AudioCodes 400HD Series of High Definition IP Phones

HD VoIP

400HD Series IP Phones for Microsoft® Skype™ for Business

Release Notes

AudioCodes 400HD Series IP Phones for Microsoft Skype for Business

Version 3.0.1

Document #: LTRT-08300

Microsoft Partner

Gold Communications

Skype for Business

AudioCodes

HD VoIP Sounds Better

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Trademarks

WEEE EU Directive

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Customer Support

Customer technical support and services are provided by AudioCodes or by an authorized AudioCodes Service Partner. For more information on how to buy technical support for AudioCodes products and for contact information, please visit our Web site at www.audiocodes.com/support.

Abbreviations and Conventions

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

Related Documentation

Document Name
405HD IP Phone User's Manual
420HD IP Phone User's Manual
430HD and 440HD IP Phone User's Manual
400HD Series IP Phone Administrator's Manual
405HD IP Phone Quick Guide
420HD IP Phone Quick Guide
430HD IP Phone Quick Guide
440HD IP Phone Quick Guide
400HD Series IP Phones for Lync Hosting Partner (LHPv2) Environment Configuration Note
IP Phone Management Server Administrator's Manual
EMS and SEM Server IOM Manual
EMS User's Manual

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1 Introduction

This document describes the new features, known constraints, and resolved constraints of AudioCodes' 400HD Series Skype for Business-compatible IP Phones for Version 3.0.1.

Note: Microsoft rebranded *Lync* as *Skype for Business* so when the term *Skype for Business* appears in this document, it also applies to Microsoft Lync.

1.1 Overview of the 400HD Series of IP Phones

AudioCodes' 400HD Series of Skype for Business-compatible IP phones offer enhanced voice quality and clarity for users of Microsoft Unified Communications. The phones' wide range of essential business features, reliability and certified interoperability make them the perfect choice for any Skype for Business deployment, from small businesses up to large multi-site enterprises.

AudioCodes is a leading Microsoft partner with a complete offering of voice-enabling solutions for the Microsoft Skype for Business Unified Communications market. In addition to IP phones, AudioCodes' portfolio for Microsoft Skype for Business includes Media Gateways, Survivable Branch Appliances (SBA), Enterprise Session Border Controllers (E-SBCs), and Call Recording.

1.2 Specifications

The table below summarizes the software specifications of AudioCodes 400HD series IP Phones for Microsoft Skype for Business.

Table 1-1: Software Specifications of 400HD Series IP Phones for Microsoft Skype for Business

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 ▪ Microsoft Skype for Business (previously Microsoft Lync) ▪ MS-ICE2 ▪ 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 to comply with Microsoft Skype for Business security requirements ▪ OpenSSL 1.0.1m, supporting SHA2 algorithms ▪ OVR (One Voice Resiliency)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.722 8000 / G.722 16000), RTA, SILK 8000/SILK 16000 (420HD GbE / 405HD GbE / 430HD GbE /440HD/450HD only), LN. ▪ Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length ▪ Adaptive Jitter Buffer 300 msec ▪ Voice Activity Detection ▪ Comfort Noise Generation

Feature	Details
	▪ Packet Lost Concealment ▪ RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC 3711) ▪ DTMF Relay (RFC 2833)

Telephony Features	▪ BLF presence on buttons; capability for 18 Multiple Points of Presence (MPOPs), including Skype for Business clients. (420HD non-GbE supportS up to 5 MPOPs).
	▪ Busy on Busy
	▪ Call Park(phone can park up to 5 calls to a parking lot)
	▪ Group Call Pickup
	▪ Call Hold / Un-Hold
	▪ Call Transfer; the hard TRANSFER key's default functionality (Blind Transfer) can be changed to Consultative Transfer.
	▪ Multi-Party Skype for Business Remote Conferencing utilizing CCCP (Centralized Conference Control Protocol)
	▪ Redial
	▪ Caller ID Notification
	▪ Call Waiting Indication, including Caller ID
	▪ Message Waiting Indication (including MWI LED)
	▪ Local and Corporate Directories
	▪ T9 predictive text for Corporate Directory search
	▪ Automatic Answering (Alert-Info header and "talk" event)
	▪ Automatic On-hook Dialing
	▪ Call Logs: Missed/Received Calls and Dialed Numbers
	▪ Speed Dials
	▪ Boss Admin: Incoming calls to a delegated line are displayed in the sidecar (440HD only)
	▪ URL Dialing
	▪ Call Forward (Do not forward, Forward to voice mail, Forward to a number)
	▪ Dial plan (supports normalization rules downloaded from the Skype for Business server via in-band provisioning)
	▪ Paging w/without Barge-in. Configurability of special keys as paging group dials.
	▪ Better Together over Ethernet (BToE) compatible with Microsoft Skype for Business
	  ✓ Automatically pairing the phone with the PC/laptop based Skype for Business client.
	  ✓ Video calls: Voice routed to phone; phone performs as a USB device.
	  ✓ Switching audio devices (when automatic pairing).
	▪ Voicemail (including capability to secure user access with PIN code)
	▪ Visual Voice Mail (all phones except 420HD)
	▪ Integration with Microsoft Exchange Server (Calendar), including meeting reminders (all phones except 420HD)
	▪ Automatic device lock
	▪ Handles up to 8 concurrent calls (450HD)
	▪ Merge option: Two separate calls can be merged into one conference
	▪ Integrated Skype for Business 'Favorites' (all phones except 420HD)
	▪ Location service
	▪ Emergency dial (911 service, etc.)

Feature	Details
Configuration / Management	▪ Device update: Skype for Business server updates the phone's firmware version if different ▪ Quality of Experience (QoE) reports sent to Microsoft's SQL server ▪ Phone User Interface Language Support (Various Languages) ▪ Web-based Management (HTTP/HTTPS) with fully integrated login ▪ IP Phone Management Server (EMS module) ▪ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and proprietary configuration file upgrade ▪ In-Band Provisioning ▪ 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 the URL of the Certificate Provisioning service ▪ DHCP option (2) for the Time Zone Offset ▪ Skype for Business Contacts ▪ LDAP (Lightweight Directory Access Protocol) ▪ Private Labeling Mechanism ▪ Configuration file encryption (Entire file and individual parameters)
Debugging Tools	▪ System Logging (Syslog) ▪ Monitoring (Ping and Traceroute) ▪ DSP Recording ▪ Crash Dump ▪ Port Mirroring ▪ Tracing ▪ Core Dump ▪ Log upload to Microsoft server (certification for 3rd party Skype for Business clients)
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 ▪ Japanese ▪ Slovak ▪ Czech ▪ Latvian (contact person information)
420HD Hardware	▪ **LCD screen:** Graphic LCD (132 X 64) ▪ Connectors interfaces: ✓ 2 x RJ-45 ports (10/100/1000 BaseT Ethernet) for WAN and LAN ✓ RJ-9 port (jack) for Headset

Feature	Details
	✓ RJ-9 port (jack) for Handset ▪ 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 1: IEEE802.3af (optional) ▪ Keys: ✓ 4 x softkeys ✓ VOICE MAIL message hotkey ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE ✓ TRANSFER ✓ VOLUME control key ✓ HEADSET ✓ SPEAKER
420HD Headset Compatibility	▪ For a comprehensive list of supported Jabra headsets, see the Jabra [Headset Compatibility Guide] ▪ For a comprehensive list of supported Plantronics headsets headsets, see http://www.plantronics.com/us/compatibility-guide/#/search/
405HD Hardware	▪ **LCD screen:** Graphic LCD (128 X 48) ▪ Connectors interfaces: ✓ 2 x RJ-45 ports (10/100/1000 BaseT Ethernet) for WAN and LAN ✓ RJ-9 port (jack) for Headset ✓ RJ-9 port (jack) for Handset ▪ 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 1: IEEE802.3af (optional) ▪ Keys: ✓ 4 x softkeys ✓ VOICE MAIL message hotkey ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE ✓ VOLUME control key ✓ HEADSET ▪ SPEAKER
405HD Headset Compatibility	▪ For a comprehensive list of supported Jabra headsets, see the Jabra Headset Compatibility Guide ▪ For a comprehensive list of supported Plantronics headsets headsets see http://www.plantronics.com/us/compatibility-guide/#/search/ ▪ These include: ✓ Jabra UC-150

Feature	Details
	✓ Jabra Speak 510+ ✓ Jabra Speak 410 ✓ Jabra MOTION OFFICE ✓ Jabra PRO 9470 ✓ Microsoft LX-3000 ✓ Plantronics C-310M ✓ Plantronics C-320M ✓ Plantronics HW720 ✓ Jabra Pro 920 EHS wireless headset ✓ Jabra Pro 9450 EHS wireless headset
430HD and 440HD Hardware	▪ **LCD screen:** Graphic LCD (132x64) monochrome (a 440HD phone hardware revision featuring an LCD resolution of 256x128 is supported from v2.0.13; a 430HD phone hardware revision featuring an LCD resolution of 256x128 is supported from 3.0.1) ▪ **BLF screen:** Graphic LCD (60x376) monochrome (applies only to the 440HD model) ▪ 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 ✓ 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 x softkeys ✓ VOICE MAIL message hotkey (including LED) ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE (including LED) ✓ TRANSFER ✓ VOLUME control key ✓ HEADSET (including LED) ▪ SPEAKER (including LED)
430HD and 440HD Headset Compatibility	▪ For a comprehensive list of supported Jabra headsets, see the Jabra Headset Compatibility Guide ▪ For a comprehensive list of supported Plantronics headsets headsets see http://www.plantronics.com/us/compatibility-guide/#/search/ ▪ For a comprehensive list of supported VXi products, see http://www.vxicorp.com/compatibility_guide/ ▪ Also the following which aren't documented online yet: ✓ Jabra UC-150 ✓ Jabra Speak 510+ ✓ Jabra Speak 410 ✓ Jabra MOTION OFFICE ✓ Jabra PRO 9470 ✓ Microsoft LX-3000

Feature	Details
	✓ Plantronics C-310M ✓ Plantronics C-320M ✓ Plantronics HW720 ✓ Jabra UC-550 ✓ Jabra Pro 920 EHS wireless headset ✓ Jabra Pro 9450 EHS wireless headset
450HD Hardware	▪ Large (800 x 400), graphical, high-resolution, 5-inch color touch (TFT) screen with an intuitive touch-oriented user interface design. ▪ 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 ✓ 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 3: IEEE802.3af (optional) ▪ Keys: ✓ 8 x softkeys ✓ VOICE MAIL message hotkey (including LED) ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE (including LED) ✓ TRANSFER ✓ VOLUME control key ✓ HEADSET (including LED) ▪ SPEAKER (including LED)
450HD Headset Compatibility	▪ For a comprehensive list of supported Jabra headsets, see the Jabra Headset Compatibility Guide ▪ For a comprehensive list of supported Plantronics headsets headsets see http://www.plantronics.com/us/compatibility-guide/#/search/ ▪ For a comprehensive list of supported VXi products, see http://www.vxicorp.com/compatibility_guide/ ▪ Also the following which aren't documented online yet: ✓ Jabra UC-150 ✓ Jabra Speak 510+ ✓ Jabra Speak 410 ✓ Jabra MOTION OFFICE ✓ Jabra PRO 9470 ✓ Microsoft LX-3000 ✓ Plantronics C-310M ✓ Plantronics C-320M ✓ Plantronics HW720 ✓ Jabra UC-550 ✓ Jabra Pro 920 EHS wireless headset ✓ Jabra Pro 9450 EHS wireless headset

1.3 Supported Models

The table below summarizes the supported AudioCodes 400HD Series IP phone models.

