Release Notes

AudioCodes High-Definition IP Phones Series

400HD IP Phones

Version 2.2.16.664

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Abbreviations and Conventions

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

Related Documentation

Document Name
405 and 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

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1 Introduction

This document describes the new features, known constraints, and resolved constraints of AudioCodes' 400HD Series IP Phones for Version 2.2.16.

1.1 400HD Series IP Phones Overview

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

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

1.2 Specifications

The phones' software specifications are summarized in the following table:

Table 1-1: Software Specifications

Feature	Details
VoIP Signaling Protocols	■ SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	■ IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling ■ SIP over TLS (SIPS) ■ 802.1p/Q for Traffic Priority and QoS ■ VLAN Discovery Mechanism (CDP, LLDP) ■ ToS (Type of Service) field, indicating desired QoS DHCP Client ■ NTP Client ■ Open SSL 1.0.1, integrated with TLS 1.2, replaced OpenSSL 0.9.8 and TLS 1.0, supporting SHA256.
Media Processing	■ Voice Coders: G.711, G.729A/B, G.722 and OPUS v1.1 (only 1 GigE phone models are supported). About OPUS v1.1: • The encoder supports any sampling frequency up to 16 kHz • The encoder supports 10 msec to 120 msec packet time • The decoder can receive any stream (all modes, mono or stereo, any sampling frequency 8 to 48 kHz) • The decoder can receive any packet time apart from 2.5 msec and 5 msec (in 'CELT only' mode only 20 msec packet time is supported) • decoder performs up/down sampling and renders the signal as wideband • Jitter Buffer size is 2 seconds ■ 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	■ Call Hold / Un-Hold ■ Call Transfer ■ Three-way Conferencing (with local mixing) ■ Remote Conference compliancy with RFC 4579, SIP Call Control, Conferencing for UAs ■ Redial ■ Auto Redial ■ Caller ID Notification ■ Call Waiting Indication ■ Message Waiting Indication (including MWI LED) ■ Local and Corporate Directories ■ Automatic On-hook Dialing ■ Automatic Answering (Alert-Info header and “talk” event) ■ CWRR (Call Waiting Reminder Ring) ■ Call Logs: Missed/Received Calls and Dialed Numbers ■ Speed Dial ■ Dial Plan ■ Call Forwarding ■ ACD Hoteling (Broadsoft and Genesys Contact Centers with 420HD phones) ■ Genesys contact center phones (405, 420HD, 430HD/440HD models) feature the BroadSoft ACD, including ACW and Reason for Not Ready. ■ BroadSoft Feature Key Synchronization for server-controlled DnD and Call Forward ■ BroadSoft Shared Call Appearance (applies only to 430HD and 440HD phones) ■ BroadSoft Device Registration Failover ■ Handles up to 8 concurrent calls
Configuration / Management	■ LCD Display User Interface Language Support (Various Languages) ■ Web-based Management (HTTP/HTTPS) ■ European date format (DDMMYYYY) and American date format (MMDDYYYY), in phone LCD and Web interface ■ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and proprietary configuration file upgrade ■ DHCP options (66, 67, and 160) for auto-provisioning ■ DHCP options (120, 60, and 77) for device information ■ DHCP option (42 or 4) for the NTP server ■ DHCP option (43) for vendor specific information ■ DHCP option (2) for the Time Zone Offset ■ Redirect server ■ LDAP (Lightweight Directory Access Protocol) ■ Private Labeling Mechanism ■ Configuration file encryption (Entire file and individual parameters)
Diagnostics and Troubleshooting Tools	■ System Logging (Syslog), Lightweight Syslog, and Tracing ■ Monitoring (Ping and Traceroute) ■ DSP Recording ■ Port Mirroring ■ VoIP Status Web page ■ Firmware Recovery ■ Packet Recording ■ Crash Dump File ■ RTCP-XR Quality of Experience Reports

Feature	Details
Supported Languages	■ English ■ Spanish ■ Russian ■ German ■ Ukrainian ■ French ■ Canadian French (Français Canadien) ■ Italian ■ Hebrew ■ Polish ■ Portuguese (displayed only if included in your Feature Key) ■ Finnish ■ Korean ■ Simplified Chinese ■ Traditional Chinese ■ Turkish ■ Japanese (Kanji, Hiragana, and Katakana input modes) ■ Hungarian ■ Slovak ■ Czech
Supported Headsets	■ Jabra UC-150 ■ Jabra Speak 510+ ■ Jabra Speak 410 ■ Jabra MOTION OFFICE ■ Jabra PRO 9470 ■ Jabra Evolve Series 20, 30, 40, 75, 80 ■ Microsoft LX-3000 ■ Plantronics C-310M ■ Plantronics C-320M ■ Plantronics HW720 ■ Plantronics Blackwire Series 300, 325, 510, 520, 710 ■ Jabra Pro 920 EHS wireless headset ■ Jabra Pro 9450 EHS wireless headset ■ Axtel For the comprehensive list of Jabra headsets, see the Jabra Headset Compatibility Guide For the comprehensive list of Plantronics headsets, see https://compatibility.plantronics.com/deskphone * * **Important!** When plugging in a USB headset, the ringer will not be heard on incoming call

1.3 400HD Series IP Phone Models

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

Table 1-2: 400HD Series IP Phone Models

Part Number	Product Description
405	405 IP Phone (black) with Power over Ethernet (PoE)
405HDEG	405HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)

Part Number	Product Description
405HDEPSG	405HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100 ports, 4 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
420HDE	420HD IP Phone PoE Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEG	420HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEW	420HD IP Phone PoE White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPS	420HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPSG	420HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
420HDEPSW	420HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 4 Programmable keys, 128x48 Graphic LCD and Power over Ethernet (PoE)
430HDE	430HD IP Phone PoE Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEG	430HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEW	430HD IP Phone PoE White 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEPS	430HD IP Phone PoE and external power supply Black 2 Ethernet 10/100 ports, 18 Programmable keys, 132x64 Graphic LCD and Power over Ethernet (PoE)
430HDEPSW	430HD IP Phone PoE and external power supply White 2 Ethernet 10/100 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)
430HDEPSG	430HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD and Power over Ethernet (PoE)
440HDEG	440HD IP Phone PoE GbE Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEWG	440HD IP Phone PoE GbE White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEPSG	440HD IP Phone PoE GbE and external power supply Black 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)
440HDEPSWG	440HD IP Phone PoE GbE and external power supply White 2 Ethernet 10/100/1000 ports, 18 Programmable keys, 256x128 Graphic LCD, 376x60 BLF LCD and Power over Ethernet (PoE)

2 Version 2.2.16.664

2.1 What's New in Version 2.2.16.664

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

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

2.2 Resolved Constraints from Previous Versions

Table 2-1: Resolved Generic Constraints

Incident	Description
IGS-3751	All sensitive information should be excluded from the configuration file (CVE-2023-22957).
IGS-3722	**Zoom**: Fail to failback from backup cloud server to primary cloud server. (This only happens when multiple lines are configured).

2.3 Known Constraints in Version 2.2.16.664

None.

3 Previous Releases

3.1 Version 2.2.16.663

3.1.1 What's New in Version 2.2.16.663

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

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

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

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

3.1.2 Resolved Constraints from Previous Versions

The table below shows the constraints that were resolved in previous versions.

Table 3-1: Resolved Constraints

Incident	Description
IGS-3724	BLF data is not retrieved from SIP messages.
IGS-3726	Incoming call number displayed full URI after upgrade from 2.2.12 to 2.2.16

3.1.3 Known Constraints in Version 2.2.16.663

None.

3.1.4 Known Constraints from Previous Versions

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

Table 3-2: Known Constraints in Previous Versions

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.

Incident	Description
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There’s an incorrect line list in the ‘Do not Disturb’ screen and the ‘Forward’ screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone’s Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.

Incident	Description
IGS-3234	Japanese characters cannot be shown correctly if the phone system's language is not Japanese.

3.2 Version 2.2.16.643

3.2.1 What's New in Version 2.2.16.643

1. Support Zoom Phone Local Survivability (ZPLS) feature.

Configuration details:

Parameter	Description
voip/signalling/sip/redundant_outbound_proxy/enabled	Enable redundant outbound proxy. Default: 0. ⓘ**Attention:** Should be enabled.
voip/signalling/sip/sip_outbound_proxy/enabled	Enable outbound proxy. Default: 0. ⓘ**Attention:** Should be enabled.
voip/signalling/sip/sip_outbound_proxy/addr	Primary Outbound proxy address. Default: (IP Address or Host Name). ⓘ**Attention:** Should be set.
voip/signalling/sip/sip_outbound_proxy/port	65535 send DNS SRV query. 1024~65534 send DNS A query. Default: 5060. ⓘ**Attention:** Should be set to 65535 for ZPLS.
voip/signalling/sip/redundant_outbound_proxy/keepalive_period	Keep Alive (OPTIONS) interval in seconds. Default: 60. Attention: Set on demand.
voip/signalling/sip/redundant_outbound_proxy/address	Redundant outbound proxy address. Default: (IP Address or Host Name). ⓘ**Attention:** Should be set.
voip/signalling/sip/redundant_outbound_proxy/port	65535 send DNS SRV query. 1024~65534 send DNS A query. Default: 5060. ⓘ**Attention:** for ZPLS no need SRV, so need to set the right port.
voip/signalling/sip/switch_redundant_to_primary/timer	The time interval before the IPP failback after the primary proxy detection succeeds, Default: 0. ⓘ**Attention:** Set on demand.

Parameter	Description
voip/signalling/sip/switch_to_DNS_primary/timer	Relevant only for failover between the servers resolved from the primary outbound proxy. The time interval before the IPP executes the failback. Default: 0. ⓘ**Attention:** Set on demand.

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

Parameter	Description
voip/dns_cache/mode	■ STATIC_DNS_CACHE_FIRST - Use static DNS cache preferentially. ■ DNS_QUERY_FIRST - Use DNS server resolution preferentially.
voip/dns_cache_A/[0-7]/name	A record name
voip/dns_cache_A/[0-7]/ip	A record IP address
voip/dns_cache_A/[0-7]/ttl	A record TTL
voip/dns_cache_srv/[0-7]/name	SRV record name
voip/dns_cache_srv/[0-7]/port	SRV record port
voip/dns_cache_srv/[0-7]/weight	SRV record weight
voip/dns_cache_srv/[0-7]/priority	SRV record priority
voip/dns_cache_srv/[0-7]/target	SRV record target
voip/dns_cache_srv/[0-7]/ttl	SRV record TTL
voip/dns_cache_naptr/[0-7]/name	NAPTR record name
voip/dns_cache_naptr/[0-7]/flag	NAPTR record flag
voip/dns_cache_naptr/[0-7]/order	NAPTR record order
voip/dns_cache_naptr/[0-7]/preference	NAPTR record preference
voip/dns_cache_naptr/[0-7]/replace	NAPTR record replace
voip/dns_cache_naptr/[0-7]/service	NAPTR record service
voip/dns_cache_naptr/[0-7]/ttl	NAPTR record TTL

3. Added Russian language to 405/405HD phones.
4. Support Intrado ERS Location Information Service (HELD)

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

Table 3-3: HELD Configuration

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

Table 3-4: HELD Status

Parameter	Description
status/diagnostics/lldp/chassis/chassisId	Chassis ID of the switch that the device is connected to. MAC address: 12-digit hexadecimal number represented by colon- hexadecimal notation. Interface Name: String format. Example: <ChassisID>AgcEiFqSa oqA</ChassisID>
status/diagnostics/lldp/chassis/portId	ID of the switch port that the device is connected to. MAC address: 12-digit hexadecimal number represented by colon- hexadecimal notation. Interface Name: String format. Example: <PortID>BAkFR2kyLzA vMTc=</PortID>
status/diagnostics/lldp/chassis/chassisIdType	ERS supports the following Chassis ID subtypes: ■ "Switch Hostname" / (6) ■ "Switch IP" /(5) ■ "Switch MAC Address"/(4)
status/diagnostics/lldp/chassis/portIdType	ERS supports the following Port ID subtypes: ■ "Port Name" (5) ■ "Port MAC Address" /(3)

3.2.2 Resolved Constraints from Previous Versions

The table below shows the constraints that were resolved in previous versions.

Table 3-5: Resolved Constraints

Incident	Description
IGS-3600	Ping function removed from Web interface due to security reasons.
IGS-3596	EAP TLS identity parameter introduced.
IGS-3542	Russian language was missing on 405/405HD.
IPPUC-8748	Removed TCP listening when phone is configured to SIP over UDP.
IGS-3641	Phone lost connection to DM after Web interface disabled.
IGS-3645	Added http authentication configuration to Web.
IGS-3639	3PCC hold stopped working after multiple switches between calls.
IGS-3653	Updated Comode CA certificate.
IGS-3604	Phone randomly rebooted.
IGS-3654	Do Not Disturb and Call Forwards Soft Keys stopped working after a while.
IGS-3703	Microsoft RSA Root Certificate Authority 2017 was added to CA certificates.

3.2.3 Known Constraints in Version 2.2.16.643

None.

3.2.4 Known Constraints from Previous Versions

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

Table 3-6: Known Constraints in Previous Versions

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.

Incident	Description
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
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104610	[SIP] The Transfer softkey appears when starting a conference.
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102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There’s an incorrect line list in the ‘Do not Disturb’ screen and the ‘Forward’ screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone’s Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.
IGS-3234	Japanese characters cannot be shown correctly if the phone system’s language is not Japanese.

3.3 Version 2.2.16.578

3.3.1 What's New in Version 2.2.16.578

Version 2.2.16.578 offers the following new feature:

- Added unique User Agent Identifier for SIP register allowing identify of an agent without using its IP address. Support requires enabling the following parameter: `voip/signalling/sip/add_sip_instance/enable`
- Add parameter /network/lan/vlan/vlan_switch/enable to enable or disable VLAN switching on the fly (Default = disabled).
- Add Danish language support

3.3.2 Resolved Constraints from Previous Versions

The table below shows the constraints that were resolved in previous versions.

Table 3-7: Resolved Constraints

Incident	Description
IGS-3439	405HD is rebooting when using USB headset
IGS-3455	Unable to resume the call after second hold
IGS-3338	EAP-TLS 1.2 support
IGS-3485	405HD are facing random reboots.

3.3.3 Known Constraints in Version 2.2.16.578

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

Table 3-8: Known Constraints

Incident	Description
IGS-3630	Unable to hear ringing on USB headset

3.3.4 Known Constraints from Previous Versions

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

Table 3-9: Known Constraints in Previous Versions

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.

