Release Note QuadroM32x 5.2.48 Edition 1

1 Introduction

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

QuadroM32x software 5.2.48 Date: September 26, 2011

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

Date: October 4, 2011

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

- A software upgrade can be made from 4.1.13 or later software.

2.3 Supported and Tested SIP phones

Below is the list of SIP phones with the corresponding firmware versions that are supported by Epygi and are recommended for optimal compatibility with QuadroM32x.

Vendor	Model	Software
snom	300	Application: snom300-SIP 8.4.32
snom	320	Application: snom320-SIP 8.4.32
snom	360	Application: snom360-SIP 8.4.32
snom	370	Application: snom370-SIP 8.4.32
snom	820	Application: snom820-SIP 8.4.32
snom	821	Application: snom821-SIP 8.4.32
snom	870	Application: snom870-SIP 8.4.32
snom	MeetingPoint	Application: snomMP-SIP 8.4.32
Aastra	480i	Application version: 1.4.3.23
Aastra	480iCT	Application version: 1.4.3.23
Aastra	9112i	Application version: 1.4.3.23
Aastra	9133i	Application version: 1.4.3.23
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	6751i	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: 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.

In the list below are the SIP phones with the corresponding firmware versions that are tested by Epygi. The document **Quadro Features on Tested IP Phones** describes the problems and limitations on the SIP phones that have been tested by Epygi.

Vendor	Model	Software
Cisco	7960	POS3-07-4.00
Cisco	SPA525G2	7.4.8
Cisco	SPA303	7.4.8
Cisco	SPA501G	7.4.8
Cisco	SPA509G	7.4.8
IpDialog	SipTone II	Application: SipTone 1.2.0 rc Z_21
Polycom	SoundPoint IP 300SIP	Application: SIP software 2.1.3
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 501SIP	Application: SIP software 3.1.3.0439
Polycom	SoundPoint IP 601SIP	Application: SIP software 3.1.3.0439
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__
SIPUra	SPA 841	3.1.4(a)
snom	PA1	8.4.32
snom	m3	Application: snom-m3-SIP 2.2
snom	190	Application: snom190-SIP 3.60s
snom	200	Application version: 3.56m
snom	220	Application version: 3.56m
Swissvoice	IP 10S	Application version:IP10 SP v1.0.1 b3
Thomson	ST2030S	APP Version: V2.69
Berkshire (ATL)	5000	SW version: 22102154
Grandstream	BT200	Program- 1.1.6.16
Grandstream	BT100	Program- 1.1.0.3
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
Grandstream	HT386	Program- 1.0.3.96
Linksys	SPA921	SW version:5.1.8
Linksys	SPA922	SW version:5.2.8
Linksys	SPA941	SW version:5.1.8
Linksys	SPA942	SW version:5.2.8
Linksys	SPA2002	3.1.20
Linksys	PAP2T	3.1.15(LS)
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	VP-2009/VP-2009P	22.21.0.40/22.30.0.60
Aastra	MBU400	Application version: 2.20
Telematrix	IP550 Spectrum Plus	SW version:1.7.252.253
Telematrix	3300IP	SW version:1.7.249.248

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 Quadro 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 QuadroM32x SW 5.2.48.

QuadroFXS 16 Gateway should have SW 5.2.1 or higher for PnP configuration with the QuadroM32x SW 5.2.48.

ActiveX Control SW 5.3.0 or higher should be used with 3pCC functions with the QuadroM32x SW 5.2.48.

Auto Dialer SW 1.0.5 or higher should be used with the Quadro32x SW 5.2.48.

Quadro SMR system SW 1.9 or higher should be used with the QuadroM32x SW 5.2.48.

To use QuadroM32x SW 5.2.48 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 QuadroM32x release.

