

# **User Manual**

**SNR-VG-3002**

**Analog Telephone Adaptor**

# Table of Content

|   |           |
|---|-----------|
| <b>1. WELCOME</b>                               | <b>3</b>  |
| <b>2. WHAT IS IN THE PACKAGE</b>                | <b>3</b>  |
| <b>3. PRODUCT OVERVIEW</b>                      | <b>4</b>  |
| 3.1. Key Features                               | 4         |
| 3.2. Hardware specification                     | 5         |
| <b>4. INSTALLATIONS</b>                         | <b>5</b>  |
| 4.1. Safety                                     | 6         |
| <b>5. BASIC OPERATIONS</b>                      | <b>7</b>  |
| 5.1. Get Familiar with Voice Prompt             | 7         |
| 5.2. Make Phone call                            | 9         |
| 5.2.1. Calling Phone or Extension Numbers       | 9         |
| 5.2.2. Call Hold                                | 9         |
| 5.2.3. Call Waiting                             | 9         |
| 5.2.4. 3-way Conferencing                       | 9         |
| Star Code Style 3-way Conference                | 9         |
| Bellcore Style 3-way Conference                 | 10        |
| 5.2.5. Direct IP-to-IP Calls                    | 10        |
| 5.2.6. Blind Transfer                           | 11        |
| 5.2.7. Attended Transfer                        | 11        |
| 5.2.8. Hunting Group                            | 12        |
| 5.2.9. Inter-port Calling                       | 12        |
| 5.3. CALL FEATURES                              | 13        |
| Call Features Tables                            | 13        |
| 5.4. FAX  | 13        |
| 5.5. Status Light Indicator                     | 14        |
| <b>6. CONFIGURATION GUIDE</b>                   | <b>14</b> |
| 6.1. Configuring SNR-VG-3002 using Web Browser  | 14        |
| 6.1.1. Accessing the Web Configuration          | 15        |
| 6.1.2. Programming SNR-VG-3002 via the LAN Port | 15        |
| 6.1.3. Programming SNR-VG-3002 via the WAN Port | 15        |
| 6.1.4. User Programming and Configuration       | 16        |
| 6.1.5. Passwords                                | 16        |
| 6.1.6. Configuration Options and Explanations   | 17        |
| 6.1.6.1 Device Status:                          | 17        |
| 6.1.6.2 Basic Options                           | 17        |
| 6.1.6.3 Super Options                           | 19        |
| 6.1.6.4 FXS PORT 1:                             | 21        |
| 6.1.6.5 FXS PORT 2:                             | 28        |
| 6.1.7. Saving the Configuration Changes         | 28        |

|           |  |           |
|-----------|--|-----------|
| 6.1.8.    | Rebooting the SNR-VG-3002 .....                  | 28        |
| 6.2.      | Configuring SNR-VG-3002 via Voice Prompt .....   | 28        |
| 6.2.1.    | DHCP Mode .....                                  | 28        |
| 6.2.2.    | STATIC IP Mode .....                             | 29        |
| 6.3.      | Configuration through a Central Server .....     | 29        |
| <b>7.</b> | <b>SOFTWARE UPGRADE .....</b>                    | <b>29</b> |
| <b>8.</b> | <b>RESTORE TO FACTORY DEFAULT SETTINGS .....</b> | <b>30</b> |
| <b>9.</b> | <b>TECHNICAL SUPPORT CONTACT .....</b>           | <b>30</b> |

## **1. WELCOME**

SNR-VG-3002 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The SNR-VG-3002 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

SNR-VG-3002 has two FXS ports.

- Special compatibility features include:
- Nortel MCS
- Standard SIP
- Broadsoft
- Howdy

## **2. WHAT IS IN THE PACKAGE**

The SNR-VG-3002 package contains:

- One SNR-VG-3002 VoIP adapter
- One universal power supply
- One Ethernet cable

## **3. PRODUCT OVERVIEW**

### **3.1. Key Features**

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPOE, STUN, TFTP, UPNP, etc.
- Built-in router, NAT.
- Device bridge mode support
- Supports dual SIP accounts via dual FXS ports
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various vocoders including G.711 (a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (32K), as well as G.729A, and ILBC ( pending ) .
- Support Caller ID/Name display or block, Hold, Call Waiting/Flash, Call Transfer, Call Forward, 3-way conferencing, Call Waiting Caller ID, in-band and out-of-band DTMF, Dial Plans, etc.
- Support fax pass through (for PCMU and PCMA) and T.38 FoIP (Fax over IP)
- Support syslog
- Support volume amplification
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Signaling and call encryption via TLS and SRTP, enabling enterprise-class security.
- Support snmp v2
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP
- Support firmware upgrade via TFTP or HTTP with encrypted configuration files.
- Ultra compact and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter.

