VS211

1 FXS / 1 FXO
SIP
VoIP Telephone Adaptor

User Manual

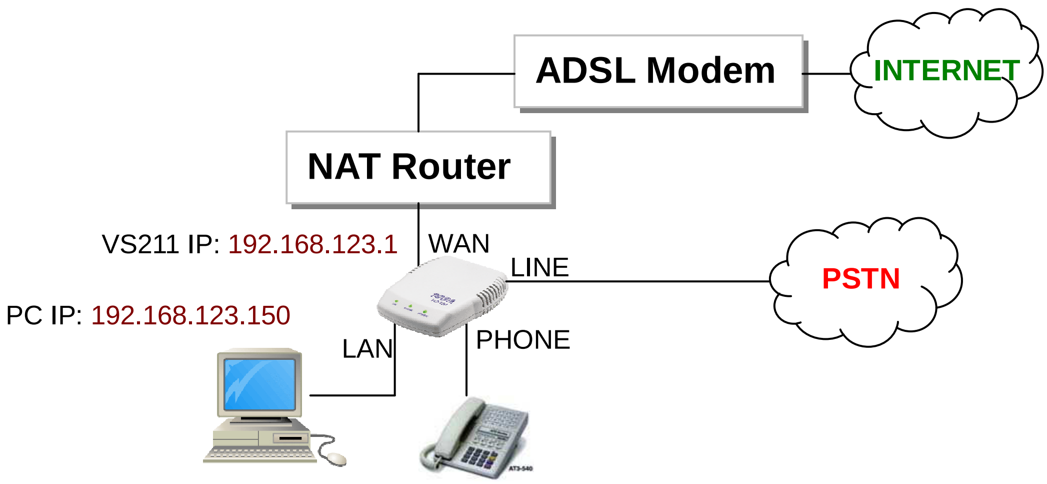
V2.1h

Quick Guide

Step 1: Broadband (ADSL/Cable Modem) Connections for VS211

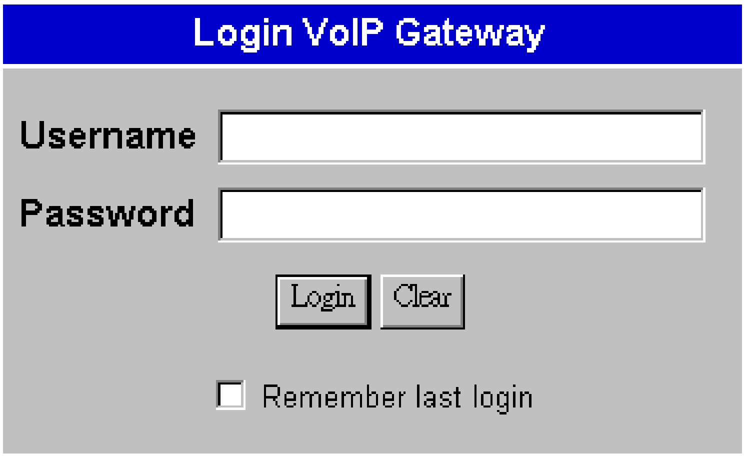
A. Connect VS211 WAN port to ADSL NAT Router as the following connection.
B. Connect VS211 LAN port to Notebook PC LAN port using a Category 5 LAN cable.
C. Connect VS211 RJ11 PHONE port to a Telephone Set.
D. Connect VS211 RJ11 LINE port to a PSTN Telephone Line.
E. Connect 12VDC Power Adaptor. After power on, the POWER LED will be **Green** ON. In 5 seconds, the PHONE LED will start flashing 5 times and be ready for configurations.
F. The PHONE LED will be **Green** flashing for a successful SIP registration. Pick up the phone, and it will be **Green** ON indicating VoIP mode.
G. Press **0*** key, and the PHONE LED will be **Yellow** ON indicating PSTN mode.
H. Press **#121#** and **#120#** from the phone to listen to IVR and to check the DHCP status and the LAN IP address (e.g. 192.168.123.1) for VS211. After the IP announcement, please hang up.

Figure A. ADSL Connections with NAT Router for VS211



Step 2: Settings for VS211 from PC Web Browser

A. VS211 is defaulted at embedded NAT mode.
B. Press **#120#** from the phone to listen to IVR and take down the IP address (e.g. 192.168.123.1) for VS211.
C. Enter the IP address from PC Web browser for configuration settings.
Example: Enter **http://192.168.123.1** from IE Web browser to display login page.
D. Enter the user name and password into the blank field. The default settings are
Username: **root**
Password: **test** .
Click the "**Login**" button to enter for configurations.



E. You need to set up the following web configurations: **Phone Settings, Network, SIP Settings, NAT Settings** for registration to a SIP server. Remember to submit, save and reboot for new configurations.

F. The PHONE LED will be **Green** flashing showing a successful registration in the SIP server. For further detail configurations, please refer to the VoIP applications chapter.

Step 3: Making Point-To-Point SIP Calls

1. While the PHONE LED is flashing continuously showing a successful registration in the SIP server, you may pick up the phone and should hear a dial tone.
2. Press **123456#** to call the party with the number **123456** registered in the SIP server. Note **#** is used to send out the call immediately. In a moment, you should hear the ring back tone, and wait for the called party to answer. For more applications, please refer to the user manual

Step 4: Making Public Switching Telephone Calls

A. Pick up the phone.

B. Press **0*** key for PSTN mode, and the PHONE LED will be **Yellow** ON, and you should hear a dial tone. Now you may dial any public phone number.

Note: Difficulties in configuring VS211? Please refer to the last chapter for trouble shootings.

TABLE OF CONTENTS

1. Introductions

The VS211 is a 1-port FXS / 1-port FXO Telephone Adaptor (TA) with SIP Protocols for Voice over IP (VoIP) applications. Connecting to the Internet and the PSTN line with an analog telephone set, the VS211 can connect a VoIP call over the Internet with extension to the public switched telephone line. VS211 provides one WAN port for Internet connections, one LAN port for Notebook PC, and two RJ11 connectors for Phone (FXS) and PSTN (FXO).   With an embedded NAT/DHCP server, VS211 can be easily configured for different network diagrams by PC Web browser and telephone set. This is very suitable for ITSP (Internet Telephony Service Providers) and SOHO users to make 2-stage VoIP with PSTN extension calls.

Note that VS211 requires an IP address, a subnet mask, and its gateway Router IP address for its own use to connect to Internet.   These three are available from your Internet service provider. VS211 may enable PPPoE or DHCP features to automatically get an assigned dynamic IP from the ITSP.   Please refer to Section 8 Configurations by Web browser for detailed information.

2. Features

The VS211 VoIP TA is equipped with two RJ11 connectors and two RJ45 connectors and is featuring as the following

- Three LED Indicators for VS211:   POWER, PHONE, LAN
- RJ45 x 2 for WAN and LAN ports + RJ11 x 2 for FXS and FXO ports
- Configurations by Web Browser and Telephone
- Embedded NAT/DHCP Server
- PPPoE/DHCP Client for Dynamic IP plus NAT, DNS, and DDNS Clients
- Support STUN server for NAT Traversal
- Support registrations for up to 3 SIP Servers.
- Hot Line Mode
- Dial Plan Settings
- T.38 FAX over IP
- Interactive Voice Recording (IVR) for telephone IP status
- Phone Book, Call Forward/Waiting, Call Transfer/Hold, and 3-Way Conference Calls
- Auto Configurations by TFTP, HTTP, or FTP server
- Remote Firmware Upgraded with HTTP or TFTP server by Web PC
- Direct IP/URL Dial without SIP Proxy or Dial number via SIP server
- Telephone features: Volume Adjustment, Phone book, and Flash
- Out-Band DTMF (RFC 2833) / In-Band DTMF / Send DTMF SIP Info

3. Standard Compliances

The VS211 VoIP TA supports for the following standards

VoIP Protocol:	IETF RFC3261 and RFC 2543 for SIP
SIP Authentication:	IETF RFC2069 and RFC 2617 for MD5
Speech Codec:	ITU-T G.711, G.723, G.729A/B, VAD and CNG
Echo Cancellation:	ITU-T G.165/168

4. Packing Contents

Inside the package you should find:

(1) One VS211 SIP TA

(2) One AC to 12VDC/1A Power Adaptor

(3) One User Manual CD

Please check if the packing is damaged or any component is missing. If so, please contact your distributor.

5. LED Indicator

On the front panel of VS211, there are three LED indicators as the following

POWER: "On" indicates the power is normal

PHONE: "**Green On**" indicates the Phone is in use (off-hook) for VoIP call.
"**Red On**" indicates the PSTN Line is in use or ringing from incoming call.
"**Yellow On**" indicates the Phone and the PSTN Line are in use for PSTN call.
"**Green Flashing**" indicates successful registration at SIP server.
"**Yellow Flashing**" indicates PSTN incoming call with successful SIP registration.

LAN: "On" indicates the WAN port is in Connection.
"Flashing" indicates the data activity of WAN port.

6. Installations & SIP Configurations

1. Connect VS211 RJ45 WAN port to ADSL Modem/Router using a Category 5 LAN cable.
2. Connect VS211 RJ45 LAN port to Notebook PC using a Category 5 LAN cable.
3. Connect VS211 RJ11 PHONE port to a Plain Old Telephone Set (POTS).
4. Connect VS211 RJ11 LINE port to a Public Switched Telephone Network (PSTN) line.
5. Connect the power adaptor (12VDC) to power on VS211, and the POWER LED will be ON.
6. The PHONE LED indicators will be OFF for about 5 seconds and start flashing for 5 times, and remain OFF for VoIP configurations. The LAN LED will be ON when RJ45 WAN port is connected. If the PHONE LED keeps flashing, it indicates that VS211 has successfully registered in the SIP server.
7. Pick up the phone, and the PHONE LED will be **Green** ON indicating VoIP mode. If you hear a busy tone, please check if the WAN port is connected properly.
8. Press **0*** keys to PSTN line. The PHONE LED will become **Yellow** ON and you should hear a PSTN dial tone. If not, please check if the PSTN Line is connected.
9. Press **#120#** to check the assigned IP address for the VS211. The default IP address is **192.168.123.1**. You may enter this IP address in IE Web browser from Notebook PC. Please refer to Chapter 8 for web configurations.
10. Register VS211 into your SIP server. Please refer to VoIP applications examples of SIP registrations. After successful registration to the SIP server, the PHONE LED will start **Green** flashing.
11. Pick up the phone, and you should hear a dial tone. Press **123456** to call the party with the number **123456** registered in the SIP server. In a moment (5 seconds), you should hear a ring back tone, and wait for answer. Note that you may press **123456#** to dial out the number immediately. Dialing without **#** will not dial out until the auto dial timer (default=5 seconds) elapsed.

7. Default Reset by Telephone

VS211 provides an easy way to reset to factory defaults by using Telephone.

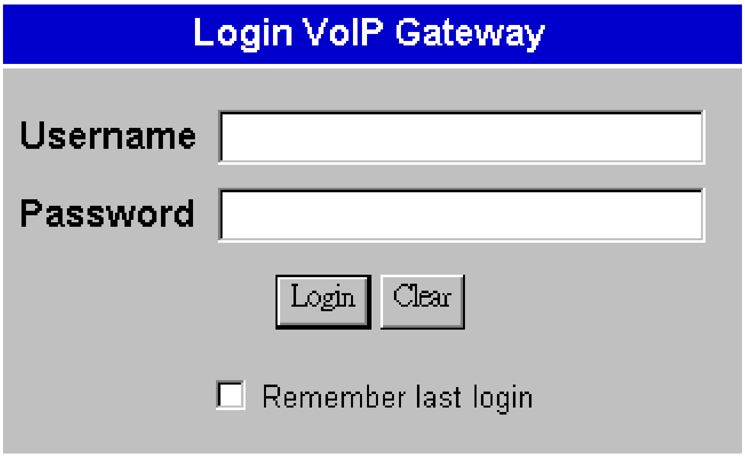
Pick up the phone and press **#198#** to reset back to factory defaults, and the VS211 will enter into POWER ON cycle. The PHONE LED indicators will be OFF for about 5 seconds and start flashing for 5 times. The POWER LED then will be lit constantly, and the PHONE LED will be OFF. If the PHONE LED keeps flashing, it indicates that VS211 has successfully registered in the SIP server.

8. Configurations by Web Browser

Login VoIP Gateway

You may enter the IP address from PC Web browser to configure VS211. For example, enter **http://192.168.123.1** from IE web browser to display login page as follows.

8.1. Please enter the default IP address **http://192.168.123.1** from PC Web browser.
The following Web page shall be displayed on PC. If you have difficulties accessing the Web page from the PC Web browser, the subnet IP of PC might be different from **192.168.123.xxx.** In this case, please refer to Chapter 11 for trouble shooting.



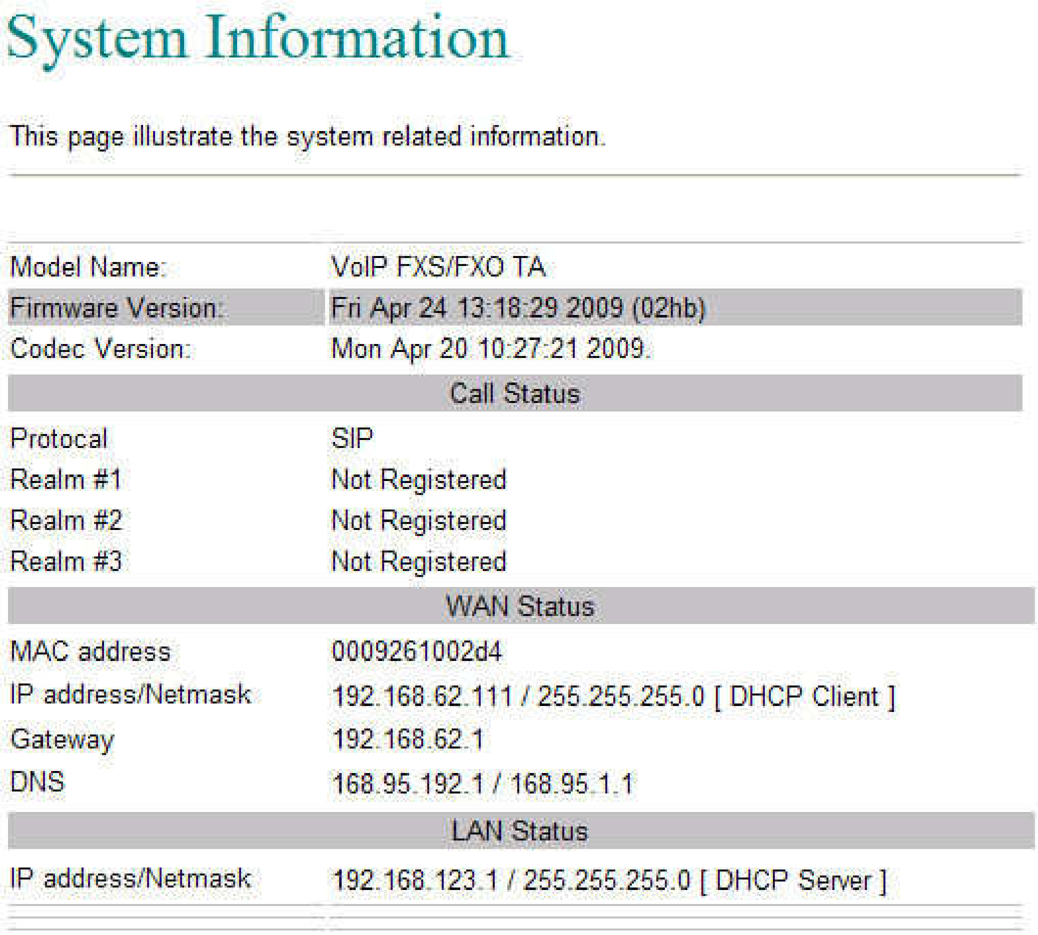
8.2. Please enter the username and password into the blank field. The default settings are:
Username: **root**
Password: **test**

8.3. Click the "**Login**" button will enter the management information page for system setup.
Note that whenever you change the setting in each Web page, please remember to click the "Submit" button in the page, and click the "Save" button to save into the non-volatile memory and click the "Reboot" button to activate the new settings.

System Information

8.4. You will see the system information such as firmware version, Codec, etc in this page.

8.5. You may click the button list at the left hand side to configure the VS211.



Model Name	Show device Model Name.
Firmware Version	Device Risc version, eg: Tue Jan 16 11:28:32 2007
Codec Version	Device DSP version, eg: Wed Dec 20 17:28:06 2006
Call Status	
Protocol	Show device VoIP protocol.
Realm #1	Show the first registry information status
Realm #2	Show the second registry information status
Realm #3	Show the third registry information status
WAN Status	
MAC address	WAN port of MAC address
IP address/Netmask	IP address/netmask
Gateway	Default router IP address
DNS	DNS IP address
LAN Status	
IP address/Netmask	Current LAN port IP address/netmask

Phone Book Settings

Phone Book

8.6. You may add/delete **Name (in numeric only)** up to maximum 140 entries in Phone book list.

8.7. To add a phone name, you need to enter the position, the name, and the phone URL. When you finished a new phone list, just click the “Add Phone” button.

8.8. To delete a phone name, please select the phone name then click “Delete Selected” button.

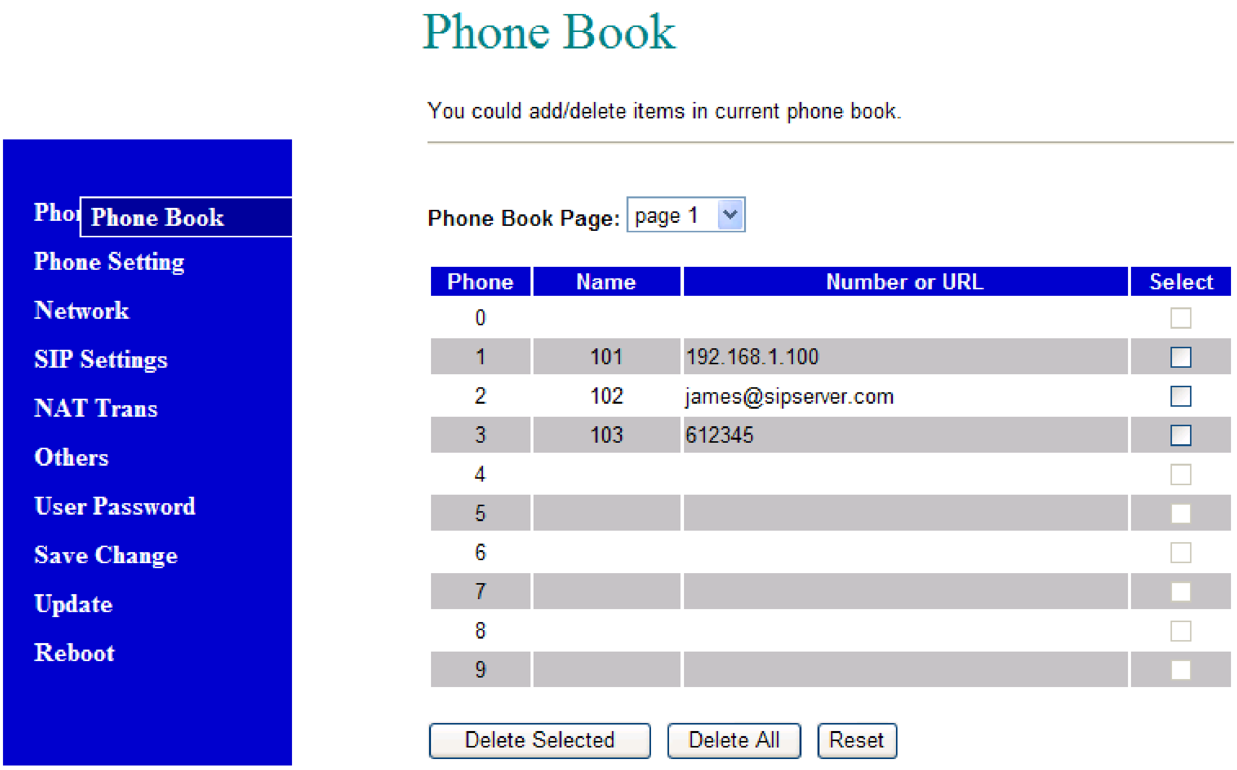
8.9. To delete all phone names, please click “Delete All” button.

8.10. After dialing a phone number, the TA will first match with the **Name** in the phone book. If matched, the TA will send out the corresponding URL. If not, the dialed phone number will be sent out.

8.11. **Example 1**: Name: **101**, URL: **192.168.1.100**
Press **101#** on telephone, and the phone at **192.168.1.100** will start ringing.

8.12. **Example 2**: Name: **102**, URL: **james@sipserver.com**
Press **102#** on telephone, and the TA will call the URL **james@sipserver.com**.

8.13. **Example 3**: Name: **103**, URL: **612345**
Press **103#** on telephone, and the TA will call the registered number **612345**.



Phone Settings

8.14. The subpages are as follows; Call Forward, SNTP, Volume, DND, Auto Answer, Caller ID, Dial Plan, Flash Time (or hook switch), Call Waiting, T.38 FAX over IP, Hot Line and Alarm settings.

Call Forward

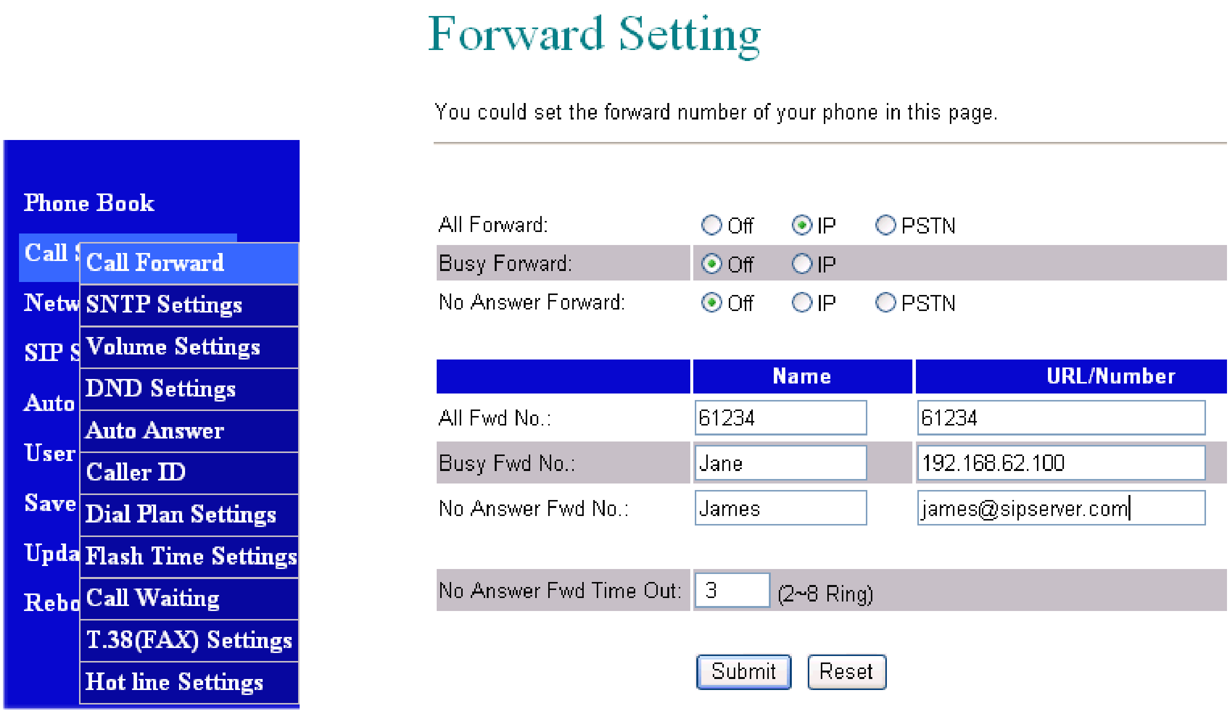
8.15. You can select the forward mode and enter the forward URL.

All Forward: All incoming call will forward to the URL or PSTN you choose.

Busy Forward: The incoming call will forward to the URL when the callee is busy.

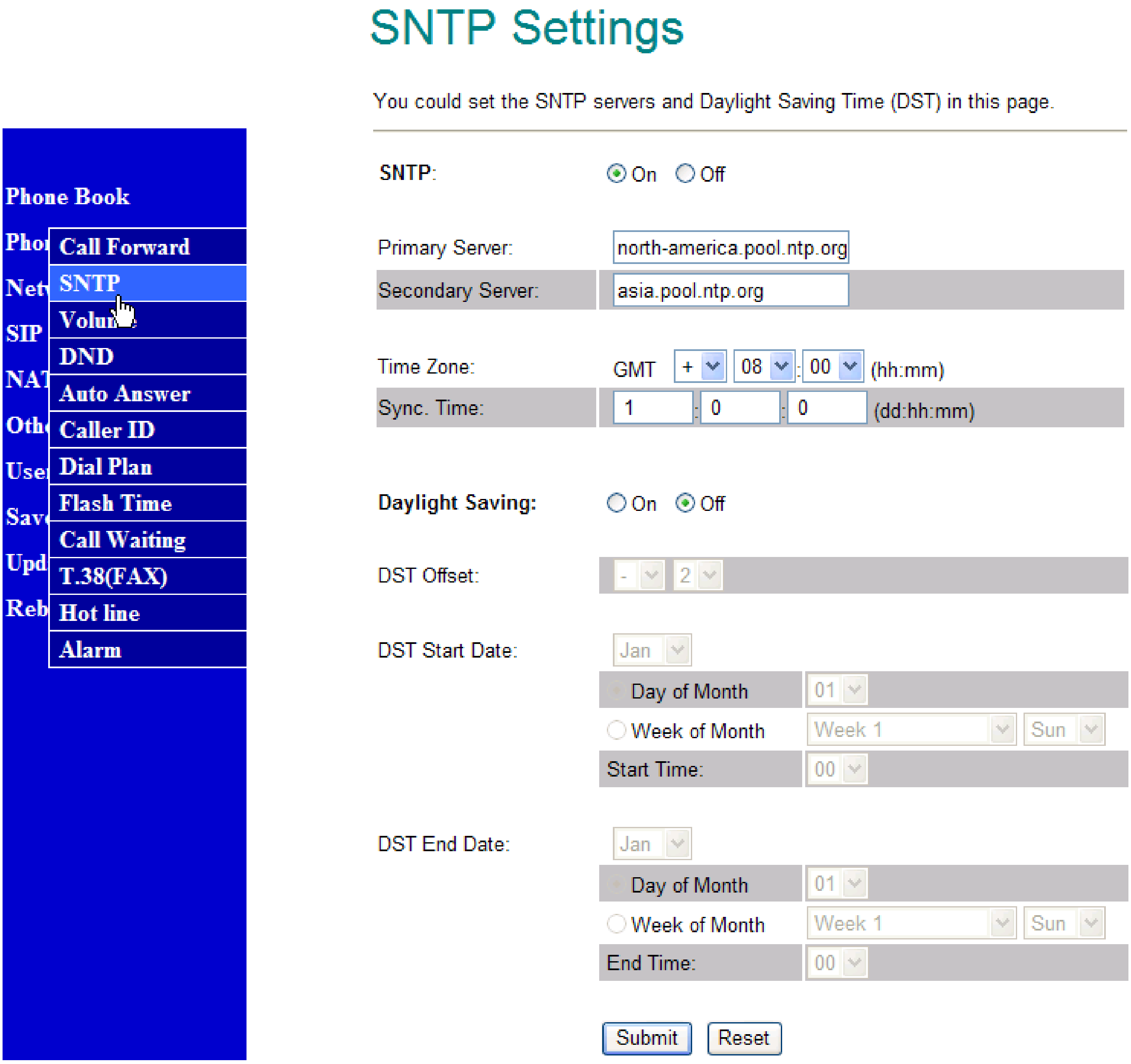
No Answer Forward: The incoming call will forward to the URLor PSTN when no answer.

8.16. You need to set the Time Out ring which will initiate No-Answer forwarding to the number you choose. When you finished the setting, please click the "Submit" button.