Release	New Features
5.2.48	PnP and auto configuration support for Grandstream GXP1400/1405/1450/2100/2110 IP phones.
	Basic auto configuration support for Polycom KIRK Wireless server 6000 IP phone.
5.2.47	
5.2.46	Security enhancement. A new security check has been added to prevent an external 3PCC or Click2Dial application from using any of the Quadro features unless specifically authorized from the user's extension settings. The new option (3pcc/Click2Dial Login Allowed) is displayed when doing an edit on the extension and is disabled by default. **Please note**. If the Quadro extension is currently being used with some 3PCC or Click2Dial application, then this option will need to be enabled after the firmware update or the feature will no longer work. Epygi applications that use 3PCC, such as HotCall and HotKeyCall, do not rely on this setting to be enabled to work.
	Auto Attendant Customized builder to create Interactive Voice Response (IVR) scenarios from the Quadro GUI.
	Support for the new Epygi AutoDialer application (with a license key).
	TDVC and H.263+ codecs support.
	Language pack support for Polycom IP phones. Polycom phones change the GUI to the installed custom language.
	Local Admin password management enhanced. The local admin login password for GUI access is alphanumeric and a second numerical password can be entered for accessing settings from the phone handset, such as the Auto Attendant greeting messages.
	The Administrator password for phones is added as an item in the security audit.
	Auto configuration support for Cisco SPA303, SPA501G, SPA509G, SPA525G2 phones.
	PnP and auto configuration support for Grandstream GXP2120 and GXV3175 phones.
	PnP and auto configuration support for Yealink VP-2009P model flavor.
5.2.40	
5.2.39	Support for Video Conferencing (with a license key).
	PnP support for Grandstream GXV3140 video phone
5.2.26	New Quadro System Security Management page added with security diagnostics and a security audit
	Secure system mail allowing to work with TLS or SSL mail servers
	A new option in Call Routing - Local AAA Table: Authentication by PIN code
	Possibility to enable/disable TLS functionality for IP Lines and extensions
	A new option in the extension voice mail settings under the Send new voicemail notifications via e-mail, allowing differentiating between fax messages and voicemail messages when deciding whether or not to receive email notifications, with or without attachments.
	Admin password management enhanced. The admin login password for GUI access is alphanumeric and a second numerical password can be entered for accessing settings from the phone handset, such as the Auto Attendant greeting messages.
	PnP and auto configuration support for snom 821
	PnP and auto configuration support for Aastra 6739i phone
	Auto configuration support for Grandstream GXV3140 video phone
	PnP and auto configuration support for Yealink VP-2009 video phone
	Auto configuration support for snom PA1 paging system
	Polycom KIRK Wireless server 300 phone support
	Call recording soft button implementation for Polycom phones

Release	New Features
	Downloaded call statistics enhancements to include detailed information regarding the used interface for each call (PSTN, ISDN, etc), if call had been forwarded, transferred, etc.
	Viewing the Call Statistics on the Quadro GUI has been updated to select the number of records per page
5.2.12	
5.2.10	Full support for Secure RTP (SRTP) to encrypt voice packets.
	Full support for TLS (Transport Layer Security) protocol (RFC 2246) to provide communications privacy for the SIP messaging port over the Internet. The protocol allows client/server applications to communicate in a way that is designed to prevent eavesdropping of the SIP messages.
	HD codecs such as G.722 and G.722.1 are now supported in pass-through mode.
	Security enhancement. Password strength checking has been added in the IP Lines settings table for configuring IP Phones. The system will provide feedback on whether the chosen password is weak. An option is also provided to let the system create a strong, random password. Password strength enforcement can be configured in the hidden menu generalconfig.cgi.
	Security enhancement. Protection against certain SIP DoS attacks has been added by enabling the "SIP IDS" feature in the hidden menu generalconfig.cgi. Enabling this option will automatically block malicious IPs in the firewall.
	Global speed dialing. Configured in Telephony/Call Routing/Global Speed Dial Directory. An Excel .CSV file can be used to upload the entries in the format: Code, Number (e.g. 44, 9726921166). Each entry should be on a new line.
	Emergency Call Interrupt feature in the Caller ID based services. A new feature that will allow specific Caller Ids to be configured so that a call received from one of these numbers will immediately disconnect the call in progress and ring the user's extension.
	System logging enhancement. If a user encounters a problem while on a phone call the pertinent details related to the call can be saved. The user will dial *82 (getting dialtone and dial *82 from the same or a different phone) as soon as the problem occurs. The details will be saved until the next time someone enters *82. When the systemlogs are downloaded it will contain the details from the most recent *82 entry.
	The voicemail to email option can now be configured in Telephony/Voice Mail Common Settings so that the subject of the email contains a user defined text along with selected system variables (e.g. caller id).
	Option to select the format of FAX in Voice Mail sent by email. Defined in Telephony/Voice Mail Common Settings. TIFF or PDF can be used.
	A prompt to the user is provided with the option to save the system configuration prior to executing a firmware update
	Authorized phones database can now be used with a custom auto attendant scenario. This allows callers to bypass authentication in custom auto attendants for user's who have been included in the authorized phone database.
	Configurable auto attendant Callback Response Delay allows users to specify a timeout before the system will try to call back the user.
	Configurable auto attendant callback activation timeout from the GUI, which allows activating automatic Call Back for trusted PSTN and SIP callers not after the one - two rings as previously defined, but during the time frame specified in the GUI. So now users can configure the activation timeout in seconds needed to wait for initiating the callback. This configuration is accessible from the hidden menu "extmanagement.cgi".

