



# Release Notes for QX2000 6.2.1, Edition 1

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## 1 Introduction

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

QX50/QX200 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

The firmware (FW) can be used on QX2000 model only.

### 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 Supported IP Phones

Listed below are the Epygi Supported IP phones with the corresponding SW (FW) versions that are tested and recommended for use with QX2000 FW 6.2.1. All the phones in this list can be automatically configured to work with QX2000 FW 6.2.1.

**Note:**

- QX2000 FW 6.2.1 supports also the Plug-and-Play (PnP) option for most IP phones. The configuration options for each specific IP phone are described in detail in the [Configuring Epygi Supported IP Phones](#) guide.
- Any known issues and limitations regarding the usage of the QX2000 FW 6.2.1 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 Support
Akuvox	R15(P)	15.0.5.235	Yes
Akuvox	SP-R53(P)	53.0.6.115	Yes
Alcatel	IP2015 (IP15)	1.0.7A-0	No
Alcatel	Temporis IP100	1.0.6A-0	No
Alcatel	Temporis IP150	1.0.6A-0	No
Alcatel	Temporis IP200	13.60.0.89	Yes
Alcatel	Temporis IP300	1.0.7B-0	No
Alcatel	Temporis IP600	14.60.0.89	Yes
Alcatel	Temporis IP700G	1.0.7A-0	No
Alcatel	Temporis IP800	15.60.0.89	Yes
AudioCodes	310HD	1.6.0_build_37	No
AudioCodes	320HD	1.6.0_build_37	No
Cisco	SPA303	7.4.9c	Yes
Cisco	SPA501G	7.4.9c	Yes
Cisco	SPA509G	7.4.9c	Yes
Cisco	SPA525G2	7.4.9c	Yes
Fanvil	C58/C58P	2.3.233.129	No
Fanvil	C62/C62P	2.3.235.128	No
Fanvil	C400	11.20.12.2.B	No
Fanvil	C600	11.20.12.2.B	No
Fanvil	F52/F52P	2.3.123.78	No
Fanvil	H2/H2S	2.0.2.2776	Yes

Vendor	Model	SW/FW Version	PnP Support
Fanvil	H3	2.0.2.2770	Yes
Fanvil	H5	2.0.2.2770	Yes
Fanvil	X3/X3P	1.3.511.1821	Yes
Fanvil	X3S/X3G	2.0.3.3049	Yes
Fanvil	X4/X4G/X4S	2.0.2.2830	Yes
Fanvil	X5/X5G	1.3.511.1821	Yes
Fanvil	X5S	R0.7.0.1	Yes
Fanvil	X6	R0.5.3	Yes
Grandstream	GXP1100	1.0.8.6	Yes
Grandstream	GXP1105	1.0.8.6	Yes
Grandstream	GXP1160	1.0.8.6	Yes
Grandstream	GXP1165	1.0.8.6	Yes
Grandstream	GXP1400	1.0.8.6	Yes
Grandstream	GXP1405	1.0.8.6	Yes
Grandstream	GXP1450	1.0.8.6	Yes
Grandstream	GXP1615/1610	1.0.4.55	Yes
Grandstream	GXP1625/1620	1.0.4.55	Yes
Grandstream	GXP1628	1.0.4.55	Yes
Grandstream	GXP1630	1.0.4.55	Yes
Grandstream	GXP1760	1.0.0.48	No
Grandstream	GXP1782/1780	1.0.0.48	No
Grandstream	GXP2100	1.0.8.6	Yes
Grandstream	GXP2110	1.0.8.6	Yes
Grandstream	GXP2120	1.0.8.6	Yes
Grandstream	GXP2124	1.0.8.6	Yes
Grandstream	GXP2130	1.0.7.99	Yes
Grandstream	GXP2135	1.0.7.99	Yes
Grandstream	GXP2140	1.0.7.99	Yes
Grandstream	GXP2160	1.0.7.99	Yes
Grandstream	GXP2170	1.0.7.99	Yes
Grandstream	GXP2200	1.0.3.27	Yes
Grandstream	GXV3140	1.0.7.80	Yes
Grandstream	GXV3175	1.0.3.76	Yes
Grandstream	GXV3240	1.0.3.62	Yes
Grandstream	GXV3275	1.0.3.62	Yes
Htek	UC924	2.0.4.2.24	No
Htek	UC926	2.0.4.2.24	No
Mitel (Aastra)	6730	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6731	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6735	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6737	3.3.1.8140-SIP	Yes
Mitel (Aastra)	6739	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6753	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6755	3.3.1.4305-SIP	Yes
Mitel (Aastra)	6757	3.3.1.4305-SIP	Yes

