Release Notes for QXISDN4+ 6.2.18, Edition 1

1 Introduction

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

QXISDN4+ firmware 6.2.18 Date: July 17, 2018

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

Date: July 17, 2018

2 Requirements

2.1 Hardware Requirements

The firmware (FW) can be used on all QXISDN4 models that are converted to QXISDN4+ with a license key.

2.2 Firmware Requirements

Attention: The firmware upgrade to **6.2.18** can **ONLY** be done from 6.0.11 and higher versions.

2.3 Supported IP Phones

Listed below are the Epygi Supported IP phones with the corresponding software (firmware) versions that are tested and recommended for use with QXISDN4+ FW 6.2.18.

Note:

- The **Auto Configuration** and **PnP** services are described in detail in the Configuring Epygi Supported IP Phones with QX IP PBXs guide.
- Any known issues and limitations regarding the usage of the QXISDN4+ FW 6.2.18 telephony services and features for each IP phone described in detail in the QX IP PBX Features on Epygi Supported IP Phones guide.

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Akuvox	R15(P)	15.0.5.235	Yes	Yes	Yes
Akuvox	SP-R53(P)	53.0.6.115	Yes	Yes	Yes
Alcatel	IP2015 (IP15)	1.0.7A-0	No	No	Yes
Alcatel	Temporis IP100	1.0.6A-0	No	No	Yes
Alcatel	Temporis IP150	1.0.6A-0	No	No	Yes
Alcatel	Temporis IP200	13.60.0.89	Yes	Yes	Yes
Alcatel	Temporis IP300	1.0.7B-0	No	No	Yes
Alcatel	Temporis IP600	14.60.0.89	Yes	Yes	Yes
Alcatel	Temporis IP700G	1.0.7A-0	No	No	Yes
Alcatel	Temporis IP800	15.60.0.89	Yes	Yes	Yes
AudioCodes	310HD	1.6.0_build_37	No	No	Yes
AudioCodes	320HD	1.6.0_build_37	No	No	Yes
Cisco	SPA303	7.4.9c	No	Yes	Yes
Cisco	SPA501G	7.4.9c	No	Yes	Yes
Cisco	SPA509G	7.4.9c	No	Yes	Yes
Cisco	SPA525G2	7.4.9c	No	Yes	Yes
Fanvil	C58/C58P	2.3.233.129	No	Yes	Yes
Fanvil	C62/C62P	2.3.235.128	No	Yes	Yes
Fanvil	C400	11.20.12.2.B	No	Yes	Yes
Fanvil	C600	11.20.12.2.B	No	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Fanvil	F52/F52P	2.3.123.78	No	Yes	Yes
Fanvil	H2/H2S	2.0.2.2776	Yes	Yes	Yes
Fanvil	H3	2.0.2.2770	Yes	Yes	Yes
Fanvil	H5	2.0.2.2770	Yes	Yes	Yes
Fanvil	X3/X3P	1.3.511.1821	Yes	Yes	Yes
Fanvil	X3S/X3G	2.0.3.3049	Yes	Yes	Yes
Fanvil	X4/X4G/X4S	2.0.2.2830	Yes	Yes	Yes
Fanvil	X5/X5G	1.3.511.1821	Yes	Yes	Yes
Fanvil	X5S	R0.7.0.1	Yes	Yes	Yes
Fanvil	X6	R0.5.3	Yes	Yes	Yes
Grandstream	GXP1100	1.0.8.6	No	Yes	Yes
Grandstream	GXP1105	1.0.8.6	No	Yes	Yes
Grandstream	GXP1160	1.0.8.6	No	Yes	Yes
Grandstream	GXP1165	1.0.8.6	No	Yes	Yes
Grandstream	GXP1400	1.0.8.6	No	Yes	Yes
Grandstream	GXP1405	1.0.8.6	No	Yes	Yes
Grandstream	GXP1450	1.0.8.6	No	Yes	Yes
Grandstream	GXP1615/1610	1.0.4.55	No	Yes	Yes
Grandstream	GXP1625/1620	1.0.4.55	No	Yes	Yes
Grandstream	GXP1628	1.0.4.55	No	Yes	Yes
Grandstream	GXP1630	1.0.4.55	No	Yes	Yes
Grandstream	GXP1760	1.0.0.48	No	No	Yes
Grandstream	GXP1782/1780	1.0.0.48	No	No	Yes
Grandstream	GXP2100	1.0.8.6	No	Yes	Yes
Grandstream	GXP2110	1.0.8.6	No	Yes	Yes
Grandstream	GXP2120	1.0.8.6	No	Yes	Yes
Grandstream	GXP2124	1.0.8.6	No	Yes	Yes
Grandstream	GXP2130	1.0.7.99	No	Yes	Yes
Grandstream	GXP2135	1.0.7.99	No	Yes	Yes
Grandstream	GXP2140	1.0.7.99	No	Yes	Yes
Grandstream	GXP2160	1.0.7.99	No	Yes	Yes
Grandstream	GXP2170	1.0.7.99	No	Yes	Yes
Grandstream	GXP2200	1.0.3.27	No	Yes	Yes
Grandstream	GXV3140	1.0.7.80	No	Yes	Yes
Grandstream	GXV3175	1.0.3.76	No	Yes	Yes
Grandstream	GXV3240	1.0.3.62	No	Yes	Yes
Grandstream	GXV3275	1.0.3.62	No	Yes	Yes
Htek	UC902	2.0.4.4.33	No	No	Yes
Htek	UC903	2.0.4.4.33	No	No	Yes
Htek	UC912G	2.0.4.4.33	No	No	Yes
Htek	UC912P	2.0.4.4.33	No	No	Yes
Htek	UC923	2.0.4.4.33	No	No	Yes
Htek	UC924	2.0.4.4.33	No	No	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Htek	UC924E	2.0.4.4.33	No	No	Yes
