PL-304 Gateway

User's Manual

<Version: V1.01 (A)>

Congratulations on your purchase of the product. Please read the manual carefully to ensure your gateway work in best status.

Security and Notes:

- **Don't use it in chemical plant、gas station or near the exploder place.**
- **Don't use it near the equipment easy to be interfered by wireless.**
- **Avoid using it in too high or low temperature, and avoid exposing the product in sun or high-humidity condition.**

Note: The specifications and information regarding the products in this manual are subject to change without notice. We reserve the right of improving the product without informing users and ultimate interpretation right of its performance.

1. Revision information

Revision History					
Versi	Note	Author	Reviewed by	Approved by	Effective date

2. Terminology

Terminology	
Name	Note

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1 Overview

The popularization of the Internet drives the rapid development of a wide variety of IP-based applications. The IP telephone technology has become the major means for operators to develop voice services now. Especially, IP technology becomes the core of the next generation network (NGN), so the IP-based voice technology will keep soaring speeding the future and become the No. 1 choice of new operators in exploring services.

Upon the market requirement, LINKSZ Co., Ltd has launched the IAD 4FXS gateway integrated access device.

As an Integrated Access Device, the upstream port of the IAD can be directly connected to the IP network and its downstream port can be connected to multiple ordinary telephone sets, to provide basic accesses for POTS users. This user port gateway can support two telephone lines and connect multiple telephone sets. At the same time, this device is capable of Internet gateway and can access data stream, such as Email or Internet information. It is also applicable for small-size enterprises and IP telephone bars.

2 Packing

The IAD is packed with color chassis. Upon receiving the product, please confirm whether the fittings are complete. The packing box contains a set of IAD, 1 piece of RJ45 network cable, 2 pieces of telephone cables, one IAD power adapter and user’s manual.

3 Safety Instructions

To ensure your safety and safe use of this product, please pay attention to the following items:

- Follow the instructions in the user’s manual.
- Keep the device far away from chemicals and regent.
- Store/use the equipment in dry and well-ventilated environment.
- Never open the chassis lest the device is short-circuited or damaged.

4 Introduction to IAD

The IAD works with the most popular LINUX embedded operation system and has special CPU and DSP compression algorithms, featuring universal functions and applicable to a wide variety of needs.

Basic features:

- One 10/100 BASE-T WAN port, used to connect broadband data network
- 4 analog loops starts the FXS interface (RJ-11), used to connect 4 telephones
- Supporting DHCP Client or static IP address allocation plan
- Supporting 802.1Q VLAN and VLAN Tag
- Mute compression and comfort tone generation technology ensure clear conversation quality.
- Self-adaptive jitter cache ensures smooth voice function
- Lost-packet compensation guarantee mechanism provides a better voice quality.
- Built-in Internet gateway function
- Supporting NAT (Network Address Translation) and NAPT
- Supporting DHCP Server, used for the dynamic address allocation plan of LAN devices
- Built-in PPPoE client, used for broadband access user authentication
- Easily-configured Console port
- Supporting remote configuration of Web mode and remote software downloading/upgrading

5 Performance Indices

Description of Product Model	
4 FXS Gateway	4-port IP voice gateway, H323/SIP protocol
Physical Specifications	
Size	150mm (L) × 97mm (W) × 29mm (H)
Power supply	AC/DC power adapter, 12V DC DC input: 12V DC/2A
Power consumption	< 15W
Weight	600g
Reliability	System availability > 99.999%, MTBF > 100,000 hours, MTTR < 5 min
Ambient requirements	
Working temperature	0°C ~ 50°C
Storage temperature	-10°C ~ 50°C
Relative humidity	5% ~ 95%, non-condensing
Technical Specifications for the interface attribute gateway	
Supporting H323/SIP protocol	
Mute processing/four wave processing	
RTP/RTCP voice channel	
Voice compression algorithm G.729, G.723,G711 and G.726	
Analog voice port (FXS), 4 ports	
Signal format: DTMF	
Echo suppression: G.165/G.168	
DTMF signal detection/generation	
Compatible to the Internet protocols, such as TCP/IP, UDP, ARP, TFTP and ICMP	
Supporting SNMP Version II	
Compatible to IEEE 802.3 10BASE-TX Ethernet	
Compatible to IEEE 802.3u 100BASE-TX fast Ethernet	

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6 Appearance Description

6.1 PL-304 gateway Front Illustration

1. DC: Power adapter: output DC 12V/800mA DC

2. Reset: press the reset key and plug the power .Reset to default set after 5 seconds

3.**Wan**:10M/100M , default IP 192.168.1.200

4: LINE1~LINE4: telephone 1 ~ 4

3.2 PL-304 Backside Illustration

1.PWR: power

2.ACTIVE: system status. Not registered in sip sever , flash 1 second and light 5 seconds ; registered , light off

3.WAN: Wan status

4.TEL1~ TEL4: telephone status

7 Configuration Description

The GATEWAY provides modify WEB parameters through WAN interface.

WAN Configuration Environment

„ Configure the "TCP/IP Protocol" of PC according to Fig. 4 with the PC and WAN interface in the same network segment.

- Configure the device according .
- Use straight-through cables in the figure.
- Configure IE according.
- After configuration, input the IAD default IP address in IE address bar. Each gateway will be allocated with an initial Wan IP address before delivery, assumed to be 192.168.0.200.

8 Configuration in WEB Mode

8.1 home

System Uptime: specifies the amount of time, which the system has been up. This time is reset every time the system is reset.

NTP time: tells System Time.

LAN IP Address: indicates the IP Address of your LAN.

MAC address: MAC address is the address of your MAC.

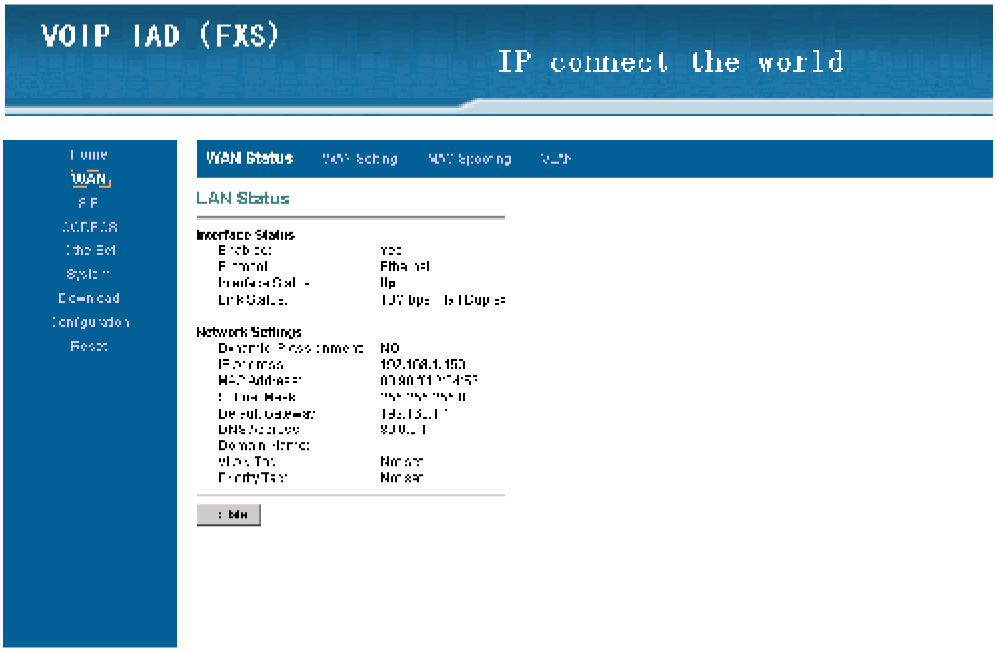
Security: for your password, which is configured in the "System" section.

