Release Notes for QXE1T1 6.2.1, Edition 1

1 Introduction

This Release Notes describes hardware and firmware requirements to use with the

QXE1T1 firmware 6.2.1 Date: December 11, 2017

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

Date: December 11, 2017

2 Requirements

2.1 Hardware Requirements

- This firmware (FW) can be used on QXE1T1 model only.
- The model name is written on the back plate of the unit and the model number is on the bottom label.

2.2 Firmware Requirements

Attention: The firmware upgrade to 6.2.1 can ONLY be done from 6.0.2 and higher versions.

2.3 Interaction with Other Epygi Software Releases

To achieve maximum compatibility with QXE1T1 FW 6.2.1, use the latest SW and FW versions:

- QX20, QX50, QX200, QX500, QX2000, QX3000, QXISDN4+ and ecQX configured in the PSTN lines sharing (master-slave) mode with QXE1T1 should be installed with 6.2.1 or higher FW version.
- QX-Quadro Configuration Console (QCC) SW 2.3 or higher should be used.
- Epygi Media Streamer (EMS) SW 2.4 or higher should be used.

3 New Features

The table below indicates a high-level list of new features that have been added beginning with the most recent QXE1T1 FW release.

Release	New Features
6.2.1	Added support to automatically archive Call History.
	Added the SSH FTP (SFTP) support, which allows to send the configuration backup files to an FTP server using the secure FTP connection.
	The Client Code Identification option can be activated and used by other billing systems as well as it is done for RADIUS server.
	Added support allowing to Restrict Simultaneous Calls for "SIP" call types.
	Added support for the SIP Registration Transport UDP/TCP/TLS options in the Extension's SIP Registration.
	Added Symmetric RTP option for IP Lines Settings. Select this option when the IP phone attached to the IP line is behind the NAT router.
	Uploading audio files for customizing any of the system or extension audio messages on the QXE1T1 has been simplified: • Apart from the files in the (*.wav) format, the system can now accept (*.mp3) files for uploading as custom messages. • The (*.wav) and (*.mp3) files can now be uploaded directly to the system without the need to convert to the proper telephony format. The uploaded files will be automatically converted to the QX supported wav format: (CCITT u-law, 8 kHz, 16-bit, Mono).
	General improvements and enhancements in the SIP TLS certificate.
	GUI enhancements on the Menu bar: • Added device's current Date/Time. • Added device's hostname.

Release	New Features
	GUI enhancements for IP Lines page: • Added support to allow quicker edits when moving between IP Lines. • Added support to allow quicker access to the attached extension's **Admin** and **User** settings.
	GUI improvements and enhancements in the **Extensions Management** page.
6.1.17	Added support for **Hosted PBX Survivability** (**HS**) feature. The feature allows your company's telephones to work, even when the broadband link or Hosted PBX are down.
	Added **IP Phones** support with license key.
	Added the **Keep Original Caller ID** option for E1/T1 trunks in the call routing wizard.
	Added a new **Search** option in the QXE1T1 **Online Help**.
6.1.10	
6.1.6	
6.1.5	Added support for **CCS DID service** in E1/T1 Trunk Settings.
6.0.13	
6.0.10	
6.0.8	
6.0.2	

4 Changed Features

The table below provides a high-level list of changed features that have been changed beginning with the most recent QXE1T1 FW release.

Release	Changed Features
6.2.1	The **PSTN Gateways Line Sharing** mechanism has been changed and updated, bringing more stability, improving the connection between PBXs and Gateways. Important Note: Please update the firmware version to 6.2.1 both on QXE1T1 and QX IP PBX to be able successfully connect the devices and share the lines.
	Hot Desking service enhanced regarding the voice notifications when login/logout on the public phones: • Added voice prompt asking user to login before using the phone. • Added voice prompt notifying user about login extensions in use. • Added voice prompt informing the user about the successful logging out.
	Added **MC-Link**, **Flowroute**, **ClarityTel**, **Adiptel** and **Fusion** as a new carrier to the VoIP Carrier Wizard list.
	Added support to download Extension's Call Detail Records for Successful, Missed and Unsuccessful Outgoing calls, when logged into the system using extension's credentials.
	Added support to exclude/include different **CDR parameters** in generated CDR reports for the Call History.
	Added support to display **SRTP** parameters in the Call History.
	Added **MO=1** parameter in the **SMS Settings**.
	Added support for the following symbols "<", ">" in the password field for **E-mail Settings**.
	The backup configuration filename format has been updated and will include the installed firmware version of the QX: **config_[Hostname]_[Firmware Version]_[Date/Time].bin**

