USER MANUAL

DPH-150S/SE

VERSION 1.0

VOIP

1. General information

1.1. Package list

DPH-150SE (with PoE) or DPH-150S(without PoE) IP Phone unit	1
Handset	1
Handset Cord	1
Power Adapter	1
Ethernet Cable	1
Quick Install Guide	1
CD-ROM	1

1.2. Power supply

- Power adapter: AC/DC Input 110-220V Output 9V~12V@0.8A, with maximum power consumption of 6 Watt.
- POE : DC/DC Input 48V Output 12V@0.5A.

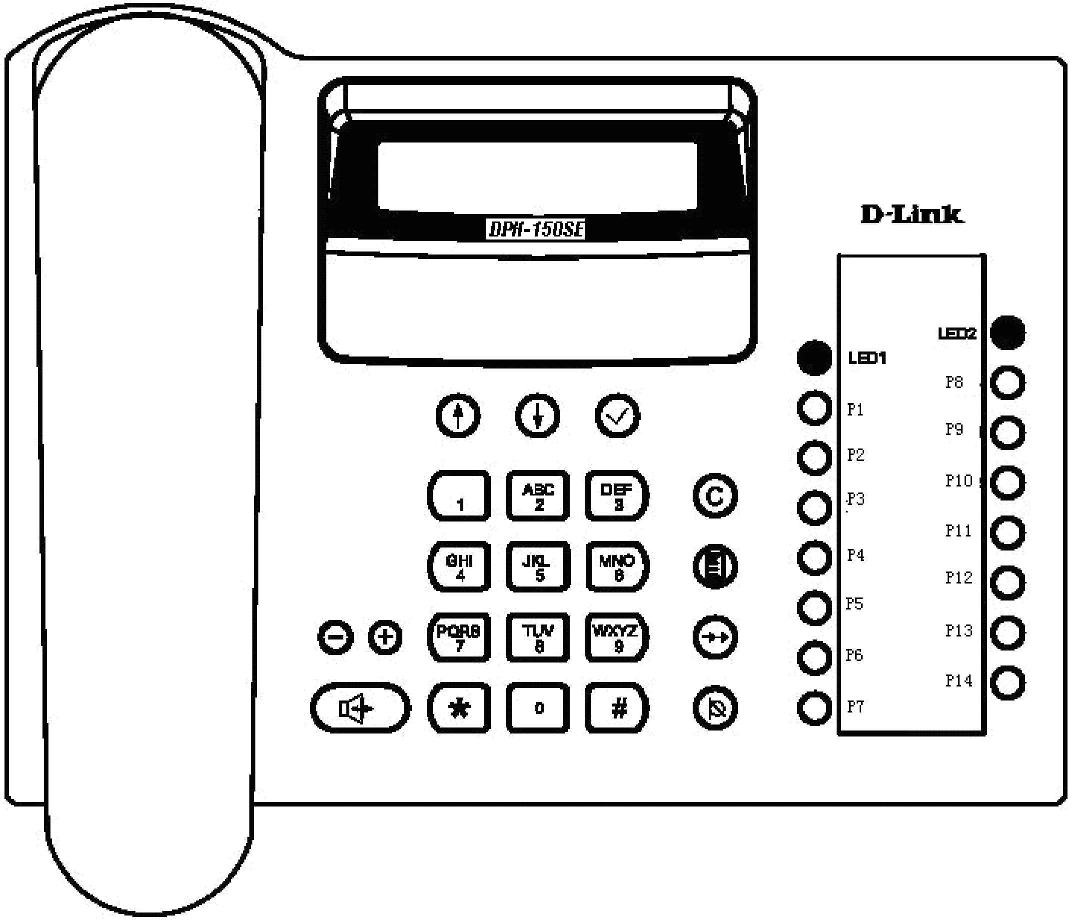
1.3. Environment condition

- Environmental temperature: -10 ºC ~ + 40ºC
- Relative humidity: 10% ~ 95%
- Environmental noise: <= 60 dB(a)
- Air pressure: 86 ~ 106 kPa

2. Your phone

2.1. Using the keys

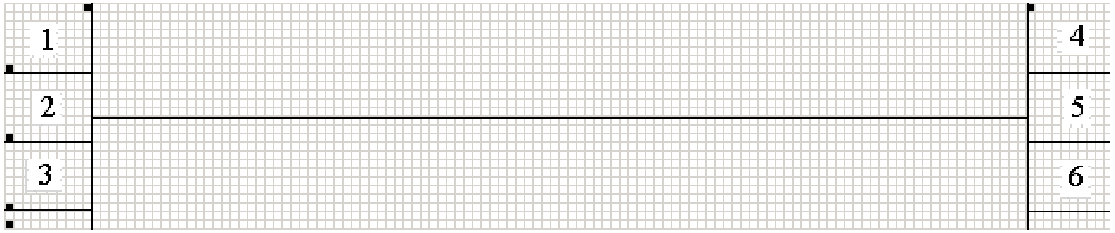
2.1.1 Telephone appearance



2.1.2 LCD

The information of system, user, etc., will be displayed clearly in two lines. (128x26 Dot matrix)

On the LCD, there are 6 icon fields.

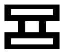


LCD will leave 20 columns for displaying icons, 10 columns on each side. Each icon size is 10x8 . So there’re six positions, numbered from No.1 to No.6, on LCD for indication of different status.

- Region No.1: Ethernet Link indication
- Region No.2: Call Forward indication
- Region No.3: Missed call indication
- Region No.4: DND/Mute indication
- Region No.5: Line1 status indication
- Region No.6: Line2 / Alarm status indication

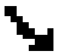
The 6 field icons are described as following:

- **Ethernet Link indicator**

  If network connector is ok, icon  will be shown at Region No.1.

  Else if network connector is disconnected, icon  will be shown.

- **Call forward**

  When call forward is set, icon  will be shown at Region No.2.

- **Missed call**

When there’re new missed calls (have not read), icon will be shown at the Region No.3.

- **DND/Mute**

If Mute is activated (only in talking), icon will be shown at region 4.

If DND is activated and not in talking state, icon will be shown at Region No.4.

- **Line1 status icon**

If the line1’s status is Call waiting, icon will be shown at region No.5.

If the line1’s status is Call holding, icon will be shown at region No.5.

- **Line2 status icon**

If the line2’s status is Call waiting, icon will be shown at region No.6.

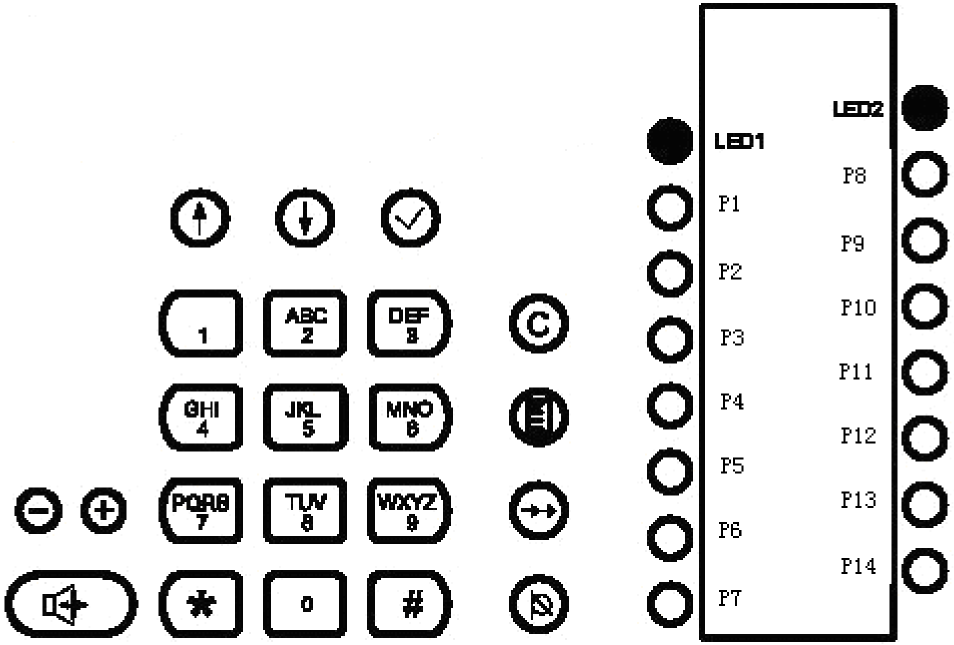
If the line2’s status is Call holding, icon will be shown at region No.6.

- **Alarm status icon**

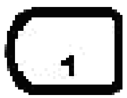
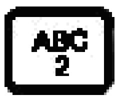
If Alarm is valid (have set alarm and any alarm time is not reached) and the state is on hook or menu operation or view missed/received/dialed calls, the alarm icon will be displayed. If hook off, the alarm icon will be off until the state returns to on hook. If an alarm time is reached and none other alarm, the alarm icon will be off.

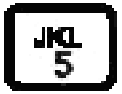
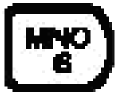
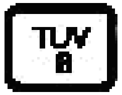
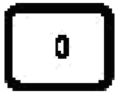
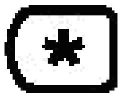
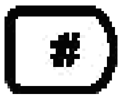
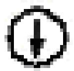
2.1.3 Key board

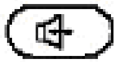
As shown in Figure DPH-150S(SE) total has 36 keys.



Their definitions are given in Table

Key button	Key definition	Function description
	1 . , : ; / - _ @ & ? ¿ ! ¡ ‘ “	
	2 a b c A B C	
	3 d e f D E F	

	4 g h I G H I	
	5 j k l J K L	
	6 m n o M N O	
	7 p q r s P Q R S	
	8 t u v T U V	
	9 w x y z W X Y Z	
	0 blank key	
	. * + / （ ）< = > % ¥ ₤ $	
	**Dial finish key/switch input mode**	**Dial finish key/switch input mode**
	shift key	1. Up arrow key; 2. Left arrow key;
	shift key	1. Down arrow key; 2. Right arrow key;

	Ok	Confirm or enter
	delete	Cancel or delete
	menu	Go to menu
	Redial key	Redial or show dialed list
	mute	Mute/Unmute the call
	Adjust volume	Reduce handset/handfree/headset volume
	Adjust volume	Enhance handset/handfree/headset volume
	handfree	On/Off handfree
P1~P14	Function key	Program keys(Function/Memory keys)

2.1.4 Text input definitions

The telephone has three text input methods: “ABC” and “abc” and “123”. Where “ABC” means uppercase character input, “abc” means lowercase character input and “123”means numeric character.

You can switch three input methods by press “#” key.

2.1.4.1 “abc” character input mode

In “abc” input mode, user can input lowercase characters. The input character will change according to the pressing times. For example, if user presses 2 times on ABC 2, it will be ‘b’. The definition of the key in “abc” method is shown in Table.

Definition of the key in “abc” method

Key	Characters in “abc” input method
1	. , : ; / - _ @ & ? ! \ " 1
ABC 2	a b c 2
DEF 3	d e f 3
GHI 4	g h i 4
JKL 5	j k l 5
MNO 6	m n o 6
PQRS 7	p q r s 7
TUV 8	t u v 8
WXYZ 9	w x y z 9
*	* + ( ) < = > % $
0	Space 0

#	Input method switch key

2.1.4.2 “ABC” character input mode

In “ABC” input mode, user can input uppercase characters. The input character will change according to the pressing times. For example, if user presses 2 times on ABC 2, it will be ‘B’. The definition of the key in “ABC” method is shown in Table.

Definition of the key in “ABC” method

Key	**Characters in “ABC” input method**
1	. , : ; / - _ @ & ? ! \ " 1
ABC 2	A B C 2
DEF 3	D E F 3
GHI 4	G H I 4
JKL 5	J K L 5
MNO 6	M N O 6
PQRS 7	P Q R S 7
TUV 8	T U V 8

WXYZ 9	W X Y Z 9
*	* + ( ) < = > % $
0	Space 0
#	Input method switch key

2.1.4.3 “123” character input mode

In “123” input mode, user can input numeric characters.. The definition of the key in “123” method is shown in Table.

Definition of the key in “123” method

Key	Characters in “ABC” input method
1	1
ABC 2	2
DEF 3	3
GHI 4	4
JKL 5	5
MNO 6	6

PQRS 7	7
TUV 8	8
WXYZ 9	9
*	.
0	0
#	Input method switch key

2.2. Display

2.2.1 Cursor

Cursor is shown as “I”， which lies at the position where you want to input . After you input a character, the cursor should locate in the right of the inputted character where the next input position is.

2.2.2 Display in view mode

In view mode, character displays from left side of LCD. When total length exceeds the max LCD content, it will show the first **18 characters.**

```
xu feng
```

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Display in the view mode

2.2.3 Display in edit mode

In edit mode, character is input from left to right. If the length exceeds the max LCD content, characters at the left of cursor should move to left side of LCD one by one.

3895499I

xu xxxxI

Display in the edit mode

2.2.4 Insert a character

Move the cursor to position where you want to insert a character. After input a character, all characters at the right of new input character will automatically move in right direction by one position. If total length reaches the buffer’s max content, then no characters can be further input.
To delete a character, user should move the cursor to the right side of position, and press [delete] key, the character before the cursor will be deleted. And all characters at the right of cursor will automatically move left by one position.
Note: If cursor is at the most left side of LCD, pressing [delete] key will delete the character at current position.

I 3895499

Display in the insert mode

2.2.5 Rapid delete

Press [Delete] key for a long time (> 2 Seconds), will cause to delete all current characters.

2.2.6 Cursor advance

In abc mode or ABC mode, user can input by rapidly press the same key. If the interval >=600 ms, cursor will advance.

2.2.7 Exit edit mode

When there’s no character left on LCD, press [delete] key will exit edit mode.

2.3. Using the menu

2.3.1.Menu operation

Press [menu] key can enter menu operation in normal state, When view missed/dialed/received call list or in menu operation, press [menu] key will quit to normal state. Entering menu operation, the LCD displays the first 2 menus (Phone book / Call records, and the line of Phone book is reverse display).
User can press [Up] / [Down] to view the menu item one by one.
In the edit mode, for example, when user input name for phone book. If the character number reaches the max length, no inserting is permitted while user presses any key. When there’s a question, ask for whether continue or cancel current operation, user can press [OK] to continue, or [Delete] to cancel. When there’s a prompt, user can press [Delete] key to exit the prompt rapidly.
In the edit mode, when input exceeds the max length, then it will not accept any input from user. It will exit menu automatically after 30 Seconds, if there’s no action on the menu.

2.3.2.Menu tree view

The menu tree total has 9 sub-menus, just as Figure 2-1 shown.

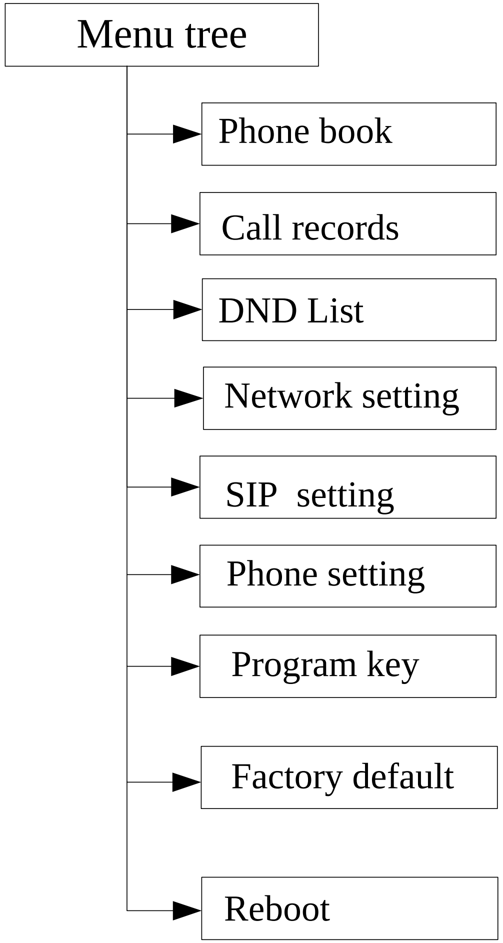
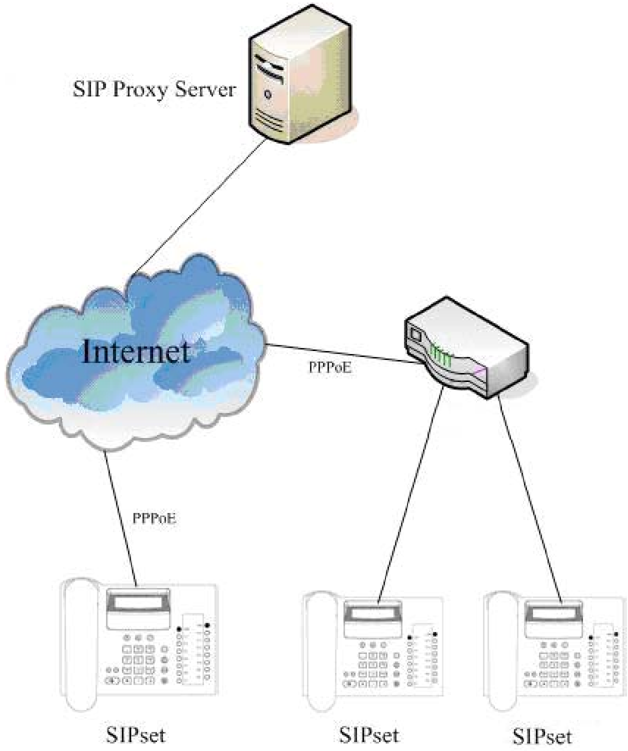


Figure 2-1 menu tree

3. Getting started

3.1. Prepare network settings

Plug reticle to Network RJ45 interface. The other side connects the SIP server's networks.



3.2. Prepare VoIP settings

1. Click  key, LCD will display config menu.

2. Select “SIP setting”→“Proxy setting”→“PxyServ”, input Proxy Server IP Address, Click to save configuration.

3. Select “SIP setting”→“Proxy setting”→“RegServ”, input Register Server IP Address, Click to save configuration.

4. Select “SIP setting”→“User Info”→“PhoneNum”, input the phone number, Click to save configuration.

5. The configuration will be effective at once. There is no need to reboot.

4. Basic functions

4.1. Making a call

You have several ways of making calls. You can:

- lift the handset, activate a headset if one is connected to your telephone, or use the Speaker,
- use an automatic dial feature like speed dial, redial, or calling directly from the Call Log.

4.1.1 Dialing

Dial a call manually by:

- OFF-HOOK or Speaker dialing.

In idle state, lift the handset, enter the call number, press ‘#’ to dial out or the call will be made automatically in 5 seconds. And the call number will be shown in LCD. You can press to delete the digits input one by one, or delete all by press for more than 2 seconds.

- Dial from call records

Press and to enter ‘Call records’, select the call list(Received calls/Dialed calls/Missed calls)you want. Press or to reach the desired number or name, press and then press ‘Dial out’ to call.

- Dial from the repertory keys

7 repertory keys can be set for the phone. Lift the handset or press the handfree key, then press the repertory key to call.

- Dial from phone book

Enter into menu of ‘phone book’, choose ‘view entry’, use down/up to select the number you want to dial, or you can choose ‘search entry’, input the name you want to search, then dial out.

- Redialing a previously called number use down/up key.
- Dialing during a call

The phone supports double line calling. You can make a new call without end the last one. Press ‘FLASH’ to hold the current conversation, and make a new call.

- Exit dialing

Replace the handset or press  to exit dialing. If you are in double line calling, when you exit the current line, the phone will ring to remind you the other call holding.

4.1.2. Handset/Hands free/Headset switching

There are four states during handset/headset/Speaker switch : handset onhook、handset offhook、handfree and headset.