### 3.2. Hardware specification

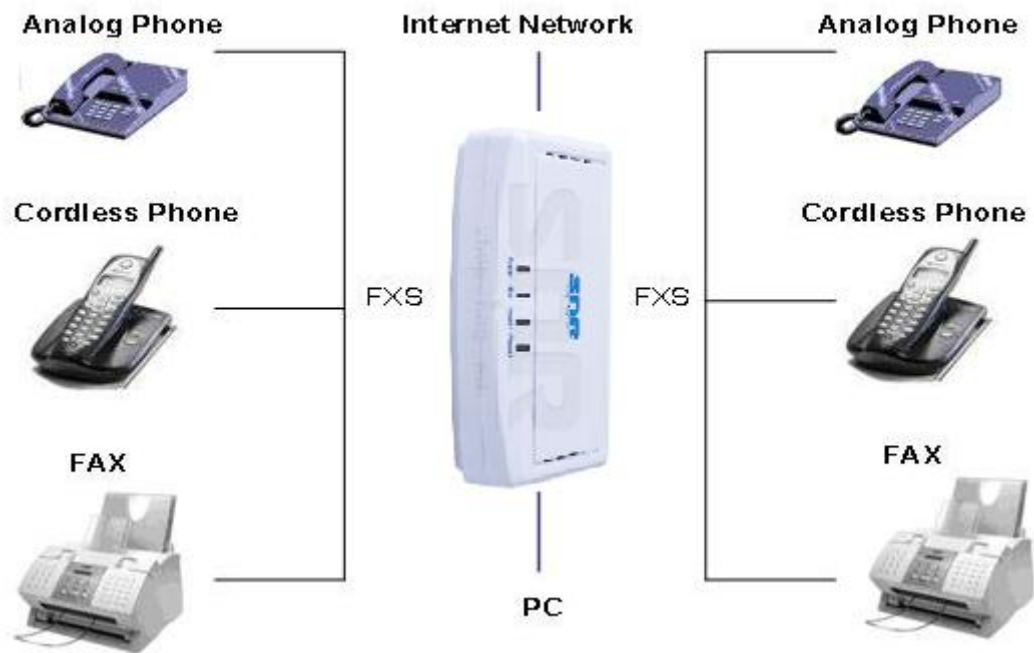
| Model                            | SNR-VG-3002  |
|----------------------------------|--|
| LAN interface                    | 1 x RJ45 10Base-T  |
| WAN interface                    | 1 x RJ45 10Base-T  |
| FXS telephone port               | 2 x FXS  |
| LED light                        | Green color  |
| Universal switching power supply | Input: 100-240VAC 50-60 Hz<br>Output: +5VDC, 1200mA,<br>UL certified |
| Dimension                        | 70mm (W) × 130mm (D) × 27mm (H)                                      |
| Weight                           | 0.30kg   |
| Temperature                      | 40 – 130 F<br>5 – 45 C   |
| Humidity                         | 10 - 90%   |

## 4. INSTALLATIONS

SNR-VG-3002 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The SNR-VG-3002 VoIP functionalities are available via regular analog telephones.

SNR-VG-3002 has two FXS ports. The PHONE1 port next to the LAN is FXS port 1 and the PHONE2 on the side of the SNR-VG-3002 is FXS port 2. Each FXS port can have a separate SIP account. This is a key feature of SNR-VG-3002. Both ports can make calls concurrently.



Following are the steps to install a SNR-VG-3002:

- Connect a standard touch-tone analog telephone (or fax machine) to PHONE1 port.
- Connect another standard touch-tone analog telephone (or fax machine) to PHONE2 port.
- Insert the Ethernet cable into the WAN port of SNR-VG-3002 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
- Connect a PC to the LAN port of SNR-VG-3002.
- Insert the power adapter into the SNR-VG-3002 and connect it to a wall outlet.

#### 4.1. Safety

The SNR-VG-3002 is compliant with various safety standards including FCC/CE and C- Tick. Its power adaptor is compliant with UL standard. The SNR-VG-3002 should only operate with the universal power adaptor provided in the package.

**Warning: Please do not use a different power adapter. Using other power adapter may damage the SNR-VG-3002 and will void the manufacturer warranty!**

Caution: Changes or modifications to this product not expressly approved by SNR, 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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mechanical, for any purpose without the express written permission of SNR.

## 5. BASIC OPERATIONS

### 5.1. Get Familiar with Voice Prompt

SNR-VG-3002 has stored a voice prompt menu for quick access to settings and simple configuration. You can enter this voice prompt menu two ways:

- Pick up the receiver (or press the Handsfree button) of the analog telephone and press “\*\*\*”,

A voice will say, “Enter the new option.” At this point, you can select from the following menu voice prompt options to begin using the SNR-VG-3002:

| Menu             | Voice Will Say the Following: |   |
|------------------|-------------------------------|---|
| <b>Main Menu</b> | “Enter a Menu Option”         | Enter “*” for the next menu option<br>Enter “#” to return to the main menu<br>Enter 01 – 07, 12 - 17, 47, 86 or 99 Menu option  |
| <b>01</b>        | “DHCP Mode”, “Static IP Mode” | Enter ‘9’ to toggle the selection<br>If user selects “Static IP Mode”, user need configure all the IP address information through menu 02 to 05.<br>If user selects “Dynamic IP Mode”, the device will retrieve all IP address information from DHCP server automatically when user reboots the device. |
| <b>02</b>        | “IP Address “ + IP address    | The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.   |
| <b>03</b>        | “Subnet “ + IP address        | Same as Menu option 02  |
| <b>04</b>        | “Gateway “ + IP address       | Same as Menu option 02  |
| <b>05</b>        | “DNS Server “ + IP address    | Same as Menu option 02  |
| <b>06</b>        | “MAC Address”                 | Announces the Mac address of the unit.  |
| <b>07</b>        | Preferred Vocoder             | Enter “9” to go to the next selection in the list: <ul style="list-style-type: none"> <li>● PCM U</li> <li>● PCM A</li> <li>● G-726</li> <li>● G-723</li> <li>● G-729</li> </ul>  |

|    |                                 |  |
|----|---------------------------------|--|
| 12 | WAN Port Web Access             | Enter "9" to toggle between enable and disable   |
| 13 | Firmware Server IP Address      | Announces current Firmware Server IP address. Enter 12 digit new IP address.   |
| 14 | Configuration Server IP Address | Announces current Config Server Path IP address. Enter 12 digit new IP address.  |
| 15 | Upgrade Protocol                | Upgrade protocol for firmware and configuration update.<br>Enter "9" to toggle between TFTP and HTTP   |
| 16 | Firmware Version                | Firmware version information.  |
| 17 | Firmware Upgrade                | Firmware upgrade mode. Enter "9" to rotate among the following three options:<br>1. always check<br>2. check when pre/suffix changes<br>3. never upgrade |
| 47 | "Direct IP Calling"             | Enter the target IP address to make a direct IP call, after dial tone. (See "Make a Direct IP Call".)  |
| 99 | "RESET"                         | Enter "9" to reboot the device; or<br>Enter MAC address to restore factory default setting (See Restore Factory Default Setting section)                 |
|    | "Invalid Entry"                 | Automatically returns to Main Menu   |

Other Menu Prompt Features:

- By pressing "\*\*\*\*" on the analog telephone, you can access the voice menu.
- "\*" shifts down to the next menu option.
- "#" returns to the main menu.
- "9" functions as the ENTER key in many cases to confirm an option.
- All entered digit sequences have known lengths-2 digits for menu option and 12 digits for IP address. Once all of the digits are collected, the input will be processed.
- Incorrect keyed entry cannot be deleted or undone. The SNR-VG-3002 will prompt you to start over by telling you that you made an error.



