User Manual

SNR-VG-60X0

Analog IP Gateway

4 or 8 FXO Ports

NAG LLC

Table of Content

1 WELCOME

Thank you for purchasing the SNR SNR-VG-60x0 IP Analog FXO Gateway. The SNR-VG-60x0 is a cost effective, easy to use and easy to configure IP communications solution for any business. The SNR-VG-60x0 supports popular voice Codecs and is designed for full SIP compatibility and interoperability with 3rd party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manages communication costs.

This manual will help you learn how to operate and manage your SNR-VG-60x0 Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, etc. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

1.1 Gateway SNR-VG-60x0 Overview

The SNR-VG-60x0 offers an easy to manage, feature rich IP communications solution for any small business or businesses with virtual and/or branch locations who want to leverage their broadband network and/or add new IP Technology to their current phone system. The SNR Enterprise Analog VoIP Gateway SNR-VG-60x0 series converts SIP/RTP IP calls to traditional PSTN calls and vice versa. There are two models - the SNR-VG-6040 and SNR-VG-6080, which have either 4 or 8 FXO ports respectively. The installation is the same for either model.

Caution: Changes or modifications to this product not expressly approved by SNR Technology, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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1.2 Safety Compliances

The SNR-VG-60x0 is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard.

Warning: use only the power adapter included in the SNR-VG-60x0 package. Using an alternative power adapter may permanently damage the unit.

1.3 Warranty

SNR has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from SNR, contact your SNR Sales and Service Representative for a RMA (Return Materials Authorization) number. SNR reserves the right to remedy warranty policy without prior notification.

2 PACKAGING

Unpack and check all accessories. The SNR-VG-60x0 package contains:

- One SNR-VG-60x0 VoIP adapter
- One universal power supply
- One Ethernet cable

2.1 Connect The SNR-VG-60x0

Managing the SNR-VG-60x0 gateway and connecting the unit to the VoIP network is very simple. Follow these four (4) steps to connect your SNR-VG-60x0 gateway to the Internet and access the unit's configuration pages.

1. Connect PSTN Line to the FXO1-FXO8 ports.

2. Insert the Ethernet cable into the WAN port of SNR-VG-60x0 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)

3. Connect a PC to the LAN port of SNR-VG-60x0 for initial configuration or if it is being used as a router.

4. Plug the power adapter into the SNR-VG-60x0 and into a power outlet.

Figure 1: Diagram of SNR-VG-60x0 Back Panel

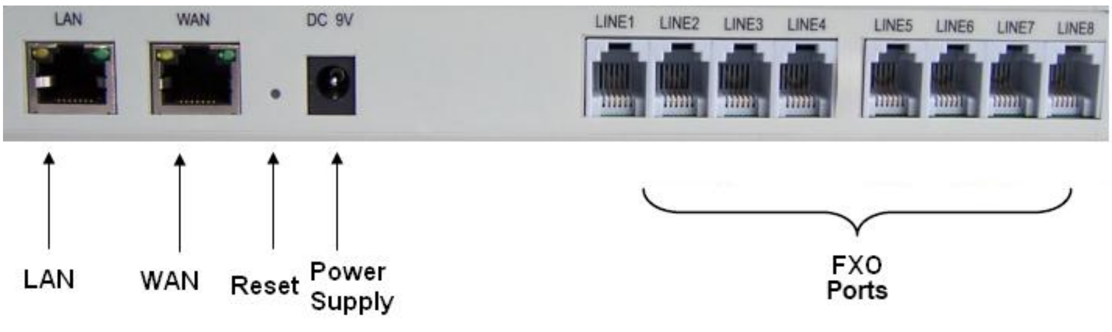


TABLE 1: Definitions Of The SNR-VG-Connectors

LAN (or PC)	Connect your PC to this port. It will then be assigned an IP address from your Router/DHCP Server. The SNR-VG-60x0 acts as a switch only.
WAN (or LAN)	Connect to the internal LAN network or Public Internet.
RESET	Factory Reset button. Press for 7 seconds to reset factory default settings.
POWER IN	Power adapter connection
FXO1 - FXO8	FXO ports to be connected to physical PSTN lines from a traditional PSTN PBX or PSTN Central Office.

Figure 2: Diagram Of SNR-VG-60x0 Display Panel

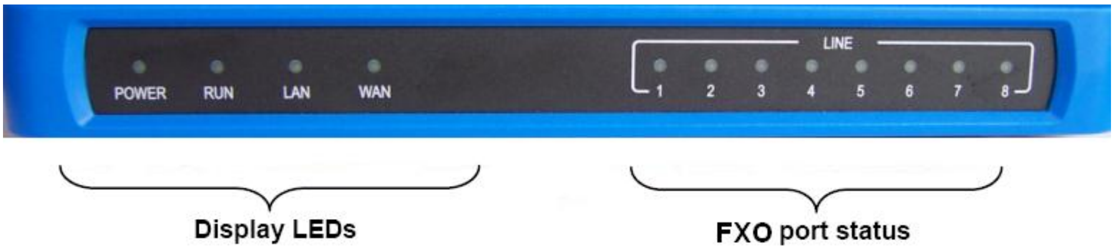


TABLE 2: Definitions Of The SNR-VG-Display Panel

Power LED	Indicates Power. Remains ON when Power is connected and turned ON.
Ready LED	Remains ON after boot-up.
LAN LED	Indicates LAN (or WAN) port activity
WAN LED	Indicates PC (or LAN) port activity
LEDs 1 - 8	Indicate status of the respective FXO Ports on the back panel Busy - ON Available - OFF.