Following are the rules for handset/headset/Speaker switch:

1. If current state is not handfree, phone will change to handfree state after users have pressed ‘handfree’ function key.
2. If current state is not headset, phone will change to headset state after users have pressed ‘headset’ function key.
3. If current state is handfree, users have pressed 'handfree' function key with the handset is off-hooked, phone will change to handset offhook state and continue the current conversation. And users have pressed 'handfree' function key with the handset is on-hooked, phone will change to handset onhook state and end the current conversation.
4. If current state is headset, users have pressed 'headset' function key with the handset is off-hooked, phone will change to handset offhook state and continue the current conversation. And users have pressed 'headset' function key with the handset is on-hooked, phone will change to handset onhook state and end the current conversation.
5. If current state is handset offhook, phone will change to handset onhook state after users have put down the handset.
6. If current state is handfree or headset, phone will not change its state after users have put down the handset.
7. Whenever phone will change to handset offhook state after users have picked up the handset.

4.1.3. Ending a call

Users can just hook on or handfree off to end the current call.
If the line2 is hold, and user end the line1's call by hooking on, the phone will automatically ring.

4.2. Answering a call

4.2.1. By operation menu

When the phone rings, press to display the operation menu. You can 'Accept' or 'Reject' the call. Also you can press to exit the menu.

4.2.2. Answering a call in idle state

When the phone rings, pick up the handset 、press headset key or press to answer.

4.2.3. Answering a call during another conversation

When receiving a new call during your conversation, you can press to display the operation menu, and 'Accept' or 'Reject' the call. Also you can replace the handset or press to end the current call. And answer the new one while the phone ring soon afterwards.

4.3. Muting a call

You can mute the phone by press while talking. And press again to continue the call.

Note:

1. If press [mute] key to mute the phone, the phone will not clear the mute function until press the [mute] key again (press [Flash]/[Conference]…keys can not unmute).
2. If Mute function is activated when talking, phone will automatically clear mute if phone go to idle state (clear mute icon display).

4.4. Holding/resuming a call

- To hold a current line, you should press [Flash] function key, the other end of the hold line will not hear any voice of local end.
- If you want to resume the call, please press [Flash] function key again, then you can go on the call.

4.5. Switching between active line

While talking, press the ‘Flash’ key which could be set in ‘Program keys’ in the operation menu to switch between active lines.

4.6. Transferring a call

To transfer a call, press the ‘Flash’ key and the number of the third party with ‘#’ for ending. If the phone is answered, you can press the “transfer” key to replace the handset and the call is delivered successfully. If the line is busy or there is no one answered, you can press the ‘Flash’ key to return.

For detail information，please refer to 7.1.2 Transfer

4.7. Setting up a conference call

4.7.1. Start a conference call

To make a conference call, press the ‘Flash’ key to hold the current call, press the number of the third party with ‘#’ for ending. When it is answered, press the ‘Conference’ which could be set in ‘Program keys’ in the operation menu to begin the conference.

Or press the ‘conference’ key to hold the current call , press the number of the third party . When it is answered, three users will be in the conference status immediately.

4.7.2. Hold a conference call

During the three-party conference, pressing the [CONF] key will cancel the conference, and show the two line’s phone number on the LCD, user can use the [OK], [UP] and [DOWN] keys to select an connecting line, while another line will be hold.

4.7.3. End a conference call

User can just hook on or handfree off to end the conference if the conference is started by the user (user press [conference] to start a conference). In this situation, the both two active lines will be cleared; the end of the active line will get busy tone.

Note:

If an active line **(**not the conference starter) disconnect from the conference, the other sides in the conference continue talking. (For example: In conference, line1 hook on, the local user can continue talking with Line2)

For detail information，please refer to 7.1.3 Conference

4.8. Volume adjustment

Handset / handfree / Headset volume have 8 levels (1~8) separately. User can adjust the volume by pressing [Volume up] / [Volume down] key in handfree/speaker/headset state.

5. Menu operation

5.1. Phone Book

DPH-150S's (DPH-150SE's) phone book can support up to 100 entries. And all these entries can be added ,modified ,deleted or dialed. If phone book is full, ip phone will alert when user want to add new entry.
When incoming call or outgoing call's number matches an entry in the phone book, the name of entry will be displayed instead of number.

5.1.1. View entry

User can view the entries in the current phone book one by one by selecting "view entry " menu item. It will show two lines at a time, and the selected line is reversed display. All entries will be sorted by ASCII ascending order. If phone book is empty, "No Record" will display on LCD.

- press to see the list,
- press or to select the record,

When you find the record, you can do the operations as below:

Dial out: select and press to call.
Modify entry: select and press to modify, and press to save.
Delete entry: select and press to delete the record.
Detail: select and press to display the name and number accordingly.
Add to DND: select and to set the name and number, and press to save.

5.1.2. New entry

User can add a new entry by selecting "New entry". The max length of name will be 18 for ASCII characters, the max length of num will be 32 digits.

- Input the name, then press
- Input the corresponding call number, then press to save.

Name or call number can not be empty, and name is the main key for storing in the phone book, so it is not permitted that two phone book items have the same name.

It will search through all phone book entries to find whether there exists an entry with the same name. If it is, it will prompt user whether he prefers to overwrite the old entry. Press [OK] can overwrite the old entry, press [Delete] can cancel the overwrite.

5.1.3. Search entry

- press to enter, input the name you want to find, press to search. Then you can press or to check the corresponding records.

For example: The phonebook has a entry (name: Abcdef, num: 123456), if inputted searching key is “Abc”, phone can display the entry (name: Abcdef, num: 123456). It means that if some first characters of any entry match the searching key, the entry is most matching entry.

When search key is empty or no matching entry is found, then it will show from first entry. If phone book is empty, whatever inputted search key, “No Record!” will display on LCD.

When you find the record, you can do the operations as below:

Dial out： select and press to call.

Modify entry: select and press to modify, and press to save.

Delete entry： select and press to delete the record.

Detail： select and press to display the name and number accordingly.

Add to DND： select and to set the name and number, and press to save.

5.1.4. Memory check

- Press to enter, you can see how many used, and how many free.

5.1.5. Delete all

Delete all of the entries if you don't want to save these records.

When LCD display “delete ok…”, the phone is deleting the entries, press [Delete] key will not have any action until the phone display the menu.

5.2. Call records

The phone provides function to record at most 20or each of dialed /missed /received calls. And the most recent call is the first entry in the list. If the call's number matches an entry in the phone book, then the name will be displayed instead of number. If more than one item with the same number occurs several times one by one, then only the last one is recorded. And you can dial or save the number to the phonebook or DND list.

5.2.1. Missed Calls

Press or to check all the missed phone numbers, then press to enter the operation menu.
Dial out: select and press to call.
Detail: select and press to display the name and number accordingly.
Add to book: select and to set the name and number, and press to save.
Add to DND: select and to set the name and number, and press to save.
Delete entry: select and press to delete the record.

5.2.2. Received calls

Press or to check all the received phone numbers, then press to enter the operation menu:
Dial out: select and press to call.
Detail: select and press to display the name and number accordingly.
Add to book: select and to set the name and number, and press to save.
Add to DND: select and to set the name and number, and press to save.
Delete entry: select and press to delete the record.

5.2.3. Dialed calls

Press or to check all the dialed phone numbers, then press to enter the operation menu.
Dial out: select and press to call.
Detail: select and press to display the name and number accordingly.

Add to book: select and to set the name and number, and press to save.
Add to DND: select and to set the name and number, and press to save.
Delete entry: select and press to delete the record.

5.2.4. Delete all

Delete all of the records if you don't want to save these records.

5.3. DND List

DPH-150S's (DPH-150SE's) DND list can support up to 100 entries. And all these entries can be added, modified, deleted or dialed.

ALL DND LIST ENTRIES WILL BE SORTED BY NUM, SO THE SORT KEY IS DND NAME. AND THE DND NAME IS EXCLUSIVE

5.3.1. View entry

User can view the entries in the current DND list one by one by selecting "view entry" menu item. It will show two lines at a time, and the selected line is reversed display. All entries will be sorted by ASCII ascending order.

- press to see the list,
- press or to select the record,

When you find the record, you can do the operations as below:

Modify entry: select and press to modify, and press to save.
Delete entry: select and press to delete the record.
Detail: select and press to display the name and number accordingly.

5.3.2. New entry

User can add a new entry by selecting "New entry". The max length of name will be 18 for ASCII characters or 9 chinese characters, the max length of num will be 32 digits. If user press [Delete] / [Del_long] when the inputted content is empty, it exits back to "New entry" menu.

- Input the name, then press

- Input the corresponding call number, then press to save.

5.3.3. Search entry

To search a entry, user need to input the name of entry. It will find out the most matching entry according to the searching key, and display that entry. User can press [Up] / [Down] to view other entries before or after that entry.
For example: The DND records has a entry(name: Abcdef, num: 123456), if inputted searching key is "Abc", phone can display the entry(name: Abcdef, num: 123456). It means that if some first characters of any entry match the searching key, the entry is most matching entry. When search key is empty or no matching entry is found, then it will show from first entry. If DND list is empty, whatever inputted search key, "No Record!" will display on LCD.

- press to enter, input the name you want to find, press to search. Then you can press or to check the corresponding records.

When you find the record, you can do the operations as below:

Modify entry: select and press to modify, and press to save.
Delete entry: select and press to delete the record.
Detail: select and press to display the name and number accordingly.

5.3.4. Memory check

- Press to enter, you can see how many used, and how many free.

5.3.5. Delete all

Delete all of the entries if you don't want to save these records.
When LCD display "delete ok…", the phone is deleting the entries, press [Delete] key will not have any action until the phone display the menu.

5.4. Network setting

5.4.1. Set IP mode

- **DHCP**
  Obtain IP address using DHCP.
- **PPPoE**
  Obtain IP address using PPPoE.
- **FIXED**
  Pre-set static IP address.

5.4.2. IP address

Set valid IP address.(Can noit modify in DHCP or PPPoE mode)
The text input mode is ‘123’, and other input modes are not permitted. And the inputted IP address should observe IP address rules.
If the inputted IP address is conflicted with other IP address on Ethernet, the phone will display “IP Conflict” on LCD, not save the IP address and exit.

5.4.3. IP mask

Set SubMask address for this phone. (Can not modify in DHCP or PPPoE mode)
The text input mode is ‘123’, and other input modes are not permitted. And the inputted IP mask should observe IP address rules

5.4.4. Default GW

Set the gateway IP address. (Can not modify in DHCP or PPPoE mode)
The text input mode is ‘123’, and other input modes are not permitted. And the inputted default GW should observe IP address rules

5.4.5. DNS setting

If User set the DNS address, user can use URL address for proxy server.

The text input mode is ‘123’, and other types of characters are not permitted. And the inputted IP address should observe IP address rules.

- **Primary DNS**
- **Secondary DNS**

5.4.6. PPPoE setting

- **User Name**

  Set the username for PPPoE. It is used when IP mode is PPPoE. Only English and digit characters are allowed as input, and the default mode is ‘abc’. (The Max numbers l is 18 characters)
- **Password**

  Set the password for PPPoE. It is used when ip mode is PPPoE. Only English and digit characters are allowed as input, and the default mode is ‘abc’. (The Max numbers l is 18 characters)

5.4.7. NTP setting

- **Set**

  Choose ON to active the NTP function, or OFF to close.
- **Server IP**

  Set NTP server IP address, which LCD display will obtain the time from the Server.
- **Zone**

  Set the zone of the current location，it could be as GMT +13:00 to -12:00.

5.5. SIP setting

5.5.1. Proxy setting

- **PxyServ**

To set the proxy server address, IP address or domain name.

When the input is in "domain name" type, then it is allowed to input all ASCII characters, and its max length is limit to 32. And if the input is an IP address, then it should observe IP address rules. The default edit mode is "123".

Note: The PxyServ can not be empty.

- **ObdProxy**

To set the outbound server address, IP address or domain name.

When the input is in "domain name" type, then it is allowed to input all kinds types of ASCII characters, and its max length is limit to 32. And if the input is an IP address, then it should observe IP address rules. The default edit mode is "123".

- **PxyPort**

To set the proxy server port.

The input should be digit value. The max input PxyPort : 65535.

- **RegServ**

To set the registration server address, IP address or domain name.

When the input is in "domain name" type, then it is allowed to input all kinds types of ASCII characters, and its max length is limit to 32. And if the input is an IP address, then it should observe IP address rules. The default edit mode is "123".

- **RegPort**

To set the registration server port.

The input should be digit value. The max input PxyPort : 65535.

5.5.2. User Info

- **Auth. name**

To set the phone name for registration, this is not necessary.

The input should be digit or English character. And also it can be empty. The default edit mode is “abc”.The max length of name can be 32.

- **Password**

To set the password for registration, this is not necessary.

The input can be digit or English character. And it can be empty. The Max password length is 18.

- **PhoneNum**

To set the line number for registration, this is necessary to fulfill.

The Max PhoneNum length is: 32. The input should be digit

5.6. Phone Setting

5.6.1. Language

It provides user with two kinds of Language to set the phone. The primary language is English, another is Deutsch. The default language is “English”.

5.6.2. Forward

This IP phone is provided three types for forwarding setting:

The default mode is “Disable all”.

5.6.2.1. Busy

Incoming call in busy status will be forwarded to assigned phone number.

- **Activate**

Input the forward number and active busy forward.

The default input mode is ‘123’ and only digit input is allowed, it max length is 32.

If the forward number is empty, the phone will display “Invalid input” for 2 Second and exit, so the the call forward function is not active.

- **Deactivate**

Deactivate the function of Busy forward.

5.6.2.2. No answer

Incoming call in no answer status will be forwarded to assigned phone number. If the incoming ring times are counted to 10, the phone will be considered as No answer status, and the incoming call will be forward.

- **Activate**

Input the forward number and active busy forward.

- **Deactivate**

Deactivate the function of Busy forward.

5.6.2.3. Unconditional

Once ‘unconditional’ is set, incoming call will be forwarded to assigned phone number.

- **Activate**

Input the forward number and active busy forward.

- **Deactivate**

Deactivate the function of Busy forward.

5.6.2.4. Disable all

Disable all forward settings.

5.6.3. Alarm

- **Add alarm item**

Add the new alarm item, you can set up to 3 alarm items at most. If there’s no room for new entry, it will give a error message and return the previous menu. When a new item is added, it will search the first free position and store it. When user input date and time, it should check to see whether the input is legal.

The alarm format will be as below:

Xxxx/xx/xx (Year/Month/date), once date set is ok, please press setting alarm time: Year: 2000 to 2099; Month: 01 to 12 ; Date: 01 to 31; Hour: 01 to 24 ; Minute: 01 to 60

If the input is illegal, the LCD will display “Invalid input…” and exit to previous menu.

You can also ignore xxxx/xx/xx (Year/Month/date), and only set alarm time for everyday alarm.

If the system time reach the alarm time when current state is on hook state, a alarm ring(ring every 30 seconds) will be emit from the speaker to inform the user, and the LCD display “Alarming” . If user not press [Delete] key, the alarm ring will ring for 30 Minutes and stop, and the LCD will go to idle display. If user press [Delete] key when alarm, the alarm ring will stop and the LCD will go to idle display.

Note:

1、If the alarm time is reached when the phone is in talking/off hook/dialing/ring state, the alarm ring will not emit until the state exit to on hook state.

2、If the alarm time is reached when in menu operation/pre dial/view call list, the current operation will cancel and exit to idle state and alarm ring start.

3、When setting the alarm date, null input is permitted. If input null alarm date, it means that the alarm is everyday alarm .When user view the alarm, “EveryDay” will display on LCD at the date position of the alarm entry.

4、Set the alarm time to null is not legal.

5、If you have set a alarm and the alarm is valid(any alarm is not reached), alarm icon will display.

- **View all**

  Display all of the current alarm setting.

- **Del all**

  Delete all alarm settings.

5.6.4. Time

Set the time for phone, The date format is:YY/MM/DD, and the time format is: HH:MM:SS.

When enter the Time setting, LCD will display the current time, and the first character is blinking. It mean the current edit number position, and you can move the blink position by press [Up]/[Down] key. After input a number [0-9], the number which is blinking will be replaced with the input number, and the blink position will move the next position (If blinking position is last position, the blinking position will not move).

5.6.5. Ring

- **Ringer volume**

This phone has 8 levels to adjust volume by pressing key up/down.

- **Ringer melody**

This phone has ten type melodies by pressing key up/down to choose the ring melody you like.

The ring melodies 9- melody 10 are user defined melody. The user defined melodies have factory setting melodies (Default is ring melody 8). User can download his/her ring melody onto the phone through WEB setting.

5.6.6. Volume

(1)Handset volume
This phone has 8 levels (1-8)to adjust volume.
(2)Speaker volume
This phone has 8 levels (1-8)to adjust volume.
(3)Headset volume
This phone has 8 levels (1-8)to adjust volume.

5.7. Program key

Program key can be divided to Function key and Memory key. So enter the Program key menu, can see the Function key and Memory key sub menu.
Refer to 2.1.3 Key board, the Keypad have 14 program keys (P1-P14). Anyone of the program keys can be defined as function key or be defined as memory key.

5.7.1. Function key

It is included seven function key to setups, which are Flash, transfer, conference, missed call, Received call, DND, Headset.
How to set function key? Refer to 7.1 Function Key
Enter the menu, and pressing key up/down to choose the function you want to set, and then press whichever function key to save it.

5.7.2. Memory key

Enter 'Memory key' menu, it will be display 'Mem number', you can input the pre-set speed dial number, then press the key of ✓, LCD will be display 'Press prog key…', then user can press one memory key to define speed dial number.
Note:
The max length of memory number is same as pre dial, please refer to 4.1.1 Dialing.

5.8. Factory default

Set SipSet Phone to factory default.

5.9. Reboot

Reboot SipSet Phone.

6. WEB SETTING

Besides using the key menu on the telephone to carry on the basic configuration, but also you may use the WEB to carry on the comprehensive configuration.

6.1. How to land web

DPH-150S(SE) has two network modes: Bridge and Router, We can switch DPH-150S’s (DPH-150SE’s) mode between the two modes.（Default mode is Bridge mode）

1、Bridge mode：

In bridge mode, two network interfaces have the same IP address called VOIP port IP address. User must ensure the interface (Network) close to DC has insert line and the status is up. Then user can land the web through either interface.

User may use the network browser to visit the VOIP port IP address, landing the WEB configuration surface.

The VOIP port IP address can obtain from the telephone LCD menu：”Network setting -> IP address”. You may input http://xxx.xxx.xxx.xxx in the browser to visit the web. “xxx.xxx.xxx.xxx” is the VOIP port IP address displayed in the telephone LCD menu and 80 is the web access port.

Also user can visit the web using https protocol, for example https://172.16.100.93.

note： When you want to visit the VOIP port through the browser, please check your computer whether has connected correctly to the VOIP port and your computer IP address whether is set correctly to the same IP subnet or has the right router.

