Safety Notices

1. Please use the specified power adapter. If you need to use the power adapter provided by other manufacturers under special circumstances, please make sure that the voltage and current provided is in accordance with the requirements of this product, meanwhile, please use the safety certificated products, otherwise may cause fire or get an electric shock.
2. When using this product, please do not damage the power cord either by forcefully twist it, stretch pull, banding or put it under heavy pressure or between items, otherwise it may cause damage to the power cord, lead to fire or get an electric shock.
3. Before using, please confirm that the temperature and environment is humidity suitable for the product to work. (Move the product from air conditioning room to natural temperature, which may cause this product surface or internal components produce condense water vapor, please open power use it after waiting for this product is natural drying).
4. Please do not let non-technical staff to remove or repair. Improper repair may cause electric shock, fire, malfunction, etc. It will lead to injury accident or cause damage to your product.
5. Do not use fingers, pins, wire, other metal objects or foreign body into
   the vents and gaps. It may cause current through the metal or foreign body, which may even cause electric shock or injury accident. If any foreign body or objection falls into the product please stop using.
6. Please do not discard the packing bags or store in places where children could reach, if children trap his head with it, may cause nose and mouth blocked, and even lead to suffocation.
7. Please use this product with normal usage and operating, in bad posture for a long time to use this product may affect your health.
8. Please read the above safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

Table of Content

1. Introducing VoIP Phone

1.1 Simple Introduction

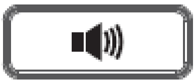
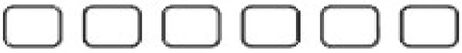
Thank you for your purchasing DPH-200SE.
DPH-200SE is a full-feature telephone that provides voice communication over the same data network that your computer uses. This phone's functions not only much like a traditional phone, allowing to place and receive calls, and enjoy other features that traditional phone has, but it also own many data services features which you could not expect from a traditional telephone.
This guide will help you easily use the various features and services available on your phone.

1.2 Delivery Content

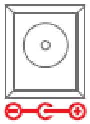
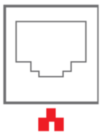
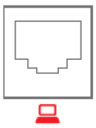
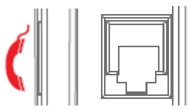
Please check whether the delivery contains the following parts:

Item	Description
IP Phone	DPH-200SE Phone with display and keypad.
Power Adapter	Power supply for telephone.
Network Cable	Used to access network for the phone.
Handset	Make phone calls with the phone's basic functions.
Handset Cable	Connected with the handset and the phone.
Quick Installation Guide	Quick install the DPH-200SE guide.
CD	Containing manual and quick installation guide.
Warranty Safety Information	Warranty Safety Information for DPH-200SE.

IP P hone d re t l esigned o p ook t fike pnventsional a b hone
overview of the IP Phone.

	Hands-free	Make the phone into hands-free mode.
 DSS keys		You can configure them in the web page.

1.4 Port for connecting

Port	Port name	Description
	Power switch	Input: 5V DC, 1A
	WAN	10/100M Connect it to Network
	LAN	10/100M Connect it to PC
	Headset	Port type: RJ-9 connector

1.5 Icon introduction

Icon	Description
	SIP Status :Green is registration; White is unregistered or failure
	Call out(handset or speaker)
	Call in
	Call hold
	Call mute
	In hand-free mode

	In headset mode
	Call transfer

1.6 LED introduction

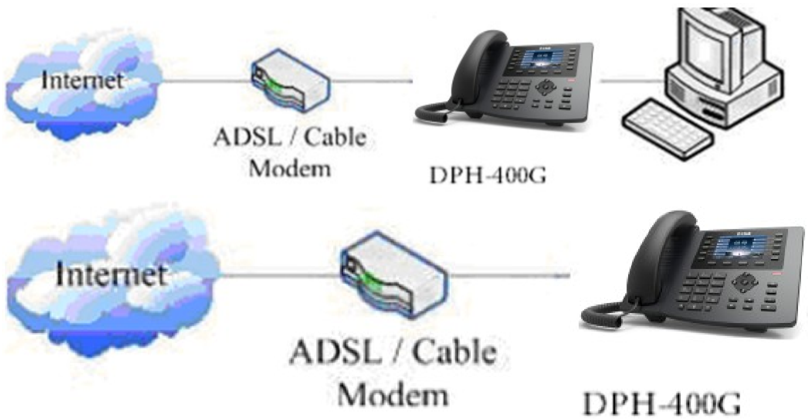
LEDs for BLF Message waiting / Incoming call - The light flashes when the telephone rings for incoming calls, and when the Message Waiting (MWI) is supported in the telephone system. The light lights up when a call is on hold.

2. Initial Connecting and Settings

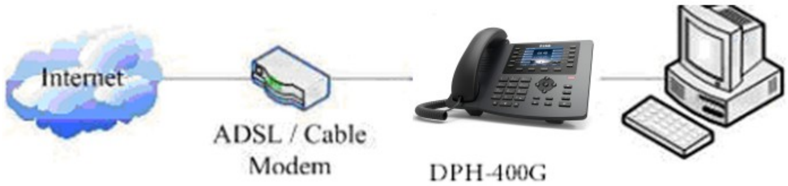
2.1 Connect the phone

2.1.1 Connect to network

Step 1: Connect the IP Phone to the corporate IP telephony network. Before you connect the phone to the network, please check if your network can work normally. You can do this in one of two ways, depending on how your workspace is set up. Direct network connection—by this method, you need at least one available Ethernet port in your workspace. Use the Ethernet cable in the package to connect WAN port on the back of your phone to the Ethernet port in your workspace. Since this VoIP Phone has router functionality, whether you have a broadband router or not, you can make direct network connect. The following two figures are for your reference.



Shared network connection—Use this method if you have a single Ethernet port in your workspace with your desktop computer already connected to it. First, disconnect the Ethernet cable from the computer and attach it to the WAN port on the back of your phone. Next, use the Ethernet cable in the package to connect LAN port on the back of your phone to your desktop computer. Your IP Phone now shares a network connection with your computer. The following figure is for your reference.



Step 2: Connect the handset to the handset port by the handset cable in the package.
Step 3: connect the power supply plug to the DC 5V adapter port on the back of the phone. Use the power cable to connect the power supply to a standard power outlet in your workspace.
Step 4: Then the phone's LCD screen displays "Dlink Logo". Later, a ready screen typically displays the date, time. If your LCD screen displays different information from the above, you need refer to the next section "Initial setting" to set your network online mode. If your VoIP phone registers into corporate IP telephony Server, your phone is ready to use.

2.1.2 Power adaptor connection

Make sure that the power you use is comply with the parameters of power adaptor.
1. Plug power adaptor to power socket.
2. Plug power adaptor's DC output to the DC5V port of DPH-200SE to start up.
3. There will be displayed blue line and "Dlink" Logo on the screen. After finishing startup, phone will show greeting, current date and time and so forth.
4. If phone has registered to the server, you can place or answer calls.

2.2 Checking IP address

There are two ways for DPH-200SE to check IP address.
1. Pick up the handset or press hands-free key, please input "# * 111" button, then you can hear the IP address voice information.
2.Long press "#" and you will see the IP address shows on screen.

2.3 Basic Initialization

DPH-200SE is provided with a plenty of functions and parameters for configuration. User needs some network and VoIP knowledge so that user could understand the meanings of parameters. In order to make user use the phone more easily and convenient, there are basic configurations introduced which is mandatory to ensure phone calls.

2.3.1 Network settings

Make sure that network is connected already before setting network of phone. DPH-200SE uses DHCP to get WAN IP configurations, so phone could access to network as long as there is DHCP server in it. If there is no DHCP server available, phone has to be changed WAN network setting to Static IP or PPPoE.

Setting PPPoE mode (for ADSL connection)

1. Get PPPoE account and password first.
2. Long press “#” at phone, you will get a IP address. Logon web page of the phone, go to Network ->Basic, choose PPPoE mode.
3. The web page will show the current information. Delete it, then input your PPPoE user and password and press Apply.
4. Refresh the web page, if it shows a PPPoE IP address, then the phone has already get IP with PPPoE.

Setting Static IP mode (static ADSL/Cable, or no PPPoE / DHCP network)

1. Prepare the network’s parameters first, such as IP Address, Net mask, Default Gateway and DNS server IP address. If you don’t know this information, please contact the service provider or technician of network.
2. Long press “#” at phone, you will get an IP address. Logon web page of the phone, go to Network ->Basic, choose Static IP mode.
3. The web page will show the current information, and then delete it. Input your IP address, Mask, Gateway, DNS and click Apply to save what you input.
4. Long press “#” at phone, you will get an IP address again. Logon web page use new IP address, go to System -> Information, check the network status, the webpage shows “Static” and shows the IP address and gateway which were set just now, if the phone could display the right time, it shows that Static IP mode takes effect.

Setting DHCP mode

1. Long press “#” at phone, you will get an IP address. Network ->Basic, choose DHCP mode, click Apply.
2. Long press “#” at phone, you will get an IP address again. Logon web page use new IP address, go to System -> Information, check the network status, t “DHCP”, if the web page shows the IP address and gateways which were set just now, it shows that DHCP mode takes effect.

3 Basic Function

3.1 Making a call

You can make a phone call via the following devices:

1. Pick up the handset, icon will be showed in the idle screen.
2. Press the Speaker button, icon will be showed in the idle screen.

You c a an d tlson iaf ahe t umber t m irsty w nd t then t t hoose other party.

3.2 Answering a call

Answering an incoming call

1. If you are not on another phone, lift the handset using, or press the Speaker button to answer using the speaker phone.
2. If you are on another call, press Speaker button or release handset to end of the first call, then you can answer the second call by pressing the Speaker button or picking up the handset.

3.3 Call Hold

1. Press the Hold button to put your active call on hold.
2. If there is only one call on hold, press the hold button to retrieve the call.

3.4 Mute

Press Mute button during the conversation, icon will be showed in the LCD. Then the called will not hear you, but you can hear the called. Press it again to get the phone to normal conversation.

3.5 Call Ended

When user finished the call, user can put the handset back to the device to hang up the call or press the hands-free button to close the audio channel to hang up.
Note: When the call is held to in holding state, the user to call again mode, putting the handset back to the hang up is not available.

3.6 Redial

Pick up the handset or hands-free key. Press Redial button to dial the last number you dialed.

4 Advanced Function

4.1 Call transfer

1. Blind Transfer
During talk, press the Transfer button, and then dial the number that you want to transfer to, and finished by Transfer button. Phone will transfer the current call to the third party. After finishing transfer, the call you talk to will be hanged up. User cannot select SIP line when phone transfers call.
2. Attended Transfer
During talk, press the Transfer button, then input the number that you want to transfer to and press Dial. After that third party answers, then press Transfer button to complete the transfer. (You need enable call waiting and call transfer first).
3. Alert Transfer
During the talk, press Transfer button firstly, and then press Dial after inputting the number that you want to transfer. You are waiting for connection, now, press Transfer button and the transfer will be done. (To use this feature, you need enable call waiting and call transfer first).

4.2 Messages waiting

When theMessagesLED lights up, you need to dial the feature access code for message retrieving. Once the messages have been retrieved, the lights up will stop. You can your messages waiting feature access code on a memory button, when labeled Messages.

4.3 Programmable Key Configuration

The phone has 6 programmable keys which are able to set up to many functions per key. The following list shows the functions you can set on the programmable keys and provides a description for each function.

1. Set the type as Memory Key

Logon web page of phone, go to Function Key, the function key type is default set to memory key. In the Dial field, you have some options, such as None, Speed Dial, Intercom, Call Park, Call forward, and MWI.

Speed dial

You can configure the key as a simplified speed dial key. This key function allows you to easily access your most dialed numbers.

Intercom

You can configure the key for Intercom code and it is useful in an office environment as a quick access to connect to the operator or the secretary.

Call Park

When the key is configured as Call Park, you can retrieve the held call by using the call park code.

Call forward

When the key is configured as Call forward, you can transfer the call to the set number.

MWI

When the key is configured as MWI, you are allowed to access voicemail quickly by pressing this key.

5 Web configuration

5.1 Introduction of configuration

5.1.1 Ways to configure

DPH-200SE has one way to different users.

- Use web browser (recommendatory way).

