

445HD, 450HD, C450HD RX50 Conference Phone

Zoom SIP IP Phones

Version 3.4.10.94



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Notice

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This document is subject to change without notice.

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WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and services are provided by AudioCodes or by an authorized AudioCodes Service Partner. For more information on how to buy technical support for AudioCodes products and for contact information, please visit our website at <https://www.audiocodes.com/services-support/maintenance-and-support>.

Stay in the Loop with AudioCodes



Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Related Documentation

Document Name
445HD IP Phone Quick Guide - Generic SIP
445HD IP Phone User's Manual - Generic SIP
450HD IP Phone Quick Guide - Generic SIP
450HD IP Phone User's Manual - Generic SIP
C450HD IP Phone Quick Guide - Generic SIP
C450HD IP Phone User's Manual - Generic SIP
RX50 Conference Phone Quick Guide - Generic SIP
RX50 Conference Phone User's Manual - Generic SIP
Device Manager Administrator's Manual
One Voice Operations Center (OVOC) IOM Manual
OVOC User's Manual

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our website at <https://online.audiocodes.com/documentation-feedback>.

1 Introduction

This document describes the new features, resolved constraints and known constraints of AudioCodes' Generic SIP IP phones for Version 3.4.8.



These *Release Notes* document only the *high-end* phone models (445HD, 450HD, C450HD, and the RX50) supported in version 3.4.8.

AudioCodes' 400HD Series of High-Definition IP Phones offers a new dimension of voice call quality and clarity for the IP Telephony market. This new series of IP Phones further expands AudioCodes' VoIP product offering for the service providers' hosted services, Enterprise IP telephony and Enterprise contact centers markets. As a natural addition to the AudioCodes Mobile Clients, Media Gateway, Media Server and Multi-Service Business Gateway products, the AudioCodes Series of High-Definition IP Phones enable System Integrators and end-customers to build end-to-end solutions that rely on AudioCodes' technological advantage and proven track record in providing state-of-the-art products.

The AudioCodes Series of High-Definition IP Phones meet a growing demand for High-Definition VoIP solutions in end-user phones and terminals, improving the productivity and efficiency of business communications with new quality standards set by the High-Definition voice technology.

1.1 Specifications

The phones' specifications are summarized in the following table:

Table 1-1: Specifications

Feature	Details
VoIP Signaling Protocols	<ul style="list-style-type: none"> ■ SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	<ul style="list-style-type: none"> ■ IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling ■ SIP over TLS (SIPS) ■ 802.1x ■ 802.1p/Q for Traffic Priority and QoS ■ VLAN Discovery Mechanism (CDP, LLDP) ■ ToS (Type of Service) field, indicating desired QoS DHCP Client ■ NTP Client ■ OpenSSL 1.0.1m, supporting SHA2 and SHA256 algorithms. ■ SHA2 - Open SSL 1.0.1 integrated with TLS 1.2, supporting SHA256, replaced the previous OpenSSL 0.9.8 and TLS 1.0 stacks ■ Wi-Fi (445HD and C450HD - BW CPN flavor): Single band 2.4GHz, Dual Band 5GHz (C450HD only), 802.11b/g/n ■ Bluetooth: BT2.1+EDR/BT3.0 and BT4.2 ■ Wideband audio support for Bluetooth headsets (Beta)
Media Processing	<ul style="list-style-type: none"> ■ Voice Coders: G.711, G.729, G.722 8000 / G.722 16000, OPUS (450HD and RX50 only). ■ Acoustic Echo Cancelation ■ Adaptive Jitter Buffer 300 msec ■ Voice Activity Detection ■ Comfort Noise Generation (RX50 is supported for OPUS and G.729 but not for G.711 and G.722) ■ Packet Lost Concealment ■ RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC 3711) ■ DTMF Relay (RFC 2833)

Feature	Details
Telephony Features	<ul style="list-style-type: none"> ■ Multi Line ■ Speed Dial and (up to 18) BLF presence buttons ■ BLF Call Pickup ■ Handles up to 8 concurrent calls ■ Multiple lines (all except RX50) ■ Call Hold / Un-Hold ■ Call Transfer: the hard TRANSFER key's default functionality (Blind Transfer) can be changed to Consultative Transfer ■ Three-way Conferencing (with local mixing) ■ Remote Conference compliancy with RFC 4579, SIP Call Control, Conferencing for UAs ■ Merge option: Two separate calls can be merged into one conference ■ Call Park (Genband only; only on 450HD / C450HD phones with Expansion Module) (all except RX50) ■ Call Forwarding (all except RX50) ■ DnD (Do Not Disturb) (all except RX50) ■ Voicemail (including capability to secure user access with PIN code) (all except RX50) ■ Message Waiting Indication (including MWI LED) (all except RX50) ■ Caller ID Notification ■ Paging w/without Barge-in. Configurability of special keys as paging group dials. ■ Call Waiting Indication ■ Personal Directory ■ Automatic On-hook Dialing (all except RX50) ■ Automatic Answering (Alert-Info header and "talk" event) ■ CWRR (Call Waiting Reminder Ring) ■ Call Logs: Missed/Received Calls and Dialed Numbers ■ Voca service to allow voice dialing ■ Redial (all except RX50) ■ Dial Plan ■ Shared Line Appearance (SLA) (Genband environment) (all except RX50)
Configuration / Management	<ul style="list-style-type: none"> ■ LCD Display User Interface Language Support (Various Languages) ■ Web-based Management (HTTP/HTTPS) ■ European date format (DDMMYYYY) and American date format (MMDDYYYY), in phone LCD and Web interface ■ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and proprietary configuration file upgrade ■ DHCP options (66, 67, and 160) for auto-provisioning ■ DHCP options (120, 60, and 77) for device information ■ DHCP option (42 or 4) for the NTP server ■ DHCP option (43) for vendor specific information ■ DHCP option (2) for the Time Zone Offset ■ Redirect server ■ LDAP (Lightweight Directory Access Protocol) ■ Private Labeling Mechanism ■ Configuration file encryption (Entire file and individual parameters) ■ One Voice Operations Center (OVOC) module IP Phone Manager Pro Express ■ Access via Telnet or SSH
Supported Environments	<ul style="list-style-type: none"> ■ Genband softswitch solution's Kandy Business Solutions (KBS) application server. ■ Zoom ■ Asterisk ■ Freeswitch
Diagnostics and Troubleshooting Tools	<ul style="list-style-type: none"> ■ System Logging (Syslog) ■ Monitoring (Traceroute) ■ DSP Packet Recording ■ Port Mirroring ■ VoIP Status Web page ■ Firmware Recovery ■ TCP Dump ■ Core Dump FileRTCP-XR Quality of Experience Reports

Feature	Details
<p>Supported Languages</p> <p>[Note that on the RX50 the virtual keypad is fully supported for all languages besides Chinese and Japanese. For non-supported languages, the virtual keypad is in English.]</p>	<ul style="list-style-type: none"> ■ English ■ Spanish ■ Russian ■ German ■ Ukrainian ■ French ■ Italian ■ Hebrew ■ Polish ■ Portuguese (displayed only if included in your Feature Key) ■ Korean ■ Finnish ■ Simplified Chinese ■ Traditional Chinese ■ Hungarian ■ Japanese ■ Slovak ■ Czech ■ Latvian (contact person information) ■ Dutch ■ Magyar
<p>Supported Headsets</p>	<ul style="list-style-type: none"> ■ For a comprehensive list of supported Jabra headsets, see the Jabra Headset Compatibility Guide ■ For a comprehensive list of supported Plantronics headsets see https://compatibility.plantronics.com/deskphone ■ Also, the following which are not documented online yet: <ul style="list-style-type: none"> • Jabra UC-150 • Jabra Speak 510+ • Jabra Speak 410 • Jabra MOTION OFFICE • Jabra PRO 9470 • Jabra Evolve Series 20, 30, 40, 75, 80 • Microsoft LX-3000 • Plantronics C-310M • Plantronics C-320M • Plantronics HW720 • Plantronics Blackwire Series 300, 325, 510, 520, 710 • Jabra UC-550 • Jabra Pro 920 EHS wireless headset • Jabra Pro 9450 EHS wireless headset
<p>RX50 Hardware</p>	<ul style="list-style-type: none"> ■ 4.3-inch touch screen (480x272) with an intuitive user interface. ■ Connectors interfaces: <ul style="list-style-type: none"> • 1 x RJ-45 ports (10/100 BaseT Ethernet) for WAN • 2 x HDMI ports for Satellite Microphones ■ Power: <ul style="list-style-type: none"> • PoE Class 3: IEEE802.3af ■ Keys: <ul style="list-style-type: none"> • MUTE (including LED) • VOLUME control keys ■ Speaker ■ Optional satellite microphone kit Room Experience Suite pickup reach by 2 meters / 6.5 feet

Feature	Details
445HD Hardware	<ul style="list-style-type: none"> ■ Color Screen 4.3": Graphic, 480x272 resolution ■ Integrated sidcar 376x60 resolution featuring 12 programmable speed dial keys with presence monitoring (BLF) ■ Connectors interfaces: <ul style="list-style-type: none"> • 2 x RJ-45 ports (10/100/1000BaseT Ethernet) for WAN and LAN (GbE support) • RJ-9 port (jack) for headset • RJ-9 port (jack) for handset • USB interface for USB headset support • RJ-11 interface for DHSG ■ Mounting: <ul style="list-style-type: none"> • Wall and desktop mounting options • One angle for desktop mount, another angle for wall mount ■ Power: <ul style="list-style-type: none"> • DC jack adapter 12V • Power supply AC 100 ~ 240V • PoE Class 2: IEEE802.3af (optional) ■ Keys: <ul style="list-style-type: none"> • 4 softkeys and 6 multifunction hard keys • VOICE MAIL message hotkey (including LED) • 4-way navigation button with OK key • MENU • REDIAL • HOLD • MUTE (including LED) • TRANSFER • VOLUME control key • HEADSET (including LED) • SPEAKER (including LED)
450HD Hardware	<ul style="list-style-type: none"> ■ Large (800 x 480), graphical, high-resolution, 5-inch color touch (TFT) screen with an intuitive touch-oriented user interface design. ■ Connectors interfaces: <ul style="list-style-type: none"> • 2 x RJ-45 ports (10/100/1000BaseT Ethernet) for WAN and LAN • RJ-9 port (jack) for headset • RJ-9 port (jack) for handset • USB interface for USB headset support • RJ-11 interface for DHSG ■ Mounting: <ul style="list-style-type: none"> • Wall and desktop mounting options • One angle for desktop mount, another angle for wall mount ■ Power: <ul style="list-style-type: none"> • DC jack adapter 12V • Power supply AC 100 ~ 240V • PoE Class 3: IEEE802.3af (optional) ■ Keys: <ul style="list-style-type: none"> • 8 x softkeys • VOICE MAIL message hotkey (including LED) • 4-way navigation button with OK key • MENU • REDIAL • HOLD • MUTE (including LED) • TRANSFER • VOLUME control key • HEADSET (including LED) ■ SPEAKER (including LED)

Feature	Details
C450HD Hardware	<ul style="list-style-type: none"> ■ Five-inch color capacitive 1280 x 720 high-resolution touch (TFT) screen ■ Connectors interfaces: <ul style="list-style-type: none"> • 2 x RJ-45 ports (10/100/1000BaseT Ethernet) for WAN and LAN • RJ-9 port (jack) for headset • RJ-9 port (jack) for handset • 2 x USB ports for headset support • - USB white port - host port • - USB black port - device port (default) but it can be configured to host port • RJ-11 interface for DHSG ■ Mounting: <ul style="list-style-type: none"> • Wall and desktop mounting options • One angle for desktop mount, another angle for wall mount ■ Power: <ul style="list-style-type: none"> • DC jack adapter 12V • Power supply AC 100 ~ 240V • PoE Class 3: IEEE802.3af (optional) ■ Keys: <ul style="list-style-type: none"> • 8 x softkeys • VOICE MAIL message hotkey (including LED) • 4-way navigation button with OK key • MENU • REDIAL • HOLD • MUTE (including LED) • TRANSFER • VOLUME control key • HEADSET (including LED) ■ SPEAKER (including LED)

1.2 IP Phone Models

The table below lists the AudioCodes 400HD Series IP Phone models.

Table 1-2: 400HD Series IP Phone Models

Part Number	Product Description
405	405 IP Phone (black) with Power over Ethernet (PoE)
405HDEG	405HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
405HDEPSG	405HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
420HDE	420HD IP Phone PoE Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEG	420HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEW	420HD IP Phone PoE White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPS	420HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPSG	420HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPSW	420HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
430HDE	430HD IP Phone PoE Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEG	430HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEW	430HD IP Phone PoE White 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEPS	430HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEPSW	430HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)
430HDEPSG	430HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)

440HDEG	440HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEWG	440HD IP Phone PoE GbE White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEPSG	440HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEPSWG	440HD IP Phone PoE GbE and external power supply White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
IPRX50	RX50 Conference Phone
IP445HDEG	445HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 480x272 Graphic Color LCD 4.3", 376x60 BLF LCD, and Power over Ethernet (PoE)
IP445HDG-R	445HD-R IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 6 Programmable keys, 480x272 Graphic Color LCD, and Power over Ethernet (PoE)
IP445HDEPSG	445HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 480x272 Graphic Color LCD, 376x60 BLF LCD, and Power over Ethernet (PoE)
IP445HDEPSG-R	445HD-R IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 6 Programmable keys, 480x272 Graphic Color LCD, and Power over Ethernet (PoE)
IP445HDEG-BW	445HD IP Phone PoE GbE black with integrated Bluetooth and Wi-Fi 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 480x272 color LCD 4.3", 376x60 BLF LCD and Power over Ethernet (PoE)
IP450HDEG	450HD IP Phone PoE GbE black 2 Ethernet 10/100/1000 ports, 800x480 5" Color Touch LCD and Power over Ethernet (PoE)
IP450HDEPSG	450HD IP Phone PoE GbE with an external power supply black 2 Ethernet 10/100/1000 ports, 800x480 5" Color Touch LCD and Power over Ethernet (PoE)
IPC450HDEG-BW	C450HD IP-Phone PoE GbE black with integrated BT and WiFi 2 Ethernet 10/100/1000 ports, 1280x720 5" Color Touch LCD and Power over Ethernet (PoE)
IPC450HDEG-DBW	C450HD IP-Phone PoE GbE black with integrated BT and Dual Band WiFi 2 Ethernet 10/100/1000 ports, 1280x720 5" Color Touch LCD and Power over Ethernet (PoE)
IPC450HDEPSG- BW	C450HD IP-Phone PoE GbE with integrated BT, WiFi and an external power supply black 2 Ethernet 10/100/1000 ports, 1280x720 5" Color Touch LCD and Power over Ethernet (PoE)
IPC450HDEPSG- DBW	C450HD IP-Phone PoE GbE with integrated BT, Dual Band WiFi and an external power supply black 2 Ethernet 10/100/1000 ports, 1280x720 5" Color Touch LCD and Power over Ethernet (PoE)

* Supported in the USA, Canada, the EU, Switzerland South Africa and Israel, Wi-Fi and Bluetooth require a specific CPN with a 'BW' suffix.

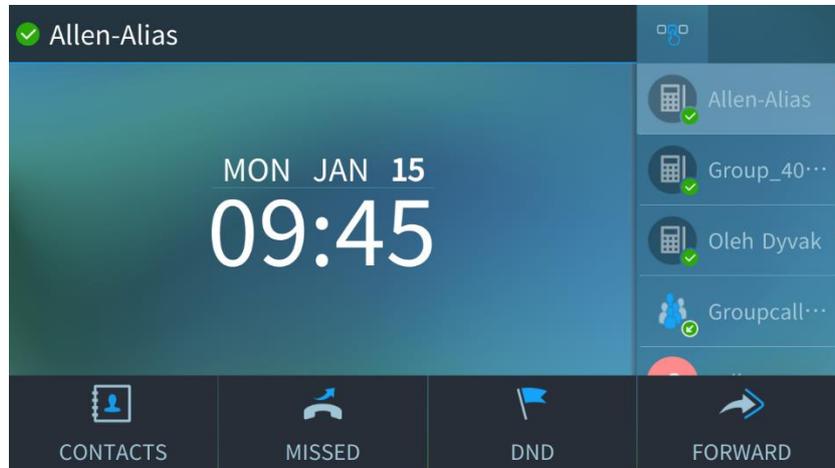


There is no CPN for the Generic SIP C450HD phone, so it is not reflected in this table even though the software release supports the model.

2 Version 3.4.10.94

2.1 What's New

- C450HD: Introduced virtual sidecar for shared lines.
 Implements virtual sidecar for IP Phones that do not have a physical sidecar.
 Figure 2-1: Virtual Sidecar



Virtual sidecar supports up to 8 Programmable/Functional keys.

Parameter	Description
system/virtual_sidecar/enabled	1 = Enable - Displays a sidecar on right of the screen. This display contains Programmable Key tab, Calls tab, and Shared Call tab. 0 = Disable. Default = 0.
system/virtual_sidecar/single_click	1 = Enable: <ul style="list-style-type: none"> • Shared Call tab - Pressing on a call performs the following: <ul style="list-style-type: none"> ◆ When call is active, performs a Call Monitoring action by select. <p>Note: When single-click is enabled, and there is one call monitoring action for Shared Line Appearance/Shared Line Group, then a single click on the call list in Shared Call tab will monitor the call directly (it does not open the Call Monitoring screen).</p> ◆ When call is on hold, performs a resume call when call is local or pickup when remote. • Calls tab - toggles hold when call is active or resume if call is on hold. Brings ongoing screen to foreground. 0 = Disable: <ul style="list-style-type: none"> • Shared Call tab - Open the shared line active calls screen. • Calls tab - Bring call ongoing screen to foreground only. Default =0.

- **Security:** Starting from this release AudioCodes firmware is now signed by AudioCodes CA. AudioCodes Signed firmware is verified during upgrade (CVE-2023-22955).