2、Router mode：

User may use the network browser to visit the LAN (PC) port or WAN (Network) port IP address, landing the WEB configuration surface. The LAN port IP address is pre-set “192.168.1.254”. Then you may input the URL address in the browser to visit the web: http://192.168.1.254

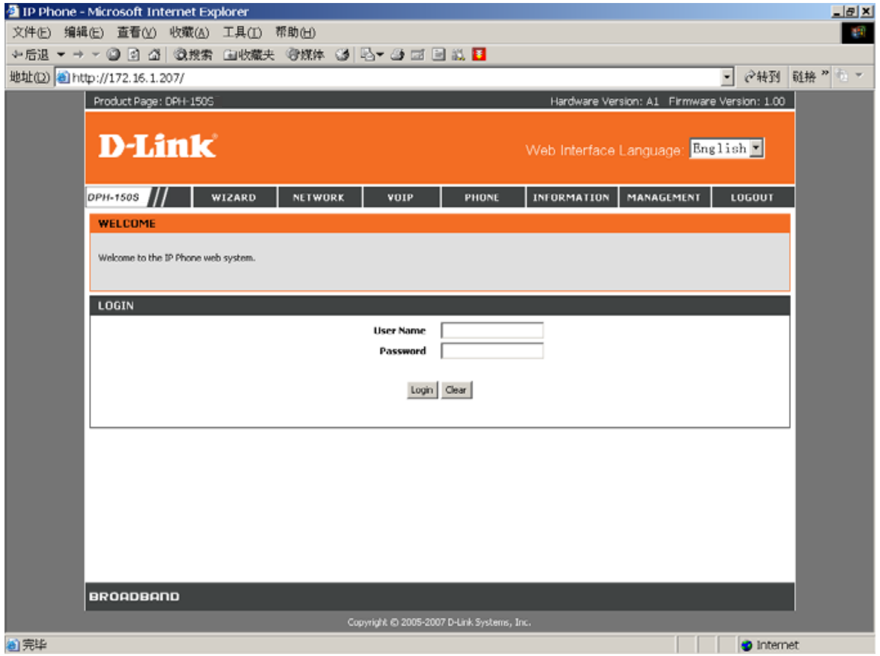
The WAN port IP address can obtain from the telephone LCD menu：”Network setting -> IP address”. Different from LAN port, when visiting the web you may input http://xxx.xxx.xxx.xxx in the browser to visit the web. “xxx.xxx.xxx.xxx” is the WAN port IP address displayed in the telephone LCD menu and 80 is the web access port.

Also whatever connecting to LAN or WAN port, user can visit the web using https protocol, for example https://172.16.100.93. Here the IP address is LAN or WAN port IP address.

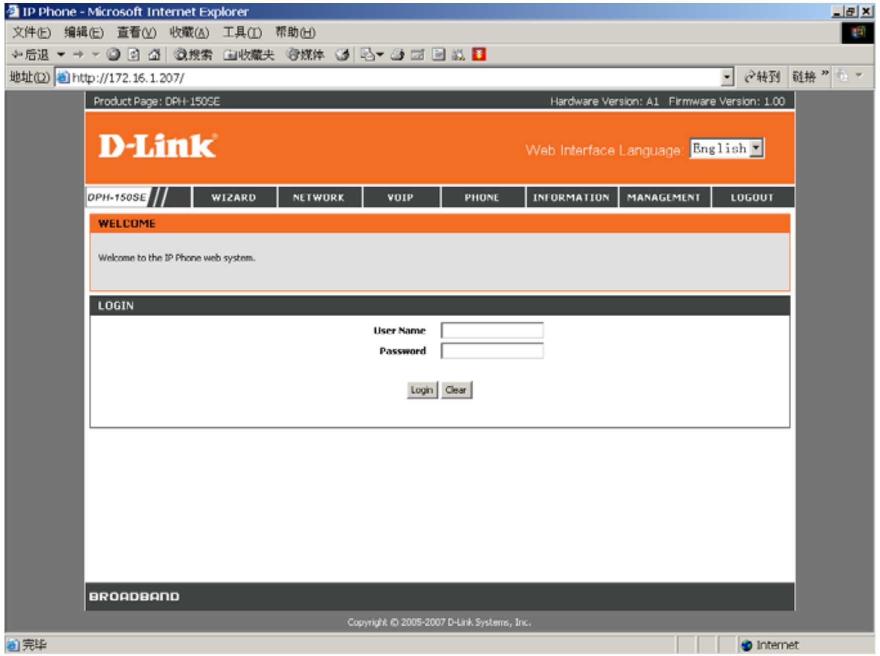
note：When you want to visit the WAN port or LAN port through the browser, please check your computer whether has connected correctly to the WAN port or LAN port, and your computer IP address whether is set correctly to the same IP subnet or has the right router.

6.2. Default User name and Password

The system default user name is “admin” and default password is null. After inputting them, clicks on “Login” button to login.



<For DPH-150S>



<For DPH-150SE>

6.3. About Note message

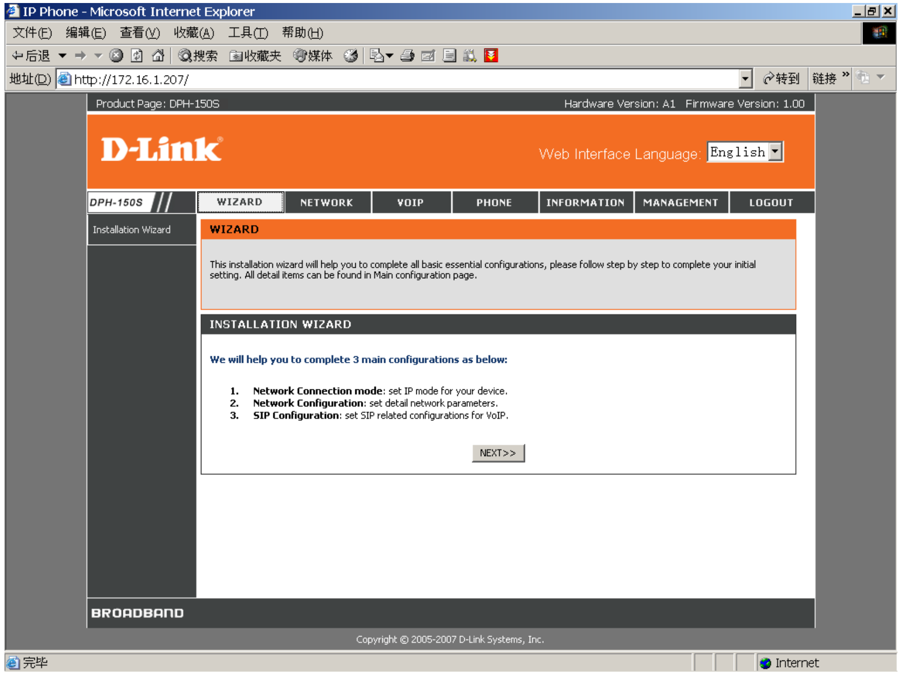
"You should reboot the system after modify these configuration."
It alarms customer to reboot phone to activate all the configurations they have changed on the web page.

"Submit may interrupt current conversation, please be sure."
It alarms customer that if they click the Submit button, the conversation may be interrupted. So if customers want to change configurations on the web page, they'd better do this when the phone is free (without conversation on it).

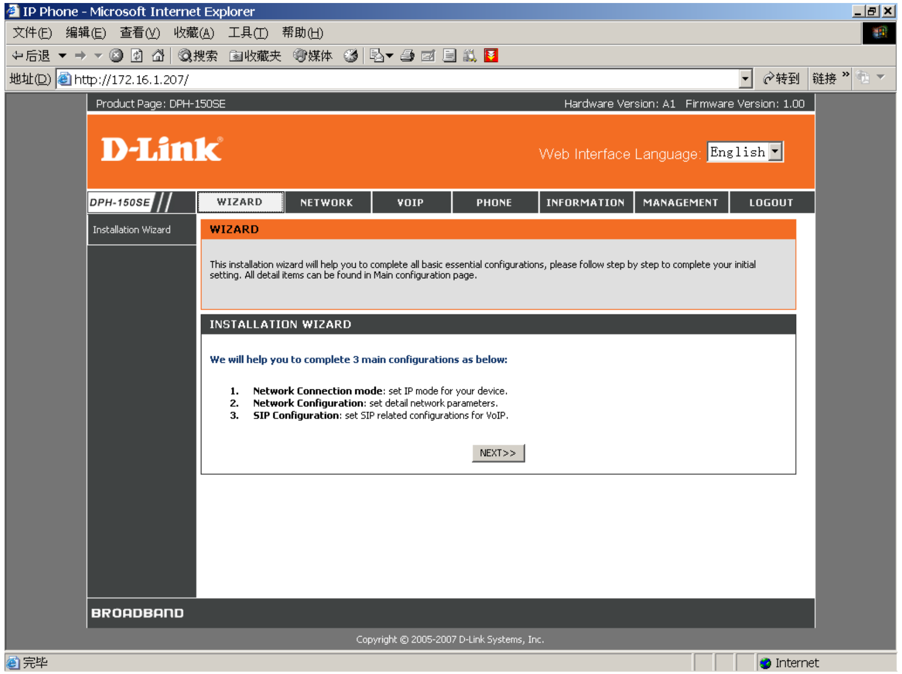
6.4. About Reset button

In many web pages, there are Reset button. The button can help customer to resume the page when he/she config a page.

6.5. WIZARD



<For DPH-150S>

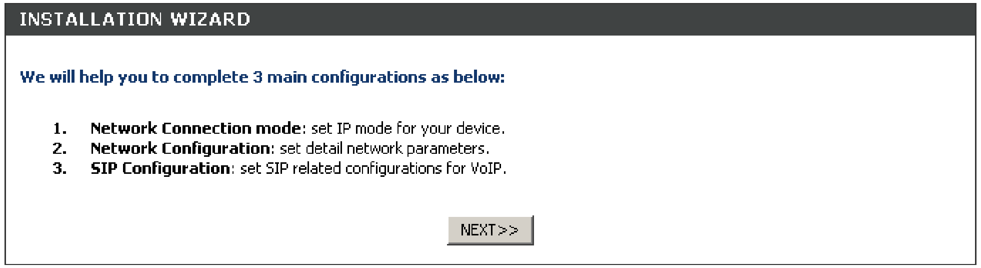


<For DPH-150SE>

Installation wizard help customer to complete all basic essential configurations. It includes three steps:

1. Network Connection mode
2. Network Configuration
3. SIP Configuration.

Select NEXT button to step. After finishing the wizard guide, the system MUST be rebooted to activate the configurations.



When entering Network Connection WEB page, there are three network modes as follows:

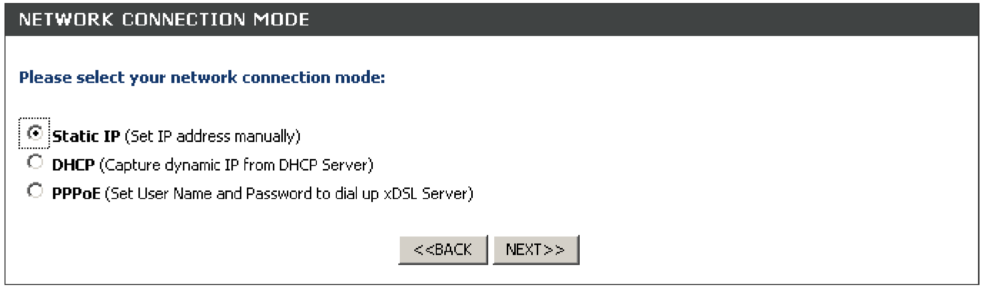
- Static IP
- DHCP
- PPPoE.

According to the mode selected, the following Network Configuration and SIP Configuration may be different.

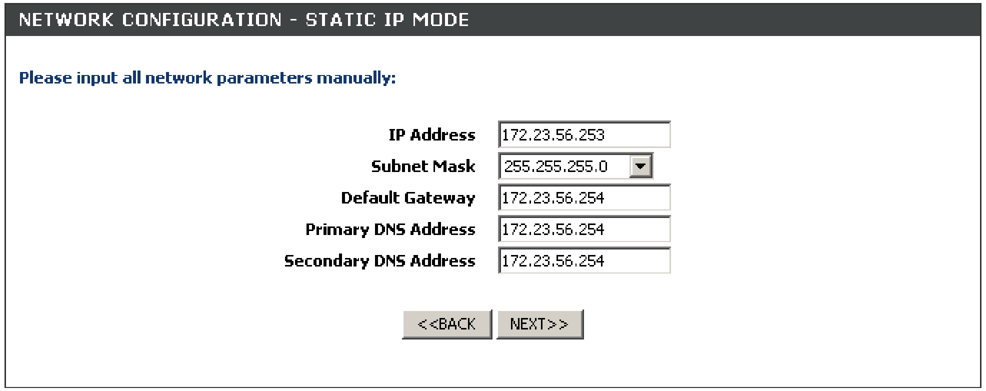
6.5.1. Static IP setting

If Static IP mode, customer needs to finish the following steps:

1. Select Static IP and press NEXT button.

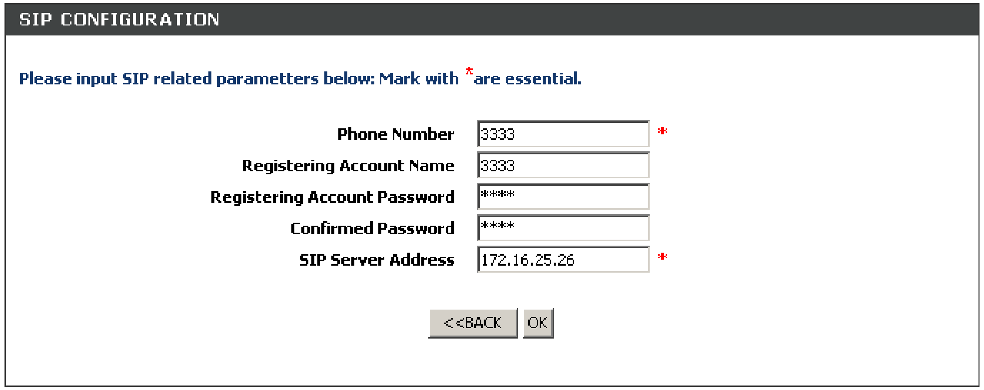


2. Input network parameters.



- IP Address: IP address of the phone
- Subnet Mask: subnet mask of the phone's IP address.
- Default Gateway: default gateway of the phone
- Primary DNS Address: primary Domain Name Service Address
- Secondary DNS Address: secondary Domain name service Address (not necessary)

3. Set SIP parameters



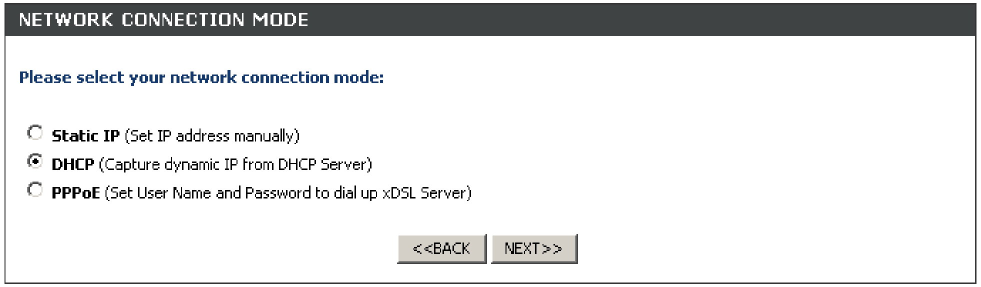
- Phone Number:
  Phone number used to call or be called. Max. length is 32 digits.
- Registering Account Name:
  Registering Account Name used to register on SIP server. Max. length is 32 characters.
- Registering Account Password:
  Registering Account Password used to register on SIP server. Max. length is 18 characters.

- Confirmed Password:
  Keep same with Registering Account Password
- SIP Server Address:
  SIP Server Address used to set both Register Server and Proxy Server at the same time in Account Settings.

6.5.2. DHCP setting

If DHCP mode, customer needs to finish two steps:

1. Select DHCP and press NEXT button.

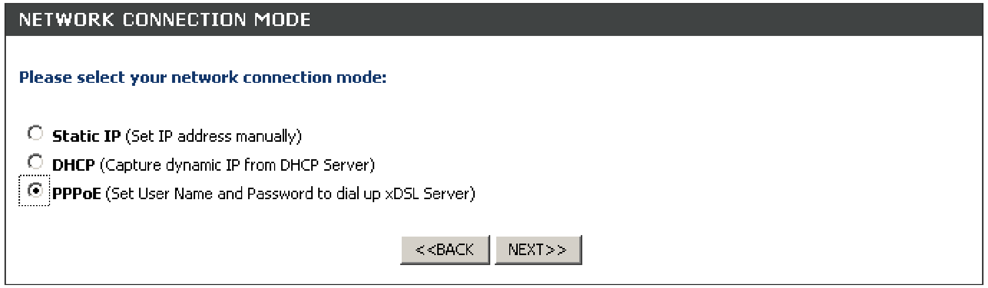


2. Set SIP parameters
   Please Refer to 6.5.1_Step 3

6.5.3. PPPoE setting

If PPPoE mode, customer needs to finish three steps:

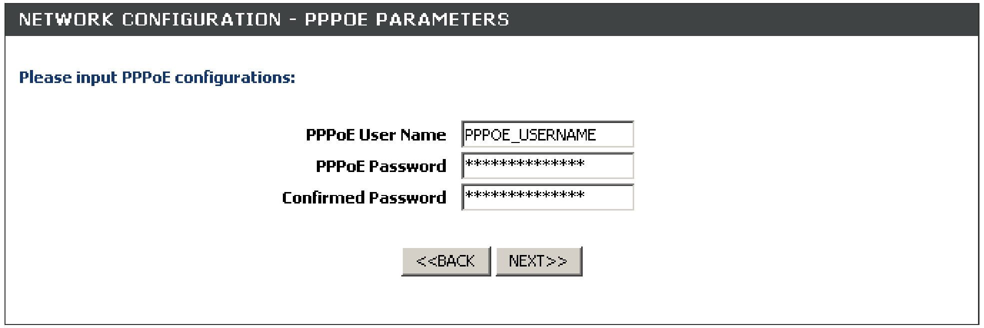
1、 Select Static IP and press NEXT button.



2、 Set PPPoE account name and password.

PPPoE User Name: Max length is 32 characters.

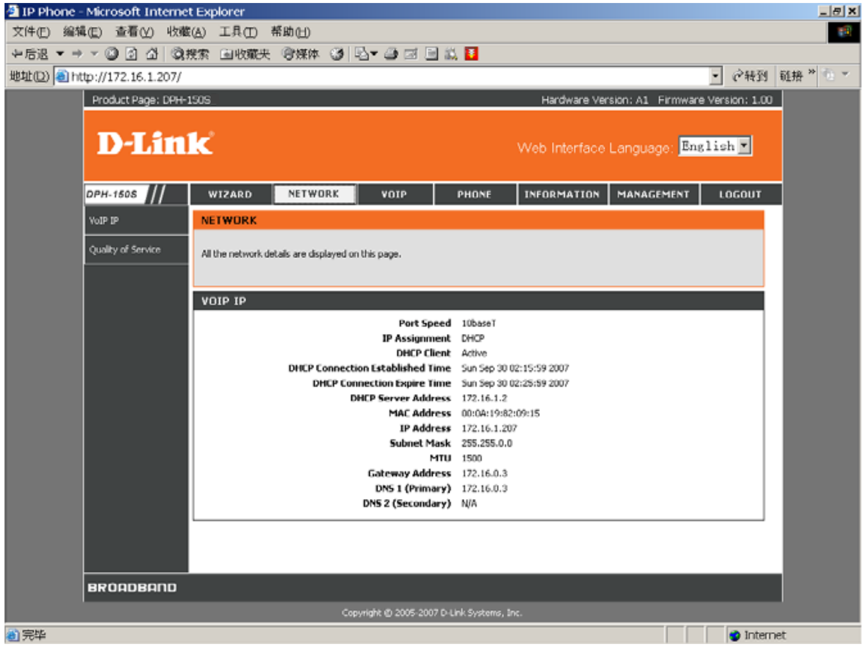
PPPoE Password: Max length is 18 characters.



