VoIP Phone
User Manual

For model:
9602IPMWD
9602IPMWD5
9600IPMWD
9600IPMWD5

Version: V2.0

The phone started, get Provisioning Server address by DHCP Server, then the phone LED lights flashing, and issued a "beep beep" prompt tone, "Config ID". Through the digital keyboard input after the ID, and then enter the "#", then the opportunity to Provisioning Server automatically load the configuration file, automatically restart after the success; if this fails, then the opportunity to enter the default standby state, after 15s can log on; if you do not want to download directly into the standby state # by default. If you do not complete the download, or download the configuration file in the AutoUpdate Settings config ID no configuration parameters, will still be asked to enter "Config ID" after the restart.
After the phone has entered the default state, you can have the phone to broadcast the IP address by pressing “**47#”.

Function

1. Support DHCP automatic distribution of IP addresses and other parameters
2. Support PPPOE agreement (ADSL, cable modem access use)
3. The program can be upgraded via HTTP, HTTPS, FTP or TFTP
4. Dynamic voice detection; comfort noise generation; voice buffer technology
5. Hold function
6. Speed dial
7. DND (Do Not Disturb), blacklist, call restriction, hotline function
8. Voicemail
9. Using a standard web browser (such as IE) for setting
10. SSH remote management function
11. Classified management for common user password and remote user password
12. Support * * code function
13. Call Waiting Feature
14. Auto answer
15. Call park
16. Call transfer
17. Tripartite conference
18. 802.1x Authentication
19. LLDP Feature

Standard and Protocols

◆ IEEE802.3/802.3u10 Base T/100Base
◆ PPPoE: Point to point protocol over Ethernet
◆ DHCP Client and Server: Dynamic Host Configuration Protocol
◆Support G.711a/u, G.729, G.723.1, G.722, iLBC speech encoding algorithm
◆ SIP RFC3261, RFC2543
◆ TCP/IP: Internet transmission control protocol
◆ RTP: Real-time Transfer Protocol
◆ RTCP: Real-time Control Protocol

◆ VAD/CNG: can save bandwidth
◆ TFTP: Trivial File Transfer Protocol

1. INTRODUCTION

This is the 9602IP network telephone user manual. Before the use of 9602IP phone, you need to make some phone configuration for normal use. This book illustrates how to use keyboard and Web phone service configuration page.

1.1 HARDWARE OVERVIEW

The default WAN port is a DHCP client, the user connects it to the ADSL or WAN port switch, LAN port connects to the computer; you can use the administrator username "admin" and password "admin" to set the login password.
Only WAN port supports POE.

1.2 SOFTWARE OVERVIEW

Network Protocol	Tone
● SIP v2(RFC3261) ● IP/TCP/UDP/RTP/RTCP ● IP/ICMP/ARP/RARP/SNTP ● TFTP Client/DHCP Client/PPPOE Client ● Telnet/HTTP Server ● DNS Clients	● Ring Tone ● Ring Back Tone ● Dial Tone ● Busy Tone
	Phone Function
	● Volume Adjustment ● Speed dial key
Codec	IP Assignment
● G.711a ● G.711u: ● G.723.1: ● G.729 ● G.722 ● iLBC	● IP (Static IP) ● DHCP ● PPPoE
Voice Quality	Security
● VAD: Voice activity detection ● AGC: Automatic Gain Control ● AEC: Automatic Echo Cancellation ● SRTP: Secure Real-time Transport Protocol	● HTTP 1.1 basic/digest authentication for Web setup ● MD5 for SIP authentication (RFC2069/RFC2617)
	QoS
	● QoS field

<table>
<tr><td>Call Function</td><td>NAT Traversal</td></tr>
<tr><td rowspan="3">● Call Hold
● Call Waiting
● Call forward
● Caller ID</td><td>● STUN</td></tr>
<tr><td>Configuration</td></tr>
<tr><td>● Web Browser
● Keypad</td></tr>
<tr><td>DTMF</td><td>Firmware Upgrade</td></tr>
<tr><td>● IN Band
● RFC2833
● SIP Info</td><td>● TFTP
● HTTP
● FTP
● HTTPS</td></tr>
</table>

2 PHONE MENU SETTINGS

Using the web configuration page: familiar with the PC user can use the method to configure the phone. Sequentially press the "**47#" button, then the phone will voice broadcast address IP. Directly in the browser address bar entering the address of the IP phone can log in web page, enter the login name: admin, password: admin

2.1 KEY FEATURES

The user can use the table below to confirm the key and hardware function.

Key function on base unit:

Key	State	Function / Display
Volume +	Conversation	Increase the volume
Volume -	Conversation	Decrease the volume
Message	Dialing	Listen to the voice message
LOCATE	Dialing	Page the handset
Speaker	Conversation	Toggle between handset and speakerphone
Mute	Conversation	Mute
Redial	Dialing	The last number redial and call
Hold	Conversation	Hold or release hold or Park key
M1~M10	Dialing	Speed dial and call or secondary function
Line1	Stand-by	Line1 state (only for two-line model)
Line2	Stand-by	Line2 state (only for two-line model)
1	Dialing	“1”
2	Dialing	“2”
3	Dialing	“3”
4	Dialing	“4”
5	Dialing	“5”
6	Dialing	“6”
7	Dialing	“7”
8	Dialing	“8”
9	Dialing	“9”
0	Dialing	“0”
*	Dialing	“*”
#	Dialing	Can be used as the first number dialing out or equivalent dial end tag

Key function on handset:

Key	State	Function / Display

Volume+	Conversation	Increase the receiver volume
	Stand-by	Increase the ringer volume
Volume-	Conversation	Decrease the receiver volume
	Stand-by	Decrease the ringer volume
Line1	Stand-by	Line1 state (only for two-line model)
Line2	Stand-by	Line2 state (only for two-line model)
Mute	Conversation	Mute
Redial	Dialing	The last number redial and call
Hold/Conf	Conversation	Hold or release hold or Park key
	Two lines on hold	Achieve Conference function (only for two-line model)
FNC/Flash	Conversation	Achieve Transfer function
Speaker	Conversation	Only M series supports it
1	Dialing	"1", press and hold for 3s to pick up the voice message
2	Dialing	"2", press and hold for 3s to dial out the number in M6
3	Dialing	"3", press and hold for 3s to dial out the number in M7
4	Dialing	"4", press and hold for 3s to dial out the number in M8
5	Dialing	"5", press and hold for 3s to dial out the number in M9
6	Dialing	"6", press and hold for 3s to dial out the number in M10
7	Dialing	"7"
8	Dialing	"8"
9	Dialing	"9"
0	Dialing	"0"
*	Dialing	"*"
#	Dialing	Can be used as the first number dialing out or equivalent dial end tag

3 THE OPERATION METHOD OF TELEPHONE

3.1 HOW TO MAKE A PHONE CALL?

You could make a phone call after the phone configuration items are set up. Please check if the cable is properly connected before use.

3.1.1 Basic call

1. Making the call by handset

   After the handset is placed off-hook, dial and use “#” key as the end dialing symbol.

2. Making the call by speakerphone

   After the phone is placed off-hook, dial and use “#” key as the end dialing symbol.

3.1.2 Hold Key

1. You can keep and release the call of current line. The only one line is presently in a call, the other line must be placed on hold.

1) Place the call of one line on hold

Make sure that the call you want to keep is enabled, then press "Hold" key.

2) Release Hold

Make sure that the call is initiated, then press "Hold" key.

2. Call Park Function

Initiate the call park function, Hold key can be used as a Park key.

3.1.3 Volume Control

Press “VOL▲” to increase the volume, while press “VOL▼” to decrease the volume.

3.1.4 Mute

During the call, if you do not want to let them hear your own voice, you can press "Mute" key, so that the other party cannot hear your voice, and you can hear the sound of other end.

3.1.5 Memory Key

In addition to serving as a storage function, but also can be used as hold, DND, transfer and conference function. See the web call feature function set.

3.1.6 Tripartite Conference Function

If the phone is Line1 hold, line2 in the call, press the conference key, which can achieve three party conference.

During the three party conference, the base unit and handset cannot be switched each other.

3.1.7 Transfer

The telephone is in conversation with A, A wants to call B, you can press the Transfer key, and then call B, press the Transfer button again after B hooking off, the transfer function can be achieved.

3.1.8 Call Park

After the call park feature is enabled, and Hold Key Active and Idle Hold Key related parameters are configured, we can perform the function of park. This function is only applicable to the base unit.

3.1.9 Redial

After the base unit is stand by or the handset is off-hooked, press Redial key, the last dialed number will be dialed out to achieve the redial function.

3.1.10 Register the Handset

Place the handset into the cradle of base unit. The “Message” LED on base unit will blink. Initiate the handset registration. The “Message” LED on handset will also begin to blink. At that time, if the base unit and handset have found out each other, the

"Message" LED on base unit will stop blinking and the "Message" LED on handset will also stop blinking and emit the prompt sound of successful registration.
Note: each base unit can register up to 5 handsets.