## **5.2. Make Phone call**

### **5.2.1. Calling Phone or Extension Numbers**

- Dial the number directly and wait for 4 seconds (Default is 4 seconds. To change the default, change the settings via the web configuration page under “No Key Entry Timeout”). Or
- Dial the number directly, and press # (assuming that “Use # as Dial Key” is set to “YES” during web configuration of your SNR-VG-3002).

Other functions available during the call are call-waiting/flash, call-transfer, and call-forward. Your SIP gatekeeper/proxy server needs to support these features in order for them to work.

### **5.2.2. Call Hold**

While in conversation, pressing the “FLASH” button on the attached phone will put the remote end on hold. Pressing the “FLASH” button again will release the previously Hold party and the bi-directional media will resume.

### **5.2.3. Call Waiting**

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User can press the flash button to put the current call party on hold and switch to the other call. Pressing flash button toggles between two active calls.

### **5.2.4. 3-way Conferencing**

SNR-VG-3002 supports 3-way conference in two styles: star code style or Bellcore style.

#### **Star Code Style 3-way Conference**

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials \*23 then C’s number then # (or wait for 4 seconds).
3. If C answers the call, then A press “flash” to bring B, C in the conference.
4. If C does not answer the call, A can press “flash” back to talk to B.

## Bellcore Style 3-way Conference

Bellcore style 3-way conference is also supported. To do this, user needs to enable “Use Bell-style 3-way Conference” in FXS web configuration.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C’s number then # (or wait for 4 seconds).
3. If C answers the call, then A press “flash” to bring B, C in the conference.
4. If C does not answer the call, A can press “flash” back to talk to B.

### 5.2.5. Direct IP-to-IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

#### Elements necessary to completing a Direct IP Call:

1. Both SNR-VG-3002 and other VoIP Device, have public IP addresses, or
2. Both SNR-VG-3002 and other VoIP Device are on the same LAN using private IP addresses, or
3. Both SNR-VG-3002 and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

SNR-VG-3002 supports two ways to make Direct IP Calling:

#### Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial “\*\*\*\*”
2. Dial “47” to access the direct IP call menu
3. Enter the IP address using format ex. 192\*168\*0\*160 after the dial tone.

#### Using Star Code

1. Pick up the analog phone then dial “\*47”
2. Enter the target IP address using same format as above.

Note: NO dial tone will be played between step 1 and 2.

Destination ports can be specified by using “\*” (encoding for “:”) followed by the port number.

#### Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is

**\*47 or Voice Prompt with option 47, then 192\*168\*0\*160.**

followed by pressing the “#” key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:

**\*47 or Voice Prompt with option 47, then 192\*168\*0\*160\*5062** followed by pressing the “#” key, if it is configured as a send key or wait for 4 seconds.

**NOTE:** When completing direct IP call, the “**Use Random Port**” should set to “NO”. You can not make direct IP calls between FXS1 to FXS2 since they are using same IP.

### **5.2.6. Blind Transfer**

Assuming that call party A and party B are talking to each other on the phone. Party A wants to transfer party B to party C:

- Party A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
- Then party A dials \*87 then dials party C’s number, and then # (or wait for 4 seconds) Party A can hang up the phone.
- Note: Call features have to be activated during web configuration by selecting YES to “Enable Call Features”. These features need to be supported by your SIP gatekeeper /proxy server in order to work.
- Party A can hold on to the phone and wait for one of the three following events:
  1. A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK signal from transfer target). At this point, party A can either hang up or make another call.
  2. A quick busy tone followed by a restored call (on supported SIP gatekeeper platforms only). This means the transferee has received a 4xx response signal for the INVITE and will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
  3. Busy tone keeps playing. This means the SNR-VG-3002 has failed to receive the second NOTIFY signal from the transferee and decided to time out.

Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee uses a device that does not support the second NOTIFY signal, this will be the case. In poor or unstable network scenarios, this could also happen, although the transfer may have been completed successfully.

### **5.2.7. Attended Transfer**

Assuming that call party A and party B are in conversation. Party A wants to Attend Transfer party B to party C:

- Party A presses FLASH (on the analog phone, or Hook Flash for old model phones)

to get a dial tone.

- Party A then dials party C's number then # (or wait for 4 seconds). Party A and party C now are in conversation.
- Party A can hang

Note: When Attended Transfer failed and if party A hangs up, the SNR-VG-3002 will ring party A again to remind party A that party B is still on the call, by pressing FLASH or Hook again will restore the conversation between party A and party B.

### 5.2.8. Hunting Group

This feature allows user to setup a single SIP account on the gateway and have the ability to use all FXS ports to make/receive calls. Using this feature, all ports active in same hunt group will have the same phone number and incoming calls will be distributed in a round robin manner among the ports active in that hunt group.

Example configuration of a hunting group:

FXS Port #1: SIP UserID and Authenticate ID entered, **Account Active** set to "Yes"

FXS Port #2: SIP UserID and Authenticate ID left blank, **Account Active set to** Hunting Group.

### 5.2.9. Inter-port Calling

In some cases a user may want to make phone calls between SNR-VG-3002 ports when the gateway will be used as stand alone unit, without any SIP server. This feature will also be applicable when the SNR-VG-3002 is used in mode Hunting Groups and will be registered to SIP server only with one master number. In such cases users still will be able to make inter-port calls by using the IVR feature. For example the user connected to port 1 can reach the user connected to port 3 by dialing \*\*\* and 72. Digit 7 indicated using inter-port calling feature, digit 2 indicates port number which should be reached. At the same manner the user connected to port 2 can reach the user connected to port 1 by dialing \*\*\* and 71.

Note: The SNR-VG-3002 support fast mode for inter-port calling by using “\*48x”. For example PORT 1 call PORT 2, just dail “\*482”, PORT 2 call PORT 1, just need to dail “\*481”.