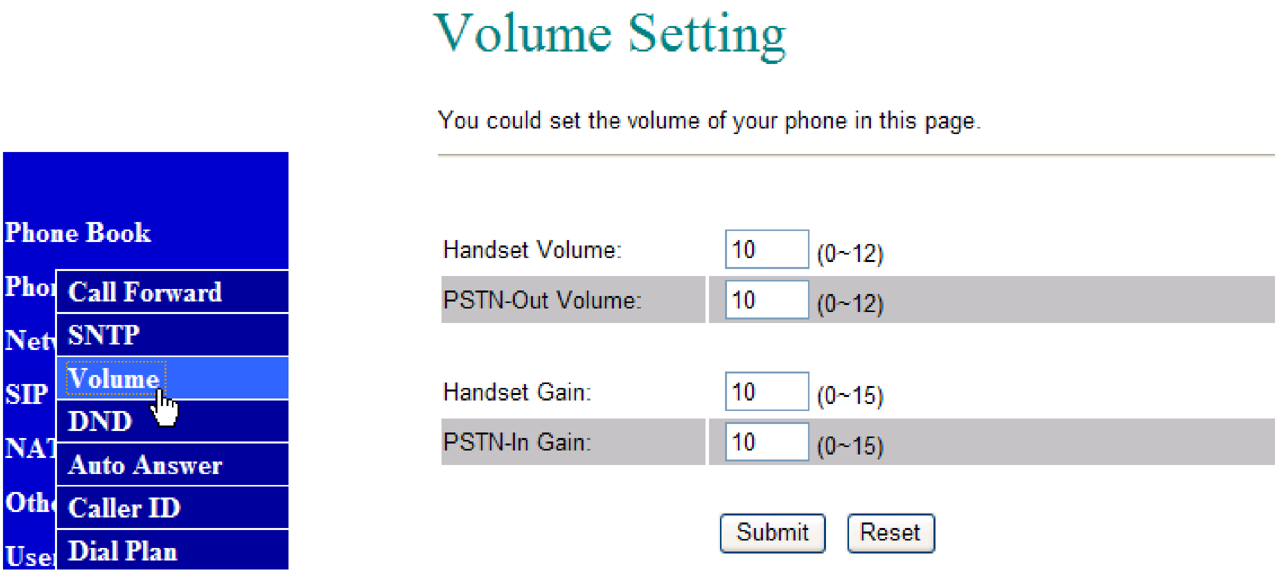
SNTP

8.17. You can setup the primary and second SNTP Server IP Address, to get the date/time information. You may also set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the "Submit" button.



Volume

8.18. You can setup the Handset Volume/Gain, PSTN-Out Volume, and PSTN-In Gain in this page. Handset Volume is to set the volume hearing from the handset. PSTN-Out Volume is to set the PSTN volume for you to hear. Handset Gain is to set the volume send out to the other side’s handset. PSTN-In Gain is to set the volume send out to the other side’s handset.



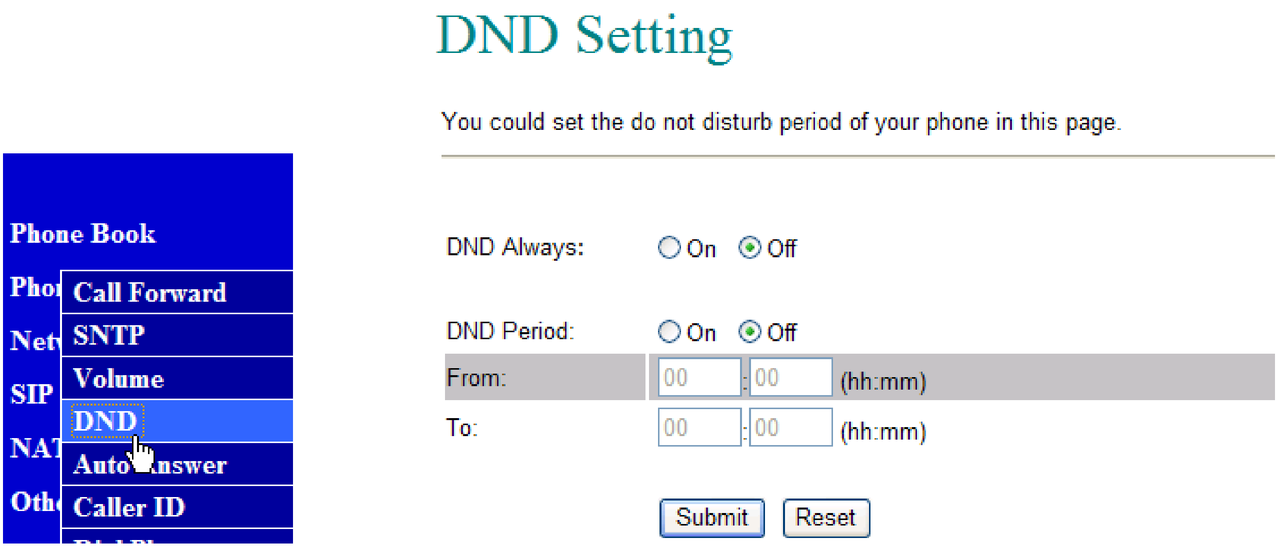
DND

8.19. You can configure the DND (Do Not Disturb) setting to keep the phone silence. You can choose either DND Always or a DND period.

8.20. DND Always: All incoming call will be blocked until this feature is disabled.

8.21. DND Period: Set a time period and the phone will be blocked during the time period. If the time in “From” is greater than that in “To” time, the DND time will be from Day 1 to Day 2.

8.22. After you finished the setting, please click the “Submit” button.



Auto Answer

8.23. Auto Answer function can be used to relay calls between VoIP and PSTN. When the ring count exceeds the number set in Auto Answer Counter, the Auto Answer function will be activated when one of the 4 Auto Answer modes is chosen.

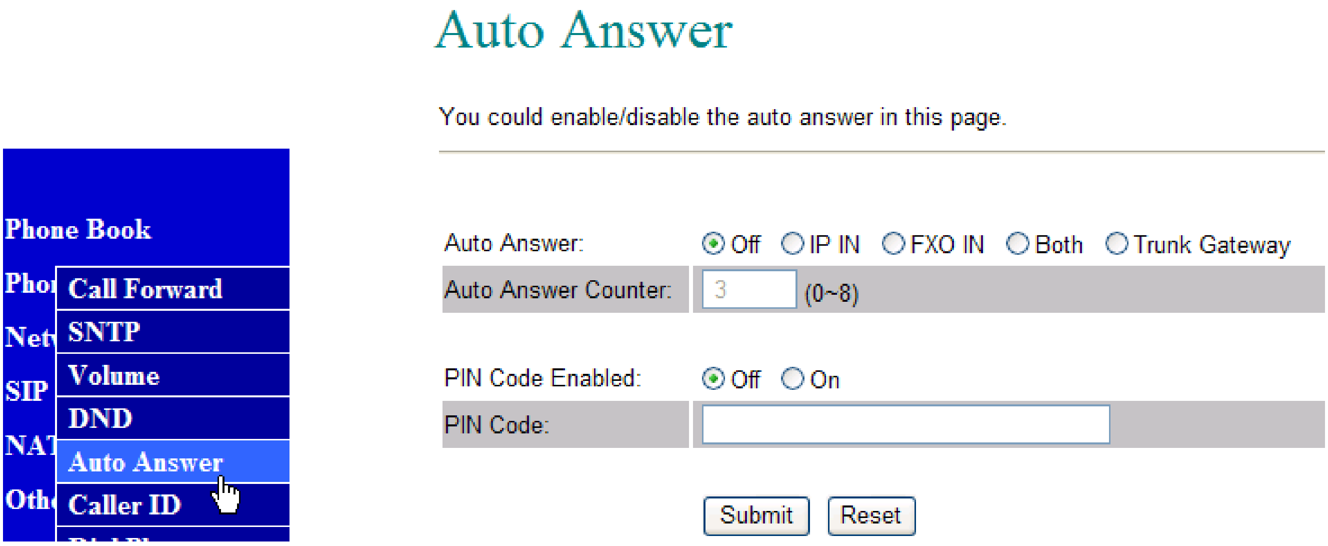
IP IN: When the incoming call is from the Internet, the FXO port will answer with a PSTN dial tone and wait the caller to dial another PSTN phone number.

FXO IN: If the incoming call is from PSTN, the VS211 will answer with a dial tone and allow caller to dial to another VoIP number.

Both: Auto Answer function are enabled for Both IP IN and FXO IN.

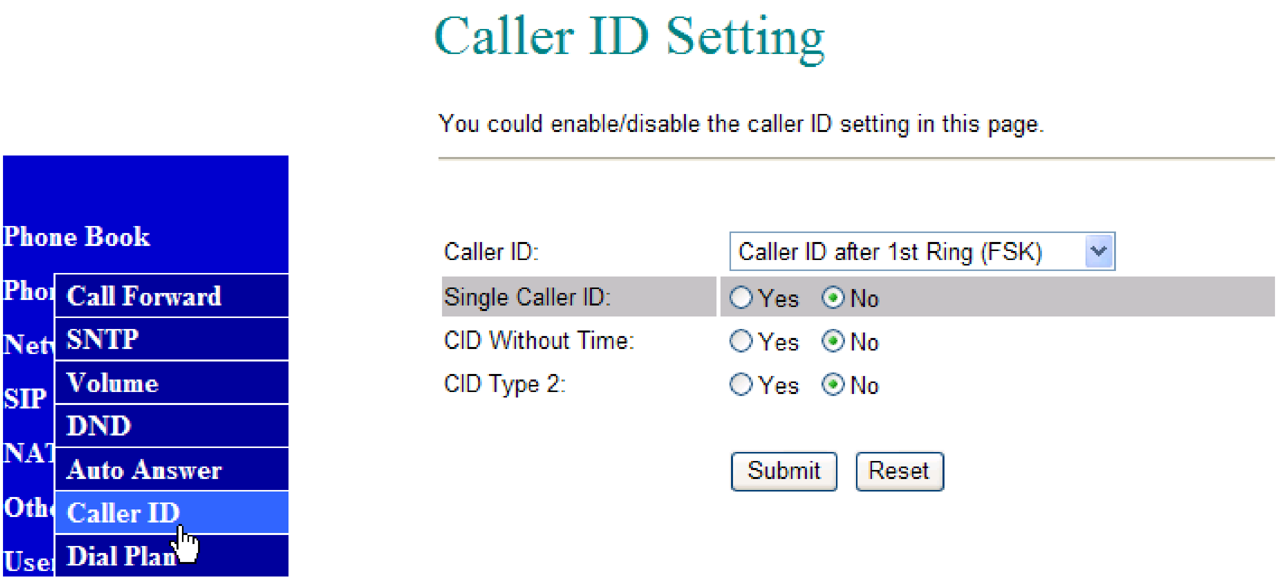
Trunk Gateway: When the incoming call is from the Internet, the FXO port will relay the calling number directly to PSTN without intermediate dial tone. This works only for incoming call from VoIP.

PIN Code: PIN Code is used to prevent from call piracy. If the PIN Code is enabled, you will hear double beep tone after calling. The caller needs to enter the right PIN code and the postfix "#" to get the PSTN dial tone. Incorrect PIN Code will result in call disconnect.



Caller ID

8.24. You may show caller ID in your PSTN Phone or IP Phone by selecting "Yes" in Single Caller ID, and the desired Caller ID option for either FSK or DTMF. After you finish the setting, please click the "Submit" button.



Dial Plan

8.25. Dial plan and auto dial timer settings can be set in this page. The dial plan allows you to map the dialing into an easy-to-remember phone number system. The auto dial timer specifies the elapse time between the dialing digits. When Drop prefix is ON and the dialing prefix is matched, the prefix will be dropped and replaced by the rule digits and followed by the rest of dialing digits. When Drop prefix is OFF and the dialing prefix is matched, the rule digits will be added before the dialing digits in accord with the settings

Routing to: For selection of FXO, the call will be routed from FXS port to FXO port (PSTN), when the dialing number matches the Routing rule. If not matched, the call will be routed to VoIP. For selection of IP, the call will be routed from FXS port to VoIP if matched, and vice versa.

Routing rule: This defines the routing rule from Telephone (FXS) to either PSTN line or VoIP. This is convenient for dial plan without using "**0***" function key.

8.26. Symbol Representations:

Symbol	Representations
x or X	0,1,2,3,4,5,6,7,8,9
+	or
D	It "Drop" the matched dialing prefix number. This is for routing rule only.

Example 1: Route to: **FXO**, Routing rule: **D02+0800**

a) Pressing **02xxxxxxxx** will result in dialing out to PSTN port **xxxxxxxx**.

b) Pressing **0800xxxxxx** will result in dialing out to PSTN port **0800xxxxxx**.

c) Other dialing number will route to VoIP and follow the rules of Drop Prefix.

Example 2: Drop Prefix: **No**, Replace rule 1: **002, 8613+8862**

a) Pressing **8613xxx** will result in dialing out **0028613xxx**.

b) Pressing **8862xxx** will result in dialing out **0028862xxx**.

Example 3: Drop Prefix: **Yes**, Replace rule 2: **006, 002+003+004+005+007+009**

a) Pressing **002xxx** will result in dialing out **006xxx**.

b) Pressing **003xxxx** will result in dialing out **006xxxx**.

Example 4: Drop Prefix: **No**, Replace rule 3: **009, 12**

a) Pressing **12xxx** will result in dialing out **00912xxx**.

Example 5: Drop Prefix: **No**, Replace rule 4: **007, 5xxx+35xx+21xx**

a) Pressing **5xxx** will result in dialing out **0075xxx**.

b) Pressing **534** will result in dialing out **534** (not matched for the rest 3 digits).

c) Pressing **35xx** will result in dialing out **00735xx**.

d) Pressing **356** will result in dialing out **356** (not matched for the rest 2 digits).

e) Pressing **35668** will result in dialing out **35668** (not matched for the rest 2 digits).

Example 6: Dial Now: ***xx+#xx+11x+1[237]xx+xxxxxxxx**

a) Pressing ***00**, ***01**, ***02 .. *99** will result in dialing out the same ***xx immediately.**

b) Pressing **#00**, **#01**, **#02 .. #99** will result in dialing out the same **#xx immediately.**

c) Pressing **110**, **111**, **.. 119** will result in dialing out the same **11x immediately**.

d) Pressing **1200**, **..** ,**1299**, **1300**, **..**,**1399**, **1700**, **..**, **1799** will result in dialing out the same **1[237]xx immediately**.

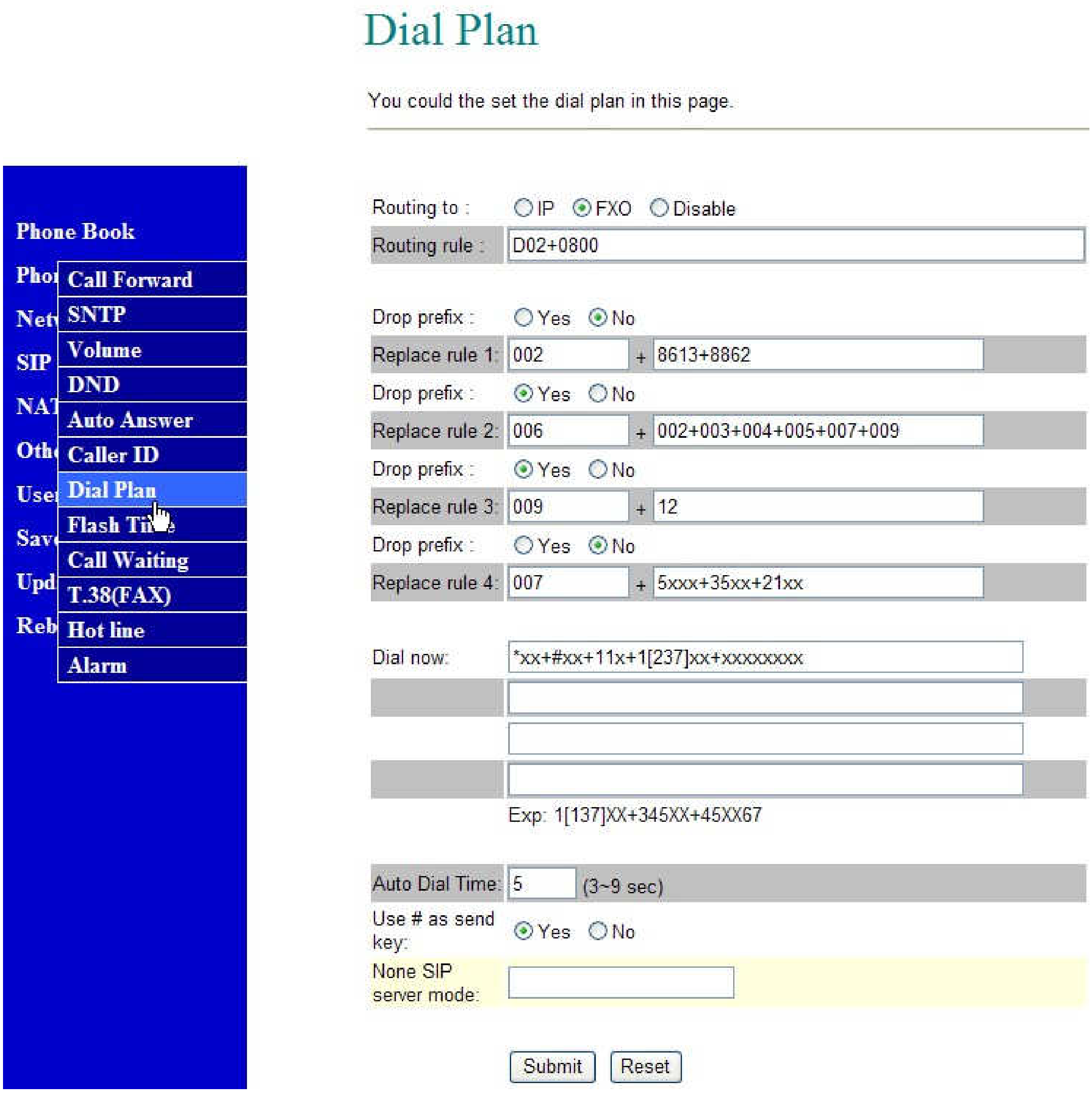
e) Pressing **12345678** (8 digits) will result in dialing out **12345678 immediately**. This implies that the phone numbers with 9 or more digits are prohibited.

8.27. Auto Dial Timer: The inter-digit timer. Default is 5 seconds.

8.28. None SIP Server Mode : When **SIP Settings > Service Domain** were left blank (that means no SIP registration available), you may enter a specific IP address (e.g. SIP gateway IP address) in this field for all dialing numbers sending to this IP address. This can be used when only SIP gateway is available. Note when at least one SIP server is successfully registered in SIP settings, this will be automatically disabled.

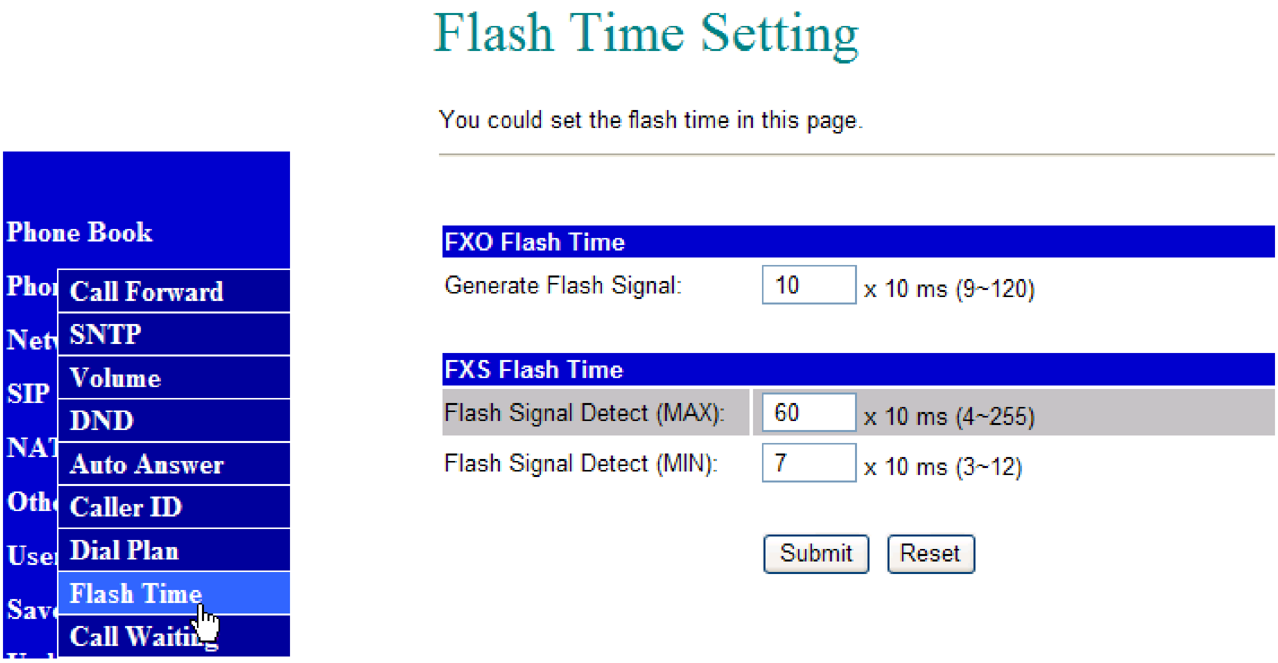
8.29. When you finish the setting, please click the Submit button.

8.30. Click the Save button. The changes you have made will be saved and the VS211 will reboot automatically.



Flash Time

8.31. You can set the flash time duration for the telephone flash key or hook switch in this page. The telephone flash key is used to switch to the other phone line or HOLD, and is quite useful for the 3-way conference call and the call waiting function. When you finish the setting, please click the “Submit” button.



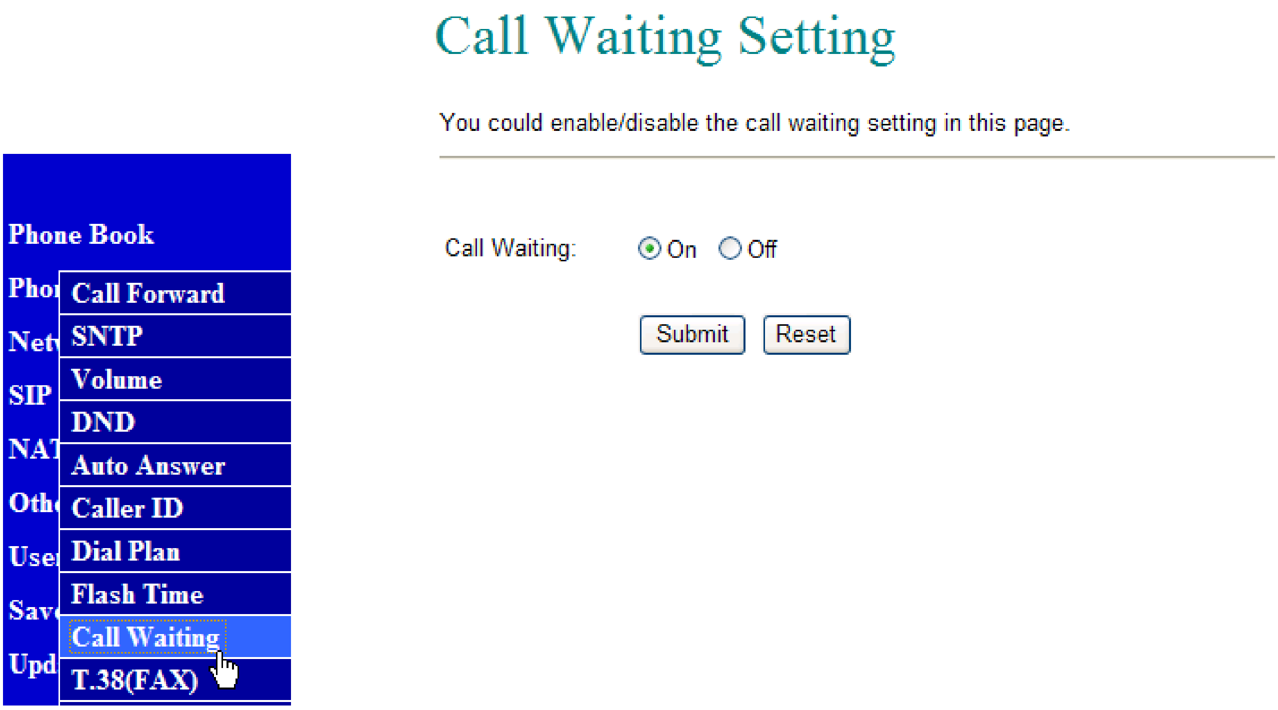
Notes:

1 Flash Signal Detect (MAX): Maximum Flash Hook duration (unit:10ms)

2 Flash Signal Detect (MIN): Minimum Flash Hook duration (unit:10ms)

Call Waiting

8.32. You can enable the call waiting function in this page. It allows answering another coming call by pressing flash key while holding the current call. You may switch back to previous call by pressing flash key again. When you finish the setting, please click the "Submit" button.

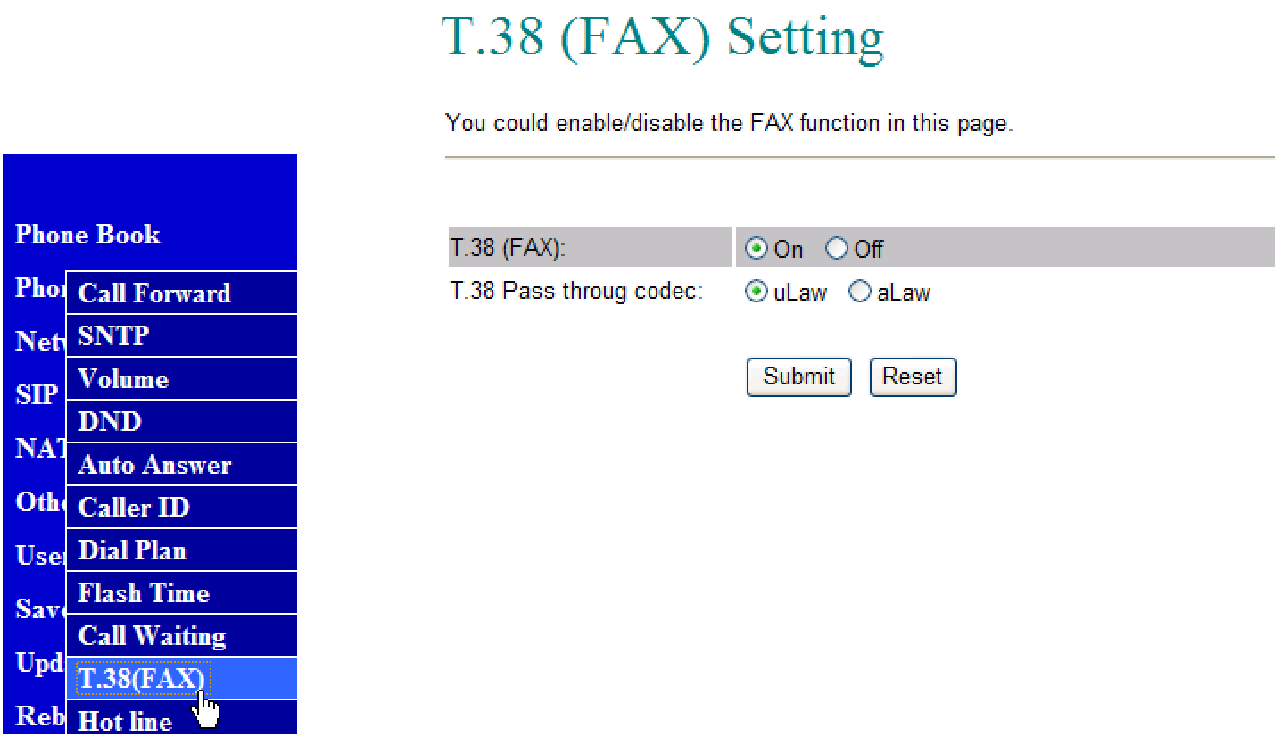


Call Transfer function

The call transfer function allows users to answer an incoming call and to hold the current call by pressing flash key, and then transfer the current call to the desired party by dialing the desired party number ended with # key. The call transfer function is exclusive with call waiting function. You may enable call transfer function by disabling the call waiting function (**#139#**), or disable call transfer function by enabling the call waiting function (**#138#**).

T.38 (FAX)

8.33. T.38 function can be used for FAX transmission over IP. Note that T.38 function must be enabled for both side of FAX over IP. You may enable or disable the T.38 function. T.38 Pass through codec for u-Law or a-Law, and make sure your SIP server/gateway also supports this T.38 function. When you finish the setting, please click the Submit button.

