

DP752/750/730/722/720/760 Firmware Release Notes

IMPORTANT UPGRADING NOTES

- **For all DP series, once upgraded to 1.0.17.1, downgrade to 1.0.15.6 or below is not supported.**
- For all DP series, once upgraded to 1.0.13.10 or above, downgrading to 1.0.13.8 or previous firmware version is not supported. It is recommended not to downgrade from 1.0.15.2 to 1.0.13.10.
- Since firmware 1.0.7.5, we have added support for new format of P values, for the old <Pxx>value</Pxx> format, in the xml config file, please use <config version="1">; for the new format of P value, that is <item name="xx.xx.xx.xx.xx">value</item>, in the xml config file, please use <config version="2">. DP7xx now support both format XML config file but will not work if the version number is wrong.
- Upgrade DP730/DP720/DP722 from 1.0.7.5 to 1.0.9.1 or higher will need update both firmware and resource images, so it will download and reboot twice. After upgraded to 1.0.9.1 or higher, the upgrade procedure will only happen once as always.
- For DP720, after the base station (DP750 or DP752) upgraded to 1.0.7.5, the handset DP720 will not automatically upgrade because of the whole GUI has been changed from 1.0.4.10. Users please manually operation the handset firmware on “Menu” -> “Settings” -> “Firmware Upgrade” to upgrade to 1.0.7.5.
- For all DP series, once upgraded to 1.0.7.5 or above, downgrading to 1.0.4.10 or previous firmware version is not supported.
- For DP750/DP720/DP760, once upgraded to 1.0.4.2 or above, downgrading to 1.0.3.40 or previous firmware version is not supported.
- For DP750/DP720, once upgraded to 1.0.3.23 or above, downgrading to 1.0.3.17 or previous firmware version is not supported.
- For DP750/DP720, once upgraded to 1.0.3.17 or above, downgrading to 1.0.2.16 or previous firmware version is not supported.
- For DP750/DP720, once upgraded to 1.0.2.16 or above, downgrading to 1.0.1.x or previous firmware version is not supported.
- For DP750/DP720, firmware 1.0.0.99 (the first public released beta firmware) and all above can upgrade to 1.0.4.7 directly, no transitional firmware is needed.

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FIRMWARE VERSION 1.0.19.2

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

02/24/2023

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 92e95bb60d96ac61f350617553322353

ENHANCEMENT

- Added support for XMPP connection on TR-069.
- Added option to disable with # key. [USE # AS REDIAL KEY]
- Added preferred primary outbound proxy feature. [PREFER PRIMARY OUTBOUND PROXY]
- Integrated new DigiCert certificates in firmware.
- Added support for the Greek language.
- Added new option "Follow SIP Transport"(P183/443/543/643) to SRTP mode to auto-switch between RTP and SRTP according to the resolved SIP transport. (Profiles->Profile x->Audio Settings->SRTP Mode-> "4 – Follow SIP Transport")
- Added the ability to configure F-timer and B-timer.[SIP TIMER F AND TIMER B]
- Added support DP75x for new Cisco/BW Failover/Failback Requests. [MAXIMUM NUMBER OF SIP REQUEST RETRIES] [FAILBACK EXPIRATION]
- Updated the default value of "Unregister on Reboot"(P81/752/552/611) to "No". (Profiles->Profile x->SIP Settings-> Basic Settings-> Unregister on Reboot-> "0 – No")
- Updated and added in XSI Settings.[XSI SETTINGS] [XSI ACCOUNT ASSIGNMENT]
- Modified "SIP User Name" to "SIP User ID"
- Added in "Missed Call Log", "Placed Call Log", and "Placed Call Log" in Network Directories. [NETWORK DIRECTORIES: CALL LOG]

BUG FIX

- Fixed the handset won't update the caller ID when Caller ID Display is set to Auto while doing a blind transfer.
- Fixed DP752/DP730 cannot add contacts from history if caller number starts with "+".
- Fixed DP720 cannot auto-upgrade firmware even if the feature is enabled on DP75x Web UI.
- Fixed when registration fails over to an SBC other than the one currently registered with, a new call-id must be used.

- Fixed account loss registration and unable to re-registered to any SBC.
- Fixed an issue that causes the on-hold reminder tone is not playing.
- Fixed the “Attended Transfer” not to use the line that is currently calling.
- Fixed an issue that the device repeats to reboot when config a domain name in outbound proxy under Telstra mode.

NEW FEATURES OVERVIEW

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

USE # AS REDIAL KEY

- **Functionality**

The device now supports to configure the Use # as Redial Key, If set to “Yes”, the “#” key will immediately redial the number of the last outgoing call, if no digits have been dialed. The setting can be found at Profiles -> Profile x -> Call Settings. By default, it is enabled.

Use # as Dial Key ☐ No ☒ Yes

Use # as Redial Key ☐ No ☒ Yes

On Hold Reminder Tone ☐ Disabled ☒ Enabled

- **New P Values**

Pvalue	Alias	Description	Value range	Default
29069	profile.1.call.pound_as_redial	Configure the Use # as Redial Key	0 – No 1 – Yes	1
29169	profile.2.call.pound_as_redial			
29269	profile.3.call.pound_as_redial			
29369	profile.4.call.pound_as_redial			

PREFER PRIMARY OUTBOUND PROXY

- **Functionality**

The device now supports to configure the Prefer Primary Outbound Proxy, if yes, the profile will register with the primary server when the failover server registration expires. The setting can be found at Profiles -> Profile x -> General Settings. By default, it is disabled.

Backup Outbound Proxy

Prefer Primary Outbound Proxy ☒ No ☐ Yes

Voice Mail Access Number

- **New P Values**

Pvalue	Alias	Description	Value range	Default
28096	profile.1.prefer_primary_outbound_proxy	Configure Prefer	0 – No	0
28097	profile.2.prefer_primary_outbound_proxy	Primary Outbound	1 – Yes	
28098	profile.3.prefer_primary_outbound_proxy	Proxy		
28099	profile.4.prefer_primary_outbound_proxy			

SIP TIMER F AND TIMER B

- **Functionality**

The device now supports to configure SIP Timer F and Timer B, which is defined in RFC 3261. The setting can be found at Profiles -> Profile x -> SIP Settings-> Basic Settings. By default, the value is 0.

SIP TIMER D

SIP TIMER F

SIP TIMER B

Enable OPTIONS Keep Alive ☒ No ☐ Yes

- **New P Values**

Pvalue	Alias	Description	Value range	Default
60075	profile.1.sip.timerf.interval	Configure the F-timer Timeout	Number:	0
60175	profile.2.sip.timerf.interval		0 – 128	
60275	profile.3.sip.timerf.interval		(second(s))	
60375	profile.4.sip.timerf.interval			
60074	profile.1.sip.timerb.interval	Configure the B-timer Timeout	Number:	0
60174	profile.2.sip.timerb.interval		0 – 128	
60274	profile.3.sip.timerb.interval		(second(s))	
60374	profile.4.sip.timerb.interval			

MAXIMUM NUMBER OF SIP REQUEST RETRIES

- **Functionality**

The device now supports to configure the maximum number of SIP request retries, which is the number of SIP retries before DNS failover. The setting can be found at Profiles -> Profile x -> Network Settings -> DNS Settings. By default, the value is 2.

Register Before DNS SRV Failover ☒ No ☐ Yes

Maximum Number of SIP Request Retries

Failback Expiration(m)

Primary IP

- **New P Values**

Pvalue	Alias	Description	Value range	Default
60055	profile.1.network.sip.number_of_retries	Number of SIP retries before DNS failover	Number: 0 – 10	2
60155	profile.2.network.sip.number_of_retries			
60255	profile.3.network.sip.number_of_retries			
60355	profile.4.network.sip.number_of_retries			

FAILBACK EXPIRATION

- **Functionality**

The device now supports to configure failback expiration, which is the expiration time in minutes when using BroadSoft R23 failback mode. The setting can be found at Profiles -> Profile x -> Network Settings -> DNS Settings. By default, the value is 60.

Maximum Number of SIP Request Retries

Failback Expiration(m)

Primary IP

- **New P Values**

Pvalue	Alias	Description	Value range	Default
60056	profile.1.network.dns.failback.expiration_timer	Expiration time in minutes when using BroadSoft R23 failback mode	Number: 1 – 64800 (minute(s))	60
60156	profile.2.network.dns.failback.expiration_timer			
60256	profile.3.network.dns.failback.expiration_timer			
60356	profile.4.network.dns.failback.expiration_timer			

XSI SETTINGS

- **Functionality**

The device now supports configuring XSI accounts 1-10 in XSI Server, XSI Port, XSI Actions Path, Broadsoft Contact Download Interval, XSI Authentication Type, Login Username, Login Password, SIP User ID, SIP Auth ID, and SIP Password. The setting can be found at Phonebook-> Global Phonebook Broadsoft XSI Settings-> XSI.

XSI

Account	Server	Port	XSI Actions Path	Broadsoft Contact Download Interval
Account 1	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 2	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 3	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 4	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 5	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 6	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 7	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 8	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 9	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>
Account 10	<input type="text"/>	<input type="text" value="80"/>	<input type="text"/>	<input type="text" value="4320"/>

Login Credentials

Account	XSI Authentication Type	Login Username	Login Password	SIP User ID	SIP Auth ID	SIP Password
Account 1	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 2	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 3	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 4	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 5	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 6	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 7	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 8	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 9	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
Account 10	Login Credentials ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

• New P Values

Pvalue	Alias	Description	Value range	Default
1591	broadsoft.xsi.server.1	Broadsoft XSI Server address with protocol.	Number: 0,1,2 0 - Login Credentials, 1 - SIP Credentials, 2 - Match SIP Account.	0
1597	broadsoft.xsi.server.2			
1603	broadsoft.xsi.server.3			
1609	broadsoft.xsi.server.4			
1615	broadsoft.xsi.server.5			
1621	broadsoft.xsi.server.6			
1627	broadsoft.xsi.server.7			
1633	broadsoft.xsi.server.8			
1639	broadsoft.xsi.server.9			
1645	broadsoft.xsi.server.10			

1592	broadsoft.xsi.port.1	Port of the Broadsoft XSI server	Number: 1 - 65535	80
1598	broadsoft.xsi.port.2			
1604	broadsoft.xsi.port.3			
1610	broadsoft.xsi.port.4			
1616	broadsoft.xsi.port.5			
1622	broadsoft.xsi.port.6			
1628	broadsoft.xsi.port.7			
1634	broadsoft.xsi.port.8			
1640	broadsoft.xsi.port.9	If it is empty, the path "com.broadsoft.xsi-actions" will be used	String	(null)
1646	broadsoft.xsi.port.10			
27937	broadsoft.xsi.XSIActions.1			
27938	broadsoft.xsi.XSIActions.2			
27939	broadsoft.xsi.XSIActions.3			
27940	broadsoft.xsi.XSIActions.4			
27941	broadsoft.xsi.XSIActions.5			
27942	broadsoft.xsi.XSIActions.6			
27943	broadsoft.xsi.XSIActions.7	Configures the broadsoft phonebook download interval (in minutes). If set to 0, automatic download will be disabled. Valid range is 5 to 4320.	Number: 0, 5 - 4320	4320
27944	broadsoft.xsi.XSIActions.8			
27945	broadsoft.xsi.XSIActions.9			
27946	broadsoft.xsi.XSIActions.10			
27947	broadsoft.xsi.downloadInterval.1			
27948	broadsoft.xsi.downloadInterval.2			
27949	broadsoft.xsi.downloadInterval.3			
27950	broadsoft.xsi.downloadInterval.4			
27951	broadsoft.xsi.downloadInterval.5	Select XSI Authentication Type. SIP User ID need to be configured if SIP account is selected	Number: 0,1,2 0 - Login Credentials, 1 - SIP Credentials, 2 - Match SIP Account	0
27952	broadsoft.xsi.downloadInterval.6			
27953	broadsoft.xsi.downloadInterval.7			
27954	broadsoft.xsi.downloadInterval.8			
27955	broadsoft.xsi.downloadInterval.9			
27956	broadsoft.xsi.downloadInterval.10			
27897	broadsoft.xsi.authType.1			
27898	broadsoft.xsi.authType.2			
27899	broadsoft.xsi.authType.3			
27900	broadsoft.xsi.authType.4			
27901	broadsoft.xsi.authType.5			
27902	broadsoft.xsi.authType.6			
27903	broadsoft.xsi.authType.7			
27904	broadsoft.xsi.authType.8			
27905	broadsoft.xsi.authType.9			
27906	broadsoft.xsi.authType.10			

1593	broadsoft.xsi.username.1	Configure the Username for the BroadWorks XSI server	String	(Null)
1599	broadsoft.xsi.username.2			
1605	broadsoft.xsi.username.3			
1611	broadsoft.xsi.username.4			
1617	broadsoft.xsi.username.5			
1623	broadsoft.xsi.username.6			
1629	broadsoft.xsi.username.7			
1635	broadsoft.xsi.username.8			
1641	broadsoft.xsi.username.9	Configure the Password for the BroadWorks XSI server	String	(Null)
1647	broadsoft.xsi.username.10			
1594	broadsoft.xsi.password.1			
1600	broadsoft.xsi.password.2			
1606	broadsoft.xsi.password.3			
1612	broadsoft.xsi.password.4			
1618	broadsoft.xsi.password.5			
1624	broadsoft.xsi.password.6			
1630	broadsoft.xsi.password.7	Configure SIP User ID for the BroadWorks XSI server.	String	(Null)
1636	broadsoft.xsi.password.8			
1642	broadsoft.xsi.password.9			
1648	broadsoft.xsi.password.10			
27907	broadsoft.xsi.SIPUsername.1			
27908	broadsoft.xsi.SIPUsername.2			
27909	broadsoft.xsi.SIPUsername.3			
27910	broadsoft.xsi.SIPUsername.4			
27911	broadsoft.xsi.SIPUsername.5	Configure SIP Auth ID for the BroadWorks XSI server.	String	(Null)
27912	broadsoft.xsi.SIPUsername.6			
27913	broadsoft.xsi.SIPUsername.7			
27914	broadsoft.xsi.SIPUsername.8			
27915	broadsoft.xsi.SIPUsername.9			
27916	broadsoft.xsi.SIPUsername.10			
27917	broadsoft.xsi.SIPId.1			
27918	broadsoft.xsi.SIPId.2			
27919	broadsoft.xsi.SIPId.3			
27920	broadsoft.xsi.SIPId.4			
27921	broadsoft.xsi.SIPId.5			
27922	broadsoft.xsi.SIPId.6			
27923	broadsoft.xsi.SIPId.7			
27924	broadsoft.xsi.SIPId.8			
27925	broadsoft.xsi.SIPId.9			
27926	broadsoft.xsi.SIPId.10			

27927	broadsoft.xsi.SIPPassword.1	Configure SIP Password for the BroadWorks XSI server.	String	(Null)
27928	broadsoft.xsi.SIPPassword.2			
27929	broadsoft.xsi.SIPPassword.3			
27930	broadsoft.xsi.SIPPassword.4			
27931	broadsoft.xsi.SIPPassword.5			
27932	broadsoft.xsi.SIPPassword.6			
27933	broadsoft.xsi.SIPPassword.7			
27934	broadsoft.xsi.SIPPassword.8			
27935	broadsoft.xsi.SIPPassword.9			
27936	broadsoft.xsi.SIPPassword.10			

XSI ACCOUNT ASSIGNMENT

- Functionality**

The device now supports configuring “Call log Type” in Account 1 to 10 on Handset 1 to 5. The setting can be found at Phonebook-> Global Phonebook Broadsoft XSI Settings-> XSI ACCOUNT ASSIGNMENT. By default, XSI Account is Account 1 and Call Log Type is “Use XSI Call Logs”

XSI Account Assignment

Handset	XSI Account	Call Log Type
HS1	Account 1 ▼	Use XSI Call Logs ▼
HS2	Account 1 ▼	Use XSI Call Logs ▼
HS3	Account 1 ▼	Use XSI Call Logs ▼
HS4	Account 1 ▼	Use XSI Call Logs ▼
HS5	Account 1 ▼	Use XSI Call Logs ▼

- New P Values**

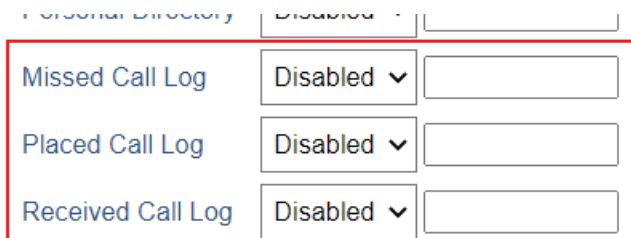
Pvalue	Alias	Description	Value range	Default
27957	broadsoft.xsi.handset.1	XSI Account Assignment	Number: 1 to 10. 1 - Account 1, 2 - Account 2, 3 - Account 2, 4 - Account 4, 5 - Account 5, 6 - Account 6, 7 - Account 7,	1
27958	broadsoft.xsi.handset.2			
27959	broadsoft.xsi.handset.3			
27960	broadsoft.xsi.handset.4			
27961	broadsoft.xsi.handset.5			

			8 - Account 8, 9 - Account 9, 10 - Account 10.	
27967 27968 27969 27970 27971	broadsoft.xsi.callLogDisplay.1 broadsoft.xsi.callLogDisplay.2 broadsoft.xsi.callLogDisplay.3 broadsoft.xsi.callLogDisplay.4 broadsoft.xsi.callLogDisplay.5	Select whether to display Base Station or XSI call logs on handset; will display Base logs if XSI not configured	Number: 0, 1 0 - Use Base Station Call Logs, 1 - Use XSI Call Logs.	0

NETWORK DIRECTORIES: CALL LOG

- Functionality**

The device now supports configuring enable or disable “Missed Call Log”, “Placed Call Log” and “Received Call log” and the name of them. The setting can be found at Phonebook-> Global Phonebook Broadsoft XSI Settings-> Network Directories. By default, it is disabled.



Pvalue	Alias	Description	Value range	Default
2981	broadsoft.xsi.enableMissedCallLog	Enable or Disable Missed Call Log	Number: 0,1	0
2982	broadsoft.xsi.missedCallLog	Name of Personal Directory	String: Max length 128	Null
2983	broadsoft.xsi.enablePlacedCallLog	Enable or Disable Placed Call Log	Number: 0,1	0
2984	broadsoft.xsi.placedCallLog	Name of Placed Call Log	String: Max length 128	Null
2985	broadsoft.xsi.enableReceivedCallLog	Enable or Disable Received Call Log	Number: 0,1	0
2986	broadsoft.xsi.receivedCallLog	Name of Received Call Log	String: Max length 128	Null

FIRMWARE VERSION 1.0.17.8

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

07/29/2022

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 5f247bdd452e2807b0b0c27a30c5a326

ENHANCEMENT

- Added support for firmware download when the server does not support “206 Partial Content”.
- Changed “broadsoft.xsi.SIPPassword” to accept up to 64 characters (previous is 32 characters).
- Updated CPE to version 1.0.4.25 for GDMS support.

BUG FIX

- Fixed some DP722 used a wrong battery calibration that may lead them cannot be fully charged.
- Fixed DNS SRV failover does not work on receiving SIP 503.
- Fixed Offhook Auto-dial option value changed to 0 after handset factory reset.
- Fixed DP75x may not register accounts with TR-069 enabled.
- Fixed issue that in a conference call after put Party A on hold, if A hang up and re-joined the conference, Party B will be displayed as “Holding” status.
- Fixed incoming call with special characters would cause the handset to restart.
- Fixed DP75x crash after receiving SIP INVITE with name including special Czech characters
- Fixed DP75x crash and account unregister upon a PBX based transfer using *230 feature code.
- Fixed phonebook contacts in Japanese displayed as garbled characters on incoming call when HS mode is set to “Parallel”.
- Fixed issue that when in parallel mode, the incoming caller firstname or lastname is too long, there is only one handset can display the name.

FIRMWARE VERSION 1.0.17.4

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

04/06/2022

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: b35a2214537c3852ce5ef8f3efd9f57a

BUG FIX

- Fixed some DP722 used a wrong battery calibration that may lead them cannot be fully charged.

FIRMWARE VERSION 1.0.17.3

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

02/22/2022

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 6feb9d08bd1ccec4a12db80b74dba32f

ENHANCEMENT

- Add support for Off-Hook Auto-Dial Delay. [OFF-HOOK AUTO-DIAL DELAY]
- Added ability to manage 3-way conference from handset. [OPERATION FOR 3-WAY CONFERENCE]

BUG FIX

- Fixed the Phonebook Download Interval does not take effect if it is set to 0.
- Updated CA certificate to fix Authenticate Server Certificate Chain enabled and account cannot be registered issue.
- Fixed DP720 cannot auto upgrade firmware even the feature is enabled on DP75x Web UI.
- Fixed DP75x has low chance to take too long to complete the upgrade process.
- Fixed when enabled both HTTP and HTTPS and set the port to be the same, access to web page is problematic.

NEW FEATURES OVERVIEW

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

OFF-HOOK AUTO-DIAL DELAY

- **Functionality**

The device now supports to configure a delay time for off-hook auto-dial, after configured, when device off-hook playing dial tone, after the configured time of seconds, the device will auto dial to the configured number. The setting can be found at DECT -> General Settings -> Handset Settings. By default, the value is 0.

Handset Settings

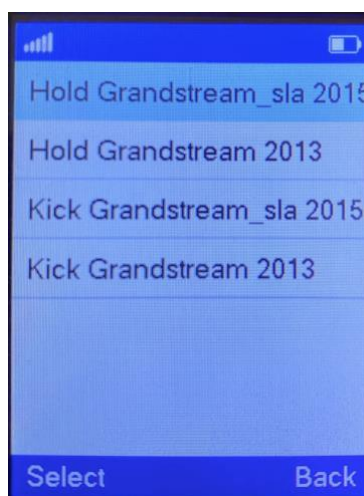
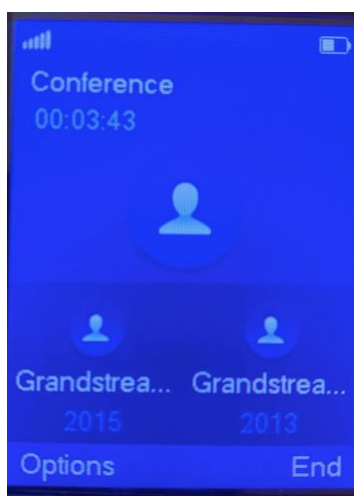
Handset	Handset Name	Handset Phonebook	Offhook Auto-dial	Off-Hook Auto-Dial Delay
HS1	<input type="text" value="HS1"/>	<input type="text" value="PB1"/>	<input type="text"/>	<input type="text" value="0"/>
HS2	<input type="text" value="HS2"/>	<input type="text" value="PB2"/>	<input type="text"/>	<input type="text" value="0"/>
HS3	<input type="text" value="HS3"/>	<input type="text" value="PB3"/>	<input type="text"/>	<input type="text" value="0"/>
HS4	<input type="text" value="HS4"/>	<input type="text" value="PB4"/>	<input type="text"/>	<input type="text" value="0"/>
HS5	<input type="text" value="HS5"/>	<input type="text" value="PB5"/>	<input type="text"/>	<input type="text" value="0"/>

- New P Values**

Pvalue	Alias	Description	Value range	Default
2296	handset.1.call.offhook_auto_dial.delay	Delay time in seconds for offhook auto-dial.	0 - 60	0
2496	handset.2.call.offhook_auto_dial.delay			
2596	handset.3.call.offhook_auto_dial.delay			
2696	handset.4.call.offhook_auto_dial.delay			
2796	handset.5.call.offhook_auto_dial.delay			

OPERATION FOR 3-WAY CONFERENCE

Since 1.0.17.3, if the handset bridged 2 calls into a conference, he now able to kick one of the participants without ending the whole conference, or he can also hold and unhold participants without kicking them off the call.



FIRMWARE VERSION 1.0.17.1

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

11/17/2021

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 399ab9ae931d29b0e37ff8d5ef32510d

IMPORTANT UPGRADING NOTES

- **For all DP series, once upgraded to 1.0.17.1, downgrade to 1.0.15.6 or below is not supported.**

ENHANCEMENT

- Add support to automatically reboot. [AUTOMATICALLY REBOOT]
- Corrected the tooltip text for "Download and Process ALL Available Config Files".

BUG FIX

- Removed XSI option from "Global Phonebook Type" because it not needed.
- Fixed TFTP upgrade of firmware still happens if DHCP options 43, 66 and 160 override the server setting. The TFTP upgrade does not support partial download which may cause the firmware download and upgrade problem that DP has already removed it.
- Fixed DP72x/DP730 cannot play distinctive ringtone for Metaswitch platform.
- Fixed DP75x Call Forward settings on DP72x/DP730 cannot synchronize with Metaswitch server when the destination field is not set to null value.

NEW FEATURES OVERVIEW

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

AUTOMATICALLY REBOOT

- **Functionality**

The device now supports to config a specific time to auto reboot by its own now. The setting can be found at Maintenance -> Automatic. By default, this feature is disabled.