Release	New Features
	Allow the Automatic Callback feature to be used to call back to the originating caller using the "Auto" call type. User can select "Auto" as the "Callback Call Type", while leaving the "Callback Destination" field empty. This will allow to callback to the caller using the Caller ID for callback.
	Quadro FXS gateway PnP support
	Full VLAN support for IP Phones (multiple subnets/DHCP)
	Support for T38 FAX proxy (from standalone IP Phone in the LAN to outside)
	Yealink IP phone enhancements. PnP and auto configuration support for T20, T22 and T26 models. Advanced button configuration added in the IP Line Settings. A custom logo image can now be displayed.
	Polycom IP phone enhancements. PnP and auto configuration for Polycom IP335 and IP450 models. Advanced button configuration added in the IP Line Settings. Receptionist configuration support for add on modules. Custom data files for Polycom IP phones can now be uploaded from the hidden menu ipphonesdirectory.cgi.
	PnP and auto configuration for Aastra 6730i phone
	PnP and auto configuration for snom 870 and snom MeetingPoint
5.1.38	Ability to configure speed-dial buttons of snom IP phones via IP phone template mechanism
5.1.31	Telematrix IP phones auto configuration support
	Call Forking (FindMe/FollowMe service)
	Call max duration (disconnecting call after the specified time)
	ACD: zero-out
	Yealink phone auto configuration
	Reason header in SIP CANCEL message (prevents the call from being reported as "missed" if it was picked up by another extension)
	Auto configuration and compatibility for Polycom VVX 1500 video phone
	ACD: agent easy login/logout buttons (including the status indication)
	Automatic logout for HotDesking phones
	Hot Desking capability enabled for remote extensions
	Line appearance added on Remote Extensions
	Park slot segmentation (access list for call parking and retrieve)
	3PCC: capability for 3pcc programmer to modify username
	Barge-in/Whisper/Listening (*93,*92,*91) (with a license key)
	ACD: ACD calls to agent should never reach their voicemail

