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 2.0.13C

Document #: LTRT-08275

Microsoft Partner
Gold Communications

Skype for Business

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HD VoIP Sounds Better

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Notice

This document presents the new features of AudioCodes' 400HD IP Phone series Version 2.0.13C for Microsoft for Skype for Business.

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Abbreviations and Conventions

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

Related Documentation

Document Name
420HD IP Phone User's Manual
430HD and 440HD IP Phone User's Manual
400HD Series IP Phone Administrator's Manual
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 2.0.13C.

Note: Microsoft has 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 ▪ Open SSL 1.0.1 integrated with TLS 1.2, compliant with Skype for Business security requirements ▪ OVR (One Voice Resiliency)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.722. ▪ Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length ▪ Adaptive Jitter Buffer 300 msec ▪ Voice Activity Detection ▪ Comfort Noise Generation ▪ Packet Lost Concealment ▪ RTP/RTCP Packetization (RFC 3550, RFC 3551), SRTP (RFC 3711) ▪ DTMF Relay (RFC 2833)

Feature	Details
Telephony Features	▪ BLF presence on buttons; capability for 18 Multiple Points of Presence (MPOPs), including Skype for Business clients. (The 420HD non-GbE phone 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 (including Blind Transfer option during calls) ▪ Three-way Conferencing (with local mixing) ▪ 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 On-hook Dialing ▪ CWRR (Call Waiting Reminder Ring) ▪ Call Logs: Missed/Received Calls and Dialed Numbers ▪ Speed Dial ▪ Boss Admin (Shared Line Appearance) (applies to 430HD/440HD; 420HD supports only 'Delegation') ▪ Dial Plan - Normalization rules ▪ URL Dialing ▪ Call Forwarding ▪ Paging w/without Barge-in and configurability of Function Keys and Programmable Keys (430HD/440HD) as paging group dials. ▪ Better Together over Ethernet (BToE) compatibility with Microsoft Skype for Business ▪ Voicemail (including capability to secure user access with PIN code)
Configuration / Management	▪ LCD Display 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	▪ Syslog and Tracing Mechanism ▪ Monitoring (Ping and Traceroute) ▪ DSP Recording ▪ Port Mirroring ▪ VoIP Status Web page

Feature	Details
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)

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

IP Phone Model	Part Number per Model Variant	Product Description
405	UC405	Lync/SfB 405 IP Phone (black) Power over Ethernet (PoE)
420HD	UC420HDE	Lync/SfB 420HD IP Phone (black) PoE
	UC420HDEPS	Lync/SfB 420HD IP Phone (black) PoE and external power supply
	UC420HDEW	Lync/SfB 420HD IP Phone (white) PoE
	UC420HDEPSW	Lync/SfB 420HD IP Phone (white) PoE and external power supply
	UC420HDEG	Lync/SfB 420HD IP Phone (black) PoE GbE
	UC420HDEPSG	Lync/SfB 420HD IP Phone (black) PoE GbE and external power supply
430HD	UC430HDE	Lync/SfB 430HD IP Phone (black) PoE
	UC430HDEG	Lync/SfB 430HD IP Phone (black) PoE GbE
	UC430HDEPS	Lync/SfB 430HD IP Phone (black) PoE and external power supply
	UC430HDEPSG	Lync/SfB 430HD IP Phone (black) PoE GbE and external power supply
	UC430HDEW	Lync/SfB 430HD IP Phone (white) PoE
440HDEG	UC440HDEG	Lync/SfB 440HD IP Phone (black) PoE GbE
	UC440HDEPSG	Lync/SfB 440HD IP Phone (black) PoE GbE and external power supply
	UC440HDEPSWG	Lync/SfB 440HD IP Phone (white) PoE GbE and external power supply
	UC440HDEWG	Lync/SfB 440HD IP Phone (white) PoE GbE

2 Version 2.0.13C

2.1 What's New in Version 2.0.13C

2.2 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 dialpad 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.
- 405
    - **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.