Parameter	Description
system/verify_firmware_signature/enable	1 = Enable - Verifies the signature during the upgrading firmware. 0 = Disable. Default = 0.

- Zoom Intercom - The Intercom feature allows hands-free user-to-user conversations between their desk phones and/or soft clients. It enables users to call another extension in the key on the IP phone. The extension phone beeps to notify the User of the incoming Intercom call, then the User's phone automatically answers in speakerphone mode.

Use cases can include announcing parked calls, notifying of visitors, and engaging in hands-free communications with colleagues.



You must configure Customer template on Zoom Portal. For details on how to add a Customer template, please see

https://support.zoom.com/hc/en/article?id=zm_kb&sysparm_article=KB0062241.

Parameter	Description
voip/auto_answer/enabled	1 = Enable, to enable intercom. 0 = Disable. Default = 0.
personal_settings/functional_key/[n]/type=Intercom	For intercom key, this parameter must be set to Intercom. Default = EMPTY.
personal_settings/functional_key/[0-55]/speed_dial_number	Define the intercom number. Default = Null.
voip/services/intercom/autoMute	1 = Enable, to enable auto mute for intercom call. 0 = Disable. Default = 1.
voip/auto_answer/support_answer_multi_call	1 = Enable, to enable auto answer second intercom. 0 = Disable to not auto answer second intercom. Default = 0.
voip/auto_answer/headset_beep/enabled	When default audio device is headset: 1 = Enable, it will beep before answering an intercom. 0 = Disable. Default = 0.
voip/auto_answer/speakerphone_beep/enabled	When default audio device is speaker: 1 = Enable, it will beep before answering an intercom. 0 = Disable. Default = 0.

- **[Feature in preview]** Google calendar: User can sign in google account and view his/her calendar. User can join directly from the calendar to Zoom or Teams meetings.



Does not support for 450HD model IP Phones.

Figure 2-2: Sign in Google Calendar

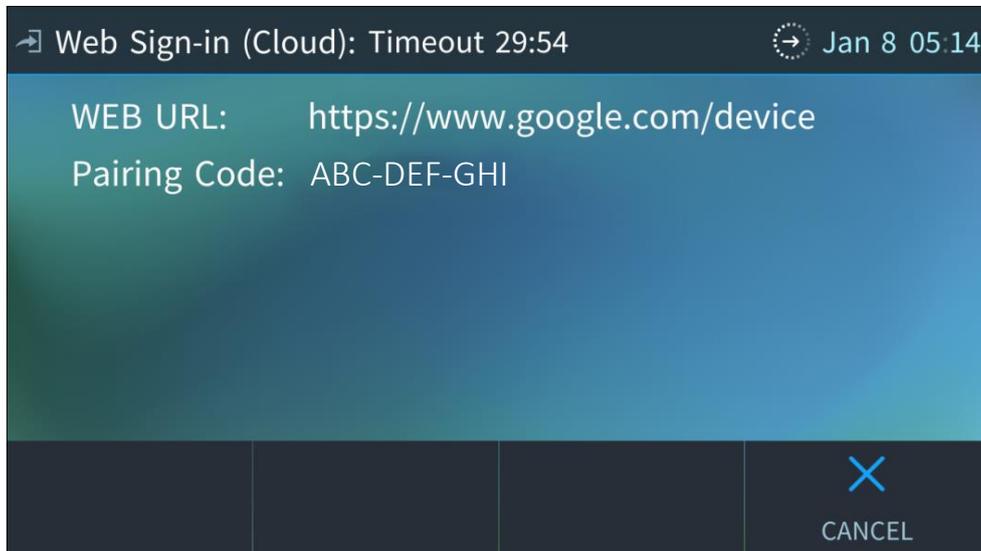
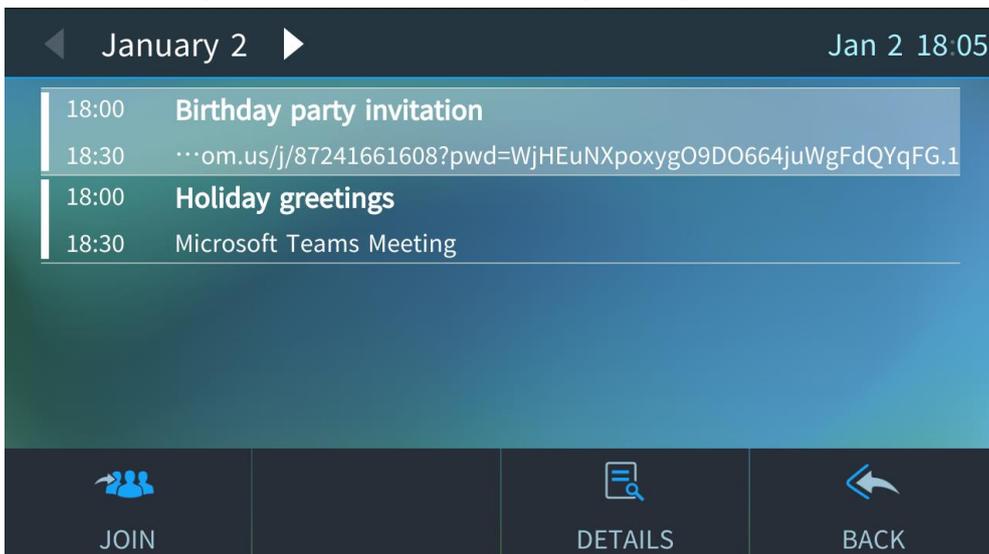


Figure 2-3: Zoom and Teams meeting on Google Calendar



Parameter	Description
account/office365/permission "USER"	Set this parameter to USER. Default = ADMIN.
lync/ews/GetEwsAddressMethod	Set this parameter to "FROM_CONFIGURATION_ONLY". Default = DYNAMIC.
account/web_signin/type	Set this parameter to GOOGLE. Default = OFFICE_365.

2.2 Resolved Constraints in Version 3.4.10.94

Resolved Zoom Constraints

Table 2-1: Resolved Zoom Constraints

Incident	Description
IPPUC-9216	Occasionally presence status does not change to "Busy" in a call when a call is occurring.
IPPUC-9264	C450HD: Call Monitoring UI does not return to Home when pressing "Back".
IPPUC-9274	445HD: IPP processes some invalid packets, cause IPP to slow down or one-way voice.
IPPUC-9320	Zoom: Fail to hold/end/make a new call in an emergency paging call.
IPPUC-9314	Redundancy proxy mode: After IP Phone switches from redundancy proxy to Primary server, it should not send registration info to Redundancy server.
IPPUC-9403	Zoom: Creating a remote conference take a long time in some Zoom environments.
IPPUC-9354	Japanese font appears different from standard font.
IPPUC-9381	Zoom: Select transfer during a call, the transfer window disappears immediately after it appears.

2.3 Known Constraints in Version 3.4.10.94

Known constraints from this release:

None

3 Previous Versions

3.1 Version 3.4.9.55



Rollbacks from Versions 3.4.9.55 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.1.1 What's New

- Configurable font size of functional keys text.
support new parameter to control the font size of functional keys text.

Parameter	Description
personal_settings/functional_key_font_size	Minimum Value = 20 Maximum Value = 43 Default = 39
personal_settings/exp_functional_key_font_size	Minimum Value = 10 Maximum Value = 30 Default = 26

- Zoom: Caller ID association per line keys.
Multiple functional keys can be assigned to the same SIP line. When explicitly dialing from a functional key (which is associated with a SIP line), we can specify an outbound caller id for this outgoing call.

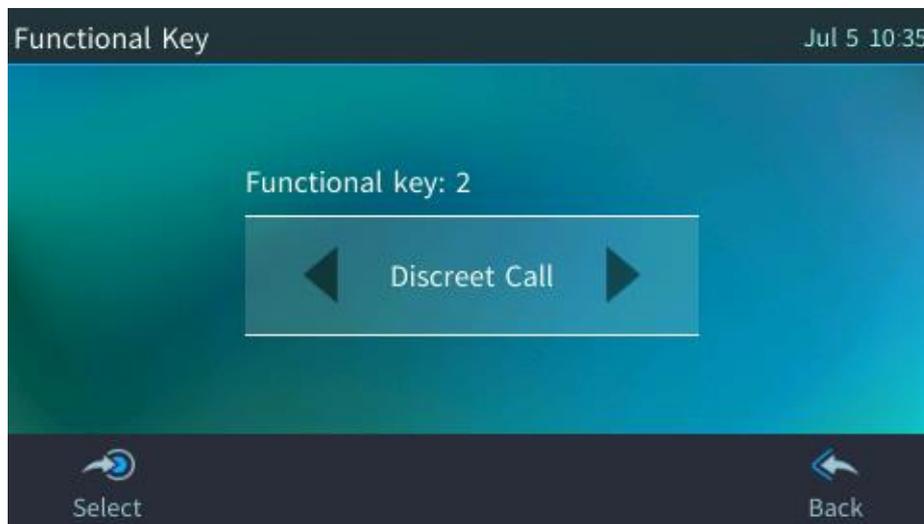
Parameter	Description
personal_settings/functional_key/n/outbound_callerId	Value = <sip:[call queue number]@[zoom server domain]> Example: <sip:+12055627793@10000698.zoom.us> Note: Applicable only for functional_key/n/type=SIP_ACCOUNT.

- Add Call Monitoring call actions for monitored call:
 - Support for switching between "Listen" "Whisper" and "Bargein" from Call Menu during monitoring call.
 - Support to "TakeOver" the call from Call Menu during monitoring call.
- Adding routingid in REFER SIP message during consult transfer.

Parameter	Description
voip/zoom/consult_transfer/add_routing_id	1 = Enable 0 = Disable Default=1

- Support for discreet call.

The feature answers a requirement for more security measures such as a silent mode call for public institutions. If a call is made in discreet mode, it's a one-way call to a remote phone. when in lock state, a discreet call does not require you to unlock the phone. The caller's phone does not indicate audially that a call is in progress. The phone's screen remains in idle mode and the backlight is not activated. The only indication that a call is in progress is the presence status of the caller changes to red (busy). Only the caller can end the call. The call is activated by a new, user-configurable function key:



It's recommended that the called party's phone be a dedicated phone to avoid the scenario of being on another call when needed for the discreet call; the phone automatically answers the discreet call; there is no need to pick up the handset. The called party then 'listens' to what's happening at the caller's end.

The following new configuration file parameters have been added to support this feature:

Parameter	Description
personal_settings/functional_key/[X]/key_label	Configure a label for the key. The label is displayed in the phone's screen next to the Functional Key. Make it intuitive to facilitate easy and quick action in an emergency. The label should be differentiated from other Speed Dial labels.
personal_settings/functional_key/[X]/line	Set to 0 for the first line to be used to make the discreet call. Default = -1
personal_settings/functional_key/[X]/speed_dial_number	Configure the emergency telephone number for the discreet call.
personal_settings/functional_key/[X]/type	Configure this parameter to DISCREETCALL.
personal_settings/discreet_call/enabled	1 = Enable 0 = Disable Default=0
personal_settings/discreet_call/type	<ul style="list-style-type: none"> ■ For SIP-GW environment, set to AC_PROPERTY_AUTO_ANSWER_HEADER. ■ For Zoom environment, set to ZOOM_PAGING.
voip/discreet/autoAnswerTime	An integer parameter which sets the auto answer timeout value. Default = 1 second.



Both caller and called party phones must be AudioCodes phones.

3.1.2 Resolved Constraints in Version 3.4.9.55

Table 3-1: Resolved Zoom Constraints

Incident	Description
IPPUC-9096	Parameter voip/paging/ring_enabled=0 doesn't work with standard cloud paging.
IPPUC-9124	Zoom remote conference: adding two of the same "entity id" participants only displays one participant in the conference.
IPPUC-8978	Zoom: Call log is removed when Enable/Disable call waiting.
IPPUC-8660	Cloud Paging: "Lock Call" option should be removed from "Call Menu" in cloud paging call.
IPPUC-8657	Need to add Zoom cloud paging softkey on UI.
IPPUC-8190	445HD: 'Entering privacy mode' screen quickly disappears from UI when enable share line privacy mode.
IPPUC-8900	Call Monitoring: Call Monitoring softkeys are unresponsive after one monitored conference call ends.
IPPUC-8894	Call Monitoring: UI display wrongly with takeover call and listen call.
IPPUC-8956	Call Monitoring: Remove irrelevant lock call softkey in Call Monitoring call.
IPPUC-9022	Call Monitoring: failed to switch from listen to whisper.
IPPUC-9027	SLG Call Monitoring: switch from listen to whisper ceases to work sometimes.
IPPUC-9016	SLG Conference Barge: the user does not exit the call after the other user pickups the hold call
IPPUC-8954	Cloud Paging: Fail to join cloud paging if enable call forward.
IPPUC-8967	SLA Call Monitoring: Second call number should change from 2 to 1 after the first call ends in Call Monitoring screen.
IPPUC-9131	Zoom DND sync: sometimes, when only DND for 1 st line (Index 0) is enabled, DND of all lines are enabled.
IPPUC-9178	Zoom: Call count displays on 1 st line key only even when the same line is assigned to several line keys.

3.1.3 Known Constraints in Version 3.4.9.55

Known constraints from this release:

None

3.2 Version 3.4.8.808



Rollbacks from Versions 3.4.8.808 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.2.1 What's New

- Support OpenSpace SIP Proxy

- Support feature key synchronization for DND and call forward using the following parameters:

Parameter	Description
system/feature_key_synchronization/enabled	1 = Enable 0 = Disable Default = 0
system/feature_key_synchronization/method	Configure this parameter to NOTIFY. Default = SUBSCRIBE.
voip/call_forward/support_multiple_type	Supports only one forward type. Set this parameter to 0. Default = 1
voip/call_forward/support_timeout	Set the parameter to 0, disable call forward timeout setting feature. Default = 1

- Support group call pickup. When one member receives an incoming call, other members are notified and can pick up the call using the following parameters:

Parameter	Description
personal_settings/functional_key/x/type	Set the function key type to GROUP_CALL_PICKUP. Default= EMPTY
voip/services/group_call_pickup/access_code	Set the parameter to empty (null). Default *98.
personal_settings/functional_key/x/speed_dial_number	Configure the pickup group number (e.g., *60).

- Support current status monitoring via BLF LEDs to monitor the state of other extensions using the following parameters:

Parameter	Description
voip/services/busy_lamp_field/enabled	1 = Enable 0 = Disable Default = 1
voip/services/busy_lamp_field/subscription_period	The interval between BLF and SIP SUBSCRIBE messages. Default = 3600
voip/services/busy_lamp_field/local_presence_for_lines	Set to 1 to use local presence. Default = 0

- Support remote conference call with four or more participants using the following parameters:

Parameter	Description
voip/services/application_server_type	Set the server type to GENERIC. Default = GENERIC
voip/services/conference/conf_ms_addr	Set the address of the server hosting the remote conference. Example: *66@sbc.teleswyz.ru
voip/services/conference/mode	Set the mode to REMOTE. Default = LOCAL

- Support for accessing Office 365 exchange services in Generic SIP mode. User can:
 - Sign-in to Office 365 exchange services via cloud signing or via username and password.
 - View calendar.
 - Join Teams/Zoom meeting via calendar.
 - Search contact from corporate directory

Required configuration:

Parameter	Description
account/office365/permission	Set this parameter to USER. Default = ADMIN
lync/ews/GetEwsAddressMethod	Set the method to FROM_CONFIGURATION_ONLY. Default = DYNAMIC



Basic Authentication in Office 365 Exchange services are no longer supported by Microsoft.

- Support Intrado ERS Location Information Service (HELD)

HELD (HTTP-Enabled Location Delivery) is a protocol that enables devices to request information about their location to a Location Information Service or LIS. Devices that support the HELD protocol are able to tightly integrate with ERS via the HELD service. The HELD Service is available in ERS Enterprise SIP Accounts and needs special activation and configuration. HELD-compliant hard-phones and softphones send their network information to the ERS in an XML request. ERS responds by determining the phone's location based on the pre-provisioned network map, sends this information to the phone in the form of locationURI or civic address. At call-time, the phone sends a SIP invite containing the locationURI, enabling ERS to retrieve the phone's location and route the call to the proper PSAP.

Table 3-2: HELD Configuration

Parameter	Description
location/HELD/server_url	Specify the HELD Server URL.
location/HELD/request_location_type	Either LocationURI, Civic, LocationURI_and_Civic.
location/HELD/nai.enable	Network Access Identifier (NAI). Default = 1 (Boolean)
location/HELD/Identity	Set the vendor-specific element to include in a location request message. Default = CompanyID
location/HELD/Identity_value	Set the value for the vendor-specific element to include in a location request message.
security/HELD_certificate_url	Certificate URL to use for server secure connection.
security/HELD_private_key_url	Private key URL to use for server secure connection.

Table 3-3: HELD Status

Parameter	Description
status/diagnostics/lldp/chassis/chassisId	Chassis ID of the switch that the device is connected to. MAC address: 12-digit hexadecimal number represented by colon- hexadecimal notation. Interface Name: String format. Example: <ChassisID>AgcEiFqSa oqA</ChassisID>
status/diagnostics/lldp/chassis/portId	ID of the switch port that the device is connected to. MAC address: 12-digit hexadecimal number represented by colon- hexadecimal notation. Interface Name: String format. Example: <PortID>BAkFR2kyLzA vMTc=</PortID>
status/diagnostics/lldp/chassis/chassisIdType	ERS supports the following Chassis ID subtypes: <ul style="list-style-type: none"> ■ "Switch Hostname" / (6) ■ "Switch IP" /(5) ■ "Switch MAC Address"/(4)
status/diagnostics/lldp/chassis/portIdType	ERS supports the following Port ID subtypes: <ul style="list-style-type: none"> ■ "Port Name" (5) ■ "Port MAC Address" /(3)

system/screenshots_path	This new configuration parameter defines a path in the file system of the IP Phone, where screenshots are saved to this specified folder in *.png format. if the folder does not exist it will be created. Default path = "/tmp/screenshots".
system/screenshots_mode	manual or auto mode doing of screenshots, it should be MANUAL or SCREENS_TRANSITION Default=NONE
system/screenshots_timer	delay(in ms) after release keys when a screenshot will be taken in auto mode Default=500

- Support for UI to return to idle screen timeout.

