Release Note QX50/QX200 6.1.17 Edition 1

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1   Introduction

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

QX50/QX200 firmware 6.1.17 Date: January 26, 2016

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

Date: January 26, 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 6.1.17. All the phones in this list can be automatically configured to work with QX50/QX200 FW 6.1.17.

Please Note: QX50/QX200 FW 6.1.17 supports both the Plug-and-Play (PnP) and the auto configuration options for most supported IP phones. For some specific phones only the auto configuration option is supported. The configuration options for each specific IP phone is described in detail in **Configuring Epygi Supported IP Phones** document.

Please Note: Any known issues and limitations regarding the usage of the QX50/QX200 FW 6.1.17 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	Software Version	PnP Support
Aastra	6757iCT(57iCT)	3.3.1.2256-SIP	Yes
Aastra	9480iCT(35iCT)	3.3.1.2256-SIP	Yes
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

Vendor	Model	Software Version	PnP Support
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	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	GXP2000	1.2.5.3	Yes
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	1.0.3.25	Yes
Grandstream	GXV3140	1.0.7.3	Yes
Grandstream	GXV3175	1.0.3.22	Yes
Mitel (Aastra)	6730i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6731i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6735i	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6737i	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6739i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6753i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6755i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6757i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6863i	4.0.0.92-SIP	Yes
Mitel (Aastra)	6865i	4.0.0.92-SIP	Yes
Mitel (Aastra)	6867i	4.0.0.92-SIP	Yes
Mitel (Aastra)	9143i	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9480i	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	Software 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
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

Please Note: QX IP PBX **IP phones firmware control** mechanism will not upgrade snom firmware version from 6.x to 7.x. This should be done manually via snom web site. Once the snom firmware version is 7.x, the QX's firmware control will automatically upgrade/downgrade the phone to Epygi's recommended version but not to 6.x.

2.4 Interaction with Other Epygi Software Releases

To achieve maximum compatibility with QX50/QX200 FW 6.1.17, use the latest software and firmware 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 1.13 or higher should be used.

- **iQall** (IOS application) version 1.1.0 and **iQall** (Android application) version 1.0.2 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 2.3 or higher should be used.

- **HotKeyCall** SW 1.11 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 6.1.17 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
6.1.17	
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.
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.

Release	New Features
	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.
	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
6.1.17	Added support to allow enabling/disabling entries in the **Authorized Phones**.
6.1.15	The recommended firmware versions for Yealink SIP-T19 E2, T21P E2, T23G(P) and T27P have been changed from xx.80.0.60 to xx.80.0.70.
	The recommended firmware versions for Grandstream GXP2130, 2140, 2160 have been changed from 1.0.3.9 to 1.0.5.23.
	The recommended firmware versions for Grandstream GXP1100, 1105, 1160, 1165, 1400, 1405, 1450, 2100, 2110, 2120, 2124 have been changed from 1.0.6.7 to 1.0.8.6.
	The recommended firmware versions for Fanvil X3/X3P and X5/X5G 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).

Release	Changed Features
	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.
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**

Release	Changed Features
	Aastra IP phones (6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i, 6863i, 6865i, 6867i, 9143i, 9480i) have been renamed to Mitel (Aastra) 6730i, 6731i, 6735i, 6737i, 6739i, 6753i, 6755i, 6757i, 6863i, 6865i, 6867i, 9143i and 9480i.
	Polycom KIRK Wireless Server 300, 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 firmware versions for Grandstream GXP 1400, 1405, 1450, 2100, 2110, 2120, 2124 have been changed from 1.0.4.9 to 1.0.6.7.
	The recommended firmware versions for snom 710, 720, 760 have been changed from 8.7.3.15 to 8.7.3.25.9.
	The recommended firmware versions for Aastra 6757iCT, 9480iCT have been changed from 2.6.0.2019-SIP to 3.3.1.2256-SIP.
	The recommended firmware versions for Mitel (Aastra) 6730i, 6731i, 6753i, 6755i, 6757i, 9143i, 9480i have been changed from 2.6.0.2019-SIP to 3.3.1.4305-SIP.
	The recommended firmware version for Mitel (Aastra) 6739, has been changed from 3.2.2.2088-SIP to 3.3.1.4305-SIP.
	The recommended firmware version for Mitel (Aastra) 6735i, 6737i have been changed from 3.2.2.7137-SIP to 3.3.1.8140-SIP.
6.0.13	The max number of Line appearance for SPA525G2 is changed from 2 to10.
	New parameters have been added to Aastra IP Phones templates.
	The recommended firmware 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 6.1.15:

T: Title
D: Description

19486	T:	The ACD Agents Status Records aren't removed from database when the records maximum size achieved, that causes system instability and the performance decrease
	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

19463	T:	3-way conference doesn't work on Grandstream GXP1100 and GXP1105 in a specific scenario
	D:	3-way conference cannot be established when Grandstream GXP1100 or GXP1105 is in the active call and tries to initiate 3-way call conference. When you press FLASH key the calls are placed on hold accordingly, thus the conference isn't established.
	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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
18707	T:	An issue with extension watching on Fanvil C62 phone
	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 the future releases.
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.
	C:	
	Fix:	Will be fixed in the future releases.
18604	T:	After power reset some system information is lost on IP PBX
	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 the future releases.
18591	T:	GXP2200, GXV3175 and GXV3140 IP phones are forced to make secure calls by default, even though in phone Web GUI the SRTP Mode is configured as "Enabled but not forced"
	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 the future releases.
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) 6739i 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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
18372, 17709	T:	A problem with configuring programmable keys in IP line Advanced Settings page for Grandstream GXP2124, GXP 2140 and GXP2160 phones
	D:	If the Programmable Keys configuration page for GXP2124, GXP2140 and GXP2160 phones has all six Line keys configured with some functions the phones becomes none functional: the dial tone is lost, the keys and buttons on phone keypad stop working.
	C:	
	Fix:	Workaround: Do not use all six line keys when configuring programmable keys. Keep at least two line keys unused to be able to make/receive calls. Or use Multi-Purpose Keys instead. Will be fixed in some future FW versions for the mentioned phones.
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 the future releases.
18112	T:	**Call Intercept does not work on Akuvox SP-R53P phone**
	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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 the future releases.
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 future releases.
15729	T:	**IP phone does not go back to the normally idle state automatically when the recording had been started via the Record button and the call was released**
	D:	The **Recording started** message remains on the phone screen after one of the parties terminates the call by going on-hook.
	C:	Only incoming calls to IP phone are possible in this state.
	Fix:	Workaround: Need to use **Exit** button to retrieve the IP phone functionality. Will be fixed in the future releases.
14797	T:	**Call Interception does not work on Grandstream GXP2000 configured as a receptionist, when extensions are watched on expansion module**
	D:	
	C:	
	Fix:	Workaround: Use basic seven keys instead of expansion module. Will be fixed in the future releases.
12190	T:	**Some of the voice mail services could be unavailable if external Voice Mail is in use for extension**
	D:	In this scenario some of the features, for example ZeroOut and entering the VMS directly with option **1** will not work.
	C:	
	Fix:	Workaround: This is normal, as those features are the QX200's internal VMS system features. If external VMS system is used, user gets the features of that external system.

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 6.1.17.

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.
- **Pending events** (when only embedded memory storage is used)

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.