

# GXP2130/2135/2140/2160/2170 Firmware Release Notes

## IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.56 to 1.0.7.25 you will lose the call history.



# **Table of Content**

IMPORTANT UPGRADING NOTE	1
Table of Content	2
FIRMWARE VERSION 1.0.8.56	11
PRODUCT NAME	11
DATE	11
FIRMWARE FILE INFORMATION	11
IMPORTANT UPGRADING NOTE	
BUG FIX	12
FIRMWARE VERSION 1.0.8.53	13
PRODUCT NAME	13
DATE	13
IMPORTANT UPGRADING NOTE	13
BUG FIX	13
FIRMWARE VERSION 1.0.8.50	14
PRODUCT NAME	14
DATE	14
IMPORTANT UPGRADING NOTE	
BUG FIX	14
FIRMWARE VERSION 1.0.8.49	15
PRODUCT NAME	15
DATE	15
IMPORTANT UPGRADING NOTE	15
ENHANCEMENT	15
BUG FIX	
NEW FEATURES OVERVIEW	
SEARCH A STRING WITH QUICK SEARCH	
DATE DISPLAY ON LCD SCREEN	
SUPPORT OF SYSLOG OVER TLS	
DISABLE CALL WAITING PER ACCOUNTSORT BROADSOFT PHONEBOOK BY LAST/FIRST NAME	
LOCK RING TONE VOLUME	
CASE INSENSITIVE IN PHONEBOOK SEARCH	
5	



CALL ON HOLD REMINDER IMPROVEMENT	21
AUTO-ANSWER RING ALERT	21
TRUSTED CA CERTIFICATE SUPPORT	21
AES-256 SUPPORT IN SRTP CALL	22
BROADSOFT CALL CENTER STATUS DISPLAY	22
GDS SUPPORT	23
USERNAME/PASSWORD AUTHENTICATION WITH OPENVPN®	25
FIRMWARE VERSION 1.0.8.47	27
PRODUCT NAME	
DATE	27
IMPORTANT UPGRADING NOTE	
ENHANCEMENT	
BUG FIX	
NEW FEATURES OVERVIEW	
CUSTOMIZE DOMAIN NAME OF THE XSI REQUEST	27
FIRMWARE VERSION 1.0.8.46	29
PRODUCT NAME	29
DATE	29
IMPORTANT UPGRADING NOTE	29
ENHANCEMENT	29
KNOWN ISSUES	31
BUG FIX	
NEW FEATURES OVERVIEW	35
SHOW/HIDE VPK LABEL ON CALL SCREEN	35
MISSED CALL NOTIFICATION	36
SYNC EXT BOARD BACKLIGHT WITH LCD	37
CERTIFICATE UPLOAD FOR OPENVPN	38
PAI UPDATE FOR CALL PARK VPK/MPK	39
DISABLE PREDICTIVE DIALING	41
MISSED CALL INDICATION	42
HIDE PROFILE PICTURE	43
NTP DEFAULT VALUE CHANGE	43
Affinity PASSWORD SUPPORT	44
HIDE/SHOW LABEL SUPPORT	44
CALL SCREEN IMPROVEMENT	
ATTENDED TRANSFER IMPROVEMENT	
ATTENDED CUSTOMIZABLE OPUS PAYLOAD TYPE	47
DND BARGE	48
EHS HEADSET RINGTONE	48



HIDE BLF REMOTE STATUS	48
ENABLE RTCP & RTCP-XR	50
LINK COMMAND	
TLS NEGOTIATION	50
VPK PAGING SUPPORT INSIDE CALL SCREEN	50
ENHANCEMENT FOR OUTBOUND NOTIFICATION	50
LINE SWITCHING IN CALL SCREEN	50
SIP LOG SUPPORT WITH KEYWORD FILTERING	51
MAC HEADER SUPPORT IN SIP HEADER	52
TARGET SOFTKEY REMOVAL	53
SIP ERROR MESSAGE DISPLAY REMOVAL	53
SENDING INSTANT MESSAGE FROM THE WEB	54
RFC2543 HOLD SUPPORT	54
CALL HISTORY MANAGEMENT	55
INTERCOM SETTINGS SUPPORT	55
SOFTKEY LAYOUT CUSTOMIZATION	56
DND ENHANCEMENT	57
Affinity SUPPORT	
MORE SOFTKEY DISPLAY MODE	58
WEB GUI DEFAULT PASSWORD WARNING MESSAGE	58
BLUETOOTH CONNECTING ENHANCEMENT	
CONTACT PICTURE SUPPORT	59
LINE KEY MODE SUPPORT	
LIVE DIALPAD FEATURE SUPPORT	61
AUTOMATIC REDIAL SUPPORT	61
PREFERRED DEFAULT ACCOUNT SUPPORT	
GXP2200EXT LCD TIMEOUT	62
SPLITTING FIRMWARE/CONFIG UPGRADE METHOD	63
FILTER CHARACTERS FROM DIALED NUMBERS	65
DISABLING LOCAL CALL FORWARDING RULES ENHANCEMENT	65
SPECIFIC MODEL CONFIGURATION FILE	66
HIDING DIALING PASSWORD	67
FIRMWARE VERSION 1.0.7.97	68
PRODUCT NAME	60
DATE	
IMPORTANT UPGRADING NOTE	
ENHANCEMENT	
BUG FIX	
NEW FEATURES OVERVIEW	
SUPPORT FOR CONFIGURING RTP PORT RANGE	
SOLLON LONGONING RIF FOR LINANGE	



SUPPORT MORE KEYS USED AS SEND	71
SUPPORT OF CONFIGURABLE HTTP/HTTPS WEB PORT.	71
SUPPORT FOR LANGUAGE INPUT SEARCH	72
SUPPORT OF SINGLE BUTTON CALL PARK	72
OPTION TO SET DTMF DELAY	74
OPTION TO DISABLE HTTP ACCESS	74
SILENT RINGTONE OPTION	74
CONFIGURATION VIA KEYPAD MENU	75
INSTANT MESSAGE DISPLAY DURATION	76
SCREENSAVER SERVER PATH	76
ENABLE DHCP VLAN	77
MPK FOR CONFERENCE	78
DIAL FROM LOCAL PHONEBOOK	78
CALL FORWARDING FROM WEB UI	79
BLF PICKUP SOUND	80
BLUETOOTH CONFIGURATIONS	80
VOICEMAIL INDICATION	81
RADOM REGISTRATION	81
RETURN CODE	82
PLANTRONICS EHS HEADSET RINGTONE	82
FIRMWARE VERSION 1.0.7.81	9.4
PRODUCT NAME	
DATE	
IMPORTANT UPGRADING NOTE	
ENHANCEMENT	
BUG FIX	
NEW FEATURES OVERVIEW	
FIRMWARE LOCAL UPGRADE	
OPUS CODEC	
TARGET SOFTKEY	
SHOW INTERNET DOWN MESSAGE	
TARGET SELECTION	
TRANSFER FROM INFOMATION	
PVALUE FOR VPK MODE IN STRING FORMAT SUPPORT.	
PREDICTIVE DIALING	
MULTIPLE NUMBERS AND OTHER INFO FOR ONE CONT	
PREDICTIVE DIALING SEARCH SUPPORT	
SEARCH FROM BROADSOFT DIRECTORIES SUPPORT	100
FIRMWARE VERSION 1.0.7.25	102
= = = =================================	



PRODUCT NAME	102
DATE	102
IMPORTANT UPGRADING NOTE	102
ENHANCEMENT	102
BUG FIX	103
NEW FEATURES OVERVIEW	104
BROADSOFT XSI AUTHENTICATION TYPE SUPPORT	104
CONFIGURE BROADSOFT XSI SIP AUTHENTICATION METHOD	105
ENABLE/DISABLE AUTO LOCATION SERVICE FROM IpVideoTalk SERVER	108
SECONDARY NTP SERVER	109
EVENTLIST BLF LISTENING TRANSPORT PROTOCOL	
SOUND NOTIFICATION	112
USER-AGENT PREFIX	113
REQUEST MODEL-SPECIFIC CONFIGURATION FILE	115
HIDE VOCODER	
CALL PICKUP BARGE-IN OPTION	
SPEAKERPHONE RX GAIN CONTROL	121
STATUS DETAIL DISPLAY	
PARTIAL MATCH IN BRAODSOFT XSI SEARCH	123
FILTERING OUT INVALID SYMBOLS	
SUPPORT FOR IEEE-48-ADDR AND IEEE-EUI-64 IN PEI HEADER	
SUPPORT FOR POPULATING THE WAN ACCESS NODE IN SIP	
DIAL FROM LDAP RESULTS	
DNS SRV FAIL-OVER MODE	124
SEPARATE SUBSCRIPTION EXPIRATION TIMER	126
LCD POPUP WINDOW	
DIAL PLAN { x+   \+x+   *x+   *xx*x+ }	128
FIRMWARE VERSION 1.0.7.15	130
PRODUCT NAME	130
DATE	
IMPORTANT UPGRADING NOTE	
ENHANCEMENT	
BUG FIX	131
NEW FEATURES OVERVIEW	132
NO TOUCH PROVISIONING	
OUTBOUND NOTIFICATION SUPPORT	
VIRTUAL MULTI-PURPOSE KEYS SUPPORT	143
PROGRAMMABLE KEYS STATUS ON WEB UI	151
AUTO PROVISION LIST STARTING POINT	152
ADDITIONAL OVERRIDE DHCP OPTION	152



ILBC AND G.723.1 SUPPORT	
ENABLE SESSION TIMER	154
RING SPEAKER FOR CALL WAITING	
CONFIGURABLE BACKLIGHT TIMER	
COLOR BACKGROUND WALLPAPER SELECTION	155
BLF LED PATTERN EXPLANATION IN WEB UI	
DISABLE SCREENSAVER WHEN VPK IS ACTIVE	
FULL BLACK SUPPORT FOR IDLE LCD BRIGHTNESS	
BLIND TRANSFER AND ATTENDED TRANSFER SOFTKEY	
DISPLAY SIP MESSAGE TEXT ON LCD	159
FIRMWARE VERSION 1.0.6.9	160
PRODUCT NAME	160
DATE	160
CHANGES SINCE 1.0.6.5	160
ENHANCEMENT	160
BUG FIX	160
FIRMWARE VERSION 1.0.6.5	161
PRODUCT NAME	161
DATE	161
CHANGES SINCE 1.0.6.2	161
ENHANCEMENT	161
BUG FIX	
FIRMWARE VERSION 1.0.5.33	163
PRODUCT NAME	163
DATE	
CHANGES SINCE 1.0.5.29	
BUG FIX	163
FIRMWARE VERSION 1.0.5.29	164
PRODUCT NAME	
DATE	
CHANGES SINCE 1.0.5.28	
BUG FIX	
FIRMWARE VERSION 1.0.5.28	165
PRODUCT NAME	165
DATE	
IMPORTANT UPGRADING NOTE	



CHANGES SINCE 1.0.5.23	165
ENHANCEMENT	
BUG FIX	
FIRMWARE VERSION 1.0.5.23	166
PRODUCT NAME	166
DATE	166
CHANGES SINCE 1.0.5.18	166
ENHANCEMENT	
BUG FIX	166
FIRMWARE VERSION 1.0.5.18	167
PRODUCT NAME	167
DATE	
IMPORTANT UPGRADING NOTE	
CHANGES SINCE 1.0.4.23	
ENHANCEMENT	
BUG FIX	
FIRMWARE VERSION 1.0.4.23	170
PRODUCT NAME	170
DATE	170
CHANGES SINCE 1.0.4.17	
ENHANCEMENT	
BUG FIX	170
FIRMWARE VERSION 1.0.4.17	172
PRODUCT NAME	172
DATE	172
IMPORTANT UPGRADING NOTE	
CHANGES SINCE 1.0.4.16	
BUG FIX	172
FIRMWARE VERSION 1.0.4.16	173
PRODUCT NAME	
DATE	
CHANGES SINCE 1.0.4.15	
ENHANCEMENT	
BUG FIX	173
FIRMWARE VERSION 1.0.4.15	174



PRODUCT NAME	1/4
DATE	174
CHANGES SINCE 1.0.4.10	174
ENHANCEMENT	174
BUG FIX	174
FIRMWARE VERSION 1.0.4.10	176
PRODUCT NAME	176
DATE	176
CHANGES SINCE 1.0.4.6	176
BUG FIX	176
FIRMWARE VERSION 1.0.4.6	177
PRODUCT NAME	177
DATE	177
CHANGES SINCE 1.0.4.5	177
BUG FIX	177
FIRMWARE VERSION 1.0.4.5	178
PRODUCT NAME	178
DATE	178
CHANGES SINCE 1.0.3.9	
ENHANCEMENT	
BUG FIX	178
FIRMWARE VERSION 1.0.3.9	181
PRODUCT NAME	181
DATE	181
CHANGES SINCE 1.0.2.9	181
ENHANCEMENT	
BUG FIX	181
FIRMWARE VERSION 1.0.2.9	184
PRODUCT NAME	184
DATE	184
CHANGES SINCE 1.0.1.14	
ENHANCEMENT	
BUG FIX	184
FIRMWARE VERSION 1.0.1.14	186
PRODUCT NAME	186



186
186
186
187
187
187
187
187
187



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### DATE

10/11/2017

#### FIRMWARE FILE INFORMATION

• GXP2130 firmware file name: gxp2130fw.bin

MD5: f435c63c93af06acdd0ba280a1b8a77d

GXP2135 firmware file name: gxp2135fw.bin

MD5: 35dc6e8782433eba803ba9774ed345cc

• GXP2140 firmware file name: gxp2140fw.bin

MD5: 6c24b3c03dce172e392437efc701be66

GXP2160 firmware file name: gxp2160fw.bin

MD5: 4bc407292def4e323465e4d825818923

• GXP2170 firmware file name: gxp2170fw.bin

MD5: 3013a219a7ac6bf4628c9a64e009d8c6

## **IMPORTANT UPGRADING NOTE**

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance:

"Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England".



Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.

• This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.56 to 1.0.7.25 you will lose the call history.

- Fixed Device should follow the SRTP Key length that was set when receiving No SDP in the Invite
- Fixed Device VPK's display couldn't be changed after getting provisioned by configuration server



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### DATE

08/11/2017

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.53 to 1.0.7.25 you will lose the call history.

- Fixed Device one-way audio issue when resuming a call on hold
- Fixed Message for outbound notification missing local name
- Fixed Garbled text in phonebook
- Fixed Contact names are lost after uploading phonebook.xml on web UI
- Fixed Device crashes after enabling "Ring For Call Waiting" then pressing headset key in call
- Fixed There is no sound in call with SRTP mode



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### DATE

06/15/2017

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.50 to 1.0.7.25 you will lose the call history.

- Fixed Instant message displaying abnormally issue
- Fixed Broadsoft phonebook only displays 50 contacts
- Fixed Device VPK keys cannot be provisioned if pvlaues that require reboot are included in configuration file



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### **DATE**

06/02/2017

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.49 to 1.0.7.25 you will lose the call history.

#### **ENHANCEMENT**

- Added Support to search a string in any position using Quick search [SEARCH A STRING WITH QUICK SEARCH]
- Added Date on the top of phone LCD Screen[DATE DISPLAY ON LCD SCREEN]
- Added Support of syslog over TLS [SUPPORT OF SYSLOG OVER TLS]
- Added Support to disable call waiting per account [DISABLE CALL WAITING PER ACCOUNT]
- Added Option to sort Broadsoft phonebook by Last/First Name [SORT BROADSOFT PHONEBOOK BY LAST/FIRST NAME]
- Added Ability to lock the phone ringing volume [LOCK RING TONE VOLUME]
- Added Ability to search with case insensitive in Web UI phonebook [CASE INSENSITIVE IN PHONEBOOK SEARCH]
- Added Call on Hold reminder improvement [CALL ON HOLD REMINDER IMPROVEMENT]
- Added Auto-answer ring alert [AUTO-ANSWER RING ALERT]
- Added Support for AES-256 in SRTP Call [AES-256 SUPPORT IN SRTP CALL]
- Added Support of Provisioning "Trusted CA Certificates" [TRUSTED CA CERTIFICATE SUPPORT]



- Added Ability to display Broadsoft call center status on idle screen[BROADSOFT CALL CENTER STATUS DISPLAY]
- Added GDS support Integration [GDS SUPPORT]
- Added Support of username/password authentication and cipher option on web with OpenVPN®
  [USERNAME/PASSWORD AUTHENTICATION WITH OPENVPN®]

- Fixed Receiver heard nothing in multicast paging calls
- Fixed VPK cannot provision successfully if configuration file had P values that required reboot
- Fixed OnHook Dialing window disappeared when there was an incoming call
- Fixed There was no Redial softkey in the More softkey list if three softkeys are configured in idle page
- Fixed LCD skipped to screensaver status directly from Bluetooth scan page in Idle brightness status
- Fixed VPK LED flashed Red if there's an incoming call to the VPK located line in non-page 1
- Fixed Line key LED turned off after switching pages in calling status of Account Mode
- Fixed Line key LED worked abnormally when locking the Default mode in Account mode
- Fixed Phone played ring back tone instead of call waiting tone
- Fixed The transferring line LED displayed abnormally
- Fixed Mute Key worked abnormally after canceling incoming call
- Fixed Uploading picture doesn't work for the first attempt
- Fixed Call recover issue with pressing hold key
- Fixed Phone failed to download Language file when options "config via" and "firmware upgrade via" were set to use different protocols
- Fixed No audio for incoming multicast paging with codecs other than PCMU
- Fixed Multicast paging call always displayed PCMU codec even if using a different codec
- Fixed Device audio loopback worked abnormally
- Fixed The picture of contact disappeared
- Fixed Auto Provisioning for Eventlist BLF became incomplete upon reboot
- Fixed Members of Eventlist BLF URI disappeared in VPK after upgrading
- Fixed Discrepancies issue with Multicast Codec Defined at the Sender Side
- Fixed Phone GUI freezing issue
- Fixed Phone redial issue when connecting to a Bluetooth headset
- Fixed Device generated core dump file after changing the option of Auto Provision List Starting Point
- Fixed Device did not display the certificate information in web page when uploading the CA certificate
- Fixed phone provisioning issue
- Fixed No audio issue after answering multiple calls
- · Fixed DUT worked abnormally with incoming call after dialing the account with picture
- Fixed GXP2130 with v1 hardware could not answer Multicast paging calls
- Improved Device BLF status update based on Call-ID of NOTIFY



#### **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

## SEARCH A STRING WITH QUICK SEARCH

#### Functionality

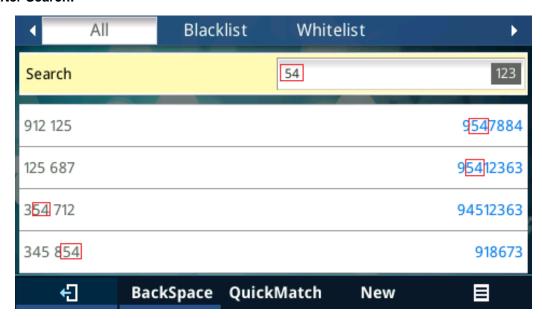
Now users are able to search part of the string and will get the search result from it.

For example, if user only remembers the first name, last name or even part of the phone number, they can still search it in the Search bar.

#### **Before Search:**



### After Search:





#### **DATE DISPLAY ON LCD SCREEN**

## **Web Configuration**

This option can be found under device web UI→ Settings → Date and Time → Show Date On Status Bar.

Show Date On Status Bar No No Yes



## **Functionality**

For GXP 2135 and 2170, if there's no VPK configured with the first VPK on the right side of the LCD screen, the time and date will be showing on the LCD. With this enhancement, we allow the users to choose if they want to display time and date on the top of the LCD screen.

#### Before:



## After:





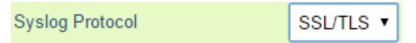
#### New Pvalue

Pvalue	Description	Value Range	Default
P8387	Show Date On Status Bar	Value = 0; No Value = 1; Yes	No

#### SUPPORT OF SYSLOG OVER TLS

#### Web Configuration

This option can be found under device web UI → Maintenance → Syslog → Syslog Protocol.



## Functionality

Now the users are able to send their syslog message through TLS server. However, they will need to upload their trusted CA certificate for the phone to be able to connect with the TLS server. It can be uploaded through web UI → Maintenance → Security Settings → Trusted CA Certificates. This option also needs a reboot to take effect.

## New Pvalue

Pvalue	Description	Value Range	Default
P8402	Syslog Protocol	Value = 0; UDP Value = 1; SSL/TLS	UDP

#### **DISABLE CALL WAITING PER ACCOUNT**

#### • Web Configuration

This option can be found under device web UI $\rightarrow$  Accounts  $\rightarrow$  Account # $\rightarrow$  Call Settings  $\rightarrow$  Disable Call Waiting.

# Disable Call Waiting



### Functionality

Instead of having call waiting option to be globally activated, now the users are able to choose to activate/deactivate with individual accounts.

Note: Use global call feature setting if set to default. This option can be found under device web UI→Call Features→Disable Call Waiting.



Pvalue (#)	Description	Value Range	Default
P26063 (Account 1) P26163 (Account 2) P26263 (Account 3) P26363 (Account 4) P26463 (Account 5) P26563 (Account 6)	Disable Call Waiting	Value = 0; Default Value = 1; No Value = 2; Yes	Default

## SORT BROADSOFT PHONEBOOK BY LAST/FIRST NAME

## Functionality

Now if you hit RMTsearch in broadsoft phonebook, you will be able to search by first/last name, phone number, or extension number.



#### LOCK RING TONE VOLUME

# Web Configuration

This option can be found under device web UI $\rightarrow$  Settings  $\rightarrow$  Preferences  $\rightarrow$  Ring Tone  $\rightarrow$  Lock Volume.

# Lock Volume



# Functionality

Allow user to Lock the ring tone volume. When this option is set to "Yes", other users cannot change the volume on the device anymore.



#### New Pvalue

Pvalue	Description	Value Range	Default
P8392	Lock Volume	Value = 0; No Value = 1; Yes	No

#### **CASE INSENSITIVE IN PHONEBOOK SEARCH**

#### Functionality

Now if the user wants to search in Web UI, the phonebook search bar is no longer case sensitive.

#### **CALL ON HOLD REMINDER IMPROVEMENT**

#### Functionality

Now the user can hear 3 beeps as hold reminder instead of 1. Also, the First on-hold reminder occurs at 1 minute and the subsequent on-hold reminder triggers every 30 seconds.

#### **AUTO-ANSWER RING ALERT**

## Web Configuration

This option can be found under device web UI→ Settings → Preference → Ring Tone → Notification Tone Volume.

16.1	100	9.00	_	30.00	
N	COTITI	cation	n Tone	Mol	li ilma 🛆
11.3	NAME OF TAXABLE PARTY.	W CHILLY II		- W 501	

5

#### Functionality

Now the users are able to configure notification tone volume. Default value is 5. Value range is from 0 to 7.

#### New Pvalue

Pvalue	Description	Value Range	Default
P8399	Notification Tone Volume	0-7	5

#### TRUSTED CA CERTIFICATE SUPPORT

## Functionality

Now users are able to provision the Trust CA Certificate with cfg.xml file, by either uploading the file directly from web, or provisioning it from their cfg.xml file.



Pvalue	Description	Value Range	Default
P8433 - P8438 (Certificate 1 to 6)	Trusted CA Certificate	String	N/A

#### **AES-256 SUPPORT IN SRTP CALL**

## Web Configuration

This option can be found under device web UI $\rightarrow$  Accounts  $\rightarrow$  Account # $\rightarrow$  Audio Settings  $\rightarrow$  SRTP Key Length.

# SRTP Key Length

AES 128&256 bit ▼

## Functionality

Allow users to choose their Key Length for their SRTP calls.

## New Pvalue

Pvalue	Description	Value Range	Default
P2383 (Account 1) P2483 (Account 2) P2583 (Account 3) P2683 (Account 4) P2783 (Account 5) P2883 (Account 6)	SRTP Key Length	Value = 0; AES 128&256 bit Value = 1; AES 128 bit Value = 2; AES 256 bit	AES 128&256 bit

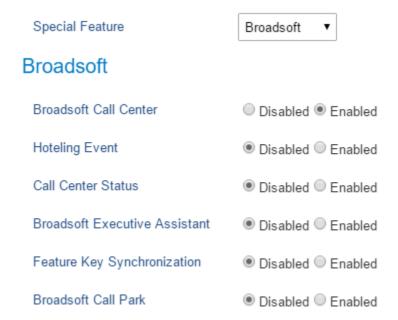
#### **BROADSOFT CALL CENTER STATUS DISPLAY**

## Web Configuration

This option can be found under device web UI $\rightarrow$  Accounts  $\rightarrow$  Account #  $\rightarrow$  SIP Settings  $\rightarrow$  Advanced Features.

To activate this feature, you will need to change Special Feature to Broadsoft and then enable the Broadsoft Call Center. Other than setting on the phone, you will also need to set up your Broadsoft Call Center for this to take effect.





## Functionality

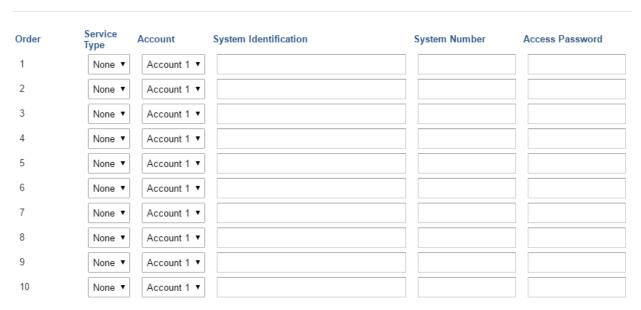
Allow users to monitor their Broadsoft Call Center status through their LCD.

#### **GDS SUPPORT**

## Web Configuration

This option can be found under device web  $UI \rightarrow Settings \rightarrow External Service \rightarrow Grandstream Door System(GDS).$ 

# Grandstream Door System





# • Functionality

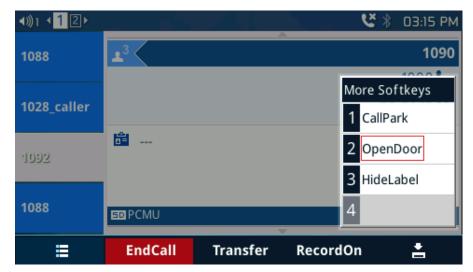
Now the user are able to link your GXP phone with your GDS.

Pvalue	Description	Value Range	Default
P32000 P32005 P32010 P32015	Service Type	Value = -1; None Value = 0; GDS	None
P32020 P32025 P32030 P32035 P32040 P32045			
P32001 P32006 P32011 P32016 P32021 P32026 P32031 P32036 P32041 P32046	Accounts	Value = 0; Account 1 Value = 1; Account 2 Value = 2; Account 3 Value = 3; Account 4 Value = 4; Account 5 Value = 5; Account 6	Account 1
P32002 P32007 P32012 P32017 P32022 P32027 P32032 P32037 P32042 P32047	System Identification	String	N/A
P32003 P32008	System Number	String	N/A



P32013 P32018 P32023 P32028 P32033 P32038 P32043 P32048			
P32004 P32009 P32014 P32019 P32024 P32029 P32034 P32039 P32044 P32049	Access Password	String	N/A

If the user has set up the GDS system correctly, when doorbell rings, user can pick up and hit OpenDoor softkey to remotely open the door for visitors.



## **USERNAME/PASSWORD AUTHENTICATION WITH OPENVPN®**

## • Web Configuration

This option can be found under device web UI → Network → OpenVPN®.



OpenVPN® Username	
OpenVPN® Password	

# • Functionality

Now users are able to configure to use optional username/password authentication if the server supports OpenVPN.

Pvalue	Description	Value Range	Default
P8394	OpenVPN® Username	String	N/A
P8395	OpenVPN® Password	String	N/A



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### **DATE**

04/13/2017

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to search the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.47 to 1.0.7.25 you will lose the call history.

#### **ENHANCEMENT**

 Added Ability to customize the domain name on the XSI request [CUSTOMIZE DOMAIN NAME OF THE XSI REQUEST]

## **BUG FIX**

 Fixed Device with Line Key Sync still did Call Forward No Answer if there is incoming call from BroadSoft Hunt Group

## **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

### **CUSTOMIZE DOMAIN NAME OF THE XSI REQUEST**

Web Configuration

This option can be found under device web UI → Settings → Broadsoft → Broadsoft XSI → XSI Actions



Path.

XSI	
Server	
Port	
XSI Actions Path	

## Functionality

Allowing Users to configure the deployment path for Broadsoft XSI Actions. If it is empty, the path "com.broadsoft.xsi-actions" will be used.

## -Steps to verify:

Step1: Access to Web GUI > Settings > Broadsoft XSI.

Step2: Configure XSI credentials (server, user ID, password, authentication type).

Step3: Set new field "Action Path" to "nl.whereiam.xsi-actions" then save the settings.

Step 4: From phone's LCD, access BS directories and check wireshark trace.

## - Actual expected results:

The phone sends request to BS server using "action path" as configured http://xsp1.iop1.broadworks.net:80/nl.whereiam.xsi-actions/v2.0/user/1356/directories/groupcommon

Pvalue	Description	Value Range	Default
P2937	<new pvalue=""> XSI Actions Path</new>	String	Null



#### PRODUCT NAME

GXP2130, GXP2135, GXP2140, GXP2160 and GXP2170

#### DATE

04/06/2017

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.8.46 to 1.0.7.25 you will lose the call history.