Incident	Description
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.

Incident	Description
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ''voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There's an incorrect line list in the 'Do not Disturb' screen and the 'Forward' screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone's Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.
IGS-3234	Japanese characters cannot be shown correctly if the phone system's language is not Japanese.

3.4 Version 2.2.16.557

3.4.1 What's New in Version 2.2.16.557

Version 2.2.16 offers the following new feature:

- New reject code parameters

Parameter	Description
voip/services/dnd_reject_code	Configures the dnd reject code that the phone sends when the **Reject** softkey is pressed or while DND is activated. Valid values are between 400 to 699 (default 603)

Parameter	Description
voip/services/other_reject_code	Configures other reject code that the phone sends when the **Reject** softkey is pressed or while DND is activated. Valid values are between 400 to 699 (default 486)
voip/services/sk_reject_code	Configures the softkey reject code that the phone sends when the **Reject** softkey is pressed or while DND is activated. Valid values are between 400 to 699 (default 603)

3.4.2 Resolved Constraints from Previous Versions

The table below shows the constraints that were resolved in previous versions.

Table 3-10: Resolved Constraints

Incident	Description
IGS-3385	Pressing DND button on the phone changes the status for that user only on the phone, and the status on the soft client remains unchanged i.e. it shows "Available"
IGS-3387	IPP failed to configure "reject code" via web
IGS-3317	IPP sends the RTP packets to the wrong port
IGS-3404	VOIP reject code backward compatibility
IGS-3356	Zoom phone sometimes sends "du..." tone when making a paging call
IGS-3409	Zoom phone can't hear the caller when a paging call barge in
IGS-3405	Phone receives UPDATE with with new number in P-Asserted-Identity header, but does not change the calling number on LCD screen.
IGS-3347	Sometimes auto-answer call has no audio
IGS-3273	Support NTP status screens
IGS-3430	Genband shareline callLog screen displays nothing during transfer
IGS-3442	Ethernet test enhancement
IGS-3398	Can't mute ringing tone
IGS-3444	GMT offset should be 4 digits
IGS-3451	Zoom phone display name (remote id) in call log is cut
IGS-3450	Zoom phone register messages printed twice in log
IGS-3441	Phone displays "Not available" notification with cropped caption
IGS-3433	Zoom phones transfer to VM using PSK refer support
IGS-3440	For SDP with two m lines, phone replies with one m line

3.4.3 Known Constraints from Previous Versions

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

Table 3-11: Known Constraints in Previous Versions

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.

Incident	Description
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ''voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There's an incorrect line list in the 'Do not Disturb' screen and the 'Forward' screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone's Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.
IGS-3234	Japanese characters cannot be shown correctly if the phone system's language is not Japanese.

3.5 Version 2.2.16.550

3.5.1 What's New in Version 2.2.16.550

Version 2.2.16 offers the following new feature:

- Support for Microsoft Teams SIP Gateway. See here for the Microsoft article 'Enable core Microsoft Teams calling functionality on compatible legacy SIP phones with SIP Gateway'). The SIP Gateway lets organizations use AudioCodes' SIP devices with Microsoft Teams to leverage existing investments in SIP devices. You can now sign-in to Teams and make and receive calls with the following AudioCodes SIP phones for personal use or for use and mainly as common area phones (CAPs):
  - 405
  - 405HD
  - 420HD
  - 430HD
  - 440HD

  Using SIP Gateway, your users can do all of the following:
  - Make calls: Users using SIP devices can make calls to the Public Switched Telephone Network (PSTN), to other SIP devices, and to Teams and Skype for Business users. Users using SIP devices can only call users who have phone numbers.
  - Receive calls: The device can receive calls from any Teams or Skype for Business client, or from other SIP devices connected to the Teams SIP Gateway as long as the calling user has a valid phone number.
  - Multiple simultaneous calls: A SIP device user in a call can put the call on hold to make or receive other calls. A SIP device user can also conference two calls.
  - Do not disturb: A SIP device user can set do not disturb on the device so that the device will not ring for incoming calls. This has no impact on the user's status on all other Teams endpoints.
  - Hold/Resume and Mute/Unmute: A SIP device user can hold and resume or mute and unmute a call by using the features for those actions on the device.
  - Voicemail: SIP device users can listen to electronically stored voice messages that callers leave for them.
  - Message waiting indicator: SIP device users can receive notifications that alert them when they have new voicemail messages.
  - Sign-in and sign-out: SIP devices users can sign in and sign out of Teams on the device.
  - Dual-tone multi-frequency: SIP device users can press number keys to provide input during interactive voice response calls.
  - Teams meetings: A SIP device user can join a Teams meeting by dialing the meeting access number. Dialing out to a same organization user's phone number is currently not supported. However, guest users from another organization can be added to a Teams meeting by a participant who dials out to a guest user's number to include that guest. NOTE: Adding a Teams meeting participant via "request to join" currently won't alert a SIP device.
  - Call transfers: SIP device users can transfer calls. SIP Gateway supports both blind and consultative transfers.
  - Local call forwarding: A SIP device user can set forwarding rules (always, on timeout, and busy) for the device. If the device is connected to the SIP Gateway, then the call will be redirected to the target address based on the rule that the device user set. To make local call forwarding work, the admin must set the AllowCallRedirect attribute in Set-CsTeamsCallingPolicy to Enabled.

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

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

3.5.2 Known SIP Gateway Constraints

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

Table 3-12: Known Constraints

<table>
<tr><th>Incident</th><th>Description</th></tr>
<tr><td>-</td><td>Teams users must have a phone number with PTSN calling enabled to use SIP Gateway.

Dialing via URI is not supported. Dialing to a different organization via URI will not be possible.
For joining a conference via dial-in, users must have a DID or phone number with PTSN calling enabled. The conference bridge must have a DID number.
</td></tr>
<tr><td>-</td><td>
The following features are by design unsupported:

Click to Join and Conference Roster
N-way conference (only three-way conference and dial-in to conference are supported)
Calendar
Visual Voicemail
Hot Desking
Search for a Contact
Presence
Discreet Call
Device Manager (OVOC plugin) (Roadmap)
Sign-in

With Username and Password
Web login

Dynamic location for E911 calls (SIP devices use static location for emergency calls, defined by DID number/user and not according to the real location defined by the switch).

</td></tr>
</table>

3.5.3 Resolved Constraints from Previous Versions

The table below shows the constraints that were resolved in previous versions.

Table 3-13: Resolved Constraints

Incident	Description
IGS-3328	There is no option to switch to "Not Ready" and select a reason during a call.
IGS-3319	When programming the softkeys for lines, you cannot “skip” a position. These must be filled in order 1-6.
IGS-3257	The called phone’s description appears on the phone screen during a call.
IGS-3323	On-call transfer scenario requires an "Ack" to delete the transfer message.
IGS-3271	Incorrect transfer translation in the Korean language.
IGS-3322	"parking_lot_number" is missing in the phone screen when loaded from configuration file.
IGS-3320	Support should exist for controlling hard keys events using configuration file parameter "system/long_key_press_timeout" [range 800-5000mSec, default:800].
IGS-3325	After calling forward is canceled, the active line is last on the list.
IGS-3255	The default “Not Ready” reason does not comply to the "unavailable_reason" parameter.
IGS-3321	Pause dialing does not function correctly.
IGS-3286	A "488 not acceptable here" code reply is sent from the phone when receiving Session Description Protocol (SDP) with both AVP and SAVP.

Incident	Description
IGS-3222	In scenarios involving two calls, pressing the "End" softkey disconnects both calls.

3.5.4 Known Constraints from Previous Versions

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

Table 3-14: Known Constraints in Previous Versions

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ▪ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ▪ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ▪ A DNS query is sent instead of an SRV query with priorities ignored. ▪ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.

Incident	Description
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ''voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There's an incorrect line list in the 'Do not Disturb' screen and the 'Forward' screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone's Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.

Incident	Description
IGS-3234	Japanese characters cannot be shown correctly if the phone system's language is not Japanese.

3.6 Version 2.2.16.538

3.6.1 What's New in Version 2.2.16.538

Version 2.2.16 offers the following new feature:

- Support for the Latvian language

3.6.2 Resolved Constraints in Version 2.2.16.538

The table below shows the constraints that were resolved in this version.

Table 3-15: Resolved Constraints

Incident	Description
IGS-3328	There is no option to switch to "Not Ready" and select a reason during a call.
IGS-3319	When programming the softkeys for lines, you cannot "skip" a position. These must be filled in order 1-6.
IGS-3257	The called phone's description appears on the phone screen during a call.
IGS-3323	On-call transfer scenario requires an "Ack" to delete the transfer message.
IGS-3271	Incorrect transfer translation in the Korean language.
IGS-3322	"parking_lot_number" is missing in the phone screen when loaded from configuration file.
IGS-3320	Support should exist for controlling hard keys events using configuration file parameter "system/long_key_press_timeout" [range 800-5000mSec, default:800].
IGS-3325	After calling forward is canceled, the active line is last on the list.
IGS-3255	The default "Not Ready" reason does not comply to the "unavailable_reason" parameter.
IGS-3321	Pause dialing does not function correctly.
IGS-3286	A "488 not acceptable here" code reply is sent from the phone when receiving Session Description Protocol (SDP) with both AVP and SAVP.
IGS-3222	In scenarios involving two calls, pressing the "End" softkey disconnects both calls.

3.6.3 Known Constraints in Version 2.2.16.538

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

Table 3-16: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.

Incident	Description
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.

Incident	Description
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There’s an incorrect line list in the ‘Do not Disturb’ screen and the ‘Forward’ screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone’s Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.
IGS-3234	Japanese characters cannot be shown correctly if the phone system’s language is not Japanese.

3.7 Version 2.2.16.501

3.7.1 What’s New in Version 2.2.16.501

Version 2.2.16 offers the following new features:

- N/A

3.7.2 Resolved Constraints in Version 2.2.16.501

The table below shows the constraints that were resolved in this version.

Table 3-17: Resolved Constraints

Incident	Description
IGS-3220	Capability was added to switch ‘Not Ready’ reasons without having to go via ‘Ready’ state.
IGS-3236	In some environments, after an incoming RE-INVITE to the phone, the phone replies with AVP instead of AVPF (SRTP).
IGS-3192	When using dual registration, the Shared Call Appearance feature functions incorrectly.
IGS-3204	Using the ‘upload.cgi’ web page for provisioning triggers a phone reboot.

3.7.3 Known Constraints in Version 2.2.16.501

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

Table 3-18: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.

Incident	Description
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There’s an incorrect line list in the ‘Do not Disturb’ screen and the ‘Forward’ screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.

Incident	Description
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone’s Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.
IGS-3234	Japanese characters cannot be shown correctly if the phone system’s language is not Japanese.

3.8 Version 2.2.16.480

3.8.1 What’s New

Version 2.2.16 includes firmware build **2.2.16.480**

Version 2.2.16 offers the following new features:

- **LDAP integration with Zoom**
- AudioCodes phones for Zoom now support Hot Desking.
- See also https://support.zoom.us/hc/en-us/articles/360043841032-Using-hot-desking-for-phones
- The Web server whitelist is now supported on phones.

3.8.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-19: Resolved Constraints

Incident	Description
IGS-2949	[420HDG] In some environments, the phone restarts at the end of the call.
IGS-2938	[405HD] The display name does not appear on the phone’s display when the LDAP-based Corporate Directory is configured for display name manipulation.
IGS-2961	In some environments, the phone loses the network connection and boots up with VLAN Data.
IGS-2973	[405HD] The phone reports an incorrect power consumption value in the LLDP packet.
IGS-2940	Changing the value of 'system/user_name' causes the phone to take VLAN DATA.
IGS-2928	In some environments, the phone sets the incorrect VLAN based on the CDP packet.
IGS-3012	The reminder beep is always played over the speaker.
IGS-3011	The phone replies with a ‘488’ to re-INVITE with SDP including two media lines: RTP/AVP and RTP/SAVP.
IGS-2988	The phone isn’t displaying the HuntGroup number once the call is answered.
IGS-3141	[420HD FE and 405] A noise is heard during MoH and a regular call.
IGS-3126	[LDAP] The contact name is not displayed in the ‘Ongoing Call’ screen.
IGS-3127	[LDAP] On some occasions, the phone doesn't display the Remote CALLER-ID as the LDAP contact name in the Call Log screen.
IGS-3128	[LDAP] The phone doesn't display an LDAP contact name in the Consultation Transfer screen.

Incident	Description
IGS-3074	[Multicast Paging] [440HD] The multicast address information is incompletely displayed.
IGS-3075	[Multicast Paging] The phone allows users to define a paging speed dial with an illegal multicast address.
IGS-3101	[Multicast Paging] [440HD] There is no option to configure a paging Programmable Key / Function Key with the default values, from the 440HD phone screen.
IGS-3130	[Supervisor listen] The remote side cannot be heard through the handset after switching the audio device while the supervisor is listening.
IGS-3106	[Supervisor listen] The agent does not hear voice via the headset after the supervisor on-hooks the handset.
IGS-3131	[Supervisor listen] The supervisor can't hear a resumed call.
IGS-3099	When the phone is configured with 'Redundant Proxy' = **Primary-fallback**, it doesn't failover to the second proxy when the first proxy is down.
IGS-3100	On some occasions during a 3-way conference, the phone shows party B twice instead of showing party B and C.
IGS-3062	The first incoming call immediately after closing a 3-way conference call doesn't appear in the phone screen.
IGS-3102	Long calls may be disconnected after 30 minutes when the phone is connected to a Proxy that needs to refresh the SIP session.
IGS-3049	On some occasions, the Call Log is empty when trying to make a call from Call Log.
IGS-3051	A bad response occurs when inputting ABC on the 420HD phone model.
IGS-3072	The date format doesn't change to American format.
IGS-3024	The mute and volume hard keys on the phone are available when the phone screen displays a greeting.
IGS-3023	No beep occurs after a greeting message.
IGS-3066	The phone sends a SUBSCRIBE message to the proxy with a lower priority even when the phone is registered to the proxy with a higher priority.
IGS-3046	[Redundant Proxy-simultaneous mode] Enabling the KeepAlive option causes the phone not to send a SIP REGISTER message to the first proxy if the first proxy goes down.

3.8.3 Known Constraints in Version 2.2.16

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

Table 3-20: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments, which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ▪ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ▪ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ▪ A DNS query is sent instead of an SRV query with priorities ignored. ▪ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.

