

Release Note QX1000 5.3.12 Edition 3

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1 Introduction

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

QX1000 software 5.3.12 Date: July 23, 2012

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

Date: February 19, 2013

2 Requirements

2.1 Hardware requirements

- The software can be used on the QX1000 model only.
- The model name and the model number are written on the connector's panel of the unit.

2.2 Software requirements

- A software upgrade can be made from 5.2.11 or later software. If the QX1000 is running on a software version lower than 5.2.11, then 5.2.11 needs to be installed from scratch before updating to the most recent release. For additional software update instructions, see **section 7**.

2.3 Supported and Tested SIP phones

Listed below are the Epygi Supported SIP phones with the corresponding firmware versions that are tested and recommended for use with QX1000 5.3.12.

Vendor	Model	Software
snom	300	Application: snom300-SIP 8.4.35
snom	320	Application: snom320-SIP 8.4.35
snom	360	Application: snom360-SIP 8.4.35
snom	370	Application: snom370-SIP 8.4.35
snom	720	Application: snom720-SIP 8.7.2.9
snom	760	Application: snom760-SIP 8.7.2.9
snom	821	Application: snom821-SIP 8.4.35
snom	870	Application: snom870-SIP 8.4.35
snom	MeetingPoint	Application: snomMP-SIP 8.4.35
Aastra	9143i (33i)	Application version: 2.5.2.1010
Aastra	9480i (35i)	Application version: 2.5.2.1010
Aastra	9480iCT (35iCT)	Application version: 2.5.2.1010
Aastra	6753i	Application version: 2.5.2.1010
Aastra	6755i	Application version: 2.5.2.1010
Aastra	6757i	Application version: 2.5.2.1010
Aastra	6757iCT	Application version: 2.5.2.1010
Aastra	6730i	Application version: 2.5.2.1010
Aastra	6731i	Application version: 2.5.2.1010
Aastra	6739i	Application version: 3.0.1.38

Please Note: Any problems and limitations on the SIP phones listed above are described in detail in the **Quadro Features on Supported IP Phones** document.

Please Note: QX1000 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 QX1000's firmware control will automatically upgrade/downgrade the phone to Epygi's recommended version but not to 6.x.

Listed below are the Epygi Tested SIP phones with the corresponding firmware versions that are tested and recommended for use with QX1000 5.3.12.

Any problems and limitations on these SIP phones are described in detail in the **Quadro Features on Tested Phones** document.

Vendor	Model	Software
Audiocodes	310HD	1.6.0_build_37
Audiocodes	320HD	1.6.0_build_37
Cisco	SPA525G2	7.4.8
Cisco	SPA303	7.4.8
Cisco	SPA501G	7.4.8
Cisco	SPA509G	7.4.8
Polycom	SoundPoint IP 330SIP*	Application: SIP software 3.3.1.0933

Polycom	SoundPoint IP 331SIP*	Application: SIP software 3.3.1.0933
Polycom	SoundPoint IP 335SIP*	Application: SIP software 3.3.1.0933
Polycom	SoundPoint IP 450SIP*	Application: SIP software 3.3.1.0933
Polycom	SoundPoint IP 550SIP*	Application: SIP software 3.3.1.0933
Polycom	SoundPoint IP 650SIP*	Application: SIP software 3.3.1.0933
Polycom	SoundStation IP 5000*	Application: SIP software 3.3.1.0933
Polycom	SoundStation IP 6000*	Application: SIP software 3.3.1.0933
Polycom	VVX 1500*	Application: 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-B	01.061
Panasonic	KX-UT136-B	01.061
snom	PA1	8.4.32
snom	m9	9.4.7
Grandstream	GXP1400	Program- 1.0.1.83
Grandstream	GXP1405	Program- 1.0.1.83
Grandstream	GXP1450	Program- 1.0.1.83
Grandstream	GXP2000	Program- 1.2.5.3
Grandstream	GXP2100	Program- 1.0.1.83
Grandstream	GXP2110	Program- 1.0.1.83
Grandstream	GXP2120	Program- 1.0.1.83
Grandstream	GXV3140	Program- 1.0.7.3
Grandstream	GXV3175	Program- 1.0.3.22
Grandstream	HT286	Program- 1.1.0.26
Yealink	SIP-T20P	SW version: 9.50.0.50
Yealink	SIP-T22P	SW version: 7.50.0.50
Yealink	SIP-T26P	SW version: 6.50.0.50
Yealink	SIP-T28P	SW version: 2.50.0.50
Yealink	SIP-T32G	SW version: 32.0.0.130
Yealink	SIP-T38G	SW version: 38.0.0.130
Yealink	VP-2009P	22.21.0.40/22.30.0.60
Yealink	VP-530	23.70.0.15

