# SmartNode™ 100 Series
# Analog Telephone Adapter

## User Manual


This is a Class B device and can be used in residential locations.

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Patton Electronics warrants all SmartNode components to be free from defects, and will—at our option—repair or replace the product should it fail within one year from the first date of the shipment.

This warranty is limited to defects in workmanship or materials, and does not cover customer damage, abuse or unauthorized modification. If the product fails to perform as warranted, your sole recourse shall be repair or replacement as described above. Under no condition shall Patton Electronics be liable for any damages incurred by the use of this product. These damages include, but are not limited to, the following: lost profits, lost savings and incidental or consequential damages arising from the use of or inability to use this product. Patton Electronics specifically disclaims all other warranties, expressed or implied, and the installation or use of this product shall be deemed an acceptance of these terms by the user.

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# About This Guide

This guide describes the SmartNode 100 Series Analog Telephone Adapter (ATA) hardware, installation, and basic configuration.

## Audience

This guide is intended for the following users:

- Operators
- Installers
- Maintenance technicians

## Structure

This guide contains the following chapters and appendices:

- Chapter 1 on page 15 provides information about SN100 features and capabilities.
- Chapter 2 on page 22 provides information about installing the SN100.
- Chapter 3 on page 29 describes how to quickly configure the SN100 for basic operation.
- Chapter 4 on page 46 describes using the SmartNode ATA home window to monitor ATA status or to use the advanced settings to configure the ATA.
- Chapter 5 on page 53 describes how to modify network settings.
- Chapter 6 on page 61 describes how to modify VoIP settings.
- Chapter 7 on page 112 describes how to modify security settings.
- Chapter 8 on page 119 describes using maintenance tools.
- Chapter 9 on page 133 describes how to contact Patton for assistance
- Appendix A on page 136 provides compliance information for the SN100
- Appendix B on page 139 provides specifications for the SN100
- Appendix C on page 143 provides example application scenarios
- Appendix D on page 146 is a frequently asked questions.

For best results, read the contents of this guide *before* you install the SmartNode 100.

# Precautions

Notes and cautions, which have the following meanings, are used throughout this guide to help you become aware of potential ATA problems. ***Warnings*** relate to personal injury issues, and ***Cautions*** refer to potential property damage.

**Note** A note presents addition.al information or interesting sidelights.

IMPORTANT

The alert symbol and IMPORTANT heading calls attention to important information.

CAUTION

The alert symbol and CAUTION heading indicate a potential hazard. Strictly follow the instructions to avoid property damage.

CAUTION

The shock hazard symbol and CAUTION heading indicate a potential electric shock hazard. Strictly follow the instructions to avoid property damage caused by electric shock.

WARNING

**The alert symbol and WARNING heading indicate a potential safety hazard. Strictly follow the warning instructions to avoid personal injury.**

WARNING

**The shock hazard symbol and WARNING heading indicate a potential electric shock hazard. Strictly follow the warning instructions to avoid injury caused by electric shock.**

### *Safety when working with electricity*

WARNING

**The SmartNode device contains no user serviceable parts, and is not be opened by the user. The equipment shall be returned to Patton Electronics for repairs or repaired by qualified service personnel.**

WARNING

**Mains Voltage: In systems without a power switch, line voltages are present in the power supply when the power cord is connected. The mains outlet used to power the SmartNode device shall be within 10 feet (3 meters) of the device, be easily accessible, and protected by a circuit breaker.**

WARNING

**For AC powered units, ensure that the power cable used meets all applicable standards for the country in which it is to be installed, and that it is connected to a wall outlet which has earth ground.**

WARNING

**For units with an external power adapter, the adapter shall be a listed Limited Power Source.**

WARNING

**Hazardous network voltages are present in WAN ports regardless of whether power to the SmartNode is ON or OFF. To avoid electric shock, use caution when near WAN ports. When detaching the cables, detach the end away from the SmartNode first.**

WARNING

**Before handling the device, disconnect the telephone network cables to avoid contact with telephone line voltages. When detaching the cables, detach the end away from the SmartNode device first.**

WARNING

**Do not work on the system or connect or disconnect cables during periods of lightning activity.**

### *Deutsch*

**Warnhinweise:**

WARNUNG

**Dieses Gerät ist NICHT für den Anschluss an das Telefonnetz (PSTN) bestimmt und auch NICHT dafür zugelassen. Es ist nur für den Anschluss an Endgeräte beim Kunden vorgesehen.**