NOTE:

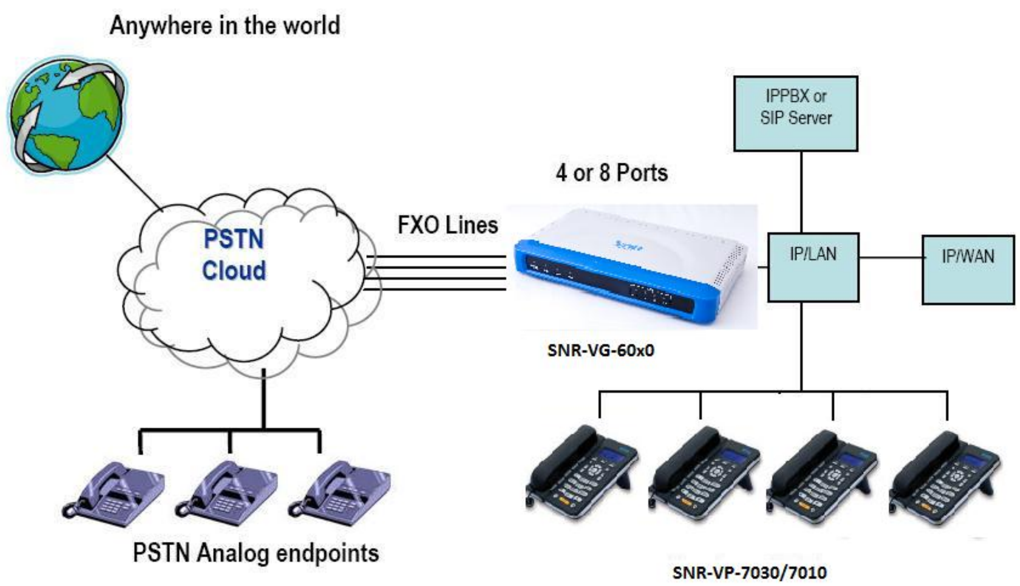
All LEDs display green when ON.

During a firmware upgrade or configuration download the following LED pattern will be observed:

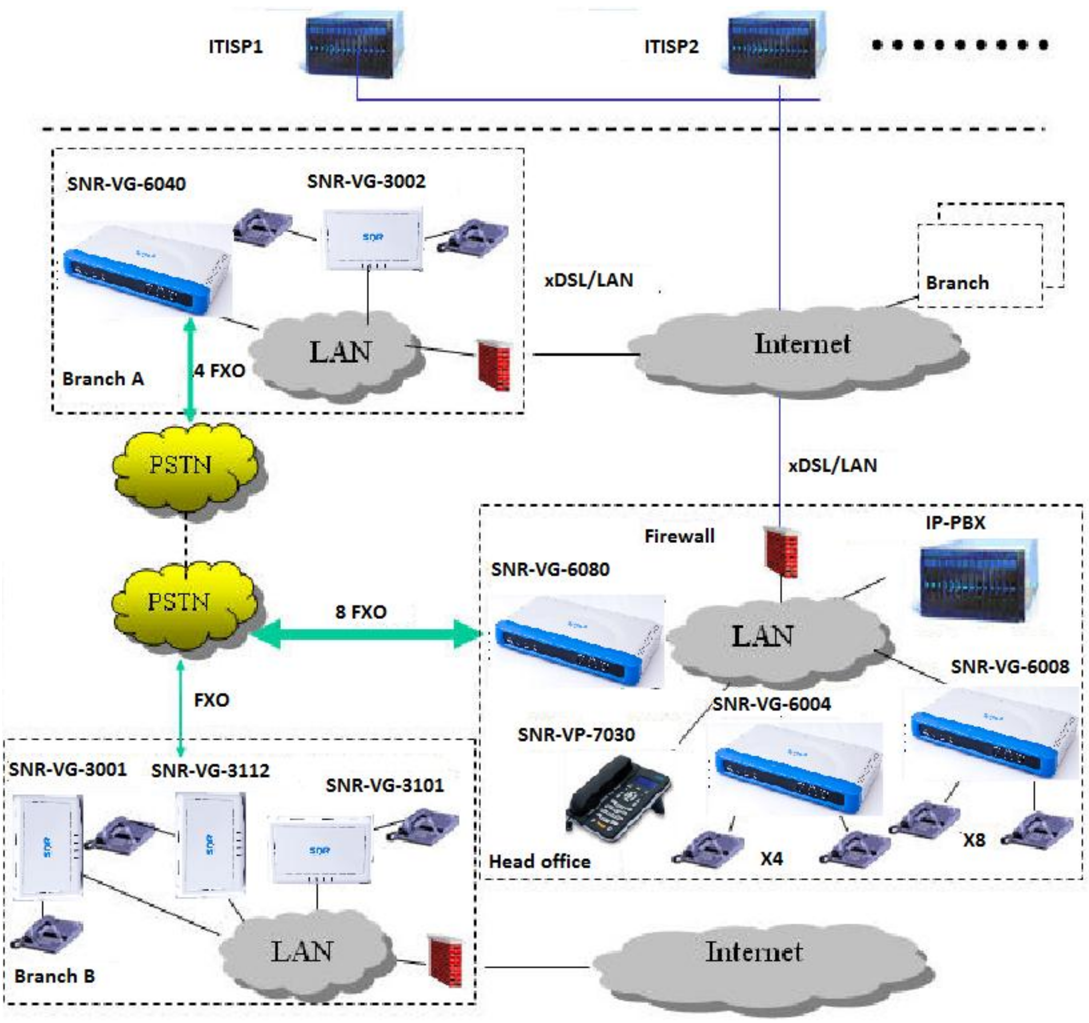
Power, WAN LEDs will be ON. The RUN LED will keep flashing during download and while the new files are written. The entire process may take between 5 to 15 minutes. The firmware upgrade is complete when you can login into the web configuration pages.