Application Code Version: tells the version of the application code which you are using.

Download Code Version: tells the version of the download code which you are using.

9.2 WAN

9.2.1 WAN status



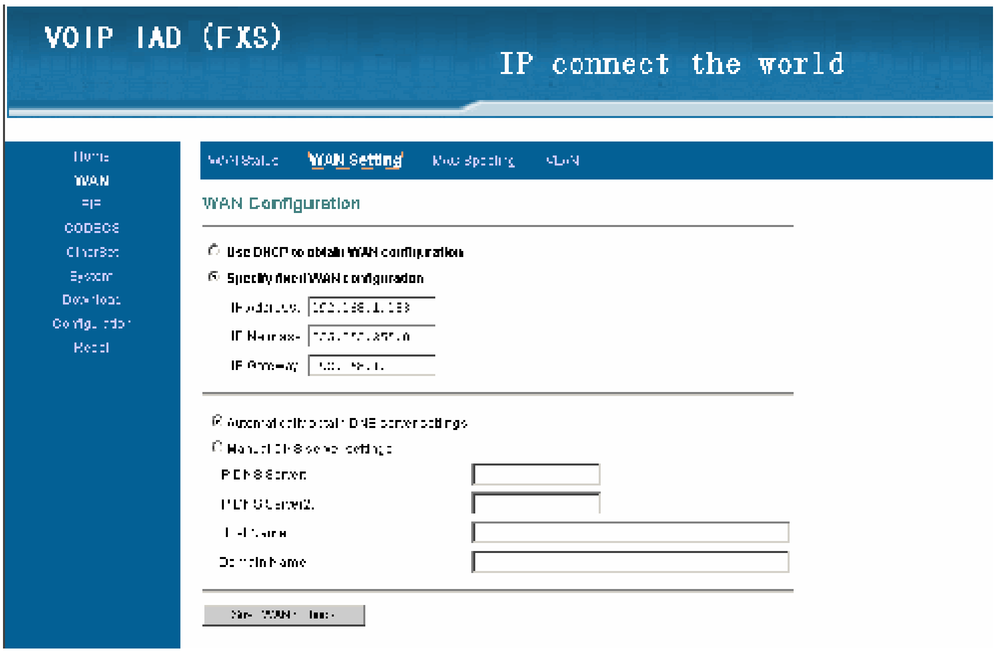
Interface Status：

1. Enable: Yes express Lan can use.
2. Protoco：Ethernet
3. Interface status：UP or Down
4. Link Status : state of connected.

Network Settings：

5. Dynamic IP Assignment: Dynamic IP
6. IP address: Local IP.
7. MAC Address: Local MAC Address.
8. Subnet Mask: Local Subnet Mask.
9. Default Gateway: Local default gateway.
10. DNS Address: local DNS server.
11. Domain Name: domain.
12. VLAN Tag: encode in each Ethernet datum
13. Priority Tag: encode in each Ethernet datum