5.1.2 Password Configuration

Default user with root level:
Username: admin
Password: admin

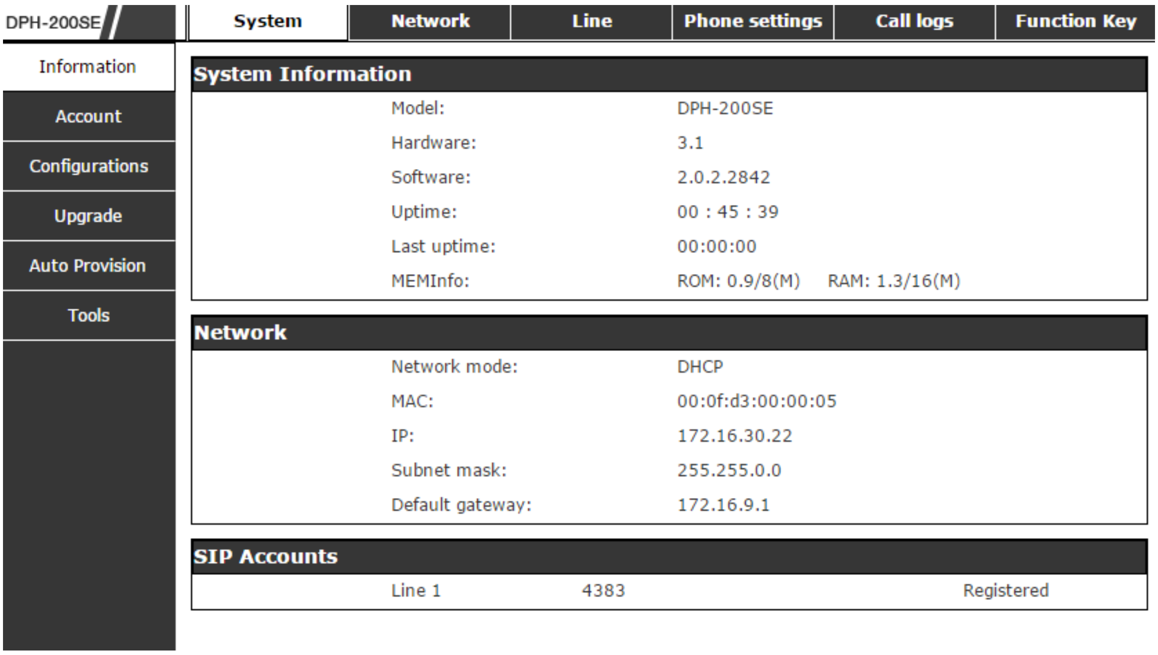
5.2 Setting via web browser

When this phone and PC are connected to network, enter the IP address of the wan port in this phone as the URL (e.g. http://xxx.xxx.xxx.xxx/ or http://xxx.xxx.xxx.xxx:xxxx/).
If you do not know the IP address, you can look it up on the phone's display by pressing Status button. The login page is as below picture.

5.3 Configuration via WEB

5.3.1 System

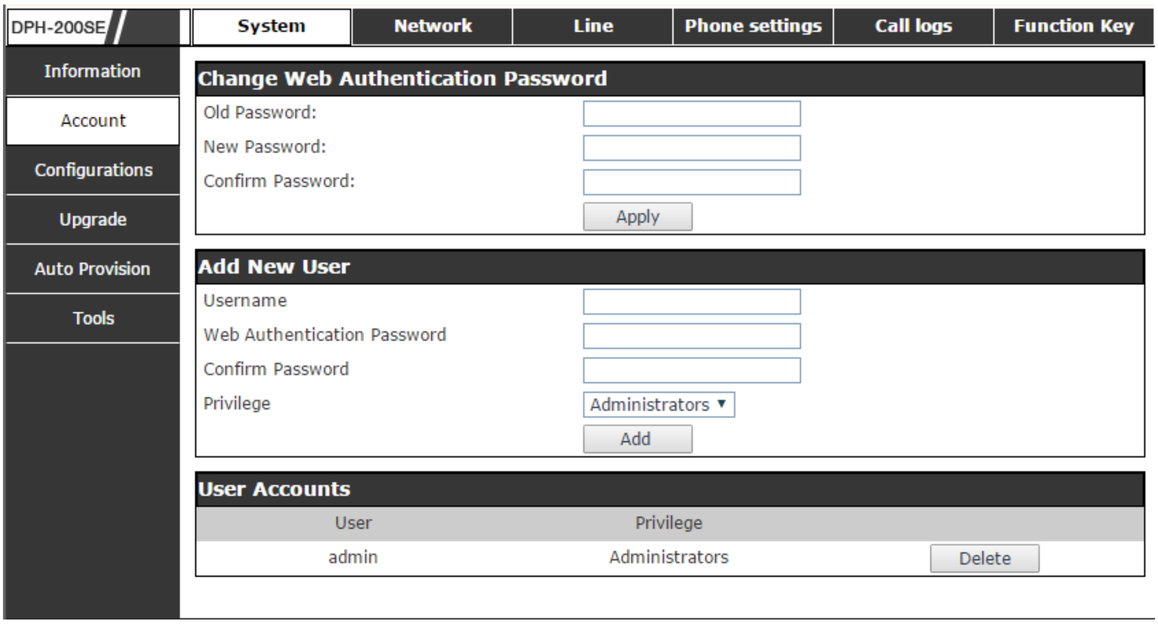
5.3.1.1 Information



Information	
Field Name	**Explanation**
System Information	Display equipment model, hardware version, of and MEMInfo.
Network	Shows the configuration information for WAN port, including connection mode of WAN port (Static, DHCP, PPPoE), MAC address, IP address of WAN port.

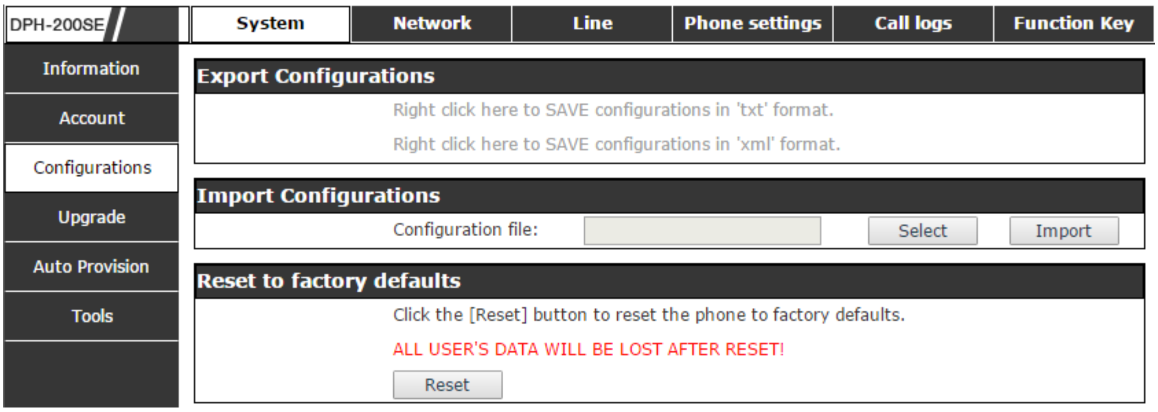
5.3.1.2 Account

Through this page, user can add or remove users depends on their needs and can modify existing user permission.



Account	
Field Name	**Explanation**
Change Web Authentication Password	
You Can modify the login password to the account	
Add New User	
You can add new user	
User Accounts	
Show the existing user information	

5.3.1.3 Configurations



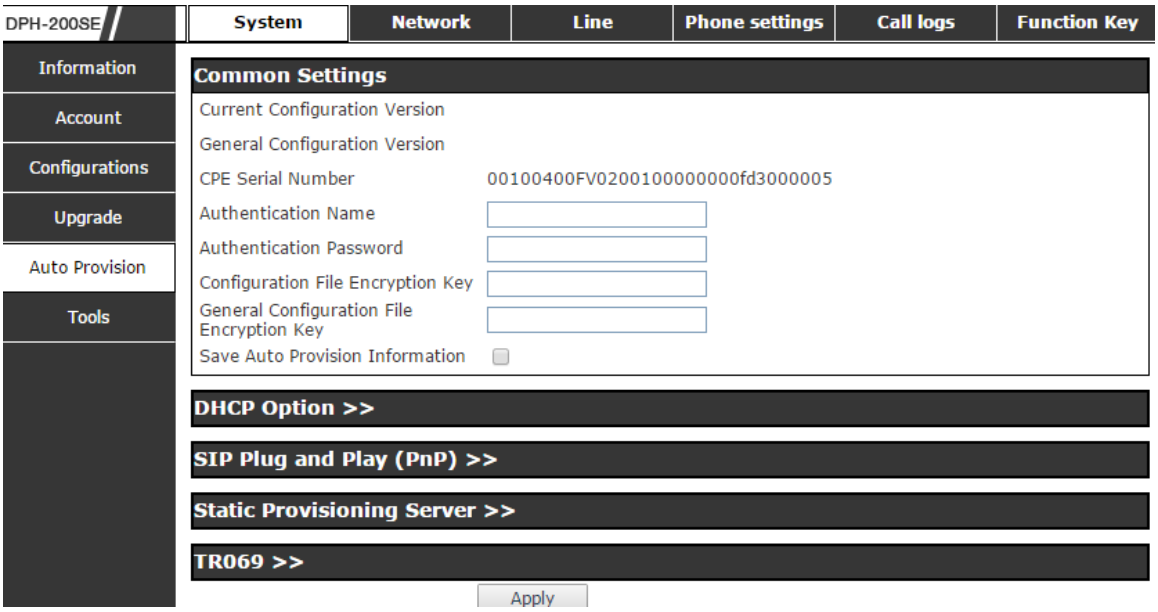
Configurations	
Field Name	**Explanation**
Export	Save t e he quipment t a t onfigurationP n t Ro c xt r

Configurations	on the choice and then choose "Save Link As."
Import Configurations	Browse to the config file, and pres equipment.
Reset to factory defaults	This will restore factory default and remove all configuration information.

5.3.1.4 Upgrade

Upgrade	
Field Name	**Explanation**
Software upgrade	
Browse to the firmware, and press Update to load it to the equipment.	

5.3.1.5 Auto Provision



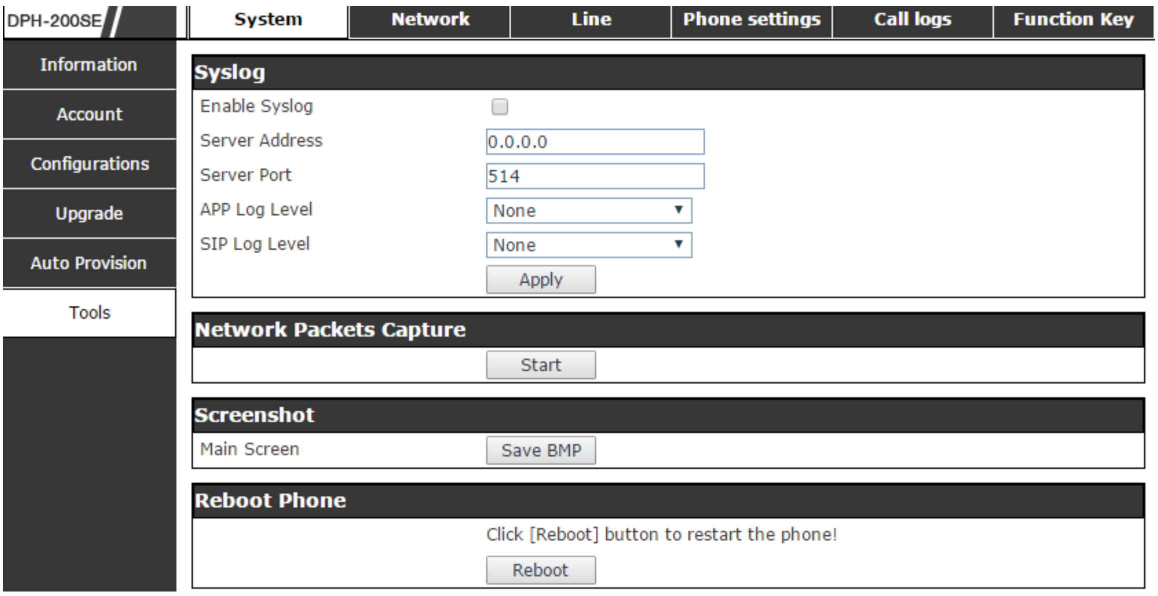
Auto Provision	
Field Name	**Explanation**

Common Settings	
Current Configuration Version	Show the current config file's version. If the version of configuration downloaded is higher than this, the configuration will be upgraded. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration
General Configuration Version	Show the common config file's version. If the configuration downloaded and this configuration is the same, the auto provision will stop. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
Authentication Name	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank the phone will use anonymous
Authentication Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Configuration File Encryption Key	Encryption key for the configuration file
General Configuration File Encryption Key	Encryption key for common configuration file

DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.
Custom Option Value	Custom option number. Must be from 128 to 254.
SIP Plug and Play (PnP)	
Enable SIP PnP	If this is enabled, the equipment will send SIP SUBSCRIBE messages to a multicast address when it boots up. Any SIP server understanding that message will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
Server Address	PnP Server Address
Server Port	PnP Server Port
Transportation Protocol	PnP Transfer protocol – UDP or TCP
Update Interval	Interval time for querying PnP server. Default is 1 hour.
Static Provisioning Server	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an

	IP address or Domain name with subdirectory.
Configuration File Name	Specify configuration file name. The equipment will use its MAC ID as the config file name if this is blank.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify the update interval time. Default is 1 hour.
Update Mode	1. Disable – no update 2. Update after reboot – update only after reboot. 3. Update at time interval – update at periodic update interval
TR069	
Enable TR069	Enable/Disable TR069 configuration
ACS Server Type	Select Common or CTC ACS Server Type.
ACS Server URL	ACS Server URL.
ACS User	User name for ACS.
ACS Password	ACS Password.
TR069 Auto Login	Enable/Disable TR069 Auto Login.
INFORM Sending Period	Time between transmissions of “Inform” Unit is seconds.

5.3.1.6 Tools



Syslog is a protocol used to record log messages using a client/server mechanism. The Syslog server receives the messages from clients, and classifies them based on priority and type. Then these messages will be written into a log by rules which the administrator has configured.

There are 8 levels of debug information.

