Release Note QX2000 6.0.6 Edition 1

1 Introduction

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

QX2000 software 6.0.6 Date: December 10, 2014

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

Date: December 15, 2014

2 Requirements

2.1 Hardware requirements

- The software (SW) can be used on all QX2000 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: A software upgrade can be made from 6.0.1 or later software. If the QX2000 is running on a software version lower than 6.0.1 then 6.0.1 needs to be installed from scratch. For details on installing procedure see section 7.

2.3 Supported SIP phones

Listed below are the Epygi Supported SIP phones with the corresponding firmware (FW) versions that are tested and recommended for use with QX2000 SW 6.0.6. All the phones in this list can be automatically configured to work with QX2000 SW 6.0.6.

Please Note: For most of the phones the QX2000 supports both the Plug-and-Play (PnP) and the auto-configuration options. For some specific phones only the auto-configuration option is supported.

Using of the configuration options on each specific SIP phone is described in detail in the **Configuring Epygi Supported IP Phones** document.

Please Note: Any known issues and limitations regarding usage of QX2000 SW 6.0.6 telephony services and features on these SIP phones are described in detail in the **Epygi IP PBX Features on Epygi Supported IP Phones** document.

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

Vendor	Model	Software
Aastra	6730i	version: 2.6.0.2019-SIP
Aastra	6731i	version: 2.6.0.2019-SIP
Aastra	6735i	version: 3.2.2.7137-SIP
Aastra	6737i	version: 3.2.2.7137-SIP
Aastra	6739i	version: 3.2.2.2088-SIP
Aastra	6753i	version: 2.6.0.2019-SIP
Aastra	6755i	version: 2.6.0.2019-SIP
Aastra	6757i	version: 2.6.0.2019-SIP
Aastra	6757iCT	version: 2.6.0.2019-SIP
Aastra	9143i (33i)	version: 2.6.0.2019-SIP
Aastra	9480i (35i)	version: 2.6.0.2019-SIP
Aastra	9480iCT(35iCT)	version: 2.6.0.2019-SIP
Akuvox	SIP-R53P	53.0.1.23
Alcatel Temporis	IP200	13.60.0.89
Alcatel Temporis	IP600	14.60.0.89
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
Fanvil	F52/F52P	2.3.123.78
Grandstream	GXP1100	Program- 1.0.6.7
Grandstream	GXP1105	Program- 1.0.6.7
Grandstream	GXP1160	Program- 1.0.6.7
Grandstream	GXP1165	Program- 1.0.6.7
Grandstream	GXP1400	Program- 1.0.4.13
Grandstream	GXP1405	Program- 1.0.4.13
Grandstream	GXP1450	Program- 1.0.4.9
Grandstream	GXP2000	Program- 1.2.5.3
Grandstream	GXP2100	Program- 1.0.4.9
Grandstream	GXP2110	Program- 1.0.4.9
Grandstream	GXP2120	Program- 1.0.4.9
Grandstream	GXP2124	Program- 1.0.4.10
Grandstream	GXP2130	Program- 1.0.2.9
Grandstream	GXP2140	Program- 1.0.2.9
Grandstream	GXP2160	Program- 1.0.2.9
Grandstream	GXP2200	Program- 1.0.3.25
Grandstream	GXV3140	Program- 1.0.7.3
Grandstream	GXV3175	Program- 1.0.3.22
Grandstream	HT286	Program- 1.1.0.26
Panasonic	KX-TGP550T04	12.17
Panasonic	KX-UT123	01.061
Panasonic	KX-UT123NE	01.221
Panasonic	KX-UT136	01.061
Polycom	KIRK wireless server 300	PCS08__
Polycom	KIRK wireless server 6000	PCS08__
Polycom	SoundPoint IP 330SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 331SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 335SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 450SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 550SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 650SIP*	UC SIP software 3.3.5.0247
Polycom	SoundPoint IP 670SIP*	UC Software 3.3.5.0247
Polycom	SoundStation IP 5000*	UC SIP software 3.3.5.0247
Polycom	SoundStation IP 6000*	UC SIP software 3.3.5.0247
Polycom	VVX 300/310*	UC SIP software 3.3.5.0247
Polycom	VVX 400/410*	UC SIP software 4.1.7.1210
Polycom	VVX 1500*	UC SIP software 4.1.7.1210
Polycom	VVX 500	UC SIP software 4.1.7.1210
Polycom	VVX 600	UC SIP software 4.1.7.1210
snom	300	snom300-SIP 8.4.35
snom	320	snom320-SIP 8.4.35
snom	360	snom360-SIP 8.4.35
snom	370	snom370-SIP 8.4.35
snom	710	snom710-SIP 8.7.3.15