3 APPLICATION DESCRIPTION

3.1 Functional Diagram of IP-PBX & SNR-VG-60x0



3.2 SNR-VG-600x&SNR-VG-60x0 Scenario/Toll-Free Calling Between Locations



4 FEATURES

SNR-VG-60x0 is a next generation IP voice and video gateway that features full interoperability with leading IP-PBXs, SoftSwitches and SIP platforms. The Gateway series offers superb voice and video quality, traditional telephony functionality, simple configuration, feature rich functionality and an additional video port that enables the gateway to act like a video surveillance gateway.

4.1 Software Features Overview

- 4 and 8 FXO port gateways

- External power supply
- Two RJ-45 ports (switched or routed)
- TFTP and HTTP firmware upgrade support
- Multiple SIP accounts, multiple SIP profiles (choice of 2 profiles per account)
- Supports Audio Codecs: G711U/A, G723, G729A/B and GSM
- G.168 – echo cancellation
- Flexible DTMF transmission: In Audio, RFC2833, SIP Info or any combination of the 3
- Selectable, multiple LBR coders per channel
- T.38 compliant

TABLE 3: SNR-VG-60x0 Software Features

	SNR-VG-60x0 FXO Analog Gateway Series
IP settings	SNR-VG-6040: 4 ports; 4 SIP accounts w/ choice of 3 SIP Server profiles SNR-VG-6080: 8 ports; 8 SIP accounts w/ choice of 3 SIP Server profiles Round-robin port scheduling to ensure available lines to access PSTN networks
Telephone Interfaces	4 or 8 FXO, RJ11
Network Interface	Two (2) 10M/100 Mbps, RJ-45
LED Indicators	Power and Line LEDs
Voice over Packet Capabilities	G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice Compression	G.711U, G711A, G.723, G.729A/B, G.726
DHCP Server/Client	Switch Mode and PPPoE
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP and RTSP
PSTN Signaling	FXO Loop start, Current Disconnect.
DTMF Method	Flexible DTMF transmission method, User interface of In-audio, RFC2833, and SIP Info
IP Signaling	SIP (RFC 3261)
Provisioning	TFTP and HTTP
Control	TLS and SIPS (pending)
Management	Syslog support,remote management using Web browser
Short and long haul	REN3: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal / Wink	Yes (Detection only). The PSTN lines will need to be subscribed to PR service from the Service Provider.
EMC	SNR-VG-60x0: EN55022 Class B, CFR Part 15 Class B,

	EN55024; SNR-VG-6040: FCC, CE (in addition)
Safety	SNR-VG-60x0: EN60950-1 SNR-VG-6080: UL60950-1 (in addition)

4.2 Hardware Specification

TABLE 4: Hardware Specification Of SNR-VG-60X0

LAN interface	2xRJ45 10/100Mbps
LED	4 or 8 LEDs (GREEN)
Universal Switching Power Adaptor	Input: 100-240V AC, 50/60Hz, 0.5A Max Output: 9V DC, 2A UL certified
Dimension	225mm (L) x 172mm (W) x 42mm (H)
Weight	0.29 lbs (3.5 oz)
Temperature	32~104°F / 0~40°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC, CE

5 CONFIGURATION GUIDE

5.1 Configuration With Web Browser

The SNR-VG-60x0 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the gateway through any common web browser.

5.1.1 Accessing The Web Configuration Menu

The SNR-VG-60x0 HTML configuration menu can be accessed via LAN or WAN port:

From the LAN port:

1. Directly connect a computer to the LAN port.

2. Open a command window on the computer

3. Type in “ipconfig /release”, the IP address etc. becomes 0.

4. Type in “ipconfig /renew”, the computer gets an IP address in 192.168.22.x segment by

default

5. Open a web browser, type in the default gateway IP address. http://192.168.22.1.

   You will see the login page of the device.

From the WAN port:

The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port:

1. Enable the “WAN Port Web Access” option via IVR option 12.

2. Find the WAN IP address of the SNR-VG-60x0 using voice prompt menu option 02.

3. Access the SNR-VG-60x0 Web Configuration page by the following URI via WAN port:

http:// SNR-VG-60x0 -IP-Address (the SNR-VG-60x0 IP-Address is the WAN IP address for the SNR-VG-60x0).

