Release Notes

AudioCodes High-Definition IP Phones Series

425HD, 445HD, and C450HD

SIP IP Phones

Version 3.5.6.13

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Notice

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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
C450HD IP Phone Quick Guide - Generic SIP
C450HD IP Phone User's Manual - Generic SIP
Device Manager Administrator's Manual
One Voice Operations Center (OVOC) IOM Manual
OVOC User's Manual

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1 Introduction

This document describes the new features, resolved constraints and known constraints of AudioCodes' Generic SIP IP phones for Version 3.5.5.14

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: Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ 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 1.0.2k, 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 ▪ Bluetooth: **425HD:** 5.0 (BR/EDR + BLE) **445HD/C450HD:** BT2.1+EDR/BT3.0 and BT4.2 ▪ Wideband audio support for Bluetooth headsets (Beta) ▪ Wi-Fi (445HD and C450HD - BW CPN flavor): Single band 2.4GHz, Dual Band 5GHz (C450HD only), 802.11b/g/n **(not applicable for 425HD)**
Media Processing	▪ Voice Coders: G.711, G.729, G.722 8000 / G.722 16000, OPUS. ▪ Acoustic Echo Cancelation ▪ Adaptive Jitter Buffer 300 msec ▪ Voice Activity Detection ▪ Comfort Noise Generation ▪ Packet Lost Concealment ▪ RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC 3711) ▪ DTMF Relay (RFC 2833)

Feature	Details
Telephony Features	■ Multi Line ■ Speed Dial and BLF presence buttons **425HD**: Speed Dial up to 8 **445HD** Speed Dial up top 39 **C450HD**: Speed Dial up to 8 or 48 with expansion module ■ BLF Call Pickup ■ Handles up to 8 concurrent calls ■ Multiple lines ■ 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 (only on 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 ■ Shared Line Appearance (SLA)
Configuration / Management	■ LCD Display User Interface Language Support (Various Languages) ■ 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 Device Manager ■ Access via Telnet or SSH
Supported Environments	■ SIP Gateway ■ Zoom ■ Asterisk ■ Freeswitch ■ Alianza Metaswitch core solution

Feature	Details
Diagnostics and Troubleshooting Tools	■ 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
Supported Languages	■ 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 (Magyar) ■ Japanese ■ Slovak ■ Czech ■ Latvian (contact person information) ■ Dutch
Supported Headsets	■ 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: • 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

Feature	Details
445HD Hardware	■ **Color Screen** 4.3": Graphic, 480x272 resolution ■ **Integrated sidecar** 376x60 resolution featuring 12 programmable speed dial keys with presence monitoring (BLF) ■ Connectors interfaces: • 2 x RJ-45 ports (10/100/1000BaseT Ethernet) for WAN and LAN (GbE support) • RJ-9 port (jack) for headset • USB interface for USB headset support • RJ-11 interface for DHSG ■ Mounting: • Wall and desktop mounting options • One angle for desktop mount, another angle for wall mount ■ Power: • DC jack adapter 12V • Power supply AC 100 ~ 240V • PoE Class 2: IEEE802.3af (optional) ■ Keys: • 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)
425HD Hardware	■ **Color Screen** 3.5 inch (480x320 resolution) ■ Connectors interfaces: • 2 x RJ-45 ports (10/100/1000BaseT Ethernet) for WAN and LAN (GbE support) • USB interface for USB headset support ■ Mounting: • Wall and desktop mounting options • One angle for desktop mount, another angle for wall mount ■ Power: • DC jack adapter 12V • Power supply AC 100 ~ 240V () • PoE Class 2: IEEE802.3af (optional) ■ Keys: • 4 softkeys and 4 physical (or 8 virtual) multifunction hard keys • VOICE MAIL message hotkey (including LED) • 4-way navigation button with OK key • MENU • REDIAL (including LED) • HOLD • MUTE (including LED) • TRANSFER • VOLUME control key • HEADSET (including LED) ■ SPEAKER (including LED)

<table>
<tr><th>Feature</th><th>Details</th></tr>
<tr><td>C450HD Hardware</td><td>

Five-inch color capacitive 1280 x 720 high-resolution touch (TFT) screen
Connectors interfaces:

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:

Wall and desktop mounting options
One angle for desktop mount, another angle for wall mount

Power:

DC jack adapter 12V
Power supply AC 100 ~ 240V
PoE Class 3: IEEE802.3af (optional)

Keys:

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)

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1.2 IP Phone Models

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

Table 2: 400HD Series IP Phone Models

Part Number	Product Description
IP425HDEG	425HD IP-Phone PoE GbE black
IP425HDEPSG	425HD IP-Phone PoE GbE black with an external power supply black
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)
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)
IPC45OHDEPSG-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 3.5.6.13

2.1 What's New – Release for Alianza Core (Metaswitch) only

- **IP Phones 400 series are compatible with Alianza Core (Metaswitch) Solution**

  AudioCodes IP Phones 400 Series are fully interoperable with the Alianza Core (Metaswitch) solution and support essential VoIP functionalities required for service provider deployments. This includes SIP-based call control, device provisioning, and standard telephony services, ensuring seamless operation within Metaswitch-based environments.

  This interoperability enables service providers to deploy AudioCodes IP phones as fully compatible and reliable endpoints, supporting a consistent user experience.

  Supported Features

- 3-way Calling
- 3-way Consultative Call Transfer
- Attended Call Transfer (no 3-way period)
- Automatic Callback
- Basic Call Functionality
- Basic Call Park and Retrieval
- Blind Call Transfer
- Bulk Provisioning Using DHCP Option 66
- Busy Lamp Field
- Busy Lamp Field Pickup
- Call Forwarding
- Call Waiting
- Calling Name Display
- Calling Number Display
- Cancel Call Waiting
- Digit Map Programming
- Distinctive Ringing - Priority Calling
- EAS Geographic Redundancy Support
- Encrypted Signaling and Media (TLS/SRTP)
- Encrypted Signaling and Media (TLS/SRTP) - Long Duration Calls
- Endpoint Voicemail - Basic Functionality
- Endpoint Voicemail - Visual Message Waiting Indicator
- Enhanced Call Park
- Enhanced Park Orbit Monitoring
- Enhanced Parked Call Retrieval
- Fault Tolerance and disconnection (Single Interface Devices)
- G.711 Codec
- G.722 Codec
- G.729 Codec
- Geographic Redundancy Support
- Hot Line
- Integrated Call Forwarding
- Integrated DND
- Multiple Line Hunt Groups
- NTP
- On-device media mixing for multi-party calls
- Originating Group Paging Calls
- Originating Intercom Calls
- Receiving Group Paging Calls
- Receiving Intercom Calls
- Receiving Push-to-Talk Calls
- RFC2833 (Out Of Band) DTMF Transmission
- Shared Line Appearance
- Shared Line Appearance Private Call Hold
- Silence Suppression with comfort noise
- Silence Suppression without comfort noise
- Simple Star Code Services
- SIP Call Forking
- TCP Communication
- Trigger config resync using SIP NOTIFY
- VoIP Voice Quality Monitoring - RTCP
- VoIP Voice Quality Monitoring - SIP PUBLISH
- Warm Line

2.2 Known Constraints in Version 3.5.6.13

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

Table 3: Known Constraints from This Release

Incident	Description
IPPUC-11353	**Metaswitch Shared Line Appearance:** There is no caller ID/Name on Call List details

2.3 Resolved Constraints in Version 3.5.6.13

Table 4: Resolved Constraints

Incident	Description
IPPUC-11352	**445HD:Security:** Serial port input completely blocked.
IPPUC-11282	When searching for a contact using Japanese Kanji (e.g., 田中一郎), the contact does not appear even with an exact match, but appears after deleting the last character.
IPPUC-11304	**Broadworks:** The device does not display the P-Asserted-Identity (PAI) name when received in a 183 Session Progress response.
IPPUC-11296	The phone intermittently fails to display the corporate logo background after reboot.
IPPUC-11279	In some cases, a call is established from default line instead of the active line.
IPPUC-11284	**Broadworks:** In some cases, the device fails to establish a remote conference call.
IPPUC-11102	Changing values of the **dial/reorder/howler tone** timeout does not take effect.
IPPUC-11257	**Broadworks:** IP Phone fails to load the XSI Personal Directory.

3 Previous Versions

3.1 Version 3.5.5.14

3.1.1 What's New

- **[445HD][BroadWorks]Call Center Availability Status**

  This feature allows enterprise front desk personnel to indicate their availability status (available or unavailable) for each assigned call center, which is communicated to the BroadWorks server. Based on this status, BroadWorks efficiently distributes incoming calls only to available personnel, reducing unanswered referrals and call disconnections.

  Users can be assigned up to three call centers, which are displayed on the right side of the phone screen and configured by the Administrator on programmable keys 4–6. The feature can be enabled or disabled by the network administrator using the configuration parameter xsi/callcenter/update (disabled by default), which controls call center availability updates via the XSI service.

Parameter Name	Description
person xsi/callcenter/update	Allows the IP Phone to receive the call center service from the BroadWorks server. ■ **1** = Enable ■ **0** = Disable (default).
personal_settings/functional_key/[15-17]/type	**Value:** BW_CALLCENTER
personal_settings/functional_key/[15-17]/ key_label	Define the string of characters used for the labels displayed on the screen next to Programmable Keys 4-6. (e.g., call center name). **Max String Length:** 128 characters

- **Use sip credentials when there is no XSI credential**
- **Improve failover/failback process based on DNS response**

Parameter Name	Description
voip/signalling/sip/dns_fail_back_reload_voip/enable	**0** = Disable (Default) **1** = Enable

- **Ability to skip DNS NAPTR queries when resolving SIP servers**

Parameter Name	Description
voip/signalling/sip/dns_no_naptr	**0** = Disable (Default) **1** = Enable

3.1.2 Resolved Constraints in Version 3.5.5.14

Table 5: Resolved Constraints

Incident	Description
IPPUC-11210	**425HD:** On rare occasions, phones may not appear in Device Manager.

Incident	Description
IPPUC-11245	**445HD:** In some cases, the phone does not send a **SUBSCRIBE** request for the BLF resource list when local presence for lines is enabled (`voip/services/busy_lamp_field/local_presence_for_lines=1`).
IPPUC-10938	No notifications when XSI server is unavailable

3.1.3 Known Constraints in Version 3.5.5.14

None

3.2 Version 3.5.4.44

3.2.1 What's New

- Reboot menu option moved from Admin menu to Settings menu to allow the non-administrative users to reboot the phone.
- Added new function key type **Speed_Dial_BLF** (which allows presence per functional key)

Parameter Name	Description
personal_settings/functional_key/x/type	Added **Speed_Dial_BLF** type

- Ability to disable the presence SUBSCRIBE message (for **Speed_Dial** functional key type only)

Parameter Name	Description
voip/services/speed_dial_subscribe/enabled	Enable/Disable sending SUBSCRIBE message for personal_settings/functional_key/x/type = Speed_Dial. **0** = Disable **1** = Enable (Default)

- Resend SUBSCRIBE message if received 404 response

Parameter Name	Description
voip/services/subs/retry_after_404	Determine whether IPP sends SUBSCRIBE retry when 404 response is received. Retry timer (If enabled) is configured in: voip/services/subs/retry/timeout. **0** = Disable (Default) **1** = Enable
voip/services/subs/retry/timeout	Configure the SUBSCRIBE retry timeout (in msec). **Range:** 10 - 10000 **Default:** 300

- Ability to disable the presence SUBSCRIBE message (for **Call Log** lists)

Parameter Name	Description
voip/services/subscribe_call_log_presence	Determine whether IPP sends SUBSCRIBE when opening the **Call Log** screen. **0** = Disable **1** = Enable (Default)

