C56 VoIP Phone User Manual

Cortelco
1703 Sawyer Road
Corinth, MS 38834
USA
www.cortelco.com
Tel: (662)287-5281
Fax:(662)287-3889

POWERED BY

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Safety Notices

Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supplies may cause damage to the device, affect the behavior or induce noise.
- Before using the external power supply, please be sure it is for use with your power voltage. Incorrect power voltage may cause fire and damage.
- Please do not damage the power cord. If the power cord or plug is damaged, do not use it. This may cause fire or electric shock.
- The power plug should be accessible at all times because this is the only way to remove power from the device.
- Handle the phone carefully. Do not drop it or shake it. Rough handling can cause internal damage.
- Do not install the device in direct sunlight. Also do not put the device on carpets or cushions, or other poorly ventilated locations. This may cause fire or overheating.
- Avoid exposure to temperatures above 40℃, below 0℃ or high humidity. Avoid wetting the unit with any liquid.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean the device. If cleaning is necessary use a soft cloth that has been slightly dampened in a mild soap and water solution.
- Do not touch the power cord or network cable during a thunderstorm. There is a slight risk of electrical shock.
- Do not attempt to open the device. Consult your authorized dealer for repair.

Table of Contents

1 Introducing C56 VoIP Phone

1.1 Thank you

Thank you for purchasing the C56(P) Voice Over Internet Protocol (VoIP) telephone. The C56(P) is a fully featured telephone that provides voice communication over the data network. This phone has all the features of a traditional telephone and gives access to many data service features. This guide will help you easily use the various features and services available on your phone.

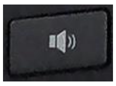
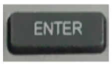
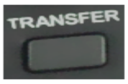
1.2 Box Contents

The following items should be packed with your telephone. Please contact your dealer if any of them are missing.

- Telephone (Main body) with display and keypad
- Handset
- Handset cord
- Power supply
- Ethernet cable

1.3 Keypad

Key	Key name	Function Description
	Navigation	Use this key to choose item in the menu, callers or phone book. Notice: the left has deleting function.
	History	Displays lists of Incoming, Outgoing, or Missed calls
	Mute	Deactivates the handset or speakerphone microphone. Allows you to talk without being heard by the distant party.
	Volume -/+	Adjust the volume by pressing these two keys.

	Redial	When off hook, this will dial the last called number. In stand-by mode, it will check the Outgoing Call.
	Speaker phone	Activate speakerphone mode.
	Indicator light	This light blinks to indicate a missed call.
	SYSINFO	Displays phone settings such as phone number, IP address, gateway address, etc.
	ENTER	Used to enter next menu or confirm settings
	MWI	Accesses voice mail system.
	TRANSFER	Performs blind or attended transfers. See Section 3.1.4 for more details.
	CONFERENCE	Creates a conference (3-Way) call. See Section 3.1.5 for more details.
	HOLD	Places caller on hold.
	EXIT	Return to a previous menu, cancel a setting or reject an incoming call
	Keyboard	Dial phone numbers

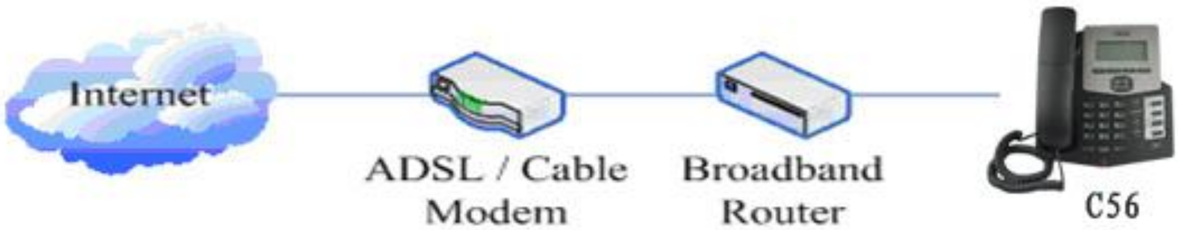
1.4 Input/Output Ports

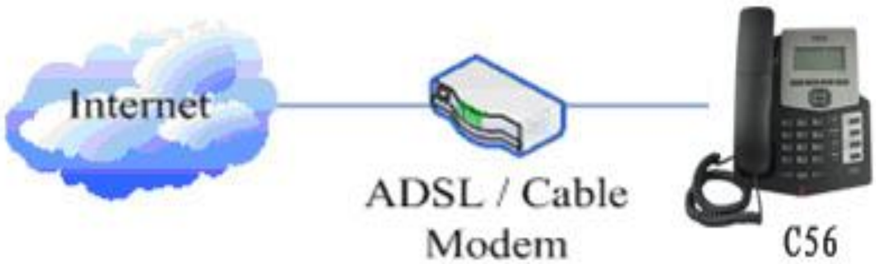
Port	Port name	Description
	Power switch	Input: 5V AC, 1A
	WAN	10/100M Connect to Network
	LAN	10/100M Connect to PC
	Handset	Port type: RJ-9 connector

2 Initial Connection and Setting

2.1 Connecting the phone

1. Connect to the network. Use the Ethernet cable in the package to connect the WAN port on the back of your phone to an Ethernet port. The following two figures show connection options.





2. Connect the handset to the handset jack using the handset cable in the package.
3. Connect the power supply to the DC port on the back of the phone. Connect the power supply to a standard power outlet. Note that the power supply will not be needed if your network provides Power over Ethernet (PoE), and you have a C56P.

4. The phone's LCD screen displays "WAIT LOGON". Later, a ready screen displays the date, time and current network mode.

If your LCD screen displays different information from the above, more information may need to be entered. Please refer to the next section. If your phone registers into your IP telephony Server, it is ready to use. If not, continue to read for more configuration information.

2.2 Network Settings

DHCP is supported by default. This allows the phone to receive an IP address and other network-related settings (Netmask, IP gateway, DNS server) from the DHCP server. If no DHCP server is available, the network connection settings must be changed. Follow the instructions below to change to either PPPoE or static IP.

2.2.1 PPPoE Mode

1. Press the 3 key for three seconds.
2. Press ENTER to confirm.
3. Press OK. The LCD will display "INPUT PASSWORD".
4. Input the password (default value is 123).
5. Press ENTER. The LCD will display "NETWORK".
6. Press ENTER. The LCD will display "WAN".
7. Press UP ARROW.
8. Press ENTER. The LCD will display "STATIC NET".
9. Press UP ARROW.
10. Press ENTER. The LCD will display "USER NAME".
11. Press ENTER.
12. Press LEFT ARROW.
13. Enter your PPPoE account number. Use LEFT ARROW to delete if necessary.
14. Press ENTER. The LCD will display the PPPoE account number.
15. Press EXIT to return to the previous menu.
16. Press UP ARROW. The LCD will display "PASSWORD".
17. Press ENTER.
18. Press LEFT ARROW.
19. Enter your PPPoE password. Use LEFT ARROW to delete if necessary.
20. Press ENTER. The LCD will display the password.
21. Press EXIT four times.
22. Press DOWN ARROW until the LCD displays "SYSTEM".
23. Press ENTER. The LCD will display "SAVE".
24. Press ENTER. The LCD will display "ARE YOU SURE".
25. Press ENTER. The LCD will display "SAVING NOW" and then display "SAVE".
26. Press EXIT twice.
27. Press and hold 3 until the LCD displays "ARE YOU SURE".
28. Press ENTER. The LCD will display "CHANGING". This means the phone is trying

to switch to PPPoE mode. When the PPPoE icon at the top of the LCD stops blinking, the mode change is complete.

2.2.2 Static IP Mode

1. Press and hold 1 for three seconds.
2. Press ENTER to confirm.
3. Press ENTER. The LCD will display “INPUT PASSWORD”.
4. Input the password (default is 123).
5. Press ENTER. The LCD will display” NETWORK”.
6. Press ENTER. The LCD will display “LAN”.
7. Press UP ARROW.
8. Press ENTER. The LCD will display “STATIC NET”.
9. Press ENTER. The LCD will display “IP”.
10. Press ENTER.
11. Press LEFT ARROW.
12. Input the IP address. Use “*” to enter the periods in the IP address.
13. Press ENTER. The LCD will display the IP address.
14. Press EXIT to return to the previous menu.
15. Press DOWN ARROW. The LCD will display “DNS2”.
16. Press ENTER.
17. Press LEFT ARROW.
18. Input the secondary DNS address.
19. Press ENTER. The LCD will display the DNS address.
20. Press EXIT to return to the previous menu.
21. Press DOWN ARROW. The LCD will display “DNS”.
22. Press ENTER.
23. Press LEFT ARROW.
24. Input the primary DNS address.
25. Press ENTER. The LCD will display the DNS address.
26. Press EXIT to return to the previous menu.
27. Press DOWN ARROW. The LCD will display “GATEWAY”.
28. Press ENTER.
29. Press LEFT ARROW.
30. Input the gateway IP address.
31. Press ENTER. The LCD will display the gateway address.
32. Press EXIT to return to the previous menu.
33. Press DOWN ARROW. The LCD will display “NETMASK”.
34. Press ENTER.
35. Press LEFT ARROW.
36. Input the netmask.
37. Press ENTER. The LCD will display the netmask.
38. Press EXIT four times.
39. Press DOWN ARROW until the LCD displays “SYSTEM”.
40. Press ENTER. The LCD will display “SAVE”.

41. Press ENTER. The LCD will display “ARE YOU SURE”.
42. Press ENTER. The LCD will display “SAVING NOW” and then display “SAVE”.
43. Press EXIT twice.
44. Press and hold 1 until the LCD displays “ARE YOU SURE”.
45. Press ENTER. The LCD will display “CHANGING”. This means the phone is trying to switch to static IP mode. When the STATIC icon at the top of the LCD stops blinking, the mode change is complete.

2.2.3 DHCP Mode

1. Press and hold 2 until the LCD displays “ARE YOU SURE”.
2. Press ENTER. The LCD will display “CHANGING”. This means the phone is trying to switch to DHCP mode. When the DHCP icon at the top of the LCD stops blinking, the mode change is complete.

3 Basic Functions

3.1 Making a call

3.1.1 Call Device

Calls can be made using either the handset or speakerphone:

1. Handset - Pick up the handset. The icon will be shown on the LCD screen.
2. Speakerphone - Press the Speaker button. The icon will be shown on the LCD screen.

The number may also be dialed first. Then the method of speaking can be chosen.

3.1.2 Call Methods

Press an available line button then use one of the following methods to place a call.

1. Dial the desired number using the keypad.
2. Press the REDIAL button to redial the last number called.

3.2 Answering a call

If the phone is idle, lift the handset, or press the Speaker button to answer using the speaker phone.

During the conversation, you can alternate between Handset and Speaker phone by pressing the speaker button or picking up the handset.

3.3 Call Hold

1. Press the Hold key to put the active call on hold.
2. While a call is on hold, you can establish another call by dialing the desired number and

confirming it with the # button.
3. Pressing the HOLD button during the second call will resume the first call.

3.4 Call Waiting

1. When a third party calls during an established call, the LCD will display the incoming call number. Press the HOLD key to place the established call on hold and answer the incoming call.
2. Press # to hang up the established call and answer the incoming call.

NOTE: Call Waiting service must be enabled.

3.5 Call transfer

3.5.1 Blind Transfer

During a conversation, press the transfer key, dial the number to which the call is to be transferred followed by "#" and then hang up.

3.5.2 Attended Transfer

During a conversation, press the hold key, dial the number to which the call is to be transferred followed by "#". After the third party answers, press transfer key to complete the transfer.
NOTE: Call waiting and call transfer must be enabled.
NOTE: The SIP server must support RFC3515.

3.6 3-way conference call

1. Press the hold key during an active call.
2. The first call will be placed on hold and dial tone will be heard.
3. Dial the number to be added to the conference.
4. Press Send.
5. When the call is answered, press CONF to add the caller to the conference.

4 Advanced Functions

4.1 Dialing Pause

In some cases, it is desirable to have the phone pause when outputting digits. For example, a call to an IVR system that requires a password should wait until the system answers before dialing the password.
To insert a pause press the HOLD key while pre-dialing. Each press of the HOLD key will insert a 2 second pause and will show on the screen as “- -“. For example, if the LCD shows 123 -- -- 45, the phone will output 123, wait 4 seconds and then output 45.

4.2 MWI(Message Waiting Indication)

This LED will flash to indicate a new voicemail. Pressing the MWI key will access the voicemail if the key has been configured correctly.

4.3 Redial / Unredial

If B is on a call when A calls, A will get busy tone. If A wants to connect to B as soon as B is available, he can use the redial function. To use this feature, A dials a prefix and then B's number.

When the redial function is activated, A will check B's calling status every 60 seconds. When B is available, A's phone will ring. When A goes off hook, the phone will call B automatically. If A does not want to call B, the redial function can be cancelled by dialing a prefix plus B's number.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 is the redial prefix code. A can dial *3* plus B's phone number to activate the redial function.

4 is the unredial prefix code. A can dial *4* to cancel the redial function.

The user can select any prefix as long as it does not interfere with dialing rules.

4.4 Click to dial

If User A browses to User B's phone number or SIP address in the contact page and clicks it, User A's phone will ring. After he goes off hook, the phone will call User B.

Note: This feature requires that the PBX support click to dial.

4.5 Auto answer

If this feature is activated, the phone will answer incoming calls after a programmable delay.

5 Web Configuration

5.1 Introduction of configuration

5.1.1 Configuration Methods

There are three methods which can be used to configure this phone:

- Phone keypad – As discussed in previous sections
- Web browser - Recommended way
- Telnet with CLI command

5.1.2 Password Configuration

There are two levels of access: root level and general level. A user with root level access can browse and set all configuration parameters, while a user with general level can set all configuration parameters except server parameters for SIP or IAX2.

- Default user with general level:
  - Username: guest
  - Password: guest
- Default user with root level:
  - Username: admin
  - Password: admin

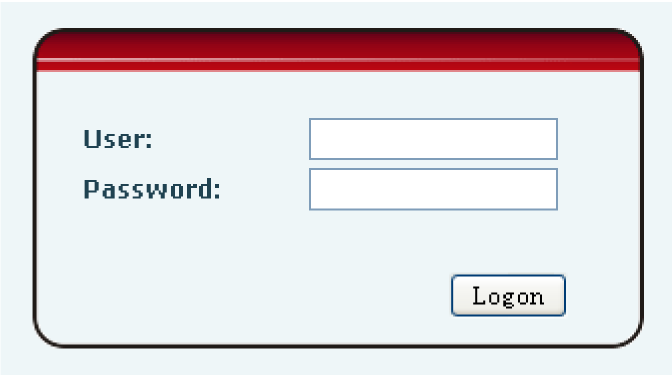
The default password for the phone screen menu is 123.

5.2 Setting via web browser

Enter the phone's IP address into the address bar of the web browser. This assumes that the pc and the phone are on the same subnet. Note: Internet Explorer, Firefox, Chrome, or Safari are supported browsers.

If the IP address is not known, it can be displayed on the phone's LCD by pressing the Menu->Status.

After entering the IP address, the following screen is displayed.

