Grandstream Networks, Inc.

DP752

Administration Guide

DP752 - Administration Guide

WELCOME

Thank you for purchasing Grandstream DP752 DECT Cordless VoIP Base Station and DP730/DP722 DECT Cordless HD Handsets.

The DP752 is a powerful DECT VoIP base station that pairs with up to 5 of Grandstream's DP series DECT handsets to offer mobility to business and residential users. It supports a range of up to 400 meters with DP730 and up to 350 meters with DP722/DP720 outdoors and 50 meters indoors to give users the freedom to move around their work or home space, delivering efficient flexibility. This DECT VoIP base station supports up to 10 SIP accounts and 5 concurrent calls while also offering 3-way voice conferencing, full HD audio and integrated PoE. A shared SIP account on all handsets will add seamless unified features that give users the ability to answer all calls regardless of location in real-time. The DP752 supports a variety of auto-provisioning methods and TLS/SRTP/HTTPS encryption security. When paired with Grandstream's DP720, DP722 or DP730 handsets, the DP752 offers a powerful cordless DECT solution for any business or residential user.

The DP730 is a DECT cordless IP phone that allows users to mobilize their VoIP network throughout any business, warehouse, retail store and residential environment. It is supported by Grandstream's DP750 and DP752 DECT VoIP base stations and delivers a combination of mobility and top-notch telephony performance. Up to five DP730 handsets are supported on each base station while each DP730 supports a range of up to 400 meters outdoors and 50 meters indoors from the base station. It touts a suite of robust telephony features including support for up to 10 SIP accounts and 2 concurrent calls per handset, full HD audio, a 3.5mm headset jack, push-to-talk, a speakerphone and more. When paired with Grandstream's DECT Base Stations, the DP730 offers a powerful cordless DECT solution for any business or residential user.

The DP722 is a DECT cordless IP phone that allows users to mobilize their VoIP network throughout any business, warehouse, retail store and residential environment. It is supported by Grandstream's DP750 and DP752 DECT VoIP base stations and delivers a combination of mobility and top-notch telephony performance. Up to five DP722 handsets are supported on each base station while each DP722 supports a range of up to 350 meters outdoors and 50 meters indoors from the base station. It touts a suite of robust telephony features including support for up to 10 SIP accounts and 2 concurrent calls per handset, full HD audio, a 3.5mm headset jack, push-to-talk, a speakerphone and more. When paired with Grandstream's DECT Base Stations.

PRODUCT OVERVIEW

Feature Highlights

The following tables contain the major features of the DP752 / DP730/DP722:

 **DP752**	• 5 Handsets. • 10 accounts. • 10 Lines. • 5 Concurrent calls. • PoE power support. • Range: Up to 400m with DP730 and up to 350m with DP722/DP720 outdoor / 50m range indoor.

P752 Features at a Glance

 **DP722**	• DECT cordless HD. • 1.8 inch (128×160) color TFT LCD • 250 hours standby / 20 hours talk time. • 15 languages embedded. • 10 accounts. • 2 concurrent calls. • 5 ring modes.

DP722 Features at a Glance

 **DP730**	• DECT cordless HD. • 2.4 inch (240×320) color TFT LCD • 500 hours standby / 40 hours talk time. • 15 languages embedded. • 10 accounts. • 2 concurrent calls. • 5 ring modes.

DP730 Features at a Glance

DP752 Technical Specifications

The following table resumes all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages, and upgrade/provisioning settings for the Base station DP752.

Air Interface	**Telephony standards:** DECT **Frequency bands:** ○ 1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US) ○ 1910 – 1920 MHz (Brazil), 1786 – 1792 MHz (Korea) ○ 1893 – 1906 MHz (Japan), 1880 – 1895 MHz (Taiwan) **Number of channels:** 10 (Europe), 5 (US, Brazil or Japan), 3 (Korea), 8 (Taiwan) **Range:** Up to 400 meters with DP730 and up to 350 meters with DP722/DP720 outdoor / 50m range indoor.
Peripherals	3 LED indicators: Power, Network, DECT. Pairing/Paging button. One 10/100 Mbps auto-sensing Ethernet port with integrated PoE
Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP

Voice Codecs	G.711μ/a-law, G.723.1, G.729A/B, G.726-32, iLBC, G.722, OPUS, G.722.2/AMR-WB (special order), in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO), VAD, CNG, PLC, AJB
Telephony Features	Hold, transfer, forward, 3-way conference, push to talk, intercom, downloadable phonebook (XML, LDAP, up to 3000 entries), call waiting, call log (up to 300 records), auto answer, flexible dial plan, server redundancy and fail-over
QoS	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control, DECT authentication & encryption
Multi-language	Chinese Simple, Chinese Tradition, Czech, Danish, Dutch, English, Estonian, Finnish, French, German, Hebrew, Hungarian, Japanese, Korean, Norwegian, Portuguese, Romanian, Spanish, Swedish, Turkish.
Upgrade/ Provisioning	Firmware upgrade via FTP/FTPS or /HTTP/HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file
Multiple SIP Accounts	Up to ten (10) distinct SIP accounts per system **Each handset** may map to any SIP account(s) **Each SIP account** may map to any handsets(s)
Ring Group	Flexible options when multiple handsets share the same SIP account ○ **Circular Mode:** All phones ring sequentially, starting with the phone after the one which rang last. ○ **Linear Mode:** All phones ring sequentially in the predetermined order, starting with the first phone each time. ○ **Parallel Mode:** All phones ring concurrently; after one phone answers, the remaining available phones can make new calls
Power & Green Energy Efficiency	Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC, 1A; Micro-USB connection; PoE: IEEE802.3af Class 1, 0.44W–3.84W
Package Content	Base unit, Universal Power Supply, Ethernet cable, Quick Installation Guide, GPL Statement
Dimensions (H x W x D)	140.31 x 64.98 x 105 mm
Weight	Base unit: 140g; Universal power supply: 50g; Package: 370g
Temperature and Humidity	Operation: -10º to 50ºC (14 to 122ºF); Storage: -20º to 60ºC (-4 to 140ºF); Humidity: 10% to 90% non-condensing

Compliance	**FCC:** FCC Part 15B; FCC Part 15D;MPE; FCC ID **CE：** EN 55032; EN 55035; EN 61000-3-2; EN 61000-3-3; EN 60950-1; EN 301 489-1/-6; EN 301 406; EN 50385; RED NB Cert **RCM：** AS/NZS 32; AS/NZS 60950.1. ANATEL, EAC, UL(adapter).

DP752 Technical Specifications

DP730 Technical Specifications

The following table resumes all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for the DP730 handsets.

Air Interface	**Telephony standards**: DECT Frequency bands: ● 1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US) ● 1910 – 1920 MHz (Brazil), 1786 – 1792 MHz (Korea) ● 1893 – 1906 MHz (Japan), 1880 – 1895 MHz (Taiwan) **Number of channels:** 10 (Europe), 5 (US, Brazil or Japan), 3 (Korea), 8 (Taiwan) **Range**: up to 400 meters outdoor and 50 meters indoor
Peripherals	2.4 inch (240×320) color TFT LCD 27 keys including 3 soft keys, 5 navigation/ menu keys, 4 dedicated function keys for SEND, POWER/END, SPEAKERPHONE, MUTE, 3 side keys including 2 volume (up and down) and 1 Push-to-Talk key 3-color MWI LED 3.5mm headset jack Proximity and accelerometer sensors Backlit keypad Removable belt clip Micro-USB port for alternative charging and non-battery operation
Protocols/Standards	Hearing Aid Compatibility (HAC) compliant
Voice Codecs	G.722 codec for HD audio and G.726 codec for narrow band audio (G.711μ/a-law, G.723.1, G.729A/B, iLBC and OPUS are supported via companion DECT base station), AEC, AGC, Ambient noise reduction on handset mic, advanced noise suppression for incoming audio
Telephony Features	Hold, transfer, forward, 3-way conference, push-to-talk, intercom, downloadable phonebook, call waiting, call log, auto answer, click-to-dial, flexible dial plan
HD Audio	Yes, in both Handsets and Speakerphone modes
Security	DECT authentication & encryption
Multi-language	English, Czech, German, Spanish, French, Hebrew, Italian, Dutch, Polish, Portuguese, Russian, Turkish, Arabic, Chinese Simple, Chinese Tradition, Japanese, Korean, Slovakian, Serbian, Greek, Swedish.
Upgrade/ Provisioning	Software Upgrade Over-The-Air (SUOTA), handsets provisioning Over-The-Air
Multiple Line Access	Each handset may access up to twenty (20) lines
Power & Green Energy Efficiency	Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; Rechargeable Li-ion battery (500 hours of standby time and 40 hours of talk time)

Package Content	Handset unit, universal power supply, charger cradle, belt clip, 1 battery, Quick Installation Guide
Dimensions (H x W x D)	Handset: 168.5 x 52.5 x 21.8mm; Charger cradle: 76 x 73 x 81mm
Weight	Handset: 180g; Charger cradle: 78g; Universal power supply: 50g; Package: 465g
Temperature and Humidity	Operation: -10° to 50°C (14 to 122°F); Charging: 0 to 45°C (32 to 113°F); Storage: -20° to 60°C (-4 to 140°F); Humidity: 10% to 90% non-condensing
Compliance	FCC, CE, RCM, IC

DP730 Technical Specifications

DP722 Technical Specifications

The following table resumes all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for the DP722 handsets.

Air Interface	**Telephony standards**: DECT Frequency bands: ● 1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US) ● 1910 – 1920 MHz (Brazil), 1786 – 1792 MHz (Korea) ● 1893 – 1906 MHz (Japan), 1880 – 1895 MHz (Taiwan) **Number of channels:** 10 (Europe), 5 (US, Brazil or Japan), 3 (Korea), 8 (Taiwan) **Range**: up to 350 meters outdoor and 50 meters indoor
Peripherals	1.8 inch (128×160) color TFT LCD 23 keys including 2 softkeys, 5 navigation / menu keys, 4 dedicated function keys for SEND, POWER/END, SPEAKERPHONE, MUTE 3-color MWI LED 3.5mm headset jack Removable belt clip Micro-USB port for alternative charging and non-battery operation
Protocols/Standards	Hearing Aid Compatibility (HAC) compliant
Voice Codecs	G.722 codec for HD audio and G.726 codec for narrow band audio (G.711μ/a-law, G.723.1, G.729A/B, iLBC and OPUS are supported via companion DECT base station), AEC, AGC, Ambient noise reduction on handset mic, advanced noise suppression for incoming audio
Telephony Features	Hold, transfer, forward, 3-way conference, push-to-talk, intercom, downloadable phonebook, call waiting, call log, auto answer, click-to-dial, flexible dial plan
HD Audio	Yes, in both Handsets and Speakerphone modes
Security	DECT authentication & encryption
Multi-language	English, Czech, German, Spanish, French, Hebrew, Italian, Dutch, Polish, Portuguese, Russian, Turkish, Arabic, Chinese Simple, Chinese Tradition, Japanese, Korean, Slovakian, Serbian, Greek, Swedish.
Upgrade/ Provisioning	Software Upgrade Over-The-Air (SUOTA), handsets provisioning Over-The-Air

Multiple Line Access	Each handset may access up to twenty (20) lines
Power & Green Energy Efficiency	Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; Rechargeable 800mAh Ni-MH Low Self-Discharge (LSD) AAA batteries (250 hours of standby time and 20 hours of talk time)
Package Content	Handset unit, universal power supply, charger cradle, belt clip, 2 batteries, Quick Start Guide
Dimensions (H x W x D)	Handset: 158 x 50 x 28.1mm; Charger cradle: 81.15 x 75.89 x 36.36mm
Weight	Handset: 110g; Charger cradle: 44g; Universal power supply: 50g; Package: 328g
Temperature and Humidity	Operation: -10° to 50°C (14 to 122°F); Charging: 0 to 45°C (32 to 113°F); Storage: -20° to 60°C (-4 to 140°F); Humidity: 10% to 90% non-condensing
Compliance	FCC, CE, RCM, IC

DP722 Technical Specifications

GETTING STARTED

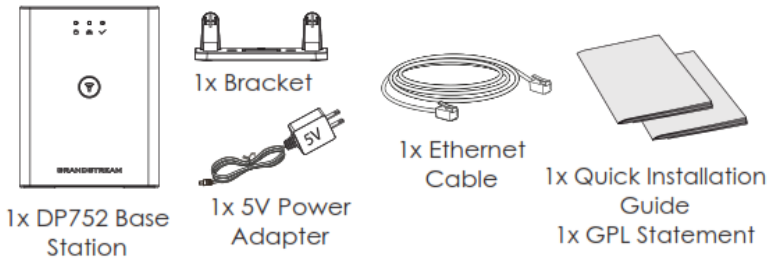
This chapter provides basic installation instructions including the list of the packaging contents and also information for obtaining best performance with the DP730/DP722 DECT Cordless HD Handsets and the DP752 DECT Cordless VoIP Base Station.

quipment Packaging

DP752

- 1 Base unit
- 1 Universal power supply 5V
- 1 Bracket
- 1 Ethernet cable
- 1 Quick Installation Guide
- 1 GPL Statement

Equipment Packaging – DP752

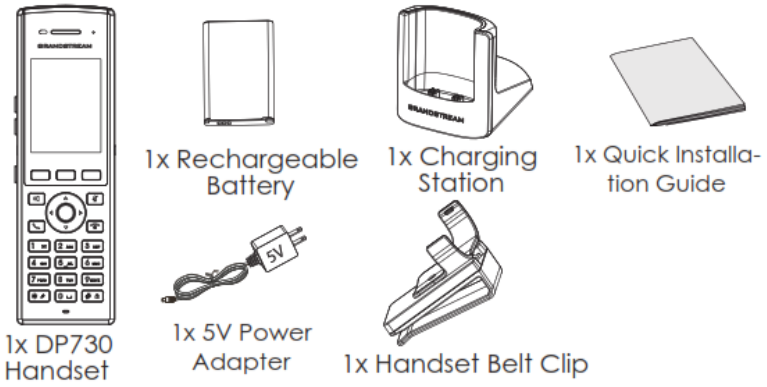


DP752 Package Content

DP730

- 1 Handset unit
- 1 Universal power supply 5V
- 1 Charging station
- 1 Handset Belt
- 1 Rechargeable Li-ion battery
- 1 Quick Installation Guide

Equipment Packaging – DP730

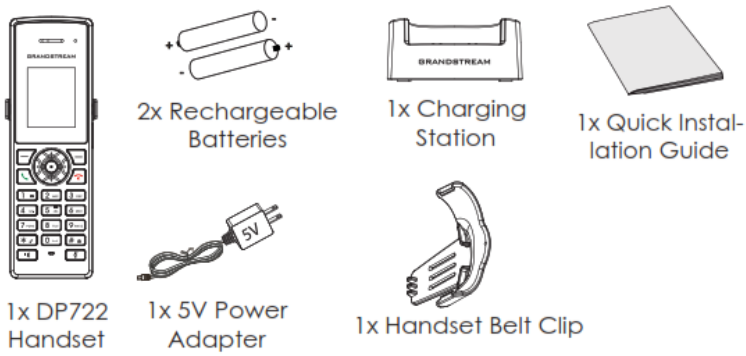


DP730 Package Content

DP722

- 1 Handset unit
- 1 Universal power supply 5V
- 1 Charging station
- 1 Handset Belt
- 2 Rechargeable batteries
- 1 Quick Installation Guide

Equipment Packaging – DP722



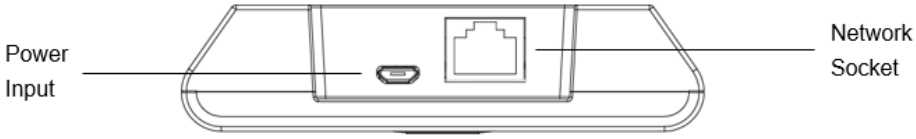
DP722 Package Content

Note

Check the package before installation. If you find anything missing, contact your system administrator.

Connecting DP752

To setup the DP752 Cordless VoIP Base Station, please follow the steps below:



DP752 Back View

You have two options for power and network connection of the base station: AC power or Power over Ethernet (PoE).

> **Note**
>
> For better signal range, we recommend installing DP752 with LED side facing toward the usage area. Ceiling mount is recommended for better coverage.

Connecting via AC power

1. Connect the micro-USB connector into the related port on the base station and connect the other end of the power adapter into an electrical power outlet.
2. Connect the supplied Ethernet cable between the Internet port on the base station and the Internet port in your network or the switch/hub device port.

Connecting via PoE

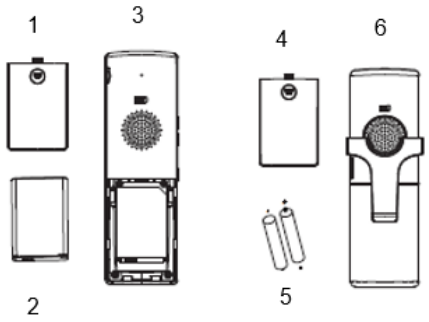
To connect the base station using PoE, you need to connect the Ethernet cable provided (or 3rd party network cable) between the Network Socket on the base station to Ethernet port of your PoE switch/hub.

Setting up DP730/DP722 Handsets

ease follow below steps to insert batteries into the Handsets:

- Open the battery compartment cover.
- **For DP730:** Inset Li-ion battery with the electrodes in the bottom left corner.
- **For DP722:** Insert AAA batteries with correct polarity (+ / -).
- Close the battery compartment cover.

> **Note**
>
> Please charge the batteries fully before using the Handsets for the first time



1. Battery compartment cover of DP730
2. Battery
3. Rear of DP730
4. Battery compartment cover of DP722
5. AAA Batteries
6. Rear of handset

Setting up the DP730/DP722

Battery Information

DP722 Batteries Specifications	DP730 battery specifications

○ **Technology:** Nickel Metal Hydride (Ni-MH) ○ **Size:** AAA ○ **Voltage:** 1.2V ○ **Capacity:** 800mAh ○ **Charging time:** 12 hours from empty to full ○ **Standby time:** up to 250 hours ○ **Talk time:** up to 20 hours' active talk time	○ **Technology:** Li-ion ○ **Nominal Voltage:** 3.8V ○ **Capacity:** 1500mAh ○ **Charging time:** 12 hours from empty to full ○ **Standby time:** up to 500 hours ○ **Talk time:** up to 40 hours' active talk time

DP722/DP730 Battery Specifications

In order to get the best performance of your DP730/DP722 Handsets, we recommend using original batteries provided in the package or batteries compliant with above specifications.

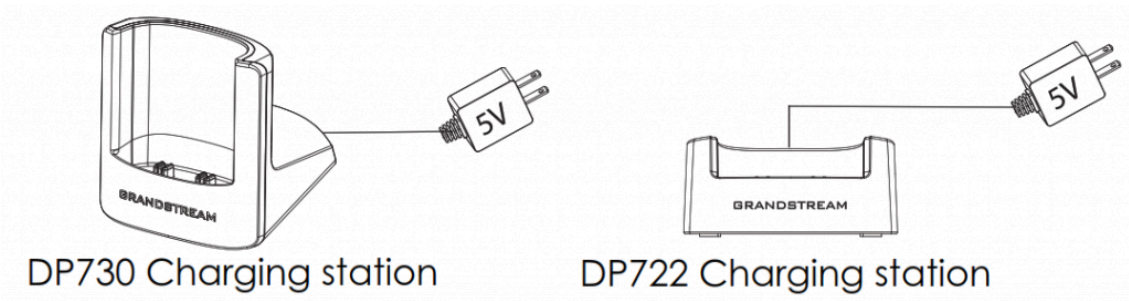
The specifications may differ depending on the age and capacity of the batteries used.

> **Important Note**
>
> Be careful when inserting the batteries into your handset to avoid any risk of short-circuit, which lead to damage your batteries and/or the handset itself. Do not use damaged batteries which can increase the risk of serious harm.

Setting up the Charge Station

Please refer to the following steps for setting up the charge station and charging the Handsets:

1. Connect the DC plug on the power adapter to the micro-USB connector on the charge station.
2. Connect the other end of the power adapter into an electrical power outlet.
3. After setting up the Handsets and the charge station, place the Handsets in the charge station.



Setting up Charging Stations

DP752 LED Patterns

The DP752 has 5 LED lights on it. Please refer to the following table for the meaning of each light.



DP752 LED Patterns

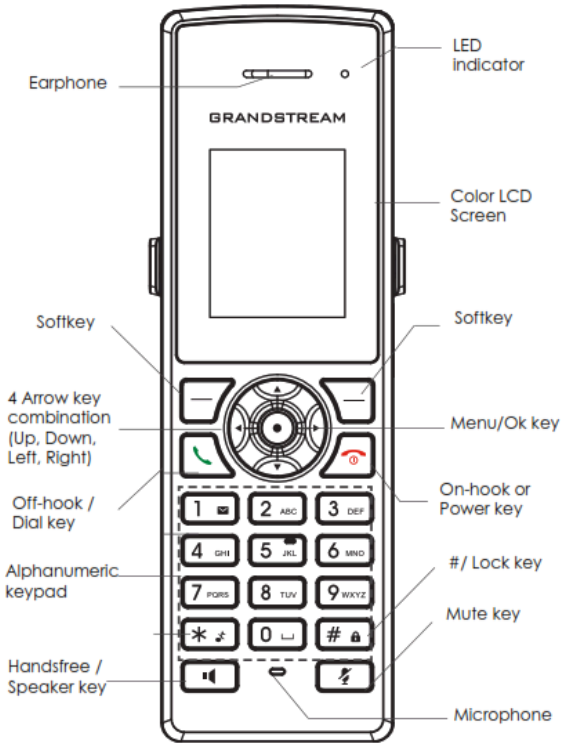
LED Light	Status
LED Light	**Status**

LED Light	Status
	Indicates Power ON/OFF.
	Indicates status of SIP account registration and network ○ **Solid ON**: SIP account registered. ○ **Blinking**: SIP account not registered or network error
	Indicates status of the DECT handset registration: ○ **Solid ON:** Handset registered to base. ○ **Fast Blinking (0.25s ON/0.25s OFF):** Paging handset. ○ **Blinking (0.5s ON/0.5s OFF):** Pairing mode.

DP752 LED Patterns

DP730/DP722 Handsets Description

The LCD screen and the Keypad are the main hardware components of the DP730/DP722.

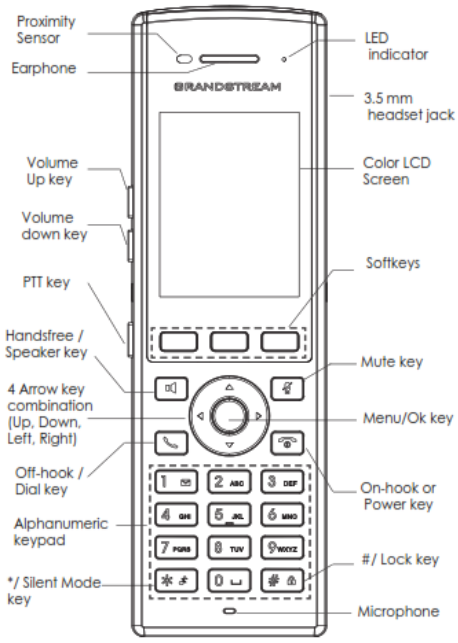


DP722 Keys Description

	Key	Description
1.	**Earphone**	Delivers audio output.
2.	**LED Indication**	**Red:** Charging. **Green:** Charge completed. **Blinking:** Missed call(s) or Voice Mail received.
3, 5	**Left and right softkeys**	Correspond to functions displayed on the LCD. These functions change depending on the current context.
4.	**LCD display**	Shows call information, handset status icons, prompt messages, etc.

	Key	Description
6.	**4 Arrow key combination**	Permits navigation of the cursor through the displayed menu options.
7.	**Men/Ok key**	Selects the option chosen by the cursor. (Enters the main menu from the home screen.)
8.	**Off-hook / Dial key**	Enters dialing mode, or dials number entered.
9.	**On-hook / Power key**	Terminates calls or turns the handset on / off.
10.	**Alphanumeric Keypad**	Provides the digits, letters, and special characters in context-sensitive applications. For + sign, press and hold key 0.
11.	**# / Lock key**	Locks keypad against unintentional entries when keep pressing #. ○ Press and hold # key for approximately 2 seconds to lock the keys. Press **Unlock** softkey and then **#** to unlock the keys.
12.	**Mute key**	Activates or deactivates the mute feature.
13.	**Hands-free / Speaker key**	Switches between handset and hands-free / speaker modes.
14.	**Microphone**	Picks up audio earpiece and hands-free calls.

Keypad Keys Description



DP730 Keys Description

	Key	Description
1.	**Proximity sensor**	The proximity senor can detect and measure gravitational acceleration, tilt, vibration, altitude changes, and static position.
2.	**Earphone**	Delivers audio output.