**WARNUNG**

- Das Gerät entält keine austauschbaren Komponenten und ist vom Benutzer nicht zu öffnen. Bei Systemen ohne Netzschalter und ohne externes Netzteil liegt Netzspannung im Gerät an, wenn das Netzkabel angeschlossen ist.
- Bei Geräten mit externem Netzteil muss das Netzteil die Anforderungen an eine zugelassene Stromquelle mit begrenzter Leistung erfüllen. Die Steckdose, die für die Stromversorgung des Gerätes verwendet wird, sollte höchstens 3 Meter vom Gerät entfernt und leicht zugänglich sein sowie durch einen den örtlichen regulatorischen Anforderungen entsprechenden Schutzschalter abgesichert sein.
- Für mit Wechselstrom betriebene Geräte muss sichergestellt sein, dass das verwendete Netzkabel alle gültigen Normen des Landes erfüllt, in dem es eingesetzt werden soll.
- Für mit Wechselstrom betriebene Geräte, die 3-polige Netzstecker haben (L1, L2 u. GND oder Phase, Neutralleiter u. Schutzleiter), muss die Steckdose geerdet sein.
- Für mit Gleichstrom betriebene Geräte muss sichergestellt sein, dass die Verbindungskabel für Spannung, Strom, erwartete Temperatur, Entflammbarkeit und mechanische Wartbarkeit geeignet sind.
- WAN-, LAN- u. PSTN-Ports (Anschlüsse) können unter gefährlicher Spannung stehen, unabhängig davon, ob das Gerät ein- oder ausgeschaltet ist. PSTN bezieht sich auf Schnittstellen wie Telefon, FXS, FXO, DSL, xDSL, T1, E1, ISDN, Voice, usw. Diese sind als „gefährliche Netzwerkspannungen“ bekannt. Um einen elektrischen Schlag zu vermeiden, muss in der Nähe dieser Anschlüsse mit Vorsicht gearbeitet werden. Werden Kabel von diesen Anschlüssen getrennt, zuerst das Kabel am anderen Ende herausziehen.
- Während eines Gewitters darf nicht am Gerät gearbeitet werden und es dürfen keine Kabel angeschlossen oder vom Netz getrennt werden.

In Übereinstimmung mit den Anforderungen der Richtlinie 2002/96/EG über Elektro- und Elektronik-Altgeräte (WEEE) muss sichergestellt sein, dass Altgeräte von anderem Abfall und Schrott getrennt werden und dem Sammel- und Verwertungssystem für Elektro- und Elektronik-Altgeräte in Ihrem Land zum Recycling zugeführt werden.

### *General observations*

**CAUTION**

Do not stack multiple SmartNode devices directly on top of one another, and do not place items on top of the device. If you will be installing equipment above the SmartNode device, leave at least 2 inches (5 cm) of clearance between the devices.

Furthermore, leave at least 2 inches (5 cm) to the left, right, front, and rear of the SmartNode device for proper ventilation.

# Chapter 1 General information

## *Chapter contents*

## Overview

Congratulations on becoming the owner of the SmartNode ATA, a device that connects a normal PSTN telephone to the Internet so that you can make telephone calls. The ATA provides an analog interface for PSTN, PBX, fax machines, telephones, and other devices that require an analog port.

This user manual describes how to install the ATA, and how to customize its configuration to get the most out of your new device.

This device has been thoroughly inspected and tested and is warranted for one year for parts and labor. If questions or problems arise during installation or use of this product, contact Patton Electronics Technical Support at +(301) 975-1007.

## Features

The SmartNode 100 ATA series are feature-rich analog telephone adapters that enable you to connect standard telephones and make VoIP calls using your broadband connection. There is no need to replace your existing analog telephones just because you want to enhance your phone system.

Easy to install and simple to use, the SN100 series connects standard telephones or fax machines to any Voice over IP (VoIP) service provider.

Well equipped with at least one FXS RJ-11 phone interface, you can speak or fax as with the old fix net telephone. Due to the small design you can easily place the adapter anywhere.

