Release Note QuadroM32x 5.3.73 Edition 2

1 Introduction

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

QuadroM32x software 5.3.73 Date: February 23, 2015

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

Date: August 31, 2015

2 Requirements

2.1 Hardware Requirements

- The software can be used on all QuadroM32x 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 to 5.3.73 can **ONLY** be done from 5.2.47 and higher 5.2.x versions. Before updating to 5.3.73 the unit should be updated to 5.2.47 or higher 5.2.x version first.

2.3 Supported IP Phones

Listed below are the Epygi Supported IP phones with the corresponding firmware versions that are tested and recommended for use with QuadroM32x SW 5.3.73. All the phones in this list can be automatically configured to work with QuadroM32x SW 5.3.73.

Please Note: For most IP phones the QuadroM32x 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 QuadroM32x SW 5.3.73 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	6730i	2.6.0.2019-SIP
Aastra	6731i	2.6.0.2019-SIP
Aastra	6735i	3.2.2.7137-SIP
Aastra	6737i	3.2.2.7137-SIP
Aastra	6739i	3.2.2.2088-SIP
Aastra	6753i	2.6.0.2019-SIP
Aastra	6755i	2.6.0.2019-SIP
Aastra	6757i	2.6.0.2019-SIP
Aastra	6757iCT	2.6.0.2019-SIP
Aastra	6863i	4.0.0.92-SIP
Aastra	6865i	4.0.0.92-SIP
Aastra	6867i	4.0.0.92-SIP
Aastra	9143i (33i)	2.6.0.2019-SIP
Aastra	9480i (35i)	2.6.0.2019-SIP
Aastra	9480iCT(35iCT)	2.6.0.2019-SIP
Akuvox	SP-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

Vendor	Model	Software Version
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	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
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*	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
Polycom	VVX 1500*	3.3.5.0247
Polycom	VVX 300/310*	4.1.7.1210
Polycom	VVX 400/410*	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	8.7.3.15
snom	720	8.7.3.15
snom	760	8.7.3.15
snom	821	8.4.35
snom	870	8.4.35
snom	m9	9.4.7

Vendor	Model	Software Version
snom	MeetingPoint	8.4.35
snom	PA1	8.4.35
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	VP-530	23.70.0.40
Yealink	W52P	25.30.0.20

Please Note: Quadro **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 Quadro'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 Softwares

- **QuadroISDN** or **QuadroFXO** gateways used in the shared mode should have SW 5.1.12 or higher to achieve maximum feature functionality with the QuadroM32x SW 5.3.73.
- **QuadroFXS 16** gateway should have SW 5.2.1 or higher for PnP configuration with the QuadroM32x SW 5.3.73.
- **QuadroMFXS 26** gateway should have SW 5.2.6 or higher for PnP configuration with the QuadroM32x SW 5.3.73.
- **ActiveX Control** SW 5.7.0 or higher should be used with **3PCC** functions with the QuadroM32x SW 5.3.73.
- **Auto Dialer** SW 1.0.11 or higher should be used with QuadroM32x SW 5.3.73.
- **Desktop Communication Console (DCC)** SW 1.13 or higher should be used with the QuadroM32x SW 5.3.73.
- **QX-Quadro Configuration Console (QCC)** SW 2.1 or higher should be used with the QuadroM32x SW 5.3.73.
- **Quadro SMR system** SW 1.9 or higher should be used with the QuadroM32x SW 5.3.73.
- **HotCall Add-In** SW 2.3 or higher should be used with the QuadroM32x SW 5.3.73.

- **HotKeyCall** SW 1.10 or higher should be used with the QuadroM32x SW 5.3.73.
- To use QuadroM32x SW 5.3.73 with a **3PCC** or **Click2Dial** application the **3pcc/Click2Dial Login 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 QuadroM32x SW 5.3.73.

3 New Features History

The table below indicates a high-level list of new features that have been added beginning with the most recent QuadroM32x SW release.