Htek	UC926	2.0.4.4.33	No	No	Yes
Htek	UC926E	2.0.4.4.33	No	No	Yes
Mitel (Aastra)	6730	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6731	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6735	3.3.1.8140-SIP	Yes	Yes	Yes
Mitel (Aastra)	6737	3.3.1.8140-SIP	Yes	Yes	Yes
Mitel (Aastra)	6739	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6753	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6755	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	6757	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel (Aastra)	9480	3.3.1.4305-SIP	Yes	Yes	Yes
Mitel	6863	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6865	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6867	4.2.0.2023-SIP	Yes	Yes	Yes
Mitel	6869	4.2.0.2023-SIP	Yes	Yes	Yes
Panasonic	KX-HDV130	03.004	Yes	Yes	Yes
Panasonic	KX-HDV130NE, KX-HDV130X	06.101	Yes	Yes	Yes
Panasonic	KX-HDV230	03.004	Yes	Yes	Yes
Panasonic	KX-HDV230NE, KX-HDV230X	06.101	Yes	Yes	Yes
Panasonic	KX-TGP550T04	12.17	No	No	Yes
Panasonic	KX-UT123 (NE/RU/X)	01.302	No	No	Yes
Panasonic	KX-UT136 (NE/RU/X)	01.302	No	No	Yes
Polycom	SoundPoint IP 330	3.3.5.0247	No	Yes	Yes
Polycom	SoundPoint IP 331	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 335	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 450	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 550	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 650	4.0.13.1445	No	Yes	Yes
Polycom	SoundPoint IP 670	4.0.13.1445	No	Yes	Yes
Polycom	SoundStation IP 5000	4.0.13.1445	No	Yes	Yes
Polycom	SoundStation IP 6000	4.0.13.1445	No	Yes	Yes
Polycom	VVX 300/310	5.7.0.11768	No	Yes	Yes
Polycom	VVX 301/311	5.7.0.11768	No	No	Yes
Polycom	VVX 400/410	5.7.0.11768	No	No	Yes
Polycom	VVX 401/411	5.7.0.11768	No	No	Yes
Polycom	VVX 500	5.7.0.11768	No	No	Yes
Polycom	VVX 600	5.7.0.11768	No	Yes	Yes
Polycom	VVX 1500	5.7.0.11768	No	Yes	Yes
QOSIP	Q7104/Q7204	1.0.3.98	No	No	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
snom	300	8.4.35	Yes	Yes	Yes
snom	320	8.4.35	Yes	Yes	Yes
snom	360	8.4.35	Yes	Yes	Yes
snom	370	8.7.5.35	Yes	Yes	Yes
snom	720	8.9.3.60	Yes	Yes	Yes
snom	760	8.9.3.60	Yes	Yes	Yes
snom	821	8.7.5.35	Yes	Yes	Yes
snom	870	8.7.5.35	Yes	Yes	Yes
snom	D345	8.9.3.60	Yes	Yes	Yes
snom	D375	8.9.3.60	Yes	Yes	Yes
snom	D710/710	8.9.3.60	Yes	Yes	Yes
snom	D715/715	8.9.3.60	Yes	Yes	Yes
snom	D725	8.9.3.60	Yes	Yes	Yes
snom	D745	8.9.3.60	Yes	Yes	Yes
snom	D765	8.9.3.60	Yes	Yes	Yes
snom	m9	9.4.7	Yes	Yes	Yes
snom	MeetingPoint	8.7.5.35	Yes	Yes	Yes
snom	M700 (M85/M65/M25)	03.24.0007	Yes	Yes	Yes
Spectralink	KIRK Wireless Server 300	PCS14C_	No	No	Yes
Spectralink	KIRK Wireless Server 6000	PCS14C_	No	No	Yes
VTech	ErisStation VCS754	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP600 (VSP601)	1.1.4.1-0	No	No	Yes
VTech	ErisTerminal VSP715	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP725	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP726	2.0.3.2-0	Yes	Yes	Yes
VTech	ErisTerminal VSP735	1.1.4.0-0	No	No	Yes
VTech	ErisTerminal VSP736	2.0.3.2-0	Yes	Yes	Yes
Yealink	CP860	37.81.0.10	Yes	Yes	Yes
Yealink	CP920	78.81.0.15	Yes	Yes	Yes
Yealink	CP960	73.80.0.25	Yes	Yes	Yes
Yealink	SIP-T19P	31.72.0.1	Yes	Yes	Yes
Yealink	SIP-T19P E2	53.81.0.25	Yes	Yes	Yes
Yealink	SIP-T20P	9.72.0.1	Yes	Yes	Yes
Yealink	SIP-T21P	34.72.0.1	Yes	Yes	Yes
Yealink	SIP-T21P E2	52.81.0.25	Yes	Yes	Yes
Yealink	SIP-T22P	7.72.0.1	Yes	Yes	Yes
Yealink	SIP-T23G(P)	44.81.0.25	Yes	Yes	Yes
Yealink	SIP-T26P	6.72.0.1	Yes	Yes	Yes
Yealink	SIP-T27G	69.81.0.25	Yes	Yes	Yes
Yealink	SIP-T27P	45.81.0.25	Yes	Yes	Yes
Yealink	SIP-T28P	2.72.0.1	Yes	Yes	Yes
Yealink	SIP-T29G	46.81.0.25	Yes	Yes	Yes