- INVITE message configuration added

Parameter name	Description
voip/signalling/invite/remove_default_port_from_request_line	Removes the default SIP port from the **Request-URI** in the INVITE message. **0** = Do not Remove (Default) **1** = Remove
voip/signalling/invite/remove_default_port_from_to_header	Removes the default SIP port from the **To** header in the INVITE message. **0** = Do not Remove (Default) **1** = Remove
voip/signalling/invite/add_port_to_contact_header	Add the SIP port to **CONTACT** header in INVITE message **0** = Do not Add (Default) **1** = Add

- Conference call configuration for Vodafone

Parameter	Description
voip/services/conference/style	Hide the **conference** softkey and **merge** calls option available in the call menu (when IPP has 2 calls) **REGULAR** - Do not hide (Default) **VODAFONE** - Hide
system/HoldOngoingCallOnInitiated	For creating a conference call, put the active call on hold before initiating a new call. **0** = REGULAR **-** keep current logic (Default) **1** = Vodafone - holds active call on initiation

- Ability to disable/enable NTP queries for public NTP servers

Parameter	Description
system/ntp/add_special_ntp_server	Disable/Enable sending NTP query for the well-known (public) NTP servers. **0** = Don’t send **1** = Send (Default)

- Display caller ID from **P-Asserted identity** header instead of **From** header in incoming call

Parameter	Description
voip/services/display_PAI_enable	Display **P-Asserted_identity** header (if exists). **0** = Disable **1** = Enable (Default)

- Firmware header download method for HTTP server to avoid downloading entire file if device version is the same as on the server

Parameter	Description
system/http/support_206	Firmware header download method for HTTP server. **0** = HTTP server that does not support 206 response **1** = HTTP server that supports 206 response (Default)

- Ability to disable **New Call** Softkey in **Incoming Call** screen

Parameter	Description
system/AllowNewCallsDuringIncoming	Disable/Enable **New Call** Softkey in **Incoming Call** screen. **0** = Disable **1** = Enable (Default)

- Vodafone Environment XSI configuration

Parameter	Description
xsi/contact/enable	Enable or disable contact sync via the xsi service. **0** = Disable (Default) **1** = Enable
xsi/contact/directory/0-7/label	Configure a label used for configuring XSI directory name
xsi/contact/directory/0-7/type	XSI directories Type: - GROUP (Default) - ENTERPRISE - GROUP_COMMON - ENTERPRISE_COMMON - PERSONAL - NONE
xsi/contact/user_id/mac_address/enabled	Add MAC address to the user ID **0** = Disable (Default) **1** = Enable
xsi/contact/search_on_empty/enabled	When enabled, an empty search box will search for all contacts. **0** = Disable **1** = Enable (Default)

- Cisco WebeX Environment - Call Log Integration based on XSI protocol

  The XSI Call Log enables users to view and manage detailed call history directly within the IP Phone interface. This integrated functionality provides a centralized record of inbound, outbound, and missed calls associated with the user's extension.

Parameter	Description
xsi/calllog/enable	Enable or disable the XSI Call Log. **0** = (Default) The XSI Call Log is disabled and the IP Phone displays the local call log. **1** = The IP Phone retrieves and displays call history from the configured XSI server.
xsi/calllog/delete/enable	Controls whether a **DELETE** softkey is shown on the *Call Log* and *Call Details* screens. **0** = (Default) The DELETE softkey is not displayed. **1** = A **DELETE** softkey is displayed (supported only if the XSI server allows call-log deletion).

- Support redundancy failback when DNS resolution of the SIP proxy server's FQDN returns a different IP address.

Parameter name	Description
voip/signalling/sip/dns_fail_back/enable	Enable or disable redundancy failback when DNS resolution of the SIP proxy server's FQDN returns a different IP address. **0** = Disable (Default) **1** = Enable

- Prevent SIP UNSUBSCRIBE on VoIP Reload upon failover/failback

Parameter	Description
voip/signalling/sip/unsubscribe_on_voip_reload	Controls whether a SIP **UNSUBSCRIBE** message (Expires: 0) is sent after a VoIP application reload upon failover/failback. ▪ **0** = Don't send an unsubscribe message when the VoIP service reloads. ▪ **1** (Default) = Sends an unsubscribe message when the VoIP service reloads.

- Exclude 'feature code dialing' from Call Log

  Feature code dialing is a PBX mechanism used for special functions (e.g., follow me, Call Forwarding, Do Not Disturb, etc.) rather than for regular call dialing.

  Feature codes typically include identifiable prefixes and/or suffixes such as `#123#`, `*678*`, `*#123456*#`, or other administrator-defined patterns.

Parameter	Description
personal_settings/call_log/exclude_pattern	Any dialed number matching one of these patterns (set by administrator) is treated as a feature code and is excluded from the call history display. You can define multiple patterns (up to 1024 characters total), separated by the '**|**' symbol. Each pattern can use the following building blocks: ■ **DTMF digits:** 0–9, *, # ■ **Wildcard x:** Matches any single digit (0–9) ■ **Range:** Square brackets (**[** and **]**) including an Asterisk (*) and one or more characters ■ **Repeat:** The character followed by a period (**.**) means the previous element can repeat any number of times (including zero) **Example:** `personal_settings/call_log/exclude_pattern = *x.|#x.|*#x.#*|#*x.*#|**#x.**#|#*[1-7]*#` The following dial strings will be recognized as feature codes and hidden from the call log: ■ *123 - (`*x.`) ■ #456 - (`#x.`) ■ *#789#* - (`*#x.#*`) ■ #*666888*# - (`#*x.*#`) ■ **#123456**# - (`**#x.**#`) ■ #*56*# - (#*[1-7]*#)

3.2.2 Resolved Constraints in Version 3.5.4.44

Table 6: Resolved Constraints

Incident	Description
IPPUC-11067	**425HD:** In certain cases, audio may be lost after prolonged idle time without rebooting the device.
IPPUC-11146	**425HD:** on rare occasions, ringing continued after answering a call.
IPPUC-11136	**425HD:** Dial tone was heard when tone volume was set to minimum.
IPPUC-11113	**425HD:** On rare occasions, a short beep was heard before the normal dial tone.
IPPUC-11099	**425HD:** Could not hear DTMF audio from remote side when voip/media/ignore_rfc_2833_packets=0.
IPPUC-11112	**425HD:** On rare occasions, a speaker had no audio after setting idle volume to minimum.
IPPUC-11185	**C450+Expansion Module:** screen-swipe behavior changed from **Right-To-Left** to **Left-To-Right**.
IPPUC-11096	**425HD / C450HD:** USB headset hang-up button did not end a remote conference call.
IPPUC-10580	When the network/lan/_802_1x/tls_identity parameter is empty, the IP phone defaults to using **<HWType_MAC>** as its EAP-TLS identity.
IPPUC-10503	When a caller disconnects the call, the call log records only the phone number and not the username.

Incident	Description
IPPUC-11161	**Proxy Failback:** After failback to Primary proxy the **SUBSCRIBE** packet kept being sent to Secondary proxy.
IPPUC-11154	Call Log entries were inconsistent depending on who ended the call.
IPPUC-11097	MISSED softkey and REDIAL hardkey still functioned after disabling the Call Log feature (system/feature/calllog/enabled=0).
IPPUC-11104	Parameter voip/dialing/timeout did not take effect after being updated via provisioning.
IPPUC-10622	Placing the handset back on the cradle failed to end an On-Hold call (voip/disconnect_call_no_unhold/enable=1).
IPPUC-11103	Incorrect default identity was used for EAP-TLS authentication.
IPPUC-11107	Maximum string length of parameter voip/dialing/dial_complete_key/key should be **1**.
IPPUC-11073	**XSI Directory:** On rare occasions, the cursor jumped when scrolling through the XSI directory.
IPPUC-11090	**XSI Directory:** Added parameter xsi/contact/search_on_empty/enabled displayed all contacts when opening an XSI contact group.
IPPUC-11074	**XSI Directory:** The displayed "Contact Number" did not match xsi/contact/number_of_results.
IPPUC-10948	**XSI Directory:** The directory screen did not display the total number of contacts.
IPPUC-10995	**XSI Directory:** On occasion, long contact names were truncated in the display.
IPPUC-11032	Failed to obtain VLAN ID from DHCP options 128 / 129 / 144 / 157 / 191 / 132.
IPPUC-11004	**Cisco Webex Call Park Extension:** Chinese characters appeared as garbled text on the phone Call Park Extension functional keys.
IPPUC-10862	Removed unnecessary indicators from System-status screen.
IPPUC-11002	On occasion, UI displayed "802.1X Authentication Failure" even when authentication succeeded.
IPPUC-11000	**FreeSWITCH:** UI did not show participants when the IPP is in a remote conference.

3.2.3 Known Constraints in Version 3.5.4.44

None

3.3 Version 3.5.2.32

3.3.1 What's New – Release for Cisco Webex Only

> To enable connection of your IP phone to the Cisco Webex environment, please ensure it is updated to version 3.5.2.32. This update includes the integration of "Identrust" trusted CA certificates, enhancing compatibility and security.

- **Interoperability with Cisco Webex calling solution**

Cisco Webex Calling is a complete secured and reliable enterprise-grade cloud calling and team collaboration solution optimized for businesses and includes centralized administration and device management. AudioCodes IP Phones are interoperable with Cisco Webex Calling and support all the essential business calling capabilities you are likely to require.

For more information on Connecting this device, please visit:

- Deployment of AudioCodes devices at:
  Cisco managed third-party device support on Webex Calling.
- Webex Center device Deployment Help at:
  Add users manually in Control Hub and Configure and manage Webex Calling devices.

- **Supported Webex Calling Features**

Anonymous Call Rejection	**Call Redial**	**N-Way Calling**
Busy Lamp Field (Monitoring) - only on 445HD & C450 (with expansion module)	Call Return	Outbound Caller ID Blocking
Call Forwarding Always	Call Transfer	Personal Phone Directory
Call Forwarding Busy	Call Waiting (for up to 4 calls)	Privacy
Call Forwarding No Answer	Call Waiting ID	Push to Talk
Call Forwarding When Not Reachable (Business Continuity)	Connected Line ID Restriction	Selective Call Acceptance
Call Forwarding Selective	Do Not Disturb	Selective Call Rejection
Call History (local)	Extension Dialing, Variable Length	Shared Call Appearance
Call Hold and Resume	Feature Access Codes	Speed Dial
Call Logs (local) with Click to Dial	Hoteling Guest	Three-Way Calling
Call Notify	Inbound Caller ID (Name)	User Web Portal
Call Queue Agent	Inbound Caller ID (Name and Number)	Voicemail
Call Recording	Multiple Line Appearance	

- **Prevent Caller from Disconnecting Emergency Calls.**

  Once an emergency call is placed, only the called emergency service can disconnect the call — the caller cannot terminate it.

Parameters	Description
personal_settings/emergency_no_disconnect/enabled	1 = Enable (Prevents caller from ending an emergency call.) 0 = Disable (Default)
voip/services/emergency_call_list	List of emergency numbers. Maximum String Length = 128 characters **example:** voip/services/emergency_call_list=911 933 (Use [space] delimiter, without commas)

- **Improved Out-of-the-Box Troubleshooting Tools**

  Improved out-of-the-box troubleshooting. Admins can now monitor the status of the various software modules of the device. If initial provisioning is unsuccessful or if admin encounters an issue related to the network / connection to Device Manager, the feature gives admin an indication as to why.

  The feature enables debugging via the phone screen without requiring external systems. Admin can check connectivity independently of external apps (select **Status** > **System State**).