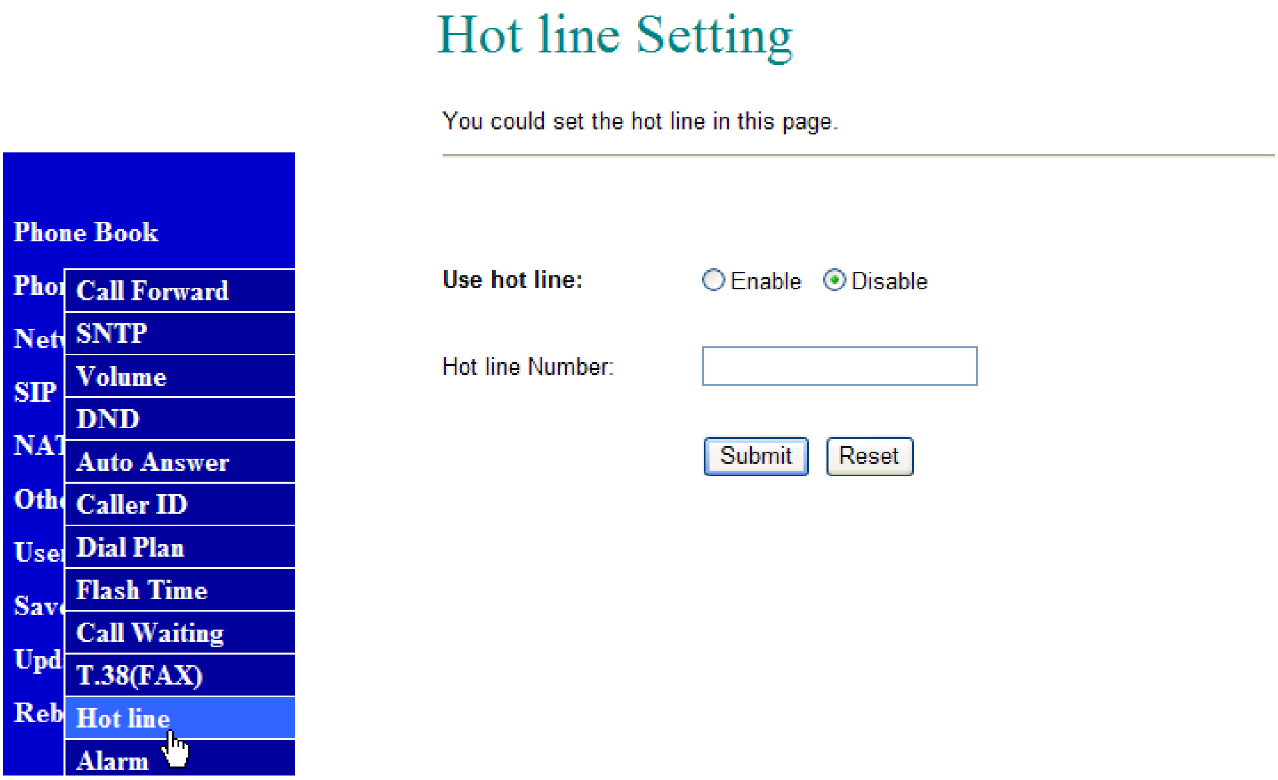


Hot Line

8.34. The Hot Line mode allows to making a direct call at the Phone Number or IP stored in this page without dialing. Hot-Line Mode is very convenient for IP calling to Public Switching Telephone Network (PSTN) number through FXO Gateway

8.35. When the Hot Line mode is enabled, you just pick up the phone and the VS211 will call the party directly to the preset IP (or URL) address. The default for Hot Line mode is disabled.

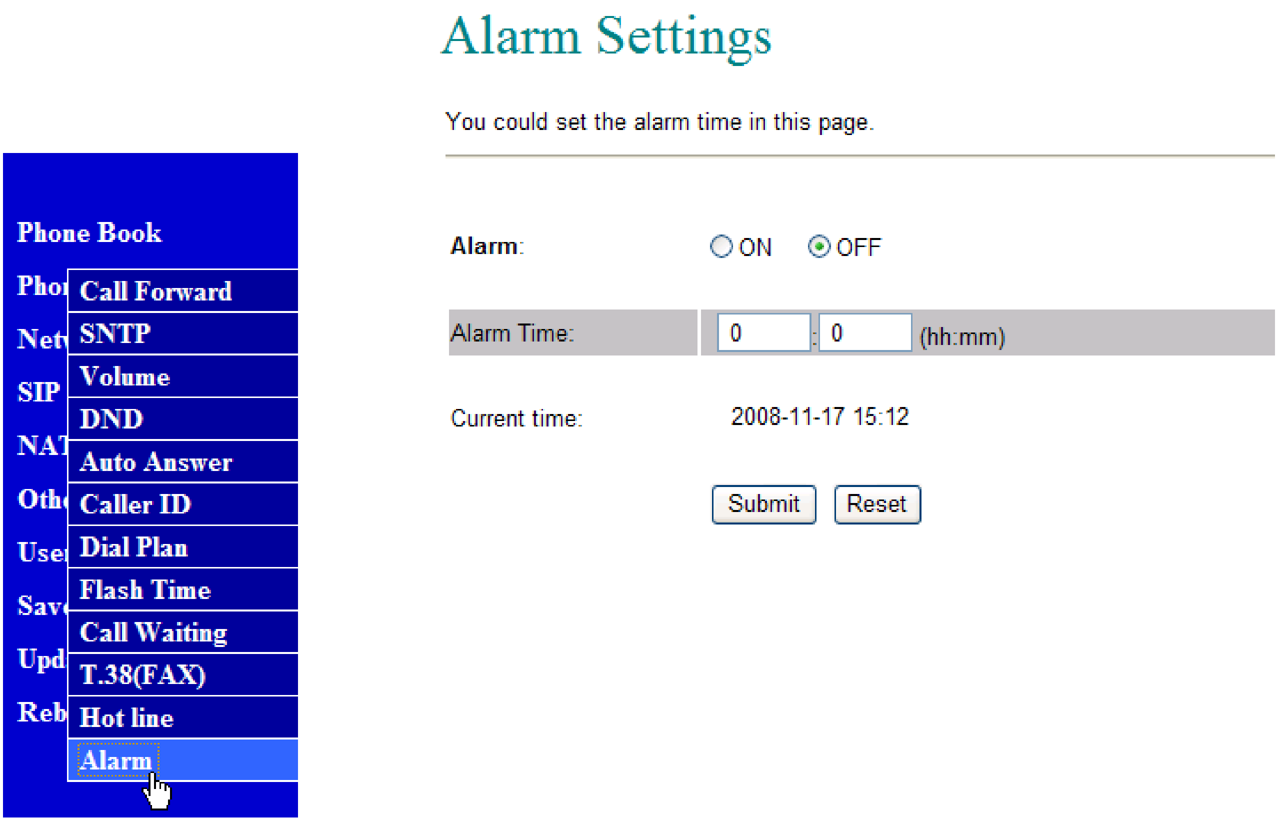
8.36. You need to “Enable” and click the “Submit” button and reboot to activate the function.



Pick up the phone. In 1-2 seconds (default wait time), the VS211 will automatically call the preset IP or phone number.

Alarm

8.37. You can configure the Alarm setting in this page..



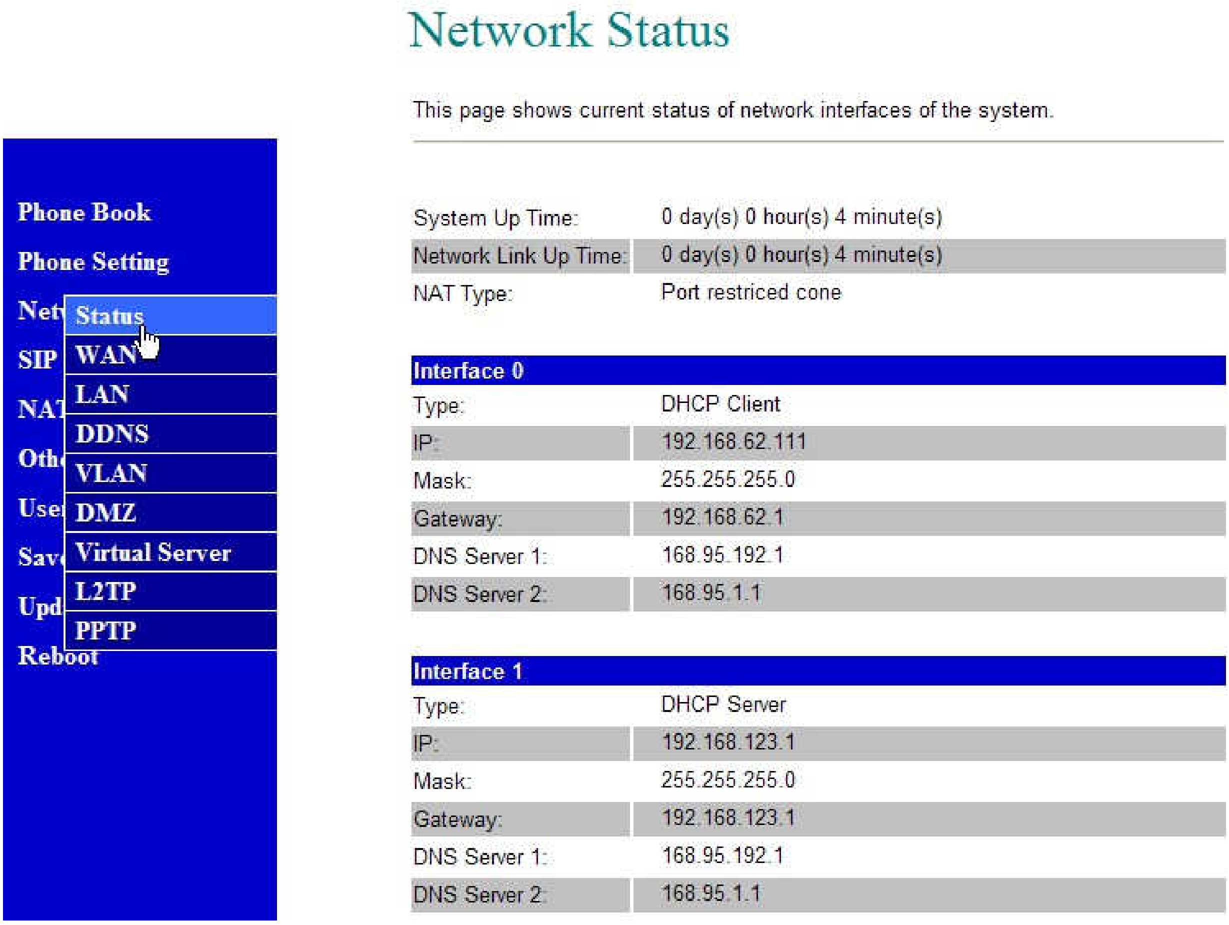
Network

8.38. VS211 is equipped with an embedded NAT router between WAN and LAN ports to meet the IP Network requirements. If you have an external NAT router, then you may select Bridge mode in WAN setting. Thus the two WAN and LAN Ethernet ports will be bridged and transparent. Otherwise, you may select NAT mode to enable embedded NAT and go on DDNS settings. The default is for NAT mode.

Network Status

8.39. You can check and show the current Network settings in this page.

Interface 0 is for WAN port Status and **Interface 1** for LAN port Status.



WAN

8.40. The WAN setting is used to configure the WAN port connects to the ADSL Modem/Router.

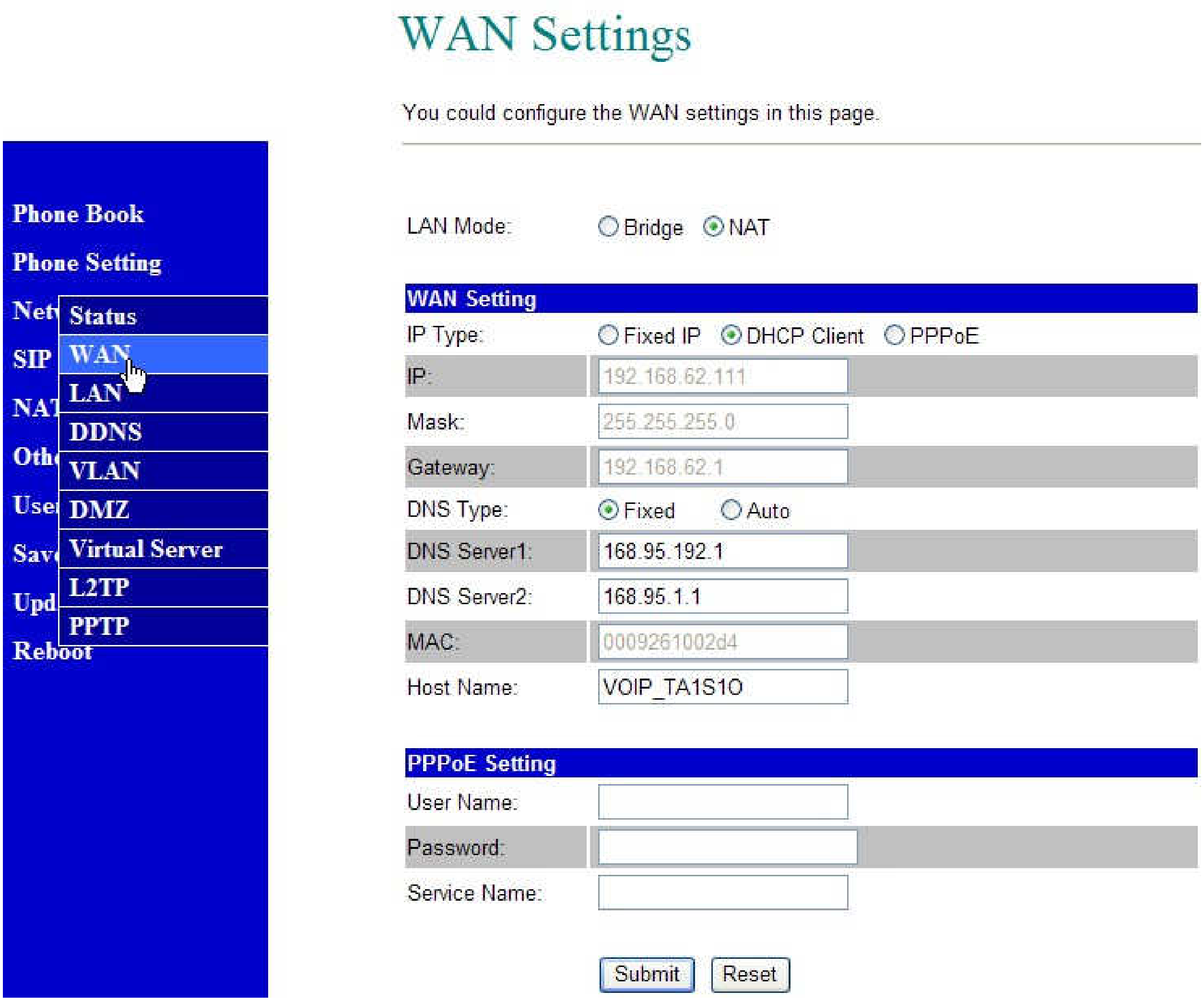
8.41. The default setting is for NAT mode to enable the embedded NAT router between the WAN port and LAN port. You may select Bridge Mode if you need NOT use the embedded NAT router. When setting to Bridge Mode, only the WAN settings will get effective and the LAN settings will be ignored.

8.42. There are three selections for WAN IP Type: Fixed IP, DHCP Client, and PPPoE modes. This WAN setting is for the WAN port when set in NAT mode. The default is at DHCP Client.

8.43. For Fix IP Mode, please make sure the IP address. Net Mask, Gateway, and DNS settings are suitable in your current network environment.

8.44. For PPPoE Mode, you have to enter correct username and password to get the IP address from your Internet Service Provider.

8.45. When you finish the settings, please click the Submit button.

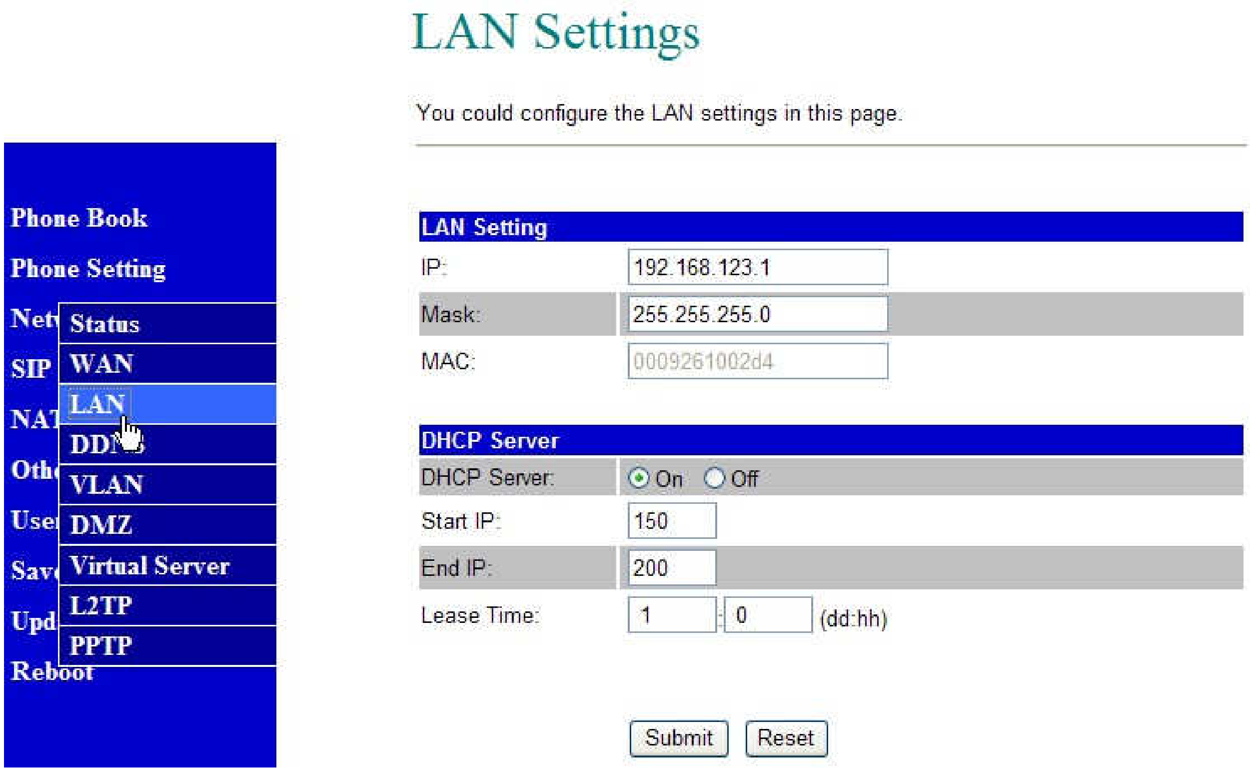


LAN

8.46. The default IP address is 192.168.123.1, with Net Mask 255.255.255.0., and DHCP Server enabled. The range of IP addresses for DHCP is from 150 to 200.

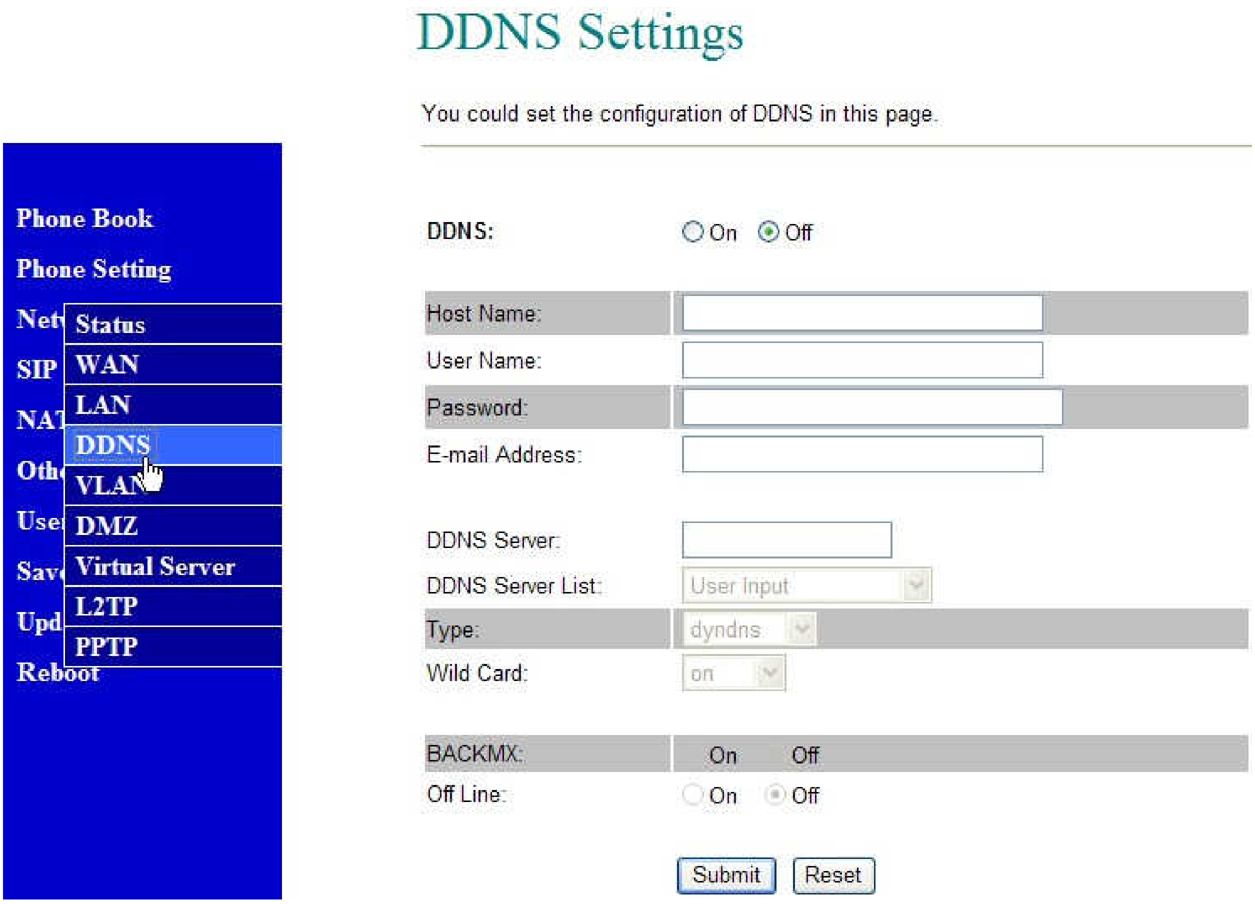
8.47. Connect your PC to the LAN port, set your PC as DHCP mode, and the PC will automatically get an IP address from the VS211.

8.48. When you finish the settings, please click the Submit button.

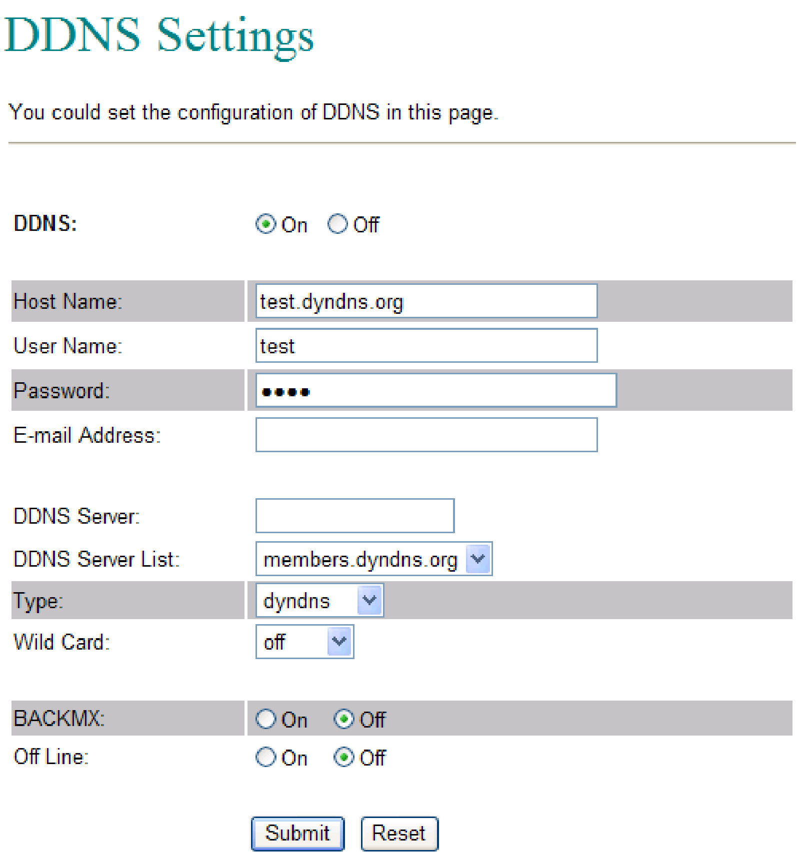


DDNS

8.49. You need to have a DDNS account before configuring the DDNS setting. Usually, most of the VoIP applications are working with a SIP Proxy Server. Nonetheless, you may have a DDNS account with a public IP address, and others can call you via the DDNS account. When you finish the setting, please click the Submit button.


DDNS Settings
You could set the configuration of DDNS in this page.
Phone Book
Phone Setting
Status
WAN
LAN
DDNS
VLAN
DMZ
Virtual Server
L2TP
PPTP
Reboot
DDNS:
On
Off
Host Name:
User Name:
Password:
E-mail Address:
DDNS Server:
DDNS Server List:
User Input
Type:
dyndns
Wild Card:
on
BACKMX:
On
Off
Off Line:
On
Off
Submit
Reset

Example:


DDNS Settings
You could set the configuration of DDNS in this page.
DDNS:
On
Off
Host Name:
test.dyndns.org
User Name:
test
Password:
E-mail Address:
DDNS Server:
DDNS Server List:
members.dyndns.org
Type:
dyndns
Wild Card:
off
BACKMX:
On
Off
Off Line:
On
Off
Submit
Reset

VLAN

8.50. The VLAN setting is for VoIP packets related to LAN port.

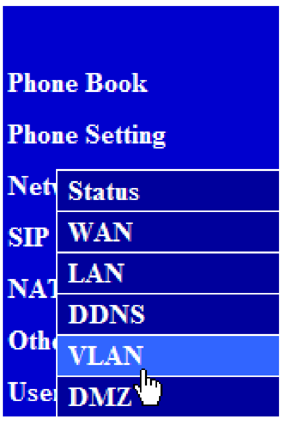
8.51. VLAN Packets: If you enable VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.

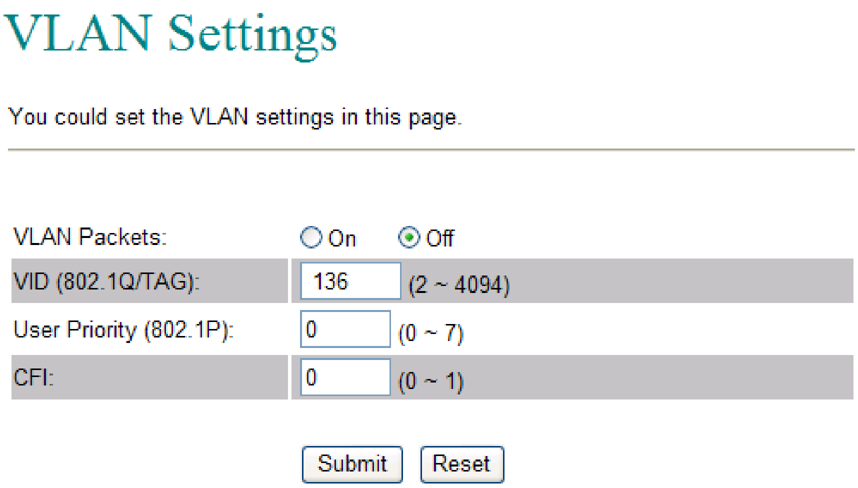
8.52. VID: Please set your VID in accordance with your service provider.

8.53. User Priority: Defines user priority with eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually, this will be defined by your service provider.

8.54. CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.

8.55. When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the TA's IP address and the same VID will be accept by the TA. If the incoming packets with TA's IP address but different VID then the packets will be discarded by the TA. The Other incoming packets with different IP address will go through the WAN port to the LAN port.