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings +

Date and Time

Syslog

Packet Capture

Factory Reset

Automatic Reboot

Support +

Automatic Reboot

No ☒

Yes, reboot daily ☐

Yes, reboot every week on ☐ Sunday v

Yes, reboot every month on day ☐ 1 v

Save
Save and Apply
Reset

- New P Values**

Pvalue	Alias	Description	Value range	Default
21929	maintenance.autoreboot.type	Configure the type of auto reboot.	0 – No 1 – Yes, reboot daily 2 – Yes, reboot every week on 3 – Yes, reboot every month on day	0
21931	maintenance.autoreboot.wday	Week day, this will take effect if type is set to Yes, reboot every week on	0 – 6: Sunday to Saturday	
28118	maintenance.autoreboot.mday	Day of a month, this will take effect if type is set to Yes, reboot every month on day	1 – 31: 1 st day to 31 st day	
21930	maintenance.autoreboot.hour	Hour of the day, this will take effect if auto reboot is enabled.	0 – 23	
43100	maintenance.autoreboot.minute	Minute of the day, this will take effect if auto reboot is enabled.	0 – 59	

FIRMWARE VERSION 1.0.15.15

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

09/24/2021

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: f754bc575501baacb1f63bdd094c76e3

ENHANCEMENT

- Add support to allow call barging by Call-Info/Alert-Info. [BARGING BY CALL-INFO/ALERT-INFO]
- Add HW version display on device web GUI status page.
- Adjust Opus to be configured with default value 123 as the same as GRP.

BUG FIX

- Fixed DP75x SIP INVITE's "Via", "Contact", "Owner" and "Connection information" headers are all IP address 0.0.0.0.
- Fixed Bypass Dial Plan not working after enabled.
- Fixed DP72x/DP730 stop receiving calls randomly.
- Fixed DP75x may repeat automatic reboot after interrupt power when updating Core or Base image.
- Fixed Russian input setting is reversed on handset.
- Fixed device sometimes does not re-register after unplugging/plugging network cable.
- Fixed the issue when the device configured to use SRV DNS mode, but the host is under A record, the custom SIP port in Outbound Proxy cannot be correctly recognized, the device will use the default SIP port.

NEW FEATURES OVERVIEW

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

BARGING BY CALL-INFO/ALERT-INFO

- **Functionality**

The device now supports to auto answer a call by Call-Info/Alert-Info, even the phone is current in middle of a call, the setting can be found at Profiles -> Profile x -> Call Settings -> Allow Barging for Auto Answer by Call-Info/Alert-Info. By default, it's enabled.

Profiles

- Profile 1
- General Settings
- Network Settings
- SIP Settings
- Audio Settings
- Call Settings
- Profile 2
- Profile 3
- Profile 4

Call Settings

Early Dial ☒ No ☐ Yes

Dial Plan Prefix

Dial Plan { x+ | *x+ | *xx*x+ }

Bypass Dial Plan ☒ No ☐ Yes

Use # as Dial Key ☐ No ☒ Yes

On Hold Reminder Tone ☐ Disabled ☒ Enabled

Match Incoming Caller ID

Matching Rule

Handset RingTone 1 ▼

Handset RingTone 1 ▼

Handset RingTone 1 ▼

Allow Auto Answer by Call-Info/Alert-Info ☐ No ☒ Yes

Custom Alert-Info for Auto Answer

Allow Barging for Auto Answer by Call-Info/Alert-Info ☐ No ☒ Yes

Enable Call Features ☐ No ☒ Yes

Disable Call Waiting Caller ID ☒ No ☐ Yes

Enable Call Waiting in Parallel Mode ☒ No ☐ Yes

- New P Values**

Pvalue	Alias	Description	Value range	Default
27859	profile.1.call.auto_answer.al	If set to "Yes", the phone will automatically answer based on SIP Call-Info/Alert-Info, even if the phone is already in another call. If Allow Auto Answer by Call-Info/Alert-Info is set to "No", this setting has no effect.	0 – No 1 – Yes	1
27860	ert_info_barging.allow			
27861	profile.2.call.auto_answer.al			
27862	ert_info_barging.allow			
	profile.3.call.auto_answer.al			
	ert_info_barging.allow			
	profile.4.call.auto_answer.al			
	ert_info_barging.allow			

FIRMWARE VERSION 1.0.15.6

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

03/18/2021

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 53393f6a0645b02f875eeca35d2c6dd

ENHANCEMENT

- Add support for Thailand language.

BUG FIX

- Fixed DP75x Dial Plan priority sequence is reversed issue.
- Fixed DP75x IP address/subnet/gateway/DNS becomes to 0.0.0.0 after set SIP TLS Private Key.
- Fixed DP75x Layer 2 QoS 802.1p Priority Value for virtual IP do not take effect.
- Fixed DP75x issue that when the SRTP is disabled but the SRTP parameters are still negotiated.
- Fixed DP75x issue when the domain name is resolved under SRV, the device will send additional registration requests after booting up.
- Fixed DP75x doesn't send a SUBSCRIBE message when DND is enabled/disabled by the mute button.
- Changed the default values of local RTP port starting from 50000.

FIRMWARE VERSION 1.0.15.3

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

12/11/2020

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 76b77dd3160de0757e2e855acd71e166

ENHANCEMENT

- Added support to have a web UI option of bypass dial plan. [BYPASS DIAL PLAN]

BUG FIX

- Fixed the bug that disconnect the upper network of the switch and STUN enabled and configured, the DP75x might restart of a hung thread issue.
- Fixed DP75x performs INVITE to Transfer-Target before REFER on Blind Transfer.
- Fixed DP75x has 6 seconds delay for sending the DHCP renewal.
- Fixed DP75x issue that when the domain name is resolved under SRV, and the device will send additional registration requests after restarting.
- Fixed some translation error in web UI.

NEW FEATURES OVERVIEW

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

BYPASS DIAL PLAN

- **Functionality**

The device now supports to bypass the dial plan by a web UI option, the setting can be found at Profiles -> Profile x -> Call Settings -> Bypass Dial Plan.

Profiles

- Profile 1
 - General Settings
 - Network Settings
 - SIP Settings
 - Audio Settings
 - Call Settings
- Profile 2
- Profile 3
- Profile 4

Call Settings

Early Dial ☒ No ☐ Yes

Dial Plan Prefix

Dial Plan { x+ | *x+ | *xx*x+ }

Bypass Dial Plan ☒ No ☐ Yes

Use # as Dial Key ☐ No ☒ Yes

On Hold Reminder Tone ☐ Disabled ☒ Enabled

Match Incoming Caller ID ☐ Matching Rule

- New P Values**

Pvalue	Alias	Description	Value range	Default
2382	profile.1.call.disable_dial_plan	Bypass Dial Plan on selected items	0 – No	0
2482	profile.2.call.disable_dial_plan		1 – Yes	
2582	profile.3.call.disable_dial_plan			
2682	profile.4.call.disable_dial_plan			

FIRMWARE VERSION 1.0.15.2

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

11/06/2020

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: eac554f1da12c935364a2b5531cc91e5

ENHANCEMENT

- Added support to input provisioning server username and password on handset UI when base station is trying to download a config file from a password protected server. [INPUT USERNAME AND PASSWORD ON HANDSET FOR PROVISION SERVER]
- Added support to configure Call Log settings. [CONFIGURE CALL LOG]
- Added DNS SRV Failover modes. [DNS SRV FAILOVER MODES]
- Added warning tone when auto answer a call or intercom. [WARNING TONE FOR AUTO ANSWER]
- Added option to automatically mute after auto answer a call or intercom. [WARNING TONE FOR AUTO ANSWER]
- Added support to disable SIP NOTIFY authentication.
- Added support for Auto Timer Zone feature.
- Added support to delete a paired base station from the handset.

BUG FIX

- Fixed a security issue that an XML field can be used as command injection.
- Fixed a security issue that user may get root access in the SSH shell with "gssu" command.
- Fixed DP75x cannot apply some GDMS settings issue.
- Fixed DP75x issue that when access to VM, it calls the number of Offhook Auto-dial number.
- Moved "No Key Entry Timeout" setting from Profile -> Call Setting to DECT -> Handset Settings.
- Fixed intercom has no missed call notification issue.
- Fixed DP75x does not report DND status to GDMS issue.
- Fixed DP75x cannot search the contacts in LDAP.
- Fixed DP72x and DP730 shows "error" when edit private contact.
- Fixed DP75x does not response 200OK after receiving the Reboot NOTIFY from UCM.
- Fixed DP75x still shows remote caller name in call history even after added it to contacts with a new name.

- Fixed DP75x still send INVITE when Dial Plan is {6868xxxx} and trying to call 68681 that does not match the Dial Plan.
- Fixed DP75x cannot dial out via Offhook Auto-dial when the line 1 is none.
- Fixed DP75x Force Timer do not take effect for session timer.
- Fixed DP75x will response 404 when it receive 'sipaaa' and enabled Validate Incoming Messages.
- Added prompt of "include the protocol header ("http://" or "https://")" for Global Phonebook Broadsoft XSI Settings.
- Fixed DP75x still shows cached Broadsoft XSI contacts after changed Broadsoft XSI settings to invalid one.
- Fixed DP75x need a reboot to take DNS Mode -> Use Configured IP to take effect.
- Fixed when DP75x dial back from missed call, the call history of the call back shows under incoming call instead of outgoing call.
- Fixed the user account cannot configure "Whitelist for WAN side" and "Blacklist for WAN side".
- Fixed DNS Server2 in SSH status result is not aligned with other values.
- Fixed DP75x "use stun to detect network connectivity" does not take effect.
- Fixed DP75x "OPTIONS Keep Alive Max Lost" does not take effect.
- Fixed DP75x OpenVPN cannot disable comp-lzo.
- Fixed DP72x and DP730 will remove account DND settings after use Mute key to disable the handset DND.
- Fixed DP75x is not ringing when the call is received from PSTN using an Alcatel PBX.
- Fixed DP75x take rejected calls to missed call history, now it will be logged as received/answered call with a different icon.
- Fixed DP75x does not accept a SIP server with more than 4 sub domains.
- Fixed DP75x cannot redial an IP call from call history.
- Fixed DP72x/DP730 still display a null home number at the contact detail view after home number been removed and a work number been configured.




NEW FEATURES OVERVIEW

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

INPUT USERNAME AND PASSWORD ON HANDSET FOR PROVISION SERVER

- **Functionality**

When the device is trying to download the config file from the config server, if the server has username and password protected, the end user needs to configure these fields in the web UI before the provisioning.

Provisioning

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings +

Date and Time

Syslog

Packet Capture

Factory Reset

Support +

XML Config File Password

HTTP/HTTPS/FTP/FTPS User Name
 HTTP/HTTPS/FTP/FTPS Password

Always send HTTP Basic Authentication Information ☒ No ☐ Yes

Verify host when using SSL ☒ No ☐ Yes

Provisioning via
☐ TFTP ☒ HTTP ☐ HTTPS ☐ FTP ☐ FTPS

Config Server Path

Config File Prefix

Config File Postfix

Now it can input the username and password on handset UI when trying to download the config file if these 2 fields is not preconfigured in Web UI.

CONFIGURE CALL LOG

- Functionality**

DP75x now support to configure the level of call logs or disable the call log. The feature can be found at Web UI
 -> Profile X -> Call Settings -> Call log:

Call Log

☒ Log All Calls
☐ Log Incoming/Outgoing Only (missed calls NOT recorded)
☐ Disable Call Log

Save

Save and Apply

Reset

- New P Values**

Pvalue	Alias	Description	Value range	Default
182	profile.1.call.call_log	Configures Call Log setting on the phone.	0 – Log All Calls	0
442	profile.2.call.call_log		1 – Log Incoming/Outgoing Only (missed call NOT recorded)	
542	profile.3.call.call_log		2 – Disabled Call Log	
642	profile.5.call.call_log			

DNS SRV FAILOVER MODES

- Functionality**

This feature is used to configure the preferred IP mode for DNS SRV. If selected "default", first IP from query result will be applied; If selected "Saved one until DNS TTL", previous IP will be applied before reaches DNS timeout; If selected "Saved one until no response", previous IP will be applied even after DNS time out until it cannot response.

When the DNS SRV Failover Mode is enabled, you can also choose to "Register before DNS SRV failover" that can waive the 3 failed tries, or still try 3 times then use the failover DNS.

This setting can be found at Web UI -> Profiles -> Network Settings:

DNS Settings

DNS Mode

DNS SRV Failover Mode

Register Before DNS SRV Failover

Primary IP

Backup IP 1

Backup IP 2

DNS SRV use Registered IP

A Record ▾

Saved one until DNS TTL ▾

☐ No
 ☒ Yes

☐ No
 ☒ Yes

• New P Values

Pvalue	Alias	Description	Value range	Default
26040	profile.1.network.dns.srv_failover_mode	Configure the DNS SRV failover mode	0 – Default 1 – Saved one until DNS TTL 2 – Saved one until no response	0
26140	profile.2.network.dns.srv_failover_mode			
26240	profile.3.network.dns.srv_failover_mode			
26340	profile.4.network.dns.srv_failover_mode			
29095	profile.1.network.dns.register_before_srv_failover	Indicates whether a REGISTER request will be initiated when a server failover occurred under DNS SRV mode	0 – No 1 – Yes	0
29195	profile.2.network.dns.register_before_srv_failover			
29295	profile.3.network.dns.register_before_srv_failover			
29395	profile.4.network.dns.register_before_srv_failover			

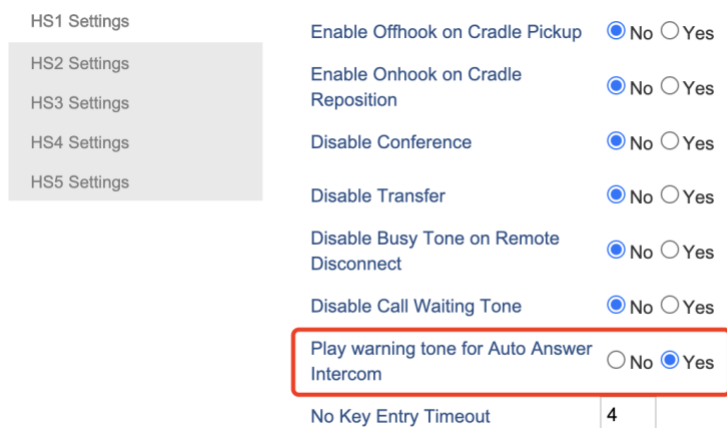
WARNING TONE FOR AUTO ANSWER

- Functionality**

DP75x support to play a warning tone for auto answer for protect the privacy. This setting can be configured by either the handset GUI or the Web UI:

Handset GUI -> Preferences -> Play Warning Tone for Auto Answer.

Web UI -> Handset Settings -> HS X Settings.



HS1 Settings

HS2 Settings

HS3 Settings

HS4 Settings

HS5 Settings

Enable Offhook on Cradle Pickup ☒ No ☐ Yes

Enable Onhook on Cradle Reposition ☒ No ☐ Yes

Disable Conference ☒ No ☐ Yes

Disable Transfer ☒ No ☐ Yes

Disable Busy Tone on Remote Disconnect ☒ No ☐ Yes

Disable Call Waiting Tone ☒ No ☐ Yes

Play warning tone for Auto Answer Intercom ☐ No ☒ Yes

No Key Entry Timeout 4

- New P Values**

Pvalue	Alias	Description	Value range	Default
26072	handset.1.call.play_warning_tone_for_auto_an	When enabled, phone will play warning tone when auto answer Intercom	0 – No 1 – Yes	0
26172	swer.enable			
26272	handset.2.call.play_warning_tone_for_auto_an			
26372	swer.enable			
26472	handset.3.call.play_warning_tone_for_auto_an			
	swer.enable			
	handset.4.call.play_warning_tone_for_auto_an			
	swer.enable			
	handset.5.call.play_warning_tone_for_auto_an			
	swer.enable			

AUTO MUTE FOR AUTO ANSWER

- Functionality**

When auto an incoming call or intercom, now DP72x/DP730 support to auto mute right after the call been answer, this can be configured by either the handset GUI or Web UI:

Handset GUI -> Preferences -> Play Warning Tone for Auto Answer.

Web UI -> Handset Settings -> HS X Settings.

DECT

- General Settings
- SIP Account Settings
- Handset Line Settings
- Handset Settings
- HS1 Settings**
- HS2 Settings
- HS3 Settings
- HS4 Settings
- HS5 Settings

HS1 Settings

Enable Auto Answer

☒ No ☐ Yes

Enable Mute for Auto Answer

☐ No ☒ Yes

Enable Offhook on Cradle Pickup

☒ No ☐ Yes

Enable Onhook on Cradle Reposition

☒ No ☐ Yes

Disable Conference

☒ No ☐ Yes

Disable Transfer

☒ No ☐ Yes

- New P Values**

Pvalue	Alias	Description	Value range	Default
27844	handset.1.call.auto_mute_for_auto_ans	Enable mute when handset is auto answering	0 – No 1 – Yes	0
27845	wer.enable			
27846	handset.2.call.auto_mute_for_auto_ans			
27847	wer.enable			
27848	handset.3.call.auto_mute_for_auto_ans			
	wer.enable			
	handset.4.call.auto_mute_for_auto_ans			
	wer.enable			
	handset.5.call.auto_mute_for_auto_ans			
	wer.enable			

FIRMWARE VERSION 1.0.13.8

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

07/10/2020

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: f54e5264efa0ed7e5702c8263a3cdc86

ENHANCEMENT

- Added support for handsets to internally page other handsets without accounts registered on menu -> Call features -> paging.
- Added support to enable or disable “On Hold Reminder Tone” in Web UI. [ON HOLD REMINDER TONE]
- Added support to configure “Caller ID Display” in Web UI.
- Updated some Chinese translations.
- Updated the CPE to 1.0.1.141 to better support GDMS.

BUG FIX

- Fixed DP7xx changes hold status to incoming call status when remote also hold the call.
- Fixed DP7xx handsets failed to ring during paging call.
- Fixed DP7xx web UI option “Authenticate Conf File” does not take effect after enabled under provision setting page.
- Fixed DP7xx drops call when 100REL is enabled and using 183 session progress on some platform.
- Fixed DP7xx handset softkey button failed to display “Cancel” when making a second call and the first call hang up by remote during the second call is in ringing.
- Fixed DP7xx intercom all only support attended transfer, while there is a blind transfer menu.
- Fixed DP7xx *16, *17 feature code not functioning issue.
- Fixed DP7xx some dial plan setting does not take effect.
- Fixed DP7xx handsets display “call released 512” instead of “line busy” when callee reject the call.
- Fixed DP7xx still able to send out INVITE when account is not registered and “Outgoing Calls Without Registration” is enabled.
- Fixed DP72x/DP730 still display call “Held” after end a held conference call to a single call.
- Fixed DP75x auto reboot after change SIP Transport from TLS to UDP with Custom TLS cert and key.

- Fixed DP75x cannot make intercom call when dial plan is set to { [*x]+ }.
- Fixed DP75x SNMP trap and trap interval do not take effect with trap version 3.
- Fixed DP75x hold reminder tone does not play after enabled the reminder tone on web UI option.
- Fixed DP75x base station may crash after provisioned by 3CX server and idled for a period of time.
- Fixed DP7xx randomly rejects incoming calls with 486 busy.
- Fixed when DP72x/DP730 is holding the call, if the remote also hold the call, the handset will falsely display a second incoming call UI.
- Fixed DP7xx handsets time cannot be updated in real time after base station time be updated.
- Fixed DP7xx send ACK for the 200OK after the re-INVITE REFER, the ACK will lead NETSAPIENS server response 491 and transfer failed.
- Fixed DP7xx dial plan working as expected issue such as {<=1>x.} and {<=1>x+} were not recognized.
- Removed invalid account status from SSH UI.
- Fixed DP72x and DP730 still retained custom LSK menu after device being factory reset.
- Fixed DP72x and DP730 still shows call in “Held” status after it has been transferred.
- Added support for “” and “_” in Contacts search box.
- Fixed DP7xx still receives PTT after enabled DND.
- Fixed DP72x and DP730 issue that when it is in ringing status, and receives another incoming call, it replies 486 busy to the second call but not generates a missed call notice.
- Fixed DP72x and DP730 issue that it still prompt missed call after viewed them in “All Calls” and rebooted.
- Fixed DP72x and DP730 cannot display the local-stored contact name when received incoming to a parallel mode account.
- Fixed DP7xx intercom call will fail if the SIP proxy is configured.
- Fixed DP72x and DP730 RSK display as “Silent” instead of “Open” when receiving GDS calls that the caller ID is stored in Contacts.

NEW FEATURES OVERVIEW

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

ON HOLD REMINDER TONE

- **Functionality**

DP75x now support disable or enable “On Hold Reminder Tone”, to set this, please go to Web UI -> Profile x -> Call Settings -> On Hold Reminder Tone.

Profiles
 Profile 1
 General Settings
 Network Settings
 SIP Settings
 Audio Settings
 Call Settings
 Profile 2
 Profile 3
 Profile 4

Call Settings

Early Dial ☒ No ☐ Yes

Dial Plan Prefix

Dial Plan { x+ | *x+ | *xx*x+ }

Use # as Dial Key ☐ No ☒ Yes

On Hold Reminder Tone ☐ Disabled ☒ Enabled

Match Incoming Caller ID Matching Rule

- New P Values**

Pvalue	Alias	Description	Value range	Default
26006	profile.1.call.on_hold_reminder_tone	Plays a reminder tone when you have a call on hold	0 – Disabled	1
26106	profile.2.call.on_hold_reminder_tone		1 – Enabled	
26206	profile.3.call.on_hold_reminder_tone			
26306	profile.4.call.on_hold_reminder_tone			

CALLER ID DISPLAY

- Functionality**

DP75x now support to configure “Caller ID Display”, to set this, please go to Web UI -> Profile x -> SIP Settings -> Basic Settings -> Caller ID Display.

Ignore Alert-Info header ☒ No ☐ Yes

Use MAC Header ☒ No ☐ Yes except REGISTER ☐ Yes to all SIP

Caller ID Display ☒ Auto ☐ Disabled ☐ From Header

Save Save and Apply Reset

- New P Values**

Pvalue	Alias	Description	Value range	Default
2324	profile.1.sip.caller.id_display	When set to "Auto", the phone will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP INVITE. When set to "Disabled", all incoming calls are displayed with "Unavailable".	0 – Auto	0
2424	profile.2.sip.caller.id_display		1 – Disabled	
2524	profile.3.sip.caller.id_display		2 – From Header	
2624	profile.4.sip.caller.id_display			

FIRMWARE VERSION 1.0.13.0

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

03/19/2020

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 0da008189083797f55ec28c681f9671e

ENHANCEMENT

- Improved the accuracy of battery detection for DP720.
- Added option to use sip or sips for SIP URI scheme when using TLS. [SIP URI SCHEME WHEN USING TLS]
- Added support for DHCP option 67.
- Added OpenVPN® Comp-lzo option under OpenVPN® Settings page. [OPENVPN® COMP-LZO]
- Added support to disable web UI access by adding the "Disable" option for Web Access Mode setting.
- Add handset option for blind transfer and attended transfer.
- Added DP720 charging current displaying in Settings -> Factory Functions -> System Monitoring.
- Updated the CPE to better support GDMS.
- Added support for SNMP. [SNMP]
- Added support for management interface settings that using a virtual network interface. [MANAGEMENT INTERFACE SETTING]
- Added support for intercom feature without SIP account. [INTERCOM FEATURE WITHOUT SIP ACCOUNT]
- Added support for transfer between HS internally without SIP. [TRANSFER BETWEEN HS INTERNALLY WITHOUT SIP]
- Added support for Broadsoft XSI phonebook. [BROADSOFT XSI PHONEBOOK]
- Added support for Feature Key Synchronization that support both Broadsoft and MetaSwitch, or any server that following the same feature key synchronize standard. [FEATURE KEY SYNCHRONIZATION]
- Added support for Russian keyboard when the language is set to Russian.
- Added support for option "Backup Outbound Proxy" in all Profiles. [BACKUP OUTBOUND PROXY]
- Added support for option "Use SBC" in all Profiles. [USE SBC]
- Added support for new xml format and text format in config files.

- Updated CPE to version 1.0.1.138 to better support GDMS.
- Display configured base station name in the handset “select base” menu instead of using RFPI number.

BUG FIX

- Fixed DP7xx will stop play ringback tone if handset enabled keypad tone and pressed a key during the ringback period.
- Fixed DP72x and DP730 have some Portuguese wrong translations.
- Fixed DP72x and DP730 do not play call waiting tone when a SIP account is parallel mode and be used in 2 or more handsets.
- Fixed handset failed to display contacts and call history if continuously off hook when connected with a DP760 repeater.
- Fixed DP72x and DP730 may fail to make calls after transfer a call via Intercom call to other handsets.
- Fixed DP75x cannot provision a config XML file that in new alias format.
- Fixed DP72x and DP730 Intercom calls sometimes has no audio.
- Fixed DP75x "Refer-To Use Target Contact" setting does not take effect, which is caused of the "Remote-Part-ID" header field in the INVITE not being correctly processed.
- Fixed DP72x and DP730 issue that after changed the line 1 account, it still uses previous line 1 account by default to dial out.
- Fixed DP75x Web UI option "Remove OBP from Route" does not work issue.
- Fixed DP75x does not send INVITE to Backup Outbound Proxy when the Outbound Proxy is not available.
- Fixed DP72x and DP730 Voicemail Access Number cannot be removed from the LCD settings.
- Fixed DP7xx automatically un-hold a call after 90 seconds.
- Correct São Paulo / Brazil time zone to be UTC-3.
- Added regex limits to SNMP trap address.
- Fixed DP75x CTI calling feature stops working after first CTI call ended.
- Fixed DP75x dial plan incorrectly matched for some cases and failed to append the prefix for dial out number.
- Fixed DP75x LDAPS protocol not working.
- Fixed TR069 ACS URL allows blank space issue.
- Fixed subscription button does not work if reset type set to disabled.
- Fixed HS displays only the first contact's number when an entry has multiple number on same attribute.
- Fixed “Transfer on Conference Hangup” does not work when the handset use different lines to make conference.
- Fixed DP75x does not take DTMF during dialing in some cases.