Table 1-2: Supported 400HD Series IP Phone Models

Part Number	Product Description
UC405HDEG	Skype for Business 405HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC405HDEPSG	Skype for Business 405HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC420HDE	Skype for Business 420HD IP Phone PoE Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC420HDEG	Skype for Business 420HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC420HDEW	Skype for Business 420HD IP Phone PoE White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC420HDEPS	Skype for Business 420HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC420HDEPSG	Skype for Business 420HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC420HDEPSW	Skype for Business 420HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
UC430HDE	Skype for Business 430HD IP Phone PoE Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC430HDEG	Skype for Business 430HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC430HDEW	Skype for Business 430HD IP Phone PoE White 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC430HDEPS	Skype for Business 430HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
UC430HDEPSW	Skype for Business 430HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)
UC430HDEPSG	Skype for Business 430HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)

Part Number	Product Description
UC440HDEG	Skype for Business 440HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
UC440HDEWG	Skype for Business 440HD IP Phone PoE GbE White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
UC450HDEG	Skype for Business 450HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 800x480 5" Color Touch LCD and Power over Ethernet (PoE)
UC440HDEPSG	Skype for Business 440HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
UC440HDEPSWG	Skype for Business 440HD IP Phone PoE GbE and external power supply White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
UC450HDEPSG	Skype for Business 450HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 800x480 5" Color Touch LCD and Power over Ethernet (PoE)

2 Version 3.0.1

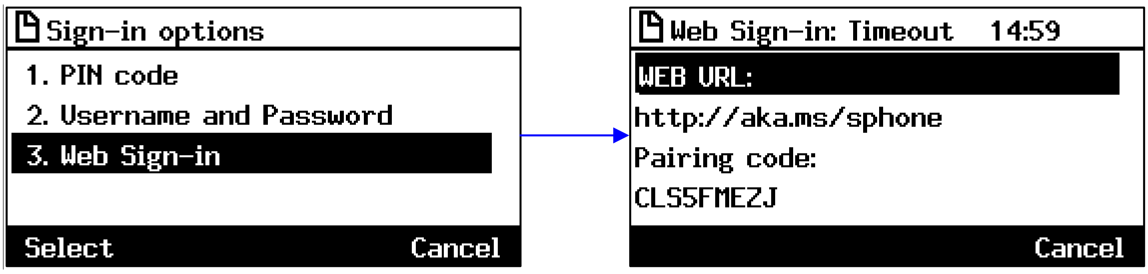
Note: Version 3.0.1 includes the following firmware builds:

- 3.0.1.**212** (all 400HD Series IP Phones except the 450HD phone)
- 3.0.1.**89.335** (only the 450HD phone)

2.1 What's New in Version 3.0.1

The following new features are introduced in Version 3.0.1:

- **Cloud PBX Web Sign-in, a.k.a. Device Pairing - connectivity to Microsoft's Cloud PBX**, Microsoft's cloud-hosted version of enterprise voice. AudioCodes' IP phone features a sign-in option allowing users to connect to Microsoft's Cloud PBX: **Web Sign-in**.



Note: This sign-in option applies only to Microsoft Cloud PBX users

The option exempts users from having to laboriously key in their user name and password using the phone keypad in order to sign in. If the option is selected, a URL and a Pairing Code are displayed, as shown in the figure above. Users must then point their browser to the URL and enter the Pairing Code in the Microsoft web page. Sign-in to Microsoft's Cloud PBX is then performed.

- **Capability to add and delete contacts to/from Skype for Business 'Favorites'**. Users can add a person to the Favorites group after (for example) a call with the person is logged. A maximum of 1,000 people can be added to the Favorites group. Users can delete the added person if necessary.
- **Join Meeting Enhancement**[1]. With this version, users can enjoy a 'join' option for online meetings even if the TNEF option is disabled on Exchange; Exchange enables sharing information between federated parties; by default, the sharing option is disabled (TNEF = disabled); the phone relies on the Join Skype Meeting link in the calendar meeting request. The join link is usually found in the body of a meeting request. However, the phone depends on this link to be present in the MAPI properties of the message. When this meeting request is sent to remote organizations (Skype for Business federated partners), the remote organization's phone by default will not show the meeting join link in the calendar because Outlook and Microsoft Exchange do not use Transport Neutral Encapsulation Format (TNEF) to package information for sending messages across the Internet. TNEF, which encapsulates MAPI message properties, is by default disabled for messages sent externally from an Exchange organization.

[1] Will be supported in next 450HD version release. Not supported in the current 450HD version release.

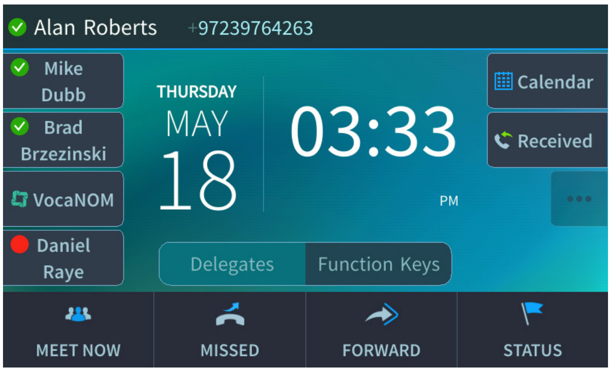
- **Secured connection to IP Phone Manager**. The connection between the phone and the IP Phone Manager can now be fully secured using HTTPS.

> **Note:** To increase the security level, it's advisable to block any HTTP connection using the configuration file parameter *security/web/https_only*.

- **New Boss-Admin experience exclusive to 440HD**. Boss and Admin can utilize the 440HD sidecar to present active and held calls in the queue.
  - Admin can see each Boss queue on the sidecar
  - Boss can see all Boss calls in the queue on the sidecar
  - A mix of Admin and Boss can be also used in this mode
  - Users can still use the sidecar for Speed Dial/BLF. The upper sidecar key allows users to switch between BLF and Boss/Admin queues.
- **Boss Privacy mode**. Conceals a remote caller's ID from the Admin's (delegate's) phone in order to protect their Boss's privacy. The feature is disabled by default; the network administrator can enable it. The feature applies to the phone's sidecar and to the Call List in the phone's screen.
- **BToE default pairing mode** is now automatic. As of this version, the default BToE pairing mode is *automatic*. Using the *manual* pairing option requires changing the phone's configuration. The new default pairing mode allows users to derive maximum benefit from the BToE feature.

> **Note:** As a result of this change (BToE default pairing mode now Automatic), the **BToE** softkey is no longer displayed after pressing the MENU hard key.

- 450HD phone's idle screen now displays both Function Keys (i.e., Speed Dial/BLF) and Delegates (i.e., delegated users):



- **Pause dialing**[1]. Pause dialing can be configured for a Speed Dial in order to create a time break, typically needed when configuring a Speed Dial to dial a destination extension number that is behind an Interactive Voice Response (IVR) system.
- **Enhanced Visual Voicemail (VVM)**. The phone updates the MWI LED and the number of messages even if the configuration between Exchange and Skype Online has not been performed correctly. This is relevant to users whose Skype or Exchange is online.
- **Users can make new calls during incoming calls**[1]. This feature is now supported on all phone models.
- **Headset ringer activated on incoming calls**[1]. The headset plays a ringer when calls come in, in addition to the phone's ring.
- **New language support**. Korean is now supported on the 405HD phone.
- **Improved debugging**[1]. DSP Packet Recording can be enabled on the fly, without requiring the network administrator to reset the phone.

- **Ability to change the DTMF level**. Network administrators can now change the DTMF level with a new configuration file parameter *voip/audio/gain/dtmf_tone_signal_level*.
- **Improved Common Area phone**. The DND (Do not Disturb) key and the Call Forward key were removed from the Common Area phone's menus to prevent Common Area phone users from making the phone 'unavailable'.
  For **backward compatibility**, a new 'voip/common_area/enhanced_mode' parameter can be set to **0** to allow Common Area phone users to be able to view the Call Forward key if necessary.
- **Improvement to the 'Locking / unlocking a paired phone' feature**. Starting from this release, **a paired phone is automatically locked 10 seconds after the PC with which it is paired, is locked**. If the user continues using the phone within 10 seconds after the PC is locked, the phone is locked 10 seconds after being in idle state. In version 3.0, when the user locked the paired PC, the phone was locked after a timeout preconfigured in the Skype for Business server lapsed.
  If the user's phone was automatically paired (by connecting its PC port to the PC/laptop 'behind' it) and if the PC/laptop was active (not locked), the phone could not be manually locked. The user could manually lock it only after locking the PC/laptop.

2.2 Resolved Constraints in Version 3.0.1

The table below shows the constraints that were known to exist in previous releases but which are now resolved in Version 3.0.1.

Table 2-1: Resolved Constraints in Version 3.0.1

Incident	Description
105871	[USB headset] Sometimes there is a voice delay on the second call.
105922	Delegate settings are removed from the Skype user if Call Forward is disabled from the phone.
104727	ToS (Type of Service) default value was set to 0xA0.
99962	[Presence] The phone sometimes shows status as 'Away' when the user is in fact available.
106086	[Multi-Party Skype for Business Remote Conferencing] Meet Now/Join does not function with extensions that are not in E164 format.
104851	[Multi-Party Skype for Business Remote Conferencing] The participant can unmute themselves when the conference host presses Mute All when the participant's status is 'Mute'.
103803	[Multi-Party Skype for Business Remote Conferencing] Phones added to the conference fail to mute themselves.
105796	The phone's user interface occasionally gets stuck on the registration message and cannot recover after pool failover. Occurs only to users who are signed in with extension number and PIN code.
102259	[EHS] No audio on Electronic Hook Switch (EHS) can be heard when answering an incoming call using the headset key.
106031	[Response Group] Attended transfer to Response Group fails.
106155	[Device Lock] 440HD phone's sidecar keys can be configured when the phone is locked.
104733	The phone doesn't support DHCP Option 42 and Option 4 (Time server) which is set with hostname. When DHCP is set with IP address, the phone successfully acquires its IP address from DHCP).
103541	The phone displays an incorrect time format when the time format is set to **12h** and the date format is set to **American**.
104465	[Transfer] Semi-attendant transfer fails when pressing the TRANSFER hard key on the phone instead of the **Dial** softkey.
104147, 104580	[Transfer] The **Transfer** softkey is missing when the phone is configured for semi-attendant transfer.
104672	[Transfer] Semi-attendant transfer cannot be performed with the ‘New Call’ option.
104927	[USB headset] Voice may not be heard when switching audio device from headset to speaker.
104469	[Boss-Admin] Admin can't make a blind transfer to the Boss from Admin's own directory.
104992	The phone removes the held far-end user from other AudioCodes phone during a call shuffle (when switching between two calls).
103883	The handset/speaker/headset volume is not saved after restarting the phone.
104728	Users who sign in with PIN code cannot perform a search for a contact in the Corporate Directory.
104539	[Paging] A paged call can be ended only by pressing the **End** softkey or by on-hooking the handset. Pressing the speaker hard key does not end the paged call.

Incident	Description
103640	In a conference call, when the phone performs a far mute, an unmute, and then a far mute, the popup message 'A presenter has muted you' is not displayed on the far phone.
103995	[405HD model phone] Korean Language is not yet supported in this version release.
100705	[USB headset] Occasionally, the phone's user interface performs slowly.
107004	[USB headset] Incorrect volume scale for some USB headset types.
107101	[USB headset] The Plantronics headset becomes unresponsive after several clicks on the new **Call/Disconnect** button.
107100	[USB headset] The Plantronics headset does not receive a dial tone the first time a new call is initiated from the headset controller.

2.3 Known Constraints in Version 3.0.1

The table below shows the constraints known to exist in Version 3.0.1.

Table 2-2: Known Constraints in Version 3.0.1

Incident	Description
-	420HD IP phone does not support: ▪ Exchange integration (Calendar) ▪ Visual Voice Mail ▪ Outlook contacts and Skype for Business 'Favorites' contacts
-	450HD IP phone does not support: ▪ Boss-Admin ▪ Multicast Paging Support for these is planned for forthcoming 450HD releases.
106815	[Boss-Admin] Second pickup sometimes fails, i.e., if Admin picks up a Boss call and then Boss tries to pick up the call back).
106161	[Boss-Admin] Skype user whose name starts with a small letter and configured as Admin, it's displayed on the Boss phone as a URI address rather than with the display name.
106517	[New 440HD Boss-Admin] When there is a crossed delegation, i.e., when Admin 1 is Boss of Admin 2 and Admin 2 is Boss of Admin 1), the Boss doesn't see the call when Admin answers the call.
106254	[New 440HD Boss-Admin] Retrieving a parked call on behalf doesn't appear on the sidecar. If the call is not parked, there is no issue.
105852	[BToE] In the scenarios listed below, the phone performs as a PC playback device and not as a standalone phone. Some call management capabilities consequently go missing, e.g., hold, transfer, conference roster screen. In these scenarios, a regular call may start similarly to a video call, i.e., the phone is used as the PC playback device. Example scenarios: ▪ RGS/Delegate call made on behalf from the PC client ▪ Call from IM (Instance Message) that was opened before the phone was paired. This is aligned with Microsoft's BToE implementation.
106432	[BToE with Delegation] When a video call comes in to the Boss, a paired Admin phone will not ring but can accept the call. This is aligned with Microsoft's BToE implementation of video calls.
105172	[Skype for Business 'Favorites'] Only the SIP URL is added to 'Favorites' when adding a contact from the Personal Directory or from Outlook.
104326	[Skype for Business 'Favorites'] The phone does not retrieve Outlook contacts from On Premises Exchange servers.
105106	[Web Sign-in] When using the phone's Web interface to perform Web sign-in, cancelling the operation via the Web is not reflected correctly on the phone's screen.
106716	[Voice Mail] On the 405HD model phone, the LED lights up to present a new voice message, but it may take few minutes for the envelope icon to be displayed on the phone's LCD screen.
103639	[Multi-Party Skype for Business Remote Conferencing] When a Conference is locked, the phone cannot add a user (Skype for Business client or other certified phone) if not admitted within 15 seconds.
103927	Sometimes, the user is unable to join a Skype for Business meeting whose configured 'End time' has passed.