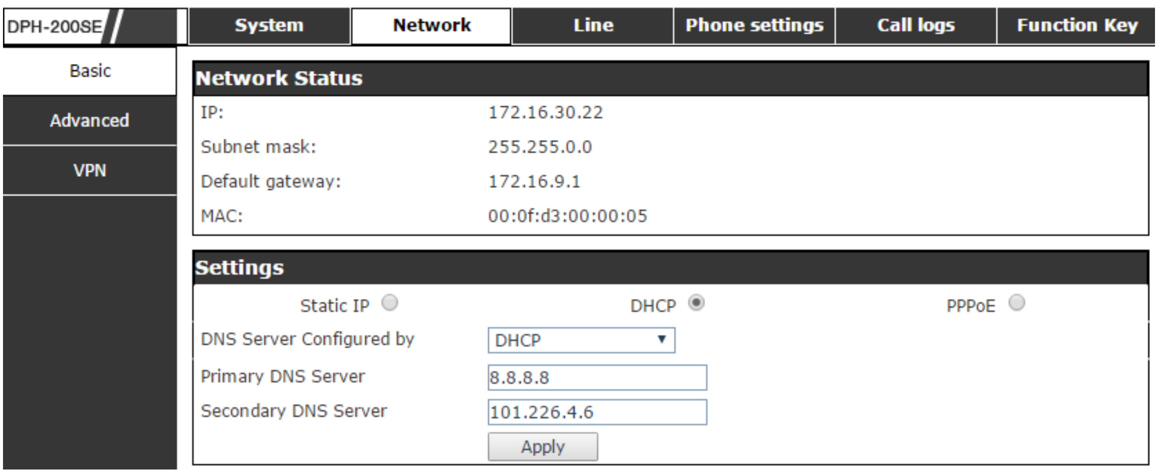
Level 0: emergency; System is unusable. This is the highest debug info level.

Level 1: alert; Action must be taken immediately.
Level 2: critical; System is probably working incorrectly.
Level 3: error; System may not work correctly.
Level 4: warning; System may work correctly but needs attention.
Level 5: notice; It is the normal but significant condition.
Level 6: Informational; It is the normal daily messages.
Level 7: debug; Debug messages normally used by system designer. This level can only be displayed via telnet.

Tools	
Field Name	**Explanation**
Syslog	
Enable Syslog	Enable or disable system log.
Server Address	System log server IP address.
Server Port	System log server port.
APP Log Level	Set the level of APP log.
SIP Log Level	Set the level of SIP log.
Network Packets Capture	
Capture a packet stream from the equipment. This is normally used to troubleshoot problems.	
Reboot Phone	
Some configuration modifications require a reboot to become effective. Clicking the Reboot button will lead to reboot immediately. Note: Be sure to save the configuration before rebooting.	

5.3.2 Network

5.3.2.1 Basic



Field Name	Explanation
Network Status	
IP	The current IP address of the equipment
Subnet mask	The current Subnet Mask
Default gateway	The current Gateway IP address
MAC	The MAC address of the equipment
MAC Timestamp	Get the MAC address of time.
Settings	
Select the appropriate network mode. The equipment supports three network modes:	
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP.
DHCP	Network parameters are provided automatically by a DHCP server.
PPPoE	Account and Password must be input manually. These are provided by your ISP.
If Static IP is chosen, the screen below will appear. Enter values provided by the ISP.	
DNS Server Configured by	Select the Configured mode of the DNS Server.
Primary DNS Server	Enter the server address of the Primary DNS.
Secondary DNS Server	Enter the server address of the Secondary DNS.
After entering the new settings, click the APPLY button. The equipment will save the new settings and apply them. If a new IP address was entered for the equipment, it must be used to login to the phone after clicking the APPLY button.	

5.3.2.2 Advanced

The equipment supports 802.1Q/P protocol and DiffServ configuration. VLAN function can support the different VLAN ID mode of processing in the WAN port and LAN port.

- Chart 1 shows a network switch with no VLAN. Any broadcast frames will be transmitted to all other ports. For example, frames broadcast from Port 1 will be sent to Ports 2, 3, and 4.

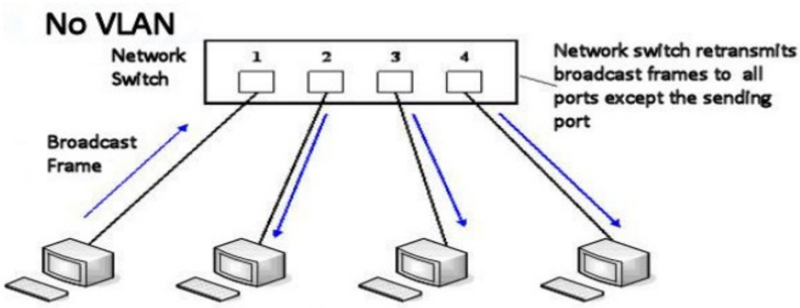


Chart 1

- Chart 2 s a hows w nt Xample i bith a wbo I tLANs f ndicated broadcast from Port 1 will only go to Port 2 since Ports 3 and 4 are in a different V VLANs can be used to divide a network by restricting the transmission of broadcast frames.

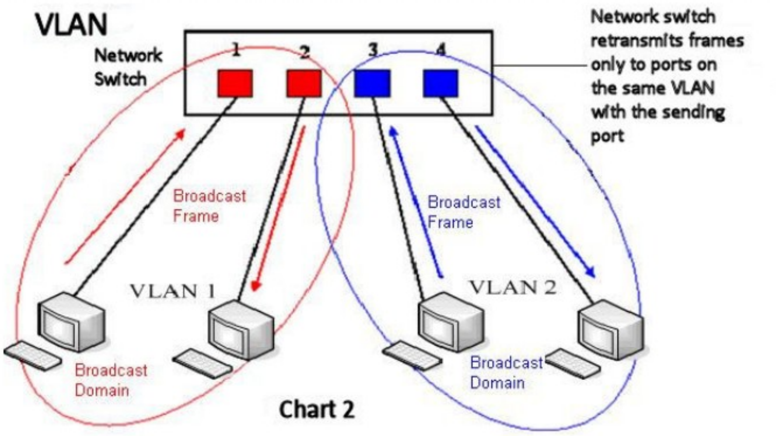
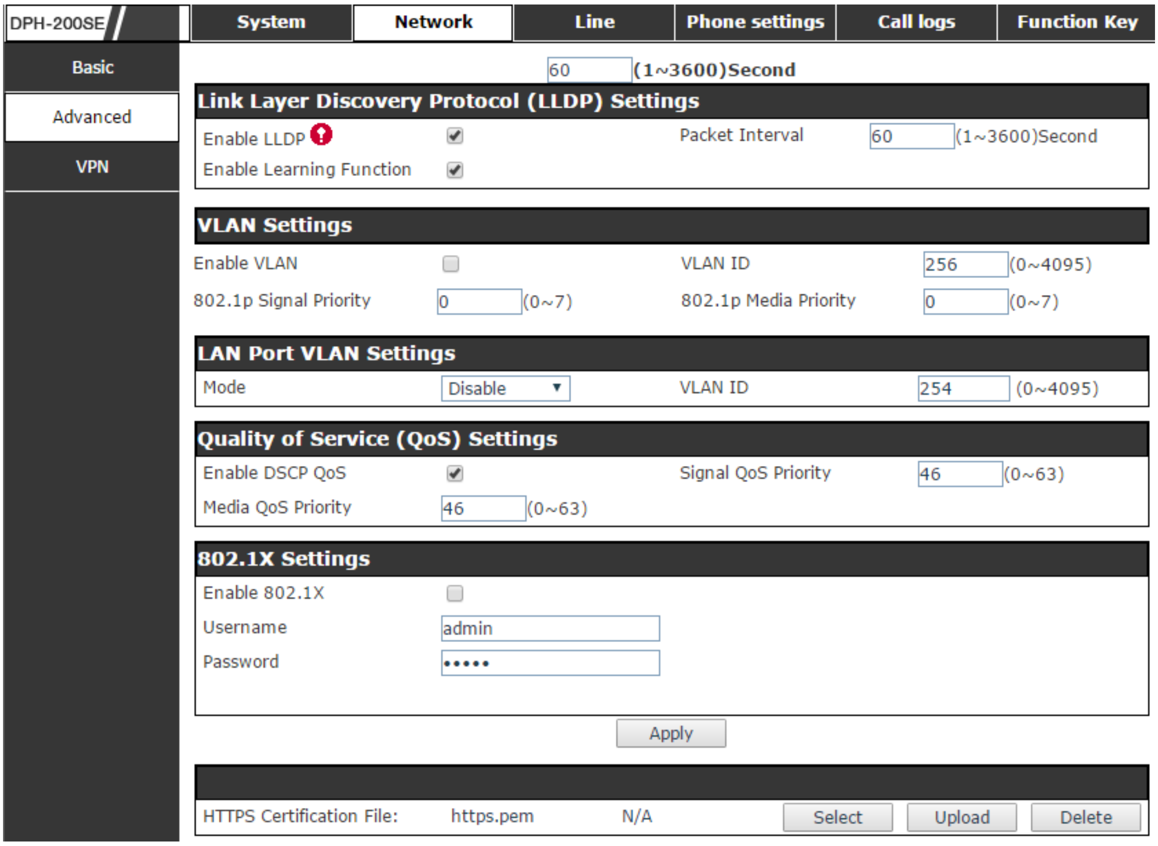


Chart 2

Note: In practice, VLANs are distinguished by the use of VLAN IDs.

