E103IP USER MANUAL

VERSION: V1.0
FOR MODELS: E103IP, E203IP

CONTENTS

After phone is boot-up and obtains the TFTP Server address through DHCP Sever, then phone will produce "toot-toot" sound to prompt entering "Config ID". After inputting ID through the numeric keyboard and "#" for ending, phone will auto-download the config file from TFTP Server. If the download is successful, phone will be re-boot; if failed, 15s later phone will enter into default standby status; if you do not want to download, you can press "#" directly for entering into the default standby status; if the download is not completed or the downloaded config file "autoupdate module" has no config parameter in the "config file name", phone will still prompt inputting "Config ID" after re-boot. After phone enters into the default standby status, pressing "47"key will make the phone auto-broadcast its own IP address.**

Function

1. Provide a Backup SIP Server
2. Support NAT, Firewall
3. Support DHCP assigning IP address, etc automatically
4. Support PPPoE (used while connecting ADSL，cable modem)
5. It can update the program through HTTP, FTP and TFTP
6. Check the dynamic voice; Soft the noise; Buffer technique of voice
7. Hold Function
8. Hotline Function
9. Speed-dial
10. Call-forward, Three-way conference call
11. DND (Do Not Disturb), Black List, Limit List
12. Auto-answer.
13. Set through standard Web Browser
14. Remote Management Function
15. Classification management for common user′s password and super user′s password.
16. Broadcast the IP address, Vlan ID, version number and phone number.
17. Cordless Handset, 1.9GHz Operation, with Handshake Technology
18. Support VLAN function
19. Call park function

Standard and Protocols

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- PPPoE
- DHCP Client and Server
- Support G.711a/u, G729, G7231 5.3/6.3 audio Codec
- SIP RFC3261, RFC 2543
- Support IAX2
- TCP/IP: Internet transfer and control protocol
- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth
- Telnet: Internet's remote login protocol
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol

1. Introduction

This is the user manual of E103 IP . Some configuration should be done before use the E103 IP phone, and then it can work normally. This manual will illustrate how to set the phone through keyboard and web service.

1.1 Overview of Hardware

1.1.1

The two RJ-45 network interface support the 10/100M Ethernet. The default WAN interface is a DHCP Client server. Users connect the WAN interface to ADSL or switch, and connect the LAN interface to the computer. You can use the administrator's user name "admin" and password "admin" to login and set.

1.1.2

Only the WAN interface supports the POE.

1.2 Overview of Software

Network Protocol • SIP v1(RFC2543) v2(RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/PPPOE Client • Telnet/HTTP Server • DNS Clients **Codec** • G.711： 64K bit/s(PCM) • G.723.1： 63k/5.3k bit/s • G.726： 16k/24k/32k/40k bit/s(ADPCM) • G.729A： 8k bit/s(CS-ACELP) • G.729B： adds VAD & CNG to G.729	**Tone** • Ring Tone • Ring Back Tone • Dial Tone • Busy Tone **Phone Function** • Volume Adjustment • Speed dial key • Phonebook **IP Assignment** • IP (Static IP) • DHCP • PPPoE
Voice Quality • VAD： Voice activity detection • CNG： Comfortable noise generator • LEC： Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	**Security** • HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/RFC2617) **QoS** • QoS field
Call Function • Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference	**NAT Traversal** • STUN **Configuration** • Web Browser • Console/Telnet • Keypad

DTMF	Firmware Upgrade
l DTMF RELAY l DTMF RFC 2833 l DTMF SIP Info	l TFTP l HTTP l FTP
SIP Server	
Provide a Backup SIP Server	

2. E103-IP phone keyboard

2.1 Keyboards functions

2.1.1 Function Table of Base Keyboard

Name	Status	Function
Store	On-hook	To enter storage mode for speed dialing
Flash	On-hook	To enter the deleting mode for pressing down 3 seconds
Redial	Dialing	Re-dial the last called number
Locate	On-hook	It's calling handset for press down for a little time and registering with handset for pressing down 3 seconds
	Call	calling the handset
Volume +	Call	Increase the volume
Volume -	Call	Decrease the volume
Speak		Hands free
Mute	Call	Mute
M1~M7	On-hook Dialing	7 speed dial numbers
Voicemail	On-hook Dialing	Pick up voicemail
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	It can be regarded as the first number being dialed out or the end mark for ending number.

2.1.2 Function table of handset keyboard

2.2 Keyboard Function

Item	Status	Function
Line1 ON/OFF	On-hook	On/off Hook switch of Handset
	Talking	Hold/retrieve the current line talking
Line2 ON/OFF	On-hook	On/off Hook switch of Handset
	Talking	Hold/retrieve the current line talking
Hold	Call	Call Waiting
Mute	Call	Mute

Redial	Dialing	Redial the number of last time
Volume +	Call	Increase the volume
Volume -	Call	Reduce the volume
FNC	Off-Hook	Quickly cut off the current line and off-hook again.
1	Dialing	"1"
	On-Hook	Long pressing can pick up the message
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	"*"
	On-Hook	Long pressing for 3 seconds entering into the registration status of the handset
#	Dialing	It can be regarded as the first number being dialed out or the end mark for ending number.

2.2.1 Store numbers

Press "STORE" key to enter into the setting state of storing keys, and input storing number, then press the corresponding storing key to store the numbers.

2.2.2 Voice Control

Pressing "VOL+" to increase the volume, and press "VOL-" to decrease the volume.

2.2.3 Hold Function

It is used for holding the current line or forwarding to other handset, for example:

When you use the #1 handset and want to use the #2 in a call, now you need to hold the #1 handset then pick up the #2 handset.

If line1/2 is in the conversation, press down "line1/2" key or "hold" key, may hold the current conversation.

2.2.4 Register handset

The handset registers to the base have two ways: Manual registration and automatic registration.

Manual registration: Under On-Hook status, pressing the "Locate" key on the base unit for three seconds to enable the handset register mode, and the Charge indicator light flashes at the same time; then press "*"key for three seconds to enable the handset register mode, then ON/OFF indicator light on the handset, and will give the prompt tone for successful registration.

Automatic registration: place the handset in the plane, the "charge" instruction light begins flash at the same time, it means enable the handset register mode, the "ON/OFF" instruction light flashes at the same time, if the handset register to the base is successful, then the "charge" instruction light on the base and the "ON/OFF" instruction light on the handset will stop flickering, and will give the prompt tone for successful registration.

Note: Each base phone can most register 5 handsets.

2.2.5 Delete handset

Keep pressing the "FLASH" key on the base unit for three seconds, the Speaker will give "beep" sound, then

Press "0" on the base unit to delete all the handset.

Press "1" on the base unit to delete the first handset.

Press "2" on the base unit to delete the second handset.

Press "3" on the base unit to delete the third handset.

Press "4" on the base unit to delete the fourth handset.

Press "5" on the base unit to delete the fifth handset.

After pressed the number keys, Charge indicator light will flashes, which means the handset is being deleted by the phone. When Charge indicator light stops flashing, it means that the deletion is finished.

2.2.6 Call forward

Choosing the "forward type" of the "Advanced Sip Setting" (the default option is "OFF"). Choose one of the three types which contain always, busy and no answer to realize call forward.

2.2.7 Three way conference call

2.2.7.1 Three way conference call of the handset

1) Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the E103 IP phone is A, and if user B calls user A through VoIP phone, user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "*", then can make 3-way conference call.

2) Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the E103 IP phone is A, and if user B calls user A through VoIP phone, user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "Conf", then can make 3-way conference call.

2.2.7.2 Three way conference call of the base unit

Choosing the "Enable Three Way Call" of "Call Service Setting" (the default option status of the phone), suppose the user of the E103 IP phone is A, and if user B calls user A through VoIP phone , user B needs to make 3-way conference call with user A and C, then user A can use the second line to call user C, the call auto-maintain with user B, during the conversation with C, pressing down "*" can make 3-way conference call.

2.2.8 Line Switch

2.2.81 Handset Line Switch

1）Suppose the line1/2 of handset is talking, the line2/1 is holding, if you want to implement line switch, first you need to press down "Hold ", then press down "ON/OFF" of the choice line. You can implement line switch.

2) Suppose the line1/2 of handset is holding, the line2/1 is dialing, if you want to switch to line1/2 talking, first hang up line2/1, then press down "ON/OFF" of Line1/2 or press "HOLD". You can implement line switch.

2.2.8.2 Base unit Line Switch

Suppose the line1/2 of the base unit is talking, line2/1 is holding, press down "line2/1" directly and switch to line2/1 talking.

2.2.9 Call park function

The "Park1" in the "Park Mode" of "Advanced SIP Setting" should be in chosen mode(the default one is chosen default mode), config the related parameter "s", "t" and "p" in memory key page, call park function may use normally afterward.