snom	720	snom720-SIP 8.7.3.15
snom	760	snom760-SIP 8.7.3.15
snom	821	snom821-SIP8.4.35
snom	870	snom870-SIP8.4.35
snom	m9	9.4.7
snom	MeetingPoint	snomMP-SIP 8.4.35
snom	PA1	8.4.35
Yealink	SIP-T19P	SW version: 31.72.0.1
Yealink	SIP-T20P	SW version: 9.72.0.1
Yealink	SIP-T21P	SW version: 34.72.0.1
Yealink	SIP-T22P	SW version: 7.72.0.1
Yealink	SIP-T26P	SW version: 6.72.0.1
Yealink	SIP-T28P	SW version: 2.72.0.1
Yealink	SIP-T32G	SW version: 32.70.0.130
Yealink	SIP-T38G	SW version: 38.70.0.125
Yealink	SIP-T41P	SW version: 36.72.0.1
Yealink	SIP-T42G	SW version: 29.72.0.1
Yealink	SIP-T46G	SW version: 28.72.0.1
Yealink	SIP-T48G	SW version: 35.72.0.34
Yealink	VP-530	23.70.0.40
Yealink	W52P	25.30.0.20

Please Note: QX2000 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 FW control will automatically upgrade/downgrade the phone to Epygi's recommended FW version but not to 6.x.

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

2.4 Interaction with other Epygi SW releases

QXISDN4, QXE1T1 or QXFXO4 external PSTN gateways used in the shared mode should have SW 6.0.1 or higher to achieve maximum feature functionality with the QX2000 SW 6.0.6.

QXFXS24 should have SW 6.0.1 or higher for PnP configuration with the QX2000 SW 6.0.6.

ActiveX Control SW 5.3.0 or higher should be used with 3PCC functions with the QX2000 SW 6.0.6.

Auto Dialer SW 1.0.5 or higher should be used with the QX2000 SW 6.0.6.

Desktop Communication Console (DCC) SW 1.8 or higher should be used with the QX2000 SW 6.0.6.

Statistics Monitoring and Reporting (SMR) system SW 1.9or higher should be used with the QX2000 SW 6.0.6.

HotCall Add-In SW 2.3 or higher should be used with the QX2000 SW 6.0.6.

HotKeyCall SW 1.10 or higher should be used with the QX2000 SW 6.0.6.

Epygi Media Streamer (EMS) SW 2.4 or higher should be used with the QX2000 SW 6.0.6.

To use QX2000 SW 6.0.6 with a 3PCCor Click2Dial application the “3pcc/Click2Dial Access Allowed” checkbox should be enabled for each extension(s) using this feature.

3 New Features History

The table below indicates a high-level list of new features that have been added beginning with the most recent QX2000 SW release.

Release	New Features
6.0.6	
6.0.5	PnP and auto configuration support for new Yealink SIP T48G phone.
	Expansion module EXP39 support for Yealink SIP-T26P, SIP-T28P and SIP-T38P phones.
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 QX2000SW release.

Release	Changed Features
6.0.6	
6.0.5	
6.0.2	

5 Fixed Issues

Issues fixed since version 6.0.5:

T: Title
D: Description

18398	T:	**The "start manually" option for call recording does not work when configuring as a function on programmable key for supported IP phones**
	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.