Parameter	Description
personal_settings/GetBackToIdleScreenTimeout	Defines the number of sec the IdleScreen will return to idle screen. If this parameter is set to 0 then timeout is disabled. Range 0 to 180 Default=0

- Support defining a recommended value for configuration parameters. If the parameter's value changes, the device will raise an alarm to the Device Manager. The alarm will only clear after the parameter's value returns to the recommended value.
- Raising an event into AudioCodes Device Manager when there is a failed/successful attempt in TELNET/SSH connection to IP Phone.

3.2.2 Resolved Constraints in Version 3.4.8.808

Table 3-4: Resolved Generic Constraints

Incident	Description
IPPUC-8541	Fail to answer 2nd call automatically when enable auto answer.
IPPUC-8576	Display name shows incorrectly when dial out via pause dialing.

Incident	Description
IPPUC-8796	Increase the max string length to 512 for all provisioning URL parameters that are currently defined to 128.
IPPUC-8958	Sometimes group call pickup fail.
IPPUC-8959	Fail to execute DTMF when dial to itself via pause dialing.
IPPUC-9154	C450HD: Significant delay in audio on USB or BT headsets.
IPPUC-9128	print user.error message appears often in the log.
IPPUC-9098	Gain/ringer parameters can be changed on the fly but will take effect after reboot.
IPPUC-9151	Remove unnecessary error messages in syslog.
IPPUC-9174	SIP-GW: Previous "Loading" screen appears when closing HeroCard window.
IPPUC-9006	Add and set configuration parameter <code>voip/services/subscribe_call_party_presence</code> to enable/disable subscribing presence of call remote party.
IPPUC-9194	Genband: IP Phone crashes during the call.
IPPUC-9179	445HD: Configure max volume of handsfree via parameter <code>voip/audio/gain/handsfree/WB_max_volume</code> .
IPPUC-9184	<code>voip/dialing/unanswered_call_timeout</code> doesn't work.
IPPUC-8751	All sensitive information should not be part of the configuration file (CVE-2023-22957).
IPPUC-8948	Encrypted Configuration File feature - allow Configurable Encryption Password (CVE-2023-22956).

3.2.3 Known Constraints in Version 3.4.8.808

Known constraints from this release:

Table 3-5: Known Generic Constraints

Incident	Description
IPPUC-9231	<p>OpenSpace SIP: Server feature key synchronization. Changing call forward type on the phone must be done via disabling call forward first.</p> <p>Example:</p> <ol style="list-style-type: none"> 1. Disable call forwarding on the phone. 2. Change from "always call forward" to "no answer call forward". 3. Enable call forwarding on the phone.

3.3 Version 3.4.8.198.32



Rollbacks from Versions 3.4.8.198.32 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.3.1 What's New

- Zoom Server Redundancy: Failback: Support using Register message to detect if primary server is active.

Parameter	Value (Default)	Description
voip/signalling/sip/detect_primary_proxy/method	OPTIONS	<p>OPTIONS - send SIP OPTIONS message to detect if primary server or primary outbound proxy (server) is active.</p> <p>REGISTER - send SIP REGISTER message to detect if primary server or primary outbound proxy (server) is active.</p> <p>(voip/signalling/sip/failback_retry_timeout configuration value must be > 0)</p>

- Zoom Do Not Disturb (DnD) sync:
Synchronizing DnD presence state between Zoom client and Zoom Phones belonging to the same user.
- DnD sync required configuration:

Parameter	Note
system/feature_key_synchronization/enabled	0 (Default) = To disable DnD sync. 1 = To enable DnD sync. This parameter must be set to 0.
system/feature_key_synchronization/local_dnd	Should be set to 1 for Zoom. Default = 0.
system/feature_key_synchronization_dnd_only/enabled	Should be set to 1 for Zoom. Default = 0.
voip/services/dnd_reject_code	Should be set to 480 for Zoom. Default = 603.
voip/services/dnd_per_line_enabled	Provides two different UI experiences: 0 = Display DnD state only on user's own line, which is typically line 0. 1 (Default) = Display DnD state on all lines.
voip/line/[0-29]/do_not_disturb/enabled	When the value of the <i>/voip/services/dnd_per_line_enabled</i> parameter (mentioned above) is set to 1, leave this <i>'do_not_disturb'</i> parameter as the default value (1). However, when the <i>/voip/services/dnd_per_line_enabled</i> parameter (mentioned above) is set to 0, set all 30 lines (0-29) except for the user's own line (typically line 0) for this parameter to 0.

3.3.2 Resolved Constraints in Version 3.4.8.198.32

Table 3-6: Resolved Constraints

Incident	Description
IPPUC-9154	C450HD: Significant delay in audio on USB or BT headsets.
IPPUC-9098	Gain/ringer parameters can be changed on the fly but will take effect after reboot.
IPPUC-9128	print user.error message appears often in the log.

Incident	Description
IPPUC-9008	Zoom: Wrong ZPLS behavior –The Zoom phone utilizes a primary proxy along with a secondary proxy. The primary proxy domain resolves to two IPs, namely IP #1 and IP #2. In the event that IP #1 encounters an issue or fails, The Zoom phone is supposed to switched to IP #2 as the backup option, rather than the secondary proxy.
IPPUC-9049	Zoom: Static DNS cache doesn't work well.
IPPUC-9038	Zoom: IP Phone plays unexpected ringtone on hang up when multicast paging packets present. The issue can be resolved by setting /voip/paging/ring_enable = 0.
IPPUC-9109	Zoom: Fail to failback from backup cloud server to primary cloud server. (This only happens when multiple lines are configured).
IPPUC-9123	Zoom: Fail to receive emergency paging call (cloud paging) when DnD enabled. (This only happens when allow emergency paging interrupts for ongoing calls is enabled).
IPPUC-9117	Zoom: IP Phone stopped sending any SIP messages after a period of time. (This situation arises only in cases where port binding may fail during failover or failback processes).
IPPUC-9092	Occasionally, the phone stops ringing when UI displays incoming call.

3.3.3 Known Constraints in Version 3.4.8.198.32

Known constraints from this release:

- None

3.4 Version 3.4.8.198.17



Rollbacks from Versions 3.4.8.198.17 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.4.1 What's New

- Support for **Zoom 4+ conference (Remote Conference)**. Allows making a conference call with four or more participants.

3.4.2 Resolved Constraints in Version 3.4.8.198.17

Table 3-7: Resolved Constraints

Incident	Description
IPPUC-8940	Remote/4+ Conference: Does not show participants in roster window (in some cases).
IPPUC-8890	Remote/4+ Conference: After ending a conference call, the line LED is still red (busy). It should return to green (idle).
IPPUC-8906	Remote/4+ Conference: Moderator(presenter) cannot remove participants.
IPPUC-8930	Remote/4+ Conference: Merging call fails.
IPPUC-8883	ZPLS: After switching back to primary server, Zoom phone does not send SIP subscribe for presence.
IPPUC-8887	ZPLS: IPP sends incorrect unsubscribe message.

Incident	Description
IPPUC-8931	Cloud Paging WEB UI: "function key" page contains a misspelling.
IPPUC-8905	Warm (consult) transfer: Add routingid in the INVITE message.

3.4.3 Known Constraints in Version 3.4.8.198.17

Known Constraints from this Release

- To use IP Phone in ZPLS environment, you need to add the following configuration to the provisioning template : voip/dns_cache/mode = STATIC_DNS_CACHE_FIRST

Known Constraints from Previous Releases

None

3.5 Version 3.4.8.198.13



Rollbacks from Versions 3.4.8.198.13 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.5.1 What's New

- Support for Enhanced Emergency Cloud Paging.

Unlike Standard Emergency Paging, this page interrupts both regular calls and standard page calls.

Enhanced Emergency Cloud Paging is configured on the Zoom Portal per page group. When a Zoom phone receives a paging call, the behavior follows the new priority definition as shown in the following table:

In progress (Receiver)	Incoming call (Receiver)	Expected behavior
Call	Enhanced Emergency Cloud Paging	Call interrupted (paging will be answered)
Standard Page	Enhanced Emergency Cloud Paging	Ongoing paging will be interrupted
Emergency Page	Enhanced Emergency Cloud Paging	Ongoing paging will not be interrupted
Emergency Page	Call	Ongoing call will not be interrupted
Emergency Page	Standard Emergency Page	Ongoing paging will not be interrupted

3.5.2 Resolved Constraints in Version 3.4.8.198.13

None

Table 3-8: Resolved Constraints

Incident	Description
IPPUC-8734	Zoom Phone Local Survivability (ZPLS): Registration fails when setting primary outbound with an invalid host name.
IPPUC-8861	The Zoom phone fails to failback to the primary proxy; the SIP option is sent to the incorrect port (port 65535).

3.5.3 Known Constraints in Version 3.4.8.198.13

Known Constraints from this Release

None

Known Constraints from Previous Releases

None

3.6 Version 3.4.8.198.9



Rollbacks from Versions 3.4.8.198.9 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.6.1 What's New

- Support Zoom Phone Local Survivability (ZPLS) feature.

Configuration details:

Parameter	Value (Default)	Description
voip/signalling/sip/keepalive_options/enabled	1	Enable –keep alive OPTIONS will be used for proxy survivability detection. Default: 0 Attention: Should be enabled
voip/signalling/sip/keepalive_options/timeout	300	Keep Alive (OPTIONS) interval in seconds Default: 300 Attention: Set on demand
voip/signalling/sip/redundant_outbound_proxy/enabled	0	Enable redundant outbound proxy Default: 0 Attention: Should be enabled
voip/signalling/sip/sip_outbound_proxy/enabled	0	Enable outbound proxy Default: 0 Attention: Should be enabled

Parameter	Value (Default)	Description
voip/signalling/sip/sip_outbound_proxy/addr	(IP Address or Host Name)	Primary Outbound proxy address  Attention: Should be set
voip/signalling/sip/sip_outbound_proxy/port	5060	65535 send DNS SRV query 1024~65534 send DNS A query Default: 5060  Attention: Should be set to 65535 for ZPLS
voip/signalling/sip/redundant_outbound_proxy/address	(IP Address or Host Name)	Redundant outbound proxy address  Attention: Should be set
voip/signalling/sip/redundant_outbound_proxy/port	5060	65535 send DNS SRV query 1024~65534 send DNS A query Default: 5060  Attention: for ZPLS no need SRV, so need to set the right port.
voip/signalling/sip/switch_redundant_to_primary/timer	0	The time interval before the IPP failback after the primary proxy detection succeeds, Default: 0  Attention: Set on demand
voip/signalling/sip/switch_to_DNS_primary/timer	0	Relevant only for failover between the servers resolved from the primary outbound proxy. The time interval before the IPP executes the failback. Default: 0  Attention: Set on demand
voip/signalling/sip/redundant_outbound_proxy/server_failure_code	null	It specifies which response code we see as a failure which requires a failover/fallback. Special values 4xx means all response codes starting with 4 will be seen as a failure. Special values 5xx means all response codes starting with 5 will be seen as a failure. Default: null  Attention: For ZPLS, leave the setting as default (Null).

3.6.2 Resolved Constraints in Version 3.4.8.198.9

The table below shows the resolved constraints from previous versions.

Table 3-9: Resolved Constraints

Incident	Description
IPPUC-8149	All parking lot programmable keys have their status turn to grey when adding/deleting a paging key.

3.6.3 Known Constraints in Version 3.4.8.198.9

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.7 Version 3.4.8.198.7



Rollbacks from Versions 3.4.8.198.7 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.7.1 What's New

- New Zoom requirements for Cloud Paging were implemented.

Configuration details:

```
personal_settings/functional_key/[0-38]/type
value: ZOOM_PAGE_STANDARD or ZOOM_PAGE_EMERGENCY
personal_settings/functional_key/[0-38]/line
value: 0 (the line to dial out the paging call, usually 0)
personal_settings/functional_key/[0-38]/key_label
value: can be : "emergency page" or "standard page"
personal_settings/functional_key/[0-38]/speed_dial_number
value: the paging number
```

3.7.2 Resolved Constraints in Version 3.4.8.198.7

The table below shows the resolved constraints from previous versions.

Table 3-10: Resolved Constraints

Incident	Description
IPPUC-8540	[RX50] when auto-answer a call is answered and placed on hold, DUT plays a beep tone endlessly.
IPPUC-8356	[Zoom remote call control] call placed on hold cannot be disconnected.
IPPUC-8616	[Zoom remote call control] Blind/Warm Transfer from client fails when DUT has multiple lines.

Incident	Description
IPPUC-8466	[Zoom] When an IP Phone is running for a long time, some shared lines cannot display the 'call list'.
IPPUC-8470	[Zoom] occasionally, LDAP contact presence cannot update in real time.
IPPUC-8575	[Zoom] While monitoring in 'Listen' mode, the "<listen>" string is missing.
IPPUC-8596	Under certain conditions, IP Phone will lose registration and then re-register.
IPPUC-8595	After a paged held call is resumed, the softkeys displayed change from paged call (limited functionality) to normal call softkeys.
IPPUC-8602	[zoom] After a long hold (e.g., 40 minutes), multiple resume/holds after, may cause one way audio.

3.7.3 Known Constraints in Version 3.4.8.198.7

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.8 Version 3.4.8.198.1



Rollbacks from Versions 3.4.8.198.1 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.8.1 What's New

- Dialing external contact from LDAP directory. Requires configuring the following configuration:

```
system/ldap/HomePhoneNumber_attr=phoneNumber
lync/contact_search_method=LDAP
```

- Zoom - Support dialing an emergency call when phone is locked. Requires configuring the following configuration:

```
voip/services/emergency_call_list=933 911
```



For more information regarding the Phone Lock feature, please contact AudioCodes.

3.8.2 Resolved Constraints in Version 3.4.8.198.1

The table below shows the resolved constraints from previous versions.

Table 3-11: Resolved Constraints

Incident	Description
IPPUC-8403	Zoom remote call control- "bound" icon on the top bar was missing when Zoom client and IPP were bound

3.8.3 Known Constraints in Version 3.4.8.198.1

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.9 Version 3.4.8.198



Rollbacks from Versions 3.4.8.198 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.9.1 What's New

Version 3.4.8.198 offers the following new features for the Zoom environment:

1. Remote Call Control

The Zoom client can control the following functions while you are using a desk phones:

- Making an outgoing call
- Holding/resuming a call
- Muting/Unmuting a call
- Syncing mute state with the Client when the mute/unmute action is taken on the phone
- Answering a call
- Ending a call
- Consult transfer
- Blind transfer
- Sending DTMF over a call



The Zoom client requires binding itself with the desk phone. The phone user must accept or reject the binding request.

No extra configuration parameters are needed to be provisioned to the desk phones.

2. Cloud Paging

New ability to configure a page group on a Zoom server. Each page group can contain one or more senders and receivers. Senders can issue a page and receivers can receive the page.

- It can be configured as a speed dial on Zoom portal and be provisioned to phones.
- The Zoom server will prompt to senders to select “standard” or “emergency” page.
- The paging receiver can treat the call (i.e., reject the call, auto answer the call, etc., according to its own status (i.e., busy, DND, idle, etc.) and the paging type (i.e., emergency or standard). See table below for an example:

Desk Phone State	Emergency	Standard
Idle	Auto-Answer	Auto-Answer
Busy on a regular call or Emergency paging	Call waiting list	Call waiting list
Busy on a Regular paging	Auto-Answer	Call waiting list

Desk Phone State	Emergency	Standard
DnD	Auto-Answer	Reject



No extra configuration parameters are needed to be provisioned to the paging receiver phones.

3.9.2 Resolved Constraints in Version 3.4.8.198

The table below shows the resolved constraints from previous versions.

Table 3-12: Resolved Constraints

Incident	Description
IPPUC-8348	Configuration parameter (voip/signalling/sip/redundant_proxy/address) for redundant proxy address field length enlarged to 128 characters.
IPPUC-8315	Support for PAI UPDATE. Warm Transfer scenario - when a transferred call includes relevant information so that the person receiving the transfer does not have to ask for the information already given.

3.9.3 Known Constraints in Version 3.4.8.198

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.10 Version 3.4.7.759.8



Rollbacks from Versions 3.4.7.759.8 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.10.1 What's New

Version 3.4.7.759.8 offers the following updates:

- Updated Icons for all call monitoring related softkeys.
- [445HD]: Short press on SharedLine programmable key opens line menu screen when personal_settings/sidecar_mode=STATIC.
NOTE: personal_settings/sidecar_mode default value changed to STATIC.

3.10.2 Resolved Constraints in Version 3.4.7.759.8

The table below shows the resolved constraints from previous versions.

Table 3-13: Resolved Constraints

Incident	Description
IPPUC-8274	Disabled line is displayed in settings/account and settings/Ringtone menus

3.10.3 Known Constraints in Version 3.4.7.759.8

The table below shows the constraints that are known to exist in this version.

Table 3-14: Known Constraints from This Release

Incident	Description
	Sometimes when you create a delegate relationship, one of the lines remains unregistered. Perform re-sync via Zoom portal or restart the phone.

Table 3-15: Known Constraints from Previous Releases

Incident	Description
None	

3.11 Version 3.4.7.759.4



Rollbacks from Versions 3.4.7.759.4 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.11.1 What's New

Version 3.4.7.759.4 offers the following new feature:

Zoom Features

- **Call monitoring for shared line group:** Allow all members of specific shared line groups to monitor each other. Monitoring occurs when the phone of a user makes or receives a call from the shared line group's extension or a direct number.
- **Conference Barge for shared line groups:** Account owners and admins can enable conference barge when setting up call monitoring for a shared line group. Conference barge allows multiple members to barge into a call that is received by a user of the shared line group.
- **Call monitoring for a shared line:** Allow a phone user (e.g., executives or assistants) to be part of a call delegation (A.K.A. a **shared line**) and allow both members of the shared line to monitor each other.
- **Privacy mode:** Account owners and admins can enable a privacy feature for shared line groups or a shared line (i.e., call delegation). This allows shared line group and shared line appearance members to lock the call, preventing others from accessing the call (e.g., picking up a held call, listening, whispering, barging, or taking over).