3.1.11 Toggle between Base Unit and Handset

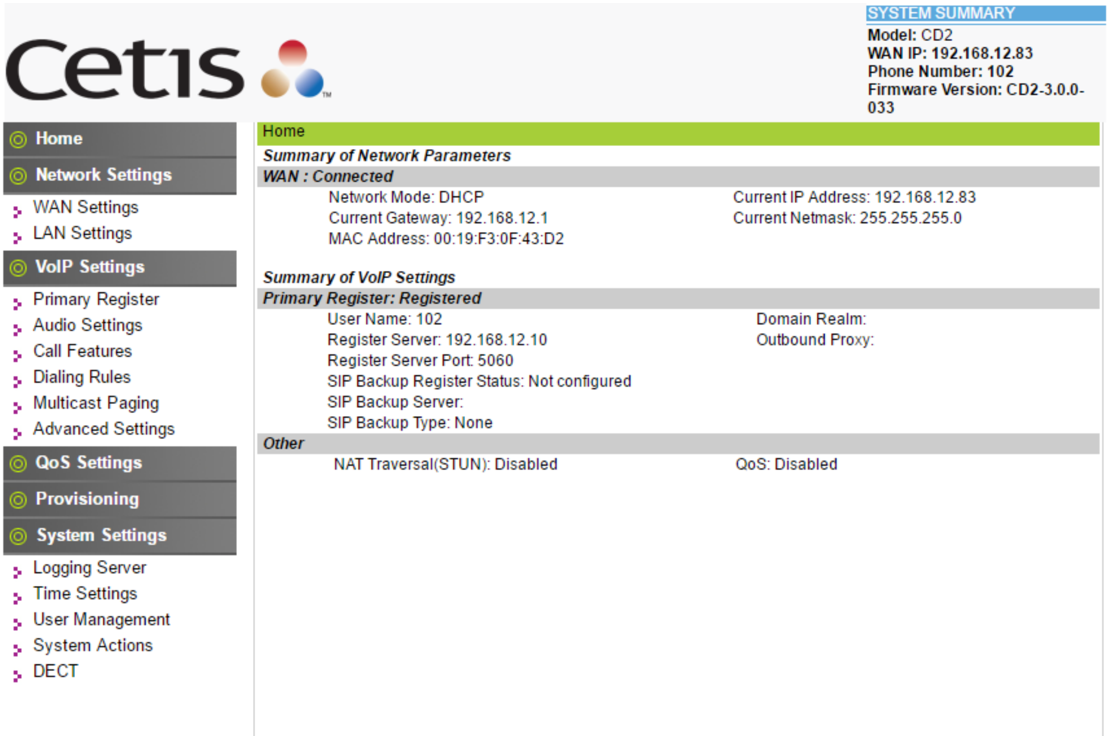
When the base unit is in conversation, press "Line1"/"Line2" key on handset, the call will switch over to the handset. If the handset is in conversation, press "SPEAKER" key, the call will switch over to the base unit.

4 WEB SETTING

The IP phone and the computer are connected to the same network (LAN), open the browser, enter the IP address of the phone, the page will request to input a username and password. Enter your username and password to login as administrator.

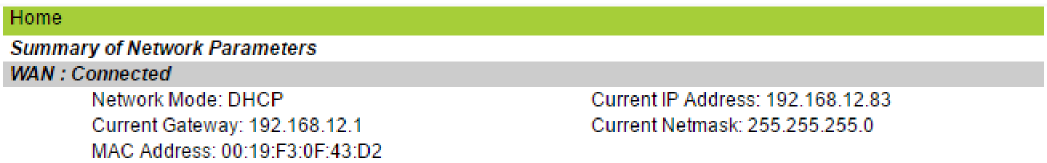
4.1 HOME PAGE

Enter the user name and password, the page is shown below:



4.2 Network Setting

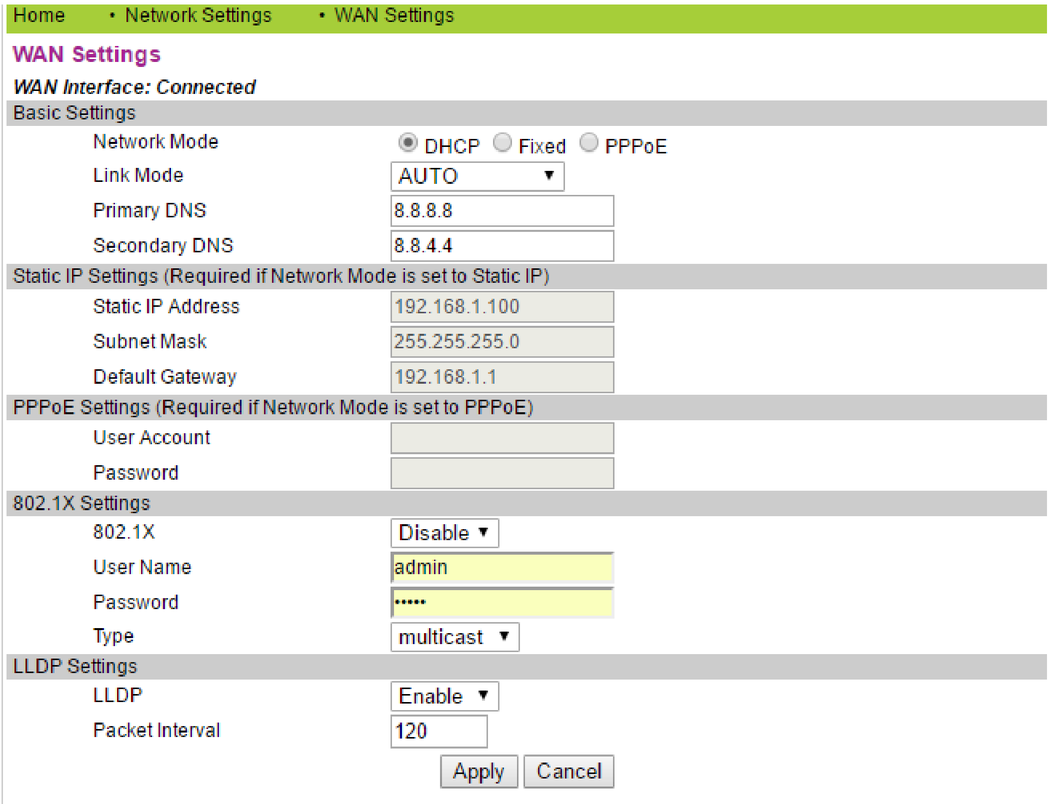
You can get the network information of phone in the page.



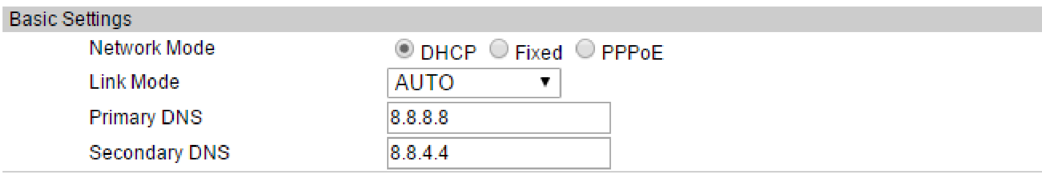
4.2.1 WAN Setting

WAN port setting page.

WAN port supports the static IP, dynamic allocation IP and PPPoE.



4.2.1.1 Basic Setting

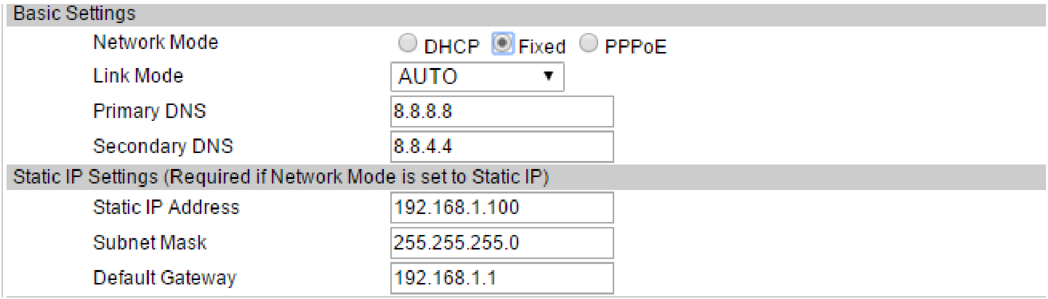


Basic Setting	
Network Mode	Select the network mode of WAN port; the default is DHCP
Link mode	Configure the WAN port network connection mode
Primary DNS	Set the main DNS address
Secondary DNS	Set the secondary DNS address

4.2.1.2 DHCP

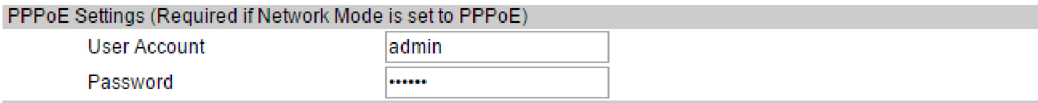
If your local network has a DHCP server, 3302IP phone can get WAN network information from the DHCP server.