## 5.3. CALL FEATURES

### Call Features Tables

Following table shows the call features of SNR-VG-3002:

| Key        | Call Features   |
|------------|---|
| *23        | 3-way conference  |
| *87        | Blind Transfer  |
| *30        | Block Caller ID (for all subsequent calls)  |
| *31        | Send Caller ID (for all subsequent calls)   |
| *67        | Block Caller ID (per call)  |
| *82        | Send Caller ID (per call)   |
| *50        | Disable Call Waiting (for all subsequent calls)   |
| *51        | Enable Call Waiting (for all subsequent calls)  |
| *70        | Disable Call Waiting. (Per Call)  |
| *71        | Enable Call Waiting (Per Call)  |
| *72        | Unconditional Call Forward.<br>To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up. |
| *73        | Cancel Unconditional Call Forward.<br>To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.                             |
| *90        | Busy Call Forward.<br>To use this feature, dial “*90” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.          |
| *91        | Cancel Busy Call Forward.<br>To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.   |
| *92        | Delayed Call Forward.<br>To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone and then hang up.         |
| *93        | Cancel Delayed Call Forward.<br>To cancel this feature, dial “*93”, get the dial tone, and then hang up.  |
| Flash/Hook | call waiting indication.<br>When in conversation without an incoming call, this action will switch to a new channel to make a new call.                   |

## 5.4. FAX

Unicorn3002 supports FAX in two modes: T.38 (Fax over IP) and fax pass through. T.38 is the preferred method because it is more reliable and works well in most network

conditions. If the service provider supports T.38, please use this method by selecting Fax mode to be T.38. If the service provider does not support T.38, pass-through mode may be used. To send or receive faxes in fax pass through mode, users will need to select all the Preferred Codecs to be PCMU/PCMA.

## 5.5. Status Light Indicator

Following tables show the SNR-VG-3002 button light pattern indication. But the Button light Show the PHONE1 status only.

Red light always indicates not normal status, Green light mostly indicate normal working status.

| Light Indicator    | Status Meaning  |
|--------------------|---|
| <b>Power Light</b> | Indicates Power. Remains ON when power is connected   |
| <b>Run Light</b>   | 1 second ON then 1 second OFF Indicates device software is running.<br>Fast blinking indicate the product in firmware upgrading or provision state                                      |
| <b>Phone Light</b> | Indicate status of the FXS Port on the back panel<br>Busy – ON (Solid Green)<br>Available – OFF<br>Fast blinking – Ringing<br>Slow blinking FXS LEDs indicates voicemail for that port. |

## 6. CONFIGURATION GUIDE

### 6.1. Configuring SNR-VG-3002 using Web Browser

SNR-VG-3002 has embedded Web server and HTML pages that allow users to configure the SNR-VG-3002 through an easy-to-use Web browser interface such as Microsoft's Internet Explorer or Netscape browser. Below is a screen shot of the SNR-VG-3002 configuration page:

| Основные настройки     |   |
|------------------------|---|
| Порт для доступа к Web | 80 (по умолчанию 80)  |
| Настройки WAN          | <input checked="" type="radio"/> DHCP<br>DHCP hostname <input type="text"/> (Опция 12)<br>DHCP domain <input type="text"/> (Опция 15)<br>DHCP vendor class ID <input type="text"/> (Опция 60)<br>DHCP user class <input type="text"/> (Опция 77)  |
|                        | <input type="radio"/> PPPoE<br>Пользователь <input type="text"/><br>Пароль <input type="text"/><br>PPPoE Service Name <input type="text"/><br>DNS-сервер <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>  |
|                        | <input type="radio"/> Статический IP-адрес<br>IP-адрес <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/><br>Маска подсети <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/><br>Шлюз <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/><br>Основной DNS <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/><br>Дополнительный DNS <input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/> |

### 6.1.1. Accessing the Web Configuration

The SNR-VG-3002 configuration page can be accessed via the LAN or WAN port.

### 6.1.2. Programming SNR-VG-3002 via the LAN Port

To program SNR-VG-3002 via the LAN port, directly connect an Ethernet cable from your PC to the LAN port of the SNR-VG-3002. After connecting the cable, confirm that the green light of the LAN port is on. If the green light is not on, this means that your PC is not yet properly connected to the SNR-VG-3002 via the LAN port.

For LAN port configuration, use the following default IP address to access the device:

<http://192.168.22.1>

### 6.1.3. Programming SNR-VG-3002 via the WAN Port

The WAN port access for web configuration is disabled by default from the factory. To access the web configuration menu from the WAN port, you must first access the device via the device LAN port (see instructions above “Programming SNR-VG-3002 via the LAN port”) and enable the “Enable WAN Web Access” option.

|   |  |
|---|--|
| Пароль  | <input type="text"/> (пароль для доступа к веб-интерфейсу)   |
| ICMP для WAN-порта  | <input type="radio"/> Выкл <input checked="" type="radio"/> Вкл (устройство не будет отвечать на PING, если установлено значение Выкл) |
| Доступ к веб через WAN  | <input type="radio"/> Выкл <input checked="" type="radio"/> Вкл  |
| <input type="button" value="Сохранить"/> <input type="button" value="Перезагрузить"/> |  |

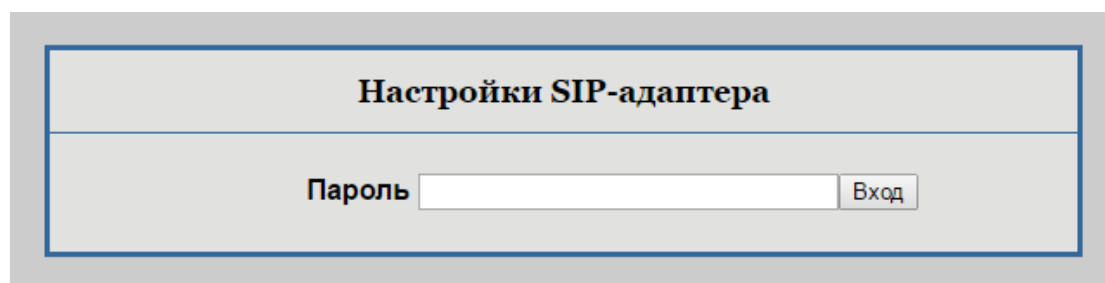
After enabling WAN access, be sure that the WAN port of the SNR-VG-3002 is connected to an uplink (i.e. router, hub, switch, etc). Then, get the WAN IP address of the SNR-VG-3002 by following section 5.1 of this user manual and selecting menu option 02. Then, access the SNR-VG-3002 via your web browser by entering the WAN IP address:

### **http://SNR-VG-3002's IP Address**

Be sure that your PC is connected to the router/hub/switch directly or via the LAN port (which also serves as a pass-through connection for internet/network access for your PC) of the SNR-VG-3002.