	Key	Description
3, 4	**Volume up / Down Keys**	Configure the handset and ringtone volume.
5.	**PTT Key**	PTT (Push-to-Talk) button, to initiate PTT call.
6.	**Hands-free / Speaker key**	Switches between Handset and Hands-free / Speaker modes.
7.	**Arrow key combination (Up, Down, Left, Right)**	Allows navigation of the cursor through the displayed menu options.
8.	**Off-hook / Dial key**	Enters dialing mode, or dials number entered.
9.	**Alphanumeric Keypad**	Provides the digits, letters, and special characters in context-sensitive applications. For + sign, press and hold key 0.
10.	*** / Silent Mode key**	Activates or deactivates the silent mode (no ringtone heard during incoming call) when keep pressing on * in idle screen.
11.	**LED indicator**	1dual-color LED indicator indicating: power, call, battery, message waiting…
12.	**3.5 mm headset jack**	Phone connector for the headphones/headsets.
13.	**Color LCD Screen**	2.4-inch (240×320) TFT color LCD
14.	**Softkeys**	Correspond to functions displayed on the LCD. These functions change depending on the current context.
15.	**Mute**	Mute microphone during conversation.
16.	**Menu/OK key**	Selects the option chosen by the cursor or enters the main menu from the home screen.
17.	**On-hook or Power key**	Terminates calls or turns the handset on / off.
18.	**# / Lock key**	Locks keypad against unintentional entries when keep pressing #. ○ Press and hold # key for approximately 2 seconds to lock the keys. ○ Press **Unlock** softkey and then **#** to unlock the keys.
19.	**Microphone**	Picks up audio earpiece and hands-free calls.

Keypad Keys Description

DP730/DP722 Icons Description

The following table contains description of each icon that might be displayed on the LCD screen of the DP730/DP722 Handsets.

	Battery status

	Not equipped with a battery
	**Battery status** Battery empty
	**Battery status** Battery low
	**Battery status** Battery normal
	**Battery status** Battery full
	**Battery status** Charging
	**Signal status** Not subscribed
	**Signal status** Not in range
	**Signal status** Signal very low
	**Signal status** Signal low
	**Signal status** Signal normal
	**Signal status** Signal good
	**Signal status** Signal very good
	**Microphone MUTE Status** OFF – Not muted ON – Muted
	**Speaker status** OFF – The speaker is inactivated ON – The speaker is activated
	Headset icon
	Missed Call icon
	Voicemail icon

	**Ringtone status** OFF – Ringtone off (Silent mode) ON – Ringtone on
	**Keypad Lock status** OFF – Keypad unlock ON – Keypad locked
	**DND Status**. OFF – Do Not Disturb disabled ON – Do Not Disturb enabled
	Call waiting
	Information
	Account not registered
	Account Registered
	Error message
	Handset number
	Incoming Call notification
	Outgoing Call notification
	Missed Call notification
	Voicemail notification
	Contacts
	Call History
	Registration
	Voice Mail
	Preferences

	Shortcut
	Call Features
	Status
	Settings

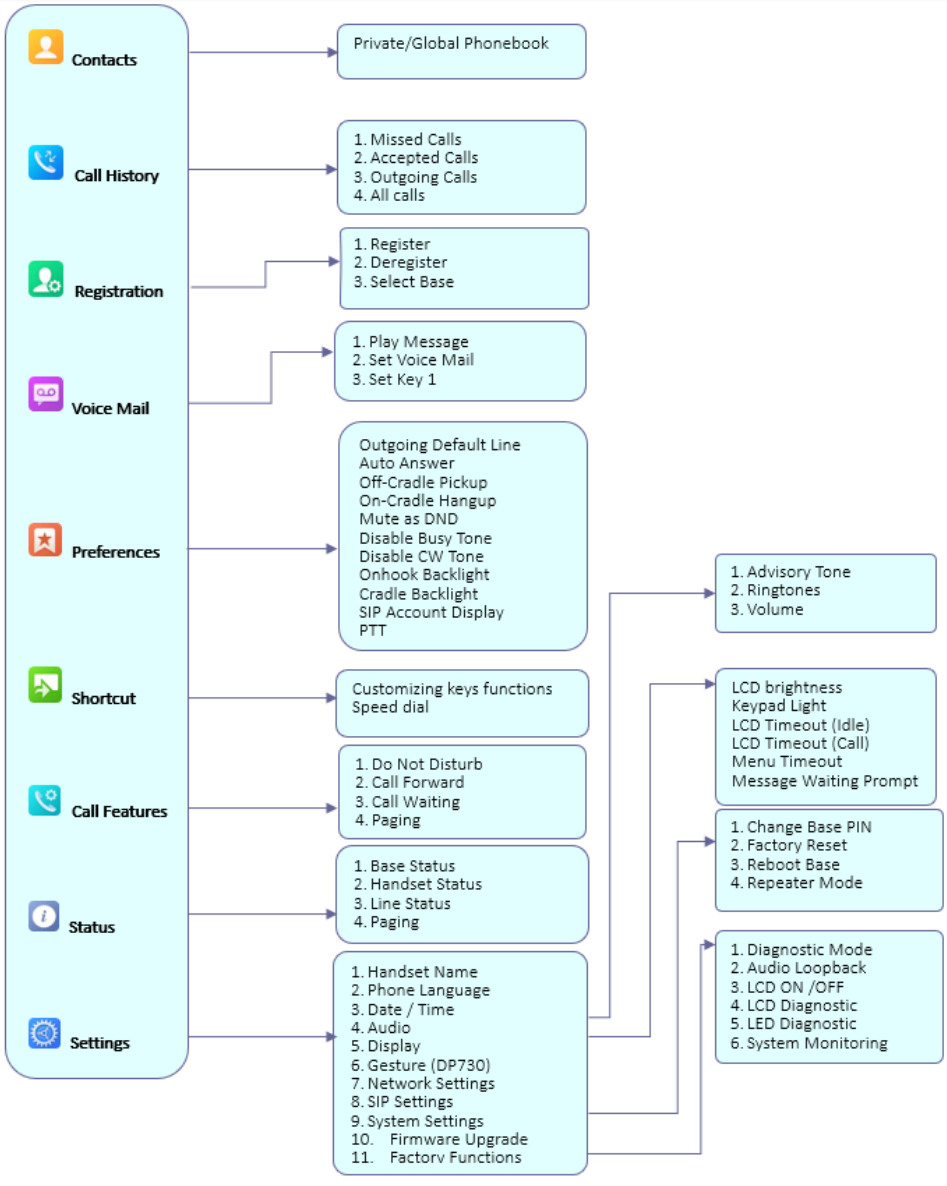
DP730/DP722 Icons Description

DP730/DP722 Handsets Menu

The Handsets has an easy-to-use menu structure. Every menu opens a list of options. To open the main menu, press “Menu” (left softkey) when the Handsets is on and in standby mode. Press Arrow keys to navigate to the menu option you require. Then press “Select” (left softkey) or **OK/Selection key** to access further options or confirm the setting displayed. To go to the previous menu item, press “Back” (right softkey). You can press **Power** key at any time to cancel and return to standby mode. If you do not press any key, the Handsets automatically reverts to standby mode after 20 seconds.

Note

Users can navigate by through handset menu by pressing menu number when displayed.



Contacts	● **Private**: Private contacts include contacts visible in the current Handsets only. ● **Global**: Global phonebook contacts are the contacts shared between the Handsets subscribed to the DP755 base station. **Note**: Private/Global Phonebooks will be merged on the handset.
Call History	Display the call history: 1. **Missed Calls.** 2. **Accepted Calls.** 3. **Outgoing Calls.** 4. **All Calls.** **Note**: You can add contacts to **Shared Contacts** directly from call logs.
Registration	● **Register**: Register your handset to base station. ● **Deregister:** Deregister your handset from base station. ● **Select Base**: Select base station.
Voice Mail	● **Play Message**: Play voice mail messages received. ● **Set Voice Mail**: Configure voice mail parameters. ● **Set Key 1**: Configure Key 1 as VM speed dial for selected account.
Preferences	● **Outgoing Default Line**: Select account to be use by default for outgoing calls. ● Auto Answer: Enable/Disable Auto Answer. (Default is Disabled). ● **Off-Cradle Pickup:** Enable/Disable Off-Cradle Pickup. If enabled, users can answer the calls by picking up the handset off-cradle. (Default is Disabled). ● **On-Cradle Hangup**: Enable/Disable On-Cradle Hangup. If enabled, users can end the call by placing the handset on-cradle. (Default is Disabled). ● **Mute as DND**: Enable/Disable Mute as DND. If enabled, pressing mute key on idle state will set the phone to DND mode. (Default is Enabled) ● **Disable Busy Tone**: Enable/Disable Busy Tone. If set to enabled, busy tone will not be played. (Default is Disabled). ● **Disable CW Tone**: Enable/Disable CW Tone. If set to enabled, Call Waiting Tone will not be played. (Default is Disabled). ● **Onhook Backlight**: Enable/Disable Onhook Backlight. If enabled, pressing "**Hangup**" key on idle screen will switch off LCD screen. (Default is Disabled) ● **Cradle Backlight**: Enable/Disable/Dim Cradle Backlight. If enabled, LCD will remain backlit when the handset is placed on-cradle/charging. If set to “Dim”, LCD brightness will be reduced when the handset is placed on-cradle/charging. (Default is Disabled) ● **SIP Account Display**: Select which SIP Account information will be displayed on the screen. ● **Name Only**: Display SIP Account Name only. (Default) ● **ID Only**: Display SIP User ID only. ● **None**: No account information will be displayed. ● **PTT (Push To Talk)**: Enable/Disable Push To Talk. If set to enabled, pressing and hold PTT hard/soft key, a PTT call will be initiated. Pressing the PTT hard/soft key, it will redirect you to the setting to enable or disable it. (Default is Disabled)
Customizing keys functions	**Customizing keys functions:** ● **L** : Configure Left Softkey function in idle. Function can be set as Menu, History, Contacts, Line or PTT. Default is Menu. ● **R** : Configure Right Softkey function. Function can be set as History, Contacts, Line or PTT Default is Contacts. ● ▲ : Configure Arrow UP Key function. Default is Outgoing Calls (Call History). ● ▼ : Configure Arrow DOWN key function. Default is Accepted Calls (Call History). ● ◄ : Configure Arrow LEFT key function. Default is Ringer Volume Down. ● ► : Configure Arrow RIGHT key function. Default is Ringer Volume Up. Select key and press OK button to configure function. Following functions are available for arrow keys: 1. Disabled, 2. Missed Calls, 3: Accepted Calls, 4: Outgoing Calls, 5: History, 6: Contacts, 7: Status, 8: Line, 9: Voice Mail, 10: Ringer Volume Up, 11: Ringer Volume Down, 12: Audio Volume Up, 13: Audio Volume Down, 14: Intercom. ● **Speed Dial:** Assign contact numbers as speed dial.

	Select a key [2], [3], [4], [5], [6], [7], [8] or [9] and press OK button. Select “Edit” to manually specify the destination number or select “From Contacts” to select a contact as speed dial destination.
Call Features	• **Do Not Disturb:** Enable/disable do not disturb mode on the phone. • **Call Forward**: Configure call forward feature. • **Call Waiting**: Configure call waiting feature. • **Paging**: Configures Inter-Handsets paging feature.
Status	• **Base Status**: Display Base status (Firmware, IP address, Subnet mask, Gateway, MAC Address) • **Handset Statut**: Display Handsets status (Model RF, Firmware, IPEI) • **Line Status:** Display Line status (Account name, Status)
Settings	• **Handset Name:** Change the Handset name. • **Phone Language:** Select the language to be displayed on the phone’s LCD. (Default is English.) • **Date/Time:** Configure date and time on the Handsets. • **Audio:** Specify ringtones for internal/external calls, the volume, advisory tones (Keypad, Confirmation, Low battery notifications) and Vibration mode (DP730 only). • **Display**: Configure backlight, LCD timeout (Idle/Call), LCD brightness, Message Waiting Prompt and menu key timeout. • **Gestures (DP730 only)**: Configure Close-to-Ear Backlight and Facedown Hangup. • **Network Settings:** Configure IP addresses and select DHCP/Static IP mode. • **SIP Settings:** Configure/View SIP accounts settings. • **System Settings**: Change Base PIN code, perform factory reset, reboot base and configure repeater mode. • **Firmware Upgrade:** Upgrade the firmware version of the Handsets. • **Factory Functions:** Diagnostic Mode (DP722) / Keypad Diagnostic (DP730) All LEDs will light up, and the LCD will display a table listing the names of all keys in red. Press any key to diagnose; the key’s name will display in blue. After all keys are diagnosed, a prompt message (“PASS”) will display; press “Back” (right softkey) to exit. **Note**: User can long press arrow UP key to exit at any time. • **Audio Loopback:** Speak to the phone using speaker/Handsets/headset. If you can hear your voice, your audio is working fine. Press “Exit” softkey to exit audio loopback mode. • **LCD ON / OFF:** Select this option to turn off LCD. Press any button to turn on LCD. • **LCD Diagnostic:** Select this option to enter LCD Diagnostic mode. Press “Next” (left softkey) to display white screen. Continue pressing the left softkey to view all remaining screens (black, blue, red, and green) and then exit. End the test early by pressing the right softkey. • **LED Diagnostic:** Enters this option and press “1” to start LED Diagnostic (you will notice that the color of the LED will be changing). Press “2” to quit. • **System Monitoring:** Displays RSSI, battery voltage and RPN information. • **Vibration (DP730 only)**: Test vibration on DP730. • **Acceleration Sensor (DP730):** Displays X, Y, Z coordinates. • **Proximity Sensor (DP730):** Test proximity sensor on DP730.

CONFIGURATION GUIDE

The DP752 can be configured using:

1. Web GUI embedded on the DP752 using PC’s web browser.
2. LCD Configuration Menu using the paired DP730/DP722 keypad.

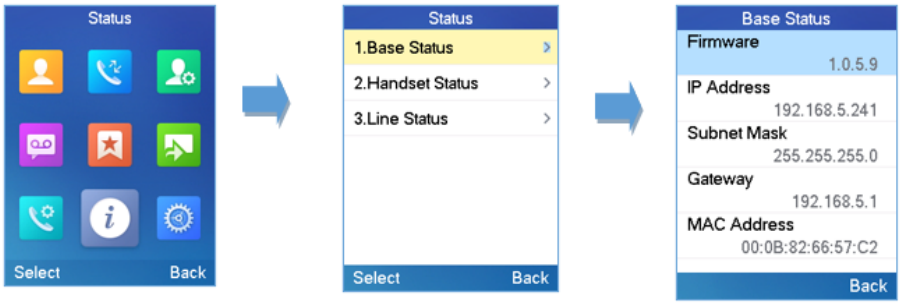
Via Web GUI you can configure all the functions supported by the DP752; while via paired DP730/DP722, you can access limited configuration and need the base station PIN code for some options.

Obtain DP752 Base Station IP Address via Paired DP730/DP722

DP752 is by default configured to obtain IP address from DHCP server where the unit is located. In order to know which IP address is assigned to your DP752, please follow below steps using a paired DP730/DP722 Handsets with your DP752 base station. Please see Register DP730/DP722 Handsets to DP752 Base Station .

1. Press “Menu” (left softkey) or **OK** button on DP730/DP722 to view operation menu.

2. Press Arrow (Up, Down, Left, Right) keys to move the cursor to **Status** icon , then press "Select" (left softkey) or **OK** button, then select **Base Status**.
3. Using Arrow keys, navigate down to view the IP address of the DP752.



Base Status

Configuration via Web Browser

The DP752 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the DP752 through a Web browser such as Google Chrome, Mozilla Firefox and Microsoft's IE.

Accessing the Web UI

1. Connect the computer to the same network as DP752.
2. Make sure the DP752 is booted up.
3. You may check DP752 IP address via a subscribed DP730/DP722 on its LCD menu at **Status → Base Status → IP Address.** Please see Obtain DP752 Base station IP Address via paired DP730/DP722
4. Open Web browser on your computer.
5. Enter the DP752's IP address in the address bar of the browser.
6. Enter the administrator's username and password to access the Web Configuration Menu.

> **Note**
>
> The computer must be connected to the same sub-network as the DP752. This can be easily done by connecting the computer to the same hub or switch as the DP752.

Web GUI Languages

Currently the DP752 series web GUI supports English, Czech, German, Spanish, French, Arabic, Hebrew, Italian, Russian, Netherlands, Japanese, Polish, Chinese Simple, Chinese Tradition, Korean, Portuguese, Slovakian, Serbian, Swedish and Turkish.

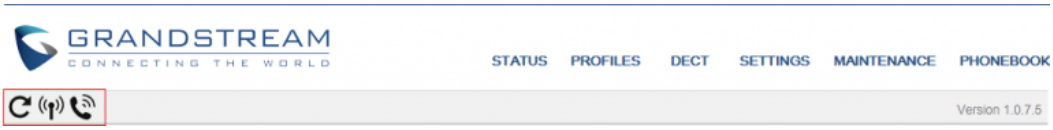
Users can select the displayed language in web GUI login page, or at the upper right of the web GUI after logging in



DP752 Web GUI Language

Icons Bar Shortcut

Users can find the icon bar right below the main menu of every page as displayed on following screenshot:



Icons Bar Shortcut

Please refer to following table describing the use of each icon:

Icon	Description
	Refresh Button: Allows users to refresh the current page.
	Subscribe Button: Allows users to open the subscription.
	Paging Button: Allows users to page all the registered DP730/DP722 Handsets.

Saving the Configuration Changes

After users makes changes to the configuration, pressing the **Save** button will save but not apply the changes until the **Apply** button on the top of web GUI page is pressed. Users can instead directly press the **Save and Apply** button. We recommend rebooting or powering cycle the phone after applying all the changes.

Web UI Access Level Management

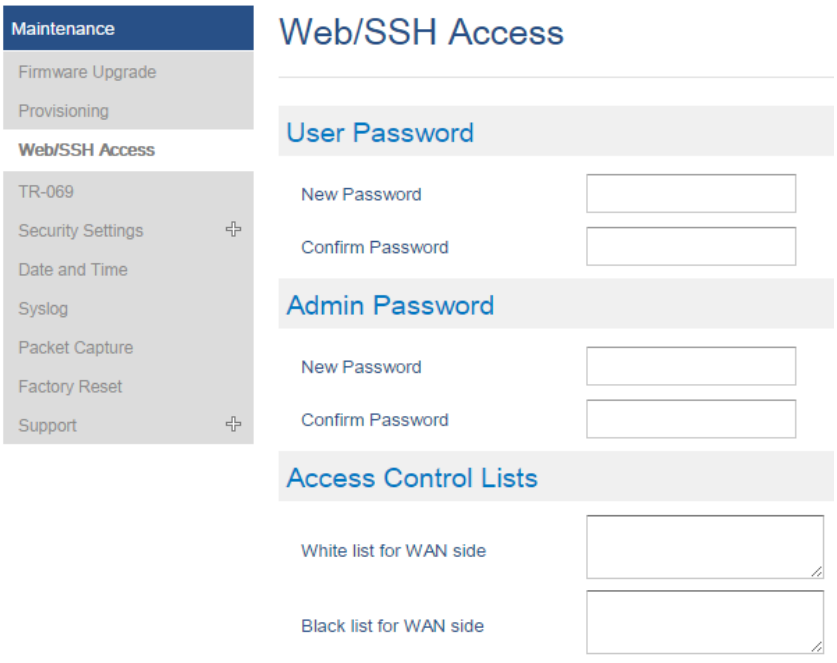
There are two default passwords for the login page:

User Level	Username	Password	Web Pages Allowed
End User Level	user	Password defined by the administrator	Only Status, Settings and Maintenance
Administrator Level	admin	Random password available on the sticker at the back of the unit.	All pages

- The password is case sensitive with maximum length of 25 characters**.**When accessing the web GUI with end-user level, "Advanced Settings" page will be hidden.
- When changing any settings, always SUBMIT them by pressing the **Save** or **Save and Apply** button on the bottom of the page. If using the **Save** button, after making all the changes, click on the **Apply** button on top of the page to submit. After submitting the changes in all the Web GUI pages, reboot DP752 to have the changes take effect if necessary; most of the options under the **Settings** page require reboot, but options under the **Accounts** and **Phonebook** pages do not.
- By Default, the user level password is disabled, and can be enabled only by the administrator.
- Once the initial Admin/user password is provided, you will be prompted to change it immediately.

Changing User Level Password

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **Maintenance → Web/SSH Access.**
4. In **Web/SSH Access** page, locate **User Password** section:
   - Type in your new user password in **New Password** field.
   - Type in again same entered password in **Confirm Password** field.
5. Press **Save and Apply** to save your new setting.



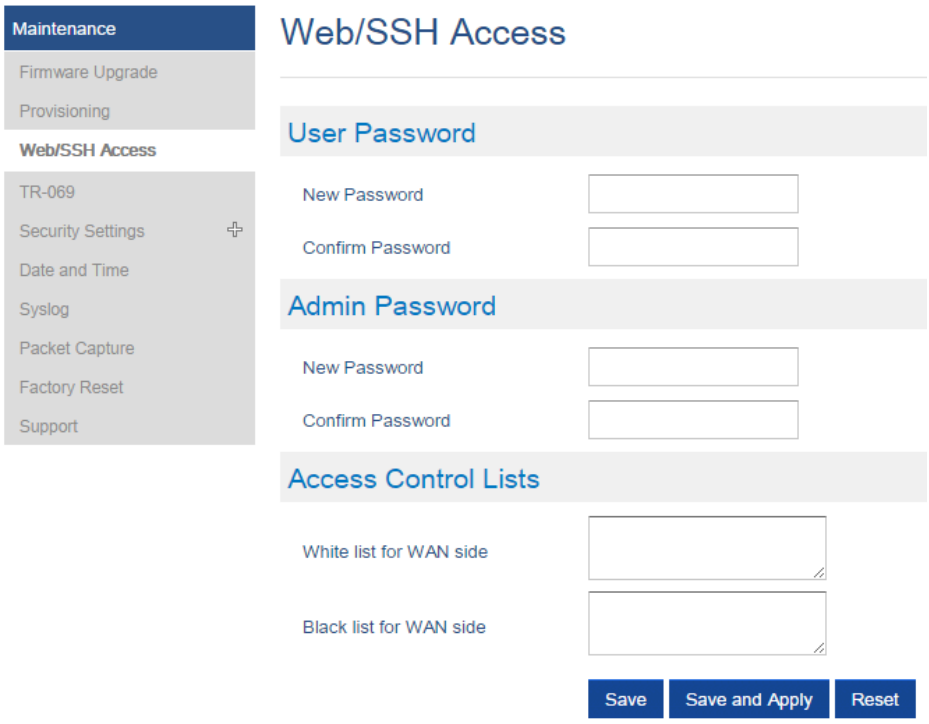
User Level Password

Note

DO NOT USE same password for both user and admin accounts.

Changing Admin Level Password

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **Maintenance → Web/SSH Access.**
4. In **Web/SSH Access** page, locate **Admin Password** section:
   - Type in your new Admin Password in **New Password** field.
   - Type in again same entered password in **Confirm Password** field.
5. Press **Save and Apply** to save your new setting.



Admin Level Password

Note

DO NOT USE same password for both user and admin accounts.

Changing HTTP / HTTPS Web Access Port

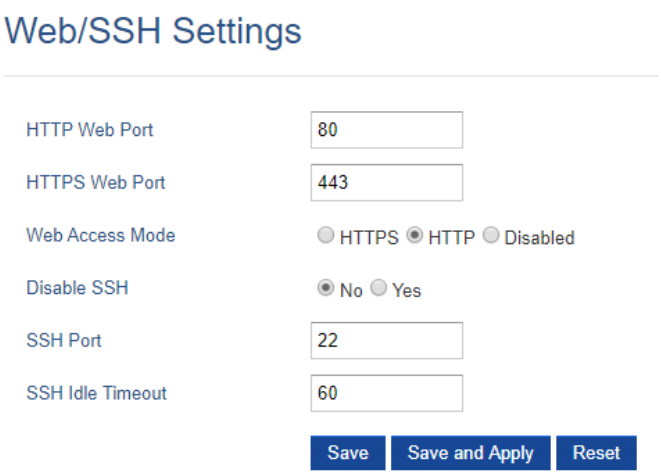
1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **Maintenance → Security Settings → Web/SSH.**
4. In **Web/SSH Settings** page, locate **HTTP / HTTPS Web Port** field and change it to your desired/new HTTP / HTTPS port.
   Note: By default, the HTTP port is 80 and HTTPS is 443.
5. Select the **Web Access Mode** depending on desired protocol (HTTP or HTTPS).
6. Press **Save and Apply** to save your new setting.

Note

A reboot is required for this change to take effect.

Note

Selecting “Disabled” for Web Access Mode will disable web UI access.



Web Access Port

Web Configuration Definitions

This section describes the options in the DP752 Web UI. As mentioned, you can log in as an administrator or an end user.