Incident	Description
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.
IGS-3157	The Gains values are different to the expected default values.
IGS-3122	There is noise in 3WC with the OPUS codec.
IGS-3104	[Genband] There’s an incorrect line list in the ‘Do not Disturb’ screen and the ‘Forward’ screen.
IGS-3090	The dial tone volume doesn't change in accord with pressing the volume buttons.
IGS-3076	When more than one paging is configured, phone 2 doesn't get incoming paging.
-	[420HD FE] Setting VLAN PC port from the phone’s Web interface doesn't affect the PC VLAN configuration; reconnecting the PC cable does.

3.9 Version 2.2.16.428

3.9.1 What’s New

Version 2.2.16 includes firmware build **2.2.16.428**.

Version 2.2.16 offers the following new features:

- The phone supports **an option to switch from ACW (After Call Work) to 'Not Ready' state**.

3.9.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-21: Resolved Constraints

Incident	Description
IGS-2949	[420HDG] In some environments, the phone restarts at the end of the call.
IGS-2938	[405HD] The display name does not appear on the phone's display when the LDAP-based Corporate Directory is configured for display name manipulation.
IGS-2961	In some environments, the phone loses the network connection and boots up with VLAN Data.
IGS-2973	[405HD] The phone reports an incorrect power consumption value in the LLDP packet.
IGS-2940	Changing the value of 'system/user_name' causes the phone to take VLAN DATA.
IGS-2928	In some environments, the phone sets the incorrect VLAN based on the CDP packet.

3.9.3 Known Constraints in Version 2.2.16

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

Table 3-22: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.

Incident	Description
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ''voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.

Incident	Description
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.

3.10 Version 2.2.16.408

3.10.1 What's New

Version 2.2.16 includes firmware build **2.2.16.408**.

Version 2.2.16 offers the following new features:

- N/A

3.10.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-23: Resolved Constraints

Incident	Description
IGS-2908	In some environments, the phone identifies a VLAN priority and therefore changes to Native VLAN.
IGS-2873	The phone fails to register to the Secondary Proxy if using the same IP address as the Primary Proxy but with a different port.
IGS-2903	For an incoming call, the phone doesn't beep neither on the speaker nor on the headset.
IGS-2835	The phone doesn't use SRV records to complete registration to the SIP proxy.
IGS-2858	The phone screen doesn't show numbers of 10 digits or more.
IGS-2801	The phone doesn't send OPTIONS/Register to the Primary Poxy after getting a 404 from the Redundant Proxy.

3.10.3 Known Constraints in Version 2.2.16

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

Table 3-24: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ▪ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ▪ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ▪ A DNS query is sent instead of an SRV query with priorities ignored. ▪ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.

Incident	Description
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ''voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.

3.11 Version 2.2.16.400

3.11.1 What's New

Version 2.2.16 includes firmware build **2.2.16.400**.

Version 2.2.16 offers the following new features:

- **A new option 'Auto Answer incoming calls' has been added** to the phone's Settings screen (MENU > Settings). The option allows users to disable/enable the auto-answer feature.

3.11.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-25: Resolved Constraints

Incident	Description
IGS-2827	The Call Forward feature doesn't function correctly if the phone has multiple lines.
IGS-2814	QoE does not function if the SIP protocol is different to the 'QoE Publish' protocol.
IGS-1806	[430HD/440HD] The 'Hebrew' language option appears incorrectly under the Language Settings menu.
IGS-2802	The BLF 'Call Pickup' feature does not function.
IGS-2766	Time Zone: GMT -09:00 is missing

3.11.3 Known Constraints in Version 2.2.16

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

Table 3-26: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.

Incident	Description
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.
IGS-2795	The Blind Transfer softkey **BXfer** is displayed in the wrong format when operating the phone in the Spanish language interface.
IGS-2799	[Genesys environment] A user to whom an incoming call from PTSN is transferred, cannot transfer the call using the Attendant Transfer method to another user.
IGS-2793	[Broadsoft environment] In an incoming PSTN call to a call center number, the name of the call center is missing if the name is configured on the server.
IGS-2573	Pentest (Penetration Testing) vulnerability to command injection. Vulnerability on command.cgi web page.

3.12 Version 2.2.16.376

3.12.1 What's New

Version 2.2.16 includes firmware build **2.2.16.376**.

Version 2.2.16 offers the following new features:

- **Remote conference call for Genband** (compliant with RFC 4579) allows 'N' participants (compared to three participants in a local 3-way conference). The network administrator must enable the feature on Genband's server and on the phone.
- **Shared Line Appearance (SLA) for Genband** [applies to 430HD and 440HD phone models]. Enables a group of user phones to share a line and for users to make/receive calls that then *appear* to be made/ended to/from that line. The phones in the group behave as extensions of that line.
  - When a user uses a shared line, all phones in the group are notified.
  - Multiple simultaneous calls are supported.
  - User status (busy or idle, for example) is displayed on all phones in the group.
  - The group has a designated Primary SLA member and one or more Secondary SLA members.
  - Any restrictions and services on the Primary SLA member apply to all group members.
  - SLA can currently be configured using Single Call Arrangement (SCA): With this method, the number of calls that can be set up simultaneously across all user phones in the group is limited to one. When one of the user phones is active in a call (incoming or outgoing), the other phones are blocked from receiving or making additional calls.
  - If an incoming call to a shared line occurs while a call is already active on it, the call receives busy treatment.
- **Option 57 in the phone's DHCP request**. Option 57 is used to define the maximum-length DHCP message that a client (phone) will accept. The minimum DHCP message size supported is 576 bytes. The maximum DHCP message size supported is 1000 bytes (DHCP Option 57).
- **'Auto Answer incoming calls' can now be disabled or enabled using the phone's programmable keys**. A new 'personal_settings/menu/callautoanswering/enabled' configuration file parameter has been added. Default: True. The setting allows network administrators to control the Call AutoAnswer option appearance in MENU > Settings.
- **A random mechanism has been added for the REST_API Keepalive** that's sent to the Device Manager. The motivation is to prevent an overload on the server if all phones go up at once, for example, after electric power is interrupted.
- **A random mechanism has been added for configuration file provisioning** (when the timer is set to check 'every x minutes' or 'hourly'). The motivation is to prevent an overload on the server if all phones go up at once, for example, after electric power is interrupted.
- **Call Park is supported on the MetaSwitch application server**.
- **BLF Subscribe is now allowed** when the phone is configured for a generic application server.
- **AudioCodes' new corporate logo is now displayed on the 405 and 405HD phones**. The logo is also displayed in these phones' management interfaces.
- The **VLAN interface can now be changed 'on the fly'** during regular phone operation.

3.12.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-27: Resolved Constraints

Incident	Description
IPPUC-2258	A call with VocaNOM disconnects before the caller's request is transferred to the requested party.
IGS-2716	In the Device Manager, the phone is displayed with the status of 'Started' in the case of incorrect credentials.
IGS-2174	The phone doesn't unmute automatically when it gets a second ongoing call.
IGS-2046	In some environments, a call using the OPUS vocoder fails due to a missing semi-colon at the end of the OPUS a=fmtp: line in SDP.
IGS-2298	The phone does not display the name of a dialed contact if the caller name info arrives as part of the SIP OK 'To' header.
IGS-2108	Calls are sporadically not saved in the Call Log.
IGS-2263	When using HTTPs and in the case of a very long provisioning URL, the phone fails to send HTTP POST to the Device Manager.
IGS-1958	The phone can't return to the first call after the second call is disconnected by the remote side.
IGS-1997	Media streaming / SRTP is incorrectly displayed in the Web interface page.
IGS-1928	The first generated SRTP packets are encrypted with a 'zero' key.
IGS-2613	The Supervisor Listen feature isn't functioning.
IGS-2680	On rare occasions the phone stops sending a keep-alive message. After a prolonged period, this causes the Device Manager to display the status as disconnected.
IGS-2338	The user is unable to change the ringtone if the OPUS codec is configured and used.
IGS-2526	A loud, continuous beep is played after a call is held for 40 seconds.
IGS-2114	[420HD] One-way voice occurs in a conference call with the PSTN using the OPUS codec.
IGS-2375	Incorrect 'Chinese traditional' is displayed during a call transfer scenario.
IGS-2426	The INVITE is followed by a BYE message when call disconnect is in hold state.
IGS-2398	When the configuration file parameter 'system/dnd/show_softkey' is disabled, DnD remains displayed in the phone's screen.
IGS-2240	The phone doesn't set VLAN priority according to the policy priority header.
IGS-2165	The phone doesn't answer incoming calls immediately if the value configured for the configuration file parameter 'voip/advanced_auto_answer/timeout' is 0.
IGS-2003	[405HD] A delay of approximately 0.5 of a second occurs on the headset when beep and auto-answer are enabled.
IGS-1877	[405 generic] The audio device changes unexpectedly from headset to speaker in the middle of a call.
IGS-1965	If the phone is left off-hook after a call is disconnected and if another call then comes in, the ringer volume differs from the volume configured.
IGS-1896	The beep reminder played when a call is put on hold for a prolonged period is enabled by default.
IGS-1913	Even though the phone receives a CANCEL almost immediately after the initial INVITE, the headset continues to ring.

3.12.3 Known Constraints in Version 2.2.16

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

Table 3-28: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer of a call from an existing three-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.

Incident	Description
102786	[440HD in a BroadSoft environment] BLF supports a list of only 23 entries; the names in the list are middle-size.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones when the ‘'voice_quality/mode' parameter is configured differently on each. This can happen for example when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.

3.13 Version 2.2.16.251

3.13.1 What’s New

Version 2.2.16 includes firmware build **2.2.16.251**.

- **OpenSSL** has been upgraded to version 1.0.2p.
- **lighttpd** has been upgraded to version 1.4.49.
- Support for **RTP/SRTP capability negotiation** according to a subset of RFC 5939, in compliance with Genesys / Avaya environment requirements. The phone sends an SDP offer with RTP and SRTP capabilities according to a new configuration parameter ‘voip/media/srtp /NegotiationMode’. Configurable parameter values:
  - **Basic** (default) [RTP/SRTP negotiation according to the document IMTC Best Practices for SIP Security. This mode is supported by Broadsoft, Microsoft and many other vendors]
  - **RFC 5939** [RTP/SRTP capability negotiation "a=tcap", "a=acap" and “a=pcfg” attributes as described in RFC 5939]
- The phone features **new capability to send UNENCRYPTED_SRTCP packets** to comply with Avaya environment requirements.

- A new option has been added to **remove the 'lifetime' parameter from the SRTP Crypto line in SDP**. According to RFC 4568, an optional 'lifetime' parameter such as "2^31" must be added to the a=crypto line. A new parameter 'voip/media/srtp/use_lifetime' has been added to allow the removal of the lifetime in all phone crypto lines in SDP. Configurable parameter values:
  - [**0**] = the lifetime is removed
  - [**1**] = the lifetime is retained (default)
- **GENBAND's softswitch solution Kandy Business Solutions (KBS)** has been added as a new application server.
  - **Users can Park Calls to other users' DNs (Directory Numbers) in a Genband environment**. For more information, see the *Kandy Business Solutions Feature Description Guide*.
  - A new **Programmable Key type allowing users to Park / Retrieve calls**, including BLF (presence), has been added to the 440HD phone in a Genband environment.
  - Caller ID has been added to the 440HD phone's sidecar display for parking calls in a in a Genband environment.
- DIGICert has been added as ROOT-CA.
- The preloaded well-known RootCA section for BroadCloud server has been updated.
- A **notification has been added** to notify users and network administrators that **the phone has entered Recovery mode**. Previously, this was supported only on the 405HD phone; now, it's supported on all phone models.
- **'USB Headset Type' has been added to the REST API status keep alive message** sent to the AudioCodes Device Manager management interface. The management interface now features new displays in the Devices Status page:
  - Column 'USB Headset Type' displays a headset connected to a phone's USB port
  - Column 'IPP Model' displays the USB icon
- The **configuration of the provisioning time parameter has been updated** to allow checking for updated files every five and 15 minutes. Previously, the minimum time was one hour.
- Network administrators can **control the DTMF tones level** through a new configuration file parameter 'voip/audio/gain/dtmf_rtp_event_signal_level' that has been added.
  - **0 db** (Minimum)
  - **31 db** (Maximum)

3.13.2 Resolved Constraints in Version 2.2.16

The table below shows the constraints that were resolved in this version.

Table 3-29: Resolved Constraints

Incident	Description
IGS-1230	[Genesys environment] A 'HOLD' action that is performed from the phone takes too long. Conversely, there's no delay when a 'HOLD' action is performed via a soft client.
IGS-989	[420HDG only] Packet Loss occurs during calls when the phone is deployed behind certain routers and is *receiving* voice.
IGS-1869	On some occasions, the Link Layer Discovery Protocol daemon (lldpd) causes the phone to respond sluggishly due to a memory leak.
IGS-1801	The phone does not feature capability to disable the 'Resume' softkey using a configuration file parameter.