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

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 UCSoftware 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

QuadroISDN or QuadroFXO external PSTN gateways used in the shared mode should have SW 5.1.12 or higher to achieve maximum feature functionality with the QX1000 SW 5.3.12.

QuadroFXS 16 Gateway should have SW 5.2.1 or higher for PnP configuration with the QX1000 SW 5.3.12.

QuadroMFXS 26 Gateway should have SW 5.2.6 or higher for PnP configuration with the QX1000 SW 5.3.12.

ActiveX Control SW 5.3.0 or higher should be used with 3pCC functions with the QX1000 SW 5.3.12.

AutoDialer SW 1.0.5 or higher should be used with the QX1000 SW 5.3.12.

DCC SW 1.8 or higher should be used with the QX1000 SW 5.3.12.

Quadro SMR system SW 1.9 or higher should be used with the QX1000 SW 5.3.12.

To use QX1000 SW 5.3.12 with a 3pcc or Click2Dial application the "3pcc/Click2Dial Login 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 QX1000 release.

Release	New Features
5.3.12	Support for the new Epygi Desktop Communication Console (DCC) application (with a license key).
	The Caller ID based services are improved by adding the presence state of the extension for use with the DCC application.
	The maximum number of active calls in Find Me/Follow is now configurable. If the number is set to 1 then only one active call will be possible and the next call will go to voice mail. If that number is >1 then the next call will ring the Find Me/Follow phones which are not on a call.
	Added option for the Auto Attendant Customized Scenario to download the generated script in VXML format.
	The Call Recording feature is improved to also allow recording for pass-through calls.
	Added capability to activate Voicemail profile based on caller id and presence state.
	PnP and auto configuration support for new IP phones: snom 720, snom 760, Yealink SIP T32G, Yealink SIP T38G, Yealink VP 530.
	The Zero-Out redirection in the Call Queue settings is modified with a new option to redirect the call to the Voice Mail of the extension, or to another destination.
	LDAP support for Yealink IP phones.
5.3.3	

4 Changed Features History

The following table provides a high-level list of changed features that have been changed beginning with the most recent QX1000 release.

Release	Changed Features
5.3.12	Call statistics archive mechanism improved to support fast and accurate loading. The maximum number of records in a page is limited to 500. Added "Clear all Records" button. Archiving should run if the maximum record count – 10 000 is reached even if the method of collecting is set to "Archive by time interval".
	All preconfigured functions have been removed from Aastras' Programmable Keys Configuration.
	Improved the voice message played when collecting the user logs using feature code *82
	Improved the voice message played when user marks the call using feature code *81
	Added possibility to capture the LAN1 interface
	The redundancy feature is available with a license key
5.3.3	

5 Fixed Issues

Issues fixed since version 5.3.3:

The Voice Mail playing order is incorrect in automatically mode		17479
D:	When accessing the voice mailbox the newly received (unread) voice mails should play first, after all received voice mails started from the first received.	
C:	However in the current FW version the voice mails in mailbox are playing in order they are received.	
An issue with adding audio files for ACD custom scenario		16817
D:	Scenario: - add an audio file with special characters in name (with slashes, dashes, brackets, etc.) for Custom Queue Messages in the ACD Customize queue scenario ; - The added file is shown as normally uploaded on the GUI.	
C:	Actually the newly added file isn't being uploaded in the system. So the callers to ACD with Customize Queue Scenario will hear nothing for that message.	
A problem with "Remove VM On Send" option in Voice mail settings		16432
D:	If this option is enabled new VMs are removed from the extension mailbox even if sending the new VM via e-mail fails because the mail server is unreachable or mail address is incorrect	
C:		
One way audio in a scenario when Holding/Retrieving the PSTN call		17245
D:	Scenario: 1. ext. A is configured to watch the ext. B 2. PSTN (E1/T1) call comes to ext. B and the functional key of the ext. A shows the ringing state for ext. B 3. ext. A presses the functional key and intercepts the call to ext. B 4. ext. A holds the call 5. ext. A retrieves the call	
C:	Result: ext. A hears the PSTN, but the PSTN doesn't hear the ext. A	
An issue with "Call to the number dialed" action in Auto Attendant Customized Scenario		17510
D:	Customized Scenario with settings: - Both Welcome and Recurring messages are configured for Customized Scenario; - The User Input is configured with "Any input other than in the list above" with action "Call to the number dialed".	
C:	This scenario doesn't work if the number is dialing during welcome message play. The call just dropped in this case. This problem concerns both menu and submenus.	
An issue with QX1000 firmware update in redundancy mode		
D:	The firmware update process couldn't be completed in redundancy mode. Image uploading will fail with message "invalid image".	
C:		

6 Known Issues

D: Description

C: Consequences

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

Call interception fails in a specific scenario with a watched extension in the many extension ring list	
9399	
D:	Scenario: <ol style="list-style-type: none"> 1. Use snom360 for receptionist extension (for example, extension #31), 2. Configure Many Extension Ringing on ext. #31 with extensions #31 and #12 in the MER list, 3. Configure extensions #11 and #12 as watched extensions, 4. Make a call to extension #31, 5. The phones for extensions # 31 and #12 will start ringing. 6. The programmable key for the ext. #12 is flashing on the receptionist's ext. #31, 7. Pressing that key the receptionist stops ringing, but ext. #12 is still ringing, 8. Hang-up the phone for that caller's extension.
C:	Call interception on the receptionist's line fails. The ring on snom phone disappears but the watched extension continues to ring.
Fix:	This problem is specific for snom phones only. When using a snom phone, calls should be answered on your own extension and not the watched extension. Or press the appropriate watching key on snom phone for second time for intercepting the incoming call.
An issue with configuring IP phones as local extensions for QX1000 IP PBX	
13471	
D:	Plug and Play and Auto configuration of IP phones is impossible with the modified "HTTP Server Port" on the QX1000.
C:	
Fix:	Add the changed http port value in the "option tftp-server-name" row ("dhcpd.conf.lan" file) for each IP phone. Example: "option tftp-server-name "http://172.30.38.1:8080". Will be fixed in the future releases.
Some of the QX1000 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 QX1000's internal VMS system features. If external VMS system is used, user gets the features of that external system.
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.
An issue with configuring Aastra IP phones as extensions for QX1000 IP PBX	
13802	
D:	After changing the Ethernet IP address for the QX1000 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.
An issue with distinctive ringing on the snom, Aastra and GrandStream IP phones	
11519	
D:	snom, Aastra and GrandStream IP phones ring only once if the distinctive ringing

	is enabled with the "winter" ringing pattern.
C:	No consequences
Fix:	Use other ringing patterns for distinctive ringing on snom and Aastra IP phones.
Call Interception doesn't work on GrandStream GXP2000 configured as a receptionist, when extensions are watched on expansion module 14797	
D:	
C:	
Fix:	Use basic 7 keys instead of expansion module. Will be fixed in the future releases.
IP phone doesn't 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 FindMe/FollowMe enabled 15942	
D:	
C:	
Fix:	Will be fixed in the next releases.
After changing QX1000 Ethernet IP configuration (IP address or subnet mask) IP phones lose registration and become unusable 16037	
D:	
C:	
Fix:	After changing QX1000 Ethernet IP configuration first reboot the QX1000 then reboot the IP phones. Will be fixed in the next release.
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 doesn't 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 isn't 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.
The configuration page for Expansion Module is missing when configuring Yealink T28 and T26 as receptionist phones 16971	
D:	In Receptionist Management wizard the Expansion module configuration pages are not available for Yealink T28 and T26 phones.
C:	
Fix:	Will be fixed in the next release.
Installing custom language pack doesn't change the language on GUI 16844	
D:	
C:	
Fix:	Will be fixed in the next releases.
A problem with SRTP calls on snom IP phones that are being recorded 16872	