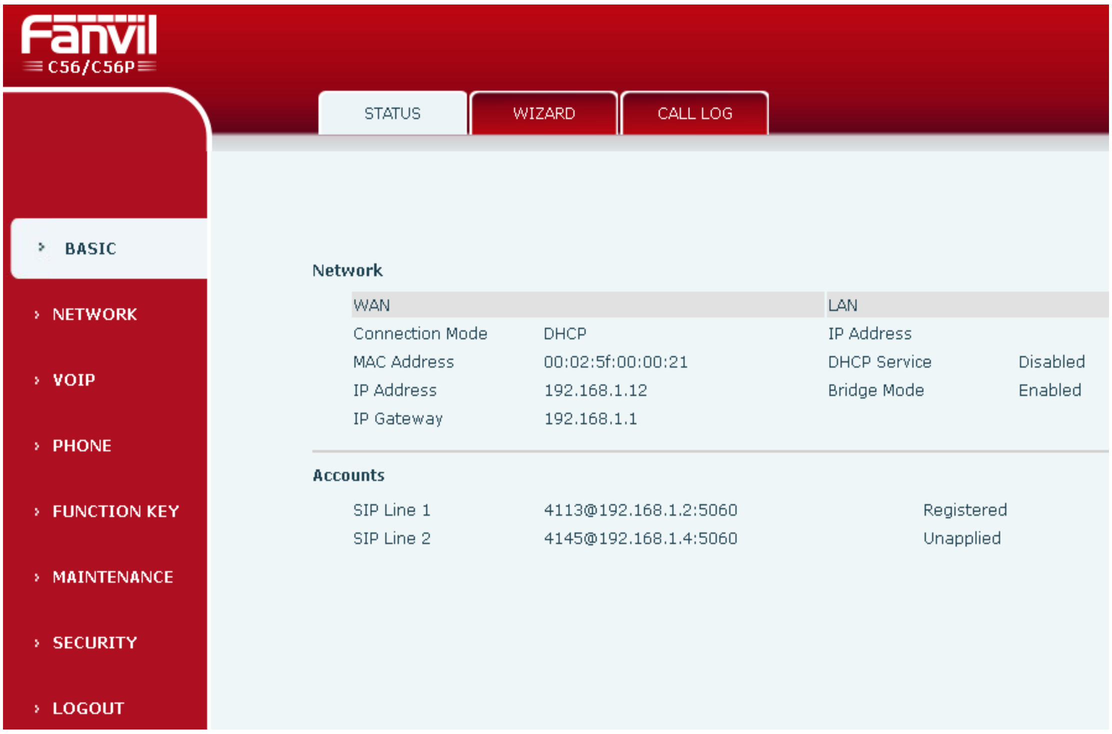


After configuring the IP phone, remember to click SAVE under the Maintenance tab. If this is not done, the phone will lose the modifications when it is rebooted.

5.3 Configuration via WEB

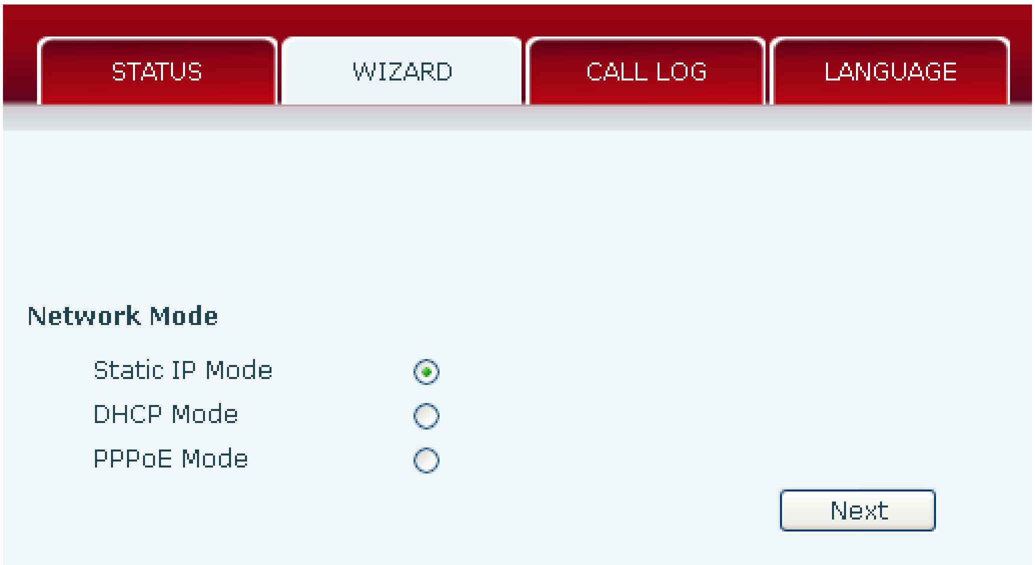
5.3.1 BASIC

5.3.1.1 Status



Field Name	Explanation
Network	Shows the configuration information for WAN and LAN port, including connection mode of WAN port (Static, DHCP, PPPoE), MAC address, IP address of WAN port and LAN port, DHCP server status for LAN port (ENABLED or DISABLED).
Accounts	Shows the phone numbers and registration status for the 2 SIP LINES.

5.3.1.2 Wizard



Select the appropriate network mode. The phone supports three network modes:

1 Static: The parameters of a Static IP connection must be provided by your ISP.

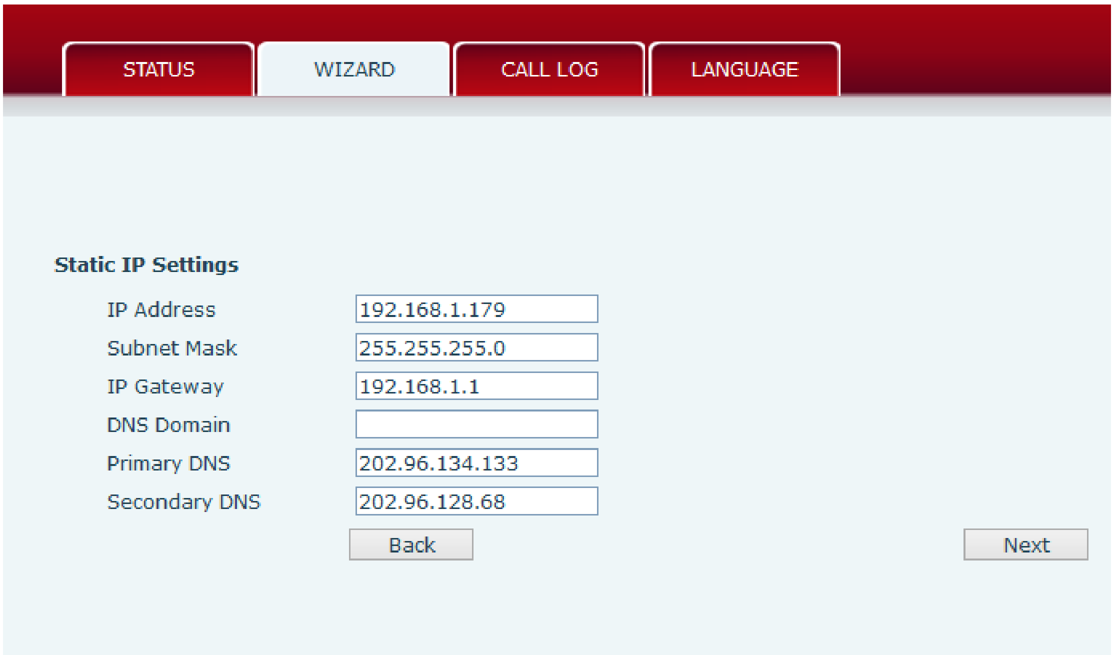
2 DHCP: In this mode, network parameter information will be obtained automatically from a DHCP server.

3 PPPoE: In this mode, you must enter your ADSL account and password.

Refer to Section 2.2 for detailed information about configuring the network parameters.

5.3.1.2.1 Static IP

If Static IP is selected, this screen will be displayed. Information provided by the ISP should be entered.



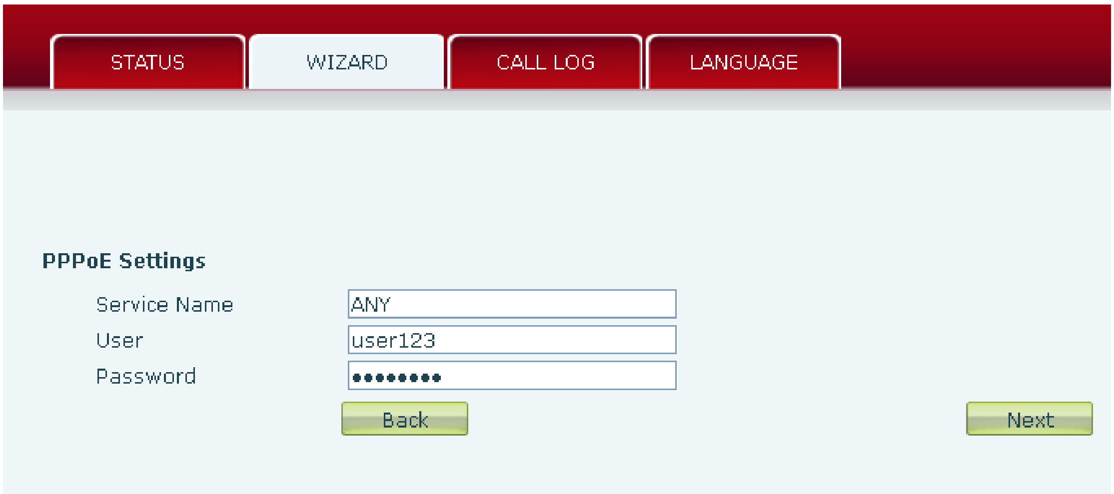
Click Back to return to the Wizard screen. Click Next to go to Quick SIP Settings

5.3.1.2.2 DHCP

After selecting DHCP and clicking NEXT, the Quick SIP Settings screen will appear. Click Back to return to the Wizard screen. Click Next to go to the Summary screen.

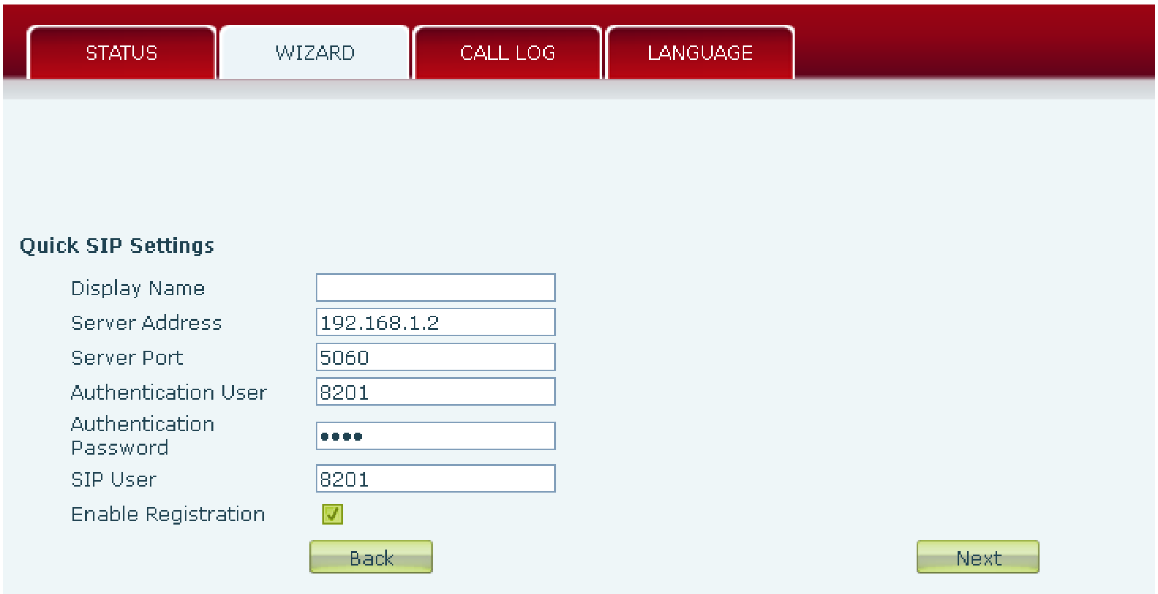
5.3.1.2.3 PPPoE

If PPPoE is selected, this screen will appear. Enter the information provided by the ISP.



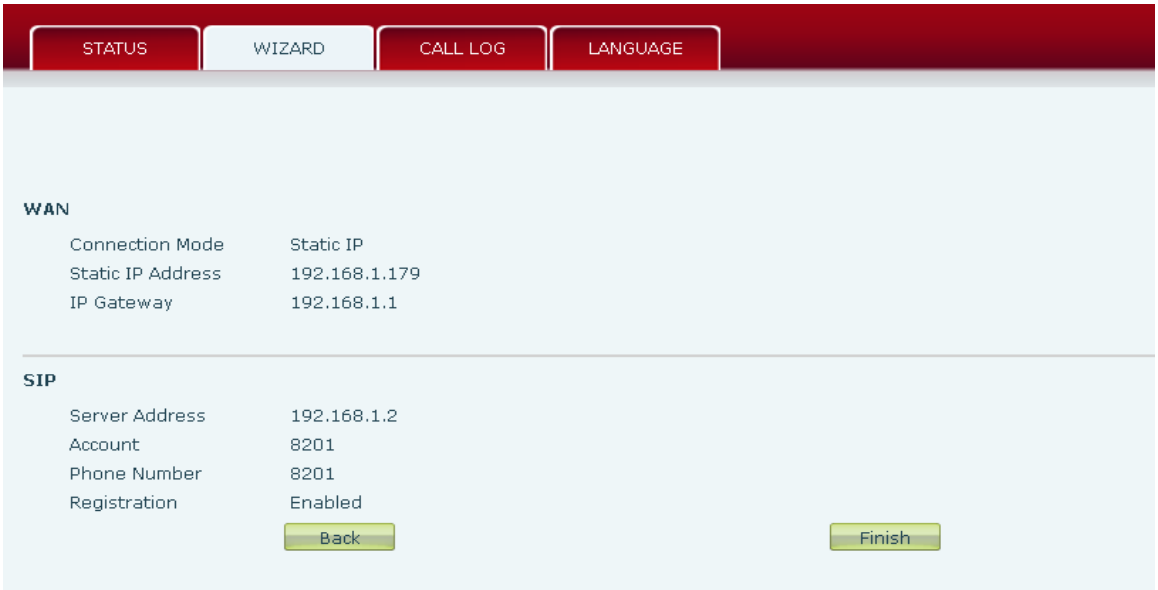
Click Back to return to the Wizard screen. Click Next to go to Quick SIP Setting.

5.3.1.2.4 Quick SIP Settings



Field Name	Explanation
Display Name	The name shown in caller ID.
Server Address	SIP server address either IP address or URI.
Server Port	SIP server port (usually 5060).
Authentication User	Login name or Authentication ID.
Authentication Password	SIP password.
SIP User	Phone number.
Enable Registration	Submits registration information. Normally checked.

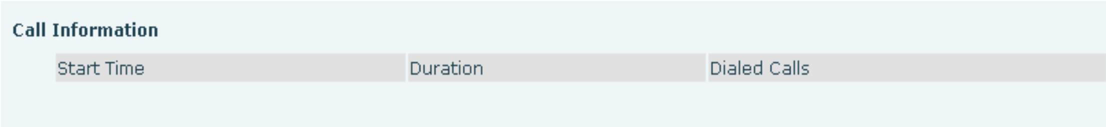
Click Back to return to the IP Address screen. Click Next to see summary screen.



Click Finish button to save settings and reboot. After the reboot, SIP calls can be made.

5.3.1.3 Call Log

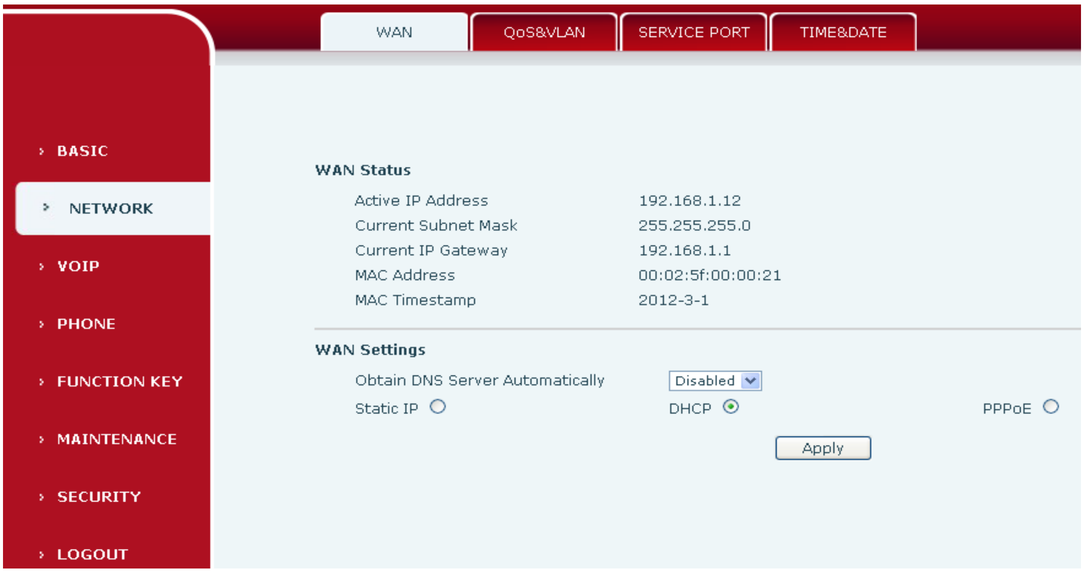
Outgoing call logs can be seen on this page.