## **6.1.4. User Programming and Configuration**

From your web browser, the SNR-VG-3002 will show the following login screen:



The screenshot shows a web browser window with a light gray background. At the top, there is a title bar with the text "Настройки SIP-адаптера" in bold black font. Below the title bar is a horizontal line. Underneath the line is a text input field with the label "Пароль" (Password) to its left. To the right of the input field is a button with the text "Вход" (Login).

Enter the password and click on the “Login” button

## **6.1.5. Passwords**

Passwords are case sensitive and all Unicorn devices come with factory default passwords as indicated below:

Advanced User Password for access to Super User Options: admin

End User Password for access to Basic User Options: 1234

Pages available for the end-user are:

- Device Status
- Basic Options

Additional pages available to super user are:

- Super Options.
- FXS 1 port
- FXS 2 port



## 6.1.6. Configuration Options and Explanations

After a correct password is entered in the login screen, the embedded web server inside the SNR-VG-3002 will show the configuration page, which is explained in details below:

### 6.1.6.1 Device Status:

| Device Status  |  |                     |            |                |                     |                        |
|--|--|---------------------|------------|----------------|---------------------|------------------------|
| Setting options  | Definitions  |                     |            |                |                     |                        |
| <b>MAC Address</b>   | The device ID, in HEX format. This is a very important ID for ISP troubleshooting.         |                     |            |                |                     |                        |
| <b>WAN IP Address</b>  | This field shows IP address of device  |                     |            |                |                     |                        |
| <b>Product Model</b>   | This product model is Unicorn3002  |                     |            |                |                     |                        |
| <b>Software Version</b>  | Information of software  |                     |            |                |                     |                        |
| <b>System Up Time</b>  | Shows system up time since the last reboot.  |                     |            |                |                     |                        |
| <b>PPPoE Link Up</b>   | Indicates whether the PPPoE connection is up if the Unicorn3002 is connected to DSL modem. |                     |            |                |                     |                        |
| <b>NAT</b>   | I Indicate the NAT type behind which the device is when the stun feature is defined.       |                     |            |                |                     |                        |
| <b>Port Status</b>   |  |                     |            |                |                     |                        |
| Shows several information regarding the individual FXS ports.  |  |                     |            |                |                     |                        |
| <b>Port</b>  | <b>Hook</b>  | <b>Registration</b> | <b>DND</b> | <b>Forward</b> | <b>Busy Forward</b> | <b>Delayed Forward</b> |
| FXS1   | On Hook  | Registered          | Yes        | 613            |                     |                        |
| FXS2   | On Hook  | Registered          | No         |                | 614                 |                        |
| <ul style="list-style-type: none"> <li>• Both FXS port1 and FXS port2 are registered with this SIP Server.</li> <li>• FXS Port 1 user has set Do Not Disturb.</li> <li>• FXS Port 1 user has set his calls to be forwarded unconditionally to ext 613.</li> <li>• FXS Port 2 user has set his calls to forward to 614 when his phone is busy.</li> </ul> |  |                     |            |                |                     |                        |

### 6.1.6.2 Basic Options

| BASIC OPTIONS SETTING |                |
|-----------------------|----------------|
| Setting options       | Definitions    |
| <b>Web Port</b>       | Default is 80. |

|  |   |
|--|---|
| <b>IP Address</b>  | <p>There are 3 modes under which the SNR-VG-3002 can operate:</p> <ul style="list-style-type: none"> <li>- 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-3002 will acquire its IP address from the first DHCP server it discovers from the office/home network it is connected to.</li> <li>-To use the PPPoE feature, the PPPoE account settings need to be set. The SNR-VG-3002 will attempt to establish a PPPoE session if any of the PPPoE fields have been entered with data.</li> <li>- 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.</li> </ul> |
| <b>Time Zone</b>   | This parameter controls how the displayed date/time will be adjusted according to the specified time zone.  |
| <b>Allow DHCP Option 2 to override Time Zone setting</b> | If set yes and under DHCP mode,the device will try to get option 2 from DHCP configure and overwrite Time Zone.   |
| <b>Daylight Savings Time</b>                             | 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.  |
| <b>Device Mode</b>                                       | 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   |
| <b>LAN Subnet Mask</b>                                   | Sets the LAN subnet mask. Default value is 255.255.255.0  |
| <b>LAN DHCP Base IP:</b>                                 | Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.22.1   |
| <b>DHCP IP Lease Time:</b>                               | Value is set in units of hours. Default value is 120hr (5 Days) The time IP address are assigned to the LAN clients   |
| <b>Port Map</b>  | Forwards a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port  |
| <b>End User Password</b>                                 | This contains the password to access the Web Configuration Menu. This field is case sensitive.  |
| <b>Reply to ICMP on WAN port</b>                         | If set to "Yes", the unicorn3002 will respond to the PING command from other computers, but it also is vulnerable to the DOS attack. Default is <b>No</b> .   |
| <b>Wan Side Http Access</b>                              | If this parameter is set to "No", the HTML configuration update via WAN port is disabled.   |