Release	New Features
	DHCP support on VLAN (including IP phones VLAN support)
	Shared Voice Mailbox
	Call routing table debug/tracing tools (new events raised from Routing rule: call failover, fail, success)
	Aastra 6731i autoconfiguration and PnP
	Polycom Soundpoint IP 331 auto configuration and PnP
	3PCC: ability to differentiate call failure reasons (different error codes)
	Added possibility to download the call statistic in CSV format
	Added possibility to play Hold Music to conference participants unless moderator will join the conference
	Added possibility for conference participants to mute/unmute their phones during the conference by pressing pound sign (#)
5.1.21	Added possibility for unregistered user to activate the conference call by entering the password for conference moderator
	Added zero-out option to the ACD call queue
5.1.18	Automatic Call Distribution support (with a license key)
	Call recording (with a license key)
	Music on hold can originate from an audio source that is streaming music or broadcasting live via an IP connection (i.e. internet radio channel).
	The Quadro Call Statistics (CDR) can be automatically sent via email or uploaded to an FTP or TFTP server.
	While playing back a voice mail the user can rewind (pressing "4") or skip forward (pressing "6"). The amount of seconds to advance or rewind can be optioned for each extension. Default = 3 secs.
	Plug and Play for Polycom IP Phones
	VLAN support
	Click To Dial (CTD) feature updated to allow calls to be initiated from a Website, connecting two parties together via the Quadro. There are two options to make the call: (a) Using the HTTP request to specific URI from a third party WEB page or application; (b) Using a special hidden page on Quadro's WebGUI (in the hidden menu callcontrol.cgi)
	New system events added to track IP phones registration status
	A custom logo image can be displayed on snom 360, 370 and 820 IP Phones
	Speed Dial numbers can now be updated from the phone handset (*76)
	Outgoing Call Blocking can be updated from the phone handset (*79)
	Hot Desking feature. This allows a single phone (IP or analogue phone) to be used by many users. Users can login to the Hot Desking phone (*78) using their extension number, and get all the credentials of their extension until they logout (*78).
	Virtual extensions can be optioned to provide a simulated ring back. The VE can be assigned to a receptionist phone and this option will now give him/her the option to answer the call before it goes to voice mail.

Release	New Features
	Zero-out on Auto Attendant
	The Automatic Call Back option has been expanded to include callback to SIP callers. Previously, this feature was specific to PSTN callers.
	TLS/SSIP support added to send and receive encrypted SIP messages from designated peers
	Video phone support (proxy mode)
	E1 MELCAS digital hookflash feature added for E1 CAS to SIP calls
	The Unconditional Call Forwarding option has been enhanced so that enabling/disabling the feature from the handset can include the designated caller Ids.
	Load balancing for the FXO lines (including FXO gateways)
	Easier for extensions to forward their calls to external numbers without compromising the security of Call Routing entries that have the PBX filtering option enabled.
	Possibility for a user to designate a bad call in the system logs by dialing *81 after releasing the bad call.
	The Quadro system voice messages can be browsed and updated from a PC using 3PCC
	New features in 3pcc: Using the SIP Dialog package the call states for extensions can be viewed and tracked.
	New features in 3pcc: This feature allows the 3pcc application to subscribe to the Quadro's extension's list and receive an event notification the list has changed.
	New features in 3pcc: The connect function has been added which allows the application to use an extension's authentication parameters (extension number, password). After successful connection establishment an application can manipulate options on the extension.
	New features in 3pcc: Capability to upload the contents of legible configuration files and do the configuration update via 3pCC
	An option to "Discard Non-Numeric Symbols" is added in the Call Routing Wizard to delete extra characters that might be received in the caller id.
	Possibility to append the called number to the caller id using the format <dialednum:range>. This will allow the user to appropriately answer a call based on the number that was dialed.
	snom M3 support
	snom820 auto-configuration support
	Ability to download "packages" of CGI's in legible format
	The configuration templates that the Quadro uses for the IP Phones can now be modified.
	Ability to remotely access the Auto Attendant and use the admin login menu (*75) to update the AA greeting messages. Previously this login was only accessible from an extension.
	Capability to configure emergency numbers from the VoIP Carrier Wizard
5.0.22	
5.0.21	Automatic call forwarding on extensions with Unregistered/Inaccessible IP phones
5.0.17	Capability to switch via phone the Unconditional Call Forwarding to "ON/OFF" for callers other than "Any/Other callers"
	Random call hunting

