Release Note QX2000 6.0.1 Edition 1

1 Introduction

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

QX2000 software 6.0.1 Date: September 26, 2014

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

Date: October 2, 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 6.0.1. All the phones in this list can be automatically configured to work with QX2000 SW 6.0.1.

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 6.0.1 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
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-SIP 8.4.35
snom	870	snom870-SIP 8.4.35
snom	MeetingPoint	snomMP-SIP 8.4.35
snom	PA1	8.4.35
snom	m9	9.4.7
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	F52/F52P	2.3.123.78
Fanvil	C58/C58P	2.3.233.129
Fanvil	C62/C62P	2.3.235.128
Polycom	SoundPoint IP 330SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 331SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 335SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 450SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 550SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 650SIP*	UC SIP software 3.3.1.0933
Polycom	SoundPoint IP 670SIP*	UC Software 3.3.1.0933
Polycom	SoundStation IP 5000*	UC SIP software 3.3.1.0933
Polycom	SoundStation IP 6000*	UC SIP software 3.3.1.0933
Polycom	VVX 300/310*	UC SIP software 4.1.4.7430
Polycom	VVX 400/410*	UC SIP software 4.1.4.7430
Polycom	VVX 1500*	UC SIP software 3.3.1.0933
Polycom	KIRK wireless server 300	PCS08__
Polycom	KIRK wireless server 6000	PCS08__
Panasonic	KX-TGP550T04	12.17
Panasonic	KX-UT123	01.061
Panasonic	KX-UT123NE	01.221
Panasonic	KX-UT136	01.061
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	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
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	VP-530	23.70.0.40
Yealink	W52P	25.30.0.20

Please Note: QX2000 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.

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/QXISDN2 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 6.0.1.

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

ActiveX Control SW 5.3.0 or higher should be used with 3PCC functions with the QX2000 SW 6.0.1.

Auto Dialer SW 1.0.5 or higher should be used with the QX2000 SW 6.0.1.

Desktop Communication Console (DCC) SW 1.8 or higher should be used with the QX2000 SW 6.0.1.

Statistics Monitoring and Reporting (SMR) system SW 1.9 or higher should be used with the QX2000 SW 6.0.1.

To use QX2000 SW 6.0.1 with a 3PCC or Click2Dial application the “3pcc/Click2Dial Access Allowed” checkbox should be enabled for each extension(s) using this feature.

3 Features

For the features available in 6.0.1 software refer to the QX50/QX200/QX2000 Manual II: Administrator's Guide.

4 Changed Features History

There are no changed features at the moment

5 Fixed Issues

No fixed issues here at the moment

6 Known Issues

D: Description
C: Consequences
Fix: How to avoid the situation, or what to do in case the situation has occurred.

Some of the voice mail services could be unavailable if external Voice Mail is in use for extension 12190	
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.
snom, Aastra, 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.
Aastra IP phone is not ringing when it is used in Many Extension Ringing list 13830	
	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.
An issue with configuring Aastra IP phones as local extensions for QX2000 IP PBX 13802	
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.
Call Interception does not work on Grandstream GXP2000 configured as a receptionist, when extensions are watched on expansion module 14797	
D:	
C:	
Fix:	Use basic seven keys instead of expansion module. Will be fixed in the future releases.
Paging and intercom services do not work on the Grandstream BT100 IP phone 14909	
D:	
C:	
Fix:	Currently BT100 has no support for paging/intercom.
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 15729	
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.
It is not possible to pickup (via pickup group) the call to extension with Find Me/Follow Me enabled 15942	
D:	
C:	
Fix:	Will be fixed in the next releases.
After changing QX2000 LAN IP configuration (IP address or subnet mask) IP phones lose registration and become unusable 16037	
D:	
C:	
Fix:	After changing QX200 LAN IP configuration first reboot the unit then reboot the IP phones. Will be fixed in the next release.
MAC address recognition problem in the FXS Gateway Management" page 16468	
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.
A problem with incoming Secure RTP call in a specific scenario 16533	
D:	When incoming Secure RTP call is connecting to the destination via Call Routing table, QX200 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.
An issue with wrongly displayed “Start Recording” message in the Active Calls 16184	
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.
Find Me/Follow Me does not work for incoming Secure RTP call 16683	
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.
ACD behavior becomes incorrect in case of troubles with 3PCC connection to an application	
D:	It affects to ACD calls
C:	
Fix:	Will be fixed in the next release.
An issue with Intercom service on snom 8xx phones 17282	
D:	When the snom phone of 8xx series (tested with snom phones 821, 870 running FW version's 8.4.32, 8.4.33) have some watching configured for it's functional keys and the status of the watching resource is 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 ring 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.