Vendor	Model	SW/FW Version	PnP Support
Mitel (Aastra)	9143	3.3.1.4305-SIP	Yes
Mitel (Aastra)	9480	3.3.1.4305-SIP	Yes
Mitel	6863	4.2.0.2023-SIP	Yes
Mitel	6865	4.2.0.2023-SIP	Yes
Mitel	6867	4.2.0.2023-SIP	Yes
Mitel	6869	4.2.0.2023-SIP	Yes
Panasonic	KX-HDV130	03.004	Yes
Panasonic	KX-HDV130NE, KX-HDV130X	06.101	Yes
Panasonic	KX-HDV230	03.004	Yes
Panasonic	KX-HDV230NE, KX-HDV230X	06.101	Yes
Panasonic	KX-TGP550T04	12.17	No
Panasonic	KX-UT123 (NE/RU/X)	01.302	No
Panasonic	KX-UT136 (NE/RU/X)	01.302	No
Polycom	SoundPoint IP 330	3.3.5.0247	Yes
Polycom	SoundPoint IP 331	3.3.5.0247	Yes
Polycom	SoundPoint IP 335	3.3.5.0247	Yes
Polycom	SoundPoint IP 450	3.3.5.0247	Yes
Polycom	SoundPoint IP 550	3.3.5.0247	Yes
Polycom	SoundPoint IP 650	3.3.5.0247	Yes
Polycom	SoundPoint IP 670	3.3.5.0247	Yes
Polycom	SoundStation IP 5000	3.3.5.0247	Yes
Polycom	SoundStation IP 6000	3.3.5.0247	Yes
Polycom	VX 300/310	4.1.7.1210	Yes
Polycom	VX 400/410	4.1.7.1210	No
Polycom	VX 500	4.1.7.1210	No
Polycom	VX 600	4.1.7.1210	Yes
Polycom	VX 1500	3.3.5.0247	Yes
QOSIP	Q7104/Q7204	1.0.3.98	No
snom	300	8.4.35	Yes
snom	320	8.4.35	Yes
snom	360	8.4.35	Yes
snom	370	8.7.5.35	Yes
snom	720	8.9.3.60	Yes
snom	760	8.9.3.60	Yes
snom	821	8.7.5.35	Yes
snom	870	8.7.5.35	Yes
snom	D345	8.9.3.60	Yes
snom	D375	8.9.3.60	Yes
snom	D710/710	8.9.3.60	Yes
snom	D715/715	8.9.3.60	Yes
snom	D725	8.9.3.60	Yes
snom	D745	8.9.3.60	Yes
snom	D765	8.9.3.60	Yes
snom	m9	9.4.7	Yes
snom	MeetingPoint	8.7.5.35	Yes

Vendor	Model	SW/FW Version	PnP Support
snom	M700 (M85/M65/M25)	03.24.0007	Yes
Spectralink	KIRK Wireless Server 300	PCS14C_	No
Spectralink	KIRK Wireless Server 6000	PCS14C_	No
VTech	ErisStation VCS754	1.1.4.0-0	No
VTech	ErisTerminal VSP600 (VSP601)	1.1.4.1-0	No
VTech	ErisTerminal VSP715	1.1.4.0-0	No
VTech	ErisTerminal VSP725	1.1.4.0-0	No
VTech	ErisTerminal VSP726	2.0.3.2-0	Yes
VTech	ErisTerminal VSP735	1.1.4.0-0	No
VTech	ErisTerminal VSP736	2.0.3.2-0	Yes
Yealink	CP860	37.80.0.30	Yes
Yealink	CP920	78.81.0.15	Yes
Yealink	CP960	73.80.0.25	Yes
Yealink	SIP-T19P	31.72.0.1	Yes
Yealink	SIP-T19P E2	53.81.0.25	Yes
Yealink	SIP-T20P	9.72.0.1	Yes
Yealink	SIP-T21P	34.72.0.1	Yes
Yealink	SIP-T21P E2	52.81.0.25	Yes
Yealink	SIP-T22P	7.72.0.1	Yes
Yealink	SIP-T23G(P)	44.81.0.25	Yes
Yealink	SIP-T26P	6.72.0.1	Yes
Yealink	SIP-T27G	69.81.0.25	Yes
Yealink	SIP-T27P	45.81.0.25	Yes
Yealink	SIP-T28P	2.72.0.1	Yes
Yealink	SIP-T29G	46.81.0.25	Yes
Yealink	SIP-T32G	32.70.0.130	Yes
Yealink	SIP-T38G	38.70.0.125	Yes
Yealink	SIP-T40G	76.81.0.110	Yes
Yealink	SIP-T40P	54.81.0.110	Yes
Yealink	SIP-T41P	36.81.0.25	Yes
Yealink	SIP-T41S	66.81.0.25	Yes
Yealink	SIP-T42G	29.81.0.25	Yes
Yealink	SIP-T42S	66.81.0.25	Yes
Yealink	SIP-T46G	28.81.0.25	Yes
Yealink	SIP-T46S	66.81.0.25	Yes
Yealink	SIP-T48G	35.81.0.25	Yes
Yealink	SIP-T48S	66.81.0.25	Yes
Yealink	SIP VP-T49G	51.80.0.100	Yes
Yealink	SIP-T52S	70.81.0.10	Yes
Yealink	SIP-T54S	70.81.0.10	Yes
Yealink	SIP-T56A	58.80.0.25	Yes
Yealink	SIP-T58A/V	58.80.0.25	Yes
Yealink	VP-530	23.70.0.40	Yes
Yealink	W52P	25.30.0.20	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 QX2000 FW 6.2.1:

- **QXISDN4, QXE1T1 or QXFXO4** gateways used in the shared mode should have FW 6.2.1 or higher.
- **QXFXS24** should have FW 6.2.1 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 1.1 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 QX2000 FW 6.2.1 with a **3PCC** or **Click2Dial** application, the **3pcc/Click2Dial Access Allowed** checkbox should be enabled for each extension(s) 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 QX2000 FW release.

Release	New Features
6.2.1	Added PnP and auto configuration support for the new <b>Yealink CP920, CP960, SIP-T40G, SIP-T52S, SIP-T54S, SIP-T56A and SIP-T58A/V</b> conference, audio and video phones.
	Added PnP and auto configuration support for the new <b>Fanvil H2/H2S, H3, H5, X3S/X3G, X5S and X6</b> IP phones.
	Added PnP and auto configuration support for the new <b>snom D745 and Akuvox R15(P)</b> IP phones.
	Added auto configuration support for the new <b>Htek UC924 and UC926</b> IP phones.
	Added PnP and auto configuration support for the new <b>Panasonic KX-HDV130 and KX-HDV230</b> IP phones.
	Added a new <b>Call Completion Fee</b> option in the Calling Cost Control allowing to calculate call cost per the number of completed calls.
	Added new <b>Auto Reload Queue Statistics</b> option in EAC settings allowing to automatically reload (refresh) Queue Summary pages.
	Added a new option for <b>Agent status by Queue, by Date (Summary)</b> report in EAC which allows to show the status duration in percentage.
6.1.50	Added PnP and auto configuration support for the new <b>Grandstream GXP1615, GXP1628, GXP1630, GXP2135 and GXP2170</b> IP phones.
	Added auto configuration support for the new <b>Grandstream GXP1760 and GXP1782/1780</b> , IP phones.
	Added PnP support for the <b>Grandstream GXP1610 and GXP1625/1620</b> IP phones.
	Added PnP and auto configuration support for the new <b>Mitel 6869</b> IP phone.
	<p>Added support for the new <b>Calling Cost Control</b> 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 <b>Extensions Settings</b> and the <b>Call Routing</b>.</p> <ul style="list-style-type: none"> <li>• You can assign a credit amount for each specific extension for making calls through the "payable" routing rules.</li> <li>• It allows to configure and use "payable" call routing rules to be used only by extensions with a calling credit assigned.</li> <li>• The overall calling costs for "payable" routing rules are calculated and reported in the call history.</li> </ul>
	<p><b>Configuration Management</b> enhancements</p> <ul style="list-style-type: none"> <li>• Added a new option to allow the <b>EAC data</b> to be backed up and saved along with the system configuration and voice data. The <b>EAC data</b> includes the EAC Chat database, Agents' Status and Call Statistics.</li> <li>• Added a new service to restore the system configuration and voice data together with the <b>EAC data</b>. <b>Note:</b> The current <b>EAC data</b> with <b>system configuration</b> will be overwritten after configuration restore.</li> </ul>
	Added a new <b>Click to Dial &amp; Announce</b> feature allowing the <b>Dial &amp; Announce</b> service to be activated on the QX extensions by using the <b>3PCC Request URI</b> method from a WEB browser.
	Added the <b>SSH FTP (SFTP)</b> 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 <b>Call History – Archiving Settings</b> page, allowing to archive immediately the available data.