4.2.1.3Static IP Setting



Static IP setting (WAN port network mode is set to Static IP)	
Static IP Address	Set static IP address
Subnet Mask	Set subnet mask with static IP
Default Gateway	Set the default gateway with static IP

4.2.1.4 PPPoE Setting



PPPoE Setting (Required if Network Mode is set to PPPoE)	
User Account	Set the PPPoE user account
Password	Set the PPPoE account password

4.2.1.5 802.1x settings

802.1x settings	
802.1x_Enable	Enable or disable 802.1x authentication
802.1x_UserName	802.1x username
802.1x_Password	802.1x authentication password
Type	Multicast/Broadcast

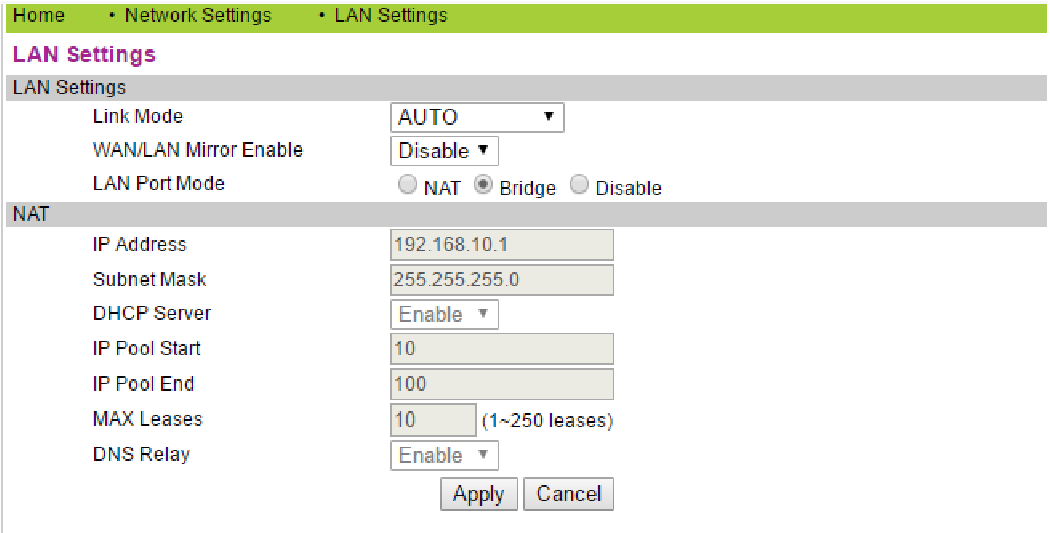
4.2.1.6 LLDP settings

LLDP settings	
LLDP Enable	Enable or disable LLDP function
Packet Interval	Packet interval

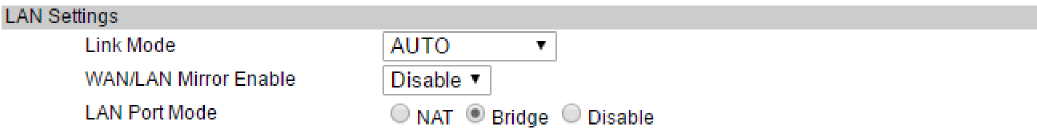
Note: if the user wants to access the phone through the WAN port, then he / she must use the new IP address to access the phone after changing IP address of WAN port.

4.2.2 LAN Settings

LAN port setting interface

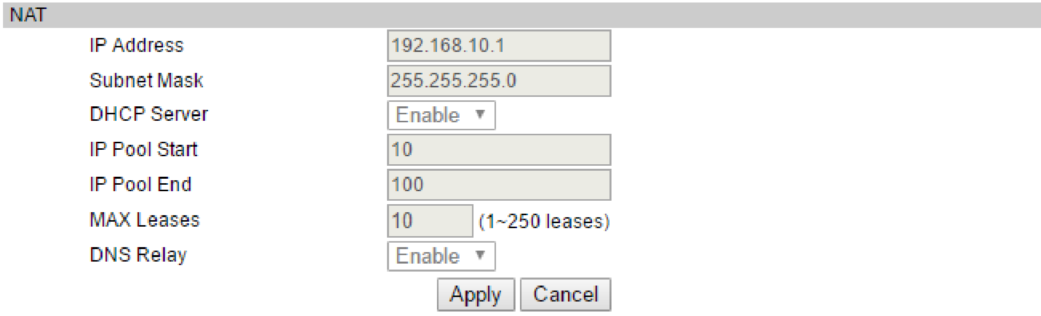


4.2.2.1 LAN Settings



LAN Settings	
Link Mode	Configure the LAN port network connection mode
WAN/LAN Mirror Enable	Whether WAN/LAN mirror mode is enabled
LAN Port Mode	The mode of LAN port is Nat/Bridge/Disable

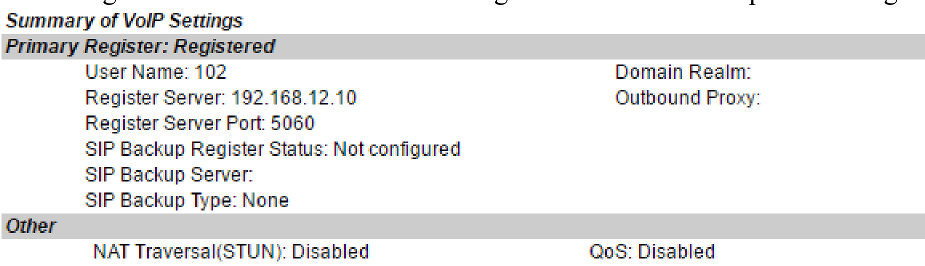
4.2.2.2 NAT



NAT	
IP Address	IP address of LAN port
Subnet Mask	Subnet Mask
DHCP Server	DHCP server is enabled or not
IP Pool Start	IP address assignment start address
IP Pool End	IP address assignment end address
MAX Leases	Maximum release time
DNS Relay	DNS relay is enabled or not

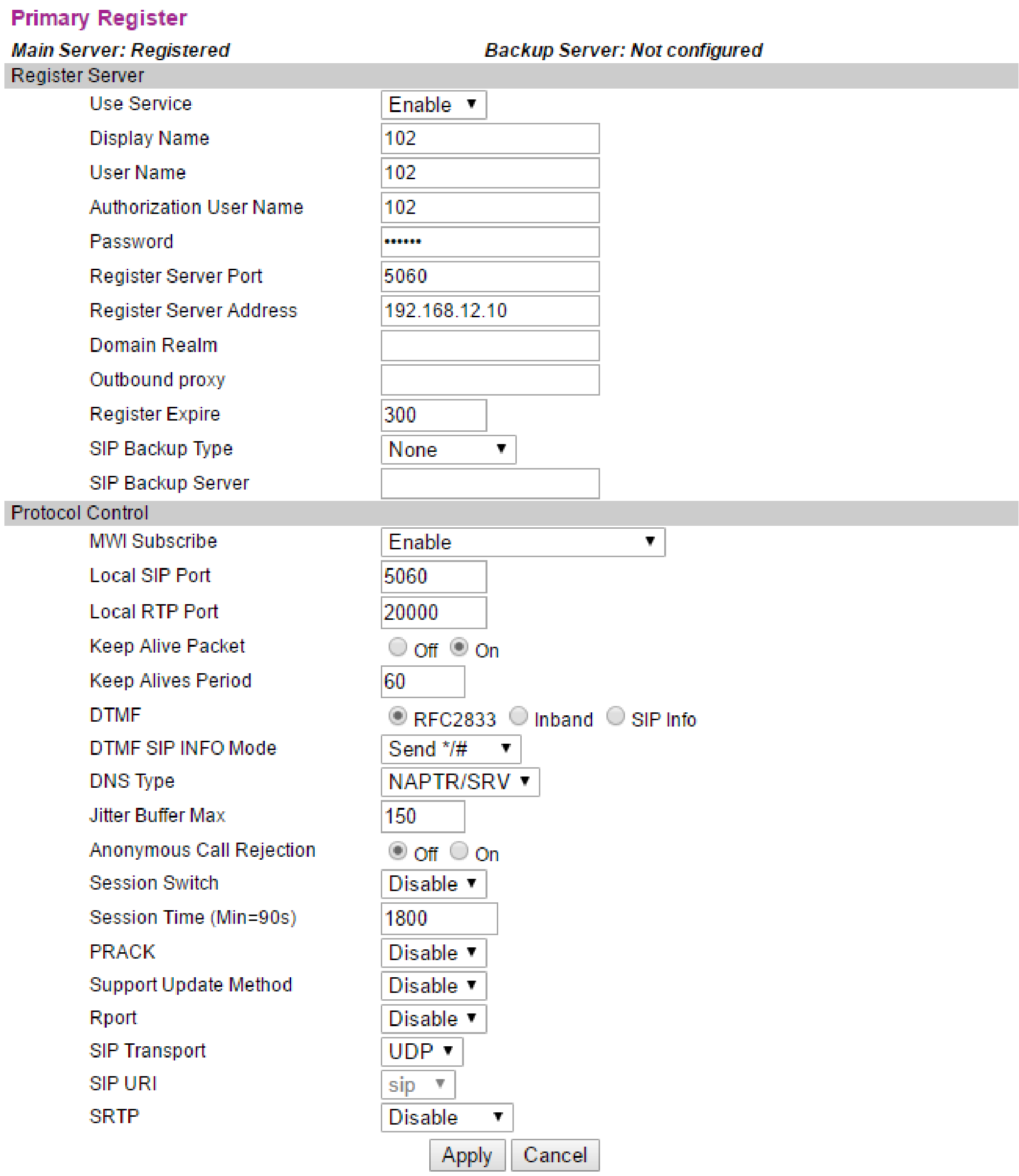
4.3 VOIP SETTING

You can get SIP account information and registration status of the phone through the page.