- **Status:** Display system info, network status, base and repeater status, account status, and line options.
- **Profiles:** Configure the profiles with general settings, network settings, SIP settings, audio settings, call settings and ring tones.
- **DECT:** Configure DECT general settings, account settings and Handsets line settings.
- **Settings:** Configure ring tones and system features.
- **Maintenance:** Configure networks, upgrading and provisioning, web/SSH access, TR-069, security settings, date and time, and syslog.
- **Phonebook:** Manage phonebooks: global (XML or LDAP) and private (XML).

Status Page Definitions

Account Status	
Account	Displays list of configured accounts’ names, from Account 1 to Account 10.
SIP User ID	Displays list of SIP user id registered.
SIP Server	Displays list of SIP Server.
SIP Registration	Shows the status of SIP registration. If the SIP account is successfully registered, it will display “YES” with green background. If the SIP account is not registered, it will display “NO” with red background.
HS Mode	Displays the HS mode configured for each account.
HS status table	Illustrates both handsets and SIP accounts statuses. Each column is dedicated to one HS; each row shows the status of the account on that HS: ● Gray: HS is not configured to use this account. ● Green: HS is idle on this account. ● Green Blinking: HS is using this account. ● Red: HS is not available.

	● Red/Orange Blinking: HS is ringing on this account. ● Brown: The line is configured, but the handset is not subscribed. For example, if accounts 1, 3 and 4 are assigned to HS3 with account 3 in use, the column for HS3 will have cell 3 with red icon, cells 1 and 4 with green icon, and cells 2 and 5 with gray icon.
DECT Base Status	
Base Station Name	Displays name of base station. Default is DP750_**[last 6 digit of MAC address]**.
Base DECT FW Version	Shows firmware version of base DECT.
Base DECT RF Region	Indicates region of base DECT RF.
Base DECT RFPI Address	Specifies DECT RFPI (Radio Fixed Part Identity) address which is a unique identity for the base.
Global Functions	– **Unsubscribe Handsets:** Unsubscribes all handsets from DECT base station. – **Upgrade Handsets:** Upgrade all handsets from DECT base station.
Handset Status	• **Handset Name:** Displays index and name (customizable) of each handset. • **IPEI:** Indicates IPEI number of each handset; this is the unique identity for the handset. If the handset is in range, the IPEI will be displayed with a green background, otherwise, it will be displayed with a red background. • **Battery icon** : Illustrates battery status for each Handset; it can be either: **Fully charged** **Not fully charged** **Low, needs to be charged or replaced** **Charging** • **Page:** Sends paging request to corresponding handset, which will receive incoming ring tone and “Paging” will be displayed on their LCD screens; this function helps you locate the handsets. • **Unsubscribe:** Unsubscribes corresponding handset from DECT base station. • **HS Firmware:** Indicates handset’s firmware version number. ● **Upgrade:** Shows handset upgrade status or trigger handset upgrade process. ● **RAM Dump:** Notifies when a debug file is available after recovery. ● **Note:** The DECT page status keeps refreshing automatically and periodically to update the information about the registered handsets.
DECT Repeater Status	
Paired Devices	Displays paired repeaters showing: ● **Model:** Displays model of repeater. ● **Name:** Displays the name of the repeater. ● **IP address:** Displays the DECT repeater IP address. ● **MAC address:** Displays the DECT repeater MAC address. ● **Status:** Displays the status of DECT Repeater: ● **Online** when the repeater is turned on and ready for use. ● **Offline** when the repeater is turned off. ● **Signal Strength:** Display link status and signal strength, different status is available (excellent, good, week and poor) depending on the distance between the repeater and base station. ● **Active Calls:** Displays number of active calls. ● **Unlink:** Click on **Unlink** button to disconnect the DECT Repeater from the base station.
Discovered Devices	Displays discovered repeaters using: ● **Model:** Displays model of repeater.

	• **Name:** Displays the name of the repeater. • **IP address:** Displays the DECT repeater IP address. • **MAC address:** Displays the DECT repeater MAC address. • **Link:** Clicks Link button to connect DECT Repeater to the base station.
Line Options	
Account	Displays list of configured accounts' names, from Account 1 to Account 10.
SIP User ID	Displays list of SIP user id registered.
DND	Shows DND (Do Not Disturb) feature status (per SIP account).
Forward	Indicates destination to which all incoming calls will be forwarded (per SIP account).
Busy Forward	Indicates destination to which incoming calls will be forwarded when the line is busy (per SIP account).
Delayed Forward	Indicates destination to which incoming calls will be forwarded if the call is not answered within a specified period of time or number of rings (per SIP account).
Network Status	
MAC Address	Shows Device ID in hexadecimal format. This is needed by network administrators for troubleshooting. The MAC address will be used for provisioning and can be found on the label on original box and on the label located on the bottom panel of the device.
IP Address Mode	Indicates used IP address mode: DHCP, Static IP or PPPoE.
IP Address	Displays assigned IP address. Example: 192.168.5.110
Subnet Mask	Displays assigned subnet mask. Example: 255.255.255.0
Gateway	Displays assigned default gateway. Example: 192.168.5.1
PPPoE Link Up	Indicates PPPoE connection status.
DNS Server 1	Shows assigned DNS server address 1. Example: 8.8.8.8
DNS Server 2	Shows assigned DNS server address 2. Example: 8.8.4.4
NAT Traversal	Indicates type of NAT for each Profile. (Based on STUN protocol.)
System Info	
Product Model	Displays product model info. Default is **DP750**.
Part Number	Shows product part number. Example: 9610003814A (last 2 digits show HW version, in this example 14A for HW version 1.4A)
Hardware Version	Shows the hardware version of the phone.
Software Version	• **Firmware Status:** Displays the status of the firmware loaded. • **Boot:** Specifies Boot version.

	• **Core:** Specifies Core version. • **Base:** Specifies Base version. • **Prog:** Specifies Prog version. This is the main firmware release number, which is always used for identifying the software system of the DP750. • **CPE:** Specifies CPE version. • **Locale:** Specifies Locale version. • **Recovery:** Specifies Recovery version. • **Handset:** Specifies Handset firmware version.
System Up Time	Indicates system uptime since last reboot.
System Time	Shows actual time and date according to your configuration.
Service Status	Reveals status of VoIP applications.

Profiles Page Definitions

General Settings	
Profile Active	Activates or deactivates SIP profile.
Profile Name	Determines the name of the profile, this profile name can also be used in Handsets config provisioning for validation. By default, the values are profile 1 – profile 4.
SIP Server	Configures SIP server IP address or domain name provided by VoIP service provider. This is the primary SIP server used to send/receive SIP messages from/to DP752/DP750. **Example:** sip.my_company.com
Failover SIP Server	Specifies failover SIP server IP address or domain name provided by VoIP service provider. This server will be used if the primary SIP server becomes unavailable.
Prefer Primary SIP Server	Prefers primary SIP server. The profile will register to primary Server if registration with Failover server expires. Default is No.
Outbound Proxy	Specifies IP address or domain name of outbound proxy, media gateway or session border controller. Used by DP752 for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work, and only outbound proxy can correct the problem.
Backup Outbound Proxy	IP address or Domain name of the Secondary Outbound Proxy which will be used when the primary proxy cannot be connected.
Voice Mail Access Number	Defines the voice mail portal access number to allow users accessing their voice messages. Max allow length 30
Network Settings	
Layer 3 QoS Settings	
SIP Diff-Serv	Defines SIP Diff-Serv value. Default is 24.
RTP Diff-Serv	Defines RTP Diff-Serv value. Default is 46.
DNS Settings	
DNS Mode	Selects DNS mode to use for the client to look up server. Default is A Record. **A Record:** resolves IP Address of target according to domain name.

	SRV: DNS SRV resource records indicate how to find services for various protocols. **NAPTR/SRV:** Naming Authority Pointer according to RFC 2915. **Use Configured IP:** If selected, please fill in Primary IP, Backup IP 1 and Backup IP 2 to be used for server look up.
DNS SRV Failover mode	This feature is used to configure the preferred IP mode for DNS SRV, the options available are: ● **Default:** first IP from query result will be applied ● **Saved one until DNS TTL:** previous IP will be applied before reaches dns timeout ● **Saved one until no response:** previous IP will be applied even after dns time out until it cannot respond ● **Failback follows failback expiration timer:** Broadsoft R23 failback mode is used. The default value is set to "Default"
Register before DNS SRV failover	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.
Maximum Number of SIP Request Retries	Defines the number of SIP retries before DNS failover. the Default value is 2.
Failback Expiration (m)	Defines the expiration time in minutes when using Broadsoft R23 failback mode. The default value is 60. The valid range is from 0-64800 minutes. **Note:** Setting the Faliback Expiration timer to 0 will be helpful only when using Telstra Special Feature.
Primary IP	Specifies primary IP address where the base sends DNS query to, when “Use Configured IP” is selected for DNS mode.
Backup IP 1	Specifies backup IP 1 address where the base sends DNS query to, when “Primary IP” is not responding.
Backup IP 2	Specifies backup IP 2 address where the base sends DNS query to, when “Backup IP 1” is not responding.
NAT Settings	
NAT Traversal	Enables/disables NAT traversal mechanism. If activated (by choosing “STUN”) and a STUN server is also specified (Maintenance → Network Settings → STUN Settings); the base performs according to STUN client specification. Under this mode, embedded STUN client will detect if and what type of firewall/NAT is being used. If detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the base will use its mapped public IP address and port in all of its SIP and SDP messages. If NAT Traversal field is set to “Keep Alive”, the base will periodically (every 20 seconds) send a blank UDP packet (with no payload data) to SIP proxy to keep the “ping hole” on the NAT open.
Use NAT IP	Defines NAT IP address used in SIP/SDP messages. It should only be used if required by ITSP.
Proxy-Require	Determines a SIP Extension to notify the SIP server that the base is behind a NAT/Firewall.
Use SBC	Indicate whether or not a SBC server is used, if users want to work under SBC associated with 3CX, they should enable this feature to have better communication with the server.
SIP Settings	
Basic Settings	
SIP Transport	Selects transport protocol for SIP packets; UDP or TCP or TLS. Make sure your SIP server or network environment supports SIP over the selected transport method. Default is UDP.
SIP URI Scheme When Using TLS	Specifies if “sip” or “sips” will be used when TLS/TCP is selected for SIP Transport. The default setting is “sips”.
Use Actual Ephemeral Port in Contact TCP/TLS	Defines whether the actual ephemeral port in contact with TCP/TLS will be used when TLS/TCP is selected for SIP Transport. If set to No, these port numbers will use the permanent listening port on the phone. Otherwise, they

	will use the ephemeral port for the connection. The default setting is “No”.
SIP Registration	Controls whether to send SIP REGISTER messages to the proxy server. Device may not be able to make/receives calls if disabled. Default is Yes.
Unregister on Reboot	Controls whether to clear SIP user’s information by sending an un-register request to the proxy server. The un-registration is performed by sending a REGISTER message with the “Contact” header set to * and Expires=0 parameters to the SIP server. This will unregister all SIP accounts under concerned Profile. The default value is "No".
Add Auth Header on Initial REGISTER	Adds “Authentication” header with blank “nonce” attribute in the initial SIP REGISTER request. Default is No.
Outgoing Calls Without Registration	Enables the ability to place outgoing calls even when not registered (if allowed by ITSP); the device will not be able to receive incoming calls. Any HS member of a hunting group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary hunting group HS. For example, HS 1, 3, and 5 are the same Hunting Group members. HS 1 is registered with a SIP account. HS 3 and 5 are not reported. HS 3 and 5 will be able to place outbound calls using the SIP account of HS 1, even if Outgoing Call without Registration is set to No. The default is Yes.
Register Expiration	Refreshes registration periodically with specified SIP proxy (in minutes). Maximum interval is 65535 minutes (about 45 days). Default is 60 minutes (or 1 hour).
SIP Registration Failure Retry Wait Time	Sends re-register request after specific time (in seconds) when registration process fails. Maximum interval is 3600 seconds (1 hour). Default is 20 seconds.
SIP Registration Failure Retry Wait Time upon 403 Forbidden	Sends re-register request after specific time (in seconds) when registration process fails due to “403 Forbidden”. Valid range is 0 to 3600 in second. 0 second means stop retry registration. Default is 1200 seconds.
Reregister Before Expiration	Sends re-register request after specific time (in seconds) to renew registration before the previous registration session expires.
Local SIP Port	Defines local port to use by the base for listening and transmitting SIP packets. Default value for Profile 1 is 5060, 6060 for Profile 2, 7060 for Profile 3 and 8060 for Profile 4.
Use Random SIP Port	Controls whether to use configured or random SIP ports. This is usually necessary when multiple base stations are behind the same NAT. Default is No.
Local RTP Port	Defines local RTP port used to listen and transmit RTP packets. Default is 5004.
Use Random RTP Port	Defines local port to use by the base for listening and transmitting RTP packets. Default value for Profile 1 is 5004, 6004 for Profile 2, 7004 for Profile 3 and 8004 for Profile 4.
SIP T1 Timeout	Defines T1 timeout value. It is an estimate of the round-trip time between the client and server transactions. For example, the base station will attempt to send a request to a SIP server. The time it takes between sending out the request to the point of getting a response is the SIP T1 timer. If no response is received the timeout is increased to (2*T1) and then (4*T1). Request re-transmit retries would continue until a maximum amount of time defined by T2. Default is 0.5 seconds.
SIP T2 Timeout	Identifies maximum retransmission interval for non-INVITE requests and INVITE responses. Retransmitting and doubling of T1 continues until it reaches T2 value. Default is 4 seconds.
SIP Timer D	Configures SIP timer D defined in RFC3261. Range of values 0-64. Default is 0.
SIP Timer F	Configures SIP timer F defined in RFC3261. Range of values 0-64. The default is 0.
SIP Timer B	Configures SIP timer B defined in RFC3261. Range of values 0-64. Default is 0.

Enable OPTIONS Keep Alive	Enables OPTIONS Keep Alive, to check SIP server. Default is No.
OPTIONS Keep Alive Interval	Time interval for OPTIONS Keep Alive feature in seconds. Range of values is 1–64800. Default is 30.
OPTIONS Keep Alive Max Lost	A maximum number of lost packets for the OPTIONS Keep Alive feature before the phone sends a re-registration. Range of values 3-10. The default is 3.
Remove OBP from Route	Removes outbound proxy information from “Route” header when sending SIP packets. Default is No.
Support SIP Instance ID	Adds “SIP Instance ID” attribute to “Contact” header in REGISTER request as defined in IETF SIP outbound draft. Default is Yes. **Note:** Users can now unregister SIP instance ID to manage SIP sessions.
Hold Target Before Refer	Sends re-INVITE to hold transfer target before sending REFER message to transferee. Default is No.
Refer-To Use Target Contact	Includes target’s “Contact” header information in “Refer-To” header when using attended transfer. Default is No.
SUBSCRIBE for MWI	Sends periodic “SUBSCRIBE” requests (depends on “Register Expiration” parameter) for message waiting indication service. Default is No.
Enable 100rel	Appends “100rel” attribute to the “required” header of the initial signaling messages. Default is No.
TEL URI	Indicates E.164 number in the “From” header by adding “User=Phone” parameter or using “Tel:” in SIP packets, if the base has an assigned PSTN Number. Disabled: Will use “SIP User ID” information in the Request-Line and “From” header. Disabled: Will use “SIP User ID” information in the Request-Line and “From” header. Enabled: “Tel:” will be used instead of “sip:” in the SIP request. Please consult your carrier before changing this parameter. The default is Disabled.
Do Not Escape ‘#’ as %23 in SIP URI	Replaces “#” by “%23” when sending SIP packets. Default is No.
Disable Multiple m Line in SDP	Sends only one m line in SDP, regardless of how many m fields are in the incoming SDP. Default is No.
Use Privacy Header	Controls whether the Privacy Header will be present in SIP INVITE message. Default is Default.
Use P-Preferred-Identity Header	Controls whether PPI Header will be present in SIP INVITE message. Default is Default.
Ignore Alert-Info Header	This option is used to configure default ringtone. If set to “Yes”, configured default ringtone will be played. The default setting is No.
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”. When set to “From Header”, the phone will display the caller ID based on the From Header in the incoming SIP INVITE. The default setting is “Auto”.
Allow SIP Reset	Allows to reset the devices directly through SIP Notify. If “Allow SIP Reset” is set to “YES”, then the base receives the NOTIFY from the SIP server with Event: reset, the base should perform a factory reset after the authentication. The authentication in this case can be either with: The admin password if no SIP account is configured on the base. With the credentials of the SIP account if configured on the base. By default, it is set to "No".
Session Timer	

Enable Session Timer	Enables/Disables the Session Timer Support. Default is Yes.
Session Expiration	Enables periodic refresh of SIP session via a SIP request (UPDATE, or re-INVITE). When the session interval expires and there is no refresh via an UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. Default is 180 seconds.
Min-SE	Defines Minimum session expiration (in seconds). Default is 90 seconds.
Caller Request Timer	Uses session timer when making outbound calls if remote party supports it. Default is No.
Callee Request Timer	Uses session timer when receiving inbound calls with session timer request. Default is No.
Force Timer	It uses a session timer even if the remote party does not support this feature. Selecting “No” will enable session timer only when the remote party supports it. The default is No. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	Specifies which end will act as refresher for outgoing calls: **UAC:** The base station acts as the refresher. **UAS:** Callee or proxy server act as the refresher. Default is Omit.
UAS Specify Refresher	Specifies which end will act as a refresher for incoming calls: **UAS:** The base station serves as the refresher. **UAC**: Callee or proxy server act as the refresher. The default is Omit.
Force INVITE	Uses INVITE message to refresh the session timer. Default is No.
Security Settings	
Validate Incoming Messages	Defines whether incoming messages will be validated or not. Default is No.
Check SIP User ID for Incoming INVITE	Checks SIP User ID in the Request URI of incoming INVITE; if it doesn’t match the base SIP User ID, the call will be rejected. Direct IP calling will also be disabled. Default is No.
Accept Incoming SIP from Proxy Only	Checks SIP address of the Request URI in the incoming SIP message; if it doesn’t match SIP server address of the account, the call will be rejected. Default is No.
Authenticate Incoming INVITE	Challenges the incoming INVITE for authentication with SIP 401 Unauthorized message. Default is No.
Authenticate Server Certificate Domain	Checks server TLS certificate to ensure that common name matches the configured SIP server. Default is No.
Authenticate Server Certificate Chain	Checks server TLS certificate to ensure that it is authorized by a known certificate authority. Default is No.
Trusted CA Certificate	Treats entered certificate as a valid CA for authenticating the server TLS certificate. The default is No.
Audio Settings	
DTMF Settings	

Send DTMF	Specifies the mechanism to transmit DTMF digits. Default is via RTP (RFC2833) **In audio:** which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs); **Via RTP (RFC2833):** permits to specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF. **SIP INFO:** uses SIP INFO to carry DTMF. The defect of this mode is that it's easily to cause desynchronized of DTMF and media packet for the reason the SIP and RTP are transmitted respectively.
Disable DTMF Negotiation	Uses above DTMF order without negotiation. Default is No.
DTMF Payload Type	Defines payload type for DTMF using RFC2833.
Vocoder Settings	
Preferred Vocoder	Configures vocoders in a preference list (up to 8 preferred vocoders) that will be included with same order in SDP message. Vocoder types are G.711 A-/U-law, G.722, G.726-32, G.723, G.729, iLBC and OPUS
Voice Frames per TX	Transmits a specific number of voice frames per packet. Default is 2; increases to 10/20/32/64 for G711/G726/G723/other codecs respectively.
G723 Rate	Operates at specified encoding rate for G.723 vocoder. Available encoding rates are 6.3kbps or 5.3kbps. Default is 6.3kbps.
G726-32 Packing Mode	Defines G726-32 packing mode ("ITU" or "IETF"). Default is ITU.
iLBC Frame Size	Specifies iLBC packet frame size (20ms or 30ms). Default is 20ms.
iLBC Payload type	Determines payload type for iLBC. The valid range is between 96 and 127. Default is 97.
Disable OPUS stereo in SDP	Disables OPUS stereo attribute in SDP header. Default is No.
OPUS Payload Type	Determines OPUS payload type. The valid range is between 96 and 127. Default is 123.
Use First Matching Vocoder in 200OK SDP	Includes only the first matching vocoder in its 200OK response, otherwise it will include all matching vocoders in same order received in INVITE. Default is No.
Callee Codec Negotiation Priority	When callee, whose codecs are given priority in Codecs negotiation, local or remote. Set it Remote will use remote codec priority, or local will use local priority. Default is Remote.
SRTP Mode	Selects the SRTP mode to use ("Disabled", "Enabled but not forced", "Enabled and forced", "Follow SIP Transport", or "Optional"). 1. **Disabled:** SRTP is not used at all. Voice communication will be transmitted without encryption or authentication, which may pose a security risk. 2. **Enabled and forced: E**nsures that SRTP is always used for voice communication. The base station enforces the use of SRTP and requires connected devices to support it. If a device doesn't support SRTP, the call may be rejected or the communication won't be established. 3. **Follow SIP Transport:** When "Follow SIP Transport" is chosen, the SRTP protocol follows the transport protocol specified in the SIP signaling messages to determine when to start and stop the encryption and decryption of the RTP traffic. 4. **Optional:** This option allows SRTP to be used, but it is not mandatory. The base station can negotiate with the devices to determine whether to enable SRTP. If SRTP is agreed on, the communication will be encrypted and authenticated. If not, the communication will proceed without SRTP. The default is Disabled.
Crypto Life Time	Adds crypto life time header to SRTP packets. Default is Yes.

Silence Suppression (VAD)	Allows detecting the absence of audio and conserves bandwidth by preventing the transmission of “silent packets” over the network. Default is No.
Jitter Buffer Type	Selects jitter buffer type (Fixed or Adaptive) based on network conditions.
Jitter Buffer Length	High (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement. Medium (initial 100ms, min 20ms, max 200ms).
Voice Monitoring	
Enable RTCP	Enables RTCP statistics and control information. Default settings is RTCP.
Call Settings	
Early Dial	Sends an early INVITE each time a key is pressed when a user dials a number. Otherwise, only one INVITE is sent after full number is dialed (the user presses Dial Key or after “no key entry timeout” expires). This option should be used only if there is a SIP proxy configured and supporting “484 Incomplete Address” responses. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). The default is No. This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.
Dial Plan Prefix	Adds specified prefix to dialed number.
Dial Plan	Dial Plan Rules: Accept Digits : +,1,2,3,4,5,6,7,8,9,0, *, #, A,a,B,b,C,c,D,d ; Grammar: x – any digit from 0-9;xx+ – at least 2-digit number; xx – exactly 2-digit number; ^ – exclude; . – wildcard, matches one or more characters [3-5] – any digit of 3, 4, or 5; [147] – any digit 1, 4, or 7; <2=011> – replace digit 2 with 011 when dialing <=1> – add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed | – or Example 1: {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 10-digit numbers of leading digits 1617 Example 2: {^1900x+ | <=1617>xxxxxxx} – Block any number with leading digits 1900 and add prefix 1617 for any dialed 7-digit numbers Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing. Default: Outgoing – {x+} Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 | +x+ } Explanation of example rule (reading from left to right): ^1900x. – prevents dialing any number started with 1900 <=1617>[2-9]xxxxxx – allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically 1[2-9]xx[2-9]xxxxxx – allows dialing to any US/Canada Number with 11 digits length 011[2-9]x. – allows international calls starting with 011 [3469]11 – allow dialing special and emergency numbers 311, 411, 611 and 911 +x+ – allow dialing any digit with leading + sign; example: +16175669300 Note: In some cases, user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature. An example dial plan will be: { *x+ } which allows the user to dial * followed by any length of numbers. Dial Plan Bypass: Users can bypass the dial plan when making a call from call history by enabling the Feature using Pvalues: Profile 1: P2382=6 Profile 2: P2482=6 Profile 3: P2582=6 Profile 4: P2682=6

Bypass Dial plan	Enable/Disable the dial plan check while making outgoing calls. The default setting is “No”.
Use # as Dial Key	Treats “#” as the “Send” (or “Dial”) key. If set to “No”, this “#” key can be included as part of the dialed number. Default is Yes.
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 default is Yes.
On Hold Reminder Tone	Supports to disable or enable “On Hold Reminder Tone” to play a reminder tone when a call is on hold.
Match Incoming Caller ID	Specifies matching rules with number, pattern or Alert Info text. When the incoming caller ID or Alert Info matches the rule, the phone will ring with selected distinctive ringtone. Matching rules: • Specific caller ID number. For example, 8321123; • A defined pattern with certain length using x and + to specify, where x could be any digit from 0 to 9. Samples: xx+ : at least 2-digit number; xx : only 2-digit number; [345]xx: 3-digit number with the leading digit of 3, 4 or 5; [6-9]xx: 3-digit number with the leading digit from 6 to 9. • Alert-Info text Users could configure the matching rule as certain text (e.g., priority) and select the custom ring tone mapped to it. The custom ring tone will be used if the phone receives SIP INVITE with Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority Selects the distinctive ringtone for the matching rule. When the incoming caller ID or Alert Info matches the rule, the phone will ring with the selected ring. **Note:** Alert info can also be input as a string.
Allow Auto Answer by Call-info/Alert-Info	If set to “Yes”, the phone will automatically turn on the speaker phone to answer incoming calls, based on the SIP Call-Info/Alert-Info header sent from the server/proxy. The default setting is “No”
Custom Alert-Info for Auto Answer	Used exclusively to match the contents of the info parameter in the Alert-Info header for auto answer. The default auto answer headers will not be matched if this is defined.
Allow Barging for Auto Answer by Call-Info/Alert-Info.	Allows to auto answer a call by Call-Info/Alert-Info, even the phone is current in middle of a call, it is disabled by default.
Enable Call Features	Enables do not disturb, call forward and other call features via the local feature codes on the base. Otherwise, ITSP feature codes can be used. Default is Yes.
Disable Call Waiting Caller ID	Disables displaying caller ID when receiving a second incoming call. Default is No.
Enable Call Waiting in Parallel Mode	Enables call waiting for accounts using this profile which are set to Parallel ring mode. Default is No.
Disable Visual MWI	Disables use of visual message waiting indicator when there is an unread voicemail message. Default is No.
Transfer on Conference Hangup	Transfers the call to the other party if the conference initiator hangs up. Default is No.
Ring Timeout	Stops ringing when incoming call is not answered within a specific period of time. Default is 60 seconds.
Hunting Group Ring Timeout	Forwards incoming call to the next member of a hunt group if not answered within a specific period of time. Default is 20 seconds.
Send Anonymous	Sets “From”, “Privacy” and “P_Asserted_Identity” headers in outgoing INVITE message to “anonymous”, blocking caller ID. Default is No.