Release	New Features
	<p>Added new <b>Reporting</b> types in EAC: <b>CDRs by Agent, by Queue, by Date</b> and <b>CDRs by Queue, by Agent, by Date</b>.</p> <p>Added the new "<b>Enable VLAN Tagging</b>" option. This option is used to enable/disable setting the <b>VLAN ID and priority</b> for IP phones. <b>Note:</b> The provided IP address will always be from the VLAN network.</p> <p>The <b>Client Code Identification</b> option can be activated and used by other billing systems as well as it is done for RADIUS server.</p>
6.1.45	<p>Added PnP and auto configuration support for new <b>Yealink SIP-T27G, SIP-T41S, SIP-T42S, SIP-T46S</b> and <b>SIP-T48S</b> IP phones.</p> <p>Added new <b>Media Streamer</b> service allowing to upload and stream uploaded audio files to the RTP destinations from the QX2000.</p> <p>Added "<b>Firmware Version Control</b>" support for Yealink phones to upgrade or downgrade the phones to the Epygi recommended firmware version.</p> <p><b>Call Recording</b> feature enhancements:</p> <ul style="list-style-type: none"> <li>• Improved wildcard character support for the <b>Call Recording</b> rules.</li> <li>• Added support to move the call recording entry up/down by keeping the checkbox selected.</li> </ul> <p>Added support allowing to <b>Restrict Simultaneous Calls</b> for "SIP" call types.</p> <p>GUI improvements and enhancements in the <b>Extensions Management</b> page.</p> <p>GUI enhancements on the <b>Menu</b> bar:</p> <ul style="list-style-type: none"> <li>• Added device's current Date/Time.</li> <li>• Added device's hostname.</li> </ul> <p>General improvements and enhancements in the <b>SIP TLS certificate</b>.</p>
6.1.41	<p>Added the new <b>Priority</b> option in the Redundancy settings, which allows the particular device to always run as a <b>Master</b>.</p>
6.1.40	<p>Added PnP and auto configuration support for the new <b>snom D345</b> IP phone.</p> <p>Added PnP and auto configuration support for the new <b>Vtech ErisTerminal VSP726</b> and <b>ErisTerminal VSP736</b> IP phones.</p> <p>Added auto configuration support for the new <b>VTech</b> phones: <b>ErisTerminal VSP715, ErisTerminal VSP725, ErisTerminal VSP735</b> IP phones, <b>ErisStation VCS754</b> conference phone and <b>ErisTerminal VSP 600 (VSP601)</b> DECT phone.</p> <p>Uploading audio files for customizing any of the system or extension audio messages on the QX2000 has been simplified:</p> <ul style="list-style-type: none"> <li>• Apart from the files in the (<b>*.wav</b>) format, the system can now accept (<b>*.mp3</b>) files for uploading as custom messages.</li> <li>• The (<b>*.wav</b>) and (<b>*.mp3</b>) 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: (<b>CCITT u-law, 8 kHz, 16-bit, Mono</b>).</li> </ul> <p><b>Dial &amp; Announce</b> service enhancements:</p> <ul style="list-style-type: none"> <li>• Added support to allow configuring the repetition count for the <b>Announcement Message</b>.</li> <li>• Added support to allow configuring the silence duration between consecutively played <b>Announcement Messages</b>.</li> </ul> <p>Add the <b>Backup Device GUI Access Port</b> option on <b>Setup→Redundancy</b> page. This option allows to access <b>Backup</b> device WEB GUI by entering the IP address of <b>Master</b> device followed by <b>Backup Device GUI Access Port</b>.</p> <p>GUI improvements and enhancements in the <b>EAC</b> application.</p>