3.3.2 Resolved Constraints in Version 3.5.2.32

Table 7: Resolved Constraints

Incident	Description
IPPUC-10879	**Zoom**: when voip/call_forward_dnd_all_extensions/enabled=0, 'all extension' option will be hidden on DND/ FORWARD screen.
IPPAN-15433	**SCEP**: Support 'POST' method to enroll device certificate.
IPPUC-10975	One-way audio after hold/resume of a 23-minute PSTN call.
IPPUC-10894	Perform immediate failover after receiving 503 from the primary server.
IPPUC-10933	**425HD**: personal_settings/GetBackToIdleScreenTimeout is not functioning.
IPPAN-15751	**SCEP**: NDES Server 2019 Enrollment failure.
IPPUC-10920	Added a **voip/services/display_PAI_enable** parameter to display info from PAI instead of FROM header.
IPPUC-10927	Default value of system/AllowCallsInLockState changed to ALLOW_INCOMING_ONLY.
IPPUC-10923	Parameter retry_after_cseq_too_small should not be a boolean.
IPPUC-10800	'NEW CALL' softkey in incoming call screen is now configurable - system/AllowNewCallsDuringIncoming.
IPPUC-10585	**425HD Zoom**: Short ringer tone plays when answering a cloud paging call.
IPPUC-10794	**USB headset**: User is able to place call from the headset’s dial button once headset is disabled(system/feature/headset/enabled=0).
IPPUC-10839	Redundancy: After receiving 401 during registration, IPP sends registration message to the second IP address.
sIPPUC-10845	Remove 'Network IPvx Type' from Web GUI.
IPPUC-10838	**Zoom discreet call:** After reconnecting network during discreet call, discreet call FK still displayed on.
IPPUC-10682	Ringer tone stops playing when receiving 2 incoming calls.
IPPUC-10813	Italian localization improvements.
IPPUC-10806	Updated some expired “well known” root CA certificates.

3.3.3 Known Constraints in Version 3.5.2.32

None

3.4 Version 3.5.1.89

3.4.1 What's New

- **Zoom Call Forwarding Setting Sync**

  When 'Call Forwarding Setting Synchronization' is enabled, the call forwarding settings on one IP phones and zoom client will synchronize in real time with other IP phones and zoom clients using the same account.

Parameters	Description
system/feature_key_synchronization/local_call_forward	Enable/Disable Local Call forward after enabling call forward sync feature. it should be 1 when call forward sync feature is enabled. **0** = Disable (Default) **1** = Enable
system/feature_key_synchronization_call_forward/enabled	Enable/Disable Call forward sync. When enabled, the **Call Forwarding** setting will be synchronized between clients (if available) and desk phones. **0** = Disable (Default) **1** = Enable

3.4.2 Resolved Constraints in Version 3.5.1.89

None

3.4.3 Known Constraints in Version 3.5.1.89

None

3.5 Version 3.5.1.75

3.5.1 What's New

- **SCEP: Certificate Signing Request Enhancements**

  Includes adding retry attempts to the certificate enrolment and renewal process and supporting Subject Alternative Name (SAN) attributes in certificate signing requests.

Parameter	Description
security/CSR_SubjectAltName_DNS/[0-9]/dns_names	Define CSR DNS name.
security/CSR_SubjectAltName_Email/[0-4]/email_addresses	Define CSR email address.
security/CSR_SubjectAltName_IP/[0-4]/ip_addresses	Define CSR ip address.
security/CSR_SubjectAltName_URI/[0-4]/uris	Define CSR URI.

- **SCEP**: Automatic Fetch of One-Time Password and Certificate Thumbprint

  Supports retrieving the SCEP one-time password and certificate thumbprint from the OTP server.

Parameter	Description
security/SCEPEnroll/otp_server_url	Set the otp server url
security/SCEPEnroll/otp_password	Set the otp server password
security/SCEPEnroll/otp_username	Set the otp server username

- **Provisioning of Corporate Directory file**

  Support uploading a local corporate directory file by provisioning. Supports **.txt** or **.cfg** format. Users can search and dial contacts from the local directory.

Parameter	Description
lync/corporate_directory/enabled	Enable/Disable **corporate directory**. **0** = Disable (Default) **1** = Enable
lync/contact_search_method	Defines contact search method for call extension. Must be set to DISABLE to use local directory. Three values can be set: ▪ DISABLE(Default) ▪ LDAP ▪ LYNC_CONTACT
provisioning/corporate_directory_uri	URI for downloading the corporate directory file (supports http, https, tftp, ftp) Default: NULL
personal_settings/max_directory_size	Define the max number of contacts for corporate directory. Range: [0->700] Default: 700

- **Ability to Disable/Enable Features and Keys on IP Phone**

  The network administrator can disable access to specific features. The feature is motivated by the requirement on the part of some enterprises to control the setting remotely to comply with company policy.

Parameters	Description
system/feature/contacts/enabled	Ability to disable the user from **accessing the Contact list**. **0** = Disable **1** = Enable (Default)
system/feature/vmail/enabled	Ability to disable the user from **accessing the Voice Mail**. **0** = Disable **1** = Enable (Default)
system/feature/menu/enabled	Ability to disable the user from **accessing the Main Menu screen**. **0** = Disable **1** = Enable (Default)
system/feature/speaker/enabled	Ability to disable the user from **accessing the speaker hard key**. **0** = Disable **1** = Enable (Default)
system/feature/headset/enabled	Ability to disable the user from **accessing the headset hard key**. **0** = Disable **1** = Enable (Default)
system/feature/handset/enabled	Ability to disable the user from **accessing the handset**. **0** = Disable **1** = Enable (Default)
system/feature/transfer/enabled	Ability to disable the user from **blind/consult transferring a call**. **0** = Disable **1** = Enable (Default)
system/feature/hold/enabled	Ability to disable the user from **holding a call**. **0** = Disable **1** = Enable (Default)

- **Supports enabling or disabling the display of DTMF digits on screen during a call.**

Parameter	Description
security/mask_dtmf_digits/enabled	Disable/Enable **mask DTMF digits** during a call. **0** = Disable (Default) **1** = Enable

3.5.2 Resolved Constraints in Version 3.5.1.75

Table 8: Resolved Constraints

Incident	Description
IPPUC-10596	**425HD**: Constant clicking noise (~1 sec interval) during ringback tone on outgoing calls

Incident	Description
IPPUC-10646	**425HD**: Increase message font size due to small LCD
IPPUC-10667	**425HD**: Poor input voice quality with Bluetooth headset or speaker
IPPUC-10668	**425HD:** Poor input voice quality with USB headset
IPPUC-10597	**425HD**: Handset sound is too sharp when call reaches voicemail
IPPUC-10633	**425HD:** Codec packetization times of 10ms and 30ms are supported (remove previous limitation)
IPPUC-10671	**425HD:** Occasional 1-second noise when ending a local conference
IPPUC-10763	**425HD**: No dial tone after quickly pressing the holder (HK) 10 times
IPPUC-10604	**425HD:** Add support for new/incoming call during a 3-way conference (remove previous limitation)
IPPUC-10750	**425HD:** Paging during active call only works on the 1st line
IPPUC-10613	RTP packets have incorrect “Differentiated Services Field (ToS)” value
IPPUC-10650	LDAP does not work when the base name contains special characters
IPPUC-10689	**Office365**: SIP line becomes unregistered when searching contacts
IPPUC-9859	**Office365**: Username not displayed on **calling screen** when Office365 account is registered
IPPUC-10672	Redundant proxy: Device fails to perform failover when session update fails
IPPUC-10503	**Zoom ICE**: One-way voice when receiving two incoming calls and answering one
IPPUC-10651	Failed to add a Speed Dial key through the Personal Directory
IPPUC-10719	**BSFT**: Feature key sync: IP phone does not sync 'Forward' configuration from server
IPPUC-10663	voip/dialing/auto_dialing/timeout value has no effect
IPPUC-10746	RTCP keep alive must be sent when call is in held state

3.5.3 Known Constraints in Version 3.5.1.75

None

3.6 Version 3.5.0.171

Version 3.5.0.171 is specifically for the 425HD model. A new release for all phone models will be available soon.

3.6.1 What’s New

- Introduction of 425HD Phone

The 425HD IP phone is an advanced, mid-range business IP phone for office workers and common area users. Designed with simplicity very much in mind, it is equipped with a large multi-lingual 3.5” color display and has 8 programmable multi-function keys.

- Older incoming calls can now get prioritized over newer incoming calls.

  When multiple incoming calls are received, they are listed in the order they arrived, from top (oldest) to bottom (newest). Older calls are prioritized to ensure they are handled first, minimizing wait times for those who have been waiting longer. This approach helps manage calls efficiently and fairly.

Parameter	Description
personal_settings/incoming_call_fairness/enabled	0 = Disable (Default) 1 = Enable

- The devices now support certificate enrollment using Simple Certificate Enrollment Protocol (SCEP) using Microsoft’s Network Device Enrollment Service (NDES) server, thereby allowing device certificates and CA certificate provisioning to be scaled to multiple devices.

  After devices are provisioned with a SCEP-related configuration, they receive a CA certificate from the NDES, issue a Certificate Signing Request (CSR) to the NDES and receive a device certificate signed by the CA certificate (the one that the device received from NDES).

  The next table shows the descriptions of the SCEP parameters.

Parameter	Description
security/SCEPEnroll/ca_fingerprint	Define the thumbprint (hash value) for the CA certificate. Default value: NULL. Network admins must set its value to (for example): 3EBE50003ABF1DF5E6B5A3230B02B856
security/SCEPEnroll/password_challenge	Define the enrollment challenge password. Default value: NULL. Network admins must set its value to (for example): 7A7F9FC4BB7625F0935E67EA6D6322ED
security/SCEPServerURL	Define the SCEP server URL. Default: NULL. If you use Microsoft NDES server, use: https://<NDES server IP address/Hostname>/certsrv/mscep/mscep.dll/pkiclient.exe
security/SCEPEnroll/renewal/advancethreshold	Define the renewal advance threshold of the device certificate. Configure between 50 and 100 (in units of percentage) Default: 80 This indicates that a renewal of the certificate (device.crt) will be initiated when 80 percent of its validity is reached.

Parameter	Description
security/SCEPEnroll/rollover/advancethreshold	Specify the threshold of the CA Root certificate's validity at which to initiate a renewal. Configure between 50 and 100 (in units of percentage). Default: 90 This indicates a renewal of the certificate (CAROOT.crt.) will be initiated when 90 percent of its validity is reached.
security/CSR/CommonName	Define a value according to the following 'wildcard' format: {mac} – the device's MAC address {IP} - the device's IP address {model} - the device model
security/CSR/Country	Define the name of the country used to generate the certificate signing request (CSR). Note: The ISO (International Organization for Standardization) code of the country / region in which the organization is located.
security/CSR/Email	Optionally, define the email address used to generate the CSR.
security/CSR/Organization	Optionally, define the legal name of the organization used to generate the CSR.
security/CSR/State	Optionally, define the name of the state / province used to generate the CSR.

- **Call History**: Ability to disable the user from accessing call history:

Parameter	Description
system/feature/callog/enabled	Disable/Enable call log access. 0 = Disable 1 = Enable (Default)

- **Call History**: Ability to disable the user from using the redial option:

Parameter	Description
system/feature/redial/enabled	Disable/ Enable redial access. 0 = Disable 1 = Enable (Default)

- **Wi-Fi:**
  - Support Wi-Fi working with EAP-TLS authentication.
  - Ability to configure Wi-Fi from configuration file.

