Release Note QX50/QX200 6.1.5 Edition 1

1 Introduction

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

QX50/QX200 software 6.1.5 Date: August 24, 2015

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

Date: August 26, 2015

2 Requirements

2.1 Hardware Requirements

- The software (SW) 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 Software Requirements

Attention: The software upgrade can be made from 6.0.2 or later SW. If the QX50/QX200 is running on a SW 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 firmware (FW) versions that are tested and recommended for use with QX50/QX200 SW 6.1.5. All the phones in this list can be automatically configured to work with QX50/QX200 SW 6.1.5.

Please Note: For most IP phones the QX50/QX200 supports both the Plug-and-Play (PnP) and the auto configuration options. 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 SW 6.1.5 telephony services and features for each IP phone is described in detail in the **Epygi IP PBX Features on Epygi Supported IP Phones List** document.

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

Vendor	Model	Software Version
Aastra	6757iCT(57iCT)	3.3.1.2256-SIP
Aastra	9480iCT(35iCT)	3.3.1.2256-SIP
Akuvox	SIP-R53P	53.0.1.23
Alcatel	IP2015	1.0.7A-0
Alcatel	Temporis IP100	1.0.6A-0
Alcatel	Temporis IP150	1.0.6A-0
Alcatel	Temporis IP200	13.60.0.89
Alcatel	Temporis IP300	1.0.7B-0
Alcatel	Temporis IP600	14.60.0.89
Alcatel	Temporis IP700G	1.0.7A-0
Alcatel	Temporis IP800	15.60.0.89
AudioCodes	310HD	1.6.0_build_37
AudioCodes	320HD	1.6.0_build_37
Cisco	SPA303	7.4.9c
Cisco	SPA501G	7.4.9c
Cisco	SPA509G	7.4.9c
Cisco	SPA525G2	7.4.9c
Fanvil	C58/C58P	2.3.233.129
Fanvil	C62/C62P	2.3.235.128

Vendor	Model	Software Version
Fanvil	C400	11.20.12.2.B
Fanvil	C600	11.20.12.2.B
Fanvil	F52/F52P	2.3.123.78
Fanvil	X3/X3P	1.3.221.1531
Fanvil	X5/X5G	1.3.115.1425
Grandstream	GXP1100	1.0.6.7
Grandstream	GXP1105	1.0.6.7
Grandstream	GXP1160	1.0.6.7
Grandstream	GXP1165	1.0.6.7
Grandstream	GXP1400	1.0.6.7
Grandstream	GXP1405	1.0.6.7
Grandstream	GXP1450	1.0.6.7
Grandstream	GXP2000	1.2.5.3
Grandstream	GXP2100	1.0.6.7
Grandstream	GXP2110	1.0.6.7
Grandstream	GXP2120	1.0.6.7
Grandstream	GXP2124	1.0.6.7
Grandstream	GXP2130	1.0.3.9
Grandstream	GXP2140	1.0.3.9
Grandstream	GXP2160	1.0.3.9
Grandstream	GXP2200	1.0.3.25
Grandstream	GXV3140	1.0.7.3
Grandstream	GXV3175	1.0.3.22
Mitel (Aastra)	6730i	3.3.1.4305-SIP
Mitel (Aastra)	6731i	3.3.1.4305-SIP
Mitel (Aastra)	6735i	3.3.1.8140-SIP
Mitel (Aastra)	6737i	3.3.1.8140-SIP
Mitel (Aastra)	6739i	3.3.1.4305-SIP
Mitel (Aastra)	6753i	3.3.1.4305-SIP
Mitel (Aastra)	6755i	3.3.1.4305-SIP
Mitel (Aastra)	6757i	3.3.1.4305-SIP
Mitel (Aastra)	6863i	4.0.0.92-SIP
Mitel (Aastra)	6865i	4.0.0.92-SIP
Mitel (Aastra)	6867i	4.0.0.92-SIP
Mitel (Aastra)	9143i	3.3.1.4305-SIP
Mitel (Aastra)	9480i	3.3.1.4305-SIP
Panasonic	KX-TGP550T04	12.17
Panasonic	KX-UT123	01.061
Panasonic	KX-UT123NE	01.221
Panasonic	KX-UT136	01.061
Polycom	SoundPoint IP 330SIP	3.3.5.0247
Polycom	SoundPoint IP 331SIP	3.3.5.0247
Polycom	SoundPoint IP 335SIP	3.3.5.0247
Polycom	SoundPoint IP 450SIP	3.3.5.0247
Polycom	SoundPoint IP 550SIP	3.3.5.0247
Polycom	SoundPoint IP 650SIP	3.3.5.0247
Polycom	SoundPoint IP 670SIP	3.3.5.0247
Polycom	SoundStation IP 5000	3.3.5.0247
Polycom	SoundStation IP 6000	3.3.5.0247