Field Name	Explanation
Start Time	Start time of the outgoing call
Duration	Duration of the outgoing call.
Dialed Calls	Account, protocol, and line of the outgoing call.

5.3.2 Network

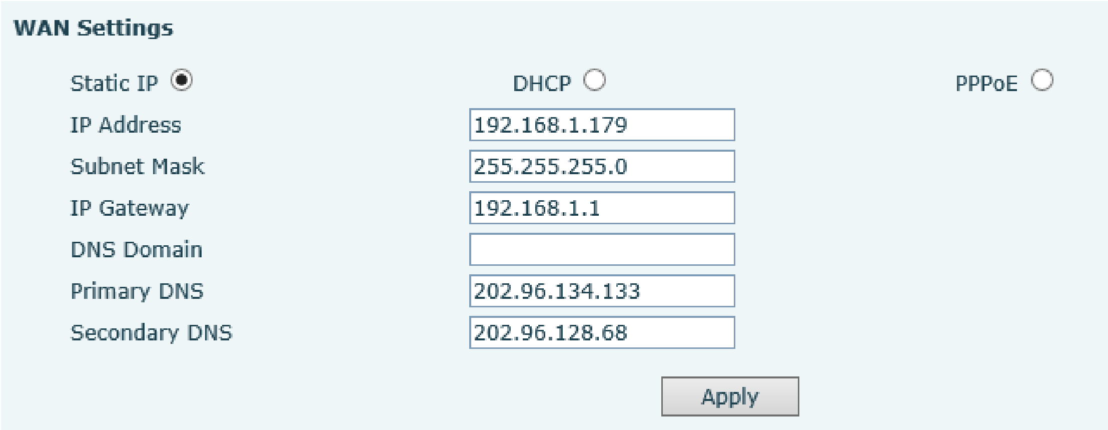
5.3.2.1 WAN Config



<table>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td>Active IP Address</td><td>The current IP address of the phone.</td></tr>
<tr><td>Current Subnet Mask</td><td>The current Subnet Mask.</td></tr>
<tr><td>Current IP Gateway</td><td>The current Gateway IP address.</td></tr>
<tr><td>MAC Address</td><td>The MAC address of the phone.</td></tr>
<tr><td>MAC Timestamp</td><td>Time the MAC address was obtained.</td></tr>
<tr><th colspan="2">WAN Settings</th></tr>
<tr><td colspan="2">The phone supports three network modes. These are also discussed in Section 2.2.
• Static: 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.</td></tr>
</table>

5.3.2.1.1 Static IP

If Static IP is chosen, the screen below will appear. Enter values provided by the ISP.

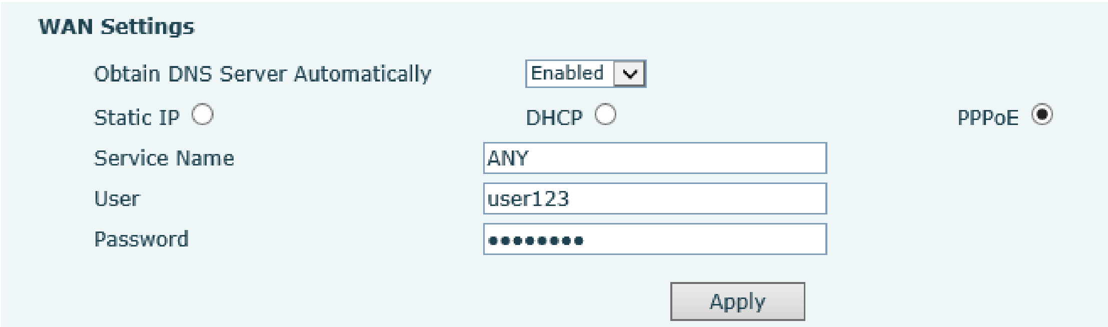


5.3.2.1.2 DHCP

If DHCP is chosen, all configuration information will be provided by a DHCP server. Contact the ISP to determine if DHCP is used.

5.3.2.1.3 PPPoE

If PPPoE is chosen, the screen below will appear. Enter the information provided by the ISP.



Service Name	IP Address or name of DSL Server
User	DSL User Name or Login ID
Password	DSL Password

After entering the new settings, click the APPLY button. The phone will save the new settings and apply them. If a new IP address was entered for the phone, it must be used to login to the phone after clicking the APPLY button.

5.3.2.2 Qos & VLAN Config

The phone supports 802.1Q/P protocol and DiffServ configuration. Use of a Virtual LAN (VLAN) allows voice and data traffic to be separated.

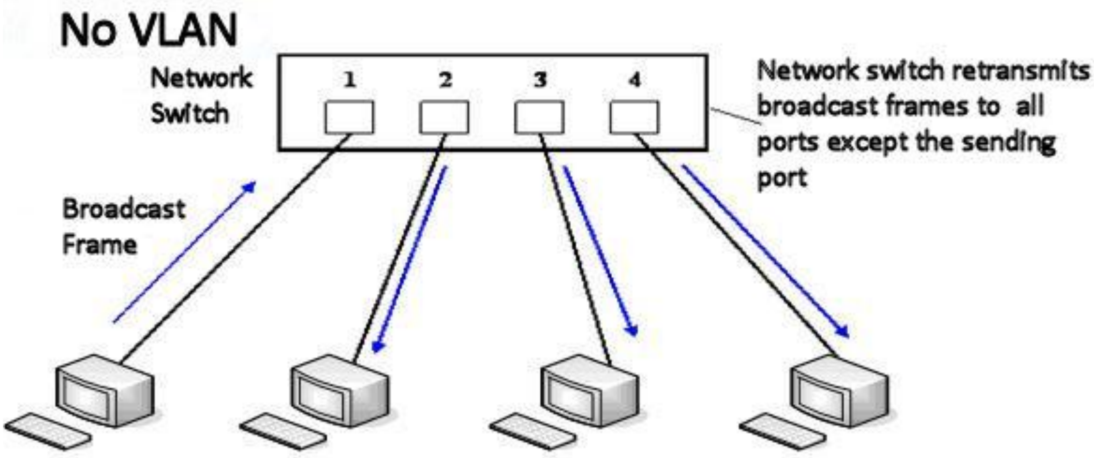


Chart 1

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

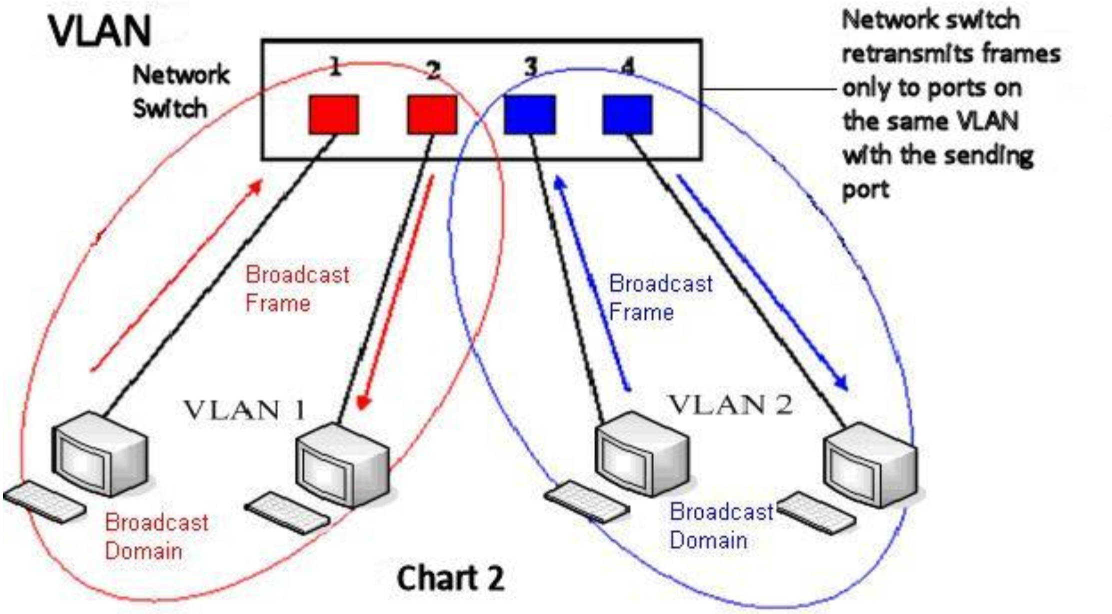
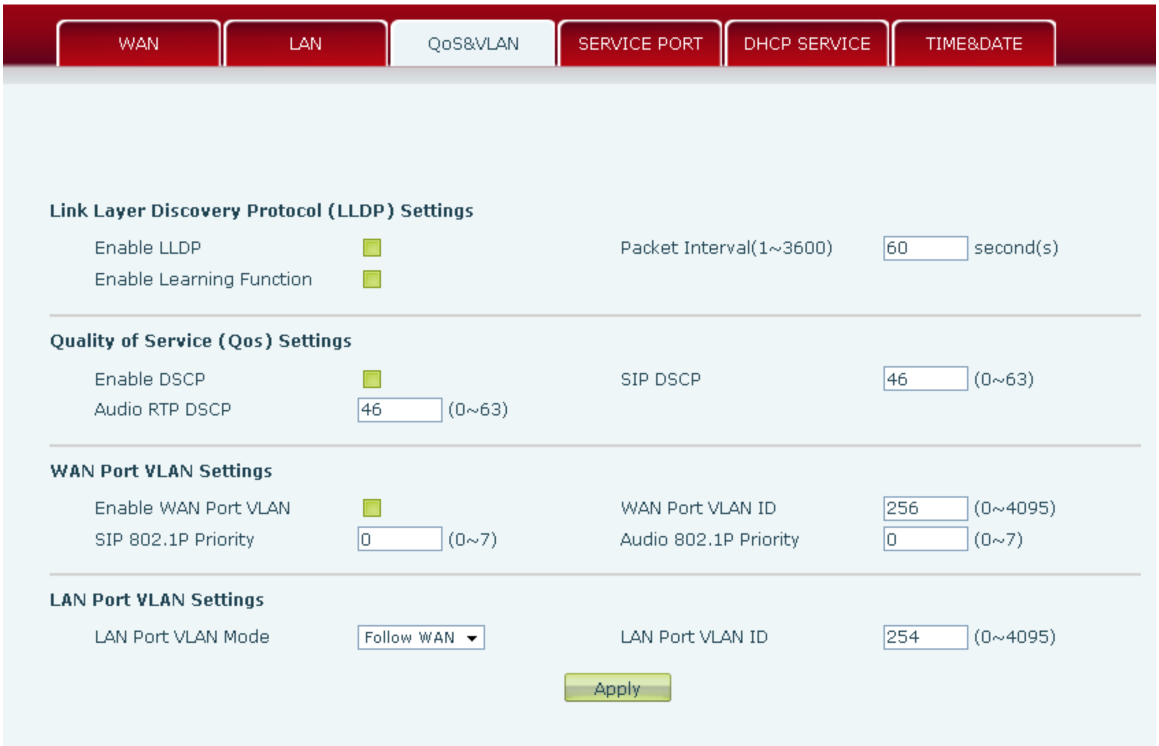


Chart 2 shows an example with two VLANs indicated by red and blue. In this example, frames broadcast from Port 1 will only go to Port 2 since Ports 3 and 4 are in a different VLAN. VLANs can be used to divide a network by restricting the transmission of broadcast frames.

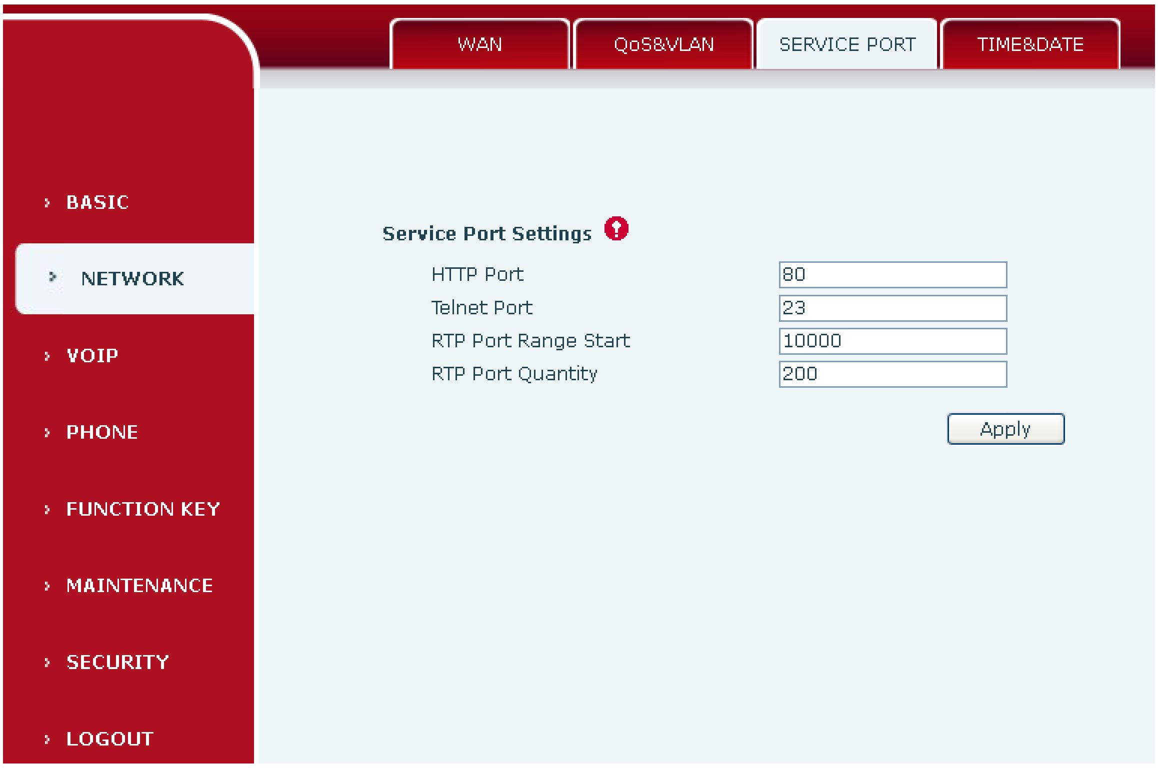
Note: In practice, VLANs are distinguished by the use of VLAN IDs.



Field Name	Explanation
Enable LLDP	Enable or Disable Link Layer Discovery Protocol (LLDP)
Packet Interval	The time interval for sending LLDP Packets
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.
Enable DSCP	Enable or Disable Differentiated Services Code Point (DSCP)
SIP DSCP	Specify the value of the SIP DSCP in decimal
Audio DSCP	Specify the value of the Audio DSCP in decimal
Enable WAN Port VLAN	Enable or Disable WAN Port VLAN
WAN Port VLAN ID	Specify the value of the WAN Port VLAN ID. Range is 0-4095
SIP 802.1P Priority	Specify the value of the voice 802.1p priority. Range is 0-7
Audio 8021P Priority	Specify the value of the signal 8021.p priority. Range is 0-7
LAN Port VLAN Mode	Follow WAN: LAN Port ID is same as WAN ID Disable: Disable Port VALN Enable: Specify a VLAN ID for the LAN port which is different from WAN ID
LAN Port VLAN ID	Used when the VLAN ID is different from WAN ID. Range is 0-4095

5.3.2.3 Service Port

Set the port values for Telnet/HTTP/RTP on this page.