Parameter	Description
network/wifi/x/verify_server_certificate	Enable/disable verify server certificate. Default = 0
network/wifi/x/eap_method	Set eap method (PEAP/ TLS/ TTLS/ PWD). Default = PEAP
network/wifi/x/auto_reconnect	Enable/disable wifi auto reconnect. Default = 1
network/wifi_enabled	Enable/disable wifi feature. Default = 0
network/wifi/x/client_cert	1 = Enable, bring a client certificate during wifi authentication (eap-tls/ttls). 0 = Disable (Default)
network/wifi/x/identity	Set ipp identity. Default = null
network/wifi/x/password	Set ssid password. Default = null
network/wifi/x/phase2_authentication	Set phase2 authentication method(NONE/PAP/MSCHAP/MSCHAPV2 /CHAP/MD5/GTC). Default = NONE
network/wifi/x/private_key	1 = Enable, bring client private key during wifi authentication (eap-tls/ttls). Default = 0
network/wifi/x/security	Set ssid security method (NONE/WPAPERSONAL/WPA2PERSONAL /WPAENTERPRISE/WPA2ENTERPRISE/WEP). Default = NONE
network/wifi/x/ssid	Set ssid name. Default = null
network/wifi/x/wps_method	Set wps method (NONE/ALL/PIN/AUTH/PBC). Default = NONE

3.6.2 Resolved Constraints in Version 3.5.0.171

Table 9: Resolved Constraints

Incident	Description
IPPUC-10237	**445HD:** in rare occasions, the presence status of all lines appears as a white circle after ending a conference call.
IPPUC-10116	**445HD-Zoom-Intercom Key**: The Zoom-Intercom key LED remains on when the key is located in the sidecar.
IPPUC-10190	**Auto-Dialing:** When auto-dial is enabled, the IP phone cannot receive incoming calls while an outgoing call is in progress.
IPPUC-10454	**Call Forward**: A blank space appears in the default field when "number of forward" is set.
IPPUC-9823	**Call Forward:** When IP Phone Call forward is enabled with a no-answer 6 second timeout, a second incoming call is forwarded prematurely after only 1 second.
IPPUC-10182	**Discreet Call:** In some cases, the first discreet call cannot be ended when a second discreet call is established.
IPPUC-10028	**Free-Switch Shared Line:** Calls on hold should not display as an active call in the Calls tab.
IPPUC-10092	**Google Calendar:** Unable to join a meeting when pressing the "Join" button on the notification screen.
IPPUC-10014	**LCD brightness:** IP Phone LCD brightness does not get brighter when getting a meeting reminder.
IPPUC-10043	**Phone Lock:** If the wrong lock PIN code is entered five times, the IP phone will only sign out the first line.
IPPUC-10553	**Virtual Sidecar**: The Call tab should automatically be displayed when making a second new call.
IPPUC-10180	**Virtual Sidecar:** The call will be automatically disconnected if hold and resume actions are performed from the virtual sidecar.
IPPUC-10487	**Virtual Sidecar**: The cursor position should default to the "active call" in the call list tab.
IPPUC-10552	**Virtual Sidecar**: The icon for an incoming call is incorrect.
IPPUC-10486	**Virtual Sidecar**: While on a call, the call list tab should automatically be displayed after answering another incoming call.
IPPUC-10584	**Wi-Fi with EAP-TLS authentication**: The IP phone fails to connect to an 802.1x Wi-Fi network on the first attempt when using a factory certificate.
IPPUC-10453	**Zoom Call Monitoring**: The user status incorrectly shows two ongoing calls as connected.
IPPUC-10130	**Zoom local conference:** sometimes IP Phone plays music on hold after joining a conference ++.
IPPUC-10404	**Zoom Remote Conference**: Failed to remove two participants with the same account from the conference.
IPPUC-10149	**Zoom:** C450 sometimes 'call count' does not display on Call Monitoring programmable key.
IPPUC-10127	**Zoom:** Zoom-name of cloud programmable key dims in brightness when disconnected and reconnected from the network.
IPPUC-10255	**Zoom-ICE**: The IP phone crashes when voip/media/ice/turn_server_password is set to NULL.

3.6.3 Known Constraints in Version 3.5.0.171

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

Table 10: Known Constraints from This Release

Incident	Description
IPPUC-10580	**425HD**: User cannot establish a new call during a local conference call.
IPPUC-10503	**ICE**: When receiving two incoming calls and answering one of them, one-way audio occurs, and the user cannot hear the person on the other end.
IPPUC-10596	**425HD:** When making an outgoing call, constant clicking noises occur approximately every second during the ringback tone.
IPPUC-10633	**425HD:** Codec packetization times of 10ms and 30ms are not supported. Only 20ms, 40ms, and 60ms are supported.
IPPUC-10668	**425HD:** USB headset is not supported in this release.

3.7 Version 3.4.11

3.7.1 What's New

- **Zoom C450HD:** Support P2P media based on ICE (Interactive Connectivity Establishment). Users or devices that have certain features like recording or monitoring enabled may not be able to use Peer to Peer Media (pending Zoom-side activation).

3.7.2 Resolved Constraints in Version 3.4.11

Resolved Zoom Constraints

Table 11: Resolved Zoom Constraints

Incident	Description
IPPUC-9461	Pause dialing from the SD key does not work during a call.
IPPUC-9599	Enabling and disabling 'DND_ALL' multiple times on a DUT with 12 shared lines causes the IP Phone to re-register occasionally.
IPPUC-9637	**Zoom Intercom:** No beep when auto-answering an incoming call.
IPPUC-9703	**Zoom Intercom:** Pressing the intercom softkey again should end the ongoing intercom session.
IPPUC-9762	**Zoom outbound redundant:** SIP message uses "sips" when registration detects primary server.
IPPUC-9525	**Google Calendar**: No meeting type icon for Teams and Zoom meetings.
IPPUC-9422	Sometimes, an 'Invalid call' screen appears when ending a cloud paging call.
IPPUC-9780	**Google Calendar**: DUT receives a new calendar after editing an existing one.
IPPUC-9688	The duration time isn't updated during a paging call.
IPPUC-9058	**Call Monitoring:** The monitored call does not display on the BMW screen occasionally.
IPPUC-9440	**Zoom Intercom:** The "Disconnected" screen flashes when receiving a second intercom call.
IPPUC-8219	**Zoom CBarge:** The IPP cannot rejoin the conference after exiting it.

Incident	Description
IPPUC-9829	**Zoom redundancy:** DUT does not fail back to the primary DNS server after a timeout.
IPPUC-9636	The virtual sidecar does not appear when system/virtual_sidecar/enabled is set to 1.
IPPUC-9827	**O365:** The "Corporate Directory" contact should not appear when the Office_365 account is signed out.
IPPUC-9687	**Zoom:** The call list does not always display under the shared line group.
IPPUC-9804	**O365:** The SIP-URI is incorrect in Outlook contacts details.
IPPUC-9675	**Google Calendar**: "all day" event only need to show data (no need to show time).
IPPUC-9814	**Google Calendar**: The calendar is not arranged in chronological order
IPPUC-9833	**Redundancy:** IP Phone should not failback to the primary outbound proxy when symmetric mode is enabled
IPPUC-9953	IP Phone should not send an additional register message to secondary primary outbound proxy after receiving 401 and 503 responses from secondary primary outbound proxy.
IPPUC-10019	**Zoom:** Failed to answer intercom call when the username of caller is long
IPPUC-10069	Broken RTP Detection fails
IPPUC-10016	When primary outbound proxy is down, IP Phone should not send unregister message to secondary primary outbound proxy before registering to secondary primary outbound proxy.

3.7.3 Known Constraints in Version 3.4.11

Known constraints from this release:

None

3.8 Version 3.4.10.94.1

3.8.1 What's New

None

3.8.2 Resolved Constraints in Version 3.4.10.94.1

Resolved Zoom Constraints

Table 12: Resolved Zoom Constraints

Incident	Description
IPPUC-9657	**ZoomPhone:** Remote Call Control cannot be enabled after guest user signed in.

3.8.3 Known Constraints in Version 3.4.10.94.1

Known constraints from this release:

None

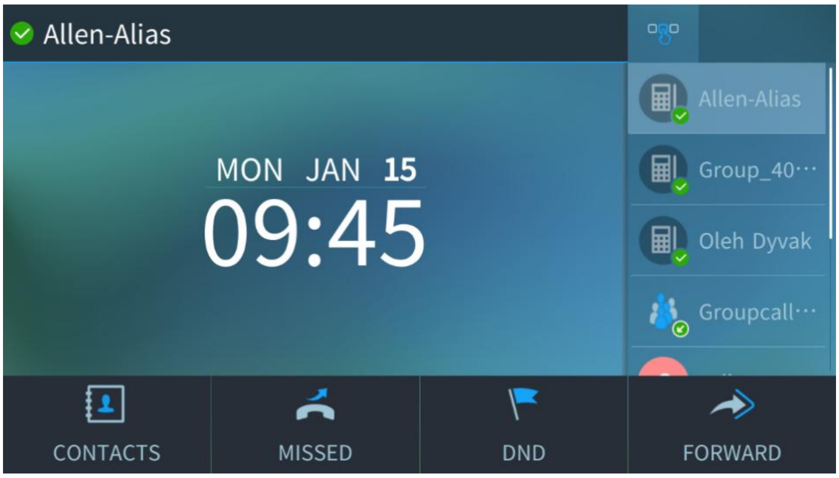
3.9 Version 3.4.10.94

3.9.1 What's New

- C450HD: Introduced virtual sidecar for shared lines.

  Implements virtual sidecar for IP Phones that do not have a physical sidecar.

  Figure 3-1: 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 **Programable Key** tab, **Calls** tab, and **Shared Call** tab. 0 = Disable. Default = 0.

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>system/virtual_sidecar/single_click</td><td>1 = Enable:
Shared Call tab - Pressing on a call performs the following:
When call is active, performs a Call Monitoring action by select.
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).
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:
Shared Call tab - Open the shared line active calls screen.
Calls tab - Bring call ongoing screen to foreground only.
Default =0.</td></tr>
</table>

- 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.

Parameter	Description
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: Sign in Google Calendar

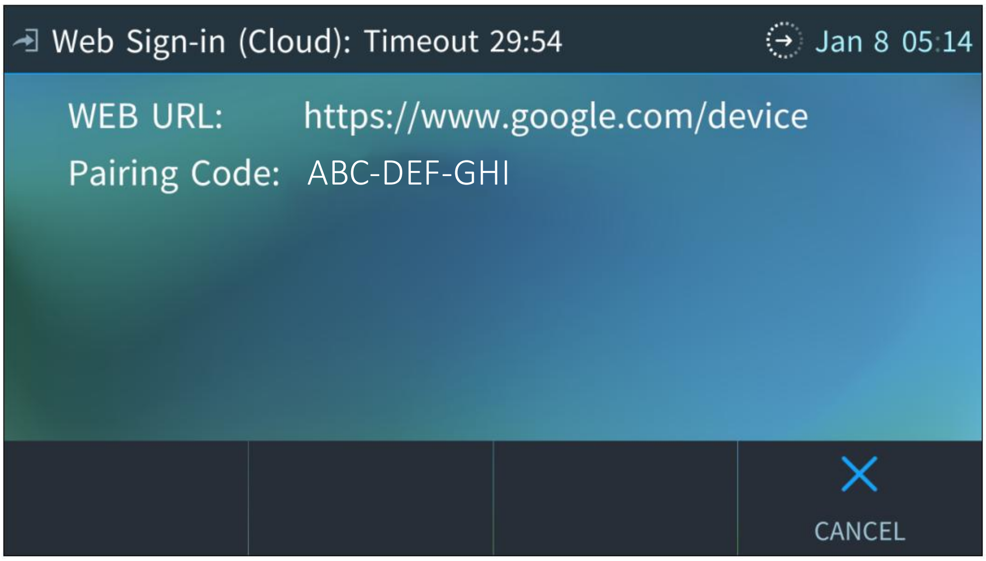
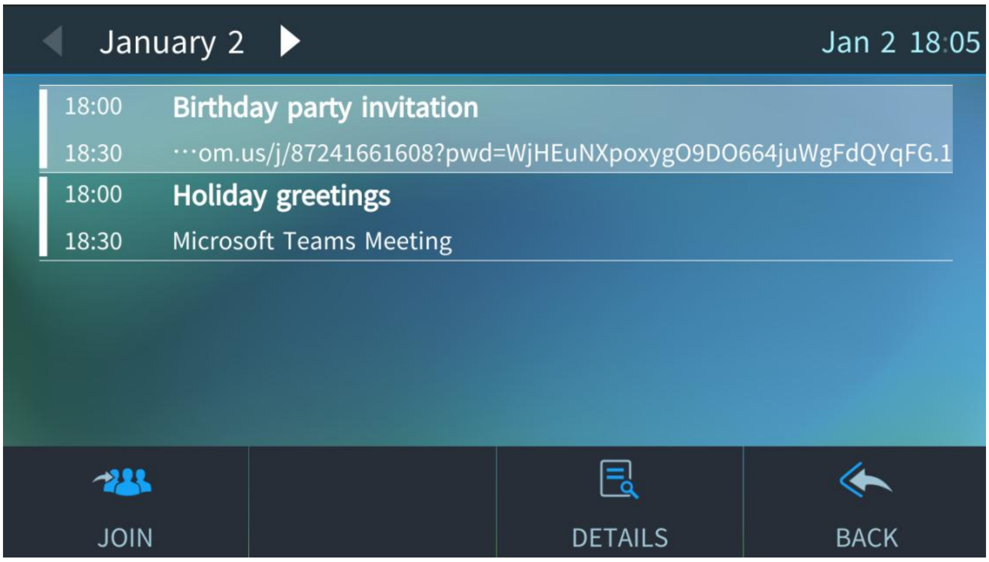


Figure 3: Zoom and Teams meeting on Google Calendar



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.