2.3 Functions and setting catalog

2.3.1 Menu catalog

1) Network
2) Call Feature
3) SIP
4) DSP
5) System
6) Other Setting

2.3.2 Network

2.3.2.1 LAN

1) Bridge Mode
2) IP
3) Netmask
4) DHCP Server
   - ◆Switch
   - ◆DNS Relay
5) NAT
   - ◆Switch
   - ◆FTPalg
   - ◆PPPTPalg

2.3.2.2 WAN

1) Status
2) Static Net
   - ◆IP
   - ◆NetMask
   - ◆Gateway
   - ◆DNS
   - ◆DNS2

3) PPPOE

◆User name

◆Password

4) QoS

2.3.3 Call Feature

23.3.1 Phone-number

1) **Public SIP**

2) **Private SIP**

2.3.3.2 Limit-List

1) Current

2) ADD

3) DEL

2.3.3.3 Black-List

1) Current

2) ADD

3) DEL

2.3.3.4 FastCall

2.3.3.5 Three Talk

2.3.3.6 Call-Waiting

2.3.3.7 Call-Forward

1) Condition

2) SIP

◆Transfer Num

◆Transfer IP

◆Port

2.3.3.8 Dial-Rule

1) End with "#"

2) Fixed Length

◆Switch

◆Length

2.3.4 SIP

2.3.4.1 Reg Status

1) Public Reg

2) Private Reg

2.3.4.1 Reg Switch

1) Public

2) Private

2.3.4.2 Server

1) Public

2) Private

2.3.4.3 Domain

1) Public

2) Private

2.3.4.4 User Agent

1) Public

2) Private

2.3.4.5 Detect-server

2.3.4.6 Dtmf-mode

2.3.4.7 Interval-time

2.3.4.8 Swap-server

2.3.4.9 RFC-version

2.3.4.10 Signal-Port

2.3.4.11 Stun

1) Switch

2) Addr

3) Port

4) Expire Time

2.3.5 DSP

2.3.5.1 Codec

2.3.5.2 Handdown-time

2.3.5.3 Dtfm-Volume

2.3.5.4 Input-Volume

2.3.5.5 Output-Volume

2.3.6 System

2.3.6.1 Save

2.3.6.2 Reboot

2.3.6.3 Set Default

2.3.7 Other Setting

2.3.7.1 Syslog

1) Switch

2) Server-IP

3) Server-Port

2.3.8 Setting catalog

1. Pressing down the "Menu" key to enter the setting status of speed dial key and input the needed saving No., then press the corresponding speed dial key to save.

2. Pressing down "Menu" above 3 seconds to enter the setting status and the default keyword is 123, and then you can press "Enter" to enter, "Menu" to exit.

3. When modifying the setting, press "Redial" to enter modification status, and "0" is to make no choice, "1" is to make choice, "Enter" is to confirm the modification, and "Menu" is to quit the modification. After finished the modification setting, we will save it on "Save" menu. After rebooting all the settings will be go into effect.

3. through web browser to set phone

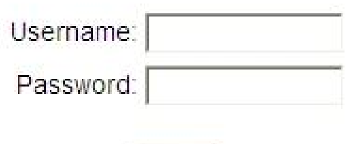
Plug one end of the network line to the network card port of the computer, the other end to Lan port of the phone, phone will obtain the IP address automatically, open IE, input the IP address on Address column, then enter into the Web Setting Page.

The method of obtaining the dynamic IP address is:

Under the on-hook status, press "**47#", then phone will broadcast the current IP address.

3.1 Logon

The default user name and password are admin/admin and guest/guest. and, admin/admin is super user name and password, guest/guest is common user name and password。Logon interface is as follows:



Current Status

Network			
WAN		LAN	
Connect Mode	DHCP	IP Address	192.168.10.1
MAC Address	00:19:f3:03:70:d2	DHCP Server	OFF
IP Address	192.168.21.107		
Gateway	192.168.21.1		

Phone Number		
SIP LINE 1	6004@61.156.234.90 :5060	Registered
SIP LINE 2	@ :5060	Unapplied

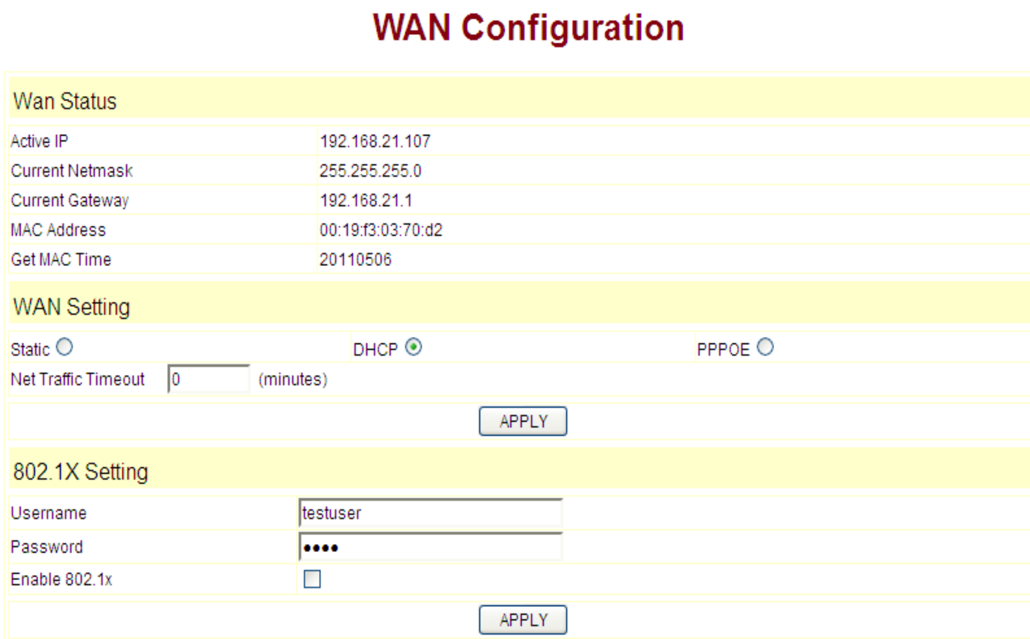
Version: SD1 V1.8.4-846 Jan 13 2011 18:35:07

3.3 Network

3.3.1 Wan Config

WAN port network setting page.

Support static IP, dynamic obtain IP and PPPoE.



Net Traffic Timeout: when wan port network emergence failure, the phone cannot obtain IP address; the phone will auto reboot after the setting time. The default time is 2mins.

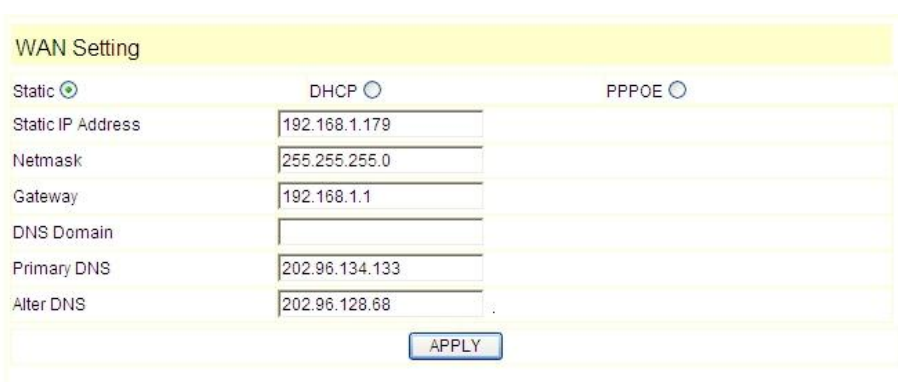
802.1x Setting: when you enable the 802.1x, after the authentication completes successfully, it should then get IP using DHCP.

Username: 802.1x server username is voip.

Password: the default password is 123456.

Enable 802.1x: enable 802.1x authentication.

Configure Static IP:



----Enable *Static*;

----Set E103-IP's IP address in the *IP Address*;

----Set netmask in the *Netmask* field;

----Set router IP address in the *Gateway*;

----DNS Domain:

----Set local DNS server in the *Preferred DNS* and the *Alternate DNS*。

Configure to dynamic obtain IP

----Enable *DHCP*;

If there is DHCP server in your local network, E103 IP will automatically obtain WAN port network information from your DHCP server.

Configure PPPoE:

----Enable *PPPoE*

----*PPPoE server*: Enter "ANY" if no specified from your ITSP.

----Enter PPPoE username and password in the *username* and *password*.

E103 IP will automatically obtain WAN port network information from your ITSP if PPPoE setting and the setup are correct.

Notice: If user accesses the IP phone through WAN port. She/he should use the new IP address to access the IP phone when the WAN port address was changed.

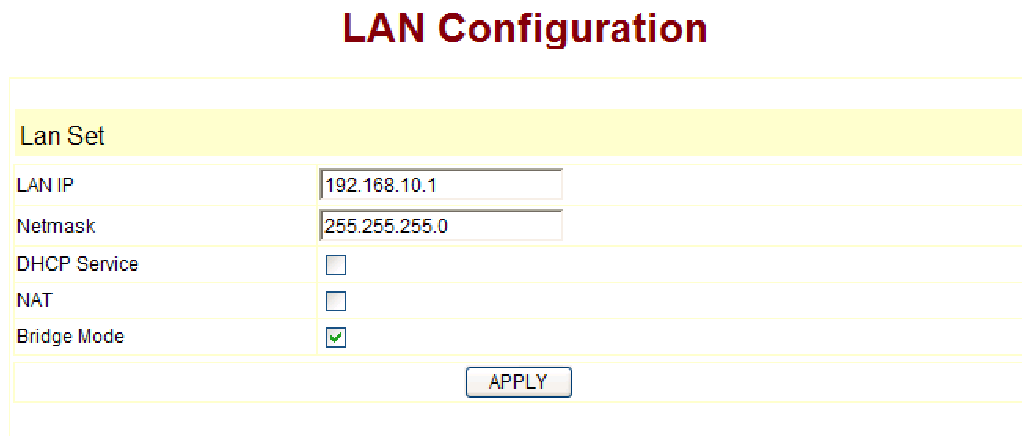
3.3.2 LAN Config

LAN IP / Netmask: Set the IP and Netmask for the LAN

DHCP Server: Enable DHCP service in LAN port; after user changed LAN IP, phone will automatically modify DHCP Lease Table and save the configure according to IP and Netmask, DHCP server configure won't take effect unless you reboot the device.

