IP 1 SIP Gateway User Manual

1． IP 1 SIP Gateway Features

1.1 Product Appearance

1. TEL: telephone status, the light is led ,when pick up the phone the light off
2. WAN: WAN port status ,when the network is ok the light will flash
3. SYS: Not registered in sip sever ,the light off， Registration lamp blinks registered, light off
4. PWR: power status led.

1.2 Backside Illustration

1. TEL: telephone interface
2. WAN: 10M/100M auto-negotiation, factory default is set to DHCP.
3. POWER: Power Jack, 9~12V,800mA
4. Reset: Please refer to "restore_to_factory_default" for the usage of this button

1.3 Software

- Configured by HTTP web browser
- Support HTTP, TFTP upgrade.
- Support major G.7XX (G711,G729,G723,G726) Codec
- Dynamic voice jitter buffer, CNG (Comfort noise generation),VAD
- G.165 compliant 16ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- E.164 dial plan and customized dial rules
- Support T38 FAX
- Support adjustable user password and super password
- IVR (Interactive Voice Response)

1.4 Protocol and standard

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX;
- SIP RFC3261
- TCP/IP: Internet transfer and control protocol

- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth
- TFTP: File Transfer protocol
- HTTP: Hyper Text Transfer protocol
- HTTPS: Secure Hypertext Transfer Protocol

1.5 Interface features

- WAN: 10M/100M auto-negotiation
- LAN: 10M/100M auto-negotiation
- FXS ports:
  Line Feed Voltage: >=42V
  Ring Voltage: >=45V.
  Ring Current: >=30mA

1.6 Electric requirements

- Voltage: 9V ~ 24V
- Power adapter: output DC 12V/450 mA

1.7 Operating requirement

- Operation temperature: 0 to 40°C (32°to 104℉)
- Storage temperature: -30°to 65°C (-22°to 149℉)
- Humidity: 10 to 90% no dew

1.8 Certificate:

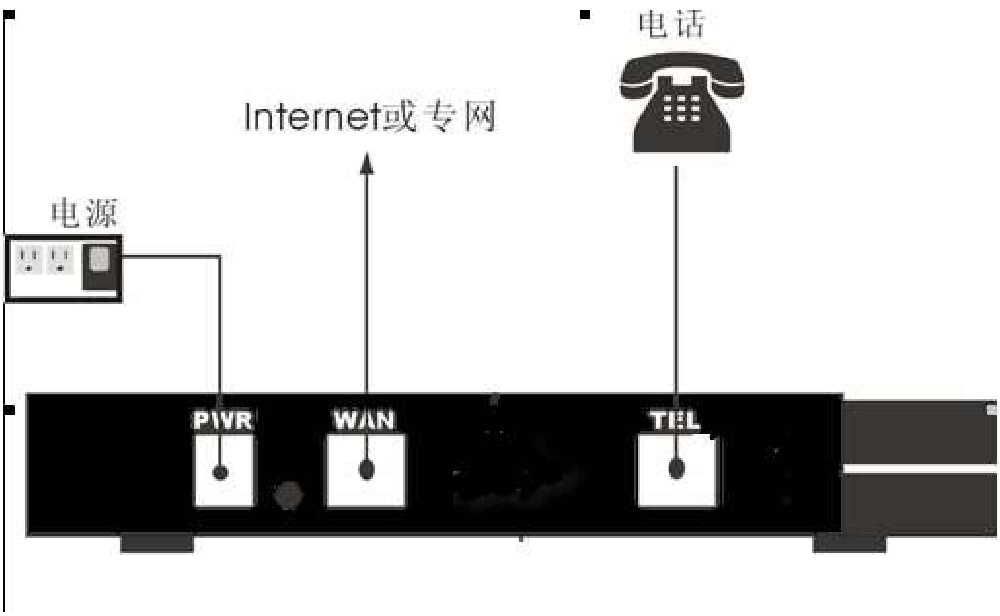
- CE, FCC Part15, RoHS

1.9 Packaging

- **Size: 22.5cm×17.0cm×7.6cm**
- **Packing list**
  - ✓ AG - 110 gateway X 1
  - ✓ Power adapter X 1
  - ✓ CD X 1

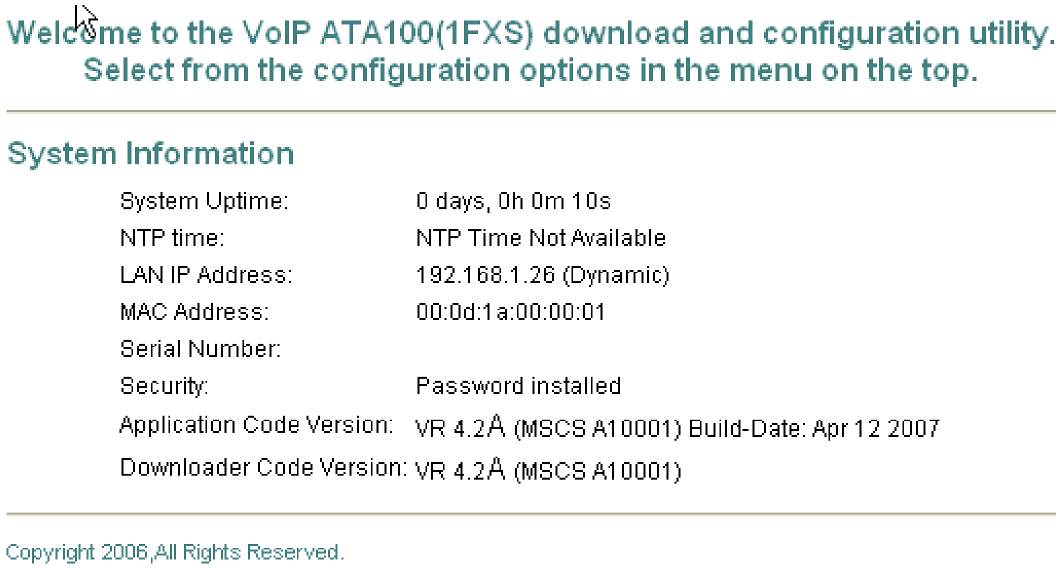
1.10 Installations

Connect the IP 1 LAN port and you computer with the RJ45 cable, and then change your computer IP to static 192.168.1.xxx or use dynamic obtain IP, type 192.168.1.1 in your IE browser to access IP 1 and change its setting.



2． Settings

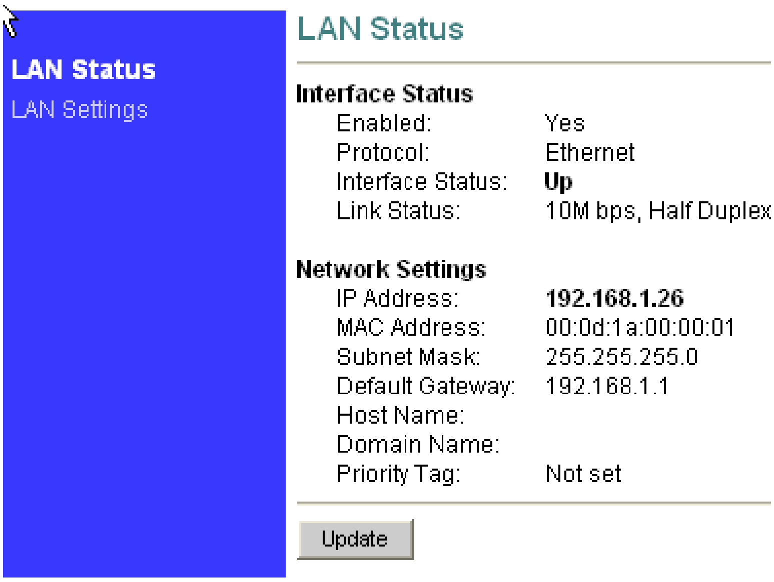
2.1 Home



1. System uptime: system running time
2. NTP time: NTP time and date
3. LAN IP Address: Gateway IP, factory default is192.168.1.1
4. MAC Address: Gateway MAC address，the same as LAN port MAC address
5. Serial Number: device serial number
6. Security: password information
7. Application Code Version: firmware version
8. Downloader Code Version: downloader version