2.3 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.
- 405 model phone:
  - The Ringtones list in the phone’s Web interface is incorrect.

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3 Previous Releases

3.1 Version 2.0.13B

3.1.1 What's New in Version 2.0.13B

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

- **405 phone model**, beta version. GA expected in version 3.0. 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).
- A **Generate Certificate, please wait** message is displayed when upgrading firmware from a previous version to this version.
- Optimized the **Lync sign-in retry mechanism,** when the network or server is inaccessible.

3.1.2 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 (mixup 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.
- Korean translation issues.
- Hungarian translation issues.
- German translation issues.
- Italian translation issues.
- Czech/Slovak translation issues.
- USB headset limitation:
  - When resuming a second call, the voice goes to speaker.
- On rare occasions, the phone may become instable when idle due to an issue with the provisioning mechanism.
- In some Skype for Business environments, the call may be rejected by the server side due to incorrect order of the remote candidates lines in the outgoing SIP message.
- Function Keys do not work for SIP addresses longer than 32 characters.
- Phone may enter a loop of unnecessary provisioning retries when an image or configuration file does not exist on the provisioning server.
- ICE credentials are refreshed every 8 hours rather than according to the value set by the server during inband provisioning. This may cause one-way voice when a call is initiated.

3.1.3 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.
- 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.
- 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.
- 405 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.

When the ringer level is set to maximum, the phone's ring is corrupted, to the extent that it becomes user unfriendly. Best practice is to temporarily refrain from setting the phone to maximum ringer level until the fix is made.

3.2 Version 2.0.13

3.2.1 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 (Shared Line Appearance) 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.

3.2.2 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.

3.2.3 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.

3.3 Version 2.0.11B

3.3.1 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.

3.3.2 Known Constraints in Version 2.0.11B

Version 2.0.11B includes the following known constraints:

- Simultaneous functioning of Boss Admin and BToE has not yet been released.
- Boss Admin:
    - It's insufficient for the Boss to turn off Call Forwarding under the Lync client's Call Forward Settings in order to remove a delegate. Under the Lync client's Call Forward Settings, the Boss must *also* remove the delegate from the Call Forwarding – Delegates list.
    - 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. This issue is planned to be resolved in the next release.
    - When the Boss configures an Admin user, the Admin's username is shown instead of the Admin displayed name.
    - Shared Line Pickup is not supported. This issue is planned to be resolved in the next release.
    - 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. This issue is planned to be resolved in the next release.
    - The **Handoff** softkey is missing on the Admin user's phone when the Boss performs a handoff to Admin.
- 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 Lync client. (
    - Audio Primary Device:
        - Switching Audio Primary Device during a call from the Lync PC client to the phone does not function. This is a limitation of Lync-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
        - During an incoming Lync call, the Primary Device cannot be changed in the Lync Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
    - Video Calls:
        - When a paired user has an active video call (e.g., answered from the Lync 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.
- The Web interface is not completely Lync-optimized; some screens inapplicable to Lync are still displayed.
- 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 doesn't read correctly when there's no "/" (forward slash) at end of sent path.
- MPOP (Multiple Points of Presence): In a Lync 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 Lync Team Call may also introduce this limitation. This issue is planned to be resolved in the next release.
- 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.
- 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.
- 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.
- 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.
- Lync mobile client: The call may disconnect when an external Lync user calls from the phone to a Local (internal) Lync user who answers the call with the Lync mobile client.
- USB headset limitations:
  - Voice may sound metallic or sometimes noises may be heard during a call. Switching the phone's audio device to Handset/Speaker and then returning to Headset resolves this issue.
  - When making a local conference call (and the phone is not in BToE pairing mode), the phone's user interface reacts slowly.