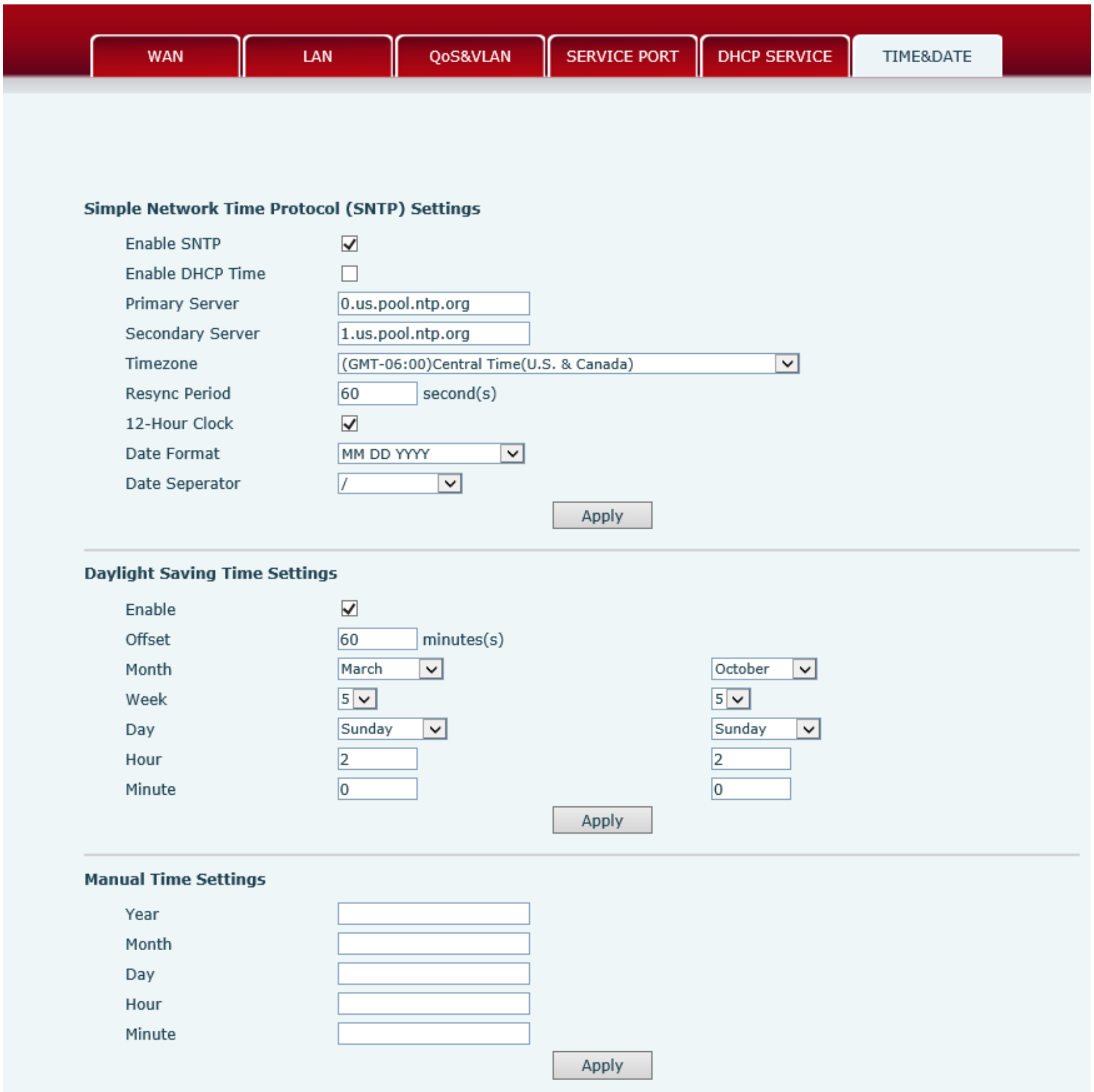
Field Name	Explanation
Web Server Type	Specify Web Server Type – HTTP or HTTPS
HTTP Port	Port for web browser access. Default value is 80. To enhance security, change this from the default. Setting this port to 0 will disable HTTP access. Example: The IP address is 192.168.1.70 and the port value is 8090, the accessing address is http://192.168.1.70:8090.
Telnet Port	Port for Telnet access. The default is 23.
RTP Port Range Start	Set the beginning value for RTP Ports. Ports are dynamically allocated.
RTP Port Quantity	Set the maximum quantity of RTP Ports. The default is 200.

Notes:

1. Any changes made on this page require a reboot to become active.
2. It is suggested that changes to HTTP Port and Telnet ports be values greater than 1024. Values less than 1024 are reserved.
3. If the HTTP port is set to 0, HTTP service will be disabled.

5.3.2.4 TIME&DATE

Set the time zone and SNTP (Simple Network Time Protocol) server on this page. Daylight savings time configuration and manual time and date entry are also done on this page



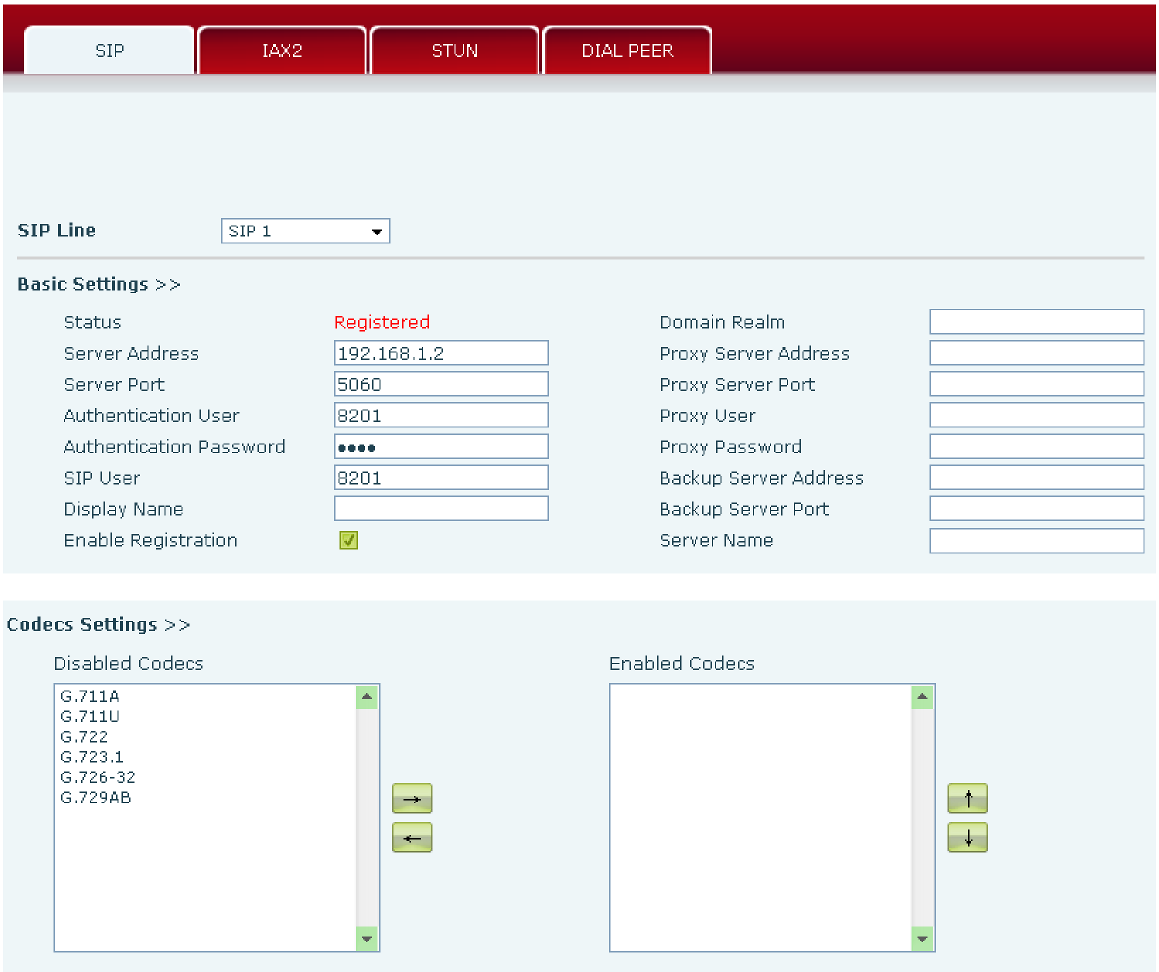
Field Name	Explanation
Simple Network Time Protocol (SNTP) Settings	
Enable SNTP	Enable or Disable SNTP
Enable DHCP Time	If this is enabled, phone will synchronize time with DHCP server.
Primary Server	IP address of Primary SNTP Server
Secondary Server	IP address of Secondary SNTP Server
Time Zone	Local Time Zone
Resync Period	Time between resync to SNTP server. Default is 60 seconds.
12 -Hour Clock	If checked, clock is 12 hour mode. If unchecked, 24 hour mode. Default is 24 hour mode.
Date Format	Specify the date format. Fourteen different formats are available.
Date Separator	Four date separators are available: /, - , . , space
Daylight Saving Time Settings	
Enable	Enable daylight saving time.
Offset(minutes)	DST offset. Default is 60 minutes.
Month	Start and end month for DST

Week	Start and end week for DST
Day	Start and end day for DST
Hour	Start and end hour for DST
Minute	Start and end minute for DST
Manual Time Settings	
Enter the values for the current year, month, day, hour and minute. All values are required.	
Note: Be sure to disable SNTP service before entering manual time and date.	

5.3.3 VOIP

5.3.3.1 SIP Configuration

Configure a SIP server on this page.



Field Name	**Explanation**
Choose the sip line to configured (SIP 1 – SIP 2). Click the dropdown arrow to select the line.	
Status	Shows registration status. Will show “Registered” if registered or “Unapplied” if not registered.
Server Address	SIP server IP address or URI.
Server Port	SIP server port. Default is 5060.
Authentication User	SIP account name (Login ID).
Authentication Password	SIP registration password.
SIP User	Phone number assigned by VoIP service provider. Phone will not register if there is no phone number configured.

Display Name	Set the display name. This name is shown on Caller ID.
Enable Registration	Check to submit registration information.
Domain Realm	SIP Domain if different than the SIP Registrar Server.
Proxy Server Address	SIP proxy server IP address or URI（This is normally the same as the SIP Registrar Server)
Proxy Server Port	SIP Proxy server port. Normally 5060.
Proxy User	SIP Proxy server account.
Proxy Password	SIP Proxy server password.
Backup Server Address	Backup SIP Server Address or URI (This server will be used if the primary server is unavailable)
Backup Server Port	Backup SIP Server Port
Server Name	Name of SIP Backup server
Codecs Settings	
Click on the desired codec to select it. Then use the Left/Right arrow keys to move to the Enabled or Disabled List. Use the Up/Down arrow to change the priority of enabled codecs.	
Advanced SIP Settings	
Forward Type	There are 3 call forwarding modes plus Disabled. Disabled：No call forwarding – Default mode Busy：If the phone is busy, incoming calls will be forwarded. No answer：If there is no answer, incoming calls will be forwarded after a specified time. Always：All incoming calls will be forwarded.
Forward Number	Number to which calls are to be forwarded.
No Ans. Fwd Wait Time	Used in conjunction with Call Forward No Answer. Wait time in seconds before call is forwarded.
Transfer Timeout	Time interval between sending “bye” message and hanging up after the phone transfers a call.
Enable Hotline	Activate Hot Line feature. Automatically call a number by going off hook.
Hotline Number	Number to be called in Hot Line Mode.
Warm Line Wait Time	Used in Hot Line Mode. Time the phone waits after off hook before dialing the hot line number.
SIP Encryption	Enable/Disable SIP Encryption.
SIP Encryption Key	SIP Encryption key.
RTP Encryption	Enable/Disable RTP Encryption.
RTP Encryption Key	RTP encryption key
Enable Auto Answer	Activate Auto Answer mode. If activated, phone will automatically answer an incoming call.
Auto Answer Timeout	Used in conjunction with Auto Answer. The phone will answer an incoming call after the Auto Answer Timeout
Enable Session Timer	If enabled, this will refresh the SIP session timer per RFC4028.
Session Timeout	Refresh interval if Session Timer is enabled.

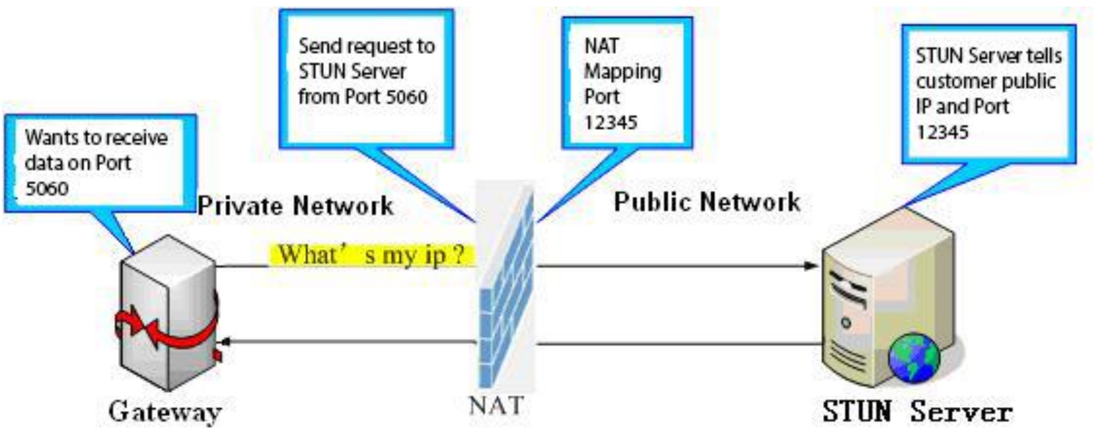
Subscribe For MWI	If enabled, the phone will send Message Waiting Indication (MWI) Subscribe message to the SIP Server
MWI Number	Specify the number to call to retrieve Voice Messages.
Subscribe Period	Time interval between MWI Subscribe Messages.
Conference Type	Choose Conference Type, either local or network
Conference Number	Number to dial to access network conference server. Not needed if Local conference mode is chosen
Registration Expires	SIP re-registration time. Default is 3600 seconds. If the server requests a different time, the phone will change to that value.
Enable Service Code	Enables or disables the services described below. These codes will be sent to the SIP server to activate or deactivate the service.
DND On Code	Do Not Disturb (DND) – When this hot key is pressed, all calls to the extension to be rejected by the server. The incoming call record will not be displayed in the Call History.
DND Off Code	Disable Server DND as described above.
Always CFwd On Code	Always Call Forward On – When this function is enabled, the server will forward all calls to a designated number. The incoming call record will not be displayed in the Call History.
Always CFwd Off Code	Disable Server Always CFwd as described above.
Busy CFwd On Code	Busy Call Forward On - When this function is enabled, the server will forward all calls to a designated number if the telephone is busy. The call record will not be displayed in Call History.
Busy CFwd Off Code	Disable Server Busy CFwd as described above.
No Ans. CFwd On Code	No Answer Call Forward On - When this function is enabled, the server will forward all calls to a designated number if there is no answer within a designated time. The incoming call record will not be displayed in the Call History.
No Ans. CFwd Off Code	Disable Server No Ans. CFwd as described above.
Anonymous On Code	Anonymous On – When this function is enabled, the server will allow the phone to make anonymous calls. In other words "Anonymous" will be transmitted for Caller ID.
Anonymous Off Code	Disable Anonymous Calling function described above.
Keep Alive Type	Specifies the NAT keep alive type. If OPTION is selected, the phone will send OPTION sip messages to the server every NAT Keep Alive Period. The server will then respond with 200 OK. If UDP is selected, the phone will send a UDP message to the server every NAT Keep Alive Period.
Keep Alive Interval	Set the NAT Keep Alive Interval. Default is 60 seconds
User Agent	Set SIP User Agent value.
DTMF Type	DTMF sending mode. There are four modes: ● In-band (Relay) ● RFC2833 ● SIP_INFO ● AUTO