#### **ENHANCEMENT**

- Added Option to Show/Hide VPK label on call screen [SHOW/HIDE VPK LABEL ON CALL SCREEN]
- Added Option to disable and enable the notification popup window for the missed call [MISSED CALL NOTIFICATION]
- Added Option to Sync Extension Board Backlight with LCD [SYNC EXT BOARD BACKLIGHT WITH LCD]
- Allowed Certificate upload for OpenVPN [CERTIFICATE UPLOAD FOR OPENVPN]
- Added Support of PAI update for CallPark VPK/MPK [PAI UPDATE FOR CALL PARK VPK/MPK]
- Added option to disable predictive dial feature [DISABLE PREDICTIVE DIAL]
- Added missed call indication popup window [MISSED CALL INDICATION]
- Added ability to hide the picture icon on call screen if not set [HIDE PROFILE PICTURE]
- Upgraded Dropbear to v2016.74.
- Updated "FR", "ES", "PT" web language.
- Removed G729 codec restriction.
- Updated NTP default value to "pool.ntp.org" [NTP DEFAULT VALUE CHANGE]



- Added backward support Affinity password and added support for passcode [Affinity PASSWORD SUPPORT]
- Added to show/hide VPK label in call screen with softkeys [HIDE/SHOW LABEL SUPPORT]
- Improved the layout for smaller LCD phones' call screen [CALL SCREEN IMPROVEMENT]
- Improved attended transfer feature [ATTENDED TRANSFER IMPROVEMENT]
- Added customizable OPUS Payload Type
- Added DND Barge Internal Calls and Paging [DND BARGE]
- Added user option to enable Plantronics EHS headset ringtone [EHS HEADSET RINGTONE]
- Hided Caller ID info on line key and BLF Presentation [HIDE BLF REMOTE STATUS]
- Enabled RTCP & RTCP-XR [ENABLE RTCP & RTCP-XR]
- Added GXP Color: Link command displays 100Mb/s for Gigabit ports [LINK COMMAND]
- Added support TLS negotiation over 1, 1.1, and 1.2 for SIP [TLS NEGOTIATION]
- Added VPK paging support inside call screen [VPK PAGING SUPPORT INSIDE CALL SCREEN]
- Added enhancement for outbound notification [ENHANCEMENT FOR OUTBOUND NOTIFICATION]
- Added support for Up/Down line switching when call screen is under active/overview state [LINE SWITCHING IN CALL SCREEN]
- Added support to send SIP log with the Keyword Filtering SIPStack [SIP LOG SUPPORT WITH KEYWORD FILTERING]
- Added the MAC address in SIP Header of Register [MAC HEADER SUPPORT IN SIP HEADER]
- Added ability to remove 'target' softkey [TARGET SOFTKEY REMOVAL]
- Added ability to remove SIP error on LCD [SIP ERROR MESSAGE DISPLAY REMOVAL]
- Added requirement sending instant message from the web [SENDING INSTANT MESSAGE FROM THE WEB]
- Added features that supports configurable option for RFC2543 HOLD (0.0.0.0) and RFC3261 (a line)
   [RFC2543 HOLD SUPPORT]
- Added the ability to manage Call History ability from Web GUI [CALL HISTORY MANAGEMENT]
- Enhanced intercom options (includes Intercom Mute, Intercom Tone, and Intercom Barge) [INTERCOM SETTINGS SUPPORT]
- Added customizable Softkey Layout [SOFTKEY LAYOUT CUSTOMIZATION]
- Added DND Authorized Number(Whitelist) & Blacklist and other enhancement [DND ENHANCEMENT]
- Added Affinity support [Affinity SUPPORT]
- Enhanced MORE softkey selection (list of the More softkey items instead of letting user keep tapping MORE to switch to different page) [MORE SOFTKEY DISPLAY MODE]
- Added the Web GUI default password warning message with password change shortcut [WEB GUI DEFAULT PASSWORD WARNING MESSAGE]
- Improved Bluetooth handsfree device connecting process [BLUETOOTH CONNECTING ENHANCEMENT]
- Enhanced Instant Message List page to display to match the rest of the menu pages
- Added contact picture/icon through SIP Call-Info header [CONTACT PICTURE SUPPORT]
- Added LINE key mode support, coexist with legacy mode[LINE KEY MODE SUPPORT]
- Added Live DialPad Feature [LIVE DIALPAD FEATURE SUPPORT]



- Added Automatic Redial Function [AUTOMATIC REDIAL SUPPORT]
- Added Preferred Default Account [PREFERRED DEFAULT ACCOUNT SUPPORT]
- Added GXP2200EXT LCD timeout [GXP2200EXT LCD TIMEOUT ]
- Split Firmware/Config Upgrade Method/Username/Password [SPLITTING FIRMWARE/CONFIG UPGRADE METHOD]
- Added ability to filter characters from dialed numbers [FILTER CHARACTERS FROM DIALED NUMBERS]
- Added ability to enable/disable local call forwarding rules without erasing configured numbers [DISABLING LOCAL CALL FORWARDING RULES ENHANCEMENT]
- Added specific model configuration file [SPECIFIC MODEL CONFIGURATION FILE]
- Added option to hide the password on the LCD display after dialing a specific prefix [HIDING DIALING PASSWORD]

#### **KNOWN ISSUES**

 When the device is using uploaded ring tone, the device randomly crashes in long time call-cancel operations.

- Fixed Device crash issue in call cancel performance test
- Fixed Local Call Feature does not work when the account has dial plan \*xxxx\*x+
- Fixed Device doesn't play ring tone after dismissing incoming call then ending current call
- Fixed Two call with different accounts could not transfer in the same platform
- Fixed DUT slowed down due to thread pool built up
- Fixed Softkeys web page title is displayed wrong
- Fixed ActionURL issue
- Fixed Illogical remote party's avatar behavior on call screen
- Fixed The picture still exists in call after deleted by the Delete button
- Fixed Headset LED displayed abnormally after enabling Live DialPad in headset toggle mode
- Fixed Web display language couldn't switch to Chinese
- Fixed Missed Call Notification can still be reset after factory reset
- Fixed Device could not get OpenVPN IP after downgrading firmware
- Fixed DUT doesn't update the CID when preforming auto-attended transfer
- Fixed Call history displayed wrong after transferring calls
- Fixed No sound in first 2 calls established using Speaker in Sound page
- Fixed History didn't show the CID info for the caller that was parked after picking up a parked call
- Fixed Monitored Call Park VPK/EXT LED displayed abnormally if connected with extension boards
- Fixed LCD displayed extra \* if pressed More softkey then checked OK in setting Forward feature
- Fixed LCD stayed in Cancel ForwardAll page 4s after pressing CancelFwd softkey
- Fixed The call screen displayed abnormally after disabling predictiveve dialing feature
- Fixed The EXT displayed abnormally when many accounts configured Eventlist BLF URI



- Fixed device played dial tip tone in Idle after CancelFwd
- Renamed "Resort" softkey in phonebook to "Sort"
- Fixed DTMF request doesn't work
- Fixed The extension board backlight was closed after switching Sync Backlight with LCD option
- Fixed phone cannot get IP if VLAN tag is set to zero and VLAN priority is set to non-zero value
- Fixed HTTP/HTTPS web port could still be reset in default status
- Fixed Pressing DTMF MPK/VPK can't play keypad tone after enabling "Live DialPad"
- Fixed Audio mode worked abnormally in Toggle mode if set 'Enable Live DialPad'
- Fixed Device works abnormally if setting Automatic Redial and Off-hook Auto Dial at the same time
- Fixed Automatic redial doesn't work in transfer process
- Fixed Device doesn't send SUBSCRIBE for Broadsoft feature after receiving 489
- Fixed Device doesn't play ring tone after pressing "Silence" softkey and then a new call comes in
- Fixed Dial plan rule doesn't take effect under account mode
- Fixed 802.1X port based authentication is not working
- Fixed LCD displayed abnormally after one conference participant ends the call
- Fixed Using "Use Configured IP" and "Accept Incoming SIP from Proxy only" feature exists conflict
- Fixed Phone crashes while trying to off-hook
- Fixed Device didn't play ring tone after cancel transfer
- Fixed Dialing numbers non-matching dial plan still could trigger Automatic Redial
- Fixed Audio wrongly switches to speaker when quickly toggling hook switch
- Fixed Device plays dial tip tone in idle if modifying web options after cancelling ForwardAll
- Fixed The ringtone is abnormal when enabling "Live Dialpad"
- Fixed "DeleteAll" softkey doesn't take effect on Broadsoft phonebook list interface
- Fixed The phone doesn't enter dialing state when "Newcall" softkey is pressed in idle state
- Fixed System time does not update with time zone change
- Fixed Custom Silence key will disable the sound of the established call
- Fixed dial plan will cause the phone crash when it's too long
- Fixed crash when downloading cfg multiple times
- Fixed GUI crashed after pressing softkeys repeatedly in UCM conference room
- Fixed Phone fails to make call when about to re-register
- Fixed Account Mode to make device handle multiple concurrent calls with only one line configured as default line mode
- Fixed device stops at "Upgrading" screen after uploading correct firmware file with wrong name
- Fixed screensaver is On but it will close when VPK is active
- Fixed LCD couldn't enter screensaver status after timeout
- Fixed presence watcher VPK lights off in hold status
- Fixed device doesn't automatically fill related info when save Broadsoft call log to local phonebook
- Fixed the phone couldn't dial the contacts in the predictive dial list
- Fixed If the NTP server does not respond, the device does not try the next IP in the list
- Fixed phone is not honoring DHCP option 66
- Fixed the default firmware Upgrade via mode should be HTTPS not TFTP after clicking reset



- Fixed VLAN DHCP does not take effect in only Option 43 of the environment
- Fixed UCM Detect doesn't work
- Fixed Accounts and VPKs disappears after setting Keypad Menu mode
- Fixed LCD displayed and skipped abnormally after switching language
- Fixed web UI is not locked out after five login failures
- Fixed failed to login SSH
- Fixed device registration fails after enabling 'Validate Certificate Chain'
- Fixed the CA certificate cannot be deleted after uploaded on Trusted CA Cert
- Fixed phonebook sorting on web page didn't work
- Fixed Eventlist BLF list should disappear after disabling Auto Provision Eventlist BLFs
- Fixed GXP2130&2135 lack Screensaver Source option on LCD display web page
- · Fixed There is no Save and Apply button on EXT web page
- Fixed VPK is displayed abnormally after switching page then pressing lines in dialing page
- Fixed some action URL did not work
- Fixed web prompts 'Phone is currently busy' if change is saved after cancelling redial
- Fixed G.723 Rate default value of Account5/Account6 is wrong after factory reset
- Fixed Account5/Account6 registered status doesn't display on Account Status web page
- Fixed Auto-Attended transfer still works even without enabling the feature
- Fixed support to stop Screensaver when VPK is active
- Fixed the volume is wrong after switching audio channel at the CALLING screen
- Fixed SCA Call-info call termination does not show active state on second DUT
- Fixed LCD doesn't skip after EXT speed Dial in Target page
- Fixed there is no Conference mode in LCD VPK/MPK settings page
- Fixed 0 changed to + if DTMF Delay set with larger value
- Fixed LCD displayed in English and switching language doesn't work after factory reset
- Fixed if VPK using call park, even if it subscribes failed, we should still support dial out
- Fixed Edit and Dial window disappears after pressing some keys
- Fixed Busy tone is abnormal after far-end and the call
- Fixed device does not play key/DTMF tone at Dialing/Call screen
- Fixed VPKs in HF line page doesn't work
- Fixed public mode login window is displayed abnormally
- Fixed Account1 registers abnormally after enabling Delay Registration
- Fixed LCD displays abnormally when there is incoming call to non-Account1 in ForwardAll page
- Fixed phone answers paging call even if custom call-info is configured
- Fixed Password Length and DTMF Delay are displayed abnormally if values are out of range
- Fixed users couldn't download devices' screenshot pictures
- Fix wrong LLDP VLAN ID is used by phone and phone's Ethernet VLAN interface is not removed when VLAN ID from network switch LLDP packet is greater than 255
- Changed "Local Call Log" to be "Local Call History"
- Fixed Monitored Call Park VPK is displayed abnormally after pickup then ending call
- Fixed Auto Provision Eventlist BLF must be filled with EXT field from XML file sent by Broadsoft server



- Fixed VPK cannot reset value and description fields via config file
- Fixed BLF MPK doesn't turn off after Reset web page
- Fixed Chinese tooltip for Local SIP Port is wrong
- Fixed device works abnormally after using random port
- Fixed click-to-dial works abnormally in Click-To-Dial disabling status
- Fixed weather doesn't work
- Fixed idle LCD brightness displays without icon
- Fixed LCD displays abnormally after pressing XML VPK in call
- Fixed LCD doesn't display UCM conference CEI info
- Fixed LCD displays abnormally in pickup call page after parking call
- Fixed audio disappears and no sound if back to home then adjusting volume in call
- Fixed LCD displays abnormally after switching pages in Transfer/CONF
- Fixed callee Request Timer doesn't work
- Fixed Blind transfer recovery feature has some issues
- Fixed LCD prompts "System Error: CSS Error" in talking status
- Fixed device can't get the weather information after disable/enable weather
- Fixed phone doesn't register account and web page phonebook click-to-dial displays error
- Fixed device works abnormally after switching calls in Random RTP port mode
- Fixed Auto Provision Eventlist BLFs fails after re-registering the account
- Fixed NAPTR/SRV mode in DNS works abnormally
- Fixed after disabling weather and currency phone still performs DNS query for Yahoo
- Fixed Line keys and VPKs couldn't be modified through config file
- Fixed LCD displays abnormally in pickup call page after parking call
- Fixed the very first LLDP packet (with TTL=0) is not sent out
- Fixed device does not send IP addr in LLDP chassis but MAC address
- Fixed Device uses wrong account to predictive dial in HF dialing page
- Fixed there is no Bluetooth phonebook on contacts
- Fixed call history group on web and LCD are not the same
- Fixed LCD doesn't update in time when call history is cleared from web page
- Fixed missed call icon doesn't disappear after the records are deleted from webUI
- Fixed contacts Search All feature interface jump error causes search failed
- Fixed the phone doesn't use default port 80 when the HTTP proxy URL doesn't include port value
- Fixed LCD does not prompt whether the input of HTTP(S) ID&PW is correct or wrong
- Fixed BLFs take long to recover after network outage
- Fixed LCD does not display error message when self-defined city code is invalid
- Fixed phone sends multiple DTMF tones when in-audio method
- Fixed LCD brightness displays wrong if there is incoming call in Appearance page
- Fixed the capture store in USB doesn't take effect
- Fixed phone doesn't reset Nonce count when authentication is required during registration
- Fixed not sending SSL certificate during wallpaper download
- Fixed Eventlist BLFs of non-related account disappears after unregistering account



- Fixed BLF MPK on the last line doesn't display status after running on/off handsfree mode
- Fixed "Check new firmware only when F/W pre/suffix changes" feature doesn't work
- Improved call screen switching performance
- Fixed BLF status not changing correctly
- Fixed predictive favorite contact not showing after offhook
- Fixed phone random crash when process config file update

## **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

## SHOW/HIDE VPK LABEL ON CALL SCREEN

## Web Configuration

This option can be found under device web UI→ Settings → Programmable Keys → Virtual Multi-Purpose Keys Settings → Show Keys Label。



#### Functionality

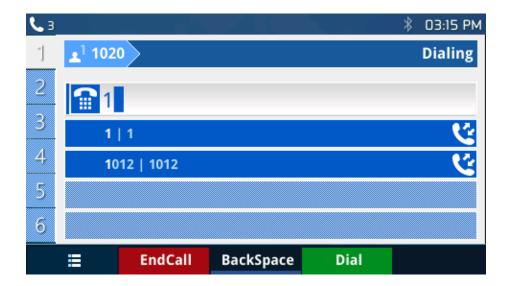
Allowing Users to either Show, Hide, or Toggle between the VPK or Line label.

## Toggle:

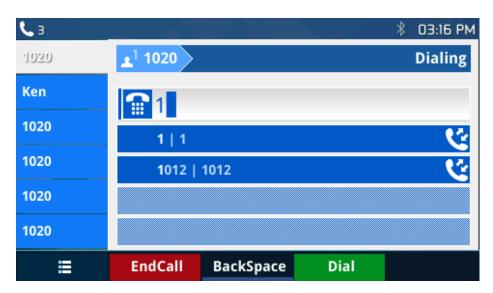


Hide:





#### Show:



## New Pvalue

Pvalue	Description	Value Range	Default
P8386	<new pvalue=""> Show Keys Label</new>	Value = "0" Toggle Value = "1" Show Value = "2" Hide	Toggle

## MISSED CALL NOTIFICATION

## • Web Configuration

This option can be found under device web  $UI \rightarrow Settings \rightarrow Call$  Features  $\rightarrow$  Enable Missed Call Notification.



# Enable Missed Call Notification



## Functionality

Allowing users to decide whether to show the notification popup window for missed call or not. (Note: currently the manually rejected calls are counted as missed calls.)

#### When set to No:



## When set to Yes:



#### New Pvalue

Pvalue	Description	Value Range	Default
P8393	<new pvalue=""> Enable Missed Call Notification</new>	Value = "0" No Value = "1" Yes	No

## SYNC EXT BOARD BACKLIGHT WITH LCD

## Web Configuration

This option can be found under device web UI→ Settings → EXT Settings → Sync Backlight with LCD.



# Sync Backlight with LCD



## Functionality

Allowing users to decide whether they wanted their Extension board (EXT2200) to be in sync with their phone's LCD or not. (Backlight will turn off while LCD goes IDLE)

#### New Pvalue

Pvalue	Description	Value Range	Default
P8400		Value = "0" No Value = "1" Yes	No

## **CERTIFICATE UPLOAD FOR OPENVPN**

## • Web Configuration

This option can be found under device web UI→Network →OpenVPN Settings.

## Functionality

Allow user to simply upload the actual file without needing to do copy & paste as before.

Old:

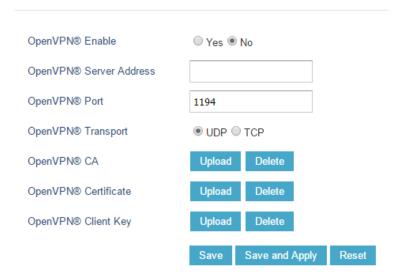
# **OpenVPN Settings**

OpenVPN Enable	○ Yes ● No
OpenVPN Server Address	
OpenVPN Port	1194
OpenVPN Transport	● UDP ○ TCP
OpenVPN CA	
OpenVPN Certificate	10
OpenVPN Client Key	<i>A</i>
	Save Save and Apply Reset

New:



# OpenVPN® Settings



#### Related Pvalue

Pvalue	Description	Value Range	Default
P9902	OpenVPN CA	String	N/A
P9903	OpenVPN Certificate	String	N/A
P9904	OpenVPN Client Key	String	N/A

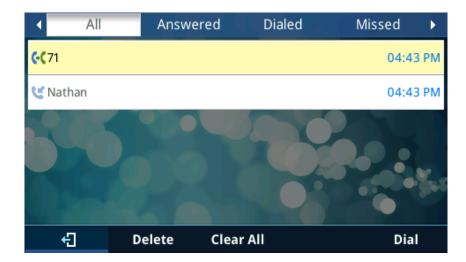
## PAI UPDATE FOR CALL PARK VPK/MPK

## Functionality

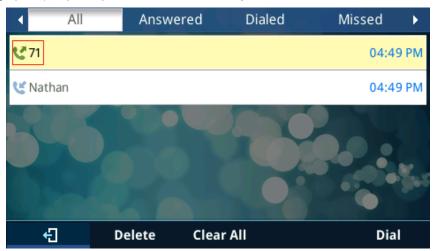
The phone will update the number in call history regarding the PAI that it received from the server. For example, when your number is parked in the callpark space, and the callpark space has been set into a VPK/MPK, if you use the VPK/MPK to retrieve the call, the number will be updated in the call history. However, if you don't use VPK/MPK but dial directly into the callpark space, the number will not be updated in the call history. In both cases, the number will be updated in the talking states.

If use VPK/MPK to park the call, you will see the dialing number (71) in call history.

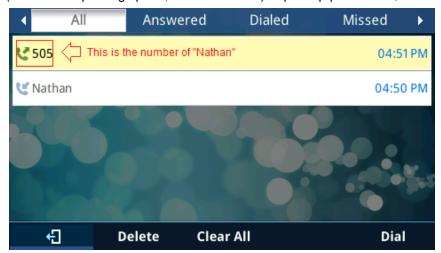




If dial 71(parking space) to pick up the call, the call history will be:



If use VPK/MPK(value set to parking space, in this case: 71) to pick up parked call, the call history:





The reason for VPK/MPK to have different call history is because when you receive SIP message, You will receive PAI. And the PAI will update the call history. On the other hand, when you directly dialed the parking space number, you will be considered as the same as calling an extension, therefore, no updates in the call history.

```
2495 12.743679
                 172.18.19.226 192.168.40.215
                                                                          589 Request: ACK sip:71@192.168.40.215:5060
                                                     SIP
2543 12.817490
                 172.18.19.226 192.168.40.215
                                                     SIP
                                                                          589 Request: ACK sip:71@192.168.40.215:5060
                                                     SIP
                                                                          590 Request: BYE sip:71@192.168.40.215:5060
3420 15.814142
                 172.18.19.226 192.168.40.215
 3421 15.814854
                  192.168.40.215 172.18.19.226
                                                                          496 Status: 200 OK
Message Header
> Via: SIP/2.0/UDP 172.18.19.226:5066;branch=z9hG4bK1025379532;received=172.18.19.226;rport=5066
> From: "506" <sip:506@lafpbx.grandstream.com>;tag=534491539
> To: <sip:71@lafpbx.grandstream.com>;tag=as4fed4346
  Call-ID: 1855369607-5066-4@BHC.BI.BJ.CCG
> CSeq: 11 INVITE
  Server: FPBX-12.0.76.2(11.21.1)
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH, MESSAGE
  Supported: replaces, timer
  Session-Expires: 1800; refresher=uas
> Contact: <sip:71@192.168.40.215:5060>
> P-Asserted-Identity: "Nathan" <sip:505@lafpbx.grandstream.com>
```

192.168.40.215 172.18.19.226

#### DISABLE PREDICTIVE DIALING

Content-Type: application/sdp

#### Web Configuration

Require: timer Content-Length: 289

2454 12.677119

This option can be found under device web UI→ Settings → Call Features → Predictive Dialing Feature



#### Functionality

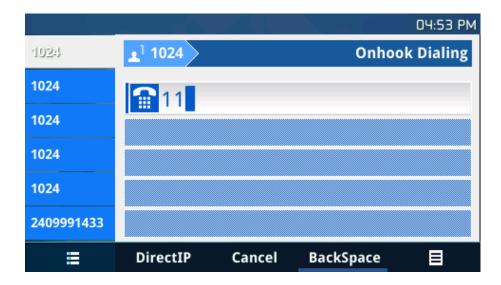
Allowing user to disable predictive dialing, if disabled, the user will no longer see any predictive number while dialing in dialing pages.

## **Enabled:**





#### Disabled:



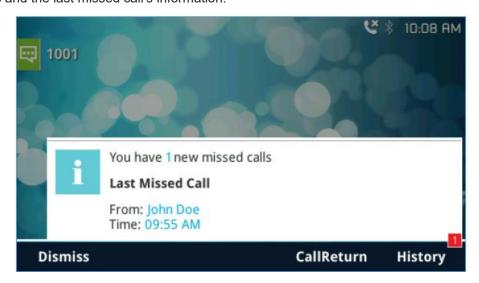
#### New Pvalue

Pvalue	Description	Value Range	Default
P22126	<new pvalue=""> Predictive Dialing Feature</new>	Value = "0" Disabled Value = "1" Enabled	Enabled

#### MISSED CALL INDICATION

## Functionality

This indication can notify users about their missed calls with popup window indicating the number of missed calls and the last missed call's information.



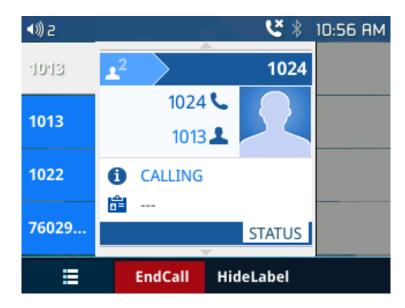


#### HIDE PROFILE PICTURE

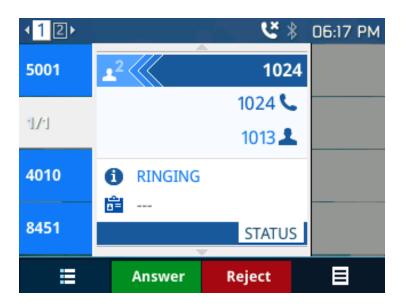
# Functionality

Now the users will no longer see the picture in calling state if the picture has not been set. This will save much space for number and name display.

## **Previous:**



## Now:



#### NTP DEFAULT VALUE CHANGE

Changed Pvalue



Pvalue	Description	Value Range	Default
P30	< Pvalue> NTP Server	String	Changed from "us.pool.ntp.org" to "pool.ntp.org"

## **Affinity PASSWORD SUPPORT**

## Functionality

Affinity (Computer-Telephony Integration) provides ability to manage phone calls from a computer including basic call operations such as making and answering calls, and advanced call operations such as call forward, conference, call hold and more.

#### Request format

The general Affinity commands request format is:

http://Phone-IP-Address/cgi-bin/function?passcode=PASSWORD&param=value

"Function" is one of Affinity functions as described in next chapter (api-get\_line\_status for example);

"Password" is the phone's admin level password. This can also be replaced with "passcode".

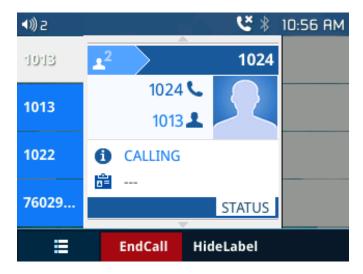
For more detailed guide about Affinity functionality, please refer to Affinity Guide: http://www.grandstream.com/sites/default/files/Resources/CTI\_Guide.pdf

## **HIDE/SHOW LABEL SUPPORT**

#### Functionality

With this softkey, users can hide all the label to free up more space on the screen to see more display of either phone numbers or names.

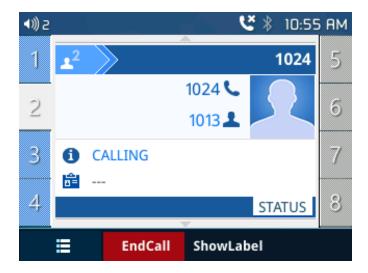
#### When the label is shown:



<sup>&</sup>quot;Param=value" is the parameter for specific Affinity function type.



#### When the label is hidden:



#### **CALL SCREEN IMPROVEMENT**

The texts will move from right to left if the texts are more than one line. Now we can see the full information without getting cutoff.

For example:



## ATTENDED TRANSFER IMPROVEMENT

#### Web Configuration

"Auto attended transfer" is renamed to "Attended Transfer Mode". The options are changed to "Static" and "Dynamic".

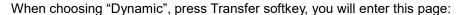
## Functionality

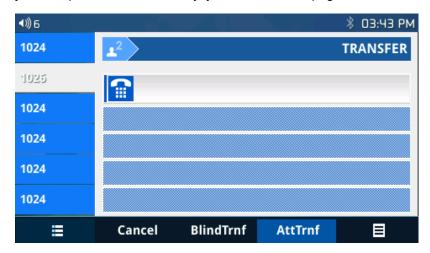
When choosing "Static", this will be acting as "No" in "Auto Attended Transfer", only Blind Transfer will be



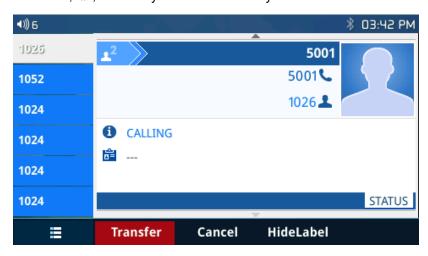
available. However, if "Dynamic" was chosen, then attended transfer will also be activate. In Dynamic mode, after pressing transfer softkey, you will have both blind and attended transfer available. The default highlight will be on AttTrnf (since the AttTrnf softkey is in MORE softkey by default, you will not be able to see the highlight unless you customize the softkey layout). The transfer mode can be identified by those two softkeys that are highlighted in blue when pressing "#" or SEND keys. However, if the user wishes they could perform a blind transfer by pressing the "BlindTrnf" softkey.

NOTE: If users press transfer hard key as "Send" after dialing the number, it will always perform as Blind Transfer regardless of modes.





By default, the highlighted softkey is AttTrnf. (You will not be able to see the highlight because it's in the MORE softkey.) In this state, "#", Send Key and AttTrnf softkey will act as Attended Transfer.



During ringing states, you can press the Transfer softkey to perform semi-auto-Attended Transfer or you can wait till the call is established to finish Attended Transfer.





If you press BlindTrnf before you input the number, BlindTrnf will be highlighted (In this states, "#", Send Key and BlindTrnf softkey will act as Blind Transfer.)

## Changed Pvalue

Pvalue	Description	Value Range	Default
P1376	< Pvalue> Attended Transfer Mode	Value = "0" changed to Static Value = "1" changed to Dynamic	0

#### ATTENDED CUSTOMIZABLE OPUS PAYLOAD TYPE

## Web Configuration

This option is under web UI->Accounts->Account#->Audio Settings "OPUS Payload Type":

123

# Functionality

Now the users can customize and specify the OPUS payload type.

## • Changed Pvalue

Pvalue	Description	Value Range	Default
P2385	< Pvalue> OPUS Payload Type	96-127	123



#### **DND BARGE**

## Web Configuration

This option can be found under device web UI→ Settings → Call Features → DND Override

DND Override	Off •	,
--------------	-------	---

## Functionality

Allowing user to set call override while the phone is under Do Not Disturb mode. By setting it off, you will not receive any calls. By setting as Allow all, all calls will be allowed. Allow Only Contacts will only allow numbers that were saved in the phonebook. Allow Only Favorites will only allow favorite numbers in the phonebook to come in.

#### New Pvalue

Pvalue	Description	Value Range	Default
P8372	<new pvalue=""> DND Override</new>	Value = "0" Off Value = "1" Allow All Value = "2" Allow Only Contact Value = "3" Allow Only Favorites	Off

## **EHS HEADSET RINGTONE**

## Web Configuration

Settings --> Preferences-->Audio Control

EHS Headset Ring Tone    Norm	mal   Plantronics EHS
-------------------------------	-----------------------

## Functionality

Allow user to enable the ringtone from Plantronics EHS headset and play the ringtone in the headset.

## New Pvalue

Pvalue	Description	Value Range	Default
P6779	<new pvalue=""> EHS Headset Ring Tone</new>	Value = 0 Normal Value = 1 Plantronics EHS	Normal

## HIDE BLF REMOTE STATUS

## Web Configuration



Settings --> Call Features

# Hide BLF Remote Status



## Functionality

Allow users to hide the Caller ID from showing at the BLF VPK and EXT. This feature only works with server side support.

## When Disabled:



The VPK will flash between the Caller ID and the BLF account.

## When enabled:



The VPK will stay under the monitored account and only notify there is an incoming call.



#### New Pvalue

Pvalue	Description	Value Range	Default
P8374	<new pvalue=""> Hide BLF Remote Status</new>	Value = "0"; No Value = "1"; Yes	No

#### **ENABLE RTCP & RTCP-XR**

#### Functionality

Allow user to have various qualities of voices for listening and/or conversational quality.

#### LINK COMMAND

```
Grandstream GXP21 Command Shell Copyright 2014
GXP21 > link
Speed: 100Mb/s
Duplex: Full
Auto negotiation: On
GXP21 >
```

#### **TLS NEGOTIATION**

#### Functionality

TLS (transport layer security) is a common protocol which provides privacy to your communication. It will also manage the communication between IP phones to prevent the communications from tampering each other.

Now the phone support TLS 1.0 (RFC2246), 1.1 (RFC4346), and 1.2 (RFC5246)

#### VPK PAGING SUPPORT INSIDE CALL SCREEN

#### Functionality

Now the user can utilize their VPK mode with their settings while in call screen. Such as dial DTMF.

#### **ENHANCEMENT FOR OUTBOUND NOTIFICATION**

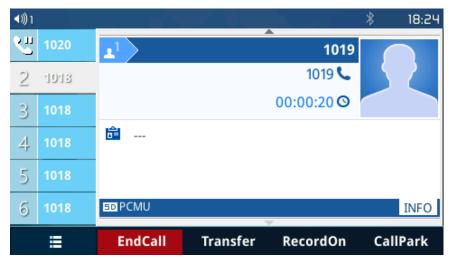
#### Functionality

Align the outbound notification with the call log change if the same account registered in multiple phones.

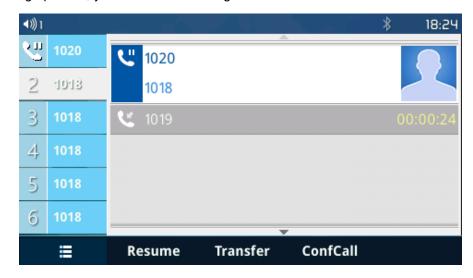
#### LINE SWITCHING IN CALL SCREEN

When having two lines(or more) active





By pressing up/down, you'll be able to manage between the calls as such:



## Functionality

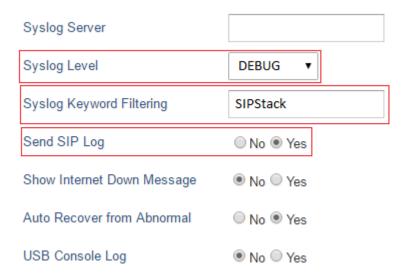
Allow the user to manipulate between calls and the select the option through softkeys.