NEW FEATURES OVERVIEW

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

SIP URI SCHEME WHEN USING TLS

- Functionality**

DP75x now support to choose "sip" or "sips" for URI scheme when using TLS. This setting can be found at Web UI -> Profile x -> SIP Settings -> Basic Settings.



The screenshot shows the 'Basic Settings' page for a SIP profile. The left sidebar lists 'Profiles', 'General Settings', 'Network Settings', 'SIP Settings', 'Basic Settings', 'Session Timer', 'Security Settings', and 'Audio Settings'. The 'SIP Settings' section is expanded, showing 'Basic Settings' selected. The 'SIP URI Scheme When Using TLS' option is highlighted with a red box, showing radio buttons for 'sip' and 'sips', with 'sips' selected. Other settings include 'SIP Transport' (UDP, TCP, TLS/TCP), 'Use Actual Ephemeral Port in Contact with TCP/TLS' (No, Yes), 'SIP Registration' (No, Yes), and 'Unregister on Reboot' (No, Yes).

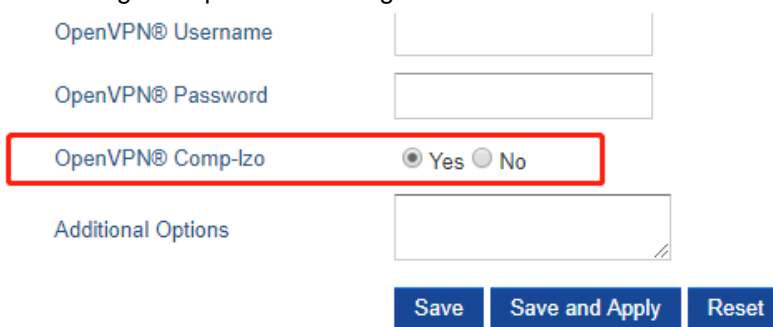
- New P Values**

Pvalue	Alias	Description	Value range	Default
2329	profile.1.sip.uri_scheme_when_using_tls	Specifies if "sip:" or "sips:" will be used when TLS/TCP is selected for SIP Transport.	0 – sip 1 – sips	1
2429	profile.2.sip.uri_scheme_when_using_tls			
2529	profile.3.sip.uri_scheme_when_using_tls			
2629	profile.4.sip.uri_scheme_when_using_tls			

OPENVPN® COMP-LZO

- Functionality**

DP75x now support enable or disable Comp-lzo option for OpenVPN. This setting can be found at Web UI -> Settings -> Network Settings -> OpenVPN Settings:



The screenshot shows the 'OpenVPN Settings' page. The 'OpenVPN® Comp-lzo' option is highlighted with a red box, showing radio buttons for 'Yes' and 'No', with 'Yes' selected. Other settings include 'OpenVPN® Username', 'OpenVPN® Password', and 'Additional Options'. At the bottom, there are buttons for 'Save', 'Save and Apply', and 'Reset'.

- **New P Values**

Pvalue	Alias	Description	Value range	Default
8508	openvpn.comp_lzo	Enable OpenVPN® Comp-lzo or Not, Default Value is YES	0 – No 1 – Yes	1

DISABLE WEB ACCESS

- **Functionality**

DP75x now support disable the Web UI access. This setting can be found at Web UI -> Maintenance-> Web/SSH Settings -> Web Access Mode, set it to “Disabled” will not allow users to access the Web UI. To reopened, can use the SSH command, config file provision, or do a factory reset.

Web/SSH Settings

HTTP Web Port
 HTTPS Web Port
 Web Access Mode ☐ HTTPS ☒ HTTP ☐ Disabled
 Disable SSH ☒ No ☐ Yes

- **New P Values**

Pvalue	Alias	Description	Value range	Default
8508	openvpn.comp_lzo	Enable OpenVPN® Comp-lzo or Not, Default Value is YES	0 – No 1 – Yes	1

SNMP

- **Functionality**

DP750/DP752 now supports SNMP. The settings can be found under web UI->Settings->Network Settings ->SNMP Settings.


Settings

Network Settings

Basic Settings

Advanced Settings

Management Interface Settings

OpenVPN® Settings

SNMP Settings

External Service

Ring Tones

System Features

SNMP Settings

Enable SNMP	<input type="radio"/> Yes <input checked="" type="radio"/> No
Version	<input type="radio"/> Version 1 <input type="radio"/> Version 2 <input checked="" type="radio"/> Version 3
Port	<input type="text" value="161"/>
Trap Community	<input type="text"/>
SNMP Trap Version	<input type="text" value="Trap Version 3"/>
SNMP Trap IP	<input type="text"/>
SNMP Trap Port	<input type="text" value="162"/>
SNMP Trap Interval	<input type="text" value="60"/>
SNMP Username	<input type="text"/>
Security Level	<input checked="" type="radio"/> noAuthUser <input type="radio"/> authUser <input type="radio"/> privUser
Authentication Protocol	<input checked="" type="radio"/> None <input type="radio"/> MD5 <input type="radio"/> SHA
Privacy Protocol	<input checked="" type="radio"/> None <input type="radio"/> DES <input type="radio"/> AES
Authentication Key	<input type="text"/>
Privacy Key	<input type="text"/>
SNMP Trap Username	<input type="text"/>
Trap Security Level	<input checked="" type="radio"/> noAuthUser <input type="radio"/> authUser <input type="radio"/> privUser
Trap Authentication Protocol	<input checked="" type="radio"/> None <input type="radio"/> MD5 <input type="radio"/> SHA
Trap Privacy Protocol	<input checked="" type="radio"/> None <input type="radio"/> DES <input type="radio"/> AES
Trap Authentication Key	<input type="text"/>
Trap Privacy Key	<input type="text"/>

Save

Save and Apply

Reset

• New P Values


Pvalue	Alias	Description	Value range	Default
21896	snmp.enable	Enable or disable SNMP	0 – No 1 – Yes	0
21904	snmp.version	SNMP version	1 – Version 1 2 – Version 2 3 – Version 3	3
21903	snmp.port	SNMP port	161, 1025-65535	161
21902	snmp.community	Name of SNMP trap community	String	Null
21899	snmp.trap.version	SNMP Trap Version	1 – Version 1 2 – Version 2 3 – Version 3	3

21897	snmp.trap.destination.address	IP address of the SNMP trap receiver	IP Address	Null
21898	snmp.trap.destination.port	Port of the SNMP trap receiver	162, 1025-65535	162
21901	snmp.trap.interval	SNMP Trap Interval	1-1440	60
21905	snmp.version3.username	Username for SNMPv3	String	Null
21910	snmp.version3.security.level	Security Level for SNMPv3	0 – noAuthUser 1 – authUser 2 – privUser	0
21906	snmp.version3.auth.protocol	Authentication Protocol for SNMPv3	0 – None 1 – MD5 2 – SHA	0
21907	snmp.version3.priv.protocol	Privacy Protocol for SNMPv3	0 – None 1 – DES 2 – AES	0
21908	snmp.version3.auth.key	Authentication Key for SNMPv3	String	0
21909	snmp.version3.priv.key	Privacy Key for SNMPv3	String	0
21911	snmp.version3.trap.username	Trap Username for SNMPv3	String	Null
21916	snmp.version3.trap.security.level	Security Level for SNMPv3 Trap	0 – noAuthUser 1 – authUser 2 – privUser	0
21912	snmp.version3.trap.auth.protocol	Authentication Protocol for SNMPv3 Trap	0 – None 1 – MD5 2 – SHA	0
21913	snmp.version3.trap.priv.protocol	Privacy Protocol for SNMPv3 Trap	0 – None 1 – DES 2 – AES	0
21914	snmp.version3.trap.auth.key	Authentication Key for SNMPv3 Trap	String	0
21915	snmp.version3.trap.priv.key	Privacy Key for SNMPv3 Trap	String	0

MANAGEMENT INTERFACE SETTING

- **Functionality**

Now DP750/DP752 now support a virtual network interface, the administrator can use it to access and manage the device.


Version 1.0.11.9

Settings

Network Settings

Basic Settings

Advanced Settings

Management Interface Settings

OpenVPN® Settings

SNMP Settings

External Service

Ring Tones

System Features

Management Interface Settings

Enable Management Interface ☐ Enabled ☒ Disabled

Management Access Management Interface only

Layer 2 QoS 802.1Q/VLAN Tag 0

Layer 2 QoS 802.1p Priority Value 0

IP Address Mode ☒ DHCP ☐ Static IP

Static IP Settings

IP Address 192.168.100.100

Subnet Mask 255.255.255.0

Gateway 192.168.100.1

DNS Server 1 0.0.0.0

DNS Server 2 0.0.0.0

Save
Save and Apply
Reset

- New P Values**

Pvalue	Alias	Description	Value range	Default
28161	network.mgmt.interface.enable	Enable or disable Management Interface	0 – No 1 – Yes	0
28162	network.mgmt.access.mode	Allow management access to management interface or both service and management interfaces	0 – Management Interface only 1 – Both Service and Management Interfaces	0
22111	network.mgmt.vlan.layer2_qos.tag	Assigns the VLAN Tag of the Layer 2 QoS packets.	161, 1025-65535	0
22112	network.mgmt.layer2_qos.priority	Assigns the priority value of the Layer 2 QoS packets. Valid range is 0 to 7.	String	0
22105	network.mgmt.type	How the IP address obtained on the gateway.	0 – DHCP 1 – Static IP	0
22106	network.mgmt.static.address.ip	Enter the IP address when static IP is used.	IP Address	192.168.100.100

22107	network.mgmt.static. netmask.ip	Enter the Subnet Mask when static IP is used.	IP Address	255.255.255.0
22108	network.mgmt.static. router.ip	Enter the Default Gateway when static IP is used.	IP Address	192.168.1.00.1
22109	network.mgmt.static. dns.1.ip	Enter DNS Server 1 when static IP is used.	IP Address	0.0.0.0
22110	network.mgmt.static. dns.2.ip	Enter DNS Server 2 when static IP is used.	IP Address	0.0.0.0

INTERCOM FEATURE WITHOUT SIP ACCOUNT

- **Functionality**

Previously, DP75x only support intercom when the destination handset has at least 1 SIP account registered and assigned. Now since 1.0.11.9, this SIP account is not required, and the intercom will work without SIP account.

TRANSFER BETWEEN HS INTERNALLY WITHOUT SIP

- **Functionality**

Previously, DP75x support transfer a call to another handset, but required to use a SIP account, now since 1.0.11.9, DP75x can transfer a call to another handset even it has no SIP account registered.

BROADSOFT XSI PHONEBOOK

- **Functionality**

Broadsoft XSI phonebook settings are under Phonebook -> Global Phonebook Broadsoft XSI Settings. It requires to configure the Global Phonebook Type to "XSI". Please follow the GXP phones XSI guide to configure the device.

http://www.grandstream.com/sites/default/files/Resources/GXP2130_2140_2160_Xsi_Guide_0.pdf



Version 1.0.11.9

Phonebook

 Global Phonebook XML
Settings

 Global Phonebook LDAP
Settings

 Global Phonebook
Broadsoft XSI Settings

 Private Phonebook
Settings

Broadsoft XSI

 Global Phonebook Type XML ▼

XSI

 Server
 Port
 XSI Actions Path
 Broadsoft Contact Download
Interval
 XSI Authentication Type Login Credentials ▼

Login Credentials

 Login Username
 Login Password

SIP Credentials

 SIP UserName
 SIP User ID
 SIP Password

Network Directories

Type	Name
Group Directory	Disabled ▼ <input type="text"/>
Enterprise Directory	Disabled ▼ <input type="text"/>
Group Common	Disabled ▼ <input type="text"/>
Enterprise Common	Disabled ▼ <input type="text"/>
Personal Directory	Disabled ▼ <input type="text"/>

Save Save and Apply Reset

• New P Values

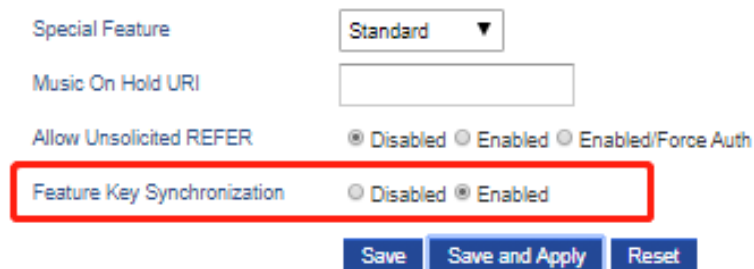
Pvalue	Alias	Description	Value range	Default
1591	broadsoft.xsi.server	Broadsoft XSI server address with protocol	String	Null
1592	broadsoft.xsi.port	Port of the Broadsoft XSI server	1 – 65535	80
2937	broadsoft.xsi.XSIActions	XSI Actions Path	161, 1025 - 65535	0
8362	broadsoft.xsi.downloadInterval	Broadsoft Contact Download Interval	5 - 4320	4320
22054	broadsoft.xsi.authType	XSI Authentication Type	0 – Login Credentials 1 – SIP Credentials	0

			2 – Account 1 3 – Account 2 4 – Account 3 5 – Account 4 6 – Account 5 7 – Account 6 8 – Account 7 9 – Account 8 10 – Account 9 11 – Account 10	
1593	broadsoft.xsi.username	Username for Broadsoft XSI server	String	Null
1594	broadsoft.xsi.password	Password for Broadsoft XSI server	String	Null
6772	broadsoft.xsi.SIPUsername	SIP User Name for Broadsoft XSI server	String	Null
22034	broadsoft.xsi.SIPId	SIP User ID for Broadsoft XSI server	String	Null
6773	broadsoft.xsi.SIPPassword	SIP Password for Broadsoft XSI server	String	Null
2971	broadsoft.xsi.enableGroupDirectory	Enable or disable Group Directory	0 – Disabled 1 – Enabled	0
2972	broadsoft.xsi.groupDirectory	Name of Group Directory	String	Null
2973	broadsoft.xsi.enableEnterpriseDirectory	Enable or disable Enterprise Directory	0 – Disabled 1 – Enabled	0
2974	broadsoft.xsi.enterpriseDirectory	Name of Enterprise Directory	String	Null
2975	broadsoft.xsi.enableGroupCommon	Enable or disable Group Common	0 – Disabled 1 – Enabled	0
2976	broadsoft.xsi.groupCommon	Name of Group Common	String	Null
2977	broadsoft.xsi.enableEnterpriseCommon	Enable or disable Enterprise Common	0 – Disabled 1 – Enabled	0
2978	broadsoft.xsi.enterpriseCommon	Name of Enterprise Common	String	Null
2979	broadsoft.xsi.enablePersonalDirectory	Enable or disable Personal Directory	0 – Disabled 1 – Enabled	0
2980	broadsoft.xsi.personalDirectory	Name of Personal Directory	String	Null

FEATURE KEY SYNCHRONIZATION

- Functionality**

To use feature key synchronization function, just enable the “Feature Key Synchronization” at Profiles -> Profile X -> Call Settings. After enabled this function, the device will sync up with the server when enabling / disabling call features like DND, call forward, call waiting. It will use NOTIFY to send the status in XML content to server and accept the NOTIFY from the server. This feature following the Broadsoft and MetaSwitch standard. Any server that following the same standard will be compatible with this feature.



Special Feature: Standard

Music On Hold URI:

Allow Unsolicited REFER: ☒ Disabled ☐ Enabled ☐ Enabled/Force Auth

Feature Key Synchronization: ☐ Disabled ☒ Enabled

Buttons: Save, Save and Apply, Reset


- New P Values**

Pvalue	Alias	Description	Value range	Default
2325	profile.1.subscribe.feature_event	Enable or disable Feature Key Synchronization for profile 1	0 – Disabled 1 - Enabled	0
2425	profile.2.subscribe.feature_event	Enable or disable Feature Key Synchronization for profile 2	0 – Disabled 1 - Enabled	0
2525	profile.3.subscribe.feature_event	Enable or disable Feature Key Synchronization for profile 3	0 – Disabled 1 - Enabled	0
2625	profile.4.subscribe.feature_event	Enable or disable Feature Key Synchronization for profile 4	0 – Disabled 1 - Enabled	0

BACKUP OUTBOUND PROXY

- Functionality**

DP75x now support the backup outbound proxy, in case when the primary outbound proxy is not working. The setting can be found at Profiles -> Profile X -> General Settings.


Version 1.0.11.9

Profiles
 Profile 1
 General Settings
 Network Settings
 SIP Settings
 Audio Settings
 Call Settings
 Profile 2
 Profile 3
 Profile 4

General Settings

Profile Active
☐ No ☒ Yes

Profile Name

SIP Server

Fallback SIP Server

Prefer Primary SIP Server
☒ No ☐ Yes

Outbound Proxy

Backup Outbound Proxy

Voice Mail Access Number

Save
Save and Apply
Reset

• New P Values

Pvalue	Alias	Description	Value range	Default
2333	profile.1.backup_outbound_proxy.1.address	Backup Outbound Proxy for profile 1	String	Null
2433	profile.2.backup_outbound_proxy.1.address	Backup Outbound Proxy for profile 2	String	Null
2533	profile.3.backup_outbound_proxy.1.address	Backup Outbound Proxy for profile 3	String	Null
2633	profile.4.backup_outbound_proxy.1.address	Backup Outbound Proxy for profile 4	String	Null

USE SBC

• Functionality

DP75x now able to configure to use SBC or not for NAT. The setting can be found at Profiles -> Profile X -> Network Settings -> NAT Settings.

NAT Settings

NAT Traversal

No ▼

Use NAT IP

Proxy-Require

Use SBC
☒ No ☐ Yes

Save
Save and Apply
Reset

- **New P Values**

Pvalue	Alias	Description	Value range	Default
26090	profile.1.use_sbc	Use SBC for profile 1	0 – Disabled 1 - Enabled	0
26190	profile.2.use_sbc	Use SBC for profile 2	0 – Disabled 1 - Enabled	0
26290	profile.3.use_sbc	Use SBC for profile 3	0 – Disabled 1 - Enabled	0
26390	profile.4.use_sbc	Use SBC for profile 4	0 – Disabled 1 - Enabled	0

FIRMWARE VERSION 1.0.11.4

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

10/25/2019

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: c1c11b2cef3c24e547b7fbe74a9ac55d

ENHANCEMENT

- Updated CPE to version 137 to improve the support on GDMS.

FIRMWARE VERSION 1.0.11.2

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

07/08/2019

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 6a5dd66836d5382f35726011df564448

ENHANCEMENT

- Added warning message for GDS door system, to show if open door successful or failed.
- Added DP75X Callee Codec Negotiation Priority option. [CALLEE CODEC NEGOTIATION PRIORITY]
- Added Chile time zone.
- Added support to bypass the dial plan check when redial / call back phone number via call history.
- Changed SIP account name length from 14 characters to 256.

BUG FIX

- Fixed handsets cannot redial unless pressing off-hook after handset been factory reset.
- Corrected DP752 UPNP friendlyName to be DP752-{MAC}.
- Fixed DP75X DSCP of SIP, change it to be CS5 instead of CS3.
- Changed DP75x certificate "issue by" to be Common Name only.
- Updated the help text of firmware upgrade and provisioning related fields to indicate the support of FTP and SFTP protocol.
- Fixed DP75X do not send REGISTER to the option 120 URL when there is one address on option 120.
- Fixed DP75X IP address/subnet/gateway/DNS changed to 0.0.0.0 after set SIP TLS Private Key.
- Fixed DP760 display "Open Subscription" and "Page All" button on the icon bar.
- Fixed gateway does not update after connected to a different network.
- Fixed after DP75x factory reset, the Option 60 been set to null instead of its default value.
- Updated CPE to 1.0.1.128 to improve the support for GDMS platform.
- Fixed DP75X Music on Hold URI option does not take effect.
- Fixed DP75X Off-hook Auto Dial option does not take effect.

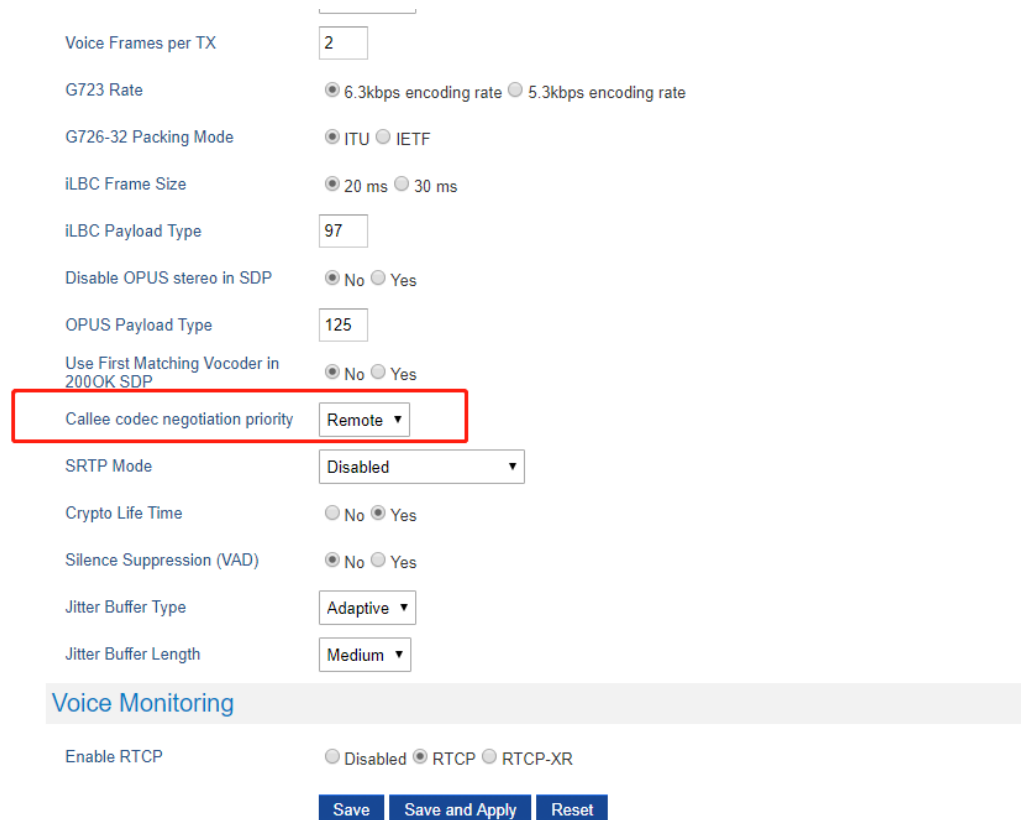
NEW FEATURES OVERVIEW

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

CALLEE CODEC NEGOTIATION PRIORITY

- Functionality**

DP750/DP752 supports configuring the callee codec negotiation priority under web UI->Profile X->Audio Settings->Callee codec negotiation priority. As callee, the codecs negotiation priority can be set to local or remote. Setting to "Remote" will use remote codec priority, or "local" to use local priority. By default, it uses Remote priority.



Voice Frames per TX: 2
 G723 Rate: ☒ 6.3kbps encoding rate ☐ 5.3kbps encoding rate
 G726-32 Packing Mode: ☒ ITU ☐ IETF
 iLBC Frame Size: ☒ 20 ms ☐ 30 ms
 iLBC Payload Type: 97
 Disable OPUS stereo in SDP: ☒ No ☐ Yes
 OPUS Payload Type: 125
 Use First Matching Vocoder in 200OK SDP: ☒ No ☐ Yes
Callee codec negotiation priority: Remote ▼
 SRTP Mode: Disabled ▼
 Crypto Life Time: ☐ No ☒ Yes
 Silence Suppression (VAD): ☒ No ☐ Yes
 Jitter Buffer Type: Adaptive ▼
 Jitter Buffer Length: Medium ▼
Voice Monitoring
 Enable RTCP: ☐ Disabled ☒ RTCP ☐ RTCP-XR
 [Save] [Save and Apply] [Reset]

- New P Values**

Pvalue	Alias	Description	Value range	Default
29061	profile.1.audio.callee_codec_negotiation_priority	Profile 1 Callee codec negotiation priority	0 – Remote 1 - Local	0
29161	profile.2.audio.callee_codec_negotiation_priority	Profile 2 Callee codec negotiation priority	0 – Remote 1 - Local	0
29261	profile.3.audio.callee_codec_negotiation_priority	Profile 3 Callee codec negotiation priority	0 – Remote 1 - Local	0

29361	profile.4.audio.callee_codec _negotiation_priority	Profile 4 Callee codec negotiation priority	0 – Remote 1 - Local	0
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FIRMWARE VERSION 1.0.9.9

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

04/23/2019

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 5b7b6c0079c3e6758f452a71da0fa90f

ENHANCEMENT

- Added DP75X Trusted CA Certificate option for HTTPS/FTPS. [TRUSTED CA CERTIFICATES]
- Added support to configure customized time zone and daylight-saving time (DST) on the web UI. [CUSTOMIZED TIMEZONE AND DAYLIGHT-SAVING TIME SETTINGS]
- Removed the limitation of Intercom call that it must enable Direct IP call. Now the Intercom call will be working regardless if Direct IP Call is disabled or not.
- Increased DP75X web UI voice mail user ID character length to 30.
- Increased SIP Certificate length limitation from 2048 characters to 8192.
- Added support to show warning messages on handset GUI after clicking "Open" button for GDS doorbell, now it will display success or failed to indicate the result.
- Added support for FTP/FTPS User Name/Password for firmware upgrade and provisioning.
- Added support to configure Voice Mail access number on DP75X web UI as the same as GXP21XX phones, now configure the access code to "*97,,password" will automatically dial the Voice Mail box and enter the password.

