Release Note QuadroM-E1T1 5.2.6, Edition 1

1 Introduction

The following release notes describe hardware and software requirements to use the

QuadroM-E1T1 software 5.2.6 Date: April 11, 2011

Enhancements, bug fixes and known issues incorporated in this software will also be listed.

Date: April 15, 2011

2 Requirements

2.1 Hardware requirements

The software can be used on all QuadroM-E1T1 models:

The model name is written on the back plate of the unit.

2.2 Software requirements

Software upgrade can be made without issues from software versions higher than **4.1.10**.

3 New Features History

The table below indicates a high-level list of new features that have been added beginning with the most recent QuadroM-E1T1 release.

Release	New Features
5.2.6	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
	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.
	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
	Added ISDN PRI Display Name support for both User/Network modes
5.2.1	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. 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.
	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".
	Allow the Automatic Callback feature to be used to call back to the originating

Release	New Features
	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.
5.1.5	Call max duration (disconnecting call after specified time)
	Call routing table debug/tracing tools (new events raised from Routing rule: call failover, fail, success)
	Music on Hold can originate from the Epygi Media Streamer which is a software application running on a PC that can have a playlist of MP3 files.
	The Quadro Call Statistics (CDR) can be automatically sent via email or uploaded to an FTP or TFTP server.
	Zero-out on Auto Attendant
	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.
	Ability to download "packages" of CGI's in legible format
	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
	Added support for P-Asserted-Identity for the calls coming from CAS to SIP and vice versa.
5.0.21	Capability to choose GUI desired theme in login
	Ability to record voice messages from a phone that are referenced from the Auto Attendant Interactive Voice Response menu. When needed, the recordings can then be easily updated with current messages.
	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.
	Added dependence for Server status and E1/T1 link status
	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
	Changed the Subscription limitations and made it configurable from generalconfig.cgi hidden page.
	New Auto Attendant or extension adding process simplified
	DHCP Advanced Settings added
	Local authentication parameters are shown in Details column of Call Statistics
	Added SNMP status and configuration

Release	New Features
	The results of system configuration backup & download are reflected in the system events
	Possibility to hide/show disabled records in the call routing table
5.0.15	Added SIP tunneling feature
	Added an option to send ISDN L3 ALERT message on call ringing
	Active calls are shown in the home page
	Added VLAN support (from the vlansettings.cgi hidden page)
	Added possibility to limit the size for View System Logs
	"Check/Update Now" option is added in Automatic Firmware Update page
5.0.8	Added VPN support
	DSP counts are shown in the Hardware Status page
	Allow E1/T1 mixed configuration for dual units with DSP card (SASY-MIDI-0101-2200 in SKU-QM-0004). In case of mixed E1/T1 configuration the user will need to deselect the "Enable TDM Pass Through" checkbox from tdmpassthrough.cgi hidden page and reboot the unit. Available limitations: both E1/T1 trunks should be configured either in CAS or CCS signaling mode.
	Added the possibility to block future calls without disconnecting the existing calls. The blocking can be activated from E1/T1 Settings page by disabling E1/T1 timeslots for the calls from SIP to ISDN PRI and for the calls from ISDN PRI.
	Added possibility to modify Caller ID based on the field "Inbound Caller Pattern"
	Added TDM switch support
	G729/G729a codec type selection is GUI configurable (in the codecsconfig.cgi hidden page)
	Added dependence for Server status and E1/T1 link status
4.1.30 (QuadroM)	Added support of P-Asserted-Identity header field (RFC 3325)
	Added possibility to configure the DTMF digit/pause durations from GUI
4.1.22 (QuadroM)	Added possibility to change static, running logos, links and browser mini logo (accessible at loadlogo.cgi hidden page)
4.1.18 (QuadroM)	Added dual E1/T1 support. **Limitations for using Dual E1/T1:** 1. Impossible to configure the first E1/T1 interface to E1 mode and the second to T1 mode or vice versa 2. Impossible to configure the first E1/T1 interface to CAS mode and the second to CCS mode and vice versa
4.1.17 (QuadroM)	SIP Address Resetting function in the Extensions Management for restoring the default SIP settings of extensions
4.1.12 (QuadroM)	Support for 30 simultaneous calls with G.729 codec

Release	New Features	
3.2.1	Interdigit service for E1/T1 CAS	
3.1.15	"Backup & Automatically download all config & voice data" feature via E-MAIL, FTP, TFTP.	
	E1/T1 settings to legible configuration file for editing.	
	Hold music options.	
	CallBack feature for E1/T1 CAS/CCS signaling.	
	Support for keypad buttons #,*, A, B, C, D for E1/T1 CAS signaling.	
	"On Channel Selection" option in CAS signalling wizard for "Voice Establishment Procedure".	
	Alternative Disconnection Mode checkbox for E1/T1 CAS User/Network mode.	
	System Remote Logging.	
3.1.7	SNMP support for event monitoring.	
	Centralized Management.	
	E1/T1 FGD (Feature Group-D) signaling type.	
	Diagnostic call loopback mode for E1/T1.	
	TOS for SIP.	
	Voice transcoding on QuadroE1T1.	
	No Answer Disconnect Timer For E1/T1 CCS Network mode.	
3.1.7	Implemented mapping of ISDN release reason to SIP response and SIP response to ISDN release reason.	
	r2settings.cgi hidden page to customize E1 CAS R2 Compelled settings.	
	Activity timeout in call routing rule for Call Type SIP, H323, IP-PSTN.	
	Implemented E1 CAS R2 MF in Network mode.	
3.0.49	Collect Call service for E1 CAS R2 Brazil.	
	E1 CAS R2 for country Mexico.	
	B groups partly enabled for E1 CAS R2	
	The r2tones.cgi hidden page moved to E1 R2 Advanced settings page to be able to change the following parameters:	ANI Request transmit {from A1 to A15}
		ANI Request receive {from A1 to A15}
3.0.49	"Generate Progress Tone to PSTN/PBX " checkbox for ISDN PRI.	