Incident	Description
-	[BToE] BToE does not function if PC users are switched without logging off first.
-	[BToE - PC application] Windows 10 users who have the BToE PC application already installed on their PCs must uninstall it before installing the new version of the application.
104040	[BTOE] When a paired user answers an incoming team-call, the phone does not present the call as a team-call.
104039	[BTOE] When a paired user answers a delegated call, the phone does not show that this is a delegated call.
103961	[BToE video] A paired user cannot answer a second incoming call via the Skype for Business client when having an active video call.
103827	[BToE - video] On a PC with low CPU resources, interference in the voice of a video call might be experienced.
104904	[BToE] Call Transfer from the Skype for Business client to another user can be performed from the phone but not from the client.
96650	[Boss Admin] When the Boss configures an Admin user, the phone's LCD displays the Admin's username instead of the Admin's regular name.
97206	[Boss Admin] When Boss performs a handoff to Admin, Admin cannot transfer the call back to Boss with a single-step Handoff softkey. Call Transfer is possible using the Transfer options.
100454	[Boss Admin] The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
100827	[Boss Admin] Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
100828	[Boss Admin] Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.
103573	[Boss-Admin in a conference scenario] The Rosters screen does not indicate if a participant is 'On-behalf '.
103572	[Boss-Admin in a conference scenario] Admin can't set up a meeting on-behalf of Boss.
101198	[405HD model phone] The Ringtones list in the phone's Web interface is incorrect.
103651	Provisioning by the Redirect Server doesn't function if the phone receives any provisioning of configuration URL via DHCP options.
93495	The Web interface displays some screens that are inapplicable to Skype for Business.
-	Call Log information is saved to the phone file system once every 6 hours. Consequently, Call Log information such as missed, received and outgoing calls may not be saved after restarting.
-	Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided if specifically requested.
96728	When performing a Consultative Transfer, the prompt 'Press **Trans** to transfer' is displayed too briefly.
96709	DHCP Option 160 does not read correctly if there is no forward slash (/) at the end of the sent path.
100478	[420HD phone] Configured Function Keys do not function after the phone parks a call.
101224	Accessing the phone's Web interface with HTTPS via Internet Explorer requires TLS 1.2 support. TLS 1.2 can be set in Internet Explorer via Tools > Internet Options > Security tab. If TLS 1.2 cannot be supported, the Chrome browser can be used instead of Internet Explorer.

Incident	Description
101269	The Chrome browser cannot be used to manually update the phone's firmware through the Web interface over HTTPS protocol. Internet Explorer can be used instead.
-	[440HD Delegated Line Feature] The phone is capable of presenting up to 12 incoming calls on the phone's sidecar. Calls over and above the 13th call will not be managed on the sidecar. They will not be displayed nor will they be capable of being picked up. Even if an index on the sidecar will free up, the call will not occupy the free index. A newly free index will be occupied by the next incoming call.
-	[440HD Delegated Line Feature] The phone is currently capable of simultaneously handling up to eight calls per line (configured with the 'number_of_calls_per_line' parameter). If eight calls are concurrently managed on the phone, attempting to pick up an additional call will fail. Other phones in the group can still pick up the call.
-	[440HD phone's 'Delegated Line' feature] Simultaneous configuration of this feature (the 440HD phone's 'Delegated Line' feature) *and* the phone's BToE feature is not supported.
-	[440HD Delegated Line Feature] Accepting, rejecting and resuming calls must be performed using the softkeys or by picking up the handset. They cannot be performed from the audio device keys – speaker/headset.
-	[440HD Delegated Line Feature] On rare occasions, switching between calls displayed in the phone's sidecar, i.e., picking up calls in succession, one after the other, may cause the phone to handle some pickups incorrectly; best practice is therefore to wait five seconds between each pickup to allow the previous action to finish.
-	[440HD Delegated Line Feature] On rare occasions, ending several calls too fast may cause unexpected behavior; best practice therefore is to have a 5-second delay between actions.
-	[440HD Delegated Line Feature] If two phones simultaneously try to pick up two calls held by a certain phone, one of the pickups may fail. Re-attempting to pick up will be successful.
105213	[Voice Mail] If the user calls their Voice Mail and then gets an incoming call, the call with the Voice Mail is ended rather than put on hold.
107516	[USB headset] Plantronics C320: Using the 'Mute' key from the USB headset is not recommended since the headset may lose synchronization with the phone. It is preferable to use the phone's mute key instead.
107485	[USB headset] Jabra HSC016: Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It is advisable to use the phone's mute key and volume up/down keys instead.
107517	[USB headset] On some Jabra USB headset models (the Evolve series, Bizz2400, though other models may also be applicable), the remote party may complain that the volume is too low. In the next release, the gains will be tuned per model. In this release, the gain can be increased, if necessary, in order to set the volume to a higher level. The following parameters must be configured via the IP Phone Manager: **voip/audio/gain/NB/headset_digital_input_gain**=10 (Default: 0) **voip/audio/gain/WB/headset_digital_input_gain**=10 (Default: 0)
107760, 107694	[Calendar - Join URL] The phone responds to Google Calendar meeting invitations with accepted/tentative/decline messages. **
107605	[BToE] Pairing is sometimes deactivated and then activated again during normal use. **
107353	OnPrem users fail to log in using PIN Authentication when OAuth is configured on the server side. **
107626	[CCCP] The remote phone can get stuck - its screen displaying 'Connecting' - when another phone admits the remote phone to the call, while conference is locked. **

Incident	Description
107825	[CCCP] When the Meet Now feature is used, the first attempt to add a user by dialing their URL fails; the second attempt to dial their URL succeeds. **
107692	[USB headset] The user may experience a short disconnection from audio (up to 8 seconds) during long calls with Jabra Evolve. **
107439	[USB headset] Disconnecting/connecting the USB headset from the phone during a call several times may cause the phone to malfunction. **
107439	[USB headset] After three Hold/Resume actions or three toggles between several existing calls, the USB headset is disconnected for up to 8 seconds. **
105881, 105954	The phone does not publish Quality of Experience reports via the QoE server if a call was a Media Bypass call. **
107305	DTMF may not function during the early media stage; DTMF is sometimes needed in order to input digits into an IVR system. **
107433	System/password is not saved when set via provisioning. **
107034	The phone gets stuck on 'Acquiring IP' if it receives a DHCP Option message which is longer than 308 chars. **

** An existing fix for this constraint can be provided on demand.

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3 Previous Releases

3.1 Version 3.0.0.575.42

3.1.1 What's New in Version 3.0.0.575.42

The following new features are introduced in Version 3.0.0.575.42:

- **Online sign-in – connectivity to Office 365**. New capability to **sign in to (connect to) and authenticate with Microsoft's Cloud PBX**, Microsoft's cloud-hosted version of enterprise voice. AudioCodes' phone features two new sign-in method options, allowing users to connect to Microsoft's Cloud PBX:
    - ADAL (Azure AD Authentication Library) that is based on OAuth 2.0 (RFC 6749). The phone always starts with ADAL and if it's unavailable on the server side, the phone moves to OrgID.
    - OrgID (Organizational ID) or LiveID is Microsoft's proprietary connectivity to Cloud services.

> **Note:** Online sign-in to the phone must be in the following format:
>
> - Sign-in address
> - Username in UPN (User Principal Name) format. UPN format is the way the user's name appears in their e-mail address listed in the Active Directory, i.e., **username@domain.com**
> - User's network IT password
>
> Signing in with a username that is a NetBIOS Domain Name, i.e., **domain\username**, as well as signing in with the phone Extension and PIN Code, are *disallowed* for Skype for Business online sign-in. They are only allowed for *on-premises* sign-in.

- **Multi-Party Skype for Business Remote Conferencing** utilizing CCCP (Centralized Conference Control Protocol) is now supported on the phone. A new **Meet Now/Conf** softkey is displayed by default in the 400HD phones. The softkey allows users to easily initiate remote multi-party Skype for Business conference calls. By pressing the new softkey, users can initiate, join or be added to a multi-party conference call while having full control and viewing capability. Users can now:
    - View the Roster – see other participants and their status (like the Mute option, Hold status)
    - View the conference PSTN dial-in number and conference ID
    - Mute/Unmute other participants
    - Manage the conference status as Lock/Unlock
    - Manage the Lobby for Conference calls that Lobby is defined – Admit/Deny other participants
    - Presenters in a conference can add users to the conference
    - Presenters in a conference can remove users from the conference
    - Presenters in a conference can change the role of a participant between 'presenter and attendant'

    In versions prior to Version 3.0.0.575.42, supported conference capability was *locally based* (*phone* based) and limited to three users in a 3-way conference, or *remote based*, with more than two parties from the Skype for Business client, using the BToE feature.

    - **Merging a call into a conference**. Two separate calls can now be merged into one conference call. This can be performed via a new **Merge** option accessed from the phone's **Call Menu** softkey, or via the Skype for Business client if the user is paired.
- **Integration with Microsoft Exchange Server (Calendar) + click to join a Skype for Business meeting**. [Applies to all AudioCodes phones except the 420HD]. Users can view their Microsoft Exchange Calendar meetings in the phone's LCD by selecting a new **Calendar** option from the MENU key. The phone by default displays meetings scheduled to commence between the present and 24 hours from the present (24H), but the network administrator can change the default and

configure the phone to display meetings scheduled to commence between the midnight of the night before the present and the midnight of the night ahead (TODAY). Via the phone, the user can join any online meeting scheduled in Skype for Business: A **Join** softkey is displayed for the user to join in the meeting online. To connect to Microsoft Exchange and receive the Calendar feature, sign-in *must be with username in UPN format*, as described in the Note above.

- **Meeting Reminder**. [Applies to all phones except 420HD]. The phone automatically pops up a Meeting Reminder for meetings scheduled in Skype for Business. The Meeting Reminder pops up in the phone's idle screen at the time defined for it in the Meeting Invitation. Users can either Snooze the Reminder, or Dismiss it. If the user selects Dismiss, the Reminder does not pop up again. The user will still be able to access the Calendar items and view the meeting. If Snooze is selected, the Reminder pops up 10 minutes before the start of the meeting, five minutes before the start of the meeting, and when the meeting starts. This provides the user advance notice and allows them to join in from the phone by selecting a new **Join** softkey.

- **Visual Voice Mail**. [Applies to all phones except 420HD]. By pressing the voicemail key on the phone, users can now see a list of voicemail messages and select which message to listen to or delete. The user's voicemail must be enabled to allow this feature. When a call comes in, the caller can be referred directly to voicemail by pressing a **To VM** softkey displayed when the phone rings.
- **BToE**
  - **Automatic Pairing** (requires BToE PC/laptop application Version 2.x). Users no longer need to *manually* pair the BToE PC/laptop application with the phone. If the laptop after automatic pairing is disconnected and moved to another location, its speaker/headset becomes the audio device associated with the Skype for Business client.
    If the laptop is *manually* paired and then relocated (manual pairing is still an option), Skype for Business audio remains through the phone. It's therefore advisable to pair *automatically*.

> **Note:** If BToE with *manual* pairing has already been performed on a PC/laptop and you want to *automatically* pair, you must delete the old pair code from the BToE PC/laptop application in order to allow BToE automatic pairing.