Vendor	Model	Software Version
Polycom	VVX 1500	3.3.5.0247
Polycom	VVX 300/310	4.1.7.1210
Polycom	VVX 400/410	4.1.7.1210
Polycom	VVX 500	4.1.7.1210
Polycom	VVX 600	4.1.7.1210
snom	300	8.4.35
snom	320	8.4.35
snom	360	8.4.35
snom	370	8.4.35
snom	710	8.7.3.25.9
snom	720	8.7.3.25.9
snom	760	8.7.3.25.9
snom	821	8.4.35
snom	870	8.4.35
snom	D715/715	8.7.5.17
snom	D725	8.7.5.17
snom	m9	9.4.7
snom	MeetingPoint	8.4.35
snom	M700	03.24.0007
snom	PA1	8.4.35
Spectralink	KIRK Wireless Server 300	PCS14C_
Spectralink	KIRK Wireless Server 6000	PCS14C_
Yealink	SIP-T19P	31.72.0.1
Yealink	SIP-T20P	9.72.0.1
Yealink	SIP-T21P	34.72.0.1
Yealink	SIP-T22P	7.72.0.1
Yealink	SIP-T26P	6.72.0.1
Yealink	SIP-T28P	2.72.0.1
Yealink	SIP-T32G	32.70.0.130
Yealink	SIP-T38G	38.70.0.125
Yealink	SIP-T41P	36.72.0.1
Yealink	SIP-T42G	29.72.0.1
Yealink	SIP-T46G	28.72.0.1
Yealink	SIP-T48G	35.72.0.34
Yealink	VP-530	23.70.0.40
Yealink	W52P	25.30.0.20

Please Note: QX IP PBX IP phones firmware control mechanism will not upgrade snom FW version from 6.x to 7.x. This should be done manually via snom web site. Once the snom FW 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 Softwares

- QXISDN4, QXE1T1 or QXFXO4 gateways used in the shared mode should have SW 6.1.5 or higher to achieve maximum feature functionality with the QX50/QX200 SW 6.1.5.
- QXFXS24 should have SW 6.1.5 or higher for PnP configuration with the QX50/QX200 SW 6.1.5.
- ActiveX Control SW 5.7.0 or higher should be used with 3PCC functions with the QX50/QX200 SW 6.1.5.
- Auto Dialer SW 1.0.11 or higher should be used with the QX50/QX200 SW 6.1.5.
- Desktop Communication Console (DCC) SW 1.13 or higher should be used with the QX50/QX200 SW 6.1.5.
- QX-Quadro Configuration Console (QCC) SW 2.1 or higher should be used with the QX50/QX200 SW 6.1.5.
- HotCall Add-In SW 2.3 or higher should be used with the QX50/QX200 SW 6.1.5.
- HotKeyCall SW 1.10 or higher should be used with the QX50/QX200 SW 6.1.5.
- To use QX50/QX200 SW 6.1.5 with a 3PCC or Click2Dial application the 3pcc/Click2Dial Access Allowed checkbox should be enabled for each extension(s) using this feature.
- Epygi Media Streamer (EMS) SW 2.4 or higher should be used with the QX50/QX200 SW 6.1.5.

Important Note: QX50/QX200 SW 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 SW release.

Release	New Features
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.

Release	New Features
	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 SW release.

Release	Changed Features
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 enhancement for FM/FM. Unanswered FM/FM call will be forwarded to No Answer Call Forwarding (NACF) destination's Voice Mailbox.
6.1.2	The **ACD** system has been completely redesigned for QX50/QX200 SW 6.1.2. The **SMR** system isn't compatible with QX50/QX200 SW 6.1.2 and higher software. 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 SW 6.1.2 and higher. The ACD settings should be reviewed before and after the update as new fields will be added in SW 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.