Incident	Description
IGS-1821	SIP user ID is shown in the DND (Do Not Disturb), CFD (Call Forward) and Voice Mail screens instead of Display Name.
IGS-1544	A SIP UPDATE message is sent even in instances where it should not be used (referred to in SIP as not allowed).
IGS-1709	One-way audio may occur when using phones that do not support OPUS but are mistakenly configured with an OPUS configuration.
IGS-1420	The phone may be displayed as disconnected in the Device Manager even though it's alive and registered.
IGS-1443	The phone encounters issues when registering to a SIP Proxy when it's behind SAS during survivability mode.
IGS-1424	SSL2 connections are open even if disabled in the configuration file.
IGS-1345	The phone is not processing the 200 OK to Register method due to a port scanner tool.
IGS-1310	[420HD] In some environments, an outgoing call from the phone shows name only and the phone number does not appear.
IGS-1222	[Genband proxy] Call transfer fails.
IGS-1284	Birthday attacks against TLS ciphers with 64 bit block size vulnerability (Sweet32).
IGS-1277	Vulnerability to Web Cross-site Scripting (XSS) attacks on mainform.cgi.
IGS-1810	The Message Waiting Indicator (MWI) port is ignored in the SUBSCRIBE message.
IGS-1862	A Core Dump is not generated correctly when the phone VoIP application is reloaded.
IGS-879	The phone occasionally incorrectly displays a 'Duplicate IP address' message.
IGS-1714	Using the Web interface to trigger 'Restore Defaults' periodically crashes the phone.
IGS-1229	[Genband environment] The phone unsuccessfully drops a participant from a conference.
IGS-724	[4xxHD] 'Forward' is incorrectly translated into Portuguese.
IGS-1231	[Brazilian Portuguese] The BXfer softkey is not translated correctly.
IGS-737	The phone rejects REST API messages from the Device Manager (previously called the IP Phone Manager) if the given username parameter differs in upper or lower case from the phone's local parameter.
IGS-1172	VLAN priority (802.1p) is absent when VLAN settings are discovered via LLDP.
IGS-1205	[420HD/405/405HD] The phone crashes when configuring a speed dial softkey from the menu.
IGS-1333	[BSFT] The phone's screen displays name and number instead of user ID.
IGS-977	[BSFT] A damaged PUBLISH occurs if there's a Local Conference on OPUS.
IGS-1776	The phone's screen does not display an incoming caller's number in the second displayed line.
IGS-1402	The SIP message 200OK reply over TLS changes to UDP.
IGS-1400	There is no support for HTTPS Security headers.
IGS-1307	One-way voice is heard on the phone after the user retrieves the first call from a Consultative Transfer call.
IGS-1326	Configuring a beep to play towards a headset upon auto-answer does not function.
IGS-1159	French translation issues.
IGS-697	The phone displays 'Transferred X to Y' instead of 'Transferring X to Y' after a '202 Accepted' arrives.
IGS-677	In a Semi-Attended Transfer scenario, the phone is missing a CWRR (Call Waiting Reminder Ringtone) indicating that the call was put on hold.

Incident	Description
IGS-1204	The phone performs restarts after making changes in the Web interface to the Programmable Keys.

3.13.3 Known Constraints in Version 2.2.16

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

Table 3-30: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD phone] [BroadSoft environment] BLF supports a list with 23 entries only with middle-size names.

Incident	Description
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones with different settings of the 'voice_quality/mode' parameter. This can happen when one phone is set to use ‘Managed OPUS’ while the other phone isn’t.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.

3.14 Version 2.2.16.142

3.14.1 What’s New

Version 2.2.16 includes firmware build **2.2.16.142**.

Version 2.2.16 offers the following new features:

- **Audible indication played after a call has been on hold for a long time**. After a call has been on hold for a long (configurable) time, a reminder tone is played every 10 seconds until the call is taken off hold.

  Two new configuration file parameters have been added to support the feature:

  - *voip/lhcwrr_enabled*
    - **[1]** (Default) Enabled. After the length of time configured for configuration file parameter *voip/lhcwrr_wait_time* lapses (see the parameter below), a reminder tone (beep) is played every 10 seconds until the call is taken off hold.
    - **[0]** Disabled. No reminder tone (beep) is played, regardless of how long the call is on hold.
  - *voip/lhcwrr_wait_time* (Default: 120 seconds) Defines the length of time that must lapse before a reminder tone (beep) is played. The tone will then be played every 10 seconds until the call is taken off hold.

- **New configuration file parameter enables the P-Asserted Identity header to be added to “18x” and “200” responses**. Parameter ‘voip/signalling/sip/PAI_On_Replay/enable’ was added to enable the header to be added to these responses, configurable as follows:
  - **[0]** (Default) P-Asserted Identity header is *not* added to “18x” and “200” responses
  - **[1]** P-Asserted Identity header is added to “18x” and “200” responses

3.14.2 Resolved Constraints in Version 2.2.16.142

The table below shows the constraints that were resolved in this version.

Table 3-31: Resolved Constraints

Incident	Description
IGS-1113	[405/405HD] The mute icon indication goes missing from the phone’s screen if any key on the keypad is pressed after a call is muted.
IGS-1104	[405/405HD] The phone sends a SIP Invite message without a port number in the “Request line” and “contact” fields.
IGS-1066	After toggling between multiple existing calls, a held call cannot be terminated by putting the handset back in the cradle (on-hooking).
IGS-995	The P-Asserted Identity header is not added to “18x” and “200” responses.
IGS-1094	[430HD/440HD] Enabling call center Agents to sign in using their PIN and their extension number results in them being unable to sign in.
IGS-1030	[405HD] The phone displays an incorrect number during a remote conference.
IGS-1015	The SIP RE-INVITE message does not include a crypto line; this results in a “488 not Acceptable here” reply from the server.
IGS-833	The phone fails to complete a Dial Plan rule when the Dial Plan starts with '0'.
IGS-985	[Voice Dialing] The voice dialing *number* rather than the target user’s *name* is saved in the Call Log in the case of call and regret.
IGS-987	[Voice Dialing] The user does not receive a warning notification in the phone's screen if a call to the voice dialing number fails due to an incorrect IP configuration.
IGS-1077	N-Way Conference (Remote Conference) in a BroadSoft environment may fail. In such a case, after pressing the **Conf** softkey, the next participant cannot be dialed, the phone gets stuck (the conference line cannot be disengaged from) and occasionally, the phone then reboots.
IGS-1024/1083	Some Web interface pages are accessible without user authentication.
IGS-857	The phone accepts HTTP and HTTPS messages sent by a REST API interface even though the connection is configured to use HTTPS only.
IGS-1064	[405HD] When a certain switch is deployed in a customer's network to enable MAC authentication, the switch checks the phone’s presence by ARP request every 120 seconds. The ARP reply is problematic because its target MAC address is the phone when it should be the MAC address of the switch.
IGS-1001	When the configuration file parameter ‘voip/voice_quality/mode’ is set to a value other than **Disabled**, Music on Hold (MoH) does not function – even though it should.
IGS-1050	Using the configuration file parameter to configure removal of the 'Missed Calls' functionality does not work.

Incident	Description
IGS-1049	Disabling 'handset_mode' (by setting the 'voip/handset_mode/enabled' configuration file parameter to **0**), still allows a call to be answered by picking up the handset.
IGS-1054	[430HD, 440HD] Greeting does not produce any audio.
IGS-934	[IP Phone Manager Pro / Express] After a prolonged period, some phones may not report the REST_API keepalive to the IP Phone Manager Pro / Express and will consequently be displayed as 'Disconnected' in the UI.
IGS-1160	Incoming calls result in a delay of up to ~1 second compared to the previous Version 2.2.12 release.
IGS-1152	[420HD] Under some special conditions that load the phone, media packets may be lost.
IGS-1102	The phone still shows the 'Missed Calls' icon even though the keys that allow viewing the calls history list are blocked.
IGS-1081	When the phone uses SIP over TLS and is configured to use a redundant Proxy, the SIP Invite to the redundant Proxy is sent without TLS.

3.14.3 Known Constraints in Version 2.2.16.142

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

Table 3-32: Known Constraints

Incident	Description
93991	[BroadSoft environment] When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	[BroadSoft environment] Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	[BroadSoft environment] Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	[Genesys environment] Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	[BroadSoft environment] When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	[440HD phone] [BroadSoft environment] BLF supports a list with 23 entries only with middle-size names.
105974	[405 and 420HD phones] When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.

Incident	Description
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] The Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	[Genesys environment] There is no voice when a call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones with different settings of the 'voice_quality/mode' parameter. This can happen when one phone is set to use 'Managed OPUS' while the other phone isn't.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It's advisable to use the phone's mute key and volume up/down keys instead.

3.15 Version 2.2.16.92

3.15.1 What's New

Version 2.2.16 includes firmware build **2.2.16.92**.

Version 2.2.16 offers the following new features:

- **USB headsets are now officially supported by AudioCodes' generic SIP IP phones**. See the specifications listed in Table 1-1 for more information about supported headsets.
- **New voice dialing capabilities from the phone to any user in the corporate directory** [Beta Version]. AudioCodes' 400HD Series of IP Phones is now directly integrated with AudioCodes' VocaNOM service to allow voice dialing to any other user in same corporate directory. To enable the service, the user must add a VocaNOM key, and IT must configure the VocaNOM IP address service on the phone. The caller hears a voice prompt requesting the callee's first and last name. When the service identifies the callee, the phone dials the callee's number just as it does in a regular call. Later, the user can press the REDIAL hard key on the phone and view the call logged in the phone's 'Dialed Calls' just like with any other call. The service is currently available in English and German only. [See also here for the *Known Constraints* related to this feature].
- **SRTP negotiation**. Support for Secure Real-Time Transport Protocol has been changed to allow a new SRTP Negotiation option. The configuration file parameter 'voip/media/srtp/mode' replaced the legacy configuration file parameter 'voip/media/srtp/enabled' to support new encryption levels. Three levels of encryption are now supported:
    - DoNotSupportEncryption [Default] [SRTP is disabled]
    - SupportEncryption [Negotiation]
    - RequireEncryption [SRTP is enabled; both sides must support encryption)

Note regarding backward compatibility: Since the legacy parameter ''voip/media/srtp/enabled' has become obsolete in this release, the network administrator must configure phones set to 'voip/media/srtp/enabled=1' in previous releases, to 'voip/media/srtp/mode=RequireEncryption' in this release, to keep forcing SRTP.

- **AudioCodes' new corporate logo** is now displayed on the 430HD phone and on the 440HD phone that features a high-resolution screen. The logo is also displayed in these phones' management interfaces.
- **Phone hard keys and softkeys can be disabled using the configuration file** [released as part of a post Version 2.2.12 release]. Hard keys that can be disabled include speaker, headset, voicemail, REDIAL, CONTACTS, MENU, TRANSFER, HOLD, VOL and mute. The feature is motivated by the requirement on the part of some enterprises to control the setting remotely to comply with company policy.
    - **Example 1**: To disable the phone's REDIAL hard key, the configuration file parameter *personal_settings/key/redial/enabled* can be set to **0**.
    - **Example 2**: To disable the option to restart the phone, the configuration file parameter *personal_settings/menu/restart/enabled* can be set to **0**.
- New configuration file parameters network administrators can use to disable phone hard keys and softkeys include:

/personal_settings/soft_keys/display_idle_screen_keys_when_dialing/enabled

/personal_settings/key/speaker_device/enabled

/personal_settings/key/headset_device/enabled

/personal_settings/key/voice_mail/enabled

/personal_settings/key/redial/enabled

/personal_settings/key/contacts/enabled

/personal_settings/key/menu/enabled

/personal_settings/key/hold/enabled

/personal_settings/key/volume/enabled

/personal_settings/key/mute/enabled

/personal_settings/menu/call_log/enabled

/personal_settings/menu/directory/enabled

/personal_settings/menu/keys_configuration/enabled

/personal_settings/menu/keys_configuration/speed_dial_keys/enabled

/personal_settings/menu/keys_configuration/soft_keys/enabled

/personal_settings/menu/keys_configuration/navigation_keys/enabled

/personal_settings/menu/settings/enabled

/personal_settings/menu/language/enabled

/personal_settings/menu/ring_tone/enabled

/personal_settings/menu/callwaiting/enabled

/personal_settings/menu/date_and_time/enabled

/personal_settings/menu/lcd_contrast/enabled

/personal_settings/menu/backlight_timeout/enabled

/personal_settings/menu/answer_device/enabled

/personal_settings/menu/restart/enabled

/personal_settings/menu/status/enabled

/personal_settings/menu/administration/enabled

/personal_settings/menu/languages/english/enabled

/personal_settings/menu/languages/spanish/enabled

/personal_settings/menu/languages/russian/enabled

/personal_settings/menu/languages/portuguese/enabled

/personal_settings/menu/languages/portuguesebrazilian/enabled

/personal_settings/menu/languages/german/enabled

/personal_settings/menu/languages/ukrainian/enabled

/personal_settings/menu/languages/french/enabled

/personal_settings/menu/languages/frenchcanadian/enabled

/personal_settings/menu/languages/italian/enabled

/personal_settings/menu/languages/hebrew/enabled

/personal_settings/menu/languages/polish/enabled

/personal_settings/menu/languages/korean/enabled

/personal_settings/menu/languages/finnish/enabled

/personal_settings/menu/languages/chinese/enabled

/personal_settings/menu/languages/chinesetraditional/enabled

/personal_settings/menu/languages/turkish/enabled

/personal_settings/menu/languages/japanese/enabled

/personal_settings/menu/languages/slovak/enabled

/personal_settings/menu/languages/czech/enabled

/personal_settings/speed_dial_programming/enabled

/personal_settings/new_call_screen/call_log_soft_key/enabled

/personal_settings/new_call_screen/directory_soft_key/enabled

personal_settings/soft_keys/incoming_call/sk_reject/enable=0

- **The phone screen's backlight timeout can be changed using the configuration file** [released as part of a post Version 2.2.12 release]. A new configuration file parameter *system/lcd/backlight/timeout* has been added to allow changing the phone's backlight timeout using the configuration file. Range: 0-6. Previously, the screen's backlight timeout could only be changed on the phone.
  - 0 = Always On
  - 1 = 10 seconds (default)
  - 2 = 20 seconds
  - 3 = 30 seconds
  - 4 = 40 seconds
  - 5 = 50 seconds

  - 6 = 60 seconds.

- **Second dial tone to receive an external line has been enhanced** [released as part of a post Version 2.2.12 release]. Until this version, when using a second dial tone, for example, when pressing **9** to get an external dial tone, the second dial tone was identical to the main dial tone, with a short break. In the current version, a different dial tone (not configurable) is played as the second dial tone.
- Applicable configuration file parameters are:
- *voip/dialing/secondary_dial_tone/enabled=1*
- *voip/dialing/secondary_dial_tone/key_sequence=<key sequence>*
- <Key sequence> can be one of the digits **1**-**9**

3.15.2 Resolved Constraints in Version 2.2.16.92

The table below shows the constraints that were resolved in this version.

Table 3-33: Resolved Constraints

Incident	Description
IGS-990	When the remote party terminates a call, the phone should play a fast busy tone for a few seconds (preconfigured to 3 seconds) before reverting to idle mode. Instead, the phone disconnects and immediately reverts to idle mode.
IGS-670	A local conference cannot be established when one party offers the OPUS vocoder and the other party does not support OPUS.
IGS-969	The phone's audio device doesn't switch back to the handset after turning off the speaker.
IGS-956	In some configurations, QoE SIP PUBLISH is sent to the Secondary Proxy instead of to the SEM server's address.
IGS-964	Enabling EHS (Electronic Hook Switch) blocks upgrading to the Skype for Business version.
IGS-1000	[405HD] The phone does not accept RFC 2833 format's payload type number 100. The phone fails to change the payload type.
IGS-996	The phones do not support the “Privacy:ID” header for anonymous calls.
IGS-993	The parameter 'Block Caller ID on Outgoing Calls' creates a non-RFC compliant P-Asserted-Identity (PAI) header. The PAI header does not include ‘sip:'.
IGS-970	Chinese characters need improvement.
IGS-394	[440HD] The phone uses the wrong port number when sending a SUBSCRIBE SIP message to retrieve BLF status. BLF doesn't function when using a non-standard SIP Proxy port.