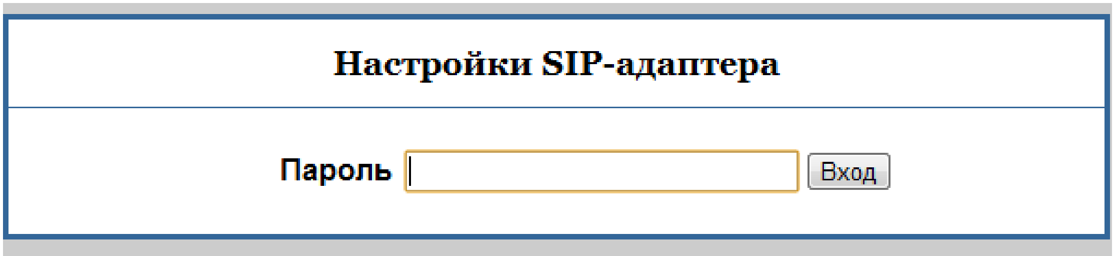
NOTE: If using a web browser to enter the configuration page, strip the leading “0”s because the browser will parse in octet. (i.e. if the IP address is: 192.168.001.014, please type in: 192.168.1.14).

5.2 End User Configuration

Once the HTTP request is entered and sent from a Web browser, the user will see a log in screen. There are two default passwords for the login page:

User	Password:	Level:
End User Level	1234	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

FIGURE 3: Screen-Shot Of SNR-VG-60x0 Log-In Screen



5.2.1 Status Page Definitions

DEVICE STATUS	
Setting Options	**Definitions**

MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
WAN IP Address	This field shows IP address of device
Product Model	Show product model of this device
Software Version	Information of software
System Uptime	Show system uptime since last reboot
PPPoE Link Up	Indicates where the PPPoE connection is up if the Uicorn2101 is connected to the DSL modem.
NAT	Indicate the NAT type behind which the device is when the stun feature is defined.
Port Status	Indicate the current status of the device such as hook (on/off),registion(registered/unregistered) etc.

5.2.2 Basic Settings Page

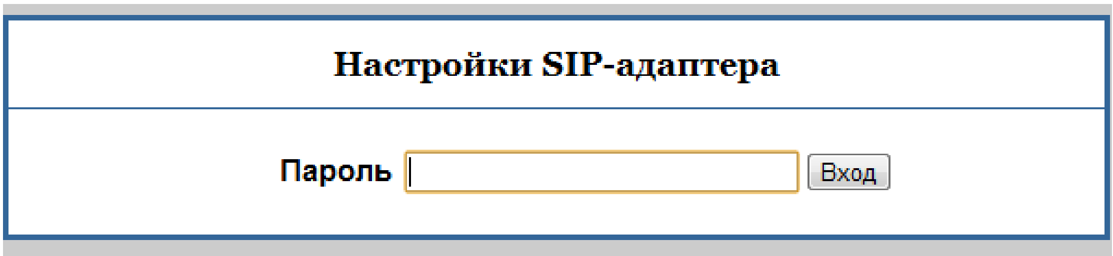
BASIC OPTIONS SETTING	
Setting options	**Definitions**
Web Port	Default is 80.
IP Address	There are 3 modes under which the SNR-VG-600x can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the chipset's memory). The SNR-VG-600x will acquire its IP address from the first DHCP server it discovers from the office/home network it is connected to. -To use the PPPoE feature, the PPPoE account settings need to be set. The SNR-VG-600x will attempt to establish a PPPoE session if any of the PPPoE fields have been entered with data. - If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured by the user. These fields are reset to zero by default.
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Allow DHCP Option 2 to override Time Zone setting	If set yes and under DHCP mode,the device will try to get option 2 from DHCP configure and overwrite Time Zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.

Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP:	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.22.1
DHCP IP Lease Time:	Value is set in units of hours. Default value is 120hr (5 Days) The time IP address are assigned to the LAN clients
Port Map	Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port
End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Reply to ICMP on WAN port	If set to "Yes", the SNR-VG-600x will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is **No**.
Wan Side Http Access	If this parameter is set to "No", the HTML configuration update via WAN port is disabled.

5.3 Super User Settings

The end-user needs to login to the Super user configuration page the same way as for the basic configuration page.