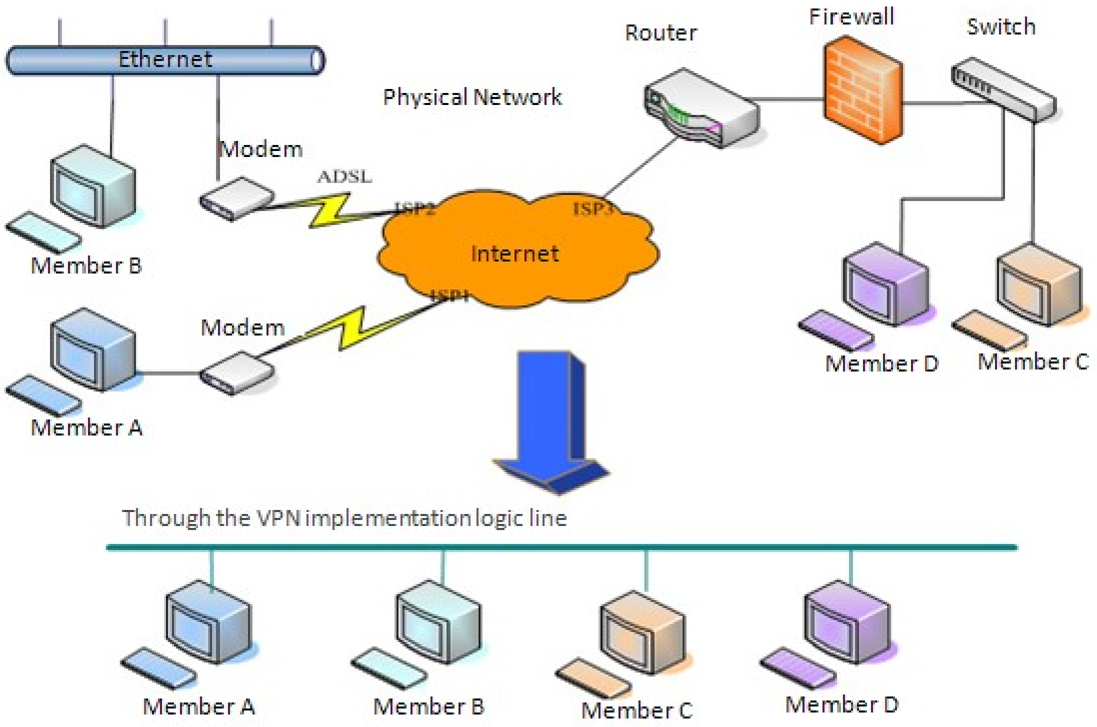


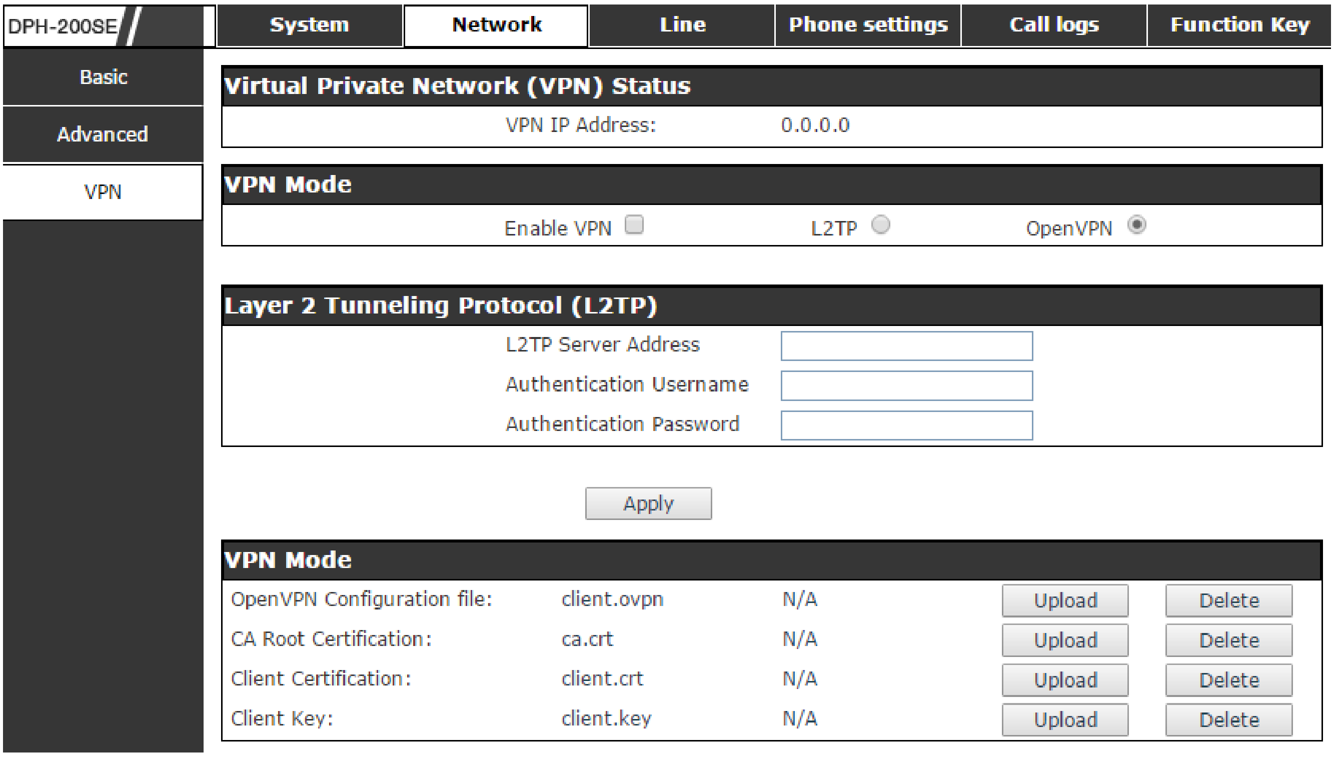
Advanced	
Field Name	**Explanation**
Link Layer Discovery Protocol (LLDP) Settings	
Enable LLDP	Enable or Disable Link Layer Discovery Protocol (LLDP)
Enable Learning Function	Enables the telephone to synchronize its VLAN data with the Network Switch. The telephone will automatically synchronize DSCP, 802.1p, and VLAN ID values even if these values differ from those provided by the LLDP server.
Packet Interval(1~3600)	The time interval for sending LLDP Packets
VLAN Settings	
Enable VLAN	Enable or Disable WAN Port VLAN
VLAN ID	Specify the value of the VLAN ID. Range is 0-4095
802.1p Signal Priority	Specify the value of the signal 802.1p priority. Range is 0-7
802.1p Media Priority	Specify the value of the voice 802.1p priority. Range is 0-7
Quality of Service (QoS) Settings	
Enable DSCP QoS	Enable or Disable Differentiated Services Code Point (DSCP)
Media QoS Priority	Specify the value of the Media DSCP in decimal
Signal QoS Priority	Specify the value of the Signal DSCP in decimal

802.1X Settings	
Enable 802.1X	Enable or Disable 812.1X
Username	802.1X user account
Password	802.1X password

8.3.2.3 VPN

The device supports remote connection via VPN. It supports both Layer 2 Tunneling Protocol (L2TP) and OpenVPN protocol. This allows users at remote locations on the public network to make secure connections to local networks.



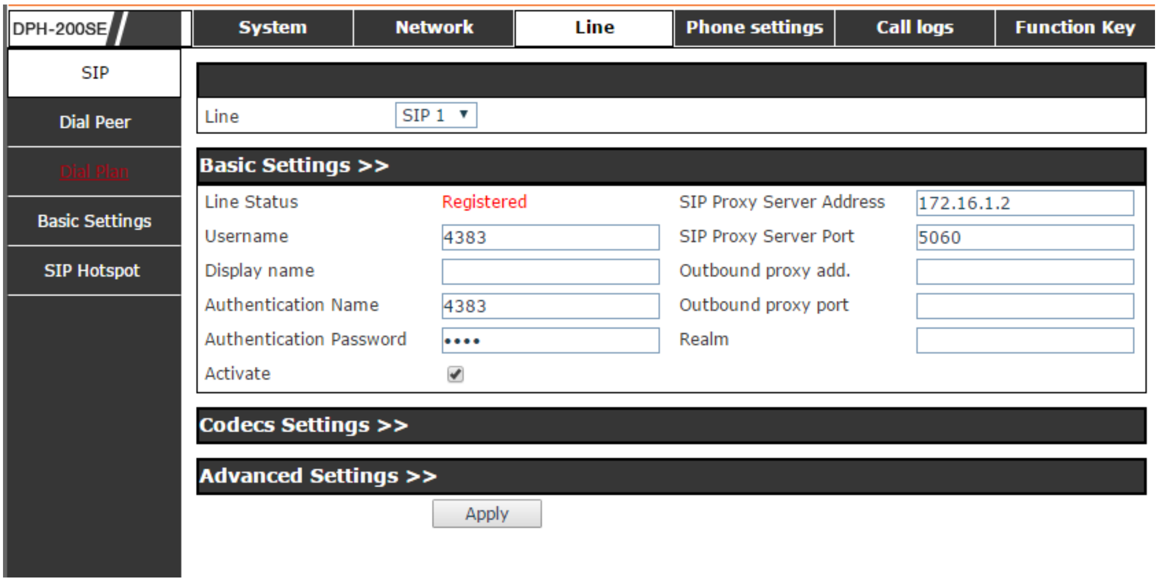


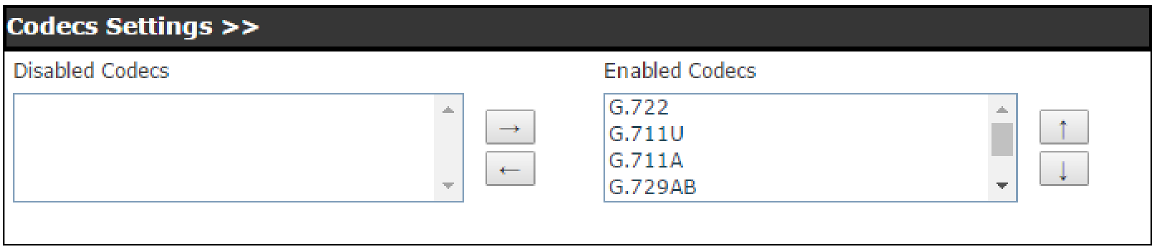
Field Name	Explanation
VPN IP Address	Shows the current VPN IP address.
VPN Mode	
Enable VPN	Enable/Disable VPN.
L2TP	Select Layer 2 Tunneling Protocol
OpenVPN	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection is made, the configuration should be saved and the phone be rebooted.)
Layer 2 Tunneling Protocol (L2TP)	
L2TP Server Address	Set VPN L2TP Server IP address.
Authentication Name	Set User Name access to VPN L2TP Server.
Authentication Password	Set Password access to VPN L2TP Server.
Open VPN Files	
Upload or delete Open VPN Certification Files	

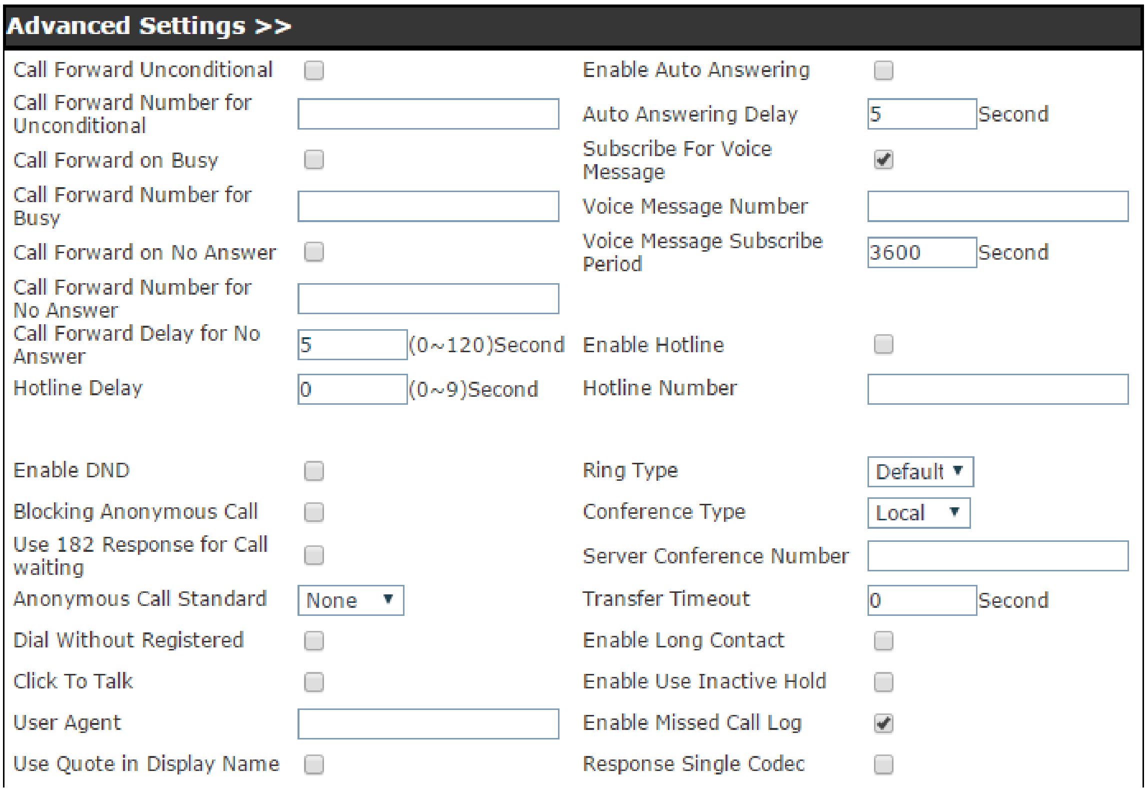
5.3.3 Line

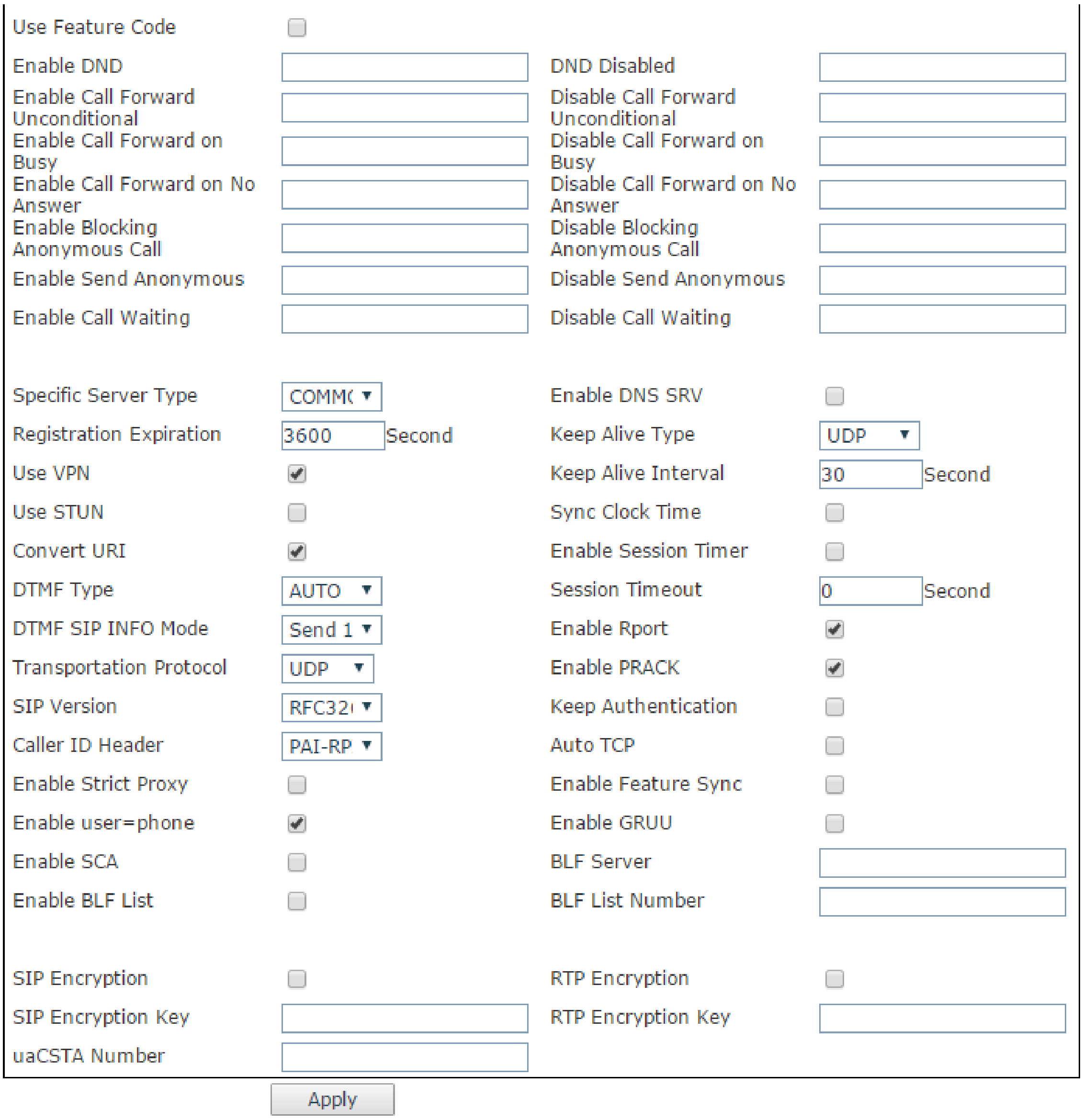
5.3.3.1 SIP

Configure a SIP server on this page.