2.2 LAN

2.2.1 LAN status



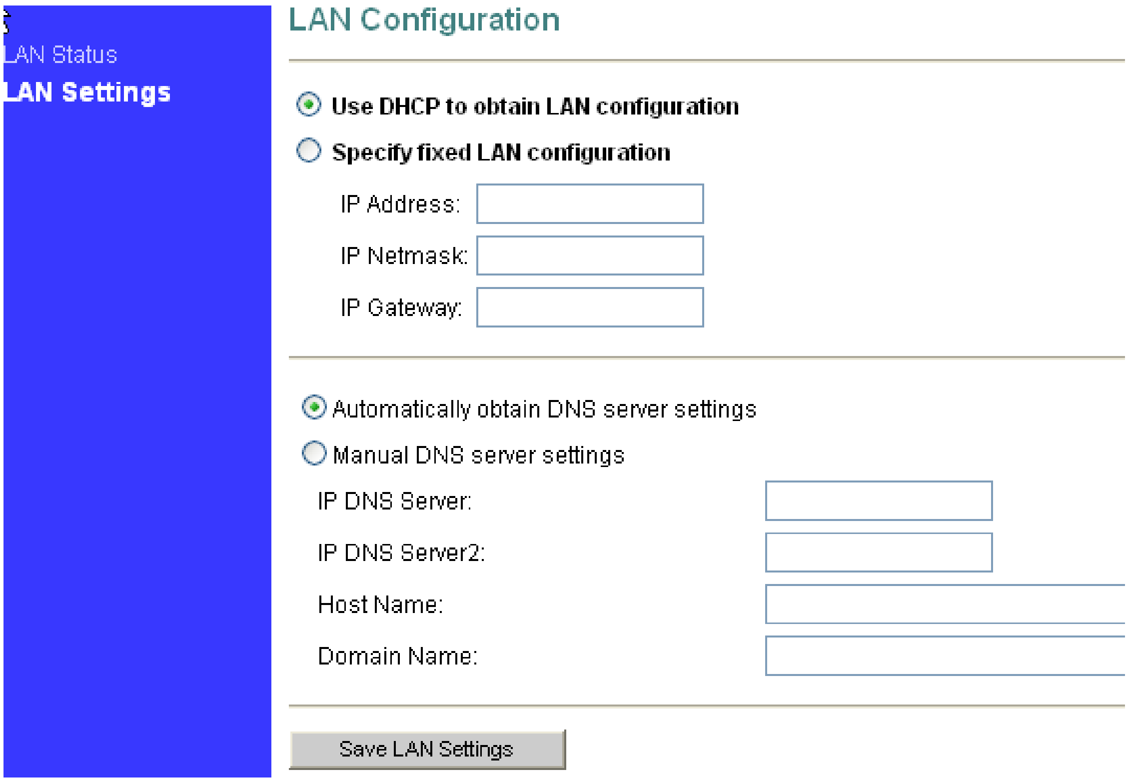
Interface Status：

1. Enable: Yes indicate LAN id ready for use
2. Protocol： Ethernet
3. Interface status: UP or Down
4. Link Status: link speed mode

Network Settings：

5. IP address: LAN port IP address,factory default is 192.168.0.1
6. MAC Address: LAN port MAC address
7. Subnet Mask: LAN port subnet mask
8. Default Gateway: default gate way IP
9. Domain Name: Domain
10. Priority Tag: Priority Tag value encoded in the Ethernet header in outgoing packets.

2.2.2 LAN settings



1、Use DHCP to obtain LAN configuration:use DHCP to obtain ip address for Ethernet

2、Specify fixed LAN configuration: fix ip address

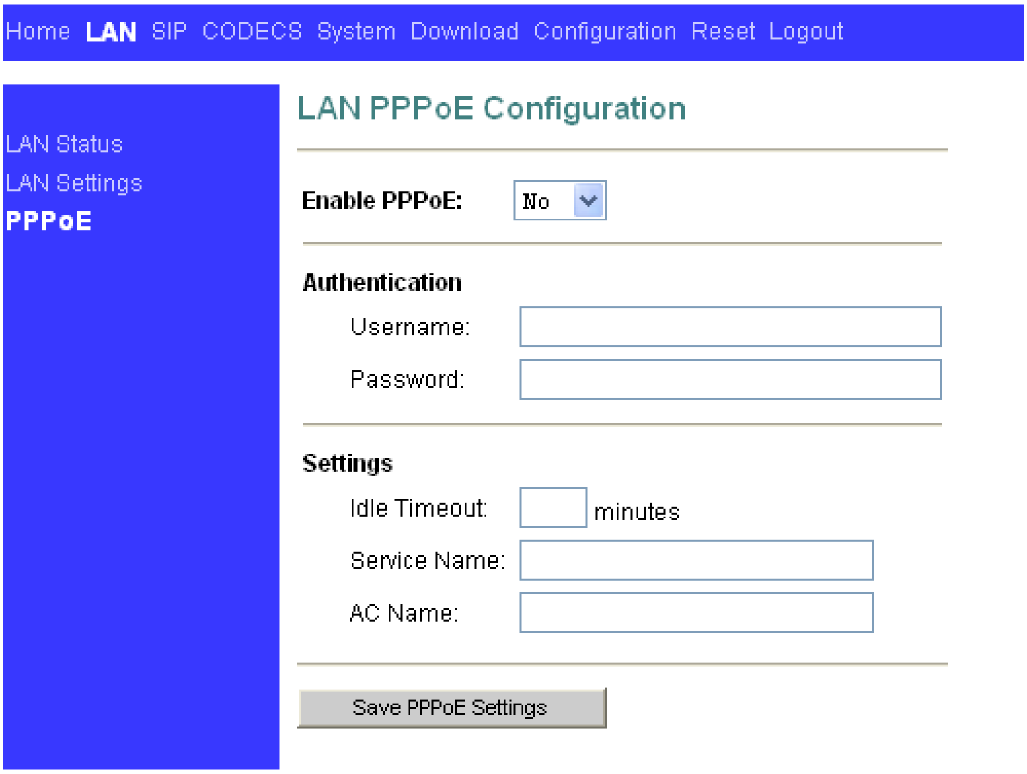
IP Address: ip address of ethernet
Subnet Mask: mask
IP Gateway: gateway