3.9.2 Resolved Constraints in Version 3.4.10.94

Resolved Zoom Constraints

Table 13: 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.

3.9.3 Known Constraints in Version 3.4.10.94

Known constraints from this release:

None

3.10 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.

3.10.1 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.

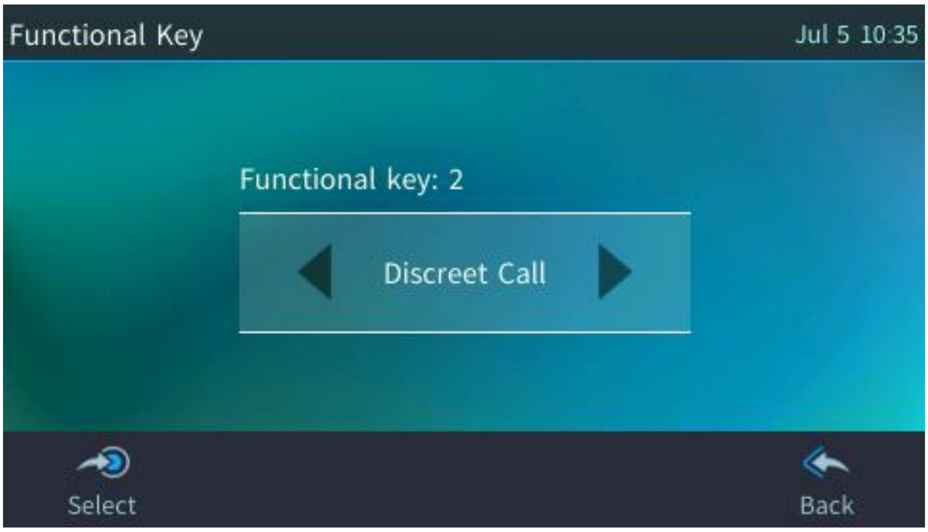
Parameter	Description
personal_settings/functional_key/n/outbound_callerId	Value = <sip:[call queue number]@[zoom server domain]> **Example:** <sip:+12055627793@10000698.zoom.us> **Note:** Applicable only for functional_key/n/type=SIP_ACCOUNT.

- 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	■ 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.

3.10.2 Resolved Constraints in Version 3.4.9.55

Table 14: 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 1st line (Index 0) is enabled, DND of all lines are enabled.
IPPUC-9178	**Zoom:** Call count displays on 1st line key only even when the same line is assigned to several line keys.

3.10.3 Known Constraints in Version 3.4.9.55

Known constraints from this release:

None

3.11 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.

3.11.1 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 15: 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 16: 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: ■ "Switch Hostname" / (6) ■ "Switch IP" /(5) ■ "Switch MAC Address"/(4)

status/diagnostics/lldp/chassis/portIdType	ERS supports the following Port ID subtypes: ■ "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.

3.11.2 Resolved Constraints in Version 3.4.8.808

Table 17: 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.
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.

Incident	Description
IPPUC-9006	Add and set configuration parameter `voip/services/subscribe_call_party_presence` 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 `voip/audio/gain/handsfree/WB_max_volume`.
IPPUC-9184	voip/dialing/unanswered_call_timeout 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).

3.11.3 Known Constraints in Version 3.4.8.808

Known constraints from this release:

Table 18: Known Generic Constraints

Incident	Description
IPPUC-9231	**OpenSpace SIP:** Server feature key synchronization. Changing call forward type on the phone must be done via disabling call forward first. Example: 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.

3.12 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.

3.12.1 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	**OPTIONS** - send SIP OPTIONS message to detect if primary server or primary outbound proxy (server) is active. **REGISTER** - send SIP REGISTER message to detect if primary server or primary outbound proxy (server) is active. (voip/signalling/sip/failback_retry_timeout configuration value must be > 0)

- 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 */voip/services/dnd_per_line_enabled* parameter (mentioned above) is set to 1, leave this '*do_not_disturb*' parameter as the default value (1). However, when the */voip/services/dnd_per_line_enabled* 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.

3.12.2 Resolved Constraints in Version 3.4.8.198.32

Table 19: 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.
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.

Incident	Description
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.

3.12.3 Known Constraints in Version 3.4.8.198.32

Known constraints from this release:

- None

3.13 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.

3.13.1 What's New

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

3.13.2 Resolved Constraints in Version 3.4.8.198.17

Table 20: 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.
IPPUC-8931	**Cloud Paging WEB UI**: "function key" page contains a misspelling.
IPPUC-8905	**Warm (consult) transfer**: Add routingid in the INVITE message.

3.13.3 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

3.14 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.

3.14.1 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

3.14.2 Resolved Constraints in Version 3.4.8.198.13

None

Table 21: 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).

3.14.3 Known Constraints in Version 3.4.8.198.13

Known Constraints from this Release

None

Known Constraints from Previous Releases

None

3.15 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.

3.15.1 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
voip/signalling/sip/sip_outbound_proxy/addr	(IP Address or Host Name)	Primary Outbound proxy address **Attention:** Should be set

Parameter	Value (Default)	Description
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).

3.15.2 Resolved Constraints in Version 3.4.8.198.9

The table below shows the resolved constraints from previous versions.

Table 22: Resolved Constraints

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

3.15.3 Known Constraints in Version 3.4.8.198.9

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.16 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.