Advanced Settings >>
Call Forward Unconditional
Call Forward Number for Unconditional
Call Forward on Busy
Call Forward Number for Busy
Call Forward on No Answer
Call Forward Number for No Answer
Call Forward Delay for No Answer
5
(0~120)Second
Hotline Delay
0
(0~9)Second
Enable Auto Answering
Auto Answering Delay
5
Second
Subscribe For Voice Message
Voice Message Number
Voice Message Subscribe Period
3600
Second
Enable Hotline
Hotline Number
Enable DND
Blocking Anonymous Call
Use 182 Response for Call waiting
Anonymous Call Standard
None
Dial Without Registered
Click To Talk
User Agent
Use Quote in Display Name
Ring Type
Default
Conference Type
Local
Server Conference Number
Transfer Timeout
0
Second
Enable Long Contact
Enable Use Inactive Hold
Enable Missed Call Log
Response Single Codec



SIP	
Field Name	**Explanation**
Basic Settings	
Line Status	Display the current line status at page loading. To get the up to date l status, user has to refresh the page manually.
Username	Enter the username of the service account.
Display name	Enter the display name to be sent in a call request.
Authentication Name	Enter the authentication name of the service account
Authentication Password	Enter the authentication password of the service account
Activate	Whether the service of the line should be activated
SIP Proxy Server	Enter the IP or FQDN address of the SIP proxy server

Address	
SIP Proxy Server Port	Enter the SIP proxy server port, default is 5060
Outbound proxy address	Enter the IP or FQDN address of outbound proxy server provided by the service provider
Outbound proxy port	Enter the outbound proxy port, default is 5060
Realm	Enter the SIP domain if requested by the service provider
Codecs Settings	
Set the priority and availability of the codecs by adding or remove them from the list.	
Advanced Settings	
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field
Call Forward Number for Busy	Set the number of call forward on busy
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field
Call Forward Number for No Answer	Set the number of call forward on no answer
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-

	free speaker or headphone
Hotline Number	Set the hotline dialing number
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Anonymous Call Standard	Set the standard to be used for anonymous
Dial Without Registered	Set call out by proxy without registration
Click To Talk	Set Click To Talk
User Agent	Set the user agent, the default is Model with Software Version.
Use Quote in Display Name	Whether to add quote in display name
Ring Type	Set the ring tone type for the line
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Transfer Timeout	Set the timeout of call transfer process
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Specific Server Type	Set the line to collaborate with specific server type
Registration Expiration	Set the SIP expiration interval
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Convert URI	Convert not digit and alphabet characters to %hh hex code

DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Transportation Protocol	Set the line to use TCP or UDP for SIP transmission
SIP Version	Set the SIP version
Caller ID Header	Set the Caller ID Header
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it will use the source IP address, not the address in via field.
Enable user=phone	Sets user=phone in SIP messages.
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
Enable BLF List	Enable/Disable BLF List
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a service list
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Keep Authentication	Keep the authentication parameters from previous authentication
Auto TCP	UsingTCP protocol to guarantee usability of transport for SIP mes above 1500 bytes
Enable Feature Sync	Feature Sycn with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
SIP Encryption Key	Set the pass phrase for SIP encryption

RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

5.3.3.2 Dial Peer

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule. When you want to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, you can set number 156 to replace 192.168.1.119 here.

Dial Peer Table

Number	Destination (Optional)	Port (Optional)	Call Mode	Alias(Optional)	Suffix (Optional)	Deleted Length (Optional)
156	192.168.1.119	5060	SIP	no alias	no suffix	0

When you want to dial a long distance call to Beijing, you need dial an area code 010 before local phone number, but you can also dial number 1 instead of 010 after we make a setting according to this dial rule. For example, you want to dial 01062213123, but you need dial only 162213123 to realize your long distance call after you make this setting.

Dial Peer Table

Number	Destination (Optional)	Port (Optional)	Call Mode	Alias(Optional)	Suffix (Optional)	Deleted Length (Optional)
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

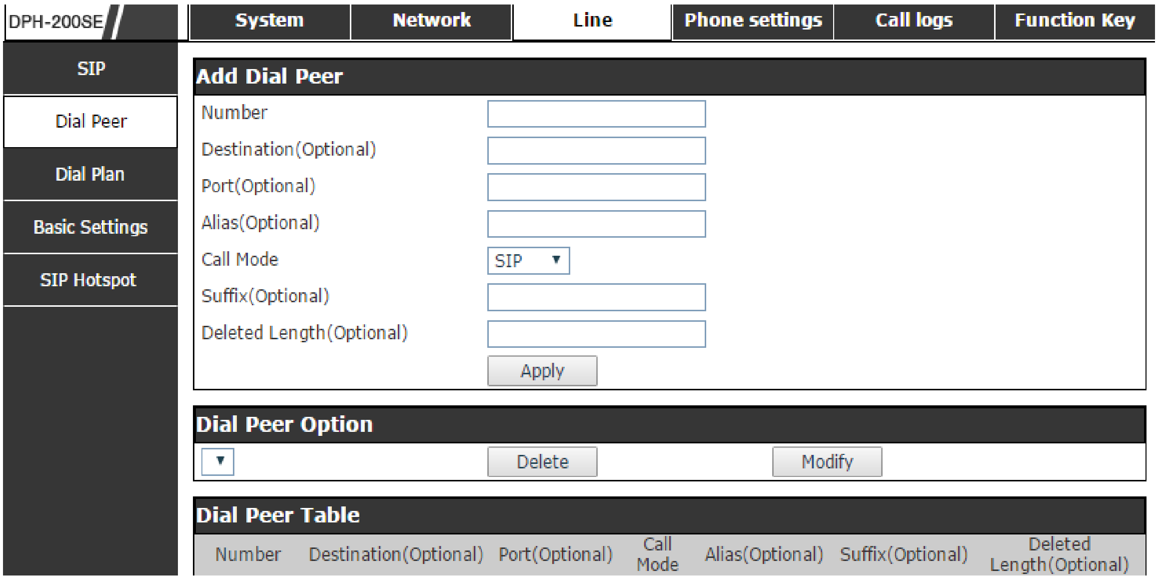
To save the memory and avoid abundant input of user, add the follow functions:

Dial Peer Table

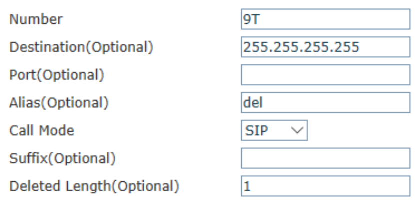
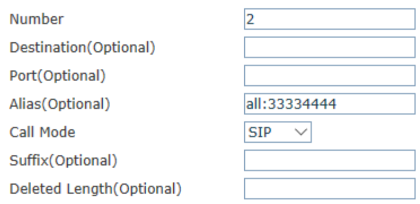
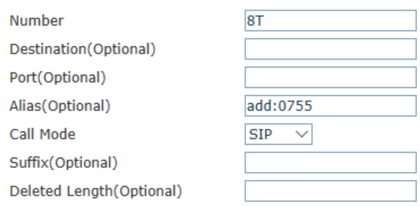
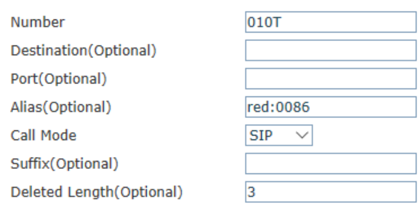
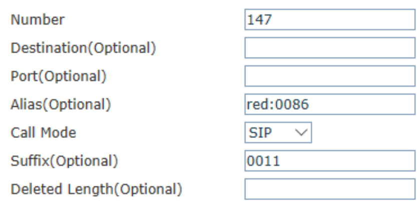
Number	Destination (Optional)	Port (Optional)	Call Mode	Alias(Optional)	Suffix (Optional)	Deleted Length (Optional)
135xxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0
13(5-9) xxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0

1. * Match any single digit that is dialed. If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.
2. [] Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.

If user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically. Use this phone you can realize dialing out via different lines without switch in web interface.



Dial Peer	
Field Name	**Explanation**
Number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, you need input your desired phone number in this blank, and then you need dial the phone number to realize calling to what the phone number is mapped. In the prefix matching, you need input your desired prefix number and T; then dial the prefix and a phone number to realize calling to what your prefix number is mapped. The prefix number supports at most 30 digits.
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, the default is 5060 for SIP.
Alias	Set alias. This is optional config item. If you don’t set Alias, it will show no alias.
Note: There are four types of aliases. 1) Add: xxx, it means that you need dial xxx in front of phone number, which will reduce dialing number length. 2) All: xxx, it means that xxx will replace some phone number. 3) Del: It means that phone will delete the number with length appointed. 4) Rep: It means that phone will replace the number with length and number appointed. You can refer to the following examples of different alias application to know more how to use different aliases and this dial rule.	
Call Mode	Select different signal protocol, SIP
Suffix	Characters to be added at the end of the phone number. It is an optional item.
Delete Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional

	item.	
Examples of different alias application		
Set by web	Explanation	Example
	You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length.	If you dial “93333”, the SIP2 server will receive “3333”.
	This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.	When you dial “2”, the SIP1 server will receive 33334444.
	The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.	When you dial “8309“, the SIP1 server will receive “07558309”.
	You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out.	When you dial “0106228”, the SIP1 server will receive “86106228”.
	If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.	When you dial “147”, the SIP1 server will receive “1470011”.

5.3.3.3 Dial Plan

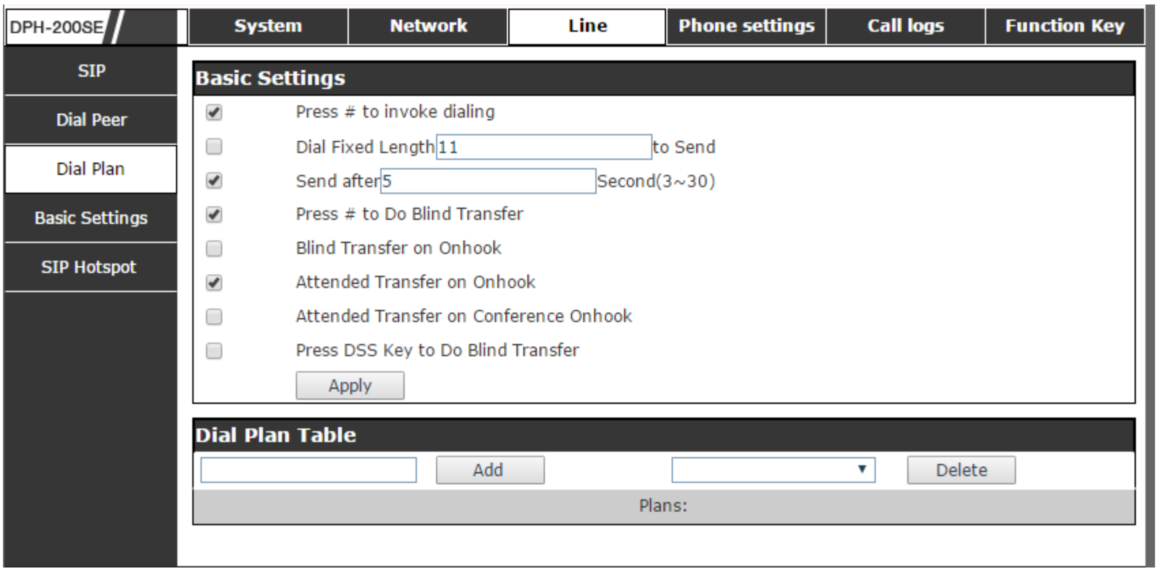
This system supports 4 dial modes:

1) End with "#": dial your desired number, and then press #.

2) Fixed Length: the phone will intersect the number according to your specified length.

3) Time Out: After you stop dialing and waiting time out, system will send the number collected.

4) User defined: you can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing. In order to keep some users' secondary dialing manner when dialing the external line with PBX, phone can be added a special rule to realize it. so user can dial a number as external line prefix and get the secondary dial tone to keep dial the external number. After finishing dialing, phone will send the prefix and external number totally to the server. For example, there is a rule 9, xxxxxxxx in the digital map table. After dialing 9, phone will send the secondary dial tone, user may keep going dialing. After finished, phone will call the number which starts with 9; actually the number sent out is 9-digit with 9.