Parameter name	Description
voip/line/[0-29]/zoom_privacy_mode/enabled	<ul style="list-style-type: none"> ■ 0 = disabled (default) ■ 1 = enabled

Generic Features

- **DNS Cache** - Allows to statically configure a set of DNS NAPTR/SRV/A records in one of two 2 modes:

Parameter name	Description
voip/dns_cache/mode	<ul style="list-style-type: none"> ■ STATIC_DNS_CACHE_FIRST - Use static DNS cache preferentially. ■ DNS_QUERY_FIRST - Use DNS server resolution preferentially.
voip/dns_cache_A/[0-7]/name	A record name
voip/dns_cache_A/[0-7]/ip	A record IP address
voip/dns_cache_A/[0-7]/ttl	A record TTL
voip/dns_cache_srv/[0-7]/name	SRV record name
voip/dns_cache_srv/[0-7]/port	SRV record port
voip/dns_cache_srv/[0-7]/weight	SRV record weight
voip/dns_cache_srv/[0-7]/priority	SRV record priority
voip/dns_cache_srv/[0-7]/target	SRV record target
voip/dns_cache_srv/[0-7]/ttl	SRV record TTL
voip/dns_cache_naptr/[0-7]/name	NAPTR record name
voip/dns_cache_naptr/[0-7]/flag	NAPTR record flag
voip/dns_cache_naptr/[0-7]/order	NAPTR record order
voip/dns_cache_naptr/[0-7]/preference	NAPTR record preference
voip/dns_cache_naptr/[0-7]/replace	NAPTR record replace
voip/dns_cache_naptr/[0-7]/service	NAPTR record service
voip/dns_cache_naptr/[0-7]/ttl	NAPTR record TTL

- [445HD] Ability to increase max volume on handset to +8dB.

Parameter name	Description
voip/audio/gain/handset/WB_max_volume	-27 to 8 (default = 3)

- [445HD] Added CallNoAnswerTimeout parameter to stop unanswered calls from ringing after a specific timeout.

Parameter name	Description
voip/services/call_no_answer_timeout	<ul style="list-style-type: none"> ■ 0 = disabled ■ 1-600 = Timeout (in seconds)

- Automatic switch of sidecar pages:
 - When a user changes a new call line on the New Call screen then the sidecar will be switched accordingly.
 - The sidecar is switched when an active call is changed.
- Default Side car page:
Default view of the sidecar to the first SCA index which is given by a config parameter.

- Configurable PK Led color

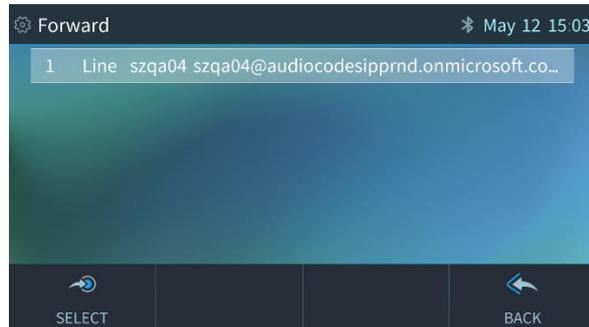
Introduce PK LED color control

Parameter name	Description
personal_settings/sidecar_mode	<p>Determines whether 445HD IPP's sidecar is automatically switched when user chooses current line (e.g., on pressing Line's PK, on answering the call etc.)</p> <ul style="list-style-type: none"> ■ [Dynamic] (Default) ■ [STATIC]
personal_settings/sidecar/default_page_line	<p>Determines what page will be displayed by default on the 445HD sidecar.</p> <ul style="list-style-type: none"> ■ [-1] (Default)(Minimum) ■ [29] (Maximum)
personal_settings/led_color_for_incoming_state	<p>Allows control the LED color for incoming state.</p> <ul style="list-style-type: none"> ■ OFF ■ RED ■ GREEN (Default) ■ ORANGE
personal_settings/led_color_for_initiated_state	<p>Allows control the LED color for initiated state.</p> <ul style="list-style-type: none"> ■ OFF ■ RED ■ GREEN (Default) ■ ORANGE

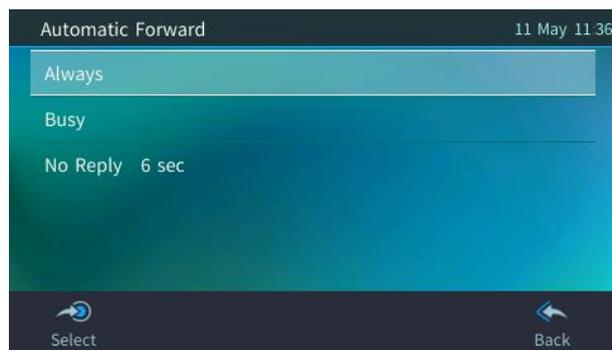
- Call forward UI improvement.

To configure call forwarding:

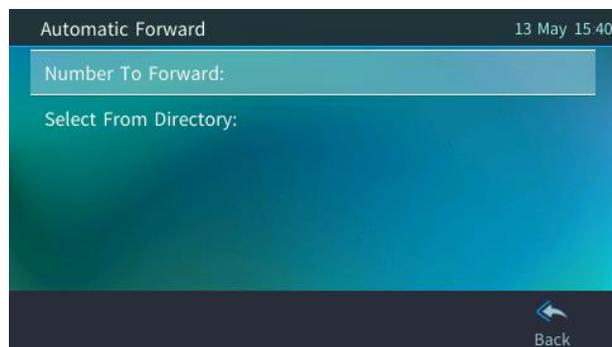
1. Open the **Automatic Forward** screen (**MENU** hard key > **Settings** > **Forward settings** -or- press the **Forward** softkey when the screen is in idle state):



2. Press the **Select** softkey.



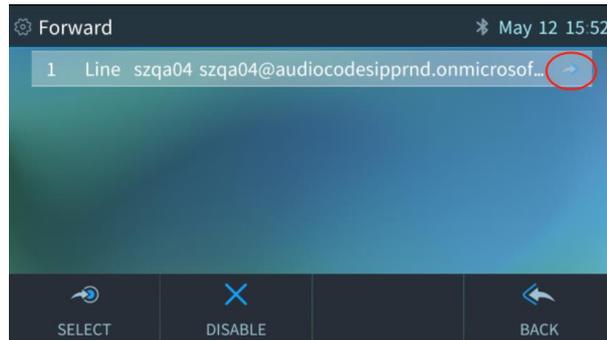
3. In the Automatic Forward screen, select either:
 - **Always:** incoming calls will always be forwarded.
 - **Busy:** incoming calls will be forwarded when the user is using the phone.
 - **No Reply 6 sec:** incoming calls will be forwarded if the user doesn't answer after a specified number of seconds; the default is **6** but you can configure up to **98** seconds.
4. After the selection, configure the phone number to which you want the calls to be forwarded.



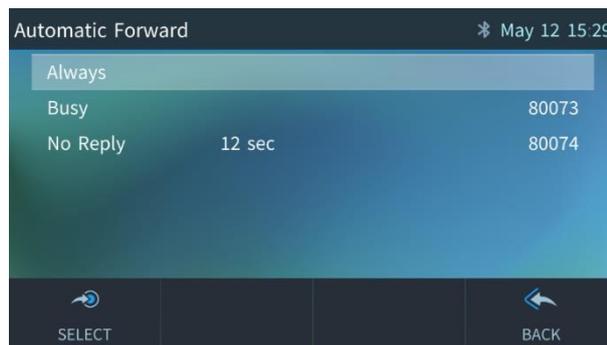
5. Alternatively, navigate to and select the **Select From Directory** option and then choose the contact to whose number you want the calls to be forwarded.

6. Press the **Start** softkey that's then activated. you will be returned to the idle screen; the forward icon  on right will appear); calls will automatically be forwarded to the configured number.

You can also see the 'forwarding' flag on this screen.

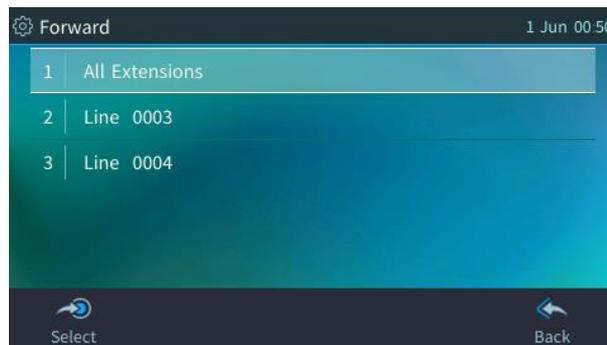


7. You can configure forward 'Busy' and 'No Reply' at the same time. In the example below, incoming calls will be forwarded to 80073 when phone is busy, and if phone is in idle state, calls will be forwarded to 80074 after ringing for 12 seconds.

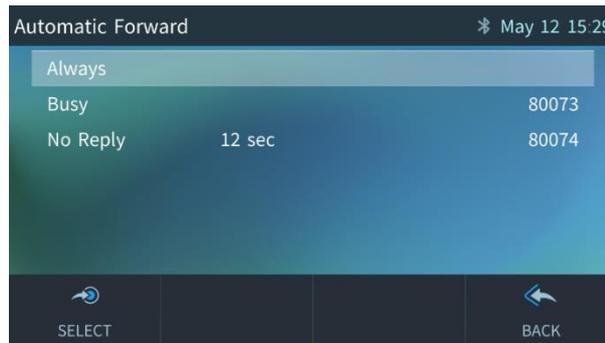


To configure call forwarding on multiple lines:

1. Open the **Forward** screen (press the **Forward** softkey when the screen is in idle state).



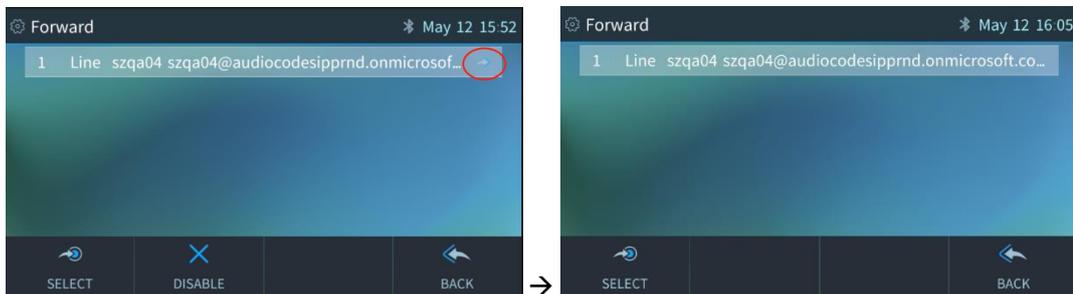
2. Navigate to a line extension on which to configure call forwarding; Select the line; the Automatic Forward screen is displayed.



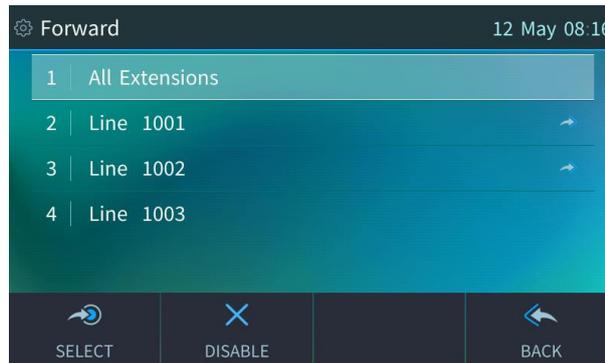
2. Configure call forwarding on that line extension.

To stop call forwarding:

- When the phone is in idle state:
 1. Press the **Forward** softkey
 2. Press the **DISABLE** softkey, the forward icon  on right will disappear.



- For multiple lines, you can select a specific line or select **All Extensions** and then press the **DISABLE** softkey; the forward icon  on right will disappear (based on your selection).



3.11.2 Resolved Constraints in Version 3.4.7.759.4

The table below shows the resolved constraints from previous versions.

Table 3-16: Resolved Constraints

Incident	Description
IPPUC-6928	Under certain conditions, the call log records the wrong extension.
IGS-3330	Disable or hide the option of factory reset for the "user" role in the HaaS mode.

Incident	Description
IPPUC-7797	Under certain conditions, the display name is truncated in the call log.
IPPUC-8063	When retrieving a parked call using a boss line, remote ID is not shown.
IPPUC-8050	Sidecar displays blank title if Zoom account has two call monitoring keys.
IPPUC-7931 IPPUC-7937	LDAP contact search does not start automatically after typing the first few characters.

3.11.3 Known Constraints in Version 3.4.7.759.4

The table below shows the constraints that are known to exist in this version.

Table 3-17: Known Constraints from This Release

Incident	Description
IPPUC-8122	[Zoom Server] RX50: DUT does not display call monitoring, group call pickup, and BLF programmable key on UI.
IPPUC-7605	[Zoom Server] Unnecessary Boss line voice mail option in the delegated phone.

Table 3-18: Known Constraints from Previous Releases

Incident	Description
	IP Phone limitation.
	Zoom: Supports multiple shared lines (to configure up to 15 shared lines).
IPPUC-6854	The maximum calls that can be displayed in “shared line call list” is 11.
IPPUC-6912	Shared line top bar does not appear after downgrading from release 3.4.7 to release 3.4.5 (Perform a “Restore to Default” to restore top bar in release 3.4.5).
	Zoom Server limitations
	Zoom: Supports multiple shared lines (to configure up to 15 shared lines)
	Number of simultaneous calls is limited to 4 per user (i.e., a user with more than one device is still limited to 4 simultaneous calls).
IPPUC-6802	The call list in the detailed call list screen disappears after a long call (about 40 minutes).
IPPUC-6814	Pick Up Call fails when IP Phone device is already managing 3 calls.
IPPUC-6804	Monitoring a call duration will becomes incorrect after the monitoring device performs a registration (due to restart or restore defaults). The call duration will display the time starting from the registration (ignoring the time prior to registration).
	General Zoom server limitation
IPPUC-6764	Username with more than 16 characters will be cut in relevant calls/call log screens.
	If user A is set as “delegated” by user B, and User B set call forwarding: incoming calls are not forwarded (i.e., User A and User B phones will ring).
	General limitation
IPPUC-6519	RTCP XR / RX50: Signal/Noise level and RERL (Residual Echo Return Loss) are not supported.

Incident	Description
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.7 using the following procedure: <ol style="list-style-type: none"> 1. Upgrade to 3.4.7 2. Restore to default 3. Downgrade to 3.4.4 or 3.4.5 4. Upgrade again to 3.4.7
	Roll backs from Versions 3.4.7.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.
	OVOC / Device Manager limitations
IPPUC-6644	No "IP Phone Speaker Firmware Download Failure" alert send out after download firmware fail.
IPPUC-6404	Change provisioning method STATIC->DYNAMIC via Web-UI triggers cached provisioning cycle.
IPPUC-2479	[RX50] does not support local 3-way conference calls.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
-	The special characters öß\$üä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.12 Version 3.4.6.687



Rollbacks from Versions 3.4.6.687 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.12.1 What's New

Version 3.4.6.687 offers the following new feature:

- Broadsoft application server Feature key synchronization for DND and Call Forward.

- Added unique User Agent Identifier for SIP register allowing identify of an agent without using its IP address. Support requires enabling the following parameter:
voip/signalling/sip/add_sip_instance/enable
- Improvements in Hebrew language translation.
- SIP gateway now supports RX50 device.

3.12.2 Resolved Constraints in Version 3.4.6.687

The table below shows the resolved constraints from previous versions.

Table 3-19: Resolved Constraints

Incident	Description
IPPUC-7855	Fixed pause dialing bugs. <ul style="list-style-type: none"> ■ Support pause dialing on SIP-URL ■ Support DTMF "A~D"
IPPUC-7829	On-call transfer scenario requires an "Ack" to delete the transfer message.
	SIP gateway now supports RX50 device.

3.12.3 Known Constraints in Version 3.4.6.687

None.

Table 3-20: Known Constraints from Previous Releases

Incident	Description
	<p style="text-align: center;">IP Phone limitation</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
IPPUC-6854	The maximum calls that can be displayed in “shared line call list” is 11
IPPUC-6912	Shared line top bar does not appear after downgrading from release 3.4.6 to release 3.4.5 (Perform a “Restore to Default” to restore top bar in release 3.4.5).
	<p style="text-align: center;">Zoom Server limitations</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
	Number of simultaneous calls is limited to 4 per user (i.e., a user with more than one device is still limited to 4 simultaneous calls).
IPPUC-6802	The call list in the detailed call list screen disappears after a long call (about 40 minutes).
IPPUC-6814	Pick Up Call fails when IP Phone device is already managing 3 calls.
IPPUC-6804	Monitoring a call duration will become incorrect after the monitoring device performs a registration (due to restart or restore defaults). The call duration will display the time starting from the registration (ignoring the time prior to registration).
	<p style="text-align: center;">General Zoom server limitation</p>
IPPUC-6764	Username with more than 16 characters will be cut in relevant calls/call log screens.
	If user A is set as “delegated” by user B, and User B set call forwarding: incoming calls are not forwarded (i.e., User A and User B phones will ring).
	<p style="text-align: center;">General limitation</p>
IPPUC-6519	RTCP XR / RX50: Signal/Noise level and RERL (Residual Echo Return Loss) are not supported.
IPPUC-6876	<p>“New Device Unlock Code” screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure:</p> <ol style="list-style-type: none"> 1. Upgrade to 3.4.6 2. Restore to default 3. Downgrade to 3.4.4 or 3.4.5 4. Upgrade again to 3.4.6
	Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.
	OVOC / Device Manager limitations
IPPUC-6644	No “IP Phone Speaker Firmware Download Failure” alert send out after download firmware fail.
IPPUC-6404	Change provisioning method STATIC->DYNAMIC via Web-UI triggers cached provisioning cycle.
IPPUC-2479	[RX50] does not support local 3-way conference calls.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2575	The user’s status indication (available / in a call) is occasionally displayed as ‘?’ in the Call Log.
-	The special characters ößüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3794	<p>In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter ‘voip/media/allow_multiple_rtp’.</p>
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).