NAT: Enable NAT.

Bridge Mode: Enable this option to switch to bridge mode. IP phone won't assign IP for its LAN port in bridge mode and its LAN and WAN port will be in the same network. (This setting won't take effect unless you save the config and reboot the device)



3.4 VoIP

3.4.1 SIP Config

Setting page of public SIP server:

Register Server Addr: Register address of public SIP server;

Register Server Port: Register port of public SIP server, default port is 5060;

Register Username: Username of your SIP account (Always the same as the phone number);

Register Password: Password of your SIP account.

Proxy Server Addr: IP address of proxy SIP server (SIP provider always use the same IP for register server and proxy server, in this case you don't need to configure the proxy server information.);

Proxy Server Port: Signal port of SIP proxy;

Proxy Username: proxy server username;

Proxy Password: proxy server password;

Domain Realm: SIP domain, enter the SIP domain if any, otherwise E103-IP will use the proxy server address as SIP domain.

Local SIP port: Local SIP register port, default 5060;

Phone Number: Phone number of your SIP account;

Enable Register: Enable/Disable SIP register.E103 IP won't send register info to SIP server if disable register.

Message Waiting Indication：Set Disable/Enable message waiting through pull-down menu, including:

Disable：MWI is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt.；

Enable (Subscribe): MWI is enabled and SUBSCRIBE will be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will blink to prompt.

Enable(No Subscribe)：MWI is enabled, but phone will not send SUBSCRIBE, if the server sends NOTIFY message indicating new voicemail received, MWI LED will also blink to prompt.

SIP1 Subscribe will still be used in config file: config option "0" means "Disable", "1" means "Enable (Subscribe)", "2" means "Enable (No Subscribe)".

Advanced SIP Setting

Advanced SIP Setting			
Register Expire Time	60 seconds	Forward Type	Off
NAT Keep Alive Interval	60 seconds	Forward Phone Number	
User Agent	TMX SD1 V1.8.4-846	Server Type	common
Signal Key		DTMF Mode	DTMF_RFC2833
Media Key		DTMF SIP INFO Mode	Send 10/11
Local Port	5060	RFC Protocol Edition	RFC3261
Ring Type	Type 1	Transport Protocol	UDP
Park Mode	Default	Subscribe Expire Time	300 seconds
Enable Keep Authentication	☐	Signal Encode	☐
NAT Keep Alive	☐	Rtp Encode	☐
Enable Via rport	☑	Enable Session Timer	☐
Enable PRACK	☐	Answer With Single Codec	☐
Long Contact	☐	Auto TCP	☐
Dial Without Register	☐	Click To Talk	☐
Enable URI Convert	☑		
	APPLY		

Register Expire Time: Register Expire time, default is 60 seconds. E103-IP will auto configure this expire time as the server recommends if it is different from the SIP server.

NAT Keep Alive Interval：NAT keep alive interval time, the default is 60 seconds;

Forward Type: the type of call forward; (the default is OFF);

OFF: Disable call forward;

Busy: If the phone is busy, it will forward to the appointed phone;

No answer：If no answer, it will forward to the appointed phone;

Always：The caller always forward to the appointed phone.

Forward Photo Number：Call the forwarded phone number.

Signal Key：Setting Signal Key; In order to prevent blocking, cooperate platform to encrypt signal, input key here,

Media Key: Setting Media Key;

Subscribe Expire Time: Config the time of sending subscription message; Each interval time, sending a subscription message. Mainly subscribe other′s state or voice message.

Enable URI Convert：Enable URI Convert function, Exchange `#′to `%23′and sending out.

Enable Keep Authentication: Whether allow the phone support register with authentication directly to send or not, Such device do not certification requirement and response with server every time, After the server received a request with the registration of certification, The server can directly reply confirmation message.

Detect Interval Time：Set server detect interval time，if phone open SIP detect server function，it will detect server every time whether or not to respond.

Signal Encode: Open signal encrypt;

Rtp Encode: Open voice encrypt;

Enable Via rport: Whether support the RFC3581 or not. Enable Via rport is used in inside network, and needs SIP server support. To maintain network equipment with net device outside of NAT connection.

Enable Session Timer: Whether the phone support the RFC4028 or not .At a certain period of time to refresh conversation ,in case a long time ,the other side drop off or break off, after refresh ,the other side did not respond ,we will hang up.

Enable PRACK:Whether the phone support SIP-prack function (used mainly in the CRBT). When you receive 183 signal, you can send prack message. Recommend to use the default configration.

Answer With Single Codec: When called to do, Only in response to Codec.

Long Contact: Config Contact field bring more parameter.

Auto TCP: When th message body exceeds 1300 bytes, automatic transmission using the TCP protocol, to protect the availability of transmission.

Detect Interval Time: Co-work with the Auto Detect Server, if Auto Detect Server is enable, E103 IP will periodically detect whether the SIP server is available according this setting.

Encrypt Key: The particular service system decrypts of the key, matching with the server Type usage, the key provide by the particular service system supplier, default is empty

Server Type：The particular service system supplier carries out the sign and speeches to encrypt, default is common；

DTMF Mode: DTMF signal sending mode: support RFC2833, DTMF_RELAY (inband audio) and SIP info.

RFC Protocol Edition: Current E103 IP SIP version. Set to RFC 2543 if the gate need to communicate to devices (such as CISCO5300) using the SIP 1.0.

Default is RFC 3261.

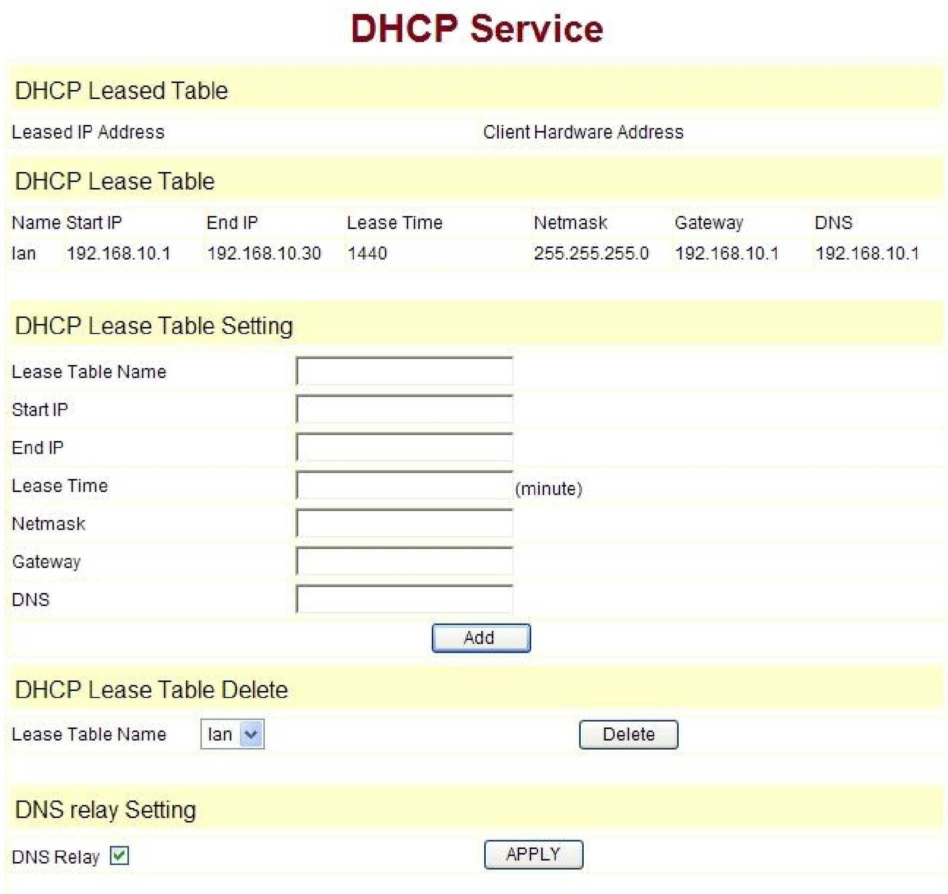
Park Mode: the default park mode is default; it means that the phone disenables call park function. If you want to enable call park function, choose"park1" mode, click "APPLY" button。

3.5 Advance

3.5.1 DHCP Server

DHCP server manage page.

User may trace and modify DHCP server information in this page.



DHCP Lease Table: display the IP—MAC corresponding table that the server distributed.

Lease Table Name: Lease table name.

Start IP: Start IP of lease table.

End IP: End IP of lease table. Network device connecting to the E103 IP LAN port can dynamic obtain the IP in the range between start IP and end IP.

Lease Time: DHCP server lease time.

Netmask: Netmask of lease table.

Gateway: Default gateway of lease table

DNS: Default DNS server of lease table.