Release	New Features
5.3.73	Added PnP and auto configuration support for the new Aastra 6863i, 6865i, 6867i IP phones.
5.3.72	An option in the **generalconfig.cgi** hidden page was added to activate **Two B-Channel Transfer** feature. By default, when an incoming call from the carrier to the Quadro is received on one of the E1/T1 PRI channels and then due to a call transfer (e.g. user transfers the call, call forwarding rule, Auto Attendant option, etc..) the Quadro places an outbound PRI call to the same carrier using a different channel, the two channels will be in use for the duration of the call. When the **Two B-Channel Transfer feature** is enabled the two channels will be released and the call will continue with the carrier maintaining the connection.
	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.
5.3.71	Added PnP and auto configuration support for the new Grandstream GXP2130 IP phone.
	Added Receptionist support for Grandstream GXP2140 IP phone.
5.3.64	Added PnP and auto configuration support for the new Yealink T19P, T21P, T41P, T42G IP phones.
	Added auto configuration support for the new Fanvil F52/F52P, C58/C58P IP phones.
	Added PnP and auto configuration support for the new Grandstream 2140, 2160 IP phones.
5.3.61	
5.3.60	Added support for the new **iQall** application.
	Added support for **Mobile Toggling**. This is a licensed feature and allows to switch the active call from the desk phone to an **iPhone/Android** running the iQall application and vice versa, without disconnecting the call in progress.
	Added PnP and auto configuration support for the new Akuvox SP-R53P IP phone.
	Added support for failover to the next DNS SRV path when making SIP calls. The **SIP DNS SRV Failover Timeout** can be configured in the hidden menu **generalconfig.cgi**.
5.3.58	Added PnP and auto configuration support for the new Grandstream GXP2200 IP phone.
5.3.55	Class of Service (CoS). Each extension can be assigned a **CoS** that will allow the use of Call Routing entries with the matching CoS assigned.
	Added PnP and auto configuration support for the new Polycom VVX 300/310, 400/410, Aastra 6735i, Yealink T46G IP phones.

Release	New Features
	The **call back number** can be configured for Call Park extensions after the park timeout expires.
	Added possibility to paste copied IP addresses, instead of typing it manually.
	Added authorization details for all **SIP request** messages.
	Auto provisioning improvement of Cisco SPAxxx phones in case of 3rd party DHCPs.
5.3.53	New **Overall Calling Time Limit** feature. This feature allows a total call duration for all calls to be configured over a specific time frame for each Call Routing entry. Once the total duration has been reached, the entry can be disabled, allowing calls to use the next available route.
	A new feature code for FXS phones. While on a 3-way call, pressing **FLASH+0** allows the user to disconnect from the call while the two remaining participants stay connected.
	Added **RTP Channel** option in the Attendant Ringing Announcement.
	An option in the generalconfig.cgi hidden page was added to remove the **Diversion** header from SIP Invite messages sent out from the Quadro. If an incoming call is being forwarded back to the PSTN, the Quadro will add the **Diversion** header into the outgoing SIP Invite. However, many ITSPs do not support this and reject the call if the **Diversion** header is present.
	Added PnP and auto configuration support for new IP phones: Aastra 6737i and Yealink W52P.
5.3.23	A new warning in the security audit for the case when the filtering on the routing rule is enabled, but the Source Call Type is set to anything but PBX.
	New **Automatic Fax Receiving** Mode.
	A new feature code ***84** for manually configuring and switching no answer call forwarding on the phone handset.
	Added PnP and auto configuration support for the new snom 710, Alcatel Temporis IP200/IP600/IP800, Grandstream GXP2124 IP phones.
	Added auto configuration support for new Fanvil C62 IP phone.
	New **URL** functionality in IP phones advance settings.
	Recording for **Barge-In** call.
	Added support for Yealink T2x's hybrid configuration (M7+M1).
5.3.19	**GUI login** for Recording Box extension. The Recording Box extension can be used to access the appropriate Recording Box via Quadro WEB interface by extension name and password.
	The **Diversion** header can be mapped from an E1/T1 connection to a SIP connection.
	Yealink IP phone configuration support enhanced to auto-detect the phone firmware and to provide the proper configuration.
5.3.13	Added support for the new **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 Me (FM/FM) 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 FM/FM 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.