9.2.2 WAN Settings



1 Use DHCP to obtain WAN configuration: Use DHCP.

2 Specify fixed LAN configuration: Configuration WAN.

3 IP Address: Input IP Address.

4 IP Netmask: Input IP Netmask.

5 IP Gateway: Input IP Gateway.

6 Automatically obtain DNS server settings: Atuomatically obtain DNS.

7 Manual DNS server settings：Input DNS.

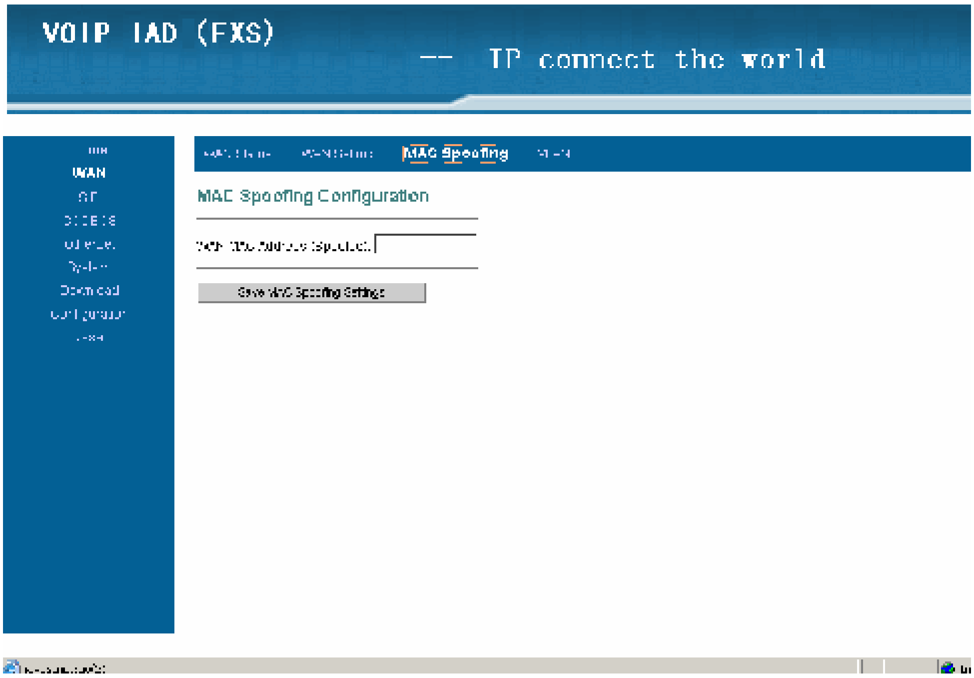
8 IP DNS Server: Primary DNS IP

9 IP DNS Server2: Secondary DNS IP.

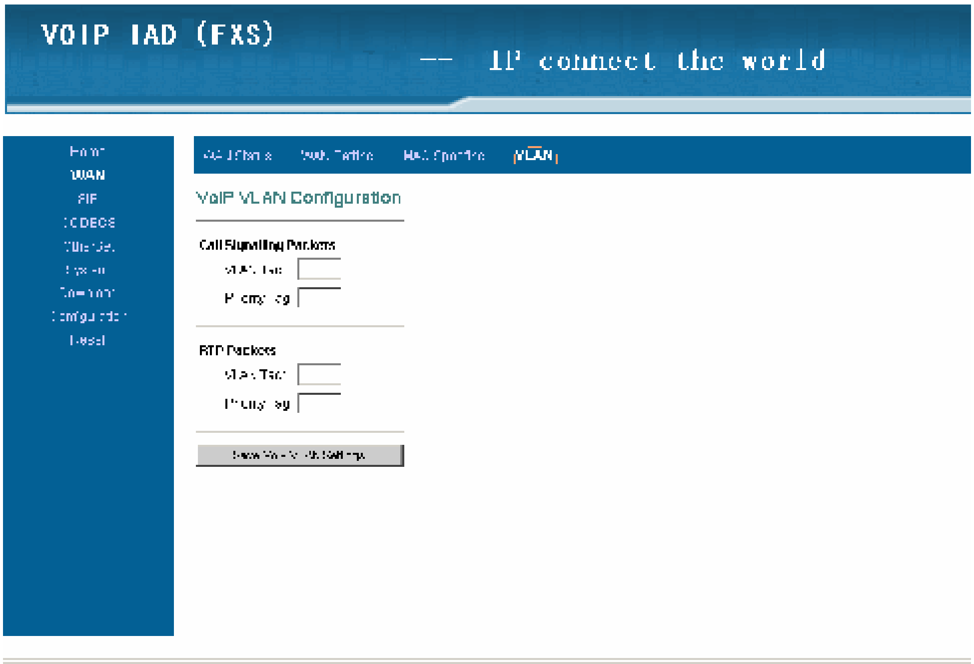
10 Host Name: Input Host Name.

11 Domain Name: Input Domain.

9.2.3 MAC



9.2.4

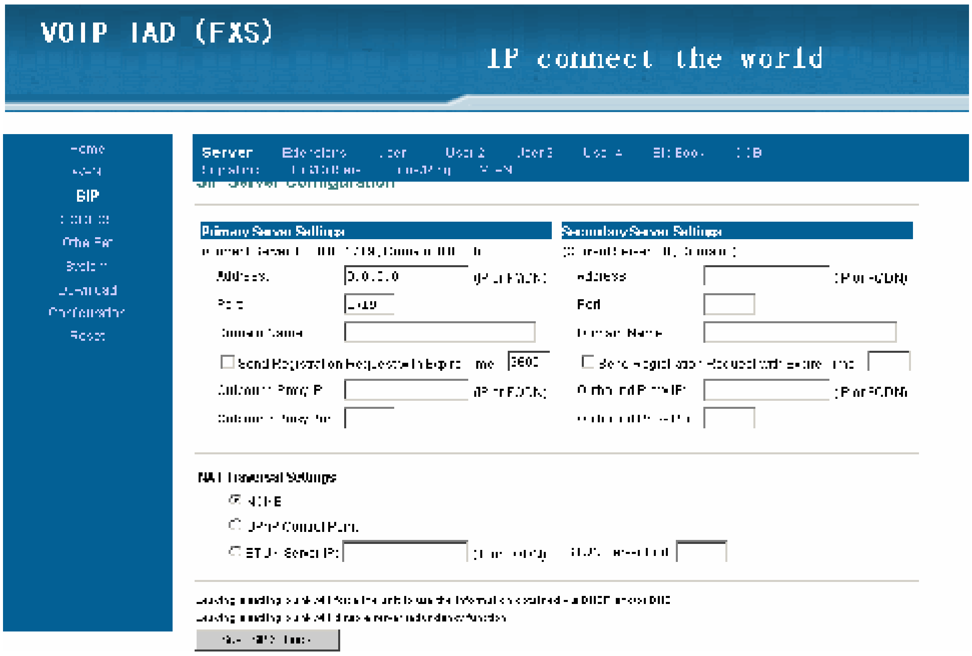


1. LAN VLAN Tag :   Value range 0-4096.

2. LAN Priority Tag: Value range 0-7.

9.3 SIP

9.3.1 Server



1. Primary Server : Primary Server Configuration.
2. Secondary Server: Secondary Server Configuration.
3. Address: Input Server IP or FQDN.
4. Port: Register Port.
5. Domain Name: Input Domain.
6. Send Registration Request with Expire Time.
7. Outbound Proxy IP: Input Proxy IP.
8. Outbound Proxy Port: Input Proxy port.

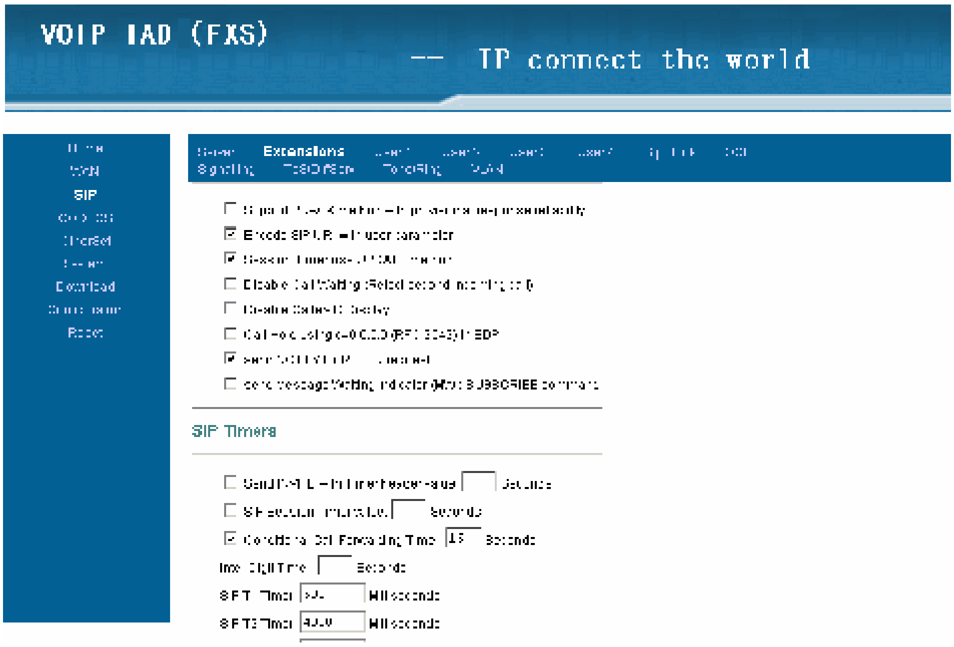
NAT Traversal

9. NONE: Not use
10. UPnP：Use UPnP
11. Stun Server IP：When Use Stun Input stun IP Address.
12. Stun Server Port：Input Stun Port.

Gateway Settings

13. Dial Plan：Appendix. Dial Plans.
14. # use as a quick dial function: When Press # dial.
15. To enable # to be recognized as dial number: use # to be recognized as dial number.
16. * use as a quick dial function: When Press * dial.
17. To enable * to be recognized as dial number: use * to be recognized as dial number.