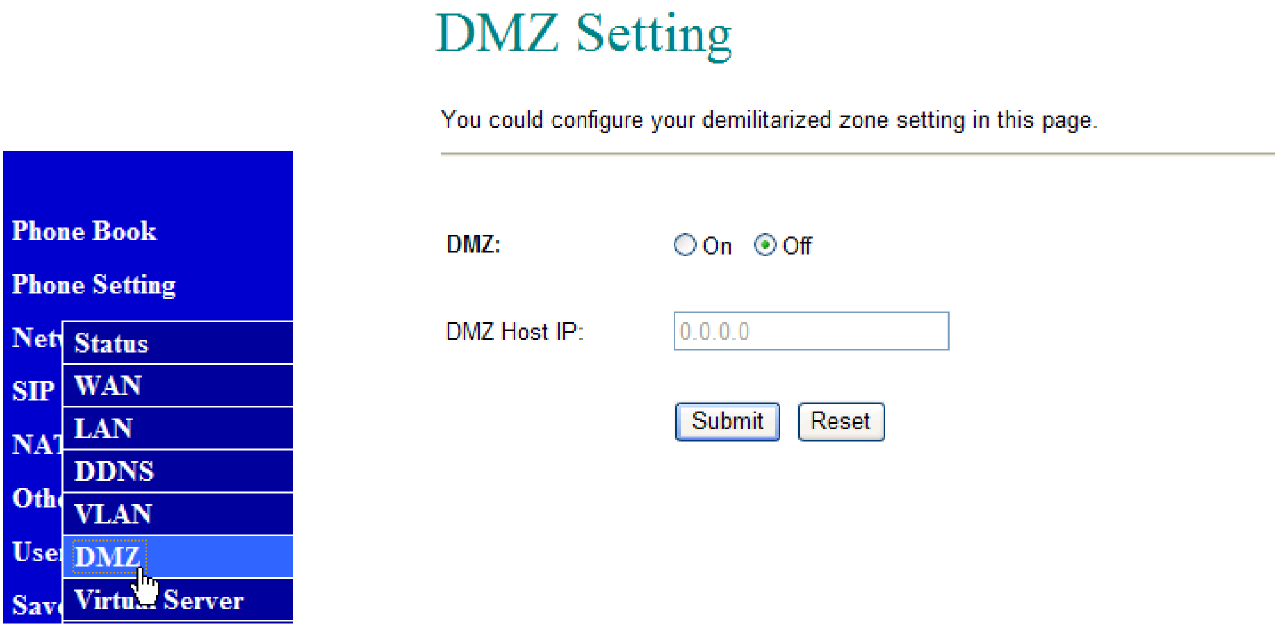
Notes:

When WAN port set to NAT Mode with VLAN enabled, please make sure the PC or network device on LAN port must support the same VLAN function to pass through WAN port.

When WAN port set to Bridge Mode with VLAN enabled, the VLAN will only function on ATA. The packets from LAN port will pass through WAN without VLAN function.

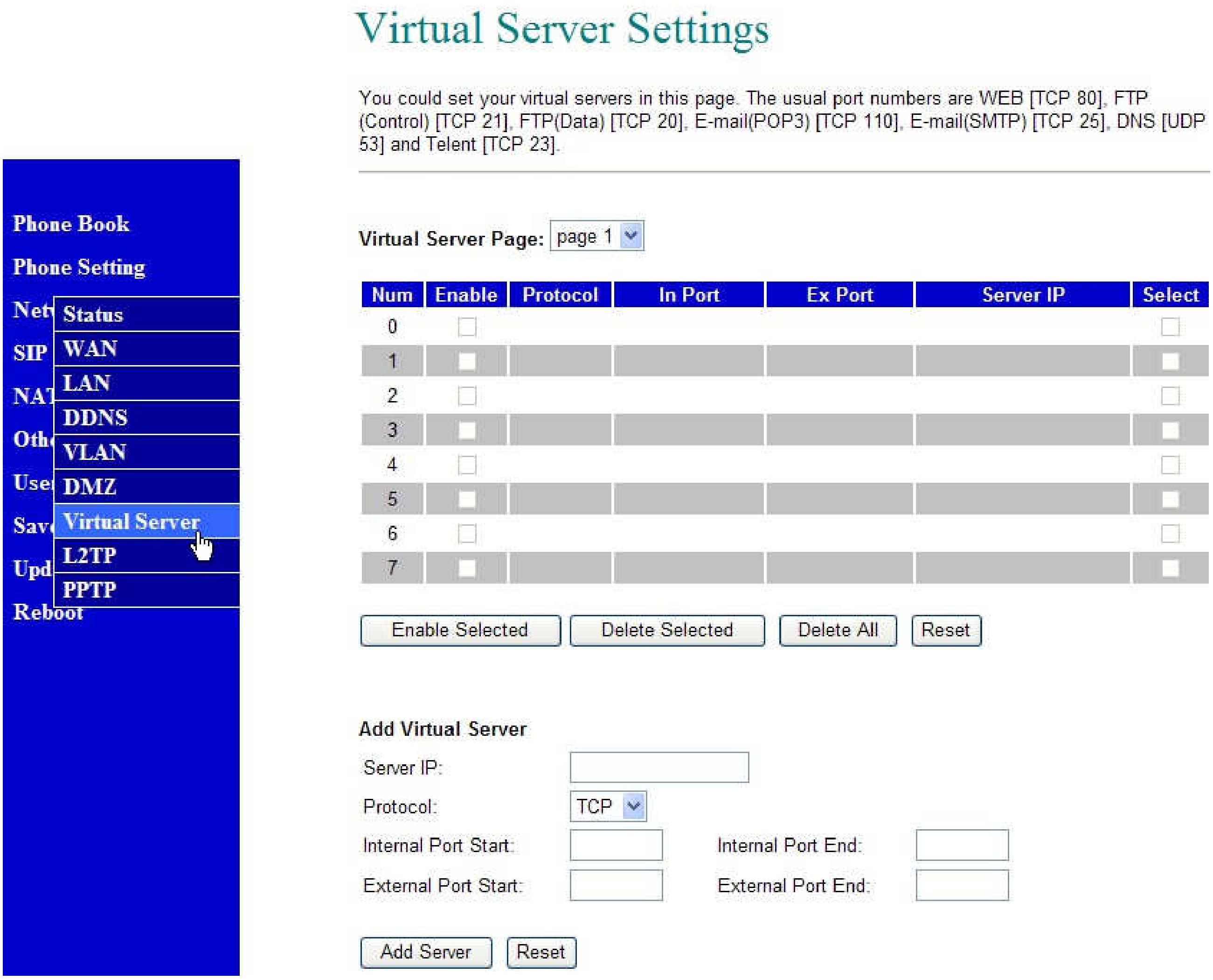
DMZ

8.56. The DMZ can be enabled/disabled and configured in this page



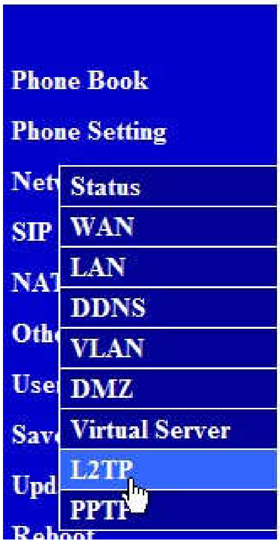
Virtual Server

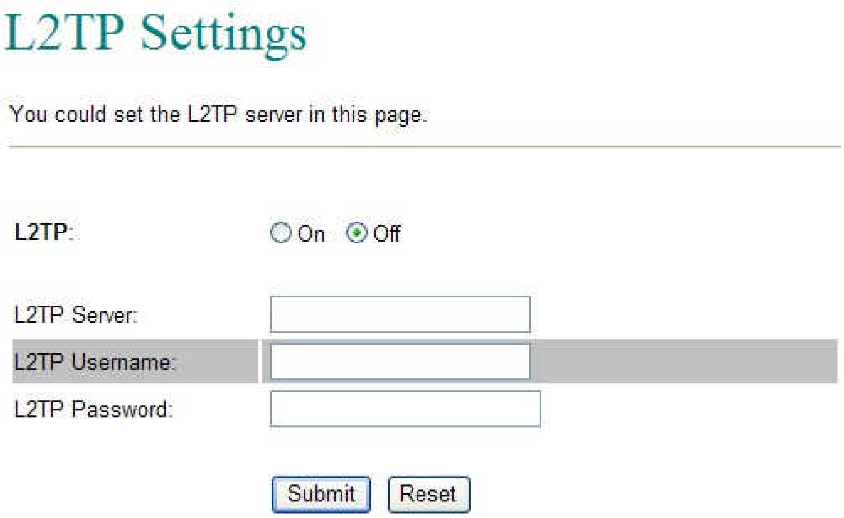
8.57. The Virtual Server IP and Port numbers can be configured in this page.



L2TP

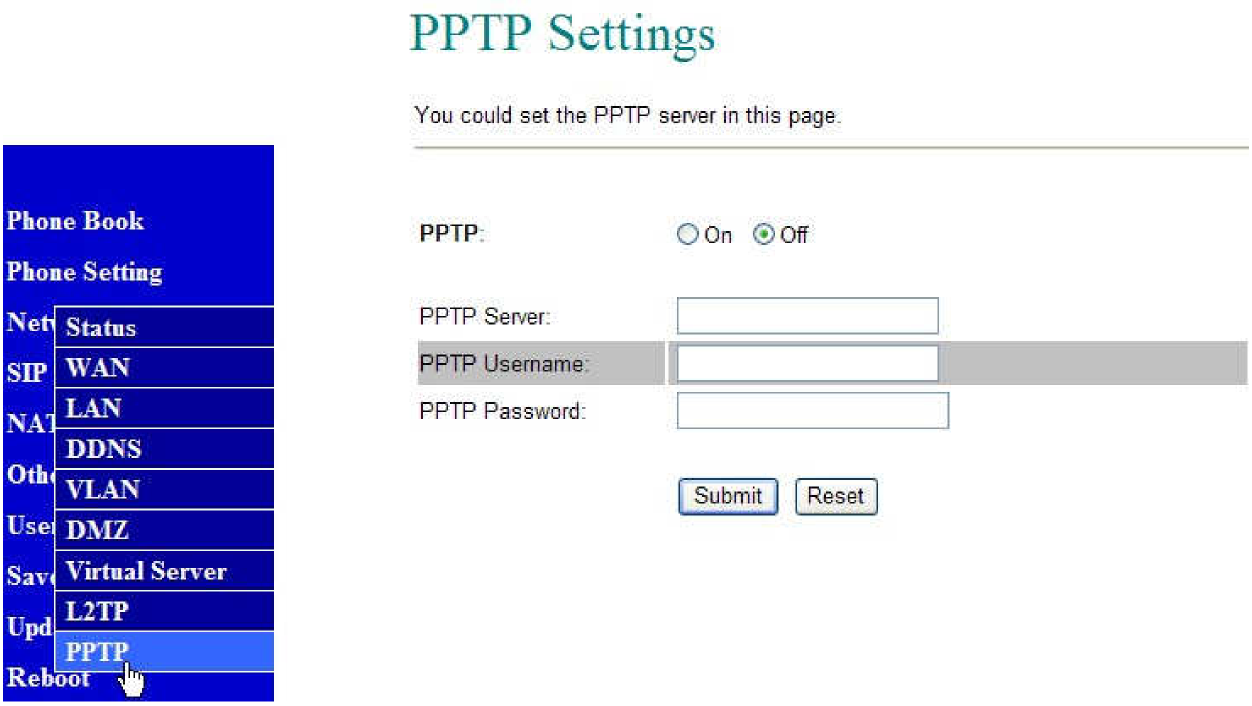
8.58. The Layer 2 Tunneling Protocol (L2TP) Server can be set ON/OFF in this page. This L2TP can be used to traverse the firewall when used with Virtual Private Network (VPN) applications.





PPTP

8.59. The Point-to-Point Tunnel Protocol (PPTP) Server can be set ON/OFF in this page. This PPTP can be used to penetrate the firewall when used with Virtual Private Network (VPN) applications.



SIP Settings

8.60. You can setup the Service Domain, Port Settngs, Codec Settings, RTP Setting, RPort Setting and Other Settings for SIP Proxy Server registrations in this page.

Service Domain

8.61. You may register up to three SIP Servers for three Realms in the VS211. You can receive the incoming calls from all the three SIP Servers. For outgoing calls, you may select the registration SIP server first, and then call the associated registration phone number.

8.62. To select SIP Server **1** (default), please pickup the phone, press **1***, then hangup.

8.63. To select SIP Server **2** (or **3**), please pickup the phone, press **2*** (or **3***), then hangup.

8.64. Click "**Active**" ON to enable the Service Domain, then enter the following items:

8.65. **Display Name:** enter the name you want to display.

8.66. **User Name:** enter the User Name given by your ITSP.

8.67. **Register Name:** enter the Register Name given by your ITSP.

8.68. **Register Password:** enter the Register Password given by your ITSP.

8.69. **Domain Server:** enter the Domain Server given by your ITSP.

8.70. **Proxy Server:** enter the Proxy Server given by your ITSP.

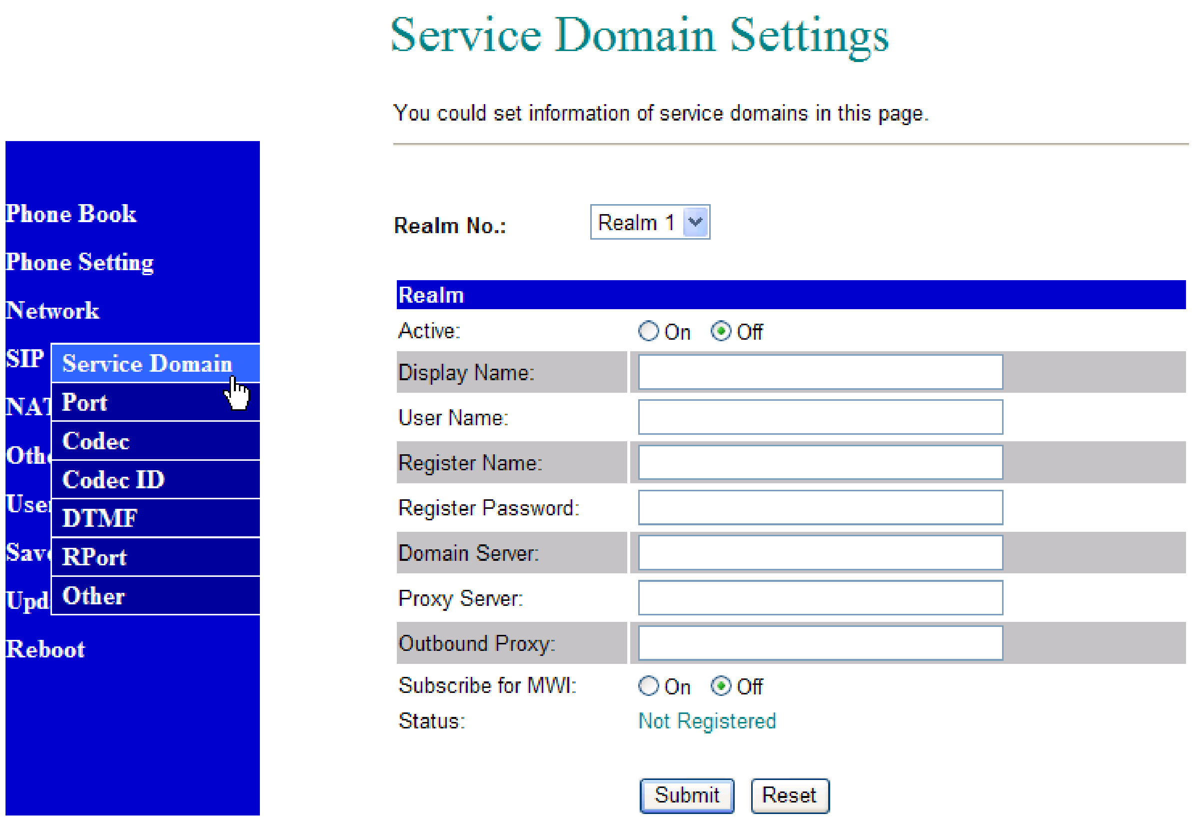
8.71. **Outbound Proxy:** enter the Outbound Proxy of ITSP. If not provided, you may skip this.

8.72. **Register Period:** enter the Register Period in minute given by your ITSP.

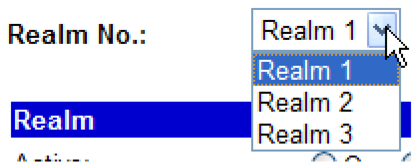
8.73. When it shows "**Registered**" in the Register Status, it indicates a successful registration to the ITSP, and the "**PHONE**" LED will start flashing. The VS211 is then ready for VoIP call

8.74. If you have more than one SIP account, please follow the steps to register to other ITSPs..

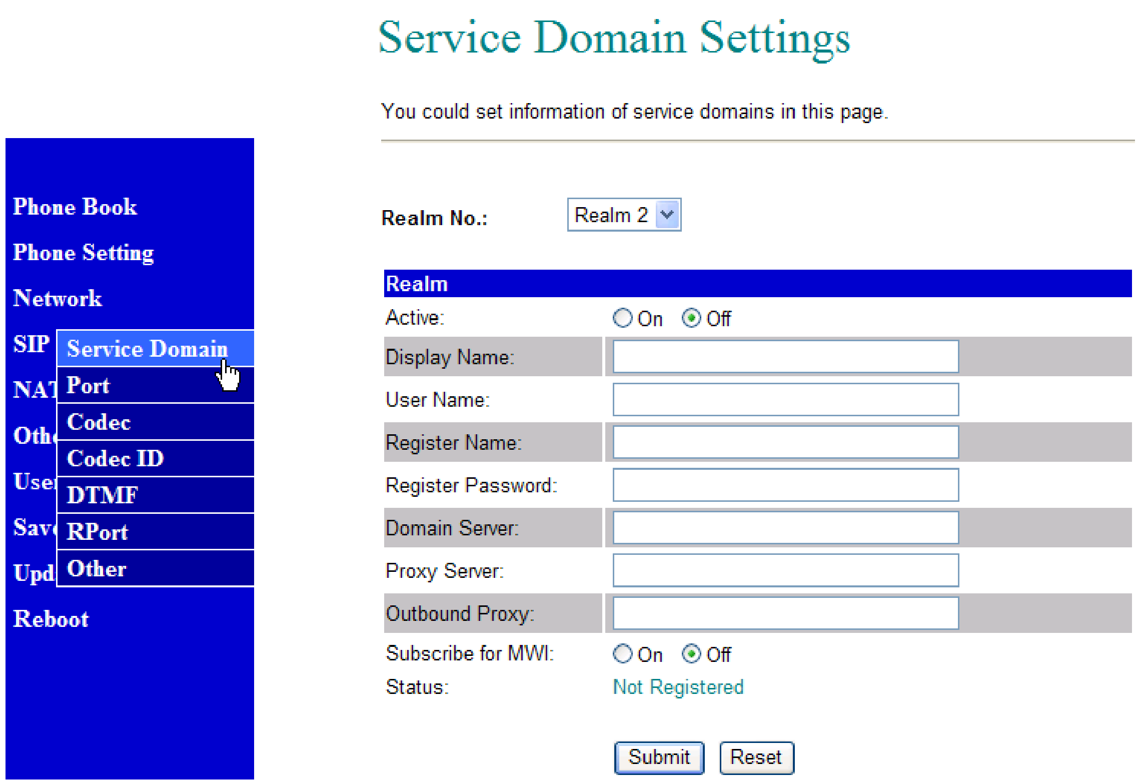
8.75. After you finish the setting, please click the "Submit" button.



Service Domain Settings
You could set information of service domains in this page.
Phone Book
Phone Setting
Network
SIP
Service Domain
Port
Codec
Codec ID
DTMF
RPort
Other
Reboot
Realm No.:
Realm 1
Realm
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Subscribe for MWI:
On
Off
Status:
Not Registered
Submit
Reset



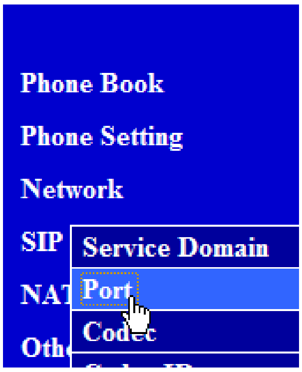
Realm No.:
Realm 1
Realm 1
Realm 2
Realm 3
Realm

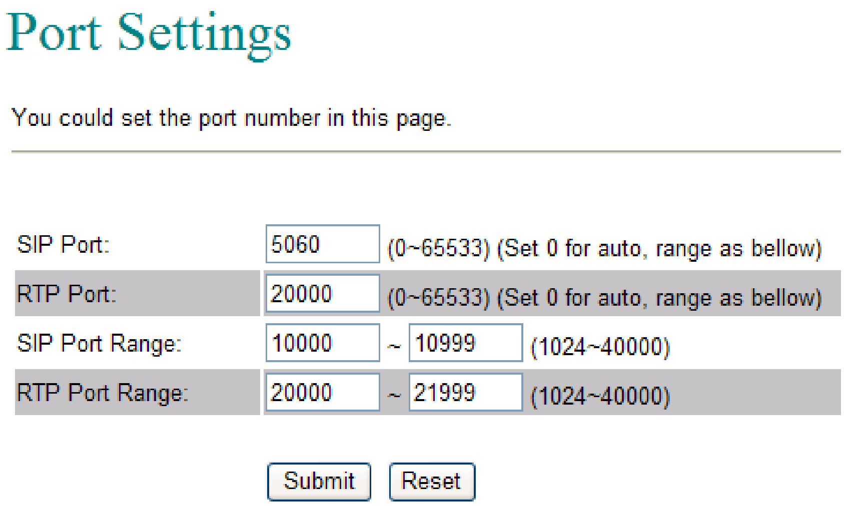


Service Domain Settings
You could set information of service domains in this page.
Phone Book
Phone Setting
Network
SIP
Service Domain
Port
Codec
Codec ID
DTMF
RPort
Other
Reboot
Realm No.:
Realm 2
Realm
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Subscribe for MWI:
On
Off
Status:
Not Registered
Submit
Reset

Port Settings

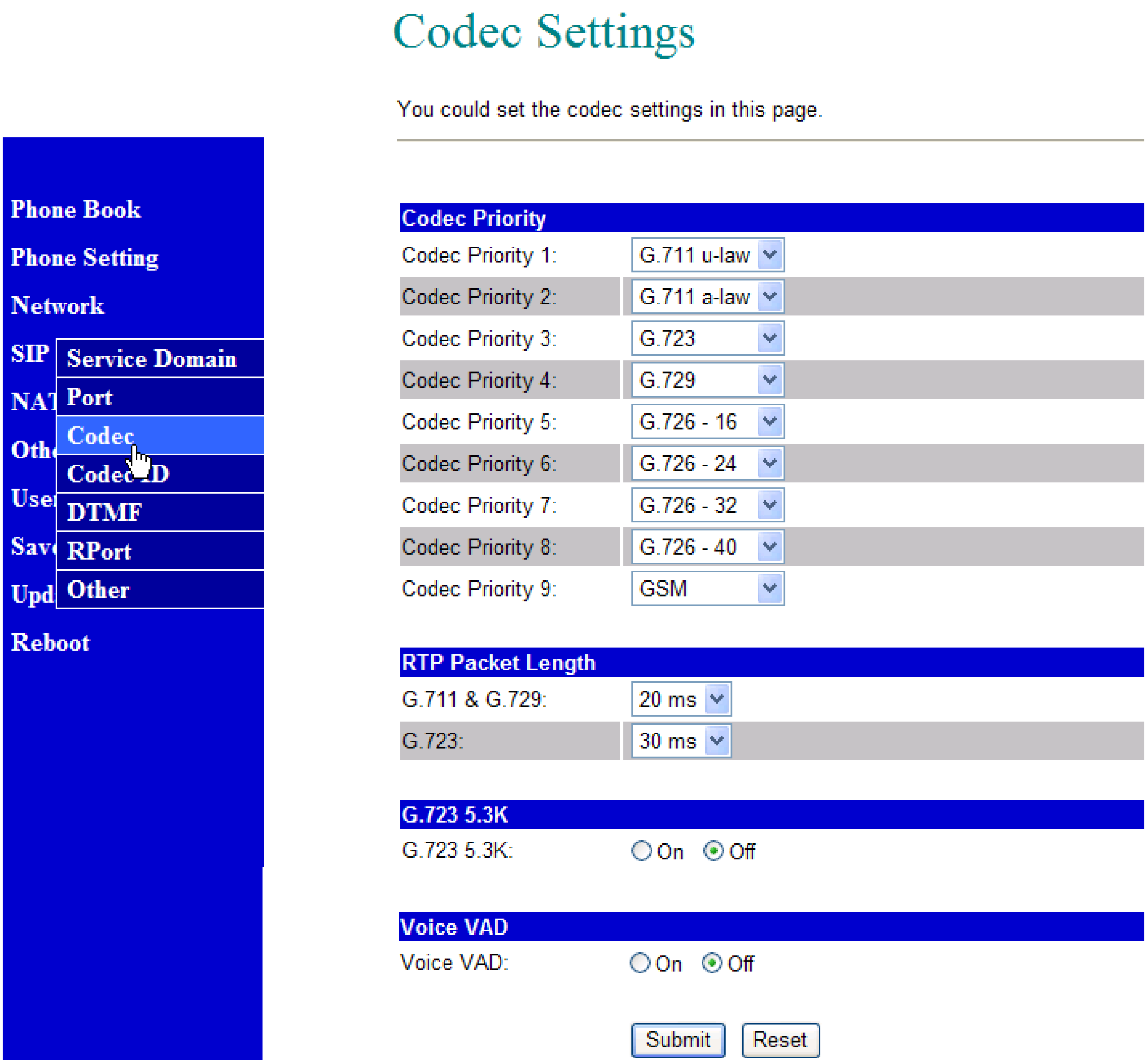
8.76. You can setup the SIP and RTP port number in this page. Each ITSP provider might have different SIP/RTP port setting, please refer to the ITSP to setup the port number correctly. When you finish the setting, please click the “Submit” button.　The defaults for SIP port and RTP port are 5060 and 20000, respectively.





