2	

5 Fixed Issues

Issues fixed since version 6.1.24:

T: Title
D: Description

19649	T:	The ACD Login/Logout BLF key on the phone doesn't get update if we change ACD Agent status from EAC
	D:	
19635	T:	Large VLAN packets (more than 1500) bytes are being dropped by QX50/QX200
	D:	Scenario: IP phones are registered and configured from WAN VLAN side. If the SIP messages sent by the IP phone are larger than 1500 bytes, the packets will be dropped by QX50/QX200.
19615	T:	Call watching and other subscriptions don't work when the Class of Service is enabled on the call routing rule(s) used for passing subscription messages
	D:	
19572	T:	"Send new voicemail notifications via e-mail" option for FAX mail doesn't work according to the selected settings
	D:	
19568	T:	An issue with forwarding voice mail to another destination using the phone handset
	D:	Scenario: When forwarding the voice message from the phone handset by pressing 2 and recording additional voice message, then pressing 9 immediately after the forwarding to remove the voice message in the mailbox, the destination(s) will only get the additional voice message but not the main message that was needed to be forwarded.
19534, 19494	T:	The call is disconnected when it is transferred from DCC application in a specific scenario
	D:	Scenario: Extension160 has FM/FM enabled and targeted to extension163 which is configured as a DCC extension. Call comes to extension 160. Extensions 160 and 163 start ringing. Extension 163 answers the call then transfers the call to extension 150 from DCC application. The call is immediately disconnected.
19512	T:	A problem with Cisco SPA501G with simply putting the call on hold and taking it off hold.
	D:	The call cannot be resumed after being put on hold.
18638	T:	When opening the Call History, sometimes it doesn't show CDR records but shows an empty page
	D:	Only after refreshing the page it shows the CDRs.
18028	T:	Customize push back number service doesn't work in a specific scenario
	D:	Scenario: Customize push back number is set to Auto. The Class of Service(CoS) is enabled globally and activated on the routing rule which is used to send the call to the customize push back number. The CoS isn't enabled on the extension that parked the call. The customize push back number doesn't work, after the call retrieve timeout expires.

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

19631	T:	Extension is not re-registered automatically when "SIP Registration Transport" is changed
	D:	If you change SIP Registration Transport on Extension from TCP or TLS to UDP, re-registration is not taking place automatically.
	C:	
	Fix:	Workaround: Uncheck Registration on SIP Server option, press Save from SIP Registration Settings section on Extension. Select the same option and press Save. Will be fixed in future release.
19591	T:	An issue with the sorting mechanism in Call Details pages (for Agents and Queues) in EAC
	D:	Sometimes, mainly during extensive calls, the sorting will stop functioning properly.
	C:	
	Fix:	Workaround: Wait for 3-5 minutes after calls (in order the database synchronization to be finished) and resort again. Will be fixed in future release.
19559	T:	ACD call statistics is displayed (calculated) incorrectly when the ACD call has been intercepted by another agent or extension
	D:	Scenario: The call comes to ACD Queue and Agent A starts ringing. Agent B intercepts the call, Agent A stops ringing, but from EAC we could see that Agent A is in busy state. After the call is terminated it is displayed in EAC statistics as if the call is answered by Agent A.
	C:	
	Fix:	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 press Play button the recording will be downloaded.
	Fix:	Workaround: Install corresponding add-ons or use other browsers (Chrome, Opera, Microsoft Edge).
19521	T:	Alarm doesn't work in the specific scenarios
	D:	Scenario 1: Alarm will not work when FM/FM service is configured on the Extension. Scenario 2: Alarm works only on Parent Extension if you have parent/child extension configuration.
	C:	
	Fix:	Will be fixed in future release.
19514	T:	Call Intercept with programmable key doesn't work on Cisco SPA phones
	D:	
	C:	