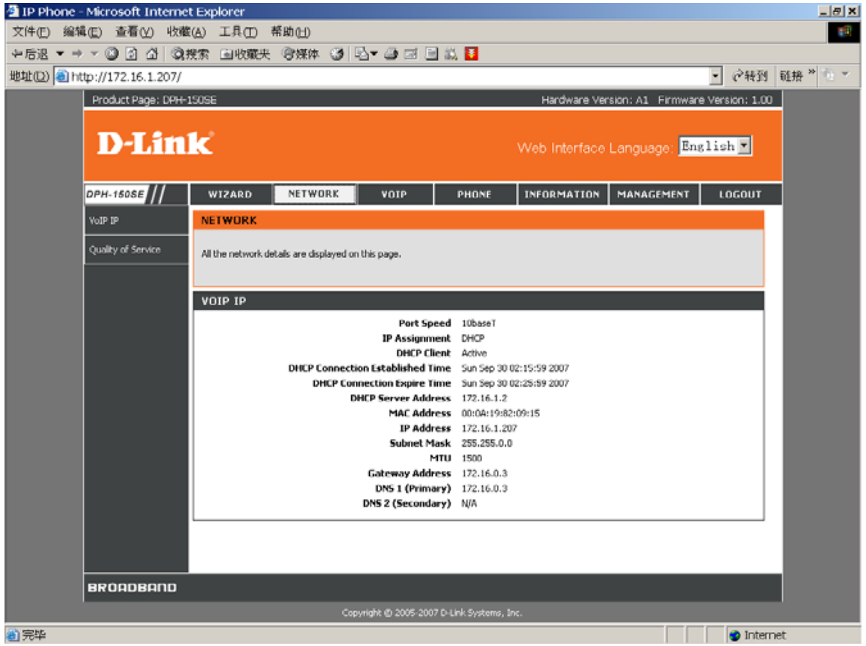
3、 Set SIP parameters

Please Refer to **6.5.1_Step 3**

6.6. Network



<For DPH-150S>



<For DPH-150SE>

DPH-150S(SE) has two network modes: Bridge and Router, We can switch DPH-150S's (DPH-150SE's) mode between the two modes. The WEB page of Bridge and Router mode are different as follows.

6.6.1. Bridge mode

Bridge mode, this section includes two web pages:

- VoIP IP
- Quality of Service

VoIP IP page are mainly used to setting network configurations.

Quality of Service are used to adjust quality of service by changing parameters.

6.6.1.1. VoIP IP

In this WEB page, we can complete the following functions:

1、 Switch Bridge mode to Router mode.

2、 Change host name of the IPPhone.
The first input is limited 32 characters; the second input is limited 32 characters.

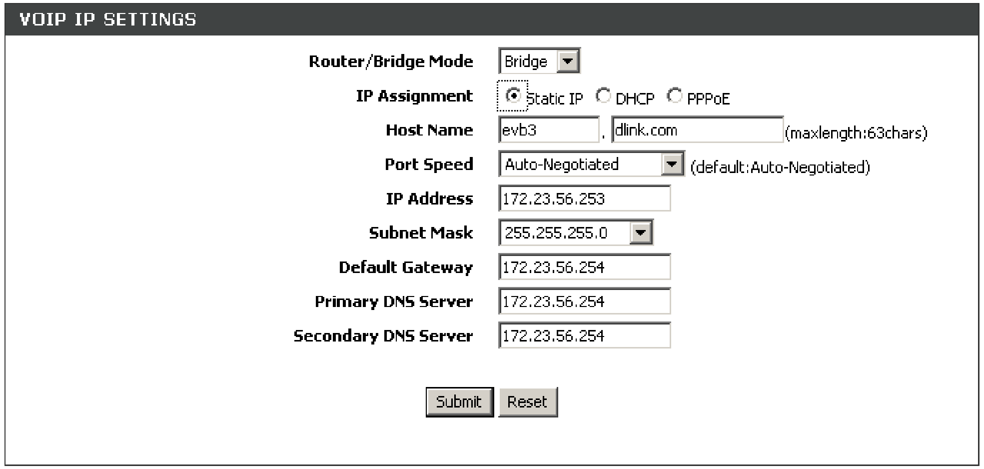
3、 Select Static IP/DCHP/PPPoE as current mode.
When clicking one of the network modes, the page will change and correspond to the network mode.

4、 Define WAN port Speed :
Auto-Negotiated(Default)/10M Bytes /100M Bytes.

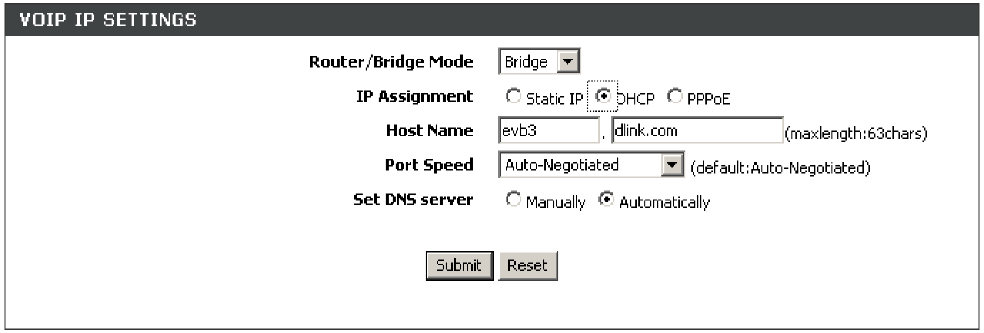
5、 Permit anybody can/cannot ping the IPPhone’s IP address.
Allowed (default)

6、 Set the IPPhone’s IP address/Net Mask /Default Gateway

7、 Set thee IPPhone’s DNS server.
primary DNS server, secondary DNS server (not necessary)



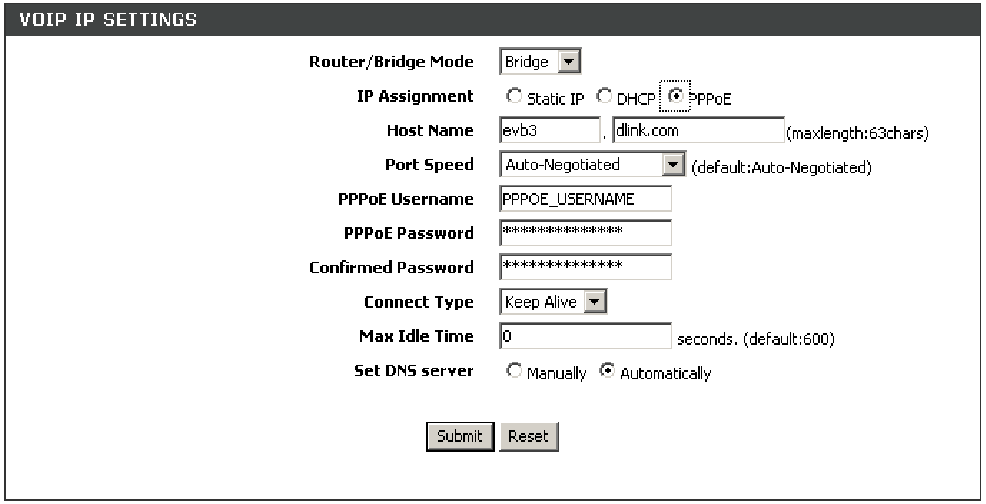
8、 In DHCP network mode, DNS server can set Automatically (default) or Manually.



9、 In PPPoE mode, PPPoE username and password can changed after submit.

Max Idle Time default is 600 seconds

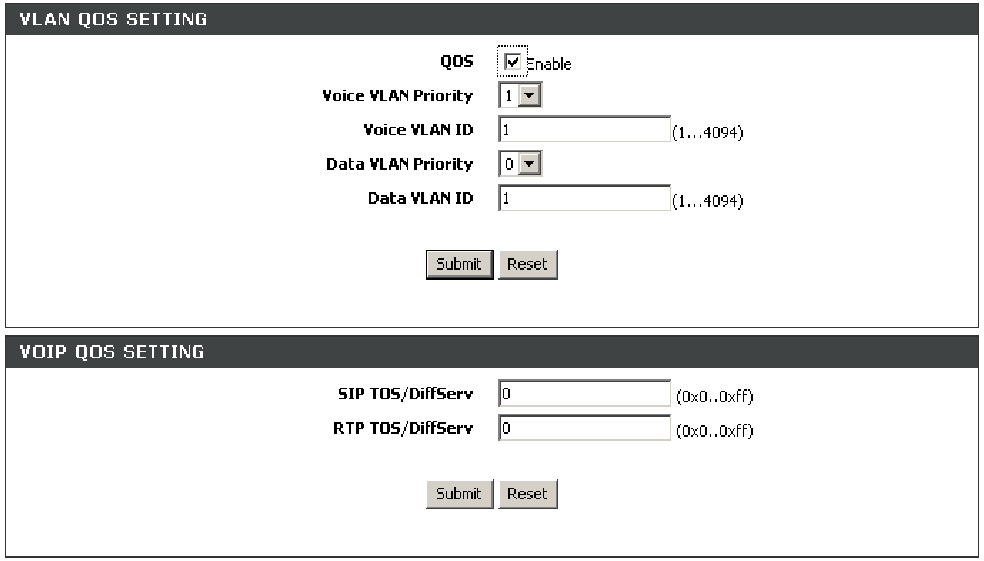
Connect Type: Keep Alive(default)/Manual On



10、There MUST be an alert to notice that the configurations will take effect after reboot.

6.6.1.2. Quality of Service

We can configure this page to active vlan qos setting. SIP flow and RTP flow can be set different vlan id or different 802.1p priority to achieve two layer qos control. We can also set TOS field in order to achieve three layers control.



QOS: enable or disable qos items to improve voice quality, default is disable

Voice VLAN Priority: 802.1p priority in SIP transaction

Voice VLAN ID: vlan id tagged in SIP transaction

Data VLAN Priority: 802.1p priority in RTP flow

Data VLAN ID: vlan id tagged in RTP flow

SIP TOS/diffserv: TOS filed value in SIP packet

RTP TOS/diffserv: TOS filed value in RTP packet

6.6.2. Router mode

Router mode contains five web pages:

1、 WAN Settings

2、 LAN Settings

3、 NAT

4、 Quality of Service

WAN Settings page are mainly used to setting WAN network configurations.

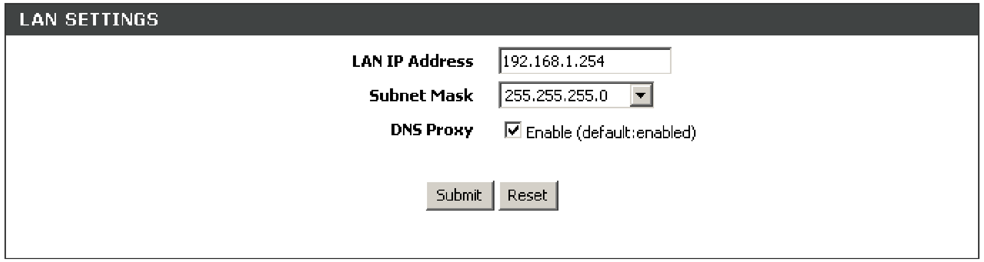
LAN Settings page are mainly used to setting LAN network configurations.
NAT Settings page are mainly used to setting NAT network configurations.
Quality of Service are used to adjust quality of service by changing parameters in the page.

6.6.2.1. WAN Settings

Please Refer to 6.6.1.1 VOIP IP

6.6.2.2. LAN Settings

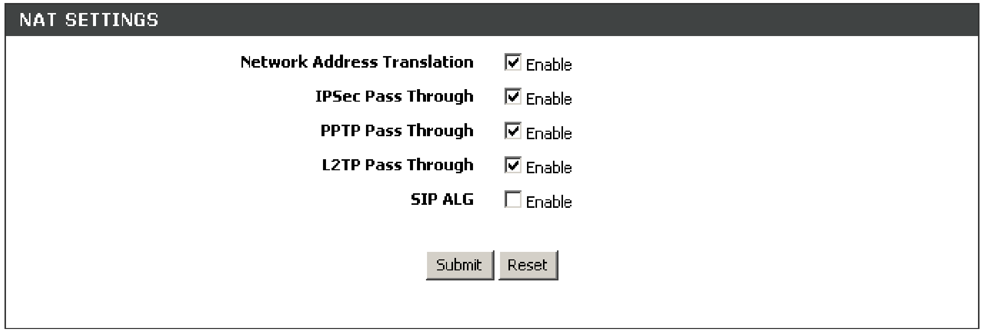
The LAN Setting page allows customers to set the configurations of LAN port.



LAN IP Address: phone’s lan IP address
Subnet Mask: phone’s lan subnet mask
DNS Proxy: set DNS Proxy enable (default) or disable.

6.6.2.3. NAT

The NAT Setting page allows customers to set the configurations of NAT

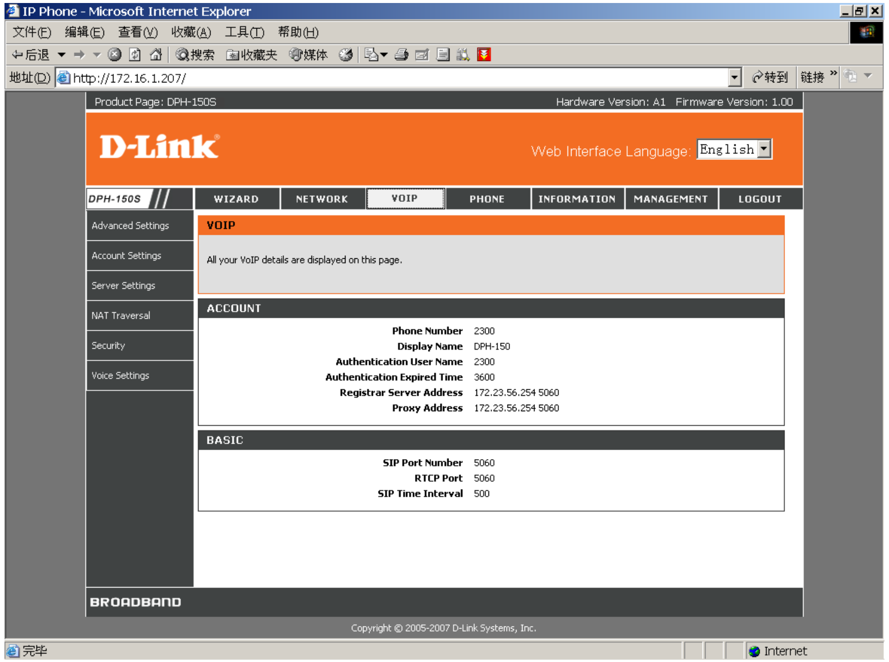


- Network Address Translation
  Enable(default)/Disable
- IPSEC Pass Through
  Enable(default)/Disable
- L2TP Pass Through
  Enable(default)/Disable
- PPTP Pass Through
  Enable(default)/Disable
- SIP ALG
  Enable/Disable(default)

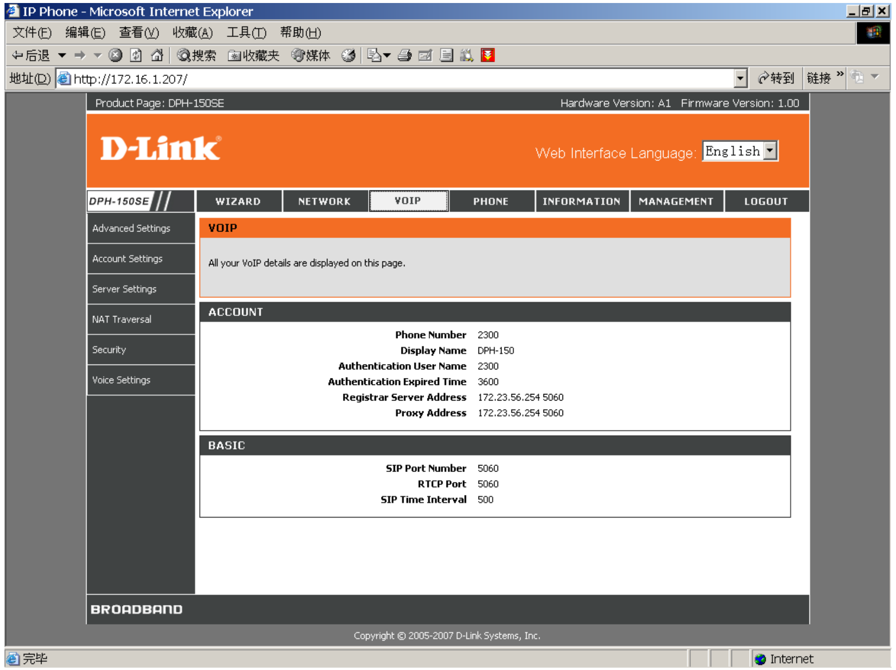
6.6.2.3. Quality of Service

Please Refer to 6.6.1.2 QOS

6.7. VOIP



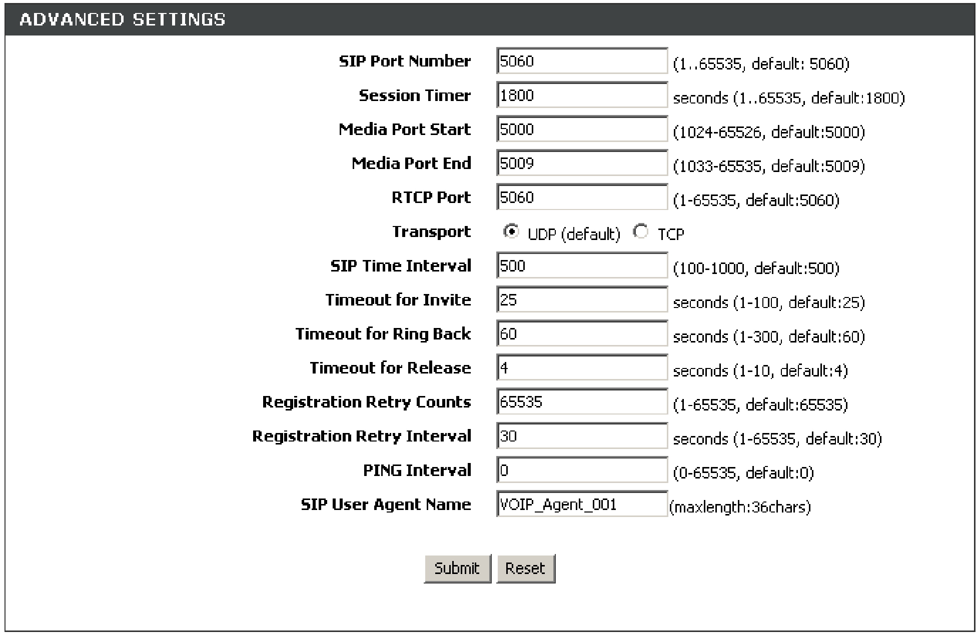
<For DPH-150S>



<For DPH-150SE>

6.7.1. Advanced Setting

The page set some basic parameters of SIP client (UA).



SIP Port Number: The port number of SIP client (UA).

Session Timer: A minimal interval of SIP session. Before the interval passes, UA should send re-INVITE or UPDATE request to other peer to keep the session alive.

Media Port Start: system will select a random port to transmit RTP flow and this item configures the start range

Media Port End: system will select a random port to transmit RTP flow and this item configures the end range

RTCP Port: RTCP port

Transport: using UDP or TCP protocol to transport control protocol data.

SIP Time Interval: Timer T1 value, T1 is an estimate of the round-trip time (RTT), and it defaults to 500 ms.(refer RFC3261 17.1.1.1)

Timeout for invite: Max periods for receiving 180 responses after sending INVITE message.

Timeout for Ring Back: Max periods for receiving 200 responses after sending INVITE msg.

Timeout for Release: Timer for releasing resource when release a call.

Registration Retry Counts: Register retry counts when register failed.