3.15.3 Known Constraints in Version 2.2.16.92

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

Table 3-34: Known Constraints

Incident	Description
93991	In a BroadSoft environment: When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	In a BroadSoft environment: Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	In a BroadSoft environment: Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.

Incident	Description
-	Reporting Quality of Service events: The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. A DNS query is sent instead of an SRV query with priorities ignored. The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	In a Genesys environment: Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	In a Broadsoft environment: When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	In a Broadsoft environment: [440HD phone] BLF supports a list with 23 entries only with middle size names.
105974	[405 and 420HD phones]: When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] the Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	In a Genesys environment: There is no voice when the call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-931	A call cannot be established between two phones with different settings of the 'voice_quality/mode' parameter. This can happen when one phone is set to use Managed OPUS while the other phone isn't.
IGS-985	[Voice Dialing] The voice dialing number rather than the target user's name is saved in the Call Log in the case of call and regret.
IGS-987	[Voice Dialing] The user does not receive a warning notification in the phone's screen if a call to the voice dialing number fails due to an incorrect IP configuration.
IGS-1054	The greeting feature doesn't function flawlessly with the 430HD and 440HD phones. It functions flawlessly with the 405, 405HD and 420HD phones.

Incident	Description
IGS-1069	[Jabra Evolve USB headset] Using the volume keys on this headset is not recommended as they send an unexpected unmute command. It’s advisable to use the phone’s mute key and volume up/down keys instead.

3.16 Version 2.2.14

3.16.1 What’s New

Version 2.2.14 includes firmware build **2.2.14.26**.

Version 2.2.14 offers the following new features:

- **Paging a group of phones**. Live announcements can be made (paged) from a phone to a group of phones, to notify a team (for example) that a meeting is about to commence. 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 screens when the paging call comes in. Before the user can configure a functional key for paging, the feature must be enabled in the Web interface by the network administrator. When it's disabled (default) and the user is in a regular call when a paging call comes in, they're prompted in the phone screen to accept or reject the paging call. If accepted, the regular call is put on hold and the paging call is heard.
  - **Barge-in**.
    - If the network administrator has *enabled* the barge-in feature and the user is in a regular call when a paged call comes in, the paged call is heard and the regular call is placed on hold. The user can opt to revert to the regular call and ignore the paged call.
    - If the network administrator has *disabled* the barge-in feature and the user is in a regular call when the paged call comes in, the user is prompted to accept or reject the paged call. If accepted, the regular call is placed on hold while the paged call is heard. The user can opt to return to the regular call. If rejected, the paged call is not heard and the regular call continues.
    - *The barge-in feature is only relevant if the paged user is in a regular call. If they're not in a regular call, the paged call is heard irrespective of whether the barge-in feature is enabled or disabled.*
- **Xsi interface connecting to BroadSoft's BroadWorks server using HTTP/S authentication**. BroadSoft environment users can enter their BroadWorks user credentials for Xsi access. The phone (430HD and 440HD) supports three Xsi services:
  - **Call Center list**
    - Users can be assigned up to three call centers that will be displayed on the right side of the user's phone screen.
    - 

  - The screen above displays three call centers: Dept. B, Dept. C and Dept. A, configured on programmable keys 4-6. Users can enable | disable each by pressing its programmable key. The network administrator can enable | disable the feature using a new configuration file parameter *xsi/callcenter/update* which by default is enabled.
  - The feature allows enterprise front desk personnel to indicate their availability status (available or unavailable), in each call center, to the BroadWorks server. The server then efficiently distributes incoming calls to front desk personnel, saving callers from the inconvenience of unanswered referrals or disconnections.
- **Contact Synchronization**
  - Contact directories are now pulled directly from the BroadWorks server. Case-insensitive Abc name search is performed instantly. Supported directories are Group Directory, Enterprise Directory, Group Common, Enterprise Common and Personal Directory. The network administrator can enable | disable the feature using a new configuration file parameter *xsi/contact/enable* which by default is enabled. The feature cannot coexist with contacts saved locally on the phone.
- **Call Log Synchronization**
  - Call Logs are pulled directly from the BroadWorks server. The phone displays the following Call Logs: All Calls, Missed Calls, Received Calls and Dialed Calls. The network administrator can enable | disable Call Log synchronization using a new configuration file parameter *xsi/calllog/enable* which by default is enabled.

3.16.2 Resolved Constraints in Version 2.2.14

The table below shows the constraints that were resolved in this version.

Table 3-35: Resolved Constraints

Incident	Description
93548	A very short tone is heard when answering an incoming call.
107440	'Gateway Name' does not function correctly for an outgoing INVITE message.
106580	Volume controls still affect the speaker even though it was disabled using parameter *voip/hands_free_mode/enabled=0*.

3.16.3 Known Constraints in Version 2.2.14

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

Table 3-36: Known Constraints

Incident	Description
93991	In a BroadSoft environment: When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	In a BroadSoft environment: Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	In a BroadSoft environment: Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.

Incident	Description
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music on Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	In a Genesys environment: Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	In a Broadsoft environment: When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	In a Broadsoft environment: [440HD phone] BLF supports a list with 23 entries only with middle size names.
105974	[405 and 420HD phones]: When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] the Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	In a Genesys environment: There is no voice when the call is made to an off-hooked line.

Incident	Description
101300	Multiple lines: The busy screen is corrupted.

3.17 Version 2.2.12

3.17.1 What's New

Version 2.2.12 includes firmware build **2.2.12.224**.

Version 2.2.12 offers the following new features:

- **Phone hard keys and softkeys can be disabled using the configuration file**. Hard keys that can be disabled include speaker, headset, voicemail, REDIAL, CONTACTS, MENU, TRANSFER, HOLD, VOL and mute. The feature is motivated by the requirement on the part of some enterprises to control the setting remotely to comply with company policy.
  - **Example 1**: To disable the phone's REDIAL hard key, the configuration file parameter *personal_settings/key/redial/enabled* can be set to **0**.
  - **Example 2**: To disable the option to restart the phone, the configuration file parameter *personal_settings/menu/restart/enabled* can be set to **0**.
- New configuration file parameters network administrators can use to disable phone hard keys and softkeys include:

  /personal_settings/soft_keys/display_idle_screen_keys_when_dialing/enabled

  /personal_settings/key/speaker_device/enabled

  /personal_settings/key/headset_device/enabled

  /personal_settings/key/voice_mail/enabled

  /personal_settings/key/redial/enabled

  /personal_settings/key/contacts/enabled

  /personal_settings/key/menu/enabled

  /personal_settings/key/hold/enabled

  /personal_settings/key/volume/enabled

  /personal_settings/key/mute/enabled

  /personal_settings/menu/call_log/enabled

  /personal_settings/menu/directory/enabled

  /personal_settings/menu/keys_configuration/enabled

  /personal_settings/menu/keys_configuration/speed_dial_keys/enabled

  /personal_settings/menu/keys_configuration/soft_keys/enabled

  /personal_settings/menu/keys_configuration/navigation_keys/enabled

  /personal_settings/menu/settings/enabled

  /personal_settings/menu/language/enabled

  /personal_settings/menu/ring_tone/enabled

/personal_settings/menu/callwaiting/enabled

/personal_settings/menu/date_and_time/enabled

/personal_settings/menu/lcd_contrast/enabled

/personal_settings/menu/backlight_timeout/enabled

/personal_settings/menu/answer_device/enabled

/personal_settings/menu/restart/enabled

/personal_settings/menu/status/enabled

/personal_settings/menu/administration/enabled

/personal_settings/menu/languages/english/enabled

/personal_settings/menu/languages/spanish/enabled

/personal_settings/menu/languages/russian/enabled

/personal_settings/menu/languages/portuguese/enabled

/personal_settings/menu/languages/portuguesebrazilian/enabled

/personal_settings/menu/languages/german/enabled

/personal_settings/menu/languages/ukrainian/enabled

/personal_settings/menu/languages/french/enabled

/personal_settings/menu/languages/frenchcanadian/enabled

/personal_settings/menu/languages/italian/enabled

/personal_settings/menu/languages/hebrew/enabled

/personal_settings/menu/languages/polish/enabled

/personal_settings/menu/languages/korean/enabled

/personal_settings/menu/languages/finnish/enabled

/personal_settings/menu/languages/chinese/enabled

/personal_settings/menu/languages/chinesetraditional/enabled

/personal_settings/menu/languages/turkish/enabled

/personal_settings/menu/languages/japanese/enabled

/personal_settings/menu/languages/slovak/enabled

/personal_settings/menu/languages/czech/enabled

/personal_settings/speed_dial_programming/enabled

/personal_settings/new_call_screen/call_log_soft_key/enabled

/personal_settings/new_call_screen/directory_soft_key/enabled

personal_settings/soft_keys/incoming_call/sk_reject/enable=0

- **The phone screen's backlight timeout can be changed using the configuration file**. A new configuration file parameter *system/lcd/backlight/timeout* has been added to allow changing the phone's backlight timeout using the configuration file. Range: 0-6. Previously, the screen's backlight timeout could only be changed on the phone.
  - 0 = Always On
  - 1 = 10 seconds (default)
  - 2 = 20 seconds

- 3 = 30 seconds
- 4 = 40 seconds
- 5 = 50 seconds
- 6 = 60 seconds.

- **Second dial tone to receive an external line has been enhanced**. Until this version, when using a second dial tone, for example, when pressing **9** to get an external dial tone, the second dial tone was identical to the main dial tone, with a short break. In the current version, a different dial tone (not configurable) is played as the second dial tone.
- Applicable configuration file parameters are:
- *voip/dialing/secondary_dial_tone/enabled=1*
- *voip/dialing/secondary_dial_tone/key_sequence=<key sequence>*
- <Key sequence> can be one of the digits **1**-**9**

3.17.2 Resolved Constraints in Version 2.2.12

The table below shows the constraints that were resolved in this version.

Table 3-37: Resolved Constraints

Incident	Description
IGS-655	Chinese language improvements
IGS-723/IGS-560	Portuguese language improvements
IGS-682	Portuguese language requires a special license key to be used
IGS-571	[3-Way Conference] [Genesys environment] Transfer from an existing local conference initiated by the phone (the originator leaves the conference leaving the two remote parties continuing talking) cannot be initiated by the phone. It can only be initiated from the Genesys soft client.
IGS-929	[3-Way Conference] The phone's VoIP application is restarted during a local conference when both remote parties close the call before the originator closes it.
IGS-830/IGS-745	[Managed OPUS] Managed OPUS does not function correctly when the phone's configuration file parameter 'voip/voice_quality/mode' is set to **Enable_RTCP**.
IGS-807	[440HD] The EHS headset controls do not function on the phone.
IGS-858	A crackling noise can be heard during calls when the phone is set to automatic answer and when the phone is set to play a beep to the user’s headset when a call comes in.
IGS-844	The phone is unable to dial to numbers with spaces taken from the LDAP server.
IGS-514	In some scenarios, when the phone gets SIP message 403 as an answer to SIP Registration to Redundant Proxy, the phone tries to switch back to the Primary Proxy.
IGS-831	System/Password can't be generated using the phone’s proprietary Encryption Tool.
IGS-792	[QoE] QoE SIP PUBLISH messages may be sent by the phone to the Proxy and not to the QoE server.
IGS-795	Issue encountered when using the 'Redial' functionality (related to VI 107281). In some environments, the proxy replies with a “Remote-Party-ID” header which previously wasn’t supported on the phone and as a result, prevents the redial call to complete successfully.
IGS-390	When using SIP over TLS, an ACK SIP Message is sent over UDP instead of TLS. This can lead to the call being dropped.

Incident	Description
IGS-689	[Genesys environment only] The phone re-registers after an SBC High-Availability (HA) switchover. In some cases (mostly during calls), the switchover doesn't complete successfully; the phone uses the incorrect destination proxy address.
VI109604	In some environments, the phone cannot get VLAN via LLDP.
VI106580	Volume controls are enabled even when speaker is disabled via the configuration file parameter *voip/hands_free_mode/enabled=0*
VI109234	When using automatic provisioning to change the timezone, the phone does not update the time.
VI107440	Gateway Name does not function correctly. The phone's configuration file parameter 'voip/signalling/sip/sip_outbound_proxy/addr' has been fixed to support Name and IP address to be used in the "From" header.

3.17.3 Known Constraints in Version 2.2.12

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

Table 3-38: Known Constraints

Incident	Description
97481, 92336, 97478, 97546	The Multicast Group Paging feature doesn't function correctly in this version.
93991	In a BroadSoft environment: When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
-	In a BroadSoft environment: Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	In a BroadSoft environment: Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music On Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.

Incident	Description
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	In a Genesys environment: Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	In a Broadsoft environment: When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	In a Broadsoft environment: [440HD phone] BLF supports a list with 23 entries only with middle size names.
105974	[405 and 420HD phones]: When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] the Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	In a Genesys environment: There is no voice when the call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
IGS-670	It's impossible to establish a local conference when a conference participant's phone uses an OPUS codec and the phone of another conference participant does not support OPUS.

3.18 Version 2.2.12.172

Version 2.2.12 includes firmware build **2.2.12.172**.

Version 2.2.12 offers the following new features:

- **Technician's digit key code**. Technicians installing phones at customer sites no longer need to connect laptops to phones to provision them. After connecting phones to the network, technicians now enter a specific digit key code which changes the phones' provisioning URL to the server's URL. If the code that the technician enters matches, the phones are automatically provisioned from that server. The feature requires software customization.

- **Configurable OPUS dynamic payload type**. Ability to configure the OPUS dynamic payload type. Previously, the OPUS dynamic payload type could not be modified.
- **Canadian French** (Français Canadien) language.

3.18.1 Resolved Constraints in Version 2.2.12.172

The table below shows the constraints that were resolved in this version.