Call which is done after Call Relay(*2) on auto attendant is not shown in the call history 17404	
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.
Shared Mailbox watching does not work when using "Allow access to Shared Mailbox for enabled extensions" option in Many Extension Ringing configuration 16635	
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.
Using Call Intercept to directly answer an incoming ACD call fails 17555	
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.
A problem with configuring programmable keys in IP line Advanced Settings page for Grandstream GXP2124, GXP 2140 and GXP2160 phones 17709, 18372	
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.
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.
A problem with uppercase letters in the skill name for the ACD configuration 18096	
D:	When calling to the ACD group, that has agents with skill defined in uppercase letter, then the skill weight for that group will be zero (0).
C:	
Fix:	Will be fixed in the next releases.
Watching does not work properly configured on Akuvox SP-R53P phone 18112	
D:	1. 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. 2. If a programmable key is configured to watch a Park Answer Ext.#, then it is not work at all. You cannot watch the call parked to the corresponding Park Answer Ext. and you cannot retrieve the calls parked to that extension.
C:	

Fix:	Will be fixed in the next releases.
VLAN DHCP continue functioning even after VLAN interface is disabled 18021	
D:	As a result the IP phones will not configured with PnP option in LAN.
C:	
Fix:	Disable the DHCP server for VLAN before disabling VLAN interface. Will be fixed in the next releases.
The "start manually" option for call recording does not work when configuring as a function on programmable key for IP phones 18398	
D:	Tested on a few phones like Yealink T41P, T32G, Aastra 6757i.
C:	
Fix:	Will be fixed in the next releases.
One way audio when calling through iLBC codec 18219	
D:	
C:	
Fix:	Will be fixed in the next releases.
After changing the Time/Date Settings manually, it takes you to the QX IP PBX login page 18397	
D:	
C:	
Fix:	Will be fixed in the next releases.
Legible configuration generates file only for FXS Lines when we choose "Line Settings" legible. 18434	
D:	
C:	
Fix:	Will be fixed in the next releases.
QX IP PBXs do not work with Quadro Configuration Console 18566	
D:	
C:	
Fix:	Will be fixed in the next releases.
PPP Interface Statistics pages are not available 18534	
D:	When we click 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.
In shared mode, E1/T1 trunk link is not available in Status ->System Status -> Lines page 18528	
D:	
C:	
Fix:	Will be fixed in the next releases.
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" 18591	
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.
The voice traffic is not encrypted when using IPSec connection between two QX IP PBXs (QX50 or QX200) 18577	
D:	
C:	
Fix:	Will be fixed in the next release.
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 18559	

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.
VLAN does not work normally	**18539**
D:	VLAN enabled for WAN interface does not work normally. IP phones could not be configured to receive SIP call from VLAN in this case, as they do not get SIP Server IP from QX200.
C:	
Fix:	Will be fixed in the next releases.
"Show Security Report" is not functioning properly	**18441**
D:	Clicking on the "Start Security Audit" button in Security Diagnostics page does not show the latest "Show Security Report" at once. Need to click the button twice in order to see the latest security report.
C:	
Fix:	Will be fixed in the next releases.
A fake error message when pressing successful calls tab in the Call History	**18186**
D:	When pressing this tab just after a successful call termination, sometimes the following error is displaying: "Log file seems to be corrupted. Please clear all records". Pressing the same tab again resolve this issue.
C:	
Fix:	Pressing this tab once more will resolve the problem. Will be fixed in the next releases. Will be fixed in the next releases.
Could not dial out(*1) or use other moderator features while message file has been playing	**18549**
D:	Could not dial out (*1) or use other moderator features while welcome message file has been playing. You should list 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 file with duration let say five minutes.
C:	
Fix:	Will be fixed in the next releases.
Cannot update company details using loadlogo.cgi hidden page	**18503**
D:	Cannot load company details.
C:	
Fix:	Will be fixed in the future 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 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, then 6.0.1 needs to be installed manually from scratch before updating to 6.0.

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.