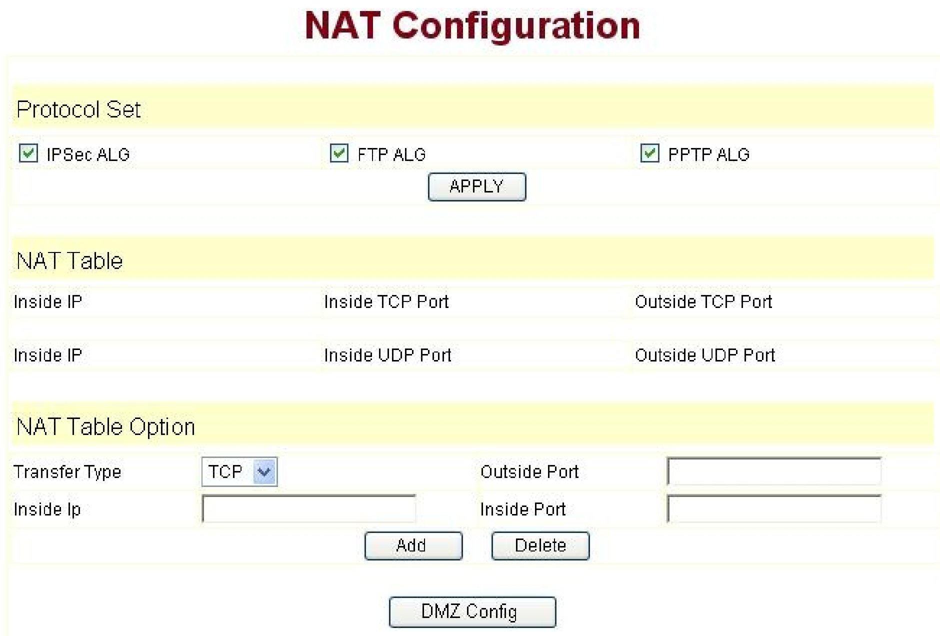
DNS Relay: Enable DNS relay function.

User may use below setting to add a new lease table.

Notice: This setting won't take effect unless you save the config and reboot the device

3.5.2 NAT

Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.



IPSec ALG: Enable/Disable IPSec ALG;

FTP ALG: Enable/Disable FTP ALG;

PPTP ALG: Enable/Disable PPTP ALG;

Transfer Type: Transfer type using port mapping.

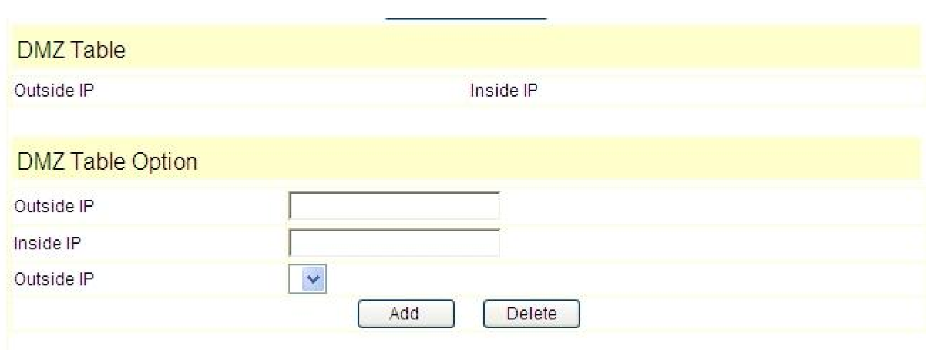
Inside IP: LAN device IP for port mapping.

Inside Port: LAN device port for port mapping.

Outside Port: WAN port for port mapping.

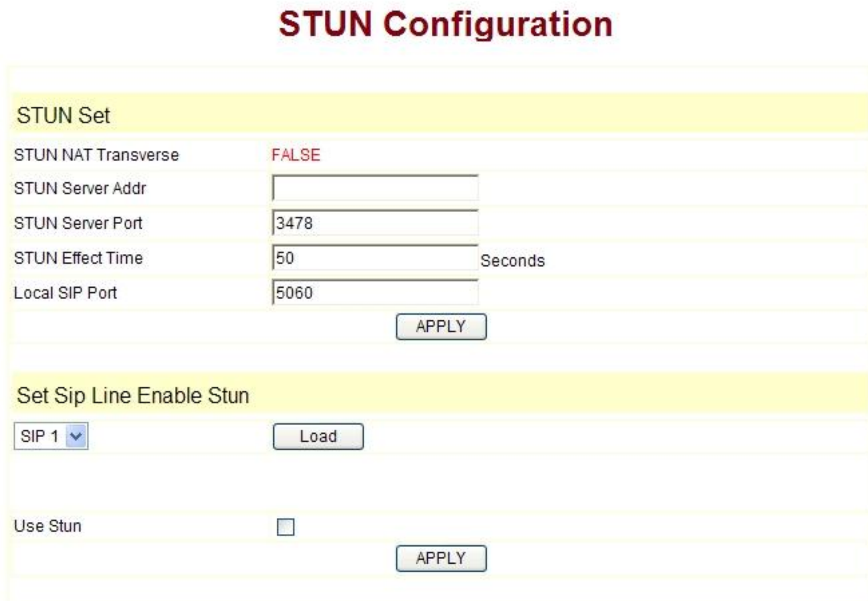
Click Add to add new port mapping item and Delete to delete current port mapping item.

DMZ Config:



3.5.3 STUN

This page is used to set the private sip server, stun server, and back up sip server information.



STUN Server setting: SIP STUN is used to realize SIP penetrates through NAT, when the phone configures IP and port of STUN server (default is 3478) and select Enable SIP Stun, common SIP server can be used to realize the phone to penetrate through NAT. In this way, If you have common SIP proxy and STUN server parked public network, it is all right, but STUN only support three NAT ways: FULL CONE, restricted, port restricted;

STUN Server Addr: configure stun server address;

STUN Server Port: configure stun server port default 3478

STUN Effect Time: stun detect NAT type circle, unit: minute.

Local SIP Port：SIP port of this phone.

Load：Load the choices of SIP line.

Use Stun：Stun. Set the Stun that allows/forbids use user setting.

3.5.4 Net Service

HTTP Port: configure HTTP transfer port; default is 80. User may change this

port to enhance system's security. When this port is changed, please use http://xxx.xxx.xxx.xxx:xxxx/ to reconnect.

Telnet Port: configure telnet transfer port, default is 23.

RTP Initial Port: RTP initial port.

RTP Port Quantity: Maximum RTP port quantity, default is 200

Notice:

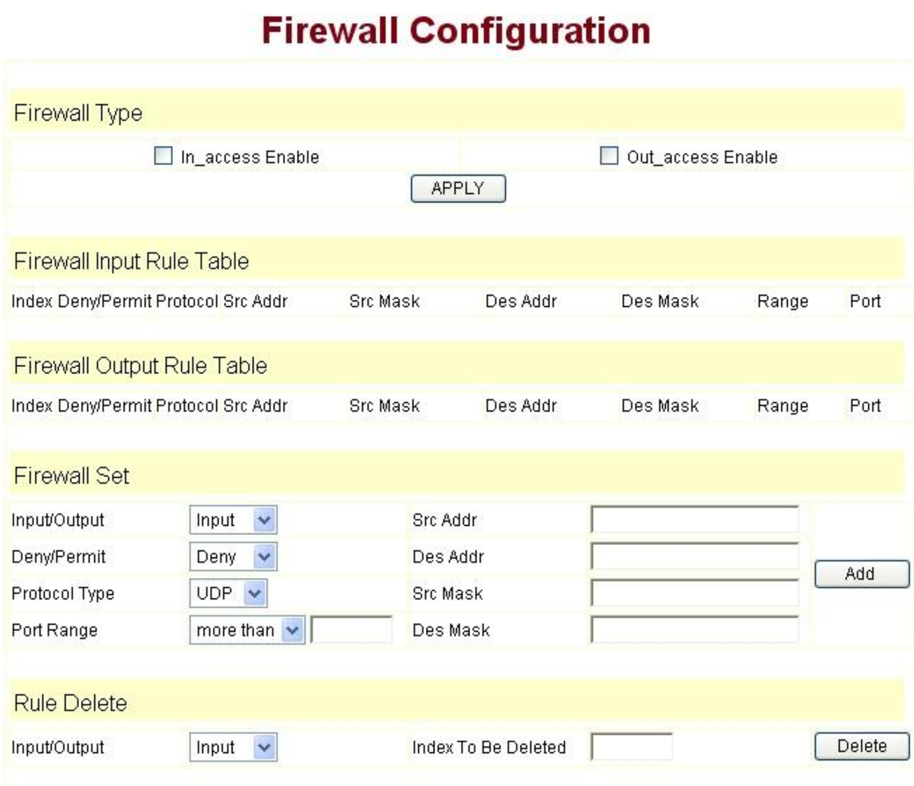
Settings in this page won't take effect unless save and reboot the device.

If you need to change telnet port or HTTP port, please use the port greater than 1024, because ports under 1024 is system remain ports.

HTTP service firbids if HTTP is set to 0.

3.5.5 Firewall settings

Firewall setting page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.



Access list support two type limits: Input access limit or output access limit. Each type supports 10 items maximum.

E103IP firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:

In_access enable: enable in_access rule;

Out_access enable: enable out_access rule;

Input/Output: Specify current adding rule is input rule or output rule.

Deny/Permit: Specify current adding rule is deny rule or permit rule.

Protocol Type: Protocol using in this rule: TCP/IP/ICMP/UDP.

Port Range: Port range of this rule.

Src Addr: Source address. It can be a specific IP address or network address.