Release	New Features
	Capability to choose desired GUI theme on login
	Capability to Show/Hide extensions attached to disabled IP lines in the Extensions Management
	Capability to Show/Hide the disabled IP lines
	Auto configuration support for the following Linksys IP phone models: SPA941, SPA942, SPA921, SPA922
	Ability to remove the Display name of the call passing through Routing Table
	New Auto Attendant or extension adding process simplified
	Add mechanism for the DND and CallFwd functionality of snom and Aastra IP phones.
5.0.10	Support for Conference Server (with a license key).
5.0.8	Ability to record voice messages from the phone handset that are referenced from the Auto Attendant Interactive Voice Response menu. When needed, the recordings can then be easily updated with current messages.
	Support for MS OCS and MS Exchange configuration with Quadro
	It is now possible to reboot each IP phone separately from the Quadro's GUI
	Added receptionist wizard and expansion module support for Grandstream GXP2000
	Programmable Keys Configuration page added in the IP Line Settings for snom, Aastra, (except for snom300 and Aastra51i, 9112i), Grandstream GXP2000. There is an advanced link on IP Lines page to configure the programmable buttons of each phone models for the "watching" and for the standard services like "DND" and others
	Extended flexibility for the Intercom feature: added PBX-Intercom call type in Call Routing wizard and 3 options to enable Intercom service
	A new checkbox called "Check with 3PCC" is added in Call Routing Wizard. When checking it will allow 3PCC application to place a hook on calls done through that rule
	Local authentication parameters are shown in Details column of Call Statistics
	"Check/Update Now" option is added in Automatic Firmware Update page
	Added SNMP status and configuration
	The results of system configuration backup & download are reflected in the system events
	snom, Aastra and Grandstream phones automatically get rebooted when the Quadro is rebooted
	Added auto configuration support for the following Polycom IP phone models: Soundpoint IP 330, Soundpoint IP 550 and Soundpoint IP 650
	Intercom and paging support for Grandstream IP phones
	Possibility to hide/show disabled records in the call routing table
	SIP P-Asserted-Identity support for SIP calls to ISDN network
	Caller ID can be used to define the call destination in call routing table
	The "Deny reinvites" option is added for ITSPs that do not support this option.

Release	New Features
	Call forwarding configuration through custom Auto Attendant
	Added two stage dialing for FXS calls
	G729/G729a codec type selection is GUI configurable
	Added dependence for Server status and E1/T1 link status
	Language pack support for the IP Phones
	Extended auto configuration capabilities for the Aastra phones. For Aastra phones the "Time-zone" setting has been added to set up this parameter according to Quadro regional settings.
	Changed the Auto Attendant custom scenario to allow multiple WAV files to be uploaded all at once in a single compressed ZIP file.
	Added possibility to change the original display name in the Caller ID
	Added Kickback feature allowing not to lose the call after blind transfer
	Changed the Subscription limitations and made it configurable from generalconfig.cgi hidden page.
4.1.15	
4.1.14	Maximum number of IP lines is 192. 32 lines by default, one expansion of 32 and 2 more expansions of 64 each
4.1.13	Receptionists' call queue
	Extension status monitoring
	Call interception
	Direct transfer to voicemail
	Simultaneous ring
	Call park
	Call pickup group
	Call paging group
	Plug and play for supported IP phones
	Simplified installation (auto configuration) for tested IP phones
	User rights management
	Support for an external voicemail system
	Silence detection on voicemail messages to truncate extended periods of silence
	QuadroFXO GW and QuadroISDN GW auto configuration
	Proprietary method for remote extension to bypass certain types of NAT (symmetric RTP)
	Status for registered IP lines and remote extensions

Release	New Features
	Readable transfer statistics
	New voice mail notifications via e-mail, via phone call and via SMS
	Universal extension recording (music on hold, VM greetings, incoming/outgoing blocking messages)
	Sending unified fax message from mailbox to fax machine
	System events reporting
	Kickback feature on IP lines
	snom and Aastra phones automatically reboot when the Quadro is rebooted
	IP phones do not lose registration after Quadro soft reboot (i.e. if the reboot is not initiated by disconnecting/reconnecting the power or pressing the reset button on the back of the Quadro)
	Possibility to upload and customize system messages (in the uploadownmessages.cgi hidden page)
	Attendant ringing announcement for both default and custom AA scenarios
	Possibility to change the default memory size to be allocated as a temporary storage for unified FAX message (in the extmanagement.cgi hidden page)
	Support for HotCall Add-In for use with Microsoft Outlook
	SIP address resetting function in the Extensions Management for restoring the default SIP settings of extensions
	Possibility to change static, running logos, links and browser mini logo (accessible from loadlogo.cgi hidden page)
	Possibility to change the signaling procedure for hold process in case the "Send Hold Music to Remote Party" is enabled on the extension
	Voicemail setup wizard to guide new users through the recording of VM greetings
	SIP trunk
	3rd party call control
	Flexible numbering plan for extensions
	Download IP phone FW and reboot of Aastra/snom IP phones
	Download Quadro directory via XML to snom/Aastra phones
	Automatic Quadro FW update
	Easier GUI for forwarding lists
	Terminate a call from Active Calls page
	Enhanced method for receptionists to watch other extensions, resulting in less networking traffic
	Distinctive ringing on snom and Aastra IP phones
	Play custom message instead of ring back tone on Auto Attendant
	Send EVENTS via SMS
	CDR tags for billing