## SIP LOG SUPPORT WITH KEYWORD FILTERING

• Web Configuration

Maintenance --> Syslog





## Functionality

By setting Send SIP Log to Yes and Syslog Level set to DEBUG, you will still get SIP log from syslog with the Keyword Filtering SIPStack. (This feature requires reboot.)

## New Pvalue

Pvalue	Description	Value Range	Default
P208	Syslog Level	Value = 0; NONE Value = 1; DEBUG Value = 2; INFO Value = 3; WARNING Value = 4; ERROR	NONE
P1387	Send SIP Log	Value = 0; No Value = 1; Yes	No
P22129	Syslog Keyword Filtering	String	N/A

# MAC HEADER SUPPORT IN SIP HEADER

## • Web Configuration

Accounts -> Account # -> SIP Settings -> Custom SIP Headers

Use MAC Header 

● No ○ Yes

If choose yes:



## Functionality

Allow user to see the MAC address of the phone they are using from the trace of the SIP Header of Register.

#### New Pvalue

Pvalue	Description	Value Range	Default
P26061	Use MAC Header	Value = 0;No Value = 1;Yes	No

#### TARGET SOFTKEY REMOVAL

#### Web Configuration

Settings --> Programmable Keys--> Softkeys Settings

Show Target Softkey No No Yes

#### Functionality

Allow user to remove target softkey by toggle Yes/No option during the off-hook dial screen and transfer screen.

#### New Pvalue

Pvalue	Descripti	on	Value Range	Default
P8377	Show Softkey	Target	Value = 0; No Value = 1; Yes	Yes

#### SIP ERROR MESSAGE DISPLAY REMOVAL

## • Web Configuration

Settings --> Call Features



Show SIP Error Response

No Yes

## Functionality

Allow user to disable the SIP error message that will show on the call screen.

## New Pvalue

Pvalue	Description	Value Range	Default
P8376	Show SIP Error Response	Value = 0; No Value = 1; Yes	No

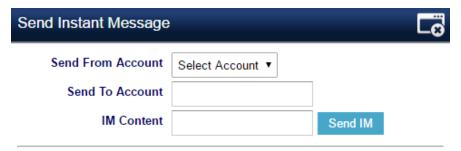
#### SENDING INSTANT MESSAGE FROM THE WEB

#### Web Configuration

Right next to the phone icon.



Then you will get to:



## Functionality

Allow users to send instant messages not only through the phone but also from the Web GUI.

## **RFC2543 HOLD SUPPORT**

## • Web Configuration

Accounts -> Account # -> Call Settings:



RFC2543 Hold



## Functionality

Allow user to toggle between RFC2543 hold and RFC3261 hold. RFC2543 hold allows user to disable the hold music send to the other side. RFC3261 will play the hold music to the other side.

#### New Pvalue

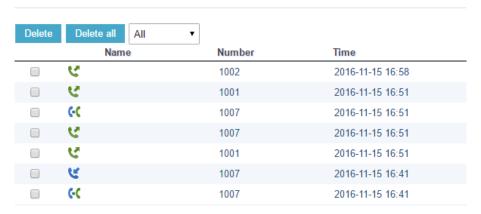
Pvalue	Description	Value Range	Default
P26062	RFC2543 Hold	Value = 0; No (RFC3261) Value = 1; Yes (RFC2543)	No

## **CALL HISTORY MANAGEMENT**

## • Web Configuration

Settings --> Call History

# **Call History**



## Functionality

Allow user to manage call histories not only through phone but also through web GUI.

#### **INTERCOM SETTINGS SUPPORT**

## • Web Configuration

Accounts -> Account # -> Intercom Settings



Allow Auto Answer by Call- Info/Alert-Info	No ○ Yes
Allow Barging by Call-Info/Alert- Info	No ○ Yes
Mute on answer Intercom call	● No ○ Yes
Play warning tone for Auto Answer Intercom	● No ○ Yes
Custom Alert-Info for Auto Answer	

# Functionality

Allow user to set-up and manage Intercom including Barge, warning tone, auto answer, and mute.

## New Pvalue

Pvalue	Description	Value Range	Default
P298	Allow Auto Answer by Call-Info/Alert-Info	Value = 0; No Value = 1; Yes	No
P26019	Allow Barging by Call- Info/Alert-Info	Value = 0; No Value = 1; Yes	No
P26071	Mute on answer Intercom call	Value = 0; No Value = 1; Yes	No
P26072	Play warning tone for Auto Answer Intercom	Value = 0; No Value = 1; Yes	No
P2356	Custom Alert-Info for Auto Answer	String	N/A

## **SOFTKEY LAYOUT CUSTOMIZATION**

# • Web Configuration

Settings --> Programmable Keys-->Softkeys Settings

After enabling the option "Custom Sotfkey Layout" (Default Disabled):



Custom Softkey Layout

• Disabled • Enabled

Enforce Softkey Layout Position

• No • Yes

The user can then customize their softkey layout:

## Custom Call Screen Softkey Layout



#### Functionality

Allow user to manage softkey layout in each available page after enabling the option. NOTE: For Some softkeys, enabling them are not a sufficient condition for them to show up, they will only show up if they satisfy certain config conditions and are enabled in web GUI.

## **DND ENHANCEMENT**

#### Functionality

Allow user to ban phone number by adding the number into the blacklist, or on the other hand, allow phone numbers to come in during DND mode by adding the number into the whitelist.

#### **Affinity SUPPORT**

#### Web Configuration

Settings --> Affinity Settings



# **Affinity Settings**

Affinity Support	Disabled      Enabled			
Preferred Account	Account	1 ▼		
	Save	Save and Apply	Reset	

## Functionality

Allow user to enable and disable CTI function and choose the account that associate with Affinity.

#### New Pvalue

Pvalue	Description	Value Range	Default
P8334	Affinity Support	Value = 0; Disabled Value = 1; Enabled	Disabled
P8335	Affinity Account	Depending on the model	Account 1

#### MORE SOFTKEY DISPLAY MODE

## Web Configuration

Settings -->Programmable Keys-->Softkeys Settings

More Softkey Display Mode 

Menu Toggle

#### Functionality

Allow user to choose from the original Toggle mode or the enhanced Menu mode. With the enhanced Menu mode, the MORE softkey now will not need the user to tap multiple times of MORE to get to the next pages, instead, pressing MORE will have a popup window and allow user to choose from the list.

#### WEB GUI DEFAULT PASSWORD WARNING MESSAGE

You are currently using the default password to login. We strongly recommend you to update your password setting.

Modify

#### Functionality

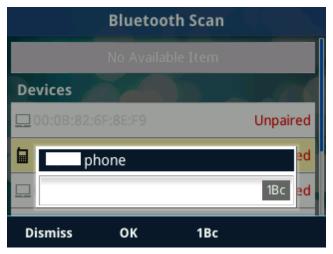
It reminds the user to change the password and allow user to modify the password by clicking the modify button.



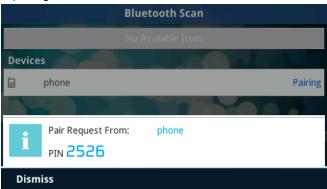
#### BLUETOOTH CONNECTING ENHANCEMENT

## Functionality

If the phone sends out the paring PIN, the user will need enter using keypad on the phone:



If the phone generated the paring PIN, and it will display on the LCD, the user need to input this PIN on the Bluetooth device to finish paring:



## **CONTACT PICTURE SUPPORT**

#### Functionality

Allow callee to see the picture/icon that the caller set. You can find the Call-Info header contains with the jpg file from sip messages. (Currently only support Openser SIP server).

#### LINE KEY MODE SUPPORT

#### Web Configuration

Settings --> Programmable Keys--> Virtual Multi-Purpose Keys Settings

# Call Screen Settings

Key Mode 

• Line Mode 
• Account Mode

Functionality

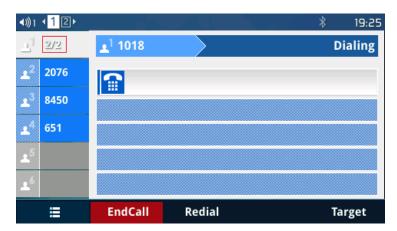


Allow users to change between Line Mode and Account Mode. Under "Line Mode", VPKs can be configured as line keys and only one call can be established per line key. Line keys are not associated with any account so multiple accounts can use any given line key to establish a call. Under Account Mode, VPKs can be configured as account keys. An account key can only be associated with one account but can have multiple calls per key.

#### Line mode:



#### Account mode:



As the screenshot showed, the Line key will represent the line itself. In the Account mode, instead of representing the line itself, the VPK represent the account, thus it can hold more than one line in the same account without using other LINE keys.

#### New Pvalue

Pvalue	Description	Value Range	Default
P8369	Key Mode	Value = 1; Line Mode Value = 2; Account Mode	Line Mode



#### LIVE DIALPAD FEATURE SUPPORT

## Web Configuration

Settings -> Call Features

Enable Live DialPad	No	O Yes
Live DialPad Expire Time	5	

## Functionality

Allow user to enable/disable Live DialPad function, when enabled, the phone will automatically dial out the number punched in after the number of seconds that the user had set.

## New Pvalue

Pvalue	Description	Value Range	Default
P8350	Enable Live DialPad	Value = 0; No Value = 1; Yes	No
P8351	Live DialPad Expire Time	2-15 (seconds)	5

## **AUTOMATIC REDIAL SUPPORT**

## Web Configuration

Settings -> Call Features

Enable Automatic Redial	No ○ Yes
Automatic Redial Times	10
Automatic Redial Interval	20

## Functionality

Allow user to enable the function, if automatic redial is enabled, when called number is busy, the phone will Redial the number X number of times set with Y seconds interval in between each redial.

New Pvalue



Pvalue	Description	Value Range	Default
P8353	Enable Automatic Redial	Value = 0; No Value = 1; Yes	No
P8354	Automatic Redial Times	1-200	10
P8355	Automatic Redial Interval	1-360 (seconds)	20

## PREFERRED DEFAULT ACCOUNT SUPPORT

#### Web Configuration

Settings -> Call Features

# Preferred Default Account

Account 1 ▼

## Functionality

Allow user to select a default account when other accounts have not been selected. When chosen an account, this account will be the account that live DialPad and auto Redial used. However, if this account is not active, then the first account that is active will be used.

#### New Pvalue

Pvalue	Description	Value Range (Depends on how many accounts)	Default
P29612	Preferred Default Account	Value = 0; Account 1 Value = 1; Account 2 Value = 2; Account 3 Value = 3; Account 4 Value = 4; Account 5 Value = 5; Account 6	Account 1

#### **GXP2200EXT LCD TIMEOUT**

## Web Configuration

Settings --> Preferences-->LCD Display

Active Backlight Timeout

1



## Functionality

When setup the timeout of the backlight of the phone, the user now also sets up the backlight for the extension board.

## New Pvalue

Pvalue	Description	Value Range (minutes)	Default
P8356	Active Backlight Timeout	1-90	1

## SPLITTING FIRMWARE/CONFIG UPGRADE METHOD

# Web Configuration

Firmware File Postfix

Before:

Maintenance -> Upgrade and Provisioning

Upgrade via	○ TFTP ○ HTTP ● HTTPS
Firmware Server Path	fm.grandstream.com/gs
Config Server Path	fm.grandstream.com/gs
Firmware File Prefix	

Config File Prefix

Config File Postfix

Now:



# Config

Config Upgrade via	○ TFTP ○ HTTP ● HTTPS
Config Server Path	fm.grandstream.com/gs
Config HTTP/HTTPS User Name	
Config HTTP/HTTPS Password	
Config File Prefix	
Config File Postfix	
Firmware	
Firmware Upgrade via	○ TFTP ○ HTTP ● HTTPS
Firmware Server Path	fm.grandstream.com/gs
Firmware HTTP/HTTPS User Name	
Firmware HTTP/HTTPS Password	
Firmware File Prefix	
Firmware File Postfix	

# Functionality

Now the users will have better idea of which section is for configuration updates and which section is for firmware updates.

## New Pvalue

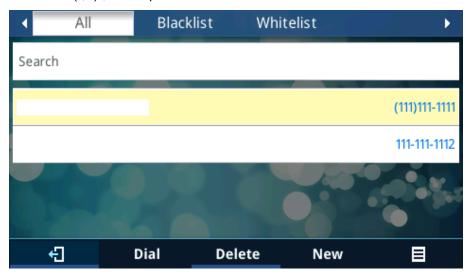
Pvalue	Description	Value Range	Default
P6768	Firmware HTTP/HTTPS Username	String	N/A
P6769	Firmware HTTP/HTTPS Password	String	N/A



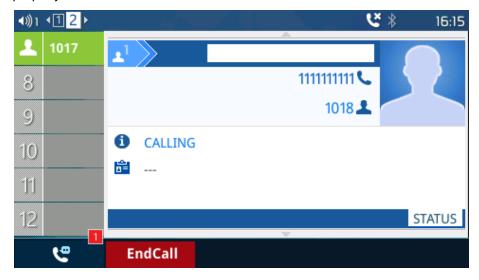
#### FILTER CHARACTERS FROM DIALED NUMBERS

## Functionality

After downloading phonebook, when dialed directly from the contact, the number may include special characters such as "(", ")", "-" or space in between number.



Now when press Dial softkey directly, the phone will filter out the special characters and make the calls send out properly.

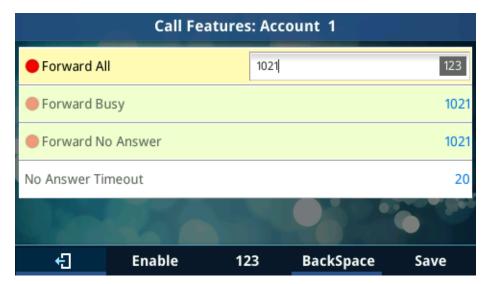


#### DISABLING LOCAL CALL FORWARDING RULES ENHANCEMENT

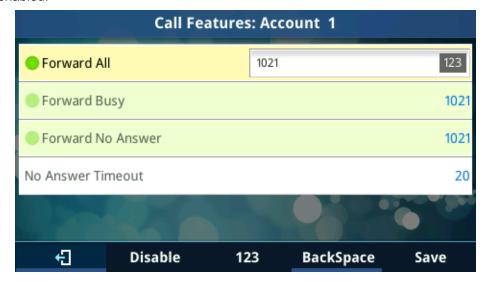
#### Functionality

Now the users can access their call forwarding from the phone without erasing the configured numbers. On the phone, you can access through Menu -> phone (6) -> Call Features -> Account #. When disabled:





#### When enabled:



#### **SPECIFIC MODEL CONFIGURATION FILE**

## Functionality

Now when the phone is provisioning, the configuration file it requested will be based on the phone's model. For example, GXP2130 will request cfggxp2130.xml file and GXP2140 will be cfggxp2140.xml.

```
237 GET /cfg000b826649c3 HTTP/1.0
66 HTTP/1.1 404 Not Found (text/html)
241 GET /cfg000b826649c3.xml HTTP/1.0
66 HTTP/1.1 404 Not Found (text/html)
236 GET /cfggxp2130.xml HTTP/1.0
66 HTTP/1.1 404 Not Found (text/html)
229 GET /cfg.xml HTTP/1.0
66 HTTP/1.1 404 Not Found (text/html)
```



## **HIDING DIALING PASSWORD**

# Web Configuration

This option can be found under web UI  $\rightarrow$  Account  $x \rightarrow$  Call settings.

Hide Dialing Password	Prefix for dialing password		
	Password length 0		

# Functionality

This option allows user to hide password from LCD display after dialing certain prefix.

## New P Value

Pvalue	Description	Value Range	Default
P26049	Account 1: Prefix for dialing password	String	N/A
P26050	Account 1: Password length	0-64	0
P26149	Account 2: Prefix for dialing password	String	N/A
P26150	Account 2: Password length	0-64	0
P26249	Account 3: Prefix for dialing password	String	N/A
P26250	Account 3: Password length	0-64	0



# FIRMWARE VERSION 1.0.7.97

#### PRODUCT NAME

GXP2130, GXP2140, GXP2160, GXP2170 and GXP2135

## **DATE**

10/03/2016

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.7.97 to 1.0.7.25 you will lose the call history.

#### **ENHANCEMENT**

- Added support for configuring RTP port range [SUPPORT FOR CONFIGURING RTP PORT RANGE]
- Added support for more keys as send [SUPPORT MORE KEYS USED AS SEND]
- Added support of configurable HTTP/HTTPS port for Web UI [SUPPORT OF CONFIGURABLE HTTP/HTTPS WEB PORT]
- Added language input search support [SUPPORT FOR LANGUAGE INPUT SEARCH]
- Added single button call parking support [SUPPORT OF SINGLE BUTTON CALL PARK]
- Added option to set the DTMF delay [OPTION TO SET DTMF DELAY]
- Added option to enable/disable HTTP access [OPTION TO DISABLE HTTP ACCESS]
- Added silent ringtone option [SILENT RINGTONE OPTION]
- Added option to lock or restrict to only call/receive functionality without menu access and ability to configure anything from phone side [CONFIGURATION VIA KEYPAD MENU]
- Added ability to change instant message display duration [INSTANT MESSAGE DISPLAY DURATION]
- Added ability to change screensaver pictures via HTTP server [SCREENSAVER SERVER PATH]
- Added support for DHCP option 132 & 133 tunneled through DHCP option 43 [ENABLE DHCP VLAN]



- Added ability to use MPK to trigger a conference [MPK FOR CONFERENCE]
- Added ability to display mobile and home number when searching in local
- Added ability to choose the dialing account to dial a contact from local phonebook [DIAL FROM LOCAL PHONEBOOK]
- Added ability to set call forwarding from the web GUI [CALL FORWARDING FROM WEB UI]
- Added ability to disable/enable a sound notification for each ringing monitored BLF [BLF PICKUP SOUND]
- Added option to allow the user to modify the configuration Bluetooth via Web UI [BLUETOOTH CONFIGURATIONS]
- Added option to enable/disable voicemail indication [VOICEMAIL INDICATION]
- Added random registration [RADOM REGISTRATION]
- Added return code when call is rejected or DND [RETURN CODE]
- Added option to enable Plantronics EHS headset ringtone [PLANTRONICS EHS HEADSET RINGTONE]
- Changed OPUS sampling rate to 48000 Hz. SDP session descriptions in invite for OPUS: a=rtpmap:123 opus/48000/2

## **BUG FIX**

- Fixed line seize failed and 481 subscription does not exist
- Fixed when screensaver is active, popup window does not work correctly, e.g. if there is new firmware found during screen saver, user cannot reject it
- Fixed if VPK is using call park, we should still support dial out even if SUBSCRIBE fails
- Fixed users cannot search for a contact without knowing the account to which it is assigned
- Fixed link command displays 100Mb/s for Gigabit ports
- Fixed call forward using forward softkey ignores the dial plan
- Fixed phone reply with 486 when the called extension is registered on the same phone
- Fixed cannot accomplish auto-attended transfer using BLF key containing "."
- Fixed device GUI crashed after pressing transfer VPK
- Fixed LCD will not skip after pressing RIGHT key to call in target page
- Fixed LCD displays wrong info after offhook timeout
- Fixed device upgrade fails after uploading firmware in serial console disabling status
- Fixed screensaver source works abnormally after changing screensaver power
- Fixed switching hands-free mode LED to re-sending BLF subscribe
- Fixed GXP2130 & 2135 has no screensaver source option on LCD display webpage
- Fixed Bluetooth name in LCD and webpage are displayed abnormally
- Fixed phone continues to send SIP options after receiving "415 Unsupported Media Type"
- Fixed no conference mode in LCD VPK/MPK settings page
- Fixed conference MPK icon is the same with speed dial mode
- Fixed conference ID is still displayed on extension board after cleaning
- Fixed some new features are still resettable even in default status
- Fixed Bluetooth power option in LCD displayed mistakenly



- Fixed screensaver settings in LCD not identical with those on webpage
- Fixed pressing RIGHT key still could modify VPK in keyboard menu lock mode
- Fixed "Enable DHCP VLAN" feature tip info displaying XXXX
- Fixed LCD does not skip after EXT speed dial in target page
- Fixed GXP color phone should wait to send REFER until the hold process is complete when doing call park
- Fixed the existed extra "Call History Flash Writing" option on call features webpage
- Fixed added phone will still resend SUBSCRIBE when receiving 489 response with non-zero Retry-After header
- Fixed there is one-way audio in Hand-free call answered by "Answer" softkey

#### **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

#### SUPPORT FOR CONFIGURING RTP PORT RANGE

#### Web Configuration

This option can be found under device web  $UI \rightarrow Settings \rightarrow General settings$ .

Local RTP Port	5004	
Max Local RTP Port	5204	Max Local RTP Port
Use Random Port	No ○ Yes	This parameter defines the Max local RTP port from 1024
Keep-Alive Interval	20	to 65400. Local RTP port will be incremented by 4 on every
Use NAT IP		new call so this number has to be greater than base RTP port by 4 for each new line
STUN server		Reset to Default
Public Mode	No ○ Yes	O Undo
Delay Registration	0	

#### Functionality

It gives users the ability to define the parameter of the local RTP port used to listen and transmit. This parameter defines the Max local RTP port from 1024 to 65400. Local RTP port will be incremented by 4 on every new call so this number has to be greater than base RTP port by 4 for each new line.

#### New P value

P value	Description	Value Range	Default
P29608	<new p="" value=""></new>	String:	5204



Local	RTP	port	1024 to 65400	
range	on gxp	21XX		
color p	hones			

#### SUPPORT MORE KEYS USED AS SEND

## • Web Configuration

This option can be found under device web UI→ Account → Account# → Call Settings

Key As Send	○ Disabled ● Pound (#) ○ Star (*)
	.,

## Functionality

Pressing selected key will immediately dial out.

## New Pvalue

P value	Description	Value Range	Default
P72	<new p="" value=""> Key As Send</new>	Value = 0; Disabled Value = 1; Pound (#) Value = 2; Star (*)	1

## SUPPORT OF CONFIGURABLE HTTP/HTTPS WEB PORT

## • Web Configuration

This option can be found under device web UI → Maintenance → Security Settings → Security.

HTTP Web Port	80
HTTPS Web Port	443

## Functionality

To configure the HTTP/HTTPS port from the web interface.

#### New P value

P value	Description	Value Range	Default
P22120	<new p="" value=""> HTTP Web Port</new>	string	80
P22121	<new p="" value=""></new>	string	443



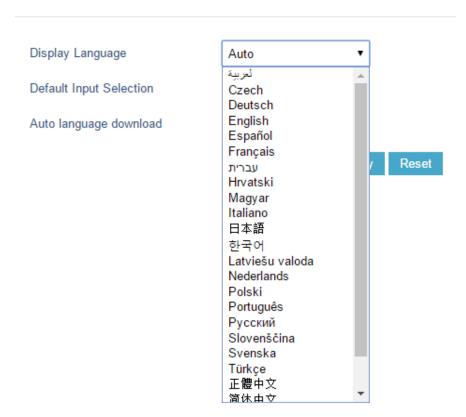
	HTTPS Web Port		
--	----------------	--	--

#### SUPPORT FOR LANGUAGE INPUT SEARCH

## Web Configuration

This option can be found under device web UI → Maintenance → Language.

# Language



## Functionality

For Latin-based languages like German and Czech, device would allow user to Input with Hrvatski, Czech, Deutsch, Spanish, French, Hungarian, Italian, Portuguese, Slovenian, and Turkish when selecting corresponding language display on LCD.

#### SUPPORT OF SINGLE BUTTON CALL PARK

## • Web Configuration

This option can be found under device web UI → Settings → Programmable Keys. → Programmable Keys.



## Multi-Purpose Keys

	Mode	Account	Description	Value
MPK 1	None ▼	Account 1 ▼	Description	Value
MPK 2	None Speed Dial	Account 1 ▼	Description	Value
MPK 3	Busy Lamp Field (BLF) Presence Watcher	Account 1 ▼	Description	Value
MPK 4	Eventlist BLF Speed Dial via Active Account	Account 1 ▼	Description	Value
MPK 5	Dial DTMF Voice Mail	Account 1 ▼	Description	Value
MPK 6	Call Return Transfer Call Park	Account 1 ▼	Description	Value
MPK 7	Monitored Call Park	Account 1 ▼	Description	Value
MPK 8	Intercom LDAP Search Conference	Account 1 ▼	Description	Value
MPK 9	Multicast Paging	Account 1 ▼	Description	Value
MPK 10	Record Call Log Menu	Account 1 ▼	Description	Value
MPK 11	None ▼	Account 1 ▼	Description	Value

Accounts → Account# → SIP Settings → Advanced Features:

Call Pickup Barge-In Code	
---------------------------	--

## Functionality

Allow users to configure the Barge-In code for the call park function from the MPK. During an active call when the idle call park MPK button (LED green) is pressed, the call will be parked. When the busy call park MPK button is red, which means a call has been parked. By pressing the Busy (LED red) MPK button, The barged-in code will be prepended to the call park number and the parked call will be retrieved.

#### New P value

P value	Description	Value Range	Default
P323	<added option=""> MPK Mode</added>	Value = 10; Monitored Call Park	None
P26046	<new p="" value=""> Call Pickup Barge-In Code</new>	String	N/A



#### **OPTION TO SET DTMF DELAY**

## • Web Configuration

This option can be found under device web UI  $\rightarrow$  Accounts  $\rightarrow$  Account#  $\rightarrow$  Audio Settings.

DTMF Payload Type	101
Send DTMF	□ in-audio  via RTP (RFC2833) □ via SIP INFO
DTMF Delay	250

## Functionality

Add ability for user to configure the delay between sending DTMF during MPK/VPK use (in milliseconds).

#### New P value

P value	Description	Value Range	Default
P26060	<new p="" value=""> DTMF Delay</new>	string	250

#### **OPTION TO DISABLE HTTP ACCESS**

## • Web Configuration

This option can be found under device web UI → Maintenance → Security Settings → Security

Web Access Mode ○ HTTPS ● HTTP ○ Disable	Web Access Mod	e 🔘	HTTPS	HTTP	Disable
--	----------------	-----	-------	------	---------

#### Functionality

Allow user to set protocol or disable access to web interface.

#### New P value

P value	Description	Value Range	Default
P1650		Value = 0; HTTPS Value = 1; HTTP Value = 2; Disabled	НТТР

## **SILENT RINGTONE OPTION**

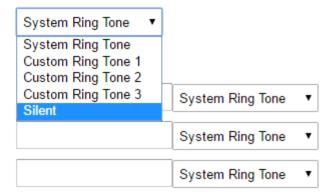
## • Web Configuration

This option can be found under device web UI→Accounts→ Account# → Call Settings.



Account Ring Tone

Match Incoming Caller ID



## Functionality

Allow user to choose silent ring tone.

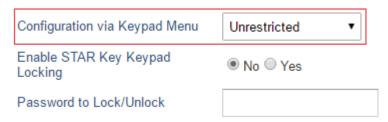
#### New P value

P value	Description	Value Range	Default
P423	<added value=""> Account Ring Tone</added>	Value = 0; System Ring Tone Value = 1; Custom Ring Tone 1 Value = 2; Custom Ring Tone 2 Value = 3; Custom Ring Tone 3 Value = 4; Silent	System Ring Tone

#### **CONFIGURATION VIA KEYPAD MENU**

#### Web Configuration

This option can be found under device web UI→Maintenance→Security Settings→Security.



## • Functionality

Allow user to lock or restrict to only call/receive functionality without menu access and ability to configure anything from phone side.

#### New P value

P value	Description	Value Range	Default
P1357	<new p="" value=""></new>	Value = 0; Unrestricted	Unrestricted



Configuration via	Value = 1; Basic settings only	
Keypad Menu	Value = 2; Constraint Mode	
	Value = 3; Locked Mode	

#### **INSTANT MESSAGE DISPLAY DURATION**

#### Web Configuration

This option can be found under device web UI→Settings→Call Features.

Instant Message Popup Timeout 10

Play Tone On Receiving IM Disabled Enabled

#### Functionality

Increase the max duration displayed to 900 and the backlight will turn on when receiving an IM while idle. Also allow the user to set sound notification when receiving an instant message.

## New P value

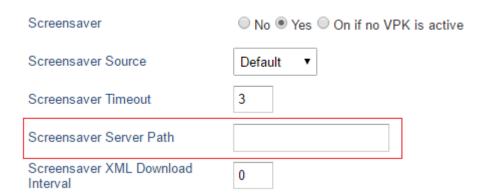
P value	Description	Value Range	Default
P26055	<new> Instant Message Popup Timeout</new>	0-900	10

#### **SCREENSAVER SERVER PATH**

## Web Configuration

This option can be found under device web UI→Settings →Preferences-->LCD Display.

## Screensaver





#### Functionality

Allow user to download their screensaver from the HTTP server.

#### New P value

P value	Description	Value Range	Default
P934	<new p="" value=""> Screensaver Server Path</new>	String	N/A

#### **ENABLE DHCP VLAN**

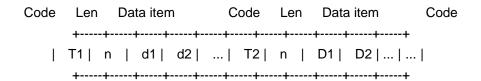
## • Web Configuration

This option can be found under device web UI →Network -> Advanced Settings

## Functionality

Added support DHCP Option 132 & 133 tunneled through DHCP Option 43 on the GXP21xx Color to deploy it on the network. By enabling the DHCP VLAN, the phone will try to read VLAN setting from DHCP option 132 and 133. If both of values in these two options are empty, phone will then read from option 43.

The format of the data in Option 43 should be like as follows:



Here is a valid example:

84:04:31:31:34:00:85:02:35:00

Which means tunnel option 132 value 114 and option 133 value 5 via option 43. Option 132 for VLAN ID and 133 for priority, or from Option 43 which encapsulates both two values.

#### New P value

P value	Description	Value Range	Default
P8300	<new p="" value=""> Enable DHCP VLAN</new>	Value = 0; Disabled Value = 1; Enabled	0



#### MPK FOR CONFERENCE

## Web Configuration

This option can be found under device web UI→Settings →Programmable Keys→ Programmable Keys.

# Multi-Purpose Keys

	Mode	Account	Description	Value
MPK 1	None ▼	Account 1 ▼	Description	Value
MPK 2	None Speed Dial Busy Lamp Field (BLF)	Account 1 ▼	Description	Value
MPK 3	Presence Watcher Eventlist BLF	Account 1 ▼	Description	Value
MPK 4	Speed Dial via Active Account Dial DTMF	Account 1 ▼	Description	Value
MPK 5	Voice Mail Call Return	Account 1 ▼	Description	Value
MPK 6	Transfer Call Park	Account 1 ▼	Description	Value
MPK 7	Monitored Call Park	Account 1 ▼	Description	Value
MPK 8	LDAP Search Conference	Account 1 ▼	Description	Value
MPK 9	Multicast Paging Record	Account 1 ▼	Description	Value
MPK 10	Call Log Menu	Account 1 ▼	Description	Value
MPK 11	None ▼	Account 1 ▼	Description	Value

#### Functionality

Allow user to set their Multi-Purpose Key to "Conference" mode to trigger a conference. By setting the extension number in the value box, the users will be able to activate a 3-way conference by simply press the assigned MPK button.