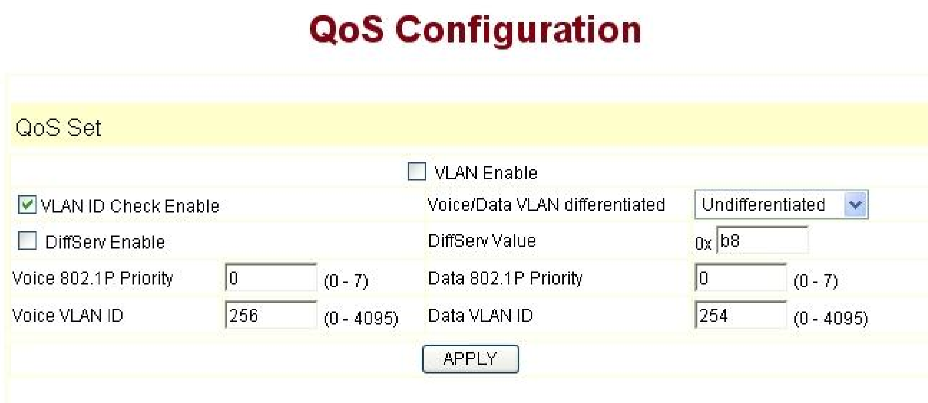
Dest Addr: Destination address. It can be a specific IP address or network address.

Src Mask: Source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

Des Mask: Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

3.5.6 VLAN Configuration

Below is QoS Configuration page:



E103-IP phone implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. E103-IP will sort the packets using the QoS and sends it to the destination.

VLAN Enable: Enable VLAN election, it can separate the voice message, signaling message and data message at the second floor, and by configuring the IP precedence which in ToS field of voice message to realize the separation of voice message and data message at the third floor, through which to allow the upper switch or router to forward the voice message firstly. (On condition that the switch or router can identify the ToS field).

VLAN ID: Dispose VLAN ID is added a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 1~4094.

DiffServ Enable: If enable the VLAN service, it indicates use DSCP mode to realize three layers QoS. This moment, the DSCP of SIP signals which between E103-IP Phone and MGC. It will use Class Selector 5 (The value is 0xA0). And the DSCP of mediums information (In RTP packets) would be used the values of DiffServ Value field.

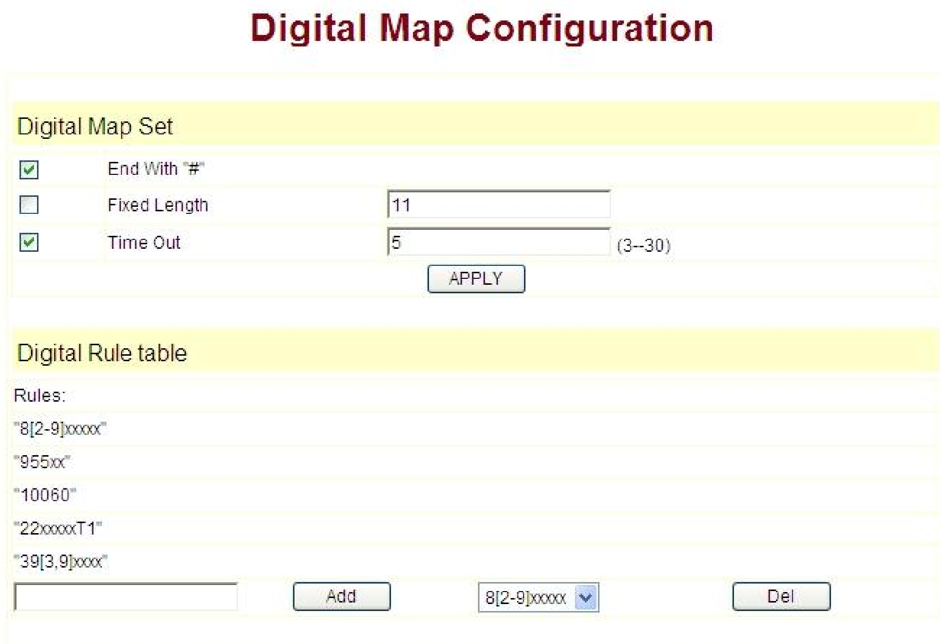
DiffServ Value: The value range:

0x28,0x30,0x38,0x48,0x50,0x58,0x68,0x70,0x78,0x88,0x90,0x98,0xb8.default is 0xb8 ,oxb8 stands for best fast transmission; 28-38 is guarantee for the transmission priority for the 1st rank , 48-58 is guarantee for the transmission priority for the 2nd rank, 68-78 is guarantee for the transmission priority for the 3rd rank, 88-98 is guarantee for the transmission priority for the 4th rank.

802.1P Priority: The priority of 802.1p

3.5.7 Digital Map

Digit map is a set of rules to determine when the user has finished dialing.



E103 IP support below digital map:

Digital Map is based on some rules to judge when user end their dialing and send the number to the server. E103 IP support following digital map:

----End With "#": Use # as the end of dialing.

----Fixed Length:: When the length of the dialing is matched, the call will be sent.

----Timeout:: Specify the timeout of the last dial digit. The call will be sent after timeout

----Prefix:: User define digital map:

[ ]: Represents the range of digit, can be a range such as [1-4], or use comma such as [1, 3, 5], or use a list such as [234]

x : Represents any one digit between 0~9

Tn: Represents the last digit timeout. n represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

Example:

8[2-9]xxxxx: : All number from 8200000 to 8899999 will be sent

immediately.

955xx：5 digits numbers begin with 9 will be sent immediately.

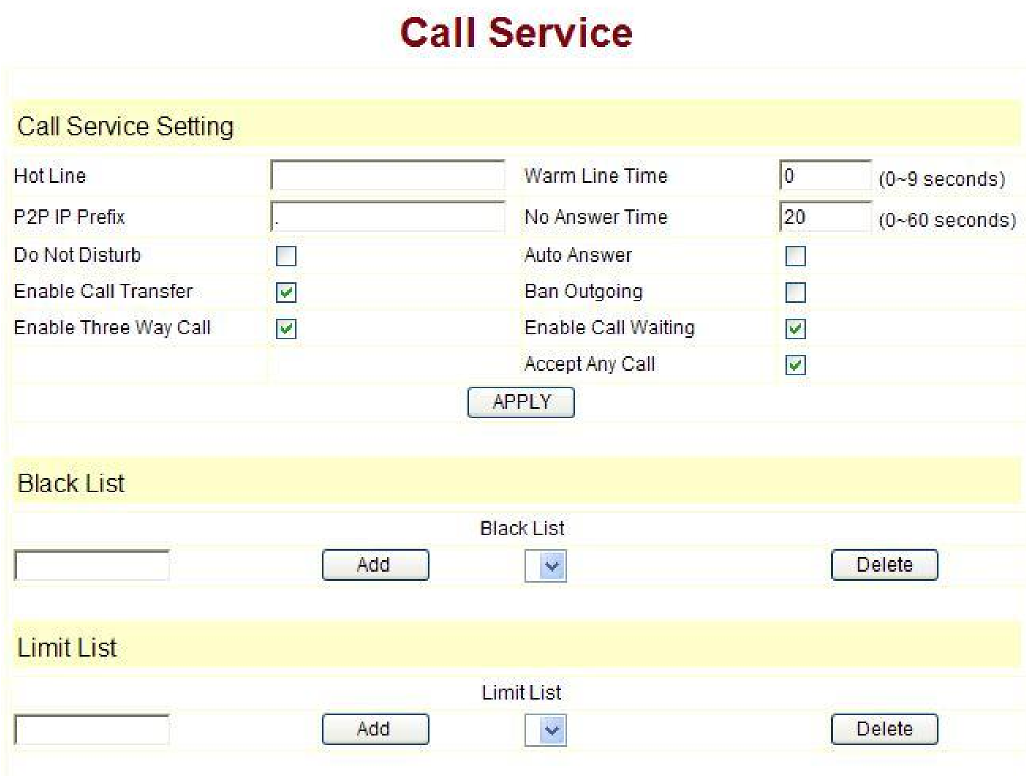
10060：Number 10060 will be sent will be immediately

22xxxxxT1：7 digits numbers begin with 22 will be sent after one second

39[3,9]xxxx：7 digits numbers begin with 393 or 399 will be sent immediately.

3.5.8 Call Service Settings

User configure the value add service such as hotline, call forward, call transfer, call waiting, 3-way conference call, auto-answer, etc in this page。



Hotline: configure hotline number.

Warm Line Time：Set waiting time for the user picking up the phone to dial hotline number, the config range is 0-9s, default is 0s.

If warm line time is 0s, then hotline number will be sent right away after off-hook;

If the range is 1-9s, take 3s as an example, hotline number will be sent immediately 3s later without pressing any key.

As long as any key is pressed within the setting time, timer will be suspended.

Auto Answer: Enable/disable auto answer function.

No Disturb: DND, do not disturb, enable this option to refuse any calls.

Ban Outgoing: Enable this to ban outgoing calls.

Enable Call Transfer: Please refer to Value_add_service for detail.

Enable Call Waiting: Enable/disable Call Waiting

Enable Three Way Call: Please refer to [Value_add_service] for detail.

Accept Any Call: If this option is disabled, E103 IP refuses the incoming call when the called number is different from E103 IP's phone number.