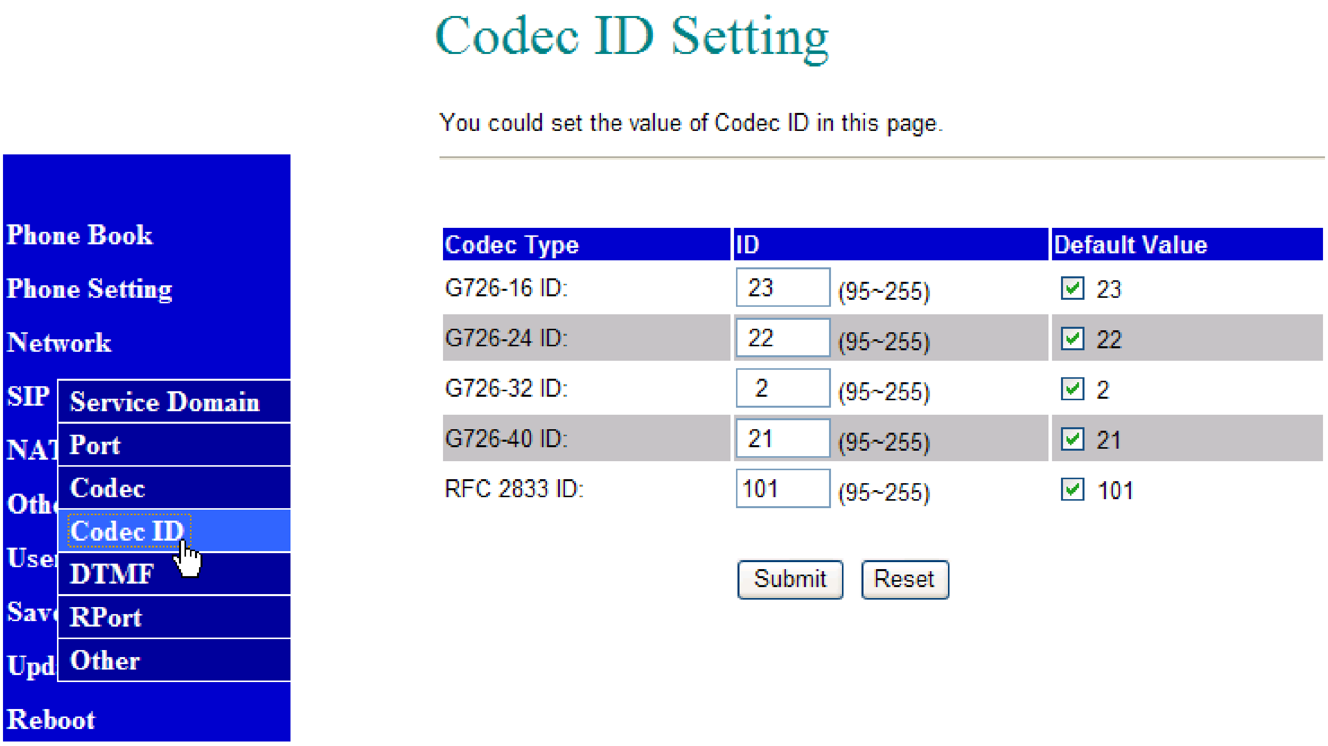
Codec Settings

8.77. You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ITSP recommendations to setup these items.



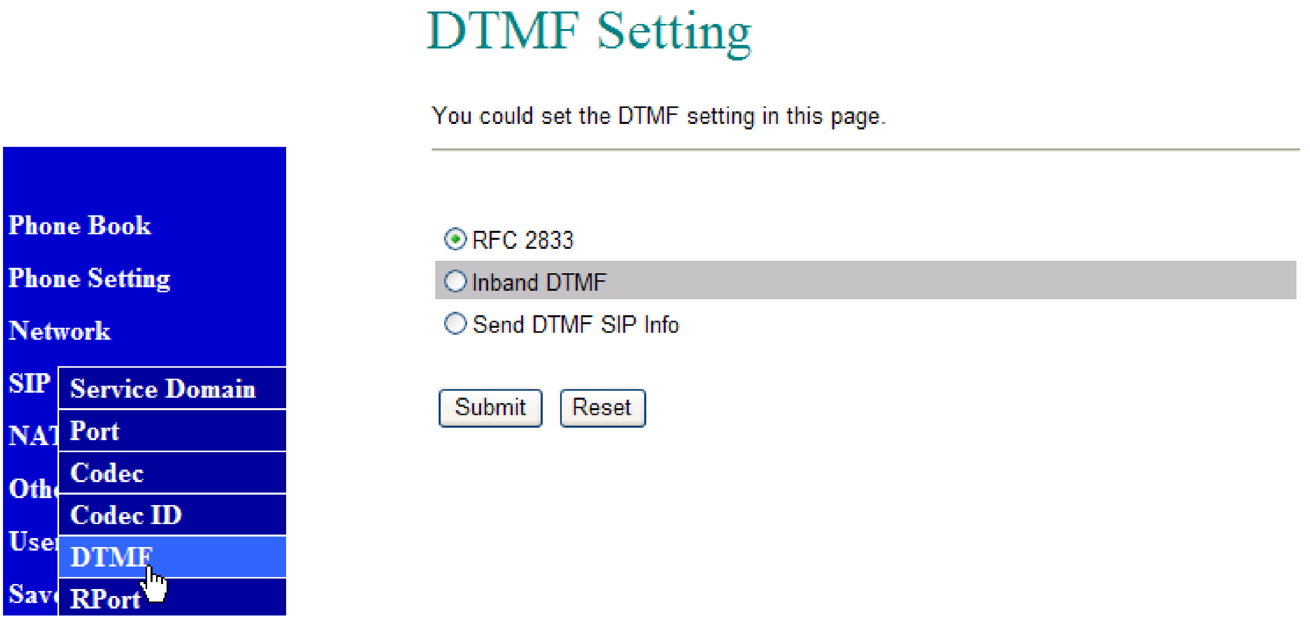
Codec ID Settings

8.78. You can setup the Codec ID in this page. You need to follow the ITSP suggestion to setup these items.



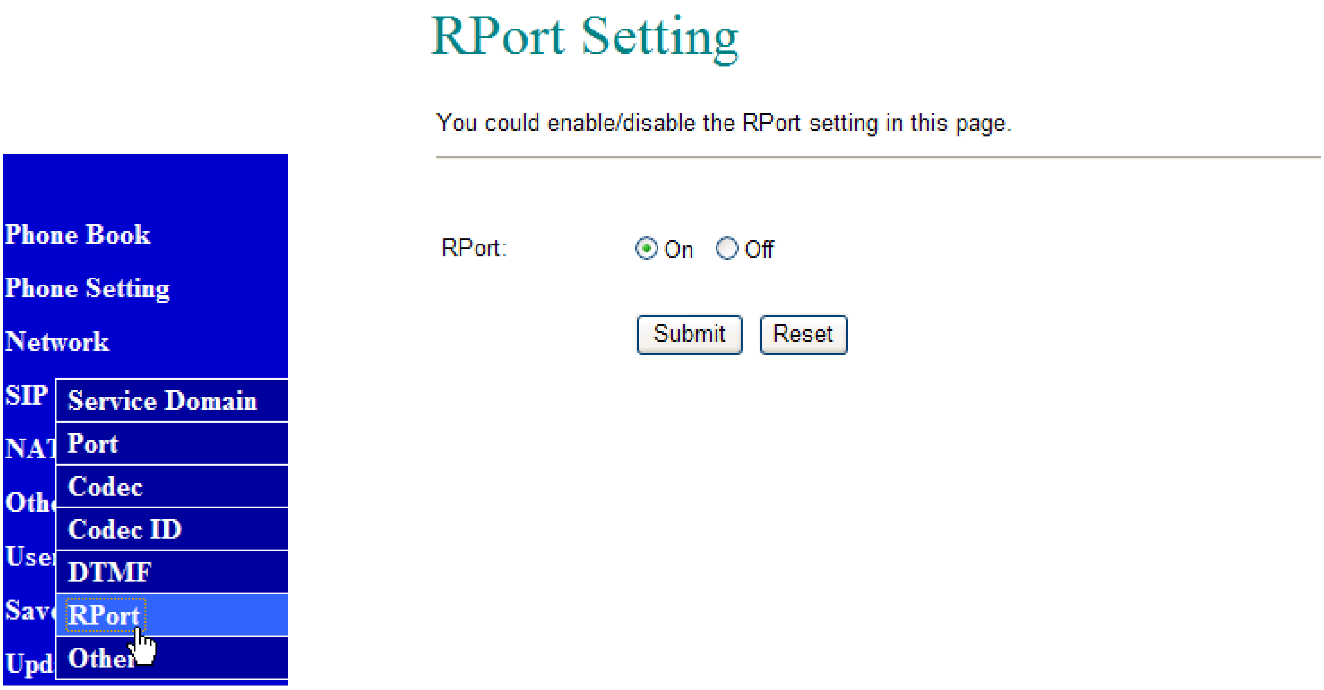
DTMF Settings:

8.79. You can setup the options for DTMF function in this page. The options include RFC2833 (Outband DTMF), Inband DTMF, and Send DTMF SIP info. The default is set at Inband DTMF. If you are making two-stage callings for extension to PSTN, you might need to select Outband DTMF option.



Rport Settings

8.80. You can enable/disable the RPort in this page. To change this setting, please follow your ITSP suggestions. When you finish the setting, please click the Submit button



Other Settings

8.81. You can setup the Hold by RFC and QoS in this page. To change these settings please follows your ITSP information. When you finish the setting, please click the Submit button. The QoS is used to set the voice packet priority. Higher value other than zero will get higher priority for the voice packets in Internet. However, the QoS function still needs to cooperate with the other Internet devices.

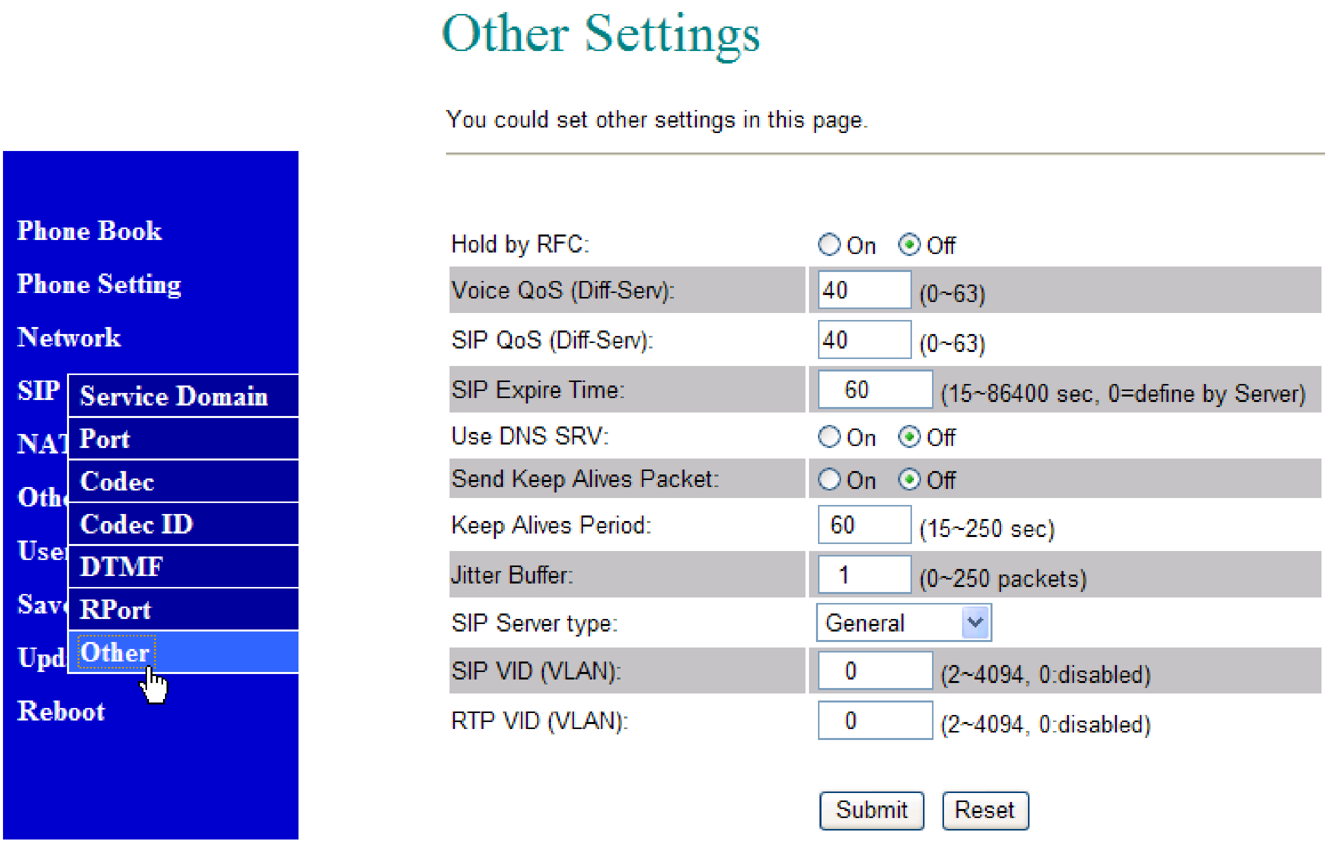
1 SIP Expire Time: 60 (15~86400 sec, 0=defined by Server).

2 Keep Alives Period: 60 (Default: 60, Range: 15 ~ 250sec) that is send keepalives messages in the RTP stream to keep NAT open.

3 Jitter Buffer: 1 (Default: 1, Range: 0~250 packets)

4 SIP Server Type: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel.

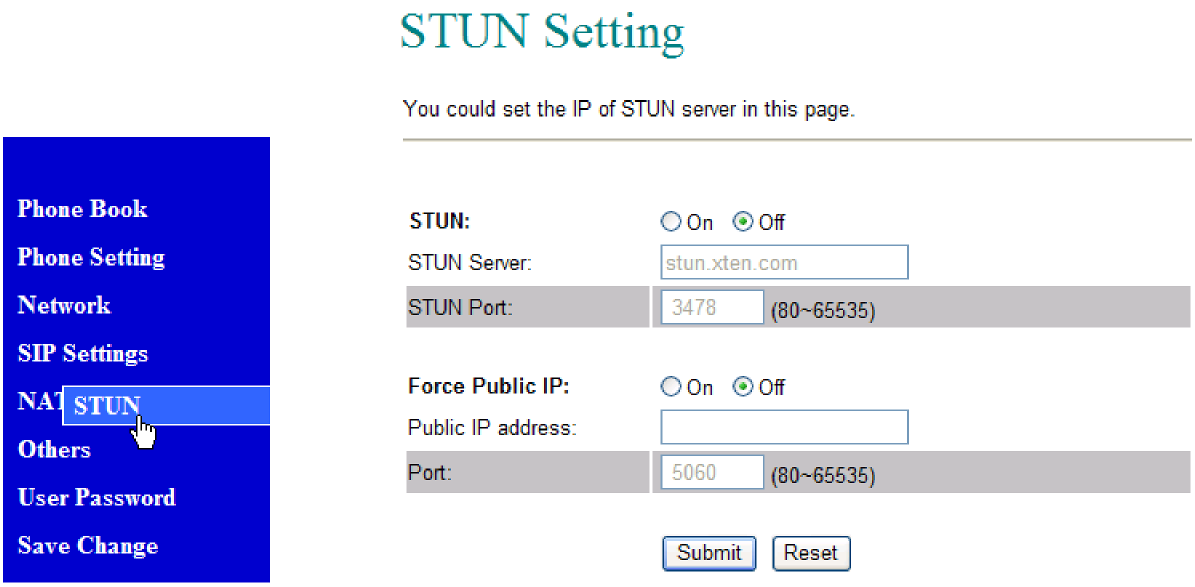
5 SIP VID: 0 (2~4094, 0: disabled). This VLAN ID is for SIP only through WAN.
RTP VID: 0 (2~4094, 0: disabled). This VLAN ID is for RTP voice packets.



NAT Trans

STUN

8.82. The STUN function must be enabled to work properly behind NAT when registered in SIP server. You may enter the STUN server IP address and the STUN port number as shown in the following example.

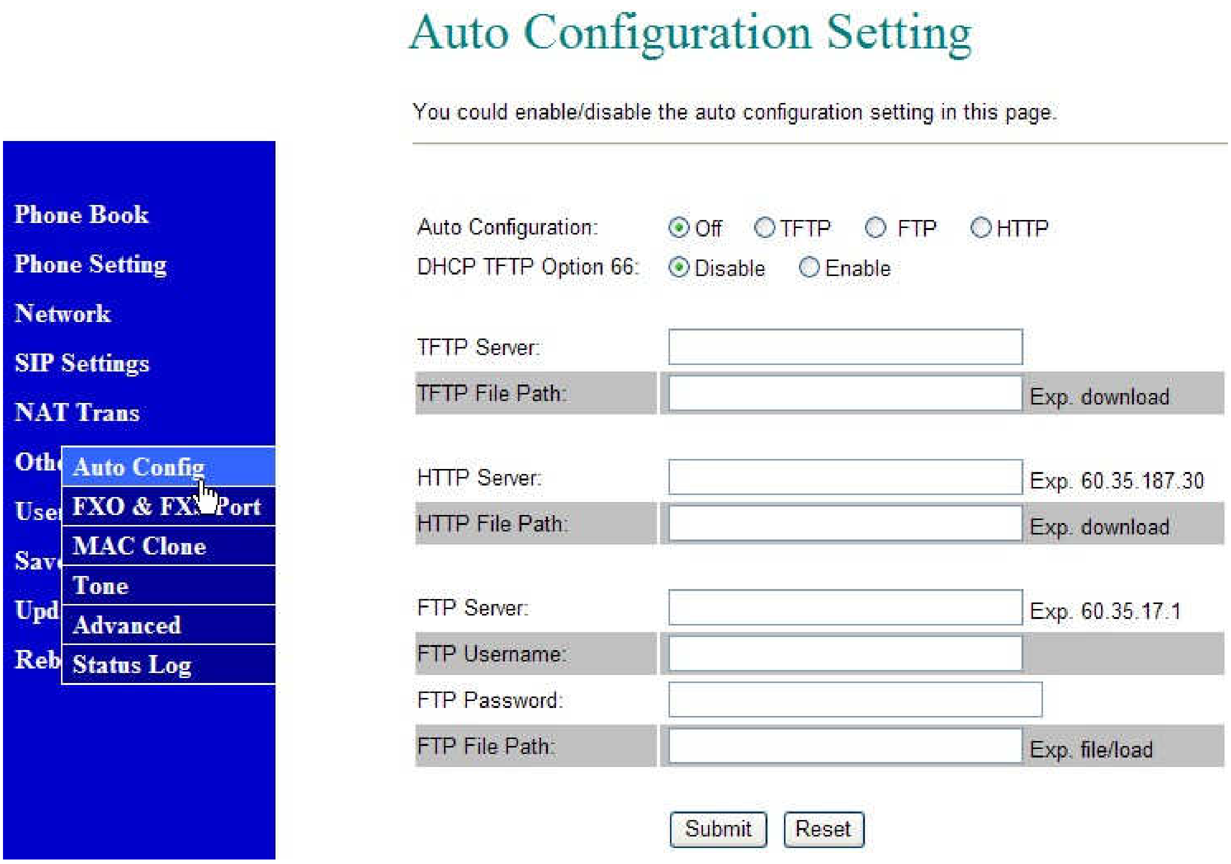


Others

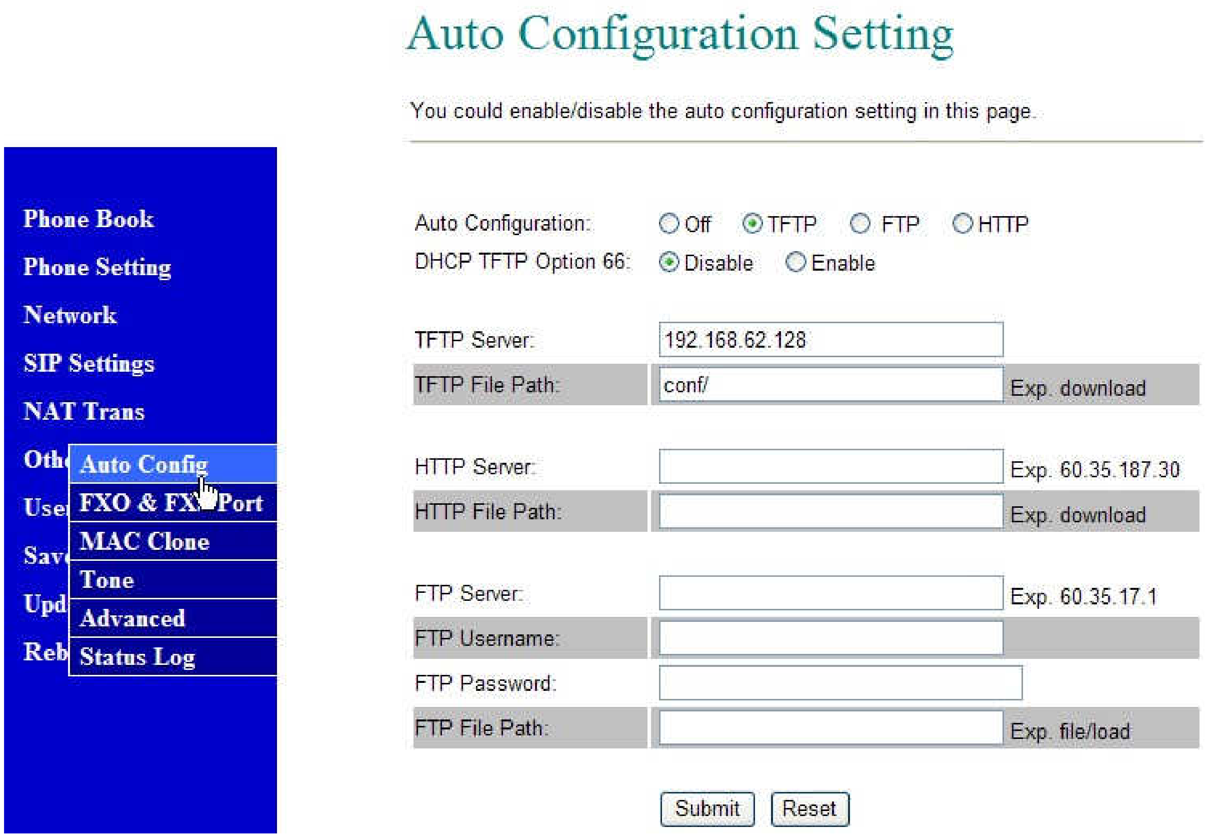
8.83. You can setup Auto Config, FXO& FXS Port MAC Clone, Tone and Some Advanced Settings and Status Log in this page

Auto Config

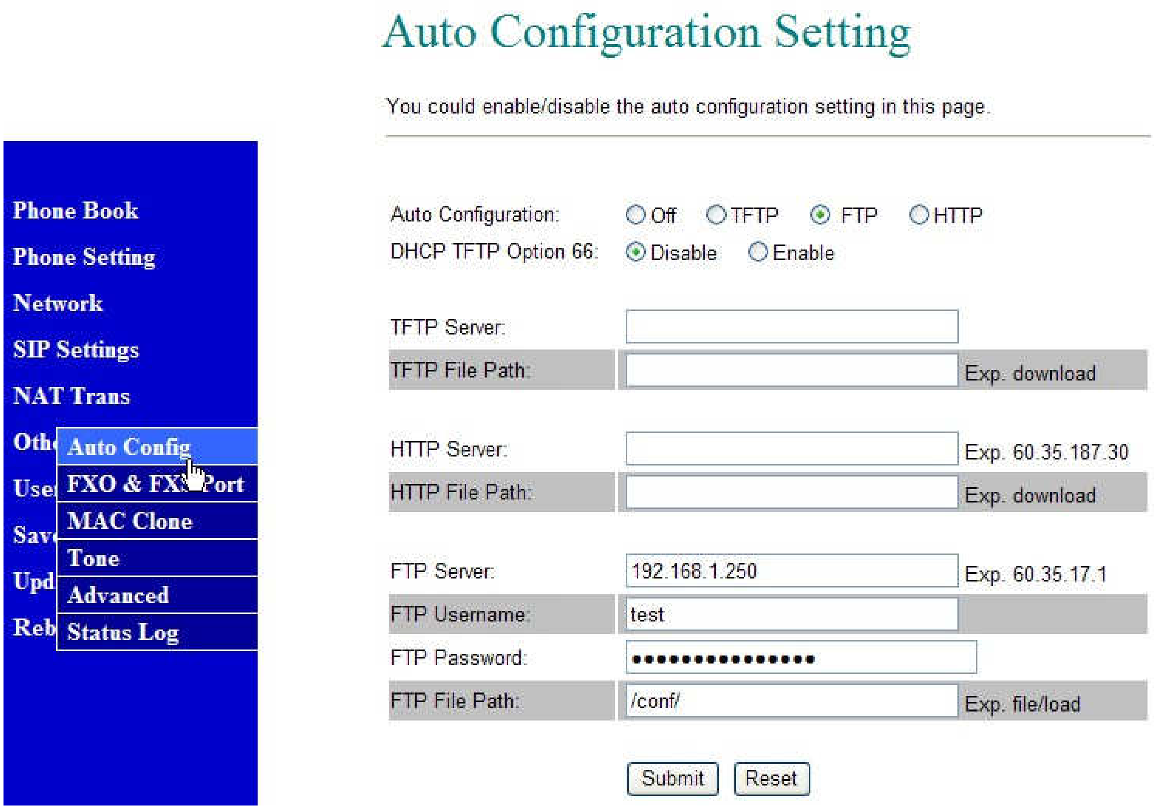
8.84. Auto Configuration function can be used to download the original configurations stored in the TFTP, HTTP or FTP server. This is useful for the new user to automatically download a predefined configuration setting. Remember to click the "Submit" button and "Save" in the **Save Change** section. The VS211 will then reboot and automatically download the original configurations from the TFTP or FTP server. Note that the TFTP download works only for public IP address


Auto Configuration Setting
You could enable/disable the auto configuration setting in this page.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Auto Config
MAC Clone
Tone
Advanced
Status Log
Auto Configuration:
Off
TFTP
FTP
HTTP
DHCP TFTP Option 66:
Disable
Enable
TFTP Server:
TFTP File Path:
Exp. download
HTTP Server:
Exp. 60.35.187.30
HTTP File Path:
Exp. download
FTP Server:
Exp. 60.35.17.1
FTP Username:
FTP Password:
FTP File Path:
Exp. file/load
Submit
Reset

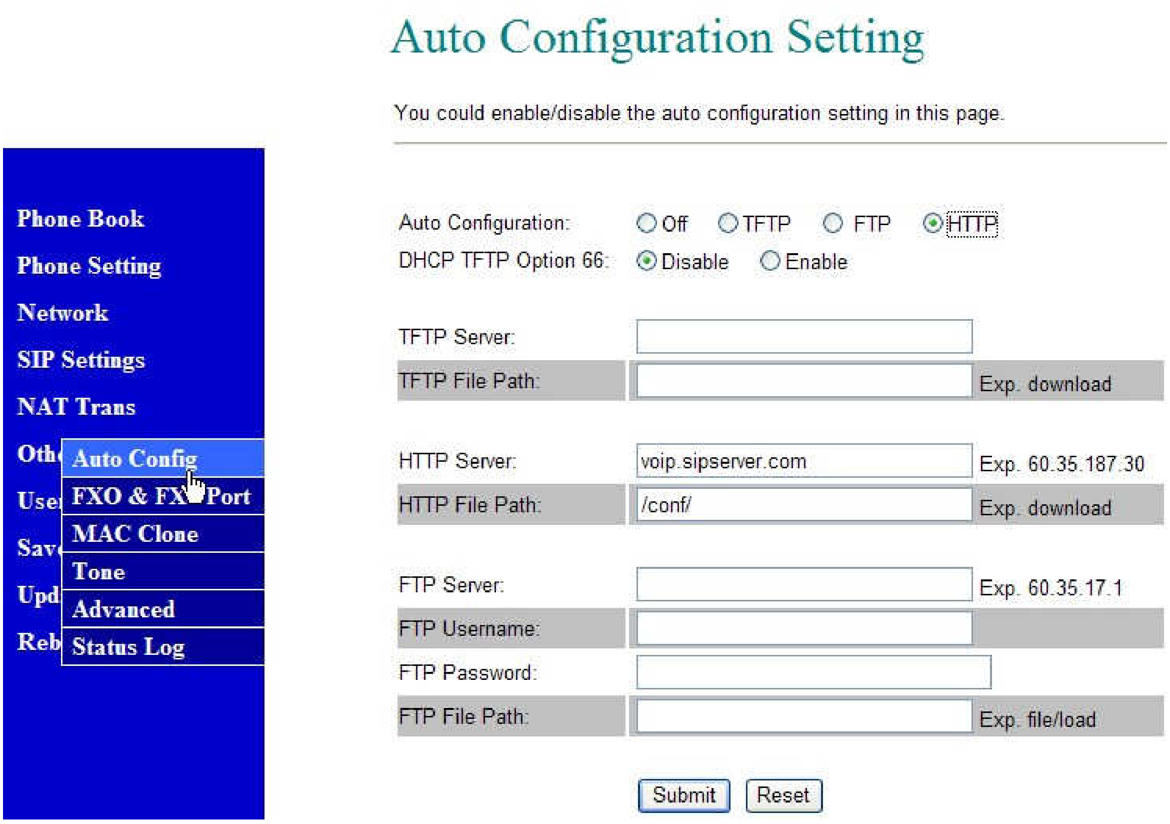
TFTP Mode


Auto Configuration Setting
You could enable/disable the auto configuration setting in this page.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Auto Config
MAC Clone
Tone
Advanced
Status Log
Auto Configuration:
Off
TFTP
FTP
HTTP
DHCP TFTP Option 66:
Disable
Enable
TFTP Server:
192.168.62.128
TFTP File Path:
conf/
Exp. download
HTTP Server:
Exp. 60.35.187.30
HTTP File Path:
Exp. download
FTP Server:
Exp. 60.35.17.1
FTP Username:
FTP Password:
FTP File Path:
Exp. file/load
Submit
Reset