Release	New Features
	Added capability to activate **Voicemail profile** based on **Caller ID** and **Presence** state.
	Added PnP and auto configuration support for the new snom 720, snom 760, Yealink T32G, Yealink T38G, Yealink VP530 IP phones.
	The **ZeroOut 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.
	Added **Collect call** feature for calls to E1/T1(CAS R2 MFC and for ISDN PRI/BRI).
	Enhancement for **E1/T1 Diagnostic** tests.
	E1/T1 settings enhanced to add the Redirecting Number for calls that have been forwarded prior to being received by the QuadroM32x. This allows the original Caller Id to be sent in the Redirect number element field and the Caller ID in the Calling Party number.
	Added **LDAP** support for Yealink IP phones.
5.3.5	The **Add Multiple Extensions** feature is improved which allows the assignment of the IP Lines to be selected when configuring multiple extensions.
	A new **MS Exchange Server** option has been added in the Use **External Voice Mail** settings for extensions. This allows voice messages to be kept in one universal inbox.
5.3.3	The **Call Park** feature has been expanded with a **Directed Call Park** method. This allows users to define the call park extension they wish to park the call on. There are two methods available: 1) Hold the current call, and then dial the call park extension number; 2) Assign the call park extension to a key on the phone and then, just press the key during an active call.
	New **Call Intercept** feature. A new feature code has been added allowing users to dial ***94** followed by the extension number to intercept an inbound call on another extension.
	IP Phone Key Assignment Templates are now available to define a set of extensions mapped to keys on IP phones. These templates can be used for multiple IP phones to streamline installation and configuration of multiple phones.
	The FM/FM announcement message is now customizable. Users can remove the message, use the default, define a new recording or use music on hold.
	Added the option to review and modify the system default FM/FM welcome message like other universal extension messages using the admin ***75** feature code on the phone.
	Added the option to review and modify the user's personal FM/FM welcome message like other personal greeting messages using the extensions ***0** feature code on the phone.
	Added support for **P-Asserted-Identity header** field defined in RFC 3325. This is important to retain original caller information even after a call is transferred within the office.
	The CDR database has been improved adding archiving up to 30 days. Additionally, CDR uploading redundancy mechanisms are added to ensure records are not getting lost during CDR server downtime or network outage.
	The maximum number of stored CDRs has been increased from 1000 to 10000.
	Statistics gathered from the **ACD** feature and FM/FM have been improved for greater visibility of call flow.
	A few new options have been added to the GUI based **Auto Attendant IVR builder** integrated into the Quadro IP PBXs. Repeated message play count, repeated message timeout duration and timeout after welcome message options are now included in the AA IVR builder tool.
	Added auto configuration support for new IP phones: snom m9, Panasonic KX-TGP550T04, KX-UT123-B, KX-UT136-B, AudioCodes 320HD and 310HD.

Release	New Features
	A special configuration page is added for simple direct mapping of **E1/T1 DID numbers** to extension numbers. The configuration screen is accessible from E1T1 Trunk settings page on CCS trunk types only. This mapping works for both incoming and outgoing E1/T1 calls, and allows for easy streamlined installation for the majority of E1T1 setups.
	A new option to use RTP streaming channel for playing Hold Music is added to the **Universal Extension Recordings** configuration page. This allows using an external RTP as default Hold Music for all extensions.
	Intercom settings are improved. Now it will be possible to define whether the phone should make a sound when Intercom is activated. The default is silence.
	Call conferencing features are enhanced regarding the **Moderator Rights Management,** conference call terminations options, and the conference statistics.
	Added support for **T.38 faxing** on IP lines for use on devices such as an ATA for faxing.
	The **Maximum recording count** option has been restored back in the Recording Box Settings.
	The **VoIP carrier wizard** is improved which allows creating the outbound routing rules not only by prefix but also by **7** and **10** digit pattern.

4 Changed Features History

The table below provides a high-level list of changed features that have been changed beginning with the most recent QuadroM32x SW releases.

Release	Changed Features
5.3.73	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.
5.3.72	The max number of **Line appearance** for SPA525G2 is changed from 2 to 10.
5.3.71	New parameters have been added to Aastra IP phones templates.
	Recommended firmware versions for Grandstream GXP2140, 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.
	The **User Name** is changed to **User Name/DID Number** in the SIP Registration Settings for extension. Beside the registration user name on the SIP server, DID number from ITSP is supported also.
5.3.64	The **SMTP Host** field in **System Mail Settings** has been extended to 48 characters long.
5.3.61	
5.3.60	Changed the **Access List** function for Call Park extensions to be similar to Call Pickup. If a password is set for the Call Park extension, when the parked call is answered the system will prompt for a password only if the user's extension is not listed in the access list.
	The method for using **VLAN**s to configure the IP phones has been enhanced to allow the phones to be switched from the **VLAN** to the **LAN** or **WAN** without having to do a factory reset on the phone, which is currently required. The enhancement will not affect currently configured IP phones using the **VLAN**.
5.3.58	New parameters have been added to Yealink IP phones templates.