Release	Changed Features
	The timezone database has been updated on QX Gateways: • The current local time has been corrected for Israel, Venezuela, Shri Lanka, Apia, Samoa and Fiji. • Added new timezone Nukualofa, Tonga (GMT+14).
	New **Date/Time** pickers have been implemented for all applicable GUI pages, allowing to select or define the date/time options easier and conveniently.
	Added support to allow/deny access to the **Diagnostics** and **Reboot** pages for QX localadmin.
	The **Network Capture** page has been moved to **Maintenance→Diagnostics→ Network Capture** page.
	The **Status→System Status→Memory** page is redesigned and modernized.
	GUI Enhancements for **E1/T1 trunk** and **DID Service** pages.
	GUI Enhancements for **Call Routing Table.**
	GUI Enhancements on the **Setup→Licensed Features** page.
	The **5061** will be used as default TLS port for SIP.
6.1.17	The maximum **Number of Call Records to Download** is increased to **10000** in the **Status→Call History→Automatic Backup.**
	The maximum length of **Connection Name** field for **PPTP** and **L2TP** has been increased up to **64** symbols. Support for the following symbols "@", "-", ".", "_" is added.
	Added support allowing to enable/disable entries in the **Authorized Phones.**
	The **Blueface Ireland, Blueface Italy and Blueface UK** carriers have been removed from the VoIP Carrier Wizard list.
	Added **SoTel/VoIPLINK** as a new carrier to the VoIP Carrier Wizard list.
6.1.10	Added support in **Extensions Multiple Editing** for the following fields: **Password** and **Confirm password** from General Settings page and **Authentication User Name** from SIP Advanced settings.
	The behavior for the VoIP Carrier Wizard is changed. The new **Authentication by IP Address** checkbox allows bypassing the **Account Name** and **Password** information and filling the **SIP Server** and **SIP Server Port** information only for the carriers not requiring account authentication.
	Added **Blueface Ireland**, **Blueface Italy**, **Blueface UK**, **BINARY NETWORKS**, **IP Directions**, **MyNetFone** and **ThinkTel Communications** as new carriers to the VoIP Carrier Wizard list.
	Kebu.it carrier has been removed from the VoIP Carrier Wizard list.
	The old password will not be required when change the **Phone Access Password** for Administrator.
	The **LAN IP Address** of the backup configuration displayed in the shell window, while restoring previously backed up configuration file.
	Loadlogo.cgi hidden page (for updating company details) has been renamed (changed) to **uploadlogo.cgi**
6.1.6	
6.1.5	
6.0.13	
6.0.10	
6.0.8	
6.0.2	

5 Fixed Issues

Issues fixed since version 6.1.17:

T: Title
D: Description

20021	T:	TCP socket of SMTP session is being closed by QX before the system receives confirmation from SMTP server about received email
	D:	
20012	T:	Unable to load the Call Routing table if there are some not allowed symbols included in the Routing Patterns
	D:	
20009	T:	The E1/T1 filtering option in the Filtering on Source section of the Call Routing Wizard doesn't show all available values
	D:	
19988	T:	Unable to modify two or more extensions at once
	D:	Multiple editing doesn't work for the same type of extensions if in the list of selected extensions there are no any user type extension.
19931	T:	The selected Tracing / Debug options aren't disabled, when you check off the "Tracing / Debug Options" from the Destination Call Type section
	D:	
19922	T:	Address update (modification) of the "Caller ID based Services" for new entries doesn't work properly
	D:	
19921	T:	The expiration/renewal isn't calculated correctly for the "Overall Call Duration Limit" service
	D:	
19917	T:	The codec information isn't shown correctly in the Call History – RTP Statistics page for calls with G726 codec
	D:	
19823	T:	The "welcome" and "recurring prompt" messages for auto attendant Standard scenario are played to caller instead of corresponding messages configured in the attendant VXML scenario
	D:	The "welcome" and "recurring prompt" messages for attendant Standard scenario are played instead of corresponding messages configured in the attendant VXML scenario. This happens only when the caller number is added in the Authorized Phones.
19814	T:	The "Username" and "Password" fields are automatically filled by default values (admin, 19) in the "Routing Call Settings - Edit Entry" section of the Call Routing Wizard
	D:	
19741	T:	The "Overall calling time renewal" for the call routing rule may be discarded because of time change on QXE1T1
	D:	
19726	T:	WEB GUI pages are loaded slowly in case of using HTTPS connection
	D:	
19567	T:	The DHCP Server will not start and work properly if MAC address is changed on QX's WAN interface
	D:	

19533	T:	Call Detail Records (in csv format) are missing the column headers when the file is archived and sent via e-mail or stored to FTP Server
	D:	
19397	T:	Incorrect ESMTP command sequence after establishing TLS negotiation
	D:	Incorrect ESMTP command sequence after establishing TLS negotiation and as a result some e-mail servers are dropping e-mail transaction.
18807	T:	The incoming calls go to the auto attendant for QX GW, instead of going to the attendant for QX IP PBX in a specific scenario with QXE1T1 and QX IP PBX configuration in share mode
	D:	When E1/T1 Route Incoming Call to **Routing with inbound destination number** then the incoming call to AA does not reach Auto Attendant for master (QX IP PBX).
18638	T:	When opening the Call History, sometimes it doesn't show CDR records but shows an empty page
	D:	Only after refreshing the page it shows the CDRs.
18758	T:	"STUN service" doesn't work when firewall level is set to "High"
	D:	
18755	T:	"SNMP Trap" for "Management Access" filtering rule doesn't work when the firewall level is set to "High"
	D:	
18726	T:	An issue with accessing to QX from LAN side with VLAN configured
	D:	If you add VLAN interface on LAN side and the Firewall level is set to **Medium**, you wouldn't have access to device from LAN side (through VLAN interface) until you enable/disable the Firewall.
18419	T:	Cannot establish call if you change signaling type for time slots using CAS Signaling Wizard
	D:	
18397	T:	When you change the Date/Time on the QX, the WEB GUI session will be automatically terminated and you will be logged out.
	D:	

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

17404	T:	Call which is done after Call Relay (*2) on auto attendant is not shown in the 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:	Workaround: Use feature code *1 instead of *2 for call relay. Will be fixed in future release.

7 General Hints

7.1 Technical Advisory

Some system information (Call History and Pending Events) may be lost when QXE1T1 is powered down. You may maximum lose the portion of the above-mentioned system information, which occurred during last hour before the QX is powered down. It's recommended to enable Call History – Archiving to minimize the loss of Call History.

7.2 Firmware Update

It is recommended to execute the update by downloading the firmware first to a PC located in the LAN side of the QXE1T1 and perform the firmware update from the LAN side. This is to ensure that the Internet connection will not affect the upgrade process.

Attention: It is recommended to back up the configuration for emergency purposes prior to upgrading the firmware. You can do that from Maintenance→Backup/Restore→Backup and download current Configuration page. The current configuration will remain after the firmware update. Moreover, all custom messages and call history will be saved during the upgrade.

To perform the manual firmware update:

1. Go to the Maintenance→Firmware→Manual Firmware Update page.
2. Click the Download Configuration link to back up the current configuration, if needed.
3. Click Choose File button to browse for image.bin file.
4. Click Save to start uploading the file.
5. Click Yes to proceed the firmware upgrade.

Note: The update process takes about 5 minutes. Normal operation will be stopped during that time.

7.3 Limitations and Restrictions

- The Network Capture size is limited to 24 MB. This will put a limitation on the duration of captured file.
- The Call Capture duration is limited to 160 seconds.
- The capture duration is limited to 160 seconds in DSP Capture hidden page.