BUG FIX

- Fixed DP75X keep using the "#" as dial key even after setting "Use # as Dial Key" as "No".
- Fixed DP75X sometimes crash when calling number as 011xxxxxx.
- Fixed DP75X does not send DNS request to the Preferred DNS Server at first.
- Fixed DP75X does not set time zone properly in the recovery image.
- Fixed DP75X setting of Validate Server Certificates does not take effect.
- Fixed DP75X fixed Web UI support documentation URL link cannot be found error.
- Improved DP72X/DP730 Intercom call feature:
 - Removed "conference" option when one of the calling is intercom call.
 - Left softkey now will display Intercom or Line correctly by matching the entered digit.

- Fixed when handsets have 2 intercom calls with different handsets, one of the call will display SIP account instead of HS name.
- Fixed DP75X web UI customized ringtones display full file path instead of just file name.
- Fixed DP75X Device Custom Certificate does not take effect after enabled.
- Fixed DP72X/DP730 failed to open GDS doorbell after first answered GDS call and then answered/rejected the second call from other non-GDS accounts.
- Fixed DP75X can be factory reset when there is active call, now it will factory reset after all calls on the base been released.
- Fixed DP75X LLDP system description and firmware revision TLV incorrect issue.
- Fixed DP75X Allow Unsolicited REFER issue that it does not transfer a call when received REFER with Allow Unsolicited REFER enabled.
- Fixed DP75X Web UI cannot set Phonebook download mode to TFTP.
- Fixed DP72X/DP730 missed call notification count number is incorrect when same account calls are merged.

NEW FEATURES OVERVIEW

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

TRUSTED CA CERTIFICATES

- **Functionality**

Now DP750/DP752 can upload Trusted CA Certificates for HTTPS/FTPS server. Customer can choose what certificates used to verify the server: default certificate only (built-in), custom certificate only or all certificates.



Maintenance

- Firmware Upgrade
- Provisioning
- Web/SSH Access
- TR-069
- Security Settings
- Web/SSH
- SIP TLS
- Radius
- Device Custom Certificate
- Security
- Date and Time
- Syslog
- Packet Capture
- Factory Reset
- Support

Security

 Validate Server Certificates ☒ No ☐ Yes

Trusted CA Certificates

CA Certificate	Issued by	Expires by
CA Certificate 1	Upload	Delete
CA Certificate 2	Upload	Delete
CA Certificate 3	Upload	Delete
CA Certificate 4	Upload	Delete
CA Certificate 5	Upload	Delete
CA Certificate 6	Upload	Delete

 Load CA Certificates
[Save](#) [Save and Apply](#) [Reset](#)

• New P Values

Pvalue	Alias	Description	Value range	Default
8433	maintenance.security.device.trusted_ca.1	Text for trusted ca content	string	null
8434	maintenance.security.device.trusted_ca.2	Text for trusted ca content	string	null
8435	maintenance.security.device.trusted_ca.3	Text for trusted ca content	string	null
8436	maintenance.security.device.trusted_ca.4	Text for trusted ca content	string	null
8437	maintenance.security.device.trusted_ca.5	Text for trusted ca content	string	null
8438	maintenance.security.device.trusted_ca.6	Text for trusted ca content	string	null
8502	maintenance.security.load_ca_certificates	Phone will verify the server certificate based on the built-in, custom or both trusted certificates list.	0 - Default Certificates 1 - Custom Certificates 2 - All Certificates	2

CUSTOMIZED TIMEZONE AND DAYLIGHT-SAVING TIME SETTINGS

- Functionality**

Now DP750/DP752 can disable session timer on web UI, which is under Profile X -> SIP Settings -> Session Timer. By default, it is enabled.

Maintenance
 Firmware Upgrade
 Provisioning
 Web/SSH Access
 TR-069
 Security Settings
Date and Time
 Syslog
 Packet Capture
 Factory Reset
 Support

Date and Time

NTP Server
 NTP Update Interval
 Allow DHCP Option 42 to NTP Server ☒ No ☐ Yes

System Time Zone Settings

Time Zone
 Self-Defined Time Zone
 Allow DHCP Option 2 to override time zone ☒ No ☐ Yes

Custom Time Zone Settings

Custom Time Zones
 UTC Offset
 Second Offset
 DST Type
 Start & End Times Start : End :
 Time zone settings by week
 Start & End Months Start End
 Start & End settings by week Start Week End Week
 Time zone settings by day
 Start & End settings by day Start End

- New P Values**

Pvalue	Alias	Description	Value range	Default
27828	maintenance.date_time.enable_time_zone	Enable or disable customized time zone.	0 - Disabled 1 - Enabled	0
27829	maintenance.date_time.time_zone_day_week	DST Type, set Daylight Savings by day or by week.	0 – by week 1 – by day	0
27830	maintenance.date_time.time_zone_offset	UTC offset	-12 ~ 14	0

27831	maintenance.date_time.time_zone_m onth	Start months for custom time zone that daylight savings type is by week.	1 ~ 12 for 12 months	4
27832	maintenance.date_time.time_zone_h our	Start hour for custom time zone that daylight savings type is by week.	0 ~ 23	2
27833	maintenance.date_time.time_zone_mi nute	Start minute for custom time zone that daylight savings type is by week.	0 ~ 59	0
27834	maintenance.date_time.time_zone_d ay_of_week	Start weekday for custom time zone that daylight savings type is by week.	0 ~ 6 (0 is Sunday, 6 is Saturday)	0
27835	maintenance.date_time.time_zone_w eek	Start week number for custom time zone that daylight savings type is by week.	1 – Week 1 2 – Week 2 3 – Week 3 4 – Week 4 5 – Week 5	1
27836	maintenance.date_time.time_zone_d ay_of_year	Start day for custom time zone that daylight savings type is by day.	0 – 365	1
27837	maintenance.date_time.second_time_ zone_offset	Second Offset	-12 ~ 14	0
27838	maintenance.date_time.second_time_ zone_month	End months for custom time zone that daylight savings type is by week.	1 ~ 12 for 12 months	9
27839	maintenance.date_time.second_time_ zone_hour	End hour for custom time zone that daylight savings type is by week.	0 ~ 23	2

27840	maintenance.date_time.second_time_ zone_minute	End minute for custom time zone that daylight savings type is by week.	0 ~ 59	0
27841	maintenance.date_time.second_time_ zone_day_of_week	End weekday for custom time zone that daylight savings type is by week.	0 ~ 6 (0 is Sunday, 6 is Saturday)	0
27842	maintenance.date_time.second_time_ zone_week	End week number for custom time zone that daylight savings type is by week.	1 – Week 1 2 – Week 2 3 – Week 3 4 – Week 4 5 – Week 5	1
27843	maintenance.date_time.second_time_ zone_day_of_year	End day for custom time zone that daylight savings type is by day.	0 – 365	1

FIRMWARE VERSION 1.0.9.1

PRODUCT NAME

DP750, DP752, DP760, DP730, DP722, DP720

DATE

03/01/2019

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: aaa031f5526d55fd630fec6a6d91d287

ENHANCEMENT

- Added option to enable or disable Session Timer for DP75x. [DISABLE SESSION TIMER]
- Added option to enable or disable DP730 keypad backlight at GUI -> Settings -> Display.
- Added "Silent" for DP730 central softkey when receiving a GDS doorbell call.
- Added support of EXTRA DEBUG level Syslog for DP75X.
- Upgraded Dropbear SSH from version 2016.74 to 2018.76 to enhance the security.
- Upgraded mini_httpd 1.19 to 1.30 to enhance the security.
- Add ability to manually configure date and time on the base Web GUI. [MANUALLY CONFIGURE DATE AND TIME]
- Added support to enable or disable "Allow Unsolicited REFER" option for DP750/DP752. [ALLOW UNSOLICITED REFER]
- Change DP720/DP730 Preferences option "Mute as DND" default to "Enabled" from "Disabled" to make enabling DND more convenient.
- Added support for Intercom feature. [INTERCOM]

BUG FIX

- Fixed DP720/DP730 stop playing dial tone when connect USB power cable during off-hook.
- Fixed DP720/DP730 speaker volume does not match configured volume if off-hook first then enable speaker mode.
- Fixed DP720/DP730 issue when have a call on hold and another incoming call rings, unhold the first call will have no audio or abnormal UI depends on the ringtone settings.
- Fixed DP720/30 default outgoing line is not the first order and be selected when choose line in the dialing interface.
- Fixed DP730 will not auto dial out after no key entered timeout that is 4 seconds by default.
- Fixed DP720/DP730 change ringtone 6, 7, 8 volume does not take effect issue.
- Fixed DP720/DP730 cannot use speaker key to call out when has multiple accounts.

- Fixed DP730 cannot use center navigation button to save SIP settings and system settings.
- Fixed DP720/DP730 keypad tone becomes abnormal after enter audio loopback diagnosis and exit before completed.
- Fixed DP720/DP730 still shows “transfer” and “cancel” option after call been hold, changed it to “select” and “unhook”.
- Fixed DP750/DP752 some STUN Server Address setting issue
- Fixed DP750/DP752 option “Layer 2 QoS 802.1p Priority Value for SIP signaling” cannot be saved after being changed.
- Fixed DP750/DP752 web status page shows account status is “in idle” while there is ongoing call.
- Fixed DP750/DP752 SSH status result return idle status while the account is being used.
- Fixed DP750/DP752 profile 4 session timer UAS Specify Refresher cannot be changed to UAS.
- Fixed DP720/DP730 “Delete All” option for contacts to only delete all private contacts.
- Fixed DP750/DP752 does not apply the configuration file when set SIP protocol to be TLS.
- Fixed DP750/DP752 semi-attended failed when enabled Refer-To User Target Contact.
- Fixed DP750/DP752 call failed when behind BroadSoft SBC because of 2 m lines for both RTP and SRTP streaming.
- Fixed DP720/DP730 stopped to calculate call talk time after it answered another call.
- Fixed DP720/DP730 Hebrew contact name is not displayed from right to left issue.
- Fixed Session Timer UAS specify refresher is not following RFC4028 issue.
- Fixed DP720/DP730 issue that the SRTP security icon should be displayed only if both TLS and SRTP are enabled.
- Fixed DP750/DP752 does not respect Session Timer when using compact header.
- Fixed DP750/DP752 unable to download firmware when trying to upgrade it via an HTTPS server with option `ssl_verify_client` enabled on the server.
- Fixed DP750/DP752 should not reboot via SSH command when there is an ongoing call.
- Fixed some inappropriate translations.


NEW FEATURES OVERVIEW

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

DISABLE SESSION TIMER

- **Functionality**

DP750/DP752 supports disabling session timer on web UI, which is under Profile X -> SIP Settings -> Session Timer. By default, it is enabled.


Version 1.0.9.1

Profiles
 Profile 1
 General Settings
 Network Settings
 SIP Settings
 Basic Settings
 Session Timer
 Security Settings

Session Timer

Enable Session Timer ☒ No ☐ Yes

Session Expiration

Min-SE

Caller Request Timer

☒ No ☐ Yes

- New P Values**

Pvalue	Alias	Description	Value range	Default
P2395	profile.1.sip.enable_session_timer	<New Pvalue>	0 – Disabled	1 – Enabled
P2495	profile.2.sip.enable_session_timer	To enable/disable	1 – Enabled	
P2595	profile.3.sip.enable_session_timer	Session Timer		
P2695	profile.4.sip.enable_session_timer	support.		

MANUALLY CONFIGURE DATE AND TIEM

- Functionality**

Instead of using NTP server to automatically get the time from internet, now DP support manually configure the date and time, the settings are under Maintenance -> Date and Time.

Maintenance
 Firmware Upgrade
 Provisioning
 Web/SSH Access
 TR-069
 Security Settings
Date and Time
 Syslog
 Packet Capture
 Factory Reset
 Support

Date and Time

NTP Server

NTP Update Interval

Allow DHCP Option 42 to NTP Server

☒ No ☐ Yes

Time Zone

Self-Defined Time Zone

Allow DHCP Option 2 to override time zone

☒ No ☐ Yes

Manual Time Settings

Manual Time

Year / Month / Day

/ /

Hour : Minute

:

- New P Values**

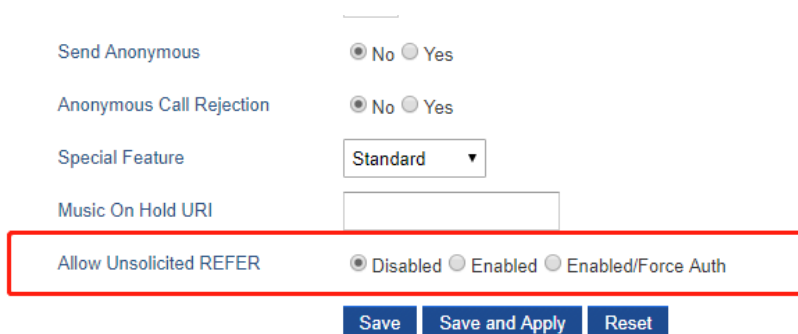
Pvalue	Alias	Description	Value range	Default
P27821	maintenance.date_time.enable_manual_time	<New Pvalue>	0 – Disabled 1 – Enabled	0 – Disabled

		Disable/Enable manual time setting		
P27823	maintenance.date_time.manual_time_year	Year	2016-2037	2019
P27824	maintenance.date_time.manual_time_month	Month	1 - 12	1
P27825	maintenance.date_time.manual_time_day	Day	1 - 31	1
P27826	maintenance.date_time.manual_time_hour	Hour	0 - 11	0
P27827	maintenance.date_time.manual_time_minute	Minute	0 - 59	0

ALLOW UNSOLICITED REFER

- Functionality**

User now can configure if allow unsolicited REFER to accomplish an outgoing call. The settings are under Profile X -> SIP Settings -> Call Settings.



- New P Values**

Pvalue	Alias	Description	Value range	Default
P26023	profile.1.sip.allow_unsolicited_refer	<New Pvalue>	0 – Disabled	0 –
P26123	profile.2.sip.allow_unsolicited_refer	Allow unsolicited	1 – Enabled	Disabled
P26223	profile.3.sip.allow_unsolicited_refer	REFER to	2 –	
P26323	profile.4.sip.allow_unsolicited_refer	accomplish an outgoing call.	Enabled/Force Auth	

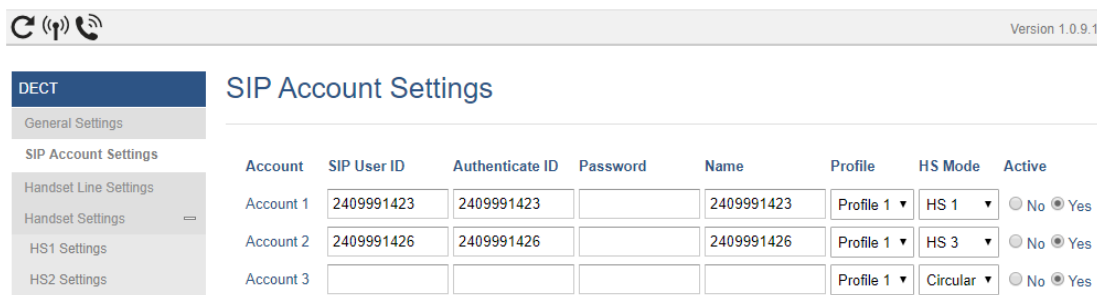
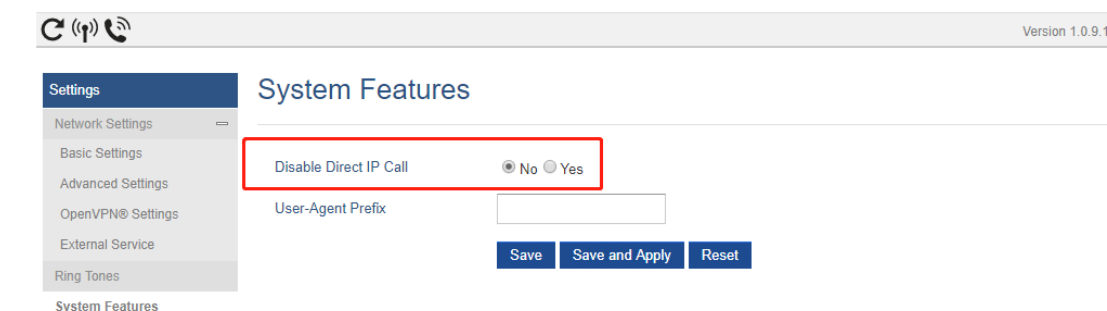
INTERCOM

- Functionality**

Intercom (intercommunication) provide users ability to make phone calls between different handsets directly without remembering each user's phone number. To use it, users need go to Handset GUI -> Shortcut to configure an navigation key (up, down, left or right key) to "Intercom". When using it, just press the key just configured, choose the handset name that you want to call.

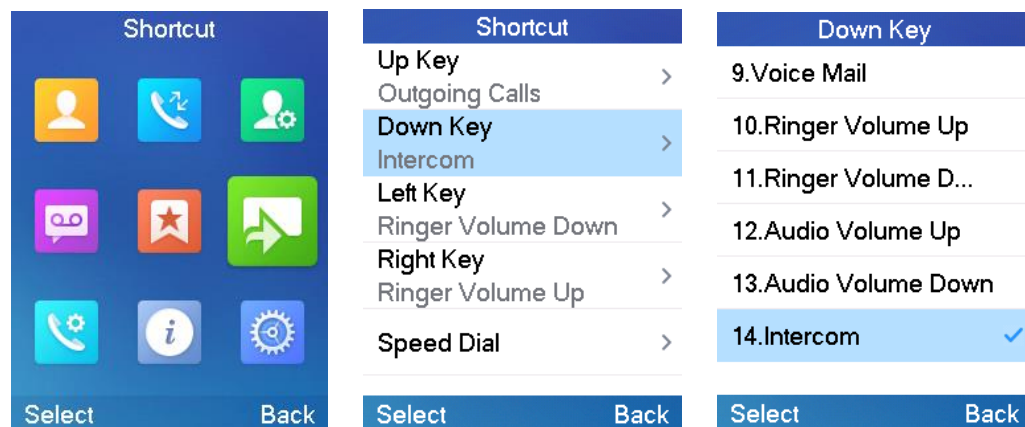
As 1.0.9.1 is the first version to support Intercom feature, there are some known limitations that we plan to remove in a future release:

1. Each handset needs at least one SIP account registered.
2. The Base need enable Direct IP Call (By default, this is enabled at Web UI -> Settings -> System Features)

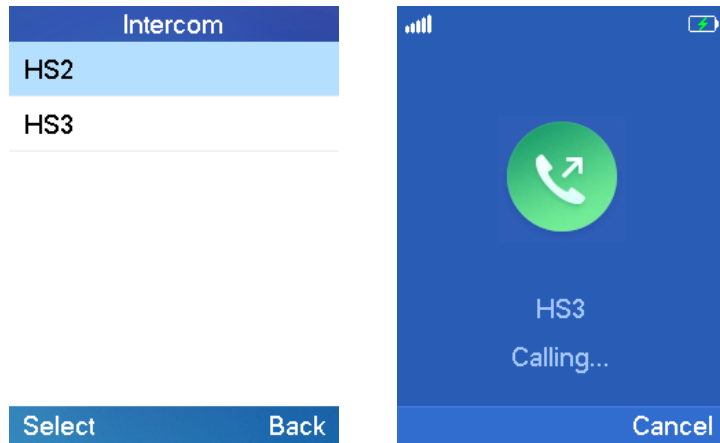


- **Configure and usage on Handset:**

For example, in below screenshots we configure the Down Key to use Intercom feature.



To use the Intercom, press Down Key, it will pop up all available handsets' name, select the one you want to dial by left softkey or green off-hook button, initiate the call.



If the Intercom call failed, please check if the handset has a valid SIP account registered and Direct IP call is enabled.

FIRMWARE VERSION 1.0.7.5

PRODUCT NAME

DP750, DP720, DP760, DP752, DP730

DATE

01/29/2019

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 86afe0c9b4c5de4df3b3e908008377a4

IMPORTANT UPGRADING NOTES

- For DP752/DP730, firmware 1.0.7.5 is the first public released beta firmware.
- For DP720, after the base station (DP750 or DP752) upgraded to 1.0.7.5, the handset DP720 will not automatically upgrade because of the whole GUI has been changed from 1.0.4.10. Users please manually operation the handset firmware on “Menu” -> “Settings” -> “Firmware Upgrade” to upgrade to 1.0.7.5.
- For all DP series, once upgraded to 1.0.7.5 or above, downgrading to 1.0.4.10 or previous firmware version is not supported.

ENHANCEMENT

- Added support for DP752/DP730 model.
- Changed DP720 GUI to new design with new style. [DP720 GUI UPDATES]
- Added support for Push-to-Talk for DP720. [DP720 PTT FEATURE]
- Added support to work with GDS3710/GDS3705 to remote open door with “Open” button. [REMOTE OPENDOOR FOR GDS3710/GDS3705]
- Added support for “Music on Hold URI” for DP750. [MUSIC ON HOLD URI]
- Added support for “Use Random SIP Port” for DP750. [USE RANDOM SIP PORT]
- Disable DP720 auto upgrade to 1.0.7.X or above.
- Improved DP720 built-in ringtones.
- Added support for new format of P values, which has more meaningful name. [NEW FORMAT OF PVALUES]

BUG FIX

- Fixed DP750 "Enable Call Waiting in Parallel Mode" need reboot to take effect.

- Fixed DP750 issue that it sometimes crash and self-generate a core dump file, only recover after reboot.
- Fixed DP720 does not display the contact name on the call history if the number starts with “+”.

NEW FEATURES OVERVIEW

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

DP720 GUI UPDATES

- **Functionality**

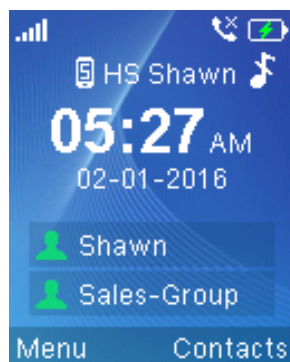
DP720 GUI has been updated on 1.0.7.5. Once upgraded to new GUI, the handset will not able to downgrade to 1.0.4.10. To use 1.0.7.5 new GUI, the user need go to handset “Menu” -> “Settings” -> “9. Firmware Upgrade” to start the upgrade. Even DP750 enabled handset automatically to upgrade, for this specific time, because of the GUI has been redesigned, the user need click the “9. Firmware Upgrade” to complete the upgrade.

Below screenshots are the noticeable changes in the new GUI.

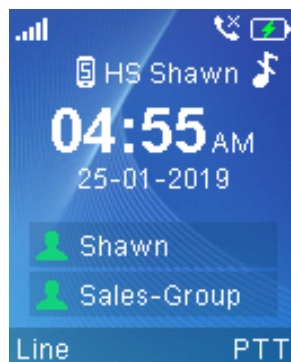
- **Idle screen:**

We changed the maximum SIP account display from 3 to 2 and changed the fonts and style, background picture.

(Default Softkeys)



(Customized Softkeys)

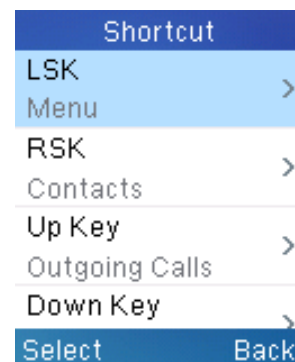
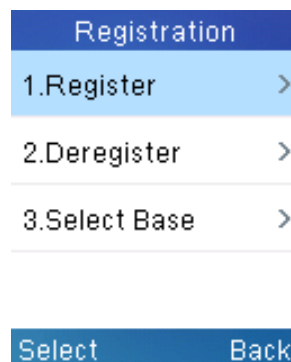
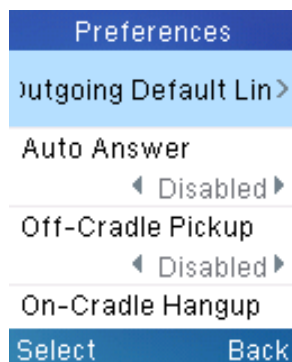
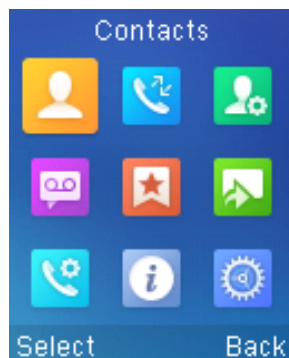


- **Menu**

We changed 6-grid menu to 9-grid. Re-arranged the order, added 3 sections as:

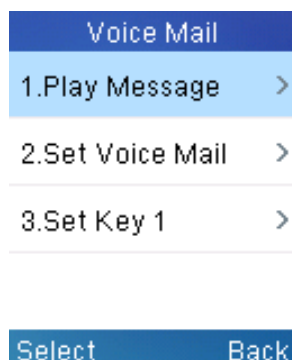
1. **Registration:** to be easier to register handset to base station for out of box experience.
2. **Preferences:** to configure the handset specific preference like auto answer, off-cradle pickup, on-cradle hang-up, enable or disable Push to Talk, etc.
3. **Shortcuts:** to configuration the action of the LSK (Left Softkey), RSK (right Softkey), navigation keys and speed dial from number key. These settings are used to under “Settings” -> “User

Settings”. this improved the setting to be more organized and also provide easier access to use PTT, because PTT feature need a dedicated button.