4.3.1 Primary Register

Configure the SIP registration information of phone in the below page.

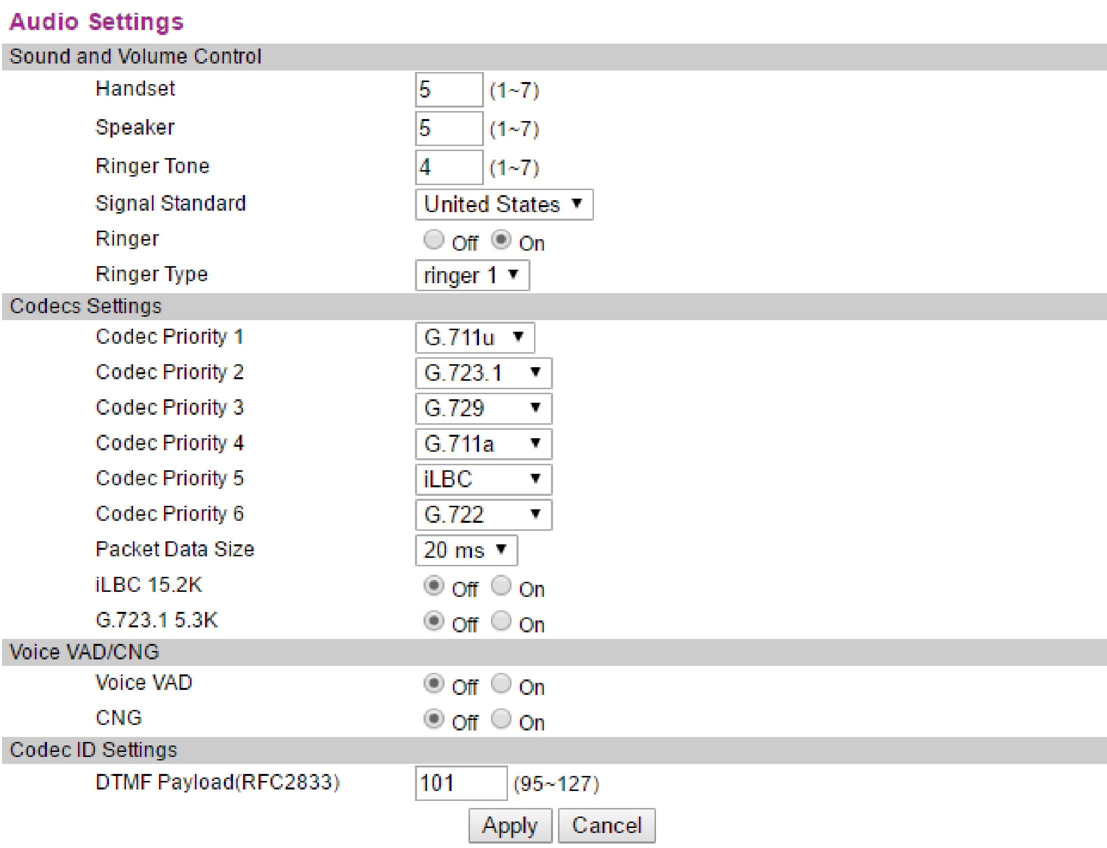


Register Server	
Use Service	Enable or disable SIP registration
Display Name	Set the displayed name of phone's SIP account
User Name	Set the username (SIP account)
Authorization User Name	Confirm the SIP account
Password	Set the password of SIP account
Register Server Port	Set the port No. of register server, the default is 5060
Register Server Address	Set the IP address or domain name of register server
Domain Realm	Set the authentication domain of server
Outbound Proxy	Set the proxy server
Register Expire	Set the register time in second, the default is 300s
Sip Backup Type	Device backup type: Failover/Redundant
Sip Backup Server	Set the address of SIP backup server

Protocol Control	
MWI Subscribe	Disable: the phone prohibits MWI function. Even if it receives a NOTIFY from server that there is a new voice mail, the phone will not have a prompt.
	Enable(Subscribe): the phone enables MWI function and will send SUBSCRIBE. If it receives a NOTIFY from server that there is a new voice mail, the MWI LED on phone will blink to give a prompt.
	Enable(No Subscribe): the phone enables MWI function but will not send SUBSCRIBE. If it receives a NOTIFY from server that there is a new voice mail, the MWI LED on phone will also blink to give a prompt.
Local SIP Port	Set the No. of local SIP port. The default is 5060.
Local RTP Port	Set the No. of local RTP port. The default is 20000.
Keep Alive Packet	Will you keep alive packet or not?
Keep Alive Period	Keep alive interval. The default is 60S.
DTMF	Select DTMF mode in 3 options: “RFC2833”, “In band” and “SIP Info”. The default is RFC2833.
DTMF SIP INFO Mode	DTMF out of band detection mode: signal=*/# or signal=10/11
DNS Type	DNS type: A request, DNS SRV, NAPTR+SRV
Jitter Buffer Max	The jitter buffer maximum. The default is 150.
Anonymous Call Pejection	Will the anonymous call be rejected? The default is disable (namely no reject).
Session Switch	Will the session switch be turned on?
Session Time(Min=90S)	Set the session time. The default is 1800S.
Prack	Temporary recovery confirmation. Ensure the reliable transfer of response of 1XX in SIP.
Support Update Method	Supports the update method.
Rport	The relocation port has penetrated NAT
Sip Transport	SIP transfer protocol: UDP/TCP/TLS
Sip URI	SIP call address uses SIP/SIPS
SRTP	The safe real-time transfer protocol mode: Optional / Mandatory

4.3.2 Audio Setting

You can adjust the volume of microphone and handset in the page, set the codec.