Features include:

- Codecs:
  - G.711-ulaw, G711-alaw, G729, G723
  - G.726: 16k/24k/32k/40k bit per second (ADPCM)
- Call Features:
  - Call Waiting, Call Transfer,
  - Call Forward as Busy forward; Non-Answer forward; unconditional forward,
  - Do-not-disturb (DND) support
  - 3-way conferencing
- Fax Modes:
  - T.38 relay support
  - G.711 pass-through

## Front Panel LED Indicators & Rear Panels

### *Front panel LEDs*

Front panel indicator lights are described for the following SmartNode SN100 Series ATA models:

- SN101/1JS/E (see figure 1). Table 1 lists the function of each front panel indicator.
- SN102/2JS/E (see figure 2 on page 17). Table 2 on page 18 lists the function of each front panel indicator.

- SN102/1JS1JO/E (see figure 3 on page 18). Table 3 on page 18 lists the function of each front panel indicator.

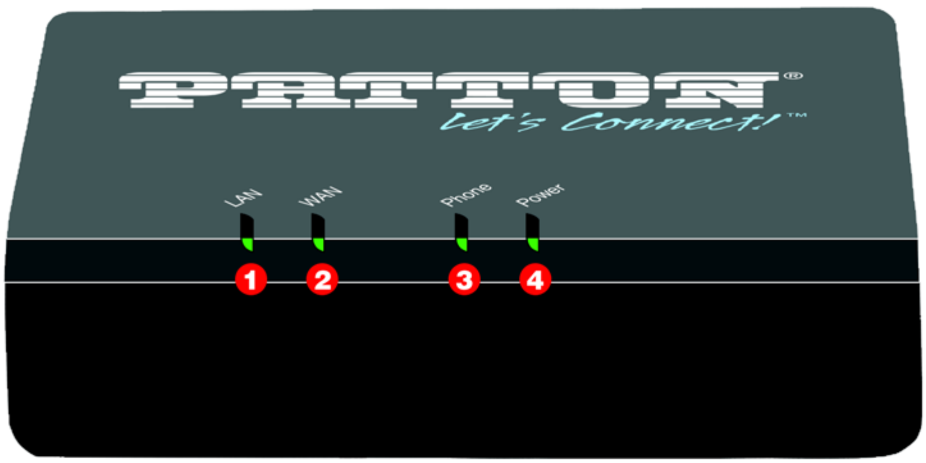



Figure 1. SN101/1JS/E ATA front panel

Table 1. SN101/1JS/E ATA front panel LED functions

| Callout | LED label | Color | Status | | |
|---|---|---|---|---|---|
| | | | On | Flashing | Off |
| 1 | LAN | Green | LAN is connected successfully | Data is transmitting | Ethernet not connected to PC |
| 2 | WAN | Green | ATA network connection established | Data traffic on cable network | Waiting for network connection |
| 3 | Phone | Green | Telephone or fax machine is on-hook | Ring Indication | Telephone or fax machine is on-hook |
| 4 | Power | Green | ATA is powered on | N/A | ATA is powered off |

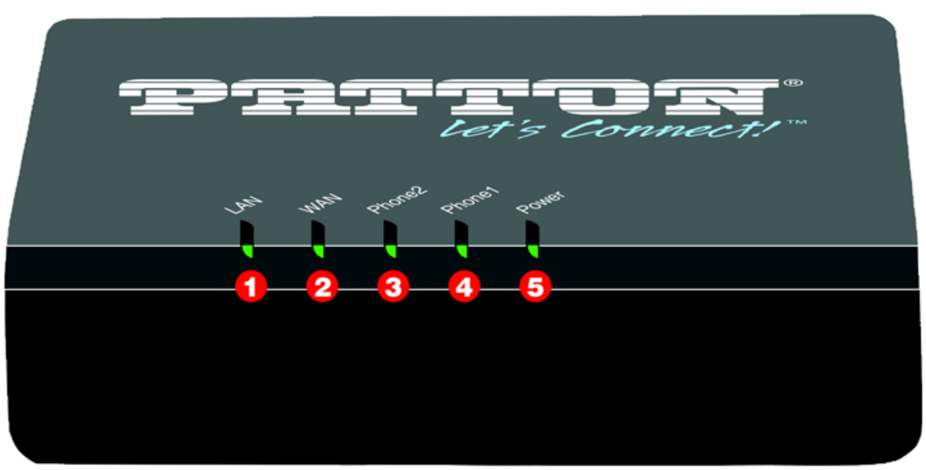



Figure 2. SN102/2JS/E ATA front panel

Table 2. SN102/2JS/E ATA front panel LED functions

| Callout | LED label | Color | Status | | |
|---|---|---|---|---|---|
| | | | On | Flashing | Off |
| 1 | LAN | Green | LAN is connected successfully | Data is transmitting | Ethernet not connected to PC |
| 2 | WAN | Green | ATA network connection established | Data traffic on cable network | Waiting for network connection |
| 3 | Phone2 | Green | Telephone or fax machine is on-hook | Ring Indication | Telephone or fax machine is on-hook |
| 4 | Phone1 | Green | Telephone or fax machine is on-hook | Ring Indication | Telephone or fax machine is on-hook |
| 5 | Power | Green | ATA is powered on | N/A | ATA is powered off |