9.3.2 Extensions



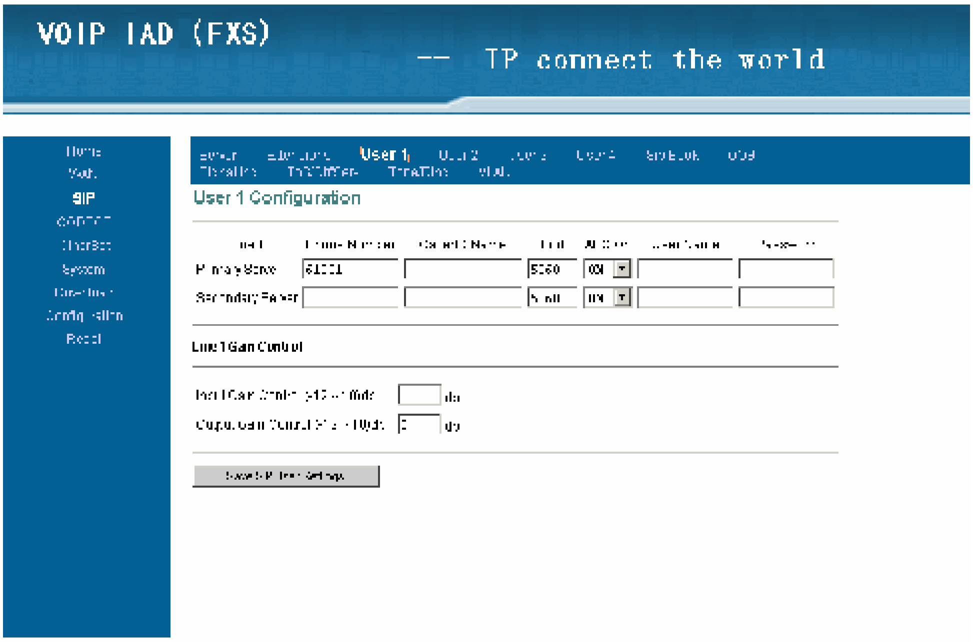
SIP Extensions：

1. Support PRACK method with provisional response reliability ：Use SIP PRACK Method.
2. Encode SIP URI with user parameter ： user = phone parameter Encode.
3. Session Timer use UPDATE method ：Use Session Timer Update method.
4. Disable Call Waiting (Reject second incoming call) ：Disable Call Waiting.
5. Disable Caller-ID Display ：Not Send CallerID Display.
6. Call Hold using c=0.0.0.0 (RFC 2543) in SDP ：Use RFC2543 for call hold using. If without selection, use RFC3263 for call hold using
7. Send NOTIFY for REFER request .

SIP Timers：

8. Send INVITE with Timer header value ：Set Send INVITE with Timer header.
9. SIP Session Timer value ： Set SIP Session timer
10. Conditional Call Forwarding Timer ：Set Conditional Call Forwarding Timer.
11. SIP T1 Timer ： Set SIP T1 Timer.
12. SIP T2 Timer ： Set SIP T2 Timer.
13. SIP T4 Timer ： Set SIP T3 Timer.

9.3.3 User1



1. **Primary Server**

   Phone Number：Input Phone Number.

   CallerID Name：Input CallerID

   Port : Input Port.

   AEC On：Use Check.

   User Name：Input User Name.

   Password：Input Password.

2. **Secondary Server**

   Phone Number：Input Secondary Phone Number.

   CallerID Name：Input Secondary CallerID.

   Port : Input Secondary Port.

   AEC On：Use Check.

   User Name：Input Secondary User Name.

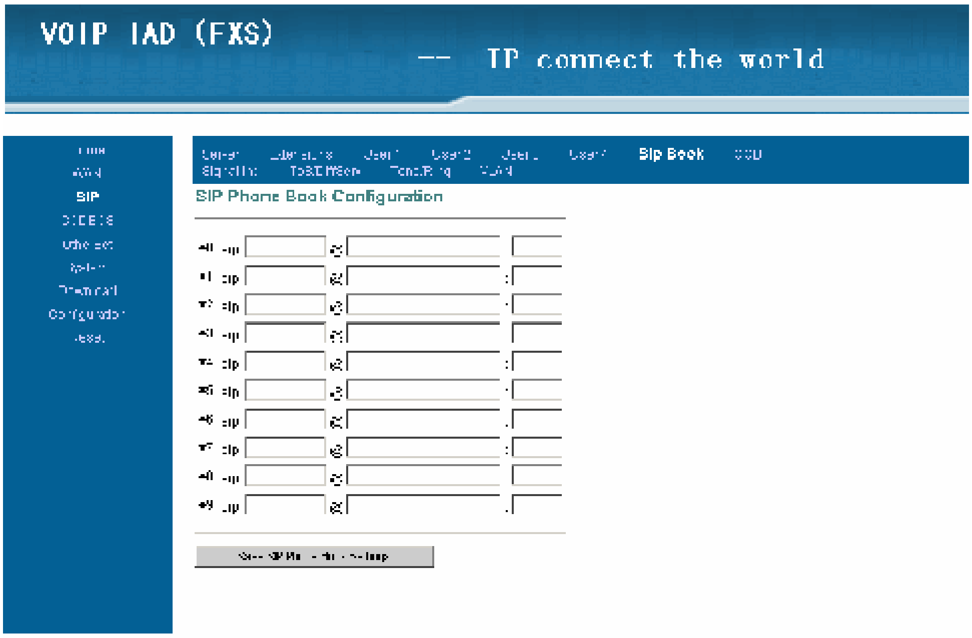
   Password：Input Secondary Password.

3．**Line1 Gain Control**：

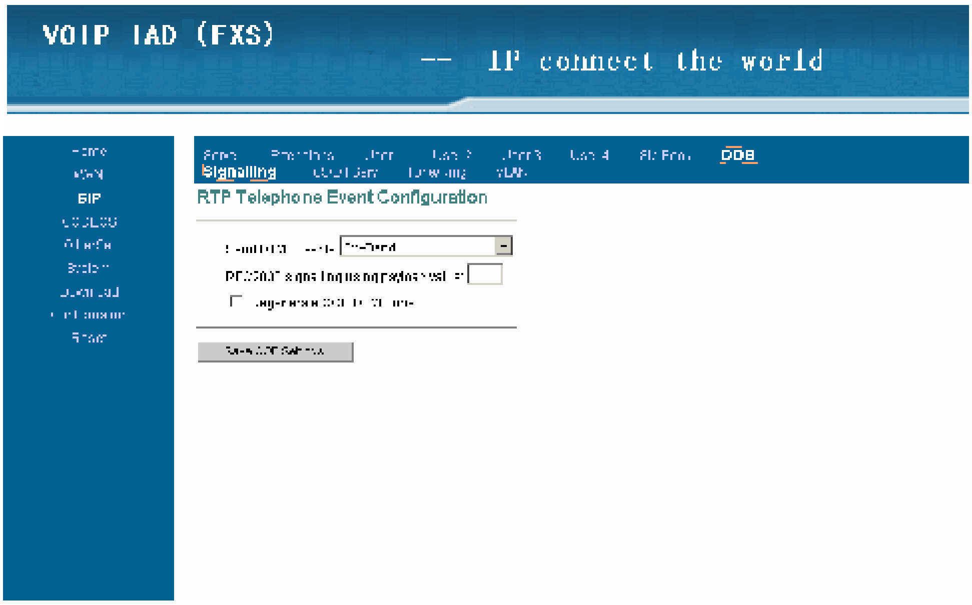
   Input Gain Control (-12 ~ 18)db：Input Gain Control.