3、Automatically obtain DNS server settings: Auto obtain DNS

4、Manual DNS server settings: setting DNS server by yourself

IP DNS Server : public DNS Server
IP DNS Server2: private DNS Server

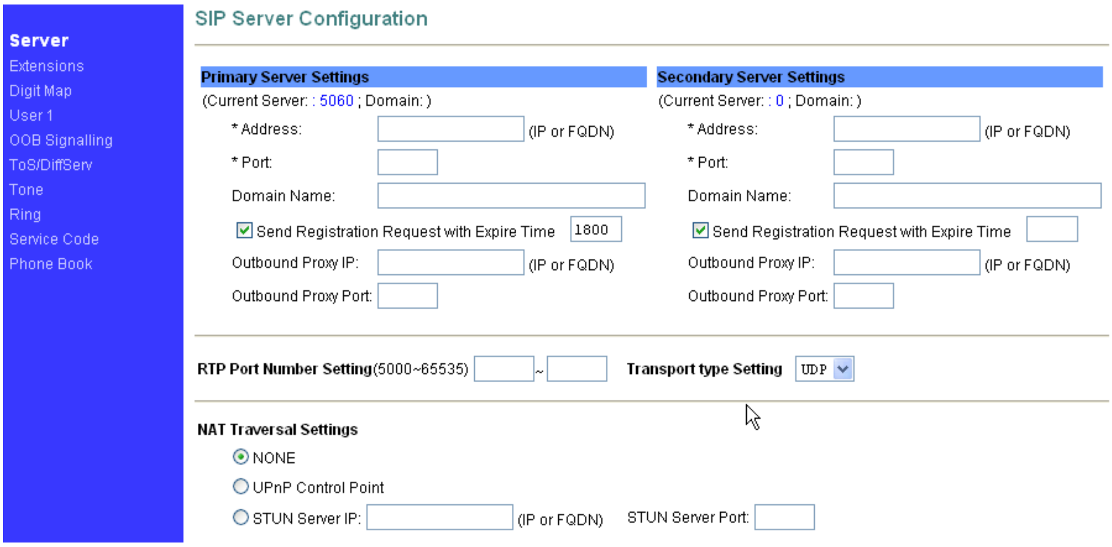
2.2.3 PPPoE setting



1. Enable PPPoE: use PPPoE to connect to the internet
2. Username&Password: PPP id and PPP pin from your ISP
3. Echo Timeout: The duration between PPP echo requests sending to server.
4. Echo Count: The number of unanswered PPP echo requests before PPP connection is closed.

2.3 SIP

2.3.1 SIP server Configuration



1. Primary Server: Primary Server, IP 1 will auto switch to Secondary server if primary server is unavailable.
2. Secondary Server: secondary server (back up function)
3. Address: SIP server IP address
4. Port: SIP server port , the well know port is 5060
5. Domain Name: server domain
6. Send Registration Request with Expire Time : Register TTL (unit: seconds).

Indicate the register period, if IP 1 always log off after some time , please set this time to a lower value.

7. Outbound Proxy IP: Outbound Proxy server IP address
8. Outbound Proxy Port: Outbound Proxy server port

RTP port Number Setting: RTP local port, the minimum value is 4

NAT Traversal

1. NONE: disable NAT traversal
2. UPnP: use UPnP , need the support of upper gateway
3. Stun Server IP: Fill in your stun server IP when using stun method
4. Stun Server Port: Fill in stun server port

Gateway Settings

1. Dial Plan: please refer to dial_plan

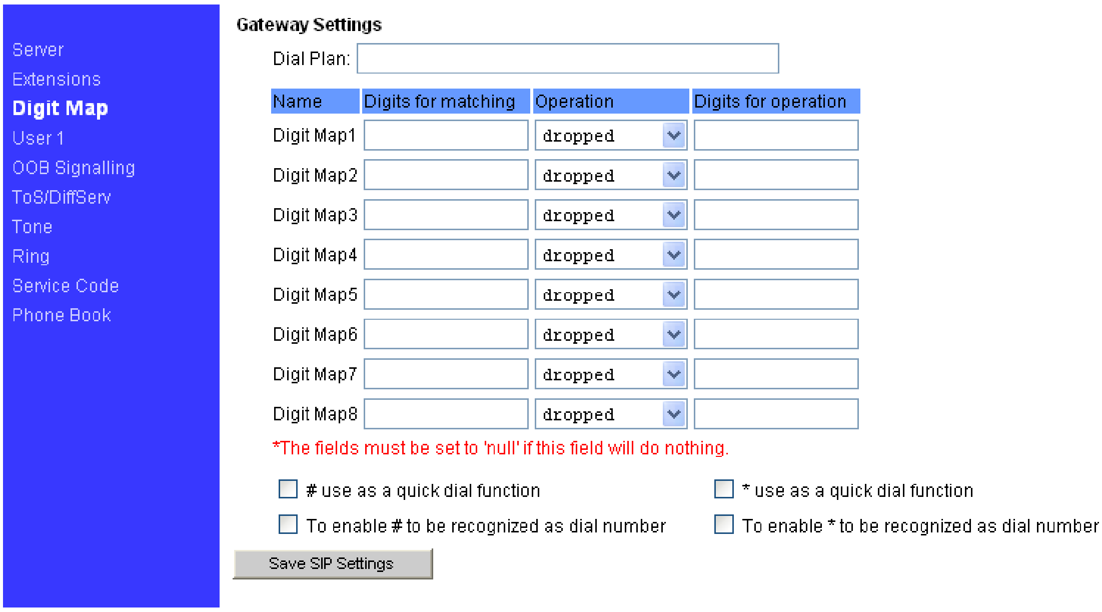
2. # use as a quick dial function: the number will send immediately after you press the # button
3. To enable # to be recognized as dial number: use # as a dial number
4. * use as a quick dial function: the number will send immediately after you press the * button
5. To enable * to be recognized as dial number: use * as a dial number

2.3.2 Extensions

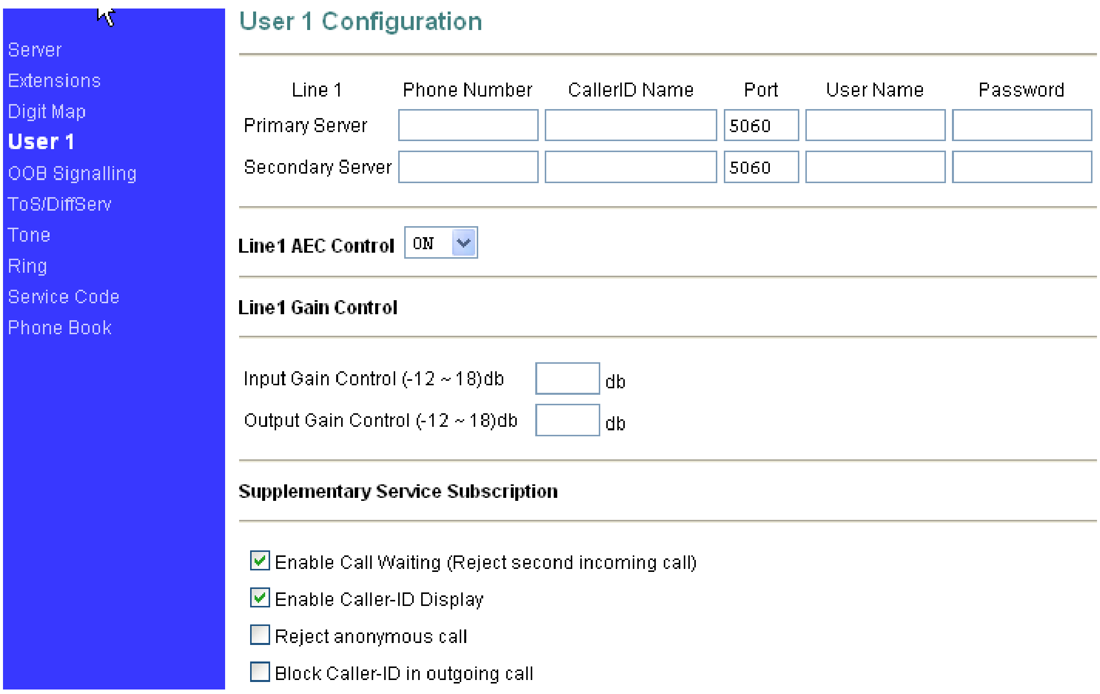
SIP Extensions:

1. Support PRACK method with provisional response reliability: enable SIP PRACK support
2. Encode SIP URI with user parameter: encode user=phone parameter in SIP URI
3. Session Timer use UPDATE method: enable SIP session timer function。
4. Enable Global Number support(E.164): enable E.164 support。

5. Call Hold using c=0.0.0.0 (RFC 2543) in SDP:using the call hold method described in RFC 2543. If unchecked, the call hold would follow RFC 3263 method
6. Send NOTIFY for REFER request: send out NOTIFY request to transfer for unattended and attended call transfer.