#### New P value

P value	Description	Value Range	Default
P323	<added p="" value=""></added>	Value = "12"; Conference	0

#### **DIAL FROM LOCAL PHONEBOOK**

## • Phone Configuration

This option can be found under device LCD Menu→Contacts→Local Phonebook→[Select number] →Edit/Dial.





#### Functionality

When the new option = Edit/Dial - is selected, user can edit the phone number and choose one account to dial to that number.

#### **CALL FORWARDING FROM WEB UI**

## • Web Configuration

This option can be found under device web UI→Accounts→ Account # →Feature Codes.

Delayed Call Forward Wait Time

# Call Forward Unconditionally (All) On Off Target Call Forward Busy On Off Target Call Forward Delayed (No Answer) On Off Target



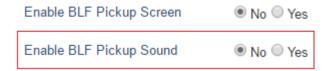
## Functionality

Allow user to configure their forward calls to forward the calls to specific target. The users will be able to set either unconditionally, Busy, or Delayed (No Answer) for the call forward destination.

#### **BLF PICKUP SOUND**

#### Web Configuration

This option can be found under device web UI→Settings→Call Features.



## Functionality

Give the user the ability to set sound notification to the monitoring BLF line.

#### New P value

P value	Description	Value Range	Default
P9906	<new p="" value=""> Enable BLF Pickup Sound</new>	Value = 1; Enable Value = 0; Disable	0

## **BLUETOOTH CONFIGURATIONS**

#### Web Configuration

This option can be found under device web UI→Settings→Bluetooth

Bluetooth Power	○ Off ○ Off & Hide Menu From LCD ● On
Handsfree Mode	○ Off ● On
Bluetooth Name	

## Functionality

Allow the user to modify the configuration of the Bluetooth via Web UI.

#### New P value

P value Description	Value Range	Default
---------------------	-------------	---------



P2910	<new p="" value=""> Enable Bluetooth (GXP21XX)</new>	Value = 0; Off Value = 1; On Value = 2; Off & Hide Menu From LCD	1
P2911	<new p="" value=""> Enable Bluetooth Hands-free feature (GXP21XX)</new>	Value = 0; Off Value = 1; On	0
P8364	<new p="" value=""> Bluetooth Device Name</new>	String	DeviceModel_MacA ddress

## **VOICEMAIL INDICATION**

## Web Configuration

This option can be found under device web UI→Settings→Preferences-->LED Control

Disable VM/MSG power light flash 

No Ves

## Functionality

Allow users to enable/disable the LED which indicates if there are any VMs/MSGs.

## New P value

P value	Description	Value Range	Default
P8371	<new p="" value=""> Disable VM/MSG power light flash</new>	Value = 0; No Value = 1; Yes	0

#### **RADOM REGISTRATION**

#### Web Configuration

This option can be found under device web UI→Settings→General Settings.

Delay Registration 0

## Functionality

Add an interval that the user can configure the time that phone will send SIP message to the register account after rebooting.



#### New P value

P value	Description	Value Range	Default
P8359	<new p="" value=""> Delay Registration</new>	string	0

#### **RETURN CODE**

## • Web Configuration

This option can be found under device web UI→Settings→ Call Features.

Return Code When Refusing Incoming Call	Busy(486)	•
Return Code When Enable DND	Busy(486)	•

## Functionality

Allow users to set up the return message to either a Refused call or when Do Not Disturb feature is active with the indicated messages and code number.

#### New P value

P value	Description	Value Range	Default
P8360	<new p="" value=""> Return Code When Refusing Incoming Call</new>	Value = 0; Busy(486) Value = 1; Temporarily Unavailable(480) Value = 2; Not Found (404) Value = 3; Decline(603)	0
P8361	<new p="" value=""> Return Code When Enable DND</new>	Value = 0; Busy(486) Value = 1; Temporarily Unavailable(480) Value = 2; Not Found (404) Value = 3; Decline(603)	0

#### PLANTRONICS EHS HEADSET RINGTONE

## • Web Configuration

This option can be found under device web UI→Settings→Preferences-->Audio Control.



# Headset

HEADSET Key Mode	Default Mode Toggle Headset/Speaker
Headset Type	● Normal ○ Plantronics EHS
EHS Headset Ring Tone	■ Normal  ■ Plantronics EHS
Always Ring Speaker	No ○ Yes
Headset TX Gain (dB)	0 decibel ▼
Headset RX Gain (dB)	0 decibel ▼

# Functionality

Allow user to enable/disable the Plantronics EHS headset and the ringtone of it.

## New P value

P value	Description	Value Range	Default
P6779	<new p="" value=""> EHS Headset Ring Tone</new>	Value = 0; Normal Value = 1; Plantronics EHS	Normal



# FIRMWARE VERSION 1.0.7.81

#### PRODUCT NAME

GXP2130, GXP2140, GXP2160, GXP2170 and GXP2135

#### DATE

09/06/2016

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.
- This release restored the Weather service. Part of the change requires users who are using "Self-Defined City Code" to change their setting. The upstream content provider has changed the API to yield a free-style search based on this argument instead of mapping to a specific code being used before. Hence the city being displayed would have been incorrect. These users should change this option to the name of the city, for instance: "Dallas"; "Dallas, NC"; "Dallas, Scotland"; "Liverpool"; "Liverpool, NY"; "Liverpool, England". Note: If you enter only the name of the city it's going to go the best match. In the case of Dallas it would be Dallas, TX; and in the case of the Liverpool it would be Liverpool, England.
- This release adds Predictive Dialing feature which included a required internal change. If you downgrade from 1.0.7.81 to 1.0.7.25 you will lose the call history.

#### **ENHANCEMENT**

- Added Support for local firmware upgrade [FIRMWARE LOCAL UPGRADE]
- Added a web option to let user chose whether or not to display internet down warning window [SHOW INTERNET DOWN MESSAGE]
- Added OPUS codec support [OPUS CODEC]
- Added target softkey for dialing status [TARGET SOFTKEY]
- Added Capability to allow DIAL softkey to dial the selected predictive dialing number under dialing state
- Added target selection for transfer and conference in call screen [TARGET SELECTION]
- Added Transfer destination should display who the call was referred by [TRANSFER FROM INFOMATION]
- Added support to accept Pvalue in string format for VPK mode configuration xml [PVALUE FOR VPK MODE IN STRING FORMAT SUPPORT]
- Added predictive dialing using call history entries [PREDICTIVE DIALING]
- Added more numbers per contact and "Company" "Title" "Job" Field on the Phonebook [MULTIPLE NUMBERS AND OTHER INFO FOR ONE CONTACT SUPPORT IN PHONEBOOK]



- Added pre-Dialing search to include Broadsoft directories [PREDICTIVE DIALING SEARCH SUPPORT]
- Added search feature on Broadsoft directories [SEARCH FROM BROADSOFT DIRECTORIES SUPPORT]

#### **BUG FIX**

- Fixed Time Zone on status page does not show correctly sometimes
- Fixed LCD brightness set as active issue when DHCP renew and other network connection
- Fixed BLF lights remain RED/GREEN after hung up issue when using LED Pattern reserved RED/GREEN
- Fixed Adding path / to server prefix will cause phone get stuck issue when downloading ringtone
- Fixed Phone does not keep assigned VLAN via LLDP issue
- Fixed Phone couldn't obtain IP after switching from LLDP to common network circumstance issue
- Fixed Phone gets stuck in Downloading firmware file page issue after uploading firmware then unplug net cable
- Fixed Accounts disappeared after connect/disconnect extension board
- Fixed MUTE LED turned off sometimes when switching to home screen (during a call when mute is on)
- Fixed Device TFTP upgraded abnormally in LLDP network environment
- Fixed LCD couldn't enter into screensaver status after timeout
- Fixed In BLF auto-provision, even if we choose the startpoint to be ext, it also shows on LCD on bootup sometimes
- Fixed GXP2140 cannot transfer using BLF issue
- Fixed Set "sort Results" option of LDAP web page to "yes" does not take effect
- Fixed Issues with call park with Freeswitch on 1.0.7.X (While on active call only)
- Fixed device generated core dump for gs\_gui on web page
- Fixed SIP log sent through syslog got truncated
- Fixed loud sound/noise is played when there is audio packets loss using G 729 audio codec
- Fixed when call Initiated within XML APP, suppress change to default phone screen
- Fixed phone stucked at base download when using LLDP-MED.
- Fixed Broadsoft phone book contact details string and formatting issue
- Fixed Broadsoft XMPP: contact list display "Unknown Contact"
- Fixed phone loss of register every 32 minutes due to failing to resolve DNS
- Fixed incoming multicast paging drops the actual ringing call
- Fixed all MPK values disappear from web GUI when one of them containing a double quote
- Enabled weather support using new yahoo API
- Fixed status of auto provision Eventlist BLF VPK/MPK showed wrongly when monitored Eventlist has parking lot extension

#### **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.



#### FIRMWARE LOCAL UPGRADE

## Functionality

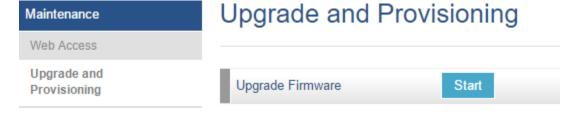
User could upload the new firmware by pressing Start, then a new pop-up windows will show like following.



After selected the correct firmware file from local storage, the phone will start the firmware upgrade automatically.

## • Web Configuration

The users could find this new feature under phone's web UI-> Maintenance->Upgrade and Provisioning.

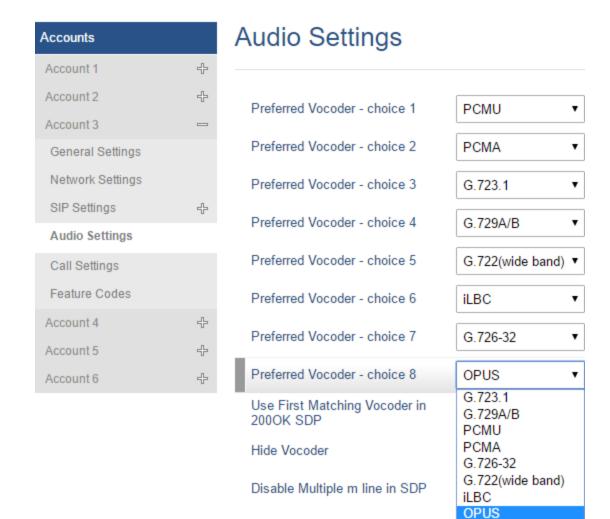


#### **OPUS CODEC**

## Web Configuration

The users could find OPUS codec available under phone's web UI→Account→Audio Settings.





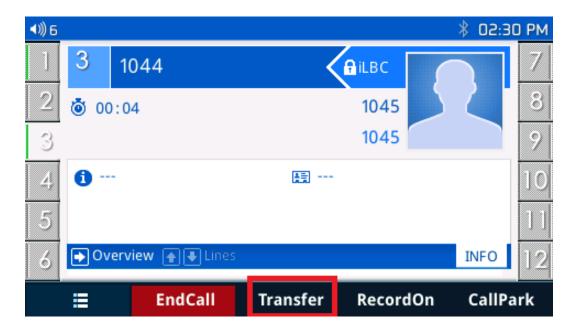
When provisioning the phone with vocoder P value, the value to set codec as OPUS is 123.

## **TARGET SOFTKEY**

#### Functionality

User can find the new target softkey for transfer status. When user pressed transfer softkey during a call.





User could press the Target Softkey:



The LCD will lead to a page showing "Local Phonebook", "LDAP Directory", "Local Call Log" and "Smart VPK Target" that the user could select and proceed the call transfer.





#### **SHOW INTERNET DOWN MESSAGE**

#### Functionality

This document provides a solution for customers who mainly use the intranet without connecting to the internet. Originally, if the internet is down, there will be a warning window displaying "Internet Down" message, user cannot operates the phone with the warning window display.

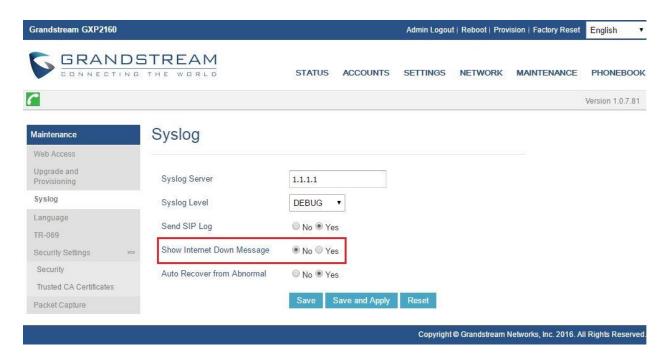
This design will provide following functionality.

- → User can chose whether or not to display the "Internet Down" message based on their environment settings.
- → Phone can still be operated even when the internet is not connected.

## • Web Configuration

The users could find OPUS codec available under phone's web UI→Maintenance→Syslog→Show Internet Down Message.





## Examples

#### → Under Idle Screen:

A warning window will display if "Show Internet Down Message" is enabled



A warning window will not display if "Show Internet Down Message" is disabled.





#### →Under Call Screen:

None

#### TARGET SELECTION

#### Functionality

With this new feature, users can realize the following function:

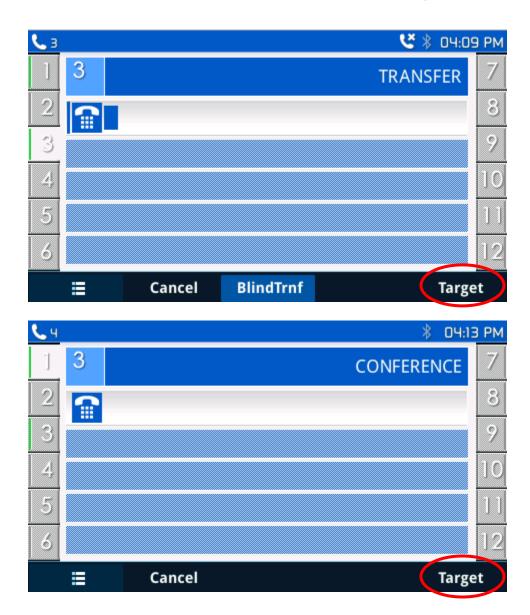
- 1) Add a new softkey "Target" under transfer/conference action page.
- 2) When click on the softkey, it will take user to a target selection page in which user can choose other possible transfer/conference targets (e.g. Local/BS Phonebook, Local/BS Call History, LDAP, Smart VPK List, or connected lines)
- 3) When accessing Local/BS Phonebook and Local/BS Call History page, the softkey will show limited options (e.g. Dial) when phone is under busy state to make it easier for user to perform the dialing task.
- 4) When accessing Smart VPK List, we will display the list of the VPK targets which are allowed to be used as the transfer/conference target based on the current account.
- 5) When accessing Line, we will list the lines which are available to be used for transfer/conference.

#### Examples

a) Under Transfer/Conference screen

A new softkey "Target" is added under transfer/conference action page.





When click on the softkey, it will take user to a target selection page in which user can choose other possible transfer/conference targets.

# b) Under target selection page screen

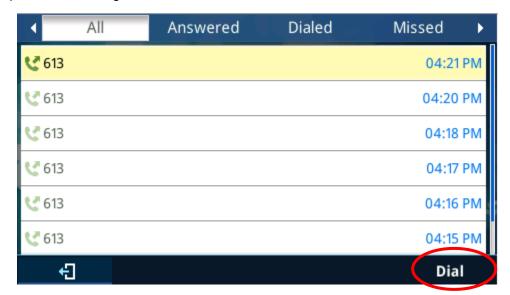
Target selection page provides a list of possible transfer/conference targets.





User can use UP/DOWN key to select target from list and use MENU key or "Select" softkey to choose the target.

By selecting Local/Broadsoft Phonebook, Local/Broadsoft Call Log and LDAP Directory, LCD will display according menu page. Also, in these pages, softkey will show limited options (e.g. Dial) to make it easier for user to perform the dialing task.



By selecting "Smart VPK Target", a list of available VPK for transfer/conference will be shown as bellow. User can use UP/DOWN key or "select" softkey to choose item to use.

Available VPK contains VPK with mode as below:

- Speed Dial (current account)
- Speed Dial via Active Account (any account)
- BLF or Eventlist BLF (current account)



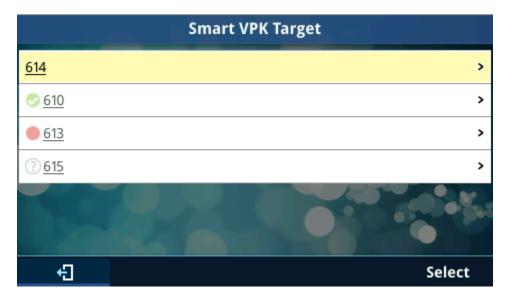
• Presence (current account)

Specially, for BLF/Eventlist BLF/Presence, an icon with status will be shown on the left side of the item. Icon here contains 3 types based on monitoring state:

• Unknown : Question Mark

Busy : RED

Available : GREEN



By selecting "line", a list of available line for transfer/conference will be shown as bellow. User can use UP/DOWN key or "select" softkey to choose item to use.

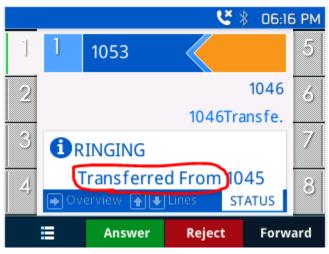




#### TRANSFER FROM INFOMATION

## Functionality

With this new feature, users can see who transferred the call to them. Example shows below:



The information of the Transferred From can be found in the "Referred-by" Header from SIP INVITE.

#### **PVALUE FOR VPK MODE IN STRING FORMAT SUPPORT**

#### Functionality

User can use the actual Virtual Multi-Purpose Key Mode in configuration xml or the Pvalue. The current mode string and Pvalue are as follows:

Mode Name	Mode String	Mode Pvalue
None	none	-1
LINE	line	0
Busy Lamp Field (BLF)	blf	1
Presence Watcher	presencewatcher	2
Eventlist BLF	eventlistblf	3
Speed Dial via Active	speeddialaa	4
Account		
Dial DTMF	dialdtmf	5
Voice Mail	voicemail	6
Call Return	callreturn	7
Transfer	transfer	8
Call Park	callpark	9
Monitored Call Park	moniteredcp	16
Intercom	intercom	10
LDAP Search	Idapsearch	11
Multicast Paging	multicastpaging	13



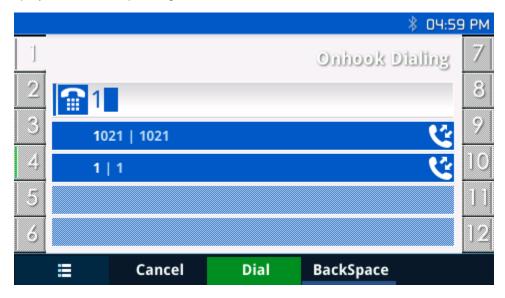
Record	record	14
Call Log	calllog	15
Menu	menu	17

The string could be capital or lower case letters, but there must no "space" in between. For example in the cfg.xml, "<P1367>Line</P1367>" or "<P1367>line</P1367>" is the same as "<P1367>0</P1367>, it will configure Virtual Multi-Purpose Key 3 as Line mode.

#### PREDICTIVE DIALING

#### Functionality

Phone can look into call history entries and if entered digits match some call history entries, list of entries should be displayed with corresponding names.



#### MULTIPLE NUMBERS AND OTHER INFO FOR ONE CONTACT SUPPORT IN PHONEBOOK

## Functionality

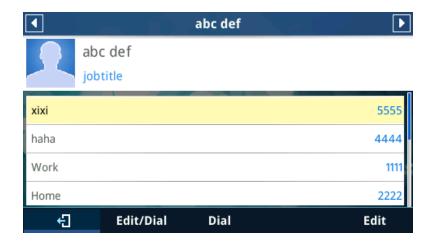
User can add Phone type in phonebook.xml to add other phone type besides existing "Work","Home" and "Cell". Here is an example for the phonebook.xml:

- <AddressBook>
- <Contact>
- <id>1</id>
- <FirstName>abc</FirstName>
- <LastName>def</LastName>
- <JobTitle>jobtitle</JobTitle>



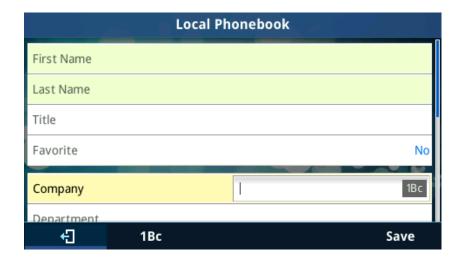
- <Frequent>0</Frequent>
- <Phone type="haha">
- <phonenumber>4444</phonenumber>
- <accountindex>0</accountindex>
- </Phone>
- <Phone type="xixi">
- <phonenumber>5555</phonenumber>
- <accountindex>0</accountindex>
- </Phone>
- <Phone type="Work">
- <phonenumber>1111</phonenumber>
- <accountindex>0</accountindex>
- </Phone>
- <Phone type="Home">
- <phonenumber>2222</phonenumber>
- <accountindex>0</accountindex>
- </Phone>
- <Phone type="Cell">
- <phonenumber>3333</phonenumber>
- <accountindex>0</accountindex>
- </Phone>
- <Primary>1</Primary>
- <Department>department</Department>
- <Job>job</Job>
- <Title>title</Title>
- <Company>company</Company>
- </Contact>
- </AddressBook>

On the display of the contact on LCD of the phone will shows as follows:





User can also input the "Company" "Title" "Job" filed through the phonebook.xml in the previous example or on the LCD of the phone like following example:

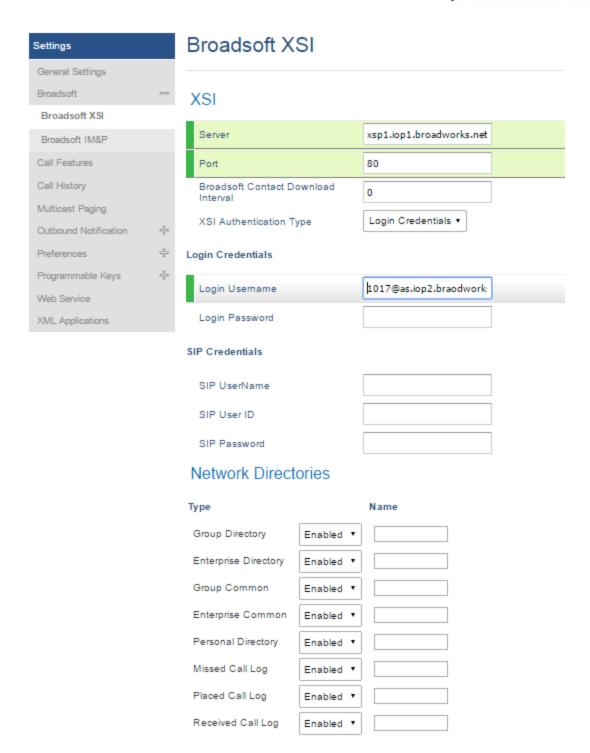


#### PREDICTIVE DIALING SEARCH SUPPORT

## Web configuration

A user has to configure Broadsoft XSI directories correctly from Web. The Web configuration path is as follows: Settings -> Broadsoft -> Broadsoft XSI.





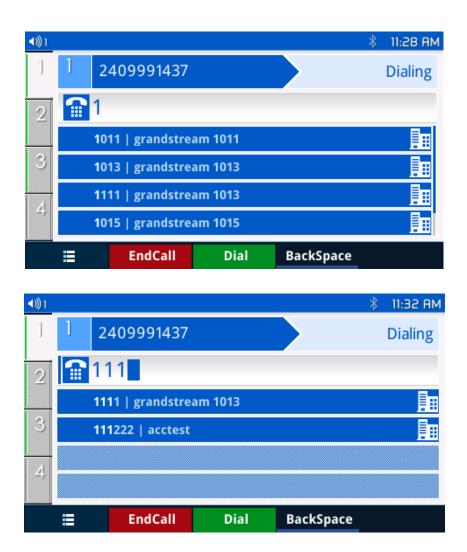
## Functionality

Phone can predict possible contact information from Broadsoft XSI directories when a user dials number. User can select the predictive dialing number to dial out even when user doesn't key in the complete number.

Predictive numbers will be shown if Broadsoft XSI directory information is loaded after pressing the related



number.

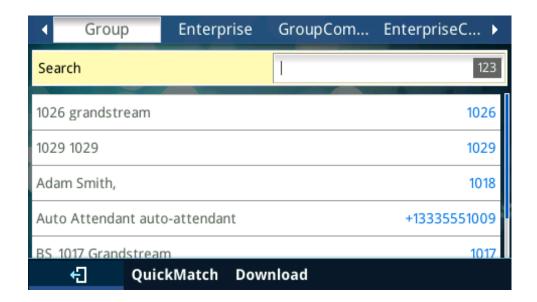


#### SEARCH FROM BROADSOFT DIRECTORIES SUPPORT

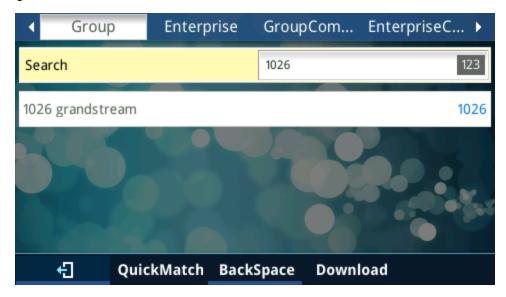
## Functionality

User can search from the Broadsoft directories when the user input the search content in the search form.





The matching list will show as follows:





# FIRMWARE VERSION 1.0.7.25

#### **PRODUCT NAME**

GXP2130, GXP2140, GXP2160, GXP2170 and GXP2135

#### **DATE**

05/10/2016

#### IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15 or above, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15 or above, downgrading to 1.0.6.9 firmware version or lower is not supported.

#### **ENHANCEMENT**

- Added support for Broadsoft XSI authentication type [BROADSOFT XSI AUTHENTICATION TYPE SUPPORT]
- Added support to configure Broadsoft XSI SIP authentication method by selecting the account [CONFIGURE BROADSOFT XSI SIP AUTHENTICATION METHOD]
- Added support to stop Screensaver when VPK is active
- Improve Auto-provision for Vonage server
- Added option to disable Auto Location Service from IpVideoTalk server [ENABLE/DISABLE AUTO LOCATION SERVICE FROM IpVideoTalk SERVER]
- Added supports for secondary NTP server [SECONDARY NTP SERVER]
- Added the ability to specify Eventlist BLF listening transport protocol which will allow the phone to listen
  on the incoming notify for the Eventlist through different transport protocol than the one used by SIP
  [EVENTLIST BLF LISTENING TRANSPORT PROTOCOL]
- Added support to play sound notification when one or more monitored BLF is ringing [SOUND NOTIFICATION]
- Added support to populate configurable User Agent field [USER-AGENT PREFIX]
- Added support to request model-specific configuration file during the provisioning process [REQUEST MODEL-SPECIFIC CONFIGURATION FILE]
- Added support to remove audio codec information on call screen [HIDE VOCODER]
- Added support of BLF call pickup with Barge-In option [CALL PICKUP BARGE-IN OPTION]
- Added option to control Speakerphone RX gain [SPEAKERPHONE RX GAIN CONTROL]
- Added support to display status detail when Ethernet not connected, account not register or configured [STATUS DETAIL DISPLAY]
- Added support of IEEE-48-addr and IEEE-EUI-64 in PEI header [SUPPORT FOR IEEE-48-ADDR AND IEEE-EUI-64 IN PEI HEADER]



- Added support to allow partial match using STAR code In Broadsoft XSI search [PARTIAL MATCH IN BRAODSOFT XSI SEARCH]
- Added support to filter out invalid symbols (ex. brackets, hyphen) from dialed numbers [FILTERING OUT INVALID SYMBOLS]
- Added support to populate the WAN access node in SIP PANI head [SUPPORT FOR POPULATING THE WAN ACCESS NODE IN SIP]
- Added ability to allow user to specify which number to dial from LDAP results [DIAL FROM LDAP RESULTS]
- Added DNS SRV Fail-over Mode option support [DNS SRV FAIL-OVER MODE]
- Added Conference CEI to support UCM conference change
- Added separate subscription expire options for each account [SEPARATE SUBSCRIPTION EXPIRATION TIMER]
- Added more information on info popup window [LCD POPUP WINDOW]
- Added support for default Dial Plan { x+ | \+x+ | \*xx+ } [DIAL PLAN { x+ | \+x+ | \*xx+ }]

#### **BUG FIX**

- Fixed Phone cannot hang up after performing Blind Transfer in speaker mode
- Fixed that GUI crashed after kicking UCM conference participant
- Fixed that if the proxy does NOT respond with Proxy DNS SRV it will jump back after 1 registration
- Fixed that during SIP fail over due to no response from the server the device does not do a DNS lookup for the SIP proxy domain
- Fixed that if the proxy responds with 503 it will not fail over to the next IP in the DNS SRV
- Fixed VONAGE-GUI crashed after manually configure Eventlist BLF
- Fixed that call will be drop automatically by the phone after 45 sec when phone accomplishes an attended transfer and receive a call after that
- Fixed that Broadsoft XSI app can't establish secure connection
- Fixed that blind transfer ignores the pattern destinations specified in the dial plan
- Fixed that device does not displaying Name field in missed calls if Call waiting is disabled
- Fixed that phone keeps sending SIP instance ID after changing its value to NO unless phone is rebooted.
- Fixed that NAPTR/SRV mode in DNS does not work
- Fixed phone rejects messages from known proxy when proxy is cleared from DNS cache
- Fixed phone rejects OPTIONS/UPDATE from known server when the option Accept Incoming SIP from Proxy only is enabled
- Fixed that CallPark Softkey works intermittently
- Fixed web GUI Edit/Delete operations fail issue when performing on contacts which appear on search results
- Fixed that Auto provision Eventlist BLF does not work with Broadsoft servers
- Fixed that Auto Provision Eventlist BLF must be filled with EXT field from XML



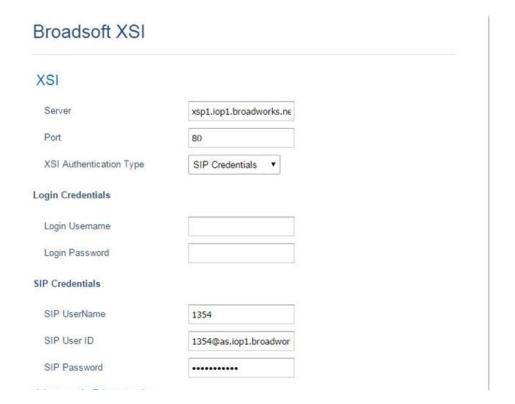
#### **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

#### **BROADSOFT XSI AUTHENTICATION TYPE SUPPORT**

#### Web Configuration

This option can be found under device web UI->Settings->Broadsoft→Broadsoft XSI.



#### Functionality

This option is used to provide SIP Authentication option for Broadsoft XSI. Configure the SIP Credentials and select SIP Credentials as XSI Authentication Type. Phone will offer SIP Credentials when connecting to Broadsoft XSI.