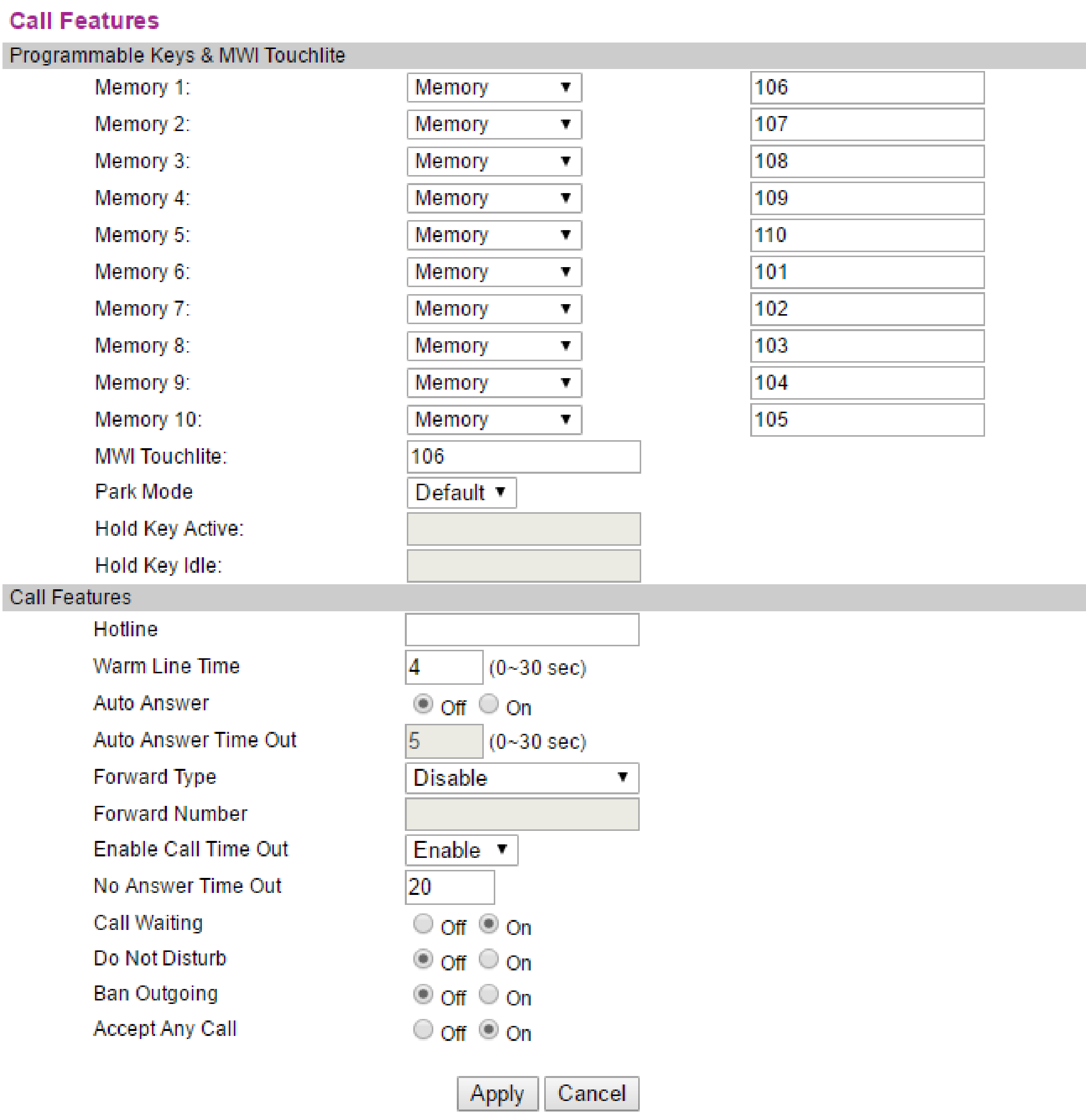


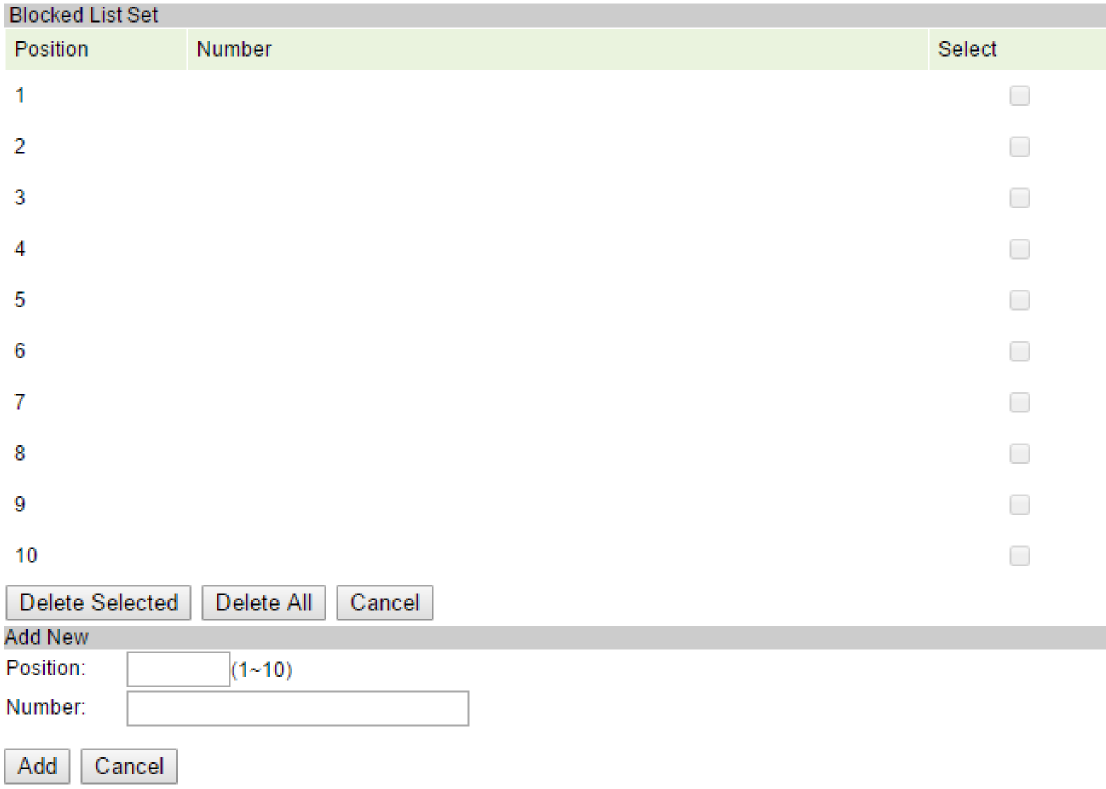
Audio Setting	
Sound and Volume Control	
Handset	Configure the handset output volume. The control range is 1~7. The default is 5.
Speaker	Configure the speakerphone output volume. The control range is 1~7. The default is 5.
Ring Tone	Configure the ringer volume. The range is 1~7. The default is 4.
Signal Standard	The signal standard. There are 12 categories in total. ＃0: Belgium; 1: China; 2: Germany; 3: Israel; 4: Japan; ＃5: Holland; 6: Norway; 7: South Korea; 8: Sweden; ＃9: Switzerland;10: Taiwan; 11: USA
Ringer	Will the ringer be enabled?
Ringer Type	There are 11 ring tones in total for selection. The default is Ringer1.
Codec Setting	
Codec Priority1~6	Set the codec priority, there are 6 modes as follows:
	l G.711a
	l G.711u
	l G.729
	l G.723.1

	l iLBC
	l G.722
Packet Data Size	The packet data size is 20mS by default.
IBLC 15.2k	iLBC 15.2kbit/s is enabled or not. The default is disable.
G.723.1 5.3k	G.723.1 5.3kbit/s is enabled or not. The default is disable.
Voice VAD/CNG	
Voice VAD	Enable or disable Mute detection function
CNG	Enable or disable the comfortable noise.
Codec ID Settings	
DTMF Payload(RFC2833)	DTMF payload. The default is 101.

4.3.3 Call Feature

You can set call feature, create the blocked list and restricted list in this page.





Call Feature	
Programmable Keys&MWI Touchlite	
Mem1~Mem10	1. Set the number in speed dial key.
	2. Set the second function. Each memory can be arbitrarily set as Hold, DND, Transfer, Conference, Multicast Paging
MWI Touchlite	Set the number in shortcut key for voice message pickup.
Park Mode	Enable or disable Park function.
Hold key Active	Set the Call Park number. In Park mode, when one line of phone is in call, press HOLD key to call the number.
Hold key Idle	Set the Call Park number. In Park mode, when the phone is idle, press HOLD key to call the number.
Call Features	
Hotline	Hotline
Warm Line Time	Set the waiting time of user taking the phone off-hook to call the hotline number. The range is 0-9s and the default is 0s. If the warm line time is 0s, the hotline number will be sent out immediately after the phone is off-hook.

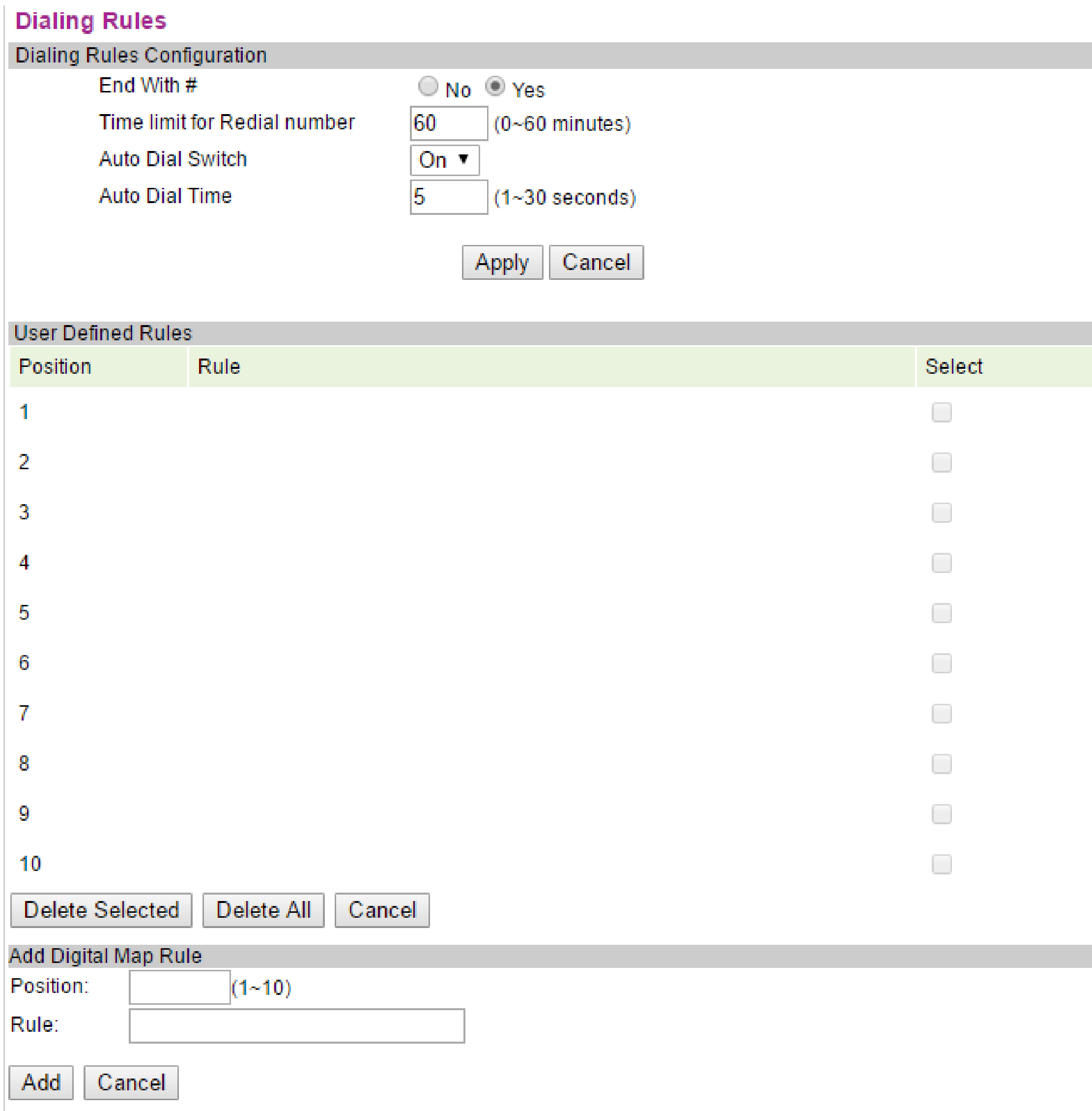
	The set range is 1-9s, for example 3s, the hotline number will be sent out immediately after 3s with the phone off-hook and without pressing any key. As long as any key is pressed within the set time, the time counting will stop.
Auto answer	Enable or disable auto answer function. If enabled, you could set 5 auto answer phone numbers for incoming call.
Auto Answer Time Out	Enable the auto answer function after timeout. The set range is 0~30s, the default is 5s.
Forward Type	Call forward type (mono-choice, the default is "Disable" type)
	Disable: disable the call forward function.
	Always Forward: all the incoming calls are forwarded to the appointed phone.
	Busy Forward: when the phone is busy, the incoming call will be forwarded to the appointed phone.
	No Answer Forward: if the phone has not answered, the incoming call will be forwarded to the appointed phone.
Forward Phone Number	Call the forwarded phone number.
Enable Call time out	Enable the no answer timeout function.
No Answer timeout	Set the no answer time. The default is 20s.
Call waiting	Enable or disable the call waiting.
Do Not Disturb	Set DND.
Ban Outgoing	Restrict any outgoing call.
Accept Any Call	Enable accepting any incoming call.