Anonymous Call Rejection	Rejects incoming calls with anonymous caller ID with “486 Busy here” message. Default is No.
Special Feature	Selects Soft switch vendors’ special mode. Example of vendors: Broadsoft, CBCOM, RNK, Huawei, ZTE IME, Phone Power, Metaswitch, Bitrix24, Telstra, and Kandy. Default is Standard.
Music On Hold URI	Configure the music on hold URI to call when a call is on hold if server support it.
Allow Unsolicited REFER	Allow unsolicited REFER to accomplish an outgoing call.
Feature Key Synchronization	When enabled, call features like DND, call forward, call waiting will be synchronized between the server and the phone. 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 following the same standard will be compatible with this feature.
Call Log	Configure the level of call logs or disable the call log.

Status Page Definitions

DECT Page Definitions

General Settings	
Base Station Name	Displays the name of the base station. Default is DP75x[last 6 digits of MAC address].
Admin PIN Code	Configures admin PIN code for authentication. Default is **0000**
Enable Repeater Mode	Enables the base station repeater mode to associate with available repeaters. Once enabled the base station starts searching for nearby repeaters and open subscription to associate with the available repeaters. This option requires rebooting the base station to take effect. Default is No.
Enable Repeater Management	Enables base station network management of discovered and paired repeaters. Once enabled, users need first to reboot the base station to take effect, then login on the web UI and browser to Status page, a new tab “DECT Repeater Status” will be available to display discovered and paired devices and also allowing users to associate / dissociate repeaters and also access their web GUI. Default is No.
DECT PTT Silence Timer	Sets timeout for PTT call (in minutes) if no handset unmutes. If set to 0, this timer will be disabled.
Clear Call Logs	Deletes call history logs of all handsets from base station
Handset Settings	**Handset:** Displays list of handsets’ indexes, from HS 1 to HS 5. **Handset Name:** Allows to customize handset name. Default is “HS1” to “HS5”. **Handset Phonebook:** Assigns private phonebook to handset. Each handset has a private phonebook or can be disabled. A private phonebook can be shared between specific handsets. Example: PB1 assigned to HS1 and HS2 **Off-hook Auto-dial:** Configures a number to auto dial when off-hook. **Off-hook Auto-dial Delay:** Configures 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.
SIP Account Settings	
Account	Displays list of accounts’ indexes, from account 1 to account 10.

SIP User ID	Enters SIP user ID provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	Enters account authenticate ID provided by VoIP service provider (ITSP). Can be identical to or different from "SIP user ID".
Password	Specifies account password provided by VoIP provider (ITSP) to register to SIP servers
Name	Chooses a name to be associated to user. Supported length is 256.
Profile	Selects the profile ID (1/2/3/4).
HS Mode	Determines HS modes; the base station supports 4 hunting group modes and 1 non-hunting group: ● **Circular mode:** all phones ring sequentially, starting with the phone after the one which rang last. ● **Linear mode:** all phones ring sequentially in the predetermined order, starting with the first phone each time. ● **Parallel mode:** all phones ring concurrently; after one phone answers, the remaining available phones can make new calls. ● **Non-Hunting Group:** an account will be assigned to a single specific handset.
Active	Activates/deactivates the account.
Handset Line Settings	
Handset Line Settings	Configures handset line settings; the base station supports up to 10 SIP accounts, 5 handsets; each Handset can be configured with up to 10 accounts. Please be aware that the handset line settings will be affected by DID settings (hunting group settings) in "DECT → SIP Account Settings". For more details about Hunting Group, please refer to [Hunting Groups].
Handset Settings (1-5)	
Enable Auto Answer	Enables / disables auto answer of incoming calls to handset. Default setting is No.
Enable Mute for Auto Answer	Enables/disables auto mute right after the call been answer, this can be configured by either the handset GUI or Web UI.
Enable Off hook on Cradle Pickup	Enables / disables off hook of handset when picked up from cradle. Default setting is No.
Enable on hook on Cradle Reposition	Enables / disables on hook of handset when repositioned on cradle. Default setting is No.
Enable Mute as DND	Enables to use Mute Key, in idle screen, as Global DND Key.
Disable Conference	Enables / disables the conference option on this handset. Default setting is No.
Disable Transfer	Enables / disables transfer option on this handset. Default setting is No.
Disable Busy Tone on Remote Disconnect	Enables / disables the busy tone heard in the handset when call is disconnected remotely.
Disable Call Waiting Tone	Disables playing call waiting tone during active call when receiving a second incoming call. The CWCID will still be displayed. Default is No.

Play warning tone for Auto Answer Intercom	Allows 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.
No Key Entry Timeout	Initiates the call within this time interval if no additional key entry during dialing stage. Default is 4 seconds.
Blind transfer wait timeout	Defines the timeout (in seconds) for no key entry during a blind transfer. Default is 30 seconds.
Custom Ringtone	Assigns custom ringtone to specific handset from the ringtones available on the base station. 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. Default is Disabled.
Time Format	Set the displayed Time Format on handsets to 12 hours or 24 hours. Default is 12hr.
Date Format	Set the displayed Date Format on handsets.

DECT Page Definitions

Settings Page Definitions

Network Settings – Basic Settings	
IP Address Mode	Selects IP address mode (DHCP, Static IP or PPPoE) for DP752 Base Station.
Preferred DNS Server	Specifies preferred DNS server to use when DHCP, PPPoE or Static mode is set.
DHCP Settings	
Host name (Option 12)	Specifies the name of the client. The name may or may not be qualified with the local domain name. This field is optional but may be required by ISP.
Vendor Class ID (Option 60)	Exchanges vendor class ID by clients and servers to convey particular configuration or other identification information about a client. Default is DP7XX.
PPPoE Settings	Configures PPPoE account ID, password and service name.
Static IP Settings	Sets IP address, subnet mask, gateway, DNS server 1, and DNS server 2.
Network Settings – Advanced Settings	
802.1X Mode	Enables/Disables 802.1X mode. To enable this mode, you should select EAP-MD5. Default is Disable.
802.1X Identity	Configures the identity for 802.1X mode.
MD5 Password	Determines the MD5 password for 802.1X mode.
802.1X CA Certificate	Uploads / deletes the 802.1X CA certificates.
802.1X Client Certificate	Uploads / Deletes the 802.1X Client Certificates.
Enable LLDP	Activates LLDP (Link Layer Discovery Protocol). Default is No.
Layer 2 QoS Settings	

Enable VLAN	Enables / Disables the VLAN mode. Default is Disabled.
Layer 2 QoS 802.1Q/VLAN Tag	Sets layer 2 QoS 802.1Q/VLAN tag. Default is 0.
Layer 2 QoS 802.1p Priority Value for SIP signaling	Sets layer 2 QoS 802.1p priority value for SIP signaling. Default is 0.
Layer 2 QoS 802.1p Priority Value for RTP media	Sets layer 2 QoS 802.1p priority value for RTP media. Default is 0.
STUN Settings	
Use STUN	Enables STUN. Default is No.
STUN server	Configures IP address or domain name of STUN server. Only non-symmetric NAT routers work with STUN.
Number of STUN Response Misses Allowed	Specifies number of STUN response misses allowed before restarting DHCP service. The minimum is 3 misses.
Keep-Alive Interval	Sends periodically a blank UDP packet to SIP server to keep “ping hole” on the NAT router open. Default is 20 seconds.
UPnP Discovery Settings	
Enable UPnP discovery	Enables/disables UPnP discovery feature. Default is Yes. After enabled, if you have a PC in the same network of DP752, you can browse it directly in your Network. Double click the device will open its web GUI in your default browser.
UPnP Discovery Notify Interval	Specifies in seconds the interval to send out SSDP notifies. Default settings is 30.
Management Virtual IP Address Settings	
Enable Management Interface	Allows administrator to setup a Virtual Network Interface on top of the physical interface for device management. Default is No.
Management Access	Chooses whether to access using “Management Interface Only” (Default) Or “Both Service and Management Interfaces”
Layer 2 QoS 802.1Q/VLAN Tag	Assigns the VLAN Tag of the Layer 2 QoS packets. Default is 0.
Layer 2 QoS 802.1p Priority Value	Assigns the priority value of the Layer 2 QoS packets. Valid range is 0 – 7.
IP Address Mode	Set the IP mode to either “DHCP” or “Static”. Default is DHCP.
Static IP Settings	
IP Address	Enter the IP address when static IP is used. Default is 192.168.100.100
Subnet Mask	Enter the Subnet Mask when static IP is used. Default is 255.255.255.0
Gateway	Enter the Default Gateway when static IP is used. Default is 192.168.100.1
DNS Server 1	Enter DNS Server 1 when static IP is used.

DNS Server 2	Enter DNS Server 2 when static IP is used.
Network → Open VPN® Settings	
OpenVPN® Enable	Enables/Disables the OpenVPN® feature. Default settings is No.
OpenVPN® Server Address	Configures the address of the OpenVPN® server.
OpenVPN® Port	Defines the port of the OpenVPN® server. Default is 1194.
OpenVPN® Transport	Determines network protocol UDP or TCP used for OpenVPN®. Default is UDP.
OpenVPN® CA	Uploads the OpenVPN® CA.
OpenVPN® Certificate	Uploads the OpenVPN® Certificate.
OpenVPN® Client Key	Uploads the OpenVPN® Client Key.
OpenVPN® Client Key Password	Password for the client.key file.
OpenVPN® Cipher Method	Must be the same cipher method used by the OpenVPN® server
OpenVPN® Username	OpenVPN® authentication username (optional)
OpenVPN® Password	OpenVPN® authentication password (optional)
OpenVPN® Comp-lzo	Enable OpenVPN® Comp-lzo or Not, Default Value is YES
Additional Options	Additional options to be appended to the OpenVPN® config file, separated by semicolon. For example: comp-lzo no;auth SHA256 Note: Please use with caution. Make sure that the options are recognizable by OpenVPN® and do not unnecessarily override the other configurations above.
SNMP Settings	
Enable SNMP	Enables/Disables the SNMP feature. Default settings is No
Version	Version SNMP version.
Port	SNMP port. Default is 161.
Trap Community	Configures the Name of SNMP trap community.
SNMP Trap Version	SNMP Trap Version. Default is Trap Version 3.
SNMP Trap IP	IP address of the SNMP trap receiver.
SNMP Trap Port	Port of the SNMP trap receiver. Default is 162.
SNMP Trap Interval	The interval between each trap sent to the trap receiver. Default is 60.
SNMP Username	Username for SNMP.

Security Level	noAuthUser: Users with security level noAuthnoPriv and context name as noAuth. authUser: Users with security level authNoPriv and context name as auth. privUser: Users with security level authPriv and context name as priv. Default is NoAuthUser.
Authentication Protocol	Select the Authentication Protocol: “None” or “MD5” or “SHA”. Default is None.
Privacy Protocol	Select the Privacy Protocol: “None” or “DES” or “AES”. Default is None.
Authentication Key	Enter the Authentication Key
Privacy Key	Enter the Privacy Key.
SNMP Trap Username	User name for SNMP Trap.
Trap Security Level	noAuthUser: Users with security level noAuthnoPriv and context name as noAuth. authUser: Users with security level authNoPriv and context name as auth. privUser: Users with security level authPriv and context name as priv. Default is NoAuthUser.
Trap Authentication Protocol	Select the Authentication Protocol: “None” or “MD5” or “SHA”. Default is None.
Trap Privacy Protocol	Select the Privacy Protocol: “None” or “DES” or “AES”. Default is None.
Trap Authentication Key	Enter the Trap Authentication Key.
Trap Privacy Key	Enter the Trap Privacy Key
External Service	
Order	Displays the order of the service. (1 – 10)
Service Type	Specifies the service’s type. Two options are available: None or GDS. Default setting is None. Note: The DP752 supports up 10 GDS items. For more details, refer to Facility Access Systems
Account	Specifies the account on which the service will be applied.
System Identification	Specifies the name to identify the service.
System Number	Specifies the system number, in case the service type option is set to GDS, the system number is the SIP user ID configured on GDS37xx, or the IP address of the GDS37xx itself if it’s using IP call.
Access Password 1	Determines the access password for Door1 defined in the GDS, in case the service type option is set to GDS, the access password 1 is the one configured on “Remote PIN to Open the Door1” field on GDS37xx settings.
Access Password 2	Determines the access password for Door2 defined in the GDS, in case the service type option is set to GDS, the access password 2 is the one configured on “Remote PIN to Open the Door2” field on GDS37xx settings.
Ring Tones	
System Ring Cadence	Sets ring cadences for all incoming calls. Syntax: c=on1/off1-on2/off2-on3/off3;) Default is set to c=2000/4000; (US standards) on1 is the period of ringing (“On time” in “ms”) while off1 is the period of silence. Up to three cadences are supported.

Call Progress Tones	Configures tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in "ms") while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeats the pattern. • "Dial tone" • "Ring back tone" • "Busy tone" • "Call-Waiting tone" Please refer to the document below to determine your local call progress tones: http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
System Features	
Disable Direct IP Call	Deactivates Direct IP-to-IP calling function. Default is No.
Enable DND Feature	If set to "No", a user cannot turn on Do Not Disturb feature via MUTE key, Softkey, or menu on LCD Set to "Yes "by Default.
Return Code When Enable DND	When DND is enabled, the phone will send the selected type of SIP message. The options are: ● Busy(486) ● Temporarily Unavailable(480) ● Not Found(404) ● Decline(603) By default, it is set to Temporarily Unavailable(480)
SIP User-Agent	Configures the SIP User-Agent header. When left blank, the default User-Agent is used. (**example** => User-Agent: Grandstream DP752 1.0.21.15)
In-call Dial Number on Pressing Transfer Key	Configures the number to be dialed as DTMF when using the TRANSFER button.

Maintenance Page Definitions

Firmware Upgrade	
Firmware Upgrade and Provisioning	Selects how firmware upgrade request will be sent: "Always Check for New Firmware", "Check New Firmware only when F/W pre/suffix changes", or "Always Skip the Firmware Check".
HTTP/HTTPS User Name	Enters user name to authenticate with HTTP/HTTPS server.
HTTP/HTTPS Password	Enters password to authenticate with HTTP/HTTPS server.
Always send HTTP Basic Authentication Information	Includes configured user name and password in HTTP request before receiving authentication challenge from the server. Default is No.
Verify host when using HTTPS	Verifies host name in server certificate when using HTTPS. Default is Yes.
Upgrade via	Selects firmware upgrade method: FTP/FTPS or HTTP/HTTPS.
Firmware Server Path	Sets IP address or domain name of firmware server. The URL of the server that hosts the firmware release. Default is fm.grandstream.com/gs. **Note:** Make sure to not prepend address with "http://" or any other protocol

Firmware File Prefix	Checks if firmware file is with matching prefix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.
Firmware File Postfix	Checks if firmware file is with matching postfix before downloading it. This field enables user to store different versions of firmware files in one directory on the firmware server.
Allow DHCP Option 43 and Option 66 to Override Server	Obtains configuration and upgrade server's information from DHCP server using options 66 and 43. **Note:** If DHCP Option 66 is enabled, the base will attempt downloading the firmware file from the server URL provided by DHCP, even though Firmware Server Path is left blank.
Automatic Upgrade	Specifies when the firmware upgrade process will be initiated; there are 4 options: ● **Yes, check for upgrade at random every X days:** User needs to specify a period in days. ● **No:** Base station will only do upgrade once at boot up. ● **Check every X minutes:** User needs to specify a period in minutes. ● **Check every day:** User needs to specify "Hour of the day (0-23)". ● **Check every week:** User needs to specify "Hour of the day (0-23)" and "Day of the week (0-6)". Note: Day of week is starting from Sunday. Default is No.
Firmware Key	Decrypts the firmware file using specified key (32-digit in Hexadecimal) when encrypted.
Handset Firmware	● **Upload:** Uploads handset firmware. Reboot the device after uploading to apply and use the new firmware. ● **Delete:** Deletes previously uploaded handset firmware. ● **Automatic Upgrade:** Enables automatic upgrade when the handset detects the new firmware.
Provisioning	
XML Config File Password	Decrypts XML configuration file when encrypted. The password used for encrypting the XML configuration file is using OpenSSL.
HTTP/HTTPS User Name	Enters user name to authenticate with HTTP/HTTPS server.
HTTP/HTTPS Password	Enters password to authenticate with HTTP/HTTPS server.
Always send HTTP Basic Authentication Information	Includes configured user name and password in HTTP request before receiving authentication challenge from the server. Default is No.
Verify host when using HTTPS	Verifies host name in server certificate when using HTTPS. Default is Yes
Provisioning via	Choose the method that the base station uses to request handset ipei config file. Can use TFTP, HTTP,FTP,FTPS or HTTPS.
Config Server Path	Sets IP address or domain name of configuration server. The server hosts a copy of the configuration file to be installed on the DP75x. Default is **fm.grandstream.com/gs**. **Note:** Make sure to not prepend address with "http://" or any other protocol.
Config File Prefix	Checks if configuration files are with matching prefix before downloading them. This field enables user to store different configuration files in one directory on the provisioning server.

Config File Postfix	Checks if configuration files are with matching postfix before downloading them. This field enables user to store different configuration files in one directory on the provisioning server.
Enable Handset Config Upgrade	Checks to allow handset config upgrade for handset related settings. Default is No.
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.
Allow DHCP Option 43 and Option 66 to Override Server	Obtains configuration and upgrade server's information from DHCP server using options 66 and 43. **Note:** If DHCP Option 66 is enabled, the DP75x will attempt downloading the firmware file from the server URL provided by DHCP, even though Config Server Path is left blank.
Allow DHCP Option 120 to Override SIP Server	Obtains configuration and update SIP server information from DHCP server using options 120. **Note:** If DHCP option 120 is enabled, the sip server setting can be changed for profile 1.
3CX Auto Provision	Sends multicast "SUBSCRIBE" message for provisioning at booting stage, used for PnP (Plug-and-Play) configuration. Default is Yes.
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to "Yes". Default setting is "No".
Automatic Provisioning	Specifies when provisioning process will be initiated; there are 4 options: ● **No:** Base station will only request configuration files once at boot up. ● **Check every X minutes:** User needs to specify a period in minutes. ● **Check every day:** User needs to specify "Hour of the day (0-23)". ● **Check every week:** User needs to specify "Hour of the day (0-23)" and "Day of the week (0-6)". Note: Day of week is starting from Sunday. Default is No.
Hour of the Day (0-23)	Defines the hour of the day to check the HTTP/TFTP server for configuration file changes. Default is 1
Day of the Week (0-6)	Defines the day of the week to check HTTP/TFTP server for configuration file changes. Default is 1
Authenticate Conf File	Authenticates configuration file before being accepted. This protects the device configuration from unauthorized modifications. Default is No.
Upload Device Config	Uploads manually device configuration (XML and TXT formats are supported) to base station.
Device Config (TXT)	Downloads actual device configuration file in .txt format.
Device Config (XML)	Downloads default device configuration file in .xml format.
Backup Configuration	Generates an XML config file storing all current configuration after pressing Backup Settings and display list of backup files available (users need to wait a few seconds and refresh the provisioning page to display the backup files). Maximum of 10 backup files are supported and when it reaches 10 entries, this feature will be disabled. **Note:** Click on Download to download the specific backup or Delete to remove all the backup files.
Web/SSH Access	

User Password	Configures user level password. Case sensitive and max. length of 30 characters.
Confirm Password	Configures the new user password again to confirm the new password.
Admin Password	Configures admin level password. Case sensitive and max length is 30 characters.
Confirm Password	Configures the new user password again to confirm the new password.
Access Control Lists	• **White list for WAN side:** If white list exists, then only these IP addresses are allowed to web and SSH access. • **Black list for WAN side:** If black list exists and white list is empty, then only these IP addresses are not allowed to web and SSH access
TR-069	
Enable TR-069	Enables / Disables TR-069 service. The default is No. **Note:** TR-069 supports Extensible Messaging and Presence Protocol (XMPP) connections.
ACS URL	Specifies URL of TR-069 Auto Configuration Server. (e.g., http://acs.mycompany.com), or IP address.
ACS Username	Enters username to authenticate to ACS.
ACS Password	Enters password to authenticate to ACS.
Periodic Inform Enable	Sends periodic inform packets to ACS. Default is No.
Periodic Inform Interval	Configures to sends periodic “Inform” packets to ACS based on specified interval.
Connection Request Username	Enters username for ACS to connect to the base station.
Connection Request Password	Enters password for ACS to connect to the base station.
Security Settings - Web / SSH	
HTTP Web Port	Customizes HTTP port used to access base station web UI. Default is 80.
HTTPS Web Port	Customizes HTTPS port used to access base station web UI. Default is 443.
Web Access Mode	Sets the protocol for the web interface, the options are: HTTPS, HTTP, Both
Enable User Web Access	The administrator can disable or enable user web access. Disabled by Default
Disable SSH	Disables SSH access. Default is No.
SSH Port	Customizes SSH access port. Default is 22.
Security Settings - SIP TLS Settings	
SIP TLS Certificate	Specifies SSL certificate used for SIP over TLS in X.509 format. Base station has built-in private key and SSL certificate. Max allowed length 8192.

SIP TLS Private Key Password	Specifies SSL certificate key used for SIP over TLS in X.509 format. Base station has built-in private key and SSL certificate.
SIP TLS Private Key Password	Specifies SSL certificate key password used for SIP over TLS in X.509 format.
Security Settings - RADIUS Server Settings	
Primary RADIUS Server	Defines the primary RADIUS server (Remote Authentication Dial-In User Service) to authenticate dial-in users and authorize their access to requested system or service.
Primary RADIUS Authentication Port	Uses specified port for authentication with the primary RADIUS server. Default is 1812.
Primary RADIUS Account Port	Specifies port to be used for the primary RADIUS account. Default is 1813.
Primary RADIUS Server Secret	Enters secret string to be used to authenticate the RADIUS connection to the primary server. It should match RADIUS configuration.
Secondary RADIUS Server	Sets IP or FQDN of the secondary RADIUS server. In case primary radius server becomes unusable, the secondary will take role and manage credit resources in the network.
Secondary RADIUS Authentication Port	Uses specified port for authentication with the secondary RADIUS authentication. Default is 1812.
Secondary RADIUS Account Port	Specifies port to be used for the secondary RADIUS account. Default is 1813.
Secondary RADIUS Sever Secret	Enters secret string to be used to authenticate the RADIUS connection to the secondary server. It should match RADIUS configuration.
RADIUS Timeout	Specifies period of time before request is cancelled if no response. Default is 2.
RADIUS Retry	Specifies amount of retry attempts if RADIUS communication failure. Default is 3.
Security Settings – Device Custom Certificate	
Device Custom Certificate	Configures the certificate used by device during provisioning and web access.
Device Custom Private Key	Configures the private key used by device during provisioning and web access.
Enable Device Custom Certificate	When enables the device will use the custom certificate, otherwise it will use the default certificates. Default is No.
Security Settings - Security	
Enable/Disable Weak Ciphers	Enable/Disable Weak Ciphers Suites, the available options are: • **Enable Weak TLS Ciphers Suites:** This option enables the use of weak TLS (Transport Layer Security) cipher suites for encryption in the DECT base station. Weak cipher suites are generally discouraged as they may not provide strong security. • **Disable Symmetric Encryption RC4/DES/3DES:** This option disables the use of symmetric encryption algorithms like RC4, DES (Data Encryption Standard), and 3DES (Triple Data Encryption Standard). These algorithms are considered weak and insecure for modern communication. • **Disable Symmetric Encryption SEED:** This option disables the use of the SEED encryption algorithm. SEED is a symmetric encryption algorithm that may be considered weak or not up to modern security standards. • **Disable All Weak Symmetric Encryption:** This option disables all weak symmetric encryption algorithms, which could include the ones mentioned in options 2 and 3, as well as others not specifically listed. • **Disable Symmetric Authentification MD5:** This option disables the use of the MD5 (Message Digest 5) algorithm for symmetric authentication. MD5 is considered weak and vulnerable to certain types of attacks.