Incident	Description
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.13 Version 3.4.6.629



Rollbacks from Versions 3.4.6.629 to versions 3.4.2 or earlier must be done via 3.4.4.xxx first.

3.13.1 What's New

Version 3.4.6.629 offers the following new feature:

- Support for Microsoft Teams SIP Gateway. See [here](#) for the Microsoft article 'Enable core Microsoft Teams calling functionality on compatible legacy SIP phones with SIP Gateway'. The SIP Gateway lets organizations use AudioCodes' SIP devices with Microsoft Teams to leverage existing investments in SIP devices. You can now sign-in to Teams and make and receive calls with the following AudioCodes SIP phones for personal use or for use and mainly as common area phones (CAPs):

- 450HD
- C450HD
- 445HD

Using SIP Gateway, users can do all of the following:

- Make calls: Users using SIP devices can make calls to the Public Switched Telephone Network (PSTN), to other SIP devices, and to Teams and Skype for Business users. Users using SIP devices can only call users who have phone numbers.
- Receive calls: The device can receive calls from any Teams or Skype for Business client, or from other SIP devices connected to the Teams SIP Gateway as long as the calling user has a valid phone number.
- Multiple simultaneous calls: A SIP device user in a call can put the call on hold to make or receive other calls. A SIP device user can also conference two calls.
- Do not disturb: A SIP device user can set do not disturb on the device so that the device will not ring for incoming calls. This has no impact on the user's status on all other Teams endpoints.
- Hold/Resume and Mute/Unmute: A SIP device user can hold and resume or mute and unmute a call by using the features for those actions on the device.
- Voicemail: SIP device users can listen to electronically stored voice messages that callers leave for them.
- Message waiting indicator: SIP device users can receive notifications that alert them when they have new voicemail messages.
- Sign-in and sign-out: SIP devices users can sign in and sign out of Teams on the device.

- Dual-tone multi-frequency: SIP device users can press number keys to provide input during interactive voice response calls.
- Teams meetings: A SIP device user can join a Teams meeting by dialing the meeting access number. Dialing out to a same organization user's phone number is currently not supported. However, guest users from another organization can be added to a Teams meeting by a participant who dials out to a guest user's number to include that guest. NOTE: Adding a Teams meeting participant via "request to join" currently won't alert a SIP device.
- Call transfers: SIP device users can transfer calls. SIP Gateway supports both blind and consultative transfers.
- Local call forwarding: A SIP device user can set forwarding rules (always, on timeout, and busy) for the device. If the device is connected to the SIP Gateway, then the call will be redirected to the target address based on the rule that the device user set. To make local call forwarding work, the admin must set the AllowCallRedirect attribute in Set-CsTeamsCallingPolicy to Enabled.

See [here](#) to find out what SIP Gateway can do for organizations and what hardware, software, and licenses organizations need for it ('Plan for SIP Gateway').

See [here](#) how to configure SIP Gateway so that organizations can use compatible SIP devices with Microsoft Teams (Configure SIP Gateway').

3.13.2 Known SIP Gateway Constraints

The table below shows the constraints that are known to exist.

Table 3-21: Known Constraints

Incident	Description
-	[RX50] Currently not supported
-	By design, Teams users must have a phone number with PSTN calling enabled to use SIP Gateway. <ul style="list-style-type: none"> ■ Dialing via URI is not supported. Dialing to a different organization via URI will not be possible. ■ For joining a conference via dial-in, users must have a DID or phone number with PSTN calling enabled. The conference bridge must have a DID number.
-	<ul style="list-style-type: none"> ■ The following features are by design unsupported: <ul style="list-style-type: none"> • Click to Join and Conference Roster • N-way conference (only three-way conference and dial-in to conference are supported) • Calendar • Visual Voicemail • Hot Desking • Search for a Contact • Presence • Discreet Call • Device Manager (OVOC plugin) (Roadmap) • Sign-in <ul style="list-style-type: none"> ◆ With Username and Password ◆ Web login • Dynamic location for E911 calls (SIP devices use <i>static location</i> for emergency calls, defined by DID number/user and not according to the real location defined by the switch).

Table 3-22: Known Constraints from Previous Releases

Incident	Description
	<p style="text-align: center;">IP Phone limitation</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
IPPUC-6854	The maximum calls that can be displayed in “shared line call list” is 11
IPPUC-6912	Shared line top bar does not appear after downgrading from release 3.4.6 to release 3.4.5 (Perform a “Restore to Default” to restore top bar in release 3.4.5).
	<p style="text-align: center;">Zoom Server limitations</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
	Number of simultaneous calls is limited to 4 per user (i.e., a user with more than one device is still limited to 4 simultaneous calls).
IPPUC-6802	The call list in the detailed call list screen disappears after a long call (about 40 minutes).
IPPUC-6814	Pick Up Call fails when IP Phone device is already managing 3 calls.
IPPUC-6804	Monitoring a call duration will become incorrect after the monitoring device performs a registration (due to restart or restore defaults). The call duration will display the time starting from the registration (ignoring the time prior to registration).
	<p style="text-align: center;">General Zoom server limitation</p>
IPPUC-6764	Username with more than 16 characters will be cut in relevant calls/call log screens.
	If user A is set as “delegated” by user B, and User B set call forwarding: incoming calls are not forwarded (i.e., User A and User B phones will ring).
	<p style="text-align: center;">General limitation</p>
IPPUC-6519	RTCP XR / RX50: Signal/Noise level and RERL (Residual Echo Return Loss) are not supported.
IPPUC-6876	<p>“New Device Unlock Code” screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure:</p> <ol style="list-style-type: none"> 1. Upgrade to 3.4.6 2. Restore to default 3. Downgrade to 3.4.4 or 3.4.5 4. Upgrade again to 3.4.6
	Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.
	OVOC / Device Manager limitations
IPPUC-6644	No “IP Phone Speaker Firmware Download Failure” alert send out after download firmware fail.
IPPUC-6404	Change provisioning method STATIC->DYNAMIC via Web-UI triggers cached provisioning cycle.
IPPUC-2479	[RX50] does not support local 3-way conference calls.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2575	The user’s status indication (available / in a call) is occasionally displayed as ‘?’ in the Call Log.
-	The special characters öß\$üä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3794	<p>In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter ‘voip/media/allow_multiple_rtp’.</p>
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).

Incident	Description
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.13.3 Resolved Constraints in Version 3.4.6.629

The table below shows the resolved constraints from previous versions.

Table 3-23: Resolved Constraints

Incident	Description
	[445HD, C450HD] Wi-Fi Driver update and bug fixes.
	SIP proxy redundancy improvements and bug fixes.
IPPUC-7194	Call Monitoring: 'Pick up' softkey should be 'Take Over'.
IPPUC-7193	Call Monitoring: The username in call list screen is incorrect.
IPPUC-7181	IP Phone does not display caller details on incoming call screen.
IPPUC-7246	Occasionally static or white noise persistent at the local side during silence from remote side
IPPUC-7220	C450HD – The maximum speaker volume is too low.
PPUC-2511	Changing line programmable key into a speed-dial key, 'little rectangle' displayed.
IPPUC-5449	Wrong Jitter calculation in QOE report.
IPPUC-5525	[445HD] loud noise occurs when adding REQUIRE ENCRYPTION phone to conference.
IPPUC-3781	[RX50] Missing Long Hold CWRR tone.
IPPUC-5804	Mismatch between paging soft key with 1st paging group.
IPPUC-5856	After ending two calls, phone stays in 'Busy' status in idle screen.
IGS-3011	Call disconnects when the re-INVITE includes both RTP and SRTP lines.
IPPUC-5984	3CX server: DTMF tones are not responding to IVR menu when phone using OPUS codec.
IPPUC-3931	On rare occasions, call/conference dropped after 30 min (phone does not refresh the session).
IPPUC-5732	USB headset Jabra Evolve's 20 key may require several presses to connect/disconnect.
IPPUC-6050	[450HD] There is noise and long voice delay in 3WC.
IPPUC-6180	BSFT: when retrieving a call from CallPark, retrieval code is displayed on the screen.
IPPUC-6281	'ABC' button is not translated in Hebrew keypad.
IGS-3187	Latvian translation for color LCD models.
IPPUC-6723	Functional keys on expansion module are displayed incorrectly when using Hebrew.
IPPUC-6820	In some environments the Re-INVITE is sent with the AVP instead of the AVPF (SRTP).
IPPUC-6749	Phone cannot register when SIP password is empty.

Incident	Description
IPPUC-6840	[RX50] soft key `Call` is not translated when using another language.
IPPUC-6688	Automatic Dialing does not work.

3.14 Version 3.4.6.583



Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.

3.14.1 What's New

Following are the new features released in Version 3.4.6.583:

- High-end model now supports interoperability with BroadSoft application servers.
- [C450HD and 445HD]: Wideband audio for Bluetooth headsets has become GA in this release.

3.14.2 Known Constraints in Version 3.4.6.583

The table below shows the constraints that are known to exist in this version.

Table 3-24: Known Constraints from This Release

Incident	Description
	BSFT limitations
IPPUC-7280	[BSFT] Call Forward cannot be configured for a shared line.
IPPUC-7288	[BSFT] If phone A is set to call forward all calls to phone B, a call to phone A (that rings on phone B) will not indicate on phone B that it is a call for phone A.
IPPUC-7364	[BSFT] A remote conference participant putting that call on hold may result in the entire conference hearing Music on Hold from the BroadSoft server.
IPPUC-7380	Outbound proxy failover is not supported.
	[BSFT] BroadSoft server supports up to 29 Busy Lamp Fields (BLFs).

Table 3-25: Known Constraints from Previous Releases

Incident	Description
	<p style="text-align: center;">IP Phone limitation</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
IPPUC-6854	The maximum calls that can be displayed in “shared line call list” is 11
IPPUC-6912	Shared line top bar does not appear after downgrading from release 3.4.6 to release 3.4.5 (Perform a “Restore to Default” to restore top bar in release 3.4.5).
	<p style="text-align: center;">Zoom Server limitations</p> <p>Zoom: Supports multiple shared lines (to configure up to 15 shared lines)</p>
	Number of simultaneous calls is limited to 4 per user (i.e., a user with more than one device is still limited to 4 simultaneous calls).
IPPUC-6802	The call list in the detailed call list screen disappears after a long call (about 40 minutes).
IPPUC-6814	Pick Up Call fails when IP Phone device is already managing 3 calls.
IPPUC-6804	Monitoring a call duration will become incorrect after the monitoring device performs a registration (due to restart or restore defaults). The call duration will display the time starting from the registration (ignoring the time prior to registration).
	<p style="text-align: center;">General Zoom server limitation</p>
IPPUC-6764	Username with more than 16 characters will be cut in relevant calls/call log screens.
	If user A is set as “delegated” by user B, and User B set call forwarding: incoming calls are not forwarded (i.e., User A and User B phones will ring).
	<p style="text-align: center;">General limitation</p>
IPPUC-6519	RTCP XR / RX50: Signal/Noise level and RERL (Residual Echo Return Loss) are not supported.
IPPUC-6876	<p>“New Device Unlock Code” screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure:</p> <ol style="list-style-type: none"> 1. Upgrade to 3.4.6 2. Restore to default 3. Downgrade to 3.4.4 or 3.4.5 4. Upgrade again to 3.4.6
	Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.
	OVOC / Device Manager limitations
IPPUC-6644	No “IP Phone Speaker Firmware Download Failure” alert send out after download firmware fail.
IPPUC-6404	Change provisioning method STATIC->DYNAMIC via Web-UI triggers cached provisioning cycle.
IPPUC-2479	[RX50] does not support local 3-way conference calls.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2575	The user’s status indication (available / in a call) is occasionally displayed as ‘?’ in the Call Log.
-	The special characters öß\$üä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3794	<p>In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter ‘voip/media/allow_multiple_rtp’.</p>
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).

Incident	Description
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.14.3 Resolved Constraints in Version 3.4.6

The table below shows the resolved constraints in this version.

Table 3-26: Resolved Constraints

Incident	Description
	[445HD, C450HD] Wi-Fi Driver update and bug fixes.
	SIP proxy redundancy improvements and bug fixes.
IPPUC-7194	Call Monitoring: 'Pick up' softkey should be 'Take Over'.
IPPUC-7193	Call Monitoring: The username in call list screen is incorrect.
IPPUC-7181	IP Phone does not display caller details on incoming call screen.
IPPUC-7246	Occasionally static or white noise persistent at the local side during silence from remote side
IPPUC-7220	C450HD – The maximum speaker volume is too low.
PPUC-2511	Changing line programmable key into a speed-dial key, 'little rectangle' displayed.
IPPUC-5449	Wrong Jitter calculation in QOE report.
IPPUC-5525	[445HD] loud noise occurs when adding REQUIRE ENCRYPTION phone to conference.
IPPUC-3781	[RX50] Missing Long Hold CWRR tone.
IPPUC-5804	Mismatch between paging soft key with 1st paging group.
IPPUC-5856	After ending two calls, phone stays in 'Busy' status in idle screen.
IGS-3011	Call disconnects when the re-INVITE includes both RTP and SRTP lines.
IPPUC-5984	3CX server: DTMF tones are not responding to IVR menu when phone using OPUS codec.
IPPUC-3931	On rare occasions, call/conference dropped after 30 min (phone does not refresh the session).
IPPUC-5732	USB headset Jabra Evolve's 20 key may require several presses to connect/disconnect.
IPPUC-6050	[450HD] There is noise and long voice delay in 3WC.
IPPUC-6180	BSFT: when retrieving a call from CallPark, retrieval code is displayed on the screen.
IPPUC-6281	'ABC' button is not translated in Hebrew keypad.
IGS-3187	Latvian translation for color LCD models.
IPPUC-6723	Functional keys on expansion module are displayed incorrectly when using Hebrew.
IPPUC-6820	In some environments the Re-INVITE is sent with the AVP instead of the AVPF (SRTP).
IPPUC-6749	Phone cannot register when SIP password is empty.

Incident	Description
IPPUC-6840	[RX50] soft key `Call` is not translated when using another language.
IPPUC-6688	Automatic Dialing does not work.

3.15 Version 3.4.6.565



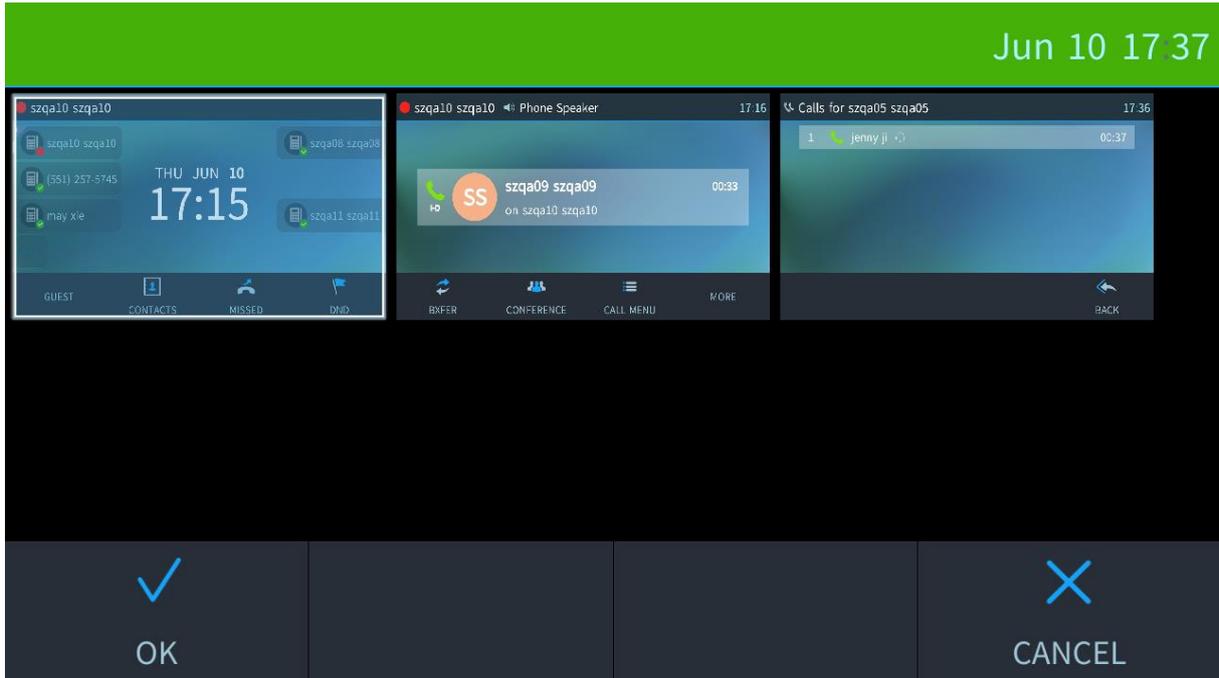
Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.