Release	Changed Features
	Possibility to pass **PIN** description to the CDR downloaded file.
5.3.55	For **Time Limited Call routing entries** the **weekly/monthly** start day has been added.
	Quadro LOGO is changed.
	Line Appearance is set to **4** for KX-UT133/KX-UT136 IP phones.
5.3.53	Added display names after extension numbers in the **Many Extension Ringing** groups, allowing users to easily find the desired extension.
	Added display names after extension numbers in the Auto Attendant **Create Scenario** page, allowing users to easily find the desired extension.
	The Grandstream GXP2124 phone is added to the phone model list for the Receptionist Configuration Wizard.
	The behavior for the announcement that informs callers that the call is being recorded has been changed so that the announcement is no longer captured in the recorded file.
	Added **VoIPVoice.it** as a new carrier to the VoIP Carrier Wizard list.
	Improvement on **Caller ID Based Services** page. Now user is able to navigate to the appropriate setting by clicking the links on the ON/OFF status of the service.
	When configuring an IP phone in the **IP Line settings**, it is now possible to paste a copied MAC address for the IP phone.
	Changed the behavior when making changes to parameters that require the IP phone to be rebooted to take effect. The system will prompt to reboot the IP phones after changing the SRTP policy, IP phones template, transport type, and the registration username and password.
	The **IP Lines Settings** table is now scrollable. Disabled IP lines are hidden from the list by default.
	Improved the way **Day/Time settings** are shown on the Call Routing GUI. When there is no rule with Time of Day set, the DT column and the DT description in the glossary are removed from the bottom of the page. Also, the Call Routing Table's brief view will now show an icon of a clock in the DT column.
	The **VLAN Settings** page has been moved from **DHCP Settings for the VLAN Interface** as a submenu to the **Network** in the main menu.
5.3.23	Added **Use RTP proxy** mode in Call routing for PBX type calls.
	Improved the GUI for **Recording Box**; recordings are displayed as tabbed page (1000 recordings per page).
	snom PA1 template enhancement: **Intercom Policy** has been added for snom PA1. Its default value is set to **off**.
	Distinctive Ring tones have been changed for Yealink phones.
	snom 7xx phones (720 & 760) have been upgraded to FW version 8.7.3.15.
	Recommended FW version for Yealink T3X and VP530 phones is updated to public Version 70 (3x.70.0.100/23.70.0.40 accordingly).
	New FW recommendations for Aastra, Grandstream, Cisco and Yealink IP phones.
	Added Speed dial based services for Polycom.
	Added **Park**, **DirPckUp**, **BargeIn** soft keys on Polycom UCSoftware.3.3.1 or higher.
	Language Pack update for Aastra, Alcatel, Grandstream GXP2000 and Polycom (FW 3.3.1.F) IP phones.
	Voice mail **Specific Profile** selection is simplified using the drop-down list of the created profiles.
	Aastra IP phone template enhancements: Added **Callers List Script** field, **Call Waiting/Hold Reminder/Stutter Tones** and **Password protected Options** checkboxes.
5.3.19	Restored the **Edit Watch Access List** link in the extension settings.
	Yealink T2x phones default settings updated.

Release	Changed Features
5.3.13	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 - 10000 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.
5.3.5	Changed the limitation for Conference ID length. Now it should be possible to create a conference with ID of 20 digit length.
	The timeout for the directed Call Park method is changed from two seconds to five seconds. To park a call, put the call on Hold and dial the Call Park extension number. This needs to be done within the five second timeout. If the five second timeout is exceeded, then the Quadro will consider it as an attempt for retrieving the parked call. Once the call is parked, it can also be retrieved by dialing the same Call Park extension.
	New FW recommendations for snom IP phones.
	Some of the system voice messages are improved.
5.3.3	Added the option to delete the uploaded Global speed dialing directory.
	Added Kebu.it as a new carrier to the VoIP Carrier Wizard list.
	The name of Login/Call Relay column in the Extensions Management is changed to External Access, which indicates whether the GUI login, 3pcc/Click2Dial login or Call Relay options are enabled on the extension.