  - **Support for video calls**. When a video call comes in, video is displayed on the PC/laptop, voice is routed to the phone. By Skype for Business design, the phone performs similarly to a USB device during this scenario. The feature is supported only if the user's phone was automatically paired (by connecting its PC port to the PC/laptop 'behind' it).
  - **Switching between audio devices**. Users can switch back and forth between audio devices. A user in an active call can switch from the phone to using a USB headset connected to the PC, for example, and then back to using the phone. In this scenario, when going back to using the phone, the phone performs similarly to a USB device (by Skype for Business design). The feature is supported only if the user's phone was automatically paired (by connecting its PC port to the PC/laptop 'behind' it).
- **Phone Automatic lock**. The Skype for Business phone now supports the capability to automatically lock after a preconfigured period of time. The feature secures phones against unwanted (mis)use. When the phone is locked:
  - Incoming calls are allowed but outgoing calls require a security PIN code
  - Without the PIN code, the Call Log, Calendar and Corporate Directory cannot be accessed but users are allowed to call preconfigured emergency numbers
- **Locking / unlocking a paired phone**: If a user's phone was automatically paired (by connecting its PC port to the PC/laptop 'behind' it) and if the PC/laptop is active (not locked), the phone cannot be manually locked. The user can manually lock it only after locking the PC/laptop. If the user doesn't manually lock the phone, it will nevertheless automatically lock after the timeout preconfigured in the Skype for Business server lapses. The phone will unlock only after the user unlocks their PC/laptop or if the user manually unlocks the phone.
- **Capability to handle multiple calls - N Concurrent calls (NCC)**. The phone is capable of managing **up to 8 concurrent calls per line**, for example, of holding multiple calls and switching between them (most relevant to the receptionist)

- **Incoming calls to a Delegated Line are displayed in the sidecar of the 440HD IP phone** (exclusively). A new option was added to display in the phone's sidecar incoming and outgoing calls which users can pick up. The phone is capable of presenting up to 12 active calls (limited by the number of the phone's sidecar keys), and of handling up to eight calls simultaneously.
- **Integrated Log Upload**. Allows uploading logs from the phone to the Microsoft server for troubleshooting/support purposes. Complies with Microsoft's certification requirements for 3rd party Skype for Business clients.
- **Device Update**. The Skype for Business server can update the IP phone firmware version. For detailed information on the update process, refer to https://technet.microsoft.com/en-us/library/gg398861.aspx/
- **Quality of Experience (QoE)** reports are now sent to Microsoft's SQL server. The phone supports QoE reporting directly to the Skype for Business monitoring tool. Supported metrics include the voice quality parameters of Jitter and Packet Loss.

> **Note:** To enable QoE reports, the Skype for Business server must be configured to enable QoE monitoring.

- **Skype for Business 'Favorites' contacts & Outlook contacts integrated with the phone**. [Applies to all AudioCodes phones except the 420HD). Contact groups defined in Skype for Business & Outlook contacts are now integrated with the phone. Pressing the CONTACTS hard key on the phone displays by default the 'Favorites' contact group defined in the Skype for Business client. The user can dial a contact directly from it. In addition, pressing the **Menu** softkey in the 'Favorites' screen provides the option to access other 'Contact groups' such as 'Outlook Contacts' or 'Family'.
- New codecs supported: Skype's **SILK 8000** and **SILK 16000**. SILK is an audio compression format and audio codec that can use a sampling frequency of 8, 12, 16 or 24 kHz and a bit rate from 6 to 40 Kbit/s. Main features:
    - Compatibility with Skype for Business
    - Flexible bitrate
    - High quality
    - Variety of sampling frequency
    - Inband FEC and good resilience to Packet Loss

> **Note:** G.722 was the first priority vocoder in version releases prior to Version 3.0.0.575.42. When upgrading from releases prior to Version 3.0.0.575.42, the list of vocoders remains unchanged. To set the SILK to be the first priority vocoder, restore the phone to its defaults or set the vocoder list differently so that SILK is added. This can be done manually or by provisioning.

- The phone’s TRANSFER hard key now by default performs Blind Transfer instead of Consultative Transfer.

  A new logging option **SIPE** (a third-party Pidgin plugin for Microsoft Skype for Business client) was added to the System Logging page. This logging level may help with the investigation of cases related to Exchange integration.
- **Forcing PIN code authentication**. Using a new configuration file parameter, the network administrator can force PIN code authentication; the only sign-in option will then be with user extension number and PIN code. Allowing only the basic PIN code option on the user's phone helps avoid user mistakes and helps avoid storing the user password on the phone.

- The **SIP User Agent** field was modified in compliance with Microsoft's UC requirements. The modified information now appears in all SIP messages that have a User Agent 'Header' field. Until now, the User-Agent was (for example):
  - **AUDC-IPPhone-405_UC_2.0.13.205.15/1.0.0000.0**

  The new User-Agent is (for example):
  - **AUDC/3.0.0.575.42 AUDC-IPPhone-405_UC_3.0.0.575.42**
- **Ability to make new calls during incoming calls**. [Applicable to all phones except the 405HD and 420HD].
  - **The procedure of making a new outgoing call is not interrupted by an incoming call**. The incoming call does not disrupt the user's number dialing process. The user can finish entering the digits and make their call, uninterrupted by the incoming call.
  - **Ability to ignore an incoming call by initiating a new one**. When the user's phone rings indicating an incoming call, they can initiate a new call, ignoring the incoming, by pressing a **New Call** softkey.

3.1.2 Resolved Constraints in Version 3.0.0.575.42

The table below shows the constraints that were known to exist in previous releases 2.0.13 and 3.0 beta which are now resolved in Version 3.0.0.575.42.

Table 3-1: Resolved Constraints in Version 3.0.0.575.42

Incident	Description
99146	[BToE] A paired user doesn't have the option to 'Reply with IM' for an incoming call from its Skype for Business client.
95497	[BToE] Call Park can be performed from the phone, not from the Skype for Business client.
-	[BToE - Audio Primary Device] Switching Audio Primary Device during a call from the Skype for Business PC client to the phone does not function. This is a limitation of Skype for Business-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
95954	[BToE - Audio Primary Device] During an incoming Skype for Business call, the Primary Device cannot be changed in the Skype for Business Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
96778	[BToE - Video Calls] Outgoing and incoming video calls are possible using the Lync PC client with the following limitation: When a paired user has an active video call (e.g., answered from the Skype for Business PC Client), attempting to answer another call (either from the Lync PC client or with the phone) disconnects the video call.
96776, 96729	[BToE - Video Calls] Outgoing and incoming video calls are possible using the Lync PC client with the following limitation: An audio call cannot switch to video using the Lync PC client's **Start My Video** option. Users trying this option (either local or remote parties) receive the notification message 'Call was not completed or has ended' and the call may be disconnected.
95582	[BToE] A paired user cannot merge two calls into a conference call using the Lync PC client's **Merge Calls** option.
104627	[BToE] When receiving a call from a Skype for Business 2016 mobile client to a paired phone that is paired with Skype for business 2016 PC client, the paired phone does not ring but its Skype for Business client does ring.
105564	[BToE] The phone occasionally fails to join a Skype for Business conference.
105677	[BToE] When the phone is automatically paired and used as the PC's default audio device (for example, during a video call), audio is functional but the phone's screen sometimes reverts to idle mode.

Incident	Description
101181	[Boss-Admin] If an Admin with two Bosses answers two incoming calls (one for each Boss), the second call can't be seen by Boss.
104992	[Boss-Admin] When the remote party (400HD phone) is on hold, trying to pick it up automatically resumes the remote party to the held call.
101011	Sign-in using Extension and PIN Code cannot be completed when the user signs in as an external user, even if the phone was already registered internally. The result on the phone may even be a loop of sign-in retries. Users must note that based on Microsoft security recommendations, a remote/external user (Skype for Business client or phone) should use user and password credentials to sign in, rather than Extension and PIN Code.
98994	Presence of Federate users is not displayed in Call Log, though BLF presence works well.
104401	The phone's location-based route is not found due to incorrect phone parsing of Chassis subtypes 6 (Interface name) and 7 (locally assigned).
104320	Japanese language: Corrupted characters
102186	When location-based route information is populated automatically to the phone, the phone's SIP INVITE doesn't include <method>manual<method>. By contrast, when the user signs in from a different location (when location information is entered manually), <method>manual<method> appears in the INVITE.
104167	[USB headset] Either one-way voice or no voice at all occurs at some point during a call. Rebooting the phone solves the problem.
-	[Calendar] The connection timeout to Microsoft Exchange server is too short (five seconds) which results in a failure to get Calendar meetings.
104879	When a call forward rule is removed via the phone, the phone sets the Skype for Business forwarding settings to their default.
105358	The Skype for Business Sign-in page in the phone's Web interface does not display the extension number. It displays 1234 instead.
105252	When a caller's phone attempts to make a connection with another phone, it continues attempting to make the connection even after more than 32 seconds pass without the recipient answering. After 32 seconds, the caller's phone should disconnect the call attempt.
105619	[Conference] The phone fails to merge calls into a conference when two PSTN numbers are involved.
104178	[Conference] The phone doesn't return to the Roster screen after merging calls.
105386	[Common Area phone] Configuring the phone to prevent the user from signing out (using the parameter 'voip/common_area/enhanced_mode') functions correctly only after booting the phone.
104896	Incorrect translation into Russian.
105518	Skype for Business online sign-in may cause the phone user interface to crash.
104774	SIPE logs will function only if the phone's settings are restored to defaults
104788	The phone displays duplicated domain on its screen for outgoing federated calls.
104147 | 104580	The **Trans** softkey does not exist when Semi Attendant Transfer is enabled.
104258	A Music on Hold (MoH) file, when loaded manually to the phone, plays less than flawlessly, but the default MoH file plays flawlessly. Users should use the default MoH file.

3.1.3 Known Constraints in Version 3.0.0.575.42

The table below shows the constraints known to exist in Version 3.0.0.575.42.

Table 3-2: Known Constraints in Version 3.0.0.575.42

Incident	Description
-	420HD IP phone does not support: ▪ Exchange integration (Calendar) ▪ Visual Voice Mail ▪ Outlook contacts and Skype for Business 'Favorites' contacts
104539	[Paging] A Paged call can be ended only by pressing the **End** softkey or by on-hooking the handset. Pressing the speaker hard key does not end the paged call.
103639	[Multi-Party Skype for Business Remote Conferencing] When a Conference is locked, the phone cannot add a user (Skype for Business client or other certified phone) if not admitted within 15 seconds.
103640	In a conference call, when the phone performs a far mute, an unmute, and then a far mute, the popup message 'A presenter has muted you' is not displayed on the far phone.
103927	Sometimes, the user is unable to join a Skype for Business meeting whose configured 'End time' has passed.
-	[BToE] BToE does not function if PC users are switched without logging off first.
-	[BToE - PC application] Windows 10 users who have the BToE PC application already installed on their PCs must uninstall it before installing the new version of the application.
104040	[BTOE] When a paired user answers an incoming team-call, the phone does not present the call as a team-call.
104039	[BTOE] When a paired user answers a delegated call, the phone does not show that this is a delegated call.
103961	[BToE video] A paired user cannot answer a second incoming call via the Skype for Business client when having an active video call.
103827	[BToE - video] On a PC with low CPU resources, interference in the voice of a video call might be experienced.
104904	[BToE] Call Transfer from the Skype for Business client to another user can be performed from the phone but not from the client.
102613	[440HD phone's Boss Admin feature] Simultaneous configuration of *both* the Boss Admin feature *and* the phone's 'Delegated Line' feature is not supported.
96650	[Boss Admin] When the Boss configures an Admin user, the phone's LCD displays the Admin's username instead of the Admin's regular name.
97206	[Boss Admin] When Boss performs a handoff to Admin, Admin cannot transfer the call back to Boss with a single-step Handoff softkey. Call Transfer is possible using the Transfer options.
100454	[Boss Admin] The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
100827	[Boss Admin] Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
100828	[Boss Admin] Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.

Incident	Description
103573	[Boss-Admin in a conference scenario] The Rosters screen does not indicate if a participant is 'On-behalf '.
103572	[Boss-Admin in a conference scenario] Admin can't set up a meeting on-behalf of Boss.
101198	[405HD model phone] The Ringtones list in the phone's Web interface is incorrect.
103995	[405HD model phone] Korean Language is not yet supported in this version release.
103651	Provisioning by the Redirect Server doesn't function if the phone receives any provisioning URL via DHCP options.
103290	Resuming from a Blind Transfer to PSTN may result in no voice if the remote user rejects the call transfer invitation.
93495	The Web interface displays some screens that are inapplicable to Skype for Business.
-	Call Log information is saved to the phone file system once every 24 hours. Consequently, Call Log information such as missed, received and outgoing calls may not be saved after restarting.
-	Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided if specifically requested.
96728	When performing a Consultative Transfer, the prompt 'Press **Trans** to transfer' is displayed too briefly.
96709	DHCP Option 160 does not read correctly if there is no forward slash (/) at the end of the sent path.
-	[USB headset] Still in beta version
100705	[USB headset] Occasionally, the phone's user interface behaves slowly.
100478	[420HD phone] Configured Function Keys do not function after the phone parks a call.
101224	Accessing the phone's Web interface with HTTPS via Internet Explorer requires TLS 1.2 support. TLS 1.2 can be set in Internet Explorer via Tools > Internet Options > Security tab. If TLS 1.2 cannot be supported, the Chrome browser can be used instead of Internet Explorer.
101269	The Chrome browser cannot be used to manually update the phone's firmware through the Web interface over HTTPS protocol. Internet Explorer can be used instead.
-	[440HD Delegated Line Feature] The phone is capable of presenting up to 12 incoming calls on the phone's sidecar. Calls over and above the 13th call will not be managed on the sidecar. They will not be displayed nor will they be capable of being picked up. Even if an index on the sidecar will free up, the call will not occupy the free index. A newly free index will be occupied by the next incoming call.
-	[440HD Delegated Line Feature] The phone is currently capable of simultaneously handling up to eight calls per line (configured with the 'number_of_calls_per_line' parameter). If eight calls are concurrently managed on the phone, attempting to pick up an additional call will fail. Other phones in the group can still pick up the call.
-	[440HD phone's 'Delegated Line' feature] Simultaneous configuration of this feature (the 440HD phone's 'Delegated Line' feature) *and* the phone's BToE feature is not supported.
-	[440HD Delegated Line Feature] Accepting, rejecting and resuming calls must be performed using the softkeys or by picking up the handset. They cannot be performed from the audio device keys – speaker/headset.
-	[440HD Delegated Line Feature] On rare occasions, switching between calls displayed in the phone's sidecar, i.e., picking up calls in succession, one after the other, may cause the phone to handle some pickups incorrectly; best practice is therefore to wait five seconds between each pickup to allow the previous action to finish.