D:	These calls are being established but no voice is heard on any party of the call.
C:	
Fix:	Will be fixed in the next releases.
The PNP process stops when enabling/disabling either "PnP for IP lines" or "firmware version control" 17183	
D:	
C:	
Fix:	Will be fixed in the next releases.
The DHCP server works on one of LAN interfaces only 17182	
D:	When Secondary LAN is enabled it is possible to enable DHCP server on the both interfaces (LAN and LAN1), but actually only DHCP server of Secondary LAN (LAN1) works.
C:	
Fix:	Will be fixed in the next releases.
An issue with Intercom service on snom 8xx phones 17282	
D:	When the snom phone of 8xx series 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 doesn't 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.
Local authentication doesn't work for a call scenario 16784	
D:	Local authentication configured in the Local AAA table by User Login or by PIN code is ignored for the following call scenario; a SIP call to QX1000 Auto Attendant, then call to a SIP destination.
C:	QX1000 doesn't ask for authentication in this case.
Fix:	Will be fixed in the next release.
The snom phone doesn't allow multiple call park/retrieve 17340	
D:	Upon successful call park/retrieve on snom the call couldn't be parked again on the same call park extension using programmable key.
C:	
Fix:	Will be fixed in some future FW release for snom.
Call which is done after Call Relay(*2) on auto attendant isn't shown in call statistics 17404	
D:	Only the call to attendant is shown in the call statistics. The call leg after call relay is missing in the call statistics 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 doesn't 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.
An issue with "Automatically attach to IP Line" option when using the "Add Multiple Extensions" feature 17535	
D:	When trying to attach the already used IP lines to newly created extensions CGI crashes instead of giving the error "One or more IP Lines in the specified range are already attached to existing extensions".

C:	
Fix:	Will be fixed in the next releases.
New voice mail indication doesn't work on snom 3xx and 8xx IP phones 17538	
D:	
C:	
Fix:	Will be fixed in some future FW release for snom.
QX1000 Public Directory doesn't work on Aastra and snom IP phones 17298	
D:	When pressing the directory hard button on Aastra and snom phones, it brings an empty list. The number of items in this blank list is equal to the number of extensions configured to "Show on public directory". Pressing the Dial on each item just does nothing.
C:	
Fix:	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.
It is impossible to park or transfer a call when there is a second incoming call in ringing state 17593	
D:	
C:	
Fix:	Will be fixed in the next releases.
Only the User type Extensions are shown on QX1000 public directory 17603	
D:	Extensions other than User type are not shown in public directory on phones even though they have "Show on public directory" checkbox available.
C:	
Fix:	Will be fixed in the next releases.
Phone parameters configured in IP Phone Template get default after renaming the Template 17588	
D:	
C:	
Fix:	Will be fixed in the next releases.
DHCP server crashes in case of enabling it for both LAN and VLAN interfaces 16418	
D:	It`s necessary to use DHCP server simultaneously on both interfaces for organizing IP Phones PnP and auto configuration functionality (when using VLAN) without third party DHCP server.
C:	
Fix:	Will be fixed in the next releases.
QX1000 crashes when using auto redial (*1) in call hunting configuration with Circular Mode 17686	
D:	Scenario: Ext. 1002 has hunt group (ext.1002,1003) with circular mode enabled, Ext. 1003 calls to ext. 1002 and hangs up the call, Ext. 1003 redials by *1 feature code. When the ext.1002 rejects the call or ringing interval of the ext.1002 is timed out QX1000 crashes.
C:	
Fix:	Will be fixed in the next releases.

7 Installing and Upgrading Instructions

7.1 General hints

Attention: The software upgrade can only be done starting from 5.2.11 and higher versions. It is recommended to backup the configuration 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 software update.

Please Note: If the QX1000 is running on a software version lower than 5.2.11, then 5.2.11 needs to be installed from scratch before updating to the most recent release. Additional details are available on the Epygi Channel Portal in the document [*QX1000 System Recovery Procedure*](#).

The steps below describe the QX1000 manual installation procedure used to install the software from scratch. This would be used for Emergency Recovery of a system or to install version 5.2.11. This procedure will result in a system that is at factory default:

- 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 QX1000 and will cause the IP phone registration failure on the QX1000.

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 3 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.