Release	New Features
	Enhanced DNS/DHCP settings
	Daylight savings adjustment
	SMS and E-mail notification when a call to a given extn. has been forwarded
	Change extension password from phone handset
	External access to the voice mailbox of users' extensions.
	Automatic backup and download of Quadro configuration
	QCM centralized licensing included
	SNMP support for event monitoring
	Intercom service (for supported IP phones)
	Paging group service (for supported IP phones)
	Redirection on timeout to the specified address if no action on the AA welcome message
	AA welcome and menu messages enabling/disabling capability
	Call queue for virtual extensions
	Toggling for call routing entries from the handset
	Retrieve timeout configuration for parked call
	SDMF caller ID standard
	Routing rules activity timeout
	Capability to make SIP calls using TCP as a transport protocol
	"Client Code Identification" feature on routing rule (with RADIUS enabled)
	Paging support with 3rd party devices
	Custom profiles for voice mail settings
	Added silent VM recording option in the Voice Mail Settings
	PSTN call instant callback
	CCS Signaling
	CAS Signaling
	Support of 2-stage dialing for E1/T1

4 Changed Features History

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

Release	Changed Features
5.2.48	New FW recommendations for snom, Grandstream and Polycom phones.
5.2.47	
5.2.46	The maximum number of Successful Call Statistics has been increased from 1000 to 10000 when a CF card is installed.
	Added option in ACD configuration to prevent the calls to agent in target group during selected period.
	Added information regarding video codec's under the Details in the call statistics
	Added P-asserted identity header in all SIP messages that contain "p-preffered identity" or "Remote-party-ID".
	Changed the SIP parser for proper processing of From: field in header if URI scheme is included.
	The SIP IDS configuration options are available in the SIP IDS settings page.
	The "3pcc/Click2Dial Login Allowed" checkbox is available when editing multiple extensions at the same time.
5.2.40	
5.2.39	Multiple SMS numbers can be allowed in SMS destination fields, separated by SMS provider required delimiter (the comma for Clickatell)
	The SoTel SIP trunk is modified in the VoIP Carrier Wizard
	The Skype service is added in the VoIP Carrier list
5.2.26	The Call Recording settings page is updated with a second, more simplistic method for enabling extensions. Either of two modes can be used: Basic view and Advanced view.
	Improvements in Call Recordings FTP retransmission mechanism.
	Extension Login permission to access the Quadro GUI. The login is disabled by default for all extensions.
	Multiple DHCP subnet support on M32x (two subnets allowed instead of one, making it possible to have about 500 devices getting DHCP from Quadro).
	LMR (Land Mobile Radio) support for QuadroM (E1/T1 to IP gateway capable of taking a channelized T1 to Multicast RTP IP streams by DS0 for use in a Radio over IP network)
	The "Send Hold Music to remote IP party" is enabled by default in the Basic services for extensions
	The Developer Logging is enabled by default
	The SIP Server text field is changed to empty by default for auto attendant (00) and two FXS extensions.
	The default for "Maximum recording duration" is changed to 1 hour in the "Recording box settings" and "Call recording settings"
	The Secure RTP options in the Extension Codecs are available when editing multiple extensions at the same time.
5.2.12	
5.2.10	Changed the behavior of intercom and paging to not interrupt current active call by default
	Intercom option has been changed so that an audible sound is now provided to the called phone to alert the person when the microphone is opened.
	Indication of BLF/MWI subscription count and raising an event if the upper limit is reached
	Allowing to have multiple recording sessions regardless of the "max mail/recording duration" setting on the VM/recording box