Incident	Description
-	[440HD Delegated Line Feature] On rare occasions, ending several calls too fast may cause unexpected behavior; best practice therefore is to have a 5-second delay between actions.
-	[440HD Delegated Line Feature] If two phones simultaneously try to pick up two calls held by a certain phone, one of the pickups may fail. Re-attempting to pick up will be successful.
104167	[USB headset] With specific headsets, two seconds of silence might occur occasionally.
105213	[Voice Mail] If the user calls their Voice Mail and then gets an incoming call, the call with the Voice Mail is ended rather than put on hold.

3.2 Version 2.0.13D

3.2.1 What's New in Version 2.0.13D

3.2.2 Resolved Constraints in Version 2.0.13D

The following constraints from the previous release have now been resolved in Version 2.0.13D:

- On rare occasions and only in specific environments, the user cannot hear the remote side at the beginning of a call for several seconds.
- Blind Transfer scenarios on rare occasions lead to dropped calls and phone re-initialization.
- A delay is noticed when answering incoming calls to a Response Group line.
- The phone configures its internal switch with VLAN tagged =1 when the external switch port is configured untagged (native) Vlan =1.
- Priority of ICE UDP host candidates in the SIP SDP packet is lower than TCP, causing the call to not be established.
- BToE
    - Users with a very specific user length (X*31 +1) cannot be paired.
    - Escalation of a call from audio to video with a paired user may cause the call to disconnect.
- When the first incoming media packet is a Comfort Noise packet, voice on the incoming path may be inaudible because the configurable parameter 'prevent_CN_in_early_media' is configured to 0.
- Although log files are supposed to be cyclic and 50 KB in size, some may exceed their size limitation and cause flawed phone performance.
- On rare occasions, softkeys disappear after 24 days.
- The phone cannot redial a number that has a non-DID extension.
- In some environments, calls from PSTN occasionally disconnect immediately due to a mismatch in SRTP format.
- Some Hungarian characters cannot be displayed.
- Some Japanese characters cannot be displayed.

3.2.3 Known Constraints in Version 2.0.13D

Version 2.0.13D includes the following known constraints:

- Boss Admin:
    - When the Boss configures an Admin user, the phone's LCD displays the Admin's username instead of the Admin's regular name.
    - When Boss performs a handoff to Admin, Admin cannot transfer the call back to Boss with a single-step **Handoff** softkey. Call Transfer is possible using the Transfer options.

- The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
- Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
- Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.
- If an Admin with two Bosses answers two incoming calls (one for each Boss), the second call can't be seen by Boss.

- BToE:
  - BToE does not function if PC users are switched without logging off first.
  - Call Park can be performed from the phone, not from the Skype for Business client. (
  - Audio Primary Device:
    - Switching Audio Primary Device during a call from the Skype for Business PC client to the phone does not function. This is a limitation of Skype for Business-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
    - During an incoming Skype for Business call, the Primary Device cannot be changed in the Skype for Business Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
  - Video Calls:

    Outgoing and incoming video calls are possible using the Lync PC client with the following limitations:
    - When a paired user has an active video call (e.g., answered from the Skype for Business PC Client), attempting to answer another call (either from the Lync PC client or with the phone) disconnects the video call.
    - An audio call cannot switch to video using the Lync PC client's **Start My Video** option. Users trying this option (either local or remote parties) receive the notification message 'Call was not completed or has ended' and the call may be disconnected.
  - A paired user cannot merge two calls into a conference call using the Lync PC client's **Merge Calls** option.
  - After installing BToE, Lync client must be used to make conference calls as the IP phone no longer displays the **Conf** softkey.
  - When a paired user disconnects a second call that they answered or made, and puts a remote conference on Hold, there's no voice when the user resumes with the remote conference.
- Sign-in using Extension and PIN Code cannot be completed when the user signs in as an external user, even if the phone was already registered internally. The result on the phone may even be a loop of sign-in retries. Users must note that based on Microsoft security recommendations, a remote/external user (Skype for Business client or phone) should use user and password credentials to sign in, rather than Extension and PIN Code.
- Phones participating in a local conference (the originator and the remote parties, assuming they have AudioCodes 400HD Series IP Phones) cannot perform Blind Transfer.
- The Web interface displays some screens that are inapplicable to Skype for Business.
- Call Log information is saved to the phone file system once every 24 hours.
- Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided upon a specific request.
- With the new dialing method, pressing **#** doesn't end the dial string. When pressing the speaker or headset button to initiate the call, the function does work.
- When performing a Consultative Transfer, the 'Press Trans to transfer' prompt is displayed briefly and then disappears.
- DHCP Option 160 does not read correctly when there is no / (forward slash) at the end of the sent path.
- Presence of Federate users is not displayed in Call Log, though BLF presence works well. (

- A phone which is in two active calls cannot be joined to a remote conference by one of the two callers' devices.
- If a call was made from the phone's Contact Search and the phone number contains parenthesis "()", the dialed number is not displayed on the phone's LCD.
- USB headset limitations:
  - USB headset is still in beta version
  - Occasionally, the phone's user interface behaves slowly.
  - When making a local conference call (and the phone is not in BToE pairing mode), the phone's user interface reacts slowly.
- 420HD phone: Configured Function Keys do not function after the phone parks a call.
- A very short beep sound is heard when answering a call using the phone headset hard key.
- Accessing the phone's Web interface with HTTPS via Internet Explorer requires TLS 1.2 support. TLS 1.2 can be set via the Tools > Internet Options > Security tab. If TLS 1.2 cannot be supported, the Chrome browser can be used instead.
- The Chrome browser cannot be used for manually updating the phone's firmware through the Web interface over HTTPS protocol. Internet Explorer can be used instead.
- The phone parameter ‘lync/moh/url’ doesn’t support FTP or HTTP transport protocols.
- 405HD model phone:
  - The Ringtones list in the phone’s Web interface is incorrect.

3.3 Version 2.0.13C

3.3.1 What's New in Version 2.0.13C

3.3.2 Resolved Constraints in Version 2.0.13C

The following constraints from the previous release have now been resolved in Version 2.0.13C:

- Response Group - in some environments, incoming calls to a phone that is part of a Response Group start with several seconds of one-way voice.
- When Media Bypass is enabled, incoming calls from PSTN may occasionally result in one-way voice.
- The phone cannot be managed from the IP Phone Management Server when it is configured to block Web HTTPS access (using the parameter security/web/https_only=1).
- When using AudioCodes' HTTP Proxy (as part of the gateway/SBC) and the IP Phone Management Server, the phone cannot be accessed via the Web interface.
- Boss Admin Delegate – the Boss phone doesn't present the Admin icon immediately after delegation if the Admin sign-in address includes capital letters.
- In some cases, the phone reports a "486 Busy Here" SIP message when requested to transfer a call. As a result, the call transfer fails.
- German language issue - the letter è is not displayed in some LCD screens.
- When logging in to a Common Area Phone with a username longer than 35 characters, the phone freezes.
- The name of the last user to log in is displayed when any key is pressed on the dial pad of an offline 420HD phone.
- The phone plays a ringtone signal when it is configured with 'Busy on Busy' and a second incoming call is routed to voicemail.
- Team-call group: When the phone is part of a team-call group, adding Call Forward and then disabling it to reconfigure the team-call group may cause the team call not to function.
- When Media Bypass is enabled, a long call from PSTN (longer than 15 minutes) may result in one-way voice.
- When Media Bypass is enabled, calls to a remote user, who is also configured with Simultaneous Ring, may result in no voice.
- Semi-Attendant Transfer functionality: In Semi-Attendant mode, it's necessary to wait until timeout (default 5 seconds) or to again press the **Transfer** button in order to invoke the transfer.
- In some environments, when the Edge Server allocates two different addresses for RTP and RTCP, calls from the phone to the Skype for Business client are occasionally disconnected with a '488 not acceptable here' SIP message.
- 405HD
    - **Directory** and **Call Log** softkeys are missing in the New Call screen.
    - The **On Behalf** softkey is displayed even though the phone has no configured Boss after trying to dial to a URL.
    - Maximum Gain Volume is poorly defined.

3.3.3 Known Constraints in Version 2.0.13C

Version 2.0.13C includes the following known constraints:

- Boss Admin:
  - When the Boss configures an Admin user, the phone's LCD displays the Admin's username instead of the Admin's regular name.
  - When Boss performs a handoff to Admin, Admin cannot transfer the call back to Boss with a single-step **Handoff** softkey. Call Transfer is possible using the Transfer options.
  - The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
  - Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
  - Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.
  - If an Admin with two Bosses answers two incoming calls (one for each Boss), the second call can't be seen by Boss.
- BToE:
  - BToE does not function if PC users are switched without logging off first.
  - Call Park can be performed from the phone, not from the Skype for Business client. (
  - Audio Primary Device:
    - Switching Audio Primary Device during a call from the Skype for Business PC client to the phone does not function. This is a limitation of Skype for Business-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
    - During an incoming Skype for Business call, the Primary Device cannot be changed in the Skype for Business Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
  - Video Calls:

    Outgoing and incoming video calls are possible using the Lync PC client with the following limitations:
    - When a paired user has an active video call (e.g., answered from the Skype for Business PC Client), attempting to answer another call (either from the Lync PC client or with the phone) disconnects the video call.
    - An audio call cannot switch to video using the Lync PC client's **Start My Video** option. Users trying this option (either local or remote parties) receive the notification message 'Call was not completed or has ended' and the call may be disconnected.
  - A paired user cannot merge two calls into a conference call using the Lync PC client's **Merge Calls** option.
  - After installing BToE, Lync client must be used to make conference calls as the IP phone no longer displays the **Conf** softkey.
  - When a paired user disconnects a second call that they answered or made, and puts a remote conference on Hold, there's no voice when the user resumes with the remote conference.
- Sign-in using Extension and PIN Code cannot be completed when the user signs in as an external user, even if the phone was already registered internally. The result on the phone may even be a loop of sign-in retries. Users must note that based on Microsoft security recommendations, a remote/external user (Skype for Business client or phone) should use user and password credentials to sign in, rather than Extension and PIN Code.
- Phones participating in a local conference (the originator and the remote parties, assuming they have AudioCodes 400HD Series IP Phones) cannot perform Blind Transfer.
- The Web interface displays some screens that are inapplicable to Skype for Business.
- Call Log information is saved to the phone file system once every 24 hours.

- Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided upon a specific request.
- With the new dialing method, pressing **#** doesn't end the dial string. When pressing the speaker or headset button to initiate the call, the function does work.
- When performing a Consultative Transfer, the 'Press Trans to transfer' prompt is displayed briefly and then disappears.
- DHCP Option 160 does not read correctly when there is no / (forward slash) at the end of the sent path.
- Presence of Federate users is not displayed in Call Log, though BLF presence works well. (
- A phone which is in two active calls cannot be joined to a remote conference by one of the two callers' devices.
- If a call was made from the phone's Contact Search and the phone number contains parenthesis "()", the dialed number is not displayed on the phone's LCD.
- USB headset limitations:
  - USB headset is still in beta version
  - Occasionally, the phone's user interface behaves slowly.
  - When making a local conference call (and the phone is not in BToE pairing mode), the phone's user interface reacts slowly.
- 420HD phone: Configured Function Keys do not function after the phone parks a call.
- A very short beep sound is heard when answering a call using the phone headset hard key.
- Accessing the phone's Web interface with HTTPS via Internet Explorer requires TLS 1.2 support. TLS 1.2 can be set via the Tools > Internet Options > Security tab. If TLS 1.2 cannot be supported, the Chrome browser can be used instead.
- The Chrome browser cannot be used for manually updating the phone's firmware through the Web interface over HTTPS protocol. Internet Explorer can be used instead.
- The phone parameter ‘lync/moh/url’ doesn’t support FTP or HTTP transport protocols.
- 405HD model phone:
  - The Ringtones list in the phone’s Web interface is incorrect.