3.15.1 What's New

Following are the new features released in Version 3.4.6.565:

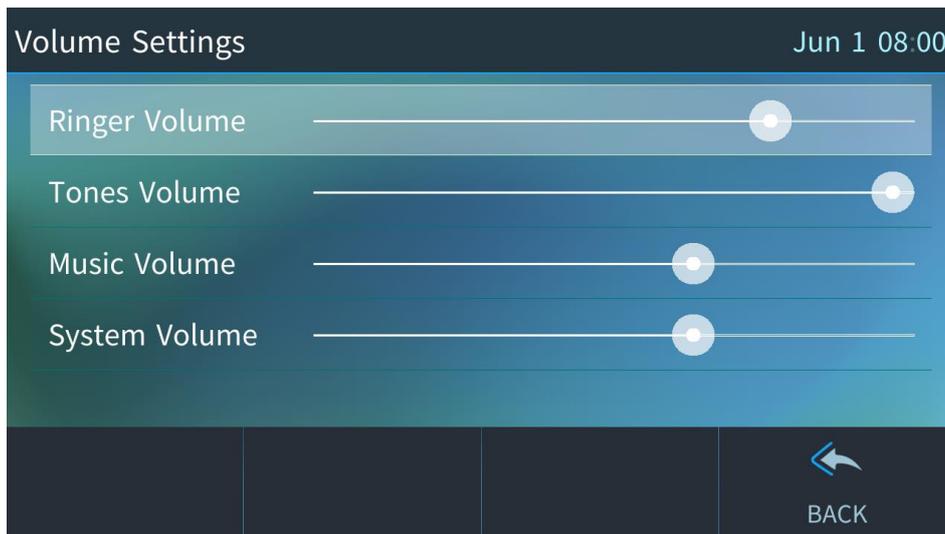
- AES256 is now supported (only applicable for 445HD, C450HD, and RX50).
The configuration file parameter 'voip/media/srtp/method' supports new values:
 - AES_256_CM_HMAC_SHA1_80
 - AES_256_CM_HMAC_SHA1_32
 - AES_256_CM_ALL_METHODS
- AudioCodes Device Manager security improvement
 - The device validates the AudioCodes Device Manager identity using known root CA.
 - The device is shipped with known Root CAs installed.
 - For the initial connection phase, the AudioCodes Device Manager should access the device using a known CA.
 - Once a successful secured connection has been established between the device and the Device Manager, the user can replace the root CA on the Device Manager and on the device. This will re-establish the connection leveraging any private root CA.
- RTCP XR related data is now supported in the QoE report for the RX50 model.
- [C450HD and 445HD]: Improved background noise block:
When a user speaks on the phone in handset mode, ambient noise (noise from the device's immediate environment) is suppressed and not heard on the far end participant/s of the call.
- RX50 - Added support for returning to factory default settings.
 1. Power up by unplugging the network (POE) cable & plugging it back in. Immediately press hard keys Mute+VolumeDown for about 10 seconds (until the LED is yellow-green or the AudioCodes logo appears).
 2. Follow Restore UI procedure.
- The Device Security feature (configured by the administrator) allows you to Lock/Unlock your phone using a PIN code.
- Screens Navigation – screen switcher
Screen switcher lists all open screens and allows switching to one of them. Windows selection screen is initiated by a long press in the Menu Hard Key or Soft Key for RX50. It can also be initiated by pressing "X" Hard Key on Idle screen.

Figure 3-1: Screen Navigation Switcher



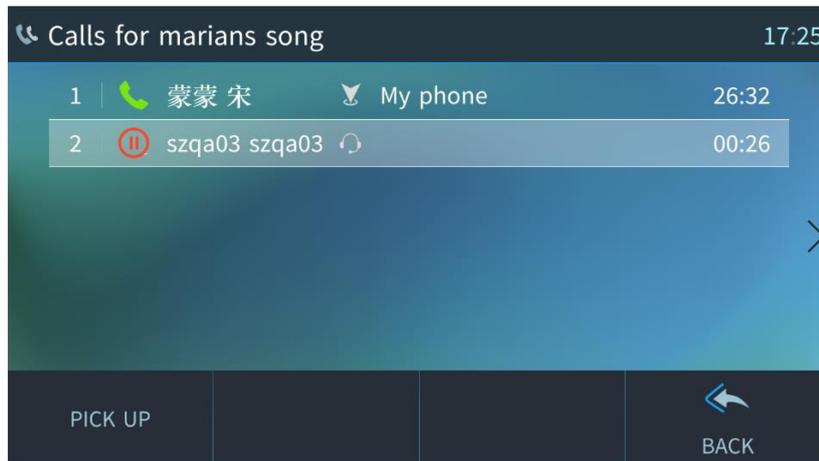
- Volume Mixer Control of multiple streams
Ability to change the volume of different phone streams (e.g., Ringer, Tones, Music, etc.)

Figure 3-2: Volume Mixer Control of multiple streams



You can pick up a call on hold (indicated by ) by pressing the “PICK UP” softkey.

Figure 3-5: Detailed Call List Screen



- Call monitoring (listen, whisper, barge, take over).
Phone users can use these call monitoring features to join or listen to calls:
 - ◆ **Listen:** Listen to a call without the parties being aware.
 - ◆ **Whisper:** Speak to a phone user in a call without other parties being aware.
 - ◆ **Barge:** Join a call and speak to all parties. The call will become a 3-way call.
 - ◆ **Take over:** Take over the call from a Zoom Phone user.

The following scenarios illustrate possible use cases:

- ◆ **Front desk receptionist:** If there is a high-priority call for an employee who is busy on another call, the receptionist can barge in and inform the employee of the important call.
- ◆ **Call center supervisor:** Supervisors can silently listen to an agent's call with a customer to ensure high-quality service, then barge in to calls to assist.
- ◆ **Group trainer:** A group trainer can listen to an agent's call with a customer then whisper to the agent to provide advice on handling the customer.

This call monitoring feature (configured by the administrator) must be enabled in the Zoom Web portal.

After enabling call monitoring, you need to create the following group types of phone users (to use this feature):

- ◆ **Monitors:** Phone users that can use call monitoring on other phone users.
- ◆ **Monitored:** Phone users that can be monitored.

References: <https://support.zoom.us/hc/en-us/articles/360044804711>.

- Supports Zoom 911 Emergency Services.
- Supports Zoom Transfer to Voice Mail.
- Supports Zoom Transfer to Call Parking.
- Support Zoom Call forwarding.
- Supports Zoom Group call pickup
 - ◆ Allows administrators to create a group and make all members a backup for each other.
 - ◆ When one member receives an incoming call, other members are notified and allowed to pick up the call.

References: <https://support.zoom.us/hc/en-us/articles/360060107472-Setting-up-and-using-group-call-pickup>.

3.15.2 Known Constraints in Version 3.4.6

The table below shows the constraints that are known to exist in this version.

Table 3-27: Known Constraints from This Release

Incident	Description
	IP Phone limitation
	Zoom: Supports multiple shared lines (to configure up to 15 shared lines)
IPPUC-6854	The maximum calls that can be displayed in “shared line call list” is 11
IPPUC-6912	Shared line top bar does not appear after downgrading from release 3.4.6 to release 3.4.5 (Perform a “Restore to Default” to restore top bar in release 3.4.5).
	Zoom Server limitations
	Zoom: Supports multiple shared lines (to configure up to 15 shared lines)
	Number of simultaneous calls is limited to 4 per user (i.e., a user with more than one device is still limited to 4 simultaneous calls).
IPPUC-6802	The call list in the detailed call list screen disappears after a long call (about 40 minutes).
IPPUC-6814	Pick Up Call fails when IP Phone device is already managing 3 calls.
IPPUC-6804	Monitoring a call duration will become incorrect after the monitoring device performs a registration (due to restart or restore defaults). The call duration will display the time starting from the registration (ignoring the time prior to registration).
	General Zoom server limitation
IPPUC-6764	Username with more than 16 characters will be cut in relevant calls/call log screens.
	If user A is set as “delegated” by user B, and User B set call forwarding: incoming calls are not forwarded (i.e., User A and User B phones will ring).
	General limitation
IPPUC-6519	RTCP XR / RX50: Signal/Noise level and RERL (Residual Echo Return Loss) are not supported.
IPPUC-6876	<p>"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure:</p> <ol style="list-style-type: none"> 1. Upgrade to 3.4.6 2. Restore to default 3. Downgrade to 3.4.4 or 3.4.5 4. Upgrade again to 3.4.6
	Roll backs from Versions 3.4.6.xxx and above to versions 3.4.2 or lower must be done via 3.4.4.xxx first.
	OVOC / Device Manager limitations
IPPUC-6644	No "IP Phone Speaker Firmware Download Failure" alert send out after download firmware fail.
IPPUC-6404	Change provisioning method STATIC->DYNAMIC via Web-UI triggers cached provisioning cycle.

Table 3-28: Known Constraints from Previous Releases

Incident	Description
IPPUC-2479	[RX50] does not support local 3-way conference calls.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
-	The special characters öß\$üä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.15.3 Resolved Constraints in Version 3.4.6

The table below shows the resolved constraints in this version.

Table 3-29: Resolved Constraints

Incident	Description
	[445HD, C450HD] Wi-Fi Driver update and bug fixes.
	SIP proxy redundancy improvements and bug fixes.
IPPUC-7194	Call Monitoring: 'Pick up' softkey should be 'Take Over'.
IPPUC-7193	Call Monitoring: The username in call list screen is incorrect.
IPPUC-7181	IP Phone does not display caller details on incoming call screen.
IPPUC-7246	Occasionally static or white noise persistent at the local side during silence from remote side
IPPUC-7220	C450HD – The maximum speaker volume is too low.
PPUC-2511	Changing line programmable key into a speed-dial key, 'little rectangle' displayed.
IPPUC-5449	Wrong Jitter calculation in QOE report.
IPPUC-5525	[445HD] loud noise occurs when adding REQUIRE ENCRYPTION phone to conference.
IPPUC-3781	[RX50] Missing Long Hold CWRR tone.
IPPUC-5804	Mismatch between paging soft key with 1st paging group.
IPPUC-5856	After ending two calls, phone stays in 'Busy' status in idle screen.
IGS-3011	Call disconnects when the re-INVITE includes both RTP and SRTP lines.

Incident	Description
IPPUC-5984	3CX server: DTMF tones are not responding to IVR menu when phone using OPUS codec.
IPPUC-3931	On rare occasions, call/conference dropped after 30 min (phone does not refresh the session).
IPPUC-5732	USB headset Jabra Evolve's 20 key may require several presses to connect/disconnect.
IPPUC-6050	[450HD] There is noise and long voice delay in 3WC.
IPPUC-6180	BSFT: when retrieving a call from CallPark, retrieval code is displayed on the screen.
IPPUC-6281	'ABC' button is not translated in Hebrew keypad.
IGS-3187	Latvian translation for color LCD models.
IPPUC-6723	Functional keys on expansion module are displayed incorrectly when using Hebrew.
IPPUC-6820	In some environments the Re-INVITE is sent with the AVP instead of the AVPF (SRTP).
IPPUC-6749	Phone cannot register when SIP password is empty.
IPPUC-6840	[RX50] soft key `Call` is not translated when using another language.
IPPUC-6688	Automatic Dialing does not work.

3.16 Version 3.4.5.8

3.16.1 What's New



- Version 3.4.5 includes firmware build **3.4.5.8**
- This version release is for a Zoom environment only

Following are the new features released in Version 3.4.5:

- **AudioCodes' Zoom Phone users can now use Hot Desking.**
- See also <https://support.zoom.us/hc/en-us/articles/360043841032-Using-hot-desking-for-phones>
- **The AudioCodes Zoom Phone now supports 'Contact directory' (LDAP).**

3.16.2 Known Constraints in Version 3.4.5

The table below shows the constraints that are known to exist in this version.

Table 3-30: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone, RX50 and HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3829	[Multicast Paging] A noise may be heard during a paging call if the phone is already occupied with more than three calls in parallel.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
IPPUC-3701	The Call Log is erased if the phone reloads the VoIP application. The VoIP application may be reloaded when some configuration parameters are changed.
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.
IPPUC-5890	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] The 'Update keys' setting on the Zoom server and some key labels are incorrect on the device and on the server.
IPPUC-6061	The speed dial BLF sometimes does not function.
IPPUC-5999	[LDAP contact] Sometimes, the phone screen does not display 'Username' in the calling screen.
IPPUC-6050	[450HD] There is a noise and a long voice delay in 3WC.
IPPUC-6139	[Zoom Server issue expected to be fixed in one of the next Zoom server releases] A Shared Line display a call as with Admin when answering an incoming call from Boss

3.16.3 Resolved Constraints in Version 3.4.5

The table below shows the resolved constraints in this version.

Table 3-31: Resolved Constraints

Incident	Description
IPPUC-4547	Improvements are required for OPUS voice quality.
IPPUC-4462, IPPUC-3936	The phone does not follow the HTTP response `301/302 Moved Permanently` from a provisioning server and as a result, does not start the provisioning process.
IPPUC-5238	Long-pressing the Clear softkey does not function in the 'Static IP' screen.
IPPUC-4442	[Wi-Fi] On some occasions, it is necessary to refresh the access points list to see the available access points.
IPPUC-1652	[Wi-Fi] Users cannot disable/enable the Wi-Fi network from the Web interface.
IPPSFB- 10024	[Wi-Fi] The Wi-Fi Protected Setup (WPS) PIN length is too long.
IPPUC-3866	[RX50, C450HD, 450HD] if one of the phone's softkeys is pressed for a prolonged period of time, the softkey remains bold.
IPPUC-3171	The uppermost part of the Contacts menu list is sometimes displayed eclipsing the status bar.
IPPUC-3254	[Virtual Keyboard] A prolonged press on the 'backspace' to partially delete a string from the middle to the start of the string, deletes the whole string.
IPPUC-1628	[Call log] The phone wrongly merges two discontinuous calls into one call.
IPPUC-2145	DTMF tones are sometimes not heard when using DTMF Transport mode 'Via SIP'.
IPPUC-3584	[SRTP] A conference call that involves parties that are configured with 'SUPPORT ENCRYPTION' and 'DO NOT SUPPORT ENCRYPTION' may result in no voice or in noises during the call.
IPPUC-5378	When two calls are in parallel and then ended and the user goes back to idle screen, the phone still appears as 'Busy'; the status displayed in the idle screen is incorrect.
IPPUC-5345	[445HD OPUS] It is not recommended to use a local 3-way conference with the OPUS vocoder. The issue will be fixed in the next version release.

3.17 Version 3.4.4.1000.52

3.17.1 What's New



Version 3.4.4 includes firmware build **3.4.4.1000.52**.

Following are the new features released in Version 3.4.4:

- **C450HD phone voice quality in handsfree mode has been improved.** A new equalizer was designed to improve the quality and the gain from the C450HD phone's microphone was increased.

3.17.2 Known Constraints in Version 3.4.4

The table below shows the constraints that are known to exist in this version.

Table 3-32: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone, RX50 and HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted per the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößšüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3829	[Multicast Paging] A noise may be heard during a paging call if the phone is already occupied with more than three calls in parallel.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and thus a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
IPPUC-3701	The Call Log is erased if the phone reloads the VoIP application. The VoIP application may be reloaded when some configuration parameters are changed.
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.

Incident	Description
IPPUC-5378	[445HD, C450HD] [Bluetooth] In some Bluetooth headset models (mainly in Plantronics headsets), a short annoying high-frequency tone is heard before the dial tone and sometimes even during the call.
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.

3.17.3 Resolved Constraints in Version 3.4.4

The table below shows the resolved constraints in this version.

Table 3-33: Resolved Constraints

Incident	Description
IPPUC-5345	[445HD OPUS] Noises are heard when the call is put on hold.
IPPUC-5752	[C450HD and C445HD] [Bluetooth] On rare occasions, no voice is heard when answering a call using the Bluetooth headset.
IPPUC-5943	[C450HD and C445HD] Bluetooth is not automatically paired after the phone is rebooted.
IPPUC-5345	[445HD OPUS] It is not recommended to use a local 3-way conference with the OPUS vocoder.

3.18 Version 3.4.4.1000.10

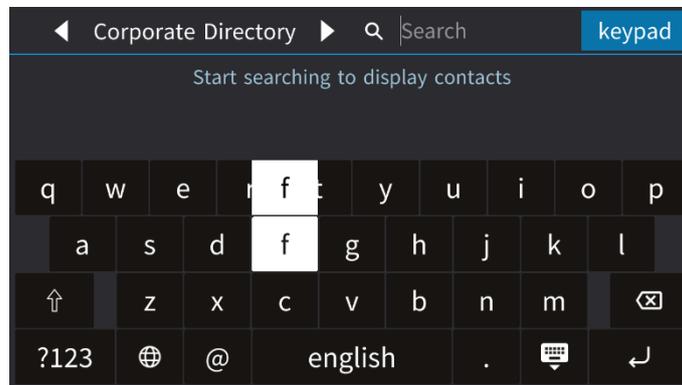
3.18.1 What's New



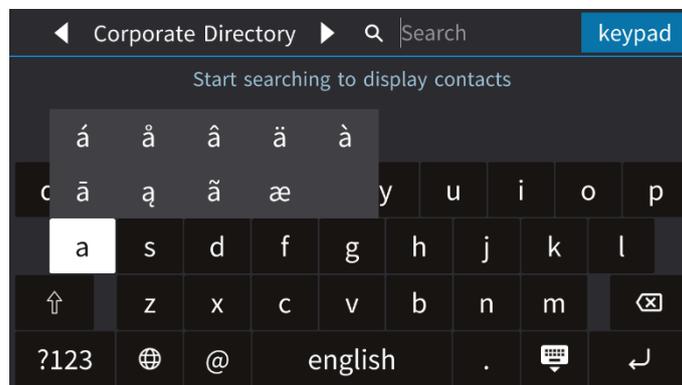
Version 3.4.4 includes firmware build **3.4.4.1000.10**.