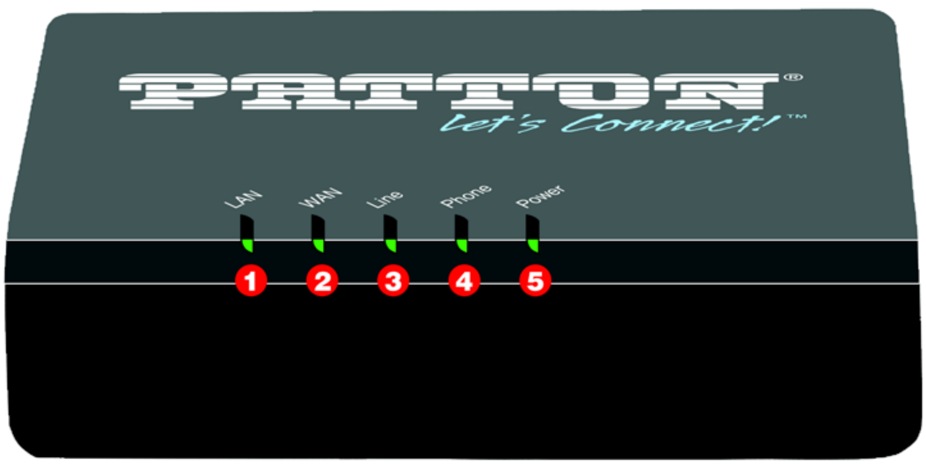



Figure 3. SN102/1JS1JO/E ATA front panel

Table 3. SN102/1JS1JO/E ATA front panel LED functions

| Callout | LED label | Color | Status | | |
|---|---|---|---|---|---|
| | | | On | Flashing | Off |
| 1 | LAN | Green | LAN is connected successfully | Data is transmitting | Ethernet not connected to PC |
| 2 | WAN | Green | ATA network connection established | Data traffic on cable network | Waiting for network connection |
| 3 | Line | Green | PBX or CO line is off-hook | Ring Indication | PBX or CO line is off-hook |
| 4 | Phone | Green | Telephone or fax machine is on-hook | Ring Indication | Telephone or fax machine is on-hook |
| 5 | Power | Green | ATA is powered on | N/A | ATA is powered off |

### Rear panel connectors

Rear panel connectors and switches are described for the following SmartNode SN100 Series ATA models:

- SN101/1JS/E (see figure 4 on page 19). Table 4 on page 19 lists the function of each connector and switch on the rear panel.
- SN102/2JS/E (see figure 5 on page 20). Table 5 on page 20 lists the function of each connector and switch on the rear panel.
- SN102/1JS1JO/E (see figure 6 on page 20). Table 6 on page 20 lists the function of each connector and switch on the rear panel.

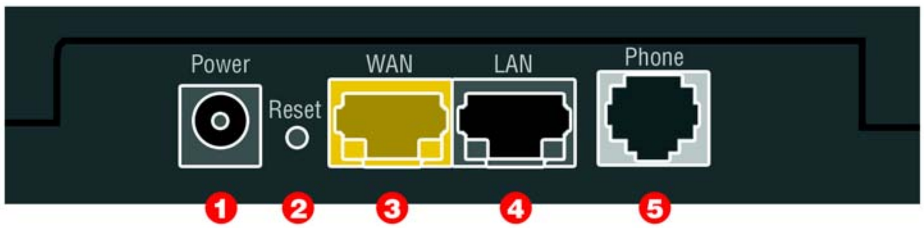



Figure 4. SN101/1JS/E ATA rear panel

Table 4. Function/description of SN101/1JS/E ATA rear panel connectors/button

| Callout | Label | Color | Description |
|---|---|---|---|
| 1 | Power | N/A | Connects to the power adapter |
| 2 | Reset | N/A | Using a paper clip or similar object, press this button briefly to restart the unit. Press and hold for 10 seconds to restore the factory default settings. |
| 3 | WAN | Gold | Connected to PC by an Ethernet cable. This port enables your PC or switch/hub to be connected to the ATA through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |
| 4 | LAN | Black | Connected to the network by an Ethernet cable. This port enables your ATA to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |
| 5 | Phone | Gray | FXS port can be connected to analog telephone or fax machine by an RJ-11 (Gray) analog line. |