Output Gain Control (-12 ~ 18)db：Output Gain Control.

9.3.4 SIP



9.3.5 OOB

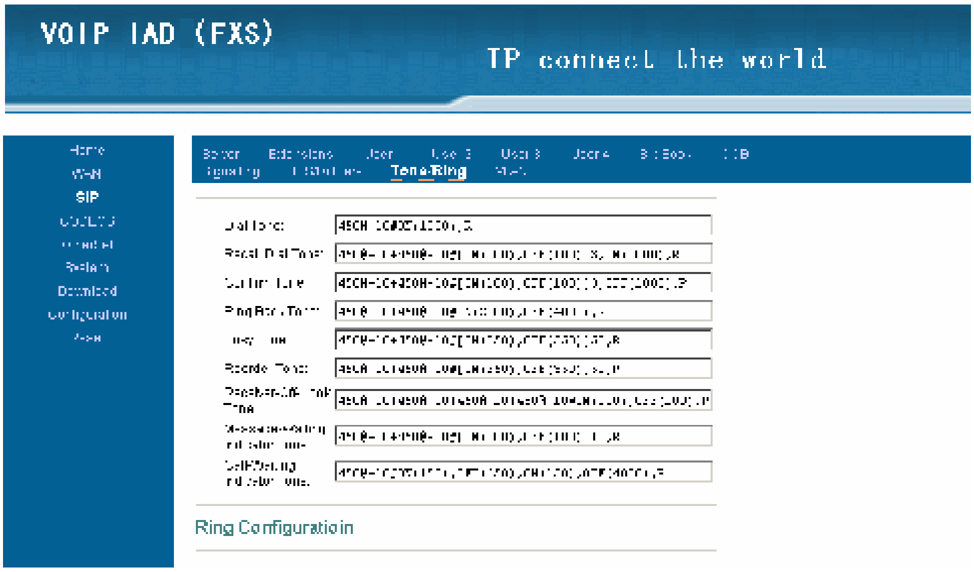


Configuration SIP Protocol DTMF Send Events: Inband、Outband RF2833,Inband&00B、RFC2976

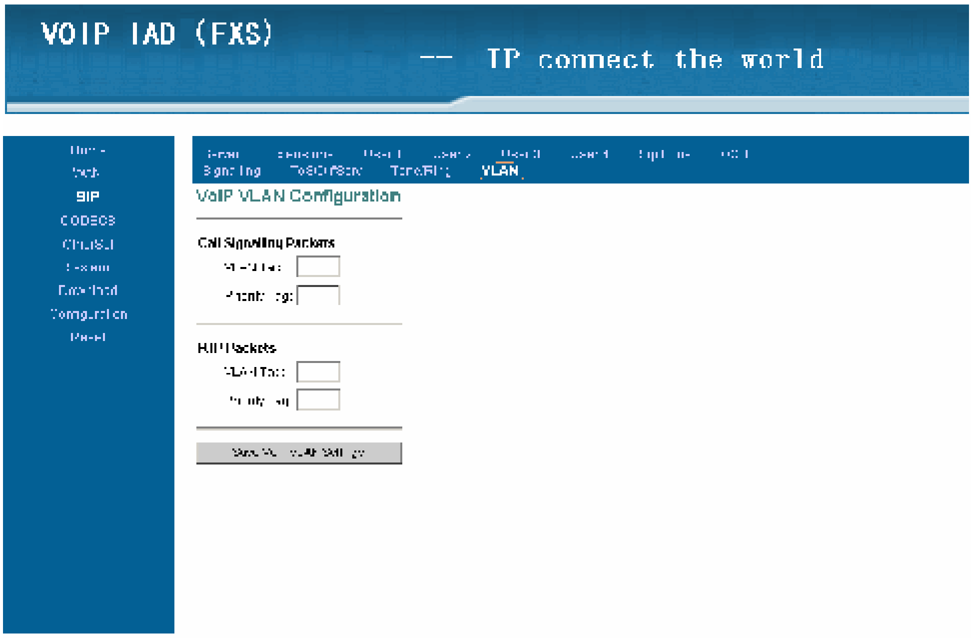
9.3.6 Tos/DiffServ

The page used for disposing Type-of-Service/Diffserv value, the digital will be used on the beginning of each IP for transmission of SIP and RTP. If no special requirement about the digital, please enter either 00 or leave it blank.

9.3.7 Tone/Ring



9.3.8 VLAN



1. Call Signalling Packets

VLAN Tag：encode in the Ethernet datum, available for 0- 4094

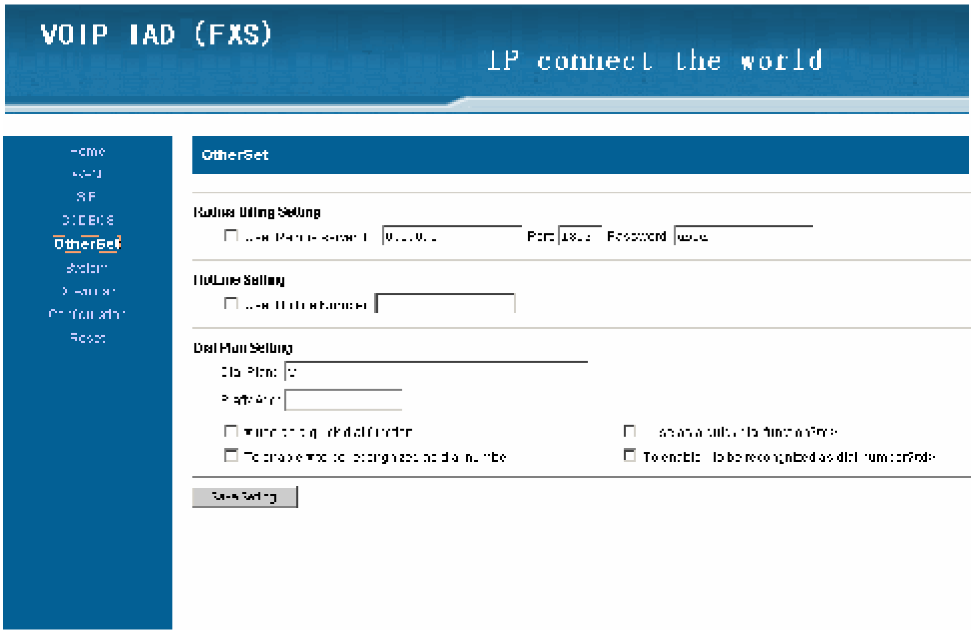
Priority Tag：encode in the Ethernet datum, available for 0- 7