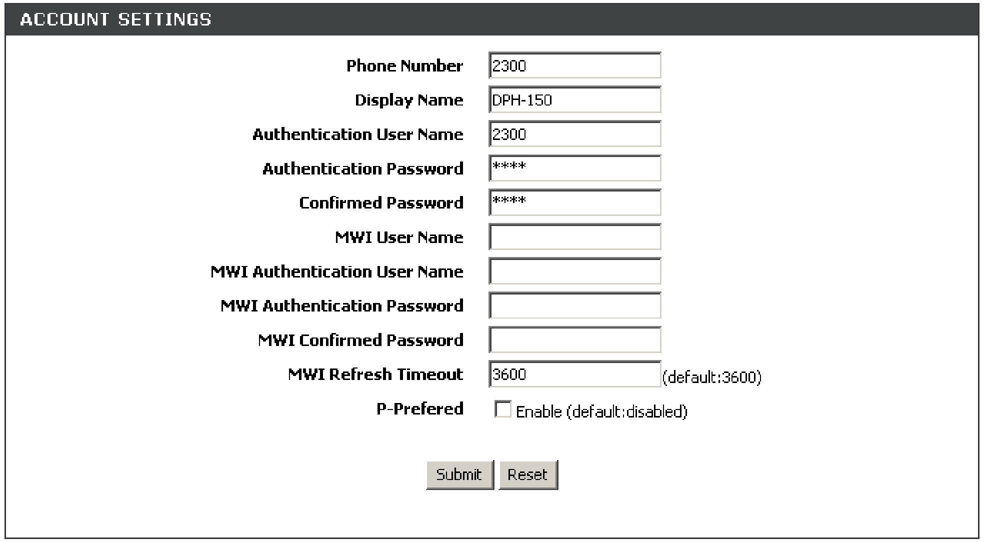
Registration Retry Interval: REGISTER request send interval when register failed.

PING interval: Time interval in secs for getting PING response and request when ping testing.

SIP User Agent Name: User agent name for User-Agent header field

6.7.2. Account Setting

The page show user's accounts of SIP register servers.



Phone Number: A user part of SIP URL, usually it's a SIP account and consists of a continuous digit string with maximum length of 32.

Display name: An optional element in SIP URL. Max length is 32 characters.

Authentication User Name: User name for authentication when receiving 401/407 for SIP request. Max length is 32 characters.

Authentication Password: Password for authentication when receiving 401/407 for SIP request. Max length is 18 characters.

Confirmed Password: Confirm for user input password

MWI User Name: User name for send SUBSCRIBE request in MWI application. Max length is 32 characters.

MWI Authentication User Name: User name for authentication when receiving 401 for SUBSCRIBE request in MWI application. Max length is 32 characters.

MWI Authentication Password: Password for authentication when receiving 401 for SUBSCRIBE request in MWI application. Max length is 18 characters.

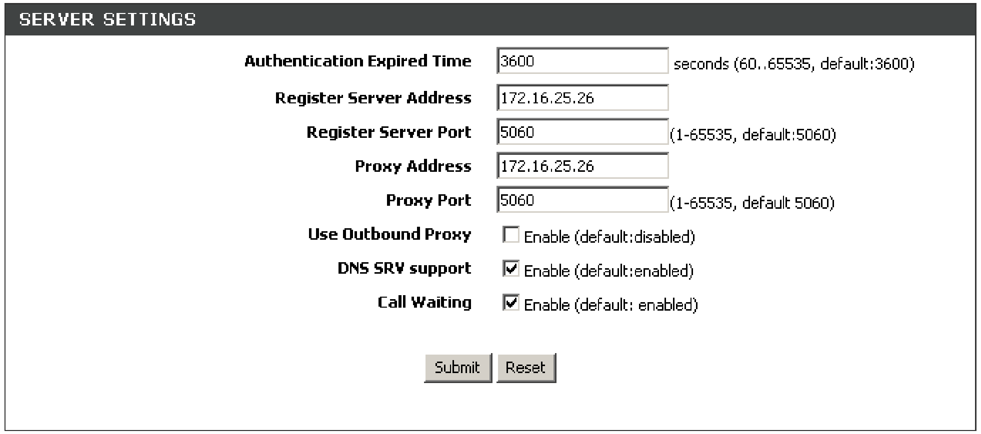
MWI Confirmed Password: Confirm for user input password.

MWI refresh Timeout: Interval of sending SUBSCRIBE request, which will be Expires header field of SUBSCRIBE request.

P-Asserted: If P-Asserted header field be enable. (P-Asserted – a new SIP header field for private extensions, refer rfc3325).

6.7.3. Server Settings

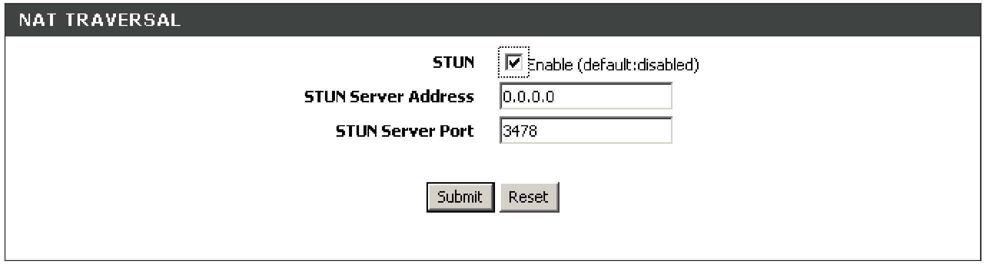
This page can be used to set the server's parameters of an account.



Authentication Expired Time: when the time expires, the client must register itself on the SIP server again.
Register Server Address: Assigns the SIP Register Server's IP address.
Register Server Port: Register server port of a certain SIP domain.
Proxy Address: Assigns the Proxy server's IP address.
Proxy Port: Assigns the Proxy server's port.
Use Outbound Proxy: If all outgoing SIP request send to outbound proxy.
DNS SRV support: If SRV DNS be support. (SRV- A DNS RR for specifying the location of services)
Call Waiting: If Call Waiting function be enabled or disabled.

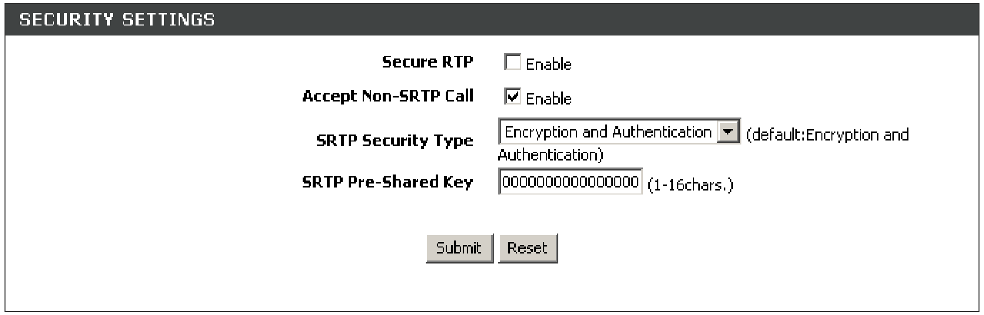
6.7.4. NAT Traversal

The page help customer to set STUN parameters.



6.7.5. Security

Setting secure RTP call of the SIP client. User can accept SRTP call or Non-SRTP call.



Secure RTP: if secure RTP be support

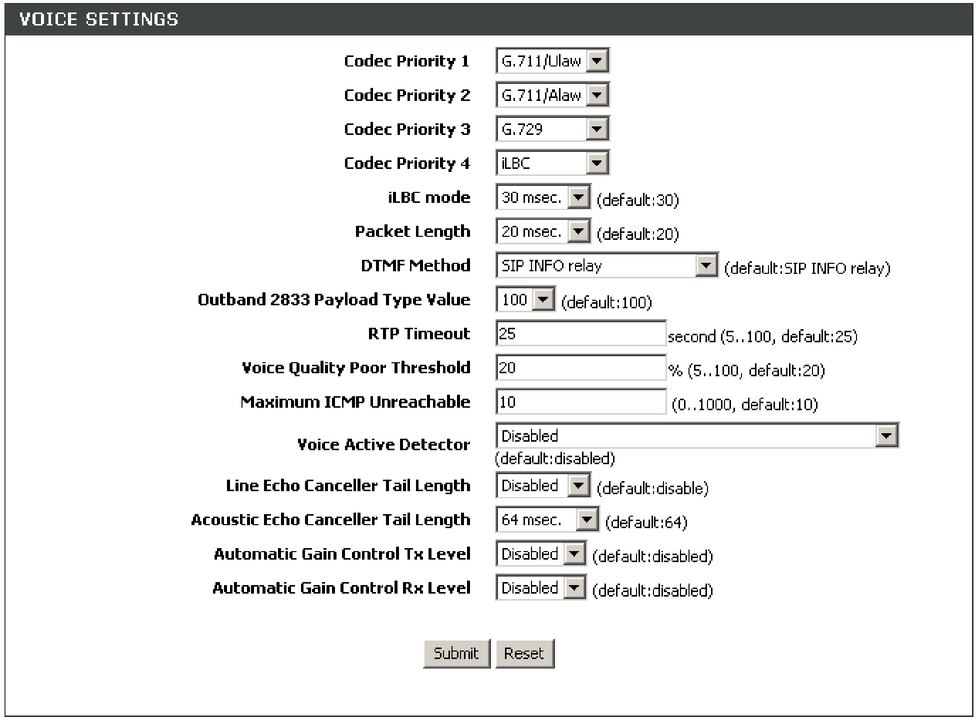
Accept Non-SRTP Call: if Non-SRTP call can be accepted when secure RTP has been enable.

SRTP Security Type: Security type of SRTP use.

SRTP Pre-shared Key: Pre-shared Key of SRTP use. The length must between 1-16chars.

6.7.6. Voice Settings

Setting the Codec priority of IP Phone and other voice settings..



Codec Priority 1-4 Codec Priority

ILBC mode: ILBC mode

Packet Length: RTP payload length. Selects a length from the pull-down menu, default setting is 20 msec.

DTMF Method: five DTMF Method to select

Outband 2833 Payload Type Value: Outband 2833 Payload Type Value

RTP Timeout: Disconnect a call after not receiving RTP packet for this time value. Assigns the time value from 1 to 100, default setting is 25 seconds.

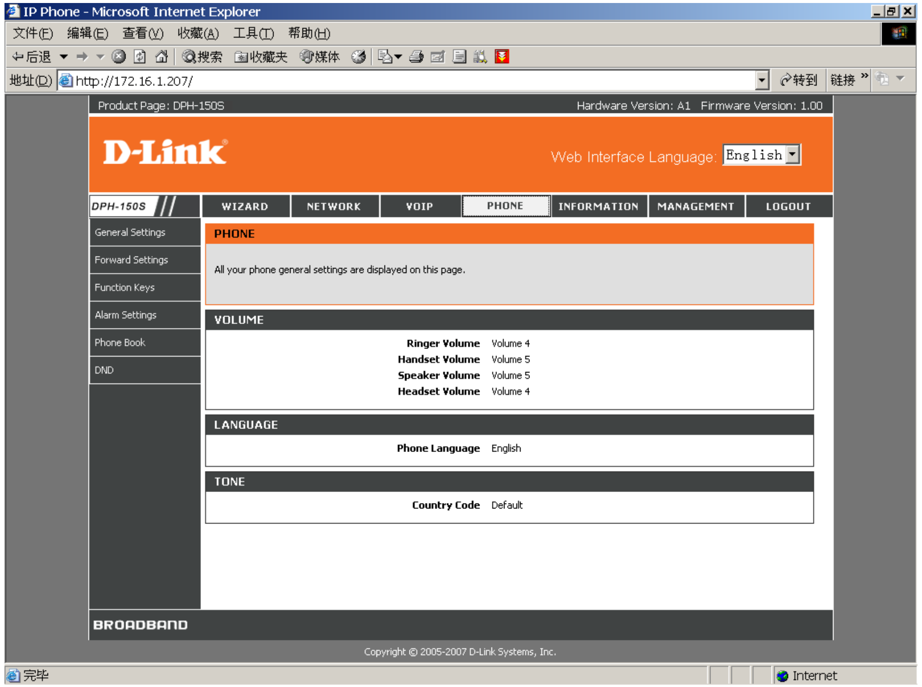
Voice Quality Poor Threshold: Allowable the maximum percentage of RTP packet loss. Assigns the percentage from 0 to 100, default setting is 20%.

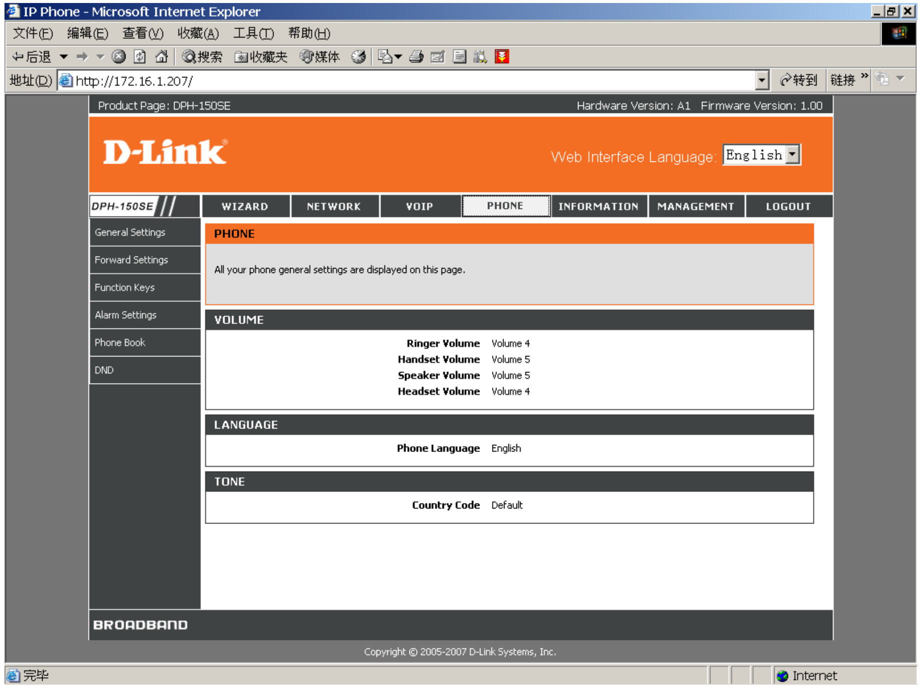
Maximum ICMP Unreachable: Allowable the maximum number of consecutive ICMP destination unreachable responses. ICMP differs in purpose from TCP and UDP in that it is usually not used directly by user network applications. One exception is the ping tool, which sends ICMP Echo Request messages(and receives Echo Response messages) to determine whether a host is reachable and how long packets take to get to and from the host. Assigns a number from 0 to 1000, default setting is 10.

Voice Active Detector: It is used in speech encoding software to determine if the voice being encoded is human speech or background noise. There are three types of silence suppression: NO CNG, Only G.711 Annex II type, and Codec Specific CN.

Line Echo Canceller Tail Length: Tail length for line echo cancellation. Default setting is in Disable mode.
Acoustic Echo Canceller Tail Length: Tail length for acoustic echo cancellation Default setting is in Disable mode.
Automatic Gain Control Tx Level: Automatic voice gain control for transmitting. Default setting is in Disable mode.
Automatic Gain Control Rx Level: Automatic voice gain control for receiving. Default setting is in Disable mode.

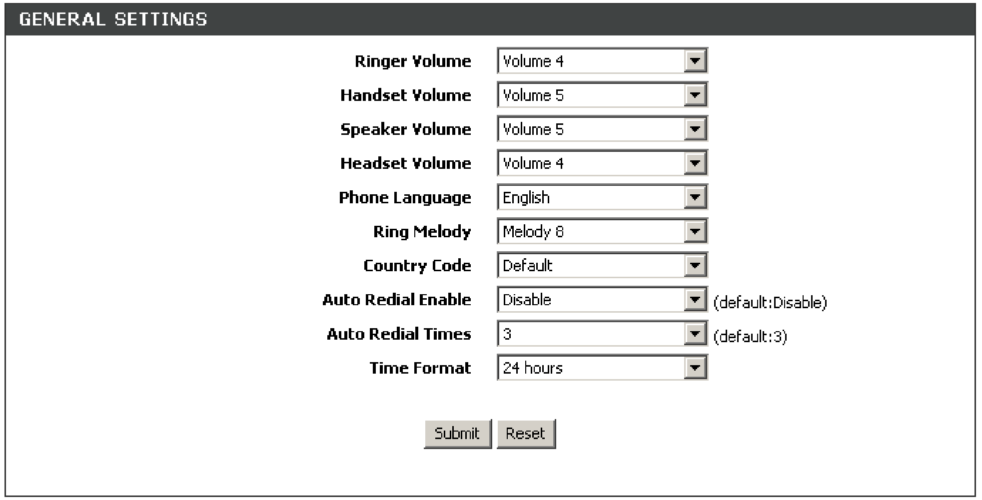
6.8. Phone





6.8.1. General Settings

This page can be used to set common parameters of the phone.



Volume Setting

set the phone 's volume of ringer/handset/speaker/handfree.

Language Setting

change the language displayed on LCD, English or Deutsch(German).

Melody Setting: change the melody of ring

Tone Setting: set tone of keystoke.

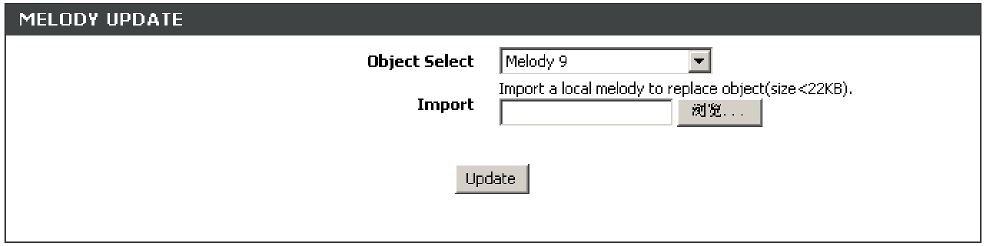
Auto Redial Setting

Auto Redial Enable: enable or disable auto redial function

Auto Redial Time: set auto redial times

Time Format Setting

Time format: time format displayed on lcd



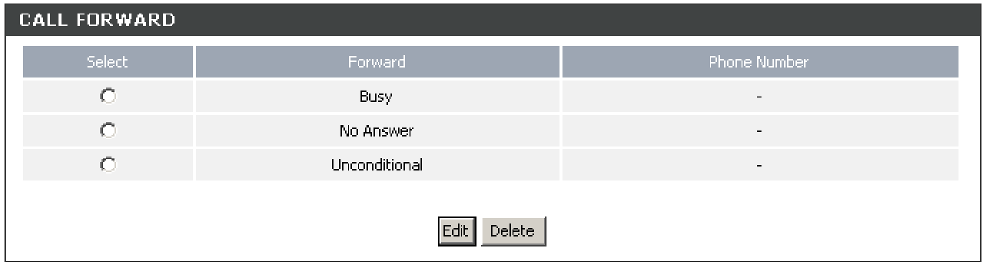
Melody Update

Object Select: select the item that will be updated.

Import: import a local melody file into the phone. The file size must be less than 22000 bytes, and its format must be PCM.

Restore Music of Hold: restore music of hold to default.

6.8.2. Forward Settings



The webinterface should provide the settings for call forwarding (call diversion) which are currently only available in the LCD menu of the telephone.

An additional parameter set by default to support persistent call forwarding with LANCOM VoIP Call Manager should be inserted.