### 6.1.6.3 Super Options

| <b>Super Options</b>                      |   |
|---|---|
| <b>Setting options</b>                    | <b>Definitions</b>  |
| <b>Admin Password</b>                     | 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.   |
| <b>Layer3 Qos</b>                         | 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.   |
| <b>Layer2 Qos</b>                         | Value used for layer 2 VLAN tag. Default setting is blank   |
| <b>Data VLAN Tag</b>                      | 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".  |
| <b>Stun sever is:</b>                     | IP address or Domain name of the STUN server.   |
| <b>Keep-alive interval</b>                | This parameter specifies how often the Unicorn3002 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.  |
| <b>Firmware Upgrade and Provisioning:</b> | Upgrade or provisioning through <b>TFTP</b> or <b>TFTP</b> server.<br>--Upgrade Via:select HTTP or TFTP mode.<br>--Allow DHCP Option: support 66,128,150. If select yes,device will get server information from DHCP option and ignore the Config Server Path.<br>Option 66--TFTP server name(if you select Upgrade Via->TFTP), HTTP server name(if you select Upgrade Via->HTTP)<br>Option 128--TFPT Server IP address.(if you select Upgrade Via->TFTP), HTTP Server IP address(if you select Upgrade Via->HTTP)<br>Option 150--TFTP server address.(if you select Upgrade Via->TFTP), HTTP server address(if you select Upgrade Via->HTTP) |
| <b>Authenticate Conf File</b>             | configure file would be authenticated before acceptance if set to Yes   |
| <b>SNMP Service Enable</b>                | If set <b>Yes</b> , The SNMP function will be work.   |

|  |   |
|--|---|
| <b>SNMP Get Community</b>                          | Set the community of read operation   |
| <b>SNMP Set Community</b>                          | Set the community of write operation  |
| <b>SNMP Manager IP 1</b>                           | Set the ip address of snmp trap   |
| <b>SNMP Manager IP 2</b>                           | Set another ip address of snmp trap   |
| <b>SNMP Manager IP 3</b>                           | Set another ip address of snmp trap   |
| <b>SNMP Manager IP 4</b>                           | Set another ip address of snmp trap   |
| <b>NTP server</b>                                  | This parameter defines the URI or IP address of the NTP server which is used by the SNR-VG-3002 to set the current date/time.   |
| <b>Allow DHCP Option 42 to override NTP server</b> | If set <b>Yes</b> , device can get NTP server from DHCP option 42.  |
| <b>Lock Keypad Update</b>                          | If set to "Yes", the configuration update via keypad is disabled.   |
| <b>Disable Voice Prompt</b>                        | Default is No   |
| <b>Syslog Sever</b>                                | The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)  |
| <b>Syslog level</b>                                | Default is blank, the feature is useful for the Internet Telephone Service Provider.  |
| <b>Download Configuration</b>                      | <b>Device</b><br>User can download configuration from the web page and save to configuration file.  |
| <b>Call Progress Tones</b>                         | 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.<br>Example configuration for N.A. Dial tone:<br>f1=350@-13,f2=440@-13,c=0/0;<br>Syntax:                    f1=freq@vol,                    f2=freq@vol,<br>c=on1/off1-on2/off2-on3/off3; [...] (Note: freq: 0 - 4000Hz; vol: -30 - 0dBm) |
| <b>Import TLS Certificate File(Optional)</b>       | User can import the TLS certification   |
| <b>Restore Configuration</b>                       | User can restore the before configuration from the configuration file saved at local pc   |

## 6.1.6.4 FXS PORT 1:

Two FXS SIP accounts each has its own configuration page. Their configurations are identical.

| <b>FXS PORT 1 SETTING</b>      |  |
|--------------------------------|--|
| <b>Settings Options</b>        | <b>Definitions</b>   |
| <b>Account active</b>          | When set to Yes the FXS port is activated  |
| <b>SIP Server</b>              | SIP Server's URI or IP address   |
| <b>Outbound Proxy</b>          | SIP Outbound Proxy Server's URI or IP address  |
| <b>SIP transport(Optional)</b> | Select transport between UDP and TLS, default is UDP.  |
| <b>NAT Traversal</b>           | <p>This parameter defines whether the SNR-VG-3002 NAT traversal mechanism will be activated or not.</p> <p>If Choosing <b>No</b>, nothing to do.</p> <p>If Choosing <b>No, but send keep-alive</b>, the SNR-VG-3002 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.</p> <p>If choosing <b>STUN</b> and a STUN server is also specified, then the SNR-VG-3002 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the SNR-VG-3002 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-3002 will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If choosing <b>STUN</b> with no specified STUN server, the SNR-VG-3002 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.</p> <p>If choosing <b>UPNP</b>, the embedded UPNP client inside the SNR-VG-3002 will attempt to mapping ports with the router by upnp protocol.</p> |
| <b>SIP User ID</b>             | SIP service subscriber's User ID   |
| <b>Authenticate ID</b>         | SIP service subscriber's Authenticate ID. Can be identical to or different from SIP User ID  |
| <b>Authenticate Password</b>   | SIP service subscriber's account password  |
| <b>Name</b>                    | SIP service subscriber's name which will be used for Caller ID display   |
| <b>Use DNS SRV</b>             | Default is No. If set to Yes the client will use DNS SRV for server lookup   |
| <b>User ID is Phone</b>        | If the SNR-VG-3002 has an assigned PSTN telephone  |

|   |   |
|---|---|
| <b>Number</b>                             | 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  |
| <b>SIP Registration</b>                   | This parameter controls whether the SNR-VG-3002 needs to send REGISTER messages to the proxy server. The default setting is “Yes”.  |
| <b>Unregister on Reboot</b>               | Default is “No.” If set to “Yes”, then the SIP user will be unregistered on reboot.   |
| <b>Register Expiration</b>                | This parameter allows the user to specify the time frequency (in minutes) the SNR-VG-3002 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).   |
| <b>Outgoing call without Registration</b> | 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.  |
| <b>Local SIP port</b>                     | This parameter defines the local SIP port the SNR-VG-3002 will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS 2 port is 5062.   |
| <b>Local RTP port</b>                     | This parameter defines the local RTP-RTCP port pair the SNR-VG-3002 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. |
| <b>Use Random Port</b>                    | This parameter, when set to <b>Yes</b> , will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple SNR-VG-3002 are behind the same NAT.   |
| <b>Refer-To Use Target Contact</b>        | Default is <b>NO</b> . If set to <b>YES</b> , then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.   |
| <b>DTMF Payload Type</b>                  | This parameter sets the payload type for DTMF using RFC2833   |
| <b>DTMF in Audio</b>                      | 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),<br>Default is <b>YES</b> .  |
| <b>DTMF via RFC2833</b>                   | This parameter specifies the mechanism to transmit DTMF digit via RTP (RFC2833).<br>Default <b>YES</b> .  |