In the Black List page, you can add blacklist number, you can also delete.

Add New	
Position	Position 1~10
Number	The number to be blocked.

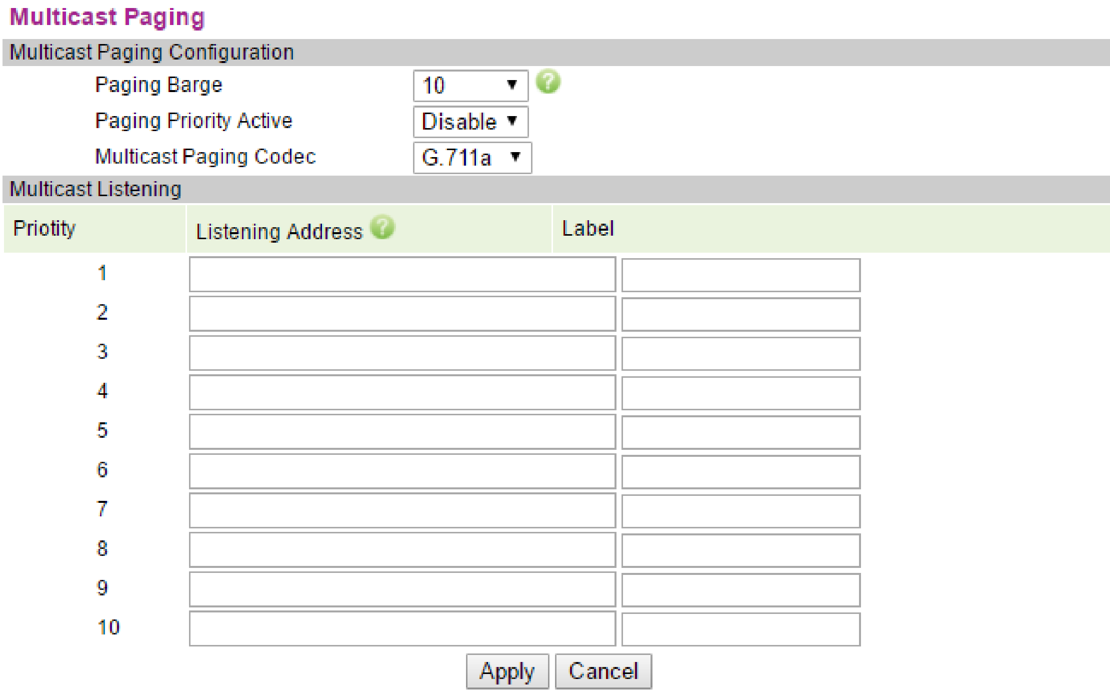
4.3.4 Dial Rule

Configure dialing rules in the page.



Dialing Rules Configuration	
Entry Name	Description
Dialing Rules Configuration	1. Set the end of dialing rules, there are 2 kinds to choose from: ● End with “#”. ● Timeout: Timeout setting. Set the waiting time for dialing end, the unit is second, the default is 5s. The default is "#" as the end of the dial. 2. 60mins。 Redialing timeliness: The default is 60mins, redial will be invalid. Maximum of 60mins can be set.
User Define Rules	Users can add 10 custom dialing rules.

4.3.5 Multicast Paging

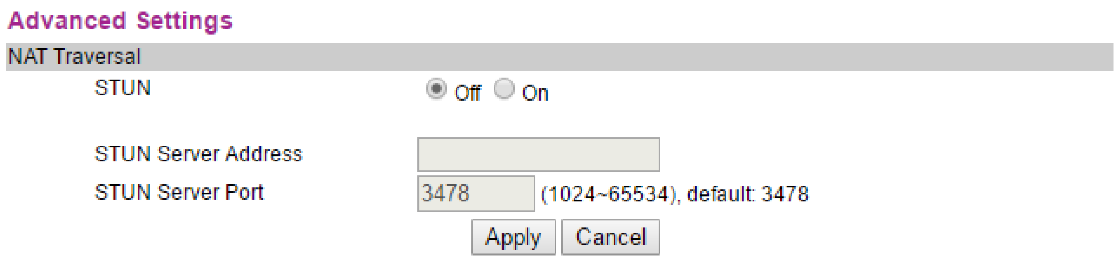


You can use the multicast function that will simply, conveniently and efficiently send the timely notice to each member of the multicast group. The multicast key is set on the telephone to send the multicast RTP stream to the pre-configured multicast address. Through the configuration monitoring multicast address on the phone, listen and play the RTP stream sent by the multicast address, the RTP stream multicast process does not involve SIP signaling. The phone can be set up to monitor 10 multicast addresses.

Multicast Paging Configuration	
Paging Barge	The common call priority in case of the multicast access. Define the call priority, 1 is the top level, 10 is the bottom level.
Paging Priority Active	Paging priority switch: you can enable or disable the paging priority switch. The function determines how to handle the newly incoming multicast RTP stream when the phone is presently performing the multicast session. If the paging priority switch is enabled, the phone will automatically ignore the multicast RTP stream with the lower priority and receive the multicast RTP stream with the higher priority and place the current multicast session on hold. If the paging priority switch is disabled, the phone will automatically ignore all the newly incoming multicast RTP streams.

Multicast Paging Codec	The multicast voice coding format: 0:G.711a; 1:G.711u; 2:G.723; 3:G.729; 4:iLBC; 5:G.722
Multicast Listening	
listening Address	You can set to listen up to 10 different multicast addresses on the phone which can be used to receive the multicast RTP stream sent by them. If the priority of incoming multicast RTP stream is lower than the priority of current call, the phone will automatically ignore the multicast RTP stream. If the priority of incoming multicast RTP stream is higher than the priority of current call, the phone will automatically receive the multicast RTP stream and place the call on hold. You can select to disable the paging priority switch, the phone will automatically ignore all the incoming multicast RTP streams.
Label	Multicast label

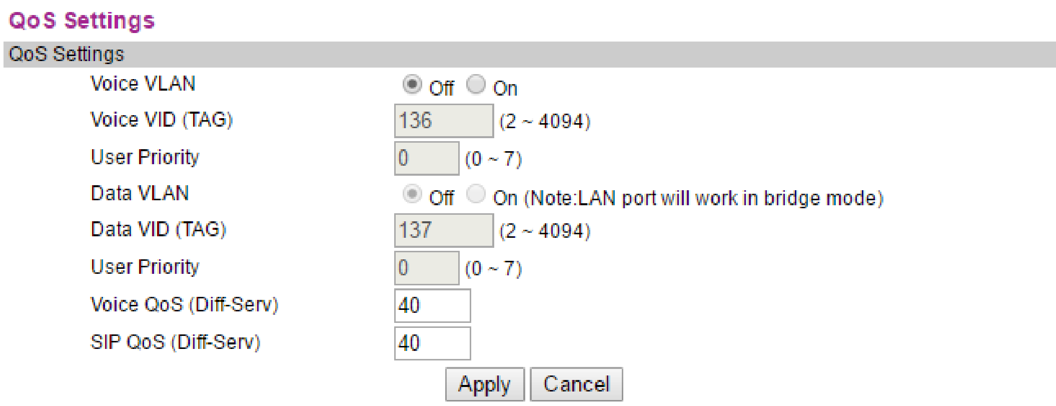
4.3.6 Advanced Settings



Advanced Setting	
Entry Name	Description
Enable	Enable or disable NAT firewall function. The default is enable.
STUN Server Address	Set the address of STUN server.
STUN Server Port	Set the port # of STUN server.