Call forwarding should have a separate configuration page in the “Phone Settings” folder:

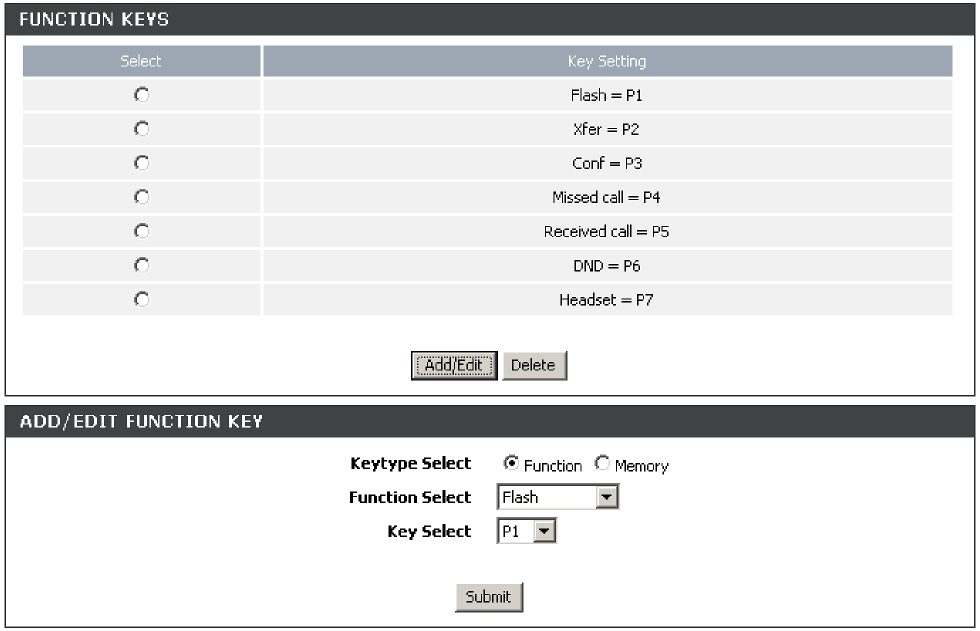
Persistent Call Forwarding at SIP Proxy	**Type**
Enable	ON/OFF (Default := ON)
Call Forwarding on Busy	**Type**

Enable	ON/OFF (Default := OFF)
Target Number	Phone number/SIP URI
Call Forwarding on No Answer	**Type**
Enable	ON/OFF (Default := OFF)
Target Number	Phone number/SIP URI
Call Forwarding Unconditional	**Type**
Enable	ON/OFF (Default := OFF)
Target Number	Phone number/SIP URI
Call Forwarding Parallel*	**Type**
Enable	ON/OFF (Default := OFF)
Target Number	Phone number/SIP URI

*CFP (Call Forwarding Parallel) is only applicable in combination with LANCOM VoIP Call Manager.

6.8.3. Function Keys

User can define some function on function keys or define phone number on memory key on the web page, and the max length of phone number is 32 digits.



A. function can only be edited; but memory can be added/deleted and edited;

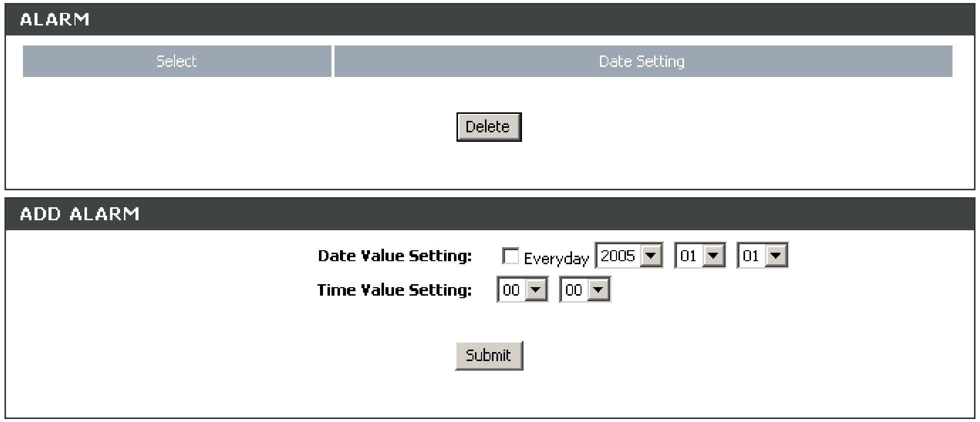
B. a key only can be set as a function or memory, or the alert message will pop out to customer ------”The Key you select has been occupied!”

C. a function only can be set on a key, but same memory content can set on different keys;

D. When editing, customer can change a function’s key; and can change the content of a memory key.

6.8.4. Alarm Settings

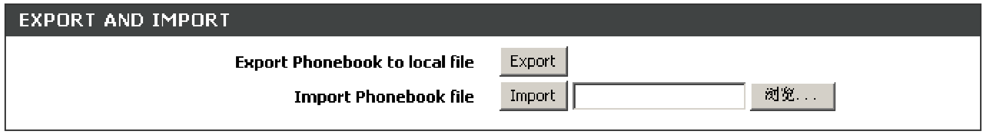
There is a function of alarm clock when you add an entry of alarm setting.



Date Value Setting: set alarm date. Empty means the alarm will be active every day.
Time Value Setting: set alarm time.

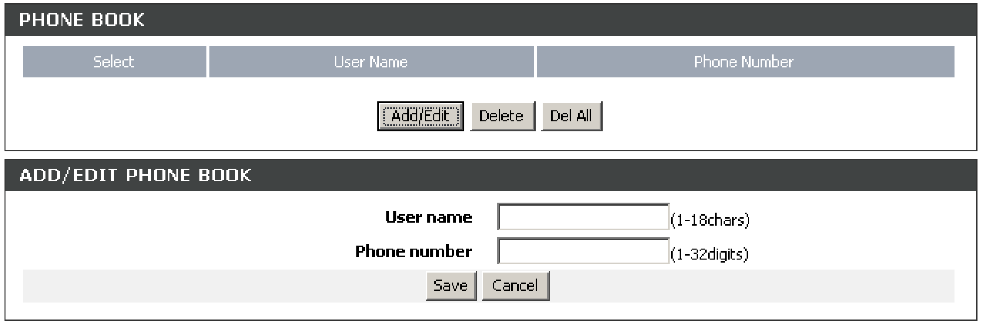
6.8.5. Phone book

1、 Save phone book of the IPPhone to a local file.
2、 Import an local phone book file to IPPhone’s phone book.(overwrite the current IPPhone’s phone book)



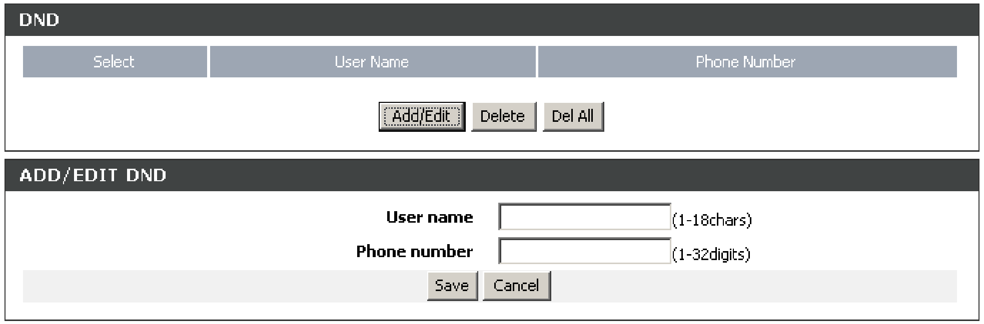
The size of import file can’t smaller than 8KB; there must be “[phone_book]” tag at the file’s head, and there must be a phone book item at least in the file.

3、 Add phone book one by one use this page.
User name: Max length is 18 characters.
Phone number: Max length is 32 digits.



6.8.6. DND

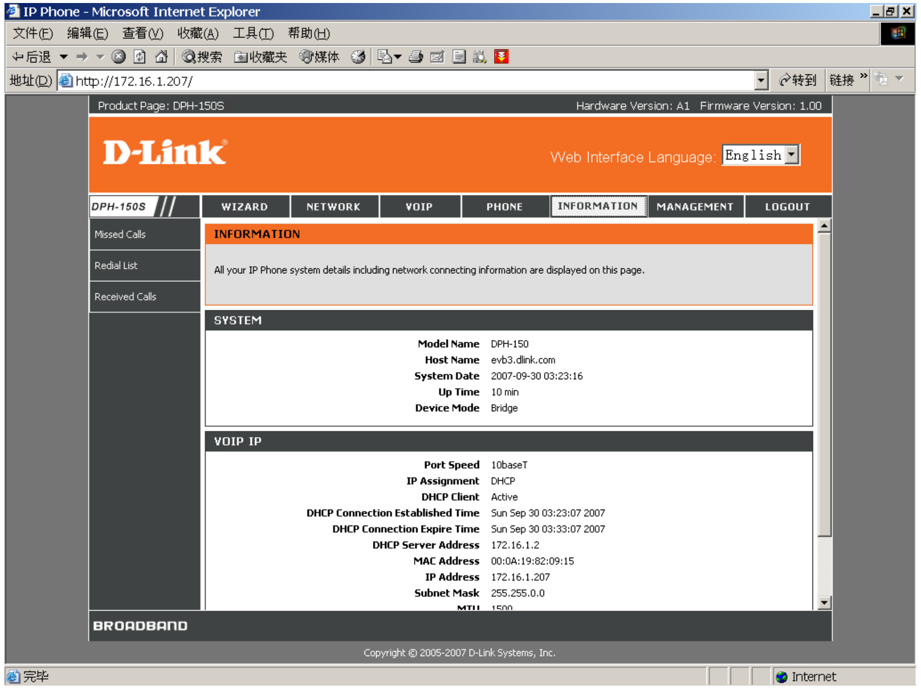
User can add/delete Do Not Disturb List entry in the page.

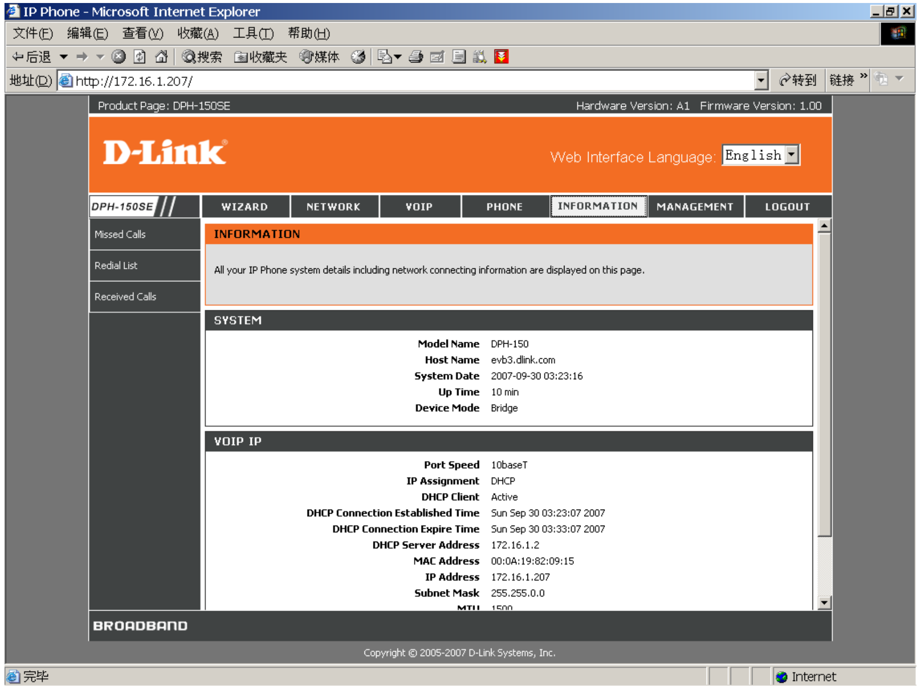


User name: user name to DND, Max length is 18 characters.

Phone number: phone number to DND, Max length is 32 digits.

6.9. Information





<For DPH-150S> <For DPH-150SE>

Information includes six web pages: System Information, Routing Table, Call Missed Record, Call Dialed Record, Call Received Record and System Log. They are used to display the main information of IP-Phone. But this information may be different according to different mode: Route/Bridge. The difference between the two modes includes two web pages: System Information and Routing Table, so when describing the two pages, we should divide it into two conditions.

6.9.1. System information

This table displays the information of system. It includes two parts: System shows the basic information of the system. The other part is about the information of network. If in router mode, this information will include Wan and LAN; if in bridge mode, it will include VOIP IP only.

6.9.1.1. Bridge mode

SYSTEM

Model Name	DPH-150
Host Name	evb3.dlink.com
System Date	2007-09-30 06:21:02
Up Time	5 min
Device Mode	Bridge

VOIP IP

Port Speed	100base-Tx
IP Assignment	DHCP
DHCP Client	Active
DHCP Connection Established Time	Sun Sep 30 06:16:15 2007
DHCP Connection Expire Time	Sun Sep 30 06:26:15 2007
DHCP Server Address	172.16.1.2
MAC Address	00:0A:19:82:09:15
IP Address	172.16.1.207
Subnet Mask	255.255.0.0
MTU	1500
Gateway Address	172.16.0.3
DNS 1 (Primary)	172.16.0.3
DNS 2 (Secondary)	N/A

6.9.1.2. Router mode

SYSTEM	
Model Name	DPH-150
Host Name	evb3.dlink.com
System Date	2007-09-30 04:28:07
Up Time	17 min
Device Mode	Router

LAN	
MAC Address	00:0A:19:82:09:15
IP Address	192.168.1.254
Subnet Mask	255.255.255.0
DHCP Server Function	Active

WAN	
Port Speed	10baseT
IP Assignment	DHCP
DHCP Client	Active
DHCP Connection Established Time	Sun Sep 30 04:26:21 2007
DHCP Connection Expire Time	Sun Sep 30 04:36:21 2007
DHCP Server Address	172.16.1.2
MAC Address	00:0A:19:82:09:15
IP Address	172.16.1.207
Subnet Mask	255.255.0.0
MTU	1500
Gateway Address	172.16.0.3
DNS 1 (Primary)	172.16.0.3
DNS 2 (Secondary)	N/A

6.9.2. Routing Table

The Routing Table doesn’t exit in bridge mode. Except for in router mode, the customer can not view the table at all.

ROUTING TABLE

Destination	Gateway	Netmask	Flags	Metric	Ref	Use	Iface
192.168.1.0	0.0.0.0	255.255.255.0	U	0	0	0	eth1
172.16.0.0	0.0.0.0	255.255.0.0	U	0	0	0	eth0
0.0.0.0	172.16.0.3	0.0.0.0	UG	0	0	0	eth0

The table shows the routing information of IP-Phone.

6.9.3. Missed Calls

MISSED CALLS

Caller	Call Time	IsReviewd

Del All

This table shows the list of missed calls. For some Record, if the link under its Caller has been clicked by customer, its reviewed table will be displayed, else not reviewed.

6.9.4. Redial List

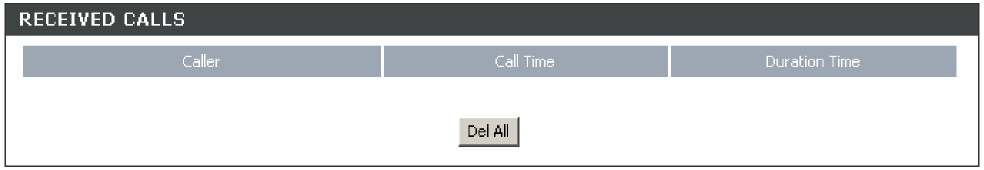
REDIAL LIST

Phone Number	Call Time	Duration Time

Del All

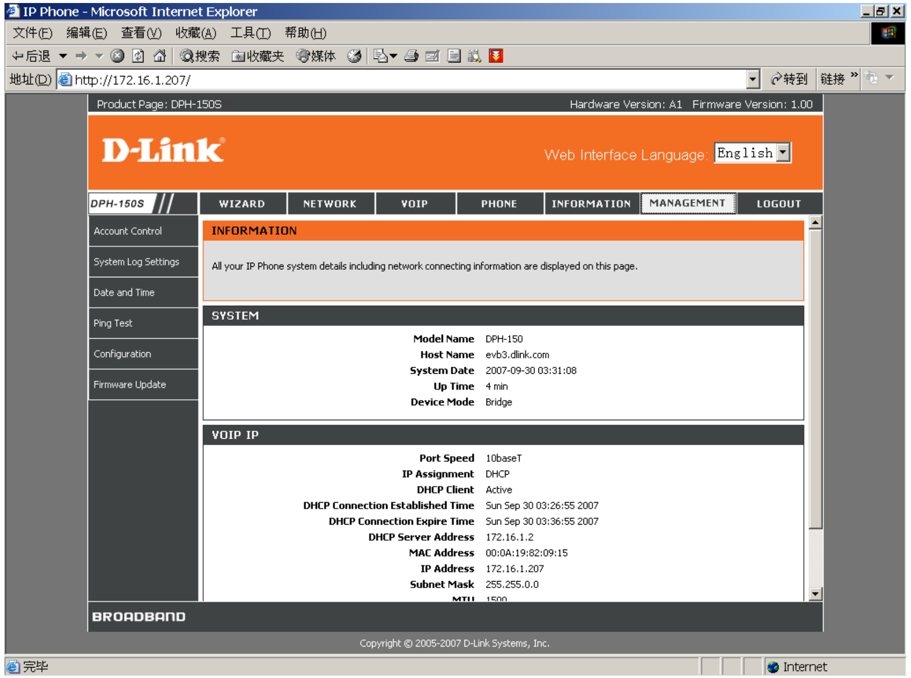
This table shows the list of dialed calls.

6.9.5. Received Calls

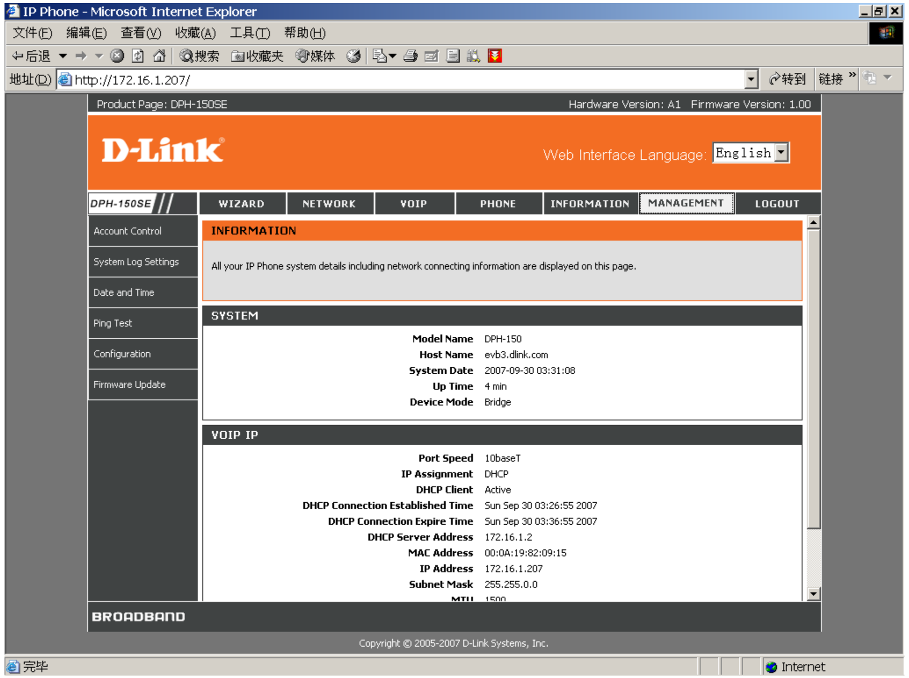


This table shows the list of received calls.

6.10. Management



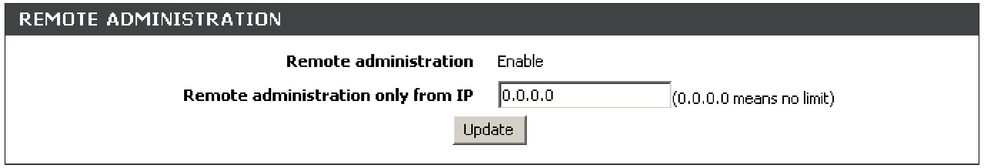
<For DPH-150S>



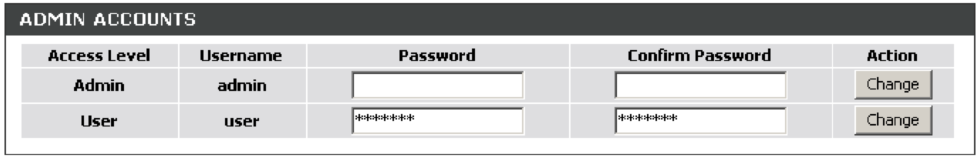
<For DPH-150SE>

6.10.1. Account Control

This page is used to display http port for remote control and to set remote administration IP address. If we set remote administration IP, only terminal with that IP can configure through web.