2.3.3 User1 Configuration(User2 is the same as User1)



Primary Server，Secondary Server

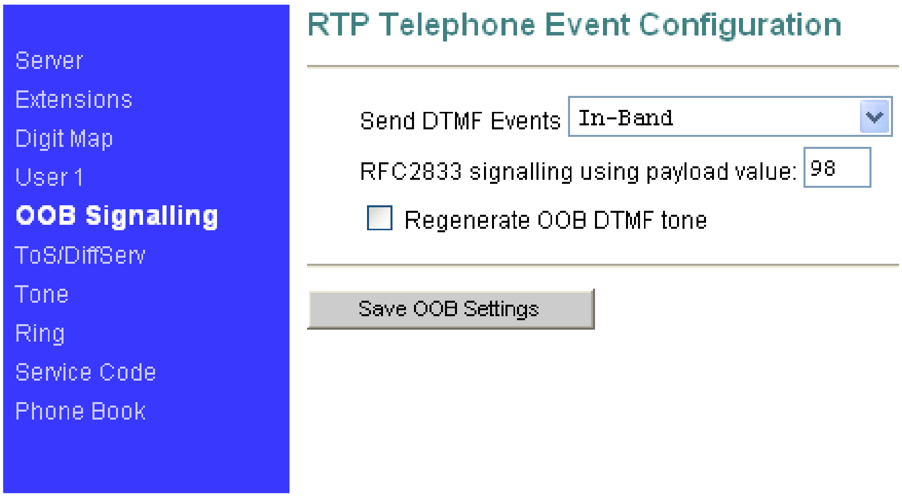
Phone Number: phone number.
CallerID Name: caller ID
Port Name: Local register port. (Note: please assign different port to different user)
User Name: user name.
Password: password.

Line1 AEC Control: enable AEC (Acoustic Echo Cancellation) function, if the other hear a significant echo, please check this option.

Line1 Gain Control:
Input Gain Control (-12 ~ 18) db: input volume control.
Output Gain Control (-12 ~ 18) db: output volume control.

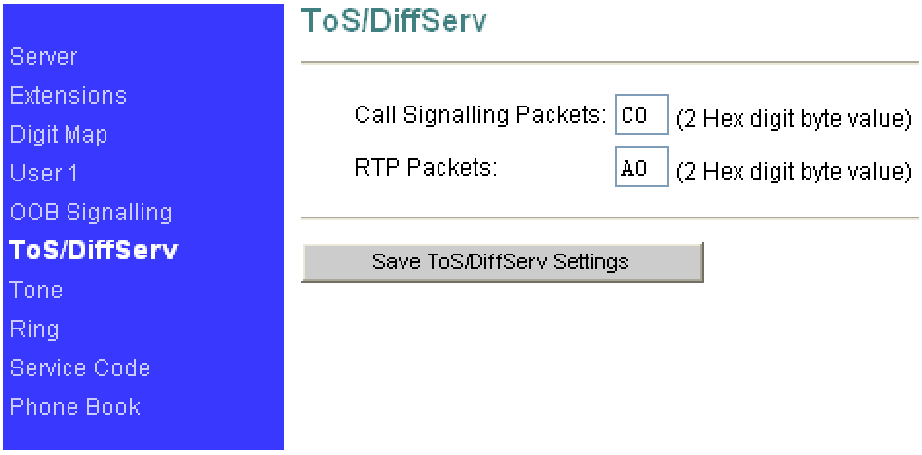
Supplementary Service Subscription:
Enable Call Waiting (Reject second incoming call): enable call waiting.
Enable Caller ID: enable caller ID display.
Reject anonymous: reject anonymous call.
Block Caller ID in outgoing call: use anonymous Caller ID.
Distinctive Ring Settings: set distinctive ring to different user.
Speed Dial Setting: speed dial number setting.

2.3.4 OOB Signalling



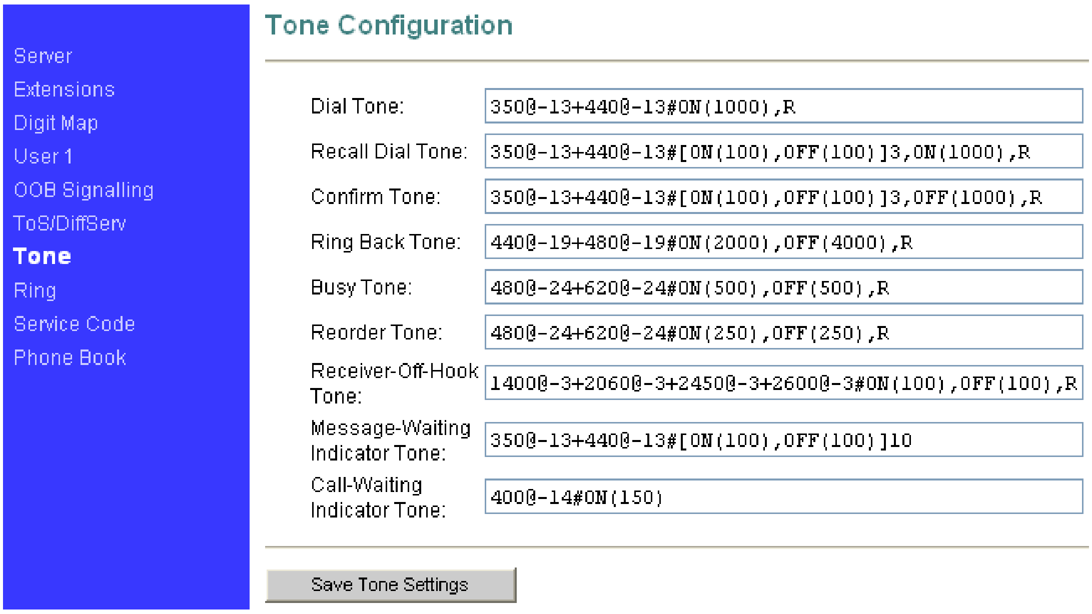
This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling. For additional information please refer RFC2833.

2.3.5 ToS/DiffServ



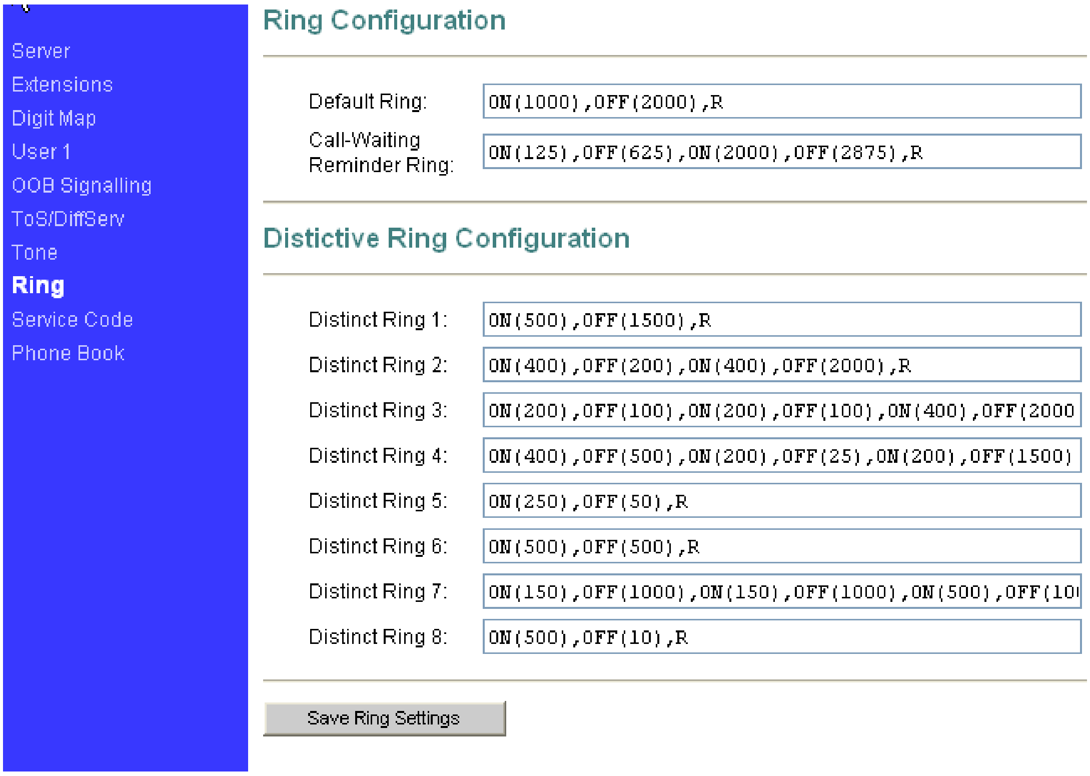
This sub-page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter “00” or leave the setting empty.
Press “Save ToS/DiffServ Settings” to save these new settings.