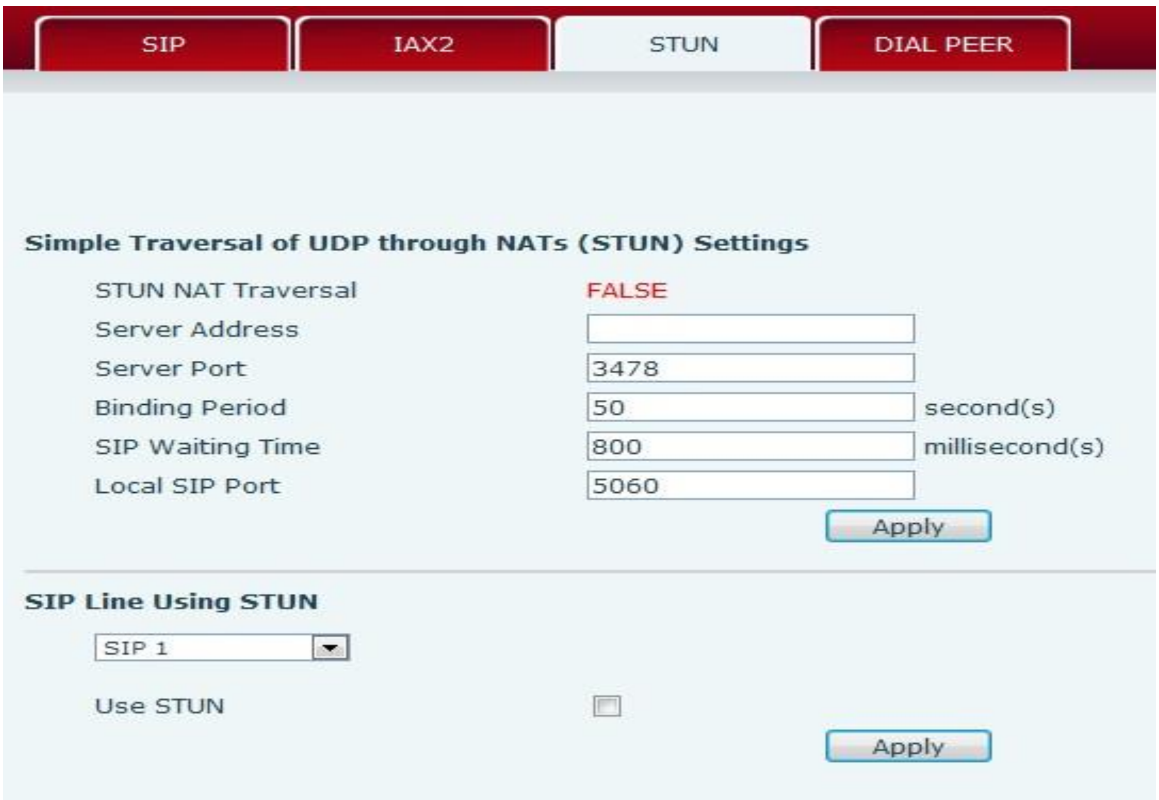
	Different VoIP Service providers may require different modes.
Local port	SIP port. Default is 5060.
Ring type	Set ring tone. There are 9 standard options and 3 user options.
Enable Rport	Enable/Disable support for NAT traversal via RFC3581 (Rport).
Enable PRACK	Enable or disable SIP PRACK function. Default is OFF. It is suggested this be used.
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Convert URI	Converts # to %23 when sending URI information.
Dial Without Registered	Allow outgoing calls without registration.
Ban Anonymous Call	Refuse Anonymous Calls
Enable DNS SRV	Enables use of DNS SRV records
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Enable BLF List	Enable/Disable BLF List
Server Type	Configures phone for unique requirements of selected server.
RFC Protocol Edition	Select SIP protocol version RFC3261 or RFC2543. Default is RFC3261. Used for servers which only support RFC2543.
Transport Protocol	Set transport protocol TCP, UDP or TLS.
Anonymous Call Edition	Set privacy support RFC3323, RFC3325 or none
Keep Authentication	Enable /disable registration with authentication. It will use the last authentication field which passed authentication by server. This will decrease the load on the server if enabled.
Ans. With a Single Codec	If enabled phone will respond to incoming calls with only one codec.
Auto TCP	Force the use of TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
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 GRUU	Support for Globally Routable User-Agent URI (GRUU)
Enable Displayname Quote	Puts quotation marks around the display-name in SIP messages. For servers that require this.
Enable user=phone	Sets user=phone in SIP messages. For compatibility with servers that require this.
Click to Talk	Set click to Talk (needs support from server).
SIP Global Settings	
Strict Branch	Enable Strict Branch - The value of the branch must be after “z9hG4bK” in the VIA field of the INVITE message received, or the phone will not respond to the INVITE. Note: This will affect all lines
Enable Group	Enable SIP Group Backup. This will affect all lines

Registration Failure Retry Time	Registration failure retry time – If registration fails, the phone will attempt to register again after registration failure retry time. This will affect all lines

5.3.3.2 STUN Config

STUN support is configured in this page.
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 phone 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.





Field Name	Explanation
STUN NAT Transversal	Shows whether or not STUN NAT Transversal was successful.
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	
SIP Line Using STUN	Select the Line for use with STUN (SIP 1 - SIP 2)
Use STUN	Enable/Disable STUN on the selected line.

5.3.3.3 DIAL PEER

This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used.
Example 1: Substitution – Assume that it is desired to place a direct IP call to IP address 192.168.119. Using this feature, 156 can be substituted for 192.168.1.119.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

Example 2: Substitution – To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.

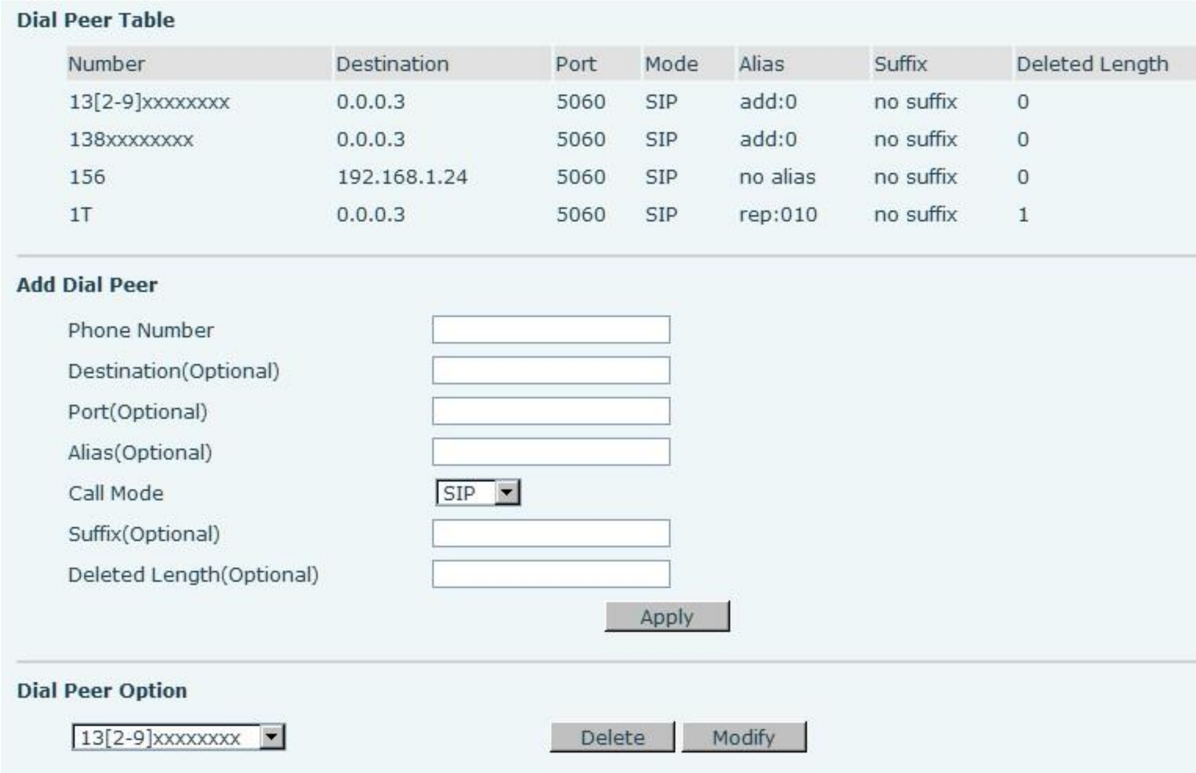
Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
13[2-9]xxxxxxxx	0.0.0.3	5060	SIP	add:0	no suffix	0
138xxxxxxxx	0.0.0.3	5060	SIP	add:0	no suffix	0

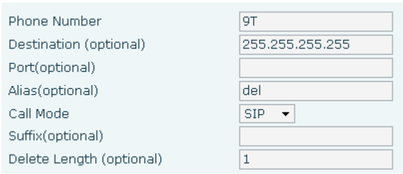
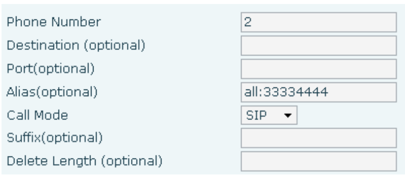
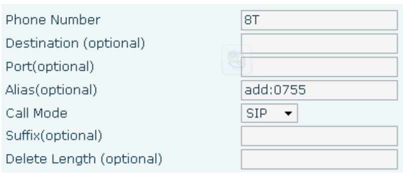
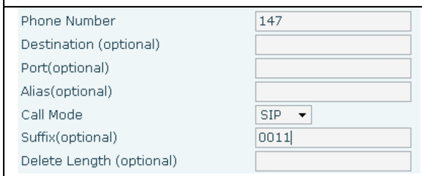
Example 3: Addition – Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.
x – Matches any single digit that is dialed.
[] – Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.



Field Name	Explanation
Phone number	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
Destination	Set Destination address. This is optional. For a peer to peer call, enter the destination IP address or domain name. To use a dial rule on the SIP2 line, enter 0.0.0.2. For SIP3 enter 0.0.0.3
Port	Set the Signaling port, the default is 5060.
Alias	Set the Alias. This is the text to be added, replaced, or deleted. It is optional.
Note: There are four types of aliases. 1) Add: xxx – xxx will be dialed before any phone number. 2) All: xxx – xxx will replace the phone number. 3) Del: The characters will be deleted from the phone number. 4) Rep: xxx – xxx will be substituted for the specified characters.	
Call Mode	Select either SIP or IAX2 protocol.
Suffix	Characters to be added at the end of the phone number. This is optional.

Delete Length	Sets 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. This is optional.

Dial Peer Examples

Web Interface	Explanation	Example
	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. Any phone number that begins with XXX will be sent via SIP2 after the first several digits are deleted depending on the delete length.	Dial “93333” The SIP2 server will receive “3333”
	This creates a speed dial function. Dialing “2”, will cause the entire alias number to be sent out.	Dial “2” The SIP1 server will receive 33334444
	The phone will add the alias to the end of the dialed number if the dialed number matches the template in the Phone Number box.	Dial “8309“ The SIP1 server will receive “07558309”
	Set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx If the dialed phone number starts with the digits in the Phone Number box, the matching digits will be replaced by the alias number.	Dial “0106228” The SIP1 server will receive “86106228”
	If the dialed phone number starts with the digits in the Phone Number box, the phone will send out the dialed phone number and add the suffix number.	Dial “147” The SIP1 server will receive “1470011”

5.3.4 Phone

5.3.4.1 AUDIO

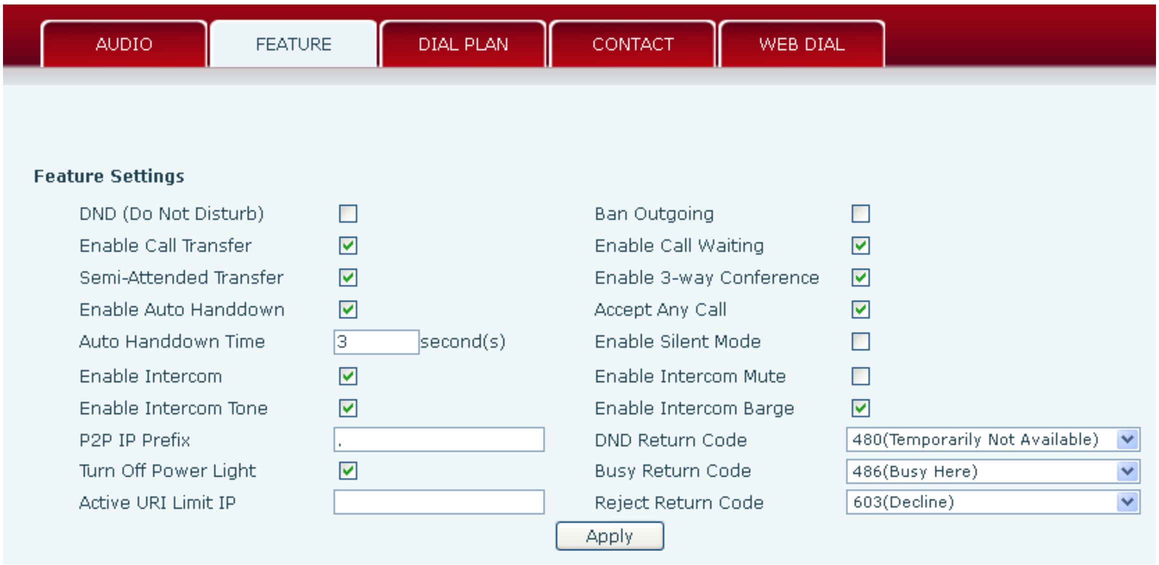
This page configures audio parameters such as voice codec, handset volume, and ringer volume.

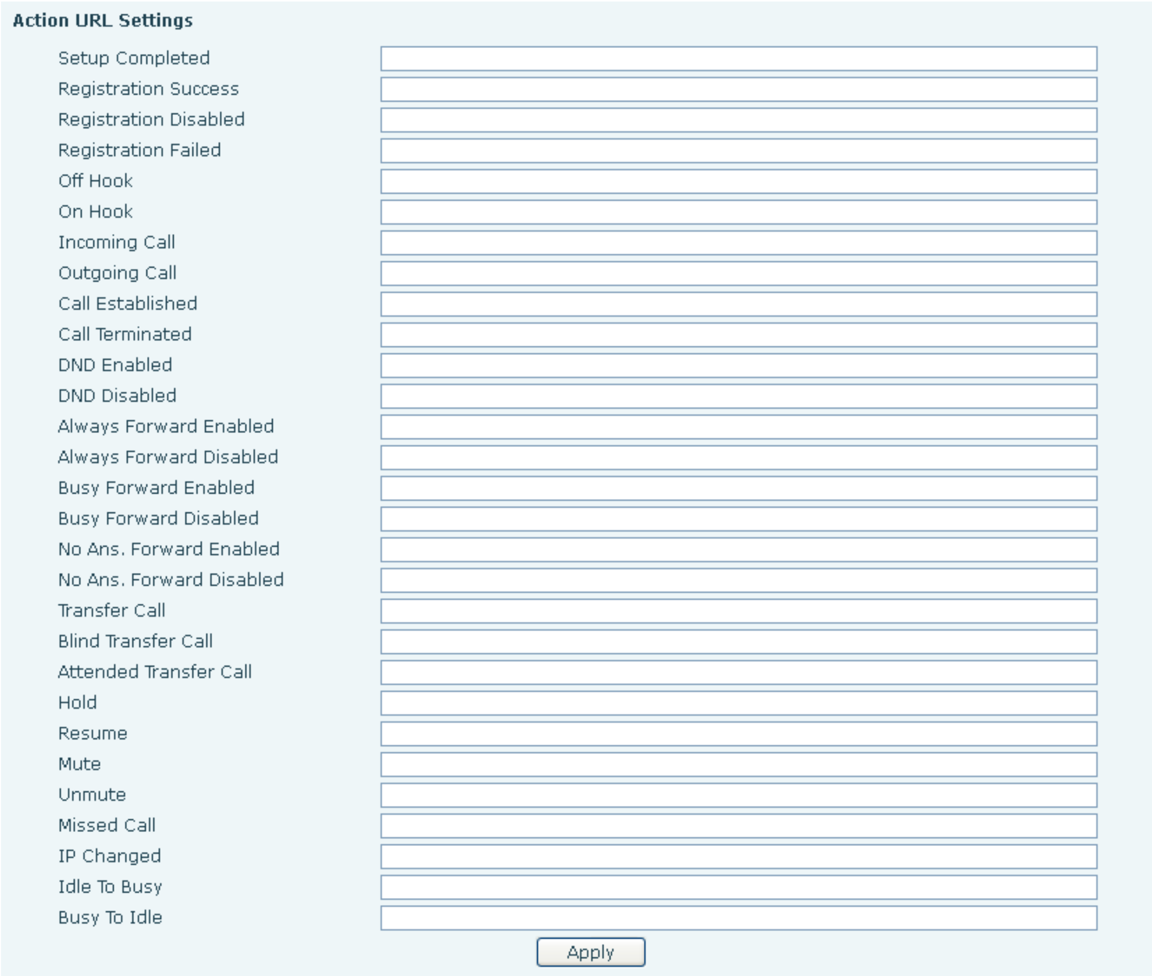
Field Name	Explanation
First Codec	The first codec choice: G.711A/u, G.722, G.723, G.729, G.726
Second Codec	The second codec choice: G.711A/u, G.722, G.723, G.729, G.726, None
Third Codec	The third codec choice: G.711A/u, G.722, G.723, G.729, G.726, None
Fourth Codec	The forth codec choice: G.711A/u, G.722, G.723, G.729, G.726, None
Fifth Codec	The fifth codec choice G.711A/u, G.722, G.723, G.729, G.726, None
Sixth codec	The sixth codec choice G.711A/u, G.722, G.723, G.729, G.726, None
Onhook Time	Time the handset must be on hook to disconnect a call. Default is 200ms.
Default Ring Type	Ring Sound – There are 9 standard types and 3 User types
Handset Input Volume	Handset Microphone volume – 9 levels
Handset Output Volume	Handset receiver volume - 9 levels
Speakerphone Volume	Speaker volume in hands free mode - 9 levels
Ring Volume	Ringer Volume - 9 levels
G729 Payload Length	G729 Payload Length – Adjusts from 10 – 60 mSec
Tone Standard	Select tone plan for the country of operation
G722 Timestamps	Choices are 160/20ms or 320/20ms