3.3.3 Resolved Constraints in Version 2.0.11B

The following constraints from the previous release have now been resolved in Version 2.0.11B:

- Calls between two phones that are signed in with the same user cannot be made. (
- Translation fixes in Hungarian language.
- In some Lync environments, a noise may be heard on the phone when calling a user that is configured with 'Simultaneously Ring'.
- In some environments, the phone may send a request for a new IP address once every three DHCP lease time cycles. A fix for this issue already exists (as part of the next release) and can be provided on request.
- Ringback may stop after only one ring when calling a user that is configured with 'Simultaneously Ring'.
- One-way voice occurs when the call is placed on-hold for more than 30 minutes and then resumed.
- BToE: No voice is experienced when a paired user joins a Lync Conference Meeting as an external user.
- The phone sometimes does not attempt to re-register when it recovers from a lengthy network disconnection.
- BLF blinks when the phone is in "Busy" state.
- Boss Admin: Boss fails to remove Admin delegation when Boss and Admin phones belong to different Lync pools.
- LLDP process crashes on rare occasions.
- Phone crashes when calling using a speed dial that was configured with a long name.
- Phone may crash on rare occasions in call transfer scenarios.
- Phone may crash on rare occasions in Remote Conference Call scenarios.

3.4 Version 2.0.11

3.4.1 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.

3.4.2 Known Constraints in Version 2.0.11

Version 2.0.11 includes the following known constraints:

- Simultaneous functioning of Boss Admin and BToE has not yet been released.
- Boss Admin:
    - It's insufficient for the Boss to turn off Call Forwarding under the Lync client's Call Forward Settings in order to remove a delegate. Under the Lync client's Call Forward Settings, the Boss must *also* remove the delegate from the Call Forwarding – Delegates list.
    - English is the only supported language. When using the phone in another language, Boss Admin LCD screens are displayed in English.
    - 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.
    - When the Boss configures an Admin user, the Admin's username is shown instead of the Admin displayed name.
    - The Admin's phone doesn't display the **Call on Behalf** option when performing a second call on behalf of the Boss, using the phone's softkeys.
    - Shared Line Pickup is not supported.
    - 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.
    - The **Handoff** softkey is missing on the Admin user's phone when the Boss performs a handoff to Admin.
    - Calls to the Boss aren't forwarded when the Admin user configures the phone to forward calls.
- BToE:
    - The new BToE PC application v1.0.8 requires users to repeat the pairing process after installation.
    - Call Park can be performed from the phone, not from the Lync client. (
    - Audio Primary Device:
        - Switching Audio Primary Device during a call from the Lync PC client to the phone does not function. This is a limitation of Lync-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
        - During an incoming Lync call, the Primary Device cannot be changed in the Lync Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
    - Video Calls:
        - When a paired user has an active video call (e.g., answered from the Lync 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, use your Lync client 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.

- The Web interface is not completely Lync-optimized; some screens inapplicable to Lync are still displayed.
- Call Log information is saved to the phone file system once every 24 hours.
- Voicemail is supported for Microsoft Exchange Server 2010 and above. 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.
- MPOP (Multiple Points of Presence) - in a Lync environment, each user should be able to have up to five devices connected simultaneously, for example, PC client, mobile application, and up to three IP phones.
- In some Lync environments, a noise may be heard on the phone when calling a user configured with 'Simultaneously Ring'.
- In some environments, the phone may send a request for a new IP address once every three DHCP lease time cycles. A fix for this issue already exists (as part of the next release) and can be provided on request.
- Presence of Federate users is not displayed in Call Log, though BLF presence works well. (
- Calls between two phones, signed in with the same user, cannot be made. (
- A phone which is in two active calls cannot be joined to a remote conference by one of the two callers' devices.
- The phone cannot perform Semi-Attendant transfer via Call Log or Corporate Directory. (

3.4.3 Resolved Constraints in Version 2.0.11

The following constraints from the previous release have now been resolved in Version 2.0.11:

- Boss Admin:
  - The Admin can't send a call to the Boss's voicemail.
- BToE:
  - When a paired user presses the phone's mute key, there's no indication in the Lync client's Conversation window. Moreover, the user cannot mute from the Lync PC client Conversation window when BToE is running.
  - An IM (Instant Message) conversation is stored as a Missed Call for a paired user. (
  - The BToE PC application switches to 'Disconnected' when performing a first call (incoming / outgoing) after an idle time.
  - An IM is not accepted when a paired user is in two active calls.
  - The BToE PC application may occasionally disconnect and reconnect. During an existing call, this occurrence may change the default audio device from the phone to the PC, and the user may anticipate that the call was disconnected.
- In an environment comprising Exchange Online (as part of Office 365), a call from the phone to a user who has Exchange Online voicemail may sometimes not reach the voicemail.
- With the new dialing method, after pressing the digits of a phone number and then picking up the

handset, or, after pressing the digits of a phone number and then pressing the speaker button on the phone, the number should be dialed.

- One second of Comfort Noise is heard when an incoming call comes in to a Response group.
- (Applies to the 420HD phone only) When using redial from the Call Log to answer a GSM\PSTN call that comes in, the phone adds a domain name to the number. (
- Incorrect Caller ID is displayed in the phone's Call Log due to **;** separations in the Caller ID.
- The phone may start a call with no audio, or even fail the call, when it is configured to operate with an O/H (After Working Hours) schedule and it receives a forwarded call from RGS.
- Incorrect handling of an incoming SIP Notify message may cause wrong BLF Presence of a Federated user after a long time (several hours).
- Inability to establish a call from the phone's Missed Calls list. (
- Pressing the **End** softkey in order to regret and disconnect a call, may not work on the first try. (
- The phone cannot choose which call to accept when two incoming calls come in at the same time.
- If a call is received while dialing to perform an outgoing call, the incoming call is answered automatically after the dialing timeout expires. The expected behavior is for the phone to make the outgoing call and ignore the incoming call when the dial timeout expires.
- The phone fails to perform a Blind Transfer to a user without an assigned number.
- The phone does not recognize an asterisk ***** as a dialing digit.
- (Applies to the 420HD phone only) The phone may crash when performing a Hold in a basic call.
- Power Consumption: the power should be 6.4W when connecting with CDP or LLDP.
- Phone presence is missing when the Lync server sends multiple presences in one BENOTIFY.
- When searching in the Corporate Directory while the phone is offline or in limited service, the phone crashes. (
- The phone may get stuck when performing toggling between two calls.
- The phone re-signs in during an incoming call if it receives a SIP 'Early Dialog' message.
- The phone may not play a ringback tone when connected over an Edge Server.
- The phone displays a wrong display name when it was signed in with a user with 'tel/ext', then re-signed in with a user without 'tel/ext'.
- Calls may fail in an environment where a DNS request for the TURN server address has resulted in two IP addresses.
- Hold and Transfer does not function well when the phone is configured with a Lync Client Policy that includes {EnableClientMusicOnHold: False}.
- When the phone is configured to work in Response Group/Attended, a delay of several seconds (up to eight seconds) may occur upon an incoming call.
- Missing digits when a number is dialed following a prolonged idle time.

3.5 Version 2.0.9.127

3.5.1 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'.

3.5.2 Known Constraints in Version 2.0.9.127

Version 2.0.9.127 includes the following known constraints:

- Boss Admin:
  - It's insufficient for the Boss to turn off Call Forwarding under the Lync client's Call Forward Settings in order to remove a delegate. Under the Lync client's Call Forward Settings, the Boss must *also* remove the delegate from the Call Forwarding – Delegates list.
- Simultaneous functioning of Boss Admin and BToE has not yet been released.
- Boss Admin:
  - English is the only supported language. When using the phone in another language, Boss Admin LCD screens are displayed in English.
  - The Admin can’t send a call to the Boss's voicemail. This issue is expected to be solved as part of the next release (April 2015).
  - 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.
  - When the Boss configures their Admin, the Admin's username is shown instead of the Admin displayed name.
  - The Admin's phone doesn't display the **Call on Behalf** option when performing a second call on behalf of the Boss, using the phone's softkeys.
  - Shared Line Pickup is not supported.