Following are the new features released in Version 3.4.4:

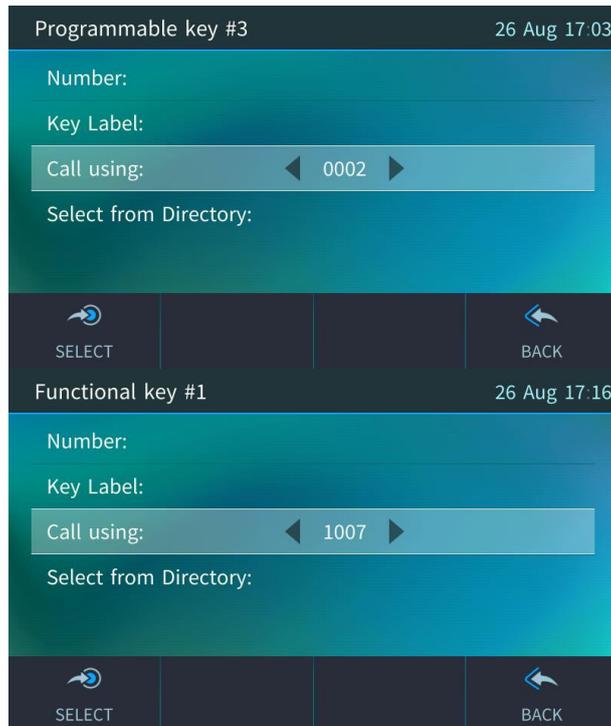
- **OPUS wideband is supported** (Applies to the 445HD and C450HD phone models).
- **Wideband audio for Bluetooth headsets** is supported as Beta. In previous releases, only narrow band audio was supported. Voice quality has consequently improved. (Applies to the 445HD and C450HD phone models).
- The phone's Expansion Module now supports **two pages and a total of 40 Functional Keys, each page displaying 20 Functional Keys (compared to 22 Functional Keys with only one page in previous releases)**. Users swipe right or left (depending on what page they are on) to transition from page to page. (Applies to the C450HD and 450HD phone models).
- **The Virtual Keyboard features new capabilities.**
 - When users press a key, the keyboard shows the key that was pressed in a popup above the key's physical location.



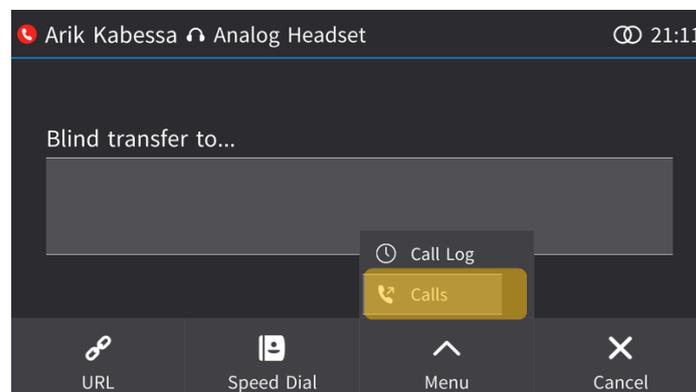
- A popup for special characters has been added for improved user experience.



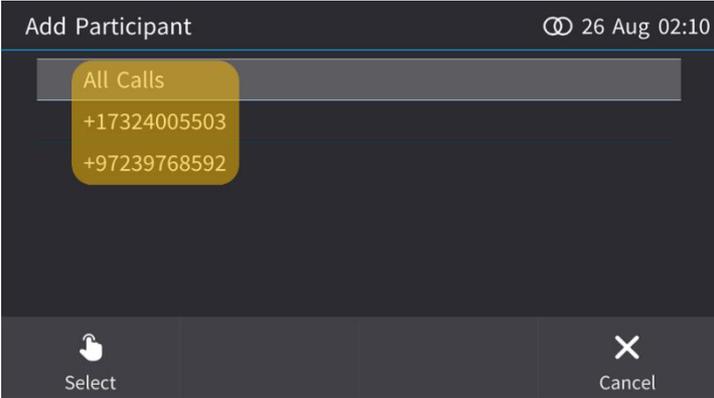
- Users can **configure a Programmable Key or Functional Key for speed dialing to be initiated by a speed dial calling line of their choice**. The feature only applies to phones configured with multiple lines. The feature determines through which line the call goes out when speed dialing. For example: A phone is configured with two lines, 0002 and 0003. When configuring a speed dial, 0002 or 0003 can be configured as the default line through which to (speed) dial out. For example, one line may be for internal calls, the other for external calls.



- The **RX50 conference phone supports Voice Quality reports** in compliance with the IETF's RFC 6035, except for the following VQ local metrics that are not provided:
 - Jitter buffer statistics
 - Burst and Gap loss
 - Signal and noise levels
 - Voice Quality estimation
- **Transfer a call to a remote party with whom a call was previously established**
 Allows a user to transfer a call to another person with whom a call was established, by selecting the call from a list of calls. The user can consult with any person with whom a call was established without needing to use the Consultation Transfer procedure. The feature is activated through the **Xfer** softkey. Users then press the **Menu** softkey, select **Calls** and then select the active call. The call is then transferred to the remote party with whom that call was established.



- **Merge a call with an existing (active) call / multiple calls / all calls** or with an existing (active) remote conference. The feature is activated through the **Call Menu** softkey. The user then selects **Merge calls** and then selects the preferred option.



3.18.2 Known Constraints in Version 3.4.4

The table below shows the constraints that are known to exist in this version.

Table 3-34: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone, RX50 and HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößšüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3829	[Multicast Paging] A noise may be heard during a paging call if the phone is already occupied with more than three calls in parallel.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
IPPUC-3701	The Call Log is erased if the phone reloads the VoIP application. The VoIP application may be reloaded when some configuration parameters are changed.
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.
IPPUC-5345	[445HD OPUS] It is not recommended to use a local 3-way conference with the OPUS vocoder. The issue will be fixed in the next version release.
IPPUC-5378	[445HD, C450HD] [Bluetooth] In some Bluetooth headset models (mainly in Plantronics headsets), a short annoying high-frequency tone is heard before the dial tone and sometimes even during the call.
-	[Applies only to 445HD / C450HD] An endless auto-negotiation with the Ethernet switch may occur if the phone is connected to the Power Supply and to PoE simultaneously. It is recommended to avoid connecting them simultaneously even though no damage occurs. If a Power Supply is used, users should disable the power from the ETH port.

3.18.3 Resolved Constraints in Version 3.4.4

The table below shows the resolved constraints in this version.

Table 3-35: Resolved Constraints

Incident	Description
IPPUC-4547	Improvements are required for OPUS voice quality.
IPPUC-4462, IPPUC-3936	The phone does not follow the HTTP response `301/302 Moved Permanently` from a provisioning server and as a result, does not start the provisioning process.
IPPUC-5238	Long-pressing the Clear softkey does not function in the 'Static IP' screen.
IPPUC-4442	[Wi-Fi] On some occasions, it is necessary to refresh the access points list to see the available access points.
IPPUC-1652	[Wi-Fi] Users cannot disable/enable the Wi-Fi network from the Web interface.
IPPSFB- 10024	[Wi-Fi] The Wi-Fi Protected Setup (WPS) PIN length is too long.
IPPUC-3866	[RX50, C450HD, 450HD] if one of the phone's softkeys is pressed for a prolonged period of time, the softkey remains bold.
IPPUC-3171	The uppermost part of the Contacts menu list is sometimes displayed eclipsing the status bar.
IPPUC-3254	[Virtual Keyboard] A prolonged press on the 'backspace' to partially delete a string from the middle to the start of the string, deletes the whole string.
IPPUC-1628	[Call log] The phone wrongly merges two discontinuous calls into one call.
IPPUC-2145	DTMF tones are sometimes not heard when using DTMF Transport mode 'Via SIP'.
IPPUC-3584	[SRTP] A conference call that involves parties that are configured with 'SUPPORT ENCRYPTION' and 'DO NOT SUPPORT ENCRYPTION' may result in no voice or in noises during the call.

3.19 Version 3.4.3.19.65



Version 3.4.3 includes firmware build **3.4.3.19.65**.

There are no new features in this release.

3.19.1 Known Constraints in Version 3.4.3

The table below shows the constraints that are known to exist in this version.

Table 3-36: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone, RX50 and HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößšüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3829	[Multicast Paging] A noise may be heard during a paging call if the phone is already occupied with more than three calls in parallel.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
IPPUC-3701	The Call Log is erased if the phone reloads the VoIP application. The VoIP application may be reloaded when some configuration parameters are changed.
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.

3.19.2 Resolved Constraints in Version 3.4.3

The table below shows the resolved constraints in this version.

Table 3-37: Resolved Constraints

Incident	Description
IPPUC-2514	Shared line. After a forwarded call is answered by another user, the LED still flashes green.
IPPUC-1395	[Wi-Fi] The MAC address field in the Device Manager keep-alive should be AudioCodes' MAC address.
IPPUC-4567 IPPUC-4600	The input gain (the gain towards the remote direction) is too low.
IPPUC-4413	[RX50] Incorrect DTMF generation during an active call prevents using the IVR.
IPPUC-4075	[802.1X] tls_identity should be used when processing TLS authentication, not eap_identity.
IPPUC-3943	[RX50] Silence compression is not optimized with the OPUS vocoder.
IPPUC-4462	The phone does not follow "301 Moved Permanently" from the provisioning server.
IPPUC-4447	[RX50 with the OPUS vocoder] A call between the RX50 conference phone and a Zoom client results in one-way voice.
IPPUC-4433	[RX50 with the OPUS vocoder] A dial-in conference call or a call between two phones registered to a Zoom environment results in poor voice quality.
IPPUC-4568	[Zoom] Zero Touch provisioning fails when the phone is connected via Wi-Fi.
IPPUC-4015	The phone crashes when registering multiple lines.

3.20 Version 3.4.3.18.70



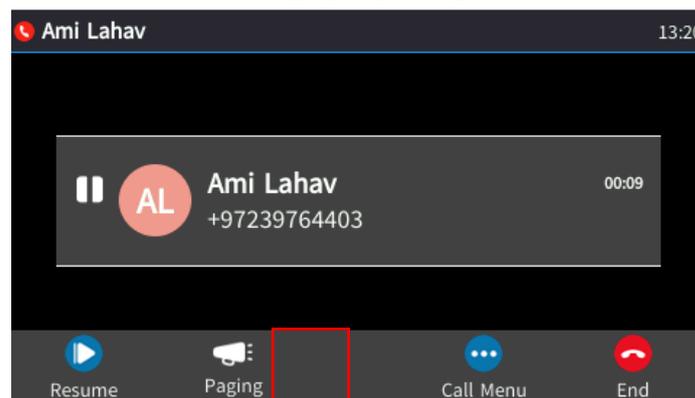
Version 3.4.3 includes firmware build **3.4.3.18.70**.

- **Paging can be performed during call hold and during an ongoing call.** To use this new feature during an **ongoing call**, users need to preconfigure a new softkey in the ongoing call screen.

Users can (for example) preconfigure the 'personal_settings/soft_keys/ongoing_call/0/key_function' configuration file parameter to PAGING (the default is BLIND_TRANSFER).

In addition, after configuring Paging (via VoIP Services), users will view a new key in the phone's Hold screen (i.e., in the screen displayed when the user holds a call).

New 'Paging' key during hold



- **Virtual QWERTY keyboard** (applies to the 450HD and C450HD phones, the HRS and the RX50 Conference Phone). The feature allows users to easily and effortlessly enter strings into fields. Users will mainly use the virtual keyboard when:

Searching for contacts



URL dialing

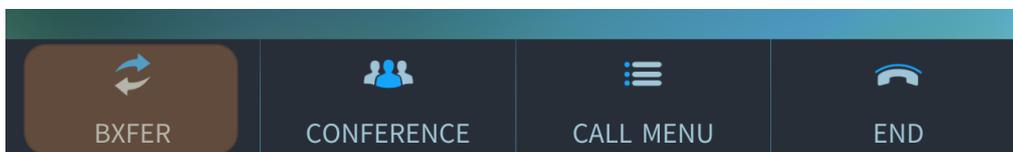


Signing in

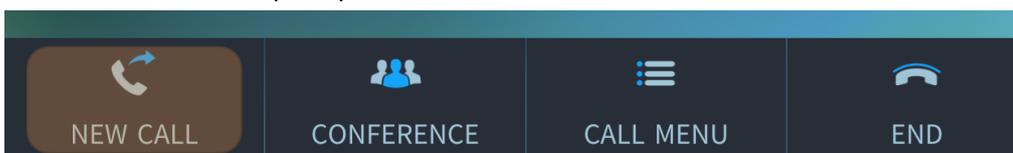


The virtual keyboard is currently supported in all languages except Chinese and Japanese (for these, the keypad will be in English). More languages will be supported in next version releases.

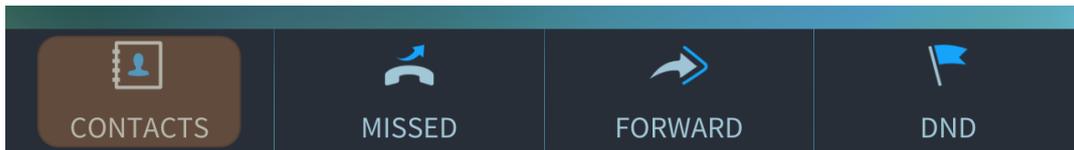
- **Customized UI experience.** Users can configure **Programmable Softkeys** for **New Call** state, **Ongoing call** state and **Idle** screen state as part of the phone's capability of allowing a customized user interface experience. See the pending *Administrator's Manual Version 3.4.2* for more information about how to customize the Programmable Softkeys.
 - **Configurable ongoing call screen.**
- Administrators can customize the ongoing call screen (shown in the figure below) in line with the preferences / requirements of enterprise management and / or the employees.



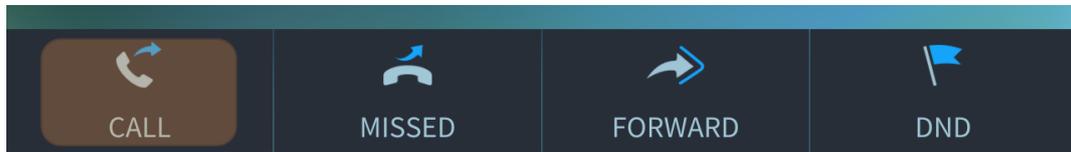
For example, the **BXfer** softkey in the ongoing call screen shown in the preceding figure can be replaced with the **New Call** softkey shown in the figure below on the phones of enterprise users who infrequently transfer calls.



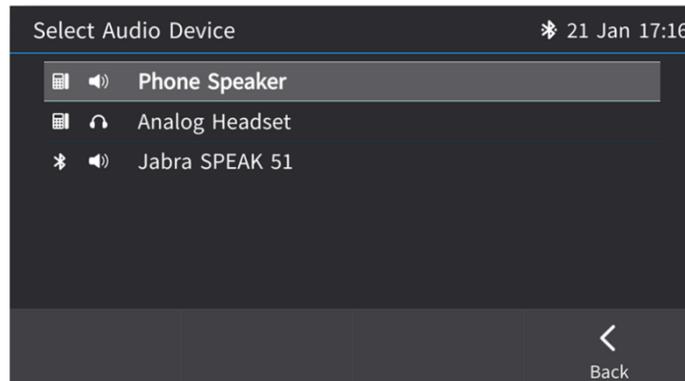
- **Configurable idle screen**
- Administrators can customize the idle screen (shown in the figure below) in line with the preferences / requirements of enterprise management and / or the employees.



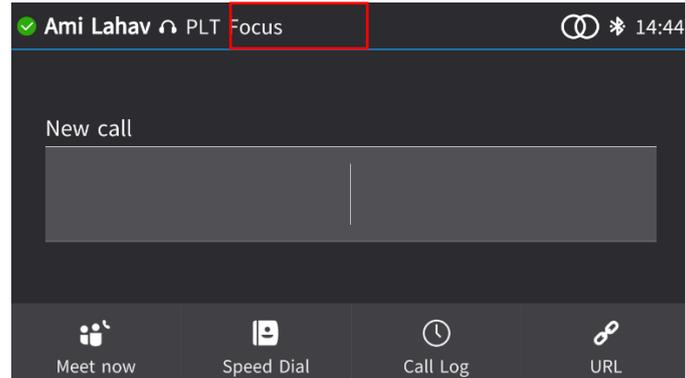
For example, the **Contacts** softkey in the idle screen shown in the preceding figure can be replaced with the **Call** softkey shown in the figure below.



- **Audio devices:** Users can switch between any available audio device either by pressing the headset / speaker key or by long-pressing the speaker / headset key and then if there are more devices, selecting the device from the list.



The device indicates the selected audio device in the screen title.



- **Multicast SUBSCRIBE provisioning method**

If provisioning information is not provided by a DHCP server (e.g., Options 66/67/160), the IP phone sends a sip SUBSCRIBE multicast message to the multicast address 224.0.1.75:5060 announcing the phone's presence within the network. An IP PBX, if it supports this provisioning method, will respond with a SIP NOTIFY message containing a provisioning server HTTP link. The phone will use this HTTP link to download a configuration file.

3.20.1 Known Constraints in Version 3.4.3

The table below shows the constraints that are known to exist in this version.

Table 3-38: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone, RX50 and HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2514	[Shared Line Appearance] In a scenario in which a forwarded call is answered by another phone that shared the same line, the local shared line LED still flashes green and does not indicate that the call was accepted by another phone.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.
IPPUC-3829	[Multicast Paging] A noise may be heard during a paging call if the phone is already occupied with more than three calls in parallel.
IPPUC-3794	In some environments, more than one RTP stream may be sent to the phone in the same call. In this case, it may handle the media incorrectly. To solve the issue, the network administrator can enable configuration file parameter 'voip/media/allow_multiple_rtp'.
IPPUC-3755	[RX50] Detection of Broken RTP is not supported and as a result, a call will not be closed due to broken RTP. The capability helps with disconnected calls (disrupted audio).
IPPUC-3701	The Call Log is erased if the phone reloads the VoIP application. The VoIP application may be reloaded when some configuration parameters are changed.
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.

3.20.2 Resolved Constraints in Version 3.4.3

The table below shows the resolved constraints in this version.