	Fix:	Workaround: Use the feature code * 94 + extension number # to intercept the call. Will be fixed in future release.
19502	T:	3-way conference cannot be established with the help of "+" sign on Grandstream GXP2200
	D:	
	C:	
	Fix:	Workaround: Tap "+ Add Call" softkey to establish 3-way conference.
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 when Grandstream GXP1100 or GXP1105 phone is in the active call and tries to initiate 3-way call conference. When you press FLASH key the first call is placed on hold thus the both calls aren't joined into a conference.
	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 QX50/QX200 LAN IP configuration, the phones configured from LAN side lose registration
	D:	After changing QX50/QX200 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 QX200 in share mode. After adding the PSTN access code from System Configuration Wizard on QX200, the default (9? * or 0? *) call routing is added in QX200's Call Routing Table only.
	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.
19167	T:	AudioCodes 310HD and 320HD IP phones are forced to make only secure calls by default, even though QX50/QX200's default SRTP settings for attached extension is "Make unsecure calls, accept anything"
	D:	
	C:	The outgoing calls cannot be established.
	Fix:	Workaround: Go to the Codec Settings for the extension attached to that phone, select Make and accept only unsecure calls option in the Secure RTP Settings and reset the phone to factory defaults. Will be fixed in future release.
18959	T:	Phone book isn't available on Yealink T4x phones at once
	D:	After factory reset on Yealink T4x phones, phone book will not appear in phone directory on the IP phones.
	C:	
	Fix:	Workaround: The phone book will appear after making changes in Phone book directory, example adding a new extension to Public Directory and rebooting phones. 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/pickup the second attempt to park the call, using the park ext. programmable key fails. The problem is happening only if we park the call to the same park extension (by pressing Call Park key).
	C:	
	Fix:	Workaround: Park the call to a different call park extension. Will be fixed in future release.
18707	T:	Call Intercept with programmable key doesn't work on Fanvil C62
	D:	If a programmable key is configured to watch an extension, you cannot pickup the call addressed to that extension by pressing the key.
	C:	
	Fix:	Will be fixed in future release.
18604	T:	After power reset some system information is lost on QX
	D:	After power reset we lose records in Call History and System Events sections. When just rebooting QX from GUI, the records in the System Events are lost.
	C:	
	Fix:	Will be fixed in future release.
18577	T:	The voice traffic is not encrypted when using IPSec connection between two QX IP PBXs (QX50 or QX200)
	D:	
	C:	
	Fix:	Will be fixed in the next release.
18559	T:	There is no audio when using service codes like *74, *75, *4 on Mitel (Aastra) 6739 in case if SRTP Policy is set as "Make and accept only secure calls" on the phone extension
	D:	
	C:	
	Fix:	Will be fixed in future release.
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.
18528	T:	In shared mode, E1/T1 trunk link is not available in Status→System Status→Lines page
	D:	
	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 QX IP PBX login page
	D:	
	C:	
	Fix:	Will be fixed in future release.
18186	T:	A fake error message when pressing successful calls tab in the Call History
	D:	When pressing this tab just after a successful call termination, sometimes the following error is displayed: Log file seems to be corrupted. Please clear all records.
	C:	
	Fix:	Workaround: Pressing this tab once more will resolve the problem. Will be fixed in future release.
18112	T:	Call Intercept with programmable key doesn't work on Akuvox SP-R53P
	D:	If a programmable key is configured to watch an extension, it allows only calling to that extension. You cannot intercept the call addressed to that extension by pressing the key.
	C:	
	Fix:	Will be fixed in future release.
17555	T:	Using Call Intercept to directly answer an incoming ACD call fails
	D:	When ACD calls to an extension of an agent and Call Intercept is used from another extension to answer the call with the feature code (*94 + extension number), the caller hears nothing and the incoming call continues ringing. Hanging up the call from caller to direct pickup, leaves an active call.
	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 for Advanced→Programmable Keys Configuration.
	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, QX200 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 future release.

7 Upgrading Instructions

7.1 Technical Advisory

Attention: For manually configured IP phones, it is now required to have the SIP Registration Timeout parameter set to 120 seconds or more on your IP phone. Values less than 120 seconds will not be accepted and will cause the IP phone registration failure on the QX50/QX200 6.1.25.

7.2 QX50/QX200 FW Requirements for Upgrading to 6.1.x

Attention: The firmware upgrade to 6.1.x can ONLY be done from 6.0.2 and higher 6.0.x versions. Before updating to 6.1.x the unit should be updated to 6.0.2 or higher 6.0.x FW version first.

7.3 General Hints

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.
In general, the configuration of a system will remain after the firmware update. Remember that some data is lost during upgrade:

- Call History (when only embedded memory storage is used)
  Workaround – to save the existing call history, download it to the PC from Status→Call History→ Settings before performing the firmware update. It is also recommended to use an external SD memory card to keep the call history safe.
- Voice mails (when only embedded memory storage is used)
  Workaround – save the recorded voice mails from the Voice Mailbox before performing the firmware update. It is also recommended to use a SD memory card to keep the voice mails safe.
- Call recordings (when only embedded memory storage is used)
  Workaround – save the recorded calls from the Recordings Box before performing the firmware update or use the recorded calls automatically upload to the FTP server option. It is also recommended to use an external SD memory card to keep the call recordings safe.
- All custom voice greetings (when only embedded memory storage is used)
  Workaround – backup and download all configuration and voice data to a PC. It is also recommended to use an external SD memory card to keep the custom voice messages safe.
- All custom recordings for the custom Auto Attendant (when only embedded memory storage is used)
  Workaround – backup and download all configuration and voice data to a PC. It is also recommended to use an external SD memory card to keep the custom recordings safe.

The following steps describe how to correctly perform the firmware update:

1. Save the current configuration by doing a Backup and Download of all current configuration and voice data (Maintenance→Backup/Restore→Backup and download current Configuration).
2. Perform the Firmware Update.
3. Is there an SD memory card installed?
   - Yes – No further action is required.
   - No – Restore the configuration that was saved in Step 1 (Maintenance→Backup/Restore→Restore previously backed up Configuration).

This is necessary to restore the custom voice messages for extensions and the custom Auto Attendants.

Please Note: When using Call Recording, ACD on the QX50/QX200 it is advisable to use an SD memory card to expand the system memory.

Attention: Always power down the QX50/QX200 before inserting/removing any SD memory card. Currently, the recommended SD card's largest capacity is 16 GB.

7.4 Limitations and Restrictions

- The memory used by Network Capture hidden page is limited to 12 MB. This will put a limitation on the duration of captured file.
- The capture duration is limited to 160 seconds in DSP Capture hidden page.
- The Call Capture duration is limited to 160 seconds.
- 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 one extension is limited to 300.
- Use Session Timer in IP Line Settings is deselected by default.