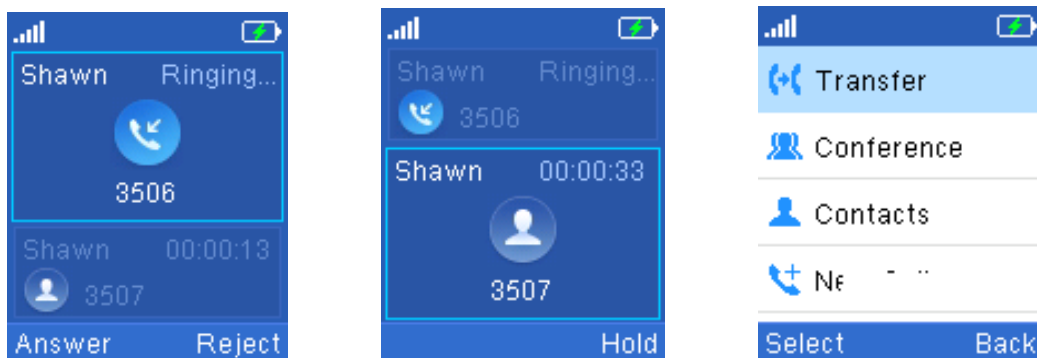
- **Voice Mail:**

On 1.0.4.10, we provided a shortcut button “Key 1” to access the voice mail and printed an icon on it. However, we are surprised to know that some users do not aware of this “key 1” shortcuts, so now we promoted this configuration, to help users more confident and convenient to use key 1 to access a preconfigured voice mail.



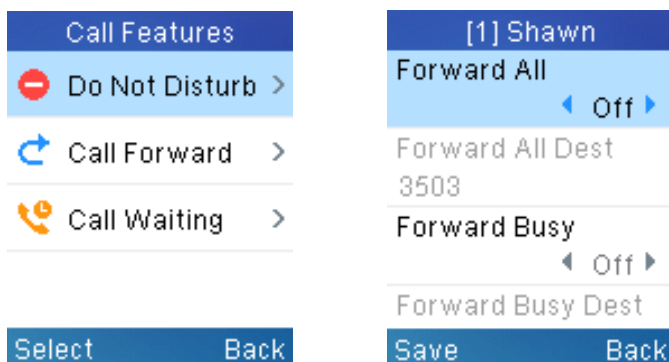
- **Calling Operation**

Now when you have 2 concurrent calls on the same handset, you will be able to select between 2 calls by Up and Down key and choose to how to operate the calls. The on-hook (red button) will always act for hang-up the current selected call. This makes the call operation more intuitive.



- **Call Features**

Call features are now all line specific settings, compare to preferences that are all handset specific settings. We also improved the Call Forward settings to allow users to quickly enable and disable the settings without wipe the forward destination number. On 1.0.4.10 when you disabled the call forward, the destination number will be also removed.

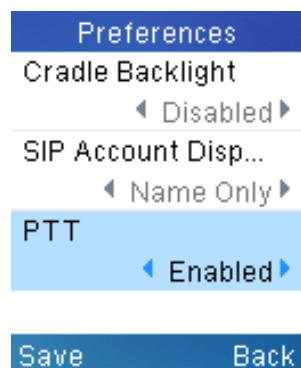


DP720 PTT FEATURE

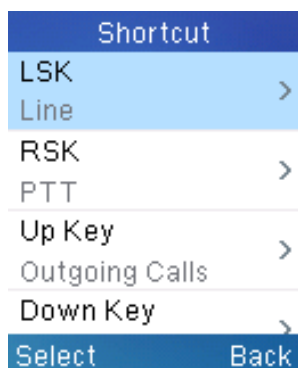
- **Web Configuration**

PTT feature is always enabled on the base station, to use the PTT feature, users need enable the PTT on handset under Handset Menu -> Preferences -> PTT -> set "Enabled". Also, to use PTT, the handset need a dedicated button to push and then to talk, user can choose to use either LSK (Left Softkey) or RSK (Right Softkey) in Shortcuts settings. (If you forgot to enable the PTT, when you press the PTT softkey, it will redirect you to the setting to enable it before you can use it.)

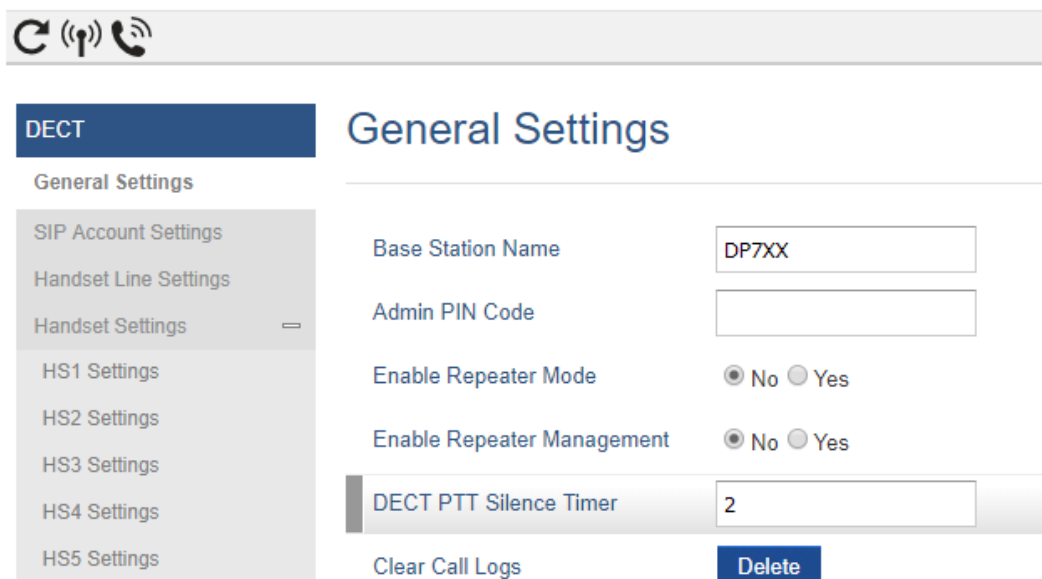
(Step1: Enable)



(Step2: Choose a softkey)



Users can configure “DECT PTT Silence Timer” on the Web UI at -> DECT -> General Settings. When a PTT call established, any handset who enabled PTT feature will automatically join the PTT call. If there is no one talking, after certain time, the PTT call will be closed. This timeout setting is to determine how long the PTT call will be closed after being idle status. If set to 0, the timer will be disabled, and the PTT will be remaining forever until all party exit the PTT call.

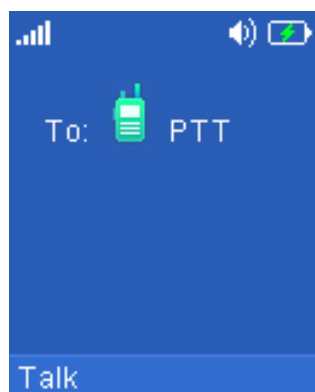


• Functionality

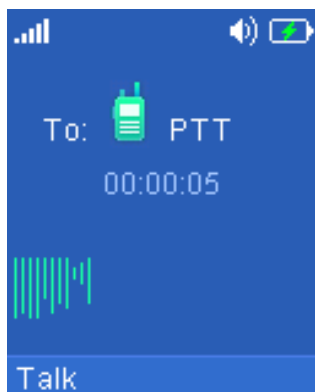
To use the PTT feature, simply press and hold the configured softkey (LSK or RSK), after you see the audio wave in green color, you can talk, and other handsets that registered on the same base station who enabled PTT feature will automatically join the PTT and play the voice on speaker.

Note: when initial a PTT call, it could take a little bit longer to establish the call for handshake, once the PTT call been established, any handset can quickly start to talk by holding the softkey.

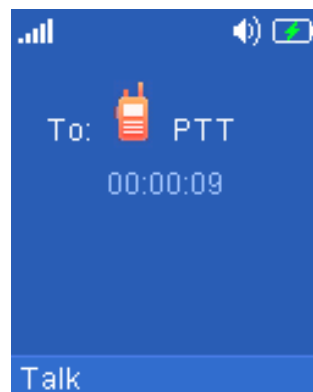
(Initialazing)



(Talking)



(Idle)



- New P Values**

Pvalue	Alias	Description	Value range	Default
P27778	Pbase.ptt.silence_timer	<New Pvalue> DECT PTT Silence Timer. Sets timeout for PTT call (in minutes) if no handset unmutes. If set to 0, this timer will be disabled.	0 ~ 60	2

REMOTE OPENDOOR FOR GDS3710/GDS3705

- Web Configuration**

The settings can be found at Web UI -> Settings -> Network Settings -> External Service. DP series phones works the same way as GXP phones. For more detail, please refer document:

http://www.grandstream.com/sites/default/files/Resources/Connecting_GDS3710_with_GXP_phones.pdf


Settings

Network Settings

Basic Settings

Advanced Settings

OpenVPN® Settings

External Service

Ring Tones

System Features

Grandstream Door System

Order	Service Type	Account	System Identification	System Number	Access Password
1	GDS ▼	Account 1 ▼	Front Door	3501
2	None ▼	Account 1 ▼			
3	None ▼	Account 1 ▼			
4	None ▼	Account 1 ▼			
5	None ▼	Account 1 ▼			
6	None ▼	Account 1 ▼			
7	None ▼	Account 1 ▼			
8	None ▼	Account 1 ▼			
9	None ▼	Account 1 ▼			
10	None ▼	Account 1 ▼			

Save

Save and Apply

Reset

• Functionality

After feature been configured, when GDS37xx doorbell been pressed, DP will receive an incoming call and play the doorbell ring. User will be able to press the “Open” button to remote open the door.

• New P Values

Pvalue	Alias	Description	Value range	Default
P32000	Pexternal_service.type.1	<New Pvalue> Service Type.	-1 – None 0 – GDS	-1 - None
P32005	Pexternal_service.type.2			
P32010	Pexternal_service.type.3			
P32015	Pexternal_service.type.4			
P32020	Pexternal_service.type.5			
P32025	Pexternal_service.type.6			
P32030	Pexternal_service.type.7			
P32035	Pexternal_service.type.8			
P32040	Pexternal_service.type.9			
P32045	Pexternal_service.type.10			
P32001	Pexternal_service.account.index.1	<New Pvalue>	0 – Account 1	0 – Account1
P32006	Pexternal_service.account.index.2		1 – Account 2	

P32011	Pexternal_service.account.index.3	Account. Specify which account to associate with the service	2 – Account 3	
P32016	Pexternal_service.account.index.4		3 – Account 4	
P32021	Pexternal_service.account.index.5		4 – Account 5	
P32026	Pexternal_service.account.index.6		5 – Account 6	
P32031	Pexternal_service.account.index.7		6 – Account 7	
P32036	Pexternal_service.account.index.8		7 – Account 8	
P32041	Pexternal_service.account.index.9		8 – Account 9	
P32046	Pexternal_service.account.index.10		9 – Account 10	
P32002	Pexternal_service.system.id.1	<New Pvalue> System Identification. to configure a unique identity to differentiate each service.	Strings	NULL
P32007	Pexternal_service.system.id.2			
P32012	Pexternal_service.system.id.3			
P32017	Pexternal_service.system.id.4			
P32022	Pexternal_service.system.id.5			
P32027	Pexternal_service.system.id.6			
P32032	Pexternal_service.system.id.7			
P32037	Pexternal_service.system.id.8			
P32042	Pexternal_service.system.id.9			
P32047	Pexternal_service.system.id.10			
P32003	Pexternal_service.system.number.1	<New Pvalue> System Number. For GDS service, this is used to configure the GDS SIP account or IP address (IP Call)	Strings	NULL
P32008	Pexternal_service.system.number.2			
P32013	Pexternal_service.system.number.3			
P32018	Pexternal_service.system.number.4			
P32023	Pexternal_service.system.number.5			
P32028	Pexternal_service.system.number.6			
P32033	Pexternal_service.system.number.7			
P32038	Pexternal_service.system.number.8			
P32043	Pexternal_service.system.number.9			
P32048	Pexternal_service.system.number.10			
P32004	Pexternal_service.password.1	<New Pvalue> Access Password. For GDS service, this is to configure the GDS remote access code which used to open the door.	Strings	NULL
P32009	Pexternal_service.password.2			
P32014	Pexternal_service.password.3			
P32019	Pexternal_service.password.4			
P32024	Pexternal_service.password.5			
P32029	Pexternal_service.password.6			
P32034	Pexternal_service.password.7			
P32039	Pexternal_service.password.8			
P32044	Pexternal_service.password.9			
P32049	Pexternal_service.password.10			

MUSIC ON HOLD URI

- Web Configuration**

The settings can be found at Web UI -> Profiles -> Profile {1-4} -> Call Settings

Anonymous Call Rejection ☒ No ☐ Yes

Special Feature Standard ▼

Music On Hold URI

Save Save and Apply Reset

- Functionality**

Configure the music on hold URI to call when a call is on hold if server support it.

- New P Values**

Pvalue	Alias	Description	Value range	Default
P2350	Pprofile.1.sip.moh_uri	<New Pvalue>	Strings	NULL
P2450	Pprofile.2.sip.moh_uri	Music on Hold URI		
P2550	Pprofile.3.sip.moh_uri			
P2650	Pprofile.4.sip.moh_uri			

USE RANDOM SIP PORT

- Web Configuration**

The settings can be found at Web UI -> Profiles -> Profile {1-4} -> SIP Settings -> Basic Settings -> Use Random SIP Port.

Reregister Before Expiration

Local SIP Port

Use Random SIP Port ☒ No ☐ Yes

Local RTP Port

Use Random RTP Port ☒ No ☐ Yes

- Functionality**

Configure to use random SIP port or not, if configured, the Local SIP Port setting will be ignored.

- New P Values**

Pvalue	Alias	Description	Value range	Default
P20501	Pprofile.1.network.use_random_sip_port	<New Pvalue>	0 – No	0 - No
P20502	Pprofile.2.network.use_random_sip_port		1 – Yes	

P20503	Pprofile.3.network.use_random_sip_port	Use a random port for SIP messages rather than the Local SIP Port.		
P20504	Pprofile.4.network.use_random_sip_port			

NEW FORMAT OF PVALUES

- Web Configuration**

The downloaded config file now will display the P value in new format, that has more meaningful name.

Parameters Config File

Upload Device Configuration

Upload Device Config

Download Device Configuration

Device Config (Text)	Download
Device Config (XML)	Download

Backup Configuration

Backup Configuration

Backup Configuration NONE

- Functionality**

Below is an example of how new P values looks like, please note that the old numeric P values will still be working, for example, in a single config file, the user can choose to use P values in the format of P9 or Pnetworks.static.address.ip.octet.1, or mixed format for different settings, the device will takes all of them. We are still improving the downloaded config file by adding the help text and re-ordering the settings to match the web.

```

7 <!--## Configuration Template ##-->
8 <!--#####-->
9 <!--#####-->
10 <!--## P values ##-->
11 <!--#####-->
12 <!--# Name: 8-->
13 <!--# Alias: network.type,NETWORK_TYPE # Help: IP Address Mode. 0 - DHCP, 1 - Static IP, 2 - PPPoE.-->
14 <!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-2] -->
15 <!--# Default Value: 0-->
16 <Pnetwork.type>0</Pnetwork.type>
17
18
19 <!--# Name: 9-->
20 <!--# Alias: network.static.address.ip.octet.1,NETWORK_STATIC_ADDRESS_OCTET1 # Help: IP Address octet 1. Ignore if DHCP or PPPoE is
used.-->
21 <!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
22 <!--# Default Value: 192-->
23 <Pnetwork.static.address.ip.octet.1>50</Pnetwork.static.address.ip.octet.1>
24
25
26 <!--# Name: 10-->
27 <!--# Alias: network.static.address.ip.octet.2,NETWORK_STATIC_ADDRESS_OCTET2 # Help: IP Address octet 2. Ignore if DHCP or PPPoE is
used.-->
28 <!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
29 <!--# Default Value: 168-->
30 <Pnetwork.static.address.ip.octet.2>84</Pnetwork.static.address.ip.octet.2>
31
32
33 <!--# Name: 11-->
34 <!--# Alias: network.static.address.ip.octet.3,NETWORK_STATIC_ADDRESS_OCTET3 # Help: IP Address octet 3. Ignore if DHCP or PPPoE is
used.-->
35 <!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
36 <!--# Default Value: 0-->
37 <Pnetwork.static.address.ip.octet.3>167</Pnetwork.static.address.ip.octet.3>
38
39

```

FIRMWARE VERSION 1.0.4.10

PRODUCT NAME

DP750, DP720, DP760

DATE

10/30/2018

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: b6945458f447ea859a85196b1731728a

ENHANCEMENT

- Added option to add MAC address to User Agent for each profile. [OPTION FOR ADDING MAC ADDRESS TO USER AGENT]

BUG FIX

- Fixed DP750 phonebook XML file downloading will fail when the server is using TLS v1.2.
- Fixed DP750 distinctive ringtones do not work with alert-info header.
- Fixed DP720 does not display the contact name on the call history if the number starts with “+”.

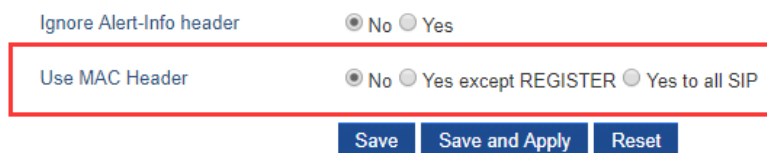
NEW FEATURES OVERVIEW

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

OPTION FOR ADDING MAC ADDRESS TO USER AGENT

- Web Configuration**

The settings can be found at Web UI -> Profiles -> Profile {1-4} -> SIP Settings -> Basic Settings -> Use MAC Header.



Ignore Alert-Info header ☒ No ☐ Yes

Use MAC Header ☒ No ☐ Yes except REGISTER ☐ Yes to all SIP

Save **Save and Apply** **Reset**

- Functionality**

By default, the settings is “No”, if set to “Yes except REGISTER”, the sip message for register or unregister will contains MAC address in the header, and all the outgoing SIP messages except REGISTER message

will attach the MAC address to the User-Agent header; If Yes to ALL, the sip message for register or unregister will contains MAC address in the header, and all the outgoing SIP message including REGISTER will attach the MAC address to the User-Agent header; If No, neither will the MAC header be included in the register or unregister message nor the MAC address be attached to the User-Agent header for any outgoing SIP message.

```
> Frame 434: 1178 bytes on wire (9424 bits), 1178 bytes captured (9424 bits) on interface 0
> Ethernet II, Src: Grandstr_87:08:a5 (00:0b:82:87:08:a5), Dst: Cisco_7c:36:33 (00:26:0b:7c:36:33)
> Internet Protocol Version 4, Src: 192.168.88.17, Dst: 192.168.41.55
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
▼ Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:013@192.168.41.55 SIP/2.0
  ▼ Message Header
    > Via: SIP/2.0/UDP 192.168.88.17:5060;branch=z9hG4bK1175063032;rport
    > From: "Remote 012" <sip:012@192.168.41.55>;tag=1063131456
    > To: <sip:013@192.168.41.55>
    Call-ID: 1182217752-5060-4@BJC.BGI.II.BH
    > CSeq: 10 INVITE
    > Contact: "Remote 012" <sip:012@192.168.88.17:5060>
    Max-Forwards: 70
    User-Agent: Grandstream DP750 1.0.4.10 000b828708a5
    Privacy: none
    > P-Preferred-Identity: "Remote 012" <sip:012@192.168.41.55>
    Supported: replaces, path, timer, eventlist
    Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE
    Content-Type: application/sdp
    Accept: application/sdp, application/dtmf-relay
    Content-Length: 453
  > Message Body
```

- **New P Values**

Pvalue	Description	Value range	Default
P26061	<New Pvalue>	0 – No	0 - No
P26161	Configure if use MAC address in the	1 – Yes except REGISTER	
P26261	User Agent	2 – Yes to all SIP	
P26361			

FIRMWARE VERSION 1.0.4.7

PRODUCT NAME

DP750, DP720, DP760

DATE

10/12/2018

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 833d7b7b226234f02892b00920165a67

ENHANCEMENT

- Added the ability to display Account Name only. [DISPLAY ACCOUNT NAME ONLY]
- Added support to provision config file by model. [PROVISION CONFIG FILE BY MODEL]
- Added support TLS protocol for Syslog. [TLS PROTOCOL FOR SYSLOG]
- Added SRTP icon on handset UI. [SRTP ICON ON HANDSET UI]
- Updated German translations for both DP750 Web UI and DP720 handset UI

BUG FIX

- Fixed DP750 CTI feature related issue that handset would not return to idle after it received api-send_key request.
- Fixed DP750 CIT feature related issue that phone number sometimes get partially missing upon api-make_call request.
- Fixed DP750 issue that under 802.1x network environment if unplug network cable and re-plug it, the following calls has no audio.
- Fixed DP750 issue that if answer a call that nearly timeout, the call has no audio.
- Fixed DP750 sometimes keep sending REGISER and cannot make calls.
- Fixed DP750 TLS registration not working properly sometimes.
- Fixed DP750 still sending codec PCMU after transferred even when the codec G729 is the only configured codec.
- Fixed DP750 "Crypto Life Time" does not reload after it enabled or disabled.
- Fixed DP750 "Handset Config File Prefix" cannot be set to empty, it always switch to default value when set to empty.
- Fixed DP750 issue that after changing SIP protocol from UDP to TCP or from TCP to UDP without reboot, the following calls might introduce big noise.
- Fixed DP750 OpenVPN set option "comp-lzo no" to disable the LZO-compression does not working.

- Fixed DP720 issue that it may fail to exit ringing interface until manually return to idle, and the DP750 responses “486 busy” for any incoming call when staying in ringing.
- Fixed DP720 right Softkey does not show “delete” when editing last name/Home/Work/Mobile in contacts edit mode.
- Fixed DP760 Repeater network LED does not on when use PPPoE network mode.

NEW FEATURES OVERVIEW

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

DISPLAY ACCOUNT NAME ONLY

- **Functionality**

On DP720 browse to Settings -> User Settings -> SIP Account Display, it can be set to “Name & ID”, “Name Only”, “ID only” or “None”. DP720 will display both Account Name and SIP User ID and, Account Name Only, SIP User ID or none accordingly.

PROVISION CONFIG FILE BY MODEL

- **Web Configuration**

There is no need to configure anything on the web UI, by default this feature is always enabled.

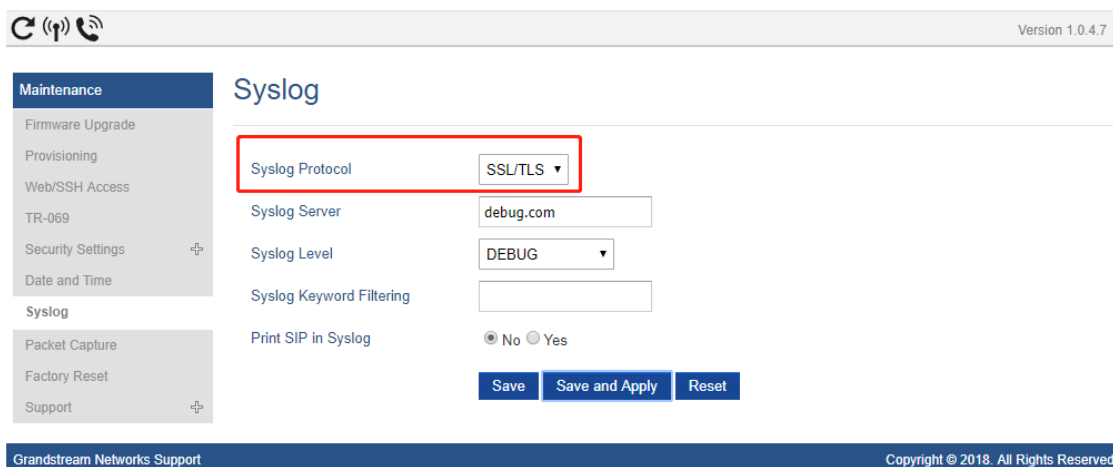
- **Functionality**

When DP750 conduct provisioning, the device will request cfg{MAC}.xml and cfg.xml file, after added this enhancement, DP750 will also request cfgdp750.xml to allow provisioning devices by a model specified xml file.

TLS PROTOCOL FOR SYSLOG

- **Web Configuration**

On DP750 Web -> Maintenance -> Syslog, added an option to set “Syslog Protocol” to “SSL/TLS” other than default “UDP”.



Version 1.0.4.7

Maintenance

- Firmware Upgrade
- Provisioning
- Web/SSH Access
- TR-069
- Security Settings
- Date and Time
- Syslog**
- Packet Capture
- Factory Reset
- Support

Syslog

Syslog Protocol: **SSL/TLS**

Syslog Server: debug.com

Syslog Level: **DEBUG**

Syslog Keyword Filtering:

Print SIP in Syslog: ☒ No ☐ Yes

Save Save and Apply Reset

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- **Functionality**

After configured, all the Syslog will be sent under SSL/TLS protocol.

- **New P Values**

Pvalue	Description	Value range	Default
P8402	<New Pvalue> Allow sending syslog through secured TLS protocol to syslog server Note: CA verification is required	0 – UDP 1 – SSL/TLS	0 – UDP

SRTP ICON ON HANDSET UI

- **Web Configuration**

There is no need to configure anything on the web UI, by default this feature is always enabled.

- **Functionality**

When using SRTP for call, there will be an icon on handset UI to indicate that.