2. RTP

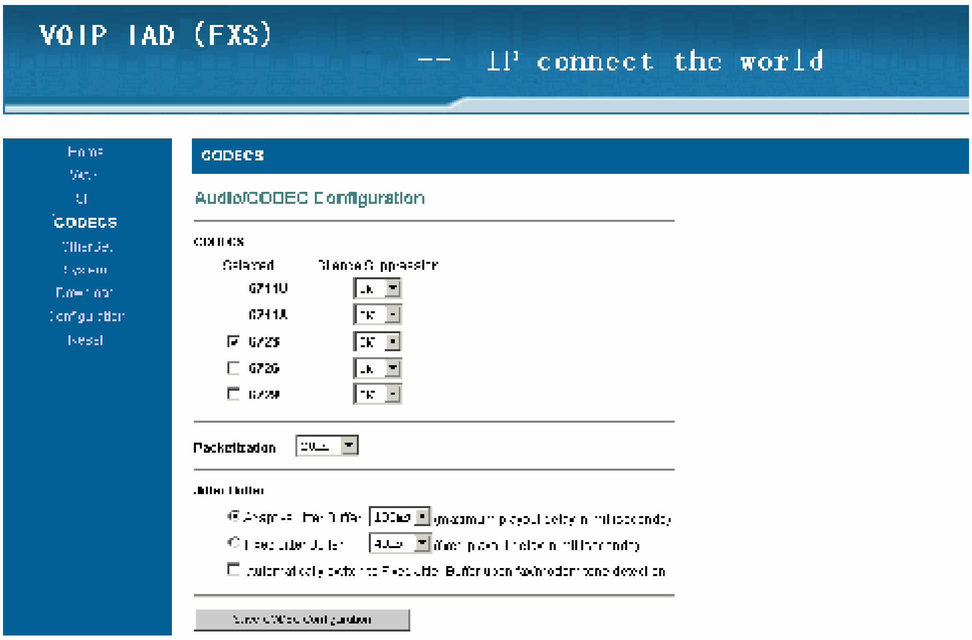
VLAN Tag：encode in the Ethernet datum, available for 0-

Priority Tag：encode in the Ethernet datum, available for 0-

9.4 Other Setting



9.5 CODECS



CODECS:

You can Selected：G711U、G711A、G723、G726、G729。

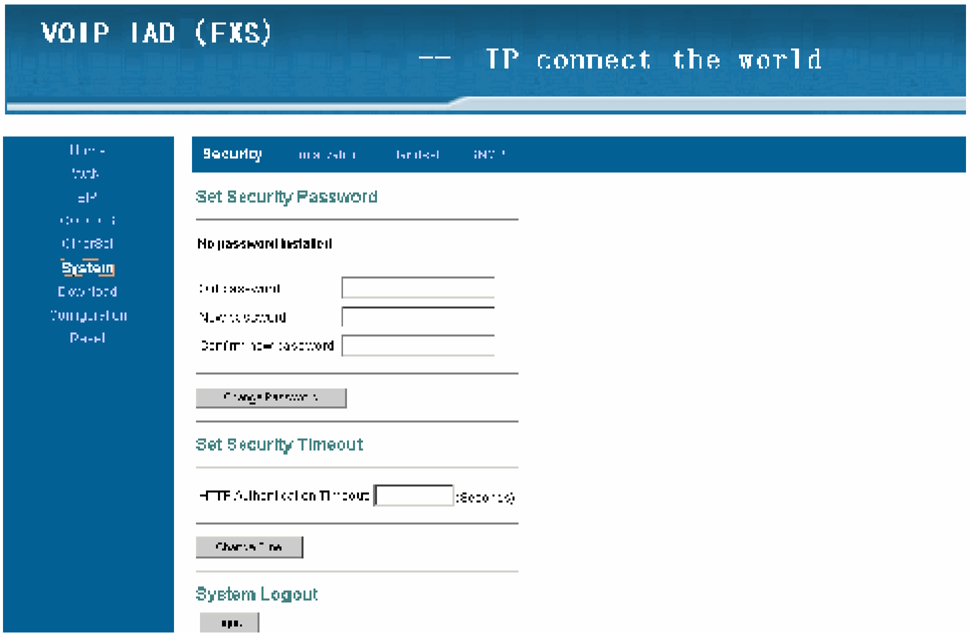
Packetization

Jitter Buffer

Config Voice Buffer Time ：select automatic or fixed， Factory Default is

9.6 System

9.6.1 Security

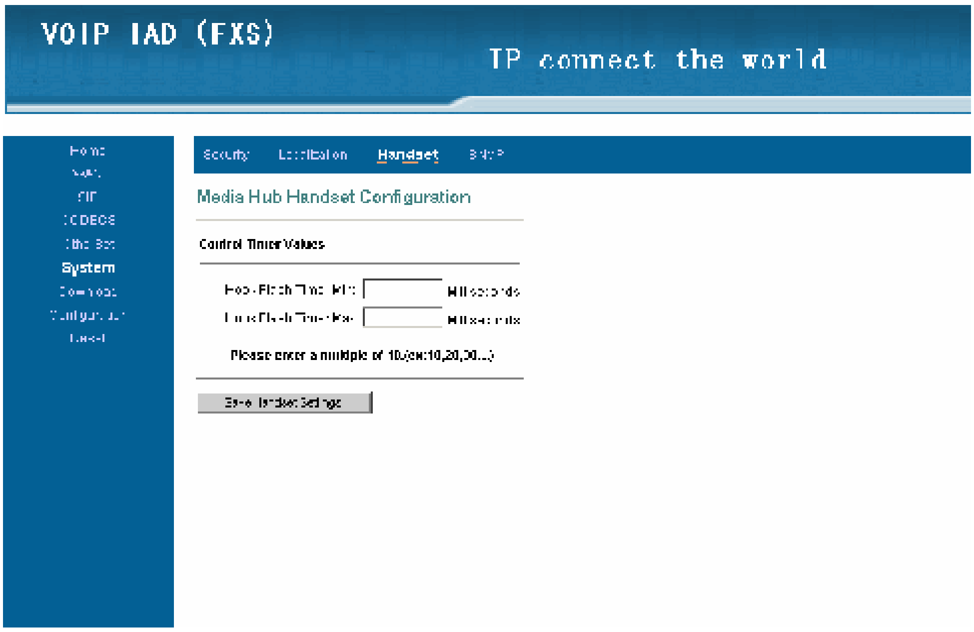


Set or Change Password Security Password.