FIGURE 4: Screenshot Of Super User Configuration



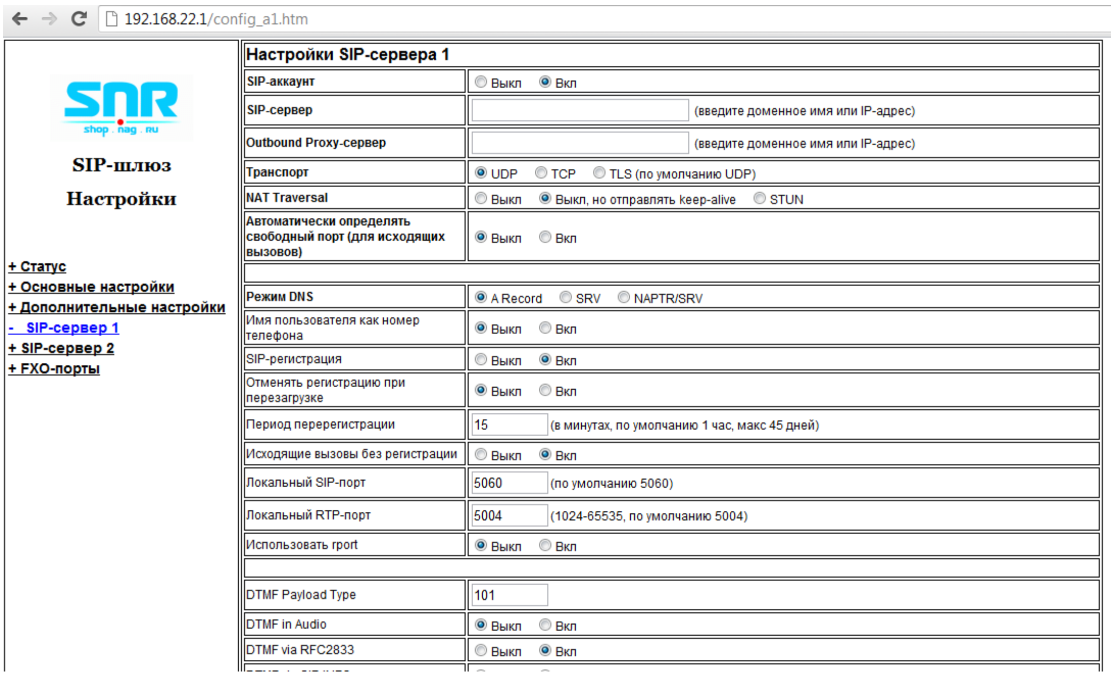
5.3.1 Super Configuration Page Definitions

Super Options	
Setting options	**Definitions**
Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for

	security reasons after clicking update and saved. The maximum password length is 26 characters,only digit or letter.
Home NPA	Local area code for North American Dial Plan.
Layer3 Qos	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer2 Qos	Value used for layer 2 VLAN tag. Default setting is blank
Data VLAN Tag	When using Bridge Mode, Data VLAN Tag is supported. when your PC connect to LAN Port, data (from your PC to switch) will be tagged with "Data VLAN Tag".
Stun sever is：	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the SNR-VG-600x sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
Firmware Upgrade and Provisioning:	Upgrade or provisioning through **TFTP** or **TFTP** server. --Upgrade Via:select HTTP or TFTP mode. --Allow DHCP Option: support 66,128,150. If select yes,device will get server information from DHCP option and ignore the Config Server Path. Option 66--TFTP server name(if you select Upgrade Via->TFTP), HTTP server name(if you select Upgrade Via->HTTP) Option 128--TFPT Server IP address.(if you select Upgrade Via->TFTP), HTTP Server IP address(if you select Upgrade Via->HTTP) Option 150--TFTP server address.(if you selectUpgrade Via->TFTP), HTTP server address(if you select Upgrade Via->HTTP)
Authenticate Conf File	configure file would be authenticated before acceptance if set to Yes
NTP server	This parameter defines the URI or IP address of the NTP server which is used by the SNR-VG-600x to set the current date/time.
Allow DHCP Option 42 to override NTP server	If set **Yes**,device can get NTP server from DHCP option 42.
Syslog Sever	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)
Syslog level	Default is blank, the feature is useful for the Internet

	Telephone Service Provider.
Download Device Configuration	User can download configuration from the web page and save to configuration file.
Call Progress Tones	Using these settings, users can configure tone frequencies and cadence according to their preference. By default they are set to North American frequencies. Configure these settings with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing (“On time” in ‘ms’) while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Example configuration for N.A. Dial tone: f1=350@-13,f2=440@-13,c=0/0; Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...] (Note: freq: 0 - 4000Hz; vol: -30 - 0dBm)
Restore Configuration	User can restore the before configuration from the configuration file saved at local pc

5.3.2 Profiles



Profiles are basically IP PBX / SIP Server configuration templates. If you have more than one IP PBX system or SIP Server that you would like to use with the SNR-VG-60x0, then you can configure Profile 1 or 2. Note – Make sure you select the correct profile for each channel under Channels WEBPAGE.

PROFILE PAGE DEFINITIONS	
Settings Options	**Definitions**
Account active	When set to Yes this profile is activated
SIP Server	SIP Server's URI or IP address
Outbound Proxy	SIP Outbound Proxy Server's URI or IP address
NAT Traversal	This parameter defines whether the SNR-VG-600x NAT traversal mechanism will be activated or not. If Choosing **No**,nothing to do. If Choosing **No, but send keep-alive**, the SNR-VG-600x will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open. If choosing **STUN** and a STUN server is also specified, then the SNR-VG-600x will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the SNR-VG-600x will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the SNR-VG-600x will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If choosing **STUN** with no specified STUN server, the SNR-VG-600x will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open. If choosing **UPNP**, the embedded UPNP client inside the SNR-VG-600x will attempt to mapping ports with the router by upnp protocol.
Ports Using The Profile Share With One Common Account	If set to "**Yes**", SNR-VG-60x0 will use the first account among the FXO PORTS of using the same profile. If set to "**No**", you need configure one port one account in FXO PORTS page.
Auto Select Idle Port (For Outgoing Call)	If set to "**Yes**", SNR-VG-60x0 will auto-select an idle Line to make outbound call to PSTN. If set to "**No**", you need configure one port one account in FXO PORTS page, then it is the business of SIP SERVER that it decide which idle FXO Port for outbound call.
Use DNS SRV	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone Number	If the SNR-VG-600x has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request