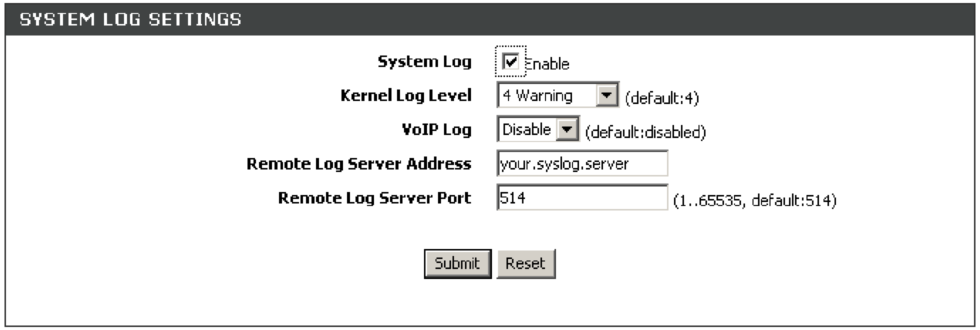
Remote administration: enable or disable remote administration
Http port for remote: http port for remote, it can not be changed on web.
Remote administration only from IP: Remote administration only from IP. 0.0.0.0 means no limit.



This web page is used to set accounts logging in web. Access Level includes admin/user/guest three different levels, admin is the highest level, an account with the level can do all the handles on web; oppositely, guest is the lowest one, the account with it can do nothing but view some pages. Clearly, user has the middle level power, the account with only can do part of handling actions.

6.10.2. System Log Settings

Customer can set system log configuration on this web page. There are two kinds of log information: Kernel log and VOIP log, and Kernel log can be set on different priority levels, on a certain level, the logs have higher levels number will not be recorded. The lower priority level, the log information recorded will be more important.



System Log: enable/disable (default) the system log function.

Kernel Log Level: 0(Emergence), 1(Alert), 2(Critical), 3(Error), 4(Warning)(default), 5(Notice), 6(Info), 7(Debug), Disabled(Kernel Log is disabled).

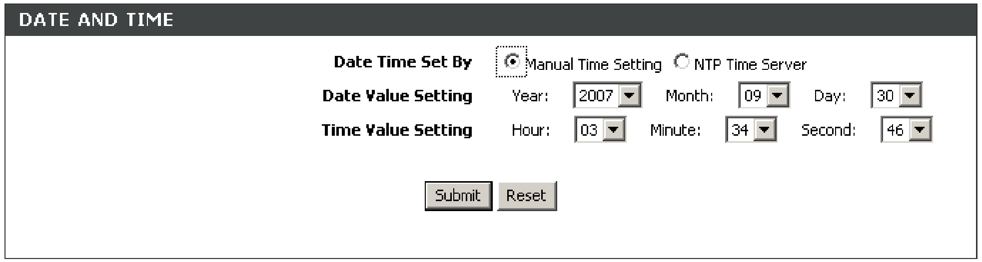
VoIP Log: enable/disable (default) VOIP Log.

Remote Log Server Address: assigns the remote log server's IP address.

Remote Log Server Port: assigns the remote log server's port.

6.10.3. Date and Time

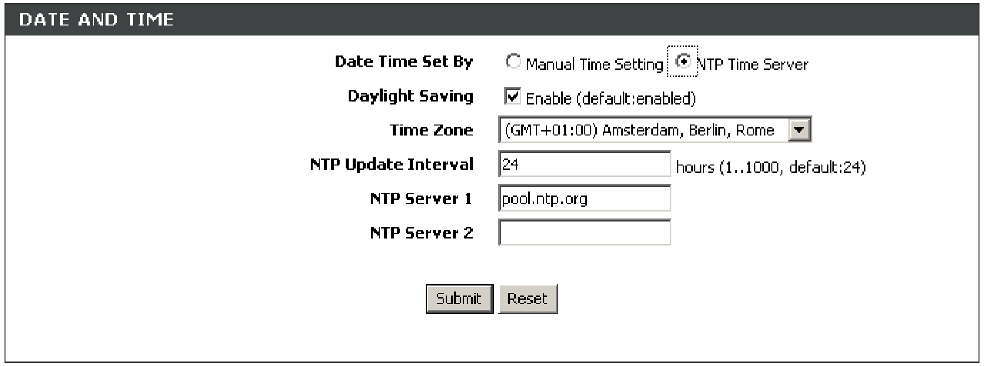
Customers can easily change displayed time on LCD by configuration this web page. There are two ways to configure current phone time on web: manual or NTP. Time zone also can be configured on web in order adapt to customers in different countries.



Date Time Set: manual set date time or via ntp server

Date Value Setting: set date in phone

Time Value Setting: set time in phone



Date Time Set: manual set date time or via ntp server

Time Zone: time zone

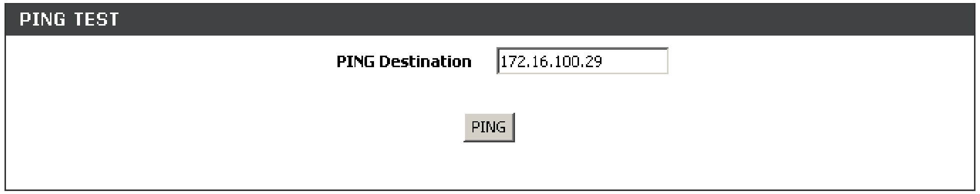
NTP Update Interval: a period to exchange ntp message.

NTP Server 1: ntp server domain or IP address

NTP Server 2: ntp server domain or IP address

6.10.4. Ping Test

This page is used to test network environment. Some time we found some problems such as can’t make a call or register fail on SIP server, and then we can use this page to test network’s connection.

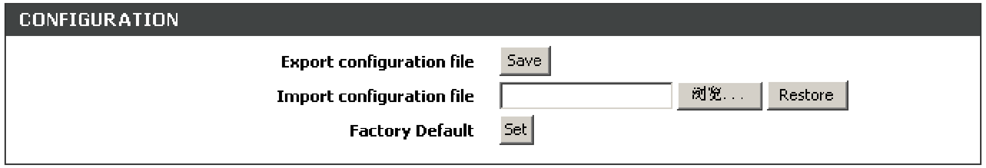



```
PING 172.16.100.29 (172.16.100.29): 56 data bytes
84 bytes from 172.16.100.29: icmp_seq=0 ttl=128 time=3.2 ms
84 bytes from 172.16.100.29: icmp_seq=1 ttl=128 time=1.3 ms
84 bytes from 172.16.100.29: icmp_seq=2 ttl=128 time=1.7 ms

--- 172.16.100.29 ping statistics ---
3 packets transmitted, 3 packets received, 0% packet loss
round-trip min/avg/max = 1.3/2.0/3.2 ms
```

PING Destination: IP address to ping. This item is in order to test network connection.

6.10.5. Configuration



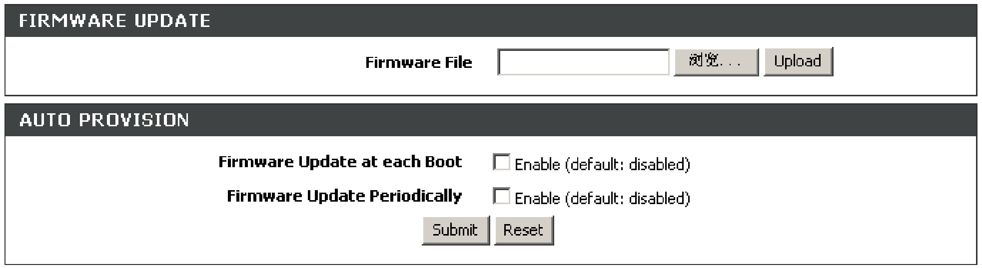
Save: save device current configuration to local file
Restore: upload a local file to restore as device configuration
Factory Default: set device configuration to factory default setting
View Current Configuration: open a static page to show current configuration of system.

6.10.6. Firmware Update

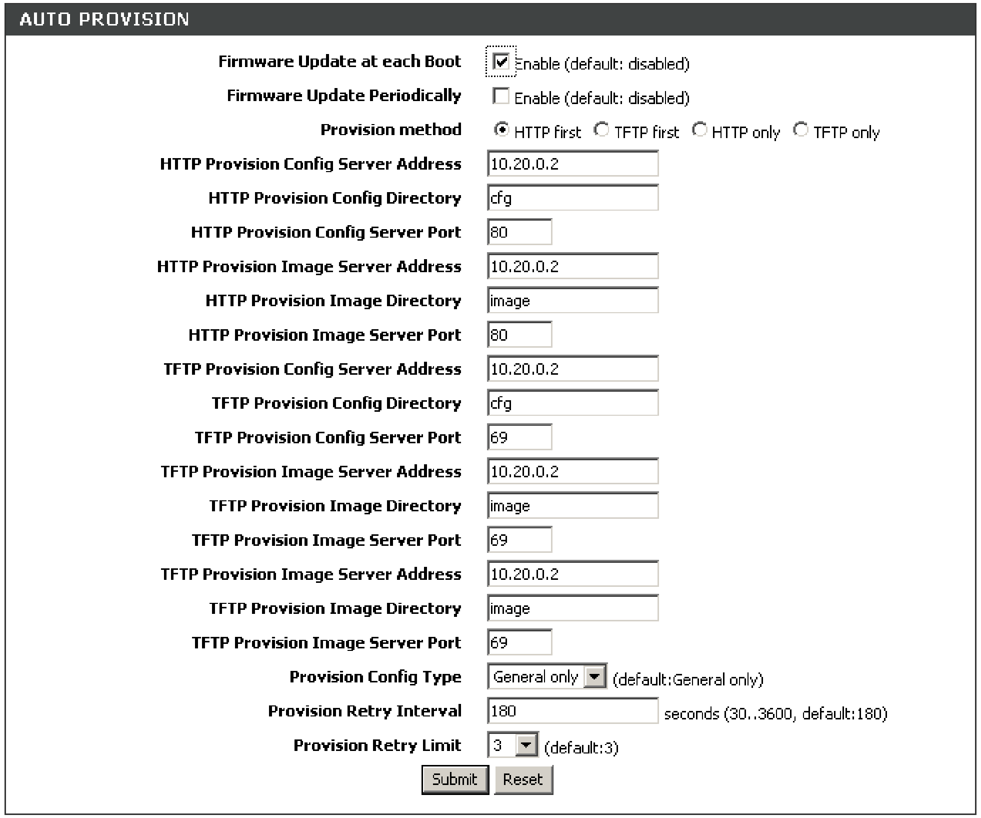
There are two ways to update firmware by web. We can update a firmware directly through web or use auto provision. If we choose auto provision, we need install http server or TFTP server on host.

Note:

We recommend user should reboot their IP Phone before Firmware Update whatever update a firmware directly through web or use auto provision.



Firmware File: firmware to update
Firmware Update at each Boot: enable or disable auto provision at each phone boot
Firmware Update Periodically: enable or disable periodical auto provision



Firmware Update at each Boot: enable or disable auto provision at each phone boot

Firmware Update Periodically: enable or disable periodical auto provision

Provision method: select auto provision method

HTTP Provision Config Server Address: IP address of http provision server which stores configuration file

HTTP Provision Config Directory: directory which stores configuration file

HTTP Provision Config Server Port: http provision configures server port number

TFTP Provision Config Server Address: IP address of TFTP provision server which stores configuration file

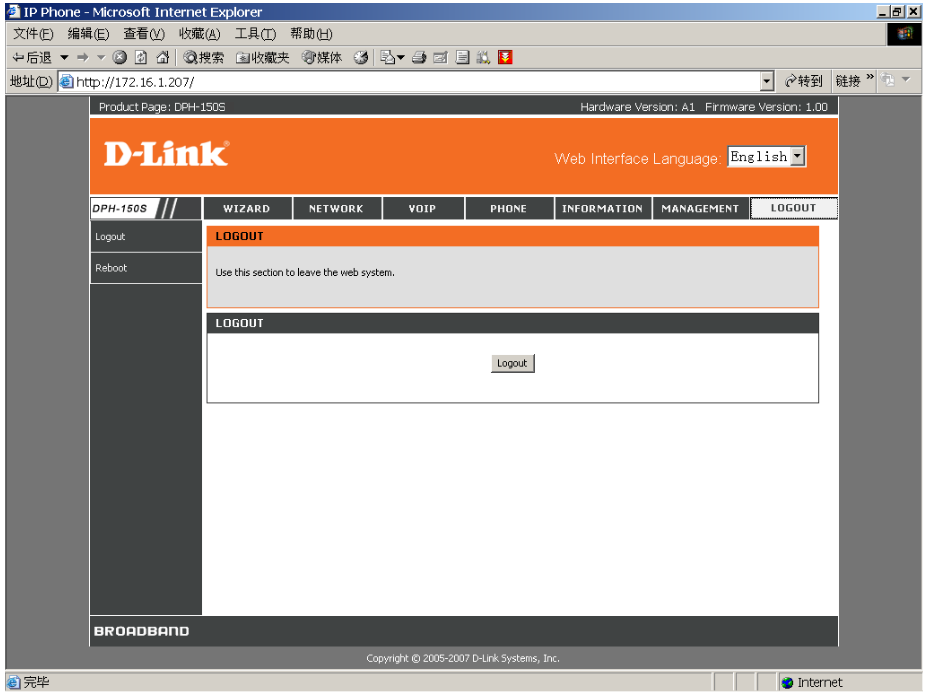
TFTP Provision Config Directory: directory which stores configuration file

TFTP Provision Config Server Port: TFTP provision configures server port number

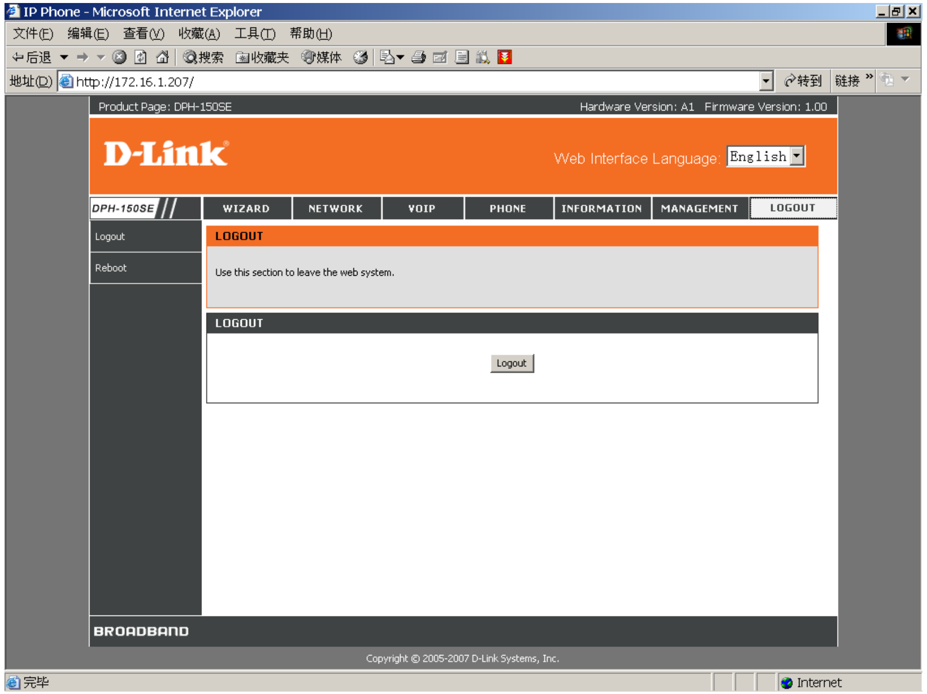
HTTP Provision Image Server Address: IP address of http provision server which stores image file

HTTP Provision Image Directory: directory which stores image file
HTTP Provision Image Server Port: http provision image server port number
TFTP Provision Image Server Address: image server IP address via TFTP to update image
TFTP Provision Image Directory: directory which placed the image
TFTP Provision Image Server Port: TFTP provision image server port number
Provision Config Type:

6.11. Logout



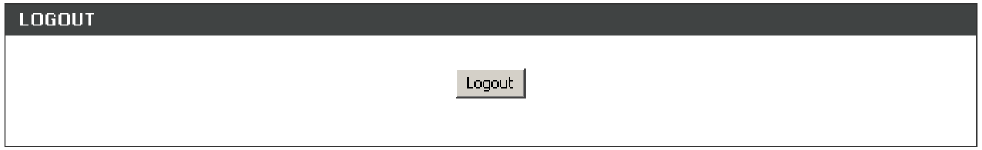
<For DPH-150S>



<For DPH-150SE>

6.11.1. Logout

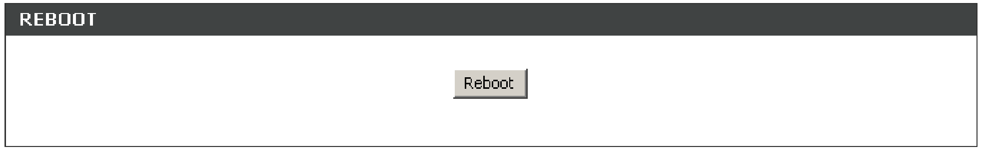
We can select this page to log out. If some people want to configure this phone, he must input user name and password to log in.



click the logout button to log out

6.11.2. Reboot

We can use this page to reboot the system. We must reboot system after config some web pages with red remarks on bottom.



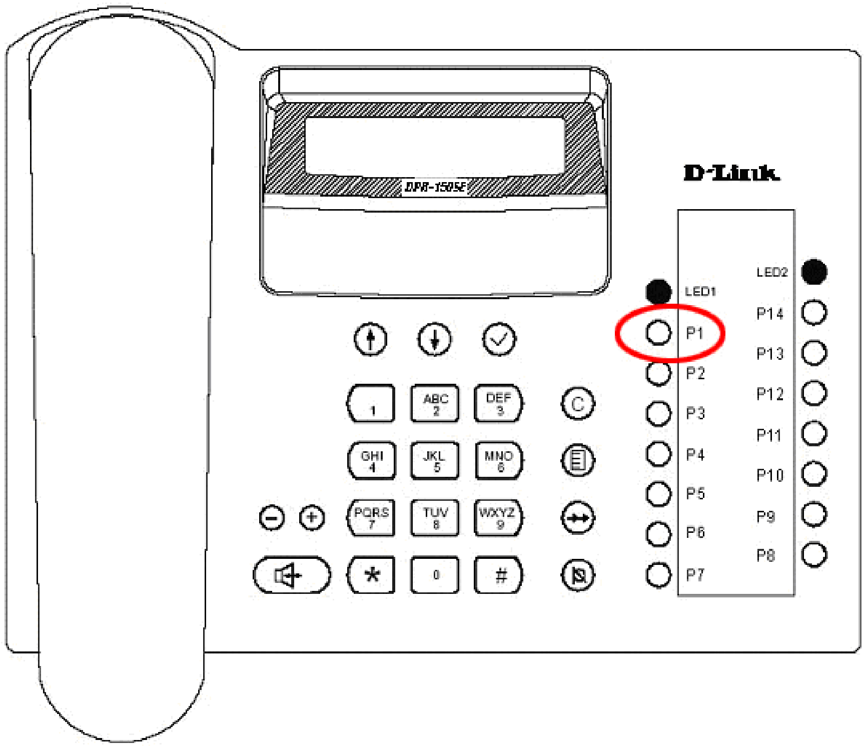
Click Reboot button to restart system.