G723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload length cannot be set greater than 20 mSec.
DTMF Payload Type	The RTP Payload type that indicates DTMF. Default is 101

5.3.4.2 FEATURE

This page configures various features such as Hotline, Call Transfer, Call Waiting, etc.





Field Name	Explanation
Do Not Disturb	If enabled, the phone will reject incoming calls. The callers receive busy tone. Outgoing calls may be made.
Enable Call Transfer	If enabled, Call Transfer is allowed.
Semi-Attended Transfer	If enabled, Semi-Attended Transfer is allowed.
Enable Auto Handdown	If enabled in speakerphone mode, the phone will automatically hang up and return to idle when the distant party terminates the call. In handset mode, it will play dial tone instead of returning to idle.
Auto Handdown Time	Wait time before the phone performs the Auto Handdown behavior described above.
Enable Auto Redial	If enabled, the phone will automatically redial a call if a busy tone is received.
Auto Redial Interval	Wait time between auto redial attempts in seconds.
Auto Redial Times	Maximum number of auto redial attempts.

Enable Intercom	If enabled, allows intercom calls.
Enable Intercom Tone	If enabled, plays intercom ring tone to alert to an intercom call.
P2P IP Prefix	Set Prefix for peer to peer IP call. For example: You wish to dial 192.168.1.119. If the P2P IP Prefix is defined as 192.168.1., it is only necessary to dial #119. The default is ".". If this box is left blank, IP dialing is disabled.
Turn Off Power Light	Disables Power Light if selected.
Emergency Call Number	The phone will dial the emergency call number even if the keyboard is locked.
Enable Password Dial	When a number is entered beginning with the password prefix, the following N numbers after the password prefix will be displayed as *. N is the value entered in the Password Length field. For example: If the password prefix is 3 and the Password Length is 2, then dialing the number 34567 will display 3**67 on the phone.
Password Dial Prefix	Prefix for password dialing as described above.
Password Dial Length	Length for password dialing as described above.
Ban Outgoing	If enabled, no outgoing calls can be made.
Enable Call Waiting	If enabled, notifies user of a second call during a call. Caller ID of the new caller will be displayed. Press HOLD button to place existing call on hold and answer new call. Press HOLD again to return to first call.
Enable 3-way Conference	If enabled, allows 3-way conference.
Accept Any Call	If enabled, the phone will accept a call even if the called number does not belong to the phone.
Enable Call Completion	This is similar to Auto Redial except that the phone detects the state of the called number before making a new call attempt.
Enable Pre-Dial	If this feature is enabled, digits dialed on-hook will be transmitted when the phone goes off-hook.
Enable Silent Mode	If enabled, the phone will not ring to indicate a new call. Instead, the light below the key pad will blink to indicate a new call.
Hide DTMF	This feature sets how DTMF digits are displayed after a call is in progress. For example, dialing a PIN code to access banking information. There are 4 choices. 4. Disabled – All the digits will be shown on the LCD. 5. All – None of the digits will be shown on the LCD. The "*" will be shown. 6. Delay – The last digit entered will be shown for a short time and then replaced by "*." 7. Last Show – The last digit entered will be shown. Previous digits are replaced by "*."
Enable Intercom Mute	If enabled, mutes incoming calls during an intercom call
Enable Intercom Barge	If enabled, the phone will auto-answer an intercom call during an

	outside call. If an intercom call is established, a second intercom call will be rejected.
DND Return Code	Specify SIP Code returned for DND. Default is 480 - Temporarily Not Available.
Busy Return Code	Specify SIP Code returned for Busy. Default is 486 – Busy Here.
Reject Return Code	Specify SIP Code returned for Rejected call. Default is 603 – Decline.
Active URI Limit IP	IP address of the server for the Action URL messages described below.
Push XML Server	IP address for XML server which can send display content to the phone.
Enable Call Waiting Tone	Enables audible notification of call waiting.
Action URL Settings	URL for various actions performed by the phone. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml
Block Out Settings	Add or Delete Blocked numbers – Enter the prefix of numbers which should not be dialed by the phone. For example, if 001 is entered, the phone will not dial any numbers beginning with 001. X and x are wildcards which match single digits. For example, if 4xxx or 4XXX is entered, the phone will not dial any 4 digit numbers beginning with 4. It will dial numbers beginning with 4 which are longer or shorter than 4 digits.

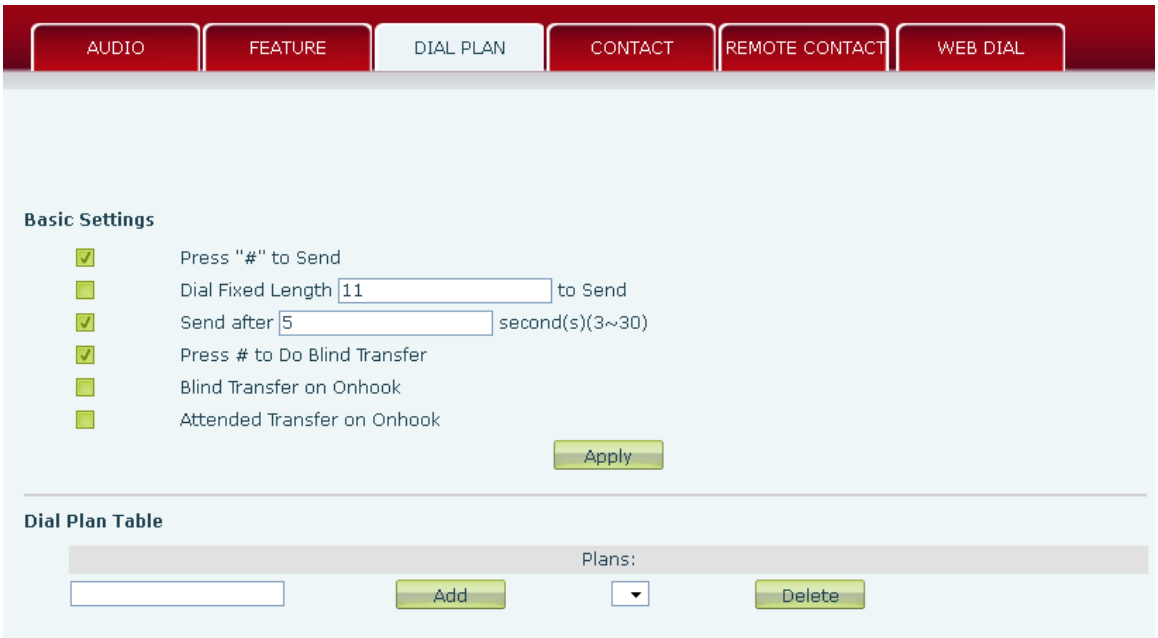
5.3.4.3 DIAL PLAN

This phone supports 7 dialing modes:

1. End with "#"– Dial the desired number, and press # to send it to the server.
2. Fixed Length – The number will be sent to the server after the specified number of digits are dialed.
3. Time Out – Number will be sent to the server after the specified time.
4. User Defined – Customized rules created by the user.

There is a special feature in the dial plan for the case where the user must dial an access code to get an external line. A digit followed by a "," will cause secondary dial tone to be generated. For example, assume a rule "9,xxxxxxx" is added. When the user dials 9, the phone will generate secondary dial tone. Then, when 8 digits have been dialed, they will all be sent to the server.

5. Press # to Do Blind Transfer - Press # after entering the target number for the transfer. The phone will transfer the current call to the third party.
6. Blind Transfer on Onhook - Hang up after entering the target number for the transfer. The phone will transfer the current call to the third party.
7. Attended Transfer on Onhook - Hang up after the third party answers. The phone will transfer the current call to the third party.



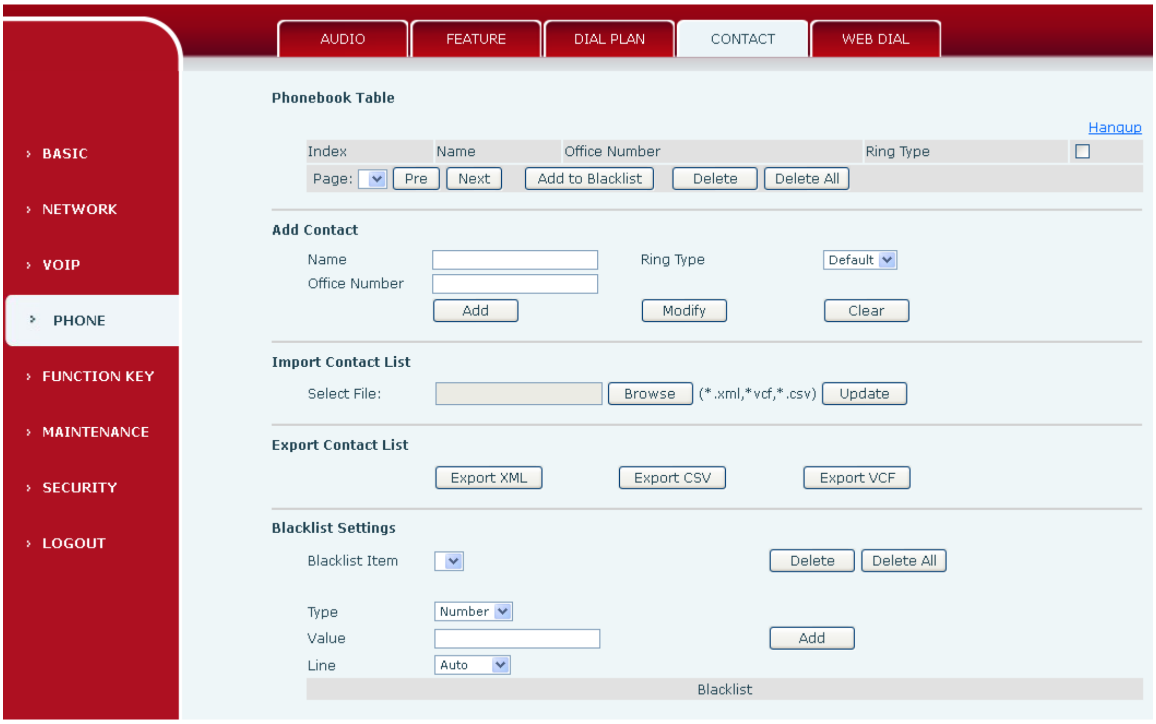
Dial Plan Special Characters	
[]	Specifies a range of digits to match. May be a range, a list of ranges separated by commas, or a list of digits.
*	Match any single digit that is dialed.
.	Match any arbitrary number of digits including none.
Tn	A time out period before digits are sent of n seconds in 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.

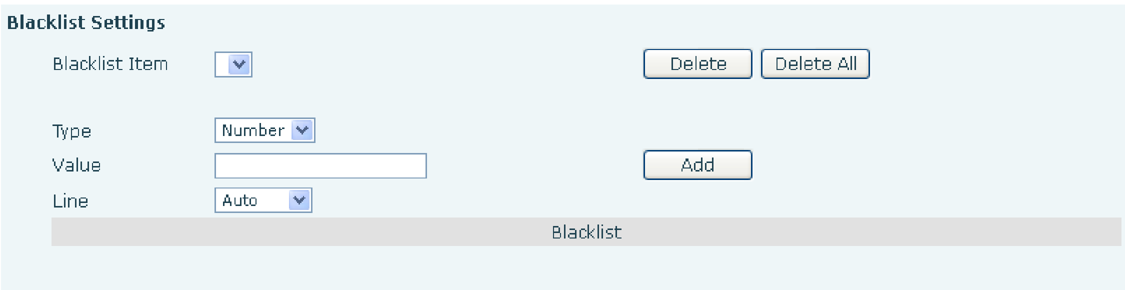
RULE
"[1-8]xxx"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

Cause extensions 1000-8999 to be dialed immediately
Cause 8 digit numbers beginning with 9 to be dialed immediately
Cause 911 to be dialed immediately
Cause 99 to be dialed after 4 seconds.
Cause any number beginning with 9911 to be dialed 4 seconds after dialing ceases.
Note: End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously.

5.3.4.4 CONTACT

Enter the name, phone number and ring type for each contact here.

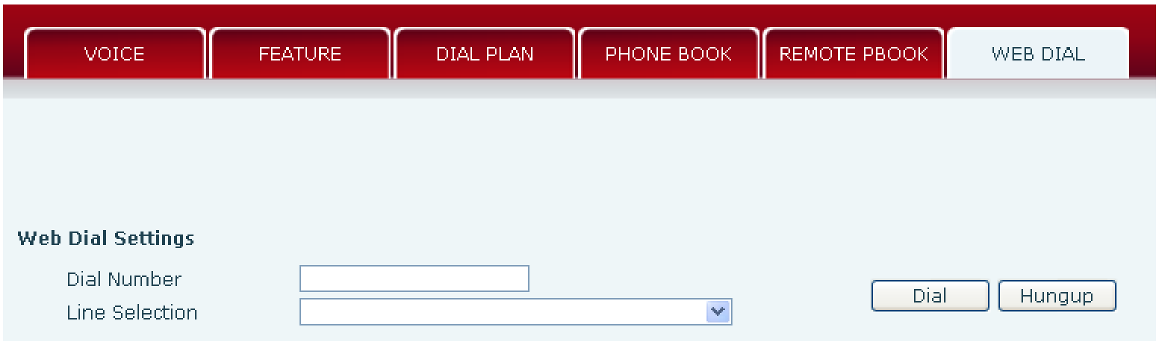




Field Name	Explanation
Phonebook Tables	
Name	Contact name
Office Number	Contact phone numbers
Ring Type	Ring type for this contact
Add Contact	
Name	Contact name
Office Number	Contact phone numbers
Ring Type	Ring type for this contact
Import Contact List	
Select File	Click the browse button to select the phonebook file to import. Then click the update button and the selected file will be added to the phone. File must be xml, vcf or csv format.