Vendor	Model	SW/FW Version	PnP		Auto Configuration
			PnP (Multicast)	Assisted PnP (DHCP options 66/67)	
Yealink	SIP-T32G	32.70.0.130	Yes	Yes	Yes
Yealink	SIP-T38G	38.70.0.125	Yes	Yes	Yes
Yealink	SIP-T40G	76.81.0.110	Yes	Yes	Yes
Yealink	SIP-T40P	54.81.0.110	Yes	Yes	Yes
Yealink	SIP-T41P	36.81.0.25	Yes	Yes	Yes
Yealink	SIP-T41S	66.81.0.25	Yes	Yes	Yes
Yealink	SIP-T42G	29.81.0.25	Yes	Yes	Yes
Yealink	SIP-T42S	66.81.0.25	Yes	Yes	Yes
Yealink	SIP-T46G	28.81.0.25	Yes	Yes	Yes
Yealink	SIP-T46S	66.81.0.25	Yes	Yes	Yes
Yealink	SIP-T48G	35.81.0.25	Yes	Yes	Yes
Yealink	SIP-T48S	66.81.0.25	Yes	Yes	Yes
Yealink	SIP VP-T49G	51.80.0.100	Yes	Yes	Yes
Yealink	SIP-T52S	70.81.0.10	Yes	Yes	Yes
Yealink	SIP-T54S	70.81.0.10	Yes	Yes	Yes
Yealink	SIP-T56A	58.80.0.25	Yes	Yes	Yes
Yealink	SIP-T58A/V	58.80.0.25	Yes	Yes	Yes
Yealink	VP-530	23.70.0.40	Yes	Yes	Yes
Yealink	W52P	25.30.0.20	Yes	Yes	Yes

2.4 Interaction with Other Epygi Software Releases

Use the latest SW and FW versions for other Epygi products to achieve maximum compatibility with QXISDN4+ FW 6.2.18:

- QXE1T1, QXFXO4 and QXISDN4 gateways used in the Share mode should have FW 6.2.18 or higher.
- QXFXS24 should have FW 6.2.18 or higher for PnP configuration.
- Auto Dialer SW 1.0.11 or higher should be used.
- Desktop Communication Console (DCC) SW 1.18 or higher should be used.
- iQall (IOS application) version 1.1.0 and iQall (Android application) version 1.0.4 or higher should be used.
- Epygi Hotel Console (EHC) SW 1.0.7 or higher should be used.
- Epygi Media Streamer (EMS) SW 2.4 or higher should be used.
- HotCall Add-In SW 2.5 or higher should be used.
- HotKeyCall SW 1.14 or higher should be used.
- Bulk User Extensions Importer version 1.4 or higher should be used.
- QX-Quadro Configuration Console (QCC) SW 2.3 or higher should be used.
- CallControl Pack SW 5.8.0 or higher should be used.
- To use QXISDN4+ with a 3PCC or Click2Dial application, the Allow 3pcc/Click2Dial Access option should be enabled for each extension using this feature.

3 New Features

The table below indicates a high-level list of new features that have been added beginning with the most recent QXISDN4+ FW release.

Release	New Features
6.2.18	Added support allowing to configure **MTU** size on LAN and VLAN interfaces.
	Added **P-Asserted Identity** (PAI) support for Mitel (Aastra) IP phones. The **PAI** option is configurable from **IP Phone Templates**. It is enabled by default.
6.2.11	Added auto configuration support for the new Htek **UC902**, **UC903**, **UC912G**, **UC912P**, **UC923**, **UC924E** and **UC926E** IP phones.
	Added **Htek UC46 (LCD)** expansion module support for **Htek UC924**, **UC924E**, **UC926** and **UC926E** IP phones.
	In the **Call Alert Settings** added a new **Leave a Voice Message** option, allowing to leave the actual message as a voice mail, available for playback on the defined extension(s).
	Added a new **Advertisement Interval** option in **Redundancy Settings** page which allows to specify the time interval between the advertisement packets that are being sent to the Backup device.
	Added a new **Allow Concurrent Calls to Parent-Child Group** option, allowing to control handling of calls to **Parent-Child** group: • If selected, incoming calls continue ringing on available phones when one of the phones in **Parent-Child** group is busy or rejects the call. • If not selected, incoming calls will follow busy state rules (Busy Call Forwarding, Call Queue, VMS, etc.) depending on what is configured, if any of the phones in the **Parent-Child** group is busy. If all extensions in the **Parent-Child** group are free and are ringing, and any of them presses **Reject** button (or somehow else declines the incoming call), then the **entire group** will be considered as busy. Therefore, incoming call will follow busy state rules depending on what is configured. Note: If the **Call Waiting Service** is enabled on the **Parent** extension, then extensions of **Parent-Child** group will receive the second call.
	Added a new **Reset All** function in the **Multi-functional Programmable Keys** (MPKs) for IP phones, allowing to clean the MPKs configuration quickly for the selected IP line, IP Phone Template and Receptionist. Note: This option is not programmed to remove already configured MPKs from the IP Phone.
	Added new failover reason – **Other**. The system will use next matching routing pattern(s) in case of **Server Failure Responses** (5xx messages) and **Global Failure Responses** (6xx messages).
	Added possibility to access QXISDN4+ WEB GUI using HTTP. Enter the following line http://xxx.xxx.xxx.xxx/unsecure in the address bar of the browser to access WEB GUI, where **xxx.xxx.xxx.xxx** is the IP address or hostname of the QX.
6.2.8	
6.2.6	
6.2.5	Added auto configuration support for the new Polycom **VVX 301/311** and **VVX 401/411** IP phones.
	Added **Phone Book** service support for Polycom phones.
	Added **Watching – Call Interception** support for Fanvil phones.
	Added support for **SNMP v3**.
	Added support for **TLSv1.1** and **TLSv1.2**.