SIP Registration	This parameter controls whether the SNR-VG-600x needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is “No.” If set to “Yes”, then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the SNR-VG-600x refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Outgoing call without Registration	Default is No. If set to “Yes,” user can place outgoing calls even when not registered (if allowed by ITSP) but is unable to receive incoming calls.even when not registered (if allowed by ITSP) but is unable to receive incoming calls.
Local SIP port	This parameter defines the local SIP port the SNR-VG-600x will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS 2 port is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the SNR-VG-600x will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value for FXS port 1 is 5004. The default value for FXS 2 port is 5008.
Use Random Port	This parameter, when set to **Yes**, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple SNR-VG-600x are behind the same NAT.
Refer-To Use Target Contact	Default is **NO**. If set to **YES**, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
DTMF in Audio	This parameter specifies the mechanism to transmit DTMF digit in audio which means DTMF is combined in audio signal(not very reliable with low-bit-rate codec), Default is **YES**.
DTMF via RFC2833	This parameter specifies the mechanism to transmit DTMF digit via RTP (RFC2833). Default **YES**.
DTMF via SIP INFO	This parameter specifies the mechanism to transmit DTMF digit via SIP INFO. Default is **NO**.

Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
USE NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Preferred Vocoder	The SNR-VG-600x supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 32K), G.723.1, G.729A/B. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 6".
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the SNR-VG-600x will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames

	respectively.
G723 Rate	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
FAX Mode	Default is **T.30**(Fax Pass-Through), or T.38 (Auto Detect) FoIP
Fax Tone Detection Mode	Default is **Callee**. This decides whether Caller or Callee sends out the re INVITE for T.38 or Fax Pass Through.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions.
Dial Plan	Dial Plan Rules: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digits number; b. xx. ?at least 2 digits number; c. ^ - exclude; d. [3-5] - any digit of 3, 4, or 5; e. [147] - any digit 1, 4, or 7; f. <2=011> - replace digit 2 with 011 when dialing Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; if leading digit is 2, replace leading digit 2 with 011 before dialing. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }

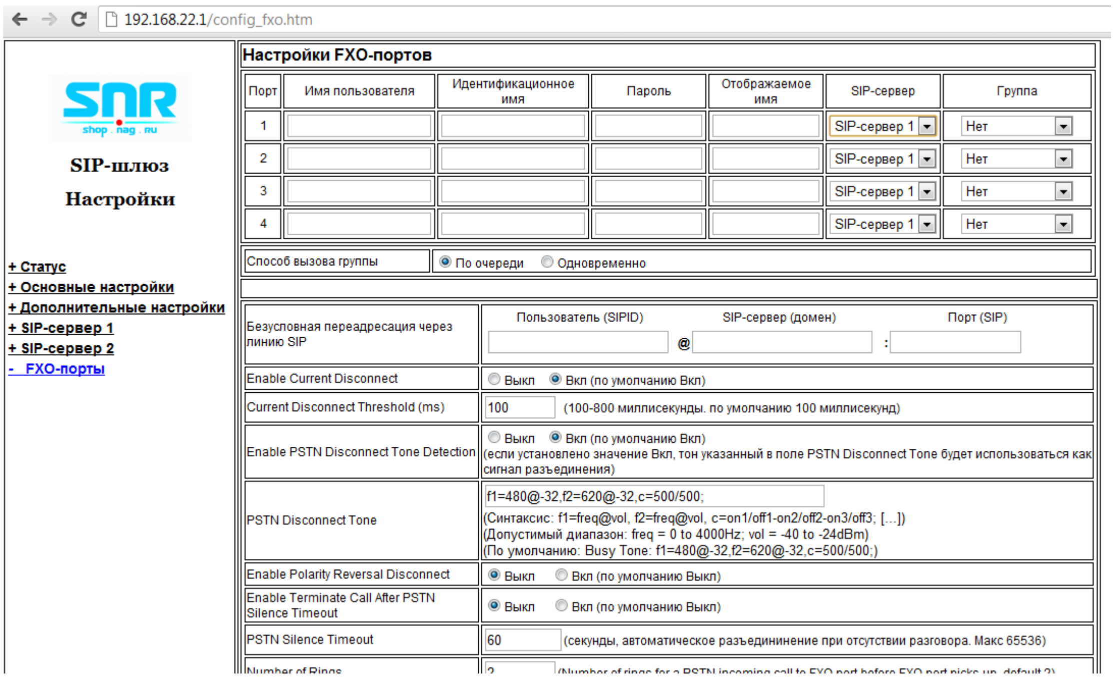
	Explanation of example rule (reading from left to right): ^1900x. - prevents dialing any number started with 1900 <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length 011[2-9]x. - allows international calls starting with 011 [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 Note: In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan should be: { [x*]+ }. More information can be availabled at Dail Plan Notes.
Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Session Expiration	The session timer extension enables SIP sessions to be periodically “refreshed” via a re-INVITE request. Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds. Default is 180 seconds.
Min-SE	The minimum session expiration (in seconds). Default is **90** seconds.
Caller Request Timer	If selecting “**Yes**” the device will use session timer when it makes outbound calls if remote party supports session timer. Default is **NO.**
Callee Request Timer	If selecting “**Yes**” the phone will use session timer when it receives inbound calls with session timer request. Default is **NO**.
Force Timer	If selecting “**Yes**” the device will use session timer even if the remote party does not support this feature. Selecting “**No**” will allow the device to enable session timer only when the remote party support this feature. To turn off Session Timer, select “**No**” for Caller Request Timer, Callee Request Timer, and Force Timer.