Export Contact File	
Export XML	Export contacts to xml file.
Export CSV	Export contacts to csv file.
Export VCF	Export contacts to vcf file.
Blacklist Settings	
Type	Select the blacklist type - number or prefix
Value	Input number or prefix
Line	Select the sip line
Note: The maximum capability of the phonebook is 500 contacts. Note: “x” and “.” are special characters in the black list. “x” matches any single digit and “.” matches any number of digits. For example, “4xxx” matches any 4 digit number beginning with 4. “6.” Matches any digit string beginning with 6. Note: There is also an allowed number list feature if the user only wants to allow a limited access to the phone. To use this, precede the number with “-“. For example, -123456, or -1234xx. Allowed number lists must end with an entry which is only a “.”	
 This will forbid incoming calls from any number except 4119.	

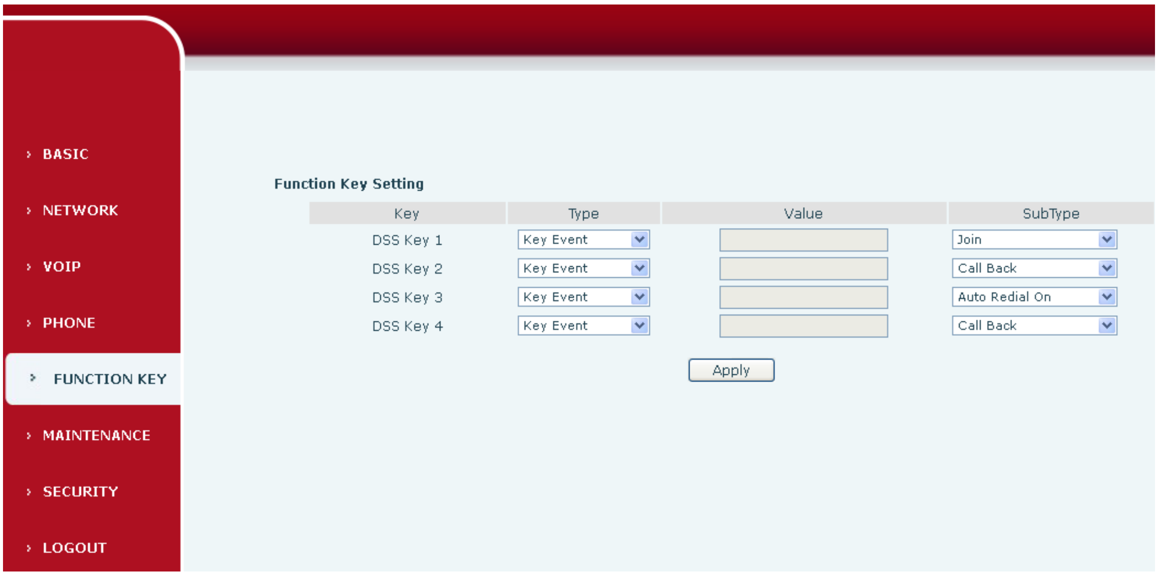
5.3.4.5 WEB DIAL



This feature allows a call to be initiated by a computer. To place a call, enter the number in the Dial Number box, select the line in the Line Selection box and press the Dial button. To end the call, press the Hangup button.

5.3.5 Function Key

The phone has 4 programmable DSS/Function keys which can be made to perform various functions. The functions are described below.



Memory Key – Select Type as Memory Key and enter the number to be dialed in the Value box. When the key is pressed, the phone will dial the programmed number.

Key Event – Select Type as Key Event and then select the SubType from the following options:

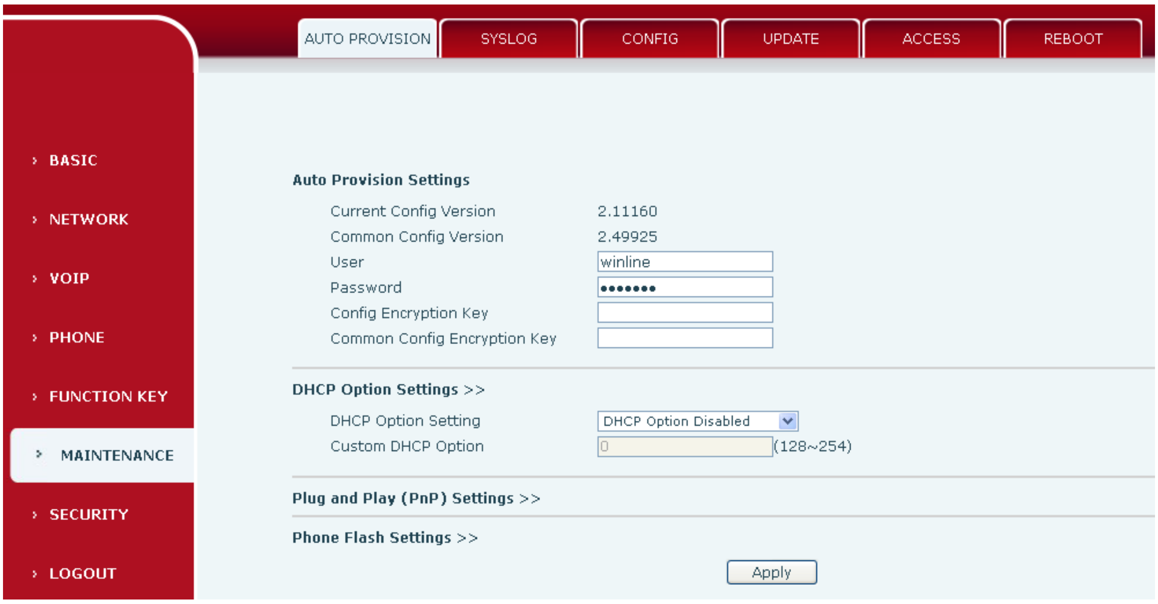
None	Message Wait Indication (MWI)	Do Not Disturb (DND)
Hold	Transfer	Phone Book
Redial	Auto redial on	Auto redial off
Call Forward	History	Flash
Headset	Call Back	

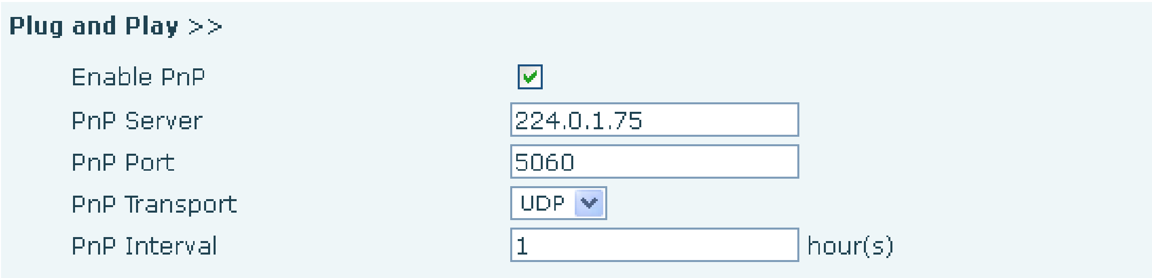
5.3.6 Maintenance

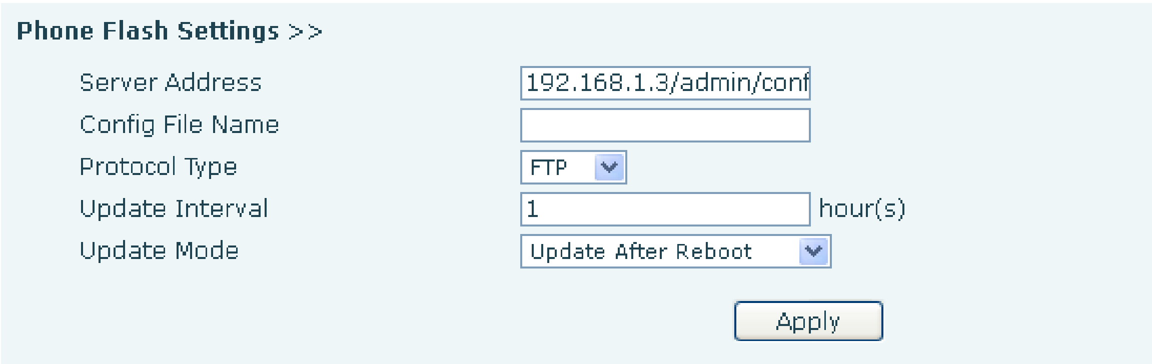
5.3.6.1 Auto Provision

The phone supports PnP, DHCP, and Phone Flash to obtain configuration parameters. They will be queried in the following order when the phone boots.

DHCP → PnP server → Phone Flash







Auto Provision Setting	
Field Name	**Explanation**
Current Config 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.
Common Config Version	Show the common config file's version. If the configuration downloaded and this configuration are 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.
User	Username for configuration server. Used for FTP/HTTP/HTTPS.

	If this is blank the phone will use anonymous.
Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Config Encryption Key	Encryption key for the configuration file
Common Config Encryption Key	Encryption key for common configuration file
DHCP Option Settings	
Field Name	**Explanation**
DHCP Option Setting	The phone supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.
Custom DHCP Option	Custom option number. Must be from 128 to 254.
Plug and Play Settings	
Enable PnP	If this is enabled, the phone 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.
PnP Server	PnP Server Address
PnP Port	PnP Server Port
PnP Transport	PnP Transfer protocol – UDP or TCP
PnP Interval	Interval time for querying PnP server. Default is 1 hour.
Phone Flash Settings	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or Domain name with subdirectory.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Config File Name	Specify configuration file name. The phone will use its MAC ID as the config file name if this is blank.
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

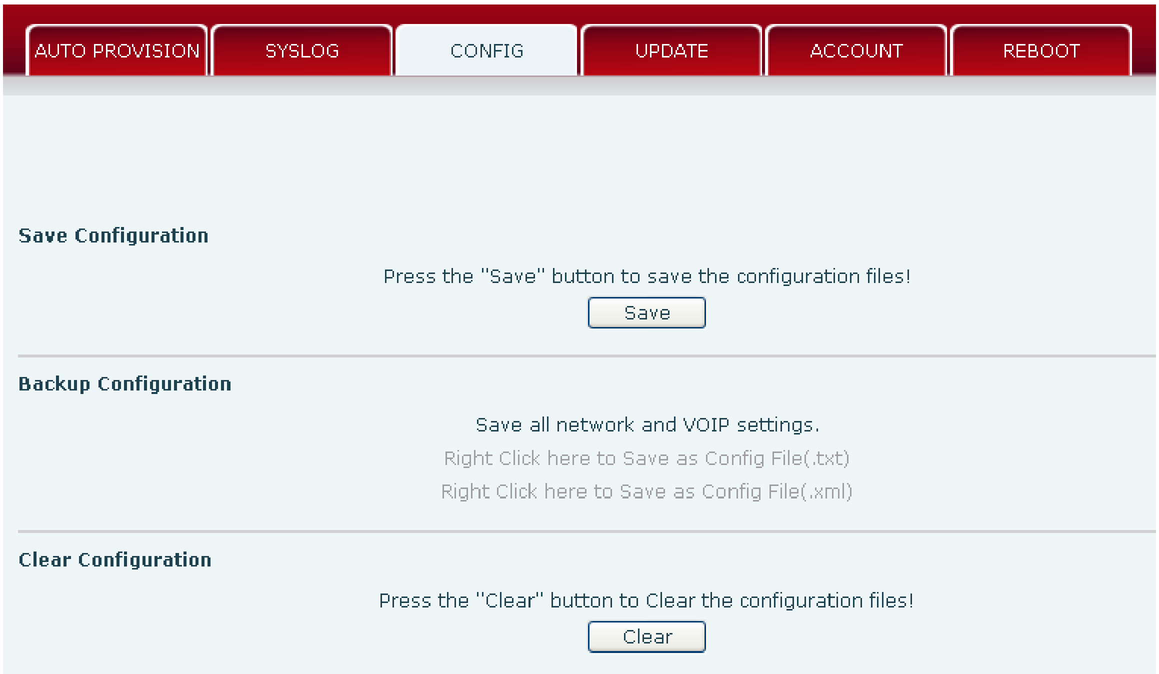
5.3.6.2 Syslog

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	Name	Description
0	Emergency	System is unusable. This is the highest debug info level.
1	Alert	Action must be taken immediately.
2	Critical	Critical conditions. System is probably working incorrectly.
3	Error	Error conditions. System may not work correctly.
4	Warning	Warning conditions. System may work correctly but needs attention.
5	Notice	Normal but significant condition.
6	Informational	Normal daily messages.
7	Debug	Debug messages normally used by system designer. This level can only be displayed via telnet.

Syslog Configuration	
Field Name	**Explanation**
Syslog Settings	
Server IP	Syslog server IP address.
Server Port	Syslog server port.
MGR Log Level	Set the level of MGR log.
SIP Log Level	Set the level of SIP log.
Enable Syslog	Enable or disable syslog.
Web Capture	
Start	Capture a packet stream from the phone. This is normally used to troubleshoot problems.
Stop	Stop capturing the packet stream

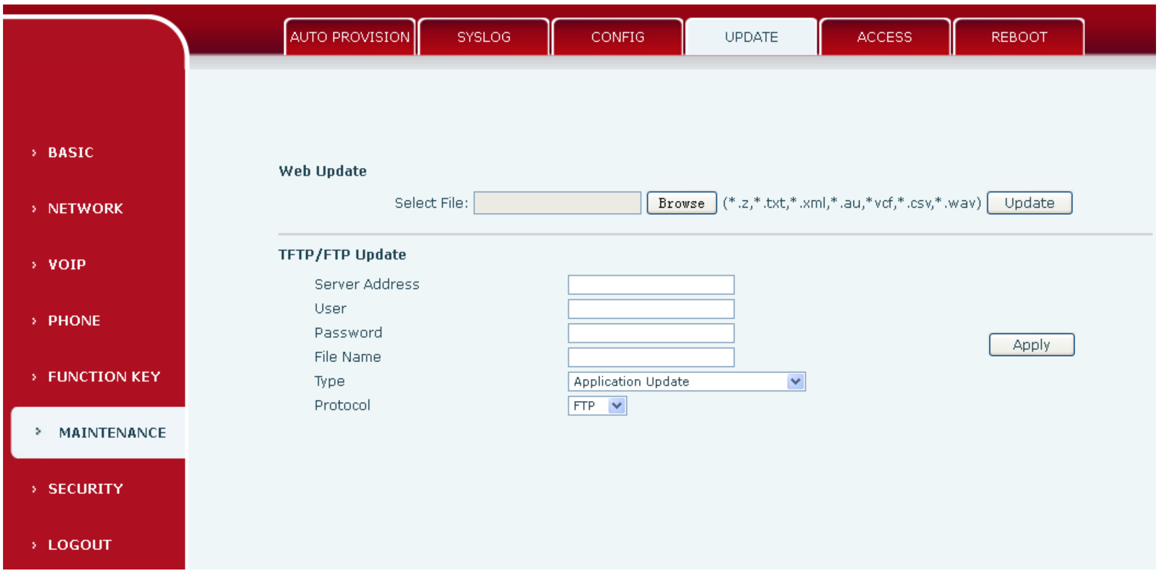
5.3.6.3 Config Setting



Config Setting	
Field Name	**Explanation**
Save Configuration	Save the current phone configuration. Clicking this saves all configuration changes and makes them effective immediately.
Backup Configuration	Save the phone configuration to a txt or xml file. Please note to Right click on the choice and then choose “Save Link As.”
Clear Configuration	Logged in as Admin, this will restore factory default and remove all configuration information. Logged in as Guest, this will reset all configuration information except for VoIP accounts (SIP1-2) and version number.

5.3.6.4 Update

This page allows uploading configuration files to the phone.