2.3.6 Tone



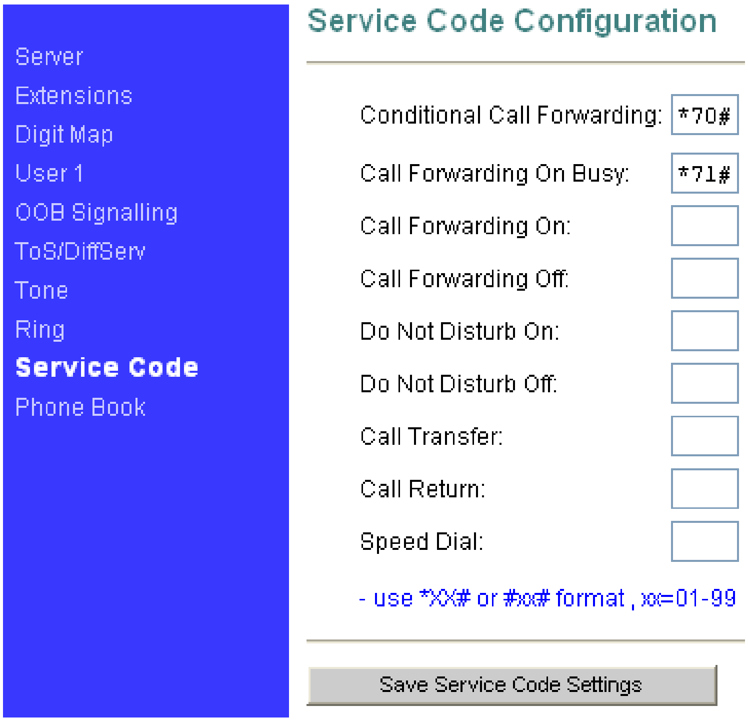
Set IP 1 ring tone for different region.

2.3.7 Ring



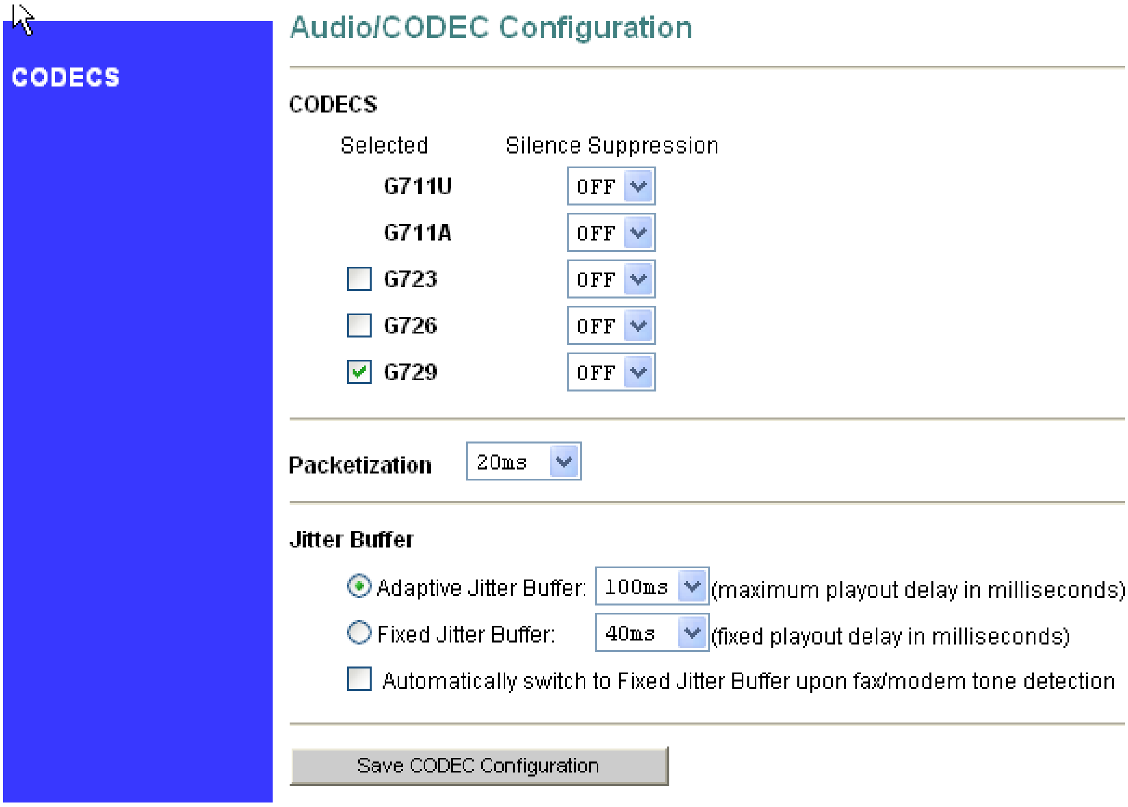
Set different user’s ring tone, co-work with “SIP→User→Distinctive Ring Settings”

2.3.8 Service Code



Please refer to value_add_service for the use of service code.

2.4 CODECS



CODECS:

Support CODEC: G711U、G711A、G723、G726、G729。

Silence Suppression: enable VAD.

Packetization:

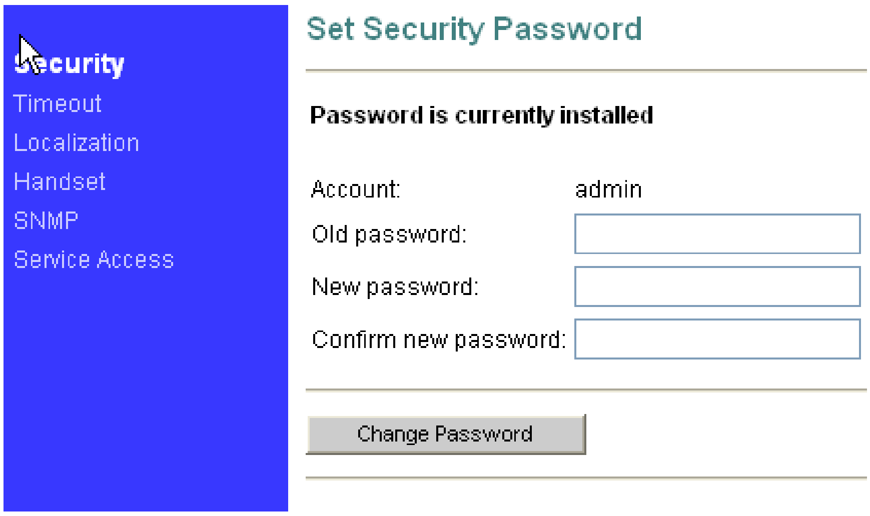
Configure the packet sending increments

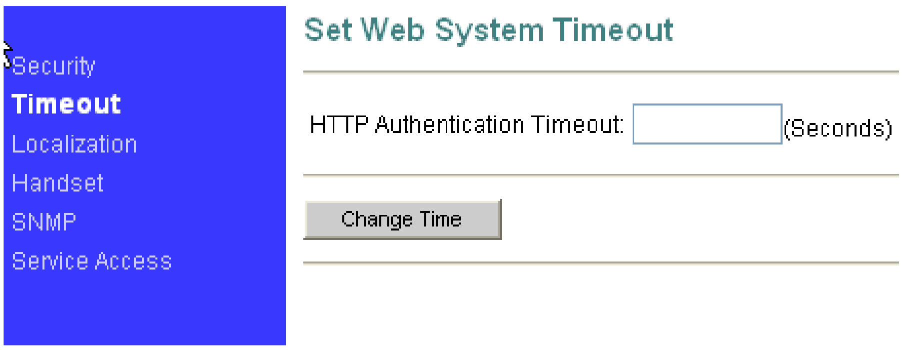
Jitter Buffer

configure the timing of the voice buffering.
Selection between adaptive or fixed jitter buffer. Default = ADAPTIVE
Set the adaptive jitter buffer maximum playout delay. Default = 100ms
or Fixed jitter buffer playout delay. Default = 40ms
Whether or not to automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection
Click on “Save CODEC Configuration” to save the configurations made.

2.5 System

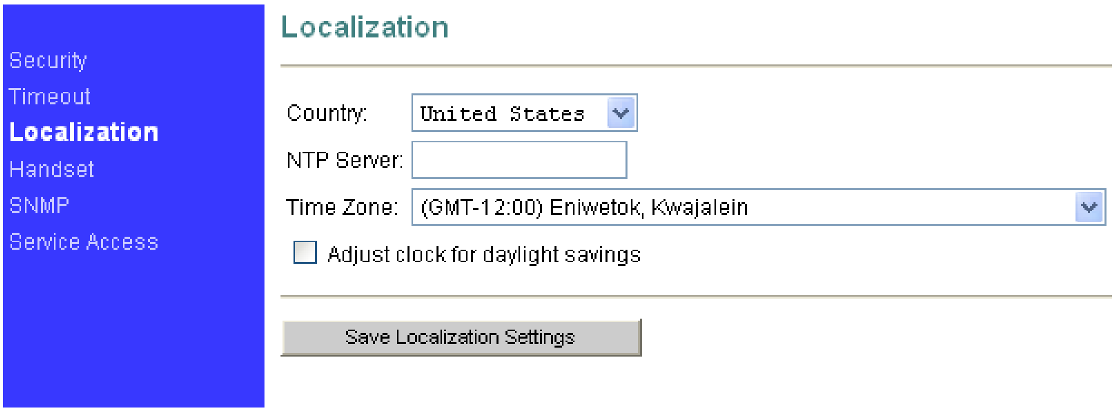
2.5.1 Security,Timeout





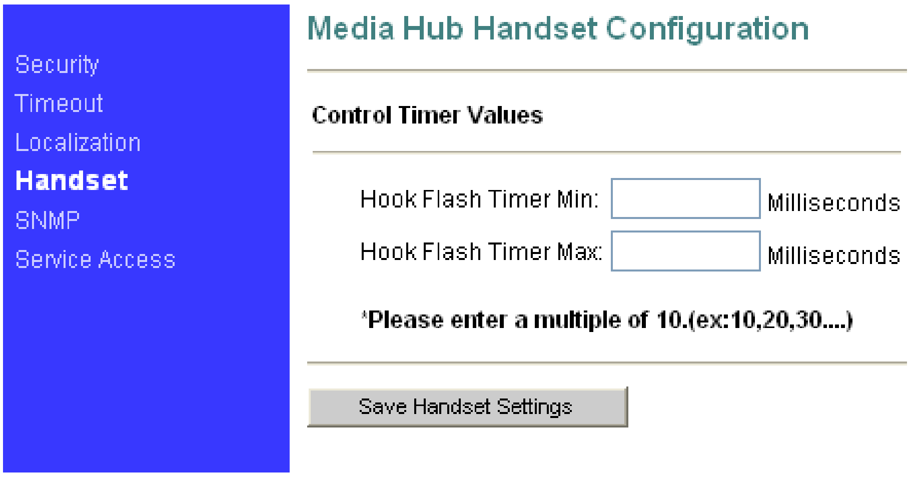
Setting web security and authentication timeout

2.5.2 Localization



Choose the correct country for a proper impedance match, as well as the NTP Server, and Time Zone. Check the “Adjust clock for daylight savings”, when applicable.
Click on “Save Localization Settings”, to save your configurations.

2.5.3 Handset



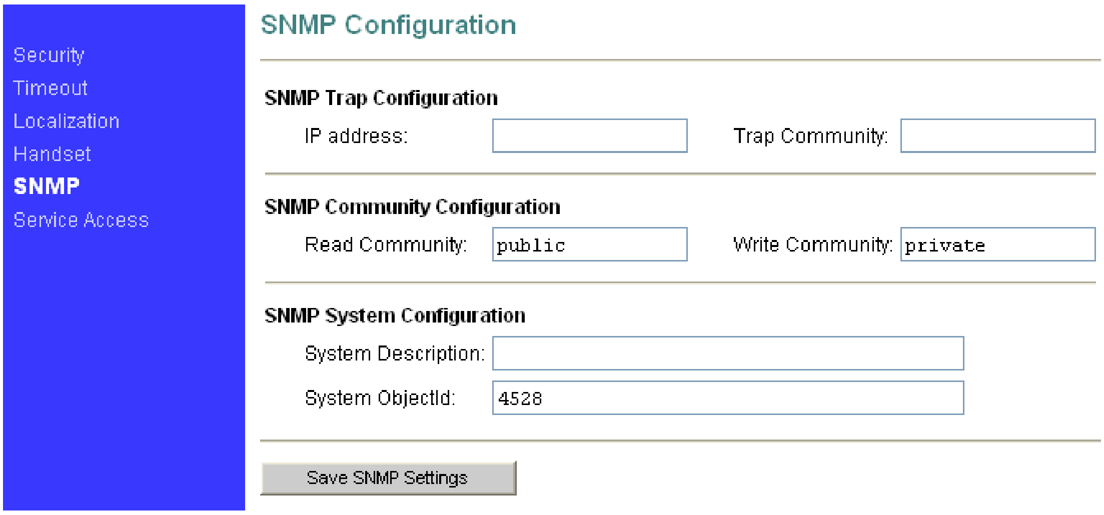
Hook Flash timing setting

Hook Flash Timer Min: minimum available time，unit: ms。

Hook Flash Timer Max: maximum available time，unit: ms。

When you press the flash during the time range you set, your action will act as hold function, and otherwise it will act as the hang up function. In some application, user wants to call another people immediately after he put down the handset, he can set the hook flash timer during the range: 10ms~20ms

2.5.4 SNMP Configuration



SNMP Trap Configuration

IP address:	Trap host IP address
Trap Community:	The community name used by the SNMP manager to verify traps. The default value is 'public'