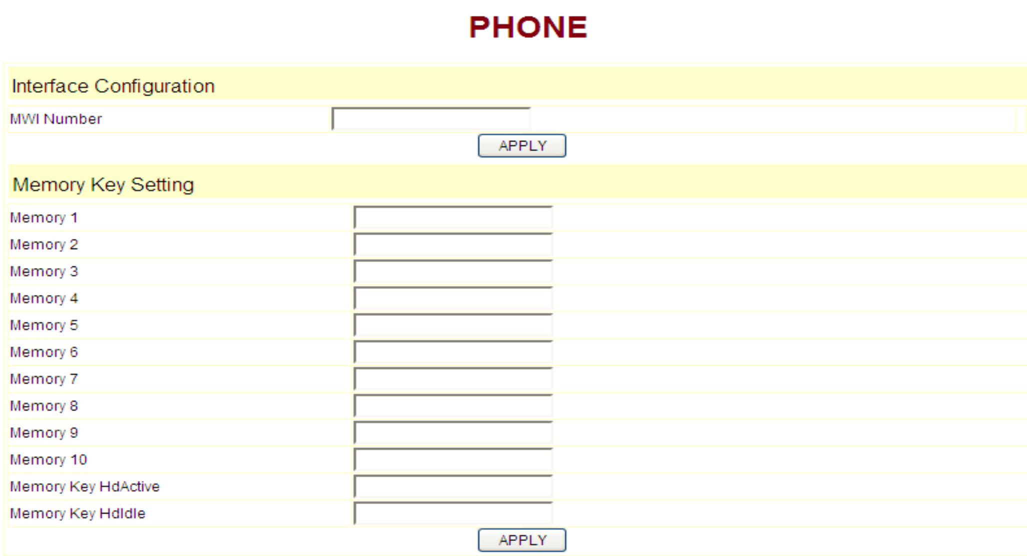
No Answer Time: no answer call forward time setting.

Black List: incoming call in these phone numbers will be refused.

Limit List: outgoing calls with these phone numbers will be refused

3.5.9 Memory Key

This page layout shows the number setting of Voice mail and speed-dial key.



MWI Number: configure the number of picking up voicemail message key.

Memory1~Memory7: configure the number of the speed dial key.

If MWI Number and Memory have disposed the number, presses down the key which corresponds on the telephone board; the telephone will dial the number automatically which the key disposed.

Memory Key HdActive: configure the number of the park key. In park1 mode, If a call is active on either line, press down "hold" key, execute SD HdActive, calling the user of this number.

Memory Key HdIdle: configure the number of the park key. In park1 mode, If no call is active and a line is available execute SD HdIdle, after pressing down "hold" key; calling the user of this number.

After call park feature is enabled, Memory Key HdActive and Memory Key HdIdle configure related parameters; you can carry out park1 function.

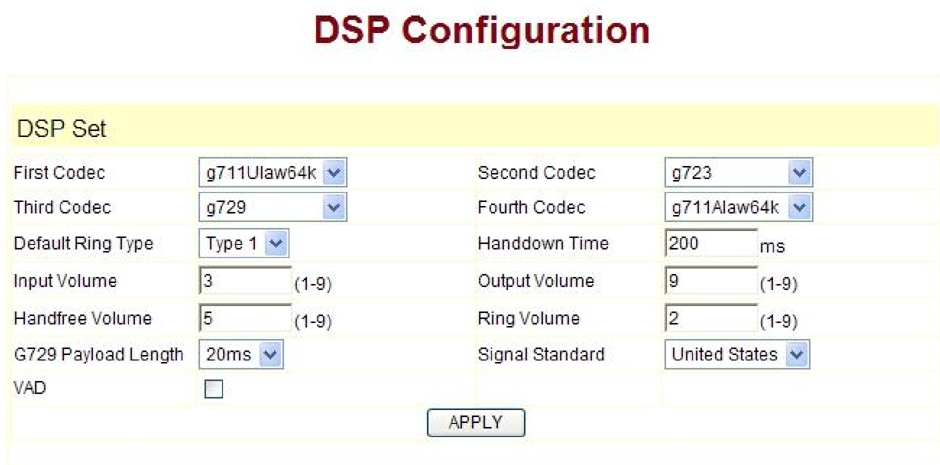
3.5.10 MMI Filter

MMI filter is used to make access limit to E103 IP phone.

When MMI filter is enable. Only IP address within the start IP and end IP can access IP phone.

3.5.11 DSP

This page mainly completes voice configuration.



CODEC: select the prefer CODEC; support ulaw, alaw, G729 and G7231 5.3/6.3

Signal Standard: Signal standard for different area.

Handdown Time: Hand down detects time.

Input Volume: Handset input volume.

Output Volume: Handset output volume.

Handfree Volume: Hand free volume

G729 Payload Length: G729 payload length

VAD: Enable/disable Voice Activity Detection

3.5.12 VPN

This page is VPN setting page, the IP phone support the VPN with UDP and L2TP protocol .The parameters is as below

VPN IP: After VPN registered successfully, VPN server will give an IP aggress

to the terminal. If there is a IP address shown on terminal (except for 0.0.0.0),it means your VPN has registered

UDP Tunnel

VPN Server Addr: Register to the address of VPN server

VPN Server Port: Register to the port of VPN server

Server Group ID: Group ID of UDP VPN

Server Area Code: Area code of VPN server

L2TP

VPN Server Addr: Register to the address of VPN server

VPN User Name: L2TP VPN username

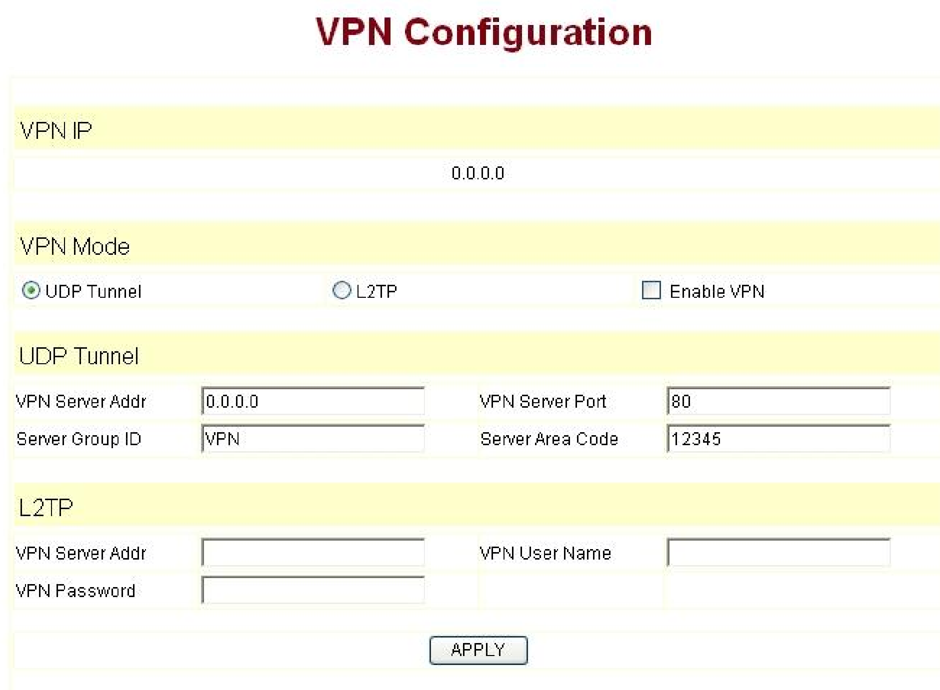
VPN Password: L2TP VPN password

UDPTunnel: use the UDP to visit VPN

L2TP: use the L2TP to visit VPN

Enable VPN: Enable the VPN server, you must choose UDP or L2TP type in advance

Notice: At the present, L2TP only support L2TP VPN server under Linux, UDP only support a private UDP VPN server.



3.6 Dial-Peer dial rule setting

Please refer to [How to use dial rule] for detail.

Dial-Peer

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
2T	255.255.255.255	5060	SIP	del	no suffix	1
3T	0.0.0.0	5060	SIP	del	no suffix	1
123	0.0.0.0	5060	SIP	all:06332221015	no suffix	0
0T	0.0.0.0	5060	SIP	rep:86	no suffix	1
11	192.168.0.11	5060	SIP	no alias	no suffix	0

Add Dial Peer

Phone Number	
Destination (optional)	
Port(optional)	
Alias(optional)	
Call Mode	SIP
Suffix(optional)	
Delete Length (optional)	

Submit

Dial Peer Option

2T Delete Modify

3.7 Config Manage

Save Config: Save current settings.

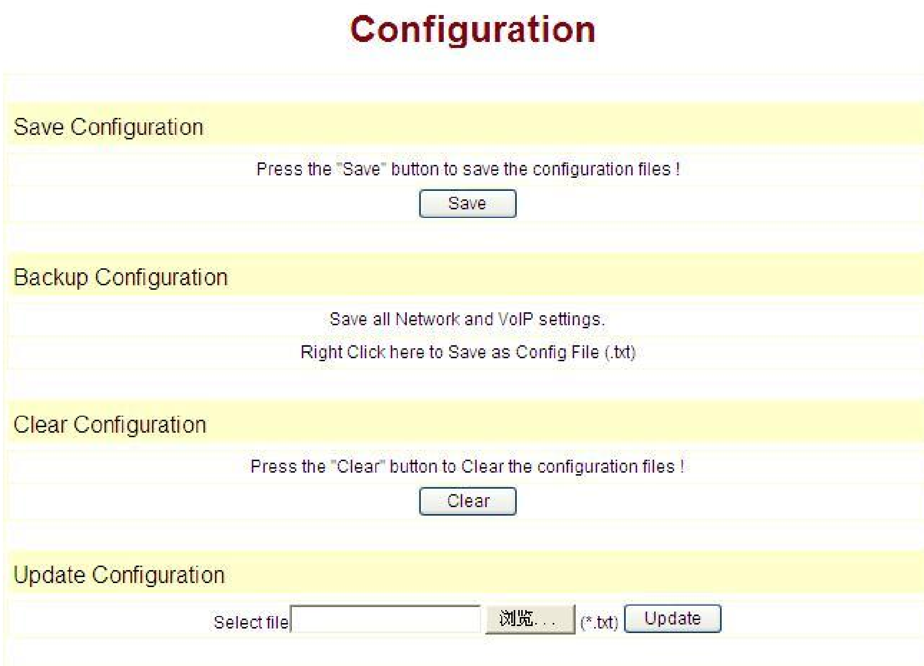
Clear Config: Restore to default settings.