Release	Changed Features
	FTP retransmission mechanism improvement and manual FTP upload request from WebGUI. Call recordings can be manually uploaded from Web GUI.
5.1.38	The SIP server value for Broadvox in the VoIP Carrier Wizard was changed
	Added an option in the Automatically Download Call Statistic page to choose the download format
	The VM-to-email text for FAX mail was changed to be different from the regular voicemail text
	An audible sound to the called phone is provided in case of intercom call
5.1.31	The common limitation of 300 as the maximum recordings count for recording boxes has been eliminated. Now the call recording count is configurable per recording box extension, the available range is of 50-2000 recordings
	The configuration for the call recording is redesigned
	After using the feature code *79 (for outgoing call blocking) from phone handset to block a dialed number the "Send Message to Caller Party " option should be enabled automatically
	The limitation for the number of entries in the local AAA table has been eliminated
5.1.21	Added new VoIP carriers to the VoIP Carrier Wizard list (Acf2i and babyTEL)
	The "Automatic Firmware Update" link is moved from "Configuration Management" to "Firmware update" page
	The auto-configuration and PnP support are added for Polycom SoundStation IP 6000
	The auto-configuration support is added for Yealink T28 IP phone
	Changed the view of advanced configuration page for the snom 820 IP phone
5.1.18	Added new VoIP carriers to the VoIP Carrier Wizard list
	Adding extensions to Paging and Call Pickup groups has been simplified
	The expiration date for trial keys are shown under the Features menu
	The Call Routing Management option for the Auto Attendant has been changed from *7 to *77. This option allows entries in the Call Routing Table to be remotely enabled/disabled.
	Possibility to multi-edit the some of the extension's Voice Mailbox Settings
	A prefix is added to the Caller ID to identify calls returned by the Kickback option
	It is possible to upload the Call Queue messages as Universal Extension Recordings
5.0.22	
5.0.21	
5.0.17	In the System-> IP routing Configuration the name of "VPN Routes" changed to "PPTP/L2TP Routes
5.0.10	Adjusted timezone parameters for Venezuela according to new standards
	The Fail Reason in the call routing changed as group of check boxes instead of drop down list
	Added status info in 3PCC settings

Release	Changed Features
	Added scrollbars in the tables for Call Routing and Extension Management pages
5.0.8	For snom phones, the "Call join on Xfer (2 calls)" setting is configurable in the phone GUI
	Adjusted timezone parameters for Venezuela according to new standards
	Added status info in 3PCC settings
	Added scrollbars in the tables for Call Routing and Extension Management pages
4.1.15	Changed default attendant name from 000 to 00
	Added PSTN Access Code and Emergency Number page in System Configuration Wizard.
	Removed PSTN Access Code from External PSTN Gateways page
	Changed default E1/T1 mode to User Mode
	Changed default VM indication for IP Lines from Tone to Lamp and for 2 FXS Lines from Lamp to Tone.
4.1.14	FXS settings(filters) changes for Brazil
4.1.13	

5 Fixed Issues

Issues fixed since version 5.2.47:

The customized queue messages are not played when the skill based call comes to ACD	17039
D:	When the call comes first to Custom Attendant, then goes to ACD with chosen skills and enters to some ACD group with customize queue scenario configured with custom messages, the custom messages are not played.
C:	

When one agent answers the call, other agents do not stop ringing	17040
D:	The ACD group is set to "All agents ringing". One agent answers the call. Other agents at this point should stop ringing, but they don't.
C:	

"Download Current CA Root Certificate" link doesn't work	17031
D:	After pressing the "Download Current CA Root Certificate" link on the SIP Settings page, system forward you to home page instead of starting the downloading.
C:	

Abnormal time shift in some rare cases	17044
D:	In rare cases when Quadro corrects the time, the time zone value is not taken into account. As a result, time shifts with the shift value equal to the time zone difference with GMT 00:00.
C:	