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 next 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 default English when using the wizards. Will be fixed in the future releases.
18638	T:	**When opening the Call History, sometimes it doesn't show CDR records at once but shows an empty page**
	D:	Only after refreshing the page it shows the CDRs.
	C:	
	Fix:	Will be fixed in the future releases.
18591	T:	**GXP2200 IP phone is 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:	
	Fix:	Login to the phone Web GUI and change the "Enabled but not forced" to "Disabled". Will be fixed in the next releases.
18566	T:	**QX product line is not supported by Quadro Configuration Console application**
	D:	
	C:	
	Fix:	Will be fixed in the next releases.
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 some future FW 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 would be very uncomfortable to wait, if you change message file from default to a custom with duration let say five minutes.
	C:	
	Fix:	Will be fixed in the next releases.
18548	T:	Part of conference recording is lost after recording pause/resume
	D:	When pause the conference recording then resume it again, the final recording contains only the part after resuming.
	C:	
	Fix:	Will be fixed in the future releases.
18534		**PPP Interface Statistics pages are not available**
	D:	When clicking on Watch PPP0 link from Status->System Status->Network page it takes to the LAN Interface Statistics page.
	C:	
	Fix:	Will be fixed in the next 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 next releases.
18503	T:	**Cannot update company details using loadlogo.cgi hidden page**
	D:	Cannot load company details.
	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 next 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 next 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:	In case if in the Programmable Keys configuration page for GXP2124, GXP2140 and GXP2160 phones all six **Line keys** are configured with some functions the phones becomes non functional: the dial tone is lost, the keys and buttons on phone keyboard 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 next FW version for the mentioned phones.
18219	T:	**One way audio when calling through iLBC codec**
	D:	
	C:	
	Fix:	Will be fixed in the next releases.
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". Pressing the same tab again resolves the issue.
	C:	
	Fix:	Pressing this tab once more will resolve the problem. Will be fixed in the next 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 next releases.
18021	T:	**VLAN DHCP continues to function even after VLAN interface is disabled**
	D:	
	C:	As a result the IP phones will not configure with PnP option in LAN.
	Fix:	Disable the DHCP server for VLAN before disabling VLAN interface. Will be fixed in the next 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 next releases.
17404	T:	**Call which is done after Call Relay(*2) on auto attendant is 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 next releases.
17282		**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.
	C:	As a result, if the calling phone is configured as a watched extension, the snom phone never activates the intercom for that caller.
	Fix:	Will be fixed in some future FW release for snom.
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 next 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 next 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, QX2000 always tries to connect it as an un-secure call and the call is being dropped due to the media parameters incompatibility.
	C:	
	Fix:	Will be fixed in the next releases.
16468	T:	**MAC address recognition problem in the FXS Gateway Management" page**
	D:	In the page Line Settings-> IP Line Settings->FXS Gateway Management, if FXS Gateway is added manually, the system cannot recognize MAC address of FXS Gateway entered in uppercase.
	C:	As a result the "Edit" and "Reboot" functional buttons become unusable and then it is not possible to edit existing records or reboot the registered FXS Gateways.
	Fix:	Enter the MAC address in lowercase. Will be fixed in the next 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 next releases.
16037	T:	**After changing QX2000 LAN IP configuration (IP address or subnet mask) IP phones lose registration and become unusable**
	D:	
	C:	
	Fix:	After changing QX2000 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 pickup (via pickup group) the call to extension with Find Me/Follow Me enabled**
	D:	
	C:	
	Fix:	Will be fixed in the next 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 terminate 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 QX2000 IP PBX**
	D:	After changing the LAN IP address for the QX2000, the 480i, 9133i and 55i Aastra phones with currently recommended 1.4.1.2000 firmware are not registering after restart.
	C:	
	Fix:	They must be factory reset to register again. Under investigation; will be fixed in the future FW versions for Aastra phones.
13380	T:	**Aastra IP phone is not ringing when it is used in many extensions ringing list**
	D:	Scenario: **1.** Many Extension Ringing is enabled on a virtual extension with an Aastra phone in the list. **2.** Distinctive Ringing is enabled on that virtual extension with Nickname that contains space.
	C:	When a call with no Caller ID comes to that extension, the Aastra phone in many extensions ringing list does not ring.
	Fix:	The problem is solved if there is a Caller ID available on the incoming call or if the Nickname does not contain spaces. This problem is limited to Aastra IP phones only.
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 Zero-out and entering the VMS directly with option “1” will not work.
	C:	
	Fix:	This is normal, as those features are the QX2000'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:	Under investigation, will be fixed in the future releases.
	T:	**ACD may not work correctly if the QX2000 encounters a problem with the 3PCC connection to an application**
	D:	ACD calls may be affected.
	C:	
	Fix:	Will be fixed in the next releases.

7 Upgrading Instructions

7.1 General hints

Attention: It is recommended to backup the configuration for **emergency purposes** prior to upgrading the firmware. You can do that by clicking the **Download Configuration** link in the **Firmware Update** page. Regardless, the configuration of a system will remain after the firmware update. Moreover, voice mails, all custom messages, call history and system events will be saved during the upgrade.

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

1. Save the current configuration by doing a Backup and Download of all configuration and voice data (**Maintenance** ->**Backup/Restore**/**Configuration Management**->**Backup and download current configuration**).
2. Perform the Firmware Update using the **Firmware Update** page.

Please Note: If the saved configuration is restored all VMs and custom messages will be lost.

Please Note: The firmware upgrade to version 6.0.x using the **Firmware Update** page can only be done starting from 6.0.1 and higher versions. If the QX2000 is running on a software version lower than 6.0.x, then 6.0.1 needs to be installed manually from scratch before updating to 6.0.x.

The steps below describe shortly the QX2000 manual installation procedure used to install the software from scratch. This would be used to install version 6.0.1 or for Emergency Recovery of a system. This procedure will result in a system that is at factory default. Additional details are available on the Epygi Channel Portal in the document *QX1000/2000 System Recovery Procedure*:

- Turn on the PC;
- Insert CD/DVD disk including installation program to the DVD ROM;
- Restart (reset) the PC;
- Installation will start automatically after PC reboot. After the successful installation the PC will automatically shut down (this may take from 10 – 15 minutes);
- The beep sound will indicate that the installation successfully completed;
- Turn on the PC and quickly remove the installation CD/DVD disk from the DVD ROM.

7.2 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 by the QX2000 and will cause the IP phone registration failure on the QX2000.

7.3 Limitations and restrictions

- The memory used by "DSP Capture", "Call Capture" and "Network Capture" hidden pages is limited to 12 MB. This will put a limitation on the duration of captured file.
- 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 500.
- Use Session Timer in IP Line Settings is deselected by default.