|                             |  |
|-----------------------------|--|
| <b>DTMF via SIP INFO</b>    | This parameter specifies the mechanism to transmit DTMF digit via SIP INFO.<br>Default is <b>NO</b> .  |
| <b>Send Flash Event</b>     | This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.  |
| <b>Enable Call Features</b> | Default is <b>No</b> . If set to <b>Yes</b> , Call Forwarding & Do-Not-Disturb are supported locally   |
| <b>Offhook Auto-Dial</b>    | This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The SNR-VG-3002 will automatically append the "@" and the host portion of the corresponding SIP address.  |
| <b>Proxy-Require</b>        | SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.   |
| <b>USE NAT IP</b>           | NAT IP address used in SIP/SDP message. Default is blank.  |
| <b>Disable call-waiting</b> | Default is No.   |
| <b>No Key Entry Timeout</b> | Default is No.   |
| <b>Preferred Vocoder</b>    | The Unicorn3002 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".   |
| <b>Voice Frames per TX</b>  | 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.<br><br>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 |

|                                |   |
|--------------------------------|---|
|                                | <p>“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-3002 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.</p> |
| <b>G723 Rate</b>               | This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.  |
| <b>SRTP Mode(Optional)</b>     | Enable SRTP mode based on selection. Default is “No”.   |
| <b>VAD</b>                     | Default is <b>No</b> . VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.  |
| <b>Symmetric RTP</b>           | Default is <b>No</b> . 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.  |
| <b>FAX Mode</b>                | Default is <b>T.30</b> (Fax Pass-Through), or T.38 (Auto Detect) FoIP   |
| <b>Fax Tone Detection Mode</b> | Default is <b>Callee</b> . This decides whether Caller or Callee sends out the re INVITE for T.38 or Fax Pass Through.  |
| <b>Jitter Buffer Type</b>      | Select either Fixed or Adaptive based on network conditions.  |
| <b>Jitter Buffer Length</b>    | Select Low, Medium or High based on network conditions.   |
| <b>Distinct Ring Tone</b>      | Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The device will ONLY use selected ring tones for particular Caller IDs. For all other calls, the device will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls..  |
| <b>Use Bell-style 3-way</b>    | Conference mode, default option is No. If set to yes, the feature   |



|                                   |   |
|-----------------------------------|---|
| <b>Conference</b>                 | code for coferece *23 would be disabled.  |
| <b>Disable Call -Waiting</b>      | Default is <b>NO</b> . User can use star codes to enable/disable call waiting.  |
| <b>Disable Call –Waiting Tone</b> | Default is <b>NO</b> . This is to disable the stutter call waiting tone when a call waiting call arrived.   |
| <b>Ring Timeout</b>               | An incoming call will stop ringing when not picked up given a specific period of time.  |
| <b>No Key Entry Timeout</b>       | Default is 4 seconds.   |
| <b>Use # as Send Key</b>          | T This parameter allows users to configure the “#” key to be used as the <b>Send</b> (or <b>Dial</b> ) key. If set to <b>Yes</b> , pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the <b>Re(Dial)</b> key. If set to <b>No</b> , this “#” key will then be included as part of the dial string to be sent out.   |
| <b>Dial Plan</b>                  | <p>Dial Plan Rules:</p> <ol style="list-style-type: none"> <li>1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , * , #, A,a,B,b,C,c,D,d</li> <li>2. Grammar: x - any digit from 0-9; <ol style="list-style-type: none"> <li>a. xx+ - at least 2 digits number;</li> <li>b. xx. ?at least 2 digits number;</li> <li>c. ^ - exclude;</li> <li>d. [3-5] - any digit of 3, 4, or 5;</li> <li>e. [147] - any digit 1, 4, or 7;</li> <li>f. &lt;2=011&gt; - replace digit 2 with 011 when dialing</li> </ol> <p>Example 1: {[369]11   1617xxxxxxx} Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617</p> <p>Example 2: {^1900x+   &lt;=1617&gt;xxxxxxx} Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers</p> <p>Example 3: {1xxx[2-9]xxxxxx   &lt;2=011&gt;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.</p> </li> <li>3. Default: Outgoing - {x+}</li> </ol> <p>Example of a simple dial plan used in a Home/Office in the US:</p> <pre>{ ^1900x.   &lt;=1617&gt;[2-9]xxxxxx   1[2-9]xx[2-9]xxxxxx   011[2-9]x.   [3469]11 }</pre> <p>Explanation of example rule (reading from left to right):</p> <p>^1900x. - prevents dialing any number started with 1900</p> <p>&lt;=1617&gt;[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically</p> |

|  |   |
|--|---|
|  | <p>1[2-9]xx[2-9]xxxxx  - allows dialing to any US/Canada Number with 11 digits length</p> <p>011[2-9]x. - allows international calls starting with 011</p> <p>[3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911</p> <p>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*]+ }.</p> |
| <b>Subscribe for MWI</b>                     | Default is No. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.  |
| <b>Send Anonymous</b>                        | 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.   |
| <b>Anonymous Call Rejection</b>              | Default is <b>No</b> . If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.   |
| <b>Check SIP User ID for incoming INVITE</b> | Check the SIP User ID in Request URI. If they don’t match, the call will be rejected.   |
| <b>Session Expiration</b>                    | <p>The session timer extension enables SIP sessions to be periodically “refreshed” via a re-INVITE request.</p> <p>Once the session interval expires, if there is no refresh via a re-INVITE message, the session will be terminated.</p> <p>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.</p>                              |
| <b>Min-SE</b>                                | The minimum session expiration (in seconds). Default is <b>90</b> seconds.  |
| <b>Caller Request Timer</b>                  | <p>If selecting “<b>Yes</b>” the device will use session timer when it makes outbound calls if remote party supports session timer.</p> <p>Default is <b>NO</b>.</p>  |
| <b>Callee Request Timer</b>                  | <p>If selecting “<b>Yes</b>” the phone will use session timer when it receives inbound calls with session timer request.</p> <p>Default is <b>NO</b>.</p>   |
| <b>Force Timer</b>                           | <p>If selecting “<b>Yes</b>” the device will use session timer even if the remote party does not support this feature. Selecting “<b>No</b>” will allow the device to enable session timer only when the remote party support this feature.</p> <p>To turn off Session Timer, select “<b>No</b>” for Caller Request Timer, Callee Request Timer, and Force Timer.</p> <p>Default is <b>NO</b>.</p>  |
| <b>UAC Specify Refresher</b>                 | As a Caller, select UAC to use the device as the refresher,   |