3.16.1 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
  ```

3.16.2 Resolved Constraints in Version 3.4.8.198.7

The table below shows the resolved constraints from previous versions.

Table 23: 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.

Incident	Description
IPPUC-8616	[Zoom remote call control] Blind/Warm Transfer from client fails when DUT has multiple lines.
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.

3.16.3 Known Constraints in Version 3.4.8.198.7

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.17 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.

3.17.1 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.

3.17.2 Resolved Constraints in Version 3.4.8.198.1

The table below shows the resolved constraints from previous versions.

Table 24: 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

3.17.3 Known Constraints in Version 3.4.8.198.1

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.18 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.

3.18.1 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
DnD	Auto-Answer	Reject

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

3.18.2 Resolved Constraints in Version 3.4.8.198

The table below shows the resolved constraints from previous versions.

Table 25: 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.

3.18.3 Known Constraints in Version 3.4.8.198

Known Constraints from This Release

None

Known Constraints from Previous Releases

None

3.19 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.

3.19.1 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.

3.19.2 Resolved Constraints in Version 3.4.7.759.8

The table below shows the resolved constraints from previous versions.

Table 26: Resolved Constraints

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

3.19.3 Known Constraints in Version 3.4.7.759.8

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

Table 27: 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 28: Known Constraints from Previous Releases

Incident	Description
None	

3.20 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.

3.20.1 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	■ 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	■ 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	■ 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

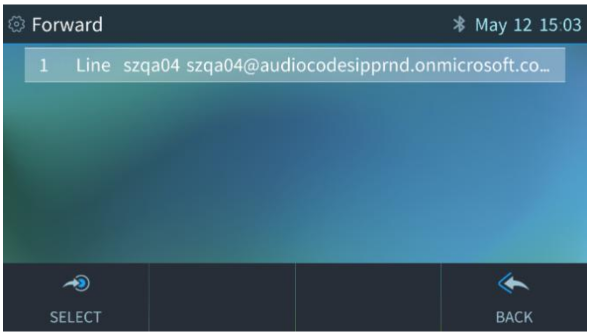
  Introduce PK LED color control

Parameter name	Description
personal_settings/sidecar_mode	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.) ■ [Dynamic] (Default) ■ [STATIC]
personal_settings/sidecar/default_page_line	Determines what page will be displayed by default on the 445HD sidecar. ■ [-1] (Default)(Minimum) ■ [29] (Maximum)
personal_settings/led_color_for_incoming_state	Allows control the LED color for incoming state. ■ OFF ■ RED ■ GREEN (Default) ■ ORANGE
personal_settings/led_color_for_initiated_state	Allows control the LED color for initiated state. ■ OFF ■ RED ■ GREEN (Default) ■ ORANGE

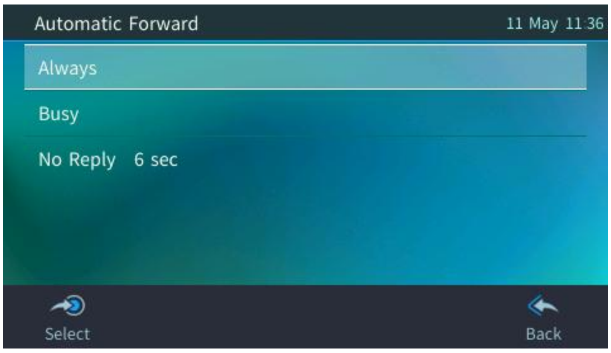
- Call forward UI improvement.

To configure call forwarding:

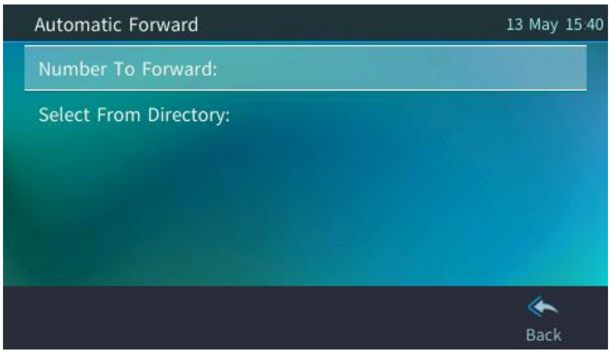
1. Open the **Automatic Forward** screen (**MENU** hard key > **Settings** > **Forward settings** -or- press the **Forward** softkey when the screen is in idle state):



2. Press the **Select** softkey.

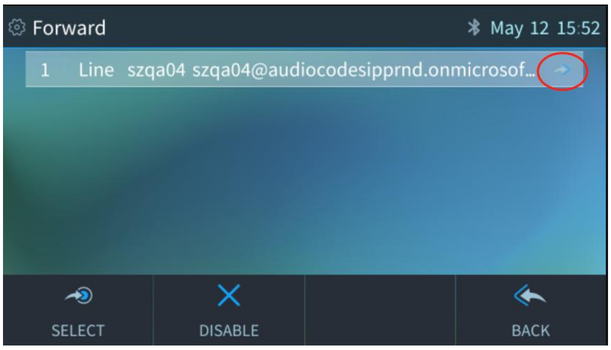


3. In the Automatic Forward screen, select either:
   - **Always:** incoming calls will always be forwarded.
   - **Busy**: incoming calls will be forwarded when the user is using the phone.
   - **No Reply 6 sec**: incoming calls will be forwarded if the user doesn't answer after a specified number 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.

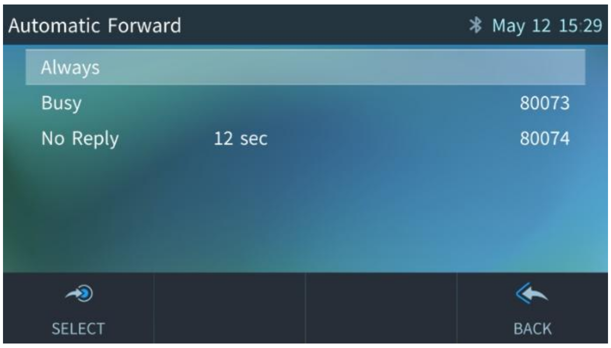


5. Alternatively, navigate to and select the **Select From Directory** option and then choose the contact to whose number you want the calls to be forwarded.
6. Press the **Start** softkey that's then activated. you will be returned to the idle screen; the forward icon  on right will appear); calls will automatically be forwarded to the configured number.

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

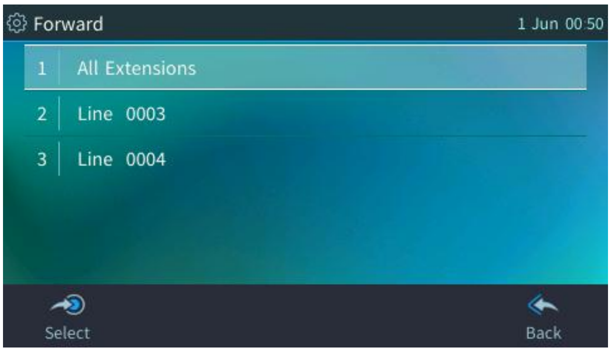


7. You can configure forward 'Busy' and 'No Reply' at the same time. In the example below, incoming calls will be forwarded to 80073 when phone is busy, and if phone is in idle state, calls will be forwarded to 80074 after ringing for 12 seconds.

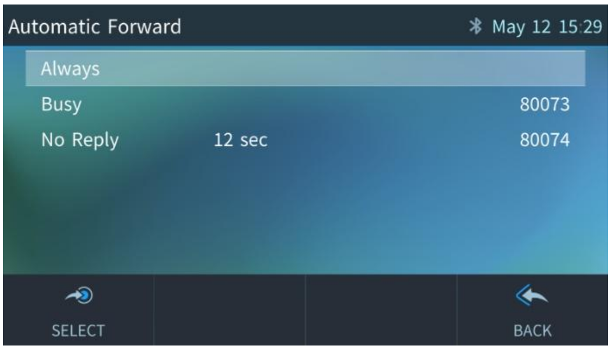


To configure call forwarding on multiple lines:

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



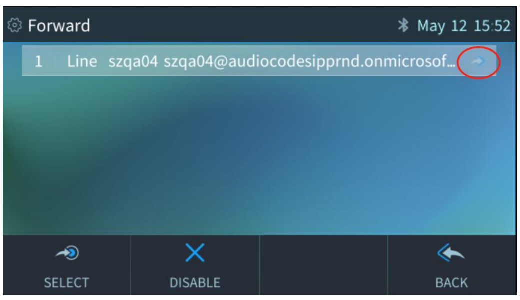
2. Navigate to a line extension on which to configure call forwarding; Select the line; the Automatic Forward screen is displayed.



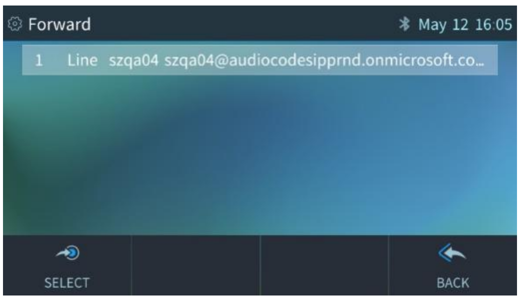
2. Configure call forwarding on that line extension.

To stop call forwarding:

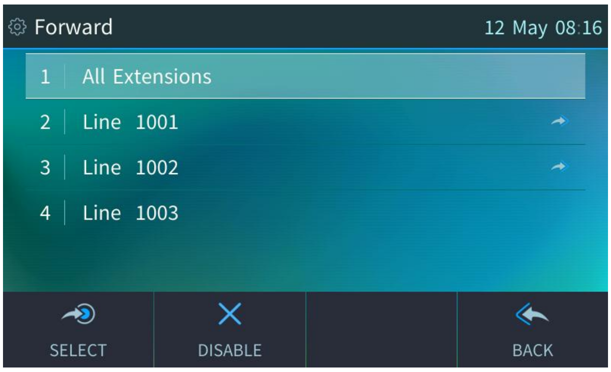
- When the phone is in idle state:
  1. Press the **Forward** softkey
  2. Press the **DISABLE** softkey, the forward icon on right will disappear.



→



- 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).



3.20.2 Resolved Constraints in Version 3.4.7.759.4

The table below shows the resolved constraints from previous versions.

Table 29: 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.
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.

3.20.3 Known Constraints in Version 3.4.7.759.4

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

Table 30: 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 31: 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).

Incident	Description
	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.
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.7 using the following procedure: 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.

Incident	Description
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

3.21 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.

3.21.1 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.

3.21.2 Resolved Constraints in Version 3.4.6.687

The table below shows the resolved constraints from previous versions.