FTP Mode

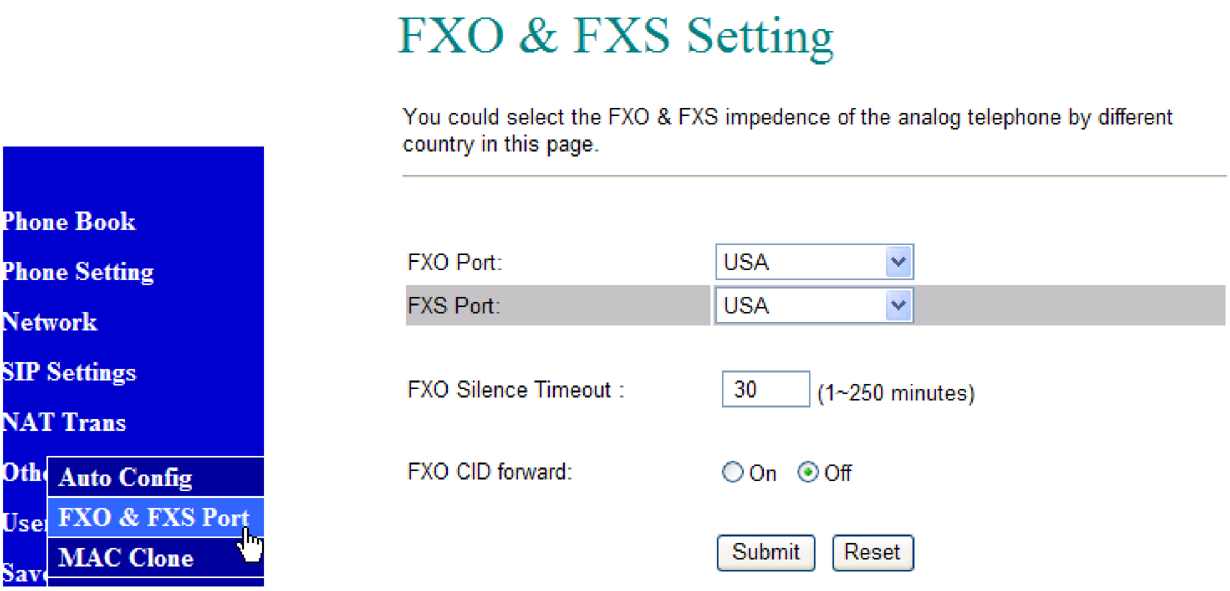


HTTP Mode



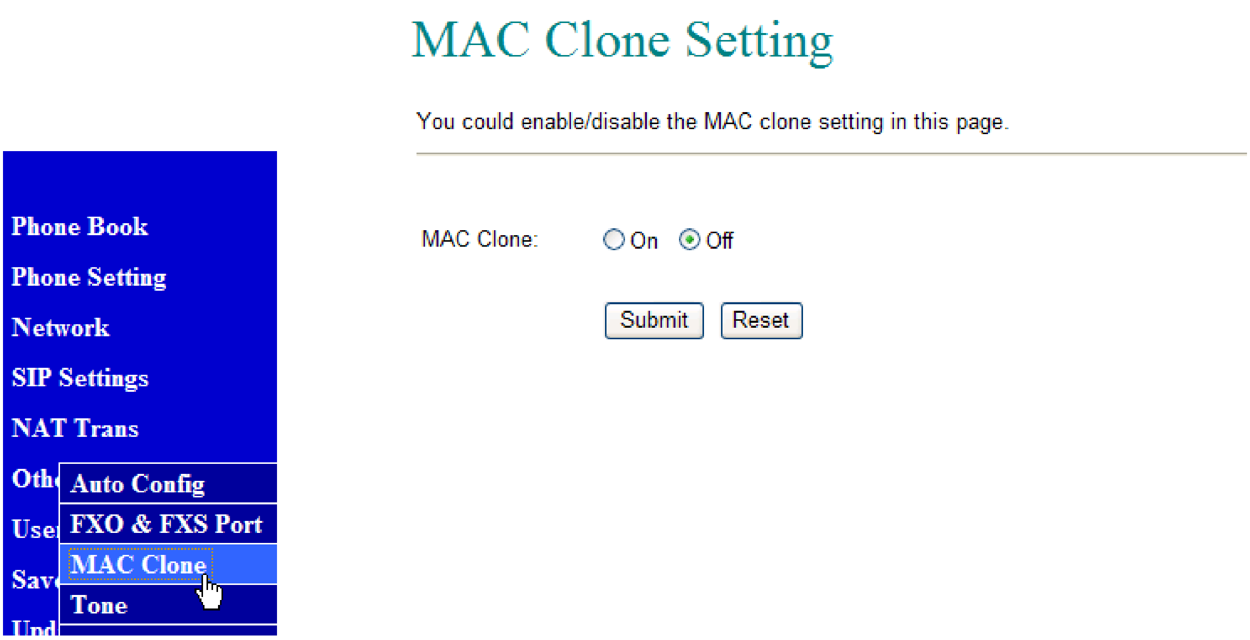
FXS & FXO Port

8.85. You may select the FXS and FXO impedence of the analog telephone by different countries



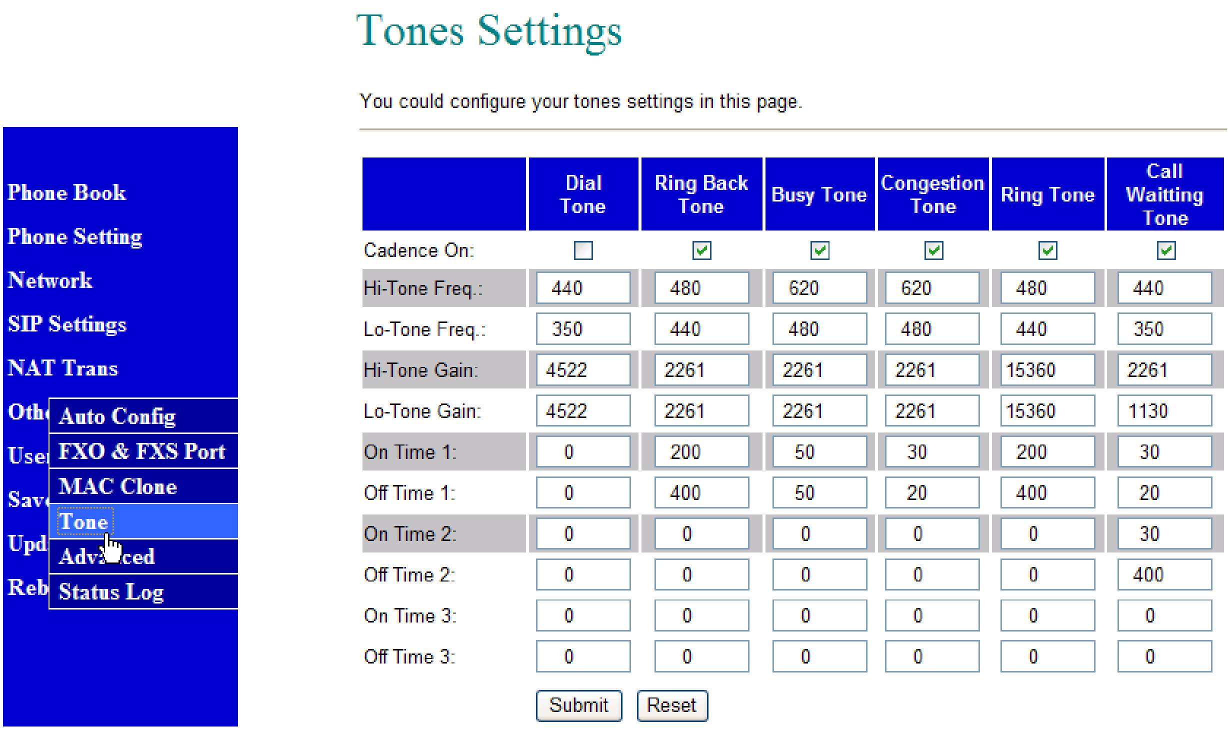
MAC Clone

8.86. The MAC Clone function is to clone the MAC when only one MAC is available from ITSP. This is to share with the PC using the same MAC. When you finish settings, please click the Submit button



Tone

8.87. .The Tone setting can be adjusted to generate Dial tone, Ring back tone, Ring tone, Congestion tone, Call waiting tone and Busy tone for different countries. When you finish with the settings, please click the Submit button.



Advanced

8.88. The advanced settings might be useful for some network requirements. The ICMP function is to echo when someone ping this device. This can prevent from hacker attacking the device by not echoing. When you finish the setting, please click the Submit button

1 IP Dialing format (P2P dialing): There are 3 options.

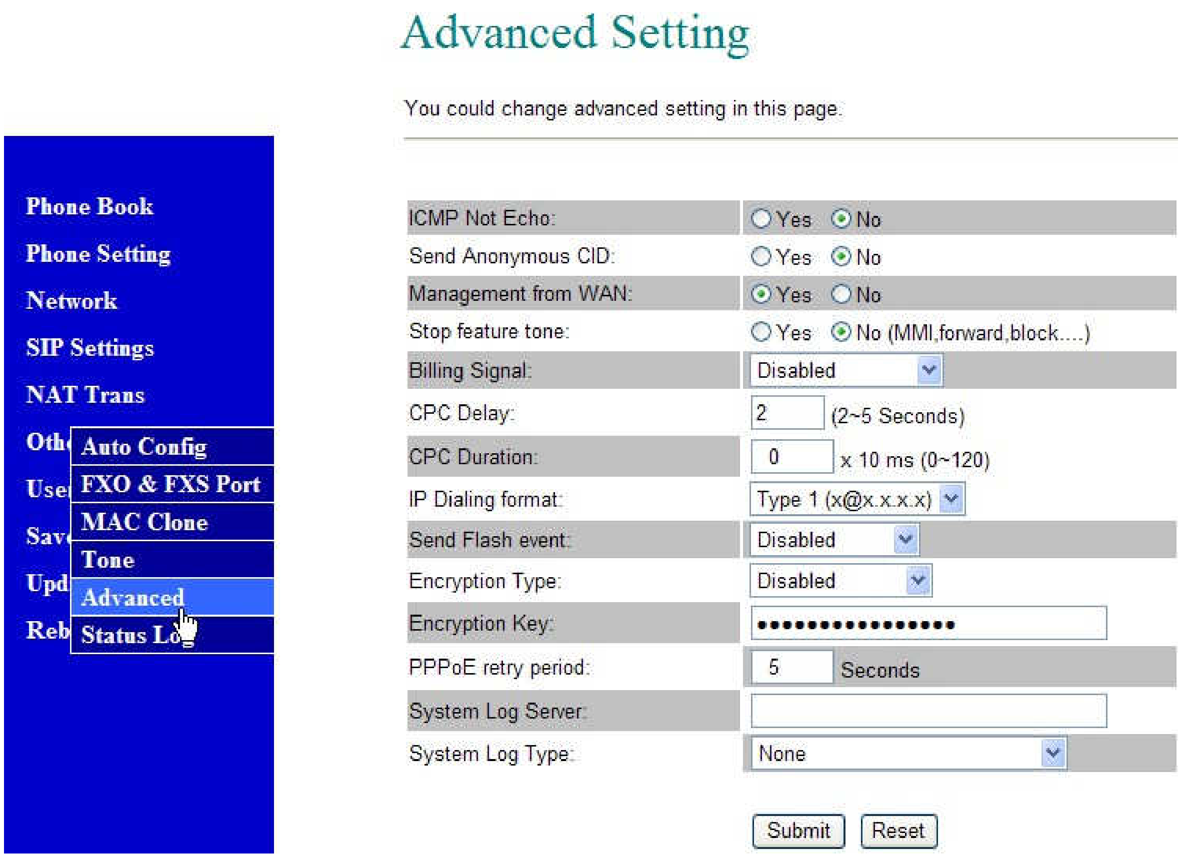
2 Type 1(xx@x.x.x), by default when you dial 192.168.1.100, VS211 will auto prefix userip@ in your dialed IP address and send out to remote end.

3 Type 2 (x.x.x.), when you dialed 192.168.1.100, it will send 192.168.1.100 out to remote end without appending anything.

4 Disable: disable IP dialing function.

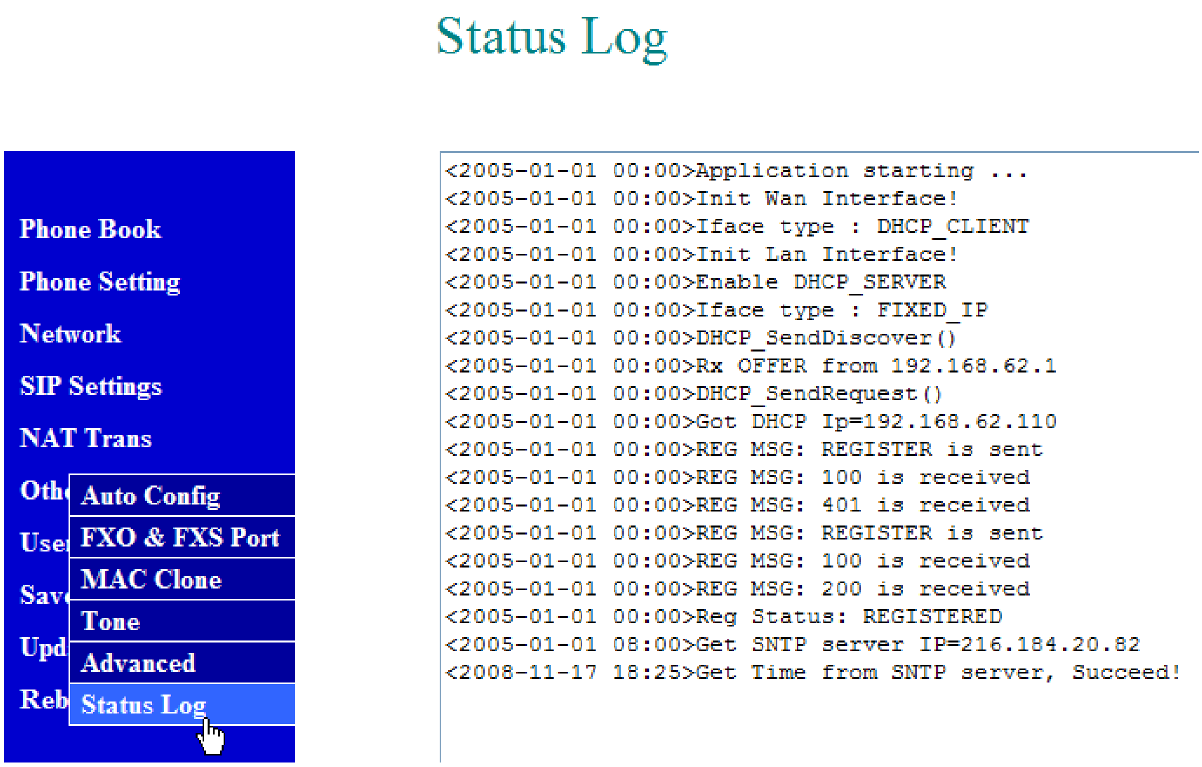
5 Encryption Type: Must be disabled for standard SIP protocol. Please consult your ITSP for encryption type when registered in ITSP.

6 Encryption Key: Special key for encryption need.



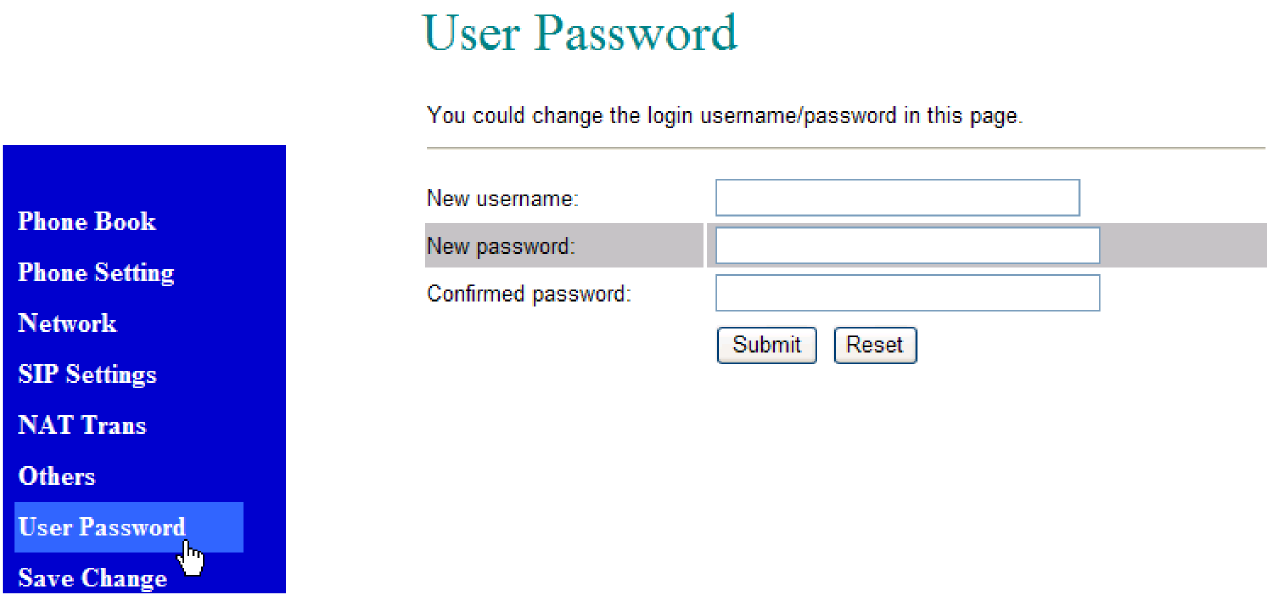
Status Log

8.89. You can check system status log



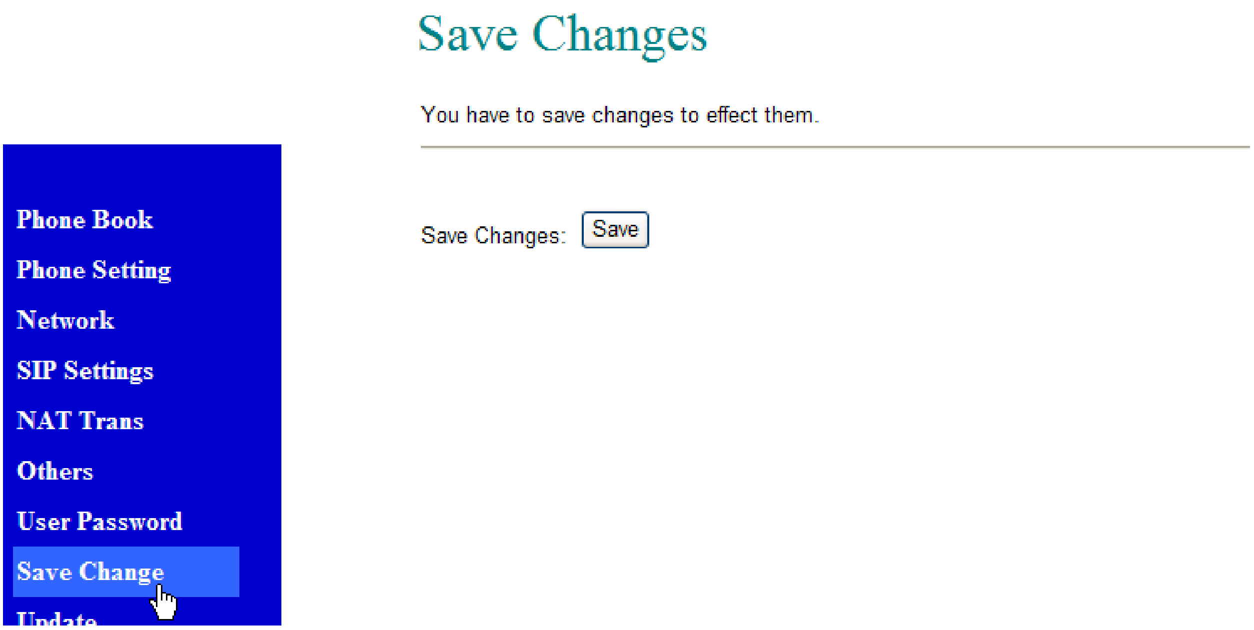
User Password

8.90. You may change the login name and password in this page.



Save Changes

8.91. You can save the changes you have made, and click the Save button. After clicking the "Save" button, the VS211 will automatically save the new settings.



Update

New Firmware

8.92. VS211 provides two methods, HTTP or TFTP, to update new firmware as the following steps

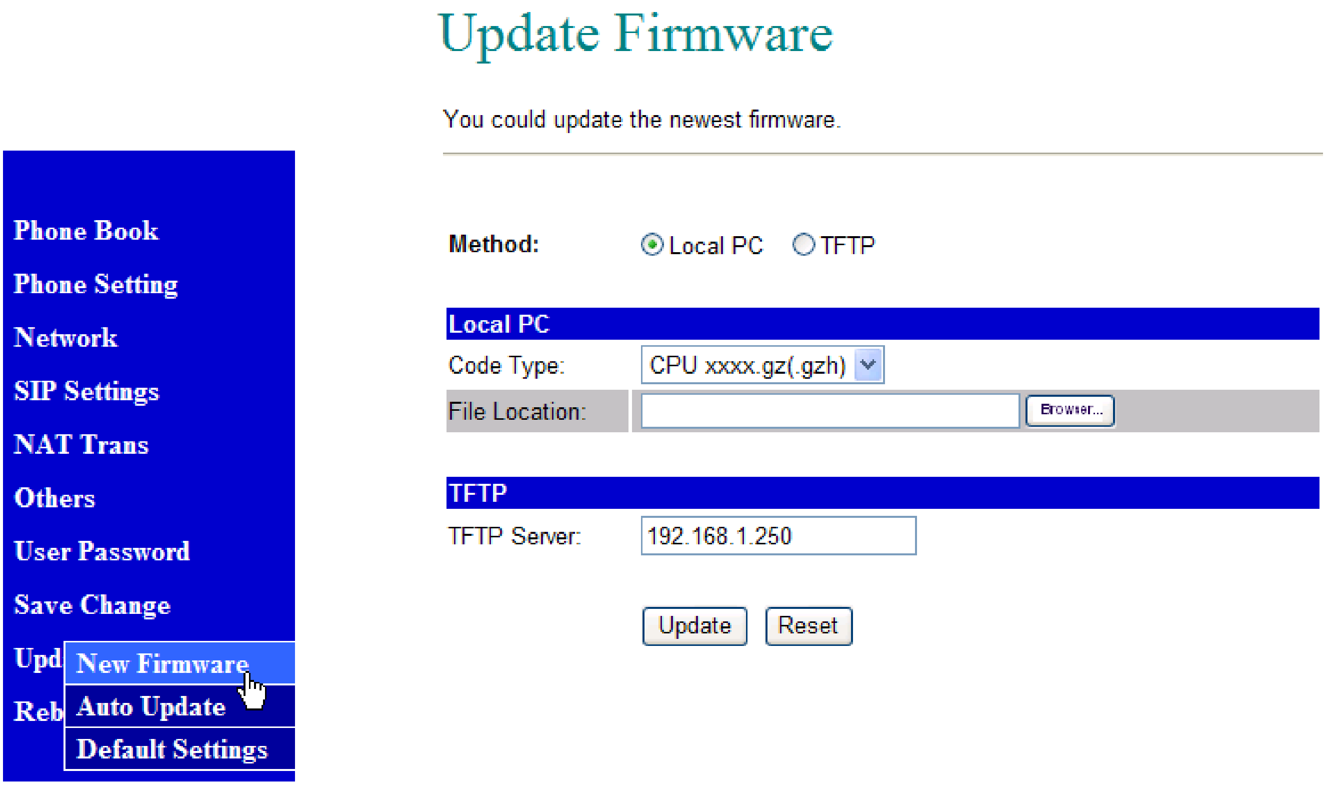
HTTP is mostly used for firmware update by using local PC.

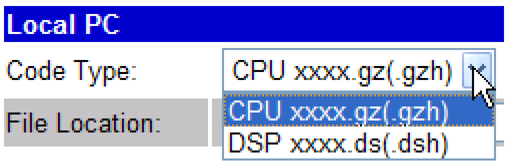
TFTP is used with TFTP server for firmwares stored in TFTP server.

1 Select the firmware code type, Risc or DSP code. (mostly for Risc code)

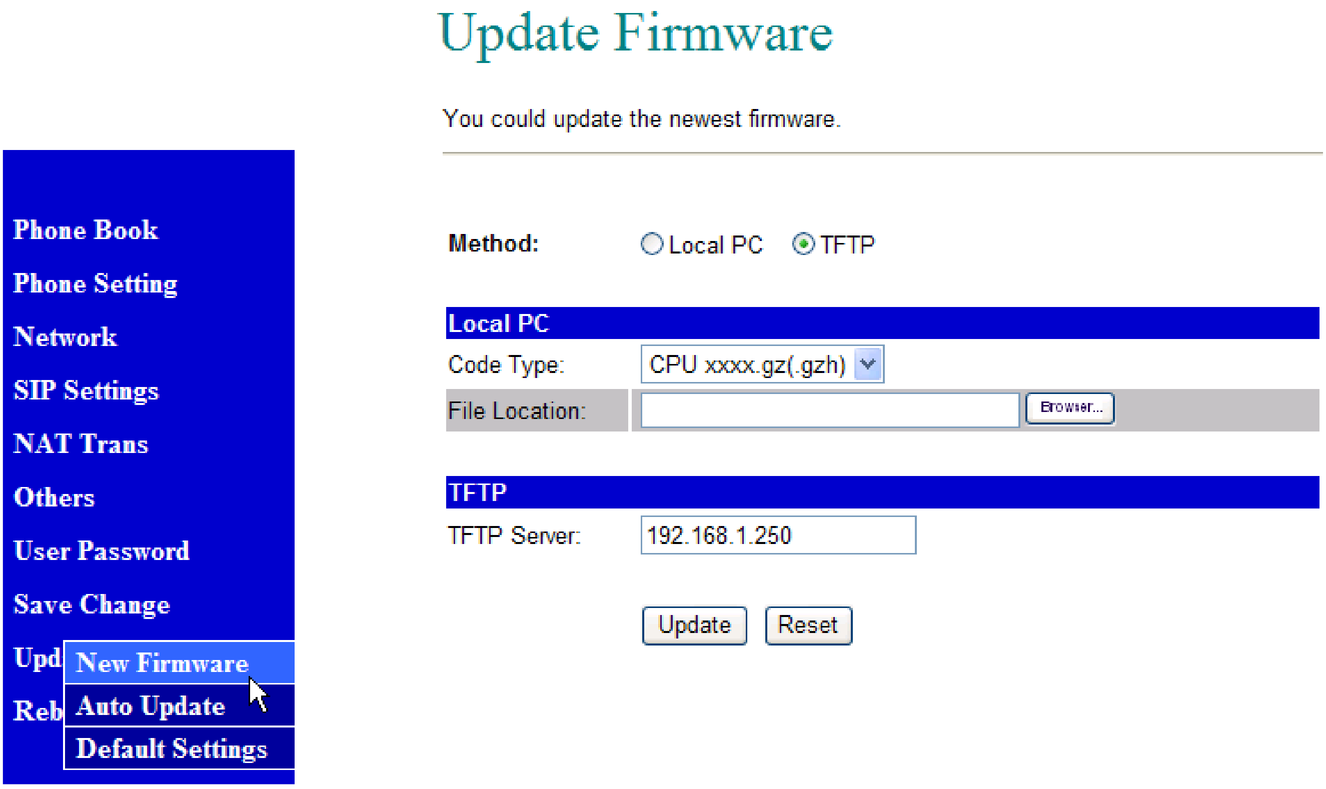
2 Click the “Browse” button to choose the updated file location for HTTP download, or

3 Select TFTP and enter the IP address of TFTP server for firmware download, then click the “Update” button.

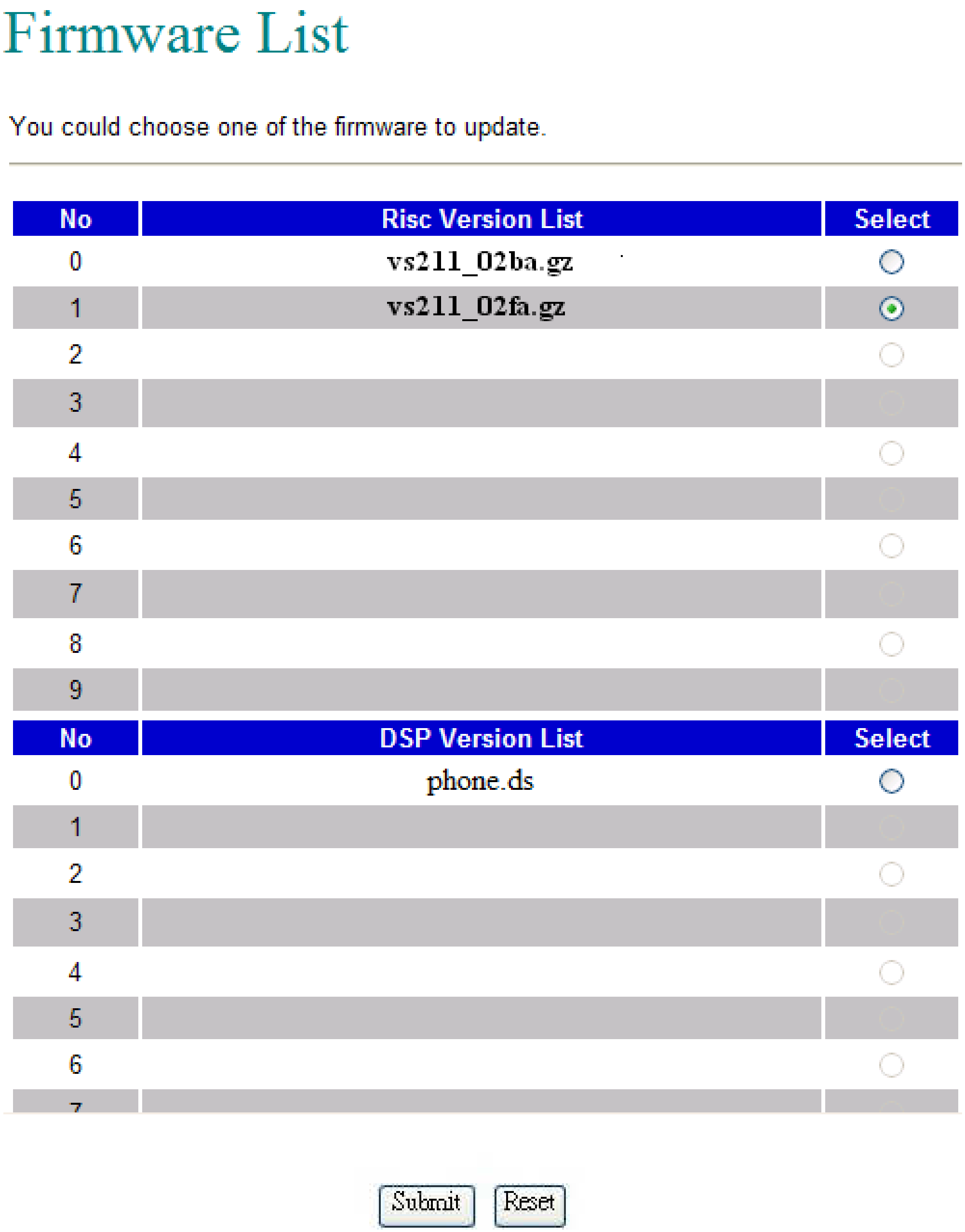




TFTP Mode


Update Firmware
You could update the newest firmware.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Others
User Password
Save Change
New Firmware
Auto Update
Default Settings
Method:
Local PC
TFTP
Local PC
Code Type:
CPU xxxx.gz(.gzh)
File Location:
Browser...
TFTP
TFTP Server:
192.168.1.250
Update
Reset


Windows Internet Explorer
NOTE:DO NOT UN-PLUG the power adapter while updating. It will take about 3 minutes to update firmware. Please wait...
Yes
Cencel



NOTE: Do NOT power OFF the VS211 after clicking the "Select" button, or you may damage the VS211. The remote TFTP download works only for public IP address.