	• **Disable All Weak TLS Ciphers Suites:** This option disables all weak TLS cipher suites, ensuring that only strong and secure TLS encryption methods are used. The default value is set to "Enable Weak TLS Ciphers Suites"
Validate Server Certificates	Validate server certificates with our trusted list for TLS connections. If set to "No", device will bypass certificate validation (not recommended)
Trusted CA Certificates	
CA Certificate	Upload Trusted CA certificate to use for HTTPS/FTPS server. Up to six CA Certificates can be uploaded. **Note:** New DigiCert certificates were added with a firmware upgrade to version 1.0.19.2
Load CA Certificates	Configures what certificates to use to authenticate with the server: • **Default Certificates (built-in)** • **Custom Certificates** • **All Certificates**
Date and Time	
NTP Server	Defines URL or IP address of the NTP (Network Time Protocol) server. Used by the base to synchronize the date and time. A list of public NTP servers can be found at pool.ntp.org.
NTP Update Interval	Contacts NTP server at specified period of time (in minutes) to obtain the date and time. Default is 60 minutes.
Allow DHCP Option 42 to NTP Server	Obtains NTP server address from a DHCP server using DHCP Option 42; it will override configured NTP Server. If set to “No”, the base will use configured NTP server to synchronize time and date even if a NTP server is provided by DHCP server. Default is No.
Time Zone	Selects time zone to define date/time on the base.
Self-Defined Time Zone	Allows users to define their own time zone.
Allow DHCP Option 2 to override time zone	Obtains time zone setting (offset) from a DHCP server using DHCP Option 2; it will override selected time zone. If set to “No”, the base station will use selected time zone even if provided by DHCP server. Default is No.
Custom Time Zone Settings	
Custom Time Zones	Enable or disable customized time zone. Default is Disabled.
UTC Offset	Offset is negative for time zone following UTC and positive for time zone leading UTC (-12 to +14). Default is 0.
Second Offset	Offset for second time zone. Offset is negative for time zone following UTC and positive for time zone leading UTC (-12 to +14). Default is 0.
DST Type	Set Daylight Savings Time by day or by week. Default is by week.
Start & End Times	Starting and ending times for the time zone. Default Start is 2:0 ; Default End is 2:0.
Time zone settings by week	• **Start & End Months:** Starting and ending months for the time zone. Default Start is April; Default End is September. • **Start & End settings by week :** Starting and ending settings by week for the

	Default Start is Sunday, Week 1 ; Default End is Sunday, Week1.
Time zone settings by day	● **Start & End settings by day:** Starting and ending settings by day (0-365) for the time zone. Default Start is 1 ; Default End is 1.
Manual Time Settings	
Manual Time	Disables/Enables manual time settings. Default is Disabled.
Year / Month / Day	Set the Year (2016-2037), Month (1-12) and Day (1-31) manually.
Hour : Minute	Enter hour (0-23), Minute (0-59) manually
Syslog	
Syslog Protocol	Allow sending syslog through secured TLS protocol to syslog server. Default is UDP. **Note:** CA verification is required
Syslog Server	Sets IP address or URL of system log server. The server collects system log information from the base station. User may configure the Syslog port by adding it after the Syslog Server address. When configured, the Syslog messages will be sent to this port instead of the default syslog port. Default is 514.
Syslog Level	Selects log level; the level is one of DEBUG, INFO, WARNING, ERROR, EXTRA DEBUG (default is NONE). Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: *GS_LOG: [device MAC address] [error code] error message.* Example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Print SIP in Syslog	Includes full SIP messages in syslog.
Packet Capture	
Status	Displays packet capture status. When user starts to capture trace file, it will show “RUNNING” status, otherwise, it will show “STOPPED”.
With RTP Packets	Defines whether packet capture file contains RTP or not. Default setting is No.
With Secret key information	If set to “Yes, the packet capture will include secret key to decrypt the capture TLS packets. By default, it is enabled.
Factory Reset	

Force Reboot	Kills active processes and forces the reboot of DP75x base station.
Configue Web UI Button	Specifies the type of reset to perform via the Web UI reset button: - There are 3 reset type for Web UI button: 1. **Full Factory Reset:** Reset all the settings 2. **NVRAM Settings Only:** 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:** Reset only the subscription (wipe out all the subscription information), but keep all the other settings like account, SIP settings. Press Reset button to reset settings based on the reset type selected.
Configure Hardware Button	Specifies the type of reset to perform via hardware button: - There are 4 reset types for hardware button: 1. **Full Factory Reset:** Reset all the settings. 2. **NVRAM Settings Only:** 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:** Reset only the subscription (wipe out all the subscription information), but keep all the other settings like account, SIP settings. 4. **Disabled:** Disable the hardware button to factory reset the base for security purpose
Automatic Reboot	
Automatic Reboot	**No:** if selected the automatic reboot will not be configured **Yes, reboot daily:** if selected, the automatic reboot will be performed daily **Yes, reboot every week on:** select the day in which you want the automatic reboot to be performed **Yes, reboot every month on day:** select the day in which you want the automatic reboot to be performed
Support - Support Documentation	
Online Support	Redirects users to tools page and DP7xx product pages available on Grandstream official website.
Offline Support	Allows users to download the drilling templates.
Support - Configuration Tools	
Download Default Device Configuration	Downloads the default device configuration file in .txt and .XML formats.
Download UCM Zero Config Template	Download the zero config templates files for UCM firmware version before and after 1.0.10.39.
Support - Debug Tools	
Remote Log Submission - Terms & Conditions	● **Agree to terms?: By clicking the submit button, you agree to terms and conditions for submitting debug logs and allow the device to send these files to Grandstream to be used for troubleshooting purposes. Note: Grandstream will not collect any information without customers acknowledged and permissions.** ● **Email Address:** Configures email field for core collection tools. Users can leave a contact email to get feedbacks. ● **Comments:** Includes 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.

Debug Log Files	Kills phone control process on base station and generates core file and other debug information and also it allows users to delete the generated log files. Reboot required to restart process.
Handset Notification	Notifies handset when a debug file is available after recovery. Default is No.
Outbound Notifications - Action URL	
Setup Completed	Configures the Action URL to send when phone finishes setup process.
Registered	Configures the Action URL to send when phone successfully registers a SIP account.
Unregistered	Configures the Action URL to send when phone unregisters a SIP account.
Off-hook	Configures the Action URL to send when phone is in off-hook state.
On-hook	Configures the Action URL to send when phone is in on-hook state.
Incoming Call	Configures the Action URL to send when phone receives an incoming call.
Outgoing Call	Configures the Action URL to send when phone places a call.
Missed Call	Configures the Action URL to send when phone has a missed call.
Established Call	Configures the Action URL to send when phone establishes a call.
Terminated Call	Configures the Action URL to send when phone terminates a call.
Enable DND	Configures the Action URL to send when phone enables DND.
Disable DND	Configures the Action URL to send when phone disables DND.
Enable Call Forward	Configures the Action URL to send when phone enables call forward function.
Disable Call Forward	Configures the Action URL to send when phone disables call forward function.
Blind Transfer	Configures the Action URL to send when phone performs blind transfer.
Attended Transfer	Configures the Action URL to send when phone performs attended transfer.
Hold Call	Configures the Action URL to send when phone places a call on hold.
Unhold Call	Configures the Action URL to send when phone resumes the call on hold.

Maintenance Page Definitions

Phonebook Page Definitions

Global Phonebook XML Settings

Global Phonebook Type	Selects type of global phonebook to use. ○ If set to **XML**, DP752 will use the configuration in Global Phonebook XML Settings page. ○ If set to **LDAP**, DP752 will use configuration in Global Phonebook LDAP Settings page.
Automatic XML Phonebook Download	
Enable Automatic XML Phonebook Download	Sends periodic requests to download XML Phonebook via HTTP, HTTPS, or TFTP.
HTTP/HTTPS User Name	Enters user name to authenticate with HTTP/HTTPS server.
HTTP/HTTPS Password	Enters password to authenticate with HTTP/HTTPS server.
Phonebook XML Server Path	Indicates server path to download XML phonebook file. This field could be IP address or URL, with up to 256 characters.
Phonebook Download Interval	Sets interval to send XML phonebook download requests (in minutes). If set to 0, automatic download is disabled. Valid range is 5 to 720. Default is **5** minutes.
Manual XML Phonebook Management	
Import XML Phonebook	**Upload:** Uploads manually global XML phonebook file to the base station. **Delete:** Clears global XML phonebook file in the base station.
Export XML Phonebook	Downloads global XML phonebook from the base station in .xml format.
Broadsoft XSI	
XSI	
Server	Configure the BroadWorks Xsi server URI. If the server uses HTTPS, please add the header “HTTPS” ahead of the Server URI. For instance, “https://SERVER_URI”.
Port	Configure the BroadWorks Xsi server port. The default port is 80. If the server uses HTTPS, please configure 443.
XSI Authentication Type	This feature allows users to configure the deployment path for Broadsoft XSI Actions. If it is empty, the path “com.broadsoft.xsiactions” will be used
Login Credentials	
Login Username	Configure the Username for the BroadWorks XSI server.
Login Password	Configure the password for the BroadWorks XSI server.
SIP Credentials	
SIP UserName	Configure SIP Username for the BroadWorks XSI server.
SIP User ID	Configure SIP User ID for the BroadWorks XSI server.
SIP Password	Configure SIP Password for the BroadWorks XSI server.

Network Directories	
Enable/Disable and choose the name of the following types: **Group Directory; Enterprise Directory; Group Common; Enterprise Common; Personal Directory.**	
Global Phonebook LDAP Settings	
Global Phonebook Type	Selects type of global phonebook to use. If set to XML, DP752 will use the configuration in Global Phonebook XML Settings page. If set to LDAP, DP752 will use configuration in Global Phonebook LDAP Settings page.
LDAP Phonebook Settings	
LDAP protocol	Chooses LDAP or LDAPS (LDAP over TLS) protocol. Default is **LDAP**.
Server Address	Configures IP address or domain name of the LDAP server.
Port	Determines LDAP server port. Default is **389**.
Base	Indicates the location in the directory where the search is requested to begin. Example: dc=grandstream, dc=com ou=Boston, dc=grandstream, dc=com
User Name	Binds "Username" for querying LDAP servers. Some LDAP servers allow anonymous binds in which case the setting can be left blank.
Password	Binds "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
LDAP Filter	LDAP filter to limit which contacts are fetched from the server. LDAP statement to limit which contacts are fetched from the server. Statement must be in parenthesis.
LDAP Version	Selects LDAP protocol version to send bind requests. Default is **Version 3**.
First Name Attribute	Defines first name attributes of each record to be returned in the LDAP search result. This field allows users to configure multiple space separated name attributes. Example: gn cn sn description

Last Name Attribute	Defines last name attributes of each record to be returned in the LDAP search result. This field allows users to configure multiple space separated name attributes. Example: gn cn sn description
Work Number Attribute	Specifies which LDAP attribute represent the contact's work number. Must be in number attributes on LDAP server.
Home Number Attribute	Specifies which LDAP attribute represent the contact's home number. Must be in number attributes on LDAP server.
Mobile Number Attribute	Specifies which LDAP attribute represent the contact's mobile number. Must be in number attributes on LDAP server.
Max. Hits	Specifies maximum number of results to be returned by LDAP server. If set to 0, server will return all search results. Valid range is 1 to 3000. Default is **500**.
Search Timeout	Sets interval (in seconds) for the server to process the request and return search results to the client. Default is **30** seconds.
Private Phonebook Settings	
Phonebook Name	Defines private phonebook name.
Import XML Phonebook	**Upload:** Uploads manually private XML phonebook file to the base station. **Delete:** Clears private XML phonebook file in the base station
Export XML Phonebook	Downloads private XML phonebook from the base station in .xml format.

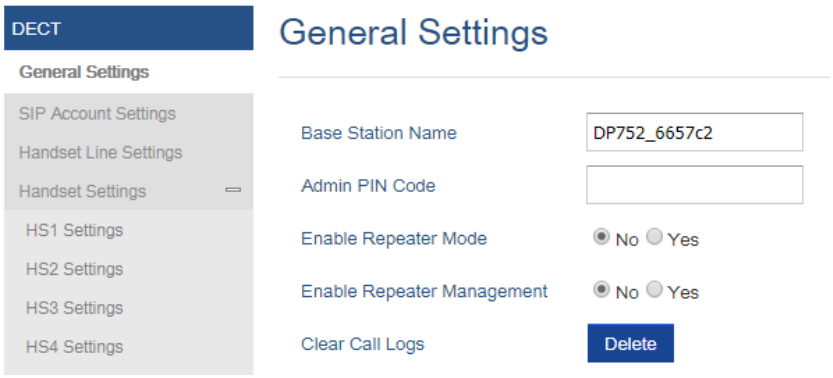
Phonebook Page Definitions

Change Base Station Admin PIN code

For security reasons, advanced settings in DP752 base station cannot be accessed from DP730/DP722 Handsets except if Admin PIN code is provided. By default, Admin PIN code is 0000.

We strongly recommend changing your Admin PIN code following below steps:

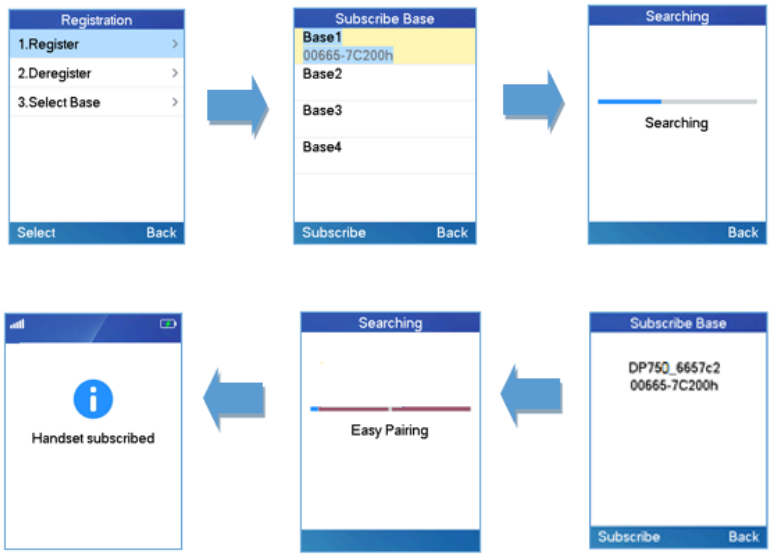
1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **DECT → General Settings** tab.
4. Enter your new **Admin PIN Code** (only digits accepted) in appropriate field.
5. Press **Save and Apply** to save your settings.



Admin PIN Code

Register DP730/DP722 Handsets to DP752 Base Station

1. On DP752 Base station, press and hold the Radio/Page  button for 4 seconds until the Radio icon starts blinking to start subscription process. Or Access web UI, and press **Subscribe** icon to **Open Subscription.**
2. On DP730/DP722, press "Subscribe" softkey if available on the main screen or access **Menu → Registration → Register** while the DP752 Radio icon is blinking.
   Note: "Subscribe" softkey appears only if DP730/DP722 is not registered to any DP752 base station.
3. Select **BaseX** (X=1-4) corresponding to the desired base station DP752, then press **Subscribe**.
4. The DP730/DP722 will search for nearby base stations and will display the RFPI code and Base station name of the discovered DP752.
5. Press **Subscribe** to pair with the displayed DP752.
6. The DP730/DP722 will display **Easy Pairing** on the LCD and play an audible buzz when successful. Then it will return to the home screen, displaying the Handsets name and number assigned by the registered base station.



Registration Process

Using DP730/DP722 with Multiple DP752 Base Stations

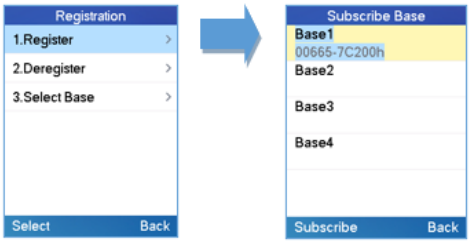
DP730/DP722 is able to be registered to up four different DP752 base stations.

Registering DP730/DP722 to an additional DP752 base station

Considering DP730/DP722 is previously registered to an initial base station, please follow below steps to register a Handsets to an additional base station:

1. Press **Menu** (left softkey or the selection key) to bring up operation menu.

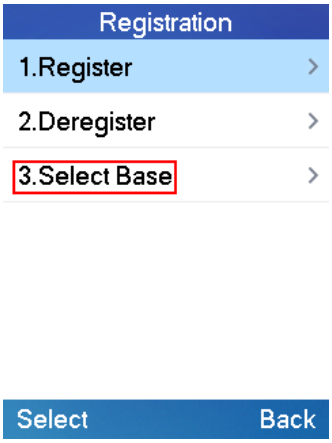
2. Use arrow keys to reach **Registration**.
3. Select **Register**.
4. Navigate to an unsubscribed base using arrow keys, and click on **Subscribe**.
5. Make sure that the subscription is opened on the new base station.



Multiple Base Stations Registration

Switching Between Different Base Stations

1. Press "Menu" (left softkey or the selection key) to bring up operation menu.
2. Use arrow keys to reach **Registration**.
3. Navigate to **Select Base** using arrow keys.
4. Select the desired base station and press **Select**.



Switching Between Base Stations

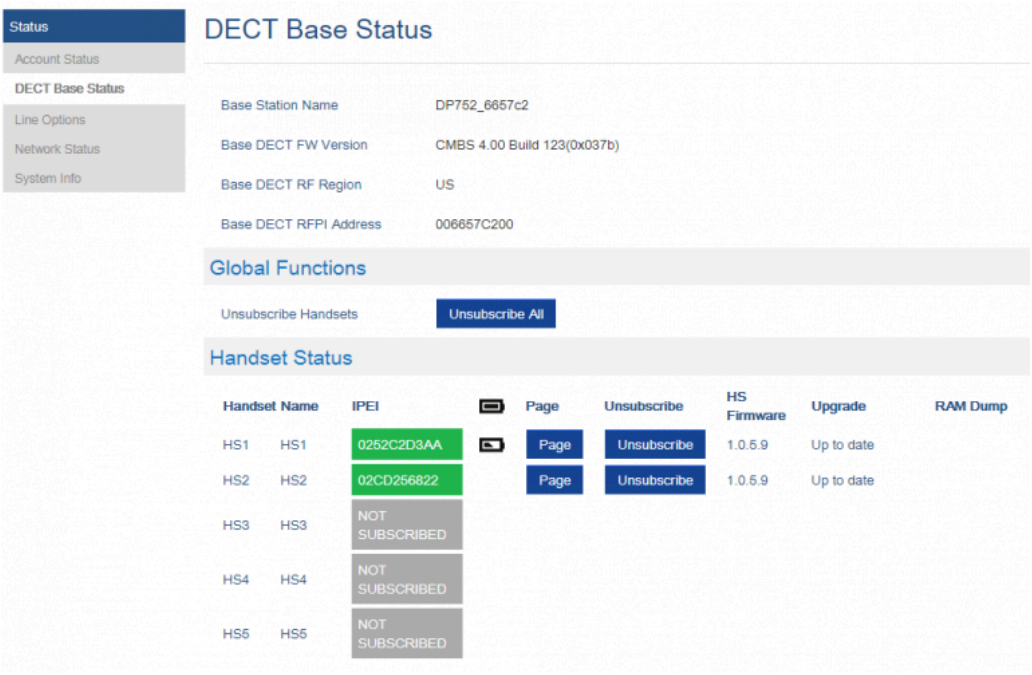
Unregister the DP730/DP722

- **Using DP730/DP722 Handsets:**

1. On DP730/DP722, press "Menu" (left softkey or the selection key) to bring up operation menu.
2. Press arrow keys to move the cursor to **Registration**, then press "Select" (left softkey).
3. Navigate to **Deregister**.
4. Select the Handsets to be unregistered and press "Deregister" (left softkey).
5. Enter the system PIN code (default: 0000).
6. Press "Done" (left softkey) to confirm or "Back" (right softkey) to cancel.

- **Using DP752 Base Station UI:**

1. Access DP752 Web Interface.
2. Go to **Status → DECT Base Status**.
3. Locate the Handsets to unregister and press "**Unsubscribe**" button.



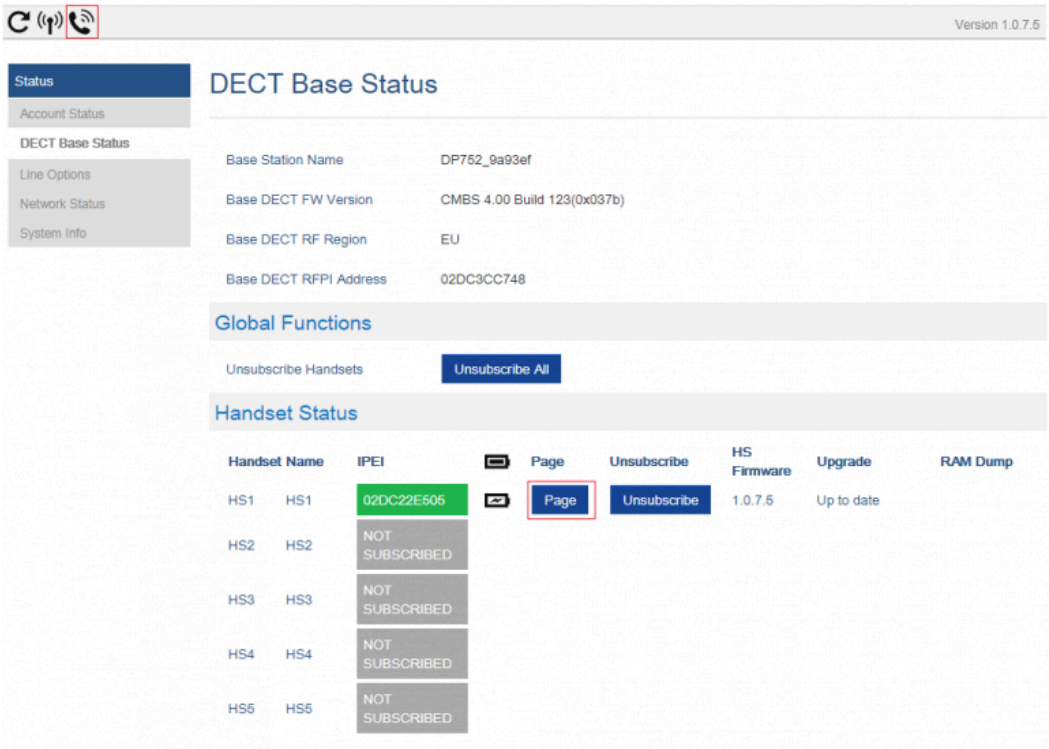
Unregister DP730/DP722 from DP752 web UI

Locating DP730/DP722 Handsets from DP752 Base station

In some situations, you may have a DP730/DP722 Handsets incorrectly positioned and you don't know its current location. You can locate a DP730/DP722 Handsets from his registered DP752 base station using below steps:

Locate via DP752 Web UI

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings and navigate to **Status → DECT Base Status** tab.
3. Choose which Handsets to locate and press its corresponding **Page** button.
4. A paging call will be received on the selected DP730/DP722 Handsets.
   Note: If you press **Page All** icon , all registered DP730/DP722 Handsets will be receiving paging call.
5. Once located, you can press End Softkey to end paging call.



Locate Handsets via Web UI

Locate via DP752 Base station

1. On DP752 Base station front side, press **Radio/Page**  button.
2. All registered Handsets will receive Paging call.
3. Once located, you can end the paging calling by pressing any key on the Handsets or by pressing again **Radio/Page**  button.