Table 32: Resolved Constraints

Incident	Description
IPPUC-7855	Fixed pause dialing bugs. ■ 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.

3.21.3 Known Constraints in Version 3.4.6.687

None.

Table 33: Known Constraints from Previous Releases

Incident	Description
	IP Phone limitation Zoom: Supports multiple shared lines (to configure up to 15 shared lines)

Incident	Description
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 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.
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure: 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	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).

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

3.22 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.

3.22.1 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').

3.22.2 Known SIP Gateway Constraints

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

Table 34: 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. ■ 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.
-	■ The following features are by design unsupported: • 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 ♦ With Username and Password ♦ Web login • Dynamic location for E911 calls (SIP devices use *static location* for emergency calls, defined by DID number/user and not according to the real location defined by the switch).

Table 35: 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 3.4.6 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 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.
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure: 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.

Incident	Description
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

3.22.3 Resolved Constraints in Version 3.4.6.629

The table below shows the resolved constraints from previous versions.

Table 36: 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.

Incident	Description
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.

3.23 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.

3.23.1 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.

3.23.2 Known Constraints in Version 3.4.6.583

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

Table 37: 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 38: Known Constraints from Previous Releases

Incident	Description
	IP Phone limitation Zoom: Supports multiple shared lines (to configure up to 15 shared lines)

Incident	Description
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 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.
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure: 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	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).

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

3.23.3 Resolved Constraints in Version 3.4.6

The table below shows the resolved constraints in this version.

Table 39: 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.

Incident	Description
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.

3.24 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.

3.24.1 What's New

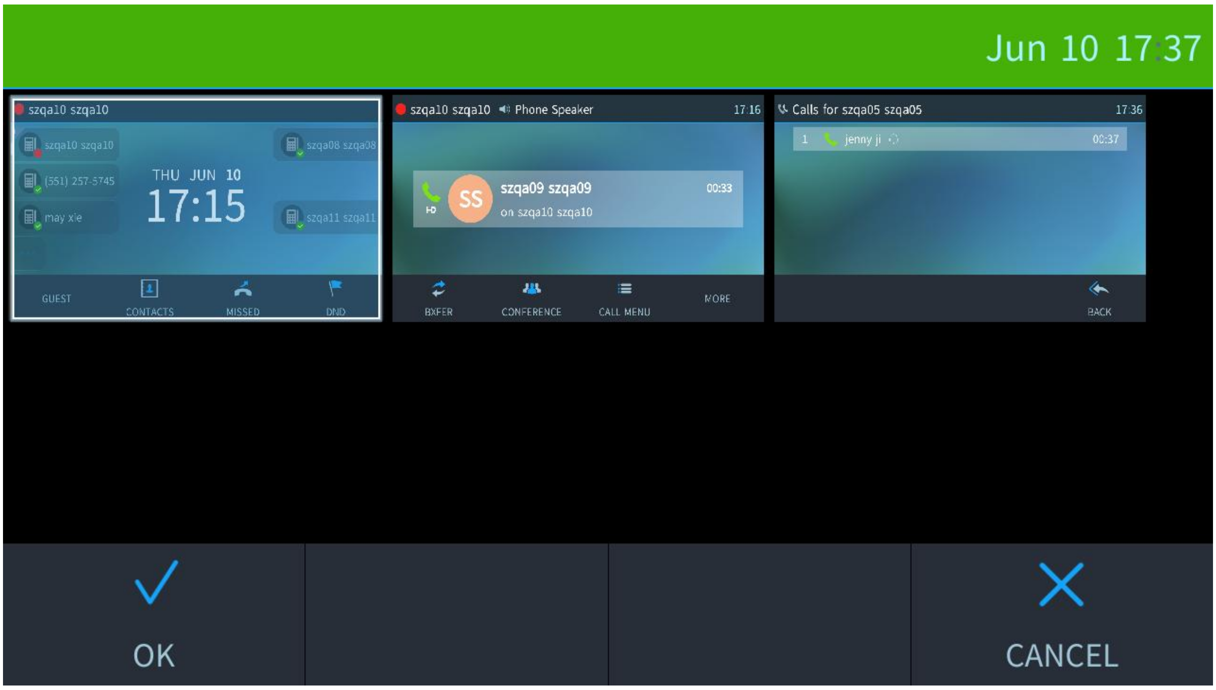
Following are the new features released in Version 3.4.6.565:

- 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.

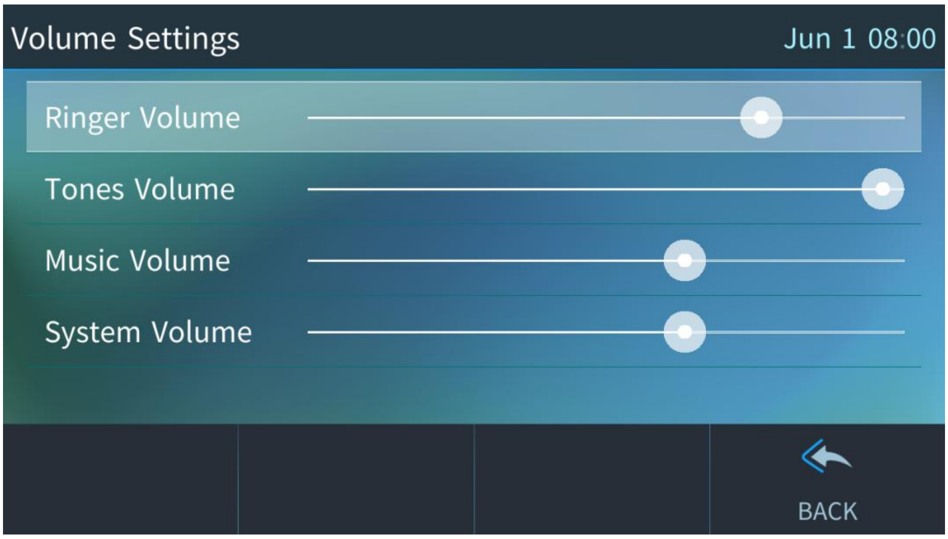
Figure 4: Screen Navigation Switcher



- Volume Mixer Control of multiple streams

  Ability to change the volume of different phone streams (e.g., Ringer, Tones, Music, etc.)

Figure 5: Volume Mixer Control of multiple streams



- New features for Zoom Environment:
    - Supports multiple shared lines (to configure up to 15 shared lines) can be configured in IP-Phone Idle screen and Sidecar (445HD) or Expansion Module (C450/450)

Figure 6: C450 Expansion Module with Multiple Shared Lines

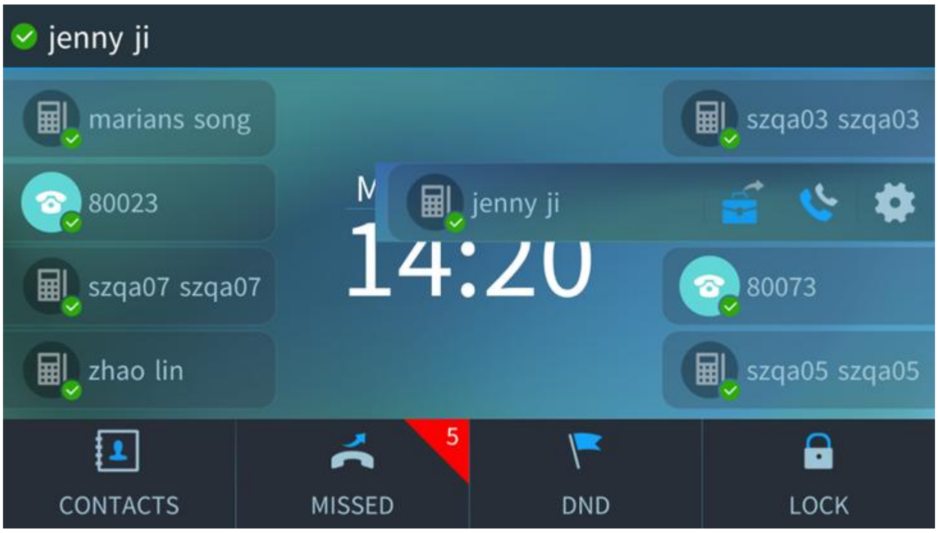
A long press on a line key on the C450/450 expands a menu and allows you to do one of the following (See Figure 7):

 Press to initiate a new call

 Press to open a detailed call list screen (see Figure 8)

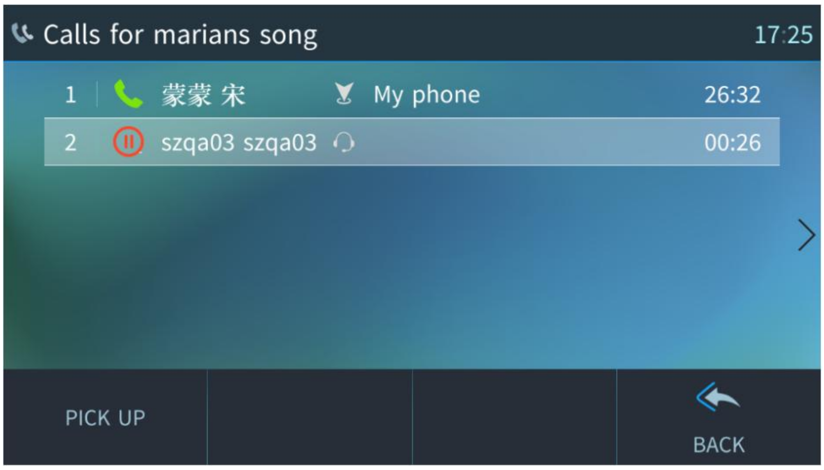
 Press to edit button configuration

Figure 7: IP-Phone idle screen with multiple shared lines



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

Figure 8: 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.

3.24.2 Known Constraints in Version 3.4.6

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

Table 40: 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 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.
IPPUC-6876	"New Device Unlock Code" screen displayed after upgrade from older firmware version to 3.4.6 using the following procedure: 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 41: 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.

Incident	Description
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

3.24.3 Resolved Constraints in Version 3.4.6

The table below shows the resolved constraints in this version.

Table 42: 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.

Incident	Description
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.

3.25 Version 3.4.5.8

3.25.1 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)**.

3.25.2 Known Constraints in Version 3.4.5

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

Table 43: 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 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-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.

Incident	Description
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

3.25.3 Resolved Constraints in Version 3.4.5

The table below shows the resolved constraints in this version.

Table 44: 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.

3.26 Version 3.4.4.1000.52

3.26.1 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.

3.26.2 Known Constraints in Version 3.4.4

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

Table 45: 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 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-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.