8.93. After clicking the "Update" button, the firmware list will be displayed from server to indicate the available firmware for download

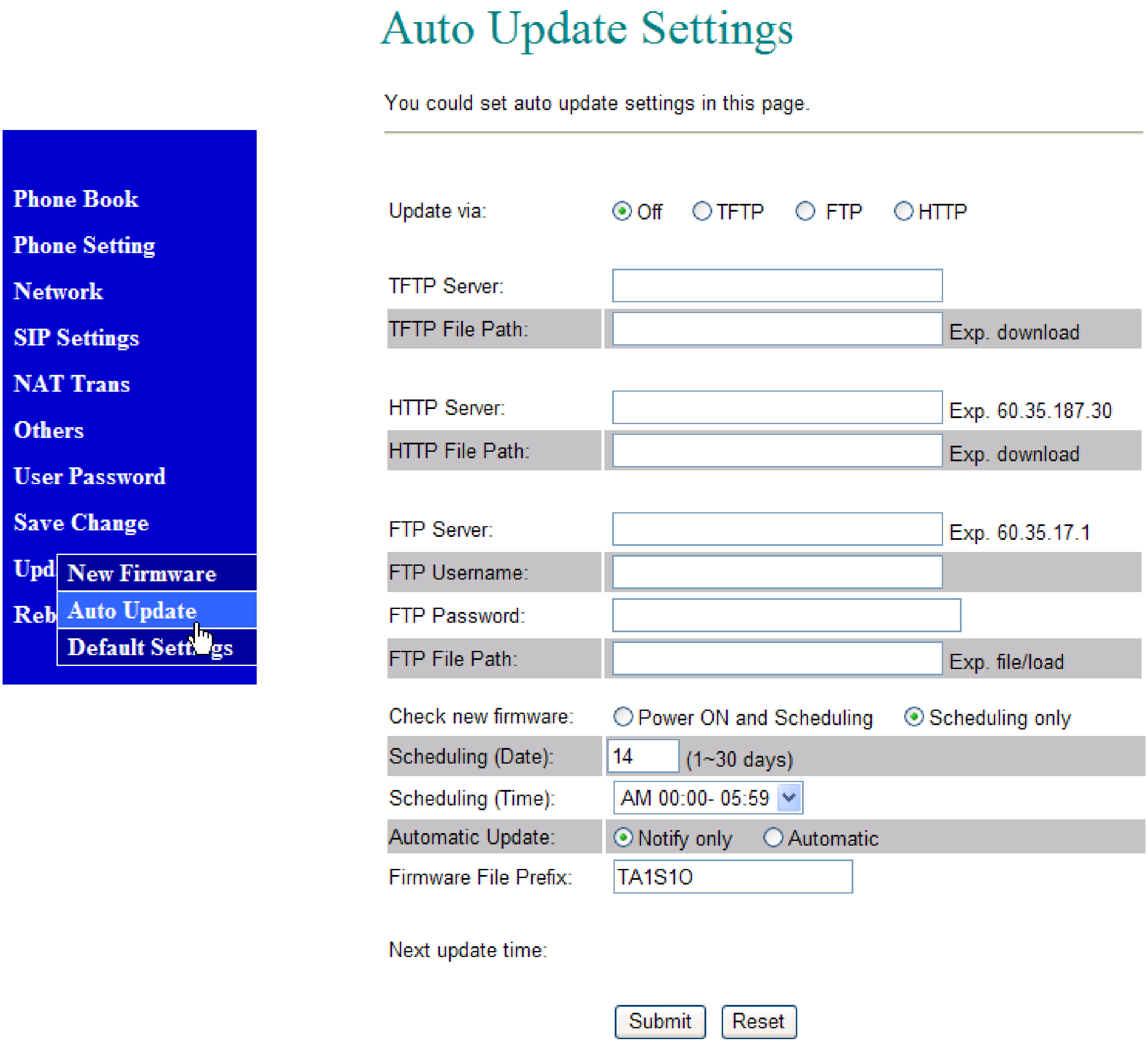
8.94. Select the new file you want to download to the VS211 then click the "Select" button
In 3 to 4 minutes, the PHONE LED indicators will start flashing 5 times to indicate successful firmware update. Then, you need to login again new IP address which is available from IVR by pressing **#120#** from phone.

Auto Update:

8.95. Auto update function can be used to auto-update the firmware stored in the TFTP, HTTP or FTP server per the schedule as the following settings:

8.96. Select the update method and enter the server IP address,

8.97. Click the “Submit” button to get auto-update effective.



Notes:

Check new Firmware:

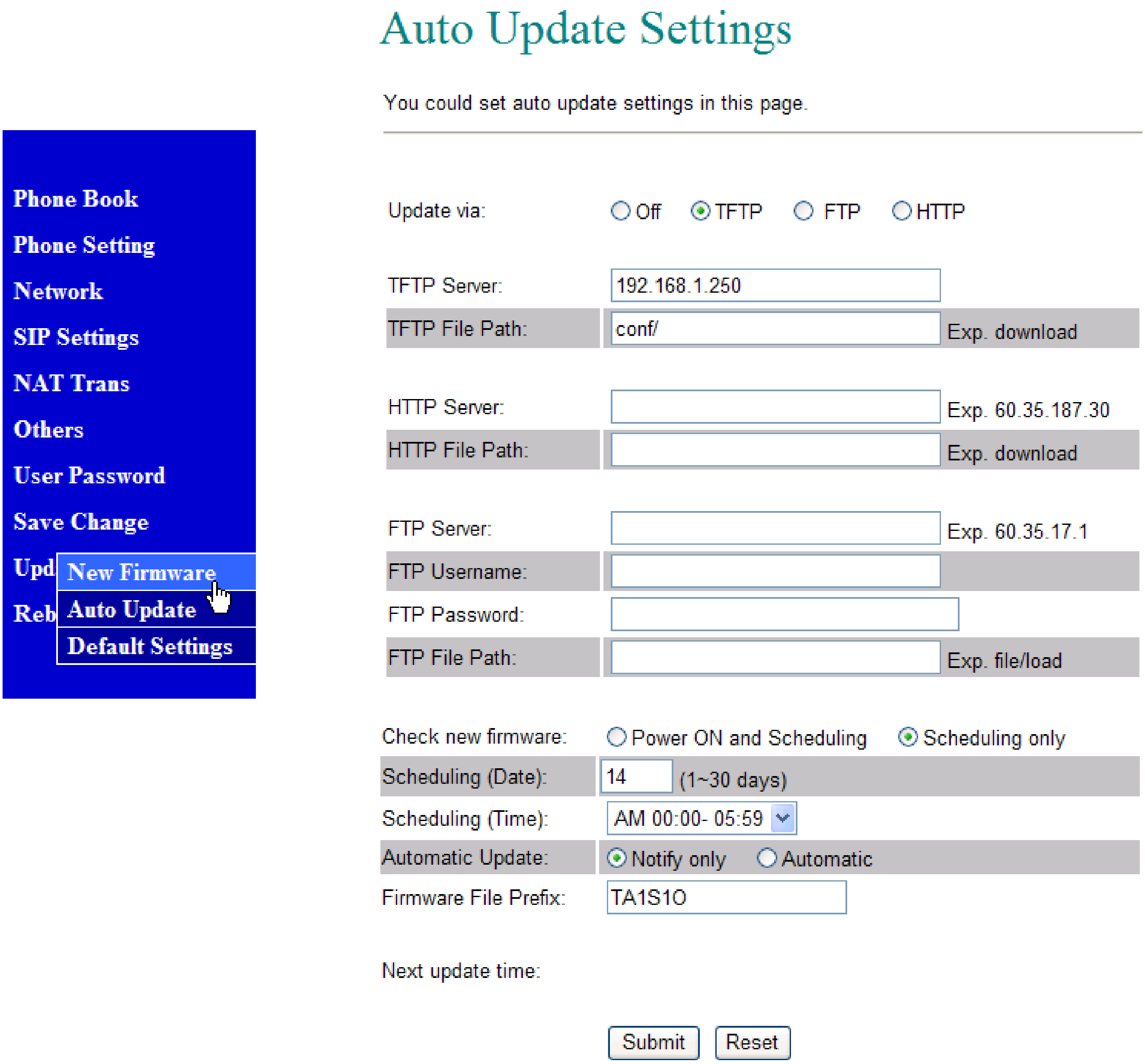
Power on and Scheduling: Check new firmware when power on and in accord with schedule. Must be manually updated.

Scheduling Only (default): Check in accord with schedule.

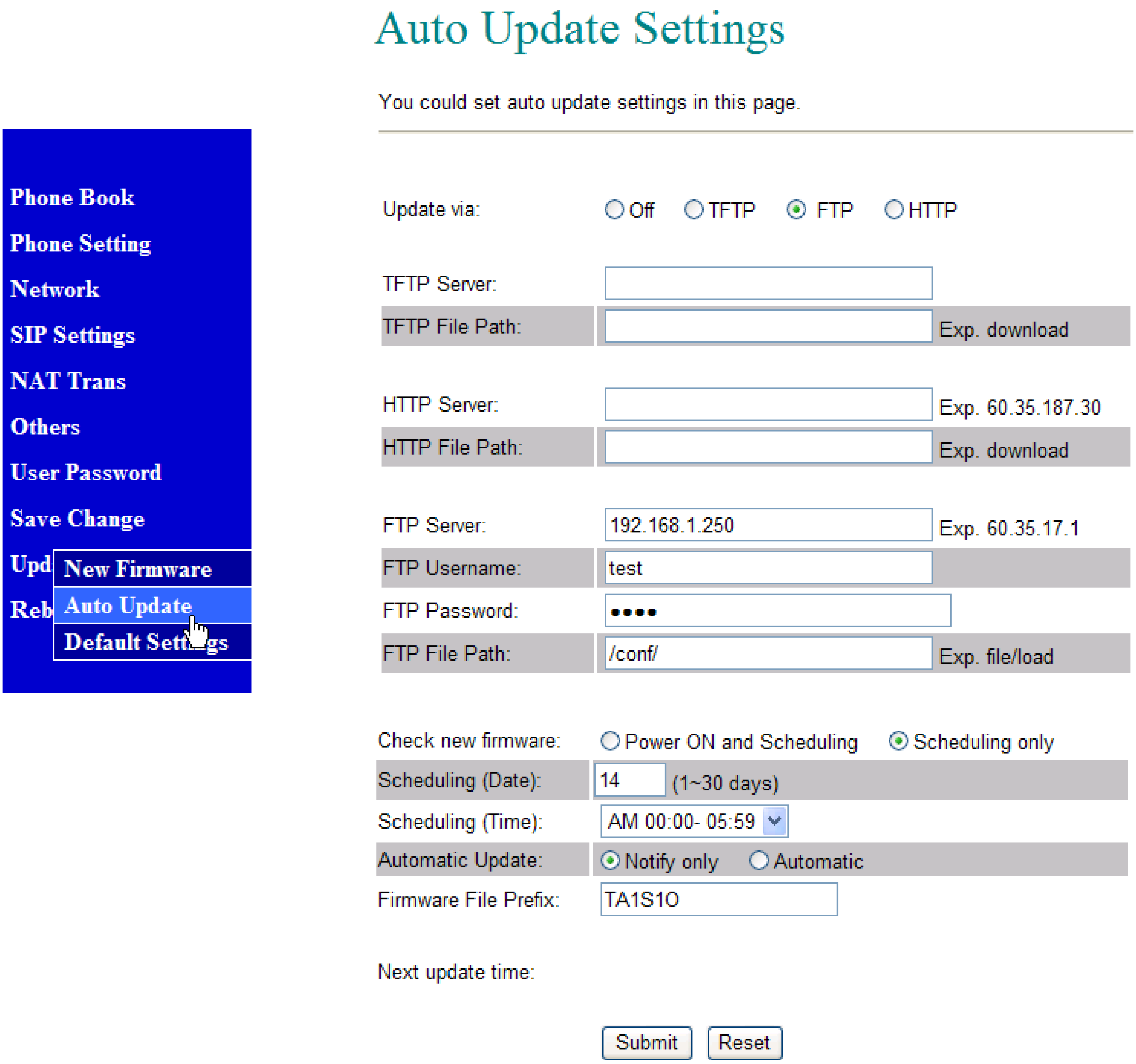
Scheduling (Date): Default at 14 days.

Scheduling (Time): Default at AM 00:00 – 05:59 randomly.

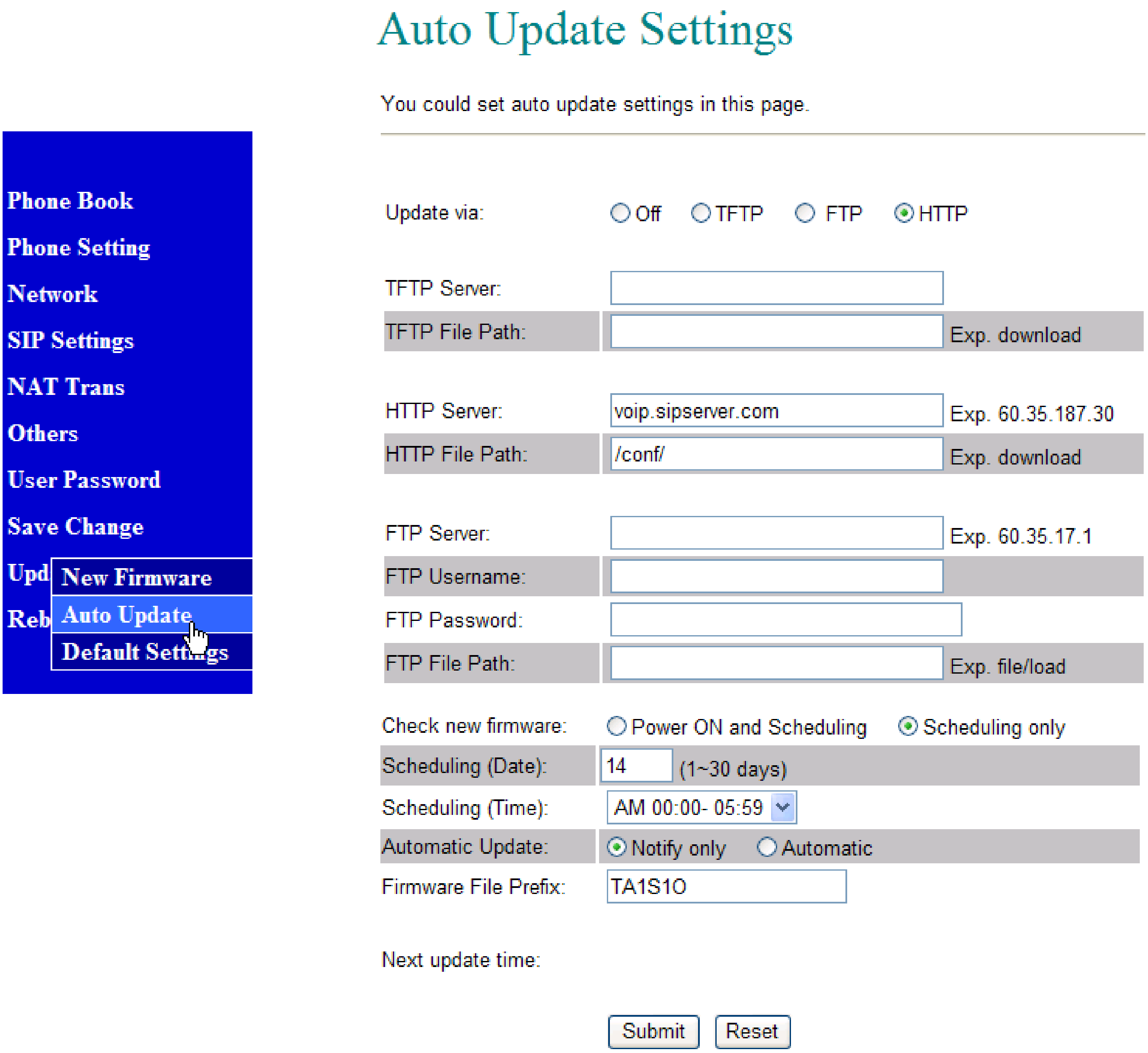
TFTP Mode


Auto Update Settings
You could set auto update settings in this page.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Others
User Password
Save Change
Upd
Reb
New Firmware
Auto Update
Default Settings
Update via:
Off
TFTP
FTP
HTTP
TFTP Server:
192.168.1.250
TFTP File Path:
conf/
Exp. download
HTTP Server:
Exp. 60.35.187.30
HTTP File Path:
Exp. download
FTP Server:
Exp. 60.35.17.1
FTP Username:
FTP Password:
FTP File Path:
Exp. file/load
Check new firmware:
Power ON and Scheduling
Scheduling only
Scheduling (Date):
14
(1~30 days)
Scheduling (Time):
AM 00:00- 05:59
Automatic Update:
Notify only
Automatic
Firmware File Prefix:
TA1S1O
Next update time:
Submit
Reset

FTP Mode

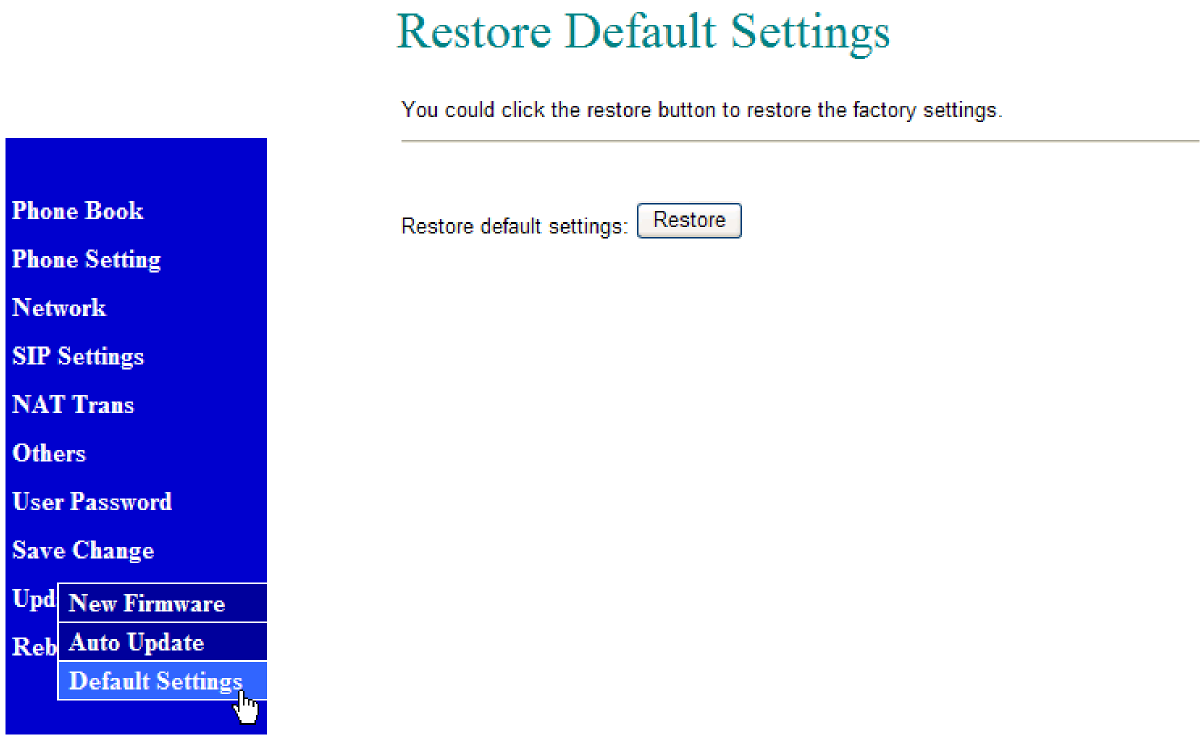

Auto Update Settings
You could set auto update settings in this page.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Others
User Password
Save Change
Upd
Reb
New Firmware
Auto Update
Default Settings
Update via:
Off
TFTP
FTP
HTTP
TFTP Server:
TFTP File Path:
Exp. download
HTTP Server:
Exp. 60.35.187.30
HTTP File Path:
Exp. download
FTP Server:
192.168.1.250
Exp. 60.35.17.1
FTP Username:
test
FTP Password:
FTP File Path:
/conf/
Exp. file/load
Check new firmware:
Power ON and Scheduling
Scheduling only
Scheduling (Date):
14
(1~30 days)
Scheduling (Time):
AM 00:00- 05:59
Automatic Update:
Notify only
Automatic
Firmware File Prefix:
TA1S1O
Next update time:
Submit
Reset

HTTP Mode


Auto Update Settings
You could set auto update settings in this page.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans
Others
User Password
Save Change
Upd
Reb
New Firmware
Auto Update
Default Settings
Update via:
Off
TFTP
FTP
HTTP
TFTP Server:
TFTP File Path:
Exp. download
HTTP Server:
voip.sipserver.com
Exp. 60.35.187.30
HTTP File Path:
/conf/
Exp. download
FTP Server:
Exp. 60.35.17.1
FTP Username:
FTP Password:
FTP File Path:
Exp. file/load
Check new firmware:
Power ON and Scheduling
Scheduling only
Scheduling (Date):
14
(1~30 days)
Scheduling (Time):
AM 00:00- 05:59
Automatic Update:
Notify only
Automatic
Firmware File Prefix:
TA1S1O
Next update time:
Submit
Reset

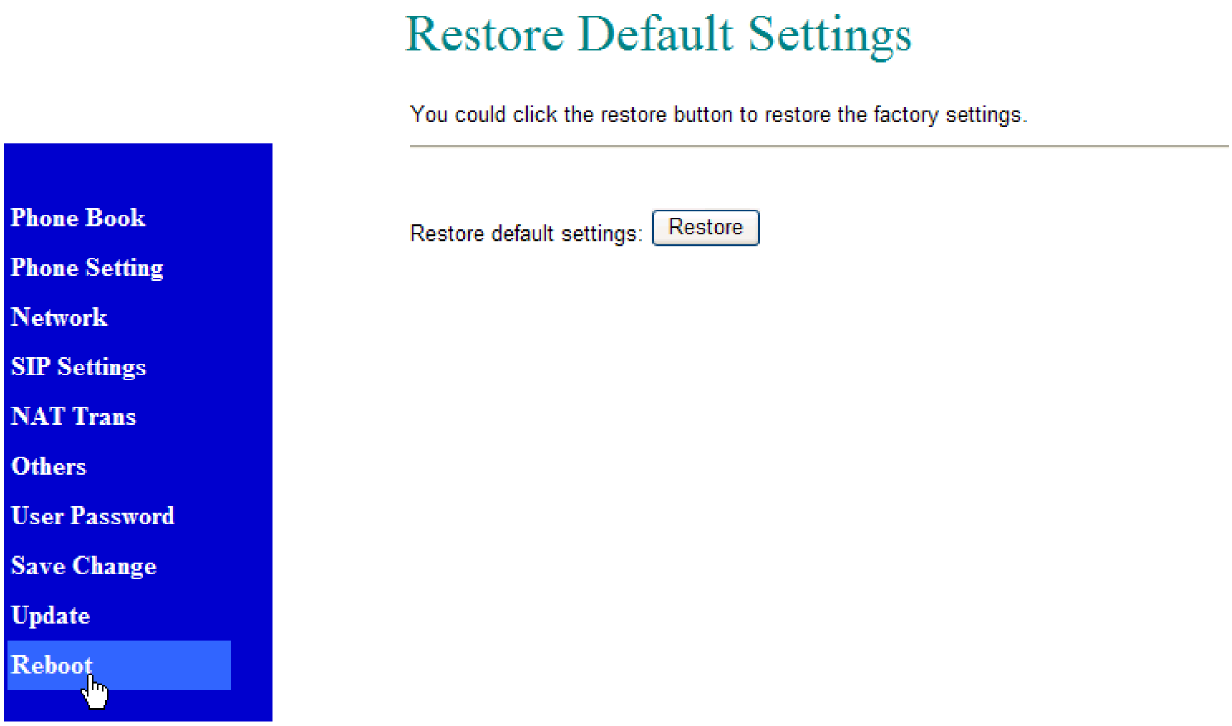
Default Setting

8.98. You can restore the VS211 to factory default in this page. By clicking the “Restore” button, the VS211 will restore to default and automatically restart again.



Reboot

8.99. You may click the Reboot button to restart, then VS211 will automatically reboot with the stored configurations.



9. Configurations by Telephone & IVR

You can use telephone to configure and to check the status of VS211. Note that the WAN port is for WAN port of VS211, and the LAN port is for LAN port of VS211.

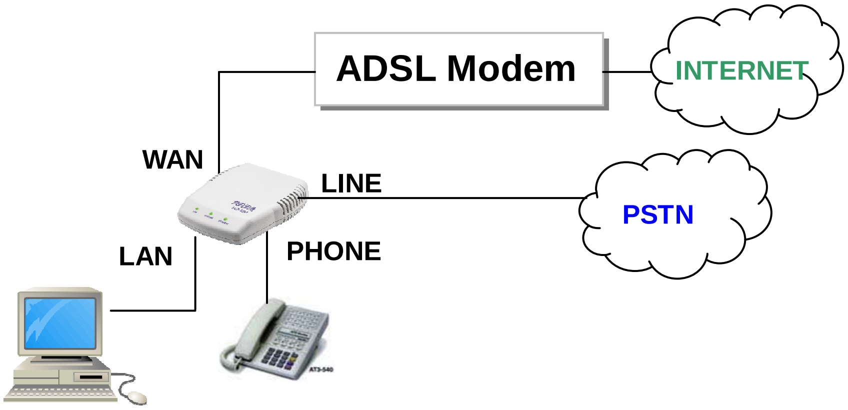
Group	IVR Action	Phone Command	Remarks
Status	Check TA LAN IP Address	**#120#**	IVR will report the current TA local IP address. Hang up while hearing end tone.
Status	Check IP Type	**#121#**	IVR will report if WAN DHCP in enabled or disabled. Hang up while hearing end tone.
Status	Check Phone Number	**#122#**	IVR will report registered phone number.
Status	Check Network Mask	**#123#**	IVR will report WAN network mask.
Status	Check Gateway IP Address	**#124#**	IVR will report WAN Gateway IP address.
Status	Check Primary DNS Server IP Address	**#125#**	IVR will report WAN Primary DNS Server IP address.
Status	Check TA WAN IP Address	**#126#**	IVR will report the TA WAN IP address.
Status	Check Firmware Version	**#128#**	IVR will announce the firmware version.
Status	Set WAN interfaces Speed	**#129+[0-4]#**	0: Auto, 1: 100M Full, 2: 100M Half, 3: 10M Full, 4: 10M Half
Setting	Set WAN DHCP client	**#111#**	This set WAN to DHCP Client mode.
Setting	Set WAN Static IP Address	**#112xxx*xxx*xxx*xxx#**	**Ex: #112061*066*159*009#** Note **xxx** must be 3 decimal digits. This setting will disable DHCP Client.
Setting	Set WAN Network Mask	**#113xxx*xxx*xxx*xxx#**	**Ex: #113255*255*255*000#** Note **xxx** must be 3 decimal digits and Must set WAN Static IP first (**#112**).
Setting	Set Router IP Address	**#114xxx*xxx*xxx*xxx#**	**Ex: #114061*066*159*254#** Note **xxx** must be 3 decimal digits and Must set WAN Static IP first (**#112**).
Setting	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	**Ex: #115159*168*001*001#** Note **xxx** must be 3 decimal digits and Must set WAN Static IP first (**#112**).
Setting	Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726-16K, 6: G.726-24K, 7: G.726-32K, 8: G.726-40K.
Setting	Set Handset Gain	**#131+[00~15]#**	**Ex: #13107#** and default is 06
Setting	Set Handset Volume	**#132+[00~12]#**	**Ex: #13209#** and Default is 10
Setting	Auto Configure Mode	**#150+[0~3]#**	**0:** Disable, **1:** TFTP mode **2:** FTP mode 3 : HTTP mode

Group	IVR Action	Phone Command	Remarks
Setting	Enable Call Waiting	**#138#**	This will disable Call Transfer.
Setting	Disable Call Waiting	**#139#**	This will enable Call Transfer.
Setting	Unlock Keypad	**#190#**	You must unlock keypad first in order to change settings by keypad.
Setting	Lock Keypad	**#191#**	Keypad can NOT be used for setting.
Setting	Reboot	**#195#**	After you hear "Option Successful," hang-up and TA will reboot automatically.
Setting	Factory Reset	**#198#**	TA will reset back to factory defaults. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST!
Setting	Blind Call Transfer	**#510#xxxxxx#**	**Ex: #510#54321#** transfer to 54321
Setting	Attendant Call Transfer	**#511#xxxxxx#**	**Ex: #511#54321#** transfer to 54321
Setting	3-Way Conference Call	**#512#xxxxxx#**	**Ex: #512#54321#** conference with 54321
Setting	Attendant Call Transfer to PSTN	**#514#xxxxxxxx#**	**Ex: #514#12345678#** transfer to 12345678 from IP to PSTN Line.
		0*	Switch to PSTN port, will hear dial tone of PSTN Line.
Setting		**1***	Select the First SIP server.
Setting		**2***	Select the Second SIP server.
Setting		**3***	Select the Third SIP server.