Register a SIP Account

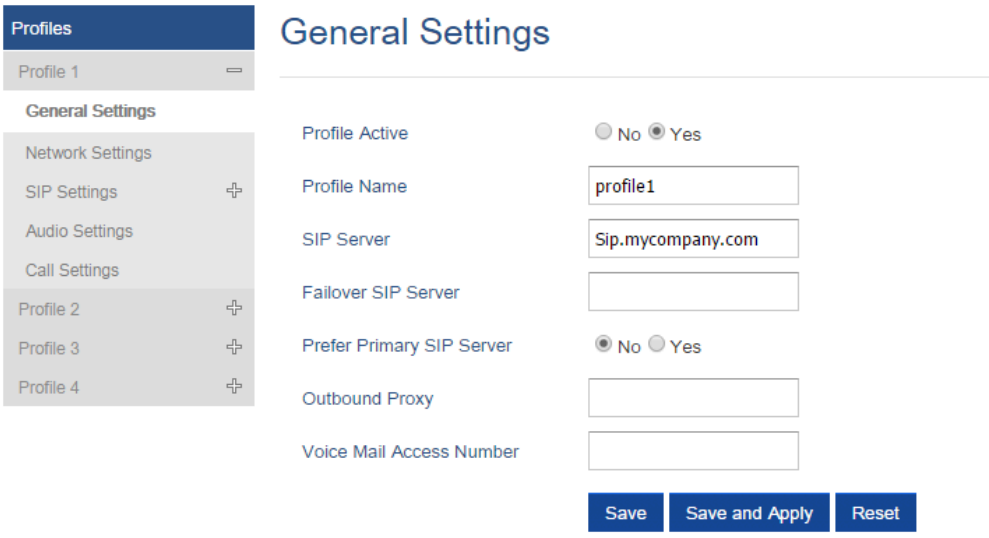
DP752 supports up to 10 SIP accounts, 5 Handsets. Each Handsets can be configured up to 10 accounts. Please be aware that line settings will be affected by DID settings (hunting group settings) in "DECT → SIP Account Settings".

Register Account via Web User Interface

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings and navigate to **Profiles** tab and select a profile to use.
   DP752 supports up to 4 profiles. A profile is a set of settings including general settings, network settings, SIP setting, audio setting, call settings and ring tones, etc.
   A profile can be used with different SIP accounts.
3. In **General Settings**, set the following:
   - **Profile Active** to **Yes.**
   - **SIP Server** field with your SIP server IP address or FQDN.
   - **Failover SIP Server** with your Failover SIP Server IP address or FQDN. Leave empty if not available.
   - **Prefer Primary SIP Server** to **No** or **Yes** depending on your configuration. Set to **No** if no Failover SIP Server is defined. If "**Yes"**, account will register to Primary SIP Server when failover registration expires.
   - **Outbound Proxy** with your Outbound Proxy IP Address or FQDN. Leave empty if not available.

For more information, related to above options please refer to [Profiles Page Definitions].

4. Press **Save and Apply** to save your configuration.

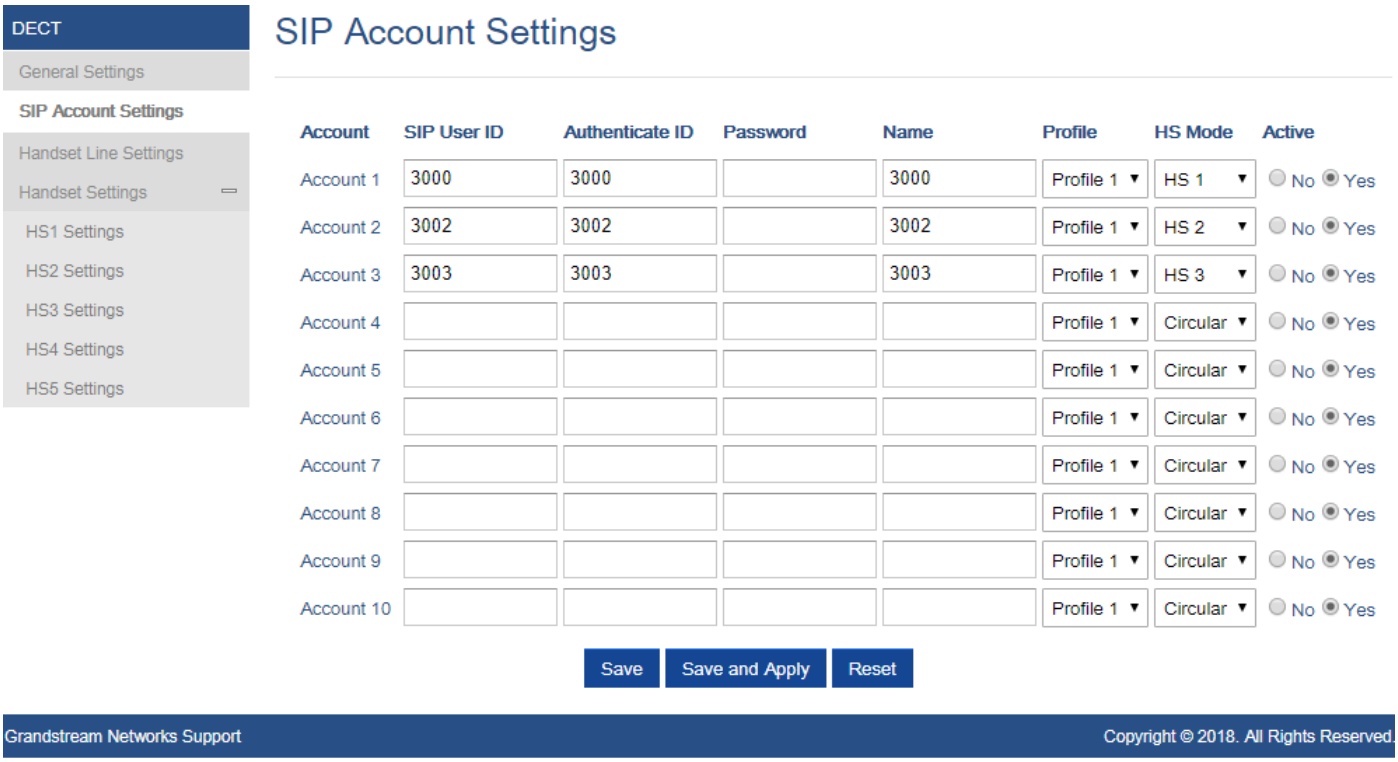


SIP Settings

5. Go to **DECT →SIP Account Settings**
6. Configure your SIP details in desired account:

- **Account:** Select Account row to configure (Account 1 – Account 10).
- **SIP User ID**: User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
- **Authenticate ID**: SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
- **Password**: SIP service subscriber's account password to register to SIP server of ITSP. For security reasons, the password will field will be shown as empty.
- **Name**: Any name to identify this specific user.
- **Profile**: Select the corresponding Profile ID (1/2/3/4).
- **HS Mode**: Assign the account to specific Handsets (HS1, HS2…) or hunting group mode.
- **Active**: Set to **Yes.** If set to **No**, account is disabled and will not register.

7. Press **Save and Apply** to save your configuration.



SIP Accounts Settings

After applying your configuration, your phone will register to your SIP Server.

You can verify if your DECT phone has registered with your SIP server from your DP752 web interface under **Status → Account Status** (a green background with Yes under SIP Registration column for corresponding account indicates the account(s) has been successfully registered).



Account Status

Multiple Lines and Hunting Groups

The DP752 Base Station has the ability to assign 10 lines to each registered DP730/DP722 Handsets (Up to 5 Handsets) to receive/make calls.

When a Handsets has many lines configured, users can select specific line for outgoing calls using **Outgoing Default Line** feature.

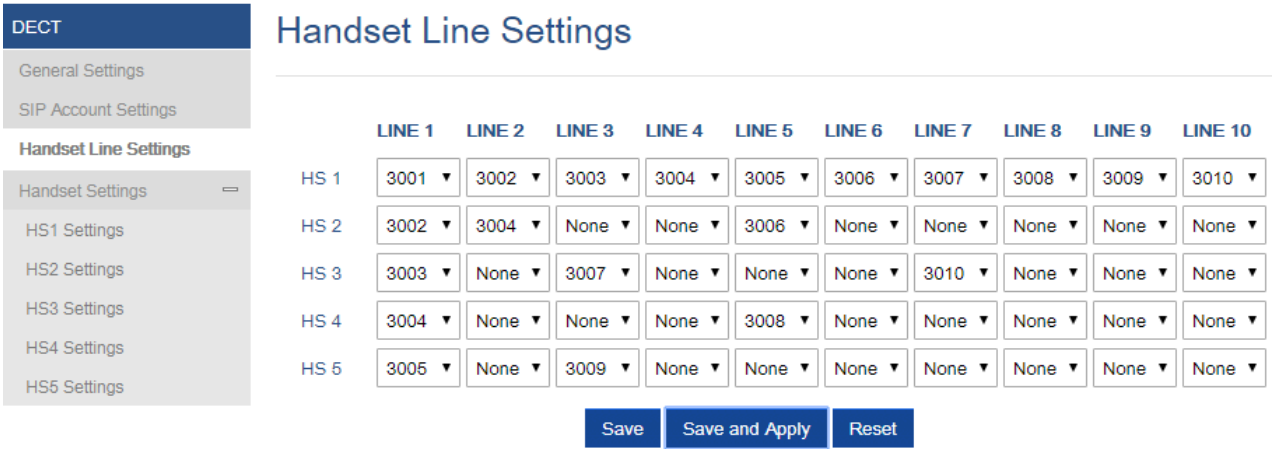
For incoming calls, users can choose either to redirect them to a specific Handsets or to many using Hunting Group feature so to have the same phone number and incoming calls will be distributed in a Linear, Circular or Parallel manner among the Handsets active in that Hunting Group. The number of hunting groups is limited by the number of SIP accounts registered to the base station (up to 10 accounts).

Hunting group feature is mainly used in office, warehouse and call center environments to distribute incoming calls in the best way depending on the type of hunt group.

In order to configure hunting groups for DP730/DP722 Handsets registered to the Base, users need first to register SIP accounts on DP752 Base Station **SIP Account Settings** and then assign accounts accordingly as lines for DP730/DP722 **Handset Line Settings.**

Handsets Line Settings
This section will describe how to assign lines for each DP730/DP722 Handsets for making calls.

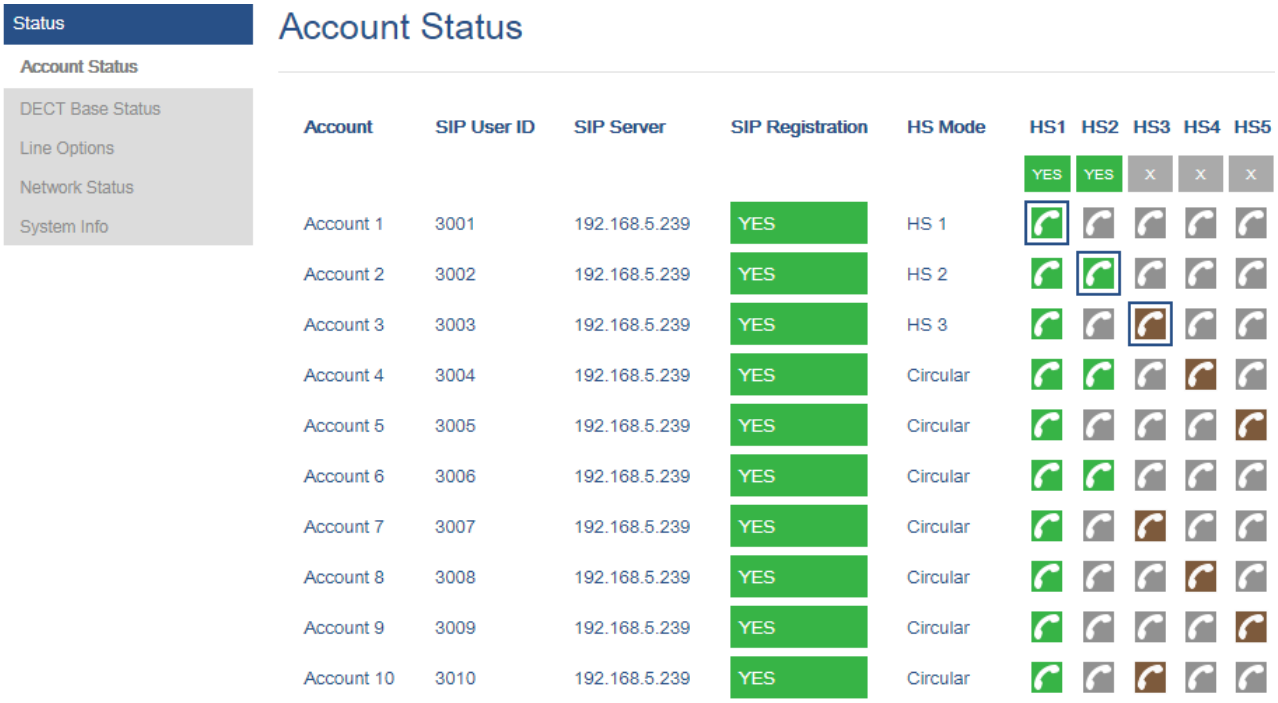
1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **DECT → Handsets Line Settings** and assign to Handsets the SIP accounts already configured. Each Handsets can be configured to use up to 10 SIP accounts.



Handsets Line Settings

After applying your configuration, **Account Status** page will display the status of Handsets along with accounts status. Each column shows one HS; each row shows if the account is assigned to a HS.

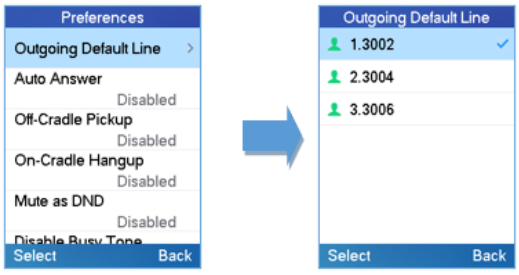
For example: If the account 2, 4 and 6 are assigned to the HS2, the column of HS2 will have cells 2, 4 and 6 in green background, and account 1, 3, 5, 7, 8, 9 and 10 are in gray background. If a line is being used it will be blinking in a green / white background.



Account Status – Line Status

Outgoing Default Line

When a Handsets is configured with more than one line, users can change the default outgoing line on DP730/DP722 Handsets using keypad **Menu → Preferences → Outgoing Default Line**.



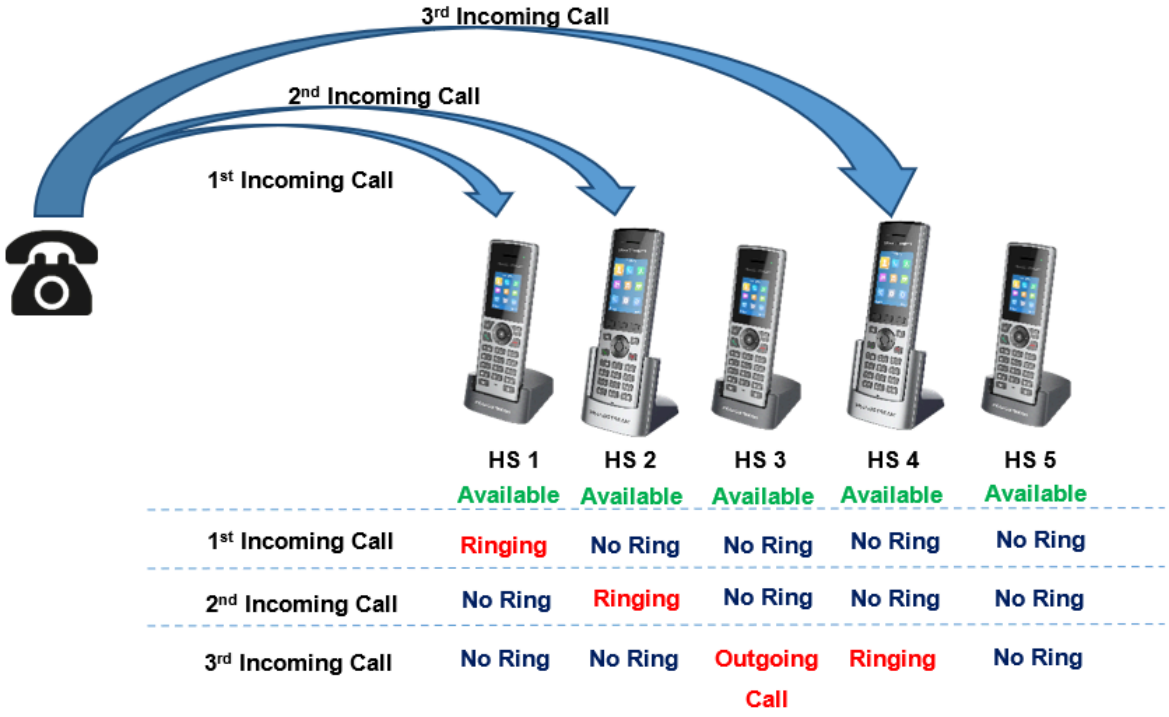
Outgoing Default Line

Hunting Groups

DP752 supports 3 types of hunting groups as described below:
In the examples below, we consider that all Handsets are in same hunting group and only the type of hunting group differs.

- **Circular:** The base station will note which Handsets answered the last call, and forward the call to the next available Handsets in round-robin fashion.

In the example below; the 1st incoming call was sent to HS1 since all Handsets were available; 2nd incoming call will be sent to HS2 since previous call was answered by HS1 and HS2 is the next available Handsets in the list; 3rd will be sent to HS4 since HS2 was the last one answering the call and HS3 is busy making an outgoing call (if HS3 was available, the call will be sent to HS3).

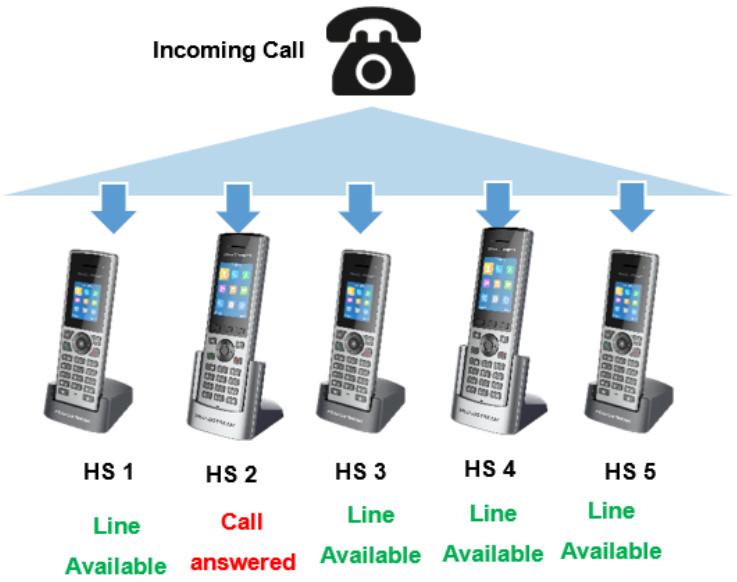


- **Linear:** The base station will distribute calls in predefined order from the lowest-numbered available Handsets, if no answer within ring timeout, the call will be sent to next available Handsets in sequence. This mode is also called "serial hunting".

In the example below; all Handsets are available, the incoming call will be sent to HS1 since it's the lowest-numbered available Handsets, if no answer within ring timeout, the call will be sent to HS2 since it's the next available lowest-numbered Handsets, and same applies for HS3 and etc.



- **Parallel:** In this mode, all phones ring concurrently. If one phone answers, the remaining available phones can make new outgoing calls.



This section will describe how to configure hunting groups for incoming calls:

Below steps are considering that SIP accounts were previously registered.

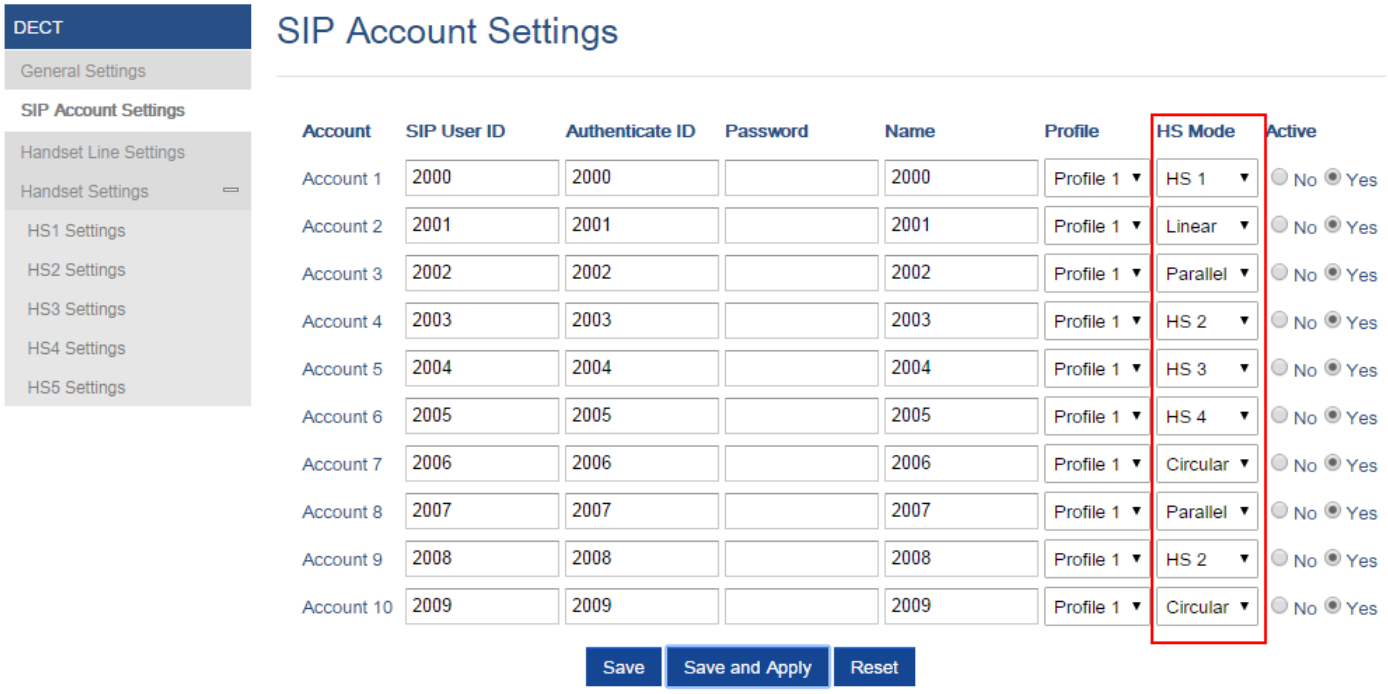
1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **DECT → SIP Account Settings.**
4. Set **HS Mode** depending on your needs to configure your hunting groups.
5. Press **Save and Apply** to save your settings.

Note

If **HS Mode** is set to **HS**, the handset can receive simultaneous calls when in ringing state.

Example:

In the example below Account 2 (3003) is assigned to HS1 and HS2, and the hunting group (HS Mode) is set to Linear, so incoming calls to that account will make HS1 ringing first then if no answer within ring timeout the HS2 will start ringing.



Hunting Group configuration



Hunting Group Status

Configuration via Keypad

To configure the LCD menu using DP730/DP722's keypad, follow the instructions below:

- Register the DP730/DP722 to DP752. Please see Register DP730/DP722 Handsets to DP752 Base Station;
- **Enter/Confirm/ selection:** Press the left softkey, right softkey, on-hook key or OK/Select key to enter the selected option, back to last layer or exit;
- **Exit:** Press "right softkey" to exit to the previous menu;
- **Return to Home page:** Press "On-hook" key to exit to the main menu.
- The DP730/DP722 automatically exits to main mode with an incoming call, when the phone is off hook or left idle for more than 20 seconds.
- When the phone is in idle, pressing the DOWN navigation key can enter the **Outgoing call log**.

Please refer to DP730/DP722 Handsets Menu Structure for more details.

Call Features

The DP752/DP730/DP722 supports traditional and advanced telephony features including caller ID, caller ID with caller Name, call forward and etc.

***30**	**Block Caller ID (for all subsequent calls)** ○ Off hook the phone; ○ Dial *30.
***31**	**Send Caller ID (for all subsequent calls)** ○ Off hook the phone; ○ Dial *31.
***67**	**Call with Caller ID Blocked (per call)** ○ Off hook the phone; ○ Dial *67 and then enter the number to dial out.