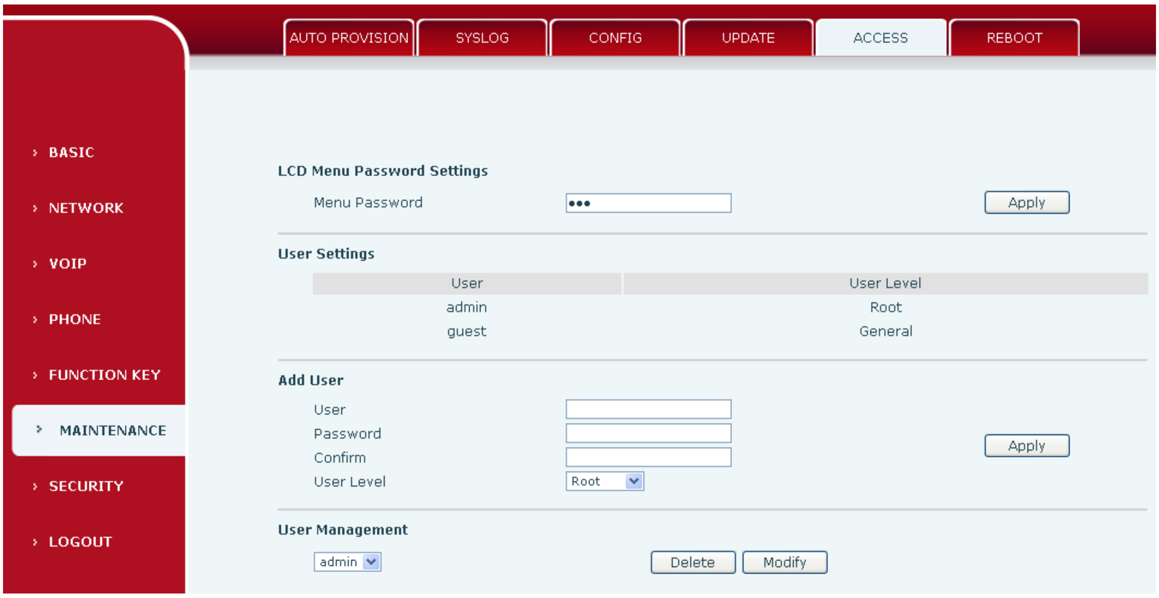


Update	
Field Name	**Explanation**
Web Update	
Web Update	Browse to the config file, and press Update to load it to the phone. Various types of files can be loaded here including firmware, ring tones, local phonebook and config files in either text or xml format.
TFTP/FTP Update	
Server Address	FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
User	FTP server Username for download/upload.
Password	FTP server password for download/upload.
File name	Name of update file or config file. The default name is the MAC of the phone.
Note: The exported config file can be modified. The config file is made up of modules. Modules which do not need changes may be deleted. For example, a config file can be downloaded and all modules removed except the SIP module. After rebooting, only the SIP settings will be changed.	

Type	Action to be executed by the phone. 1. Application update - download system update file 1. Config file export - Upload config file to FTP/TFTP server. It can then be named and saved. 2. Config file import - Download the config file from FTP/TFTP server. The configuration will be effective after the phone is reset. 3. Phone book export (.vcf, .csv, .xml) - Upload the phonebook file to FTP/TFTP server. It can then be named and saved. 4. PhoneBook import (.vcf, .csv, .xml) - Download phonebook file from FTP/TFTP server.
Protocol	Select FTP/TFTP server.

5.3.6.5 Access

User accounts can be added or deleted from this page. The authority of accounts can also be changed.



Access Configuration	
Field Name	**Explanation**
LCD Menu Password Settings	
Menu Password	Sets the password for entering the setup menu from the phone keypad. The password must be only digits.
User Settings	
This table shows the current user accounts	
Add User	
User	Set User Account name
User Level	There are two levels. Root user can modify the configuration. General user can only read the configuration.

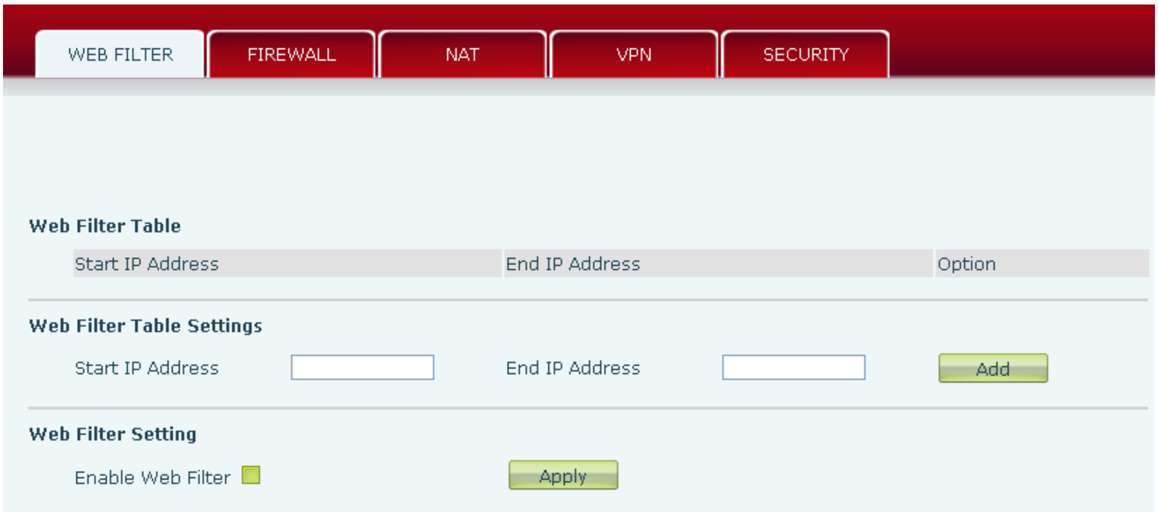
Password	Set the password
Confirm	Confirm the password
User Management	
Select the account and click Modify to modify the selected account. Click Delete to delete the selected account. A General user can only add another General user.	

5.3.6.6 Reboot

Some configuration modifications require a reboot to become effective. Clicking the Reboot button will cause the phone to reboot immediately.
Note: Be sure to save the configuration before rebooting.

5.3.7 Security

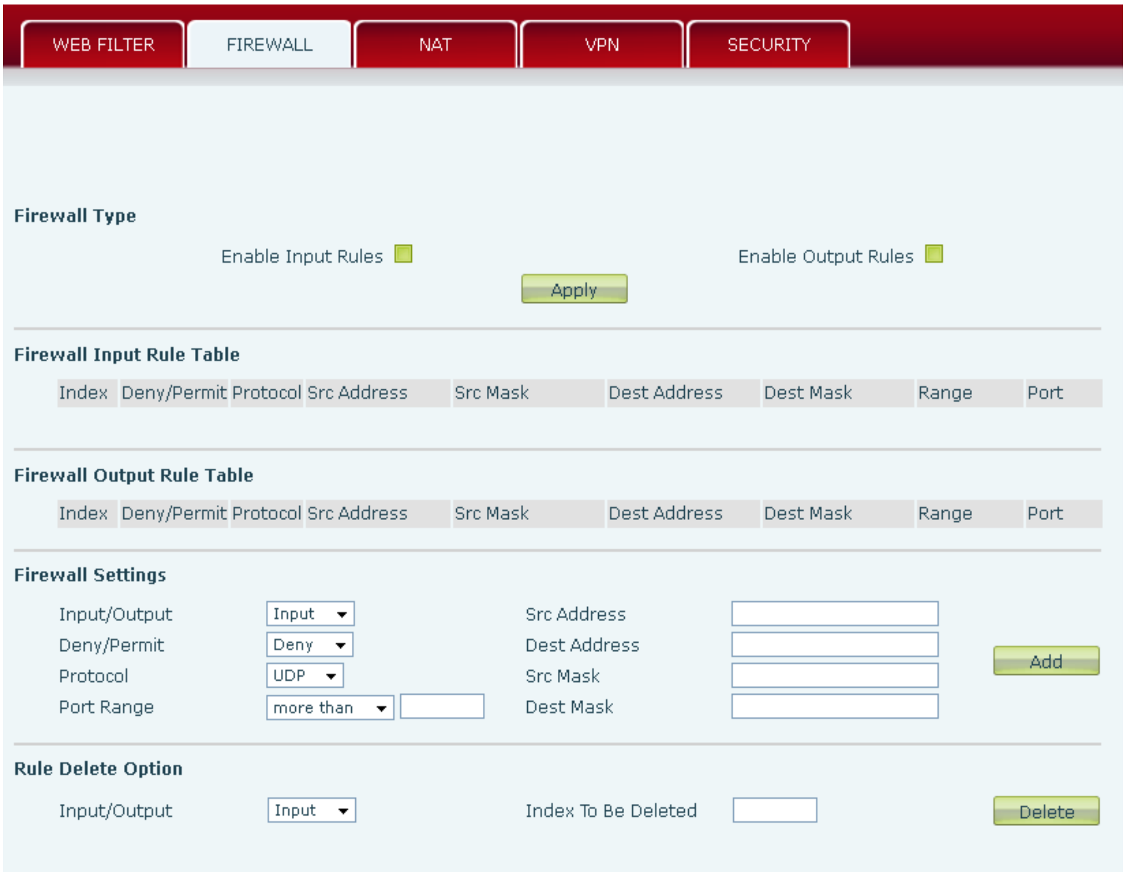
5.3.7.1 WEB FILTER



WEB Filter	
The Web filter is used to limit access to the phone. When the web filter is enabled, only the IP addresses between the start IP and end IP can access the phone.	
Field Name	**Explanation**
Start IP Address	Beginning IP Address for MMI Filter
End IP Address	Ending IP Address for MMI Filter

Add	Add this filter range to the Web Filter Table
Enable Web Filter	Select to enable MMI Filter.
Apply	Make filter settings effective.
Note: Once a range is added, it can be modified or deleted.	
Note: Be sure that the filter range includes the IP address of the configuration computer.	

5.3.7.2 Firewall



Firewall Configuration	
Firewall rules can be used to prevent unauthorized Internet users from accessing private networks connected to this phone (input rule), or prevent unauthorized devices connected to this phone from accessing the Internet (output rule). Each rule type supports a maximum of 10 items.	
Field Name	**Explanation**
Enable Input Rules	Enable rules limiting access from the Internet.
Enable Output Rules	Enable rules limiting access to the Internet.
Input/Output	Specify if the current rule is input or output.
Deny/Permit	Specify if the current rule is Deny or Permit.
Protocol	Filter protocol type (TCP/ UDP/ ICMP/ IP)
Port Range	Set the filter Port range
Src Address	Set source address. It can be a single IP address or use * as a wild card. For example: 192.168.1.14 or *.*.*.14.

Dest Address	Set destination address. It can be a single IP address or use * as a wild card. For example: 192.168.1.14 or *.*.*.14.
Src Mask	Set the source address mask. For example: 255.255.255.255 points to one host while 255.255.255.0 points to a C type network.
Dest Mask	Set the destination address mask. For example: 255.255.255.255 points to one host while 255.255.255.0 points to a C type network.

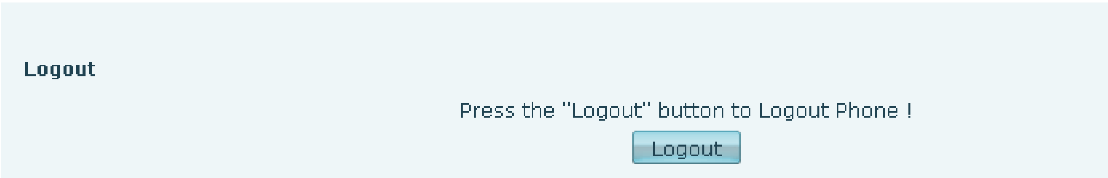
Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Dest Address	Dest Mask	Range	Port
1	Deny	UDP	192.168.1.14	255.255.255.0	192.168.1.118	255.255.255.0	More than	1

When a connected device tries to access 192.168.1.118, the phone will deny the request because of the out_access rule. Access to any other IP address will be allowed.

Click the **Delete** button to delete the selected rule.

5.3.8 Logout



Click **Logout** to exit the phone web page.

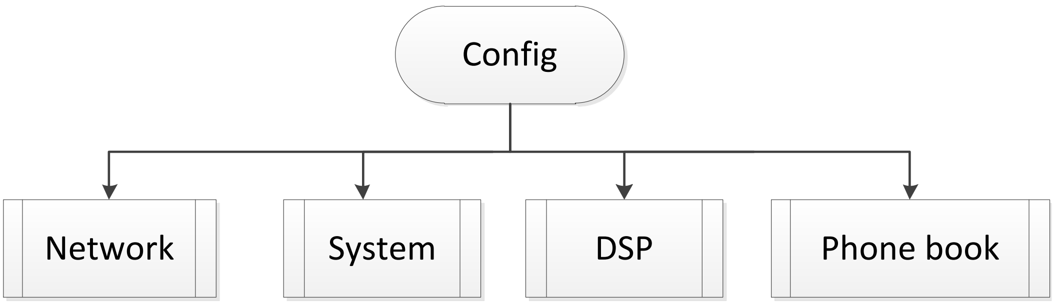
5.4 Settings via Phone's Keyboard

5.4.1 Procedure

- Use the Menu key to begin configuration from the keypad.
- Use the Up/Down key to browse menus and submenus.
- Use the ENTER key to enter submenus and confirm operations.
- Use the EXIT key to go back or to cancel operation.

5.4.2 Phone menu

Phone main menu:



6 Appendix

6.1 Specification

6.1.1 Hardware

Item		Specification
Power Adapter		Input: 100-240V
		Output: 5V 1A
Port	WAN	10/100Base- T RJ-45 1 PORT
	LAN	10/100Base- T RJ-45 1 PORT
Power Consumption		Idle: 1.5W
		Active: 1.8W
LCD Size		74x28mm
Operation Temperature		0～40℃
Relative Humidity		10～65%
CPU		Broadcom
SDRAM		8MB
Flash		2MB
Dimension(L x W x H)		20 X 18.5 X 19.3cm
Weight		0.99kg

6.1.2 Voice Features

- Supports 2 SIP servers
- Supports RFC3261
- Codecs
    - G.711A/u
    - G.723.1 high/low
    - G.729a/b
    - G.722
    - G.726
    - Codec Setting per SIP line
- Echo cancellation: G.168 Compliance in LEC, additional acoustic echo cancellation(AEC) can reach 96ms max filter length in hands-free mode
- Supports Voice Gain Setting, VAD, CNG
- Full duplex hands-free
- SIP support
    - SIP domain
    - SIP authentication
        - none
        - basic
        - MD5

- DNS
- Peer to Peer/ IP call
- Automatic line selection
- 9 Standard ring tones
- DTMF
  - SIP info
  - DTMF Relay (In-Band)
  - RFC2833
  - AUTO
- SIP applications
  - Call Forward
  - Call Transfer（Blind/Attended）
  - Hold
  - Call Waiting
  - 3 Way Conference
  - Redial
  - paging
  - Intercom
  - Auto Redial
- Call control features
  - Flexible dial plan
  - Hotline
  - Anonymous Call Reject
  - Black List (Reject Authenticated Call)
  - Approved Caller List
  - Limit Call
  - Do Not Disturb
  - Caller ID
  - CLIR (reject anonymous call)
  - CLIP(make anonymous call)
  - Dial without Registration
- Phonebook 500 records
  - Incoming Calls
  - Outgoing Calls
  - Missed Calls
  - Max of 300 Records Each
  - Supports vCard/XML/CSV
- 4 DSS keys
- Time Display
  - 12/24 Hour
  - Support Daylight Saving Time
- Supports Path, Group
- Supports SIP Privacy
- Supports MWI

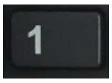
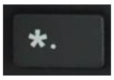
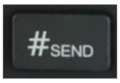
6.1.3 Network Features

- WAN/LAN
    - Bridge
    - Bridge with port mirror
- Supports PPPoE for xDSL
- Supports VLAN
    - 802.1Q
    - 802.1P
- Supports STUN
- Wan Port Supports Main DNS and Secondary DNS
- Supports DNS via DHCP or Static DNS
- Supports DHCP client on WAN
- QoS with DiffServ
- Network Tools in Telnet Server
    - Ping
    - Trace Route
    - Telnet Client

6.1.4 Maintenance and management

- Firmware Upgrade
    - POST
    - HTTP
    - FTP
    - TFTP
- Configuration
    - Web
    - Telnet
- Two Account Levels
- Supports Syslog
- Supports Auto Provisioning
    - Firmware Upgrade
    - Auto-Provisioning

6.2 Digit-character map table

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