4.4 QoS SETTING

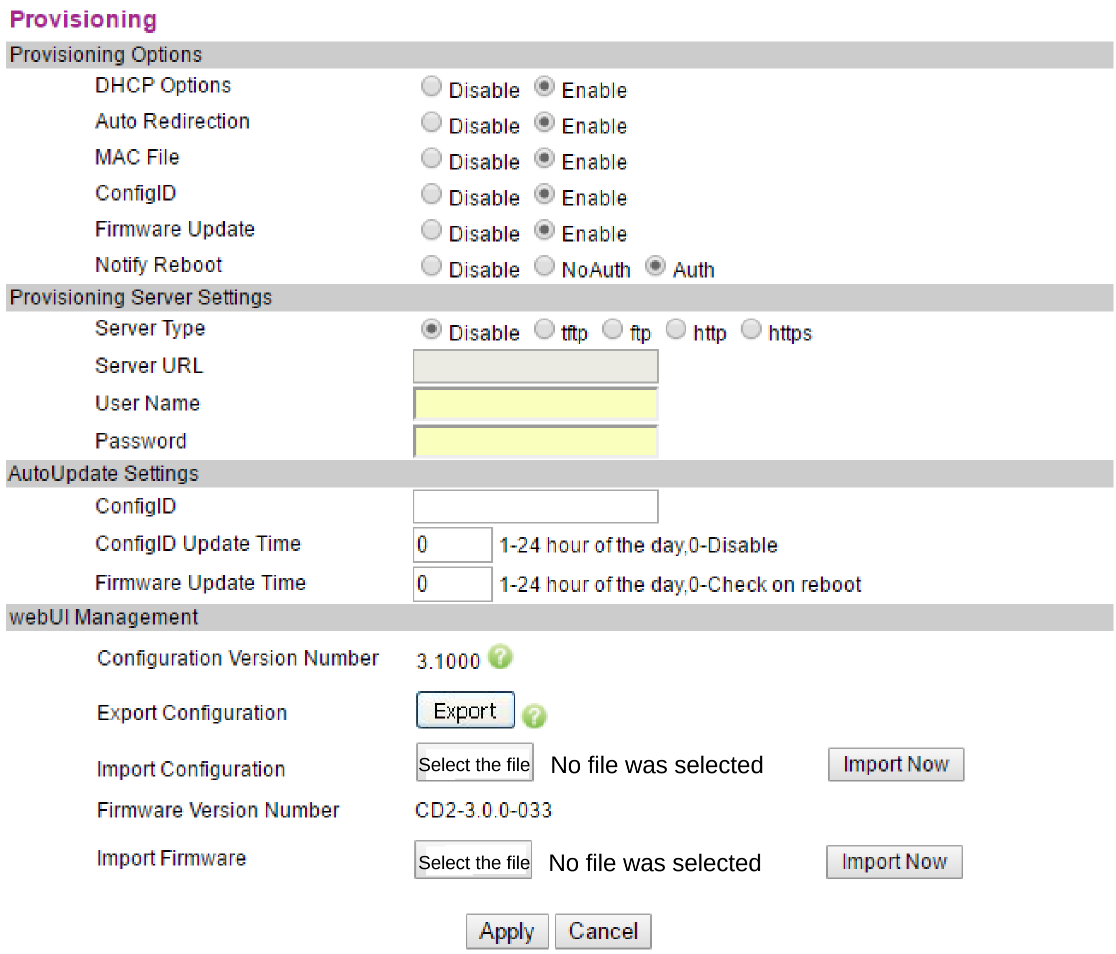
You can get QoS information in the page.



QoS Setting	
Entry Name	Description
Voice VLAN	Enable or disable Voice VLAN function. The default is disable.
Voice VID(TAG)	The Voice Video Tag. The range is 2~4094. The default is 136.
User Priority	User priority. The default is 0.
Data VLAN	Enable or disable Data VLAN function. The default is disable. When it is enabled, LAN port will operate in the bridge mode.
Data Priority	Data label. The range is 2~4094. The default is 137.
User Priority	User priority. The default is 0.
Voice QoS (Diff-Serv)	Voice interval service priority: the default is 40.
SIP QoS (Diff-Serv)	SIP interval service priority: the default is 40.

4.5 PROVISIONING

You can set the configuration information of phone in the page.



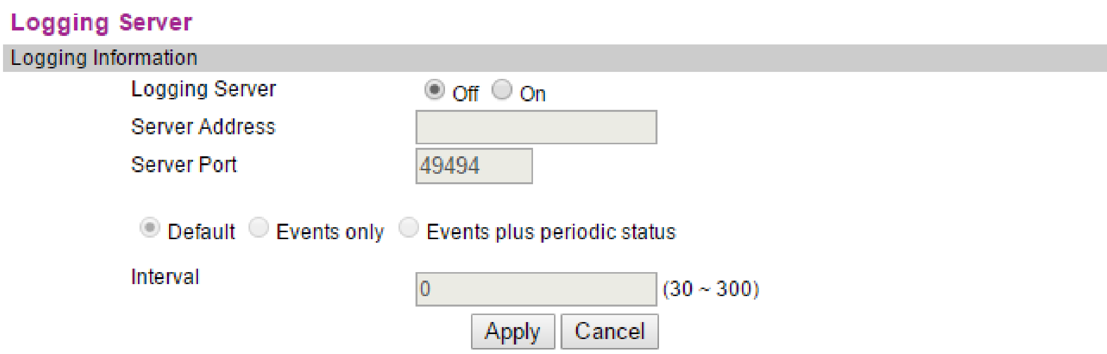
Provisioning	
Provision Options	
DHCP Options	Support DHCP Options parameter or not.
Auto Redirection	Support Auto Redirection or not
MAC File	Support that the Config. filename is MAC address or not
Config ID	Support that the Config. filename is config ID or not
Fireware Update	Support the firmware upgrade
Notify Reboot	Enable or disable Notify Reboot. After enabled, it is divided into 2 cases, one needs the authentication, the another does not need.
Provisioning Server Settings	
Server Type	Configure the server type: disable /TFTP/FTP/HTTP/HTTPS
Server URL	Configure the server address: IP address or domain name
User Name	User name
Password	Password
AutoUpdate Settings	
Config ID	Config ID

ConfigID Update Time	0-24, 0 - Disable, 1-24 hour selects any hours among 1-24 and generates a random number as the minute of upgrade among 0~60 and delays a few seconds to begin to detect if Config ID file is updated.
Firmware Update Time	0-24, 0 - Only check at reboot, 1-24 hour selects any hours among 1-24 and generates a random number as the minute of upgrade among 0~60 and delays a few seconds to check if there is any firmware update.
WebUI Management	
Configuration Version Number	Configure the version number of file
Export Configuration	Export the Config. file to local
Import Configuration	Import the Config. file from local, press "Import Now" to do import
Firmware Version Number	The version number of firmware
Import Firmware	Import the firmware version from local, press "Import Now" to do import

4.6 System Settings

4.6.1 Syslog Server

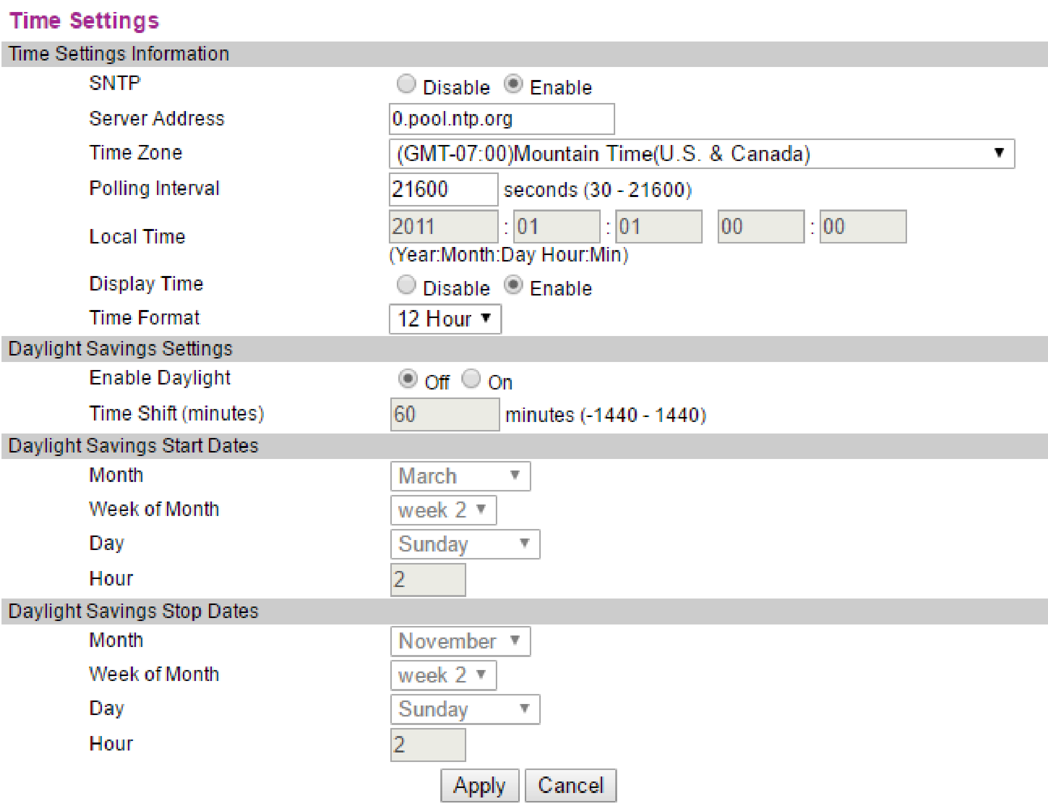
Set the information of Syslog server.