Table 3-39: Resolved Constraints

Incident	Description
107881	Pressing a Programmable Key to return from hold may cause the held call to be disconnected and the phone's VoIP application to restart.
109043	[Genesys] [420HD] In ACD mode, the phone screen doesn't display the current agent's status.
108984 / 108710	In some environments, when the operator attempts to leave a 3PCC call it may drop the original call.
107281	Redialing sometimes does not function flawlessly.
109015	Voice VLAN may not be configured correctly.
107824	DHCP replies with a 'Destination Unreachable' when an ACK is received for an INFORM message because the 'Client ID' header is missing in the INFORM message.
108730	Establishing a local conference is not possible when the phone is configured with OPUS and one of the remote parties doesn't support OPUS.
108932	A local conference cannot be established using the OPUS vocoder. The conference is initiated using the G.711 vocoder instead.
108187	[QoE] The phone sometime reports an incorrect call duration.
107898	[IP Phone Manager Pro] The phone fails to create a REST_API connection to IP Phone Manager Pro via HTTPS.
107906	The phone gets stuck on 'Acquiring IP' if it receives a DHCP Option message longer than 308 chars.
107113	A codec negotiation issue occurs when using the G722 vocoder.
107754	[405HD] The Programmable Keys page is missing from the phone's Web interface.
107724	Conference involving OPUS and SRTP: A local conference results in no voice.
104457	A call may not be established via TLS when the SIP proxy is configured with an IP address rather than configured with a domain.

3.18.2 Known Constraints in Version 2.2.12.172

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

Table 3-40: Known Constraints

Incident	Description
97481, 92336, 97478, 97546	The Multicast Group Paging feature doesn't function correctly in this version.
93991	In a BroadSoft environment: When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.

Incident	Description
-	In a BroadSoft environment: Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	In a BroadSoft environment: Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn’t function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn’t function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music On Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	In a Genesys environment: Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	In a Broadsoft environment: When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	In a Broadsoft environment: [440HD phone] BLF supports a list with 23 entries only with middle size names.
105974	[405 and 420HD phones]: When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.

Incident	Description
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] the Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	In a Genesys environment: There is no voice when the call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.

3.19 Version 2.2.12.126

Version 2.2.12 includes firmware build **2.2.12.126**.

3.19.1 What's New in Version 2.2.12.126

Version 2.2.12 offers the following new features:

- **New IP phone models**: Two new IP phone models are supported in this release:
  - 405G phone
  - 430HD phone featuring a high-resolution LCD
- **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. The feature is most relevant to the enterprise front desk.
- **Shared Calls Appearance** enables multiple 440HD phones in a Broadsoft environment (exclusively) to be associated in an **SCA** group so that calls can be made or received on any phone in the group. If a call comes in on a phone in the group, all phones in the group ring simultaneously. The first to answer is connected to the caller. All other phones then stop ringing. The recipient can then opt to put the call on hold. All calls put on hold and all active calls are displayed in all phones' sidecars. An SCA group user can pick up a call by pressing their sidecar BLF LED.
  - 440HD - full support
  - 405 / 420HD / 430HD - support as participant only
- **BroadSoft's N-Way Calling feature** which allows users to set up ad-hoc conferences with multiple parties.
  - All phone models support the feature
  - Exclusive to BroadSoft environments
  - One conference per line can be established, for example, when three lines are configured, three conference calls can be established
  - N calls can be merged into a conference call
  - New participants can be added into an existing conference call
- **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 headphone's port. The management connector connects to the Auxiliary port using a DHSG cable which can be ordered from AudioCodes.
- **Call Centers**
  - A beep can be played to headsets when a call comes in, instead of ringing. The beep is heard even if 'Auto answer' is configured to **0**. Two new configuration file parameters were added:
    - */voip/beep_to_ringing_device/enabled* (enables/disables beeping the device) (Default: 0)
    - /voip/beep_to_ringing_device/number_of_beeps (if the feature is enabled, the number of beeps must be configured) (Default: 3)
- **Call Center Queue** using SIP Authentication for Xtended Services Interface (Xsi). BroadSoft environment users can enter their BroadWorks user credentials for Xsi access. The phones use SIP authentication data to authenticate Xsi access. The phones send the BroadWorks user ID to the Xtended Services Platform (Xsp) to identify the user, along with the SIP authentication user name and password to authenticate access to the Xsi.
- **Enhanced quality of experience (QoE)**. Reports (SIP PUBLISH) were improved. Fixes to QoE-related issues were implemented.
- **New capability to provide a provisioning path via DHCP for VLAN configuration**. VLAN can be configured using (1) Link Layer Discovery Protocol (LLDP) (2) Cisco Discovery Protocol (CDP) (3) manually. If (1) is unsuccessful, (2) is attempted, etc. The new capability provides another VLAN configuration option.
- **New method to refresh an existing call: SIP UPDATE**. A SIP UPDATE message is now used instead of a SIP Re-Invite message to refresh an existing call.
- **New ring tones**. Three new ring tones were added:
  - Office Classic – ring 5
  - Home Classic – ring 10
  - Business Special – ring 9
- **OPUS configuration management for enhanced** v**oice quality despite poor network conditions**. The feature allows the OPUS audio codec's configuration to be changed on the fly when poor conditions such as packet loss or jitter are detected in the network. The OPUS functions at a lower channel bit rate and consumes less bandwidth, delivering better voice quality in spite of the poor network conditions.
- **New configuration file parameter value on the 405HD phone** (exclusively):
- *personal_settings/soft_keys/ongoing_call/0/key_function=BLIND_TRANSFER*
- The new value enables the same functionality as that enabled by the TRANSFER hard key on the other phones in the 400HD IP Phone Series, but from a **BXfer** (blind transfer) softkey instead. The value of the parameter on the other phones is **HOLD**.
- **The phone plays a fast busy tone when it is automatically disconnected on the remote side**. When the phone is automatically disconnected from the remote side, it not only displays a 'Disconnected' message for three seconds (default) but also plays a fast busy tone that can be configured with parameter 'voip/dialing/automatic_disconnect_delay_timer'.
  When the parameter 'enable_remote_disconnect_warningTone' is configured to **1** and the phone accepts an incoming call, if the remote side automatically ends the call (disconnects) the phone plays a fast busy tone.

- **A new timeout parameter 'Interdigit Short Timeout' has been added to the Web interface** (Configuration > Voice over IP > Dialing) below the parameter 'Dialing Timeout'. The new parameter is shorter than 'Dialing Timeout'. Default: 3 seconds. It was implemented as **0S** for the Dial Map. If a user wants to make an international call by dialing **00** and wants to dial the secretary/operator by dialing **0**, the user can do both by adding **0S** to the Dial Map. For example, if the digit map string= *xx|[2-9]11|0S|[2-9]xxxxxxxxx|1xxx[2-9]xxxxxx, it has **0S** in it. When the user dials **0**, **0** will match **0S** and will therefore start the 'Interdigit Short Timeout' timer. After this timeout, **0** is dialed out. User can dial **00** or **0123** within the 'Interdigit Short Timeout'. After the 'Dialing Timeout', the string is dialed out.
- **The phone's mute key can be disabled with a new configuration file parameter**. A new configuration file parameter *voip/block_mute_key* allows network administrators to configure enabling or disabling the mute key.
- Improved **Japanese language** phone version.
- The **registration expired time is now configurable**. The registration expired time is that time that lapses before the refresh registration message is sent. A new 'register_before_expires_percent' parameter has been added to the configuration file. Default (in percentage): 15%. Non-percentage values are 5-85. These represent the time that must lapse before the new registration message is sent, for example, 15% means that if the expiration time is 100 seconds, the registration refresh message will be sent after 85 seconds. In previous releases, it was 33%.

3.19.2 Resolved Constraints in Version 2.2.12.126

The table below shows the constraints that were resolved in this version.

Table 3-41: Resolved Constraints

Incident	Description
102548	Telnet access is sometimes denied after disconnecting and then reconnecting the network cable.
104405	The phone’s volume resets to the default value when rebooting.
105972	The phone publishes an incorrect DHCP Option 12 (hostname). The DHCP Option 12 value changes from <model>_<MAC> to line ID.
102991	The phone gets a data VLAN instead of a voice VLAN from some L2 switches, due to incorrect device ID parsing.
107498	[LLDP] The phone configures its internal switch with VLAN tagged =1 when the external switch is configured as untagged (native) VLAN =1.
104726	The phone's default ToS value is incorrect.
104523	In a BroadSoft environment: The phone doesn't enter into Held state because the SIP ‘sendonly’ INVITE is sent to the incorrect proxy.
104513	In a BroadSoft environment: The phone sometime sends a SIP BYE to the secondary proxy though it is registered to the primary proxy.
104504	In a BroadSoft environment: SIP ACK message is sent to the secondary proxy when the phone is registered to the primary proxy.
102163	In a BroadSoft environment: Call Forward No Answer (CFNA) ring count is set to 0 when disabling the call forward.
104470	Calls from an environment with SRTP to an environment with RTP fails as the phone rejects the call with a SIP 488 ‘Not acceptable here’ message.
104457	A SIP ACK message is sent over UDP instead of over TLS and the call drops.

Incident	Description
107047	Upgrade from the IP Phone Manager starts even though the phone is in an active call.
105082	The phone causes an incorrect status indication in the IP Phone Manager when it is set to work with a Redundant proxy.
104524	Japanese Language: The incorrect date is displayed (one month ahead).
104215	In a Genesys environment: Consultative Transfer fails when working with a soft client (in auto-answer mode).
103388	The Redundant Proxy cannot be set to a value of more than 32 characters.
102941	Attended Transfer when using Speed Dial while another call already exists is not working.
102885	The phone occasionally doesn't display a name in the Call Log if the call is unanswered (and the name is saved in the Personal Directory).
101662	[405 only] DTMF is not sent in Early Media state.
101214	The phone gets stuck if the LDAP is set to 'Enabled' and there is no LDAP server.
101141	The PC connected behind the phone is unable to perform EAL-TLS authentication.

3.19.3 Known Constraints in Version 2.2.12.126

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

Table 3-42: Known Constraints

Incident	Description
97481, 92336, 97478, 97546	The Multicast Group Paging feature doesn’t function correctly in this version.
93991	In a BroadSoft environment: When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
	In a BroadSoft environment: Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's screen but they cannot be configured via the phone's Web interface.
97578	In a BroadSoft environment: Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the navigation key.
97610	It's recommended to configure 802.1x EAP-TLS with the configuration file rather than from the phone screen.
106693	802.1x EAP-TLS is disabled by default. For environments which require 802.1x EAP-TLS, a special version can be provided.
97969	[Asterisk environment only] When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
96418	HTTPS provisioning is unsupported when server-side authentication (mutual authentication) is enabled.

Incident	Description
-	Reporting Quality of Service events: ■ The SIP PUBLISH message doesn't function correctly in a conference call (conference holder or remote parties). ■ The SIP PUBLISH message doesn't function correctly when two concurrent calls exist. ■ A DNS query is sent instead of an SRV query with priorities ignored. ■ The Jitter Buffer increases when Music On Hold is played.
98765	[SIP 100 phones] After TCP (TLS) retransmissions, the device is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
99861	CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
99157	RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
101270	Firmware cannot be updated manually from Chrome when accessing the Web interface over HTTPS.
107189	When the phone is set to Off-Hook dialing (which allows dialing all digits in idle mode until pressing 'Dial'), the phone collects the digits but does not display them.
106766	Contact Center: Fails to log in the ACD when SIP Transport Protocol is set to TCP.
106692	XSI – 'Reject' incoming call is not functioning.
106658	In a Genesys environment: Transfer a call from an existing 3-way conference - to 'drop' and leave the two remote parties in the call - may fail.
106434	In a Broadsoft environment: When the phone makes a second call, the conference index appearance LED is updated incorrectly.
102786	In a Broadsoft environment: [440HD phone] BLF supports a list with 23 entries only with middle size names.
105974	[405 and 420HD phones]: When phones are set to static IP address and provisioning is static, the phones do not perform provisioning after a reset.
105651	[SCA] After several SCA scenarios with barge-in, the sidecar records display empty even if there are active calls in the phone's main screen.
105371	[SCA] After a barge-in to a second call, the Index appearance LED is incorrect.
105589	[EHS] When pressing mute on the headset, the mute LED on the phone doesn't light up.
104850	BLF doesn't work if the SIP Proxy port that is used is not standard.
104390	TLS does not initiate a handshake when a static IP address is configured.
104610	[SIP] the Transfer softkey appears when starting a conference.
104587	[QoE] A SIP PUBLISH message isn't created when there is a SIP BYE message.
102782	In a Genesys environment: There is no voice when the call is made to an off-hooked line.
101300	Multiple lines: The busy screen is corrupted.
107113*	Codec negotiation may fail with some remote parties when the phone uses codec G.722/16000.
107724	Conference (involving OPUS and SRTP): A local conference results in no voice.
107754*	405HD Web interface: The 'Programmable Keys' page is missing.

* The constraint has been solved; the solution will be part of the next official release.

3.20 Version 2.2.8

3.20.1 What's New in Version 2.2.8

Version 2.2.8 offers the following new features:

- **Auto redial**. If a called party is unavailable because they're busy (for example), the caller phone's LCD prompts **Extension Busy. Activate auto redial on busy?** If the caller then activates the feature, the busy extension is automatically redialed every *n* seconds (configured by the network administrator). The caller can stop the redialing at any time and (re)activate it at any time.
- **A total of 33 Speed Dials+BLFs can now be configured when the phones are deployed in a BroadSoft environment**. The 33 SDs+BLFs are configured on pages 1, 2 and 3 of the phone's BLF sidecar. Users define 12 SDs+BLFs and then when defining the 13th, the 12th SD+BLF shows **Next** and the name in the 12th moves to the 13th. Applies only to the 440HD model.
- **Open SSL 1.0.1**, integrated with TLS 1.2, replaces OpenSSL 0.9.8 and TLS 1.0.
- **OPUS v1.1 voice coder is now part of the phone generic firmware** and does not require dedicated firmware. The OPUS voice coder is not supported on all phone models (see also Table 1-1 under 'Media Processing').
- **Call Centers**
  - **New 3PCC support for DTMF**. Users can press DTMF keys on a keyboard of a remote computer application to enter digits into a phone's LCD. The feature does not require additional configuration (it's exclusively for Genesys' Call Center environment).
  - Disabling handset mode. Administrators can now disable handset mode using a new configuration file parameter 'voip/handset_mode/enabled' (Default=Enabled). Some call centers don't want agents to work with any device other than headsets. Some prefer not to even *connect* the handsets to the phones. In this case, their administrators can set the configuration file parameter to disabled.
  - Displaying a message in agents' phone LCDs - Call Center administrators can define a message using a new configuration file parameter 'system/display/message_on_screen' that will be displayed in agents' phone LCDs, for example, 'Reminder: Your calls might be recorded'. Agents will then see this message displayed in their LCDs when their phones are in idle state.
  - Hide the ACW (After Call Work) softkey. Administrators can now hide the ACW (After Call Work) softkey. The softkey is by default shown in the phone's LCD after logging in to the call center's Automatic Call Distributor (ACD) server. Administrators can change the default and hide it using a new configuration file parameter 'voip/services/ACD/show_acw_softkey/enabled'.
  - The headset LED can now stay on when the phone is on standby *and* when it is in conversation mode (if the headset is configured as the default audio device). IT administrators can set a new configuration file parameter 'voip/highlight_audio_device' to HEADSET (default=NONE) to enable the feature.
- Administrators can remotely enable or disable the Do not Disturb functionality on users' phones using a new configuration file parameter 'voip/line/0/do_not_disturb/activated', where **[0]** = Disable (default) and **[1]** = Enable.
- **QoE**: General improvements and fixes.
- If a VoIP application needs to be reloaded, the application by default sends a SIP Registration message with **Expires:0** which means unregister.
  By setting a new parameter 'voip/signalling/sip/unregister_on_voip_reload' to **1**, the application will not send the unregistration message when its reloaded.