9.6.2

Selected Country and time Zone.

9.6.3 Handset

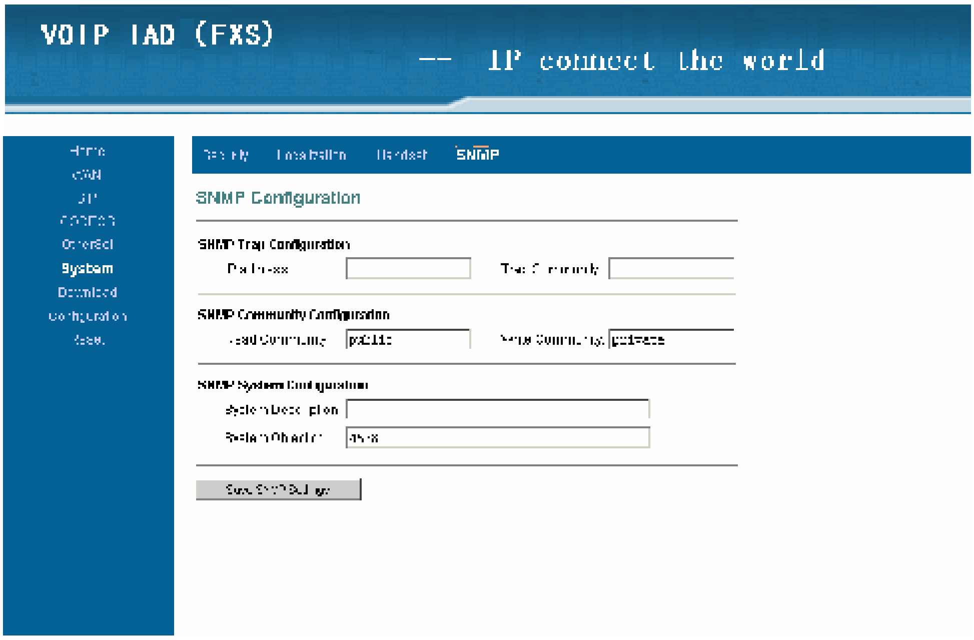


Set Hook Flash Control Timer

Hook Flash Timer Min: Set Hook Flash Min availability Timer.

Hook Flash Timer Max: Set Hook Flash Max availability Timer.

9.6.4 SNMP Configuration



SNMP Trap Configuration

1. IP address：Trap host IP address.

2. Trap Community: The community name used by the SNMP manager to verify traps. The default value is ‘public’.

SNMP Community Configuration

1. Read Community：The community name used by the SNMP manager when reading SNMP data items from a client MIB. The default value is ‘public’

2. 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’

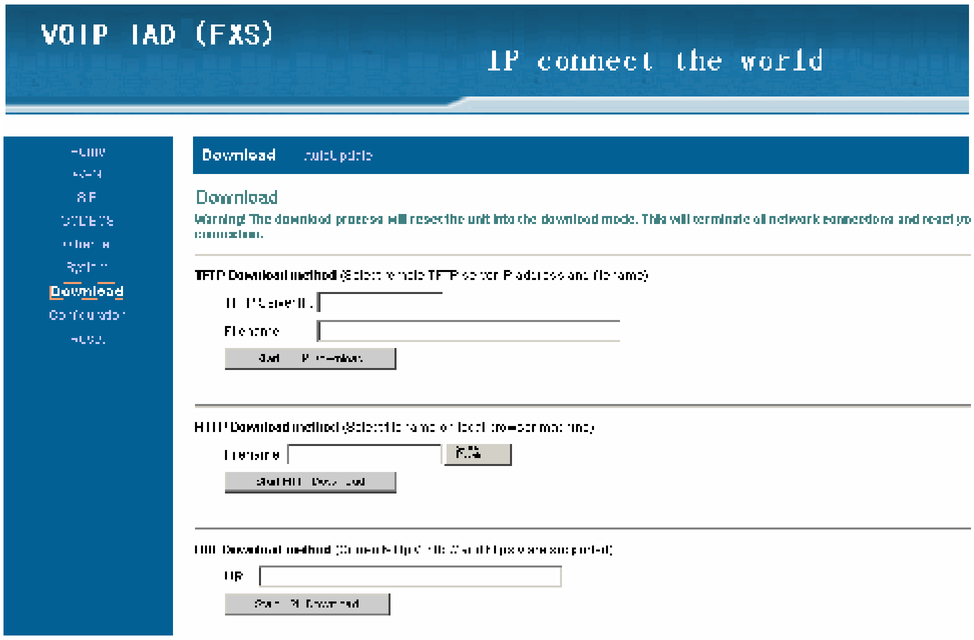
SNMP System Configuration

1. System Description：Description of the unit (e.g. “John’s phone”)

2. System ObjectId:：A vendor’s enterprise ID

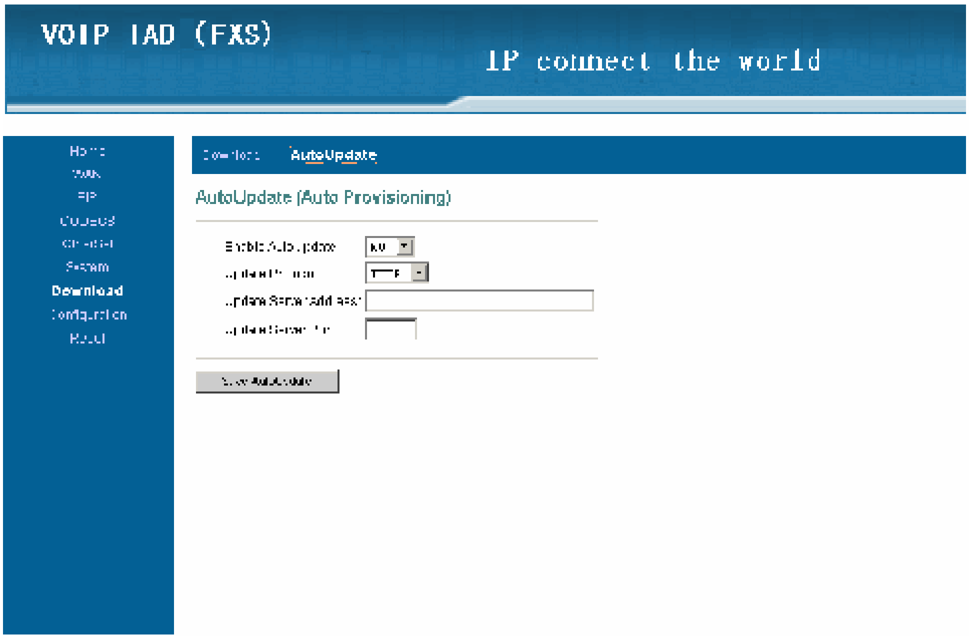
9.7 Download

9.7.1 Download



For both **HTTP** /**TFTP** and **URL** 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.