Incident	Description
IPPUC-3709	[Virtual Keypad] When long-pressing a key on the Virtual Keypad, the key may get stuck in long-press mode.
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.

3.26.3 Resolved Constraints in Version 3.4.4

The table below shows the resolved constraints in this version.

Table 46: 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.

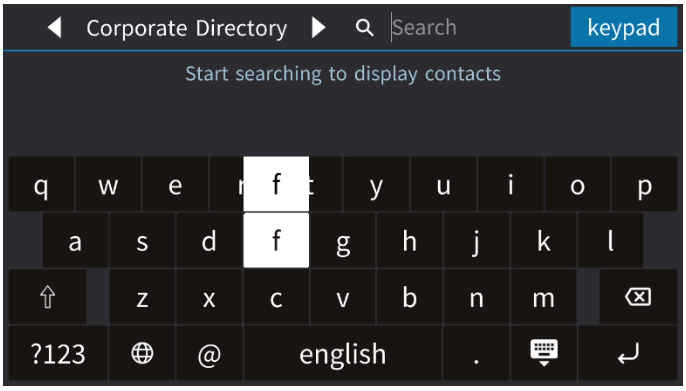
3.27 Version 3.4.4.1000.10

3.27.1 What's New

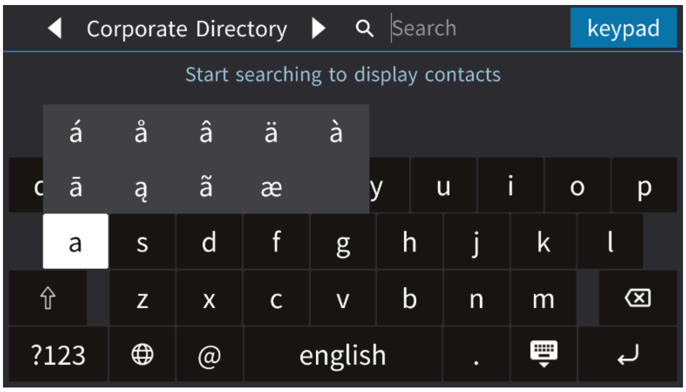
Version 3.4.4 includes firmware build **3.4.4.1000.10**.

Following are the new features released in Version 3.4.4:

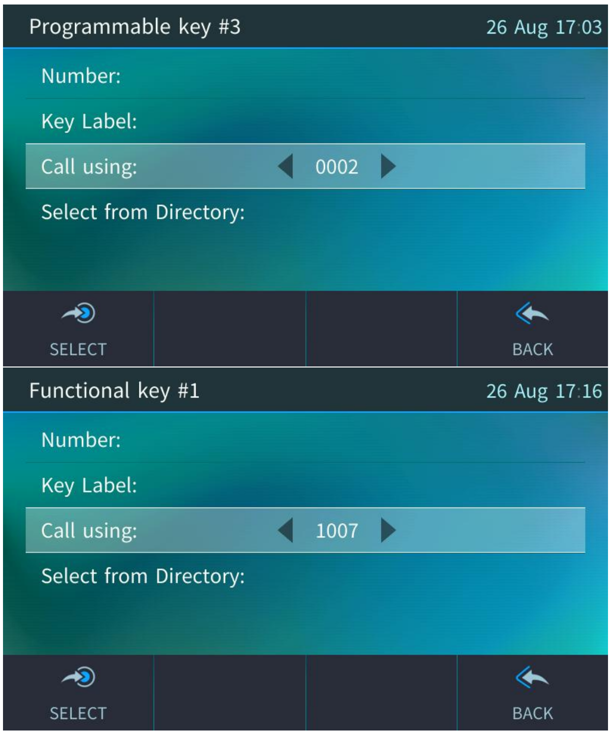
- **OPUS wideband is supported** (Applies to the 445HD and C450HD phone models).
- **Wideband audio for Bluetooth headsets** is supported as Beta. In previous releases, only narrow band audio was supported. Voice quality has consequently improved. (Applies to the 445HD and C450HD phone models).
- The 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.

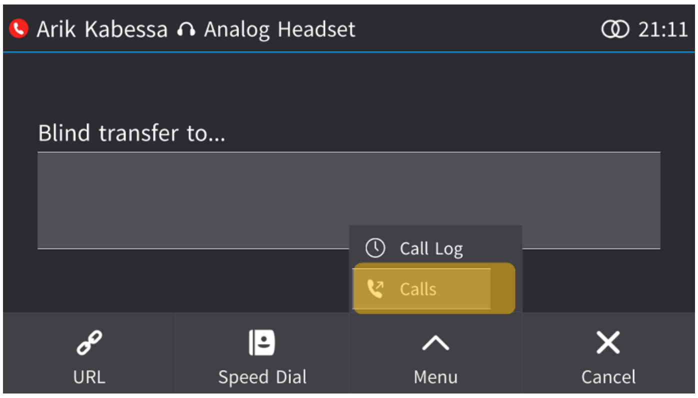


- Users can **configure a Programmable Key or Functional Key for speed dialing to be initiated by a speed dial calling line of their choice**. The feature only applies to phones configured with multiple lines. The feature determines through which line the call goes out when speed dialing. For example: 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.

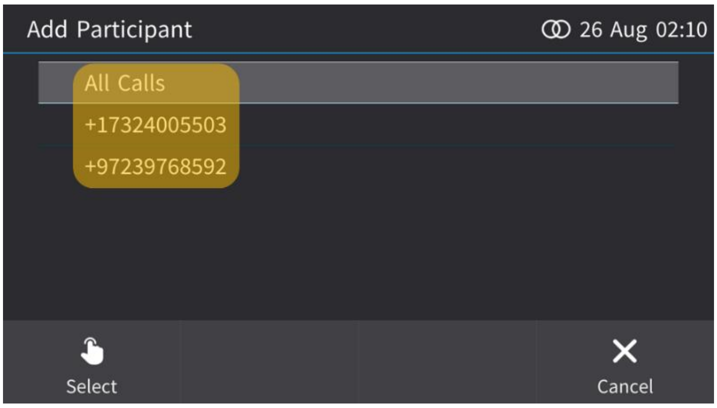


- The **RX50 conference phone supports Voice Quality reports** in compliance with the IETF's RFC 6035, except for the following VQ local metrics that are not provided:
  - Jitter buffer statistics
  - Burst and Gap loss
  - Signal and noise levels
  - Voice Quality estimation
- **Transfer a call to a 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.


Add Participant
26 Aug 02:10
All Calls
+17324005503
+97239768592
Select
Cancel

3.27.2 Known Constraints in Version 3.4.4

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

Table 47: 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 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-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.

3.27.3 Resolved Constraints in Version 3.4.4

The table below shows the resolved constraints in this version.

Table 48: 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.

3.28 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.

3.28.1 Known Constraints in Version 3.4.3

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

Table 49: 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 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-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.

3.28.2 Resolved Constraints in Version 3.4.3

The table below shows the resolved constraints in this version.

Table 50: 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.

3.29 Version 3.4.3.18.70

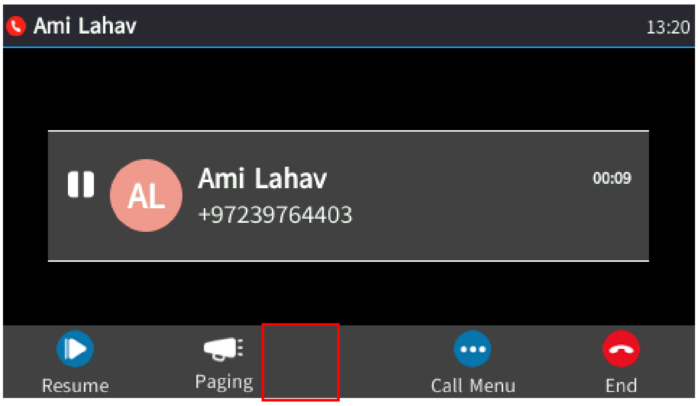
Version 3.4.3 includes firmware build **3.4.3.18.70**.

- **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:

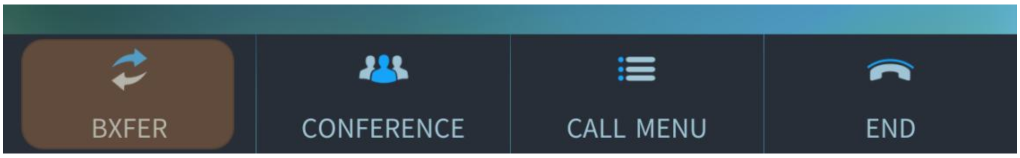
Searching for contacts

URL dialing

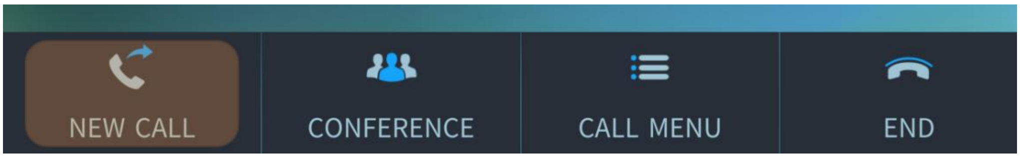
Signing in

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.

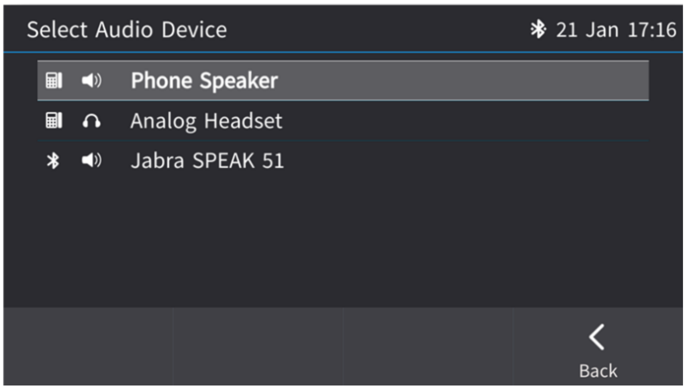


- **Configurable idle screen**

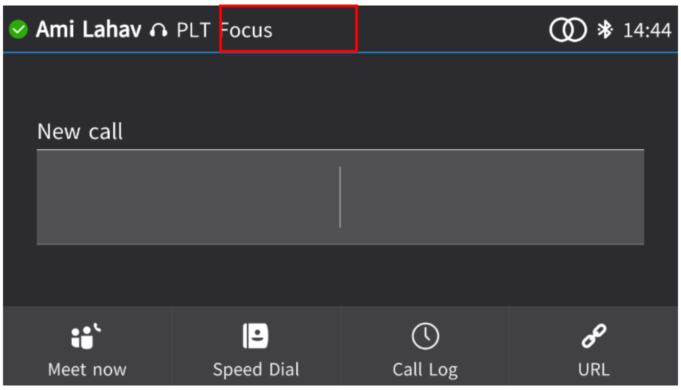
Administrators can customize the idle screen (shown in the figure below) in line with the preferences / requirements of enterprise management and / or the employees.

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.

3.29.1 Known Constraints in Version 3.4.3

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

Table 51: 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 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-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.

3.29.2 Resolved Constraints in Version 3.4.3

The table below shows the resolved constraints in this version.

Table 52: 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.

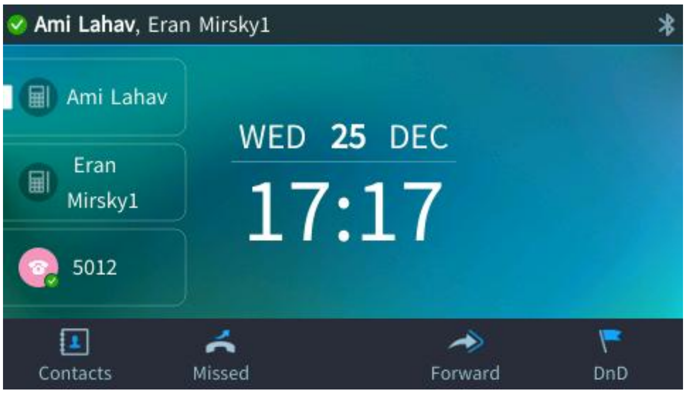
3.30 Version 3.4.1.565

3.30.1 What’s New

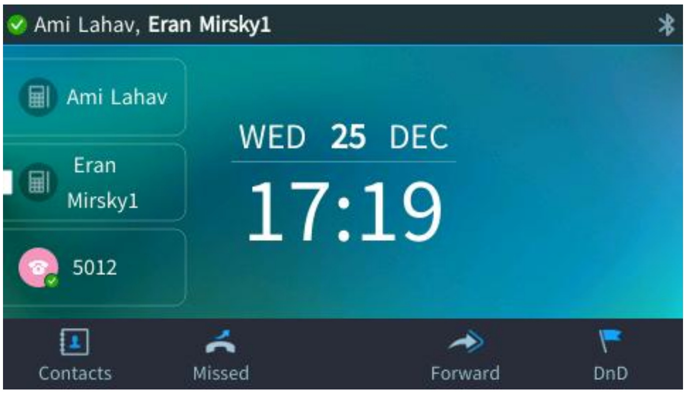
Version 3.4.1 includes firmware build **3.4.1.565**.

- **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'.

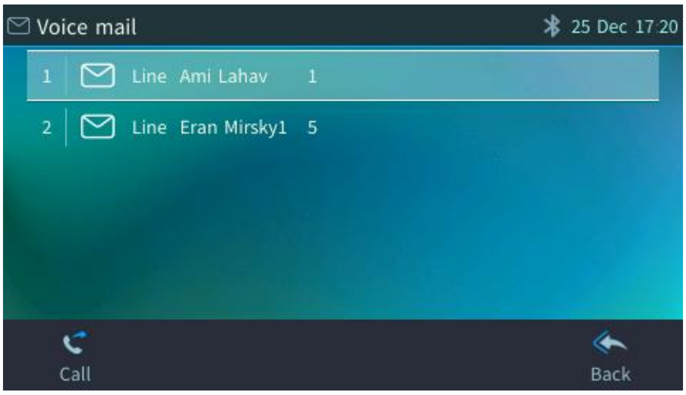
- **Support for multiple lines**. Account status (default line, Call Forward and DND) is visually indicated for each account adjacent to the line's Programmable Key. To decide which line will be used:
  - Press the hard key next to the line.



A rectangle icon indicates the current line:



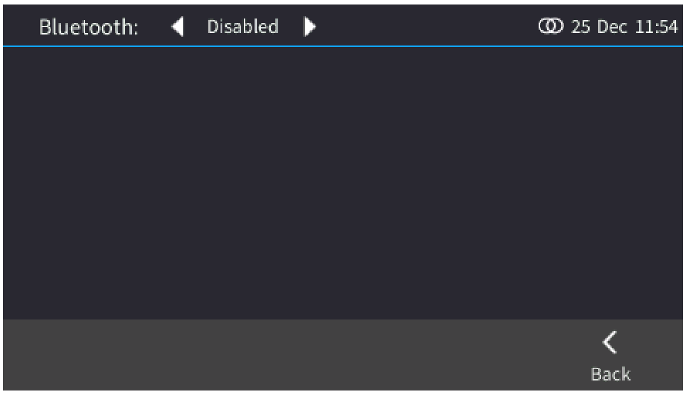
  - Press the voicemail hard key; the option to switch between lines is displayed.



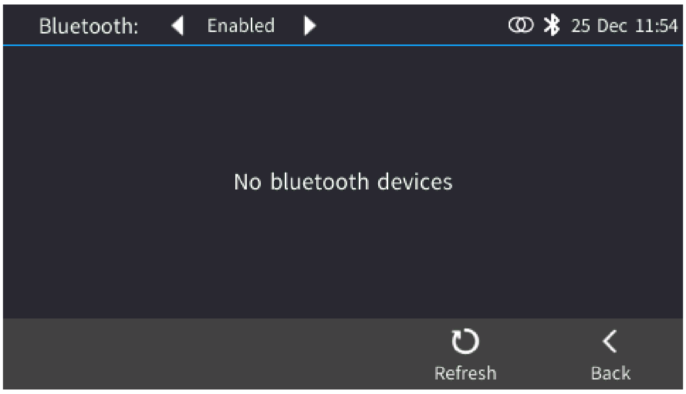
  - The same applies to call Forward (see the figure below) and DND.

- **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.

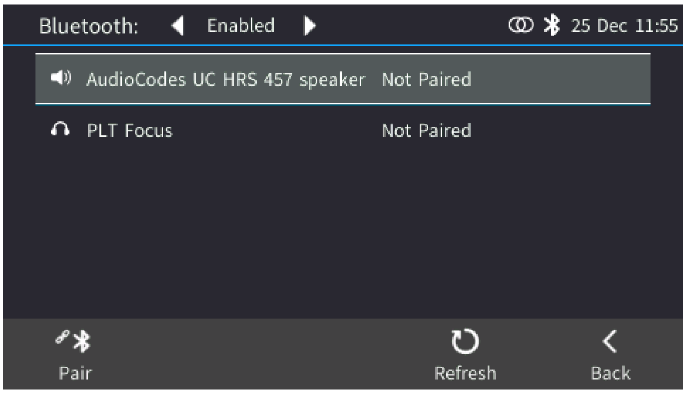
  To enable Bluetooth on the phone:
    - Access the Bluetooth screen (**Menu** > **Settings**):



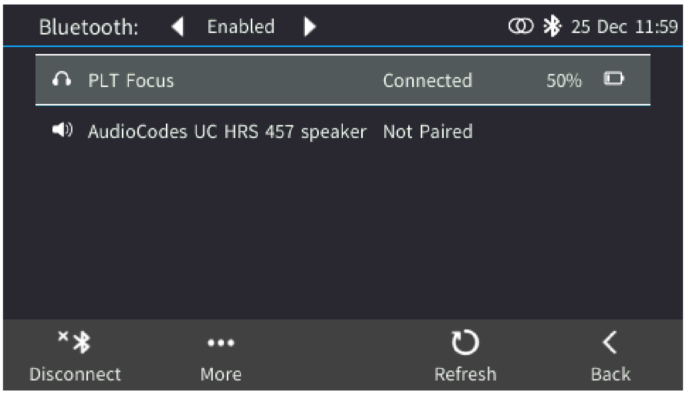
    - Press the right/left rim of the navigator key to configure **Enabled**:



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



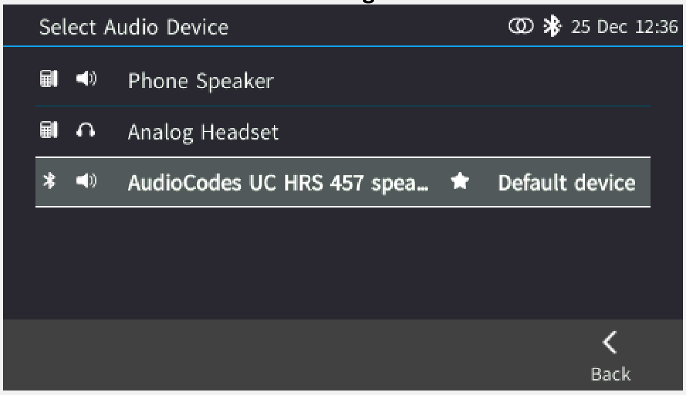
- Press the **Pair** softkey to pair the device. After pairing is complete, the phone displays 'Connected'.



- 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).

3.30.2 Known Constraints in Version 3.4.1

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

Table 53: 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 öß§üä are not supported in password usage.
IPPUC-2894	[Local Conference] The phone cannot hear DTMF when DTMF Transport mode is via SIP.

3.30.3 Resolved Constraints in Version 3.4.1

The table below shows the resolved constraints in this version.

Table 54: 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.

3.31 Version 3.4.0.14

3.31.1 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.

3.31.2 Known Constraints in Version 3.4.0.14

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

Table 55: 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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