FIRMWARE VERSION 1.0.4.2

PRODUCT NAME

DP750, DP720, DP760

DATE

07/19/2018

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: e46cd212301a5ebbd30231765e2da6b4

IMPORTANT UPGRADING NOTES

- **For DP750/DP720/DP760, once upgraded to 1.0.4.2 or above, downgrading to 1.0.3.40 or previous firmware version is not supported.**
- **For DP750/DP720, firmware 1.0.0.99 (The first public released beta firmware) and all above can upgrade to 1.0.4.2 directly, no transitional firmware is needed.**

ENHANCEMENT

- Added support for CTI (Computer-Telephony Integration) feature as the same as GXP phones. [CTI FEATURE]
- Improve the support to customize the Syslog port. [CUSTOMIZE THE SYSLOG PORT]
- Added support for DP720 to use Audio Loopback test when it is not registered
- Play keypad tone during the call if it is enabled

BUG FIX

- Fixed DP750 User Agent header uses wrong MAC format when downloading phonebook.xml
- Fixed P6007 doesn't return configure value in SSH
- Fixed DP750 cannot access web and cannot upgrade/download when using OpenVPN
- Fixed the default values for Verify host when using HTTPS that under Provision and Upgrade/Downgrade are different, set both value to enabled
- Fixed DP750 always registers to Failover SIP Sever First if configured, and fails to register to the second IP of SRV Records of Primary SIP server
- Fixed DP750 web display a crash warning while there isn't a core dump
- Fixed DP750 circular mode sometimes two sides have no sound when the second incoming call timeout

- Fixed DP750 under repeater mode, when two handsets set up 2 concurrent calls on the same repeater DP760, a third handset can still switch to this repeater
- Fixed DP720 customized ringtone file will be corrupt when rebooting during downloading
- Fixed DP720 does not play confirmation tone when saving language settings
- Fixed DP720 does not play the ringtone when change the volume after received a missed call.
- Fixed DP720 mute key misfunction if mute by long pressing 3 second and unmute by a quick pressing, the handset LCD shows unmuted, but the handset still stay in mute mode.
- Fixed DP720 system ring become abnormal after playing ringtone on handset and configuring tone on DP750 Web at the same time
- Fixed DP720 does not play keypad tone if unsubscribed and re-subscribed and press off-hook or speaker key first
- Fixed DP720 answer incoming call during searching base will lead the call become abnormal
- Fixed DP720 show "Error" and back to idle automatically after access and exit to menu quickly
- Fixed DP720 cannot switch to the last call record when pressing up key on the first record

NEW FEATURES OVERVIEW

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

CTI FEATURE

- **Web Configuration**

There is no need to configure anything on the web UI, by default this feature is always enabled.

- **Functionality**

Here is the full user manual for CTI feature of GXP phones:

http://www.grandstream.com/sites/default/files/Resources/CTI_Guide.pdf

For DP750/DP720/DP760, all the usage is the same except:

1. GXP is single phone, while DP750 could register 5 handsets, therefore, for some URL, it will need specific which handset it is.
2. DP720 does not support as many keys as GXP phones. Here are all the keys that DP720 support: 0-9, HASH, *, LEFT_SOFT_KEY, RIGHT_SOFT_KEY, MUTE, SPEAKER, STAR, ONHOOK, OFFHOOK, OK, LOCK, UNLOCK, UP, DOWN, LEFT, RIGHT.

Command and Example:

- `api-get_account_status`

Accepted arguments: account

Example: http://192.168.5.135/cgi-bin/api-get_account_status?passcode=admin&account=2

- `api-get_hs_status`

Accepted arguments: hs

Example: http://192.168.5.135/cgi-bin/api-get_hs_status?passcode=admin&hs=1

- api-get_hs_line_status

Accepted arguments: hs

Example: http://192.168.5.135/cgi-bin/api-get_hs_line_status?passcode=admin&hs=1

- api-make_call

Accepted arguments: hs, phonenumber

Example: http://192.168.5.135/cgi-bin/api-make_call?passcode=admin&hs=1&phonenumber=3501

- api-device_operation

Accepted arguments: hs, line, cmd(cmd can be endcall, holdcall, unholdcall, acceptcall, rejectcall, cancel, speaker, mute, setdnd, unsetdnd)

Example:

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=endcall

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=holdcall

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=unholdcall

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=acceptcall

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=rejectcall

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&line=1&cmd=cancel

For mute, speaker and setdnd, unsetdnd command, there is no "line=x" argument.

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&cmd=mute

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&cmd=speaker

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&cmd=setdnd

http://192.168.5.135/cgi-bin/api-device_operation?passcode=admin&hs=1&cmd=unsetdnd

- api-send_key

Accepted arguments : hs, keys(keys are in format <key1>:<key2>:<key3>:....., keys can be 0-9, HASH, *, LEFT_SOFT_KEY, RIGHT_SOFT_KEY, MUTE, SPEAKER, STAR, ONHOOK, OFFHOOK, OK, LOCK, UNLOCK, UP, DOWN, LEFT, RIGHT)

Example: http://192.168.5.135/cgi-bin/api-send_key?passcode=admin&hs=1&keys=3:3:0:9:HASH

Please be aware that # key need be sent as HASH on DP750/DP720.




All the response from HTTP request, please check the GXP CTI user manual.

CUSTOMIZE THE SYSLOG PORT

- **Web Configuration**

To customize the Syslog port, login DP750 Web, and just add the port after the Syslog Server address.

Previously we added a separated option and P value to configure the Syslog port, which is not consistent with other products and since we already support port after Server address like the SIP server and other similar settings, we decide to remove the delicate settings for Syslog port, but just use it along with the Syslog Server address. Now you can still configure this value by using the P207 which is for Syslog Server address.



Version 1.0.4.2

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings

Date and Time

Syslog

Packet Capture

Factory Reset

Support

Syslog

Syslog Server

1.2.3.4:1401

Syslog Level

DEBUG

Print SIP in Syslog

☒ No ☐ Yes

Save

Save and Apply

Reset

FIRMWARE VERSION 1.0.3.40

PRODUCT NAME

DP750, DP720, DP760

DATE

05/22/2018

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: a719a1bb6c9b385e2414e7591406486a

ENHANCEMENT

- Added support to allow configure prefix for User-Agent in SIP header. [USER-AGENT PREFIX]
- Increased the length of Authenticate ID, SIP User ID, Password from 32 to 128.
- Added support for “.#”, “.T” and “T” and alphanumeric characters in the Dial Plan.
- Added support for SIP OPTIONS Keep Alive. [USE SIP OPTIONS TO KEEP ALIVE]
- Added support for automatic upgrade and provisioning be triggered at a random time in every certain days. [RANDOM UPGRADE AND PROVISION]
- Removed force core generation from web
- Added OpenVPN feature as the same as GXP phones. [OPENVPN]
- Added the option to set Date and Time Display Format via the web GUI and Provisioning. [CONFIGURE DATE AND TIME DISPLAY]
- Increased the number of digits that can be dialed on the handset

BUG FIX

- Fixed sometime DP750 failed register accounts sometimes when using SIP/TLS
- Fixed DP750 does not save certain allowed inputs in the Dial Plan
- Fixed DUT crashed after downloaded some files via TR069
- Fixed the SIP Authentication password is visible from the handset
- Fixed When handset line1 set to none, handset off hook didn't have default outgoing line configured
- Fixed Pvalue template that downloaded from device Web UI have some Pvalues missing.

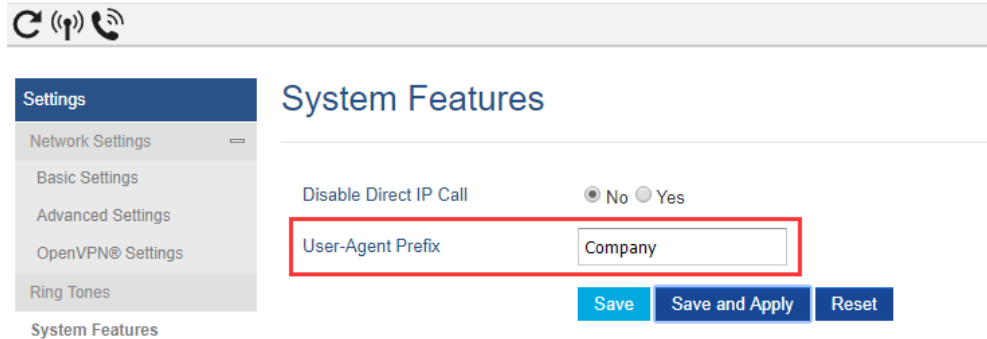
NEW FEATURES OVERVIEW

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

USER-AGENT PREFIX

- Web Configuration**

Add support to allow configure customized User-Agent Prefix. The configuration is under Web UI -> Settings -> System Features -> User-Agent Prefix.



The screenshot shows the 'System Features' configuration page. On the left is a sidebar with 'Settings' expanded, showing 'Network Settings', 'Basic Settings', 'Advanced Settings', 'OpenVPN® Settings', and 'Ring Tones'. Below this is 'System Features'. The main area has a toggle for 'Disable Direct IP Call' set to 'No'. Below that, the 'User-Agent Prefix' field is highlighted with a red box and contains the text 'Company'. At the bottom are three buttons: 'Save', 'Save and Apply', and 'Reset'.

- Functionality**

```

v Session Initiation Protocol (REGISTER)
  > Request-Line: REGISTER sip:192.168.254.5 SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 192.168.219.10:5064;branch=z9hG4bK642335446;rport
    > From: "3503" <sip:3503@192.168.254.5>;tag=742784024
    > To: <sip:3503@192.168.254.5>
    > Call-ID: 295784746-5064-1@BJC.BGI.CBJ.BA
    > CSeq: 2000 REGISTER
    > Contact: <sip:3503@192.168.219.10:5064>;reg-id=3;+sip.instance="urn:uuid:00000000-0000-1000-8000-000B828A4768"
    > Max-Forwards: 70
    > User-Agent: Company Grandstream DP750 1.0.3.40
    > Supported: path
    > Expires: 3600
    > Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE
    > Content-Length: 0
  
```

- New P Values**

Pvalue	Description	Value range	Default
P8358	<New Pvalue> Configure the prefix in the User-Agent header.	String	null

USE SIP OPTIONS TO KEEP ALIVE

- Web Configuration**

Add option to use SIP OPTIONS to Keep Alive. It's under Web UI -> Profiles X -> SIP Settings -> Basic Settings.



Profiles

Profile 1 ⌵
 General Settings
 Network Settings
 SIP Settings ⌵
 Basic Settings
 Session Timer
 Security Settings
 Audio Settings
 Call Settings
 Profile 2 ⌵
 Profile 3 ⌵
 Profile 4 ⌵

Basic Settings

SIP Transport	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS/TCP
SIP Registration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Unregister on Reboot	<input type="radio"/> No <input checked="" type="radio"/> Yes
Add Auth Header On Initial REGISTER	<input checked="" type="radio"/> No <input type="radio"/> Yes
Outgoing Calls Without Registration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Register Expiration	<input type="text" value="60"/>
SIP Registration Failure Retry Wait Time	<input type="text" value="20"/>
SIP Registration Failure Retry Wait Time upon 403 Forbidden	<input type="text" value="1200"/>
Reregister Before Expiration	<input type="text" value="0"/>
Local SIP Port	<input type="text" value="5060"/>
Use Random SIP Port	<input checked="" type="radio"/> No <input type="radio"/> Yes
Local RTP Port	<input type="text" value="5004"/>
SIP T1 Timeout	<input type="text" value="0.5 sec"/>
SIP T2 Timeout	<input type="text" value="4 sec"/>
SIP TIMER D	<input type="text" value="0"/>
Enable OPTIONS Keep Alive	<input checked="" type="radio"/> No <input type="radio"/> Yes
OPTIONS Keep Alive Interval	<input type="text" value="30"/>
OPTIONS Keep Alive Max Lost	<input type="text" value="3"/>
Remove OBP from Route	<input checked="" type="radio"/> No <input type="radio"/> Yes
Support SIP Instance ID	<input type="radio"/> No <input checked="" type="radio"/> Yes

• New P Values

Pvalue	Description	Value range	Default
P2397 (Profile 1) P2497 (Profile 2) P2597 (Profile 3) P2697 (Profile 4)	<New Pvalue> Enable OPTIONS Keep Alive, to check SIP server	0 – No 1 – Yes	0 – No
P2398 (Profile 1) P2498 (Profile 2) P2598 (Profile 3) P2698 (Profile 4)	<New Pvalue> OPTIONS Keep Alive Interval: Time interval for OPTIONS Keep Alive feature in seconds	1 – 64800	30
P2399 (Profile 1) P2499 (Profile 2) P2599 (Profile 3)	<New Pvalue>	3 – 10	3

P2699 (Profile 4)	OPTIONS Keep Alive Max Lost: Maximum number of lost packets for OPTIONS Keep Alive feature before the phone re-registration		
-------------------	---	--	--

RANDOM UPGRADE AND PROVISION

- Web Configuration**

The setting is under Web UI -> Maintenance -> Firmware Upgrade / Provisioning:

Verify host when using HTTPS ☐ No ☒ Yes

Upgrade via ☐ TFTP ☒ HTTP ☐ HTTPS

Firmware Server Path

Firmware File Prefix

Firmware File Postfix

Allow DHCP Option 43 and Option 66 to Override Server ☐ No ☒ Yes

Automatic Upgrade ☒ No

☐ Yes, check for upgrade at random every day(s)

☐ Yes, check for upgrade every minute(s)

☐ Yes, check for upgrade every day

☐ Yes, check for upgrade every week

Random Interval Start Time :

Random Interval Stop Time :

Hour of the Day(0-23)

Day of the Week (0-6)

Firmware Key

Handset Firmware

Handset firmware

Automatic Upgrade ☒ No ☐ Yes

- Functionality**

This feature is used to set DP750 or DP760 automatic upgrade or provisioning at a random time in a certain period. You can set the device randomly automatic upgrade in X days and run upgrade / provisioning between the start time and stop time. Reboot the device will not lose the pending upgrade / provisioning.

- New P Values**


Pvalue	Description	Value range	Default
P27816 (Upgrade)	<New Pvalue>	1 – 3650	1

P27811 (Provisioning)	check for upgrade at random every X day(s).		
P27817	<New Pvalue>	0 – 23	1
P27818	Random upgrade start hour and start	0 – 59	1
P27819	minutes, stop hour and stop minutes.	0 – 23	1
P27820		0 – 59	1
P27812	<New Pvalue>	0 – 23	1
P27813	Random provisioning start hour and start	0 – 59	1
P27814	minutes, stop hour and stop minutes.	0 – 23	1
P27815		0 – 59	1

OPENVPN

- **Web Configuration**

The setting is under Web UI -> Settings -> Network Settings -> OpenVPN Settings. After configured OpenVPN, you can also choose to use VPN as NAT Traversal at Profile X -> Network Settings -> NAT Traversal.



Settings
 Network Settings
 Basic Settings
 Advanced Settings
OpenVPN® Settings
 Ring Tones
 System Features

OpenVPN® Settings

OpenVPN® Enable ☐ Yes ☒ No

OpenVPN® Server Address

OpenVPN® Port

OpenVPN® Transport ☒ UDP ☐ TCP

OpenVPN® CA

OpenVPN® Certificate

OpenVPN® Client Key

OpenVPN® Cipher Method ☒ Blowfish ☐ AES-128 ☐ AES-256 ☐ Triple-DES

OpenVPN® Username

OpenVPN® Password

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Profiles

Profile 1

General Settings

Network Settings

SIP Settings

Audio Settings

Call Settings

Profile 2

Profile 3

Profile 4

Network Settings

Layer 3 QoS Settings

SIP Diff-Serv: 24

RTP Diff-Serv: 46

DNS Settings

DNS Mode: A Record

Primary IP:

Backup IP 1:

Backup IP 2:

NAT Settings

NAT Traversal: No

Use NAT IP:

Proxy-Require:

Save Save and Apply Reset

- New P Values**

Pvalue	Description	Value range	Default
P7050	<New Pvalue> Enable or disable OpenVPN®	0 – No 1 – Yes	0
P7051	<New Pvalue> OpenVPN® Server Address	String	Null
P7052	<New Pvalue> OpenVPN® Port	1 – 65535	1194
P2912	OpenVPN® Transport Determines network protocol used for OpenVPN®	0 – UDP 1 – TCP	0 – UDP
P8394	<New Pvalue> OpenVPN® username	String	Null
P8395	<New Pvalue> OpenVPN® password	String	Null
P8396	<New Pvalue> OpenVPN® cipher Method	0 – Blowfish 1 – AES-128 2 – AES-256 3 – Triple-DES	0

CONFIGURE DATE AND TIME DISPLAY

- Web Configuration**

The Setting is under Web UI -> DECT -> Handset Settings -> HS X Settings, to configure the display format of Time and Date.

DECT

- General Settings
- SIP Account Settings
- Handset Line Settings
- Handset Settings
- HS1 Settings**
- HS2 Settings
- HS3 Settings
- HS4 Settings
- HS5 Settings

HS1 Settings

Disable Private Phonebook

☒ No ☐ Yes

Enable Auto Answer

☒ No ☐ Yes

Enable Offhook on Cradle Pickup

☒ No ☐ Yes

Enable Onhook on Cradle Reposition

☒ No ☐ Yes

Disable Conference

☒ No ☐ Yes

Disable Transfer

☒ No ☐ Yes

Disable Busy Tone on Remote Disconnect

☒ No ☐ Yes

Disable Call Waiting Tone

☒ No ☐ Yes

Custom Ringtone

Disabled ▾

Time Format

12hr ▾

Date Format

DD-MM-YYYY ▾

Save

Save and Apply

Reset

- New P Values**

Pvalue	Description	Value range	Default
P30080 (HS 1)	<New Pvalue> Set the displayed Time Format on handsets.	0 – 12hr	0 – 12hr
P30081 (HS 2)		1 – 24hr	
P30082 (HS 3)			
P30083 (HS 4)			
P30084 (HS 5)			
P30060 (HS 1)	<New Pvalue> Set the displayed Date Format on handsets.	0 – DD-MM-YYYY	0
P30061 (HS 2)		1 – MM-DD-YYYY	
P30062 (HS 3)		2 – YYYY-MM-DD	
P30063 (HS 4)			
P30064 (HS 5)			

FIRMWARE VERSION 1.0.3.37

PRODUCT NAME

DP750, DP720, DP760

DATE

03/16/2018

FIRMWARE FILE INFORMATION

- Firmware file name: dp75xfw.bin

MD5: 7f2e584240b0d230079fd92b6355799a

ENHANCEMENT

- Added support for DP750 to configure device with custom certificate signed by custom CA certificate. [DEVICE CUSTOM CERTIFICATE SETTINGS]
- Added support for DP750 to customize the Syslog port. [CUSTOMIZE THE SYSLOG PORT]
- Reset DP760 repeater region to AUTO when pressing page and reset button more than 2s. [DP760 AUTO REGION]
- Increased all volume levels by +4dB

BUG FIX

- Issued a new DP750 HTTPS default certificate for it had expired. New issue certificate will be expired in year 2038.
- Fixed DP750 issue that it cannot register the SIP account when using SRV.
- Fixed DP720 issue that after handset selected another base the original base still marked it as online.
- Fixed DP750 issue that it fails lookup LDAP contact names during ringing if the number starts with +.
- Fixed DP720 keep rebooting automatically when battery is low.
- Fixed DP750 crashes and generates core dump randomly when transferring a call.
- Fixed DP720 automatically reboot after loading phonebook that included specific Japanese/Russian character.
- Fixed DP750 issue that handset fails to receive incoming call but response SIP 486 error
- Fixed DP750 cannot send BYE message when using TLS in some specific SIP proxy.
- Fixed DP720 would not play ringtone after pressed speaker to answer a call.

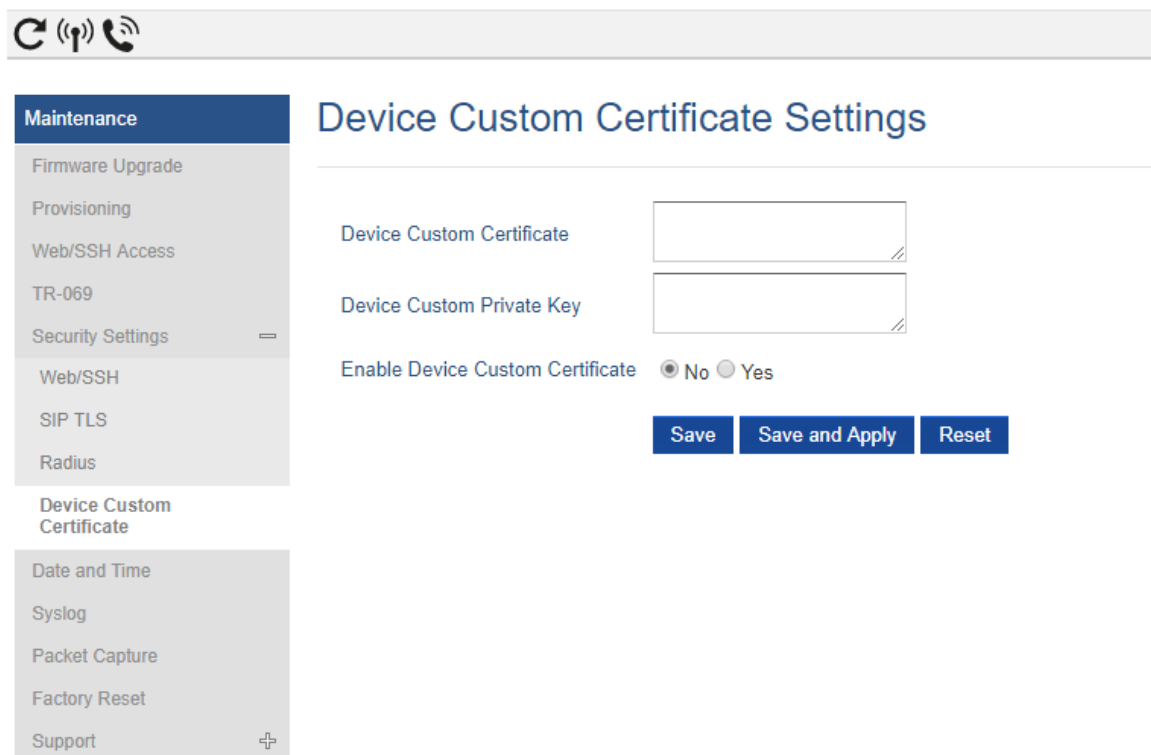
NEW FEATURES OVERVIEW

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

DEVICE CUSTOM CERTIFICATE SETTINGS

- Web Configuration**

Add custom certificate support, now the user can configure their own custom certificate and private key for their own HTTPS server. It's under Web UI -> Maintenance -> Security Settings -> Device Custom Certificate Settings.






- New P Values**

Pvalue	Description	Value range	Default
P27776	<New Pvalue> Device Custom Certificate: Certificate used by device during provisioning and web access	String	null
P27775	<New Pvalue> Device Custom Private Key: Key used by device during provisioning and web access	String	null
P27777	<New Pvalue> Enable Device Custom Certificate: Enable it will use custom certificate else use default certificate	0 – No 1 – Yes	0 – No

CUSTOMIZE THE SYSLOG PORT

- Web Configuration**

Add option to configure Syslog port. It's under Web UI -> Maintenance -> Syslog.

Version 1.0.3.37

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings

Date and Time

Syslog

Packet Capture

Factory Reset

Support

Syslog

Syslog Server

192.168.219.5

Syslog port

514

Syslog Level

DEBUG ▼

Print SIP in Syslog

☐ No
 ☒ Yes

Save

Save and Apply

Reset

- New P Values**

Pvalue	Description	Value range	Default
P8472	<New Pvalue> Syslog port: Syslog messages will be sent to this port instead of the default syslog port of 514	Integer (1 – 65535)	514

DP760 AUTO REGION

Now DP760 supports auto region for US/Europe/Brazil mode.

After pressing Page/Reset button for more than 2 seconds, DP760 will enter AUTO region mode, you can use any region DP750 to pair with it, after the DP760 paired with a base station, it will set it's region to the same as the base station (DP750).

Note:

1. If you have a DP760 that has FW before 1.0.3.34, you will need upgrade it to 1.0.3.34 first, then do a factory reset. After that, the unit will support Auto-Region feature.
2. By default, after factory reset, the DP760 will enter Auto Region mode.

FIRMWARE VERSION 1.0.3.34

PRODUCT NAME

DP750, DP720

DATE

02/05/2018

IMPORTANT UPGRADING NOTES

- **Upgrading handsets from web UI is disabled while in repeater mode. Please move the handset in range of DP750 base station and start upgrading from handset.**

ENHANCEMENT

- Enhanced the stability when working with DP760 repeater.

FIRMWARE VERSION 1.0.3.31

PRODUCT NAME

DP750, DP720

DATE

11/28/2017

IMPORTANT UPGRADING NOTE

- **Upgrade of handsets from web UI is disabled while in repeater mode. Please move the handset in range of DP750 base station and start upgrade from handset.**

ENHANCEMENT

- Added language support for Slovakian, Swedish.
- Hide RPN from web UI.
- Add icon or symbol to indicate if the handset is using repeater or not

BUG FIX

- Change NTP Server default value to pool.ntp.org.
- Fixed Phonebook contacts with numbers leading with "0" are not resolved in incoming/outgoing calls.
- Fixed the receiver cannot hear the voice if it answered the call from DP720 quickly.
- Fixed DP720 cannot play ring-back tone if it as transferee in blind transfer.
- Fixed DP750 didn't send RTP and there is no audio in the call after the receiver of transfer was ringing timeout.
- Fixed DP720 can still receive incoming call after long pressing mute key to enable DND function.
- Fixed DP750 issue of CPE closes the TCP connection before sending empty HTTP POST request.
- Fixed DP750 crashed after downloaded some files via TR069.
- Fixed DP720 can't search base even that base have opened subscribe
- Fixed DP760 not able to be managed by DP750 via web when using VLAN
- Fixed DP750 still allows more than 3 current calls under repeater mode, that lead more than 5 channels be used and causes device crashes, now DP750/DP720 will only able to make 2 current calls when enabled Repeater Mode.