- The Dual Registration feature has been optimized with the now configurable parameter 'voip/signalling/sip/redundant_proxy/dual_reg/t1' (default = 20ms), which was previously hard-coded and unconfigurable.
- 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 interface's 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').
- 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.

3.20.2 Resolved Constraints in Version 2.2.8.60

The following constraints were resolved in this version:

- On rare occasions, usually when a call is started, the phone LCD process crashes, causing the phone to unregister from the Genesys server
- . If LDAP is enabled and an LDAP server is not connected, the phone crashes. (vi101214)
- TLS and ACD functionality don't work with Genesys Windows Server. (vi101572)
- Genesys Hoteling ACD. The phone logs in without the correct user ID if the user sets 'State After Login' to **Not Ready** (only) and the phone is then disconnected from the network (LAN Link Failure). (vi99126)
- Pressing the **End** softkey to end a local conference and then replacing the handset causes dialing failure the next time dialing is performed. Pressing the **Speaker** button works around the issue. The issue occurs only when using the handset. (vi98883)
- Genesys:
  - ACD
    - Subscribe SIP Message after login sends an incorrect value for 'Not Ready' Reason Code.
    - 'Not Ready' Reason Code is incorrect after network disconnection and reconnection, or after updating a parameter via the phone's Web GUI.
    - Phone does not display the agent 'Not Ready' Reason Code in some scenarios.
  - Phone may crash upon an incoming call when it is configured for auto answer with playing the beep tone.
  - Call is dropped during Call Transfer scenario in some specific environments due to incorrect 'automatic disconnect' detection.
  - Call Retrieve may be delayed by a few seconds if the phone is configured for auto answer and to play a beep when a call comes in.
  - If the audio device was changed during the previous call, the phone does not answer the next call automatically despite the fact that it is configured for 'auto answer'.
- Incorrect UDP/TCP port is set in MWI (Message Waiting Indication) Subscribe message.
- When changing the system/web/enabled parameter via the configuration file and loading the file via provisioning, it does not take effect the first time, only after an additional restart.
- In some scenarios where the first call is disconnected from the remote side during another incoming call, the 'Speaker' button must be pressed to bring the phone to idle mode.
- When very short times are defined for the dial tone and howling timeouts (shorter than the automatic disconnect setting), the phone does not go back to Standby (idle) mode after a dial tone and howling timeout.

- In some environments, the TLS handshake occasionally fails on SIP signaling.
- Secretary Transfer. When trying to transfer a second incoming call using the Transfer button, the button does not respond.
- Incorrect Caller ID during 'Secretary Transfer' scenario.
- Secretary Transfer scenario fails when trying to transfer the second incoming call using the Speed Dial.
- Delayed response to incoming SIP Invite message may occasionally result in a delay with opening the calls.
- Metaswitch
    - One-way voice during conference calls.
    - Shared line isn't supported when using TLS.
    - When using TLS and SRTP, incoming calls are disconnected after a few seconds.
- Multiple Lines: Call between two phones with two lines each may cause the phone to crash.
- Phone Function Keys do not work for SIP addresses longer than 32 characters.

3.20.3 Known Constraints in Version 2.2.8

The following known issues exist in this version:

- Phone tone volumes revert to their default values after a firmware upgrade is performed.
- Incoming DTMF tones are played at 300 ms (hardcoded value) via SIP INFO, instead of ideally being set according to the 'Duration' field in the incoming SIP INFO packet.
- When in an existing call, you cannot make a second call by dialing via the Call Log and Corporate/Personal Directory.
- 430HD/440HD IP phones: The phone's Dialed Calls screen is sometimes updated incorrectly; new calls are displayed lowermost in the list of Dialed Calls.
- The Multicast Group Paging doesn't function correctly in this version.
- BroadSoft environment:
    - When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
    - Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's LCD screen; however, they cannot be configured via the phone's Web interface.
    - Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the Navigation Key.
- It is recommended to configure 802.1x EAP-TLS from the configuration file and not from phone UI.
- Asterisk environment only – When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
- HTTPS provisioning is not supported when server-side authentication (mutual authentication) is enabled.
- Reporting Quality of Service Events:
    - The SIP PUBLISH message doesn't function correctly in a Conference call (Conference holder or remote parties).
    - The SIP PUBLISH message doesn't function correctly when two concurrent calls exist.
    - A DNS A query is sent instead of an SRV query with priorities ignored.

    - The Jitter buffer increases when Music On Hold is played.
- After TCP (TLS) retransmissions, the device (only SIP 100 phones) is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
- The phone cannot perform a redial during an existing call, unless the user makes a new call first.
- CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
- RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
- Manual firmware update cannot be performed from Chrome when accessing the Web GUI with HTTPS.

3.21 Version 2.2.4

3.21.1 What's New in Version 2.2.4

Version 2.2.4 offers the following new features:

- AudioCodes' 405 IP phone is now supported.
- **Call Centers**
    - **Supervisor Listen** - New feature allows supervisors to pick up an agent's handset (in **Mute only** mode) and listen in on a conversation that the agent is conducting on headphones with the customer.
    - **Select Ring Audio Device** – The phone can be configured to ring on one of the following audio devices: speaker, headset, both, or no ring. For more information, see the *Administrator's Manual*, section 'Configuring Ringing on the Default Audio Device'.
    - **Disable hands-free mode** - Instead of using 'headset only' mode the phone can be configured to disable the default 'hands-free' mode so that when the agent presses the speaker key it won't have any effect. For more information, see the *Administrator's Manual*, section 'Configuring Ringing on the Default Audio Device'.
    - **Greeting Recording** - Agents can record personal voice greetings directly on their phones which play automatically when calls come in. For more information, see the *Administrator's Manual*, section 'Allowing Call Center Agents to Record Welcome Greetings'.
    - **ACD (BroadSoft-based ACD method) Hoteling supported on all models** - ACD hoteling is supported for all 400HD models (405, 420HD, 430HD and 440HD). All AudioCodes' phones can therefore now be used in Genesys call centers. For more information, see the *Administrator's Manual*, appendix 'Configuring Automatic Call Distribution (ACD)'.
- **SHA2 Support** - OpenSSL was upgraded from 0.99.8 to 1.0.1m. The newer open source supports SHA2 algorithms.
- **Blind Transfer** – The phone can be enabled to support Blind Transfer by adding a **BXfer** softkey. For more information, see the *Administrator's Manual*, section 'Configuring Blind Transfer'.
- **Drop From Local Conference** - The initiator of a call conference can drop out of the conference by on-hooking the phone, without disrupting parties B and C. Previously, only a softkey option was available for dropping out. For more information, see the *Administrator's Manual,* section 'Allowing the Initiator to Drop out of the Conference'.

- **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.
- The status of factory-installed certificates is now displayed.
    - The phone start up (boot) process now visually displays the following certificate status indications:
        - **Installed**
        - **Self-Signed**
        - **Not-Installed**
    - The Web interface now provides visual indication that factory certificates are installed.
        - 'MAC Address' and 'Device Certificate' parameters have been added to the Web interface's System Information page.
        - The values for the 'Device Certificate' parameter can be **Installed**, **Self-Signed**, or **Not Installed**.
    - The phone's LCD now visually indicates that factory certificates are installed.
        - The 'Device Certificate' parameter has been added to the Release Information menu.
        - The values of the 'Device Certificate' parameters can be **Installed**, **Self-Signed**, or **Not Installed**
- **Send DTMF via SIP and via RTP together** – the phone can be set to send DTMF 'via SIP' together with DTMF via RTP (inband or RFC 2833).
- **http/s provisioning** - Phones can get their HTTP/S provisioning URL from DHCP Option 43, as well as from DHCP Option 160 and DHCP Option 66/67.
- **CDP Enhanced** - Enhanced support for Cisco-proprietary Cisco Discovery Protocol (CDP).
- Restoring phone settings to defaults, without requiring access to the 'Administration' menu or to the Web interface, is extended.
- Slovak, Czech and Turkish languages are now supported.

3.21.2 Resolved Constraints in Version 2.2.4

The following constraints were resolved in this version:

- The phone cannot hold a call with a user whose phone is configured with a fun tone (via Early Media).
- When the phone is set to 1000 Mb/s, it correctly negotiates Speed and Duplex when connected to Cisco switch (CDP) but due to wrong publish, the Cisco switch discovers a duplex mismatch.
- Genesys ACD Hoteling – The agent's status doesn’t function correctly in TLS protocol.
- Korean translation issues.
- If the phone receives a call over SDP with 'video' media capabilities, the phone cannot terminate the call.

- Voice is unclear if the phone switch calls when one call is made with OPUS and the other with G.711.
- Setting some parameters via configuration file provisioning does not take effect unless a reboot is performed (most important is enabling Telnet).
- The phone's LCD may blink during DHCP lease time.
- In some environments, the phone may send a request for a new IP address once every three DHCP lease time cycles..
- The LCD turns on when there is a SIP Notify update for BLF status changes.
- When using the BLF to monitor two users who call each other, there is no indication of who is the calling party and who is the called party.
- If the VOICEMAIL hardkey is pressed when the phone is ringing, it causes the phone to answer.
- BroadSoft
  - Remote Conference fails when uploading the fourth user to the media server.
  - BLF Populated
    - The phone will not accept a 'BLF Resource List' exceeding 12 users.
    - When the phone's 'BLF Resource List' exceeds 12 users, the **Next** button is missing.
- The phone does not Re-Register according to the expiry value in the SIP OK Message from the server.
- Calling a non-existent number results in an 'automatic disconnect' instead of a busy tone.
- [420HD only] In rare scenarios, the phone UI may show an existing call in the LCD in Idle mode, although the call has ended. The phone doesn't send a SIP REGISTER request on Registration expire time if the NTP server reverts the time setting.

3.21.3 Known Constraints in Version 2.2.4

The following known issues exist in this version:

- Phone tone volumes revert to their default values after a firmware upgrade is performed.
- Incoming DTMF tones are played at 300 ms (hardcoded value) via SIP INFO, instead of ideally being set according to the 'Duration' field in the incoming SIP INFO packet.
- When in an existing call, you cannot make a second call by dialing via the Call Log and Corporate/Personal Directory.
- 430HD/440HD IP phones: The phone's Dialed Calls screen is sometimes updated incorrectly; new calls are displayed lowermost in the list of Dialed Calls.
- The Multicast Group Paging doesn’t function correctly in this version.
- BroadSoft environment:
  - When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
  - Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's LCD screen; however, they cannot be configured via the phone's Web interface.
  - Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the Navigation Key.
- It is recommended to configure 802.1x EAP-TLS from the configuration file and not from phone UI.

- Asterisk environment only – When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
- HTTPS provisioning is not supported when server-side authentication (mutual authentication) is enabled.
- Reporting Quality of Service Events:
  - The SIP PUBLISH message doesn't function correctly in a Conference call (Conference holder or remote parties).
  - The SIP PUBLISH message doesn't function correctly when two concurrent calls exist.
  - A DNS A query is sent instead of an SRV query with priorities ignored.
  - The Jitter buffer increases when Music On Hold is played.
- After TCP (TLS) retransmissions, the device (only SIP 100 phones) is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
- The phone cannot perform a redial during an existing call, unless the user makes a new call first.
- CDP Enhanced functions well but publishes incorrect values. The value is always 02 01, which means 10M Half-Duplex.
- RFC 2833 functions well only with the default payload type value (101). Changing the payload type using the configuration file is not recommended.
- Pressing the **End** softkey to end a local conference and then replacing the handset causes dialing failure the next time dialing is performed. Pressing the **Speaker** button works around the issue. The issue occurs only when using the handset.
- Genesys Hoteling ACD - The phone logs in without the correct user ID if the user sets 'State After Login' to **Not Ready** (only) and the phone is then disconnected from the network (LAN Link Failure).

3.22 Version 2.2.2

3.22.1 What's New in Version 2.2.2

Version 2.2.2 offers the following new features:

- European and American date formats are both now supported. The format is configurable in the phone's LCD as well as in the Web interface.
  The European date format is DDMMYYYY. The American format is MMDDYYYY.
- Japanese is now supported. Japanese features three different input modes: Kanji, Hiragana, and Katakana. A submenu for Kanji opens, depending on what base characters are entered.
- If the receiver is placed on-hook after a call is put on hold, the call is by default put on speaker rather than being disconnected. The default can be changed so that if the receiver is placed on-hook after the call is put on hold, the call is disconnected.
- Icons in the phone's LCD now indicate if line keys are configured in a Shared Call Appearance group, or as private lines. A hollow icon indicates a phone configured in an SCA group. A solid icon indicates a phone configured as private. Applies only to 430HD and 440HD phones.
- The **Restart** option is now also available under the 'Settings' menu, as well as under the 'Administration' menu.
- When a phone is in a Multi Line configuration, VOICEMAIL indication *per line* is now supported.