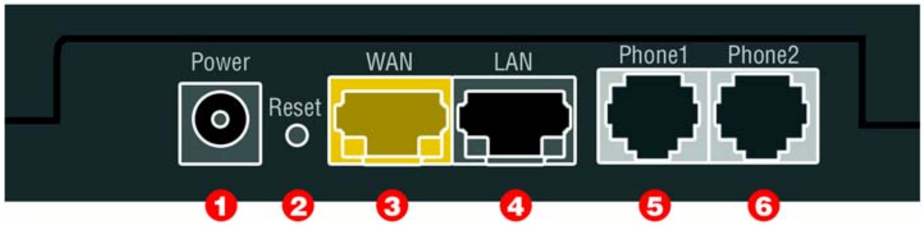



Figure 5. SN102/2JS/E ATA rear panel

Table 5. Function/description of SN102/2JS/E ATA rear panel connectors/button

| Callout | Label | Color | Description |
|---|---|---|---|
| 1 | Power | N/A | Connects to the power adapter |
| 2 | Reset | N/A | Using a paper clip or similar object, press this button briefly to restart the unit. Press and hold for 10 seconds to restore the factory default settings. |
| 3 | WAN | Gold | Connects to a PC by an Ethernet cable. This port enables your PC or switch/hub to be connected to the ATA through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |
| 4 | LAN | Black | Connects to the network by an Ethernet cable. This port enables your ATA to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |
| 5 | Phone1 | Gray | FXS port can be connected to analog telephone or fax machine by an RJ-11 (Gray) analog line. |
| 6 | Phone2 | Gray | FXS port can be connected to analog telephone or fax machine by an RJ-11 (Gray) analog line. |

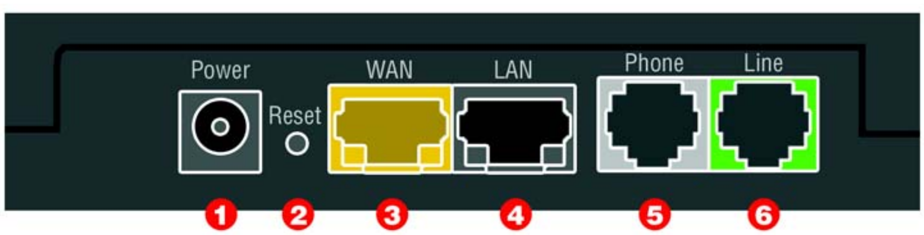



Figure 6. SN102/1JS1JO/E ATA rear panel

Table 6. Function/description of SN102/1JS1JO/E ATA rear panel connectors/button

| Callout | Label | Color | Description |
|---|---|---|---|
| 1 | Power | N/A | Connects to the power adapter |
| 2 | Reset | N/A | Using a paper clip or similar object, press this button briefly to restart the unit. Press and hold for 10 seconds to restore the factory default settings. |
| 3 | WAN | Gold | Connects to a PC by an Ethernet cable. This port enables your PC or switch/hub to be connected to the ATA through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |

Table 6. Function/description of SN102/1JS1JO/E ATA rear panel connectors/button (Continued)

| Callout | Label | Color | Description |
|---|---|---|---|
| 4 | LAN | Black | Connects to the network by an Ethernet cable. This port enables your ATA to be connected to an Internet Access device, e.g. router, cable modem, ADSL modem, through a networking cable with RJ-45 connectors used on 10Base-T and 100Base-TX networks. |
| 5 | Phone | Gray | FXS port can be connected to analog telephone or fax machine by an RJ-11 (Gray) analog line. |
| 6 | Line | Green | Connected to PBX or CO line by an RJ-11 (Gray) analog line. The PSTN (FXO) port can be connected to the extension port of a PBX or directly connected to a carrier's PSTN line. |

# Chapter 2 Installation

## Chapter contents

## Introduction

This chapter describes installing the following SmartNode 100 Series devices:

- "Installing the SN101/1JS/E (1 FXS port) ATA" on page 23
- "Installing the SN102/2JS/E (2 FXS ports) ATA" on page 25
- "Installing the SN102/1JS1JO/E (1 FXS port and 1 FXO ports) ATA" on page 26

Refer to the appropriate section to install the ATA as shown in figure 7.