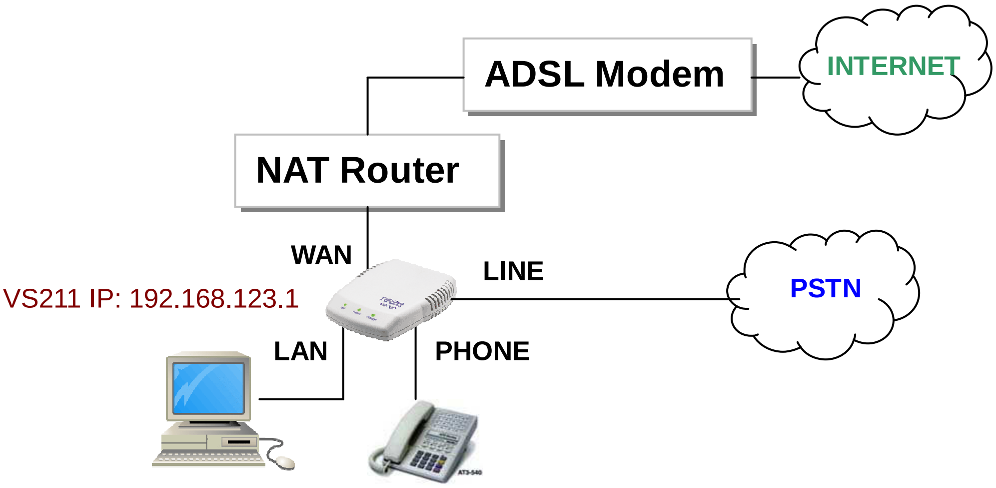
10. VoIP Applications Examples

You can use PC Web browser to configure VS211.
For example, enter **http://192.168.123.1** from PC web browser.

A. ADSL Connections without NAT Router for VS211



B. ADSL Connections with NAT Router for VS211



Example 1: Public Switched Telephone Network (PSTN) Calling/Answering

Applications:

VS211 is default at the VoIP mode. For PSTN calls, you may just pick up the phone, press **0*** key, and dial directly to the PSTN number like a normal telephone.

Configurations:

1. Select “**ON**” for the “**Auto Answer**” in Phone settings.
2. Select a value for Auto Answer Ring Counter, and the default value is set at **3**.
3. Click the “**Submit**” button.

Calling/Answering

4. Pick up the phone and press **0*** key for PSTN mode, and you should hear a dial tone.
5. Press, e.g. **1234567**, to call the PSTN party with **1234567.** In a moment, you should hear a ring back tone, and wait for the called PSTN party to answer.
6. When receiving PSTN incoming call, you must pick up the phone within **4** rings to answer, otherwise the VoIP mode will answer automatically for IP extension call. Please refer to **Example 5** for more details of PSTN to SIP extension calls.
7. If the “**Auto Answer**” is OFF, the FXO port will become PSTN only, and the function of extension call from PSTN to SIP call will be disabled.

Example 2: SIP-to-SIP Calling/Answering

Applications:

The applications can be for ADSL connections as in either Diagrams A or B. Both parties are registered to SIP server with private IP under NAT router. The SIP-to-SIP calling works when both calling and answering parties are registered to SIP server with given registered phone numbers. For Diagram A without NAT router, you may select NAT mode to enable the embedded NAT router. For Diagram B with external NAT router, you may select Bridge mode to disable the embedded NAT. Please refer to **Example 11** for more detailed SIP server registrations.

Configurations:

1. Select either “**NAT**“ or “**Bridge**” in accord with your network in “**WAN settings**” page,
2. Select “**DHCP Client**” to automatically get an IP address from NAT router.
3. Remember to click the “**Submit**” button,
4. Select Active “**ON**” in the “**SIP settings / Service Domain**” page,

5. Enter the items of **Register Name, Register Password, Proxy Server, and Outbound Proxy,**
6. Select "**ON**" in the "**STUN setting**", if required by your ITSP.
7. Upon successful SIP registration, the PHONE LED will start **Green** flashing.

Callings:

8. Pick up the phone, and you should hear a dial tone for VoIP mode.
9. Press **1688#** or **1688** to call the party with the registered SIP phone number **1688.** Note **#** key will dial out the number immediately. Dialing without **#** will not dial out until the auto dial timer (default=5 seconds) elapsed.

Example 3: SIP to PSTN Calling

Applications:

The applications can be for ADSL connections as in both Diagrams A and B. Both parties are registered to SIP server with either fixed real IP or private IP under NAT router. The SIP-to-PSTN calling works when both calling and answering parties are registered to SIP server with given registered phone numbers.

Configurations:

1. Same as in **Example 2.**
2. Select "**ON**" in the "**STUN setting**" page, if required by your ITSP.
3. Select "**ON**" for the "**Auto Answer**" and "**PIN Code**"in Phone settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**
4. After registration, the PHONE LED will be flashing to indicate a successful SIP registration.

Callings:

5. Pick up the phone for VoIP mode, and press **1688#** or **1688** to call the party with the registered SIP phone number **1688.**
6. After **4** rings of Auto Answer, the FXO port will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for PSTN. Note you must add the postfix "**#**". PIN Code is used to prevent from call piracy. Incorrect PIN Code will result in call disconnect. If PIN code is OFF, the caller may press PSTN number directly.
7. Press **1234567** to call the PSTN party number of **1234567.**

Example 4: SIP to Direct IP Calling

Applications:

The application is for the calling party with ADSL connection as in either Diagrams A or B. The calling party is registered to SIP server with either fixed real IP or private IP under NAT router. The answering party is with fixed real IP.

Configurations:

1. Same as in **Example 2.**
2. Make sure the PHONE LED is flashing continuously with a successful SIP registration.

Callings:

3. Pick up the phone for VoIP mode.
4. Press **211*21*191*4#** or **211*21*191*4** to call the party with the real IP address of **211.21.191.4.** In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

Example 5: PSTN to SIP Calling

Applications:

The applications can be for ADSL connections as in both Diagrams A and B. Both parties are registered to SIP server with either fixed real IP or private IP under NAT router.

Configurations:

1. Same as in **Example 2.**
2. Select "**ON**" for the "**Auto Answer**" and "**PIN Code**"in Phone settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234.**
3. Make sure the PHONE LED is flashing continuously with a successful SIP registration.

Callings:

4. Make a Call from PSTN line to the VS211 FXO number, e.g. **7654321.** In a moment, you should hear a ring back tone, and wait for the VS211 to answer. After **4** rings, the VoIP mode will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for VoIP mode. Incorrect PIN Code will result in call disconnect. Note that if PIN code is OFF, there will be not short beep tone and the caller may press SIP number directly.
5. Press **1688#** or **1688** to call the party with the registered SIP phone number **1688.** In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

Example 6: Direct IP to Direct IP Calling/Answering

Applications:

The applications are for ADSL connection without NAT router as in Diagram A. Both parties are with fixed real IP. The Direct IP calling works when both calling and answering parties are with known fixed IP. SIP server registrations are not required in this application.

Configurations:

1. Select "**Fixed IP**", and bridge "**ON**" in the "**Network / WAN settings**" page,
2. Enter the items of **IP, Subnet Mask, Gateway IP,**
3. Click the "**Submit**" button.
4. Make sure the SIP server is OFF (default is OFF) and PHONE LED is NOT flashing.

Callings:

5. Pick up the phone for VoIP mode.
6. Press **211*21*191*4#** or **211*21*191*4** to call the party with the real IP address of **211.21.191.4.** Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

Example 7: Direct IP to Direct IP Calling within NAT Router

Applications:

For the calling party in ADSL connection with NAT router as in Diagram B, this Direct IP calling can work when the answering parties are with fixed **private** IP addresses within the same VPN network, or with fixed **real** IP addresses.

Configurations:

1. Select "**Fixed IP**", and bridge "**ON**" in the "**Network / WAN settings**" page,
2. Enter the items of **IP, Subnet Mask, Gateway IP,**
3. Click the "**Submit**" button.
4. Make sure the SIP server is OFF (default is OFF) and PHONE LED is NOT flashing.

Callings:

5. Pick up the phone for VoIP mode.
6. Press **192*168*1*51#** or **192*168*1*51** to call the party with the private IP address of **192.168.1.51.** Press **211*21*191*4** to call the party with the real IP address of **211.21.191.4.** In a moment, you should hear a ring back tone, and wait for the called party to answer.

Example 8: Direct IP to PSTN Calling

Applications:

The Direct IP to PSTN calling is for the application when the answering parties are with known fixed real IP addresses. The SIP server registrations may not be necessary.

Configurations:

1. Same as in **Example 6**.
2. Select "**ON**" for the "**Auto Answer**" and "**PIN Code**" in Call settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234.**

Callings:

3. Pick up the phone for VoIP mode.
4. Press **211*21*191*4#** or **211*21*191*4** to call the party with the real IP address of **211.21.191.4.** After **4** rings, the FXO port will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone from PSTN. PIN Code is used to prevent from call piracy. Incorrect PIN Code will result in call disconnect. Note that if PIN code is OFF, the caller may press PSTN number directly.
5. Dial directly the number, e.g. **1234567** to call the PSTN party number of **1234567.** In a moment, you should hear a ring back tone, and wait for the called PSTN party to answer.

Example 9: PSTN to Direct IP Calling

Applications:

The PSTN to Direct IP calling is for the application when the answering parties are with known fixed private or real IP addresses. The SIP server registrations may not be necessary.

Configurations:

1. Same as in **Example 6**.
2. Select "**ON**" for the "**Auto Answer**" and "**PIN Code**" in Call settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**.

Callings:

3. Make a call from PSTN line to the VS211 FXO number, e.g. **7654321.** In a moment, you should hear a ring back tone, and wait for the VS211 to answer. After **4** rings of Auto Answer, the FXO port will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for VoIP mode.
4. Press **192*168*1*51#** or **192*168*1*51** to call the party with the private IP address of **192.168.1.51.** Press **211*21*191*4** to call the party with the real IP address of **211.21.191.4.** In a moment, you should hear another ring back tone, and wait for the VoIP called party to answer.

Example 10: 3-Way Conference Call, Call Transfer, Call Waiting, Hold

3-Way Conference Call, Call Transfer Applications:

These are for call transfer and conferencing among Parties A, B, and C. Three parties are registered to SIP server with either fixed real IP or private IP. There are two kinds of call transfer; Blind Transfer and Attendant Transfer.

Blind Transfer:

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#510# [Party C number] #** to transfer to Party C.

Attendant Transfer:

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#511# [Party C number] #** to call and talk to Party C. Hang up from Party B, then Party A will transfer and connect to Party C.

3-Way Conference Call:

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#512# [Party C number] #** to call and talk to Party C. Press Flash key from Party B, then Party C will join for the conferenc call.

Attendant Transfer to PSTN:

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#514# [PSTN number of Party C] #** to call and talk to Party C. Hang up from Party B, and Party A will be transferred from FXS port to FXO port and connect to PSTN number of Party C.

Call Waiting Application:

When a new call is coming while you are talking, you will hear an interrupt tone and you can push the Flash key to switch to answer the new call. You may push the Flash key to switch between the two calls.

Call Hold Application:

You may push the Hold (Flash) key to hold the current call for a while, then push Hold key again to resume talking.

Example 11: SIP-to-SIP Calling for http://www.inphonex.com/

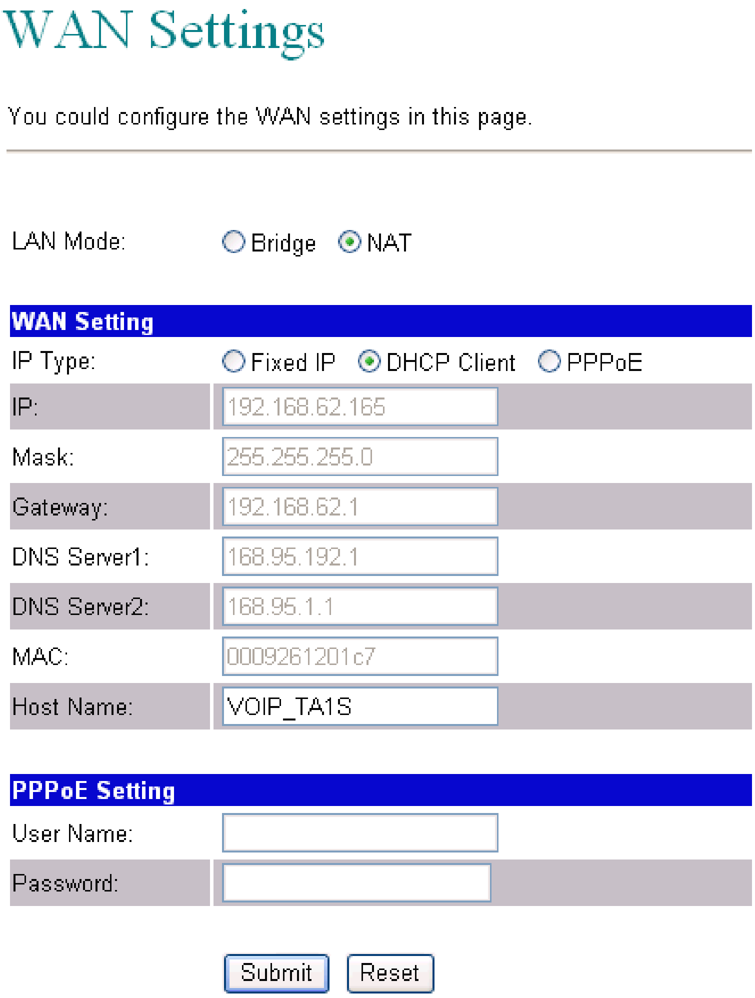
Applications:

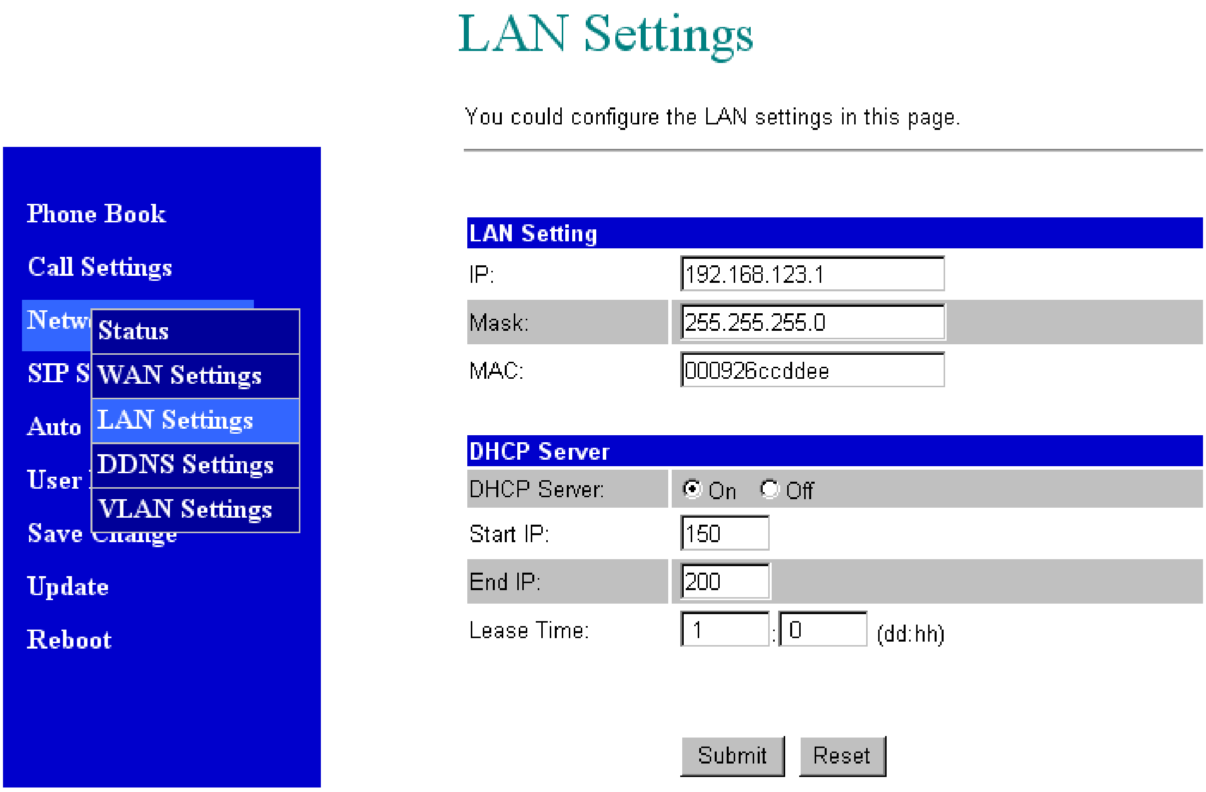
This shows how to use INPHONEX as an example for free ITSP provider. The applications are for both parties registered to INPHONEX SIP server.

1. Visit http://www.inphonex.com and sign up for a new registered account number. Follow the instructions for registration.
2. After finished, you will receive a mail sent by the INPHONEX mail system, and you will get one INPHONE phone number and password in the mail. For example, the register name/phone number is **7123456** with password **xxxx**.
3. Login to the Web configuration page.

Configurations:

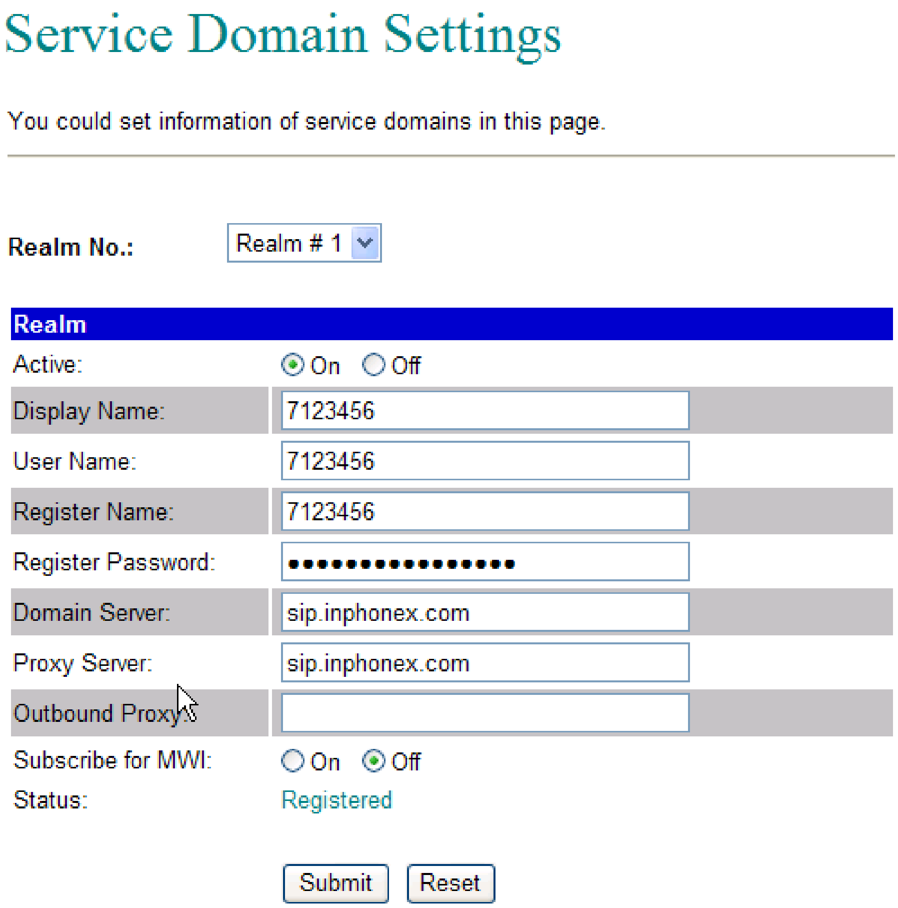
4. WAN and LAN Settings





5. SIP Settings

You have to enter the Display Name, User Name, Registered Name, Registered Password, Domain Server (sip.inphonex.com), Proxy Server (sip.inphonex.com), Outbound Proxy (sip.inphonex.com). After finished the setting, click the Submit button and the Save Change button. The system will reboot automatically. After system boot up, the SIP setting page will show “Registered”, and the PHONE LED will start flashing.



INPHONEX SIP Server Register Name: **7123456**
Password: **xxxx**
Domain Server: **sip.inphonex.com**
Proxy Server: **sip.inphonex.com**

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.729
Codec Priority 2:	G.711 u-law
Codec Priority 3:	G.711 a-law
Codec Priority 4:	G.723
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40
Codec Priority 9:	GSM

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	On / Off (selected)

Voice VAD	
Voice VAD:	On / Off (selected)

Submit Reset

Callings:

6. Pick up the phone for VoIP mode. (Your INPHONEX phone number 7123456).
7. Press **7123455** to call the party with the registered INPHONEX phone number **7123455.** In a moment, you should hear the ring back tone, and wait for the called party to answer.

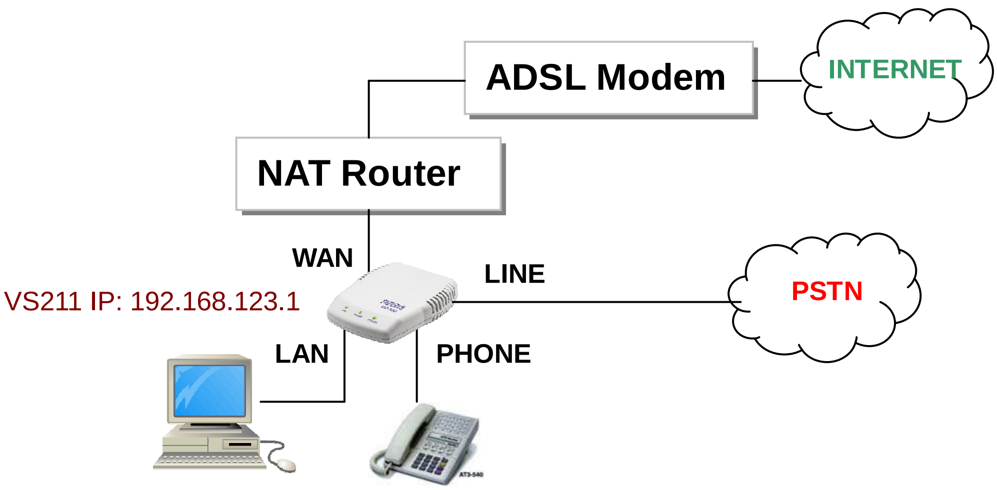
11. Trouble Shooting for Web Configurations

11.1. DO NOT HEAR DIAL TONE?

The phone port of VS211 is set to VoIP mode at default. When you pick up the phone and hear a busy tone, it indicates the WAN port is NOT connected. Make sure the ADSL Ethernet cable is connected to the WAN port of VS211 and Power Reset again. You may press **0*** key to switch to PSTN line, and you should hear another dial tone from PSTN. If not, please make sure the PSTN line is connected to the LINE port.

11.2. CAN NOT ACCESS WEB PAGE?

IE Web Browser is a useful tool to configure VS211. When you have difficulties in accessing the default IP address **http://192.168.123.1** of VS211 as in the following figure, the most possibility is that the PC might have different subnet IP settings from **192.168.123.xxx**. In this case, you must change VS211 IP address to the same subnet as PC and NAT router.

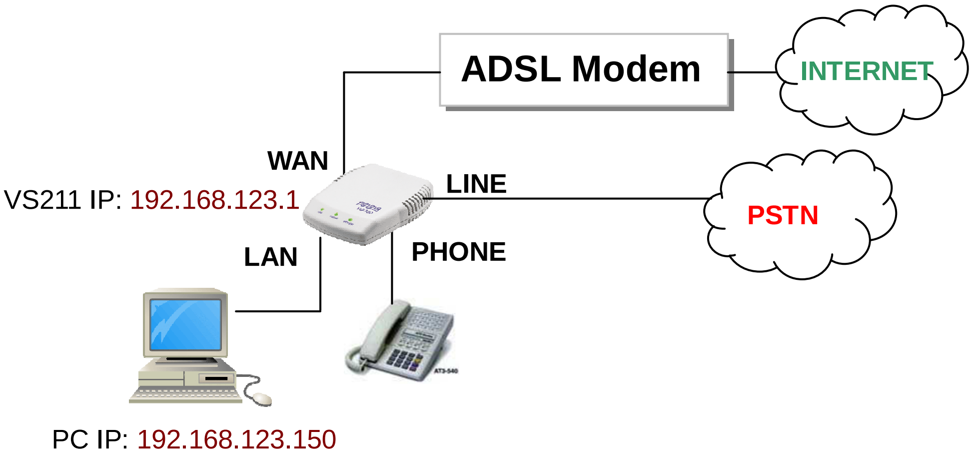


Example: To change VS211 IP address to the same subnet as PC and NAT router

1. Pick up the phone and press **#111#** from the phone to enable DHCP Client mode. VS211 will reboot, and LED will start flasing to get an IP address from NAT DHCP server.
2. Pick up the phone and press **#120#** to obtain the VS211 IP address from telephone IVR, for example, **192.168.62.51.**
3. Enter from IE web browser **http://192.168.62.51** to login VS211 web page for configurations.

11.3. ONLY ONE IP AVAILABLE FROM ADSL/CABLE SERVICE PROVIDER?

Sometimes only one IP address is available from Internet Service Provider (ISP) without NAT router as the following figure. Usually, a DHCP or PPPoE server at the central side of ISP is used to assign one IP address to each user. In this case, you may need to enable the embedded NAT router of VS211 to provide more than one IP addresses for PC and VS211.



Example: To change PC IP address to the same subnet as **192.168.123.1** for VS211

1. As in Window 2000 (my computer),
   - At "Network and Dialup Connections", right click on "Local Area Connection", then click on property.
   - The "Local Area Connection Properties" window will pop up.
   - Double click on "Internet Protocol (TCP/IP)".
   - The "Internet Protocol (TCP/IP) Properties" window will pop up.
   - Click on "Use the following IP Address".
   - Enter IP: **192.168.123.150** (150 can be any number other than 1, which is used by VS211).
   - Enter Subnet mask: **255.255.255.0**
   - Enter Default gateway: **192.168.123.1**
   - Click on OK button.
3. You will lose internet connection at this time.
4. At IE browser, type **http://192.168.123.1**
5. Follow the example in "Advanced Settings for Embedded NAT" for web login.
6. At LAN setting, turn on DHCP server.
7. At WAN setting, choose "DHCP client" to work with your ADSL/Cable modem.
8. Save change, wait for VS211 to reboot.
9. Change your PC's "Internet Protocol (TCP/IP) Properties" back to "obtain an IP address automatically".
10. You may press **#120#** and **#126#** to listen to the IVR for LAN and WAN IP addresses.

11.4. VOIP EXTENSION CALLS TO PSTN ARE NOT WORKING?

You must enable the Auto Answer function in Call Settings in order to answer automatically by VoIP mode. The Auto Answer is disabled at default. Make sure the PSTN is connected to LINE port. When the ring count exceeds the number set in Auto Answer Counter, the FXO port will auto answer and allow for VoIP extension call.

Extension Calls from Internet VoIP to PSTN

If the incoming call is from VoIP, then FXO port will answer with a PSTN dial tone and allow caller to redial to PSTN phone number.

Extension Calls from PSTN to Internet VoIP

If the incoming call is from PSTN, then VS211 FXO port will answer with a short beep tone and allow caller to redial to VoIP Phone number.