#### New P Values

Pvalue	Description	Value range	Default
P6772	<new pvalue=""></new>	String	None
	Pvalue to set SIP UserName for		
	Broadsoft XSI		
P22034	<new pvalue=""></new>	String	None
	Pvalue to set SIP User ID for		
	Broadsoft XSI		



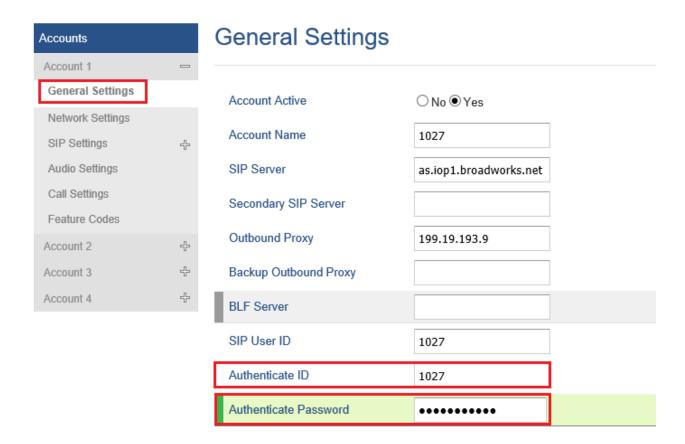
P6773	<new pvalue=""></new>	String	None
	Pvalue to set SIP Password for		
	Broadsoft XSI		
P22054	<new pvalue=""></new>	0 for Login Credentials	0
	Pvalue to set XSI Authentication Type	1 for SIP Credentials	
	for Broadsoft XSI		

#### CONFIGURE BROADSOFT XSI SIP AUTHENTICATION METHOD

#### Web Configuration

User can use Authenticate ID and Authenticate Password as SIP Username and SIP Password respectively account this feature from Web configuration

In Accounts Tab -> Account # -> General Settings:

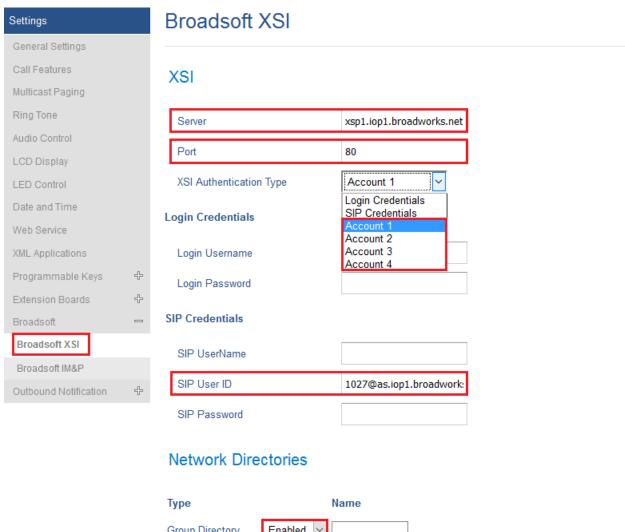


Authenticate ID and Authenticate Password must be configured if a user select that account for accessing Broadsoft XSI Directory and Call Log.

In Settings Tab -> Broadsoft -> Broadsoft XSI Settings, Users need to configure Server Path, Port, account



number, SIP User ID and enable the XSI directories they would like to access.





#### Functionality

This new method for Broadsoft XSI authentication is selecting SIP account instead of manually input SIP Username and SIP Password. By this way, users can save time on configuration.

Phone can retrieve Authenticate ID and Authenticate password information from the selected account and use them as SIP credential for accessing Broadsoft XSI Directory and Call Log information.

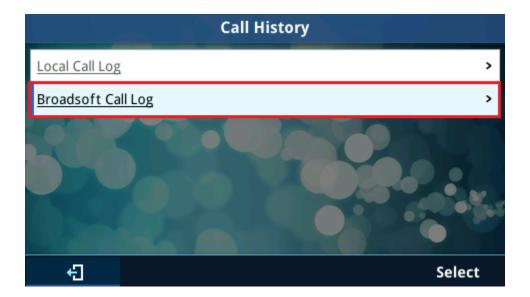
Check functionality under Menu Screen: when all required information are set, Broadsoft Phonebook option



will appear in Contacts tab.



Broadsoft Call Log option will appear in Call History tab.



# Changed P Values

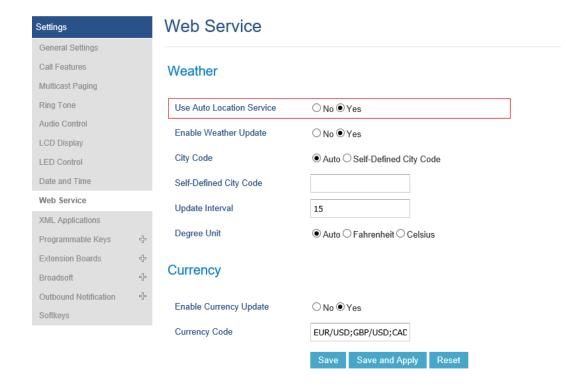
P value	Description	Value range	Default
P22054	<changed pvalue=""></changed>	2 to 7 for SIP Credentials	0
	Pvalue to set XSI Authentication Type	of account 1 to 6.	
	for Broadsoft XSI		



#### ENABLE/DISABLE AUTO LOCATION SERVICE FROM IpVideoTalk SERVER

#### • Web Configuration

This option can be found under device web UI->Settings→Web service:



#### Functionality

User could enable or disable auto location services on the phone to decide whether phone could get the location information from IPVideoTalk service server. Reboot required to take effect.

In the original design, the phone will automatically send DNS request to IPVideoTalk server for location information at phone booting.

The option give users a choice to disable auto location feature. If users disable the service, the phone will stop sending DNS request for location information. It can reduce network traffic and speed up the provisioning process as well.

To check the functionality, we use Wireshark to trace DNS packets.

When Use Auto Location Service option is set as Yes, user can find DNS request send to service.ipvideotalk.com.



No.	Time	Source	Destination	Protocol	Lenath	Info			
109	25.68982	8172.18.	1192.168.	4 DNS	83	Standard	query	0x2786 A	service.ipvideotalk.com
110	25.70723	6192.168	.172.18.2	3 DNS	543	Standard	query	response	0x2786 A service.ipvideotalk.com CNAME r
111	25.71098	7172.18.	7192.168.	4 DNS					AAA service.ipvideotalk.com
112	25.71175	4192.168	.172.18.2	3 DNS					Ox7d8d AAAA service.ipvideotalk.com CNAM
113	25.72053	6172.18.	7192.168.	4 DNS					service.ipvideotalk.com
114	25.72136	3192.168	.172.18.2	3 DNS	543	Standard	auerv	response	Oxfc36 A service.ipvideotalk.com CNAME r
964	37.82365	4172.18.	7192.168.	4 DNS					fm.grandstream.com
965	37.82444	5192.168	.172.18.2	3 DNS					Oxbc49 A fm.grandstream.com A 54.169.67.
984	40.17662	5172.18.	7192.168.	4 DNS					fm.grandstream.com
985	40.17738	5192.168	.172.18.2	3 DNS					0x181d A fm.grandstream.com A 54.169.67.
1006	42.47322	2172.18.	1192.168.	4 DNS	78	Standard	query	0x69b4 A	fm.grandstream.com
1007	42.47399	9192.168	.172.18.2	3 DNS					0x69b4 A fm.grandstream.com A 54.169.67.
1048	44.40531	3172.18.	7192.168.	4 DNS	78	Standard	query	0xbb9c A	fm.grandstream.com

When Use Auto Location Service option is set as No, after rebooting the phone, user can find out there is no DNS request send to service.ipvideotalk.com in the Wireshark trace.

	Time								
1086	53.819271	172.18.	192.168.	4 DNS	78	Standard	query	0xae39 A	fm.grandstream.com
1087	53.820039	192.168	.172.18.2	3 DNS	190	Standard	query	response	Oxae39 A fm.grandstream.com A 54.169.67.73
1128	56.274590	172.18.	7192.168.	4 DNS	78	Standard	query	0xb2c8 A	fm.grandstream.com
	56.275389								Oxb2c8 A fm.grandstream.com A 54.169.67.73
1150	58.258002	2172.18.	7192.168.	4 DNS	78	Standard	query	0xa48a A	fm.grandstream.com
1151	58.258779	9192.168	.172.18.2	3 DNS	190	Standard	query	response	Oxa48a A fm.grandstream.com A 54.169.67.73
1168	60.184733	3172.18.	7192.168.	4 DNS	78	Standard	query	0x9f6e A	fm.grandstream.com

# New P Values

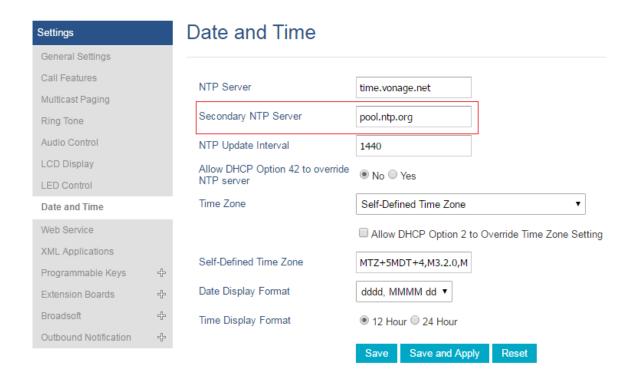
Pvalue	Description	Value range	Default
P8338	<new pvalue=""></new>	0: No	1
	Pvalue to disable or enable Auto	1: Yes	
	Location Service		

## **SECONDARY NTP SERVER**

# • Web Configuration

User can find the new option at Web configuration Settings  $\rightarrow$  Preferences-->Date and Time.





New design will improve current NTP server's behavior in two ways bellow.

GXP should try all of the IP address that DNS server provided. Previously, GXP doesn't try next IP in DNS resolution if there is no response from current IP.

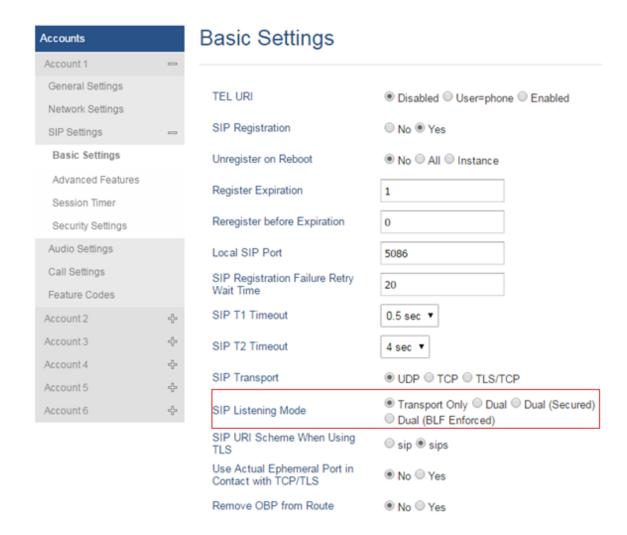
Allow user to configure 2 NTP server domain names. GXP will loop through all of the IP addresses resolved from them.

### EVENTLIST BLF LISTENING TRANSPORT PROTOCOL

## Web Configuration

User can find the new option at Web configuration Settings→Accounts→Account Number→SIP Settings→Basic Settings.





Based on option "SIP Transport" and new option "SIP Listening Mode", GXP will decide which transport protocol it should listening to from the incoming request.

SIP						
Transport	UDP		TCP		TLS/TCP	
Mode						
SIP						
Listening Mode						
	Accept	incoming	Accept	incoming	Accept	incoming
Transport Only	request using	UDP.	request using	TCP.	request using	TLS/TCP.
	All outgoing re	quest will	All outgoing r	equest will	All outgoing re	equest will
	go out using U	DP.	go out using	ГСР.	go out using 7	LS/TCP.



	T			
	Accept incoming	Accept incoming	-	
Dual	request using both TCP	request using both TCP		
	and UDP.	and UDP.		
	All outgoing request will	All outgoing request will		
	go out using UDP.	go out using TCP.		
	Accept incoming	-	Accept incoming	
Dual (Secured)	request using both		request using both	
	TLS/TCP and UDP.		TLS/TCP and UDP.	
	All outgoing request will		All outgoing request will	
	go out using UDP.		go out using TLS/TCP.	
	Accept incoming	Accept incoming	-	
Dual (BLF	request using both TCP	request using both TCP		
Enforced)	and UDP.	and UDP.		
	All outgoing request will	All outgoing request will		
	go out using UDP	go out using TCP		
	except for the	except for the		
	BLF/Eventlist	BLF/Eventlist		
	subscription the phone	subscription the phone		
	will add Transport=TCP	will add Transport=TCP		
	into the contact header.	into the contact header.		

# New P values

Pvalue	Description	Value range	Default
P26024	<new pvalue=""></new>	0: Transport Only	0
P26124	Pvalue to set SIP Listening mode for	1: Dual	
P26224	each account Pvalue to disable or	2: Dual (Secured)	
P26324	enable Auto Location Service	3: Dual (BLF Enforced)	
P26424			
P26524			

# **SOUND NOTIFICATION**

# • Web Configuration

User can find the new option at Web configuration Settings→Call Features.





New options added to allow user to enable sound and LCD notification when one or more monitored BLF is ringing.

By Enable BLF Pickup Screen, when monitored BLF is ringing, GXP should pop up a BLF information window.

By Enable BLF Pickup Sound, when monitored BLF is ringing, GXP should play a sound to inform user.

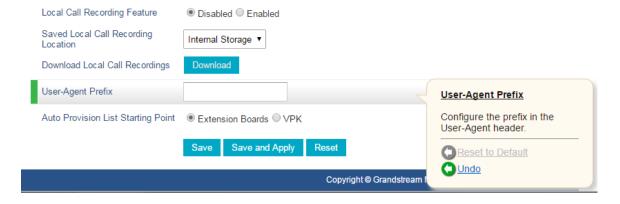
## New P Values

P value	Description	Value range	Default
P9905	<new p="" value=""></new>	0 or 1	0
	Pvalue to enable BLF pickup screen	0: No	
		1: Yes	
P9906	<new p="" value=""></new>	0 or 1	0
	P value to enable BLF pickup Sound	0: No	
		1: Yes	

## **USER-AGENT PREFIX**

# • Web Configuration

User can find the new option at Web configuration Settings → Call Features. A new option "User-Agent Prefix" is added





Add a new option for input the user agent field with operator configurable value or value that identifies the device. The option should be configurable to give the end point device specific identification. Ex. The value could be Mobile, Fixed, Desktop, and etc. The configured User Agent should be prepend to vendor's default User

The new design can be briefly summarized as bellow.

A new P value will be assigned to set User-Agent Prefix. User will be able to configure this option through web, or configuration file. The Prefix User-Agent configured will be prepend to default user Agent.

For example, default User-Agent field: Grandstream GXP2170 1.0.7.25

User-Agent Prefix: Work-Phone

New User-Agent field: User-Agent: Work-Phone Grandstream GXP2170 1.0.7.25

To check the functionality, we make a phone call from DUT and check the SIP INVITE message sent out. Before setting User-Agent Prefix

INVITE sip:741@172.18.32.58 SIP/2.0

Via: SIP/2.0/UDP 172.18.21.34:5060;branch=z9hG4bK669509149;rport

From: "742" <sip:742@172.18.32.58>;tag=1287565619

To: sip:741@172.18.32.58

Call-ID: 448810970-5060-3@BHC.BI.CB.DE

CSeq: 20 INVITE

Contact: "742" sip:742@172.18.21.34:5060

Max-Forwards: 70

User-Agent: Grandstream GXP2170 1.0.7.25

Supported: replaces, path, timer

Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE,

**MESSAGE** 

Content-Type: application/sdp

Accept: application/sdp, application/dtmf-relay

Content-Length: 210

After setting User-Agent Prefix to Work-Phone



INVITE sip:741@172.18.32.58 SIP/2.0

Via: SIP/2.0/UDP 172.18.21.34:5060;branch=z9hG4bK252838042;rport

From: "742" <sip:742@172.18.32.58>;tag=85821971

To: sip:741@172.18.32.58

Call-ID: 1513280525-5060-4@BHC.BI.CB.DE

CSeq: 31 INVITE

Contact: "742" sip:742@172.18.21.34:5060

Max-Forwards: 70

User-Agent: Work-Phone Grandstream GXP2170 1.0.7.25

Supported: replaces, path, timer

Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE,

**MESSAGE** 

Content-Type: application/sdp

Accept: application/sdp, application/dtmf-relay

Content-Length: 210

### New P Values

P value	Description	Value range	Default
P8358	<new p="" value=""></new>	String	NONE
	P value to set User-Agent Prefix		

## REQUEST MODEL-SPECIFIC CONFIGURATION FILE

## Functionality

Add model specific configuration file to be requested during the provisioning process.

For example: When GXP2170 request the provisioning files it needS to also request the following file: cfggxp2170.xml.

```
548 2016/127 17:10:57.542685000 172.18.21.5 54.169.67.73 HTTP 237 GET /gs/cfggxp2170.xml HTTP/1.0

Frame 2548: 237 bytes on wire (1896 bits), 237 bytes captured (1896 bits) on interface 0

Ethernet II, Src: Grandstr_87:a3:b3 (00:0b:82:87:a3:b3), Dest: All-HSRP-routers_02 (00:00:0c:07:ac:02)

Internet Protocol Version 4, Src: 172.18.21.5 (172.18.21.5), Dest: 54.169.67.73 (54.169.67.73)

Transmission Control Protocol, Src Port: 52605 (52605), Dest Port: 80 (80), Seq: 1, Ack: 1, Len: 171

Hypertext Transfer Protocol

GET /gs/cfggxp2170.xml HTTP/1.0\r\n
User-Agent: Grandstream Model Hw GXP2170 Sw 1.0.7.25 Devid 000b8287a3b3\r\n
Accept: */*\r\n
HOSt: fm.grandstream.com\r\n
Connection: Keep-Alive\r\n
\r\n
[Full request URI: http://fm.grandstream.com/gs/cfggxp2170.xml]
[HTTP request URI: http://fm.grandstream.com/gs/cfggxp2170.xml]
[Response in frame: 2555]
```

## **HIDE VOCODER**

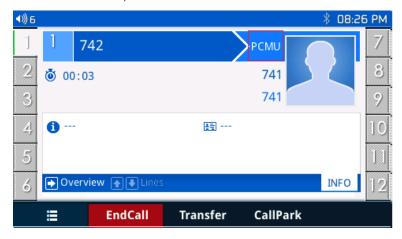
## Web Configuration

User can find the new item at Web configuration Accounts → Account Number → Audio Settings. A new option "Hide Vocoder" is added.

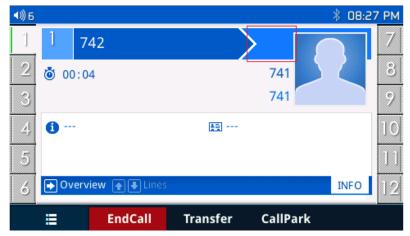




To check the functionality, we establish a phone call from DUT and check the call screen. When option Hide Vocoder is set as No, user can check the codec on the call screen as bellow.



When option Hide Vocoder is set as Yes, the coded will be hidden from call screen as bellow.





#### New P Values

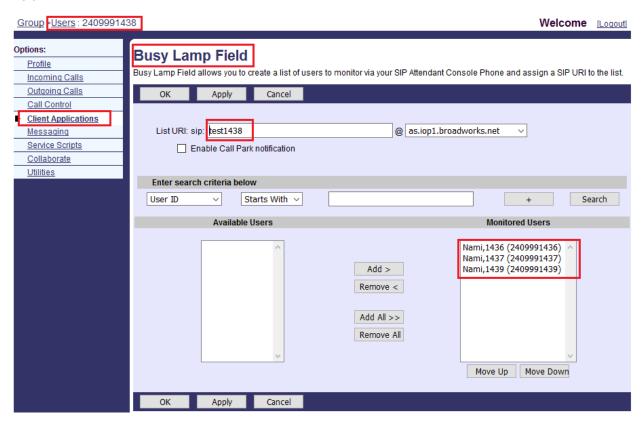
P value	Description	Value range	Default
P26047	<new pvalue=""></new>	Number, 0 or 1	0
P26147	Pvalue to set Hide Vocoder for account	No:0	
P26247	1-6	Yes:1	
P26347			
P26447			
P26547			

#### **CALL PICKUP BARGE-IN OPTION**

# • Web configuration

We take Broadsoft service server configuration for example:

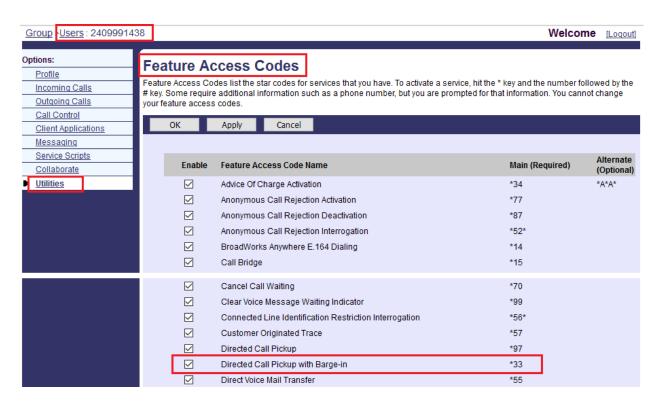
Enable BLF Eventlist function: Configuration Path: Group → Users → Client Applications → Busy Lamp Field



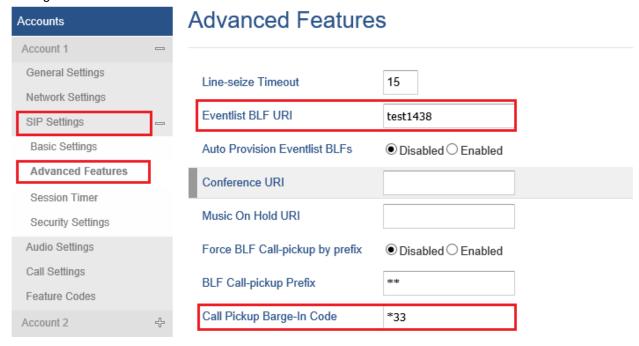
The information List URI sip and Monitored Users field will be configured in Grandstream Phone.

Check Call Pickup with Barge-in code: Configuration Path: Group →User → Utilities → Feature Access Codes



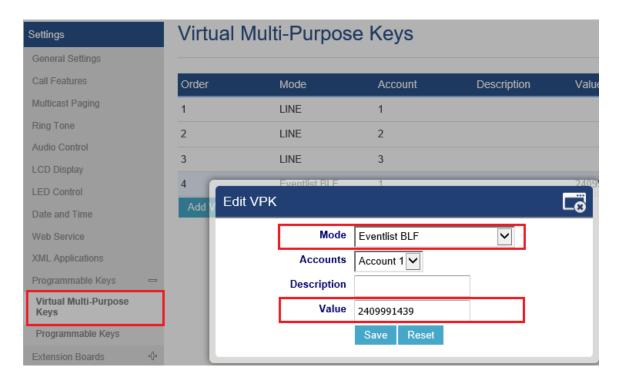


Based on Broadsoft Service server, the default Feature Access Code for Directed Call Pickup With Bargein is \*33. This information will be configured in Grandstream Web Setting  $\rightarrow$  Account #  $\rightarrow$ SIP Settings  $\rightarrow$  Advanced Features



Configure Eventlist BLF for one of monitored extension in Visual MPK





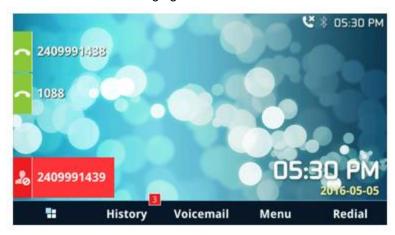
This new feature will provide the following functionality:

Allow users to dial a feature access code (FAC) followed by an extension number to pick up or barge-in a call to the monitored extension.

If the monitored extension is ringing state (LED indicator is flashing red), a user can pick up that call before the call is answered by monitored side.

If the monitored extension is in talk (LED indicator is solid red), a user can barge-in that existing call. That is, monitored side will release the call, the barge-in user become the controller of this call. When a barge-in occurs and the user's warning tone is enabled on server side, a user would hear barge-in warning tone.

Monitored Extension 2409991429 is Call Ringing State



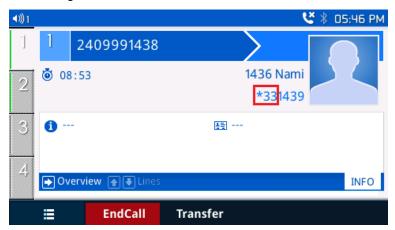


Monitored extension 2409991429 is busy, which is on talk state



In the above two states, User can use Call Pickup with Barge-in by pressing the key, which is configured as Eventlist BLF function key.

When the phone performs Call Pickup with Barge-in, the call screen would be likes as the follow image and the user might hear the warning tone if the function is enabled.



# New P Values

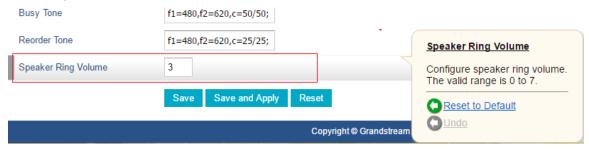
Pvalue	Description	Value range	Default
P26046	Call Pick up with Barge-In Code	String	NONE
P26146			
P26246			
P26346			
P26446			
P26546			



#### SPEAKERPHONE RX GAIN CONTROL

## • Web Configuration

User can find the new item at Web configuration Settings → Preferences--> Ring Tones Speaker, Ring Volume option is added.



## Functionality

User can configure speaker ring volume. The valid range is 0 to 7. The speaker ring volume will change after applying the changes.

## New P Values

P value	Description	Value range	Default
P8352	<new p="" value=""></new>	0 to 7.	5
	P value to set speaker ring volume		

## STATUS DETAIL DISPLAY

## Functionality

User can see more accurate display on LCD when Ethernet is not connected, account is not registered or configured.

## 1) Network Down:





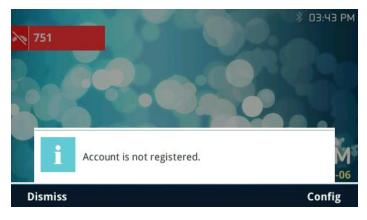
# 2) No Cable:



# 3) Internet Down:



# 4) Account is not registered:



Detailed warning window display condition can be summarized as bellow.

## No Bluetooth Handsfree connected:

--Network Down or Cable not connected

Loading popup window will be displayed indicates "Network Down" or "No LAN Cable". Config Softkey: Visible when Keypad Menu mode is not "basic settings only".



-- Internet Down and no account registered (Incorrect configuration of DNS will lead to Internet Down)

Loading popup window will be displayed indicates "Internet Down".

Config Softkey: Visible when Keypad Menu mode is not "basic settings only".

DirectIP Softkey: Visible when direct IP call is not disabled.

Network Softkey: Always visible to check network status.

-- Internet Down and has account registered

Popup window will be closed.

## With Bluetooth Handsfree connected:

No network related warning window will be displayed.

For Hard key handling during network warning window shown:

- -- Menu key: User can access menu module.
- -- Phonebook key: User can access Phonebook module.
- -- Other keys are blocked.

#### PARTIAL MATCH IN BRAODSOFT XSI SEARCH

## Functionality

Added supports for searching partial match for the phonebook entry by adding STAR code in Broadsoft XSI search.

Type a string contained for a search field, for example "TE" in First Name field of Broadsoft XSI search. Phone will send the following:

1355@as.iop1.broadworks.net/directories/group?lastName=&firstName=\*TE\*&number=

This will help to displayed all the entry with first name including "TE".

## FILTERING OUT INVALID SYMBOLS

### Functionality

When dialing contact number either from local Phonebook or LDAP book which contains some characters like '(' or ')' or spaces and dashes, the call will be established and these characters will be removed before sending digits.

### For example:

When creating phonebook or use LDAP contact with a number in the following form "(234) 567-8901". Make a call from phonebook or LDAP book contact, 2345678901 will be sent out.



#### SUPPORT FOR IEEE-48-ADDR AND IEEE-EUI-64 IN PEI HEADER

## Functionality

Add support of IEEE-48-addr and IEEE-EUI-64 in SIP header for emergency calls. Phone will populate the IEEE-48-addr, IEEE-EUI-64 (Example: IEEE-EUI-48; eui-48-addr=00-02-3F-1F-B2-42) in the PEI (Pemergency info) SIP header.

From Wireshark, we could find this info in PEI Invite SIP header

#### SUPPORT FOR POPULATING THE WAN ACCESS NODE IN SIP

## Functionality

Phone will include the MAC of Access Point in P-Access-Network-Info header. From Wireshark, we could find this info in P-Access-Network-Info header.

```
    □ P-Preferred-Identity: <sip:3003@192.168.92.200>
    □ SIP PPI Address: sip:3003@192.168.92.200
    SIP PPI User Part: 3003
    SIP PPI Host Part: 192.168.92.200
    P-Access-Network-Info: IEEE-EUI-48; eui-48-addr=9C-37-F4-84-05-56
```

## **DIAL FROM LDAP RESULTS**

## Functionality

Add the support let the user could dial the selected number from the LDAP results.

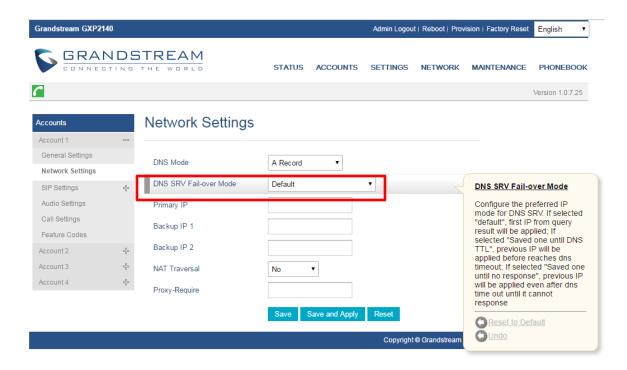
Configure the phone with correct LDAP info, then fill Name attribute with a value, such as CallerID Name, and Number attribute with Account Number, Mobile Number, Home Number etc. Make an LDAP search on the phone. Select an Entry, then press Dial Soft Key. Phone will show a screen to let user select the number to call from LDAP entry attributes.

## **DNS SRV FAIL-OVER MODE**

## Web Configuration

User can find the new option at Web configuration ACCOUNTS ->Account 1(1-6) ->Network Settings





The option will decide which IP is going to be used in sending SIP packets after IPs for SIP server host are resolved with DNS SRV. When a phone sends register sip requests, the phone will resolve the configured SIP server via DNS SRV. Normally, this will result in multiple IP address.

Regardless of which option set, it will try to send register messages to one IP at a time for each IP until it receives response. Then the phone is going to be registered with the server that received responses. However, its renewal registration will be sent to IP based on the configured option.

If the option is set with "default", it will again try to send register messages to one IP at a time, and the process repeats.

If the option is set with "Saved one until DNS TTL", it will send register messages to the previously registered IP first. If no response, it will try to send one at a time for each IP. This behavior lasts as long as DNS TTL (time-to-live) is up.

If the option is set with "Saved one until no responses", it will send register messages to the previously registered IP first, but this behavior will persist until the registered server does not respond.