Release	New Features
6.1.35	<p>Added support for the new <b>Day/Night Switching</b> service for the Scheduling feature. This service allows the state of the schedules to be manually controlled by either using the MPK keys on the phones or the new feature code <b>*86</b>, available also remotely on the QX Auto Attendant. Schedules are assigned Schedule IDs. The <b>Schedule IDs are assigned</b> to MPK keys in the <b>IP Lines→Programmable Keys Configuration</b> page.</p> <p><b>Attention:</b> After FW update to 6.1.35 the <b>Schedules</b> comes up with blank <b>Schedule IDs</b>. Thus, the <b>Day/Night Switching</b> service won't work until the schedule(s) are updated with a <b>Schedule ID</b>. Therefore, after FW update select schedules one by one in the <b>Telephony→Schedules</b> page, edit and update with <b>Schedule IDs</b>.</p>
	<p>Added PnP and auto configuration support for the new <b>Yealink SIP-T40P</b> IP phone.</p>
	<p>Added PnP and auto configuration support for the new <b>Fanvil X4/X4G</b> IP phone.</p>
	<p>Added support allowing the QX2000 to act as an <b>OpenVPN</b> client.</p>
	<p>Added <b>OpenVPN</b> support for the following IP phones:</p> <ul style="list-style-type: none"> <li>• snom 370, 720, 760, 821, 870, D375, D710/710, D715/715, D725, D765 IP phones and snom MeetingPoint Conference phone. <b>Note:</b> OpenVPN support will only work for the phone after applying <b>small patch firmware</b> (<a href="http://wiki.snom.com/VPN_Support">http://wiki.snom.com/VPN_Support</a>).</li> <li>• Grandstream GXV3240 and GXV3275 Video phones.</li> <li>• QOSIP Q7104/Q7204 IP phone.</li> </ul>
	<p>Added <b>Use OpenVPN Settings</b> option for <b>IP Lines Settings</b>. Select this option when you want to configure IP Phone using OpenVPN settings. <b>Note:</b> This option is available only for some phones.</p>
	<p>Added <b>Symmetric RTP</b> support for IP Lines. Select this option when the IP phone attached to the IP line is behind the NAT router.</p>
	<p><b>ACD &amp; EAC</b> enhancements:</p> <ul style="list-style-type: none"> <li>• Added support to display information about the agents logged into EAC on <b>Extensions→ACD→EAC Status</b> page.</li> <li>• Added support to terminate the EAC session for the logged in agent(s) from QX GUI.</li> <li>• Added support to display "<b>Wait Time in Queue</b>" for each call in EAC.</li> <li>• Added support to select "<b>Duration Format</b>" from <b>EAC→Settings</b> page for EAC reports.</li> </ul>
	<p><b>Dial &amp; Announce</b> service enhancements:</p> <ul style="list-style-type: none"> <li>• Added support to allow the <b>Dial &amp; Announce</b> service to be automatically activated on a schedule.</li> <li>• Added a new option to interrupt the active call and play an announcement to the users in the dial &amp; announce list. This option is available for <b>QX extensions</b> only.</li> <li>• Added <b>Auto Answer</b> option to page the extension's IP phone by forcing the phone to go off-hook and play the announcement. This option is available for <b>QX extensions</b> only.</li> </ul>
	<p>GUI enhancements for <b>Voice Mail Settings</b> and <b>Profiles for Voice Mail Settings</b> pages.</p>

## 4 Changed Features

The table below provides a high-level list of changed features that have been changed beginning with the most recent QX2000 FW release.