***82**	**Call with Caller ID Enabled (per call)** ○ Off hook the phone; ○ Dial *82 and then enter the number to dial out.
***72**	**Unconditional Call Forward**. To set up unconditional call forward: ○ Off hook the phone; ○ Dial *72 and then enter the number to forward the call; ○ Press OK softkey or SEND key.
***73**	**Cancel Unconditional Call Forward**. To cancel the unconditional call forward: ○ Off hook the phone; ○ Dial *73;
***90**	**Busy Call Forward**. To set up busy call forward: ○ Off hook the phone; ○ Dial *90 and then enter the number to forward the call; ○ Press OK softkey or SEND key.
***91**	**Cancel Busy Call Forward**. To cancel the busy call forward: ○ Off hook the phone; ○ Dial *91;
***92**	**Delayed Call Forward**. To set up delayed call forward: ○ Off hook the phone; ○ Dial *92 and then enter the number to forward the call; ○ Press OK softkey or SEND key.
***93**	**Cancel Delayed Call Forward**. To cancel the delayed call forward: ○ Off hook the phone; ○ Dial *93;

Call Features

DP752 Phonebook Management

DP752/DP730/DP722 support Private and Global Phonebooks; both phonebook types can be used at same time:

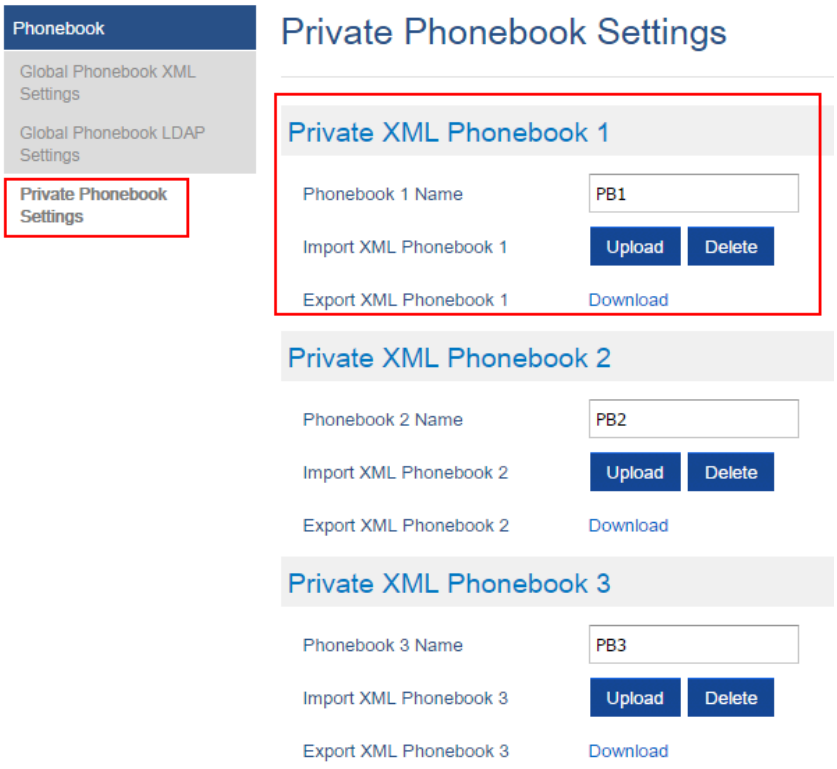
Private Phonebook

Private phonebook allows you to manage your contacts on each registered Handsets; each Handsets can have his own private phonebook with his own contacts. DP752 supports up to 5 private phonebooks.

A private phonebook can be assigned to one or more Handsets registered to the base.

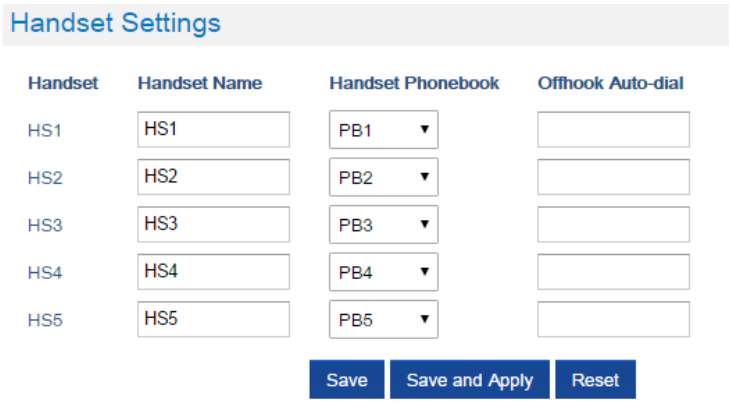
The following steps explain how upload your private phonebook and assign it to a specific Handsets:

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings and go to **Phonebook → Private Phonebook Settings.**



Private Phonebook Settings

3. In **Private XML Phonebook X** section (X from 1 to 5):
    1. Enter **Phonebook X Name** (default value is PB1 for first Handset, PB2 for second Handset, etc.).
    2. Press **Upload** button to **Import XML Phonebook X.**
    3. Browse your computer files and select your desired **phonebook.xml** file.
    4. Press **Save and Apply** to save your settings.
4. Go to **DECT → General Settings** tab.
5. In **Handsets Settings** section, select your **Handsets Phonebook** to assign it to a specific Handsets as shown below where **PB1** is assigned to **HS1**, **PB2** is assigned to **HS2**...

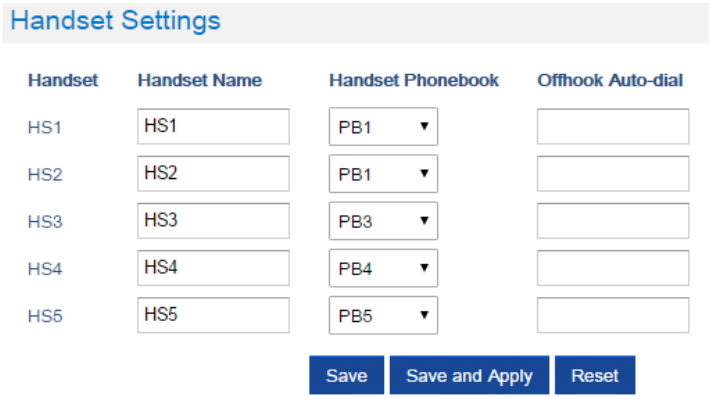


Handset Phonebook Settings

You can assign same Private Phonebook to more than one Handsets.

For example, we can assign **Handsets Phonebook** named **PB1** to **HS1** and **HS2.**

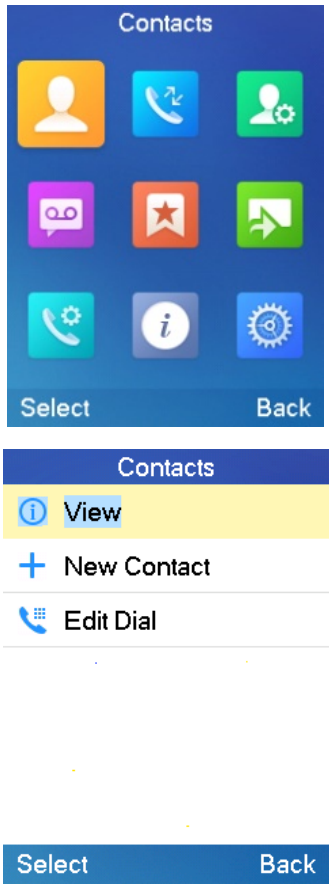
Any change in **PB1** contacts will be applied to both **HS1** and **HS2** private phonebooks.



Handsets Phonebook Selection

6. Press **Save and Apply** to save your configuration.

After applying your configuration, your DP730/DP722 Handsets will display uploaded phonebook contacts. You can access your private phonebook by pressing **Contacts** on your DP730/DP722 Handsets. You can press Option Softkey in order to view, Create New Contact or Edit Dial if adding changes on a contact number is required before dialing.



Contacts

Global Phonebook

Global phonebook allows to manage contacts and use them in all registered Handsets. The contacts can be imported either via XML or via LDAP. Follow steps below to upload your shared phonebook:

Global Phonebook via XML

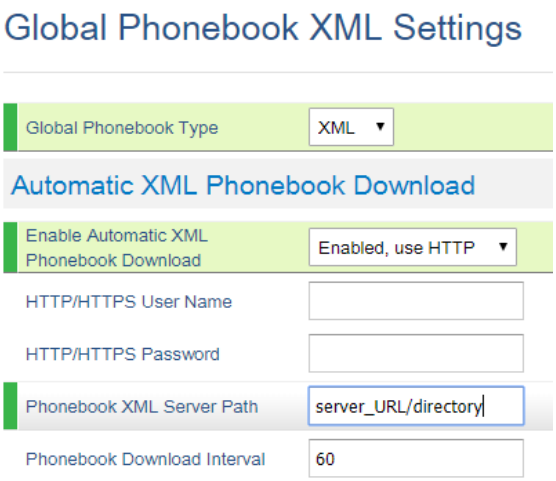
1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **Phonebook → Global Phonebook XML Settings** tab.
4. Set **Global Phonebook Type** to **XML** (in this case, LDAP phonebook will not be available).



Global Phonebook XML Settings

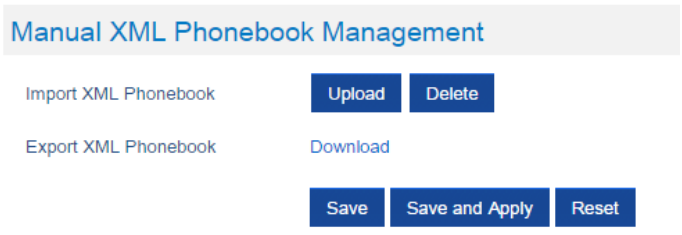
5. There are two methods to import/download your XML Phonebook:

- **Automatic XML Phonebook Download**
  For this method, you need to use a TFTP or HTTP or HTTPS server and make your *phonebook.xml* file is available in your preferred server.
  1. Set **Enable Automatic XML Phonebook Download** to **Enabled, use TFTP/HTTP or HTTPS** depending on your server.
  2. If using HTTP or HTTPS server and User Name and Password are required to connect to the server, set **HTTP/HTTPS User Name** and **HTTP/HTTPS Password** fields with appropriate values.
  3. Configure **Phonebook XML Server Path** field. This field could be IP address or URL, with up to 256 characters. The phone will request file named **phonebook.xml** from provided directory. Example: *server_URL/directory*
  4. Configure the **Phonebook Download Interval** (in minutes) to periodically contact your server to download new phonebook file version if available. If set to 0, automatic download will be disabled. Valid range is 5 to 720.



Automatic XML Phonebook Download

- **Manual XML Phonebook Management**
  1. Press **Upload** in **Import XML Phonebook.**
  2. Browse your files and select your **phonebook.xml** file.



Manual XML Phonebook Management

XML Phonebook file format

```
<?xml version="1.0" encoding="UTF-8"?>
<AddressBook>
    <Contact>
        <FirstName>First name</FirstName>
        <LastName>Last name</LastName>
        <Ringtone>Ringtone ID (default 0)</Ringtone>
        <Phone type="Home">
            <phonenumber>Home phone number</phonenumber>
        </Phone>
        <Phone type="Work">
            <phonenumber>Work phone number</phonenumber>
        </Phone>
        <Phone type="Mobile">
            <phonenumber>Mobile phone number</phonenumber>
        </Phone>
    </Contact>
</AddressBook>
```

Object	Position	Type	Values	Comments
AddressBook	Root element	Mandatory	–	Root element of the XML document
Contact	Child element	Mandatory	–	Each contact is an entry
LastName	Child element	At least one of them present	String	Last name of the contact
FirstName	Child element	String	First name of the contact	
Phone	Child element	Mandatory	–	Phone number
PhoneNumber	Child element	At least one present	Int	Type=”Home” or Type=”Work” or Type=”Mobile”

XML Phonebook Example:

```
<?xml version="1.0" encoding="UTF-8"?>
<AddressBook>
    <Contact>
        <FirstName>John</FirstName>
        <LastName>Doe</LastName>
        <Ringtone>0</Ringtone>
        <Phone type="Home">
            <phonenumber>1000</phonenumber>
        </Phone>
        <Phone type="Work">
            <phonenumber>1001</phonenumber>
        </Phone>
        <Phone type="Mobile">
            <phonenumber>1002</phonenumber>
        </Phone>
    </Contact>
    <Contact>
        <FirstName>Alice</FirstName>
        <LastName>Beck</LastName>
        <Ringtone>0</Ringtone>
        <Phone type="Home">
            <phonenumber>2000</phonenumber>
        </Phone>
        <Phone type="Work">
            <phonenumber>2001</phonenumber>
        </Phone>
        <Phone type="Mobile">
            <phonenumber>2002</phonenumber>
        </Phone>
    </Contact>
 </AddressBook>
```

Global Phonebook via LDAP

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Press **Login** to access your settings.
3. Go to **Phonebook → Global Phonebook LDAP Settings** tab.
4. Set **Global Phonebook Type** to **LDAP** (in this case, XML phonebook will not be available).



Global Phonebook LDAP Settings

5. Under **LDAP Phonebook Settings**, set your LDAP parameters to connect to your LDAP server.
Refer to [Phonebook Page Definitions] for parameters explanation.

6. Press **Save and Apply** to save your configuration.

- **Example of configuration:**

```
LDAP protocol: LDAP
Server Address:  192.168.1.100
Port: 389
Base: dc=pbx,dc=com
User Name:
Password:
LDAP Filter: (mobile=%)(sn=%)
LDAP Version: Version 3
First Name Attribute: sn
Last Name Attribute: cn
Work Name Attribute:
Home Name Attribute:
Mobile Number Attribute: mobile
Max. Hits: 500
Search Timeout: 30
```

After applying your configuration, your global phonebook will be synchronized with all registered Handsets and contacts will be displayed on your DP730/DP722 Handsets screens.

DP752 ASSOCIATION WITH DP760 DECT REPEATER

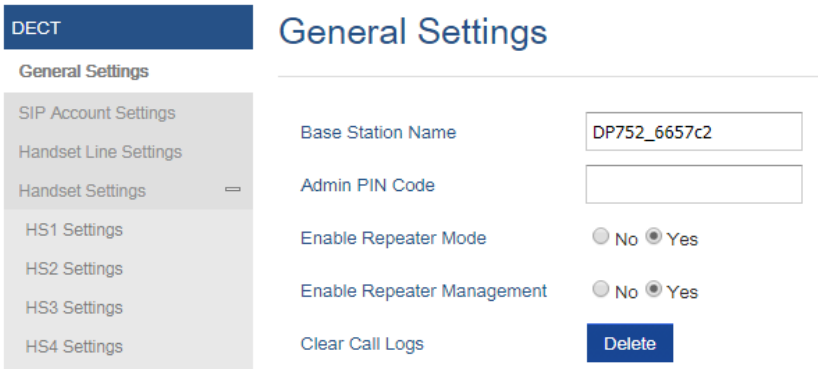
Important Note

- DP760 can relay up to 2 concurrent calls.
- After pressing Page/Reset button for more than 2 seconds on the DP760, it will enter AUTO region mode, in this mode, the three LEDs of right side keep quick blinking, then the DP760 will search the base signal in the current environment to auto associate with it and then auto switch to the same region (EU, US or BR) of the base station.
- 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 and it will enter Auto Region mode.

Enabling Repeater Mode on DP752

Before associating the DP760 DECT Repeater to your DP752 Base Station, you should firstly enable the repeater mode on your base station. Please refer following steps to enable the repeater mode on the DP752 base station:

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Navigate to **DECT → General Settings** and set **Enable Repeater Mode** to **Yes**.



Enable Repeater Mode

3. Press **Save and Apply**, then **Reboot** the base to apply the new settings.

DP760 DECT Repeater Association

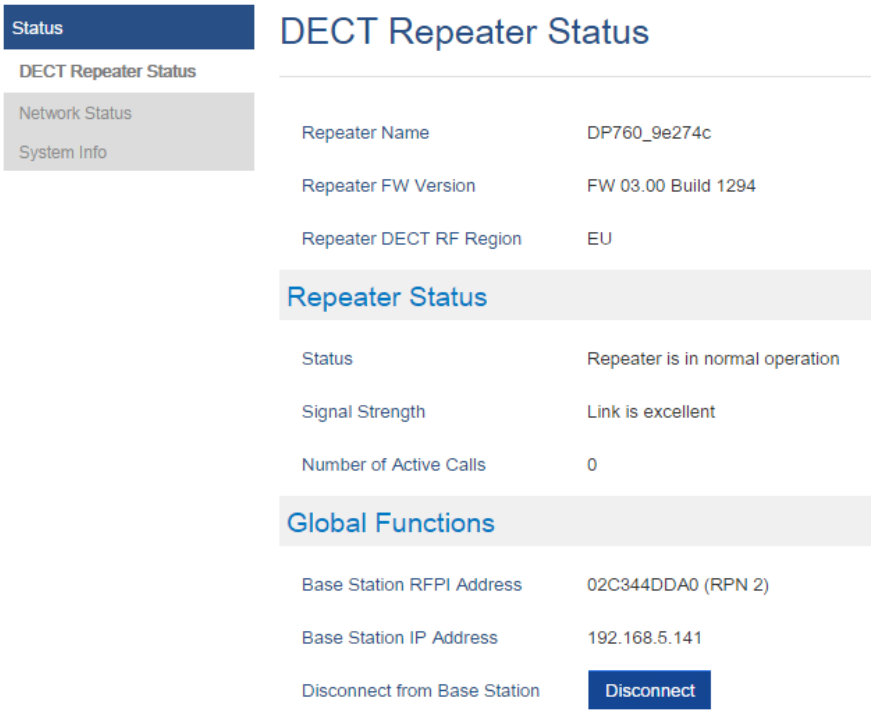
After enabling the repeater mode on DP752 Base Station, you can easily associate it with your DP760 DECT Repeater using one of the following methods:

- **Auto Association**
- **Manual Association**
- **Use DP752 Repeater Management to associate DP760**

Auto Association

To auto associate your DP760 DECT Repeater with DP752 Base Station:

1. Power on the DP760 DECT Repeater.
2. After the DP760 finish booting, the association LED will be blinking to indicate that the repeater is searching for nearby base stations.
   Note: If all the LEDs remain on, it indicates that the DP760 Repeater has successfully associated with a DP752 Base and ready for use. You may need to dissociate it before continuing (Please refer to [DP760 DECT REPEATER DISSOCIATION]).
3. Open subscription on your DP752 Base Station using one of following methods:
   1. Rebooting the Base station.
   2. Long press on page/subscribe  button on DP752 for 4 seconds.
   3. Access DP752 Web GUI and press subscribe button .
4. The DP760 DECT Repeater will automatically pair to DP752. Upon successful association with the DP752 Base station all LEDs will turn ON and the repeater status will be updated as following.



Repeater Status – Associated (Auto Association)

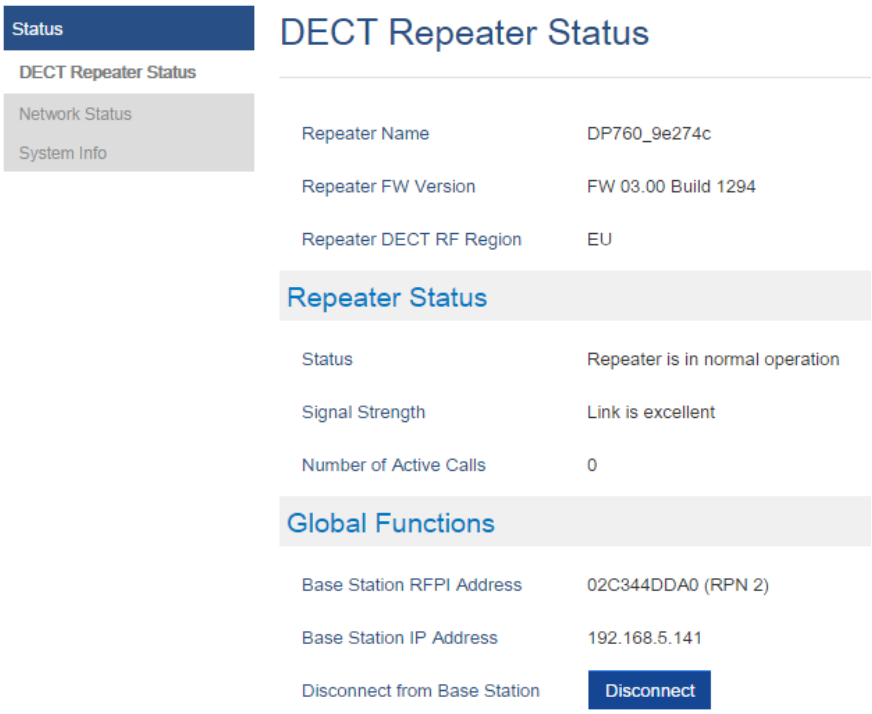
Notes

- For auto association, the option **Use manually configured RFPI** under DP760 Web GUI should be disabled (by default disabled).
- The RF auto association requires the repeater to be close to the base during the pairing process to work properly (this is a standard DECT repeater pairing limitation).

Manual Association

To manually associate your DP760 DECT Repeater to your DP752 Base Station:

1. Access the Web GUI of your DP760 using the admin's username and password.
2. Navigate to **DECT → General Settings** and set **Use Manually Configured RFPI** to **Yes**.
3. On **Manually Configured Base RFPI** field, configure your DP752 Base Station RFPI address.
   Note: The RFPI Address of your DP752 Base Station is available under the Web GUI → **Status → DECT Status → Base DECT RFPI Address**.
4. Click **Save and Apply** to confirm the new settings.
5. Open subscription on your DP752 Base Station using one of following methods:
   - Rebooting the Base station
   - Long press on page/subscribe  button of DP752 for 7 seconds
   - Access DP752 Web GUI and press subscribe button .
6. The DP760 DECT Repeater will automatically pair to DP752. If the association is correctly associated, all the LEDs will remain solid on and the repeater status will be updated as following.



Repeater Status – Associated (Manual Association)

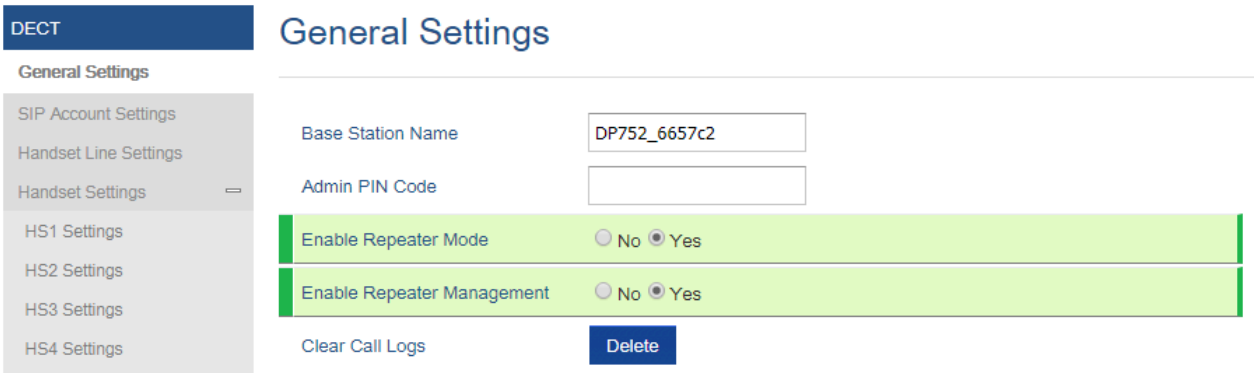
> **Note**
>
> Make sure that **Repeater Mode** is enabled on your DP752 Base Station.

Use DP752 Repeater Management to Associate DP760

- **Enabling Repeater Management Mode**

The following steps illustrate how to enable and use the Repeater Management mode on DP752:

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Navigate to **DECT → General Settings** and set **Enable Repeater Management** to **Yes**.
3. Press **Save and Apply**, then reboot the unit to apply the new settings.



Repeater Associated Status

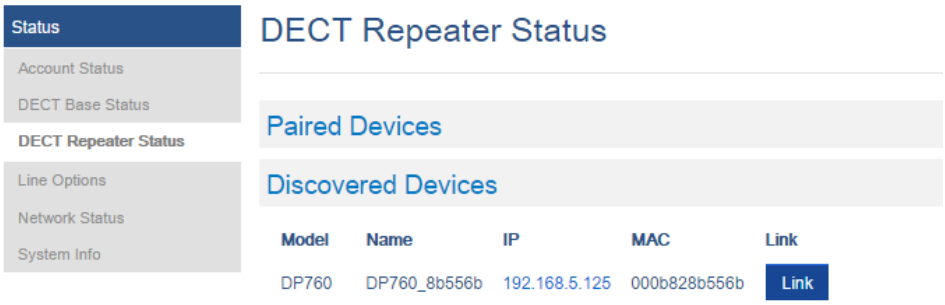
- **Associating DP760 using Repeater Management Mode on DP752**

The repeater management mode displays discovered and paired devices using their name, IP and Mac addresses and also gives the possibility of associating the DP752 Base Station with DP760 DECT Repeater.

Please refer to following steps in order to associate DP760 with DP752 using repeater management mode:

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Navigate to **Status → DECT Repeater Status** and select the repeater station from **Discovered Devices**.
3. Clink **Link** button to associate the base and repeater stations.

Note: The DP760 Repeater stations associated with DP752 Base Station are displayed and available under **Status → DECT Repeater Status → Paired Devices** with their status, signal strength, number of active calls and possibility to disconnect them from the paired base station.