The Advanced configuration page of snom821 phone is incorrect	16954
D:	The advanced page of snom821 appears like for 360/370 with 12 configurable buttons, but the snom821 is like snom820 and has only 4 buttons on its case.
C:	

The call will fail when pressing hold or transfer on snom IP phones using firmware version 8.x.x	16734
D:	This problem will happen if one of options: "make unsecure calls, accept anything" or "try to establish secure calls, accept anything" is used in Secure RTP settings for the extension.
C:	

It isn't possible to pickup the call addressed to watched extension using the programmable key on Grandstream GXP2120	16997
D:	
C:	

6 Known Issues

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

It is impossible to login as 'admin' after making changes on "Advanced PPP Settings" page	**9549**
D:	You cannot login to the Quadro as 'admin' after making changes on the "Advanced PPP Settings" page. You can only login as 'user'.
C:	No consequences
Fix:	Restart the Quadro to resolve this.
One way audio with IPSec VPN in the scenario "Gateway<->Remote Subnet"	**9327**
D:	Scenario: **1.** Establish an IP-Sec connection between two Quadros (connection type is "Quadro <-> Remote Subnet"). **2.** Make an SIP call from Quadro1 (gateway) to Quadro2 (remote subnet).
C:	The call is established, but there is no audio in Quadro1. The same result will occur when calling from Quadro2 to Quadro1. A direct call cannot be established at all.
Fix:	Add an additional "Quadro <> Remote Gateway" tunnel in IPSec configuration. With two tunnels: "Quadro <> Remote Gateway", "Quadro <> Remote Subnet" in IPSec VPN configuration this issue will resolved.
Call interception fails in a specific scenario with a watched extension in the many extension ring list	**9399**
D:	Scenario: **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 light for ext. #11 will stay ON on the ext. #31.
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.
An issue with configuring IP phones as local extensions for Quadro IP PBX	**13471**
D:	Plug and Play and Auto configuration of IP phones is impossible with the modified "HTTP Server Port" on the Quadro.
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 Quadro 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 Quadro'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 extensions 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 doesn't contain spaces. This problem is limited to Aastra IP phones only.

An issue with configuring Aastra IP phones as local extensions for Quadro IP PBX
13802

D:	After changing the LAN IP address for the Quadro 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 and Aastra 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.

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 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 Quadro LAN IP configuration (IP address or subnet mask) IP phones lose registration and become unusable 16037	
D:	
C:	
Fix:	After changing Quadro LAN IP configuration first reboot the Quadro then reboot the IP phones. Will be fixed in the next release.

An issue with Emergency Interrupt feature on snom IP phones 16391	
D:	The Emergency Interrupt fails when snom phone is in a 3-way call. Instead of ringing the phone call goes to VM directly
C:	
Fix:	Will be fixed in the next releases.

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:	
Fix:	Will be fixed in the next releases.

An issue when using the variable "Date" in the subject of voice mail send by email 16150	
D:	If the subject $[VM_DATE] is placed at the end of the subject line, then Voice Mail will not send as an attachment.
C:	Add an underscore (_) after the $[VM_DATE]. Or put the $[VM_DATE] in the beginning or between other subjects in the subject line.
Fix:	Will be fixed in the next releases.

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 can't 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, Quadro 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 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.

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 by the Quadro and will cause the IP phone registration failure on the Quadro.

7.2 General hints

It is recommended to execute the update by downloading the software first to a PC located in the LAN side of the Quadro 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**
- **DHCP leases**

The following steps describe how to correctly perform the firmware update to avoid losing the custom voice greetings and recordings for custom auto attendants:

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.

All custom voice greetings and all recordings for the custom Auto Attendants can be restored by uploading the configuration that was saved in Step1 (**System** -> **Configuration Management** -> **Upload and Restore all config and voice data**).

Attention: Power down the Quadro before inserting/removing Compact Flash card.

7.3 Limitations and restrictions

- The memory used by "DSP Capture", "Call Capture" and "Network Capture" hidden pages is limited to 3 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 300
- Use Session Timer in IP Line Settings is deselected by default