	Default is **NO**.
UAC Specify Refresher	As a Caller, select UAC to use the device as the refresher, or UAS to use the Callee or proxy server as the refresher. Default is **Omit**
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the device as the refresher. Default is **UAC**.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "**Yes**" to use INVITE method to refresh the session timer. Default is **NO.**
Send 200 OK Until FXO Has Detected Polarity Reversal	Default No. Check with your PSTN carrier before set to Yes
FXO Pick Up Incoming Call After Receive 200 OK Form Server	Default No.
Special Feature	Choose the selection to meet some special requirements from Soft Switch vendors. Default is **standard.**
Volume Amplification	Voice path volume adjustment. Rx is a gain level for signals transmitted by FXS Tx is a gain level for signals received by FXS. Default = 0dB for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the Tx Gain Level parameter located on the FXS Port Configuration page. If call volume is too low when using the FXS port (ie. the ATA is at user site), adjust volume using the Rx Gain Level parameter under the FXS Port Configuration page. If voice volume is too low at the other end, user may increase the far end volume using the Tx Gain Level parameter under the FXS Port Configuration page.
PSTN AC Termination	You can select the AC termination by Country or by Impedance.
Current Disconnect	Set it to "Yes" of the traditional PBX you are using with ATA/Gateway uses this method for signaling call termination. Default is No.
Current Disconnect Threshold(ms)	A configurable period of time in which the FXS port will drop off voltage on the line to indicate to the local party that the call is disconnected from the remote side. (100-800 ms.

	Default 100 ms)
Caller ID Minimum RX Level (dB)	An adjustable value for the Caller ID signal to help this device to recognize Caller ID from different networks. (-50 -0dB. Default -30dB)
Caller ID Transport Type	According to customer’s choice CID information will be transferred from PSTN network to VoIP network using following rules: 1. via SIP from - PSTN CID is in the SIP From field 2. via P-Asserted-Identity - SIP From field uses the pre-configured account user Id. PSTN CID is in the P-Asserted-Identity field 3. Send anonymous - SIP From field uses "anonymous". PSTN CID is put in the P-Asserted-Identity field 4. Disable - PSTN CID will not be sent. SIP From field uses the pre-configured account user ID
PIN for PSTN Calls	Enter digits to authorize calling PSTN numbers from VOIP, default is no.
PIN for VOIP Calls	Enter digits to authorize calling VOIP terminals from PSTN, default is no.

5.3.3 Configuring The FXO Channels

Configuring the FXO channels on the SNR-VG-60x0 is an easy process. Follow the GUI interfaces. The Device Status page terms are defined in **FXO Ports page**



FXO PORT SETTING

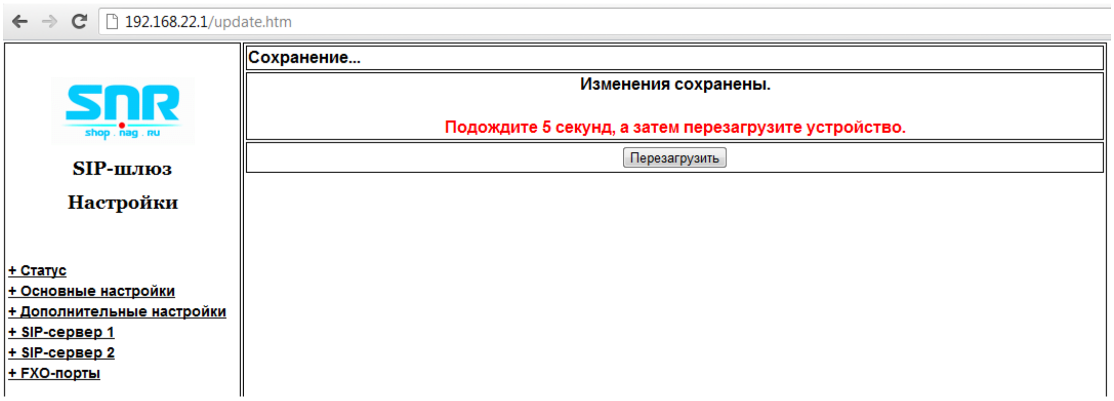
Setting Options	Meaning
FXO Port	FXS Port Number
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for SNR-VG-600x to register to (SIP) servers of ITSP.
Name	Name
Profile ID	Select the corresponding Profile ID (1/2)
Unconditional Call Forward to VOIP	Calls are unconditionally forwarded to the specified VoIP phone number once users dial the FXO port PSTN number
Enable PSTN Disconnect Tone Detection	If set to **Yes**, arrived Busy Tone is used as the disconnect signal.
PSTN Disconnect Tone	This configuration should be configured by the VoIP service provider. Some country use single frequency tone to signal PSTN disconnection, some country use double frequency tone. This setting can be configured to suit the telephone company's standard in different country.
Enable Polarity Reversal Disconnect	If set to **Yes**, the Polarity Reversal is used as the disconnect signal.
Enable Terminate Call After PSTN Silence Timeout	If set **Yes**, the device terminate the call when Silence timer expire.
PSTN Silence Timeout	Silence timer value, default is 60 minutes.
Number of Rings	Number of rings for a PSTN incoming call to FXO port before FXO port picks up, default 2
Min Delay Before Dial PSTN(ms)	Default is 500ms. This needs to be equal to or greater than the Current Disconnect threshold setting. Once the threshold is reached the gateway can dial out. This parameter should only be used if there are PSTN line detection issues.
DTMF Digit Volume(dB)	Default value is 11dB.
DTMF Digit Length(x10ms)	Digit length and Dial Pause are port digit dialing configurations; FXO needs to dial out digits for VOIP to PSTN 1 stage calls, and unconditional call forward to