Release	New Features
	Display Name recognition for PRI 5ess switch type when it is being delivered in the Facility Message.
	Capability of forcing its own list of preferred codecs for inbound calls.
	Several AAA combinations for accounting.
	Increased SIP username field length from 20 to 32.

4 Changed Features History

The following table provides a high-level list of changed features that have been changed beginning with the most recent QuadroM-E1T1 release.

Release	Changed Features
5.2.6	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
5.2.1	
5.1.5	The limitation for the number of entries in the local AAA table has been eliminated
	Terminology corrections in the Call Routing WEB GUI
	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 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.
	The "MAC address" widget behavior is changed to be like IP address widget: if user presses 2 digits/letters, the cursor moves to the next text field.
	CAS and ISDN agents should send to CM CallErrorCode_TemporaryUnavailable instead of CallErrorCode_LineBusy in case of no free channels.
	Added ringing Timeout in case of E1 CAS User/Network
	The maximum number of call records in the Call Statistics tables increased up to 1000
5.0.21	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 scrollbars in the tables for Call Routing and Extension Management pages
5.0.15	Allow E1/T1 mixed configuration for all types of dual units. In case of mixed E1/T1 configuration the user doesn't need to deselect the "Enable TDM Pass Through" checkbox from tdmpassthrough.cgi hidden page. Available limitations: both E1/T1 trunks should be configured either in CAS or CCS signaling mode.
5.0.8	Support for P-Asserted-Identity header field (RFC 3325) implemented fully
4.1.30 (QuadroM)	
4.1.22 (QuadroM)	
4.1.18 (QuadroM)	
4.1.17 (QuadroM)	

Release	Changed Features
4.1.12 (QuadroM)	Two new IP-PSTN providers in the "VoIP Carrier Wizard" list (AGN and Netlogic)
3.2.4	Added country specific advanced Answer/Busy signal(s) for E1 CAS R2 Compelled signalling
3.2.1	DSP capture simplified.
3.1.15	Removed Overlap Mode functionality for ISDN Switch Types 5ESS, DMS100.
	Echo Tail default value changed from 128 to 64.
3.1.7	E1 CAS R2 default settings for Paraguay, Peru, Uruguay, Venezuela.
3.0.49	Channel Selection Ordering for E1/T1 CAS outgoing calls.
	ISDN PRI Network mode for sending Dial tone in Overlap mode.
	For ISDN service do not send STATUS message when PROGRESS message is coming after ALERT message.
	Changed ISDN PRI DMS100 Switch type functionality.

5 Fixed issues

Issues fixed since version 5.2.1:

ISDN L3 problem in case of receiving RELEASE_COMPLETE message in DISCONNECT REQUEST state	16721
D:	
C:	
Outgoing call problem with signalling type (R2 Non-Compelled, R2 Non-Compelled with ANI) on CAS	**16491**
D:	In this scenario the call will be dropped in about 15 seconds
C:	
Problem with signalling type E&M Immediate Start	**16613**
D:	When using this signalling type call doesn't close on calling phone after call terminating it by called phone.
C:	

6 Known Issues

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

Problem with the E1/T1 Diagnostics	11922
D:	E1/T1 diagnostics does not provide correct values for the Rx/Tx buffers.
C:	
Fix:	Under investigation, will be fixed in the future releases.
It is recommended to restart the Quadro after Interface Type change for the E1/T1 trunk	**12795**
D:	Scenario: **1.** Go to Telephony – E1/T1 Settings, select a Trunk to edit **2.** Change the Interface type from "User" to "Network" or vice versa
C:	In very rare cases all trunks may become unusable until Quadro is rebooted.
Fix:	Restart the Quadro after interface change. Under investigation, will be fixed in the future releases.

7 Upgrading Instructions

7.1 General hints

It is recommended to execute the update by downloading the software first to a PC located in the LAN 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.

Remember that some data is lost during upgrade:

- **Call Statistics**
  Workaround - to save the existing call statistics, download it to the PC from Call Statistics-Statistics Settings before performing the firmware update.
- **All custom voice greetings**
  Workaround – backup and download all configuration and voice data to a PC.
- **All custom recordings for the custom Auto Attendant**
  Workaround – backup and download all configuration and voice data to a PC
- **Transfer statistics for the network**
- **Event log**
- **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:

- 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).
- 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 Step 1 (System/Configuration Management- Upload and Restore all config and voice data).