3.4 Version 2.0.13B

3.4.1 What's New in Version 2.0.13B

The following new features have been introduced in Version 2.0.13B:

- **SHA2 Support** - Open SSL 1.0.1, integrated with TLS 1.2, replaced OpenSSL 0.9.8 and TLS 1.0 stacks, supporting SHA256, to comply with Skype for Business security requirements.
- **405HD phone model**, beta version. GA expected in version 2.0.15. Contact your local AudioCodes sales representative regarding Microsoft certification.
- **Ringtone** can be provisioned from the configuration file. Administrators can choose a ringtone from a selection of ringtones, which will ring when a call comes in.
- **Voice Dialing capability** - **VocaNOM** speed dial functionality, for quicker and friendlier calling capability.
- From this version on, **core dump files will be compressed** before they are saved to the phone's flash memory, allowing for safer use of the core dump.
- **Improved Web interface security**. Logging in to a phone from the phone's Web interface is now performed exclusively in the 'Lync sign in' page (see resolved constraint below).
- A **Generate Certificate, please wait** message is displayed when upgrading firmware from a previous version to this version (see resolved constraint below).

3.4.2 Resolved Constraints in Version 2.0.13B

The following constraints from the previous release have now been resolved in Version 2.0.13B:

- Skype for Business mobile client:
  - The call may disconnect when an external Skype for Business user calls from the phone to a local (internal) Skype for Business user, who answers the call with the Skype for Business mobile client.
  - An incoming call from the Lync Mobile client to the phone cannot be established in some environments due to DNS resolving issues.
- Boss Admin:
  - An Admin who is in two ongoing calls or in a local conference call cannot pick up a call Boss holds.
  - The incorrect pickup screen may appear when Admin has more than one Boss (mix-up between Bosses).
  - When Admin receives a second call (for Boss), the **BossVM** softkey is not functioning.
  - Call pickup after Handoff of the same call and Handoff after call pickup of the same call are currently not possible and targeted to be supported in the next release.
  - Boss Voice Mail status appears in both Admin and Boss phones.
  - When redirecting a call to Boss Voice Mail, the 'to' number/user/text is empty.
  - BTOE (Admin phone is paired)
    - Lync client doesn't indicate that a call is made 'on behalf'.
    - Lync client doesn't indicate that this is an incoming call from Boss.
    - Lync client doesn't indicate to whom the call is being handed off.
  - Programmable Key of the Boss in Admin's phone is not removed after moving Boss' phone to a different pool.
  - When configuring a Boss on an Admin phone that already had a different Boss who was deleted, the Programmable Key of the Boss is not located in the correct place.
  - Admin refers a regular call (not a call to Boss) to Boss' Voice Mail when changing focus between two incoming call (and one of the calls is to Boss).

- Internet Explorer with HTTPS cannot be used to access the phone's Web interface; the option is blocked as part of the security changes related to SSL/TLS upgrade.
- When accessing the Web interface's Line Setting page, the password might be sent as unsecured clear text, so this page, as well as the 'Line Settings' section in the Quick Setup page, are exposed to threat.
- When a user line number URI without 'tel:' is received from the Active Directory, both BLF presence and the user's name are not displayed.
- Paging
    - Incoming ring continues to be played even after the phone rejects a paging call.
    - Phone crashes when trying to end a paging call received while signing in.
    - Phone initiates a new call if the speaker button is pressed while a paging call via the handset is active.
- One-way-voice occurs after 30 minutes in a call, in calls between the phone and the PSTN, if the SRTP Key is re-generated upon a SIP Refresh Reinvite from the remote side.
- Phone uses the same cookie session ID to access the Web interface.
- Phone plays a Ringback tone during a call if voip/services/call_waiting/generate_tone/enable=0.
- Calls made through Media Bypass are disconnected after 15/30/45 minutes due to bad handling of SIP Refresh Reinvite.
- Can't perform Blind Transfer when the phone is in remote conference; the phone's LCD gets stuck on 'transferring'.
- When the phone is in limited service, the BLF LEDs still stay on.
- Consulted Transfer doesn't function if, after receiving a second incoming call, the **Trans** softkey is pressed.
- On rare occasions, the phone fails to sign-in, and the following message is received: 'Fail to obtain user certificate'.
- In some environments (in which a significant delay is noticed), a call that comes in during an existing call may be disconnected if it is answered immediately after the first Call Waiting beep.
- Blind Transfer cannot be performed for a Lync user with a line URI that does not include the ‘+’ sign.
- On rare occasions, the phone may crash during the provisioning process.
- Korean language translation issues.
- USB headset limitations:
    - When resuming a second call; the voice goes to speaker.

3.4.3 Known Constraints in Version 2.0.13B

Version 2.0.13B includes the following known constraints:

- Boss Admin:
    - When the Boss configures an Admin user, the Admin's username is shown instead of the Admin displayed name.
    - When the Boss performs a handoff to Admin, Admin cannot transfer the call back to the Boss with a single step ‘**Handoff**’ softkey. Call Transfer is possible using the Transfer options.
    - The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
    - Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
    - Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.

- If an Admin with two Bosses answers two incoming calls (one for each Boss), the second call can't be seen by Boss.

- BToE:
  - BToE does not function if PC users are switched without logging off first.
  - Call Park can be performed from the phone, not from the Skype for Business client.
  - Audio Primary Device:
    - Switching Audio Primary Device during a call from the Skype for Business PC client to the phone does not function. This is a limitation of Skype for Business-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
    - During an incoming Skype for Business call, the Primary Device cannot be changed in the Skype for Business Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
  - Video Calls:

    Outgoing and incoming video calls are possible using the Lync PC client with the following limitations:
    - When a paired user has an active video call (e.g., answered from the Skype for Business PC Client), attempting to answer another call (either from the Lync PC client or with the phone) disconnects the video call.
    - An audio call cannot switch to video using the Lync PC client's **Start My Video** option. Users trying this option (either local or remote parties) receive the notification message "Call was not completed or has ended" and the call may be disconnected.
  - A paired user cannot merge two calls into a conference call using the Lync PC client's **Merge Calls** option.
  - After installing BToE, Lync client must be used to make conference calls as the IP phone no longer displays the **Conf** softkey.
  - When a paired user disconnects a second call that they answered or made, and puts a remote conference on Hold, there's no voice when the user resumes with the remote conference.
- Sign-in using Extension and PIN Code cannot be completed when the user signs in as an external user, even if the phone was already registered internally. The result on the phone may even be a loop of sign-in retries. Users must note that based on Microsoft security recommendations, a remote/external user (Skype for Business client or phone) should use user and password credentials to sign in, rather than Extension and PIN Code.
- Phones participating in a local conference (the originator and the remote parties, assuming they have AudioCodes 400HD Series IP Phones) cannot perform Blind Transfer.
- The Web interface displays some screens that are inapplicable to Skype for Business.
- Call Log information is saved to the phone file system once every 24 hours.
- Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided upon a specific request.
- With the new dialing method, pressing **#** doesn't end the dial string. When pressing the speaker or headset button to initiate the call, the function does work.
- When performing a Consultative Transfer, the 'Press Trans to transfer' prompt is displayed briefly and then disappears.
- DHCP Option 160 does not read correctly when there is no "/" (forward slash) at the end of the sent path.
- Presence of Federate users is not displayed in Call Log, though BLF presence works well.
- A phone which is in two active calls cannot be joined to a remote conference by one of the two callers' devices.
- If a call was made from the phone's Contact Search and the phone number contains parenthesis "()", the dialed number is not displayed on the phone's LCD.
- Skype for Business mobile client:
  - No voice may occur after Blind Transfer.

- USB headset limitations:
  - USB headset is still in beta version
  - Occasionally, the phone's user interface behaves slowly.
  - When making a local conference call (and the phone is not in BToE pairing mode), the phone's user interface reacts slowly.
- 420HD phone: Configured Function Keys do not function after the phone parks a call.
- A very short beep sound is heard when answering a call using the phone headset hard key.
- Accessing the phone's Web interface with HTTPS via Internet Explorer requires TLS 1.2 support. TLS 1.2 can be set via the Tools > Internet Options > Security tab. If TLS 1.2 cannot be supported, the Chrome browser can be used instead.
- Team Call Group: Disabling Call Forward from the phone's LCD causes the Team Call Group setting to be deleted.
- The phone parameter 'lync/moh/url' doesn't support FTP or HTTP transport protocols.
- 405HD model phone:
  - Directory and Call Log softkeys are missing in the New Call screen.
  - When dialing to URL 'On Behalf', the softkey is displayed even though the phone is not configured for Boss-Admin.
  - The Ringtones list in the phone's Web interface is incorrect.

3.5 Version 2.0.13

3.5.1 What's New in Version 2.0.13

The following new features have been introduced in Version 2.0.13:

- **New hardware revision of the 440HD** phone model, in addition to the existing 440HD model, featuring enhanced LCD resolution (from the existing resolution of 132x64 to a resolution of 256x128). Contact AudioCodes regarding availability of the new 440HD hardware revision.
- **New Boss-Admin (Delegated Line) functionalities** (430HD and 440HD phones only)
  - Call Pick-up:
    - Boss can monitor Admin when Admin is in a call on Boss' behalf
    - Boss can pick up a call from Admin when Admin is in a call on Boss' behalf and has put the call on hold
    - Admin can pick up a call from Boss if Boss puts the call on hold
    - Admin 1 can pick up a call from Admin 2 (if Admin 2 is in the call on Boss' behalf and then puts the call on hold)
  - **Admin can forward to Boss' voicemail** without picking up Boss' line
- **Multicast Paging, including Barge-in**. Multicast Paging Allows a live announcement to be made (paged) from a phone to a group of phones, to notify a team (for example) that a meeting is about to commence at a certain venue. The paged announcement is multicast via a designated group IP address, in real time, on all idle phones in the group, without requiring listeners to pick up their receivers. The name of the group is displayed on phone LCDs when the paging call comes in.
  - **Barge-in**. This feature, when enabled, allows paging calls to interrupt (barge in on) phone conversations that are in progress, without prompting recipients with an option to accept or reject the paging call. When disabled (default), those who are in regular calls when a paging call comes in are prompted in their phones' LCDs to choose whether or not to accept or reject the paging call. If it's accepted, the regular call is put on hold and the paging call is heard.
- **User access to voicemail can now be secured by PIN code authentication** so that when users press the voicemail button, they're prompted to enter their PIN code. By default, the phone skips PIN code authentication and allows users direct access to voicemail. IT administrators can secure voicemail access by disabling the parameter 'voip/services/vm_skip_pin_code/enabled'.
- **New default sign-in method with extension number and PIN code**: The default sign-in method presented in the phone's LCD is now extension number and PIN code. This is the most preferred sign-in method used by users. The default sign-in method presented previously in the phone's LCD was Username and Password, which is a more complicated method than the present default.
- **Blind Transfer** softkey. During calls, the Blind Transfer **(BXfer)** softkey is by default now displayed in the phone's LCD because the 'voip/signalling/sip/sk_blind_transfer/enable' configuration file parameter is by default now enabled (**1**). IT administrators can disable the softkey (**0**), in which case it will not be displayed during calls. The softkey provides a convenient way for users to perform Blind Transfer.
- **One Voice Resiliency (OVR) support**. The version was tested with the OVR support feature enabled on AudioCodes' Media Gateway/SBC products.
- **Open SSL 1.0.1 integrated with TLS 1.2** has replaced the previous OpenSSL 0.9.8 and TLS 1.0 stacks to comply with Skype for Business security requirements.
- **Added capability to display contact person information in the Latvian language**.
- **Better Together over Ethernet (BToE) PC application**. Tested to be compatible with Microsoft Skype for Business.
- **Web Login is now fully integrated into the IP phone's Web interface** as a screen displaying 'User Name' and 'Password' fields. The screen is displayed when logging off and then logging in again, or when logging in for the first time. Previously, the Web interface presented a login popup prompt.