Release	Changed Features
6.2.1	The <b>PSTN Gateways Line Sharing</b> mechanism has been changed and updated, bringing more stability, improving the connection between PBXs and Gateways. <b>Important Note:</b> Please update the firmware version to <b>6.2.1</b> both on QX2000 and QX Gateway(s) to be able successfully connect the devices and share the lines.
	Added option allowing to share and synchronize the configured Incoming Interdigit Service settings with QXISDN4 and QXE1T1 gateways when connected with QX2000 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. <b>Note:</b> 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: <ul style="list-style-type: none"> <li>• The current local time has been corrected for Israel, Venezuela, Shri Lanka, Apia, Samoa and Fiji.</li> <li>• Added new timezone Nukualofa, Tonga (GMT+14).</li> </ul>
	New <b>Date/Time</b> pickers have been implemented for all applicable GUI pages, allowing to select or define the date/time options easier and conveniently.
	Enhancements for the <b>Call History – Call Cost</b> page: <ul style="list-style-type: none"> <li>• Added new filtering options supporting multicriteria searching for payable call records.</li> <li>• Added support to download the displayed CDRs in the (*.log) and (*.csv) formats respectively.</li> </ul>
	The recommended FW version has been changed for <b>Yealink SIP-T40P</b> from 54.81.0.25 to 54.81.0.110.
	The recommended FW version has been changed for some of <b>snom</b> phones. For <b>snom 720, 760, D710/710, D715/715, D725, D765</b> from 8.7.5.35 to 8.9.3.60 and for <b>D345, D375</b> from 8.9.3.35 to 8.9.3.60.
	<b>Panasonic KX-UT123</b> and <b>KX-UT123NE</b> IP phones have been merged and renamed to <b>KX-UT123 (NE/RU/X)</b> .
	<b>Panasonic KX-UT136</b> IP phone has been renamed to <b>KX-UT136 (NE/RU/X)</b> .
	The recommended FW version has been changed for some <b>Panasonic</b> phones. For <b>KX-UT123 (NE/RU/X)</b> and <b>KX-UT136 (NE/RU/X)</b> from 01.221 to 01.302.
	<b>Akuvox SP-R53P/SP-R53</b> IP phone has been renamed to <b>Akuvox SP-R53(P)</b> .
	The recommended FW version has been changed for <b>Akuvox SP-R53(P)</b> IP phone from 53.0.1.23 to 53.0.6.115.
	Added <b>nexogy, ClarityTel</b> and <b>Adiptel</b> as the new carriers to the VoIP Carrier Wizard list.
Added a new option allowing to select Conference extensions from the <b>Unconditional, Busy, No Answer</b> and <b>Unregistered</b> 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 <b>Grandstream</b> IP phones. For <b>GXP1610</b> and <b>GXP1625/1620</b> from 1.0.2.27 to 1.0.4.55, for <b>GXP2130, GXP2140</b> and <b>GXP2160</b> from 1.0.5.23 to 1.0.7.99.
	The recommended FW versions have been changed for <b>Mitel</b> IP phones. For <b>6863, 6865</b> and <b>6867</b> from 4.0.0.92-SIP to 4.2.0.2023-SIP.

Release	Changed Features
	<p>The maximum number of Watched Extensions for <b>DCC Pro</b> has been increased: for <b>QX20</b> from 30 to 32, for <b>QX50</b> and <b>QXISDN4+</b> from 30 to 50, for <b>QX200</b> from 100 to 200, for <b>QX500</b> and <b>QX2000</b> from 100 to 300.</p> <p>The HTML5 <b>Date/Time picker</b> is implemented for Date/Time selection.</p> <p>The backup configuration filename format has been updated and will include the installed firmware version of the QX: <b>config_[Hostname]_[Firmware Version]_[Date/Time].bin</b></p> <p>Added option allowing to display Media Streamer's allocated and used memory space on the <b>Status→System Status→Memory</b> page.</p> <p>Added new option allowing to select and change <b>Schedule State</b> from WEB GUI.</p> <p>The <b>Network Capture</b> page has been moved to <b>Maintenance→Diagnostics→Network Capture</b> page.</p> <p>GUI Enhancements for <b>Call Routing Table</b>.</p> <p>GUI Enhancements on the <b>Setup→Licensed Features</b> page.</p> <p>GUI Enhancements for <b>IP Phone Templates</b>.</p>
6.1.45	<p>The recommended FW versions have been changed for some <b>Yealink</b> phones. For <b>CP860</b> from 37.80.0.10 to 37.80.0.30, for <b>T19 E2</b> from 53.80.0.130 to 53.81.0.25, for <b>T21 E2</b> from 52.80.0.130 to 52.81.0.25, for <b>T23G</b> from 44.80.0.130 to 44.81.0.25, for <b>T27P</b> from 45.80.0.130 to 45.81.0.25, for <b>T29G</b> from 46.80.0.130 to 46.81.0.25, for <b>T40P</b> from 54.80.0.130 to 54.81.0.25, for <b>T41P</b> from 36.80.0.130 to 36.81.0.25, for <b>T42G</b> from 29.80.0.130 to 29.81.0.25, for <b>T46G</b> from 28.80.0.130 to 28.81.0.25 and for <b>T48G</b> from 35.80.0.130 to 35.81.0.25.</p>
6.1.41	<p>Added the "<b>Electronic Hookswitch</b>" option for Polycom IP phones into the <b>IP Phone Templates</b>. When you use a headset that supports electronic hookswitch (EHS), you can place, answer, and end calls by using controls on your headset.</p>
6.1.40	<p>Changes in behavior for the <b>Parent-Child</b> extensions configuration:</p> <ul style="list-style-type: none"> <li>• If any of the extensions in the <b>Parent/Child</b> group are busy, then entire group will be considered busy, therefore the incoming call will follow the busy state rules (busy forwarding, call queue, VMS, etc.) depending on what is configured. <b>Note:</b> If the "<b>Call Waiting Service</b>" is enabled on the <b>Parent</b> extension, then extensions of <b>Parent/Child</b> group will receive the second call.</li> <li>• If all extensions in the <b>Parent/Child</b> group are free (not busy) and are ringing, and any of them presses the "<b>Reject</b>" button (or somehow declines the incoming call), then the entire group will be considered as busy. Therefore, incoming calls will follow busy state rules (busy forwarding, queue, VMS, etc.) depending on what is configured.</li> </ul> <p>Added the <b>Transfer on Conference Hang Up</b> option for Grandstream IP phones into the <b>IP Phone Templates</b>. This allows the conference initiator to be disconnected from the conference while keeping the other two parties in the call.</p> <p>Added <b>MC-Link</b> and <b>Flowroute</b> as the new carriers to the VoIP Carrier Wizard list.</p>
6.1.35	<p>The recommended FW versions have been changed for some <b>Yealink</b> phones. For <b>C860</b> from 37.72.0.10 to 37.80.0.10, for <b>T41P</b> from 36.72.0.1 to 36.80.0.130, for <b>T42G</b> from 29.72.0.1 to 29.80.0.130, for <b>T46G</b> from 28.72.0.1 to 28.80.0.130 and for <b>T48G</b> from 35.72.0.34 to 35.80.0.130.</p> <p>The max number of <b>Line appearance</b> has been increased for some <b>Yealink</b> phones. For <b>T29G</b> from 10 to 16, for <b>T41P</b> from 6 to 10, for <b>T42G</b> from 6 to 12, for <b>T46G</b>, <b>T48G</b> and <b>T49G</b> from 10 to 16.</p> <p>The recommended FW versions have been changed for some <b>snom</b> phones. For <b>MeetingPoint</b>, <b>370</b>, <b>821</b> and <b>870</b> from 8.4.35 to 8.7.5.35.</p> <p><b>QOSIP Q7104</b> IP phone has been renamed to <b>QOSIP Q7104/Q7204</b>.</p>