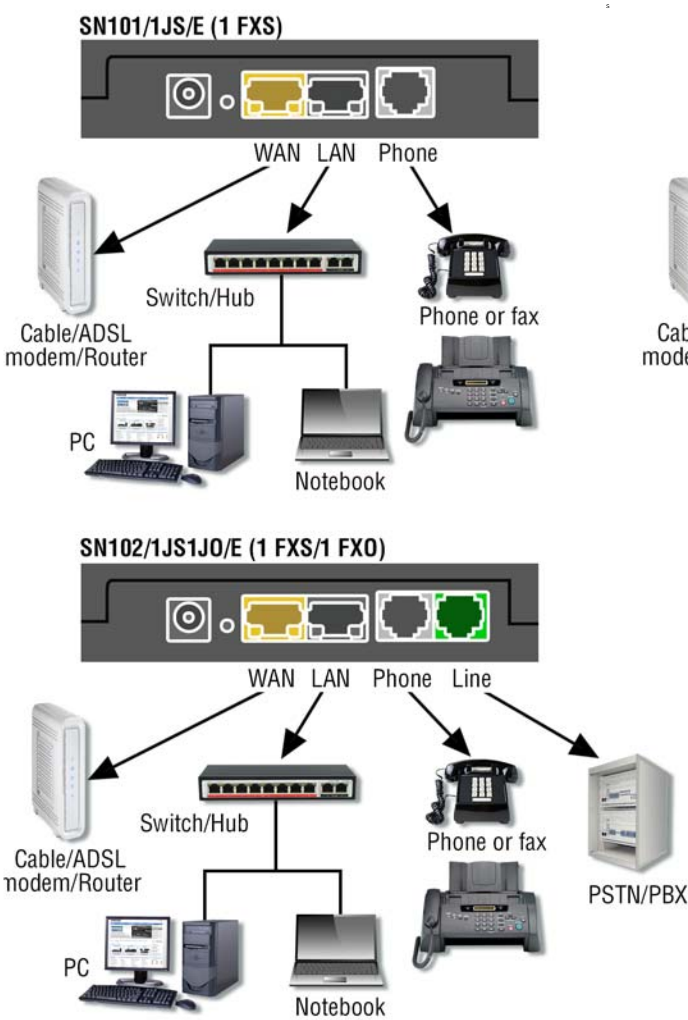



Figure 7. Typical SN100 Series applications

## Installing the SN101/1JS/E (1 FXS port) ATA

Installation consists of the following:

- Verifying the package contents (see section "Package contents" on page 24)
- Connecting the ATA *LAN* port directly to a computer or to a computer via a switch/hub (see section "" on page 24)

- Connecting the ATA *WAN* port to a cable/DSL modem or router (see section "Connecting the ATA to a cable/DSL modem or router" on page 25)
- Connecting the ATA *Phone* port to an analog telephone or fax machine (see section "Connecting the ATA to a telephone or fax machine" on page 25)
- Connecting power to the ATA (see section "Connecting power to the ATA" on page 25)

### Package contents

- SmartNode 100 Series ATA
- Ethernet cable (Cat. 5, UTP, unshielded twisted pair, Color: yellow, RJ-45 connectors, Length 5 feet/1.5 m) (see figure 8)
- Phone line cable (Color: gray, RJ-11 connectors, Length 5 feet/1.5 m)
- Power adapter (12VDC/1A)


Figure 8. SN100 ATA cables and 12VDC/1A power adapter

### Connecting the ATA to a computer

1. Connect one end of the Ethernet cable included in the SN100 package to the *LAN* port on the ATA. If you are connecting the ATA directly to a computer, go to step 2. Otherwise, go to step 4 to connect multiple computers to the ATA via a hub, switch, or router.
2. Disconnect power from your computer.
3. Connect the other end of the Ethernet cable to your computer's network interface card (NIC). Go to section "Connecting the ATA to a cable/DSL modem or router".
4. Disconnect power from your hub, switch, or router.
5. Connect the other end of the Ethernet cable to your hub, switch, or router's WAN/Internet port.