- **New capability for phones to operate with 18 Multiple Points of Presence (MPOPs)** including Microsoft Skype for Business clients. (The 420HD non-GbE phone supports up to 5 MPOPs).
- **Block User Sign-Out**. Administrators can disable users from signing out. A new configuration file parameter 'lync/userSetting/prevent_user_sign_out' has been added. Default=0 (disabled). When enabled, the **Sign out** softkey is not displayed in the LCD. Identical to the previous version's configuration file parameter 'voip/common_area/enhanced_mode', which was specific for Common Area users, the new configuration parameter applies to *all* users. The new parameter takes precedence over 'voip/common_area/enhanced_mode'.
- IT administrators can **disable user phone microphones** using a new configuration file parameter 'voip/audio/microphone/enable' which by default is enabled.

3.5.2 Known Constraints in Version 2.0.13

Version 2.0.13 includes the following known constraints:

- Boss Admin:
  - When the Boss configures an Admin user, the Admin's username is shown instead of the Admin displayed name.
  - When the Boss performs a handoff to Admin, Admin cannot transfer the call back to the Boss with a single step '**Handoff**' softkey. Call Transfer is possible using the Transfer options.
  - The list in the Dialed Calls screen, shown after pressing the REDIAL key, is incorrect after a call to Boss is made and the call is answered by Admin. When the caller presses REDIAL, they see Admin's phone number instead of Boss'.
  - Admin who is in two ongoing calls or local conference call, cannot pick up another call Boss holds.
  - The incorrect pickup screen may appear when Admin has more than one Boss (mixup between Bosses).
  - Response Group - A call initiated by Admin on Boss' behalf to Response Group cannot be picked up by Boss.
  - Response Group - Incoming call from Admin (on behalf of Boss) appears in Response Group agent's phone LCD as a regular call and not as a Response Group call.
  - When Admin receives a second call (for Boss), the **BossVM** softkey is not functioning.
  - Call pickup after Handoff of the same call and Handoff after call pickup of the same call are currently not possible and target to be supported in next release.
- BToE:
  - BToE does not function if PC users are switched without logging off first.
  - Call Park can be performed from the phone, not from the Skype for Business client.
  - Audio Primary Device:
    - Switching Audio Primary Device during a call from the Skype for Business PC client to the phone does not function. This is a limitation of Skype for Business-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
    - During an incoming Skype for Business call, the Primary Device cannot be changed in the Skype for Business Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
  - Video Calls:

    Outgoing and incoming video calls are possible using the Lync PC client with the following limitations:
    - When a paired user has an active video call (e.g., answered from the Skype for Business PC Client), attempting to answer another call (either from the Lync PC client or with the phone) disconnects the video call.
    - An audio call cannot switch to video using the Lync PC client's **Start My Video** option. Users trying this option (either local or remote parties) receive the notification message "Call was not completed or has ended" and the call may be disconnected.

- A paired user cannot merge two calls into a conference call using the Lync PC client's **Merge Calls** option.
- After installing BToE, Lync client must be used to make conference calls as the IP phone no longer displays the **Conf** softkey.
- When a paired user disconnects a second call that they answered or made, and puts a remote conference on Hold, there's no voice when the user resumes with the remote conference.

- Phones participating in a local conference (the originator and the remote parties, assuming they have AudioCodes 400HD Series IP Phones) cannot perform Blind Transfer.
- The Web interface displays some screens that are inapplicable to Skype for Business.
- Call Log information is saved to the phone file system once every 24 hours.
- Voicemail is supported for Microsoft Exchange Server 2010 and later. A version supporting Voicemail for Microsoft Exchange Server 2007 will be provided upon a specific request.
- With the new dialing method, pressing **#** doesn't end the dial string. When pressing the speaker or headset button to initiate the call, the function does work.
- When performing a Consultative Transfer, the 'Press Trans to transfer' prompt is displayed briefly and then disappears.
- DHCP Option 160 does not read correctly when there is no "/" (forward slash) at the end of the sent path.
- Presence of Federate users is not displayed in Call Log, though BLF presence works well.
- A phone which is in two active calls cannot be joined to a remote conference by one of the two callers' devices.
- If a call was made from the phone's Contact Search and the phone number contains parenthesis "()", the dialed number is not displayed on the phone's LCD.
- Skype for Business mobile client:
  - The call may disconnect when an external Skype for Business user calls from the phone to a local (internal) Skype for Business user, who answers the call with the Skype for Business mobile client.
  - No voice may occur after Blind Transfer.
- USB headset limitations:
  - USB headset is still in beta version
  - Occasionally, the phone's user interface behaves slowly.
  - When making a local conference call (and the phone is not in BToE pairing mode), the phone's user interface reacts slowly.
- 420HD phone: Configured Function Keys do not function after the phone parks a call.
- A very short beep sound is heard when answering a call using the phone headset hard key.
- Can't perform group call pick up for incoming calls from federated Lync users.

3.5.3 Resolved Constraints in Version 2.0.13

The following constraints from the previous release have now been resolved in Version 2.0.13:

- Boss Admin:
  - In scenarios in which the Admin uses the **Call on Behalf** softkey or the **Directory** softkey, the next outgoing call may be performed as a 'Call On Behalf' call even if it wasn't.
  - In scenarios where the Admin uses the **Handoff** softkey to handoff the call to the Boss and then the Boss tries to handoff the call back to the Admin, the call may fail.
  - Domain is added to the caller ID name when making an 'On Behalf Of' call.
  - Phone may crash when Boss puts another phone in parking lot and then presses the ‘Handoff’ softkey.
  - When the Boss configures an Admin, the Programmable Key and display name in the Boss' phone flash 3-4 times until stable.

  - Incorrect input in the phone LCD when calling Boss via the Directory (CONTACTS key).
  - When configuring more than five Bosses to one Admin, the sixth Boss overrides the first Boss.
- MPOP (Multiple Points of Presence): In a Skype for Business environment, each user can have only up to five devices connected simultaneously (for example, PC client, mobile application, and up to three IP phones). Adding users using the Skype for Business Team Call may also introduce this limitation.
- The phone cannot perform Semi-Attendant Transfer via Call Log or Corporate Directory.
- RTCP-XR is not supported in this version and is planned to be supported as part of the QoE Reporting feature in the next release.
- On rare occasions, the phone enters “sign out” state and does not recover automatically, but requires the user to sign in manually by pressing the **Sign in** softkey.
- If the Active Directory search fails and PAID is included in the SIP INVITE message, the phone does not display the callee / caller name.
- In some environments, the phone may send a request for a new IP address once every three DHCP lease time cycles.
- Phone may crash after pressing 'Hold' during a local conference.
- In some scenarios, ‘Call log’ information is not deleted after restoring the phone to its default settings.
- EMS
  - Phone doesn't send an ‘alarm clear’ notification to the EMS when signing in successfully, if the phone previously failed to sign in due to wrong user/password.
  - On rare occasions, the phone sends a 'Registered' status even though it failed to connect.
- Blind transfer is not performed automatically and requires pressing the ‘Transfer’ key again to complete the transfer.
- Phone cannot display alphanumeric characters in Em size, i.e., the point size of the font in its language context).
- In some cases, the phone fails to get information from the Location server (for E911 service).
- The phones does not get Chassis and Port ID information from LLDP-MED, leading to an incorrect Location lookup.
- The phone's Audio ports do not correspond with the Skype for Business configuration.
- The sign-in process takes too long when using a username/password with special characters (‘$’ for example). This issue does not occur in most Skype for Business environments.
- Missed call display is cut if more than 100 calls are missed.
- Fast busy tone (to signal that the remote party is busy) isn’t heard on the phone in hands-free mode. Instead, the phone just shows 'Extension busy' for a short period and then reverts to the idle screen.
- When entering characters using the phone dial pad (e.g., when entering a contact in the Directory), standard characters should appear prior to special characters (for example, when pressing ‘1’,’ a, b, c, á, ä,..’ and not ‘a, á, ä, b, c,...’).
- Japanese language
  - The phone crashes when performing a search in the Corporate directory.
  - The phone crashes during a Consultative Transfer.
  - The display in the main LCD of an incoming 'team call' is incorrect.
- Czech & Slovak languages
  - The user status is incorrect.
- Korean Translation languages issues.
- Directory lookup does not function in some environments.
- Auto Registration fails after recovery from a WAN failure.
- Pool pair failover fails in environments that contain multiple front ends per pool.

- Semi-Attendant Transfer via Call Log or Corporate Directory can only be performed using the CONTACTS hard key on the phone (the 'Contacts' softkey cannot be used).
- A call is disconnected after 75 seconds when it is established with Early Media (SIP Message 183) and without 200 OK.
- The phone crashes if a Blind Transfer is performed while a call is in held state.
- Incorrect display in the phone's LCD when calling an undefined user (non-existent number).
- Blind Transfer doesn't function when transferring to numbers that are normalized, i.e., without a '+' symbol.
- The phone displays an incorrect number for non-DID users.
- When a user logs out from the phone's Web interface, there is no requirement to enter a username / password to log in again.
- USB headset limitations:
    - Voice may sound metallic or noises may sometimes be heard during a call. Switching the phone's audio device to Handset/Speaker and then returning to Headset resolves this issue.

3.6 Version 2.0.11B

3.6.1 What's New in Version 2.0.11B

The following new features were introduced in Version 2.0.11B:

- **USB Headset beta** (supported only by 430HD and 440HD IP Phone models). The following USB headsets are supported:
  - Jabra UC-150
  - Jabra Speak 510+
  - Jabra Speak 410
  - Jabra MOTION OFFICE
  - Jabra PRO 9470
  - Microsoft LX-3000
  - Plantronics C-310M
  - Plantronics C-320M
  - Plantronics HW720
- **Better Together over Ethernet (BToE) PC Application (1.0.20):**
  - Support for Microsoft Windows 10
  - Compatible with Skype for Business
  - GUI enhancements
- **Block Sign-Out for Common Area Users:** Administrators can now disable end users from signing out of a common area phone. To support the feature, the new configuration file parameter, *voip/common_area/enhanced_mode* has been added. The default is 0 (disabled). When enabled, the **Sign out** soft key is not displayed on the LCD.
- **Sign-in through Web:** A new Web page has been added that allows users to sign-in through the phone's Web-based management interface.
- **AudioCodes' enterprise voicemail servers are now supported** as an alternative option to Microsoft Exchange Server.
- **Disable Local Three-way Conferencing:** Administrators can now disable the local three-way conference capability. By default (when not in BToE pairing mode), when phones are in call state, the phone's LCD displays options to enable local three-way conferencing. To support the feature, a new configuration file parameter, *lync/local3wayConf/enabled* has been added.
- **Allow Users to Display Phone or Extension Number:** Administrators can now allow users to define whether to display their telephone or extension number on the phone's LCD. This is only possible if the enterprise's Active Directory includes both telephone and extension numbers. To support the feature, the new configuration file parameter, *lync/sign_in/line_type_display/ext* has been added. The default is 1 (extension number is displayed).
- Core Dump file generation can be enabled and downloaded through the phone's Web-based management interface.

3.7 Version 2.0.11

3.7.1 What's New in Version 2.0.11

The following new features have been introduced in Version 2.0.11:

- **Factory-Set Certificates and AudioCodes Trusted Root CA**. AudioCodes IP phones are now loaded with factory-set preinstalled certificate files: private key file, certificate file and a Trusted Root CA file that is signed by AudioCodes. Whenever the IP phone authenticates with a remote server, it can be authenticated using these certificate files. Each IP phone receives a uniquely generated private key certificate file based on its MAC address. If the remote server is configured to authenticate the client and AudioCodes factory-set certificates are used for authentication, then the AudioCodes Certificate and AudioCodes Trusted Root CA must be downloaded to the remote server. These files can be downloaded from the AudioCodes Web site. For more information, contact your local AudioCodes sales representative. If you use the AudioCodes Redirect server to obtain firmware and configuration files, then the factory-set certificates are used to authenticate the connection with this server. If default certificate files are missing or deleted, the phone will regenerate these files automatically the next time it is powered up.
- **New dialing feature**. After entering the digits of a phone number on the keypad, users can now also pick up the receiver or press the **Speaker** button for dialing to occur. This is the new default behavior. Previously, users needed to press the **Dial** softkey.
- When using the handset in a call, if the handset is on-hooked after putting the call on hold, the call is not disconnected and the audio is switched to the speaker. To maintain backward compatibility, users can set the configuration file parameter 'voip/onhook_disconnect_when_held/enabled' to **1**. This causes the call to be disconnected in the above scenario, as it was in earlier versions.
- **New Presence State** is now supported: The red BLF LED flashes on and off to indicate when a contact is in a call. When a contact, who is listed in a user's phone's BLF list, is in a call, the red BLF LED on the user's phone flashes on and off.
- When two incoming calls occur simultaneously, the phone's LCD displays them in a graphically user-friendly way: the names of the calling parties appear on the LCD and the adjacent incoming call icons flash. If a user is in a call and a third party calls, the name of the calling party appears on the LCD and the adjacent incoming call icon flashes.
- BToE
    - **Automatic mute when a paired phone joins a Lync meeting**. When a user, whose phone is connected to BToE, joins a Lync meeting, the phone is muted and the red **MUTE** button is lit. The user can unmute using the phone's **MUTE** button or using the Lync client.
    - When answering an incoming video call with a paired phone, the call is established. The default device is the PC speaker/microphone rather than the phone. Subsequent audio calls will be unaffected; the paired phone will still be the default device.
- **New Blind Transfer (BXfer) softkey**. A new softkey, **BXfer**, provides a more convenient way for users to perform a Blind Transfer.
  The feature must be enabled using the 'voip/signalling/sip/sk_blind_transfer/enable' configuration file parameter. The default softkey is **Hold**.
- **Support for Electronic Hook Switch (EHS) DHSG**. Answering calls and changing volume level with EHS-capable headsets is now supported. This newly supported capability can be enabled by setting the configuration file parameter 'voip/services/electronic_hook_switch/enabled' to **1**.