Release	Changed Features
	The recommended FW version has been changed for <b>QOSIP Q7104/Q7204</b> from 1.0.3.97 to 1.0.3.98.
	The default <b>Digitmap</b> for Polycom phones has been changed. <ul style="list-style-type: none"> <li>• Added support allowing user to dial out automatically after <b>Digitmap Timeout</b>, without pressing <b>#</b> sign or <b>Send</b> softkey.</li> <li>• Added support allowing to accelerate dial out (don't wait for <b>Digitmap Timeout</b>) by pressing <b>#</b> sign.</li> </ul>
	Programmable key support (DSS Keys) has been increased up to <b>37</b> keys for <b>Fanvil X5/X5G</b> .
	<b>Hot Desking</b> feature enhanced regarding the voice notifications when login/logout on the public phones: <ul style="list-style-type: none"> <li>• Added voice prompt asking user to login before using the phone.</li> <li>• Added voice prompt notifying user about login extensions in use.</li> <li>• Added voice prompt informing the user about the successful logging out.</li> </ul>
	Added support to allow/deny access to the <b>Diagnostics</b> and <b>Reboot</b> pages for QX localadmin.
	The <b>Status→System Status→Memory</b> page is redesigned and modernized.
	The <b>Subscriptions Count</b> section ( <b>Status→System Status→IP Line Registration</b> page) is enhanced to display correct and informative data for used/allowed subscriptions.

## 5 Fixed Issues

Issues fixed since version 6.1.50:

T: Title  
D: Description

20024	T:	Unable to select the "Expires on" option for calling cost expiration/renewal date
	D:	
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:	
20011	T:	Input checking mechanism doesn't work for Remote Extension Settings – Line Appearance section
	D:	
20008	T:	The successful calls of the selected Conference will not be shown when you click the Conference hyperlink from Conference History - Conferences page
	D:	
19993	T:	Unable to disconnect EAC session from Extensions→ACD→EAC Status page when the browser where it was logged in is unreachable
	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 is no user type extension.
19980	T:	Unable to download and upload the "Custom Messages of the extension" from the "Account Settings" page
	D:	
19978	T:	The "Callee" field is missing in the filter section of the Call and Conference History pages
	D:	
19974	T:	The (R) is not shown next to the IP Line on the Extensions Management page when Remote Extension is activated on the Extension
	D:	
19947	T:	Unable to park calls using "Call Park" MPK on Yealink SIP-T40P IP phone
	D:	The issue is solved in the latest recommended firmware version (54.81.0.110) for the Yealink SIP-T40P.
19805	T:	The BLF indication (for programmable keys) on snom phones (7xx and D7xx series) is switched off after the subscription timeout expires, regardless of the actual state of the BLF event
	D:	The issue is solved in the latest recommended firmware version (8.9.3.60) for the snom 7xx and D7xx series.