6. Connect your computer(s) to the available hub, switch, or router LAN ports using separate Ethernet cables.

### Connecting the ATA to a cable/DSL modem or router

1. Connect one end of an Ethernet cable (with RJ-45 connectors) to the *WAN* port on the ATA.
2. Disconnect power from your cable/DSL modem or router.
3. Connect the other end of the Ethernet cable to a DSL/cable modem or router.

### Connecting the ATA to a telephone or fax machine

Connect your analog telephone or fax machine to the *Phone* port on the ATA.

### Connecting power to the ATA

1. Connect the power adapter's barrel jack to the *Power* port on the ATA.
2. Connect the other end of the power adapter to the AC power outlet.
3. Verify that the ATA's *Power* LED is lit.
4. Connect power to the computer or hub, switch, or router.
5. Connect power to the cable/DSL modem or router.

The ATA is installed. Proceed to Chapter 3, "Quick Start" on page 29.

## Installing the SN102/2JS/E (2 FXS ports) ATA

Installation consists of the following:

- Verifying the package contents (see section "Package contents" on page 25)
- Connecting the ATA *LAN* port directly to a computer or to a computer via a switch/hub (see section "Connecting the ATA to a computer" on page 26)
- Connecting the ATA *WAN* port to a cable/DSL modem or router (see section "Connecting the ATA to a cable/DSL modem or router" on page 26)
- Connecting the ATA *Phone1* and *Phone2* ports to two analog telephones, two fax machines or one phone and one fax machine (see section "Connecting the ATA to telephones or fax machines" on page 26)
- Connecting power to the ATA (see section "Connecting power to the ATA" on page 26)

### Package contents

- SmartNode 100 Series ATA
- Ethernet cable (Cat. 5, UTP, unshielded twisted pair, Color: yellow, RJ-45 connectors, Length 5 feet/1.5 m) (see figure 8 on page 24)
- Phone line cable (Color: gray, RJ-11 connectors, Length 5 feet/1.5 m)
- Power adapter (12VDC/1A)

### Connecting the ATA to a computer

1. Connect the one end of the yellow Ethernet cable included in the SN100 package to the *LAN* port on the ATA. If you are connecting the ATA directly to a computer, go to step 2. Otherwise, go to step 4 to connect multiple computers to the ATA via a hub, switch, or router.
2. Disconnect power from your computer.
3. Connect the other end of the Ethernet cable to your computer's network interface card (NIC).
4. Disconnect power from your hub, switch, or router.
5. Connect the other end of the Ethernet cable to your hub, switch, or router's WAN/Internet port.
6. Connect your computer(s) to the available hub, switch, or router LAN ports using separate Ethernet cables.

### Connecting the ATA to a cable/DSL modem or router

1. Connect one end of an Ethernet cable (with RJ-45 connectors) to the *WAN* port on the ATA.
2. Disconnect power from your cable/DSL modem or router.
3. Connect the other end of the Ethernet cable to a DSL/cable modem or router.

### Connecting the ATA to telephones or fax machines

The two ATA FXS phone ports enable you to connect two analog telephones, two fax machines, or one telephone and one fax machine to the ATA.

1. Connect an analog telephone or fax machine to the *Phone1* port on the ATA.
2. Connect an analog telephone or fax machine to the *Phone2* port on the ATA.

### Connecting power to the ATA

1. Connect the power adapter's barrel jack to the *Power* port on the ATA.
2. Connect the other end of the power adapter to the AC power outlet.
3. Verify that the ATA's *Power* LED is lit.
4. Connect power to the computer or hub, switch, or router.
5. Connect power to the cable/DSL modem or router.

The ATA is installed. Proceed to Chapter 3, "Quick Start" on page 29.