  The feature was verified using the following headsets:
    - Jabra® PRO 920
    - Jabra® PRO 9450

  The headset's base unit connects to the phone's headphone port. The Audio connector connects to the headphones port. The management connector connects to the Auxiliary port using a DHSG cable which can be ordered from AudioCodes.
- Call waiting audial indication (beep progress tone), which can disrupt a conversation, can be disabled so that only visual indication is enabled. If a user is in a call and a third party calls that

user, the called user's LCD visually indicates that a calling party is waiting. The feature is configurable in the Web interface's Services page, under the Call Waiting screen section, with the 'Generate Tone' parameter, which must be disabled. The feature is also configurable in the cfg file with the 'voip/services/call_waiting/generate_tone/enable' parameter, which must be disabled.

- Capability to remove ('kill') the Location popup. Until now, if Location was not configured, a popup opened in the phone's LCD enabling users to either Set Location, or Skip. If Skip was selected, the popup would close, but would pop up every few hours. Users can now kill the popup, or opt (still) to manually set Location, or Skip (every few hours the popup returns).
- When a user signs out and another signs in, the phone presents empty Speed Dials and empty Call Logs to the newly signed in user. The Speed Dials and Call Logs of the signed out user are not saved on the phone.

3.8 Version 2.0.9.127

3.8.1 What's New in Version 2.0.9.127

The following new features have been introduced in Version 2.0.9.127:

- A new logging mechanism 'Lightweight Syslog' allows the user to perform phone logging without affecting the phones' performance.

  To enable the Lightweight Syslog:

  Access the phone's Web interfaces' System Logging page (**Status &Diagnostics** tab > **Diagnostics** > **Logging**), set the 'Activate' parameter to **Network** and provide a valid IP address and server port. Do not set any of the options (keep all as 'None').

> **Note:** When Lightweight Syslog is enabled as described above, the IP Phone Trace feature is also automatically enabled. Therefore, it is highly recommended to disable this feature (**Status &Diagnostics** tab > **Diagnostics** > **Tracing**).

- When the Active directory includes the telephone number and extension number, a new parameter 'lync/sign_in/line_type_display/ext' allows the user to define whether to display a telephone or extension number. The Default setting of this parameter keeps the previous phone behavior (Extension number).
- The IP Phones now support the Cisco Discovery Protocol (CDP) Enhanced. Whenever a PC is connected or disconnected from the phone, CDP messages are sent from the phone to the connected Cisco switch. For example, when a PC is disconnected from the phone, a CDP message is sent and this PCs' MAC address entry is removed from the Cisco switch, thereby enabling a different PC to connect to the phone.
- When searching in a Corporate Directory, the comma character is moved to be one of first characters to appear when pressing on digit '1'.
- An option has been added to display the caller name from the incoming SIP message "From" header instead of the caller name from the Active Directory.

  By default, the parameter 'lync/contact_name_priority=CONTACT_SEARCH' and the phone display the caller name from the Active Directory info.

  To use the name from the Invite "From" header, set the parameter 'lync/contact_name_priority=CALL_DESCRIPTION'.
- A new parameter 'voip/media/prevent_CN_in_early_media' when set ignores RTP with Comfort Noise when it is received as the first packet from the network. To enable it, set the parameter to 1 (default is 0). This feature is related to the following resolved limitation:
  - Upon an outgoing call, the Ringback tone is not heard if the first packet received on the RTP socket is STUN/Turn or Comfort Noise.
- Increase the time value for the 'Away' state of the current User Presence from 18 hours to 24 days. Setting the User Presence is implemented using the following parameter: 'lync/presence/state_change_timeout'.

- Version 2.0.9.93

3.8.2 What’s New in Version 2.0.9.93

The following new features have been introduced in Version 2.0.9:

- Added support for the US date format.
- Multi-language support:
  - Korean was improved.
  - Japanese and Hungarian are now also supported.

3.9 Version 2.0.9.65

3.9.1 What’s New in Version 2.0.9.65

The following new features have been introduced in Version 2.0.9:

- T9 predictive text for Corporate Directory search. When searching for a contact in the Corporate Directory, users can press dial pad keys to input letters. Only a single press on any key, regardless of the letter's position on the key, is necessary.
- Corporate Directory search - an **OK** softkey was added to enhance the searching experience.
- A new 420HD model with Gigabit Ethernet (GbE) was added.
- Incoming call ring can be reduced to silence (mute) using the volume-down button.

3.10 Version 2.0.7

3.10.1 What's New in Version 2.0.7

- Boss Admin feature
  - Allows users to establish a relationship between a boss's phone and an administrative secretary's phone.
  - Streamlines office workflow and enhances efficiency.
  - Each phone can support up to five bosses or admins. One boss can have up to five admins. One admin can have up to five bosses. A many-to-many configuration is also supported.
  - Applies to AudioCodes' 430HD and 44HD IP phone models.
  - Delegation must be allowed for the user in the environment, else configuration will not work.
    All users must be allowed to configure all users as delegates.
- Better Together over Ethernet (BToE) enhancements
  - Allows video calls. When there's an incoming video call, users can:
    - Answer from the phone; the call will fall back to audio.
    - Answer from the PC client and have full audio and video using the PC audio device.
  - BToE TCP port can be configured to communicate between the BToE PC application and the phone.
- Automatic mass provisioning and management using the AudioCodes Element Management System (EMS)
  - Automatic mass provisioning of IP phones using the DHCP provisioning method can now be performed from the AudioCodes EMS Provisioning Server in the IP Phones Management Server, accessible from the EMS.
  - For detailed information, see the *IP Phone Management Server Administrator's Manual*.
- New Acoustic Echo Canceller for significantly improved IP telephony voice quality.
- Login to the Web interface is now performed pulling the 'User Name' and 'Password' parameter values configured in the configuration file.
  - The default 'User Name' and 'Password' values are **admin** and **1234** respectively.
  - If either or both of these parameter values are unconfigured in the configuration file, users can log in to the Web interface using the same Microsoft password/PIN they used to sign in to the IP phone (to maintain backward compatibility).
- New ways of dialing. Users can now:
  - Press the speaker or headset button, or pick up the handset, and when the dial tone is heard, start dialing; touch tones are played. Users can then press either the **#** (digit) key or the **Dial** softkey to 'send' the number, or wait a few seconds for the number to automatically be sent.

    -OR-
  - On the phone's dial pad, press the digit keys of the called party's number (there are no touch tones) and then press the **Dial** softkey to trigger the call. This is the equivalent of the SEND key on mobile devices.

    Backward compatibility is supported by setting the 'voip/dialing/on_hook_dialing' configuration file parameter to **OPEN_DEFAULT_AUDIO_DEVICE**

3.11 Version 2.0.5 - BToE

3.11.1 What's New in Version 2.0.5

- Better Together over Ethernet (BToE):
  - Lets users pair their AudioCodes IP phone with their Microsoft Lync client on their PC or laptop, over Ethernet, and from their PC or laptop control phone operations such as answer incoming calls, make outgoing calls (click-to-dial), hold and resume calls, and initiate/join an online meeting or Lync conference using their Lync client.
  - Allows mirroring of each call on both the AudioCodes IP phone and the PC, so that calls can be controlled from either the IP phone or the PC, adding substantial value to AudioCodes unified communications.
  - Allows the phone to be automatically signed in (from the PC client, after you set the pair code and then your Password in the Lync PC client login).
  - Functions even when phone and PC are in different subnets. The 'lync/BToE/CheckNetwork' parameter, when set to **0** (default), enables this. Users who used the previous BToE version must set this parameter to **0** in the configuration file.
- The key sequence for call transfer was modified. In previous versions, the TRANSFER key on the device was used to perform a blind transfer - the established call was not put on hold before transferring. In the current release, the TRANSFER key on the device is used to perform attended transfer - the established call is put on hold in order to transfer the call. Blind transfer is performed using the Call Menu softkey. See the *User's Manual* for details. (95518).
  - Note: Semi-attended transfer using the TRANSFER key on the phone is disabled by default in order to avoid confusion. To enable it, the 'system/semi_attended/enable' parameter can be set to **1** (can be added to global.cfg).
- 802.1X Authentication. IEEE Standard for Port-based Network Access Control (PNAC). Part of IEEE 802.1 group of networking protocols. Provides an authentication mechanism for devices joining a LAN or WLAN.

3.12 Version 2.0.3 - BToE

3.12.1 What's New in Version 2.0.3 - BToE

Version 2.0.3 - BToE offers the following new features:

- Lets users pair their AudioCodes IP phone with their Microsoft Lync client on their PC or laptop, over Ethernet, and from their PC or laptop control phone operations such as answer incoming calls, make outgoing calls (click-to-dial), hold and resume calls, and initiate/join an online meeting or Lync conference using their Lync client.
- Allows mirroring of each call on both the AudioCodes IP phone and the PC, so that calls can be controlled from either the IP phone or the PC, adding substantial value to AudioCodes unified communications.

In addition:

- The key sequence for Call Transfer was modified. In previous versions, the TRANSFER button on the device was used to perform a Blind Transfer (the established call was not put on hold before transferring). In the current release, the TRANSFER button on the device is used to perform Attended / Semi-Attended Transfer (the established call is put on hold in order to transfer the call). Blind Transfer still can be performed from the **Call Menu** softkey (95518).

3.13 Version 2.0.3

3.13.1 What's New in Version 2.0.3

Version 2.0.3 offers the following new features:

- 430HD and 440HD IP Phones for Lync with Speed Dial and BLF support, in addition to the 420HD IP Phone that was already supported in Version 2.0.3.
- Direct access to the Lync contact list.
- Multi-Tenant (LHP) - Lync Server Multitenant Hosting Pack is a Microsoft Unified Communications (UC) hosting solution for telecommunications and hosting providers. The solution enables Microsoft hosting partners to deploy a single instance of the Lync Server software to securely and economically host multiple tenants with a rich, fully integrated UC solution.
- For details, see the 400HD Series IP Phones for Lync Hosting Partner (LHPv2) Environment Configuration Note.
- IP Phone level MOH (Music On Hold) support. The IP Phone is responsible for sending MOH.
- Busy On Busy. The IP Phone signals a "Busy Here" message when the end user who is being called has an active Lync call (an active call using the IP Phone or any other client the user is logged in with).
- Lync 2013 support, including features such as Call Pickup and Call Park (requires Lync CU01).
- Multi-language support. Korean and Chinese are now also supported.

3.14 Version 2.0.1.44.21

3.14.1 What's New in Version 2.0.1.44.21

This version offers:

- Web and display phone user interface (UI) login credentials are identical to the sign-in credentials.

Notes:

- When logging in to the IP Phone's Web interface, enter the same credentials as those used for logging in to the Lync client. The login method can be username or PIN code.
- To access and view the Web interface GUI as an administrator, log in as a user admin type and password, according to credentials.
- To access and view the Web interface GUI as a user, log in as user <username> and password, according to credentials.
- When a user is signed out from their phone, the Web GUI can be accessed with the default username (admin,1234).

- Optimized volume gain performance: enabling AGC optimizes outgoing voice volume. In the Web (VoIP > Voice), set 'Automatic Gain Control' to **Enable**; in the configuration file (provisioning), set voip/audio/gain/automatic_gain_control/enabled=1.
- Improved IP phone performance by disabling debug tracing. In the Web (Status & Diagnostics > Diagnostics > Tracing), set trace level to **None**; in the configuration file, set system/trace/level=None.

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AudioCodes 400HD Series of High Definition IP Phones

HD VoIP

400HD Series IP Phones for Microsoft® Skype™ for Business

Release Notes

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