Table 3-39: Resolved Constraints

Incident	Description
IPPUC-2890	[RX50] The DSP version has been added to the Release Information screen on the phone.
IPPUC-3739	The phone does not renew the BLF Subscription after the BLF Subscription self-terminates.
IPPUC-3603	[C450HD/445HD] The ringer stopped functioning.
IPPUC-3707	In the Dutch language interface, the 'No matches' message is displayed only as an 'S'.
IPPUC-3395	The C450HD phone's Bluetooth feature sometimes enters an infinite 'Connecting' state.
IPPUC-2514	Shared line. After a forwarded call is answered by another user, the LED still flashes green.
IGS-2814	[QoE] A SIP PUBLISH is sent out in UDP protocol regardless of the SIP transport being used.
IPPUC-3129	The Clear softkey does not support long-press.
IPPUC-3556	With multiple lines, the line in a DND state becomes DND-deactivated if All extensions are selected for the parameter 'DND Activated'.
IPPUC-3555	[445HD] The phone displays the incorrect Paging title.
IPPUC-3353	The phone sends a corrupted keep-alive to the OVOC during the upgrade firmware phase. This leads to a duplicated registration of the same unit on the OVOC with incorrect parameters (occurs only on the Wi-Fi unit).
IPPUC-2941	[RX50] It is not possible to select calls from the Call List by touch.
IPPUC-3508	Parking Lot: There is no BLF monitoring when "park_prefix" and "retrieve_prefix" are empty.

3.21 Version 3.4.1.565

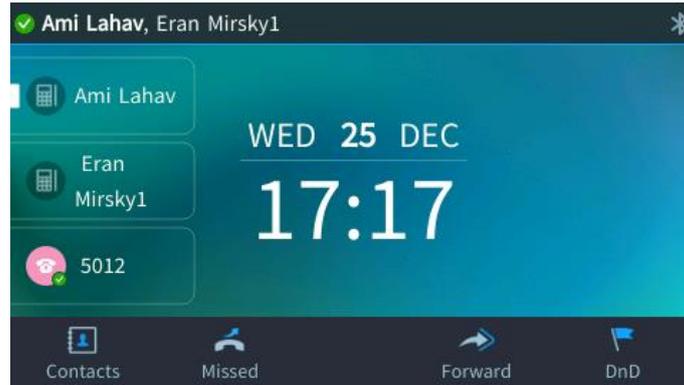
3.21.1 What's New



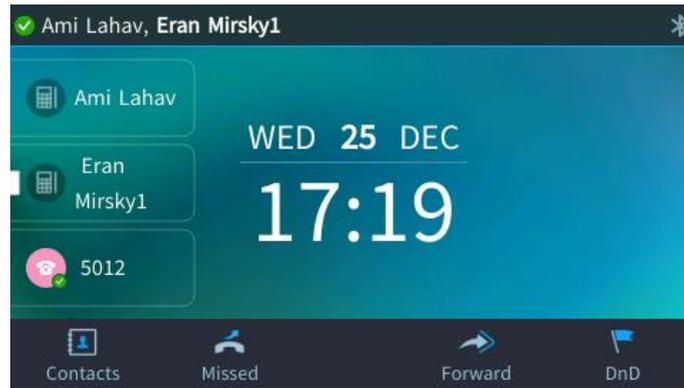
Version 3.4.1 includes firmware build **3.4.1.565**.

- **Support for AudioCodes' entire high end IP phone portfolio** including the 445HD, 450HD and C450HD phones and the HRS conference device. Support for the 450HD / C450HD phone models also includes support for the Expansion Module.
- **The expanded list of supported SIP environments is:**
 - Genband
 - Zoom
 - Asterisk
 - Freeswitch
- **Drop from 3-way conference.** The initiator of a call conference can drop out of the conference by on-hooking the phone, without disrupting parties B and C. Previously, only a softkey option was available for dropping out. For more information, see the *Administrator's Manual*, section 'Allowing the Initiator to Drop out of the Conference'.
- **Support for multiple lines.** Account status (default line, Call Forward and DND) is visually indicated for each account adjacent to the line's Programmable Key. To decide which line will be used:

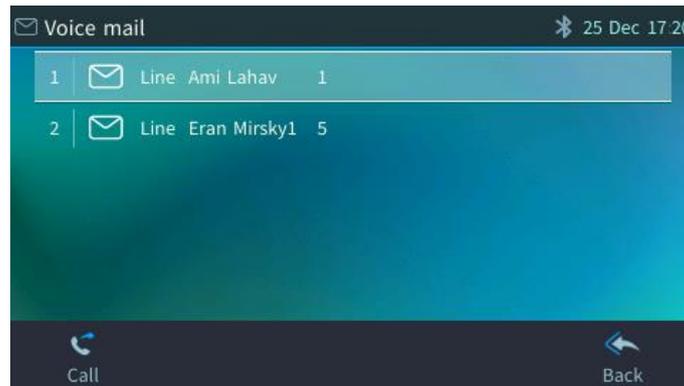
- Press the hard key next to the line.



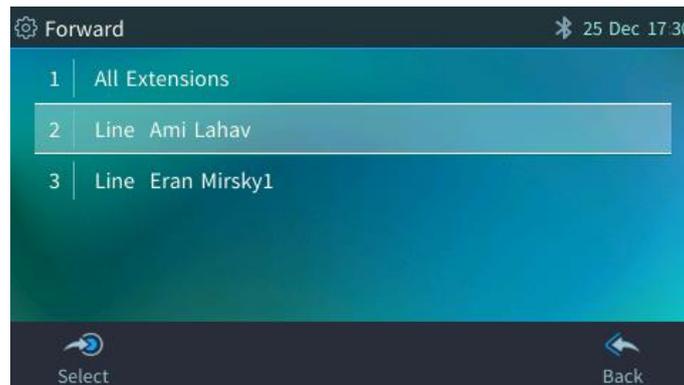
A rectangle icon indicates the current line:



- Press the voicemail hard key; the option to switch between lines is displayed.



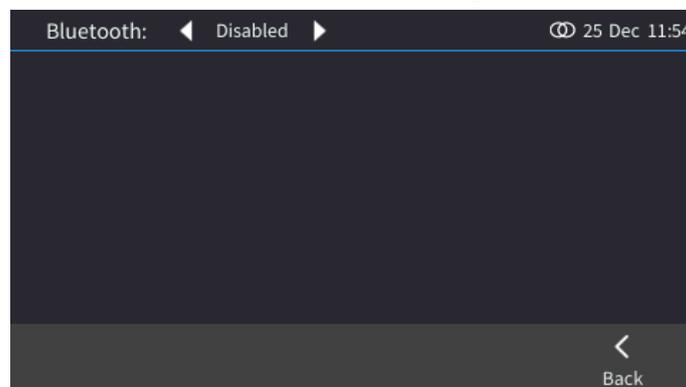
- The same applies to call Forward (see the figure below) and DND.



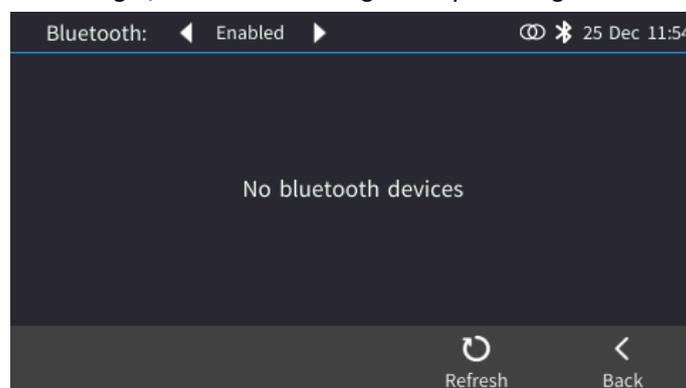
- **Shared Line Appearance (SLA) for Genband.** Enables a group of user phones to share a line and for users to make/receive calls that then *appear* to be made/ended to/from that line. The phones in the group behave as extensions of that line.
 - When a user uses a shared line, all phones in the group are notified.
 - Multiple simultaneous calls are supported.
 - User status (busy or idle, for example) is displayed on all phones in the group.
 - The group has a designated Primary SLA member and one or more Secondary SLA members.
 - Any restrictions and services on the Primary SLA member apply to all group members.
 - SLA can currently be configured using Single Call Arrangement (SCA): With this method, the number of calls that can be set up simultaneously across all user phones in the group is limited to one. When one of the user phones is active in a call (incoming or outgoing), the other phones are blocked from receiving or making additional calls.
 - If an incoming call to a shared line occurs while a call is already active on it, the call receives busy treatment.
- **Support for Bluetooth** (beta level) Integrated Bluetooth for wireless headset connectivity (applicable to the 445HD and C450HD phones as detailed above). Bluetooth is supported in specific regions such as the USA, Canada, the European Union, Switzerland, South Africa, and Israel, and requires a specific CPN with a 'BW' suffix when ordering. For an updated list of supported regions, contact AudioCodes.

To enable Bluetooth on the phone:

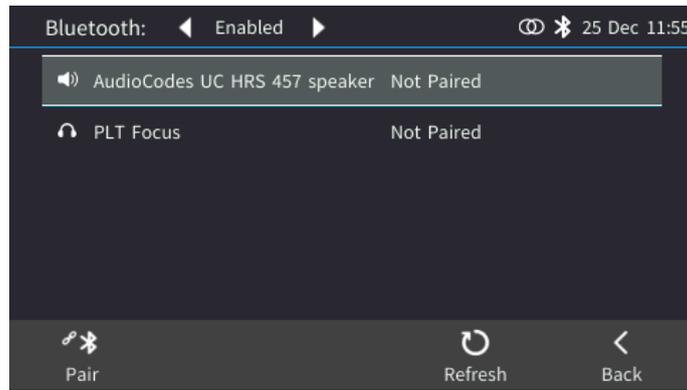
- Access the Bluetooth screen (**Menu > Settings**):



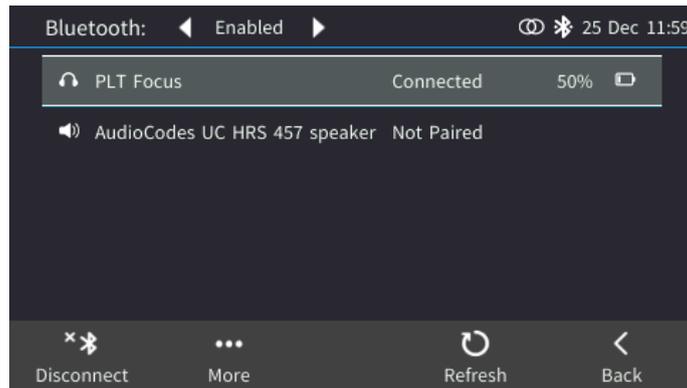
- Press the right/left rim of the navigator key to configure **Enabled**:



- Configure the device (Bluetooth headset or speaker) to allow pairing and then press the **Refresh** softkey; the phone attempts to discover available devices:



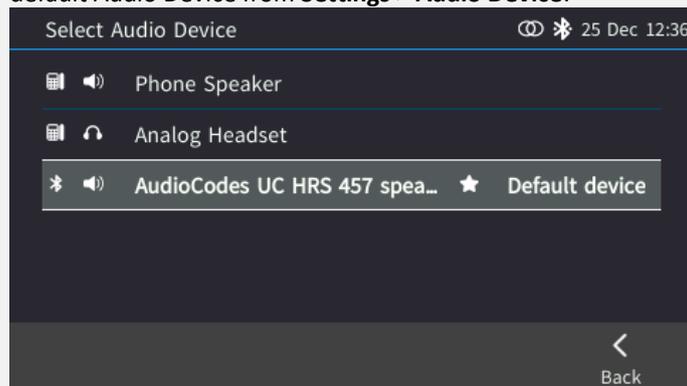
- Press the **Pair** softkey to pair the device. After pairing is complete, the phone displays 'Connected'.



- When the phone is connected via Bluetooth, its battery level is visually indicated adjacent to the 'Connected' indication. Bluetooth connectivity is indicated on the upper bar by the Bluetooth icon.
 - ◆  indicates Bluetooth is enabled, not paired
 - ◆  indicates that the device is connected
- Start using the device.



- All Bluetooth headsets are defined by the phone as headsets and the phone's headset hard key onhooks / offhooks the headset.
- Connecting both the USB headset and the Bluetooth headset is currently not recommended.
- Known speakers such as the HRS 457, Jabra 710 and Jabra 510 are not defined as Bluetooth headsets. Users can define a known Bluetooth speaker as the phone's default Audio Device from **Settings > Audio Device**:



- After it is selected, the Bluetooth speaker will be used whenever a call is answered via the **Accept** softkey or initiated via the **Dial** key. The phone's hard speaker and headset keys are used for its speaker and connected headset, for example, USB headset.

- **Up to 33 Speed Dials are supported on the 445HD sidecar.**
- **Programmable Softkeys can be configured for 'New Call' state.** In previous releases (including version 2.2.16), programmable Softkeys could be configured only for 'Ongoing call' and 'Idle screen' states
- **Improved IP phone security.** Web server access is only allowed on port 80 (HTTP) and port 443 (HTTPS) to the One Voice Operations Center (OVOC).

3.21.2 Known Constraints in Version 3.4.1

The table below shows the constraints that are known to exist in this version.

Table 3-40: Known Constraints

Incident	Description
IPPUC-2479	The 450HD phone and the HRS do not support local 3-way conference calls.
-	SIP Redundancy methods are not fully supported.
IPPUC-2514	[Shared Line Appearance] In a scenario in which a forwarded call is answered by another phone that shared the same line, the local shared line LED still flashes green and does not indicate that the call was accepted by another phone.
IPPUC-2576	[Multiple lines] The calls in the Call Log are not sorted according to the lines from which the calls were made and accepted by.
IPPUC-2536	[450HD/C450HD Expansion Module] A Line Function Key configured on the Expansion Module does not display the selected line indication (the line that was chosen as the default line from which to initiate calls).
IPPUC-2575	The user's status indication (available / in a call) is occasionally displayed as '?' in the Call Log.
IPPUC-2511	A Programmable Key (PK) configured to 'Line' and currently the default line changes its settings to 'Speed-dial'; the 'little rectangle' representing the default line is remains displayed for this PK.
IPPUC-2720	[445HD Shared Line] A Shared Line configured on the sidecar functions correctly, but the LED's functioning is faulty.
IPPUC-2775	[450HD/C450HD] On rare occasions, the numeric keypad is "locked" in idle mode and prevents the user from making a call. The alternative method of using the speaker of the handset and then pressing on the numeric keypad functions correctly.
-	The special characters ößüä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.

3.21.3 Resolved Constraints in Version 3.4.1

The table below shows the resolved constraints in this version.

Table 3-41: Resolved Constraints

Incident	Description
IPPUC-32	During a local conference call (445HD, C450HD), phone A's audio device switches from headset to speaker when phone B drops out.
IPPUC-13	[Device Manager]: The phone does not send a keepalive (KA) message with the status 'upgrading' before upgrading.
IPPUC-895	When making a paging call or when receiving an incoming paging call, the phone's screen display is incorrect.
IPPUC-909	The phone does not play a ringing tone when a paging call comes in.
[IGS-2578]	The user cannot make a paging call when the phone displays the New Call screen.
IPPUC-933	Long-pressing the Clear SK key to clear the entire string at once, does not function.
IGS-2482	The 'HD' logo is not displayed when toggling between two calls that use different vocoders.
IGS-2485	The speaker plays no voice when unplugging the USB headset device during a call.
IGS-2552	After a Consultation Transfer, the transferred call displays an incorrect screen on the phone.
-	'Drop from 3-way conference' is currently not supported.

3.22 Version 3.4.0.14

3.22.1 What's New



Version 3.4.0.14 includes firmware build **3.4.0.14.3**.

- This is the first version release for the 445HD | 445HD-R Generic SIP IP Phone.
- Supported SIP Environments:
 - Ribbon Communications (formerly GENBAND) softswitch solution Kandy Business Solutions (KBS) application server
- Supported features in this release include:
 - Single Line
 - Speed Dial and (up to 18) BLF presence buttons
 - BLF Call Pickup
 - Handles up to 8 concurrent calls
 - Call Hold / Un-Hold
 - Call Transfer: the hard TRANSFER key's default functionality (Blind Transfer) can be changed to Consultative Transfer
 - Three-way Conferencing (with local mixing)
 - Remote Conference compliancy with RFC 4579, SIP Call Control, Conferencing for UAs
 - Merge option: Two separate calls can be merged into one conference
 - Call Park (Genband only; only on 450HD / C450HD phones with Expansion Module)
 - Call Forwarding
 - DnD (Do Not Disturb)
 - Voicemail (including capability to secure user access with PIN code)
 - Message Waiting Indication (including MWI LED)
 - Caller ID Notification
 - Paging w/without Barge-in. Configurability of special keys as paging group dials.
 - Call Waiting Indication
 - Personal Directory
 - Automatic On-hook Dialing
 - Automatic Answering (Alert-Info header and "talk" event)
 - CWRR (Call Waiting Reminder Ring)
 - Call Logs: Missed/Received Calls and Dialed Numbers
 - Voca service to allow voice dialing
 - Redial
 - Dial Plan
 - Wi-Fi capability (see [IP Phone Models](#) above)
 - USB headset
 - Electronic Hook Switch (EHS) DHSG. Calls can be answered, and volume level can be changed with EHS-capable headsets.
 - Power Saving (LCD Brightness)

- Up to 20 configurable Softkeys / Programmable Softkeys (PSKs) in Idle and Ongoing Call States.

3.22.2 Known Constraints in Version 3.4.0.14

The table below shows the constraints that are known to exist in this version.

Table 3-42: Known Constraints

Incident	Description
IPPUC-32	During a local conference call, phone A's audio device switches from headset to speaker when phone B drops out.
IPPUC-13	[Device Manager]: The phone does not send a keepalive (KA) message with the status 'upgrading' before upgrading.
IPPUC-895	When making a paging call or when receiving an incoming paging call, the phone's screen display is incorrect.
IPPUC-909	The phone does not play a ringing tone when a paging call comes in.
[IGS-2578]	The user cannot make a paging call when the phone displays the New Call screen.
IPPUC-933	Long-pressing the Clear SK key to clear the entire string at once, does not function.
IGS-2482	The 'HD' logo is not displayed when toggling between two calls that use different vocoders.
IGS-2485	The speaker plays no voice when unplugging the USB headset device during a call.
IGS-2552	After a Consultation Transfer, the transferred call displays an incorrect screen on the phone.
-	SIP Redundancy methods are not fully supported.
-	'Drop from 3-way conference' is currently not supported.

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