  - 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.
- BToE:
  - When a paired user presses the phone's mute key, there's no indication in the Lync client's Conversation window. Moreover, the user cannot mute from the Lync PC client Conversation window when BToE is running.
  - The new BToE PC application v1.0.8 requires users to repeat the pairing process after installation.
  - Call Park can be performed from the phone, not from the Lync client. (
  - Audio Primary Device:
    - Switching Audio Primary Device during a call from the Lync PC client to the phone does not function. This is a limitation of Lync-compatible IP phones. (Audio Primary Device can be switched from the phone to the PC client.)
    - During an incoming Lync call, the Primary Device cannot be changed in the Lync Conversation window. It can be changed when making outgoing calls, and for incoming PSTN calls.
  - Video Calls:
    - When a paired user has an active video call (e.g., answered from the Lync 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, use your Lync client to make conference calls as the IP phone no longer displays the **Conf** softkey.
- In an environment comprising Exchange Online (as part of Office 365), a call from the phone to a user who has Exchange Online voicemail may sometimes not reach the voicemail. This issue is expected to be solved as part of the next version release (May 2015).
- The Web interface is not completely Lync-optimized; some screens inapplicable to Lync are still displayed.
- Call Log information is saved to the phone file system once every 24 hours.
- Voicemail is supported for Microsoft Exchange Server 2010 and above. 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.
- With the new dialing method, after pressing the digits of a phone number and then picking up the handset, or, after pressing the digits of a phone number and then pressing the speaker button on the phone, the number should be dialed. This issue is expected to be solved as part of the next version release (May 2015).
- MPOP (Multiple Points of Presence) - in a Lync environment, each user should be able to have up to five devices connected simultaneously, for example, PC client, mobile application, and up to three IP phones.
- In some Lync environments, a noise may be heard on the phone when calling a user configured with 'Simultaneously Ring'.

3.5.3 Resolved Constraints in Version 2.0.9.127

The following constraints from the previous release have now been resolved in Version 2.0.9.127

- Japanese language:
  - Incorrect syntax in UI was fixed.
  - Support Japanese input on the phone keypad for searching in the Corporate directory.
  - Cannot use the phones' Voicemail when the phone is set to the Japanese language.
- Hungarian language:
  - The username is not displayed when calling to the Lync Team Call Group.
  - An additional letter is added when transferring a call.
- Non-Latin languages can't be set on the Web Interface for the Contact Name.
- The phone ringer volume does not save and returns to the default settings after reboot.
- Retrieving a parked call does not function using the # key. The * key, or any digit key can be used instead.
- 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.
- Call Park doesn't work when 'callParkServerUri' is set to a length larger than 128 characters.
- The phone does not failover to the Redundant pool when the first pool is down.
- Setting the Call Forwarding option in the Lync client deletes the call pickup code from the phone.
- Boss-Admin – Boss presence remains after Admin delegation is cancelled in Lync 2010.
- Presence indication is missing when searching in the Corporate directory contacts from different Lync pools.
- Configuration of VLAN priority on LAN & PC ports doesn’t work well. Packets are always tagged with priority 0 even if VLAN priority was set to a different value.

3.6 Version 2.0.9.93

3.6.1 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.

3.7 Version 2.0.9.65

3.7.1 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.

3.8 Version 2.0.7

3.8.1 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**

3.9 Version 2.0.5 - BToE

3.9.1 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.

3.10 Version 2.0.3 - BToE

3.10.1 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).

3.11 Version 2.0.3

3.11.1 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.

3.12 Version 2.0.1.44.21

3.12.1 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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