<table>
<tr><th>Release</th><th>New Features</th></tr>
<tr><td rowspan="5"></td><td>Security enhancements: Users will be redirected to HTTPS for the QX Login and Logout pages. This will allow to encrypt traffic between user's device (PC, smartphone, etc.) and the QX.
Note:
• Check and reconfigure Port Forwarding settings on the router, if the QX is located behind router to make sure that there is also Port Forwarding for HTTPS.
• If you have already configured Port Forwardings to access the devices located on the QX LAN side, then check the entered address link to be with HTTP (instead of HTTPS) or reconfigure the Port Forwarding to HTTPS.</td></tr>
<tr><td>Added a new Deactivate button on the IP Lines page allowing to change the status for selected group(s) of IP lines to inactive (free).</td></tr>
<tr><td>Added a new Use Epygi SIP Server button on the Conference Management and ACD Queues pages to allow quick SIP registration of Conference extensions and ACD queues on Epygi SIP Server.</td></tr>
<tr><td>Added a new Billed Extension column in the Call History pages to provide information about the extensions that are charged for the calls.</td></tr>
<tr><td>Added support to provide QX users with e-mail, sms and event notifications in case of calls (emergency calls, etc.) completed through the respective call routing rules.</td></tr>
<tr><td rowspan="7">6.2.1</td><td>Added PnP and auto configuration support for the new Yealink CP920, CP960, SIP-T40G, SIP-T52S, SIP-T54S, SIP-T56A and SIP-T58A/V conference, audio and video phones.</td></tr>
<tr><td>Added PnP and auto configuration support for the new Fanvil H2/H2S, H3, H5, X3S/X3G, X5S and X6 IP phones.</td></tr>
<tr><td>Added PnP and auto configuration support for the new snom D745 and Akuvox R15(P) IP phones.</td></tr>
<tr><td>Added auto configuration support for the new Htek UC924 and UC926 IP phones.</td></tr>
<tr><td>Added PnP and auto configuration support for the new Panasonic KX-HDV130 and KX-HDV230 IP phones.</td></tr>
<tr><td>Added a new Call Completion Fee option in the Calling Cost Control allowing to calculate call cost per the number of completed calls.</td></tr>
<tr><td>Added a checking mechanism for emergency codes:
• Prevents an emergency code from being added if there is already an extension with the same number.
• Prevents an extension from being created if there already exists an emergency code with the same number.</td></tr>
<tr><td rowspan="4">6.1.50</td><td>Added PnP and auto configuration support for the new Grandstream GXP1615, GXP1628, GXP1630, GXP2135 and GXP2170 IP phones.</td></tr>
<tr><td>Added auto configuration support for the new Grandstream GXP1760 and GXP1782/1780, IP phones.</td></tr>
<tr><td>Added PnP support for the Grandstream GXP1610 and GXP1625/1620 IP phones.</td></tr>
<tr><td>Added PnP and auto configuration support for the new Mitel 6869 IP phone.</td></tr>
</table>

Release	New Features
	Added support for the new **Calling Cost Control** licensable feature. This feature allows to limit and control the cost of calls through the routing rules. The following changes are done concerning mainly the **Extensions Settings** and the **Call Routing**. • You can assign a credit amount for each specific extension for making calls through the "**payable**" routing rules. • It allows to configure and use "**payable**" call routing rules to be used only by extensions with a calling credit assigned. • The overall calling costs for "**payable**" routing rules are calculated and reported in the call history.
	Added a new **Click to Dial & Announce** feature allowing the **Dial & Announce** service to be activated on the QX extensions by using the 3PCC **Request URI** method from a WEB browser.
	Added the **SSH FTP** (**SFTP**) support, which allows to send the configuration backup files to an FTP server using the secure FTP connection.
	Added a new "**Archive Now**" option on the **Call History – Archiving Settings** page, allowing to archive immediately the available data.
	Added the new "**Enable VLAN Tagging**" option. This option is used to enable/disable setting the **VLAN ID and priority** for IP phones. Note: The provided IP address will always be from the VLAN network.
	The **Client Code Identification** option can be activated and used by other billing systems as well as it is done for RADIUS server.

4 Changed Features

The table below provides a high-level list of changed features that have been changed beginning with the most recent QXISDN4+ release.