5 Fixed Issues

Issues fixed since version 5.3.72:

T: Title
D: Description

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

19200	T:	iQall toggling doesn't work when you talk on desk phone and want to toggle to mobile
	D:	
	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 QuadroM32x'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.
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.
18759	T:	The STUN service doesn't work when the firewall level is set to "High".
	D:	When the firewall level is set to **High** QuadroM32x doesn't release STUN packets, because firewall prohibits them to be created and sent.
	C:	
	Fix:	To make STUN service working need to add the STUN server IP address in the **SIP access** filtering rule. 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.
18651	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.
18638	T:	**When opening the Call Statistics, 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.
18556	T:	**PPTP fails connecting to Windows 8/8.1 when MSCHAPv2 Encryption type is something else than NO MPPE**
	D:	
	C:	
	Fix:	Will be fixed in the future releases.
18549	T:	**Could not dial out (*1) or use any other moderator feature while welcome message file has been playing**
	D:	Could not dial out **(*1)** or use other moderator features while welcome message file has been playing. You should listen to the whole welcome message file first, after that use moderator features. It is recommended to keep the welcome message to a short duration.
	C:	
	Fix:	Will be fixed in the future releases.
18548	T:	**Part of conference recording is lost after recording pause/resume**
	D:	When pausing the conference recording and then resuming it again, the final recording contains only the part after resuming.
	C:	
	Fix:	Will be fixed in the future releases.
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 Statistics**

	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.
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 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, QuadroM32x 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 QuadroM32x LAN IP configuration (IP address or subnet mask) IP phones lose registration and become unusable
	D:	
	C:	
	Fix:	After changing QuadroM32x LAN IP configuration first reboot the unit then reboot the IP phones. Will be fixed in the future release.
15942	T:	It is not possible to pick up (via pickup group) the call to extension with FM/FM enabled
	D:	
	C:	
	Fix:	Will be fixed in the future releases.
15729	T:	IP phone does not go back to the normally idle state automatically when the recording had been started via the Record button and the call was released
	D:	The Recording started message remains on the phone screen after one of the parties terminates the call by going on-hook.
	C:	Only incoming calls to IP phone are possible in this state.
	Fix:	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 QuadroM32x
	D:	After changing the QuadroM32x'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.
13471	T:	An issue with configuring IP phones as local extensions for Quadro IP PBX
	D:	PnP and auto configuration of IP phones is impossible with the modified HTTP Server Port on the QuadroM32x.
	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.

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 ZeroOut and entering the VMS directly with option **1** will not work.
	C:	
	Fix:	This is normal, as those features are the QuadroM32x's internal VMS system features. If external VMS system is used, user gets the features of that external system.
11519	T:	An issue with distinctive ringing on the snom and Aastra IP phones
	D:	Snom and Aastra phones ring only once if the distinctive ringing is enabled with the "winter" ringing pattern.
	C:	
	Fix:	Use other ringing patterns for distinctive ringing on snom and Aastra IP phones.
	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 QuadroM32x 5.3.73.

7.2 QuadroM32x SW Requirements for Upgrading to 5.3.x

Attention: The software upgrade to 5.3.x can **ONLY** be done from 5.2.47 and higher 5.2.x versions. Before updating to 5.3.x the unit should be updated to 5.2.47 or higher 5.2 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 QuadroM32x 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 Statistics** (only when embedded memory storage is used)
  Workaround - to save the existing call statistics, download it to the PC from **Call Statistics→Statistics Settings** before performing the firmware update. It is also recommended to use an external Compact flash memory to keep the call statistics 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 an external Compact flash memory 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 Compact flash memory to keep the voice mails 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 Compact flash memory 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 Compact flash memory 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 config and voice data (**System→Configuration Management→Backup and download all config and voice data**).
2. Perform the Firmware Update.

3. Is there a **Compact Flash** memory card installed?
   - **Yes** - No further action is required.
   - **No** - Restore the configuration that was saved in Step 1 (**System→Configuration Management→Upload and Restore all config and voice data**).

   This is necessary to restore the extension custom voice messages and the custom Auto Attendant messages.

Attention: Always power down the Quadro before inserting/removing any **Compact Flash** memory card. The largest recommended **Compact Flash** memory card is **8GB**.

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.