## New P Values

P value	Description	Value range	Default
P26040	Account 1: P value to DNS SRV Fail-over Mode	0: Default	0



		1: Saved one until DNS TTL	
		2: Saved one until no response	
P26140	Account 2: P value to DNS	0: Default	0
	SRV Fail-over Mode	1: Saved one until DNS TTL	
		2: Saved one until no response	
P26240	Account 3: P value to DNS	0: Default	0
	SRV Fail-over Mode	1: Saved one until DNS TTL	
		2: Saved one until no response	
P26340	Account 4: P value to DNS	0: Default	0
	SRV Fail-over Mode	1: Saved one until DNS TTL	
		2: Saved one until no response	
P26440	Account 5: P value to DNS	0: Default	0
	SRV Fail-over Mode	1: Saved one until DNS TTL	
		2: Saved one until no response	
P26540	Account 6: P value to DNS	0: Default	0
	SRV Fail-over Mode	1: Saved one until DNS TTL	
		2: Saved one until no response	

# SEPARATE SUBSCRIPTION EXPIRATION TIMER

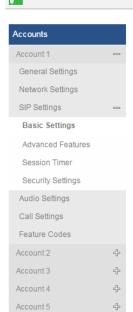
# • Web Configuration

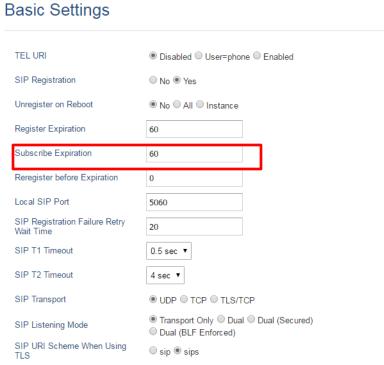
User can find the new option at Web configuration ACCOUNTS  $\rightarrow$  Account 1(1-6)  $\rightarrow$  SIP Settings  $\rightarrow$  Basic Settings  $\rightarrow$  Subscribe Expiration





STATUS ACCOUNTS SETTINGS NETWORK MAINTENANCE





# Functionality

Account 6

New design separate expiration timer between registration and subscription. This will reduce unnecessary network traffic.

## New P Values

P value	Description	Value range	Default
P26051	Account 1: P value to set	0-64800(minutes)	60
	Subscribe Expiration		
P26151	Account 2: P value to set	0-64800(minutes)	60
	Subscribe Expiration		
P26251	Account 3: P value to set	0-64800(minutes)	60
	Subscribe Expiration		
P26351	Account 4: P value to set	0-64800(minutes)	60
	Subscribe Expiration		
P26451	Account 5: P value to set	0-64800(minutes)	60
	Subscribe Expiration		
P26551	Account 6: P value to set	0-64800(minutes)	60
	Subscribe Expiration		



#### LCD POPUP WINDOW

## • LCD info popup window

In addition to IP Setting and IP Address, more information including MAC address and Software Version were added into info popup window.



# Functionality

Same as original design, user can press "UP" key to show Info popup window and press Dismiss to close the window. A new softkey "More" is added to bring user to Status menu page directly.

## **DIAL PLAN {** x+ | \+x+ | \*x+ | \*xx\*x+ **}**

The original default dial plan value was  $\{x+ \mid *x+ \mid *xx*x+ \}$ , even there was no "\+x+" included, the phone still allowed dialing any number with a leading +. We fixed this issue on this version, and added rule "\+x+" to represent "dialing any number with a leading +". The new default dial plan value is changed to  $\{x+ \mid \+x+ \mid *xx*x+ \}$  accordingly, the explanation for this default dial plan are as follows:

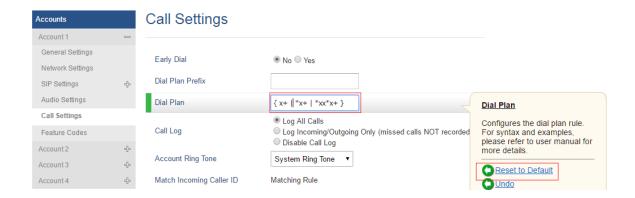
Allow any number of digits, OR any number with a leading +, OR any number with a leading \*, OR any number with a leading \* followed by a 2 digits number and a \*.

## Note:

In this version, user need to reset to new default dial plan value or add "+x+" to dial plan to allow "dialing any number with a leading +", to dial + from keypad, press on 0 until + appears on LCD.

Reset to default on Web → Accounts → Account Number → Call Settings:





# Changed P Values

P Value	Description	Value range	Default
P290	Account 1: Pvalue to set Dial	String	{ x+   \+x+   *x+
P459	Plan		*xx*x+ }
P559			
P659			
P1759			
P1859			



# FIRMWARE VERSION 1.0.7.15

### PRODUCT NAME

GXP2130, GXP2140, GXP2160, GXP2170 and GXP2135

## **DATE**

03/12/2016

## IMPORTANT UPGRADING NOTE

- For GXP2130/2140/2160, once upgraded to 1.0.7.15, downgrading to 1.0.5.x firmware version or lower is not supported.
- For GXP2170/2135, once upgraded to 1.0.7.15, downgrading to 1.0.6.9 firmware version or lower is not supported.

### **ENHANCEMENT**

- Added support for No Touch Provisioning to prompt for username and password for XML config file download for Broadsoft server [NO TOUCH PROVISIONING]
- Changed the default provisioning protocol to HTTPS. This option "Upgrade via" is under phone's web UI->Maintenance->Upgrade and provisioning.
- Added support for outbound notification [OUTBOUND NOTIFICATION SUPPORT]
- Added support for Virtual Multi-Purpose Keys [VIRTUAL MULTI-PURPOSE KEYS SUPPORT]
- Added support to show programmable keys status on web UI [PROGRAMMABLE KEYS STATUS ON WEB UI]
- Added option "Auto Provision List Starting Point" on web UI [AUTO PROVISION LIST STARTING POINT]
- Added additional ability to customize DHCP option for provisioning server [ADDITIONAL OVERRIDE DHCP OPTION]
- Added support for iLBC and G723 [ILBC AND G.723.1 SUPPORT]
- Added options for G723 rate, iLBC frame size and payload type [ILBC AND G.723.1 SUPPORT]
- Added option to enable and disable session timer [ENABLE SESSION TIMER]
- Added option to ring speaker for call waiting [RING SPEAKER FOR CALL WAITING]
- Added configurable backlight timer [CONFIGURABLE BACKLIGHT TIMER]
- Added color background wallpaper selection [COLOR BACKGROUND WALLPAPER SELECTION]
- Added BLF LED Pattern Explanation Form on web UI [BLF LED PATTERN EXPLANATION IN WEB UI]
- Disable screen saver when VPK is active [DISABLE SCREENSAVER WHEN VPK IS ACTIVE]
- Added fully black support for the idle screen LCD brightness (i.e., allow idle brightness to be 0) [FULL BLACK SUPPORT FOR IDLE LCD BRIGHTNESS]



- Add Blind and Attended Transfer softkey options [BLIND TRANSFER AND ATTENDED TRANSFER SOFTKEY]
- Add ability to display SIP MESSAGE text on LCD [DISPLAY SIP MESSAGE TEXT ON LCD]

## **BUG FIX**

- Fixed Line keys' LED doesn't provide visual indications of Shared call appearances
- Fixed GXP2170 Call Center Guest Login box does not display the cursor
- Fixed VPK does not change status if manually configure Eventlist BLF
- Fixed Line LED doesn't light up in transferring page
- Fixed Line LED turns off after pressing Home softkey in NO RESPONSE call page
- Fixed Line LED turns off after switching to home screen then entering other screen during the call
- Fixed GXP2130/40/60's some VPK modes doesn't work or display in LCD
- Fixed LCD does not respond after off-hook at the broken network state
- Fixed Preferred vocoder could be reset after device factory reset
- Fixed Phone cannot auto answer by the call-info provided in certain scenarios
- Fixed "Alert-Info" for ring tone is ignored
- Fixed Phone doesn't update the CID when preforming auto-attended transfer
- Fixed Phone keeps sending "SUBSCRIBE" after receiving "NOTIFY" with reason=noresource
- Fixed Dialing screen display mobile number instead of home number
- Fixed Transferred call logs do not display the name of the transferred number
- Fixed Device works abnormal after pressing EHS headset key at the on-hook dialing screen
- Fixed Bluetooth hands-free line flashes in red when there is no incoming call on the Bluetooth handsfree line
- Fixed Bluetooth hands-free call screen displays "PrivateHold" softkey which should be invalid
- Fixed Device uses Bluetooth hands-free line to call voicemail user ID
- Fixed Plantronics EHS compatibility issue
- Fixed "Confcall" softkey in Bluetooth hands-free line and SIP calls doesn't work
- Fixed Display language of softkeys on "Keypad Locking Password" box does not update
- Fixed Time zone string is incorrect for New Foundland, Canada
- Fixed Instant message is displayed abnormally
- Fixed Audio is cut off in the beginning of the outgoing RTP stream when receiving an incoming call
- Fixed Phone failed to retrieve the parked call through the call park feature
- Fixed Phone lost network access when enabling LLDP/LLDP-med
- Fixed Phone is not recognized by some TP-LINK switch using LLDP
- Fixed "Confirm" softkey is displayed abnormally when deleting phonebook entry through dialing page
- Fixed LCD does not jump to dialing screen after pressing "NewCall softkey" in the conference
- Fixed Broadsoft IM&P feature that the phone uses wrong DNS query type
- Fixed Extra route header on ACK request
- Fixed Adding "Save confirmation" message when saving settings on phone GUI
- Changed default dial plan to { x+ | \+x+ | \*x+ | \*xx\*x+ }



## **NEW FEATURES OVERVIEW**

This section lists major new features and describes how to use it from the user's point of view.

#### NO TOUCH PROVISIONING

After the phone sends config file request to the Broadsoft provisioning server via HTTP/HTTPS, if the provisioning server responds 401 Unauthorized asking for authentication, the phone's LCD will prompt a window for user to enter username and password. Once correct username and password are entered, the phone will send config file request again with authentication. Then the phone will receive the config file to download and get provisioned automatically.

Besides manually entering the username and password in LCD prompt, users can save the login credentials for provisioning process as well. The username and password configuration is under phone's web UI->Maintenance->Upgrade and provisioning page: "HTTP/HTTPS Username" and "HTTP/HTTPS Password". If the saved username and password saved are correct, login window will be skipped. Otherwise, login window will be popped up to prompt users to enter correct username and password again.

#### **OUTBOUND NOTIFICATION SUPPORT**

Outbound notification options can be found under device web UI->Settings->Outbound Notifications. In the web UI, there are three sections under Outbound Notifications: "Action URL", "Destination" and "Notification".

### Action URL

To use Outbound Notification->Action URL, users need to know the supported events and the dynamic variables for the supported events. The dynamic variables for the supported events will be replaced by the actual values on the phone in order to notify the event to SIP server.



Settings		Action URL	
General Settings			
Call Features		Setup Completed	
Multicast Paging		Setup Completed	
Ring Tone		Registered	
Audio Control		Unregistered	
LCD Display		Off Hook	
LED Control		OII HOOK	
Date and Time		On Hook	
Web Service		Incoming Call	
XML Applications		0.4	
Programmable Keys	4	Outgoing Call	
Extension Boards	4	Missed Call	
Broadsoft	4	Established Call	
Outbound Notification		T	
Action URL		Terminated Call	
Destination		Open DND	
Notification		Close DND	
		0 5 1	
		Open Forward	
		Close Forward	
		Blind Transfer	
		Attended Transfer	
		Hold Call	
		UnHold Call	
			Save Save and Apply Reset

Supported Events:
Setup Completed
Registered
Unregistered
Off Hook
On Hook
Incoming Call
Outgoing Call
Missed Call
Established Call
Terminated Call
Open DND
Close DND



Open Forward	
Close Forward	
Blind Transfer	
Attended Transfer	
Hold Call	
UnHold Call	

Supported Dynamic Variables		
Dynamic Variable	Description	
\$phone_ip	The IP address of the phone	
\$mac	The MAC address of the phone	
\$product	The product name of the phone	
\$program_version	The software version of the phone	
\$hardware_version	The hardware version of the phone	
\$language	The display language of the phone	
\$local	The called number on the phone	
\$display_local	The display name of the called number on the phone	
\$remote	The call number on the remote phone	
\$display_remote	The display name of the call number on the remote phone	
\$active_user	The account number during a call on the phone	

After the user finishes setting Action URL on phone's web UI, when the specific phone event occurs on the phone, phone will send the Action URL to the specified SIP server. The dynmic variables in the Action URL will be replaced by the actual values.

### Here is an example:

Configure the following Action URL on the phone's web UI->Settings->Outbound Notification->Actial URL:

Incoming Call: 172.18.24.103/mac=\$mac&local=\$local

Outgoing Call: 172.18.24.103/remote=\$remote&phone\_ip=\$phone\_ip
On hold: 172.18.24.103/program\_version=\$program\_version

During incoming call, outgoing call and call hold, capture the trace on the phone and exam the packets. We can see the phone send Action URL with actual values to SIP server to notify phone events. In the following screenshot, from top to bottom, the phone events for each HTTP message are: Outgoing Call, Incoming Call and On Hold in the format of the defined action URL with the parameters replaced with actual values.



```
Source
                          Destination
                                              Protocol
                                                            Length Info
                                                               150 GET /mac=00:0B:82:67:0D:6E&local=2071 HTTP/1.1
000
      172.18.23.173
                          172.18.24.103
                                              HTTP
                                                               152 GET /remote=2071&phone_ip=172.18.23.173 HTTP/1.1
                           172.18.24.103
      172.18.23.173
                                              HTTP
                                                               144 GET /program_version=0.10.5.111 HTTP/1.1
      172, 18, 23, 173
                          172.18.24.103
                                              HTTP
```

```
B Frame 457: 150 bytes on wire (1200 bits), 150 bytes captured (1200 bits) on interface 0
⊕ Ethernet II, Src: Grandstr_67:0d:6e (00:0b:82:67:0d:6e), Dst: Grandstr_64:e3:12 (00:0b:82:64:e3:12)
⊞ Internet Protocol Version 4, Src: 172.18.23.173 (172.18.23.173), Dst: 172.18.24.103 (172.18.24.103)
⊞ Transmission Control Protocol, Src Port: 50668 (50668), Dst Port: 80 (80), Seq: 1, Ack: 1, Len: 84

    ∃ Hypertext Transfer Protocol

  GET /mac=00:0B:82:67:0D:6E&local=2071 HTTP/1.1\r\n
    □ [Expert Info (Chat/Sequence): GET /mac=00:0B:82:67:0D:6E&local=2071 HTTP/1.1\r\n]
        [GET /mac=00:0B:82:67:0D:6E\&local=2071 HTTP/1.1\r\n]
        [Severity level: Chat]
        [Group: Sequence]
     Request Method: GET
     Request URI: /mac=00:0B:82:67:0D:6E&local=2071
     Request Version: HTTP/1.1
    Host: 172.18.24.103\r\n
    Accept: */*\r\n
    \r\n
    [Full request URI: http://172.18.24.103/mac=00:0B:82:67:0D:6E&local=2071]
    [HTTP request 1/1]
    [Response in frame: 462]
```

The P values listed in below table are for the options under phone web UI->Settings->Outbound Notification->Action URL.

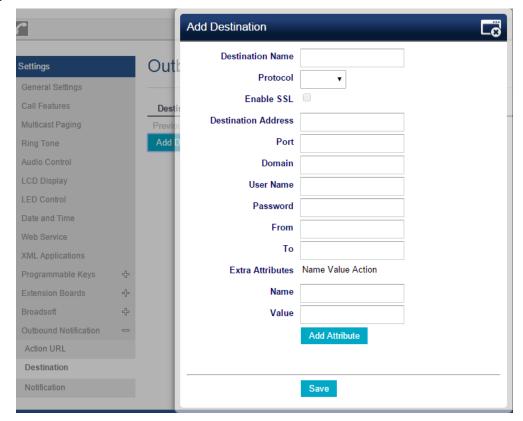
P Value	Web UI Option	Value Format
P8304	Setup Completed	
P8305	Registered	
P8306	Unregistered	
P8308	Off Hook	
P8309	On Hook	
P8310	Incoming Call	
P8311	Outgoing Call	
P8312	Missed Call	
P8313	Established Call	String
P8314	Terminated Call	Stillig
P8316	Open DND	
P8317	Close DND	
P8318	Open Forward	
P8319	Close Forward	
P8320	Blind Transfer	
P8321	Attended Transfer	
P8324	Hold Call	
P8325	UnHold Call	

## Destination

The options under phone's web UI->Settings->Outbound Notification->Destination configures outbound



notification destination server information. Click on "Add Destination" and users will see following window to configure destination server information.



The following table describes each option in the above interface.

<b>Destination Server Option</b>	Description
Destination Name	Identify the destination name. It must be unique.
Protocol	Configure the protocol associated with the destination server. Currently
	XMPP and SMTP are supported.
Enable SSL	Configure whether to use SSL to encrypt for SMTP protocol. This option
	is not editable for XMPP.
Destination Address	Configure destination server address, e.g., talk.google.com.
Port	Configure destination server port, e.g., 5222.
Domain	Configure the destination server domain for XMPP protocol. This option is
	not editable for SMTP.
User Name	Configure the authorization user name of the destination server.
Password	Configure the authorization user password for the destination server.
From	Configure the sender name for SMTP protocol. This option is not editable
	for XMPP.
То	Configure the receiver's address.
Extra Attribute Name	Configure extra attribute's name reserved for protocol specific attributes
	such as "jid" for XMPP protocol. If "jid" is specified, user name and domain



	will be overridden.
Extra Attribute Value	Configure extra attribute's value reserved for protocol specific attributes
	such as "abc@gmail.com" for "jid" of XMPP protocol. If it's specified, user
	name and domain will be overridden.

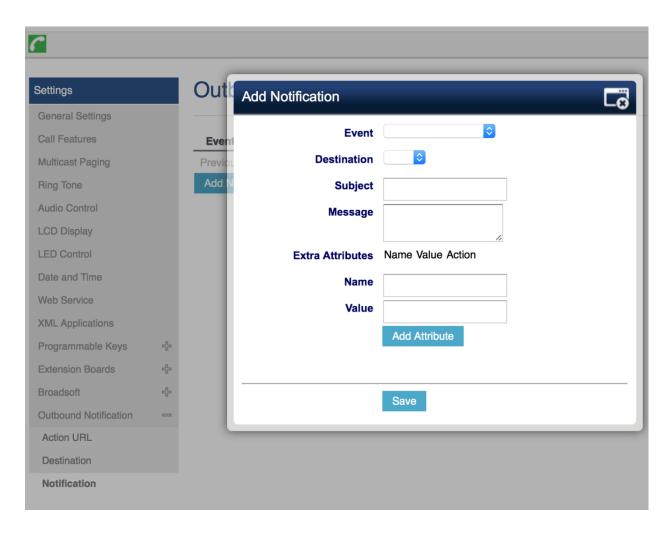
Up to 10 destinations can be configured here. The P values are listed in below table.

P Value	Destination	Value Format
P9910	Destination 1	String. Each P value consists of all the options configured for this destination.
P9911	Destination 2	Example 1 - Destination 1 with protocol XMPP and 2 extra Attributes
P9912	Destination 3	configured: P9910=serverName= <i>destination1</i> &protocol= <i>XMPP</i> &serverAddress= <i>talk.go</i>
P9913	Destination 4	ogle.com&port=5222&user=username1&password=password1&from=&to=t
P9914	Destination 5	o1&domain=gmail.com&extraAttrName1=extraAttrValue1&extraAttrName2= extraAttrValue2
P9915	Destination 6	Example 2 - Destination 2 with protocol SMTP and 3 extra Attributes
P9916	Destination 7	configured: P9911=serverName=destination2&protocol=SMTP&serverAddress=smtps://
P9917	Destination 8	smtp.gmail.com&port=465&user=username2&password=password2&from= username2&to=to2&domain=&extraAttrName1=extraAttrValue1&extraAttrN
P9918	Destination 9	ame2=extraAttrValue2&extraAttrName3=extraAttrValue3
P9919	Destination 10	The <i>highlighted strings</i> in above examples are the actual values configured in each field for the destination.

## Notification

After configuring detination server, users can configure notification information under phone's web UI->Settings->Outbound Notification->Notification. Click on "Add Notification" and users will see following window to configure notification.





Notification Option	Description
Event	Configure the event which will trigger an outbound notification.
Destination	Configure the name of the destination where the outbound notification will
	be sent to.
Subject	Configure the subject of Email notification. This option is only applicable
	to SMTP protocol and it's not editable for other protocols.
Message	Configure the message body or the outbound notification.
Extra Attribute Name	Configure extra attribute's name reserved for specific attributes for a given
	notification in the future.
Extra Attribute Value	Configure extra attribute's value reserved for specific attributes for a given
	notification in the future.

The message body of the notification for each event can be customized with dynamic attributes embedded. The following table shows the mapping between event and dynamic attribute.



Line number associated with the call account   Account number associated with the call remoteNumber   The remote party number   The remote party number   The remote party number   IocalNumber   The local party number   IocalNumber   The SIP dialog call ID   The SIP dialog call ID   The time stamp when the missed call event occurs   This is for DND status. The value can be "enabled" or "disabled"   This is for DND status. The value can be "enabled" or "disabled"   This is for the type of the call. The value can be "incoming" or "outgoing"   Iine   Line number associated with the call   account   Account number associated with the call   account   Account number associated with the call   TemoteNumber   The remote party number   The remote party number   IocalNumber   The local party number   IocalNumber   The siP dialog call ID   The SIP server address of the account   The SIP dialog call ID   Time   The timestamp when the call is forwarded   fwNumber   Call Forward destination number   fwReason   Call Forward destination number   fwReason   Call Forward destination number   Incomplete   Call Forward destination number   The didness of the PC who will log in phone's web UI or   SSH   OAMLoginFormIP   OAM login From IP address. The value can be "SSH" or "WebGUI"   OAMLoginCode   OAM login result code. The value can be "succeeded" or "failed"   Time   OAMLogin From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH   OAMLoginFromIP   OAMLoginFrom IP address. The value is the IP address o	Event	Dynamic Attribute Name	Dynamic Attribute Description	
remoteNumber The remote party number remoteName The remote party name localNumber The local party number The SIP server address of the account Call-id The SIP dialog call ID The time Status This is for DND status. The value can be "enabled" or "disabled"  Call_Forward CallType This is for the type of the call. The value can be "incoming" or "outgoing" line Line number associated with the call account Account number associated with the call remoteNumber The remote party number remoteNumber The remote party number localNumber The Iocal party number IocalNumber The Iocal party number SipServer The SIP server address of the account The SIP dialog call ID time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason  OAM_Login  OAM_Login OAMUser OAM login reme such as "admin" OAMLoginFromIP OAM login From IP address. The value can be "SSH" or "WebGUI"  OAMLoginCode OAM login result code. The value can be "succeeded" or "falled" time OAM login result code. The value can be "succeeded" or "falled" time OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or SSH OAMLoginCode OAM login result code. The value can be "succeeded" or "falled" time OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or "SUCCEEDED" OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI" OAMLoginFoomIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI" OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI" OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"	Call_Missed	line	Line number associated with the call	
remoteName   The remote party name     localNumber   The local party number     localName   The local party number     localName   The socal party number     sipServer   The SIP server address of the account     time   The time stamp when the missed call event occurs     This is for DND status. The value can be "enabled"     Call_Forward   CallType   This is for DND status. The value can be "enabled"     Call_Forward   CallType   This is for the type of the call. The value can be "incoming" or "outgoing"     line   Line number associated with the call     account   Account number associated with the call     remoteNumber   The remote party number     remoteName   The remote party number     remoteName   The local party number     localName   The local party number     localName   The local party number     localName   The sIP server address of the account     call-id   The SIP dialog call ID     time   The timestamp when the call is forwarded     fwNumber   Call Forward destination number     fwReason   Call Forward reason     OAM Login Source   OAM user name such as "admin"     OAMLoginSource   OAM login source. The value can be "SSH" or "WebGUI"     OAMLoginCode   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH     OAMLoginCode   OAM login time stamp     OAMLoginSource   OAM login result code. The value can be "succeeded" or "failed"     time   OAM login time stamp     OAMLoginSource   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH     OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH     OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH     OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH		account	Account number associated with the call	
localNumber		remoteNumber	The remote party number	
localName   The local party name   sipServer   The SIP server address of the account   call-id   The SIP dialog call ID   time   The time stamp when the missed call event occurs   This is for DND status. The value can be "enabled" or "disabled"   Call_Forward   call_Type   This is for the type of the call. The value can be "incoming" or "outgoing"   line   Line number associated with the call   account   Account number associated with the call   remoteNumber   The remote party number   remoteName   The remote party number   localName   The local party number   localName   The local party number   localName   The Iosal party number   localName   The SIP server address of the account   call-id   The SIP dialog call ID   time   The timestamp when the call is forwarded   fwNumber   Call Forward destination number   fwReason   Call Forward reason   Call Forward reason   OAM_Login Source   OAM user name such as "admin"   OAMLoginSource   OAM login source. The value can be "SSH" or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH   OAMLoginCode   OAM login time stamp   OAM login time stamp   OAM login source. The value can be "SSH" or "WebGUI"   OAMLoginSource   OAM login source. The value can be "SSH" or "WebGUI"   OAMLoginSource   OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAMLoginFromIP   OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"   OAM l		remoteName	The remote party name	
SipServer   The SIP server address of the account		localNumber	The local party number	
Call-id   The SIP dialog call ID		localName	The local party name	
time The time stamp when the missed call event occurs  DND status This is for DND status. The value can be "enabled" or "disabled"  Call_Forward callType This is for the type of the call. The value can be "incoming" or "outgoing"  line Line number associated with the call account Account number associated with the call remoteNumber The remote party number remoteName The remote party number localNumber The local party number localName The local party number incalName The SIP server address of the account call-id The SIP dialog call ID time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward destination number fwReason OAM user name such as "admin"  OAM_Login  OAM_Login  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUl"  OAMLoginFomIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginSource OAM login result code. The value can be "succeeded" or "failed"  time OAM login source. The value can be "SSH" or "WebGUl"  OAM_Lockout OAM login result code. The value can be "succeeded" or "failed"  time OAM login source. The value can be "SSH" or "WebGUl"  OAM_Lockout OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUl"  OAM_Lockout OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUl"		sipServer	The SIP server address of the account	
DND status This is for DND status. The value can be "enabled" or "disabled"  Call—Forward CallType This is for the type of the call. The value can be "incoming" or "outgoing"  line Line number associated with the call account Account number associated with the call remoteNumber The remote party number remoteName The local party number localNumber The local party number  IocalNumber The local party number Intel local p		call-id	The SIP dialog call ID	
Call_Forward  CallType  This is for the type of the call. The value can be "incoming" or "outgoing"  line  Line number associated with the call account Account number associated with the call remoteNumber The remote party number remoteName IocalNumber Ine olocal party number IocalName SipServer The SIP server address of the account Call-id The SIP dialog call ID The timestamp when the call is forwarded fwNumber Gall Forward destination number fwReason Call Forward reason  OAM_Login  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginCode OAM login result code. The value can be "succeeded" or "failed" time OAMLogin FromIP OAM login from IP address. The value can be "SSH" or "WebGUI"  OAMLoginFormIP OAM login source. The value can be "SSH" or "WebGUI"  OAM login result code. The value can be "succeeded" or "failed"  time OAM login source. The value can be "SSH" or "WebGUI"  OAM login from IP address. The value can be "SSH" or "WebGUI"  OAM login from IP address. The value can be "SSH" or "WebGUI"  OAM login from IP address. The value can be "SSH" or "WebGUI"  OAM login from IP address. The value can be "SSH" or "WebGUI"  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"		time	The time stamp when the missed call event occurs	
Call_Forward    Call_Forward   CallType	DND	status	This is for DND status. The value can be "enabled"	
line   Line number associated with the call			or "disabled"	
line Line number associated with the call account Account number associated with the call remoteNumber The remote party number remoteName The remote party number The local party number IocalNumber The local party number IocalName The local party name IocalName The local party name SipServer The SIP server address of the account Call-id The SIP dialog call ID Time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason  OAM_Login  OAM_Login  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login result code. The value is the IP address of the PC who will log in phone's web UI or SSH  time OAM login time stamp  OAM_Lockout  OAM_Lockout  OAM_LoginSource OAM login source. The value can be "succeeded" or "failed"  time OAM login time stamp  OAM_Lockout  OAM_LoginSource OAM login source. The value can be "SSH" or "webGUI"  OAM_LoginSource OAM login source. The value can be "succeeded" or "failed"  time OAM login from IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"	Call_Forward	callType	This is for the type of the call. The value can be	
account Account number associated with the call remoteNumber The remote party number remoteName The remote party name localNumber The local party number localName The local party name sipServer The SIP server address of the account call-id The SIP dialog call ID time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason  OAM_Login  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAM_Lockout OAM login source. The value can be "succeeded" or "failed"  time OAM login time stamp  OAM_Lockout OAM login source. The value can be "SSH" or "WebGUI"  OAM login source. The value can be "succeeded" or "failed"  time OAM login time stamp  OAM login source. The value can be "SSH" or "WebGUI"  OAM login source. The value can be "SSH" or "WebGUI"  OAM login source. The value can be "SSH" or "WebGUI"  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or			"incoming" or "outgoing"	
remoteNumber The remote party number remoteName The remote party name localNumber The local party number localName The local party name sipServer The SIP server address of the account call-id The SIP dialog call ID time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason  OAM_Login OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAM_Lockout OAM login source. The value can be "succeeded" or "failed" time OAM login time stamp  OAM_Lockout OAM login source. The value can be "SSH" or "WebGUI"  OAM login from IP address. The value can be "succeeded" or "failed"  OAM_Lockout OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"		line	Line number associated with the call	
remoteName localNumber localName The local party name The SIP server address of the account The SIP dialog call ID The time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason OAM_Login  OAM_Login OAM_Login Source OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAMLogin From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAM_Lockout OAM_Lockout OAM_LoginFromIP OAM_Lockout OAM_LoginFromIP OAM_Login From IP address. The value can be "succeeded" or "failed"  time OAM_Login From IP address. The value can be "SSH" or "WebGUI"  OAM_LoginFromIP OAM_Login From IP address. The value is the IP address of the PC who will log in phone's web UI or "WebGUI"  OAM_LoginFromIP OAM_Login From IP address. The value is the IP address of the PC who will log in phone's web UI or		account	Account number associated with the call	
localNumber   The local party number		remoteNumber	The remote party number	
IocalName   The local party name		remoteName	The remote party name	
SipServer   The SIP server address of the account		localNumber	The local party number	
Call-id  time  The timestamp when the call is forwarded  fwNumber  Call Forward destination number  fwReason  Call Forward reason  OAM_Login  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAMLogin From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode  OAM login result code. The value can be "succeeded" or "failed"  time  OAM login time stamp  OAM_Lockout  OAM login source. The value can be "SSH" or "webGUI"  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		localName	The local party name	
time The timestamp when the call is forwarded fwNumber Call Forward destination number fwReason Call Forward reason  OAM_Login  OAM_Login  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode OAM login result code. The value can be "succeeded" or "failed"  time OAM login time stamp  OAM_Lockout OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		sipServer	The SIP server address of the account	
fwNumber         Call Forward destination number           fwReason         Call Forward reason           OAM_Login         OAM user name such as "admin"           OAMLoginSource         OAM login source. The value can be "SSH" or "WebGUI"           OAMLoginFromIP         OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH           OAMLoginCode         OAM login result code. The value can be "succeeded" or "failed"           time         OAM login time stamp           OAM_Lockout         OAM user name such as "admin"           OAMLoginSource         OAM login source. The value can be "SSH" or "WebGUI"           OAMLoginFromIP         OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		call-id	The SIP dialog call ID	
TwReason  OAM_Login  OAMUser  OAM user name such as "admin"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode  OAM login result code. The value can be "succeeded" or "failed"  time  OAM login time stamp  OAM_Lockout  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		time	The timestamp when the call is forwarded	
OAM_Login  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAMLoginFromIP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode  OAM login result code. The value can be "succeeded" or "failed"  time  OAM login time stamp  OAM_Lockout  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		fwNumber	Call Forward destination number	
OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode OAM login result code. The value can be "succeeded" or "failed"  time OAM login time stamp  OAM user name such as "admin"  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		fwReason	Call Forward reason	
OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH  OAMLoginCode OAM login result code. The value can be "succeeded" or "failed" time OAM login time stamp  OAM_Lockout OAM user name such as "admin" OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or	OAM_Login	OAMUser	OAM user name such as "admin"	
OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH OAMLoginCode OAM login result code. The value can be "succeeded" or "failed" time OAM login time stamp OAM_Lockout OAMUser OAM user name such as "admin" OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI" OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		OAMLoginSource	OAM login source. The value can be "SSH" of	
address of the PC who will log in phone's web UI or SSH  OAMLoginCode OAM login result code. The value can be "succeeded" or "failed" time OAM login time stamp  OAM_Lockout OAM user name such as "admin" OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or			"WebGUI"	
OAMLoginCode OAM login result code. The value can be "succeeded" or "failed" time OAM login time stamp OAM_Lockout OAMUser OAM user name such as "admin" OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI" OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		OAMLoginFromIP	OAM login From IP address. The value is the IP	
OAMLoginCode  OAM login result code. The value can be "succeeded" or "failed"  time  OAM login time stamp  OAM_Lockout  OAM user name such as "admin"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or			address of the PC who will log in phone's web UI or	
"succeeded" or "failed"  time OAM login time stamp  OAM_Lockout OAMUser OAM user name such as "admin"  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or			SSH	
time OAM login time stamp  OAM_Lockout OAM user name such as "admin"  OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		OAMLoginCode	OAM login result code. The value can be	
OAM_Lockout  OAMUser  OAM user name such as "admin"  OAMLoginSource  OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or			"succeeded" or "failed"	
OAMLoginSource OAM login source. The value can be "SSH" or "WebGUI"  OAMLoginFromIP OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or		time	OAM login time stamp	
"WebGUI"  OAMLoginFromIP  OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or	OAM_Lockout	OAMUser	OAM user name such as "admin"	
address of the PC who will log in phone's web UI or		OAMLoginSource		
address of the PC who will log in phone's web UI or		OAMLoginFromIP	OAM login From IP address. The value is the	
			1	
			SSH	