Release	Changed Features
6.2.18	Major **Security** Enhancements
	The default **MTU** size for VLAN interfaces has been decreased from **1500** to **1432** bytes.
	The configured **SRTP policy** of PBX extension will be provided to the Yealink IP phones during configuration.
	The configured **DTMF parameters** of PBX extension will be provided to the IP phones during configuration.
6.2.11	The maximum length of **API ID** field for **Clickatell** SMS Gateway has been increased up to **128** symbols.
	The recommended FW version has been changed for some of supported **Htek** phones. For **UC924** and **UC926** from 2.0.4.2.24 to 2.0.4.4.33.
	GUI Enhancements for the following pages: • Admin Settings of the extensions (user, auto attendant, etc.) • User Settings of the extensions • Call History • Conference History
	Depending on the IP phone model, the **Use Session Timer** option will be enabled for the configured IP line.
	The default **Line Appearance** has been increased from **2** to **5** for each IP line.
6.2.8	
6.2.6	**Network** and **Broadcast** IP addresses will not be included into **Usable Host IP Range.** These IPs will be reserved for network purposes.
6.2.5	The function of **Mixed** mode for **Recording Storage Settings** has been updated to keep the call recordings safe in case of FTP failure. Now this mode allows to send recordings to FTP server immediately together with keeping a copy in the local storage.
	The recommended FW version has been changed for **Yealink CP860** from 37.80.0.30 to 37.81.0.10.
	The recommended FW version has been changed for some of the **Polycom** phones. For **Polycom SoundPoint IP 331, IP 335, IP 450, IP 550, IP 650, IP 670** from 3.3.5.0247 to 4.0.13.1445, for **SoundStation IP 5000** and **IP 6000** from 3.3.5.0247 to 4.0.13.1445, for **VVX 300/310, VVX 400/410, VVX 500** and **VVX 600** from 4.1.7.1210 to 5.7.0.11768, for **VVX 1500** from 3.3.5.0247 to 5.7.0.11768.
	The first programmable key on **Polycom** phones is reserved for the phone account.
	The **Call Quality Warning** in the System Events has been modernized to show information about the callee, caller and call date/time.
	GUI enhancements for **Call History** and **Conference History** pages.
	Redundancy feature has been redesigned to allow sending VRRP packets to unicast IP address (virtual IP) of slave device. Note: Having VRRP packets sent via unicast IP address will allow Redundancy feature to work in scenario when the master and backup devices are located in different places, meaning that there is router between master and backup.
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 QXISDN4+ and QX Gateway(s) to be able successfully connect the devices and share the lines.

Release	Changed Features
	Added option allowing to share and synchronize the configured Incoming Interdigit Service settings with QXISDN4 and QXE1T1 gateways when connected with QXISDN4+ in shared mode.
	The allowed duration of recorded voice mail sent as attachment via e-mail has been increased from 3 to 5 minutes, when G729a codec is used for recording voice mails. Note: If G711u codec is used for recording, the attached voice mail will not be truncated before being sent via e-mail.
	The timezone database has been updated on QX IP PBXs: • 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.
	Enhancements for the Call History – Call Cost page: • Added new filtering options supporting multicriteria searching for payable call records. • Added support to download the displayed CDRs in the (*.log) and (*.csv) formats respectively.
	The recommended FW version has been changed for Yealink SIP-T40P from 54.81.0.25 to 54.81.0.110.
	The recommended FW version has been changed for some of snom phones. For snom 720, 760, D710/710, D715/715, D725, D765 from 8.7.5.35 to 8.9.3.60 and for D345, D375 from 8.9.3.35 to 8.9.3.60.
	Panasonic KX-UT123 and KX-UT123NE IP phones have been merged and renamed to KX-UT123 (NE/RU/X).
	Panasonic KX-UT136 IP phone has been renamed to KX-UT136 (NE/RU/X).
	The recommended FW version has been changed for some Panasonic phones. For KX-UT123 (NE/RU/X) and KX-UT136 (NE/RU/X) from 01.221 to 01.302.
	Akuvox SP-R53P/SP-R53 IP phone has been renamed to Akuvox SP-R53(P).
	The recommended FW version has been changed for Akuvox SP-R53(P) IP phone from 53.0.1.23 to 53.0.6.115.
	The Emergency number configured on the QX IP PBX will be added in the provisioning file for snom IP phones.
	Added nexogy, ClarityTel and Adiptel as the new carriers to the VoIP Carrier Wizard list.
	Added a new option allowing to select Conference extensions from the Unconditional, Busy, No Answer and Unregistered Call Forwarding lists.
	The default TLS port number (5061) will be selected for SIP.
6.1.50	The recommended FW versions have been changed for some Grandstream IP phones. For GXP1610 and GXP1625/1620 from 1.0.2.27 to 1.0.4.55, for GXP2130, GXP2140 and GXP2160 from 1.0.5.23 to 1.0.7.99.
	The recommended FW versions have been changed for Mitel IP phones. For 6863, 6865 and 6867 from 4.0.0.92-SIP to 4.2.0.2023-SIP.
	The maximum number of Watched Extensions for DCC Pro has been increased: for QX20 from 30 to 32, for QX50 and QXISDN4+ from 30 to 50, for QX200 from 100 to 200, for QX500 and QX2000 from 100 to 300.
	The HTML5 Date/Time picker is implemented for Date/Time selection.
	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
	Added option allowing to display Media Streamer's allocated and used memory space on the Status→System Status→Memory page.