Backup Config: Backup the config file, via point the right key of mouse- à save target as....- à will pop a save window, then type the config file name in the File name (the file type is text file)

Update Configuration: Update the current configuration through configuration files.

Notice:

Clearing config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except sip, advance sip restore to factory default.



3.8 Update Firmware

3.8.1 Update

Web Update：

Update the application or configuration files of the phone. The application document is .z format, and the configuration file is .cfg format.

Through clicking on the "browse" button to open the upgrade file or configuration file, then click on "Update" button. After the upgrade, the phone will automatically restart.

Notice：When upgrading, WEB page cannot be closed.

FTP Update:

Upload/download the configure file with FTP or TFTP server, or download firmware from FTP or TFTP server

Back up configure file to your FTP/TFTP server.

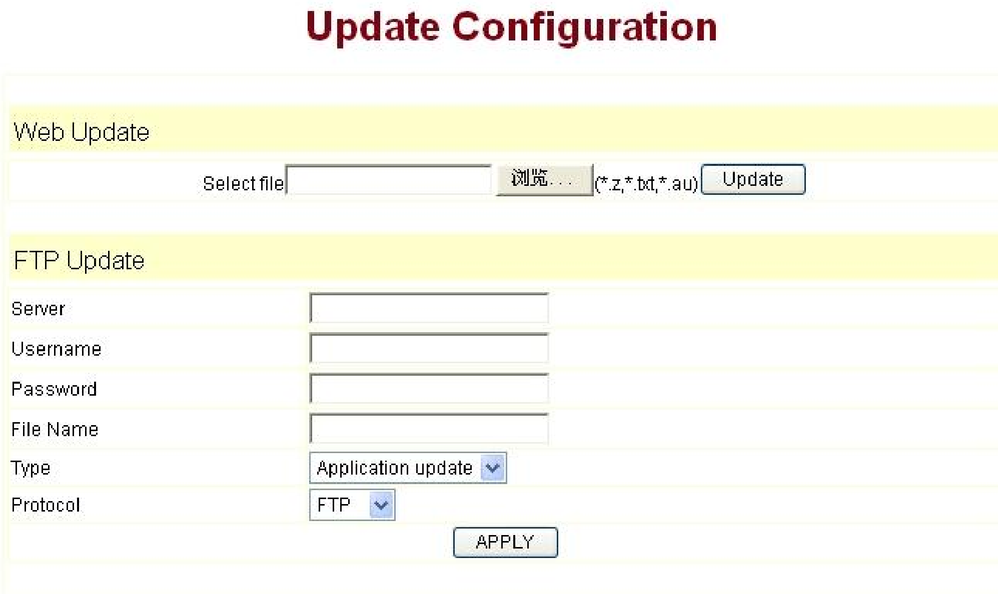
Configure use .cfg extension.

The Type includes two parts of config file export and config file import

Config file export: export the config file

Config file import: import the config file

The phone phone support FTP and TFTP auto update, the gateway will auto obtain the configure file from your update server if configured. To obtain the original configure file, you can use the FTP/TFTP back up as describe above. Configure file using module structure, user may remain the concerned modules and remove other modules. Put the configure file in the root directory of update server when finish editing.



3.8.2 Auto Update

Current Version: the system will display the current version number

Server Address: FTP/TFTP server address

Username: FTP server user name

Password: FTP server password

Config File Name: The name of configuration file

Config Encrypt Key: The encrypt key of confirmation file

Protocol Type: The protocol type that used for upgrading

Update Interval Time: The interval time that the terminals search for new configuration file.

Update Mode: auto provision mode; Disable: not auto update, Update after reboot: auto update after reboot, Update at time interval: auto update after a certain time

Configure file version was in the <<VOIP CONFIG FILE>> Version 1.0007 and <GLOBLE CONFIG MODULE> CongFile Version

For instance:

Gateway original version is:

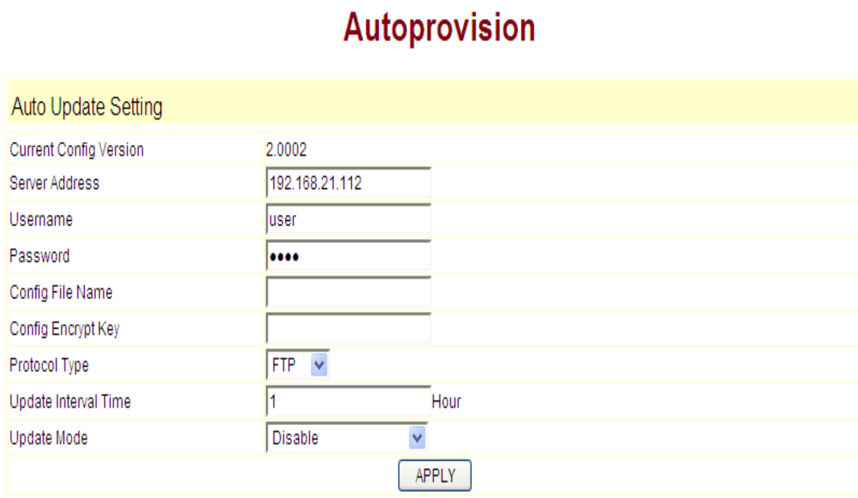
<<VOIP CONFIG FILE>>Version: 1.0000

<GLOBLE CONFIG MODULE> CongFile Version: 6

User may edit the configure file version to:

<<VOIP CONFIG FILE>>Version: 1.0007

<GLOBLE CONFIG MODULE> CongFile Version: 7



3.9 System Manage

3.9.1 Account Manage

Set web access account or keypad password of E103 IP.

3.9.2 Syslog config

Set the system log

Server IP: set the syslog server address

Server Port: set the syslog server port

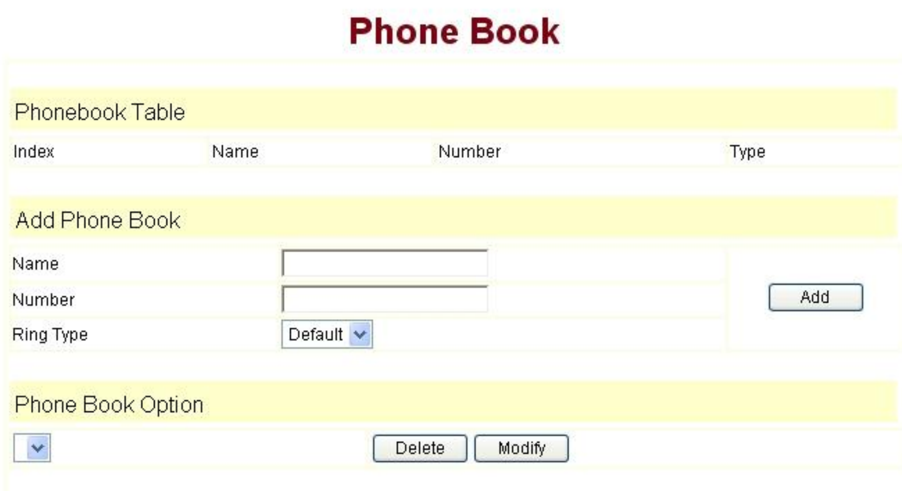
MGR Log Level: set the MGR log level

SIP Log Level: set the SIP log level

IAX2 Log Level: set the IAX2 log level

Please click "apply" after setting

3.9.3 Phone Book



3.9.4 Time Set

This page layout is the setting of time of phone.

Server: type the IP address of time server

Timezone: select correct time zone in list box

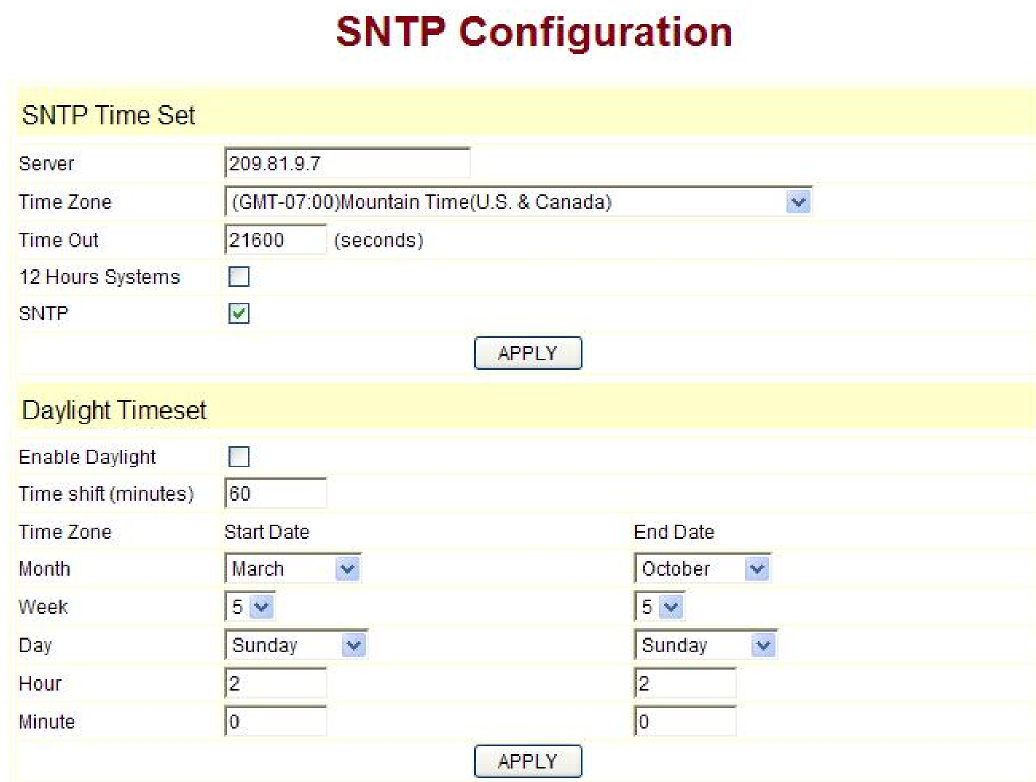
Timeout: longest response time for SNTP

Daylight: daylight saving time

SNTP: select SNTP server

12 Hours Format: select 12 hours format

Manual Config: time setting



3.9.5 MMI SET

Set the greeting information on LCD.