Syslog Server	
Entry Name	Description
Syslog Server	Enable or disable the syslog function. The default is disable.
Server Address	Set the IP address or domain name of syslog server. The default is empty. It could be loaded from option43.

Server Port	Set the port # of syslog server. The default is 49494.
default	The default of logintelval is 0.
Events only	Log information print interval is 1min.
Events plus periodic status	Logint is the setting range of lower interval.
inteval	Log interval time setting.

4.6.2 Time Settings

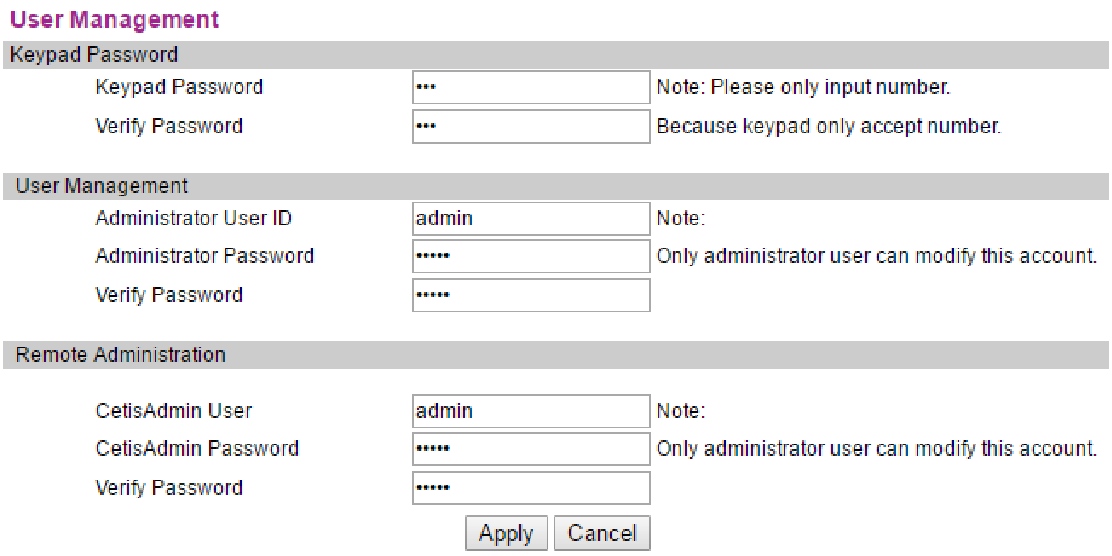


Time Settings	
Time Settings Information	
SNTP	SNTP server enable or disable.
Server Address	SNTP server address: the default is 0.pool.ntp.org
Time Zone	Time zone selection
Polling Interval	Polling interval
Local Time	Local time
Display Time	Display the time or not
Time Format	Time format: 12 hour/24 hour
Daylight Savings Settings	

Enable Daylight	Daylight savings enable or disable.
Time Shift(minutes)	Time difference (minute)
Daylight Savings Start Dates	
Month	Daylight Savings Start Month
Week of Month	Week of Month
Day	Day of Week
Hour	Hour of Day
Daylight Savings Stop Dates	
Month	Daylight Savings Stop Month
Week of Month	Week of Month
Day	Day of Week
Hour	Hour of Day

4.6.3 User Management

Set the user information.

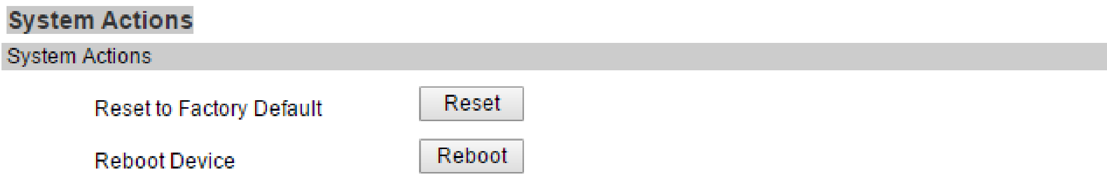


User Management	
Keypad Password	
Keypad Password	Set the keypad access password. The default is 123.
Verify Password	Input the set new password again for verification.
User Management	
Administrator User ID	Set the administrator ID as the username for webpage login. The default is admin.

Administrator Password	Set the password for webpage login in the identity of administrator. The default is admin.
Verify Password	Input the administrator password again for verification.
Remote Administration	
CetisAdmin User	Set the username of remote administrator. The default is admin.
CetisAdmin Password	Set the login password of remote administrator. The default is admin.
Verify Password	Input the administrator password again for verification.

4.6.4 System Actions

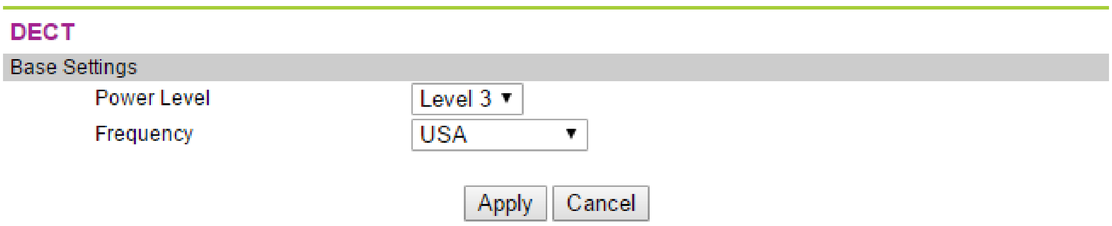
System operation.



System Action	
Reset Factory Default	Click【Reset】button to recover factory setting of phone.
Reboot Device	Click【Reboot】button to reboot the phone.

4.6.5 DECT

DECT configuration.



DECT
Base Settings

Power Level	Select the power level of registered handset. The power level is related to the receiving range. The level 0 is minimum, level 7 has the maximum receiving range.
Frequency	Select the DECT frequency band: select the different countries.

5 SHORTCUT KEYS

1. * * 47 #, Broadcast the current ip address of the phone.

2. * * 39 #, Broadcast the current software version of the phone.

3. * * 85 #, Broadcast the current phonevlan ID.

4. * * 83 #, Broadcast current tftp server address.

5. * * 72 #, Restart the phone.

6. * * 36 #, Broadcast the current account of the phone.

7. * * 33 * password #, Clear all the current configuration of the phone, and automatically restart.

8. * * 77 * password * config ID #, The phone downloads the configuration file from the tftp server and restarts automatically after the download is successful.

9. * * 87 * password * VLAN ID #, Modify the vlan ID of the phone; modify the vlan id success, the prompt success, and broadcast the modified vlan. ID, and then restart the phone.

10. * * 89 * < keypad password > * < TFTP server IP address > * < configid> #, The phone downloads the configuration file from the tftp server and restarts automatically after the download is successful.

11. The following ways: the phone is connected to the POE static settings after the start

    * * 73 * 123 #   Set the phone wan port to a fixed ip address mode.

    * * 74 * 123 * 192.168.18.111 #   Set a fixed ip address, I heard ip broadcast voice after the success of the amendment.

    * * 76 * 123 * 255.255.255.0 #   Set subnet, I heard the broadcast ip address of the voice after the success of the amendment.

    * * 49 * 123 *192.168.18.1 #   Set the gateway, I heard the ip address of the broadcast voice after the success of the amendment.

    **72# after the phone restarts, input IP address in the PC's LAN browser, enter the WEB setup IP account settings.

FCC Statement

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference, and

(2) This device must accept any interference received, including interference that may cause undesired operation.

FCC Radiation Exposure Statement:

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .This base unit should be installed and operated with minimum distance 20cm between the radiator& your body.

ISED RSS Warning/ISED RF Exposure Statement

This device complies with Innovation, Science and Economic Development Canada licence-exempt RSS standard(s). Operation is subject to the following two conditions: (1) this device may not cause interference, and (2) this device must accept any interference, including interference that may cause undesired operation of the device.

Le présent appareil est conforme aux CNR d'ISED applicables aux appareils radio exempts de licence. L'exploitation est autorisée aux deux conditions suivantes: (1) l'appareil ne doit pas produire de brouillage, et (2) l'utilisateur de l'appareil doit accepter tout brouillage radioélectrique subi, même si le brouillage est susceptible d'en compromettre le fonctionnement.

This equipment complies with ISED radiation exposure limits set forth for an uncontrolled environment. This base unit should be installed and operated with minimum distance 20cm between the radiator& your body.This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter. Le rayonnement de la classe b repecte ISED fixaient un environnement non contrôlés.Installation et mise enœuvre de ce matériel devrait avec échangeur distance minimale entre 20 cm ton corps.Lanceurs ou ne peuvent pas coexister cette antenne ou capteurs avec d'autres.

For Handset part

This device has been tested and meets applicable limits for Radio Frequency (RF) exposure. The SAR limit of FCC/ISED is 1.6W/Kg averaged over one gram of tissue. The highest SAR value reported under this standard during product certification for use at the head is 0.03 W/Kg.