SNMP Community Configuration

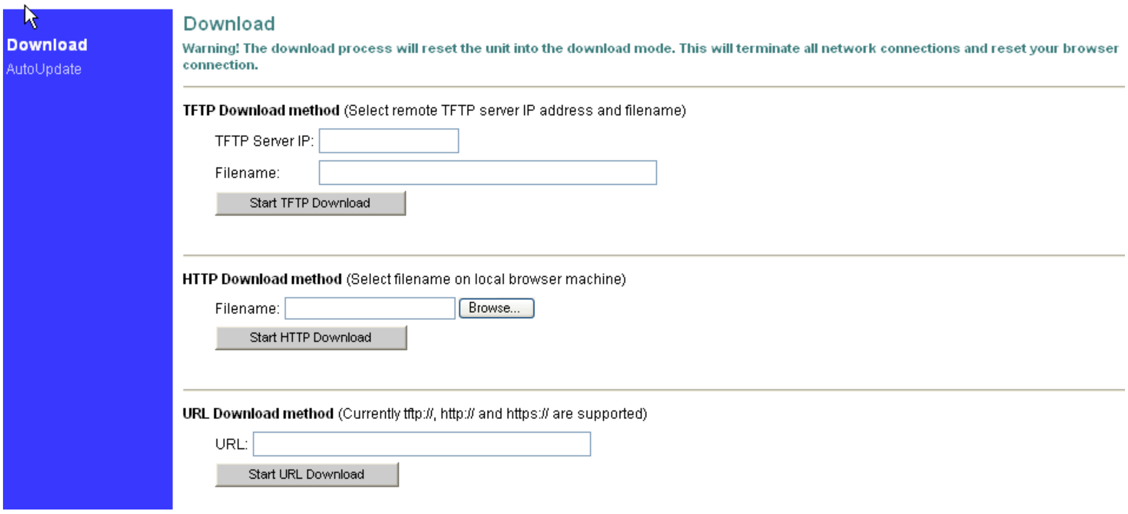
Read Community:	The community name used by the SNMP manager when reading SNMP data items from a client MIB. The default value is 'public'
Write Community:	The community name used by the SNMP manager when setting SNMP data items in a client's MIB. The default value is 'public'

SNMP Community Configuration

System Description:	Description of the unit (e.g. "John's phone")
System Object Id:	A vendor's enterprise ID

2.6 Download

2.6.1 Download

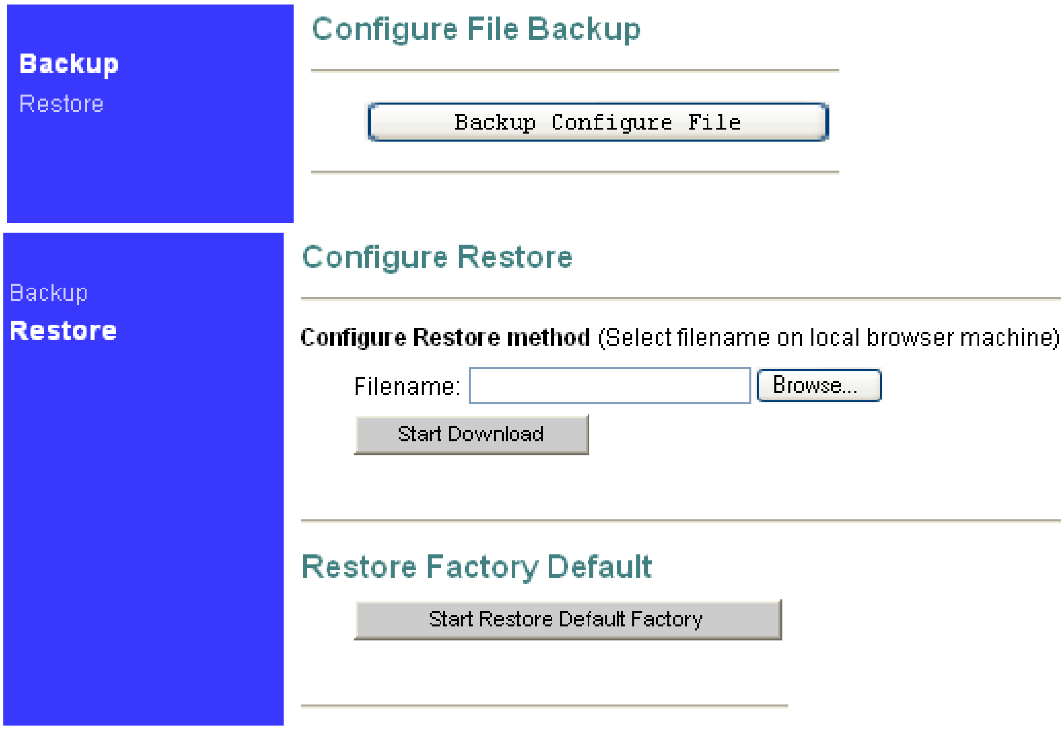


For both HTTP and TFTP methods, the device will reboot itself into the downloader mode if the main application is executing, and proceed with the ROM file download and permanent write of the application to the device's flash memory. After the download is completed, the download status page will be displayed.

Note: The available upgrade firmware is in .r0 extension; make sure you use the correct firmware before you update the firmware.

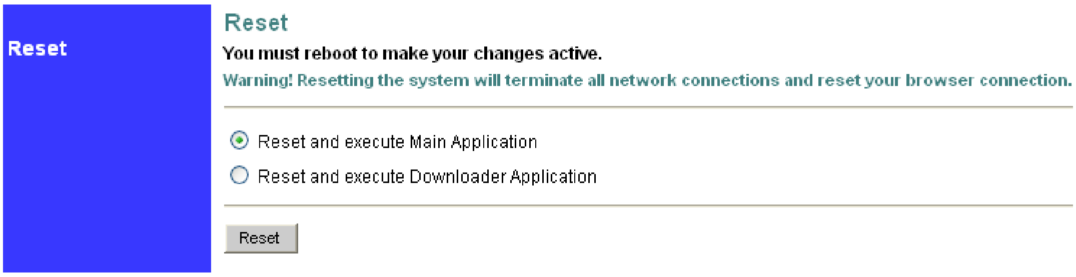
2.7 Configuration

2.7.1 Backup and restore settings



Back up and restore the configure files.

2.8 Reset



IP 1 will save the current settings and reset by clicking the “reset” button

3 . Restore to factory default

If your IP 1 settings is in chaos or you can’t get the IP 1 IP to access it, you can reset the device to factory default:

a) Power off
b) Press reset button and power on
c) The PWR led will light →Then PWR , SYS, WAN and LAN led will blink→ Then the PWR and SYS led light.
d) Release the reset button after about 15~20 seconds,
e) The PWR, SYS, WAN and LAN led will blink for a while and then the PWR and SYS led will light.

IP 1 will be reset to factory default after the above procedure, you can then access IP 1 through its LAN port, please refer to access_AG110 for details.

4 . FAQ

Q1 What is the default account of IP 1?

A1 : The default account is:

Administrator:	user name: admin	password: voip;
User:	user name: user	password: voip

Q2 How to use the IVR function of IP 1?

A2 :

The IVR function is record in G729 codec, so you have to choose G729 codec to active the IVR.

You can use IVR function to observe and set the WAN port network parameters pick up the handset and dial **** to enter IVR mode。

Key	Function	Input
****	Enter main menu	Submenu
100#	Check network state	
110#	DHCP Settings	1# Enable DHCP function 2# Disable DHCP function or# back to main menu
120#	Static IP address Settings	Use "*" replace ".", and "#" as end. For example: 172*16*230*227# or # back to main menu
130#	Gateway IP settings	Use "*" replace ".", and "#" as end. For example: 172*16*230*1# or # back to main menu
140#	Subnet mask settings	Use "*" replace ".", and "#" as end. For example: 255*255*255*255# or # back to main menu

Q3 How can I know the IP address of IP 1?