DECT Repeater Status on DP752

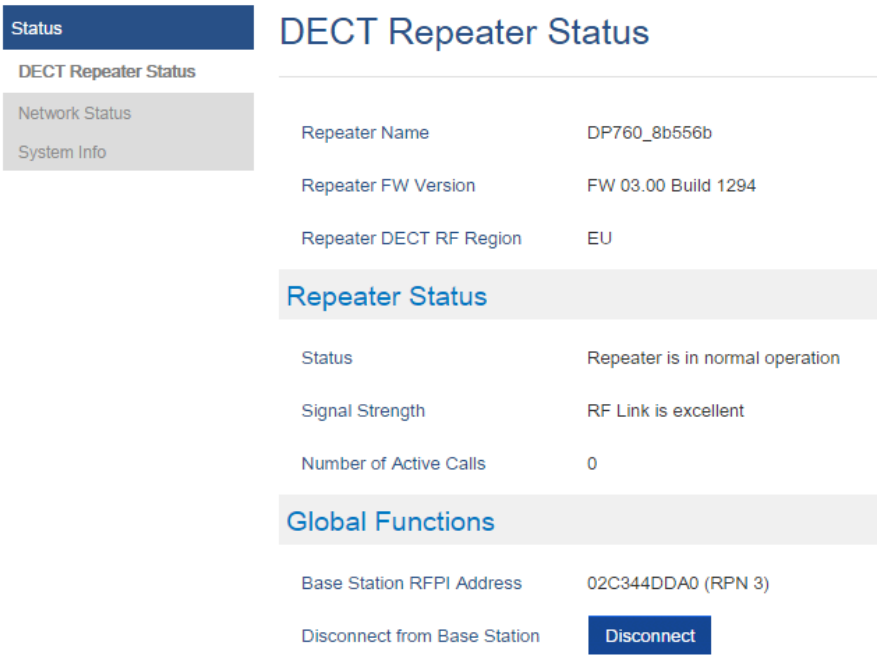
DP760 DECT REPEATER DISSOCIATION FROM DP752

To disconnect DP760 DECT Repeater from the Base Station DP752, please refer to following options:

Using DP760 Web GUI

To disconnect your DP760 DECT Repeater via Web GUI, please refer to following steps:

1. Access the Web GUI of your DP760 using the admin's username and password.
2. Enter the administrator's username and password to access the Web Configuration Menu.
3. Navigate to **Status → DECT Status → Global Functions** and press **Disconnect**.
4. The DP760 DECT Repeater will be automatically disconnected from the DP752 and it will start again scanning for available RFPs.



DECT Repeater Status on DP752

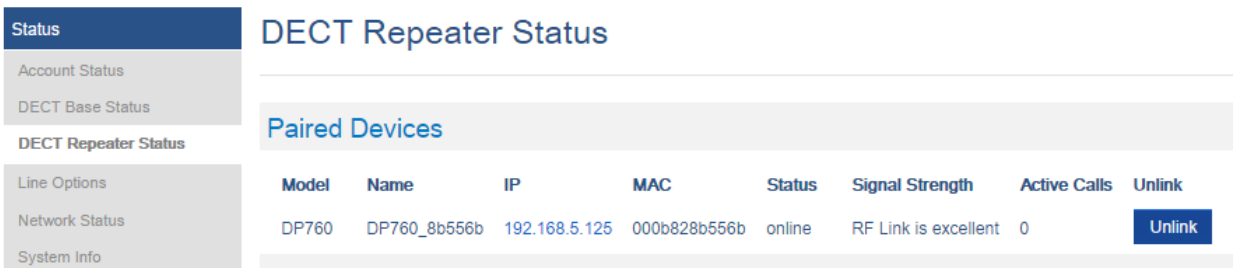
Using Page Button

Long press on page button on the back panel of DP760 for 7 seconds, it will remove the association values of the previous base station.

Using Repeater Management Mode on DP752

To disconnect your DP760 DECT Repeater using the repeater management mode on DP752:

1. Access the Web GUI of your DP752 using the admin's username and password.
2. Navigate to **Status → DECT Repeater Status**.
3. Select the repeater station from **Discovered Devices**.
4. Clink **Unlink** button to dissociate the repeater from the base station.



DECT Repeater Status on DP752

UPGRADING AND PROVISIONING

The DP752 can be upgraded via FTP/FTPS/HTTP/HTTPS by configuring the URL/IP Address for the FTP/FTPS/HTTP/HTTPS server and selecting a download method. Configure a valid URL for FTP/FTPS or HTTP/HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:

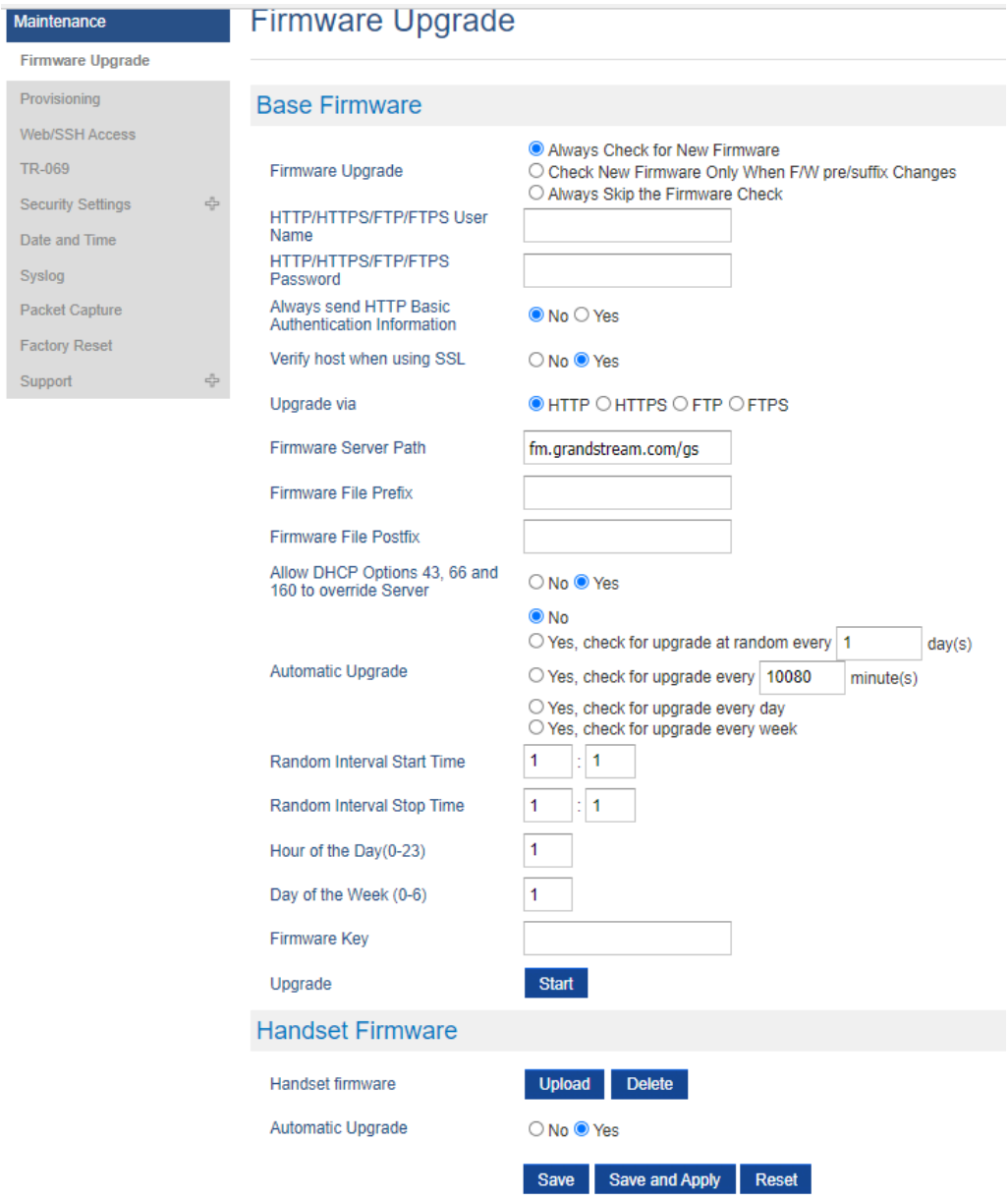
firmware.grandstream.com

fw.ipvideotalk.com/gs

DP752 Firmware Upgrade Procedure

Please follow below steps in order to upgrade the firmware version of your DP752 base station:

1. Access DP752 Web GUI.
2. Press **Login** to access your settings.
3. Go to **Maintenance → Firmware Upgrade** page, enter the IP address or the FQDN for the upgrade server in "**Firmware Server Path**" field and choose to upgrade via FTP/FTPS or HTTP/HTTPS.
4. Make sure to configure Firmware **Upgrade and Provisioning** to **Always Check for New Firmware.**
5. Update the change by clicking the "**Save and Apply**" button. Then "**Reboot**" or power cycle the base station to update the new firmware.



Firmware Upgrade Page

Upgrading via Local FTP/HTTP Servers

For users that would like to use remote upgrading without a local FTP/HTTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their devices via this server. Please refer to the webpage: http://www.grandstream.com/support/firmware

Alternatively, users can download a free FTP or HTTP server and conduct a local firmware upgrade. A free windows version FTP server is available for download from:

https://filezilla-project.org/download.php?type=server

Instructions for local firmware upgrade via FTP:

1. Unzip the firmware files and put all of them in the root directory of the FTP server.
2. Connect the PC running the FTP server and the phone to the same LAN segment.
3. Start the FTP server and configure the FTP server in the phone's web configuration interface.
4. Configure the Firmware Server Path to the IP address of the PC.
5. Configure the Admin and password authentication Credentials set on your FTP server.
6. Save and apply the changes and reboot the base station.

End users can also choose to download a free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

Upgrading DP730/DP722 Handsets

User could upgrade their Handsets either using the LCD menu or via the Web GUI

Using the LCD menu

1. On DP730/DP722, press "Menu" (left softkey or the selection key) to bring up operation menu.
2. Press arrow keys to move the cursor to **Settings** and then press "Select"**.**
3. Navigate to **Firmware Upgrade** and press "Select" (left softkey), the Handsets will upgrade the firmware available on the base station.

Using the Web GUI

1. Access DP752 Web GUI.
2. Press **Login** to access your settings.
3. Go to **Maintenance → Firmware Upgrade** page, and enable the **Automatic Upgrade** option so the Handsets will upgrade the firmware automatically once it detects the new firmware.



Handset Firmware Upgrade

4. Update the change by clicking the "**Save and Apply**" button.

> **Notes**
>
> - The DP730/DP722 Handsets detects automatically if the base station is upgraded with a new firmware and display on the LCD a prompt message inviting the user to initiate the upgrade, more over under the Web GUI of the base station DP752 → Status → DECT Status → Handsets Status, appears an **Upgrade All** button once the base detect that several registered Handsets are using an old firmware version so the user could upgrade all Handsets by pressing **Upgrade All** button.
> - After downloading firmware, DP730/DP722 screen will be black and red LED will be blinking showing that firmware upgrade process is ongoing.

DP752/DP730/DP722 Provisioning

Configuration File Download

Grandstream SIP Devices can be configured via the Web Interface as well as via a Configuration File (binary or XML) through TFTP/FTP/FTPS or HTTP/HTTPS. The **Config Server Path** is the TFTP/FTP/FTPS or HTTP/HTTPS server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The **Config Server Path** can be the same or different from the **Firmware Server Path**.

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with the "New Password" in the Web GUI→Maintenance→Web/SSH Access page→Admin Password. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

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

```
<!--##  Configuration Template                                          ##-->
<!--#########################################################################-->
<!--#################-->
<!--##  P values  ##-->
<!--#################-->
<!--# Name: 8-->
<!--# Alias: network.type,NETWORK_TYPE # Help: IP Address Mode. 0 - DHCP, 1 - Static IP, 2 - PPPoE.-->
<!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-2] -->
<!--# Default Value: 0-->
    <Pnetwork.type>0</Pnetwork.type>


<!--# Name: 9-->
<!--# Alias: network.static.address.ip.octet.1,NETWORK_STATIC_ADDRESS_OCTET1 # Help: IP Address octet 1. Ignore if DHCP or PPPoE is used.-->
<!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
<!--# Default Value: 192-->
    <Pnetwork.static.address.ip.octet.1>50</Pnetwork.static.address.ip.octet.1>


<!--# Name: 10-->
<!--# Alias: network.static.address.ip.octet.2,NETWORK_STATIC_ADDRESS_OCTET2 # Help: IP Address octet 2. Ignore if DHCP or PPPoE is used.-->
<!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
<!--# Default Value: 168-->
    <Pnetwork.static.address.ip.octet.2>84</Pnetwork.static.address.ip.octet.2>


<!--# Name: 11-->
<!--# Alias: network.static.address.ip.octet.3,NETWORK_STATIC_ADDRESS_OCTET3 # Help: IP Address octet 3. Ignore if DHCP or PPPoE is used.-->
<!--# Attributes: Category=NUMBER Mandatory=1 Base=10 Range=[0-255] -->
<!--# Default Value: 0-->
    <Pnetwork.static.address.ip.octet.3>167</Pnetwork.static.address.ip.octet.3>
```

New P values format

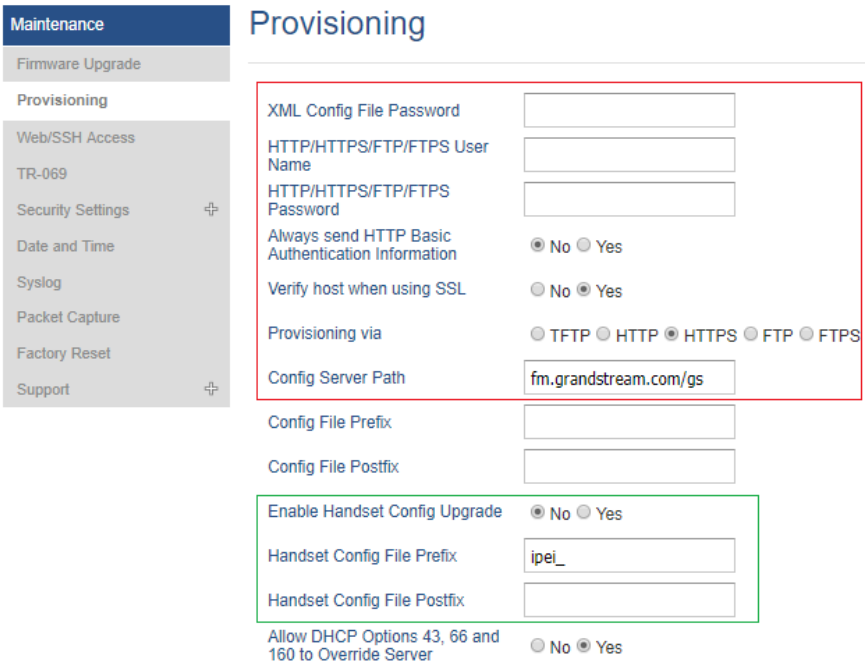
When the DP752 boots up or reboots, it will send a request to download a file named "cfgxxxxxxxxxxx" followed by a configuration XML file named "cfgxxxxxxxxxxx.xml", where "xxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If the download of "cfgxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to:

https://documentation.grandstream.com/knowledge-base/sip-device-provisioning-guide/

Handsets Provisioning

To configure the Handsets provisioning, please browse the base station Web GUI at Maintenance → Provisioning. This is the same page used for base station provisioning. Handsets 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 Handsets Config Upgrade, need reboot the DP752 to take effect.



Provisioning Feature

- **Provisioning via:** Choose the method that the base station uses to request Handsets ipei config file. Can use TFTP, FTP, FTPS, 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 Handsets configuration file with the matching encrypted prefix will be downloaded and flashed into the device.
- **Handset Config File Postfix:** If configured, only the Handsets configuration file with the matching encrypted postfix will be downloaded and flashed into the device

When a DP730/DP722 registers to a DP752 for the first time or an already registered Handsets boots up, the DP752 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 Handsets'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.

For more detailed information, please refer to Handset provisioning Guide.

RESTORE FACTORY DEFAULT SETTINGS

Warning

Restoring the Factory Default Settings will delete all configuration information on the phone. Please backup or print all the settings before you restore to the factory default settings. Grandstream is not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

Resetting the DP752 Base Station

Using Hardware Button

Long press the page button  on the device about 30 seconds, until the device reboot.

Notes

- The device will enter subscribe mode after about 8 seconds first, need keep pressing until factory reset.
- Reset Type can be configured from **[Configure Hardware Button]**.

From Web Interface

1. Access DP752 Web GUI.
2. Navigate to **Maintenance → Factory Reset**.
3. Select the **Reset Type** from the reset drop down list.
4. Press **Reset** button and the unit will prompt a confirmation message, press **OK** to reset, then the unit restart automatically with parameters restored to default values or press **Cancel** to annul the reset.

Resetting the DP730/DP722 Handsets

Please follow the instructions below to reset the DP730/DP722 Handsets:

1. Press "Menu" (left softkey or the selection key) to bring up operation menu.
2. Press arrow keys to move the cursor to **Settings** and then press "Select".
3. Navigate to **System Settings → Factory Reset**, a warning window will pop out to make sure a reset is requested and confirmed;
4. Press "**Yes**" (left softkey) to confirm and the phone will reboot.

To cancel the Reset, press "**No**" (right softkey) instead.

Users also can perform factory reset using combo keys on DP730/DP722 during booting up prior it's fully booted, to give DP730/DP722 the ability to recover from some extreme cases like self-reboot or stuck right after booted up.

Please refer to following procedure for DP730/DP722 factory reset using combo key:

1. Connect the USB cable and power off DP730/DP722.
2. Power on DP730/DP722 and wait for Green LED.
3. Press 3 times Central Key (OK/Selection Key) quickly after Green LED on.
4. Press Speaker Key  first and then Mute Key  after Red LED on.
5. DP730/DP722 LED will start blinking in 3 colors: Green/Red if factory reset successful triggered.
6. Wait for about 1-3 minutes, the DP730/DP722 will boot up with setup wizard for initial language and date format settings.

CHANGE LOG

This section documents significant changes from previous firmware versions. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.21.16

- Added SN information to the Web UI. [Serial Number]

Firmware Version 1.0.21.15

- Added the ability to handle simultaneous incoming calls to Broadworks HuntGroup while the handset is at ringing state. [Hunting Groups]
- Added CPE version in the system info page. [CPE]
- Added the option "Return Code When Enable DND". [Return Code When Enable DND]
- Added the option to define the User-Agent. [SIP User-Agent]
- Added client key password field for OpenVPN settings. [OpenVPN® Client Key Password]
- Added the option "Enable Mute as DND. [Enable Mute as DND]

Firmware Version 1.0.21.12

- No Major changes

Firmware Version 1.0.21.10

- Added prompt to change the default password upon first login. [Web UI Access Level Management]
- Disabled non-admin account by default. [Enable User Web Access]

Firmware Version 1.0.21.6

- Added support for Action URL. [Action URL]

Firmware Version 1.0.21.4

- No Major changes

Firmware Version 1.0.21.2

- Added support to unregister SIP Instance ID only. [Instance ID]
- Added Pin for GDS to open 2nd Door [Access Password 2]

Firmware Version 1.0.19.8

- Added "Enable DND Feature" [Enable DND feature]
- Added ability to use "_" (underscore)in the SIP Server. [SIP Server]
- Enhanced security with SSL Medium Strength Cipher Suites Supported.[Enable/Disable Weak Ciphers]
- Added not failover/failback if current SBC has same priority as top choice on same SRV result. [Failback follows failback expiration timer]
- Added allow failback timer setting of 0 to disable failback mode (only necessary when using Telstra Special Feature) [Failback Expiration (m)]

Firmware Version 1.0.19.6

- Add SRTP mode "Optional" under SRTP options. [SRTP Mode]
- Added "Kandy" Special Feature mode for ITSP. [Special Feature]

Firmware Version 1.0.19.5

- Added support for "Allow SIP Reset". [ALLOW SIP RESET]

rmware Version 1.0.19.4

- Added ability to handle all SRV responses when using NAPTR/SRV as DNS mode for Telstra ITSP. [Special Feature]

Firmware Version 1.0.19.3

- Added support for SSL Key Log File. [WITH SECRET KEY INFORMATION]
- Added support for Alert-info in string format. [Alert Info]

Firmware Version 1.0.19.2

- Added support for XMPP connection on TR-069. [XMPP]
- Added option to use # as Redial key. [Redial key]
- Added preferred primary outbound proxy feature. [Outbound Proxy]
- Integrated new DigiCert certificates in firmware. [DigiCert certificates]
- Added new option "Follow SIP Transport to SRTP mode" to auto-switch between RTP and SRTP according to the resolved SIP transport. [Follow SIP Transport]
- Added the ability to configure F-timer and B-timer. [F-Timer] [B-Timer]
- 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" to "No". [Unregister on Reboot]
- Added support for the Greek language.

Firmware Version 1.0.17.8

- No major changes.

Firmware Version 1.0.17.4

- No major changes.

Firmware Version 1.0.17.3

- Added support for Off-Hook Auto-Dial Delay. [DECT Page Definitions]

Firmware Version 1.0.17.1

- Add support to automatically reboot. [Automatic Reboot]

Firmware Version 1.0.15.15

- Added support to allow call barging by Call-Info/Alert-Info. [Allow Barging for Auto Answer by Call-Info/Alert-Info]
- Added HW version display on device web GUI status page. [Hardware Version]
- Adjust Opus to be configured with default value 123 as the same as GRP. [OPUS Payload Type]

Firmware Version 1.0.15.6

- No major changes.

Firmware Version 1.0.15.3

- Added support to have a web UI option of bypass dial plan. [Bypass Dial plan]

Firmware Version 1.0.15.2

- Add 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. [User Name] [Password].
- Added support to configure Call Log settings. [Call Log]
- Added DNS SRV Failover modes. [DNS SRV Failover mode]
- Added warning tone when auto answer a call or intercom. [Play warning tone for Auto Answer Intercom]
- Added option to automatically mute after auto answer a call or intercom. [Enable Mute for Auto Answer]
- Added support to disable SIP NOTIFY authentication. [Disable SIP NOTIFY Authentication]
- Added support for Auto Timer Zone feature. [Time Zone]
- Added support to delete a paired base station from the handset.

Firmware Version 1.0.13.8

- Added support to enable or disable “On Hold Reminder Tone” in Web UI. [On Hold Reminder Tone]
- Added support for handsets to internally page other handsets without accounts registered. [Paging]
- Added ability to configure Caller ID Display. [Caller ID Display]
- Updated the CPE to 1.0.1.141 to better support GDMS.

Firmware Version 1.0.13.0

- No major changes.

Firmware Version 1.0.11.12

- Added support for SNMP. [SNMP Settings]
- Added support for management interface settings that uses a virtual network interface. [Management Interface Settings]
- Added support for Broadsoft XSI phonebook. [Broadsoft XSI]
- 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 option “Backup Outbound Proxy” in all Profiles. [Backup Outbound Proxy]
- Added support for option “Use SBC” in all Profiles. [Use SBC]
- Added option to use sip or sips for SIP URI scheme when using TLS. [SIP URI Scheme When Using TLS]

- 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. [Web Access* Mode]
- Updated the CPE to better support GDMS.

Firmware Version 1.0.11.4

- No major changes.

Firmware Version 1.0.11.2

- Added DP75X Callee Codec Negotiation Priority option. [Callee Codec Negotiation Priority]
- Added Chile time zone. [Time Zone]
- Added support to bypass the dial plan check when redial / call back phone number via call history.[Dial Plan Bypass]
- Changed SIP account name length from 14 characters to 256. [Name]

Firmware version 1.0.9.9

- 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. [Custom Time Zone Settings]
- Added support to configure Voice Mail access number on DP75X web UI. [Voice Mail Access Number]
- Increased DP75X web UI voice mail user ID character length to 30. [Voice Mail Access Number]
- Increased SIP Certificate length limitation from 2048 characters to 8192. [SIP TLS]
- Added support for FTP/FTPS User Name/Password for firmware upgrade and provisioning. [Firmware Upgrade via]
- 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.

Firmware version 1.0.9.1

- Added option to enable or disable Session Timer for DP75x. [Enable Session Timer]
- Added option to enable or disable DP730 keypad backlight. [Display]
- Added support of EXTRA DEBUG level Syslog for DP75X. [Syslog Level]
- Add ability to manually configure date and time on the base Web GUI. [Manual Time Settings]
- Added support to enable or disable “Allow Unsolicited REFER” option for DP750/DP752. [Allow Unsolicited REFER]
- Change DP730 Preferences option “Mute as DND” default to “Enabled” from “Disabled” to make enabling DND more convenient. [Mute as DND]
- Added support for Intercom feature. [Shortcut]

Firmware version 1.0.7.5

- This is the initial version for DP752/DP730/DP722.

Need Support?

Can’t find the answer you’re looking for? Don’t worry we’re here to help!

CONTACT SUPPORT