	OAMLockoutCode	OAM lockout result code. The value can be "locked"	
	OAMLOCKOULCODE	or "unlocked"	
	OAM a also stime a		
La cancia de Oall	OAMLockoutTime	OAM lockout time stamp	
Incoming_Call	callingNumber	Calling party number	
	callType	Type of the call. The value can be "incoming" or	
		"outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName	The local party name	
	sipServer	The SIP server address of the account	
	call-id	The SIP dialog call ID	
	time	The time stamp when the incoming call event occurs	
Outgoing_Call	callType	Type of the call. The value can be "incoming" or	
		"outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName	The local party name	
	sipServer	The SIP server address of the account	
	time	The time stamp when the outgoing call event occurs	
Call_Established	callType	Type of the call. The value can be "incoming" or "outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName		
		The local party name  The SIP server address of the account	
	sipServer		
	call-id	The SIP dialog call ID	
Call Taxasis 4 1	startTime	The time stamp when the outgoing call event occurs	
Call_Terminated	callType	Type of the call. The value can be "incoming" or "outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	



	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName	The local party name	
	sipServer	The SIP server address of the account	
	call-id	The SIP dialog call ID	
	startTime	The time stamp when the call is established	
Call_Forward_Status	duration	The call duration	
	account	The account number associated with the call forward	
		status change	
	forwardNumberAll	The forward number for Call Forward All	
	forwardNumberBusy	The forward number for Call Forward Busy	
	forwardNumberNoAns	The forward number for Call Forward No Answer	
Call Hold	callType	Type of the call. The value can be "incoming" or	
		"outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName	The local party name	
	sipServer	The SIP server address of the account	
	call-id	The SIP dialog call ID	
	startTime	The time stamp when the call is on hold	
Call_Resume	callType	Type of the call. The value can be "incoming" or "outgoing"	
	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	localName	The local party name	
	sipServer	The SIP server address of the account	
	call-id	The SIP dialog call ID	
	startTime	The time stamp when the call is resumed	
Blind_Transfer	line	Line number associated with the call	
	account	Account number associated with the call	
	remoteNumber	The remote party number	
	remoteName	The remote party name	
	localNumber	The local party number	
	•		



	localName	The local party name		
	sipServer The SIP server address of the account			
	call-id	The SIP dialog call ID		
	time	The time stamp when the call is transferred		
	transferName	Transferred party name		
	transferNumber	Transferred party number		
Attended_Transfer	Line	Line number associated with the call		
	account	Account number associated with the call		
	remoteNumber	The remote party number		
	remoteName	The remote party name		
	localNumber	The local party number		
	localName	The local party name		
	sipServer	The SIP server address of the account		
	call-id	The SIP dialog call ID		
	Time	The time stamp when the call is transferred		
	transferName	Transferred party name		
	transferNumber	Transferred party number		
Register_Status	registerStatus	Account register status. The value can be		
		"registered" or "unregistered"		
Bootup_Complete	N/A	N/A		
The dynamic	mac	MAC address of the phone		
attributes in this row	phone_ip	IP address of the phone		
are common	program_version	Software version of the phone		
attributes that can be	hardware_version	Hardware version of the phone		
applied to all events	product	Product name of the phone		
	language	Display language on the phone		

All above dynamic attributes' value is generated by phone system and can be used as dynamic attributes with a pair of curved braces around them. For example, if the message body is specified as following:

Your call from {remoteName}:{remoteNumber} to {localName}:{localNumber} was forwarded to {fwdNumber} by reason {fwdReason}.

Then the message received in the outbound notification will look like this:

Your call from Daniel:2070 to Jasmine:2071 was forwarded to 777777 by reason unconditional.

Only attributes in curved braces will be replaced by the run time value. Other content will remain the same as static text.

For each event, at most 3 notifications can be configured. In total, up to 75 notifications can be configured.



The P value for each notification is listed in below table.

P Value	Notification	Value Format
P9920	Notification 1	String. Each P value consists of all the options configured for this
P9921	Notification 2	notification.
P9922	Notification 3	Example 1 – Notification 1 for event "Call_Missed" to destination 1, with 2 extra Attributes configured:
P9923	Notification 4	P9920=eventName=Call_Missed&destName=destination1&subject=&msg= You have a missed call from {remoteName}:{remoteNumber} on Line {line},
P9924	Notification 5	account {account} at
P9925	Notification 6	{time}.&extraAttrName1=extraAttrValue1&extraAttrName2=extraValue2
P9926	Notification 7	Example 2 – Notification 2 for event "Incoming_Call" to destination 2, with 2 extra Attributes configured:
P9927	Notification 8	P9921= eventName=Incoming_Call&destName=destination2&subject=Incoming
P9928	Notification 9	Call Alert&msg=You have an {callType} call from
P9929	Notification 10	{remoteName}:{remoteNumber} on Line {line}, account {account} at
		{time}.&extraAttrName1=extraAttrValue1&extraAttrName2=extraAttrValue2
P9993	Notification 73	
P9994	Notification 74	The highlighted strings in above examples are the actual values configured
P9995	Notification 75	in each field for the notification.

## **VIRTUAL MULTI-PURPOSE KEYS SUPPORT**

# • Web UI Configuration

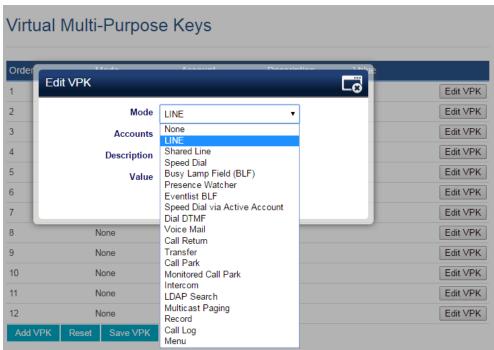
Users can find new Virtual Multi-Purpose Keys (VPK) configuration under phone's web UI->Settings->Programmable Keysp->Virtual Multi-Purpose Keys tab. It is recommended to select "Reset" on this page before configuring VPK here. By default, all fixed VPKs are listed.



# Virtual Multi-Purpose Keys

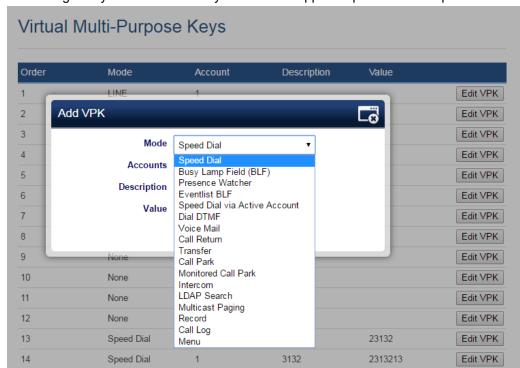
Order	Mode	Account	Description	Value	
1	LINE	1			Edit VPK
2	LINE	2			Edit VPK
3	LINE	3			Edit VPK
4	LINE	4			Edit VPK
5	LINE	5			Edit VPK
6	LINE	6			Edit VPK
7	None	1			Edit VPK
8	None	1			Edit VPK
9	None	1			Edit VPK
10	None	1			Edit VPK
11	None	1			Edit VPK
12	None	1			Edit VPK
Add VPK	Reset Save VPK				

Click on "Edit VPK" for the line (fixed VPK) you would like to configure. A new window will pop up for VPK configuration. Users can configure Mode, Account, Description and Value for the VPK. Up TO 20 mode options can be selected for the VPK. Once done, press "Save" on this window and press "Save VPK" on the bottom of the Virtual Multi-Purpose Keys page again to apply the change.



If users would like to configure more VPKs than the ones displayed on the page, the users can click on





"Add VPK" to configure dynamic VPK. The dynamic VPK supports up to 17 mode options.

#### Please note:

- 1. Dynamic VPK doesn't support LINE and Shared LINE mode. These two mode options are only available for fixed VPKs.
- 2. Dynamic VPK doesn't support NONE mode. If users do not need this VPK, click on "Edit VPK" for it and select "Delete" to remove this VPK.
- 3. All settings require user to click on "Save" on the prompted window and also "Save VPK" button on the bottom of Virtual Multi-Purpose Keys page to take effect.

#### • LCD Indication and Configuration

Save VPK

The configured fixed VPKs are displayed next to the corresponding line. If dynamic VPKs are configured, the users can see a page number shown on the upper left corner on the LCD.

The following figures show page 1 and page 2 of the VPKs on LCD. Pressing "RIGHT" arrow key or "Next" softkey will switch to the next page; pressing "LEFT" arrow key will switch back to the preivous page.





The users could also edit and add VPK from LCD.

- 1. To edit (fixed) VPK, press and hold the line key for about 4 seconds, a configuration window will pop up for the user to configure.
- 2. To add (dynamic) VPK, press and hold the RIGHT arrow key for about 4 seconds, a configuration window will pop up for the user to configure.

Up to 20 mode can be supported on fixed VPK and up to 17 mode can be supported on dynamic VPK. Each mode is indicated by a different icon on the LCD and the icon will be different when in different status. Please find the icon indications below for different mode of VPK.

VPK Mode	State	Icon	LED Status
LINE	Unregistered (No IM, Voice mail, No Call Forward)	*	OFF
	Registered + Idle (No IM, Voice mail, No Call Forward)	^	OFF
	Unregistered + IM + Voice mail	4	OFF
	Registered + IM + Voice mail	4	OFF
	Unregistered + IM (No Voice mail)	2	OFF
	Registered + IM (No Voice mail)	<b>2</b>	OFF
	Unregistered + Voice Mail (No IM)	<u> </u>	OFF



	Registered + Voice Mail (No IM)	<b>=</b>	OFF
	Unregistered + Call Forward All (No IM, No Voice Mail)	(*	OFF
	Registered + Call Forward All (No IM, No Voice Mail)	(+	OFF
	Unregistered + Call Forward Delay + Call Forward Busy (No IM, No Voice Mail)	<b>€</b> ¢	OFF
	Registered + Call Forward Delay + Call Forward Busy (No IM, No Voice Mail)	<b>€</b> Φ	OFF
	Unregistered + Call Forward Delay (No IM, No Voice Mail, No Call Forward Busy)	(¢	OFF
	Registered + Call Forward Delay (No IM, No Voice Mail, No Call Forward Busy)	(¢	OFF
	Unregistered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)	(+	OFF
	Registered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)	(¢	OFF
	Registered + Ringing	0	Flashing RED
	Registered + On Hold	<u>"</u>	Flashing GREEN
	Registered + Connected + Incoming Call	C	GREEN
	Registered + Connected + Outgoing Call	U	GREEN
Shared Line	Unregistered	20	OFF



	Registered + Not support SCA Call-info header	<b>C</b> ?	OFF
	Registered + Not support SCA or SCA Failed	£:	OFF
	Registered + Idle	<sup>1</sup>	OFF
	Registered + Seized	2	RED
	Registered + Processing	2	Flashing GREEN
	Registered + Alert	<b>2</b>	Flashing RED
	Registered + Hold by user	2	Flashing GREEN
	Registered + Hold by the other party	2	Flashing RED
	Registered + Connected	2	GREEN
BLF/ Eventlist BLF	Offline, Unknown	20	OFF
	Terminated	1	GREEN
	Proceeding	20	RED
	Ringing (Early)	2	Flashing RED
	Trying	<b>4</b> 00	Flashing GREEN
	Confirmed	<b>4</b> 00	RED
Presence Watcher	Offline, Unknown	40	OFF



	Avaliable	1	GREEN
Handsfree	Unpair	■×	OFF
	Paired, but not connected	<b>⊟</b> ?	OFF
	Connected	<b>□</b> \$	OFF
Speed Dial	Account Unregistered	Cs	OFF
	Account Registered	CF	OFF
Speed Dial Via Active Account		Cs	OFF
Dial DTMF			OFF
Call Return		ಲ	OFF
Transfer	Account Unregistered	(+	OFF
	Account Registered	(+	OFF
Call Park	Account Unregistered	•	OFF
	Account Registered	•	OFF
Intercom	Account Unregistered	((*))	OFF
	Account Registered	((•))	OFF
LDAP Search		ID:	OFF



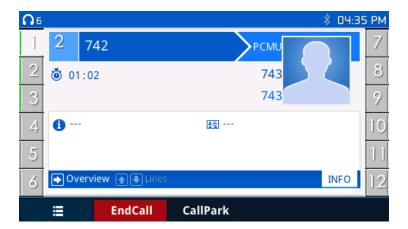
Multicast Paging		ۺ	OFF
Record	Idle	REC	OFF
	Recording	REC	Flashing
Call Log		<b>©</b>	OFF
Menu	-	ď	OFF
Voice Mail	Account not registered	<b>C</b>	OFF
	Account Registered (No new voice mail)	<u></u>	OFF
	Account Registered (Have new voice mail)	<u> </u>	OFF

Please note no matter how each line is configured on the idle screne, all the lines in call screen will keep line or shared line displayed for the corresponding accounts. For example, even if the user has configured all lines as VPK (with non-LINE mode), he/she can still use the configured account to dial out by offhook or pressing SPEAKER, HEADSET or any other unconfigured LINE key to go to call screen.



When the user is in call screen (during a call), he/she can press softkey to switch back to VPK screen.





When the user is in VPK screen during a call, he/she can press softkey or corresponding line key to switch back to call screen.

#### PROGRAMMABLE KEYS STATUS ON WEB UI

Users could access programmable key status under phone's web UI->Status.

Web III > Status > Programmable Keye Status	Virtual Multi-purpose Keys
Web UI->Status->Programmable Keys Status	Multi-purpose Keys
Web UI->Status->Extension Boards Status	Extension 1 keys
	Extension 2 keys
	Extension 3 keys
	Extension 4 keys

Select the tab you would like to check the status, the status of the specific keys will display. The screenshot below shows virtual Multi-purpose keys status.



# Virtual Multi-purpose keys Status

	Mode	Account	Description	Value
VPK 1	LINE	Account 1	No Description	No Value
VPK 2	LINE	Account 2	No Description	No Value
VPK 3	LINE	Account 3	No Description	No Value
VPK 4	LINE	Account 4	No Description	No Value
VPK 5	Speed Dial	Account 1	No Description	No Value
VPK 6	Speed Dial	Account 1	No Description	No Value
VPK 7	None	Account 1	No Description	No Value
VPK 8	None	Account 1	No Description	No Value
VPK 9	None	Account 1	No Description	No Value
VPK 10	None	Account 1	No Description	No Value
VPK 11	None	Account 1	No Description	No Value
VPK 12	None	Account 1	No Description	No Value
VPK 13	None	Account 1	No Description	No Value
VPK 14	None	Account 1	No Description	No Value
VPK 15	None	Account 1	No Description	No Value
VPK 16	None	Account 1	No Description	No Value

#### AUTO PROVISION LIST STARTING POINT

The users could find new option "Auto Provision List Starting Point" under phone's web UI->Settings->Call Features. Users could select "Extension Boards" or "VPK" which will be used first when the phone is being automatically provisioned with eventlist BLF.

Auto Provision List Starting Point 

© Extension Boards 

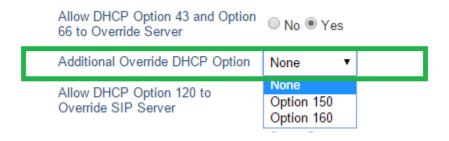
VPK

The P value for this option is P8349. 0 – Extension Borads, 1 – VPK. The default value is 0.

### ADDITIONAL OVERRIDE DHCP OPTION

The users could find new option "Additional override DHCP option" under phone's web UI->Maintenance->Upgrade and Provisioning. When enabled, users could select Option 150 or Option 160 to override the firmware server instead of using the configured firmware server path or the server from option 43 and option 66 in the local network. Please note this option will be effective only when option "Allow DHCP Option 43 and Option 66 to Override Server" is enabled.





The P value for this option is P8337. 0 – None, 1 – Option 150, 2 – Option 160. The default value is 0.

#### **ILBC AND G.723.1 SUPPORT**

The users could find iLBC and G.723.1 codecs available under phone's web UI->Account->Audio Settings.

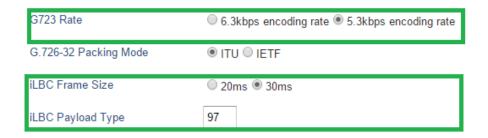
# Audio Settings



When provisioning the phone with vocoder P value, the value to set codec as iLBC is 98 and the value to set codec as G.723.1 is 4.

Under the same web page, the options "G723 rate", "iLBC Frame Size" and "iLBC Payload Type" are also added for thoses two codecs.





For complete P value list, please refer to configuration template.

#### **ENABLE SESSION TIMER**

Users could enable/disable session timer under phone's web UI->Account->SIP Settings->Session Timer. The new P value is P2395. 0 – No, 1 – Yes. By default it's enabled.

# Session Timer



#### RING SPEAKER FOR CALL WAITING

Users could find option "Ring For Call Waiting" under web UI->Settings->Call Features. When enabled, the phone will ring ringing tone instead of call-waiting tone when the audio is on handset or headset. The new P value is P22102. 0 – No, 1 – Yes. The default value is 0.

# Call Features



#### **CONFIGURABLE BACKLIGHT TIMER**

Users could find option "Active Backlight



Timeout" under web UI->Settings->LCD display. This will configure the minute of active backlight timeout with valid range from 1 to 90.

# LCD Display



#### **COLOR BACKGROUND WALLPAPER SELECTION**

Users could find option "Color Background" under web UI->Settings->LCD Display: Wallpaper category. Enter any HEX color code based on your preference. The color codes could be found here: http://htmlcolorcodes.com/

If an invalid code is configured, the phone will use default value #000000 instead.

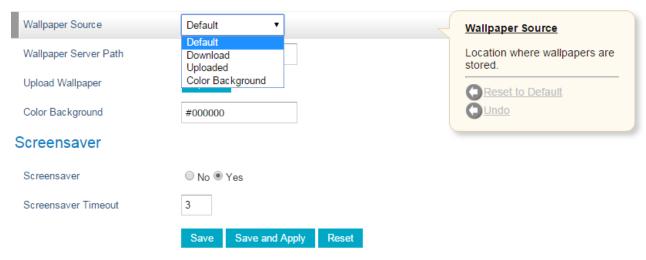
# Wallpaper



Please note the user must select "Color Background" in "Wallpaper Source" option in order to use the configurable color background code.

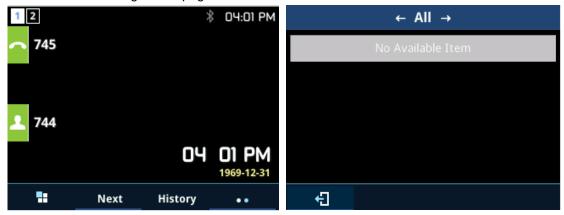


# Wallpaper



#### Example:

If the user users default color code #000000, the idle screen will load "black" as background. This color will also affect MENU configuration page.



#### New P value:

P2916: Wallpaper Source. 0 – default, 1 – download, 2 – USB, 3 – uploaded, 4 – Color background. Default value is 0.

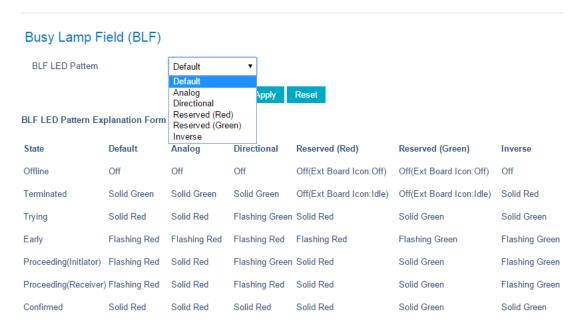
P2921: Background color. This value can be a string with max length of 16 characters. Default value is #000000.

#### **BLF LED PATTERN EXPLANATION IN WEB UI**

Users could view and configure the color and pattern of the LED status based on the BLF status update. This feature is under web UI->Settings->LED Control.



# **LED Control**



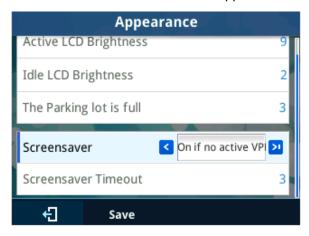
#### DISABLE SCREENSAVER WHEN VPK IS ACTIVE

This option is under web UI->Settings->LCD Display: Screensaver category. A new option "On if no VPK is active" is added there.

# Screensaver



This option is also available under LCD->MENU->Preference->Appearance.





The phone will consider the page active if VPK is in Early (ringing), Trying (dialing) and Confirmed (talking) status when VPK is configured with mode "BLF", "Eventlist BLF" or "Presence".

The P value is P2918 (existing) and value 2 is added for "On if no VLK active". P2918: Screensaver. 0 – Off, 1 – On, 2 – On if no VPK active. The default value is 1.

#### FULL BLACK SUPPORT FOR IDLE LCD BRIGHTNESS

This option can be found under web UI->Settings->LCD display. Users will be able to configure the LCD brightness to 0 when the phone is idle. Valid range is 0 to 12 where 0 is off and 100 is the brightest.



#### **BLIND TRANSFER AND ATTENDED TRANSFER SOFTKEY**

This feature works when option "Enabled Auto-Attended Transfer" under web UI->Call Features is set to "Yes". When the user tries to transfer an ongoing call, after pressing "Transfer" softkey and entering the number to be transferred to, the user will be able to select softkey "BlindTrnf" for blind transfer or softkey "AttTrnf" attended transfer.







#### **DISPLAY SIP MESSAGE TEXT ON LCD**

During an active call, if the phone receives SIP message REQUEST that has message body with line-based text data defined, the content will be displayed on the phone's LCD. In the following example, the phone LCD will display "Total \$5" as defined in the SIP message text.

10021 2013-00-02 00:02:12.004889000 209.190.121.194 192.108./8.139	51P 030 Request: BYE STP:1014202@192.108./8.139:3004
10793 2015-06-02 06:02:13.096212000 209.190.121.194 192.168.78.139	SIP 636 Request: BYE sip:1014202@192.168.78.139:5064
136 2015-06-02 06:00:52.348419000 192.168.78.139 209.190.121.194	SIP/SDF 1015 Request: INVITE sip:2418712216@209.190.121.194
165 2015-06-02 06:00:52.486721000 192.168.78.139 209.190.121.194	SIP/SDF 1192 Request: INVITE sip:2418712216@209.190.121.194
1110 2015-06-02 06:01:01.412646000 209.190.121.194 192.168.78.139	SIP 456 Request: MESSAGE sip:1014202@192.168.78.139:5064   (text/plain)
1746 2015-06-02 06:01:06.407798000 209.190.121.194 192.168.78.139	SIP 460 Request: MESSAGE sip:1014202@192.168.78.139:5064   (text/plain)
2386 2015-06-02 06:01:11.409775000 209.190.121.194 192.168.78.139	SIP 460 Request: MESSAGE sip:1014202@192.168.78.139:5064   (text/plain)
3035 2015-06-02 06:01:16.405856000 209.190.121.194 192.168.78.139	SIP 459 Request: MESSAGE sip:1014202@192.168.78.139:5064   (text/plain)
3704 2015-06-02 06:01:21.389838000 209.190.121.194 192.168.78.139	SIP 459 Request: MESSAGE sip:1014202@192.168.78.139:5064   (text/plain)
4775 2015 06 03 06:01:36 201772000 200 100 131 104 103 160 70 130	ern 456 housest Necesco cin:101470704107 160 70 170:5064 1 (toyt/plain)
Frame 1110: 456 bytes on wire (3648 bits), 456 bytes captured (3648 bits) on interface (	
Ethernet II, Src: Dell_04:85:71 (00:11:43:04:85:71), Dst: Grandstr_5e:66:c3 (00:0b:82:5e	e:66:c3)
Internet Protocol Version 4, Src: 209.190.121.194 (209.190.121.194), Dst: 192.168.78.139	9 (192.168.78.139)
3 User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5064 (5064)	
Session Initiation Protocol (MESSAGE)	
⊞ Request-Line: MESSAGE sip:1014202@192.168.78.139:5064 SIP/2.0	
⊞ Message Header	
■ Message Body	
□ Line-based text data: text/plain	
Total \$5	



#### **PRODUCT NAME**

GXP2170

#### **DATE**

01/14/2016

#### **CHANGES SINCE 1.0.6.5**

#### **ENHANCEMENT**

- Improved DND indication to cover transparent background
- Added support to display date time/weather when RIGHT VPK panel is empty
- Added support to configure whether to show label background on VPK
- Added support to show long label on VPK
- Added support to hide softkeys on main page
- · Added secondary softkey FIRST to quickly return to VPK page one
- Added NewCall softkey on call screen for user to initiate new phone call
- Marked softkey EndCall to RED style

- Fixed Device BLF is not updated correctly issue
- Fixed Phone randomly doesn't send voice when using Plantronics EHS CS540A headset with APV-63 EHS adapter
- Fixed Line key LEDs don't provide visual indications of Shared call appearances
- Fixed Device fails to displays BLF and Call park subscription status
- · Fixed The mode of EXT is changed to Speed Dial after reset on web
- Fixed DND indication unfriendly issue
- Fixed Device work abnormal after press TRANSFER key then press NewCall softkey
- Fixed The Back softkey at the Bluetooth Scan screen cannot switch to Home softkey
- Fixed Device possible crash when drawing DND screen



#### PRODUCT NAME

GXP2170

#### **DATE**

12/18/2015

#### **CHANGES SINCE 1.0.6.2**

#### **ENHANCEMENT**

- Added VPK support for eventlist auto-provision. If there are more BLFs in the eventlist than idle VPK keys, extra BLFs will be auto-provisioned to EXT board
- Improved the display format of missed calls
- Added display of extension Forward All in idle screen
- Added German language input in all configuration fields where the user is required to enter alphanumeric input
- Added 12 lines support (with 6 accounts)
- Added "None" mode for VPK

- Fixed LCD displays abnormal after changing language on the VPK editing screen
- Fixed Broadsoft Silent Alerting feature doesn't work
- Fixed GUI crashes after downloading a large XML application file
- Fixed The device calls the wrong number after entering number in onhook dialing screen then pressing MPK
- Fixed The device does not send DHCP DISCOVER after factory reset
- Fixed LCD doesn't have prompt when adding an invalid local group
- Fixed Obtaining the wallpaper from USB flash drive requires reboot to take effect
- Fixed Voicemail VPK is invalid after configuring its value
- Fixed The HOLD screen displays abnormal
- Fixed LCD does not display the RESUME softkey when the call is on hold
- Fixed LCD does no display MENU mode on editing screen of VPK
- Fixed Device fails to set LINE to Menu mode
- Fixed Device fails to download wallpaper after repeating downloading and uploading
- Fixed LCD does not display softkey-MPK in time when setting the MPK on the weather or currency screen
- Fixed The editing screen of VPK is displayed wrong after setting VPK on LCD
- Fixed The label of LINE 6 is displayed abnormal after disabling Bluetooth Hands-free mode
- Fixed Device speed dial with Bluetooth Hands-free account after setting VPK account to account 6



- Fixed The Save and Apply button on VPK web page does not work for other web pages
- Fixed LCD pops up prompt after pressing VPK which is not configured
- Fixed The Save and Apply button on VPK web page works abnormal
- Fixed LCD is displayed abnormally when adding the 47<sup>th</sup> and 48<sup>th</sup> VPK
- Fixed The device is displaying extra currency entries when less than 5 entries are queried
- Fixed Currently page does not update periodically



# **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

# **DATE**

02/09/2016

# **CHANGES SINCE 1.0.5.29**

- Fixed Device crashing issue after cleaning up eventlist BLF when LED light is abnormal
- Fixed Device is losing network access when enabling LLDP/LLDP-MED
- Fixed Device is not recognized by TP-LINK switch using LLDP
- Fixed Device showing no IP issue



# **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

# **DATE**

01/08/2016

# **CHANGES SINCE 1.0.5.28**

- Fixed Device BLF is not updated correctly
- Fixed Phone randomly doesn't send voice when using Plantronics EHS CS540A headset with APV-63 EHS adapter



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

12/30/2015

#### IMPORTANT UPGRADING NOTE

Once upgraded to 1.0.5.28, downgrading to any previous firmware version is not supported.

#### **CHANGES SINCE 1.0.5.23**

#### **ENHANCEMENT**

- Added Dual outbound SIP proxy support
- Added support to show SIP message text content on LCD screen
- · Added option to ignore Alert-info header when used for distinctive ringtone
- · Added support Broadsoft XSI authentication type
- Added option to filter characters from dialed numbers
- Added ability to customize DHCP option for provisioning server
- · Added ability to prevent the phone from sending DNS then HTTPS requests to IPVideoTalk media server
- Added ability to search in Broadsoft XSI search by using part of the strings
- · Added BLF Call pickup with Barge-In option
- Added Broadsoft XSI SIP Authentication method by selecting the account

- Fixed BLF keys not showing correct status for monitored extension with multiple registrations
- Fixed Phone cannot auto answer by call info provided by PortaOne
- Fixed DUT sends SIP instance ID when set to No, unless DUT is rebooted
- Fixed DUT doesn't update the CID when performing auto-attended transfer
- Fixed DUT keeps sending SUBSCRIBE after receiving NOTIFY with reason=noresource



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

# **DATE**

09/25/2015

#### **CHANGES SINCE 1.0.5.18**

#### **ENHANCEMENT**

- Added support to allow line labeling by setting Line Key to Account X
- Added support to have auto answer audio source configurable
- Added support for re-conference after split
- · Added support for secondary preferred DNS
- Added fail-safe if server does not include "Expires" field in SUBSCRIBE response
- Updated logo for web UI

- Fixed Phone GUI slow performance issue
- Fixed Audio mode is wrong after auto answer call in Toggle mode
- · Fixed MPK light indicator doesn't work for call park
- Fixed Phone does not update BLF status when the version on the phone is greater than the version in the received NOTIFY message



#### PRODUCT NAME

GXP2130/GXP2140/GXP2160

#### **DATE**

08/12/2015

#### IMPORTANT UPGRADING NOTE

Once upgraded to 1.0.5.18, downgrading to any previous firmware version is not supported.