|                              |   |
|------------------------------|---|
|                              | <p>or UAS to use the Callee or proxy server as the refresher.<br/>Default is <b>Omit</b></p>  |
| <b>UAS Specify Refresher</b> | <p>As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the device as the refresher.<br/>Default is <b>UAC</b>.</p>  |
| <b>Force INVITE</b>          | <p>Session Timer can be refreshed using INVITE method or UPDATE method.<br/>Select “<b>Yes</b>” to use INVITE method to refresh the session timer.<br/>Default is <b>NO</b>.</p>  |
| <b>Special Feature</b>       | <p>Choose the selection to meet some special requirements from Soft Switch vendors.<br/>Default is <b>standard</b>.</p>   |
| <b>FAS Impedance</b>         | <p>Select the impedance of analog telephone connected to phone port</p>   |
| <b>Caller ID Scheme</b>      | <p>select caller ID to suit standard of different area.</p>   |
| <b>Onhook Voltage</b>        | <p>Select onhook voltage to suit standard of different area or PBX.</p>   |
| <b>Polarity Reversal</b>     | <p>Select Polarity Reversal to adapt some call charge/billing system. Default is <b>NO</b>.</p>   |
| <b>Hook Flash Timing</b>     | <p>Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value.</p>   |
| <b>Volume Amplification</b>  | <p>Voice path volume adjustment.<br/>Rx is a gain level for signals transmitted by FXS<br/>Tx is a gain level for signals received by FXS.<br/>Default = 0dB for both parameters. Loudest volume: +6dB<br/>Lowest volume: -6dB.<br/>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.<br/>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.<br/>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.</p> |
| <b>Ring Tones</b>            | <p>This function lets you configure ring tone cadence preferences. User has 10 choices.<br/>The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.</p>  |

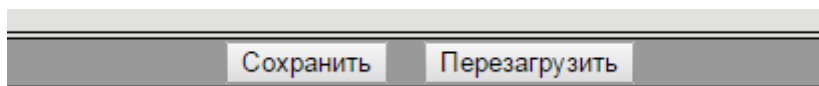
### 6.1.6.5 FXS PORT 2:

The FXS PORT 2 configure almost same as FXS PORT 1 page.

| FXS PORT 2 SETTING    |  |
|-----------------------|--|
| Settings Options      | Definitions  |
| <b>Account active</b> | When set to Yes the FXS port is activated.<br>When set to Hunting Group, the FXS Port 2 share the same number with the FXS Port 1. |

### 6.1.7. Saving the Configuration Changes

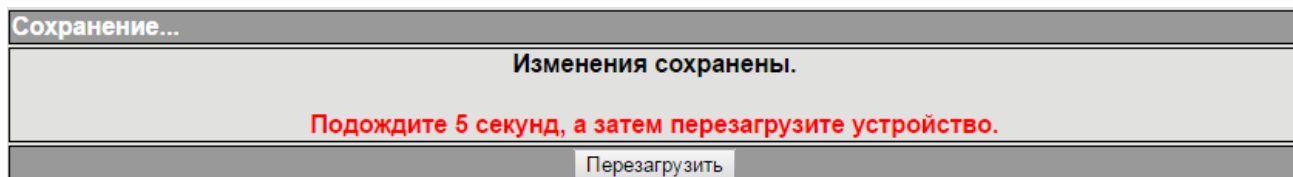
Once a change is made, users should click on the “SaveSet” button in the Configuration page, as follow:



The SNR-VG-3002 will then display the following screen to confirm that the changes have been saved. Please allow 5 to 10 seconds before rebooting the device.

### 6.1.8. Rebooting the SNR-VG-3002

You can reboot the SNR-VG-3002 by clicking on the “Reboot” button after each update to the configuration page. Alternatively, you can reboot by unplugging the power supply of the SNR-VG-3002 and then powering it on again. If your SNR-VG-3002 ever becomes “stuck” or un-responsive, you can unplug the power supply to reboot it. Frequent rebooting by unplugging the power supply is not recommended and should not be necessary.



## 6.2. Configuring SNR-VG-3002 via Voice Prompt

### 6.2.1. DHCP Mode

Follow section 5.1 with voice menu option 01 to enable SNR-VG-3002 to use DHCP

### **6.2.2. STATIC IP Mode**

Follow section 5.1 with voice menu option 01 to enable SNR-VG-3002 to use STATIC IP mode, then use option 02, 03, 04 to set up SNR-VG-3002's IP, Subnet Mask, Gateway respectively.

### **6.3. Configuration through a Central Server**

SNR-VG-3002 devices can be automatically configured from a central provisioning system.

When SNR-VG-3002 boots 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 "cfg001fc1xxxxx", where "001fc1xxxxx" is the MAC address of the SNR-VG-3002.

For more information regarding configuration file format, please refer to the related technical documentation.

The configuration file can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of SNR-VG-3002s can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. The central provisioning system uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual SNR-VG-3002 for firmware upgrade, etc.

## **7. SOFTWARE UPGRADE**

To upgrade software, SNR-VG-3002 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-3002.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the SNR-VG-3002'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-3002. 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-3002.

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, ISP 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 "cfg001fc1xxxxx", where "001fc1xxxxx" is the MAC address of the SNR-VG-3002. 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.

## **8. RESTORE TO FACTORY DEFAULT SETTINGS**

### **Warning:**

Restoring to the factory default settings will delete all configuration information of the device.

Steps to follow in restoring to factory default settings:

- a) Press "\*" for voice prompt.
- b) Enter "99" and then you will hear the voice prompt "Reset".
- c) Enter the number "862584658050". A "click" sound will be heard.
- d) Wait for 15 seconds.

The device is now restored to the factory default setting.

## **9. TECHNICAL SUPPORT CONTACT**

Email: [support@nag.ru](mailto:support@nag.ru)