Release	Changed Features
	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) 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.
	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.
	Recommended firmware versions for snom 710, 720, 760 have been changed from 8.7.3.15 to 8.7.3.25.9.
	Recommended firmware versions for Aastra 6757iCT, 9480iCT have been changed from 2.6.0.2019-SIP to 3.3.1.2256-SIP.
	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.
	Recommended firmware version for Mitel (Aastra) 6739, has been changed from 3.2.2.2088-SIP to 3.3.1.4305-SIP.
	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.
	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 6.1.2:

T: Title
D: Description

19200	T:	iQall toggling doesn't work when you talk on desk phone and want to toggle to mobile
	D:	
19175	T:	After time limited key expiration, the access to QX50/QX200 can be denied
	D:	
18667	T:	An issue with Caller ID on FXO calls
	D:	Neither DTMF nor FSK Caller ID that is sent prior to the first ring is detected by QX IP PBX.
18534	T:	PPP Interface Statistics pages are not available
	D:	When clicking on Watch PPP0 link from Status→System Status→Network page it takes you to the LAN Interface Statistics page.
17282	T:	An issue with Intercom service on snom 8xx phones
	D:	When the snom 8xx phone series (tested with snom models 821, 870 running FW version's 8.4.32, 8.4.33) have watched extensions configured and the status of the watched extension has changed (e.g. the watching phone receives or makes a call), immediately the next intercom call to the snom phone does not activate the intercom but continuously rings the phone. The issue has been fixed in recommended snom FW version 8.4.35.

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

19179	T:	SRTP support on snom D715/715 and D725 has been temporarily disabled.
	D:	Snom D715/715 and snom D725 forced to make only secure calls by default, even though QX50/QX200's default SRTP settings for Extension is **Make unsecure calls, accept anything** and snom's default SRTP policy set to **Optional**.
	C:	
	Fix:	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:	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.
18986	T:	Paging doesn't work for Parent/child extensions
	D:	
	C:	
	Fix:	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.
18682	T:	All configuration wizards are broken when Espanol_intl_x3 is selected as the GUI language

	D:	None of the configuration wizards, like call routing, VoIP Carrier, System (LAN), Internet (Uplink), are working when Espanol_intl_x3 is used as GUI language. Page two for the wizards becomes empty and the next/previous buttons stop functioning. The wizards work after switching to English.
	C:	
	Fix:	Use the latest Espanol_intl_x9 language pack to solve the issues.
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:	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 Astra 6739i IP phone in case if SRTP Policy is set as "Make and accept only secure calls" on the phone extension**
	D:	There is no audio when using service codes like *74, *75, *4, or even when calling to local auto attendant on Astra 6739i IP phone in case if the **Make and accept only secure calls** option is selected as SRTP Policy on the phone extension. No such problem with the same settings on other Aastra and Yealink phones.
	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:	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:	Pressing this tab once more will resolve the problem. Will be fixed in the future releases.
18112	T:	Watching does not work properly configured 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 pickup the call addressed to that extension by pressing that 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:	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:	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.
16184	T:	An issue with wrongly displayed "Start Recording" message in the Active Calls
	D:	The recording type is set to **start automatically**; the number of allowed parallel call recordings is exhausted, therefore the recording for the next call cannot be started, but the **Start Recording** is wrongly shown in the Active Calls page.
	C:	
	Fix:	Will be fixed in the future releases.
16037	T:	After changing QX50/QX200 LAN IP configuration (IP address or subnet mask) IP phones lose registration and become unusable
	D:	
	C:	
	Fix:	After changing QX50/QX200 LAN IP configuration first reboot the unit then reboot the IP phones. Will be fixed in the next release.
15942	T:	It is not possible to pick up (via pickup group) the call to extension with Find Me/Follow Me 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:	Need to use **Exit** button to retrieve the IP phone functionality. Will be fixed in the future releases.
14909	T:	Paging and intercom services do not work on the Grandstream BT100 IP phone
	D:	
	C:	
	Fix:	Currently BT100 has no support for paging/intercom.
14797	T:	Call Interception does not work on Grandstream GXP2000 configured as a receptionist, when extensions are watched on expansion module
	D:	
	C:	
	Fix:	Use basic seven keys instead of expansion module. Will be fixed in the future releases.
13802	T:	An issue with configuring Aastra IP phones as local extensions for QX50/QX200
	D:	After changing the QX50/QX200's LAN IP address, the Aastra 480i, 9133i and 55i phones with 1.4.1.2000 firmware are not registering after restart.
	C:	
	Fix:	They must be factory reset to register again. Will be fixed in the future FW versions for Aastra phones.
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:	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.
	T:	Aastra, snom, Grandstream and Thomson IP phones may disconnect if you press "Mute" button for a long time (60 min)
	D:	
	C:	
	Fix:	Will be fixed in the future releases.

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 6.1.5.

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

Attention: The software 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 SW version first.

7.3 General Hints

It is recommended to execute the update by downloading the software 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 software update. Remember that some data is lost during upgrade:

- Call History (only when 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 card memory to keep the call history safe.
- Voice mails (only when 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 (only when 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 (only when 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 (only when 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 (only when embedded memory storage is used)
- Transfer statistics for the network interfaces
- DHCP leases

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. The largest currently recommended SD card is 16GB.

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.