#### **CHANGES SINCE 1.0.4.23**

#### **ENHANCEMENT**

- Added Auto provision the Multi-purpose keys based in the SIP NOTIFY messages from the Eventlist BLF feature
- Updated Time settings for Moscow, Russia
- Added SSL encryption key random generate feature
- Added Auto-scale feature to fit more characters in the LCD account label area
- Added Option to disable SIP NOTIFY Authentication
- Added Feature to display BLF status on LCD and Extension board
- Added Automatic UCM Call Features Integration
- Added Features as a softkey after configuring feature codes for phone
- Added Option to allow hands-free conference with SIP line
- Added Phonebook HTTP/HTTPS Authentication
- Added Configuration to enable/disable host verification
- Added Feature to enable bypass dial plan through call history and directories
- Added Option to unregister instance on reboot, which only clears the registration of the phone's own contact before the phone gets rebooted
- Added One-button (Redial) showing recent dialed list
- Added More obvious indication for DND display
- Re-grouped the status information from LCD to make it the same as status page on WEB UI
- Added Feature to display IP2Location information on both Web GUI and LCD
- Added Time setting into LCD menu
- Added Factory reset and provision now function to web
- Added Audio recording application
- Added OnHold Reminder tone
- Added Call Log as a new mode of programmable keys for users to access to the call history directly
- Added Feature to allow configuration of MPK or Line key to provide MWI for other extension



- Added Configurable option to change the status of BLF LED indicator
- Added Option to customize the On/Off code for DND/call forwarding
- Added Reset phone through SIP notify
- Added Option to enable priority to Intercom Call
- Added Support for Hebrew characters displayed on EXT board
- Added "Transfer to Voicemail" softkey support for FreePBX
- Added Broadsoft UC-One feature: Special handling for numbers ending with \*
- Added "Trusted CA Certificates" option to allow user to upload certificate
- Added Redirect HTTPS firmware/Config file upgrade request when receiving HTTP 302 response
- Hid returned values for passwords (eg. Admin password, SIP passwords) when P-value is received
- Added Option to set the default call log type to be used with the phone
- Modified Feature for auto-check language files update from server and added options to download
- Improved Forward All timeout behavior

- Fixed Phone fails to upgrade and gets stuck at booting screen due to firmware mask check fail after incomplete firmware upgrade
- Fixed When call is answered using handset, subsequent calls will not ring Bluetooth headset
- Fixed Ethernet link shows disconnected with no IP in mixed network environment with IPv4 and IPv6
- Fixed User can retrieve any configuration value without authentication
- Fixed Phone couldn't dial DTMF correctly via MPK buttons
- Fixed Bluetooth Name is always updated along with other variables, such as power and discoverable
- Fixed Device connected with UCM auto-provision server fails to be provisioned using HTTPS method
- Fixed Configure Eventlist BLF URL of other accounts will cause Eventlist BLF MPK key has no light
- Fixed LDAP Search failed when using Windows Active Directory/LDAP service
- Fixed Can hear clicking sound at the beginning of the call
- Fixed Click-to-Dial issue after enabling Hands-free mode
- Fixed Long pressing blank GXP2140EXT MPKs couldn't skip to edit page
- Fixed Call waiting tone disappears after far end cancels call
- Fixed Duplicate P-value in exporting config.txt file
- Fixed Device works abnormally when there is incoming call in call features page
- Fixed Audio mode works abnormally when using EHS headset for the call
- Fixed GXP2140 audio mode is changed to speaker after canceling Transfer
- Fixed Line LED doesn't light up during the call
- Fixed Off-hook auto dial works abnormally in FwdAll page
- Fixed Mute is not disabled after hands-free device ends the call
- Fixed Phone is muted after pressing Mute key in ringing status
- Fixed Call couldn't be retrieved when auto-attended transfer fails
- Fixed Displaying invalid Conf call softkey after 2nd party ending call
- Fixed Speaker key ending call abnormally in Toggle mode
- Fixed LCD display is abnormal after ending auto-answer call in FwdAll page



- Fixed Mute key and icon display issue after enabling Idle Mute
- Fixed Upgrade default method becomes TFTP after factory reset through TR069
- Fixed Phonebook softkey doesn't work
- Fixed There is no Hands-free line account display after connecting with hands-free device
- Fixed Device plays dial tip tone in idle status
- Fixed Issue of Hands-free line before logging in public mode
- Fixed Phone does not accept BLF NOTIFY if SUBSCRIBE is not yet accepted
- Fixed Issue when adding contacts in public mode
- Fixed Last account display issue after enabling Hands-free mode and configuring with MPK
- Fixed There is no "1Bc" softkey on number input box of new contact
- Fixed LCD displays abnormally after far end cancelling call in Keypad lock mode
- Fixed Star key Keypad/Restrict mode Lockout Duration doesn't work
- Fixed Some contents are displayed incompletely in public mode login page
- Fixed Hands-free call is not terminated when the cellphone is disconnected while dialing
- Fixed LCD doesn't skip to dialing page after pressing Headset key in Menu page
- Fixed Gs\_pcap crashes after removing USB when capturing the trace for the call
- Removed Broadsoft Phone book/Call Log from LCD when not in use
- Fixed SIP TLS Private key and certificate cannot be provisioned using XML
- Fixed Phone fallback to PCMU upon re-invite during the session
- Fixed Incompatibility issue with Plantronics M165 Bluetooth headset
- Fixed Device doesn't write in contacts after clearing then downloading the same phone book.xml file
- Fixed Audio codec choice 3 and choice 4 are both G.729 after reset those options
- Fixed Device upgrade abnormally after downloading files then powering off in HTTPS mode
- Fixed Call Forward indication
- Fixed Memory leak while parsing UCM Keep Alive Option
- Fixed HTTP request for currency does not return correct values
- Fixed Hebrew language missing strings



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

08/02/2015

#### **CHANGES SINCE 1.0.4.17**

#### **ENHANCEMENT**

- Added LDAPS support
- Hided LDAP config when constraint mode activated
- Added option to enable/disable Crypto life time when using SRTP
- Enabled MetaSwitch configuration file download authentication feature in general build
- Added support of DHCP OPTION 160 on Metaswitch FW
- Updated New Zealand time zone
- Added SW version and DevID to HTTP requests for phonebook.xml and language.txt
- Added support for x-gs-screen event from SIP NOTIFY message
- Added option to enable/disable the "Active MPK Info Page"
- Added display info from Diversion in received INVITE
- Added scaling the input on account description to fit in the LCD account description area
- Fixed Bad quality the multi-purpose keys from the Eventlist BLF
- Added ability to control NTP update interval
- Simplified the procedure to make N-Way conference under BroadSoft server
- Added Hebrew input
- Added support for BLF offline state
- Added Quick/Exact match search method on Handsfree mobile phonebook
- Added printing syslog level on syslog messages
- Added option 160 support
- Added option to enable/disable "Active MPK Info Page"

- Fixed LDAP doesn't support display in Chinese
- Fixed LDAP Lookup for incoming call does not display Russian characters
- Fixed call park pick-up with replace header
- Fixed NOTIFY from call group pickup not applied
- Fixed phone not displaying contact with Polish characters in LDAP
- Fixed devices will call wrong number after DTMF in talk and then pressing Speed Dial MPK



- Fixed auto answer works abnormally when there are multiple calls
- Fixed dial DTMF with line keys works abnormally after pressing twice in idle
- Fixed GUI crashed after pressing Send key in Bluetooth phonebook page
- Fixed firmware upgrade issue with option 160
- Fixed Secondary dial tone dial plan with insert/replace
- Fixed G722 encoded audio prompt files are distorted during playback
- Fixed Early Conference and Ring-back Anomaly issue
- Fixed Headset cannot be put on mute when Bluetooth is enabled
- Separated QoS SIP and RTP options
- Fixed Onhook dialing with secondary dial tone causes "No dial plan rules matched"
- Fixed CONF+BLF use Broadsoft ID instead of Extension number
- Fixed Syslog output not including MAC address in all messages
- Fixed Syslog output not showing correct firmware version
- Fixed Bad quality audio using G.722
- Fixed Phone does not ask for configuration file, firmware and ring tones when LLDP is used to discover VLAN
- · Fixed Mute doesn't work after holding the talk in Bluetooth audio mode
- · Fixed Device worked abnormally after answering multicast paging call in transfer page
- Fixed Phonebook export/upload are accessible without authentication request
- Fixed Configuration value might be retrieved without authentication
- Fixed LDAP Search only by string beginning
- Fixed Phone crashed after multicast paging calling in account1 inactive status
- Fixed SSH root access doesn't work.



# **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

# **DATE**

02/19/2015

# **IMPORTANT UPGRADING NOTE**

• Once upgraded to 1.0.4.17, downgrading to any previous firmware version is not supported.

# **CHANGES SINCE 1.0.4.16**

# **BUG FIX**

• Fixed potential security issue for unauthorized access to product configuration



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

09/30/2014

#### **CHANGES SINCE 1.0.4.15**

#### **ENHANCEMENT**

• Added support to configure phone's MPK from phone GUI

- Fixed Disable call waiting also worked in ringing status
- Fixed Bluetooth enabled/disabled is not indicated properly on the phone screen
- Fixed Phone hangs while transferring calls
- Fixed Phone cannot initiate multicast paging call when the handset is offhook from idle state
- Fixed Bluetooth: Line toggling causes audio loses
- Fixed Preferred vocoder option default value error after resetting audio settings page
- Fixed Conference works abnormal when using MPK to transfer
- Fixed BYE request sent to the wrong proxy/IP based on the route set
- Fixed Device randomly crashed after removing network cable in talking status



#### PRODUCT NAME

GXP2130/GXP2140/GXP2160

#### **DATE**

09/16/2014

#### **CHANGES SINCE 1.0.4.10**

#### **ENHANCEMENT**

- · Added support to play short reminder beep when performing auto answer
- Added HTTP CTI support
- Added HTTP/HTTPS authentication support for phonebook download
- Added ability to select call pickup mode
- Added support to apply dial plan when dialing from call history/phonebook
- Added support to unregister on URI based and clear only the contact header from the phone when using "Unregister on Reboot" option
- Added XML application support
- Added Direct IP Call support on MPK and Phonebook
- Added ability to dial the digits faster when using MPK as Dial DTMF
- Added ability to show text pop-up window via SIP MESSAGE
- Added option "Disable Busy Tone on Remote Disconnect" to disable playing busy tone upon receiving BYE
- Added option to enable/disable Broadsoft call park

- Fixed 3CX call park retrieval for shared parking is not working
- Fixed MPK Call Park mode is not working correctly on 3CX platform
- Fixed HTTP GET provisioning request displays HTTP username/password in plain text
- Fixed Device plays ringback tone in talking and audio loopback mode
- Fixed Device doesn't apply configurations after enabling 3CX auto provision
- Fixed Feature code for "Busy Call Forward" cannot be enabled/disabled when handset is on hook
- Fixed Phone that initiates multicast paging cannot listen to the same multicast IP:Port
- Fixed After receiver ends the multicast paging call, the receiver can still listen when off hook to dial screen
- Fixed Paging barge of multicast paging doesn't work
- Fixed Multicast paging barge doesn't have audio



- Fixed there is no sound in talking and audio loopback mode after network is down->initiating multicast paging->reconnecting network
- Fixed SIP Server and outbound proxy cannot be configured with host address in format x.x.x.0
- Fixed IP call failed by using "Edit and Dial" option in call history
- Fixed phone crashes after clearing the listened multicast paging IP:Port
- Fixed phone doesn't answer the call when it's the first time to enable Auto Answer
- Fixed phone failed to connect to UCM6100 to pick extension for auto provisioning if using HTTPS method
- Fixed G.722 encoded audio prompt files are distorted during playback
- Fixed tooltip for option "Unregister on Reboot" is not correct
- Fixed phone frequently sends SUBSCRIBE for DND when using Broadsoft platform



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

06/19/2014

#### **CHANGES SINCE 1.0.4.6**

- Fixed There is no dial tone, dialing DTMF tone, ring back tone, call waiting tone when using Bluetooth headset
- Fixed MPK mode "Call Park" is not working on 3CX platform
- Fixed DTMF digits are not transmitted "In audio" when using Bluetooth headset
- Fixed Device is using the next account when dialing through predictive dial list
- Fixed SPLIT softkey in auto-attended transfer is not working properly
- Fixed MPK does not send DTMF using "Enable Sending DTMF via Speed Dial" during a call when configured as BLF
- · Fixed DTMF is not recognized
- Fixed Phone crashed after dialing conference extension on UCM61xx
- Fixed Last 2 accounts on the phone displays abnormal after setting Account Active to No
- Fixed MPK mode "Record" is not working



# **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

# **DATE**

06/05/2014

# **CHANGES SINCE 1.0.4.5**

#### **BUG FIX**

• Fixed Bluetooth hands-free: device ends call after pressing other line in HF line talk



#### PRODUCT NAME

GXP2130/GXP2140/GXP2160

#### **DATE**

05/30/2014

#### **CHANGES SINCE 1.0.3.9**

#### **ENHANCEMENT**

- Added Option to enable/disable widget display
- Added Support to insert pauses into speed dials and phonebook entries
- Added Web interface to capture trace. New P values added:
  - P2999: capture location. 0 internal storage, 1 USB
  - P6007: capture with RTP packets. 0 No, 1 Yes
  - P6008: USB filename for the captured trace
- Added Multicast paging support
- Added Broadsoft IM&P feature: Dial Conference
- Added New P values for Broadsoft IM&P feature:
  - P6005: Broadsoft IM&P port number
  - P6006: Broadsoft IM&P host address
- Added Option to show account name only and not the User ID on the LCD screen
- Added Phone screen should display labels for line keys when configured as Call Park/BLF
- Changed display "Group ID" block name in Broadsoft call log
- Added "SpeedDial" as "Action" to softkeys
- Added "Forward" option in ringing screen to transfer incoming call without need to answer it

- Fixed IP display abnormal under voice VLAN circumstance
- Fixed LCD display abnormal when switching line in Bluetooth Hands-free line calling out status
- Fixed End the first call then the audio mode icon would disappear
- Fixed IP address doesn't display when assigned VLAN via LLDP
- Fixed Phone shows wrong registration status when phone's IP is 0.0.0.0
- Fixed 2-way audio keeps open during a blind transfer
- Fixed Adjusting account index to begin from 1 for phonebook xml file
- Fixed Dial softkey displays even there is no entry in UC-One
- Fixed Device couldn't log in UC-One after pressing LogIn softkey repeatedly
- Fixed There is no cancel softkey in Transfer interface after Bluetooth Hands-free line ends the call



- Fixed Device works abnormal sometimes after downloading config file
- Fixed BTOff softkey doesn't work after switching line
- Fixed Device works abnormal after pressing BTOff softkey when there are 2 incoming calls
- Fixed Audio mode changed after the other party answers the call when using Bluetooth Hands-free line
- Fixed Distinctive ringtone only reads last line when multiple alert-info headers are sent
- Fixed UC-One softkey icon on LCD screen can't show up or hide automatically
- Fixed There is no Resume softkey after pressing UP key in HOLD screen
- Fixed Phone doesn't handle multiple m lines in SDP correctly
- Fixed Contacts disappeared in UC-One screen
- Fixed Public mode is abnormal
- Fixed Bluetooth: couldn't redial via Bluetooth headset
- Fixed Softkey displays incorrectly if switching to another line when ringing under Bluetooth Handsfree mode
- Fixed Bluetooth hands-free: call is established abnormally when the connected mobile phone is in a call with another phone
- Fixed Bluetooth hands-free: audio sound performs abnormally after pressing HF line key in Bluetooth headset talk
- Fixed Bluetooth hands-free: LCD displayed abnormally after ending Bluetooth hands-free account talk
- Fixed Bluetooth hands-free: Device performed abnormally after off/onhook in HF line calling out status
- Fixed Bluetooth displayed wrong in pair request window interface
- Fixed Bluetooth hands-free: pair with hands-free device abnormal after disconnecting in talk
- Fixed Bluetooth ring tone issue
- Fixed Bluetooth hands-free: LCD displayed wrong when there is incoming call of HF account line in IP display interface
- Fixed Star key lock has no password limit on LCD
- Fixed Softkey5 on the 1st entry in Bluetooth scanning screen displayed wrong
- Fixed LCD displayed no talk duration and swap softkey disappeared after answering second call
- Fixed Bluetooth hands-free: all softkeys in incoming call page didn't work
- Fixed Weather application language cannot be customized
- Fixed Star keypad lock failed in screensaver mode
- Fixed Broadsoft phonebook sends search inquiry to wrong URL
- Fixed Device connects to Bluetooth hands-free device automatically after disconnecting manually
- Fixed Device couldn't dial through Bluetooth phonebook when talking in Bluetooth hands-free mode
- Fixed GXP2130 Broadsoft executive assistant page displayed incomplete
- Fixed LCD displayed abnormally when there is incoming call in Bluetooth phonebook screen
- Fixed Bluetooth phonebook entry disappeared after clearing search filter
- Fixed Cursor moved abnormally when turning pages in Call history
- Fixed Last call wasn't held after dialing through Broadsoft Phonebook
- Fixed There is no audio icon displayed after dialing through Broadsoft phonebook
- Fixed Pressing line key couldn't answer incoming call in pickup call screen



- Fixed Broadsoft UC-One feature: fails to display all contact entries in Group Directory
- Fixed When switching line, instead of showing dial screen, it shows an overview screen
- Fixed Input Shiftable mode is not functioning correctly
- Fixed Bluetooth hands-free: first incoming call after connection will not play out ring tone when using IPhone
- · Fixed Bluetooth hands-free: LCD displayed abnormally after HF line incoming call cancelling
- Fixed Bluetooth hands-free: LCD displayed abnormally after ending HF line talk
- Fixed Bluetooth hands-free: One-way audio when incoming call in HF line talking
- Fixed Bluetooth hands-free: No Answer or Reject softkey on incoming call after ending HF line talk



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

05/05/2014

#### **CHANGES SINCE 1.0.2.9**

#### **ENHANCEMENT**

- Updated OpenSSL library from 1.0.1e to 1.0.1g
- Added PhonePower special feature
- Added P value for HTTP Proxy. P1552 HTTP Proxy Server
- Added Broadsoft IM&P features: Contact/Buddy List and Presence
- Added Screensaver options to LCD under "Preference"->"Appearance"
- Added Web UI option to select default search mode for phonebook
- Added Support to display extension number in searching result from Broadsoft Group Directory
- Added GUI interface for users to download language files
- Added Bluetooth softkey in Hands-Free dial screen
- Added Backspace softkey will change to clear when long pressing it in UCM Detect screen
- Added Second dial tone support
- Added Input character selection window
- Added BLF server support
- Added New P values:

P2375/P2475/P2575/P2675/P2775/P2875 - BLF Server option for Account 1/2/3/4/5/6

- P2920 Input Selection Mode
- P2964 Enable Broadsoft IM&P
- P2965 Broadsoft IM&P Account
- P2966 Broadsoft IM&P Username
- P2967 Broadsoft IM&P Password
- P2968 Broadsoft IM&P Autologin
- P2969 Broadsoft IM&P Display Non XMPP Contacts
- P2970 Default Phonebook Search Mode

- Fixed Fetching data using cURL vulnerability
- Fixed HTTP/HTTPS proxy settings not respected
- Fixed DSP flow buffer index in TX path



- Fixed Part of the call history logs lost
- Fixed GXP2130 Home screen time display will shift when ticking
- Fix set date for 802.1X certification
- Fixed Bluetooth Hands-Free: no pairing request window pops up and Bluetooth crashes when using with certain mobile devices
- Fixed Bluetooth Hands-Free: device couldn't manually connect with hands-free device after disabling Hands-Free mode
- Fixed Bluetooth Hands-Free: SWAP softkey displays after ending one talk in hands-Free mode
- Fixed Bluetooth Hands-Free: device couldn't switch connection with different Hands-Free device directly
- Fixed Bluetooth Hands-Free: pressing Speaker key couldn't end call for Hands-Free account
- Fixed Bluetooth Hands-Free: device performs abnormally after reboot device in Hands-Free talking
- Fixed Bluetooth Hands-Free: device works abnormally after Hands-Free device switches audio source
- Fixed Bluetooth Hands-Free: audio mode is changed after redialing in Headset dialing mode
- Fixed Bluetooth status display is wrong in scanning screen
- Fixed GUI crashed after searching and pressing Down key twice in local phonebook
- Fixed GUI crashed after incoming call and then pressing Right and Left keys on local call log screen
- Fixed Cursor disappears after switching language in LCD
- Fixed Pressing digits when the phone is on hook doesn't go to dial screen if the screensaver is active
- Fixed TR-069: Removed TR-069 enable/disable option. TR-069 is enabled by default.
- Fixed TR-069: CPE replies with error code 9016 for firmware upgrade
- Fixed Reduced audio buzz when receiving high amplitude, distorted RTP
- Fixed Implemented Handset TX (MIC path) user gain
- Fixed LCD didn't skip to dialing screen after off hook in screensaver process
- Fixed No Connect softkey for paired devices in Bluetooth scanning screen
- Fixed "Custom Call-Info for Auto Answer" doesn't work
- Fixed Device couldn't be called in and one-way audio happens after calling out
- Fixed No SignIn softkey on LCD when account has been added to a call queue
- Fixed The call is dropped by the phone after 37 seconds
- Fixed Device couldn't download Bluetooth phonebook automatically after connecting with Galaxy series mobile devices
- Fixed Softkeys in idle screen is displayed incompletely for some languages
- Fixed LCD returns to wrong directory after checking detail info in call history
- Fixed Intercom MPK didn't work when the phone is in talking status
- Fixed LCD displays abnormally after pressing LDAP search MPK in screensaver
- Fixed Changed title "Last Name" to "Name" in Broadsoft call log
- Fixed LCD displays screensaver after pressing HOLD+DOWN keys in talking status
- Fixed Uploading wallpaper doesn't take effect unless the wallpaper source is uploaded
- Fixed Wallpaper doesn't scale to screen size
- Fixed Input selection window disappears too soon on ExactMatch phonebook search mode



- Fixed There is no "Pickup" and "Call Park" softkey
- Fixed It can only display 15 numbers in "Edit and Dial"
- Fixed Bluetooth headset key has to be pressed twice to answer incoming call in holding status
- Fixed Bluetooth headset and speaker mode are used at the same time after resuming the call on hold
- Fixed Two calls are initiated on redial after Account 1 is configured as SCA
- Fixed Phone couldn't release the call via Bluetooth headset after back from another audio mode
- Fixed Pressing Bluetooth headset key couldn't end call
- Fixed LCD doesn't skip to dialing screen after pressing Hands-Free line key in FwdAll screen
- Fixed There is no system ring tone when adjusting volume in ringing screen
- Fixed Must press Headset key twice to switch to Toggle mode
- Fixed Broadsoft Call Center status is displayed abnormally
- Fixed LCD is returned to idle screen before dialing out from call history
- Fixed Adjusting volume on screen dialing screen won't take effect after establishing call



#### **PRODUCT NAME**

GXP2130/GXP2140/GXP2160

#### **DATE**

03/27/2014

#### **CHANGES SINCE 1.0.1.14**

#### **ENHANCEMENT**

- Added Bluetooth handsfree profile support
- Added Bluetooth phonebook download support
- Added Predictive dial support for Bluetooth handsfree account
- Added Broadsoft Flexible Seating support
- Added Broadsoft Executive/Assistant support
- Added 3CXv12 CTI integration
- Added Provider lock support
- Added Configuration file upload support via web GUI
- Added Screen saver support
- Added Wallpaper support
- Added STAR key keypad lock feature
- Added Keypad shortcut to reboot (HOLD + LEFT) and provisioning (HOLD + RIGHT)
- Added Missed call softkey on HOME screen
- Added Phonebook search using characters by pressing number keys
- Added Option to disallow turn on DND function
- Added Bluetooth options "Device name" "Bluetooth MAC" and "Discoverable timeout"
- Added Special symbol input method
- Added Hide currency and weather screen if not enabled

- Fixed Lack of side tone on the phone
- Fixed "Configuration via Keypad Menu" doesn't take effect
- Fixed Handset mode not working if SCA is enabled
- Fixed Pressing line key shouldn't take effect if SCA is enabled
- Fixed Bluetooth scan result still shows after Bluetooth is turned off
- Fixed Bluetooth has no ring back tone or DTMF tone
- Fixed Bluetooth has no sound in audio loopback
- Fixed Handset icon displays after answering call using Bluetooth headset



- Fixed Phone is automatically disconnected from Bluetooth headset
- Fixed Bluetooth headset has no ring tone after pressing BT ON/BT OFF softkey
- Fixed Phone doesn't play failure treatment after line seize on active line
- Fixed Bluetooth discoverable option abnormal
- Fixed Wrong audio channel after hold and unhold in headset toggle mode
- Fixed EHS headset has no dial tone when switching line
- Fixed The number entered after 0 changes to + in transfer or conference screen
- Fixed Bluetooth audio mode is still on after answering call from EHS headset
- Fixed No sound after pressing another line key in incoming call
- Fixed Increased ringer maximum loudness by 5dB
- Fixed Phone can't dial from Broadsoft phonebook if account 1 is not registered
- Fixed GUI crash using weather after configuring certain self-defined city code
- Fixed Idle mute is disabled after pressing MUTE in talking
- Fixed "Enable Idle Mute" option in web GUI doesn't take effect
- Fixed Transfer not working if BLF MPK has alphanumeric characters in user ID configured
- Fixed Pressing RIGHT key doesn't take effect in LCD menu->UCM Detect: "Account to Apply"
- Fixed LCD switches to wrong screen after accessing call history detail information
- Fixed LCD menu and keypad selection related issues



#### **PRODUCT NAME**

GXP2140/GXP2160

#### **DATE**

03/13/2014

#### **CHANGES SINCE 1.0.1.6**

- Fixed Conference participant cannot hear the ring-back tone when the third-party is ringing
- Fixed Network connectivity lost when changing the Layer 2 QoS 802.1p Priority
- Fixed Phone doesn't send 200OK to reply NOTIFY for eventlist BLF in IPv6
- Fixed Self-defined ring tone plays when switching account in dialing page
- Fixed Hoteling status update in LCD idle screen not working properly
- Fixed Phone doesn't send out Unavailable code info in SUBSCRIBE for Broadsoft call center
- Fixed Phone crashes sometimes on parsing NOTIFY messages with Registration event
- Fixed NTP server path modification couldn't be saved
- Fixed GUI crash in UCM detect



#### **PRODUCT NAME**

GXP2140/GXP2160

#### **DATE**

02/07/2014

#### **CHANGES SINCE 1.0.0.17**

#### **ENHANCEMENT**

- Added Broadsoft call park notification support
- Added Broadsoft shared call appearance support
- Added Broadsoft call center support
- Added Eventlist BLF update support for Broadsoft
- Added Broadsoft network-based 3-way/N-way call support
- Added Local group and Broadsoft phonebook in phonebook support
- Added Instant message

- Fixed Numbers typed in "LDAP" will disappear after pressing volume keys
- Fixed Auto location abnormally after factory reset
- Fixed No DTMF tone in Bluetooth headset
- Fixed No ring tone can be heard from headset under "Headset" mode
- Fixed Cursor skipped abnormally after LDAP searching
- Fixed Disabling "Always Ring Speaker" invalid and no headset icon on LCD when toggling on headset mode
- Fixed "Always Ring Speaker" abnormal in Toggle mode
- Fixed Noise exists in ring tone after receiving 486
- Fixed Invalid web UI options "Intercom" and "Disable DND"
- Fixed Extension board won't reboot with GXP2140
- Fixed Wrong language display after modifying in config file
- Fixed Phone can hear itself in Bluetooth headset mode after the far end puts the call on hold
- Fixed LCD won't switch to transferring interface after conference is ended
- Fixed There is no voicemail icon next to account bar when receiving voicemail
- Fixed Cannot adjust volume when there is an incoming 3rd party call
- Fixed LCD language doesn't update immediately
- Fixed "Disable Missed Call Backlight" doesn't take effect
- Fixed Audio mode is wrong or no sound in talking after disconnecting with Bluetooth headset



- Fixed Phone won't automatically connect to Bluetooth headset at bootup
- Fixed LCD displays wrong ID if returning from an unsuccessful transfer
- Fixed LDAP search result cannot be fully displayed if pressing UP key
- Fixed Device submitted wrong PPPoE service name
- Fixed Phone cannot establish second conference call
- Fixed Subscribe update doesn't synchronize with Registration
- Fixed Device ends call automatically when switching lines in EHS mode
- Fixed Search filter is not cleared after exit from LDAP directory
- Fixed Last name information is lost after saving LDAP searching result to local phonebook
- Fixed Audio mode won't switch to headset mode after off-hook on volume adjusting screen
- Fixed Pressing BT headset call button invalid on incoming call during an active call
- Fixed Pressing BT headset call button cannot end call
- Fixed Pressing line key can't switch to dialing interface in audio loopback mode
- Fixed LCD displays abnormal after pressing BT headset call button to end call
- Fixed Cannot end call after pressing Headset key in CONF interface
- Fixed No ring tone after far end hangs up in CONF or Transfer interface
- Fixed Device is abnormal after modifying LDAP settings in talking
- Fixed LCD displays + if pressing 0 for a long time as DTMF during talking
- Fixed Volume adjusting issues in audio loopback mode
- Fixed Extra softkey "Uncheck" displays after deleting phonebook group
- Fixed Input mode can be toggled when editing LDAP directory number
- Fixed MPK LED doesn't update status upon receiving BLF eventlist NOTIFY on extension board
- Fixed Forward icon displays after disabling account 1
- Fixed Phone ignores characters after # when sending DTMF during the call
- Fixed Incoming calls display elapsed time of the last call
- Fixed Phone won't return to active call if answering a new call during transfer operation
- Fixed Call-Info header is in INVITE after switching to second account
- Fixed Redial abnormal after configuring account active to No
- Fixed Feature code \*72 takes effect even if account 1 is not registered
- Fixed GUI crash after pressing Speed Dial MPK with no available lines
- Fixed Off-hook auto dial not working after configuring EHS headset
- Fixed Switching audio mode issues in off-hook auto dial
- Fixed Device cannot automatically switch audio mode after turning off Bluetooth headset
- Fixed Wrong audio codec is negotiated after resuming the conference on hold
- Fixed Wrong LCD display when pressing CONF key if conference is disabled
- · Fixed Wrong audio mode after disabling toggle mode
- Fixed Device cannot dial or reboot after switching MPK mode from BLF to Presence Watch
- Fixed Provisioning doesn't work via HTTP after reboot