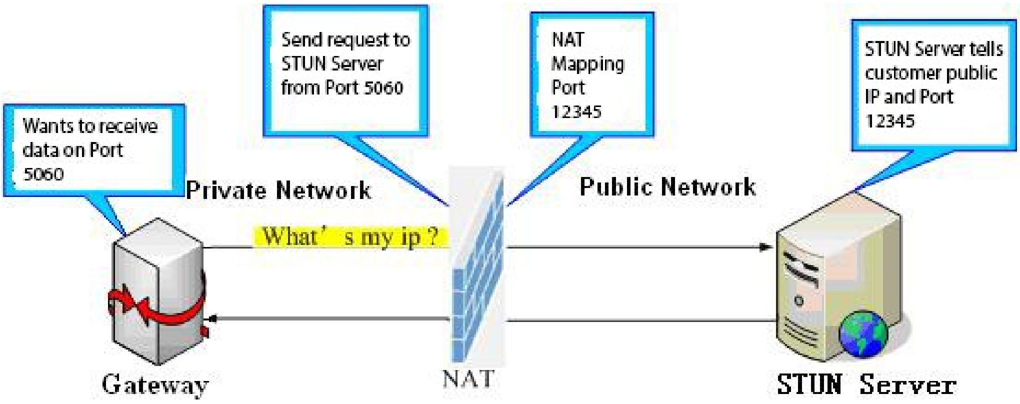


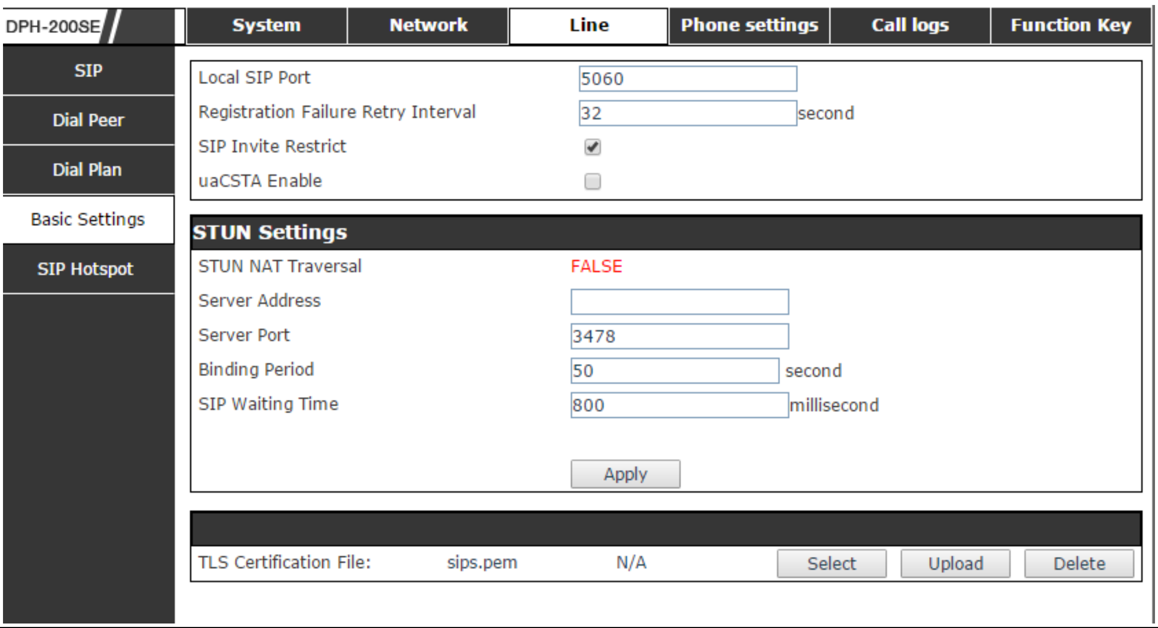
Dial Plan	
Field Name	**Explanation**
Basic Setting	
Press # invoke dialing	Set Enable/Disable the phone ended with "#" dial.
Dial Fixed Length	Specify the Fixed Length of phone ending with.
Send after (3-30)seconds	Set the timeout of the last dial digit. The call will be sent after timeout.
Press # to Do Blind	Enable Blind Transfer On Hook, when executing Blind Transfer End with

Transfer	#, press # after inputting the number that you want to transfer, the phone will transfer the current call to the third party.
Blind Transfer on OnHook	Enable Blind Transfer on On Hook, when executing Blind Transfer, hang up after inputting the number that you want to transfer, the phone will transfer the current call to the third party.
Attend Transfer on OnHook	Enable Attend Transfer on On Hook, when executing Attended Transfer, hang up after the third party answers, the phone will transfer the current call to the third party.
Attended Transfer on Conference Onhook	Attended Transfer on Conference Onhook - Hang up during a 3-way conference call, the other two ways will make a call.
Press DSS Key to Do Blind Transfer	Press DSS Key to Do Blind Transfer – When user is in the 'XFER' screen, user can fulfill Blind Transfer by pressing DSS Key.
Dial Plan Table	
Below is user-defined digital map rule: [] Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits. * Match any single digit that is dialed. . Match any arbitrary number of digits including none. Tn Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans. "RULE" "[1-8]XXX" "9XXXXXXX" "911" "99T4" "9911x.T4" Cause extensions 1000-8999 to be dialed immediately. Cause 8 digit numbers started with 9 to be dialed immediately. Cause 911 to be dialed immediately after it is entered. Cause 99 to be dialed after 4 seconds. Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.	
Notice: End with "#", Fixed Length, Time out and Digital Map Table can be	

5.3.3.4 Basic Settings

STUN – Simple Traversal of UDP through NAT –A STUN server allows a phone in a private network to know its public IP and port as well as the type of NAT being used. The equipment can then use this information to register itself to a SIP server so that it can make and receive calls while in a private network.





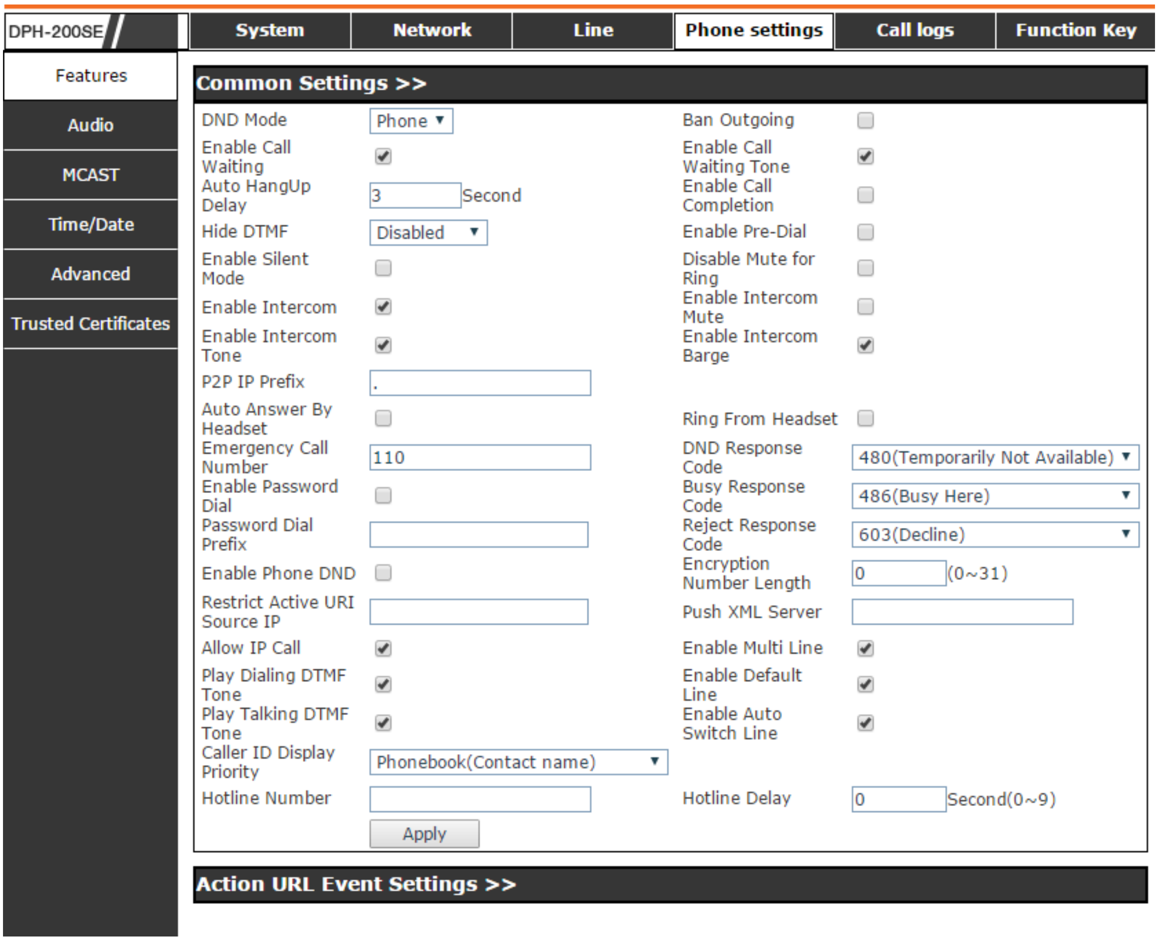
Basic Settings	
Field Name	**Explanation**
SIP Settings	
Local SIP Port	Set the local SIP port used to send/receive SIP messages.
Registration Failure Retry Interval	Set the retry interval of SIP REGISTRATION when registration failed.
STUN Settings	
Server Address	STUN Server IP address
Server Port	STUN Server Port – Default is 3478.
Binding Period	STUN blinding period – STUN packets are sent at this interval to keep the NAT mapping active.
SIP Waiting Time	Waiting time for SIP. This will vary depending on the network.

SIP Line Using STUN(SIP1 or SIP2)	
Use STUN	Enable/Disable STUN on the selected line.
TLS Certification File	
Upload or delete the TLS certification file used for encrypted SIP transmission.	
Note: the SIP STUN is used to achieve the SIP penetration of NAT, is the realization of a servi when the equipment configuration of the STUN server IP and port (usually the default is 3478), and select the Use Stun SIP server, the use of NAT equipment to achieve penetration.	

5.3.4 Phone Setting

8.3.4.1 Features

In this web page, you can configure Hotline, Call Transfer, Call Waiting, Black List, white list Limit List and so on.



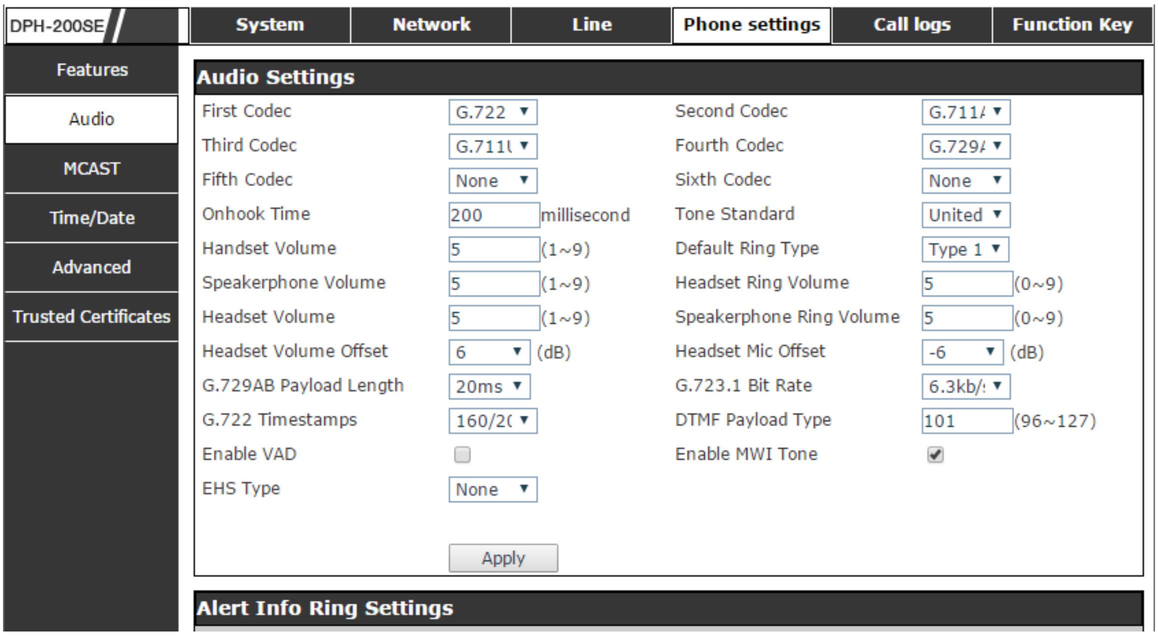
Common Settings	
Field Name	**Explanation**
DND Mode	DND might be disabled phone for all SIP lines, or line for SIP individually

	But the outgoing calls will not be affected
Ban Outgoing	If enabled, no outgoing calls can be made.
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Waiting Tone	Turn off this feature, and you will not hear a 'beep' sound in talking mode when there is another incoming call
Auto HangUp Delay	Set the Auto HangUp Delay time.
Enable Call Completion	Enable Call Completion by selecting it.
Hide DTMF	Specify the hide DTMF mode.
Enable Silent Mode	Enable Silent Mode by selecting it, the phone light will red blink to remin that there is a missed call instead of playing ring tone.
Disable Mute for Ring	Disable Mute for Ring
Enable Intercom	Enable Intercom by selecting it
Enable Intercom Mute	If enabled, mutes incoming calls during an intercom call.
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the interco call during a call. If the current call is intercom call, the phone will reject the second intercom call.
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you wan 192.168.1.119, I y d P f I P ou a define y 2Pd o P # refix s reach 192.168.1.119. Default is ".". If there is no "." Set, it means to disable dialing IP.
Auto Answer By Headset	When this item is checked, the device will auto-answer phone calls by headset if the auto-answer or intercom is enabled.
Ring From Headset	Enable Ring From Handset by selecting it, the phone plays ring tone handset.
Emergency Call Number	Specify t E he C mergency N D t all umber. y c espite dial the emergency call number.
DND Response Code	Specify DND Return code.
Enable Password Dial	Enable P assword D ial s i W n y electing i b t, w hen the password prefix, the following N numbers After the password prefix will be hidden as *, N stand for the value which you enter in the Password Length

	field. For example: you set the password prefix is 3, enter Length is 2, then you enter the number 34567, it will display 3**67 on phone.
Busy Response Code	Specify Busy Return Code.
Password Dial Prefix	Specify the prefix of the password call number.
Reject Response Code	Specify Reject Return Code.
Enable Phone DND	Enable Phone DND
Encryption Number Length	Set the Encryption Number Length.
Restrict Active URI Source IP	Specify the server IP that remote control phone for corresponding operation.
Push XML Server	Specify the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
Allow IP Call	Set the Enable or Disable IP Call.
Enable Multi Line	Set the Enable or Disable Multi Line.
Play Dialing DTMF Tone	Set the Enable or Disable Play Dialing DTMF Tone.
Enable Default Line	Set the Enable or Disable Default Line.
Play Talking DTMF Tone	Set the Enable or Disable Play Talking DTMF Tone.
Enable Auto Switch Line	Set the Enable or Disable Auto Switch Line.
Caller ID Display Priority	Set the Caller ID Display Priority
Hotline Number	Set the Hot line Number
Hotline Delay	Set the Hot line Delay time.
Action URL Event Settings	
Specify the Action URL that Record the operation of phone; send this corresponding information to server, url: http://InternalServer /FileName.xml? (Internal Server is server IP. Filename is nam xml that contains the action message).	