Release	Changed Features
	Added new option allowing to select and change Schedule State from WEB GUI.
	The Network Capture page has been moved to Maintenance→Diagnostics→Network Capture page.
	GUI Enhancements for Call Routing Table.
	GUI Enhancements on the Setup→Licensed Features page.
	GUI Enhancements for IP Phone Templates.

5 Fixed Issues

Issues fixed since version 6.2.11:

T: Title
D: Description

20292	T:	P-Asserted-Identity parameter isn't transferred to the IP phones, when the call comes from External Party
	D:	
20290	T:	Shared FXO Lines don't appear on the FXO Line Settings page (Master device) after successful PSTN Line Sharing connection in a specific scenario
	D:	
20262	T:	Conference server is not detecting pressed "in-band" DTMFs if the initial call is offering "out-band"
	D:	
20257	T:	PSTN Line Sharing mechanism stops working in a specific scenario
	D:	
20179	T:	Call Routing rules are not being removed after deleting the PSTN Line Sharing entry
	D:	
19621	T:	Some Configuration and data are left after QX Factory Reset
	D:	The recorded Voice Mails, Call History, Recorded Calls and Custom Voice Messages have been left after QX Factory Reset.

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

20074	T:	Fanvil IP Phones have issue with firmware downgrade in general. Fanvil Phones stop working when downgrading the firmware, even if you downgrade to Epygi recommended version
	D:	
	C:	
	Fix:	Don't downgrade the firmware on Fanvil IP Phones. Will be fixed by in next version.
20036	T:	Sometimes the "Transfer Failed" notification is raised (shown) on Fanvil X6 display, though the transfer is successful
	D:	
	C:	No consequences, as actually the transfer (Blind and Consultative Transfer) is successfully completed.
	Fix:	Will be fixed in the next release.
19446	T:	After changing QXISDN4+ LAN IP configuration, the phones configured from LAN side lose registration
	D:	After changing QXISDN4+ LAN IP configuration (changing the network part of the IP address) the system doesn't reboot phones automatically.
	C:	IP phones lost registration.
	Fix:	Workaround: Reboot phones manually. Will be fixed in future release.
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/pick up the second attempt to park the call, using the park ext. programmable key fails. The problem is happening only if you park the call to the same park extension (by pressing Call Park key).
	C:	
	Fix:	Workaround: Park the call to different call park extension.
18577	T:	The voice traffic is not encrypted when using IPSec connection between two QXISDN4+
	D:	
	C:	
	Fix:	Will be fixed in future release.
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 future release.

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 future release.
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:	Workaround: Use feature code *1 instead of *2 for call relay. Will be fixed in future release.
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 future release.
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 on IP Lines→MPK page.
	Fix:	Workaround: Use the Shared Mailbox: Edit Voice Mailbox Access List link in the Voice Mailbox Settings for extension. Will be fixed in future release.
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, QXISDN4+ 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 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 future release.

7 General Hints

7.1 QXISDN4 Conversion to QXISDN4+ with License Key

Conversion from QXISDN4 to QXISDN4+ (GW to PBX) can be made from 6.0.11 or later firmware. If the QXISDN4 is running on a firmware version lower than 6.0.11 then 6.0.11 needs to be installed first.

1. Go to the *http://xxx.xxx.xxx.xxx/conversionkey.cgi* hidden page, enter the conversion key and apply. The device will continue to function as QXISDN4.
2. Firmware update the device with a FW version dedicated for QXISDN4+. After successful update and reboot the device will function as QXISDN4+.
3. After the update, the default LAN IP will be changed from http://172.28.0.1 to http://172.30.0.1.

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 QXISDN4+ 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, the locally saved voice mails and call recordings, 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, (recommended).
3. Click the 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.
- In case if Voice Mail Recording Codec is other than PCMU, the maximum length of voice message sent by email is limited to 5 minutes.
- The Voice Mailbox size is limited to 300 voice mails for each extension.