	PSTN, and route to PSTN. Digit Length is the play time for each digit. Note: In order to receive the caller ID information, the delay should be set to a value larger than the delay required to complete the PSTN caller ID delivery. Please note that the value will be multiplied by 10ms.
DTMF Dial Pause(x10ms)	Dial pause is the time between 2 digits for the same scenario as explained above. Please note that the value will be multiplied by 10ms.
Stage Method(1/2)	This configuration is applicable for VoIP to PSTN calls and indicates one or two stage dialing methods.
Unconditional Call Forward to PSTN	Calls are unconditionally forwarded to the specified PSTN phone number once users dial the FXO port VoIP number. Each port can be setted independence.

5.4 Saving The Configuration Changes

Once a change is made, press the “Update” button in the Configuration Menu. The SNR-VG-60x0 will display the following screen to confirm that the changes have been saved. To activate changes, reboot or power cycle the SNR-VG-60x0 after all changes are made.

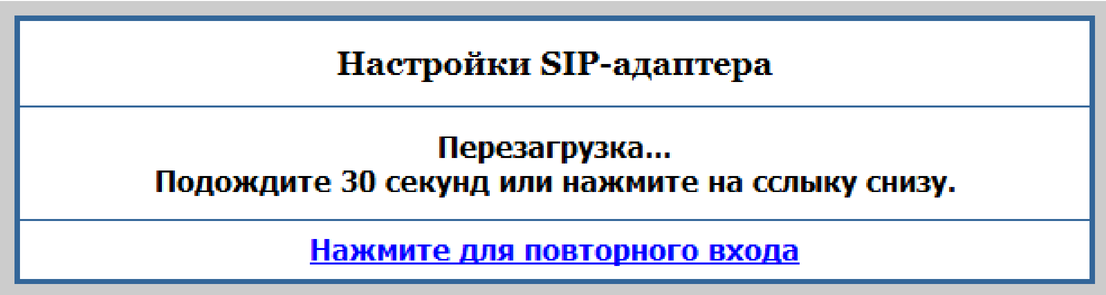
FIGURE 5: Screen-Shot Of Save Configuration



5.5 Rebooting From Remote

The administrator can remotely reboot the unit by pressing the “Reboot” button at the bottom of the configuration menu. The following screen will indicate that rebooting is underway.

FIGURE 6: Screen-Shot Of Rebooting



The user can re-login to the unit after waiting for about 30 seconds.

6 FIRMWARE UPGRADE

To upgrade software, SNR-VG-60x0 can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the SNR-VG-60x0.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the SNR-VG-60x0's Web configuration interface. To configure the TFTP server via voice prompt, follow section 5.1 with option 06, once set up the TFTP IP address, power cycle the ATA, the firmware will be fetched once the ATA boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the SNR-VG-60x0. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, please power cycle the SNR-VG-60x0.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For those who do not have a local TFTP server, SNR technology provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Service section of SNR's Web site to obtain this TFTP server's IP address.

NOTES:

When SNR ATA boot up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is "cfg.txt" and the other is "cfg001fc1xxxxxx", where "001fc1xxxxxx" is the MAC address of the SNR-VG-60x0. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.

7 RESTORE FACTORY DEFAULT SETTINGS

There are two (2) methods for resetting your unit:

Reset Button

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.

2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.

3. Insert a pin in this hole, and press for about 8 seconds.

4. Take out the pin. All unit settings are restored to factory settings.

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.

2. Enter "99" and wait for "reset" voice prompt.

3. Enter 862584658050

NOTE:

1. Factory Reset will be disabled if the **"Lock keypad update"** is set to "Yes".

2. Please be aware by default the SNR-VG-60x0 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port.

8 TECHNICAL SUPPORT CONTACT

Email: Support@nag.ru