5.3.4.2 Audio

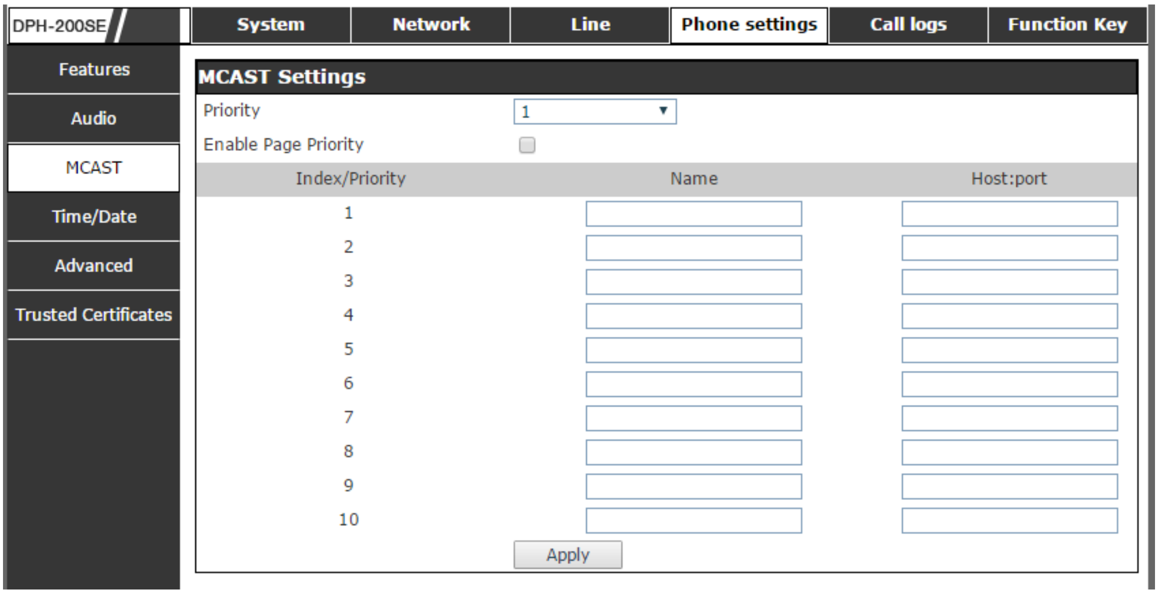
In this page, you can configure voice codec, input/output volume and so on.



Audio Setting	
Field Name	**Explanation**
First Codec	The first codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB
Second Codec	The second codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Third Codec	The third codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Fourth Codec	The forth codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Fifth Codec	The third codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Sixth Codec	The forth codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None
Onhook Time	Specify the least reflection time of Hand down, the default is 200ms.
Tone Standard	Configure tone standard area.
Handset Volume	Set the Headset calls the volume level.
Default Ring Type	Ring Sound – There are 9 standard types and 3 User types.
Speakerphone Volume	Set the speaker calls the volume level.
Headset Ring Volume	Set the Headset ring the volume grade.
Headset Volume	Set the headset calls the volume level.
Speakerphone Ring Volume	Set the speaker ring the volume grade.
Headset Volume Offset	Set the headset the Volume the Offset.
Headset Mic	Set the headset MIC the Offset.

Offset	
G.729AB Payload Length	G.729AB Payload Length – Adjusts from 10 – 60 m Sec.
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s.
G.722 Timestamps	Choices are 160/20ms or 320/20ms.
DTMF Payload Type	Choices are 160/20ms or 320/20ms.
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload length cannot be set greater than 20 m Sec.
Enable MWI Tone	Enable MWI Tone by selecting it

5.3.4.3 MCAST



It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, monitor and play the RTP stream which sent by the multicast address.

MCAST Settings

Equipment can be set up to monitor up to 10 different multicast addresses, us receive the multicast RTP stream sent by the multicast address.

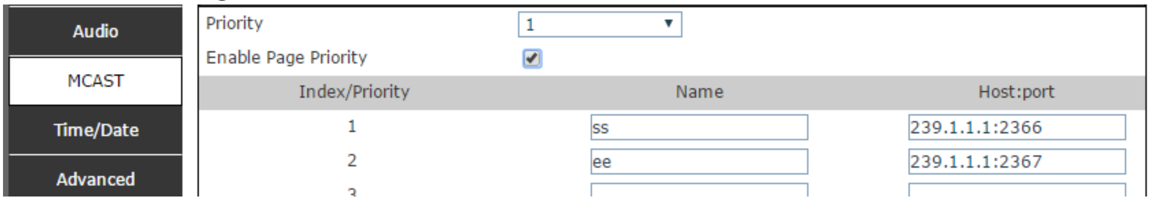
Here are the ways to change equipment receiving multicast RTP stream proc mode in the Web interface: set the ordinary priority and enable page priority.

- Priority:

In t he rop-down c p ox o o hoose t p riority t p f o rdinary the incoming flows of multicast RTP, lower precedence than the current comm device w a ill uitomaticatllyg R s gnolret p he o t roiup fTP tream of m ulticaRt i h TP t cs iglher c hanp hed urwenta ommon receive the group RTP stream, and keep the current common calls in state. You can a choose to disable in the receiving threshold drop-down box, the device will automatically ignore all local network multicast RTP stream.

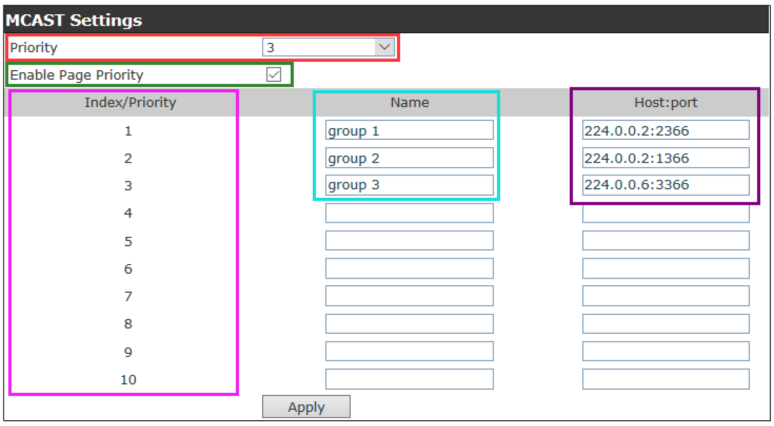
- The options are as follows:
  - ◇ 1-10: To definite the priority of the common calls, 1 is the top level while 10 is the lowest
  - ◇ Disable: ignore all incoming multicast RTP stream
  - ◇ Enable the page priority:

    Page priority determines the device how to deal with the new receiving multicast RTP stream when it is in multicast session currently. When Page priority enabled, t d hw a evice i ill t lutopaticallym R s gnore b he receive top-level priority multicast RTP stream, and keep the current multicast session in state; If it is not enabled, the device will automatically ignore all receiving multicast RTP stream.
- Web Settings:



The multicast SS priority is higher than that of EE, which is the highest priority.

Note: when pressing the multicast key for multicast session, both multicast sender and receiver will beep.



- **Blue part (name)**

  "Group 1","Group 2" and "Group 3" are your setting monitoring multicast name.The group name will be displayed on the screen when you answer the multicast. If you have not set, the screen will display the IP: port directly.
- **Purple part (host: port)**

It is a set of addresses and ports to listen, separated by a colon.

- **Pink part (index / priority)**

Multicast is a sign of listening, but also the monitoring multicast priority. The smaller number refers to higher priority.

- **Red part (priority)**

It is the general call, non multicast call priority. The smaller number refers to high priority. The followings will explain how to use this option:

- ◇ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" launched a multicast call.
- ◇ All equipment has one or more common non multicast communication.
- ◇ When you set the Priority for the disable, multicast any level will not answer multicast call is rejected.
- ◇ when you set the Priority to a value, only higher than the priority of multicast can come in, if you set the Priority is 3, group 2 and group 3 for priority level equal to 3 and less than 3 were rejected, 1 priority is 2 higher than ordinary call priority device can answer the multicast message at the same time, keep the hold the other call.

- **Green part (Enable Page priority)**

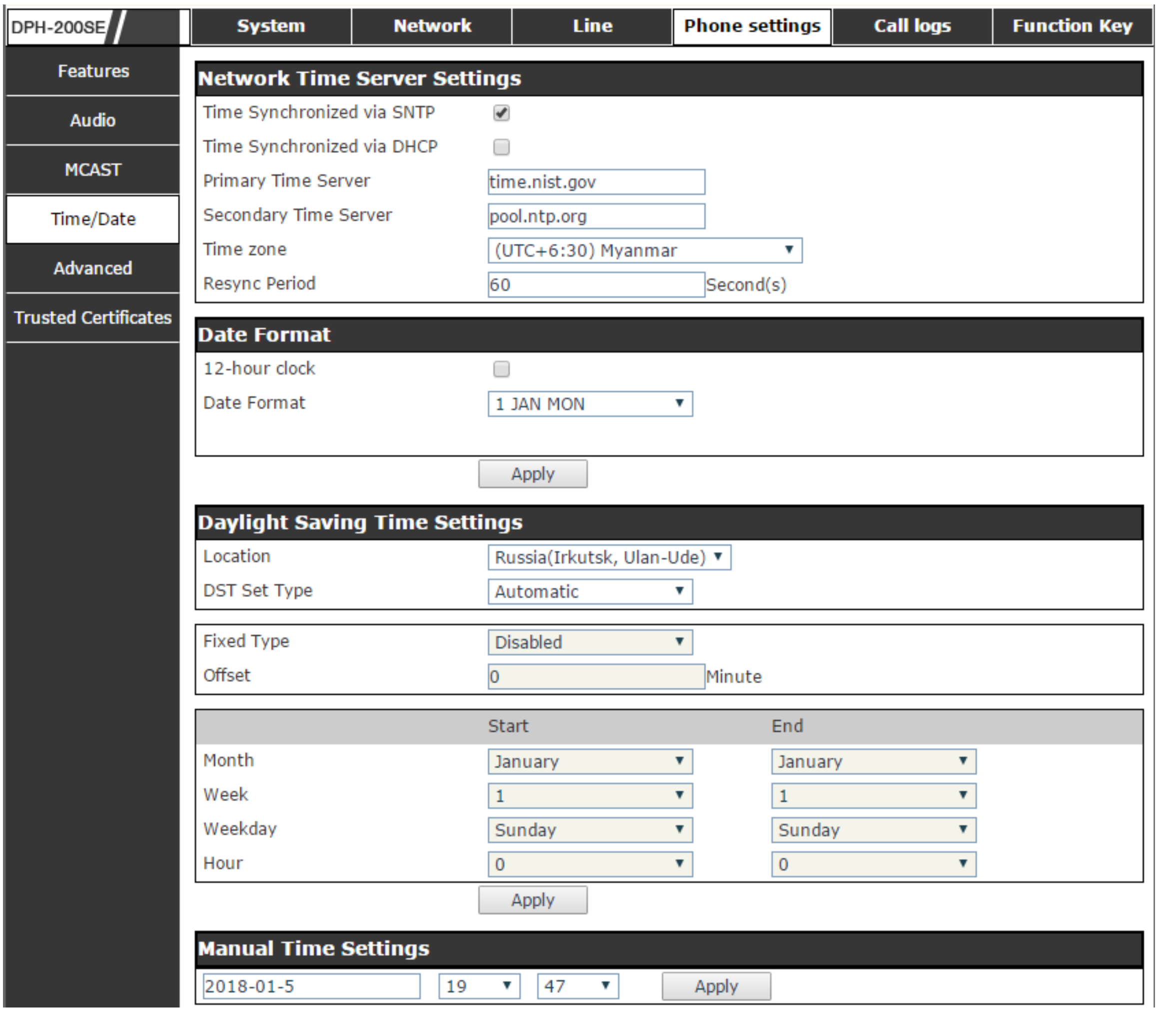
Set whether to open more priority is the priority of multicast, multicast is pink part number. Explain how to use:

- ◇ The purpose of setting monitoring multicast "group 1" or "3" set up listening "group of 1" or "3" multicast address multicast call.
- ◇ All equipment has been a path or multi-path multicast phone, such as listening to "multicast information group 2".
- ◇ If multicast is a new "group of 1", because "the priority group 1" is 2, higher than the current call "priority group 2" 3, so multicast call can come in.
- ◇ If multicast is a new "group of 3", because "the priority group 3" is 4, lower than the current call "priority group 2" 3, "1" will listen to the equipment and maintain the "group of 2".

Multicast service

- **Send:** when configured ok, our key press shell on the corresponding equipment, equipment directly into the Talking interface, the premise is to ensure no current multicast call and 3-way of the case, the multicast can be established.
- **Lmonitor:** IP port and priority configuration monitoring device, when the call is initiated and incoming multicast, directly into the Talking interface equipment.