7. EXAMPLE CONFIGURE

7.1. Function Key

7.1.1. Hold

This phone is supported two lines call. Once line 1 is talking, line 2 is on hold status, you can use [Flash] function key to switch line.



NOTE:

If only one line in talking, once you press [Flash] function key, due to the other line is free, so you will hear the dial tone. It can build the second line through dial, then press [Flash] function key to switch.

7.1.2. Transfer

This phone support two kinds transfer function: attended transfer and unattended transfer.

- Attended Transfer - You control the transfer. Press Transfer to put the caller on hold and dial the party to which you want to transfer the call. Either announce the transfer or hang up after pressing the Transfer button again. That's means when transferring a call, you can optionally announce the call to the transfer recipient.
- Unattended Transfer - The SIP telephone controls the transfer after you dial the number to which you want to transfer the call. Press the number of the third party ended with the "transfer" key to which you want to transfer the call and hang up or go on-hook. The transfer automatically occurs without speaking to the receiving party.

The following procedures describe each transfer type in detail.

Phone A, phone B, phone C

Sending a call to another phone (attended transfer)

To perform attended transfers:

1. A, as a caller, to call B, A and B in talking, A want to transfer active to C, A press the function key [Flash] button, then A hear a dial tone.

2. A Dial the number to C. it can be pressed [#] or time out 5 seconds when A finish inputting the call number to dial, A can hear ring back if C is available.

3. If A want to transfer the call to C, press the function key [Transfer] button, then A can press [Transfer] to transfer the call to C, and A will hear busy tone, B and C is in talking; If A do not want to transfer the call to C, press the function key [Flash] button again to return to original line B to continue talking with A.

4. Hang up A's handset.

Sending a call to another phone (Unattended Transfer)

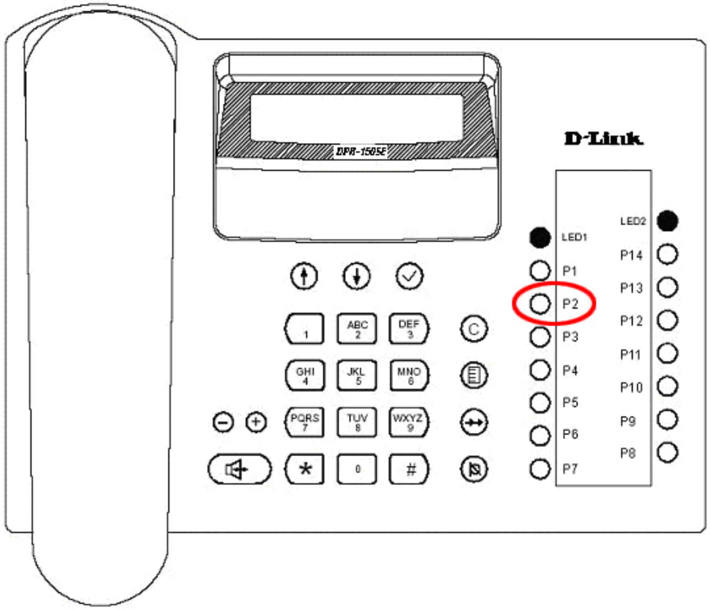
To perform unattended transfers.

1. With the call active and one other line available, press the function key[Flash].

2. Dial the number to which you want to transfer the call.

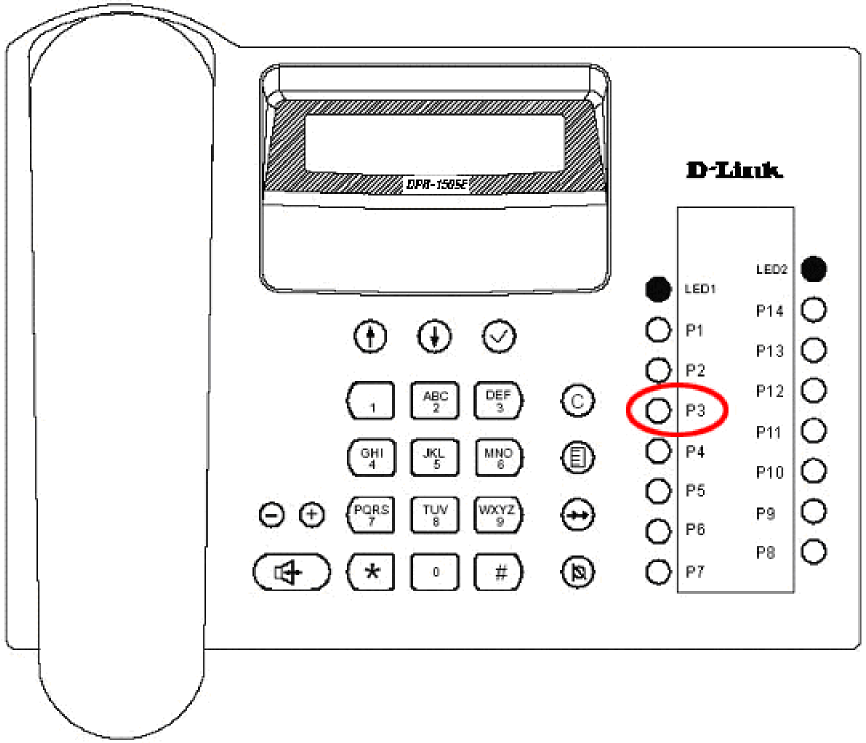
The call is sent to the extension or number you dialed.

3. Hang up your handset.



7.1.3. Conference

This phone support two kinds conference function:
(1)To make a conference call, press the 'Flash' key to hold the current call, press the number of the third party with '#' or time out 5s for ending. When the third party is answered, press the function key [Conference] to begin the conference.
(2) To make a conference call, press the 'Flash' key to hold the current call, then press Conference key, hearing the dialling tone to press the number of the third party with '#' or time out 5s for ending. When the third party is answered, beginning the conference.



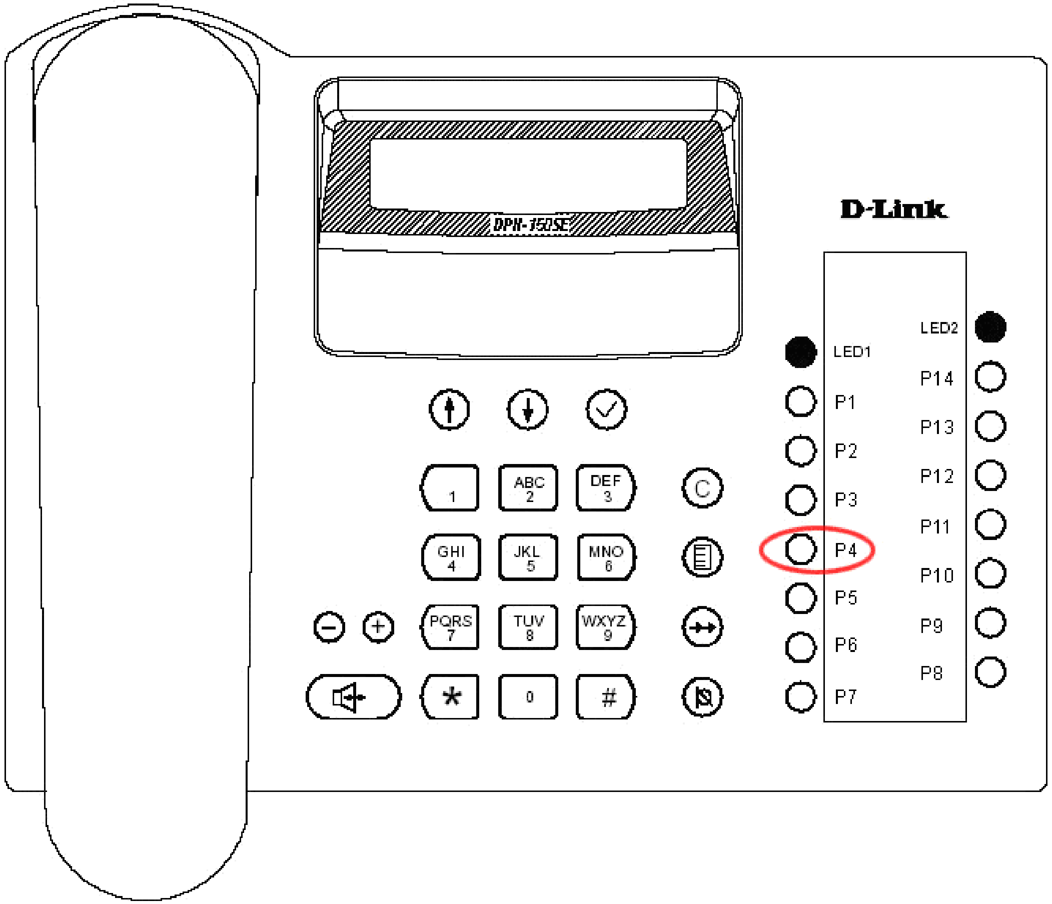
NOTE:

When in a conference call, the middle party hooks on, the conference call will be finished.

If the initial party presses the conference key again, the telephone will require you to choose line1 or line2. Once the initial party chooses one line，this line call will be held and the other line call will be break down。

7.1.4. Missed calls

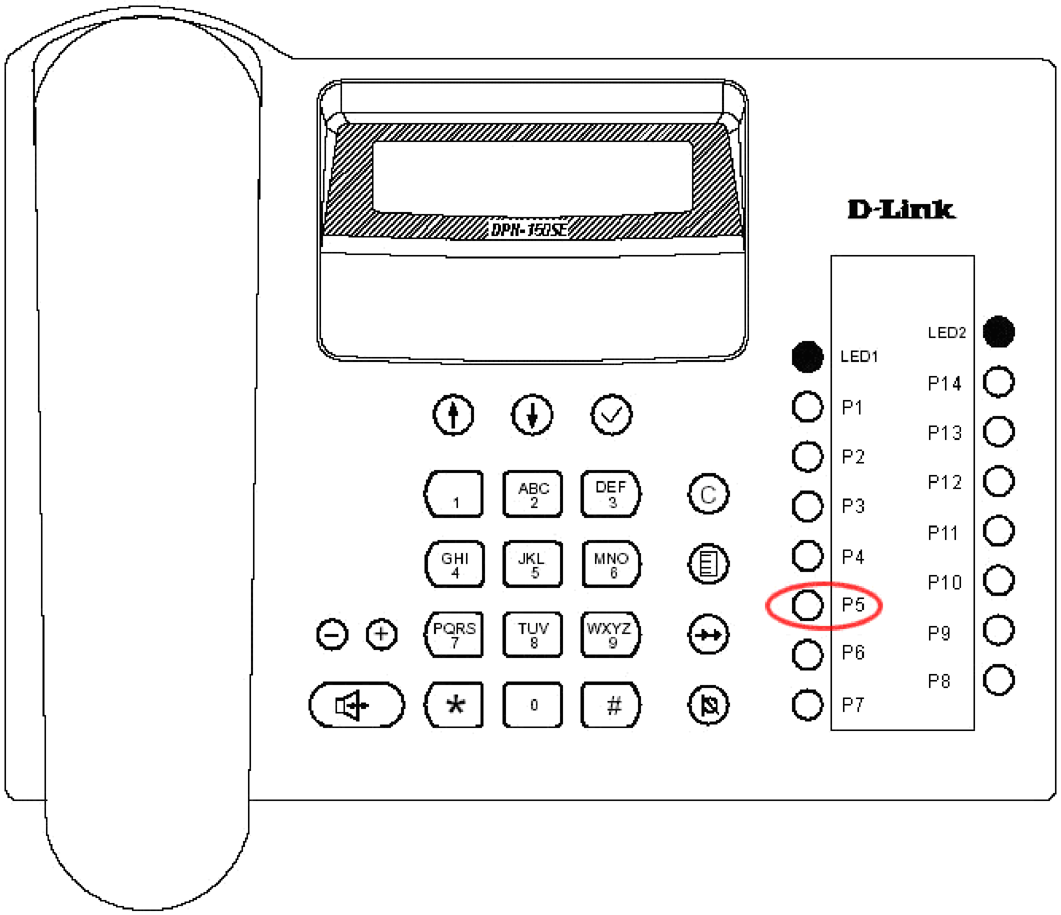
When press the function key [MissedCall], it will view missed call list. Also user can view the list by pressing [Up]/ [Down] key(or key).
If there is no missed call, the LCD screen will show “No Missed Call”.



NOTE:
When there’re new missed calls, the LED1 indicator will flash. After user check all of the new missed call, the LED1 indicator will turn off.

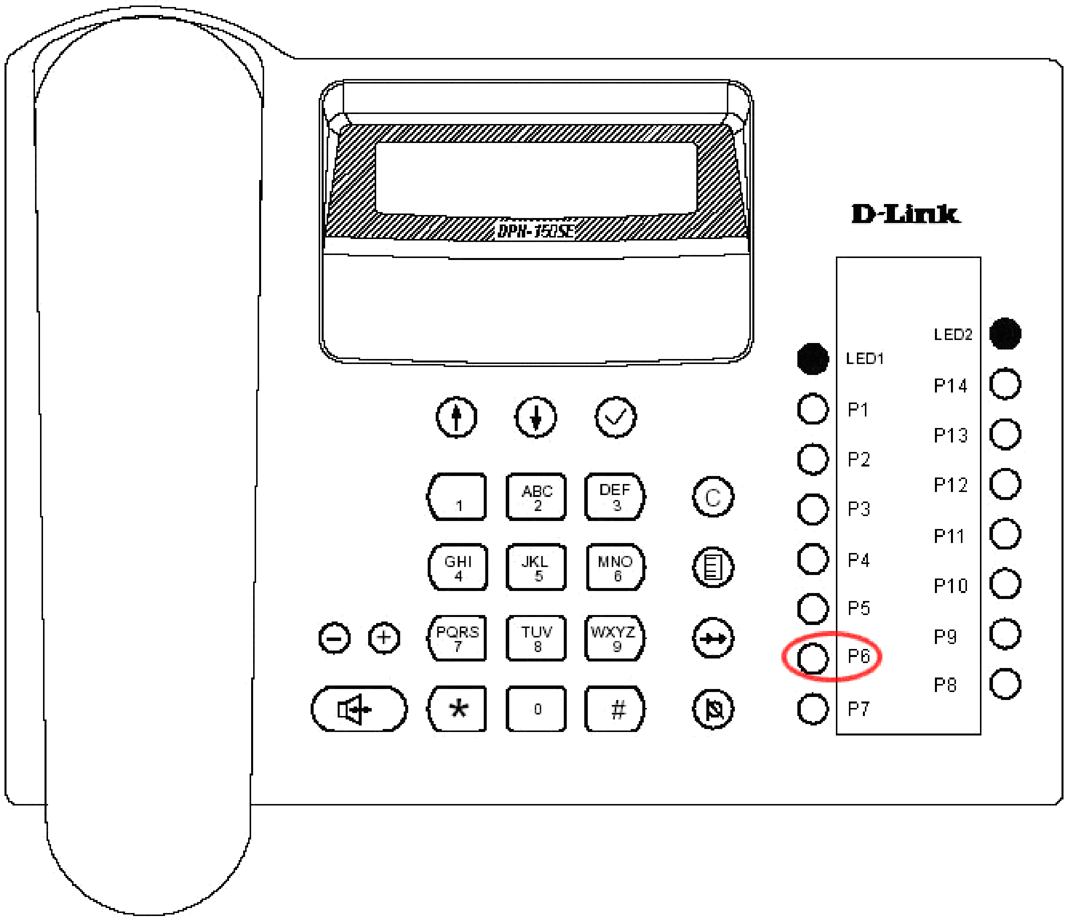
7.1.5. Received calls

When press the function key [ReceivedCall], it will view received call list.
If there is no received call, the LCD screen will show “No Received Call”.



7.1.6. DND

DND means Do Not Disturb from any other call. If you want to have this function, please press [DND] function key; if you don't want to have this function, press [DND] function key again.

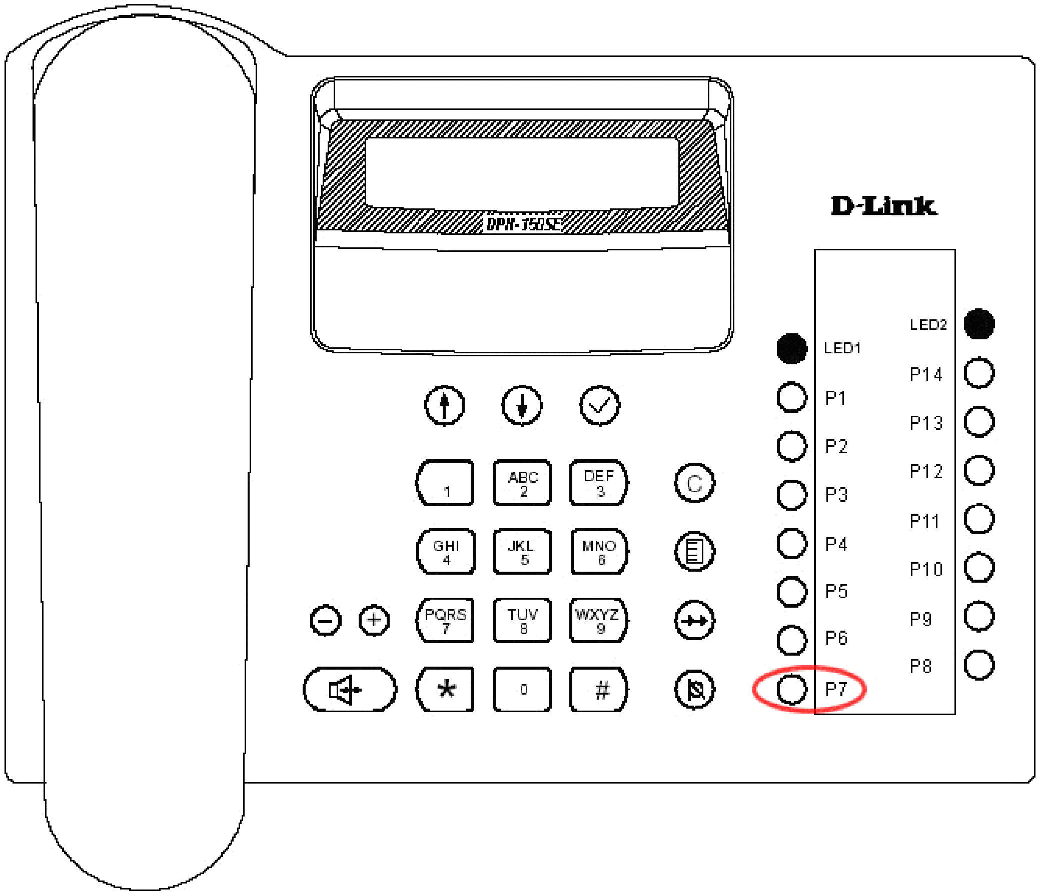


NOTE:

If DND function is active, LED1 indicator will be turned on, in this state, you can call anyone or everybody can't call you.

7.1.7. Headset

This phone has headset and Mic jacks, you can connect the headset used the jacks. If you plug in the headset, whatever the current state, press [Headset] function key, and then switch voice path to the headset or close the voice path of headset to handset or handfree.



NOTE:

When headset is in active, LED2 indicator will be turn on.

8. Troubleshooting

1. How to input ‘a, b, c…’ character or ‘1, 2, 3…’ digital number?
   You can press the key [#] to switch.
2. Why dial a phone number, the call dial out after a moment?
   This phone has two ways to dial out the call, one is wait 5 seconds then phone will dial out automatically, the other way is press [#] after you finish inputting dial number, the call will dial out immediately.
3. If ADSL user can use this phone to make a call?
   Yes, this phone supports the user for ADSL, please configure your networking setting or call your service.
4. Why finish this phone setting, such function doesn’t in active?
   Once we finished setting, You should reboot system then function will be active.