## 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.
20035	T:	Call disconnect in some calling scenarios with snom phones
	D:	In some calling scenarios, like picking up the parked call or using the local authentication in routing, the call might be disconnected if the call originator is a snom phone.
	C:	
	Fix:	Will be fixed in the next release.
19894	T:	Automatic "Daylight Saving Time" doesn't work on Fanvil IP phones
	D:	
	C:	
	Fix:	Workaround: Create an IP Phone Template for Fanvil phones, select the "manual" option for "Daylight Saving Time". Attach this template to the IP lines for Fanvil phones. Will be fixed in future release by Fanvil.
19805	T:	The BLF indication (for programmable keys) on snom phones is switched off after the subscription timeout expires, regardless of the actual state of the BLF event
	D:	The issue appears on snom 3xx and 8xx series running 8.7.5.35 firmware version.
	C:	
	Fix:	Workaround: The issue is solved in snom 8.7.5.44 beta firmware.
19537	T:	ACD call recordings cannot be played from EAC when using the Mozilla Firefox browser
	D:	The Mozilla Firefox browser doesn't have native support for (*.wav) audio format.
	C:	When you click <b>Play</b> instead of playing the recording, it will be downloaded.
	Fix:	Workaround: Install corresponding add-ons or use other browsers (Chrome, Opera, Microsoft Edge).
19463	T:	3-way conference doesn't work on Grandstream GXP1100 and GXP1105 IP phones in a specific scenario
	D:	3-way call conference cannot be established on Grandstream GXP1100 or GXP1105 phones when they receive a call.
	C:	
	Fix:	Workaround: Login into WEB GUI of the phone and assign 3-way conference key as a MPK. Use this key to initiate 3-way call conference when the phone is already in the active call. Will be fixed in future release.

19446	T:	<b>After changing QX2000 LAN IP configuration, the phones configured from LAN side lose registration</b>
	D:	After changing QX2000 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:	<b>It's not possible to park a call twice to the same call park extension by using programmable key on Yealink T32G and T38G</b>
	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.
18549	T:	<b>Could not dial out (*1) or use any other moderator feature while welcome message file has been playing</b>
	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:	<b>Part of conference recording is lost after recording pause/resume</b>
	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:	<b>Calls which are done using Call Relay (*2) on the auto attendant are not shown in Call History</b>
	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:	<b>Find Me / Follow Me does not work for incoming Secure RTP call</b>
	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:	<b>Shared Mailbox watching does not work when using "Allow access to Shared Mailbox for enabled extensions" option in Many Extension Ringing configuration</b>
	D:	Extension has Many Extension Ringing enabled with a few extensions configured for Shared Mailbox.
	C:	However, in the IP Line settings, the <b>Shared VMail Ext. xxx</b> option is not listed in the drop-down list on <b>IP Lines→MPK</b> page.
	Fix:	Workaround: Use the <b>Shared Mailbox: Edit Voice Mailbox Access List</b> link in the Voice Mailbox Settings for extension. Will be fixed in future release.
16533	T:	<b>A problem with incoming Secure RTP call in a specific scenario</b>
	D:	When incoming Secure RTP call is connecting to the destination via Call Routing table, QX2000 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 the next releases.

## 7 General Hints

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### 7.1 Firmware Installation and Update

The steps below describe shortly the QX2000 manual installation procedure used to install the firmware from scratch. This would be used to install version 6.0.1 or for **Emergency Recovery** of a system. This procedure will result in a system that is at factory default. Please refer to [QX1000/2000 System Recovery Procedure](#) document for more details.

1. Turn on the PC.
2. Insert CD/DVD disk including installation program to the DVD ROM.
3. Restart (reset) the PC.
4. Installation will start automatically after PC reboot. After the successful installation, the PC will automatically shut down (this may take from 10-15 minutes). The beep sound will indicate that the installation successfully completed.
5. Turn on the PC and quickly remove the installation CD/DVD disk from the DVD ROM.

**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, voice mails, 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.

### 7.2 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 VM sent by email is limited to 5 minutes.
- The number of **VMs in the mailbox** for an extension is limited to 500.
- **Use Session Timer** in IP Line Settings is deselected by default.