FIRMWARE VERSION 1.0.3.23

PRODUCT NAME

DP750, DP720

DATE

03/14/2017

IMPORTANT UPGRADING NOTE

- For DP750/DP720, once upgraded to 1.0.3.23 or above, downgrading to 1.0.3.17 firmware version or lower is not supported.
- For DP750/DP720, once upgraded to 1.0.3.17 or above, downgrading to 1.0.2.16 firmware version or lower is not supported.
- For DP750/DP720, once upgraded to 1.0.2.16 or above, downgrading to 1.0.1.x firmware version or lower is not supported.
- For DP750/DP720, firmware 1.0.0.99 (The first public released beta firmware) and all above can upgrade to 1.0.3.23 directly, no transitional firmware needed.

ENHANCEMENT

- Improved maximum concurrent calls from 4 wide-band calls to 5.
- Added Email address to core debug page. [CORE COLLECTION EMAIL]
- Added DP720 factory reset combo key to be able to recover the device during booting up. [DP720 FACTORY RESET DURING BOOTING UP]
- Implemented Handset Provisioning feature. Add warning information to SIP Account Settings and Line Settings Pages if Handset Provisioning enabled. [HANDSET PROVISIONING FEATURE]
- Implemented DP750 Debug Tools that allow generate and submit debug files to Grandstream debug server directly. Add handset notification for base core dump files. [DP750 DEBUG TOOLS]
- Added web UI support to DP750 for repeater management such as display/link/unlink, etc. [REPEATER MANAGEMENT PAGE]
- Removed Tools/alarm menu from DP720 GUI, replaced by Status menu.
- Added keypad tones for non DTMF keys on DP720.
- Added subscribe Softkey in idle screen if HS no subscribed on DP720
- Added backup configuration feature. [BACKUP CONFIGURATION]
- Added UPnP discovery feature. [UPNP DISCOVERY]
- Added support for more languages and updated translation.
- Added DP750/DP720 product resource link to DP750 web UI footer at left side.
- Added support for page refresh by Refresh Page button, and added the icons of Refresh Page, Open Subscription and Page All button to icon bar. [ICON BAR WITH SHORTCUT ICON BUTTON]
- Added the ability to customize DP720 ringtones. [CUSTOMIZE DP720 RINGTONES]

- Enhanced factory reset feature by adding options to select Full RESET, RESET without deleting handsets subscription or RESET only deleting handset subscription. [ENHANCED FACTORY RESET]
- Added capability to rename the profile. [RENAME THE PROFILE]
- Added support for Distinctive Ringtone based on a customized SIP Alert-info header
- Added support to upload TXT configuration file, now user can upload TXT configuration file as the same as XML configuration files.
- Added support to auto-refresh on dect status web UI
- Added a new tab named "Support" under Maintenance page for support document, configuration support and debug tools.
- Added DP720 walk through setup wizard after factory reset.
- Added DP750 the ability to send SIP INFO and RFC2833 at same time. [SEND SIP INFO AND RFC2833 AT SAME TIME]
- Added force reboot button to web UI to force reboot DP750 in factory reset page under maintenance section.
- Improved power savings and stability.

BUG FIX

- Fixed DP720 did not play busy tone in speaker when handset is in speaker mode
- Fixed DP750 UPnP only able to display one DP device at the same time
- Fixed DP750 IP call cannot be redialed
- Removed "Disable Call Waiting Tone" from Profile page, the settings are now under DECT -> Handset Settings now
- Fixed DP750 Music on hold is not sending correct signal for resume
- Fixed DP750: Self-defined time zone field length is too short, increased it from 32 characters to 64
- Fixed DP750 SIP Registration before expiration field value limit is too small, increased it to 64800
- Fixed DP750 SIP Registration failure retry sending duplicated REGISTER packets
- Fixed DP750 cannot unsubscribe the handset after factory reset via TR069
- Fixed DP750 could not make outbound call after handset stay in the off-hook status more than two minutes
- Removed UCM Zero Config template offline download from DP750 web GUI, because DP has official released and the template now can be accessed directly on UCM
- Fixed DP750 cannot factory reset when reset device by TR069
- Fixed DP750 "support tools" online support links are open in self-page and removed offline user manual
- Fixed DP750 Change web port did not tip reboot
- Fixed DP720 contacts cannot display all after browse more than 10 entries
- Fixed DP720 key mapping settings does not working properly
- Fixed DP720 reboot automatically when receiving a call at the same time of successfully saving a new contact
- Fixed DP720 LSK/RSK becomes invalid in Settings menu after fast repeat enter and exit User Settings
- Fixed DP720 ring5 cannot be save and apply to melody

- Fixed DP720 cannot access to submenu or back to idle when unsubscribe it during in the call history interface
- Fixed DP720 did not play busy tone in speaker when handset is in speaker mode
- Fixed DP720 keypad tone is very small
- Fixed HS prompt "Unknown User" when it is not subscribed and pressing keypad from main interface, update the prompt to "Not subscribed"
- Fixed "Check for New Firmware" button does not prompt "the operation successfully" after being clicked
- Fixed handset will lose connection after conference member end the call sometimes
- Fixed when set the SRTP Mode to "enabled and forced", after answering the second calls, the status of handset will become abnormal
- Fixed handset to prompt "Current account was disabled" instead of "User unknown" when calling using disabled account
- Fixed VLAN settings sometimes does not working properly issue
- Fixed incoming account misrepresented in Circular/Parallel/Shared hunt mode
- Fixed "Disable Call Waiting Caller ID" option does not working
- Fixed reject incoming call from single handset will terminates the call instead of advancing to the next handset in circular/linear mode
- Fixed DP750 is providing incorrect Line status to ACS server
- Fixed the confirmation tone is not played by speaker
- Fixed DP750 inactive SIP accounts triggers No Registration LED
- Fixed the System Info page does not refresh after clicking Check Firmware button
- Changed the "Upgrade" button to "Check for New Firmware" and added firmware version information after the button being pressed
- Fixed after changed IP, user could not access web UI
- Fixed that DP750 can reboot when DP720 is upgrading
- Fixed the value cannot be saved if typing Return key on White/Black list for WAN side
- Fixed account info mismatch when in Parallel or Shared mode
- Fixed DP750 doesn't accept dial plan that works fine on GXP
- Fixed 750 crashes after 24-36 hours of operating time which is caused by incoming parallel call been canceled too soon.

NEW FEATURES OVERVIEW

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



CORE COLLECTION EMAIL

- **Web Configuration**

Add email field for core collection tools. Users can leave a contact email to get feedbacks.

 Your device recovered from a crash! Submit logs to Grandstream for troubleshooting? Continue Dismiss


Maintenance

- Firmware Upgrade
- Provisioning
- Web/SSH Access
- TR-069
- Security Settings 
- Date and Time
- Syslog
- Packet Capture
- Factory Reset
- Support 
- Support Documentation
- Configuration Support
- Debug Tools**

Debug Tools

Remote Log Submission - Terms & Conditions

The log files listed below are generated automatically if the usability of the device is affected. By clicking the Submit button, you agree to allow the device to send these files to Grandstream to be used for troubleshooting purposes. If possible, please include a comment with information regarding the use of the device when the problem occurred.

As an alternative to the device submitting the files, you may download the files below and submit them to Grandstream manually.

Agree to Terms? ☐ Yes, allow sending of crash logs to Grandstream.

Email Address

Comments

Submit Files

Debug Log Files

forced_core_cmb5_nvram_db_DP750_FW_R1_0_3_21_MAC_ core.tar Tue Feb 28 15:14:02 2017 Download

Delete All Files

Delete

Terminate Phone Process

Generate Debug File

Handset Notification

Prompt Handsets After Recovery? ☐ Yes ☒ No

Save

Save and Apply

Reset

- **New P Values**

Pvalue	Description	Value range	Default
P27774	<New Pvalue> Contact email address for core dump files	String max 256 bit	null

DP720 FACTORY RESET DURING BOOTING UP

- **Web Configuration**

No web configuration is needed, this feature is a built-in for all DP720 since 1.0.3.22.

- **Functionality**

This feature uses combo keys to factory reset DP720 during booting up prior it's fully booted, to give DP720 the ability to recover from some extreme cases like self-reboot or stuck right after booted up.

The procedure for DP720 factory reset using combo key:

1. Connect the USB cable and power off DP720.
2. Power on DP720 and wait for Green LED.
3. Press 3 times Central Key (OK/Selection Key) quickly after Green LED on.
4. Press Speaker Key (left bottom) first and then Mute Key (right bottom) after Red LED on.
5. DP720 LED will start blinking in 3 colors: Green, Red and Orange if factory reset successful triggered.
6. Wait for about 1-3 minutes, the DP720 will boot up with setup wizard for initial language and date format settings.

HANDSET PROVISIONING FEATURE

- **Web Configuration**

To configure the handset provisioning, please browse the base station Web GUI at Maintenance -> Provisioning. This is the same page used for base station provisioning. Handset shares the settings of base station provisioning in the field in the red rectangle, and has its own settings in the blue rectangle. After enabled Handset Config Upgrade, please reboot the DP750 to take effect.

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings +

Date and Time

Syslog

Packet Capture

Factory Reset

Support +

Provisioning

XML Config File Password
 HTTP/HTTPS User Name
 HTTP/HTTPS Password
 Always send HTTP Basic Authentication Information ☒ No ☐ Yes
 Verify host when using HTTPS ☒ No ☐ Yes
 Provisioning via ☐ TFTP ☒ HTTP ☐ HTTPS
 Config Server Path

Config File Prefix
 Config File Postfix

Enable Handset Config Upgrade ☐ No ☒ Yes
 Handset Config File Prefix
 Handset Config File Postfix

Allow DHCP Option 43 and Option 66 to Override Server ☐ No ☒ Yes
 Allow DHCP Option 120 to Override SIP Server ☒ No ☐ Yes

Provisioning via: Choose the method that the base station uses to request handset ipei config file. Can use TFTP, HTTP or HTTPS.

Config Server Path: Defines the server path for provisioning. Do not prepend address with "http://" or any other protocol.

Handset Config File Prefix: If configured, only the handset configuration file with the matching encrypted prefix will be downloaded and flashed into the device.

Handset Config File Postfix: If configured, only the handset configuration file with the matching encrypted postfix will be downloaded and flashed into the device.

• Functionality

When a DP720 registers to a DP750 for the first time or an already registered handset boots up, the DP750 will send a provisioning request for a file on the configured provisioning server, with location "Config_Server_Path/[Prefix][HS IPEI].xml[Postfix]". For example, if handset's IPEI number is 02c332b510,

prefix is "ipei_", postfix is null, and config server path is "fw.grandstream.com/gs", the request URL is "fw.grandstream.com/gs/ipei_02c332b510.xml.

Example Configuration File

```
<?xml version="1.0" encoding="UTF-8"?>
<hs_config>
  <ipei>02c332b510</ipei>
  <hs_name>HS 1FEDE</hs_name>
  <sip>
    <line>
      <display_name>1FEDE_Line_1</display_name>
      <profile_name>profile1</profile_name>
      <user_id>5328</user_id>
      <auth_id>5328</auth_id>
      <password>password</password>
      <ring_type>DID</ring_type>
    </line>
  </sip>
  <hs_settings>
    <force_apply>1</force_apply>
    <admin>
      <disable_private_phonebook>0</disable_private_phonebook>
      <disable_shared_phonebook>0</disable_shared_phonebook>
      <disable_conference>0</disable_conference>
      <disable_transfer>0</disable_transfer>
      <custom_ringtone>2</custom_ringtone>
    </admin>
    <user>
      <auto_answer>0</auto_answer>
      <cradle_answer>0</cradle_answer>
      <cradle_answer>0</cradle_answer>
      <remote_disconnect_tone_disable>0</remote_disconnect_tone_disable>
      <disable_callwaiting_tone>0</disable_callwaiting_tone>
      <lcd_backlight_cradle>0</lcd_backlight_cradle>
      <key_up_idle_function>4</key_up_idle_function>
      <key_down_idle_function>3</key_down_idle_function>
      <key_left_idle_function>12</key_left_idle_function>
      <key_right_idle_function>11</key_right_idle_function>
      <key_up_call_function>0</key_up_call_function>
      <key_down_call_function>0</key_down_call_function>
      <key_left_call_function>14</key_left_call_function>
      <key_right_call_function>13</key_right_call_function>
```

```

<left_softkey_idle_function>1</left_softkey_idle_function>
<right_softkey_idle_function>7</right_softkey_idle_function>
<left_softkey_call_function>16</left_softkey_call_function>
<right_softkey_call_function>17</right_softkey_call_function>
<mute_as_dnd>0</mute_as_dnd>
</user>
</hs_settings>
</hs_config>
  
```

For more detailed information, please go to DP750/DP720 [product page](#), at How to Guide section, found the [Handset Provisioning Guide](#).

• New P Values

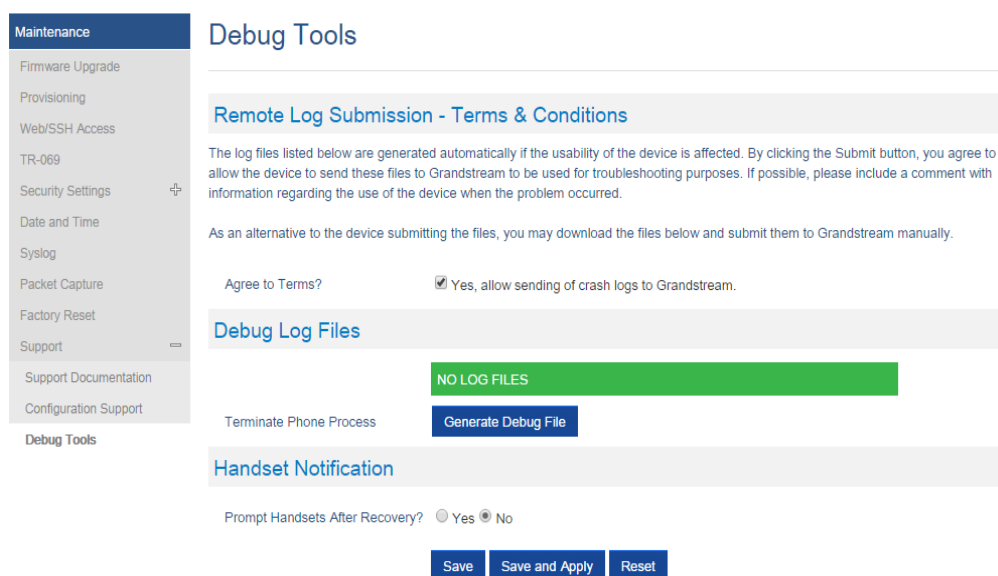
Pvalue	Description	Value range	Default
P27014	<New Pvalue> Enable Handset Config Upgrade	0 – No 1 – Yes	0 – No
P27015	<New Pvalue> Handset Config File Prefix.	String	ipei_
P27016	<New Pvalue> Handset Config File Postfix.	String	NULL

DP750 DEBUG TOOLS

• Web Configuration

User can find the new option at Web -> Maintenance -> Support -> Debug Tools.

Screenshot without debug log file:



The screenshot shows the 'Debug Tools' page in the Grandstream web configuration interface. On the left is a sidebar menu with options: Maintenance, Firmware Upgrade, Provisioning, Web/SSH Access, TR-069, Security Settings, Date and Time, Syslog, Packet Capture, Factory Reset, Support, Support Documentation, Configuration Support, and Debug Tools. The main content area is titled 'Debug Tools' and includes a section for 'Remote Log Submission - Terms & Conditions' with a checkbox for 'Agree to Terms?' (checked) and a 'Generate Debug File' button. Below this is a 'Debug Log Files' section showing 'NO LOG FILES' and a 'Generate Debug File' button. At the bottom is a 'Handset Notification' section with a radio button for 'Prompt Handsets After Recovery?' (set to 'No') and 'Save', 'Save and Apply', and 'Reset' buttons.

Screenshot with debug log file:

Maintenance
 Firmware Upgrade
 Provisioning
 Web/SSH Access
 TR-069
 Security Settings
 Date and Time
 Syslog
 Packet Capture
 Factory Reset
 Support
 Support Documentation
 Configuration Support
Debug Tools

Debug Tools

Remote Log Submission - Terms & Conditions

The log files listed below are generated automatically if the usability of the device is affected. By clicking the Submit button, you agree to allow the device to send these files to Grandstream to be used for troubleshooting purposes. If possible, please include a comment with information regarding the use of the device when the problem occurred.

As an alternative to the device submitting the files, you may download the files below and submit them to Grandstream manually.

Agree to Terms? ☒ Yes, allow sending of crash logs to Grandstream.

Comments

Something abnormal happens to my DP750 and DP720, please take a look of my debug log files, thanks. --Shawn

Submit Files

Debug Log Files

forced_core_cmbs_nvram_db_DP750_FW_R1_0_3_17_MAC_000B82870905_core.tar Tue Jan 24 10:25:29 2017

Download

Delete All Files

Delete

Terminate Phone Process

Generate Debug File

Handset Notification

Prompt Handsets After Recovery? ☐ Yes ☒ No


Save

Save and Apply

Reset

• Functionality

Allow user to generate debug log file and submit it to Grandstream debug server, or submit existing debug log file that self-generated from device crash. Grandstream will not collect any information without customers acknowledged and permissions. It will only send debug log file every time customer clicks the “Submit Button”, after file successfully been sent, the old one will be deleted. Also when there is any debug log files exist, the web UI will prompt it to user every time login:


Your device recovered from a crash! Submit logs to Grandstream for troubleshooting?

Continue


Dismiss

Grandstream DP750

Admin Logout

Reboot

English


GRANDSTREAM
 CONNECTING THE WORLD

STATUS


PROFILES

DECT

SETTINGS

MAINTENANCE

PHONEBOOK



Version 1.0.3.17

Pvalue	Description	Value range	Default
P27751	<New Pvalue> Agree to terms to allow sending crash logs to Grandstream.	0 – No 1 – Yes	1 – Yes

P27753	<New Pvalue> Configure if prompt handset if there are debug log files exist.	0 – No 1 – Yes	0 – No
--------	---	-------------------	--------

REPEATER MANAGEMENT PAGE

- Web Configuration

DECT

General Settings

General Settings

SIP Account Settings

Handset Line Settings

Handset Settings

HS1 Settings

HS2 Settings

HS3 Settings

HS4 Settings

General Settings

Base Station Name

DP750

Admin PIN Code

Enable Repeater Mode

☐ No
☒ Yes

Enable Repeater Management


☐ No
☒ Yes

Clear Call Logs

Delete

- Functionality

After enabled Repeater Management, please reboot the base station first to take effect. Then login web UI and browse to Status page, you will find the new tab names: DECT Repeater Status pages. In which you can find all the paired devices and discovered devices, you can simply link and unlink the repeater by click the button. You can also access the web UI of the repeaters by click the IP links.


Version 1.0.3.17

Status

Account Status

DECT Base Status

DECT Repeater Status

Line Options

Network Status

System Info

DECT Repeater Status

Paired Devices

Model	Name	IP	MAC	Status	Signal Strength	Active Calls	Unlink
DP760	DP760_9e2764	192.168.219.32	000b829e2764	online	RF Link is excellent	0	Unlink

Discovered Devices

Model	Name	IP	MAC	Link
DP760	DP760_9e2768	192.168.219.34	000b829e2768	Link
DP760	DP760_9e2762	192.168.219.31	000b829e2762	Link
DP760	DP760_9e2767	192.168.219.33	000b829e2767	Link
DP760	DP760_9e276e	192.168.219.35	000b829e276e	Link

Grandstream Networks Support
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Pvalue	Description	Value range	Default
P27019	<New Pvalue> Enable Repeater Management feature, need reboot device to take effect.	0 – No 1 – Yes	0 – No

BACKUP CONFIGURATION

- Web Configuration**

User can find the new option at Web -> Maintenance -> Provisioning. (at the bottom of the page)

Backup Configuration

Backup Configuration

Backup Settings

cfg000B82870905-2017-01-05-16-54-13.xml Thu Jan 5 16:54:28 2017 [Download](#)

Delete

- Functionality**

Click “Backup Settings” Button, device will generate an xml config file storing all the current configuration on device itself. After clicked, please wait a few seconds then click the refresh button on the page icon bar or wait for the page self-refresh, you will found the xml list to download then. And support maximum 10 backup files. When it reaches 10 entries this feature will be disabled and the web UI will looks like below:

Backup Configuration

Backup Configuration

DISABLED (MAX OF 10 CONFIG FILES REACHED)

cfg000B82870905-2017-01-05-16-54-13.xml Thu Jan 5 16:54:28 2017 [Download](#)

cfg000B82870905-2017-01-24-11-20-59.xml Tue Jan 24 11:21:14 2017 [Download](#)

cfg000B82870905-2017-01-24-11-22-17.xml Tue Jan 24 11:22:32 2017 [Download](#)

cfg000B82870905-2017-01-24-11-23-41.xml Tue Jan 24 11:23:57 2017 [Download](#)

cfg000B82870905-2017-01-24-11-24-08.xml Tue Jan 24 11:24:24 2017 [Download](#)

cfg000B82870905-2017-01-24-11-24-55.xml Tue Jan 24 11:25:10 2017 [Download](#)

cfg000B82870905-2017-01-24-11-26-38.xml Tue Jan 24 11:26:52 2017 [Download](#)

cfg000B82870905-2017-01-24-11-26-59.xml Tue Jan 24 11:27:15 2017 [Download](#)

cfg000B82870905-2017-01-24-11-27-15.xml Tue Jan 24 11:27:31 2017 [Download](#)

cfg000B82870905-2017-01-24-11-27-39.xml Tue Jan 24 11:27:55 2017 [Download](#)

Caution: Click Delete button will delete all the configuration, please download all the configuration files before you click the “Delete” button, after configuration files been delete, the UI will looks like below:

Backup Configuration

Backup Configuration

Backup Settings

Backup Configuration

NONE

UPNP DISCOVERY

- Web Configuration**

User can find the new option at Web -> Settings -> Networks Settings -> Advanced Settings.

UPnP Discovery Settings

Enable UPnP discovery

☐ No ☒ Yes

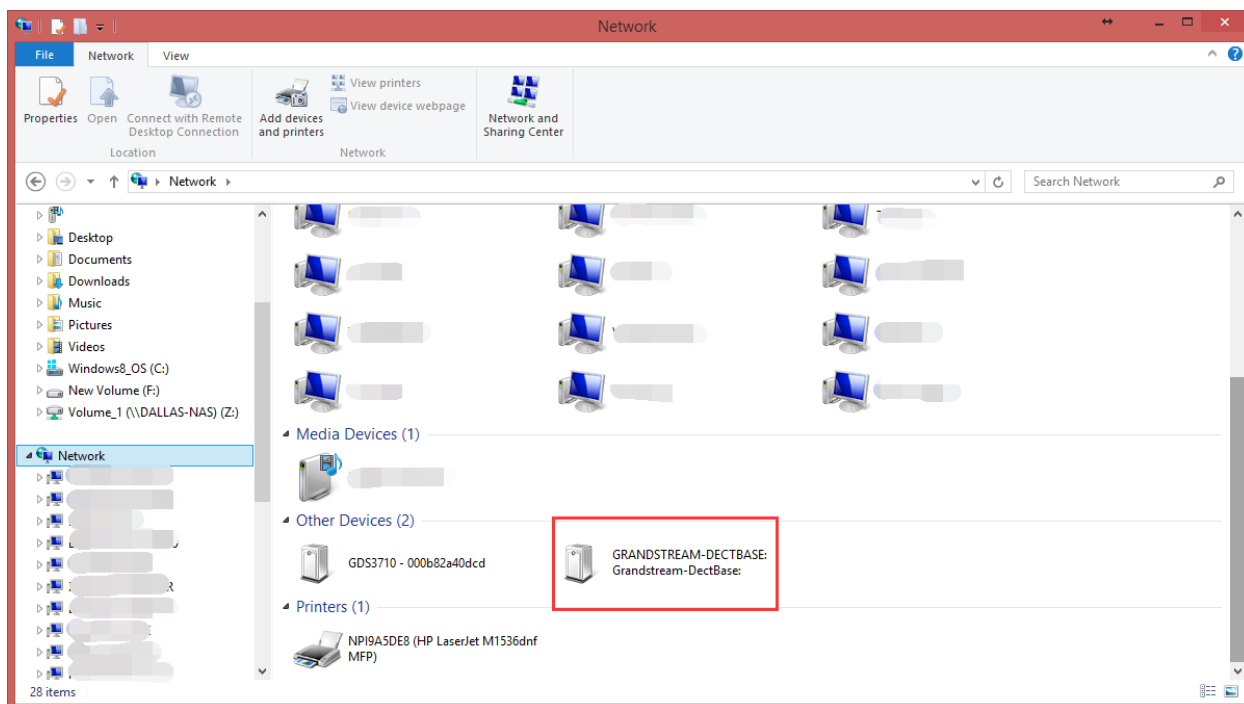
UPnP discovery notify interval

30

- Functionality**

Enables UPnP discovery feature. After enabled, if you have a PC in the same network of DP750, you can browse it directly in your Network. Double click the device will open its web GUI in your default browser.

The UPnP Discovery notify interval is set how often it will refresh to find the device, default interval is 30 seconds.