- Direct access to MENU key items is now available when pressing the menu item's number. Users can for example press the **4** key on the phone after pressing the MENU key, for direct access to the 'Settings' options (Language, Ring Tone, Call Waiting, etc.).
- Users are disabled from configuring Programmable Keys as Line Keys. The first two Programmable Keys are by default set to lines. The other four Programmable Keys are by default empty. Users cannot configure all six as Line Keys, neither in the phone LCD nor in the Web interface. The option to assign a line (SIP account) is removed. Applies only to 430HD and 440HD phones.
- BroadSoft
    - 'Forward No Reply' timeout is now also configurable as 'number of rings' by configuring the ini file parameter 'voip/line/0/call_forward/timeout_mode' to **RINGS_COUNT** instead of to the default **SECONDS**. If configured as **RINGS_COUNT**, the phone will by default ring 2r (2 rings) before the call is forwarded. The setting can be changed according to user preference to (for example) 4r (4 rings). The feature allows compliance with BroadSoft's Feature Key Synchronization method.
    - The local 'Forward' functionality accessed from the phone's LCD idle screen when Feature Key Synchronization is disabled, can also be configured as rings or seconds in the same way.
- A new format has been applied to the 'user agent' values, and new hardware information has also been added, namely, '-RevX' string.
  Previously, the 'user agent' value was **AUDC-IPPhone/2.0.8.49 (420HD; 00908F480AFC)**.
  The following information has been added to it:
    - **Exact model** and **Rev** (added to the 420HD model running on DSP AC494 with a fast clock)
    - AUDC-IPPhone/2.0.8.49 (420HD-RevX; 00908F480AFC) (resistors information, added to the 420HD model)
    - **AUDC-IPPhone/2.0.8.49 (420HD-RevX-AC494; 00908F480AFC)** (added to the 420HD model running on DSP AC494 in fast clock)
    - AUDC-IPPhone/2.0.8.49 (420HDG-RevX; 00908F480AFC) (added to the 420HD GbE model)
    - AUDC-IPPhone/2.0.8.49 (430HDG-RevX; 00908F480AFC) (added to the 430HD GbE model)

3.22.2 Known Constraints in Version 2.2.2

The following known issues exist for this version:

- 420HD IP phone: Upgrading from a version earlier than 2.0.2.22 forces a Restore to Default. To upgrade to version 2.2.0 without losing the phone's configuration, a “via version” called 420HD_2.0.2.22.6 can be used. This file is available from: ftp://vop-c11:IPP420@ftp.audiocodes.com/Release/420_via_ver/
- Phone tone volumes revert to their default values after a firmware upgrade is performed.
- TR-069: The phone supports TR-098 (Internet Gateway Device Data Model for TR-069) instead of TR-181 (Device Data Model for TR-069).
- Incoming DTMF tones are played at 300 ms (hardcoded value) via SIP INFO, instead of ideally being set according to the 'Duration' field in the incoming SIP INFO packet.
- When in an existing call, you cannot make a second call by dialing via the Call Log and Corporate/Personal Directory.
- 430HD/440HD IP phones: The phone's Dialed Calls screen is sometimes updated incorrectly; new calls are displayed lowermost in the list of Dialed Calls.
- The Multicast Group Paging doesn’t function correctly in this version.

- BroadSoft environment:
  - When 'SIP Proxy' and 'Default Gateway' are configured with an IP address instead of a Hostname, the Blind Transfer feature does not function correctly.
  - Feature Key Synchronization - the Call Forward and Do Not Disturb functionalities can be configured from the BroadSoft Server Web Interface or from the phone's LCD screen; however, they cannot be configured via the phone's Web interface.
  - Shared Call Appearance - the user cannot toggle between two incoming calls using Programmable Key 1 and Programmable Key 2. Toggling can only be performed using the Navigation Key.
- It is recommended to configure 802.1x EAP-TLS from the configuration file and not from phone UI.
- Asterisk environment only – When the phone has an active call and the call is placed on hold, the caller may hear the remote party speaking while the call is still on hold.
- 420HD – In rare scenarios, the phone UI may show an existing call in the LCD in Idle mode, although the call has ended. HTTPS provisioning is not supported when server-side authentication (mutual authentication) is enabled.
- Reporting Quality of Service Events:
  - The SIP PUBLISH message doesn't function correctly in a Conference call (Conference holder or remote parties).
  - The SIP PUBLISH message doesn't function correctly when two concurrent calls exist.
  - A DNS A query is sent instead of an SRV query with priorities ignored.
  - The Jitter buffer increases when Music On Hold is played.
- After TCP (TLS) retransmissions, the device (only SIP 100 phones) is unanswered and no failover or any other frame is sent. The device's LED flashes blue, indicating that it is trying to register.
- 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.
-

3.22.3 Resolved Constraints in Version 2.2.2

The following constraints were resolved in this version:

- The calling party now hears a busy tone if the called party's line is busy. Previously, if the called party's line was busy, the calling party's phone's LCD reverted to idle without any indication that the called party's line was busy. The new busy tone complies with international telecom standards in traditional non-VOIP telephony systems.
- BroadSoft environment:
  - *Automatic switching* is now implemented by default, i.e., users do not need to press the **9** key, for example, to "get an external line", they dial directly the number of the party they want. To allow the feature, the ini file parameter 'voip/dialing/secondary_dial_tone/enabled' is by default now set to **0** and the Web interface parameter 'Secondary Dial Tone' to **Disable**. *Manual switching* is implemented if the default is changed, i.e., users will need to press the **9** key, for example, to get a line to outside the enterprise; after pressing the key, they'll hear a secondary tone, and then they'll dial the number they need.
  - When using the new 'Populate the BLF List' feature, the phone couldn't acquire more than 12 extensions on the BLF Resource List, or couldn't acquire 12 or fewer extensions which were too long.

- Multiple lines configuration - only the first line could use Remote Conferencing. Other lines could only use Local Conferencing.
- When canceling a "Remote Conference" call, the phone continued to ring after the call had ended (Idle mode).
- Failover: Phone did not diagnose the 50x SIP message response as "timeout". This did not result in a retry as it should have.

- Drop From Conference failed when the first call to the originator was incoming and the second outgoing.
- ACD-Hoteling (420HD only): After recovering from LAN link failure, the phone displayed the host name and not the agent username.
- When using Handsfree mode and both parties tried to end the call together, one side would get a dial tone (their speaker would be activated and they'd need to deactivate it).
- 802.1X - after reconnecting the LAN cable, the phone didn't perform CDP/LLDP.
- When disconnecting an Ethernet cable and then connecting it again, the phone did not send an HTTP request to the provisioning HTTP server.
- CWRR (Call Waiting Reminder Ringtone) did not ring when the call to the third party was regretted (before the third party answered) and the phone returned to the call on hold.
- Calling a non-existent number caused an "automatic disconnect" instead of a busy tone. As a result, users missed the reason why the call failed to establish.
- The clock shown in the phone's idle screen was incorrect when using AM/PM format.
- A DNS SRV query was not sent when using any UDP port except the default port (5060).

3.23 Version 2.2.0

3.23.1 What's New in Version 2.2.0

Version 2.2.0 offers the following new features:

- Factory-Set certificates and AudioCodes Trusted Root CA:

  AudioCodes IP phones are now loaded with factory-set preinstalled certificate files: Client certificate and a Trusted Root CA that are signed by AudioCodes. Each IP phone receives a uniquely generated private key.
- Extended softkey functionality:
  - Up to 20 (0-19) softkey events can be configured in the Idle State.
  - Up to 20 (0-19) softkey events can be configured in the Ongoing Call State.
  - Up to 12 (0-11) programmable softkey (PSKs) events can be configured to either an Ongoing Call State softkey or an Idle State softkey (see description below).

  When more than four softkey events are configured, the user can scroll to additional pages.
  - Programmable Softkeys (PSK):

    You can configure a programmable key event and assign it to a softkey (Programmable Softkey-PSK) in either the Idle Call State or the Ongoing Call State. The PSK can be used for performing actions, such as connecting to a Voice Mail server or returning the details of the last call. The purpose of these softkeys is to instruct the Enterprise's server to perform these actions. The user can also be prompted to enter a personal code before the softkey functionality can be activated.
- User Interactive login to a remote HTTP or HTTPS server:

During the Automatic Provisioning process, the user can optionally be prompted on the phone to enter the Provisioning server login credentials (username and password). This occurs when during the server's authentication process, it is recognized that an http username and/or password has not been specified or that these credentials are incorrect.

- Additional language support:

  Korean and Japanese have now been added as supported Phone Display languages.

- Reporting Quality of Service events using SIP PUBLISH messages:

  You can configure the phone to send Quality of Experience reports to a QoE collecting server, such as the AudioCodes SEM server. This mechanism is implemented using RTCP-XR (RTCP Extended Reports). These extended reports include voice quality metric data events, such as Jitter Buffer, Packet Loss, Delay and Burst, which are collected by the phone during the VoIP session. At the end of the VoIP session e.g. call disconnect or Hold states, values are calculated for each voice quality data event and sent to the QoE server in a SIP PUBLISH message.

- Redirect server:

  You can now use the AudioCodes Redirect server to direct you to the appropriate Provisioning server URL to download the relevant configuration and firmware files.

  Once the IP phone is powered up and network connectivity is established, it automatically requests provisioning information. In case it does not obtain these files according to the regular provisioning hunt order methods e.g. DHCP, it sends an HTTP request to the AudioCodes HTTPS Redirect server for the appropriate Provisioning server URL.

- Blind transfer:

  You can now configure a softkey with Non-Consultative (Blind) call transfer functionality.

- ACD hoteling (420HD phones):

  This feature automatically distributes incoming calls to agents' phones on the basis of agent availability and unavailability. In contact centers, ACD is a key feature of CTI (Computer Telephony Integration). The feature automatically distributes incoming calls to a specific group of terminals that are used by contact center agents.

  The user agent can use the 420HD phone’s UI to update the SIP server for the following presence events:

  - Whenever the call center representative logs in or out.
  - Whenever the call center representative indicates whether they are ready or not to take a call. When the BroadSoft server is configured, the user can also specify the reason for their unavailability e.g. Lunch break.
  - Whenever the user is busy with After Call Work (ACW) (for BroadSoft SIP server only).

- The call waiting beep progress tone can now be disabled.

- The following files are now supported for debug usage:

  - phone/etc/voip_task_reload_counter.txt – this file includes information on the VoIP Task Reload reason.
  - phone/etc/watchdog_reload_process_counter.txt – this file includes information on the Watchdog reload.

  These files can be accessed via telnet.

3.24 Version 2.0.8

3.24.1 What’s New in Version 2.0.8

Version 2.0.8 offers the following new features:

- Automatic mass provisioning and management of IP phones in enterprises, using AudioCodes' Element Management System (EMS). Automatic mass provisioning of IP phones using the DHCP provisioning method can now be performed from AudioCodes' EMS Provisioning Server in the IP Phones Management Server, accessible from the EMS. For detailed information, refer to the *IP Phone Management Server Administrator's Manual*.
- HTTP Authentication for BroadSoft Device Management:

  When IP phones attempt to connect to the BroadWorks Device Management Provisioning server to download software and/or configuration files, HTTP authentication (per RFC 2617) is used to authenticate this connection. In this case, one of two HTTP authentication methods can be used: 1) *Basic* – username and password are sent in plain text over the network or 2) *Digest* - a hash function is applied to the password before sending it over the network (i.e., more secure than Basic).
- Remote conference capabilities (for detailed information on this new feature, refer to RFC 4579, Session Initiation Protocol (SIP) - Call Control - Conferencing for User Agents.
- BroadSoft environment:
  - Feature Key Synchronization: This new feature synchronizes the Do not Disturb (DnD) and Call Forward functionalities with the BroadSoft BroadWorks server. After activating the feature, the DnD and Call Forward functionalities are performed by the BroadSoft BroadWorks server and not by the IP phone.
  - Shared Call Appearance: Applies only to 430HD and 440HD phones. This new feature delivers "shared call appearances" on a "shared line" in SIP.

    "Call appearance" is the presentation of a call on a line. "Shared call appearance" is a call appearance that is visible and (optionally) accessible through the original endpoints in the call, as well as through an authorized set of other endpoints in the network. "Private call appearance", by contrast, is a call appearance that is only visible and accessible through the original endpoints involved in setting up the call. A "shared line" is a line that is *only presented with shared call appearances*. A "private line", by contrast, is a line that is *only presented with private call appearances*.
  - Device Registration Failover (Applicable in BroadCloud environment):

    This feature enables a secondary server to take over the functions of the primary server on the enterprise network if SIP communication between the SIP access device and the primary proxy server is blocked or delayed. No phone functionality is lost when the secondary server takes over.
  - Monitored Lines (a.k.a. BLF populated), based on BroadSoft's BroadWorks BLF service, lets executive assistants or front desk operators monitor lines in the network.
- Enhanced Provisioning URL capability: a new option to include 400HD specific model (using the format <MODEL>) has been added in the configuration file provisioning URL.
- OPUS v1.1 voice coder (for detailed information, see the 'Specifications' table above, in the 'Media Processing' features). Requires a dedicated firmware file with OPUS support.
- Major Transport Layer Security (TLS) improvements.
- Additional Function Keys (440HD phone only): A total of 33 Function Keys can be configured on the 440HD phone. Of these, you can configure up to 12 Function Keys as Speed Dials+BLFs. In this case, the dedicated integrated Side-Car Contacts LCD screen can scroll to up to two more additional pages; the Speed Dials+BLFs functionality can only be configured on the first page, and on the second and third pages, you can configure Speed Dials functionality.

3.25 Version 2.0.6

3.25.1 What's New in Version 2.0.6

- Version 2.0.6 offers the following new features:
- Remote conference: For detailed information on this new feature, refer to RFC 4579, Session Initiation Protocol (SIP) - Call Control - Conferencing for User Agents.
- BroadSoft Feature Key Synchronization: This new feature synchronizes the Do not Disturb (DnD) and Call Forward functionalities with the BroadSoft server. After activating the feature, the DnD and Call Forward functionalities are performed by the BroadSoft server and not by the IP phone.
- BroadSoft Shared Call Appearance: Applies only to 430HD and 440HD phones. This new feature delivers "shared call appearances" on a "shared line" in SIP. “Call appearance” is the presentation of a call on a line. "Shared call appearance" is a call appearance that is visible and (optionally) accessible through the original endpoints in the call, as well as through an authorized set of other endpoints in the network. "Private call appearance", by contrast, is a call appearance that is only visible and accessible through the original endpoints involved in setting up the call. A "shared line" is a line that is only presented with shared call appearances. A "private line", by contrast, is a line that is only presented with private call appearances.
- BroadSoft Device Registration Failover: This feature enables a secondary server to take over the functions of the primary server on the enterprise network if SIP communication between the SIP access device and the primary proxy server is blocked or delayed. No phone functionality is lost when the secondary server takes over.

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