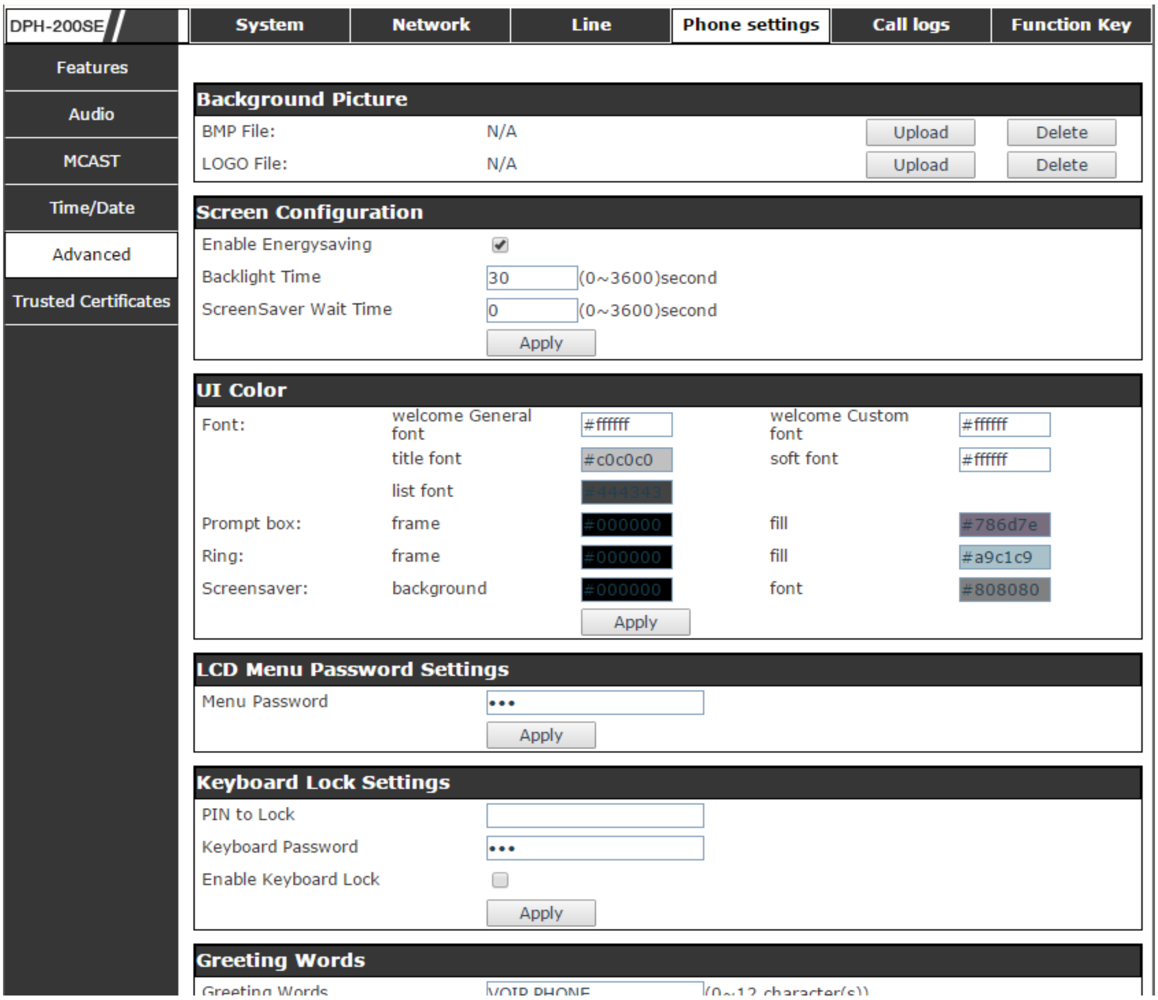
5.3.4.4 Time/Date



Time/Date	
Field Name	**Explanation**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time zone	Select the time zone
Resync Period	Time of re-synchronization with time server
Date Format	
12-hour clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format

Daylight Saving Time Settings	
Location	Select the user's time zone specific area
DST Set Type	Select automatic DST according to the preset rules of DST, or the manually input rules
Offset	The DST offset time
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Manual Time Settings	
The time set by hand, need to disable SNTP service first.	

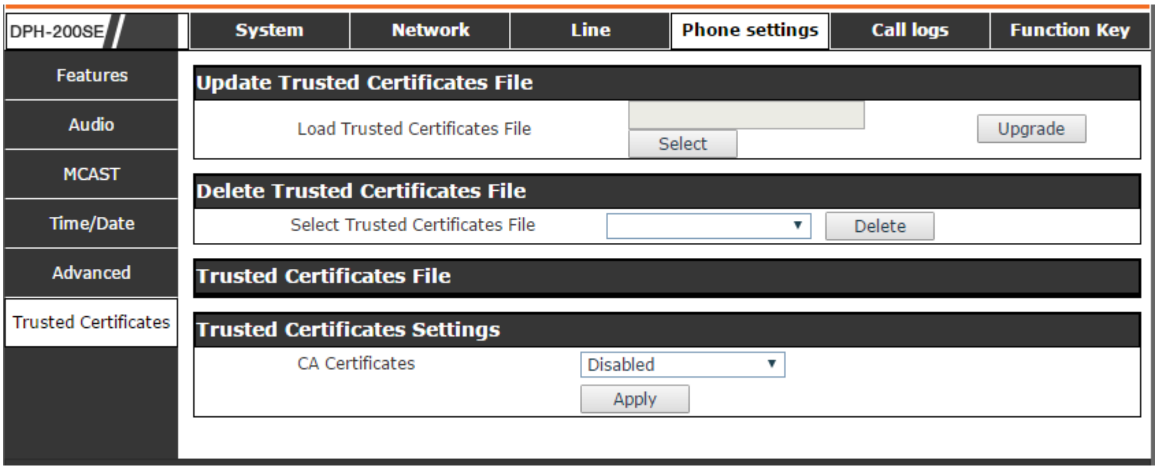
5.3.4.5 Advanced



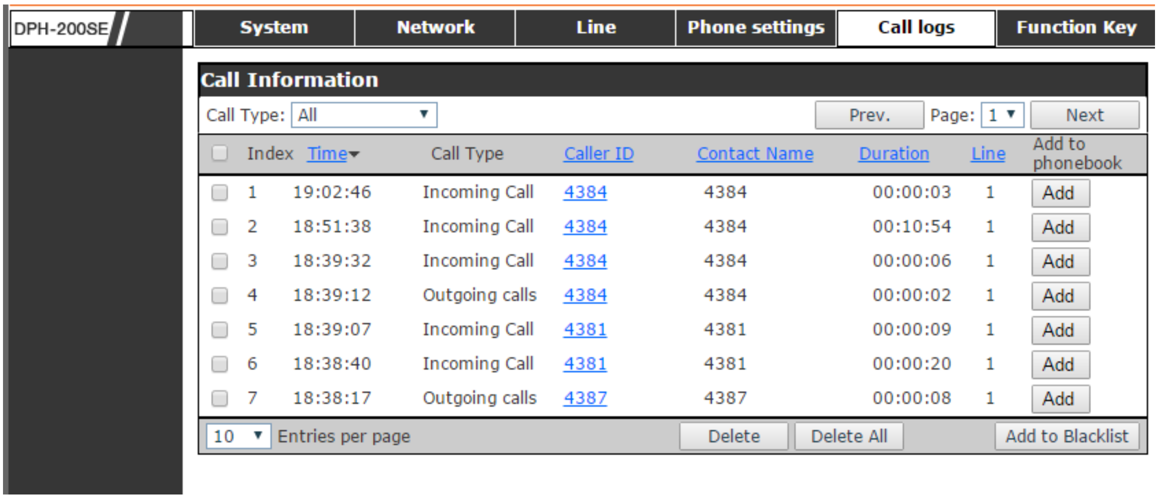
Advanced	
Field Name	**Explanation**
Screen Configuration	
Enable Energysaving	Enable Energysaving by selecting it.
Backlight Time	Set the Backlight Time.
LCD Menu Password Settings	
Menu Password	Set the password for entering the Advanced setting menu of the phone. The password is digit. The password is 123 by default.
Keyboard Lock Settings	
PIN to Lock	Set the PIN to Lock.
Keyboard Password	Set the password for entering the setting menu of the phone by the phone's key board. The password is digit.
Enable Keyboard Lock	Enable Keyboard Lock by selecting it.
Greeting Words	
The greeting message will display on the top left corner of the LCD when the device is idle, which is limited to 16 characters. The default chars are 'VOIP PHONE'.	

5.3.4.6 Trusted Certificates

User may Update or Delete Certificates File in this webpage.



5.3.5 Call logs



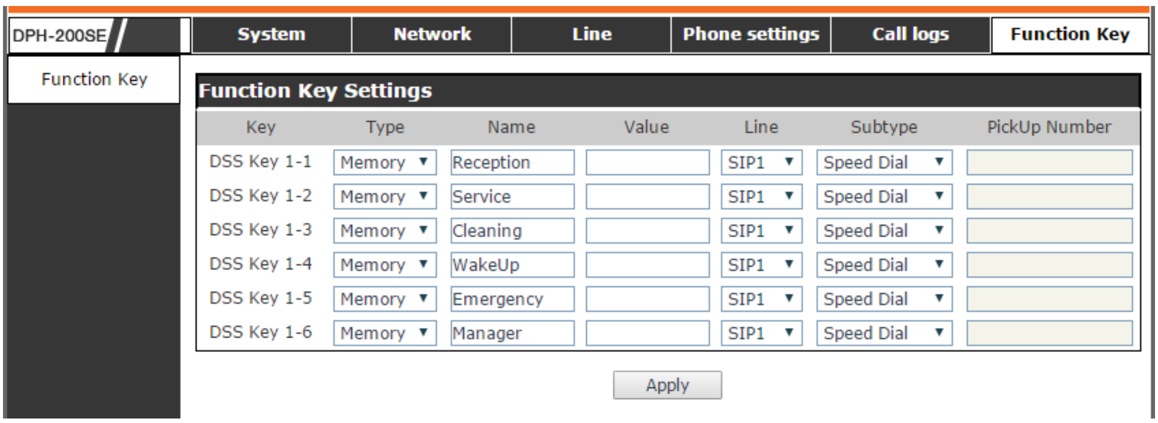
User can browse complete call logs in this page, order the call logs by time, ca contact n d ame, o l uration, a f rt ine,l b nd c ah t lsoi o ilter he missed, or all.

User can save a call log into his/her phonebook or add it to the blacklist.

User can also make web call by click on the number of a call log.

5.3.6 Function Key

The device provides 6 user-define DSS Keys at most. User may configure/customize each DSS key in this webpage.



Function Key	
Field Name	**Explanation**

Function Key Settings	
Memory Key	**SpeedDial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to conn phone quickly; it is widely used in office environments. **Call Park**: When the key is configured as Call Park, call by using the call park code. **Call forward**: When the key is configured as Call forward, you can transfer the call to the set number. **MWI**: When the key is configured as MWI, you are allow voicemail quickly by pressing this key.

6 Appendix

6.1 Specification

6.1.1 Hardware

Item		**DPH-200SE**
Adapter (Input / Output)		Input: 100-240V Output: 5V 1A DC
Port	WAN	10/100Bace-T RJ45 1 PORT
	LAN	10/100Bace-T RJ45 1 PORT
Power Consumption		IDLE: 2.5W Active: 2.8W
LCD size		480x320,TFT color LCD 3.5”
Operation Temperature		0~40℃
Relative Humidity		10~65%
CPU		Broadcom VoIP chipset
SDRAM		16MB
Flash		8MB
Dimension(L x W x H)		29 x 26 x 6 cm
Weight		1.16Kg

6.1.2 Voice features

- HD voice: HD handset
- Codec: G.711A/U, G.723.1 high/low, G.729AB, G.722, G.726-32
- DTMF: in-band, RFC2833 and SIP INFO
- Full-duplex Acoustic Echo Canceller (AEC) – Hands-free Mode, 96ms tail-length
- Voice Activity Detection (VAD) / Comfort Noise Generation (CNG) / Background Noise Estimation (BNE) / Noise Reduction (NR)
- Packet Loss Concealment (PLC)
- Dynamic Adaptive Jitter Buffer up to 300ms
- DTMF: In-band, Out-of-Band – DTMF-Relay(RFC2833) / SIP INFO
- Call transfer (unattended/ attended/ semi-attended)

- Call holding
- Call waiting
- Redial
- Call completion
- Predial
- MWI
- Flexible dial plan
- Barring function for outgoing calls
- Do not disturb
- Auto answer
- CLIR(rejects anonymous calls)
- CLIP(to make an anonymous call)
- Dial without registration
- Call logs with missed calls/ incoming calls/ outgoing calls. Each support 300 records(Web)
- Speed Dial
- Hotline/Warm-line
- Password dial
- DTMF hidden
- Action URL/ active URI
- Multicast
- Web dial
- Emergency call

6.1.3 Network features

- WAN / LAN: 10/100M Ethernet ports (Bridge mode)
- IP Configuration: Static / DHCP / PPPoE
- Network Access Control: 802.1x
- VPN: L2TP (Basic Unencrypted) / OpenVPN
- VLAN
- QoS

6.1.4 Maintenance and management

- Auto-Provisioning via FTP/TFTP/HTTP/HTTPS/DHCP OPT66/SIP PNP/TR069
- Web Management Portal
- Web-based Packet Dump

- Configuration Export / Import
- Firmware Upgrade
- Syslog

6.2 Digit-character map table

Keypad	Character	Keypad	Character
1		7 PQRS	7 P Q R S p q r s
2 ABC	2 A B C a b c	8 TUV	8 T U V t u v
3 DEF	3 D E F d e f	9 WXYZ	9 W X Y Z w x
4 GHI	4 G H J g h i	*.	* .
5 JKL	5 J K L j k l	0	0
6 MNO	6 M N O m n o	#	#