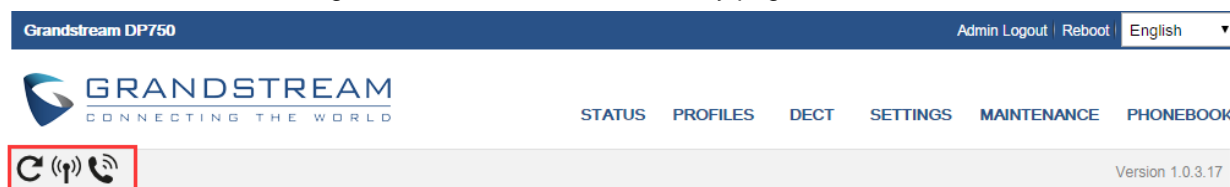
- **New P Values**

Pvalue	Description	Value range	Default
P27756	<New Pvalue> P value to enable/disable UPnP discovery	0 or 1 0: No (Disabled) 1: Yes (Enabled)	1
P27757	<New Pvalue> Pvalue to set the interval of UPnP discovery notify	15 - 86400 (seconds)	30

ICON BAR WITH SHORTCUT ICON BUTTON

- **Web Configuration**

User can find the icon bar right below the main menu of every page.



- **Functionality**

The first icon is to refresh the current page. The second icon is used to enable subscription. The third icon is used to page all the handset.

CUSTOMIZE DP720 RINGTONES

- **Web Configuration**

Once use firmware provisioning method uploaded .bin ringtone files, users can assign custom ringtone to specific handset at Web -> DECT -> Handset Settings.

DECT

HS1 Settings

General Settings
 SIP Account Settings
 Handset Line Settings
 Handset Settings

HS1 Settings

HS2 Settings

HS3 Settings

HS4 Settings

HS5 Settings

Disable Private Phonebook

☒ No ☐ Yes

Enable Auto Answer

☐ No ☒ Yes

Enable Offhook on Cradle Pickup

☒ No ☐ Yes

Enable Onhook on Cradle Reposition

☒ No ☐ Yes

Disable Conference

☒ No ☐ Yes

Disable Transfer

☒ No ☐ Yes

Disable Busy Tone on Remote Disconnect

☒ No ☐ Yes

Disable Call Waiting Tone

☒ No ☐ Yes

Custom Ringtone

Disabled ▼

Save

Save and Apply

Reset

• Functionality

DP750/DP720 now support customized ringtones now. It takes up to 10 ringtone files which have be named as ring1.bin to ring10.bin, and you can assign one ringtone to each handset in the web UI as the screenshot above.

Steps of configuring a custom ringtone to a handset:

1. Get ringtone files that accepted by DP devices. DP750 and DP720 only support .bin ringtone file. Please use the ring generator tools named “Grandstream IP Phone Custom Ringtone Generator” at our official website: <http://www.grandstream.com/support/tools>
Each ringtone files can maximum be 128KB and have to be named in ringX.bin, X can be 1 to 10.
2. Put the ringX.bin file under the firmware provisioning path, reboot DP750, it will fetch the .bin ringtone file as the same way as fetch firmware.
3. After DP750 booted up, browser to Web -> DECT -> HS Settings, at the Cutom Ringtone, you will see the uploaded ringtone listed under the drop down box. Select the one you want to use for each handset.
4. Save and Apply, wait the DP720 fully downloaded the ringtone, you will be able to set DP720 to use that ringtone.

Each handset only able to use one custom ringtone and DP750 maximum support 10 custom ringtone. You might need refresh the page of HSx Settings page to display the uploaded ringtone. Below is screenshot of how it looks like when there is a custom ringtone.

DECT

General Settings

SIP Account Settings

Handset Line Settings

Handset Settings ≡

HS1 Settings

HS2 Settings

HS3 Settings

HS4 Settings

HS5 Settings

HS1 Settings

Disable Private Phonebook

☒ No ☐ Yes

Enable Auto Answer

☐ No ☒ Yes

Enable Offhook on Cradle Pickup

☒ No ☐ Yes

Enable Onhook on Cradle Reposition

☒ No ☐ Yes

Disable Conference

☒ No ☐ Yes

Disable Transfer

☒ No ☐ Yes

Disable Busy Tone on Remote Disconnect

☒ No ☐ Yes

Disable Call Waiting Tone

☒ No ☐ Yes

Custom Ringtone

Disabled ▼

Disabled

/var/user/handset/ringtones/ring2.bin

ENHANCED FACTORY RESET

- Web Configuration**

User can set DP750 reset button behavior at Web -> Maintenance -> Factory Reset:

Maintenance

Firmware Upgrade

Provisioning

Web/SSH Access

TR-069

Security Settings +

Date and Time

Syslog

Packet Capture

Factory Reset

Support +

Force Reboot/Factory Reset

Force Reboot

Reboot

Configure Web UI Button

Reset Type

Full Factory Reset ▼

Perform Selected Reset

Reset

Configure Hardware Button

Reset Type

Full Factory Reset ▼

Save

Save and Apply

Reset

- Functionality**

Allow user to configure the behavior of Web UI reset button and hardware button on device itself.

There are 3 reset type for Web UI button:

1. Full Factory Reset - This will reset all the settings.

2. NVRAM Settings Only - This will reset all the settings except subscription information, so the handset will stay in subscribed after reset, but all the other settings like account info, call settings, audio settings, etc. will be reset.
3. DECT Settings Only – This will reset only the subscription (wipe out all the subscription information), but keep all the other settings like account, SIP settings.

There are 4 reset types for Hardware button:

1. Full Factory Reset - This will reset all the settings.
2. NVRAM Settings Only - This will reset all the settings except subscription information, so the handset will stay in subscribed after reset, but all the other settings like account info, call settings, audio settings, etc. will be reset.
3. DECT Settings Only – This will reset only the subscription (wipe out all the subscription information), but keep all the other settings like account, SIP settings.
4. Disabled – This will disable the hardware button to factory reset the base for security purpose.

- **New P Values**

Pvalue	Description	Value range	Default
P27766	<New Pvalue> Pvalue to set the behavior of Web UI reset button.	0: Full Factory Reset 3: NVRAM Settings Only 4: DECT Settings Only	0
P88	<New Pvalue> Pvalue to set the behavior of hardware reset button.	0: Full Factory Reset 3: NVRAM Settings Only 4: DECT Settings Only 1: Disabled	0

RENAME THE PROFILE

- **Web Configuration**

User can set DP750 each profile's name at Web -> Profiles -> each profile tab.



The screenshot shows the 'Profiles' section on the left with 'Profile 1' selected. The 'General Settings' tab is active. The 'Profile Name' field is highlighted with a red box and contains the text 'profile1'. Other fields include 'Profile Active' (radio buttons for No and Yes), 'SIP Server' (192.168.254.5), and 'Failover SIP Server' (empty).

- **Functionality**

Custom the name of the profile, this profile name also been used in handset config provisioning for validation. The default values are profile1 – profile4.

- **New P Values**

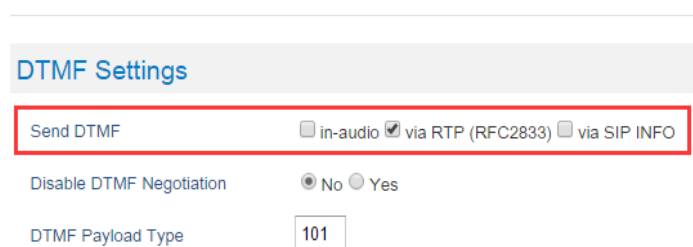
Pvalue	Description	Value range	Default
P27762	<New Pvalue> Pvalue to set the name of profile 1	String	profile1
P27763	<New Pvalue> Pvalue to set the name of profile 2	String	profile2
P27764	<New Pvalue> Pvalue to set the name of profile 3	String	profile3
P27765	<New Pvalue> Pvalue to set the name of profile 4	String	profile4

SEND SIP INFO AND RFC2833 AT SAME TIME

- **Web Configuration**

User can set DP750 each profile's name at Web -> Profiles -> each profile tab -> Audio Settings.

Audio Settings



- **Functionality**

DP750 used to be able to select one of the DTMF type out of in-audio, RFC2833 and SIP INFO, now the option changed from drop down box single-select to multiple check box. So user can choose to use multiple types at the same time.

- **New P Values**

Pvalue	Description	Value range	Default
P2301	<New Pvalue> Pvalue to set if in-audio DTMF type enabled for profile 1	0 – not checked 1 - checked	0
P2302	<New Pvalue> Pvalue to set if RFC2833 DTMF type enabled for profile 1	0 – not checked 1 - checked	1

P2303	<New Pvalue> Pvalue to set if SIP INFO DTMF type enabled for profile 1	0 – not checked 1 - checked	0
P2401	Pvalue to set if in-audio DTMF type enabled for profile 2	0 – not checked 1 - checked	0
P2402	<New Pvalue> Pvalue to set if RFC2833 DTMF type enabled for profile 2	0 – not checked 1 - checked	1
P2403	<New Pvalue> Pvalue to set if SIP INFO DTMF type enabled for profile 2	0 – not checked 1 - checked	0
P2501	Pvalue to set if in-audio DTMF type enabled for profile 3	0 – not checked 1 - checked	0
P2502	<New Pvalue> Pvalue to set if RFC2833 DTMF type enabled for profile 3	0 – not checked 1 - checked	1
P2503	<New Pvalue> Pvalue to set if SIP INFO DTMF type enabled for profile 3	0 – not checked 1 - checked	0
P2601	Pvalue to set if in-audio DTMF type enabled for profile 4	0 – not checked 1 - checked	0
P2602	<New Pvalue> Pvalue to set if RFC2833 DTMF type enabled for profile 4	0 – not checked 1 - checked	1
P2603	<New Pvalue> Pvalue to set if SIP INFO DTMF type enabled for profile 4	0 – not checked 1 - checked	0

FIRMWARE VERSION 1.0.2.16

PRODUCT NAME

DP750, DP720

DATE

08/11/2016

IMPORTANT UPGRADING NOTE

- **For DP750/DP720, once upgraded to 1.0.2.16 or above, downgrading to 1.0.1.x firmware version or lower is not supported.**

ENHANCEMENT

- Added support for entering Plus Button '+' in DP720
- Added DP720 LCD auto turn on feature when pickup from or reposition to charger
- Added Access Control List to control access to web and SSH interfaces [ACCESS CONTROL LIST]
- Added option to enable/disable TR-069 [ENABLE/DISABLE TR-069]
- Added status display as "Normal" when core dump list is empty [CORE DUMP LIST STATUS]
- Added support for HTTPS web access [HTTPS WEB ACCESS]
- Added initial support for Repeater mode [INITIAL SUPPORT FOR REPEATER]
- Added Admin/User Guides and Drilling Template to web GUI support page [DOCUMENTS ADDED TO SUPPORT PAGE]
- Added headset detection and icon for DP720 dial mode screen
- Added Serbian, Slovakian and Swedish languages. Improved support for Czech, Dutch, German, Hebrew, Japanese, Korean, Turkish, Polish languages. [LANGUAGES UPDATED]

BUG FIX

- Fixed DP750 will be "no service" if handset receives incoming call just when make new conference call
- Fixed enable Automatic XML Phonebook Download function was invalid
- Fixed after dial out code star feature, FXS LED keep blinking
- Fixed Priority value for RTP media for layer 2 QoS Settings always be 0 after changed values
- Fixed when handset default line set to none at base side, then change default line from handset side will make base server down.
- Fixed DP750 does not taking IP address from networks with LLDP option enabled
- Fixed incoming call not always rejected when HS is off hook
- Fixed NAPTR/SRV setting does not work
- Fixed DP750 does not taking IP address from networks with LLDP option enabled.
- Fixed "Add Auth Header On Initial REGISTER" option does not work

- Fixed DUT will leave a hold call when transfer failed
- Fixed after receiving incoming call with parallel/share mode some handsets cannot ring in page function
- Fixed call forwarding settings only allow 8 - 10 digits
- Changed DP720 menu "Line Status" in Call Settings to "Lines" for more accurate description
- Fixed handset cannot auto answer paging
- Fixed handset cannot display incoming calls when stay in factory reset confirm interface
- Change menu "Number" in Set Voice Mail to "VM Access Number" for more intuitive description
- Fixed call log sometimes record wrong duration time
- Fixed change ring tone would lead to the volume of call become bigger
- Fixed in contact 'search' interface, "delete" option displays as "back"
- Fixed call waiting tone only play once
- Fixed if click "reboot" option during calling, device does not reboot after called automatically
- Fixed "Number" field is empty when editing XML phonebook contact if phone type is "WORK"
- Fixed deleting all shared contacts might delete other handsets' private contacts
- Fixed Broadsoft IOT related bug that Invite with Alert-info is not playing desired tone
- Fixed Broadsoft IOT related bug that Inband DTMF not working
- Fixed Broadsoft IOT related bug that add ability to bridge conference before third party answers
- Fixed Broadsoft IOT Call transfer issue
- Fixed Broadsoft IOT related issue that in Call Manager "answer call" and "hold" function not working correctly
- Fixed handset can press conference or other options when it is in early media
- Fixed DUT could not call out after removing second line with same account
- Fixed when answer the 2nd call, remote side would hear a serious noise.
- Fixed call manager "answer call" and "hold" function not working correctly
- Fixed ringtone volume cannot restore in next incoming call after adjust ringtone volume to minimum during incoming call
- Fixed if the length of sip settings value is bigger than 12 characters in web, the handset side would display blank
- Fixed handset cannot make new call directly on contacts.
- Fixed no clock display during calling
- Fixed DP720 does not play out audio with professional Sennheiser mono headset, neither single SC230 nor dual SC262
- Fixed if click press "speaker" key quickly after off hook, audio channel will not switch to the speaker
- Fixed handset does not play ringtone when receiving incoming call in parallel or shared mode, and it cannot answer the call
- Fixed when transfer a call, device will send INVITE to the third party before sending REFER
- Fixed device will fail to transfer if the first session is hold, then handset will always prompt user unknown when calling
- Fixed handset will not play CW tone if it is in select contact UI
- Fixed digits not sent to base when 24-digit number entered via on-hook dialing (limit to 20 digits)
- Fixed speaker icon was not disappeared but actually HS was in earpiece mode

- Fixed handset lacks characters "" and "&"
- Fixed handset auto-save the volume after adjusted the alarm volume without saving
- Fixed melody setting in individual contact is not working
- Fixed handset will crash if pressing '1' or '4' key just when booting up
- Fixed handset info displayed new IP address after saving new static IP in handset without reboot
- Fixed "Delayed Call Forward Wait Time" option on web is invalid
- Fixed sometimes base will still ring next handset but server already cancels the call
- Fixed direct dial on contact menu will only dial out via contact's home number
- Fixed DNS Settings is inconsistency between DNS Server 1, DNS Server 2 and Preferred DNS Server
- Fixed after back to main menu, the icon used to indicate there are Missed Calls and Voice Messages disappeared
- Fixed DP720 Speed dial only display first name which should be both first name and last name
- Fixed if handset play standard ring tone before off hook it will in speaker mode
- Fixed the contact cannot be saved with more than 35 characters
- Fixed DUT can re-dial with outgoing number is empty
- Fixed the redial option cannot be used if "Use # as Dial Key" is set to NO
- Fixed after handset selected another base the original base still marked this handset as online
- Fixed the session will be hung up if unplug headset during the call
- Fixed when handset hold the call, FXS led did not blink
- Fixed DUT as transferee cannot back to call if transfer failed
- Fixed if press back key when loading contact in speed dial menu HS will back to main interface
- Fixed if handset generate two missed call history at the same time, the first missed call history name would display blank
- Fixed the call history display order is not from latest to oldest
- Fixed handset does not display LDAP contacts
- Fixed after receiving incoming call with parallel/share mode device always responded 408
- Fixed handset will display error when search LDAP contacts
- Fixed Web GUI cannot be accessed when use PPOE mode
- Fixed Web GUI cannot be accessed when use Qos2
- Fixed handset search contacts function does not work
- Fixed if the third party answers & ends the call quickly the original session will be abnormal
- Fixed handset will auto exit new call UI if receiving incoming call during new call process
- Fixed HS did not auto answer the call from contact that has been configured melody
- Fixed user cannot check sip server name in sip settings by pressing right Softkey
- Fixed handset status did not update base mac address after subscribed another base
- Fixed the name of right Softkey is displayed as "delete" but actually there is no entry
- Fixed missed call LED still flash after viewing all missed call logs
- Fixed Session Expiration value can set to small then Min-SE value
- Fixed HS cannot update CID while in alerting state
- Fixed it will not record missed call in Parallel and Shared mode
- Fixed SIP accounts cannot register to Failover SIP Server except account 1

- Fixed it will not record accepted call in Parallel and Shared mode
- Fixed DUT will not play melody ringtone when handset stay in contact list interface
- Fixed handset popped up menu for a moment every time it received incoming call
- Fixed if edit dialing number that more than 20 characters on call history HS will be block in blank interface
- Fixed remote end also heard the CW tone which was generated by HS
- Fixed device does not respond 486 if HS calls to internal HS with the fourth call
- Fixed if HS is in status UI when BS boots up the HS will not register to BS
- Fixed battery icon need change to white color icon
- Fixed green button is not always act as "Send" when making transfer and conference call

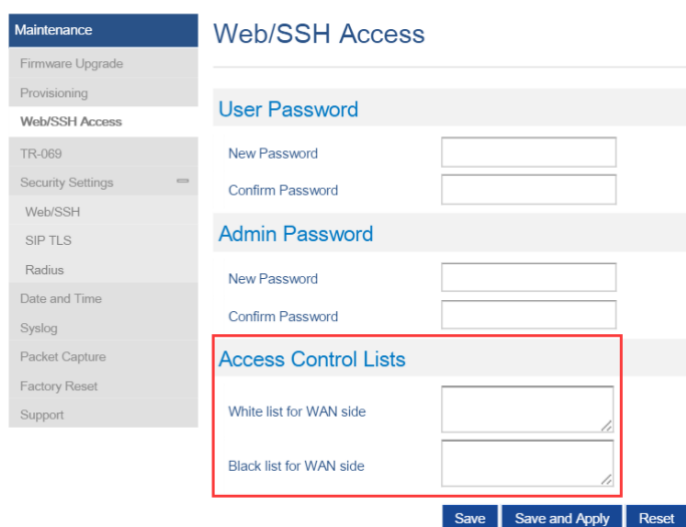
NEW FEATURES OVERVIEW

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

ACCESS CONTROL LIST

• Web Configuration

These settings can be found at Web -> Maintenance -> Web/SSH Access.



• Functionality

If added IP address to White list for WAN side field, then only the IP address in white list can access Web and Telnet.

If added IP address to Black list for WAN side field, then The IP address in black list will cannot access Web or Telnet.

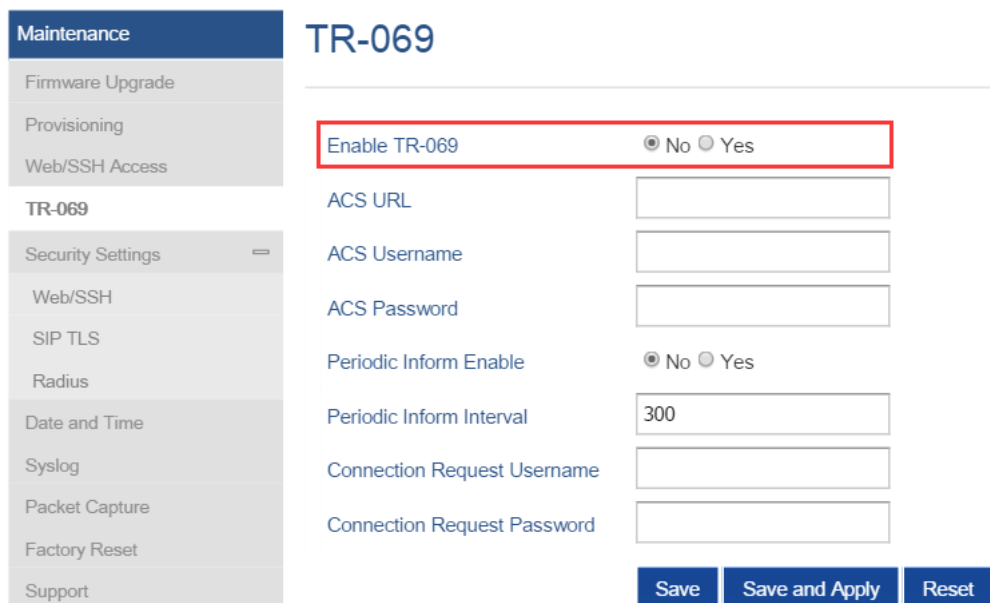
• New P Values

Pvalue	Description	Value range	Default
P20701	<New Pvalue> White list for WAN side.	ipaddress	
P20702	Black list for WAN side.	ipaddress	

ENABLE/DISABLE TR-069

- Web Configuration**

User can find the new option at Web ->Maintenance -> TR-069.



- Functionality**

Allow user to enable and disable TR-069 function.

Pvalue	Description	Value range	Default
P1409	<New Pvalue> Enable or disable TR-069.	0 - No 1 - Yes	0 - No

CORE DUMP LIST STATUS

- Web Configuration**



- Functionality**

If there is no Core Dump file, display a "Normal" Status.

HTTPS WEB ACCESS

- Web Configuration**

User can find the new option at Web -> Maintenance -> Security Settings -> Web/SSH.

Maintenance

- Firmware Upgrade
- Provisioning
- Web/SSH Access
- TR-069
- Security Settings

Web/SSH

- SIP TLS
- Radius
- Date and Time
- Syslog
- Packet Capture
- Factory Reset
- Support

Web/SSH Settings

HTTP Web Port

80

HTTPS Web Port

443

Web Access Mode

☐ HTTPS
 ☒ HTTP

Disable SSH

☒ No
 ☐ Yes

SSH Port

22

Save

Save and Apply

Reset

- Functionality**

New options added to allow user to enable HTTPS Web Access Mode and config HTTPS Web Port. After enabled HTTPS mode, will need reboot the DP750 to take effect the settings. Then can access the device by <https://IPADDRESS:PORTS>, for example: <https://192.168.1.18:443>

- New P Values**

Pvalue	Description	Value range	Default
P27010	<New Pvalue> Port for HTTPS web access. The default setting is 443	1 - 65535	443
P1650	<New Pvalue> Sets the protocol for the web interface.	0 or 1 0: HTTPS 1: HTTP	1

INITIAL SUPPORT FOR REPEATER

- Web Configuration**

User can find the new option at Web -> DECT -> General Settings. A new option "Enable Repeater Mode" is added.

DECT

General Settings

General Settings

SIP Account Settings

Handset Line Settings

 Base Station Name

 Admin PIN Code

 Enable Repeater Mode ☒ No ☐ Yes

Clear Call Logs

Delete

Handset Settings

Handset	Handset Name	Handset Phonebook	Offhook Auto-dial
HS1	<input type="text" value="HS1"/>	<input type="text" value="PB1"/>	<input type="text"/>
HS2	<input type="text" value="HS2"/>	<input type="text" value="PB2"/>	<input type="text"/>
HS3	<input type="text" value="HS3"/>	<input type="text" value="PB3"/>	<input type="text"/>
HS4	<input type="text" value="HS4"/>	<input type="text" value="PB4"/>	<input type="text"/>
HS5	<input type="text" value="HS5"/>	<input type="text" value="PB5"/>	<input type="text"/>

Save

Save and Apply

Reset

• Functionality

Add initial support for repeater function. Add option to enable /disable repeater mode, the change for this option will need reboot DP750 to take effect.

Right now GS's repeater hasn't official released and DP750 only support EU repeater, we will add support for US repeater later.

Step to Pair a repeater:

1. Make sure you base is in EU region, display in Web GUI as below:

Base DECT RF Region

EU

2. Make sure your repeater is EU standard.
3. Set Enable Repeater Mode to "Yes", Click Save and Apply, then reboot base take effect the settings.
4. Click "Subscribe" button in Web -> Status -> DECT Status as below screenshot or long press Page button on DP750 for 7 7 seconds to open subscription:

Open Subscription

Subscribe

5. Repeater will automatically pair to DP750
6. We don't have feature to display repeater connection status right now, but repeater LED should show the connection established.

- **New P Values**

Pvalue	Description	Value range	Default
P20002	<New Pvalue> Pvalue to enable/disable base station repeater mode.	0 or 1 0: No (Disabled) 1: Yes (Enabled)	0

DOCUMENTS ADDED TO SUPPORT PAGE

- **Web Configuration**

User can find the new item at Web -> Maintenance -> Support. Offline Admin/User Guide and Drilling Template can be download from DP750 now.

Maintenance

[Firmware Upgrade](#)

[Provisioning](#)

[Web/SSH Access](#)

[TR-069](#)

[Security Settings](#)

[Web/SSH](#)

[SIP TLS](#)

[Radius](#)

[Date and Time](#)

[Syslog](#)

[Packet Capture](#)

[Factory Reset](#)

Support

Support

Online Support

[DP750 Product Page](#)
[Cordless IP Phones | Grandstream Networks](#)

[Support Tools](#)
[Tools | Grandstream Networks](#)

Offline Support

[Administration Guide](#)
[DP750/DP720 Administration Guide](#)

[User Guide](#)
[DP750/DP720 User Guide](#)

[Drilling Template](#)
[MD Hole Location for Wall Mount](#)

Download Default Device Configuration

[Default Device Config \(Text\)](#)
[Download](#)

[Default Device Config \(XML\)](#)
[Download](#)

Download UCM Zero Config Template

[Versions 1.0.10.43 to latest](#)
[Download](#)

[Versions 1.0.10.38 to 1.0.10.42](#)
[Download](#)

[Versions 1.0.10.1 to 1.0.10.37](#)
[Download](#)

Save

Save and Apply

Reset

LANGUAGES UPDATED

- **Web Configuration**

User can set DP750 language at Web right-top side. User can set DP720 language at handset GUI -> Settings -> Phone Languages.