3.9.6 Logout & Reboot

Logout：Exit the Web entry.

Reboot Phone：Logout the entry, and reboot the phone. When user modifies any config of the phone, it will take effect after being rebooted, you can enter into this layout and click "Reboot". And the phone will be rebooted automatically.

Note：Reboot IP phone, some setting needs to reboot to make it works. Please always save configuration before reboot, otherwise the setting will return to previous setting.

4. Operating Method for Dialing

4.1 How to dial IP Phone

You can make a call after being made a proper setting on your phone. Please confirm whether all the net wires are connected correctly.

If you want to make a call, you can make it after dialing the number and then pressing "#".

You can find IP address through the menu.

Modify the IP address of the computer, and making it the same net as the phone.

Inputting the IP address of E103 IP in the browser, and then you can visit the setting layout of E103 IP after pressing the Enter key; super user account is admin/admin; common user account is guest/guest.

4.2 Set the phone being connected to server

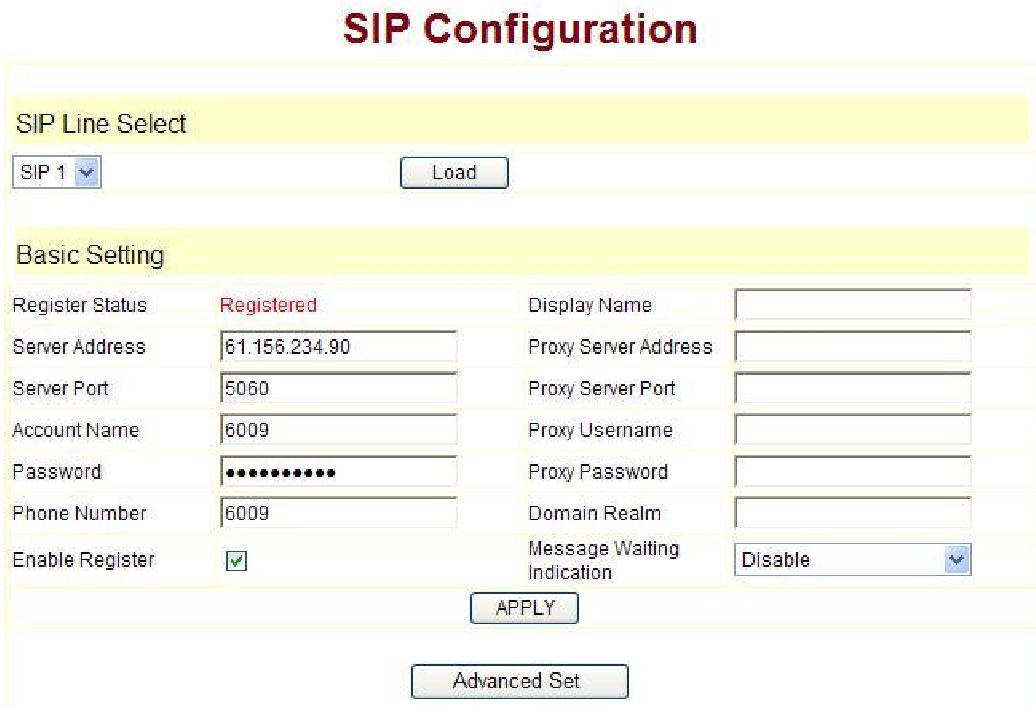
4.2.1 Set the WAN interface

The connection ways of entering the Network à WAN Config layout phone of the net port:

E103-IP could be connected to Internet by using the static IP, DHCP IP, or PPPoE dialing.

As for the specific configuration please see 3.3.1 WAN Config.

4.2.2 SIP setting：



Enter into the *VoIP à SIP Config* to set the layout config and sip account information:

Register Server Addr: Register address of public SIP server

Register Server Port: Register port of public SIP server，default port is 5060.

Register Username: Username of your SIP account (Always the same as the phone number)

Register Password: Password of your SIP account.

Phone Number: Phone number of your SIP account

----choose Enable Register；

You can dial VoIP phone when the WAN interface and IAX protocol are being set correctly.

Message Waiting Indication：Set Disable/Enable message waiting through Pull-down menu, including:

Disable：MWI is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt.；

Enable (Subscribe): MWI is enabled and SUBSCRIBE will be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will blink to prompt.

Enable(No Subscribe)：MWI is enabled, but phone will not send SUBSCRIBE, if the server sends NOTIFY message indicating new voicemail received, MWI LED will also blink to prompt.

SIP1 Subscribe will still be used in config file: config option 〝0〞means 〝Disable〞, 〝1〞means 〝Enable (Subscribe)〞, 〝2〞means 〝Enable (No Subscribe)〞.

4.3 How to use the dial rule?

E103-IP provide flexible dial rule, with different dial-rule configure, user can easily implement the following function:

----Replace, delete or add prefix of the dial number.

----Make direct IP to IP call

----Place the call to different servers according the prefix.

You can click "Add" to add a new dial rule. Below is the detail setting of the dial-rule:

Phone Number: The Number suit for this dial rule, can be set as full match or prefix match. Full match means that if the number user dialed is completely the same as this number, the call will use this dial-rule. Prefix match means that if prefix of the number that the user dials is the same as the prefix, the call will use this dial-rule, to distinguish from the full match case; you need to add "T" after the prefix number in the phone number setting.

Call Mode: support SIP.

Destination (optional): call destination, can be IP or domain. Default is 0.0.0.0; in this case the call will be routed to the Public SIP server. If you set the destination to 255.255.255.255, then the call will be routed to the private SIP server. Also you can key other address here to make direct IP calls

Port (optional): Configure the port of the destination, default is 5060 in SIP

Alias (optional): Set up the Alias. We support four Aliases as below. Alias need to co-work with the *Del Length*:

add:xxx, add prefix to the phone number, can set to reduce the dial length.

All: xxx, replace the phone number with the xxx, can use as speed dial function.

Del: delete the first N numbers. N is set in the *Del Length*

rep:xxx，replace the first N numbers. N is set in the Del Length. For Example: User wants to place a call 86633-8215555, then can set the *phone number* in the dial rule as 0633T, and set the *Alias* as rep: 86633, and set the *Del Length* to 4. Then all calls begin with 0633 will be changed to 86633 xxxxxxxx.

Suffix (optional): Configure suffix, show no suffix if not set

Instance:

2T rule: If the call starts with 2, the first 2 will be deleted, and the rest number will be sent to private SIP server.

3T rule: If the call starts with 3, the first 3 will be deleted, and the rest number with be sent to public SIP server.

123 rule: Dial 123 and will send 06332221015 to your server. It is used as speed dial function.

0T rule: If the call begins with 0, the first 0 will be replaced by 86. It means that if you dial 06332221015 and AG-188 will send 866332221015 to your server.

11 rule: when you dial 11, the call with send to 192.168.0.11, suit for LAN application without set up a sip server.

4.4 Voice mail

When there are new messages, the MWI will give out prompt according to the options of "Message Waiting Indication" in web page layout.

If "Disable" is selected, MWI function is disabled, even if received NOTIFY message from the server indicating new voicemail, phone will not prompt;

If "Enable (Subscribe)" is selected, MWI function is enabled and Subscribe will be sent, if the server sends NOTIFY message indicating new voicemail received, MWI LED will blink to prompt;

If "Enable (No Subscribe)" is selected, MWI is enabled, but Subscribe will not be sent, if the server sends NOTIFY message indicating new voicemail received,

MWI LED will also blink to prompt. After the voicemails are picked-up, MWI LED will stop blinking.

IC Warning:

This device complies with Industry Canada license-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 under operation of the device.

Privacy of communications may not be ensured when using this telephone.

Under Industry Canada regulations, this radio transmitter may only operate using an antenna of a type and maximum (or lesser) gain approved for the transmitter by Industry Canada. To reduce potential radio interference to other users, the antenna type and its gain should be so chosen that the equivalent isotropically radiated power (e.i.r.p.) is not more than that necessary for successful communication.

FCC Caution:
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:

The base complies with FCC radiation exposure limits set forth for uncontrolled environment.

The base should be installed and operated with minimum distance 20cm between the radiator& your body.

The handset compliances with the RF Exposure Evaluation.

This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

---Reorient or relocate the receiving antenna.

---Increase the separation between the equipment and receiver.

---Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

---Consult the dealer or an experienced radio/TV technician for help.