A3： you can use the following methods to obtain IP 1's IP ：

1. Use IVR function**;**

2. Observe the IP from the upper gateway；

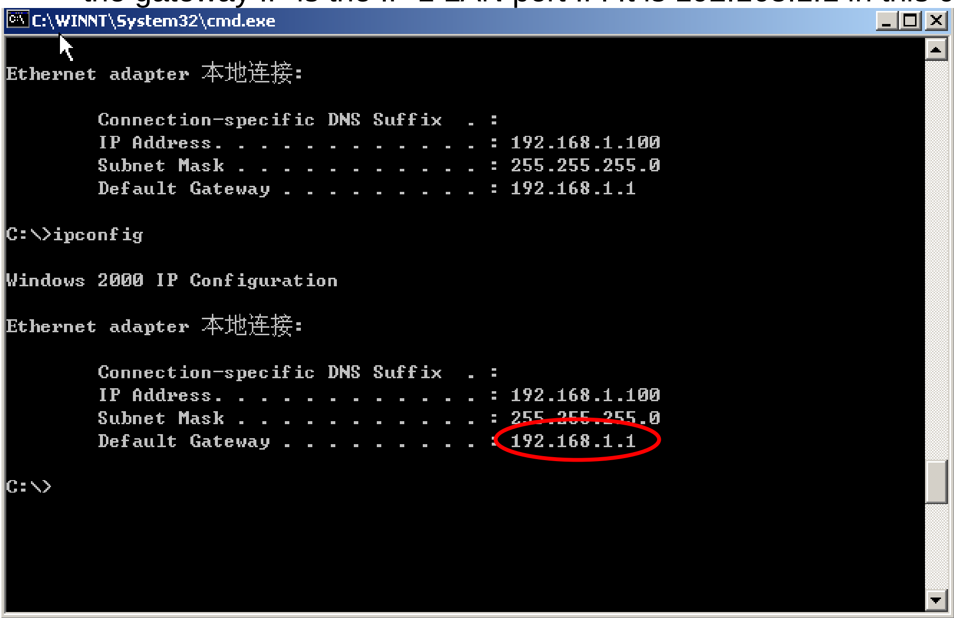
3. Use the DHCP assignment function of IP 1

i. Reset IP 1 to factory_default ，

ii. Directly connect your computer and IP 1' LAN port，

iii. Set your computer to dynamic obtain IP，

iv. Use command "ipconfig" to view your computer network status, and the gateway IP is the IP 1 LAN port IP. It is 192.168.1.1 in this case.



v. Then you can use this IP address to access IP 1 through its LAN port the default account is admin/voip

Q4 How to update IP 1 firmware?

A4： Go to Download→Download, press “browse” in the http download method，and choose the correct firmware file (a 1.5M file in .r0 extension),and press the “Start HTTP Download” to perform updating。

Q5 How to use dial plan?

A5：

A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

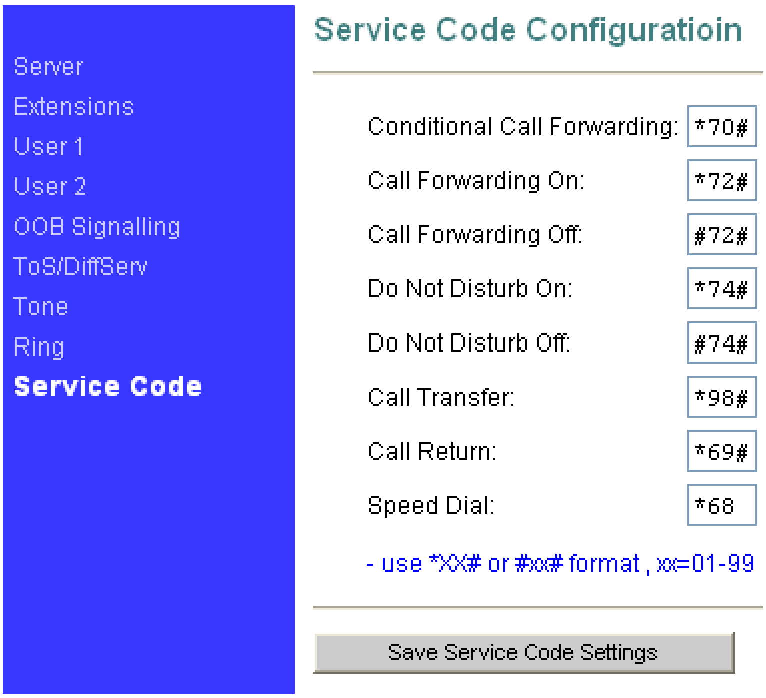
Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"
Timer ::= "T" | "t"
Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c"| "D" | "d"
Range ::= "X" | "x" -- matches any digit
| "[" Letters "]" -- matches any of the specified letters
Letters::= Subrange | Subrange Letters
Subrange::= Letter -- matches the specified letter
| Digit "-" Digit -- matches any digit between first and last
Position::= Letter | Range
StringElement::= Position -- matches any occurrence of the position
| Position "." -- matches an arbitrary number of occurrences
including 0
String ::= StringElement | StringElement String
StringList::= String | String "|" StringList

DialPlan::= String | "(" StringList ")"
A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is underqualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string. The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxxx) will match immediately if 5 digits are entered, it will also match after a 4 second pause when 3 digits are entered.

Simple Dial Plan
Allows dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

Complex Dial Plan
Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.
Dial plan for this is:
(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

Q6 How to use the value add service of IP 1?



You need to set the service code for using the IP 1 value add service. For example, I set the service code as the above picture。

Condition Call Forwarding: (the call will transfer if no one answer)

a) Set forwarding number: pick up the handset→press *70# →then you will hear the dial tone→press the forwarding number→then you will here three beeps indicating setting finish。
b) Set the timeout: go to the “sip extensions→Conditional call

Forwarding timer" and set the timeout before forwarding，unit: second, and then active this option.

c) Then the call will automatically transfer to the forwarding number if no one answers the call in the timeout.

Call Forwarding: (forwarding always)

a) Enable call forwarding: pick up the handset→press *72# →then you will hear the dial tone→press the forwarding number→then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically。
b) Disable call forwarding: pick up the handset → press #72# →then you will here three beeps indicating setting finish

Do not disturb: (DND)

a) Enable DND: pick up the handset → press *74# →then you will here three beeps indicating setting finish→then the phone won't ringing when there is an incoming call。
b) Disable DND: pick up the handset → press #74# →then you will here three beeps indicating setting finish

Call transfer:

a) Unattended transfer: A call B→B press *98# and then enter C number→then B will hear three beeps indicating the transfer successfully。
b) Attended transfer：A calls B→B push the hook flash to hold A→B then dial C number to talk with C → then B press *98# to transfer the call→then A can talk with C。

Call Return:

Pick up the handset →and then press *69# to dial the latest received call

3 way conference call:

A calls and talks with B → B push the hook flash to hold A B then dial C number to talk with C →B then push the hook flash again to enable three way conference call→ C will leave the call is B push the hook-flash again.

Q7 How to configure IP 1?

A7 please refer to “IP 1 quick start guide”

Q8 How to change IP 1 LAN port MAC address?

A8 please access http://IP_1ip/burn.htm and change the MAC address, after you have changed it, clap the reset button to save your setting.

Q9 Why does my IP 1 always drop off from the server?

A9

You can find the register TTL in the “SIP→server→ Send Registration Request with Expire Time”, if this time is longer then the system require register time, IP 1 will always drop off from the server, please set this time to a suitable value, (unit: seconds).

Q10 How to use the speed dial function?

A10

You need to set the speed dial number in the “SIP→User→ Speed Dial Settings”, and then set the operation code in the “Server→Service Code”, for example *68, then you can dial *681 to replace the speed dial number in the speed dial settings.