9.7.2 AutoUpdate



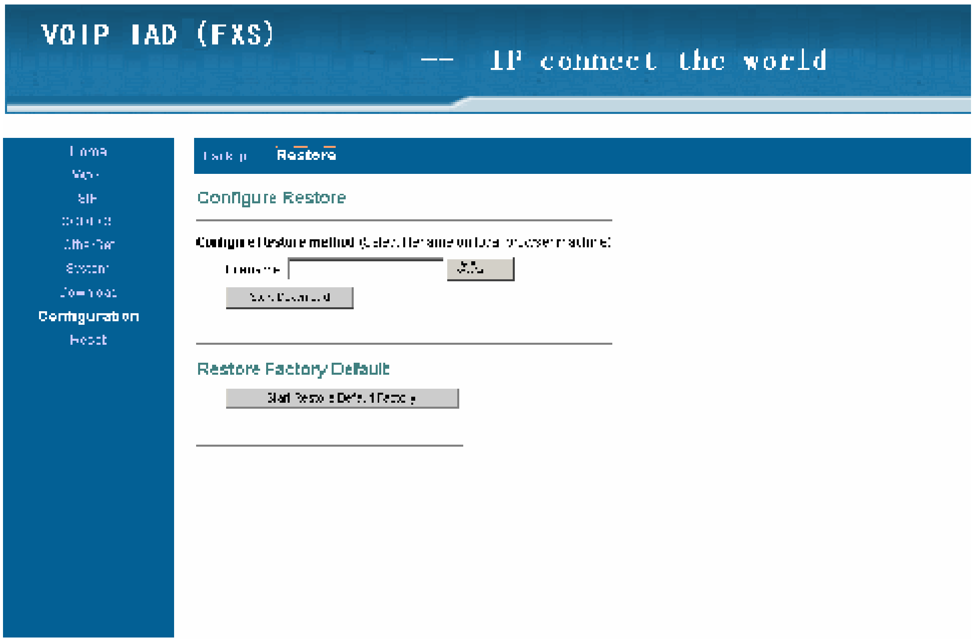
Configuration AutoUpdate Download.

9.8 Configuration

9.8.1 Backup

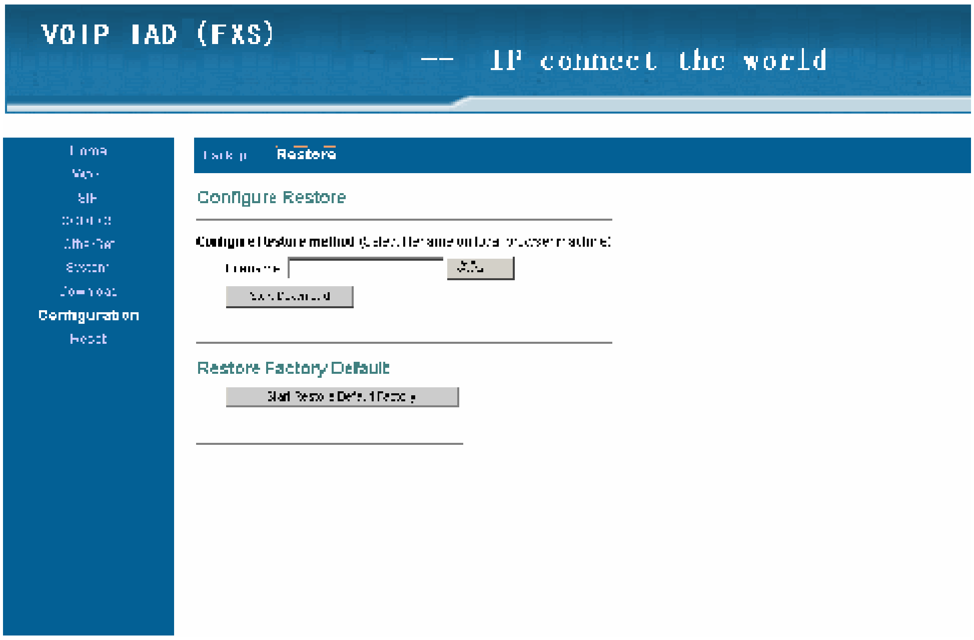
Configure file Backup

9.8.2 Restore



Restore Factory Default.

9.9 Reset



Chose the “**Reset and execute Main Application**” option, for execution of the main application which you have configure, once you reset the system.

Chose the “**Reset and execute Downloader Application**” option, to being downloading, once you reset the system.

9 Notes

9 .1 Appendix. Dial Plans

The H.323 and SIP code will allow provisioning (via web browser) of the dial plan. 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)

Non-dialed Line Dial Plan

As soon as handset is lifted the unit contacts the gatekeeper (used for systems where dtmf detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

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)

9.2 What About Reset Default Setting?

Press WT-3204 backside Reset Key till 5 second.

9.3 Other Features:

Conditional Call Forwarding

(Enter ＊70＃ and hear **Dialtone** , enter guest phone number, hang up after you hear the

After the seconds you set on web page, an unanswer incoming call will be transfer to the guest phone number.

To enable this function, check the web page -> sip extensions -> conditional call Forwarding timer checkbox.

Call Forwarding On

(Enter ＊72＃and hear **Dialtone ,** enter guest phone number, hang up after you hear the

Any incoming call will be immediately transfer to guest number.

Call Forwarding Off

(Enter ＃72＃, hang up after you hear the **Confirm**

Do not Disturb On

(Enter ＊74＃, hang up after you hear the **Confirm**

Incoming calls can be blocked , and the caller will hear busy tone from you.

Do not Disturb Off

(Enter ＃**74**＃, hang up after you hear the **Confirm**

Call Hold / Retrieve

(When in conversations,you may press flash-hook to hold this call / retrieve this call

By flash-hook ,You can dail to another after hearing the **Redial tone.**

Call Transfer **-Blind transfer**

(A call B, B pick up this call, B enter ＊98＃ then enter **C's**

B hang up after hearing **Confirm tone,** A will be transfer to C.)

-Attendant transfer

(A call B, B pick up this call, B can hold this call and redial C by flash-

after C is acknowledged,B enter ＊98＃ to

3-way Conference calling

(While A,B in talk,A press flash-hook to hear **Redial tone,** then A call to C, after AC in talk, A press

flash-hook again to add B into the conference)

Leave Conference call

(When in 3-way Conference,the master can drop the 3rd call by flash-hook)

Answer Call waiting/Call Alternate

(When hearing a **Call-Waiting-Indicator-Tone** during conversation,you can press flash-hook to

hold current conversation and pickup the incoming call)

A can switch conversation from B to C ,or C to B ,by pressing flash-hook

2nd incoming call can be set to reject on web pgae -> sip extensions -> Disable call waiting checkbox.

Call Return

(Enter ＊69＃ to call return to the last engage phone