## Installing the SN102/1JS1JO/E (1 FXS port and 1 FXO ports) ATA

Installation consists of the following:

- Verifying the package contents (see section "Package contents" on page 27)
- Connecting the ATA *LAN* port directly to a computer or to a computer via a switch/hub (see section "Connecting the ATA to a computer" on page 27)
- Connecting the ATA *WAN* port to a cable/DSL modem or router (see section "Connecting the ATA to a cable/DSL modem or router" on page 27)

- Connecting the ATA *Phone* FXS port to an analog telephone or fax machine (see section "Connecting the ATA to a telephone or fax machine" on page 27)
- Connecting the ATA *Line* FXO port to the PSTN or a PBX (see section "Connecting the ATA to the PSTN or a PBX" on page 27)
- Connecting power to the ATA (see section "Connecting power to the ATA" on page 27)

### Package contents

- SmartNode 100 Series ATA
- Ethernet cable (Cat. 5, UTP, unshielded twisted pair, Color: yellow, RJ-45 connectors, Length 5 feet/1.5 m) (see figure 8 on page 24)
- Phone line cable (Color: gray, RJ-11 connectors, Length 5 feet./1.5 m)
- Power adapter (12VDC/1A)

### Connecting the ATA to a computer

1. Connect the one end of yellow Ethernet cable included in the SN100 package to the *LAN* port on the ATA. If you are connecting the ATA directly to a computer, go to step 2. Otherwise, go to step 4 to connect multiple computers to the ATA via a hub, switch, or router.
2. Disconnect power from your computer.
3. Connect the other end of the Ethernet cable to your computer's network interface card (NIC).
4. Disconnect power from your hub, switch, or router.
5. Connect the other end of the Ethernet cable to your hub, switch, or router's WAN/Internet port.
6. Connect your computer(s) to the available hub, switch, or router LAN ports using separate Ethernet cables.

### Connecting the ATA to a cable/DSL modem or router

1. Connect one end of an Ethernet cable (with RJ-45 connectors) to the *WAN* port on the ATA.
2. Disconnect power from your cable/DSL modem or router.
3. Connect the other end of the Ethernet cable to a DSL/cable modem or router.

### Connecting the ATA to a telephone or fax machine

Using the included gray phone cable, connect your analog telephone or fax machine to the *Phone* port on the ATA.

### Connecting the ATA to the PSTN or a PBX

Connect the PSTN or a PBX to the *Line* port on the ATA.

### Connecting power to the ATA

1. Connect the power adapter's barrel jack to the *Power* port on the ATA.
2. Connect the other end of the power adapter to the AC power outlet.
3. Verify that the ATA's *Power* LED is lit.

4. Connect power to the computer or hub, switch, or router.
5. Connect power to the cable/DSL modem or router.

The ATA is installed. Proceed to Chapter 3, "Quick Start" on page 29.

# Chapter 3 Quick Start

## Chapter contents

## Introduction

This chapter describes using the Connection Setup and VoIP Setup wizards to quickly configure the ATA for operation.

## Configuring the SN100 to communicate with your computer

By default, the SN100 is configured for the following network environment:

- WAN type: **DHCP client**
- LAN IP: **192.168.1.1**
- LAN subnet mask: **255.255.255.0**

Do the following:

1. Configure the network interface of your computer to be the DHCP client. It will obtain an IP address from the SmartNode ATA automatically.
2. Using a Web browser (Microsoft Edge, Google Chrome, etc.) link to URL **http://192.168.1.1**. A login window displays (see figure 9).

**Note** Most examples of SN100 screens in this manual are from an SN102/2JS/E (ATA with 2 FXS ports). Where differences are relevant, screens from other SN100 models will be shown.

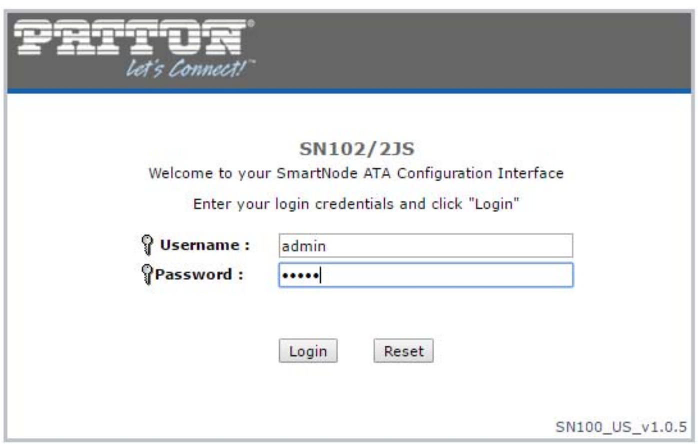



Figure 9. Login window

3. Type the default user name in the *Username* box: **admin**
4. Type the default password in the *Password* box: **admin**
5. Click the *Login* button. The *Please select Wizard or Advanced mode* window displays (see figure 10 on page 31).

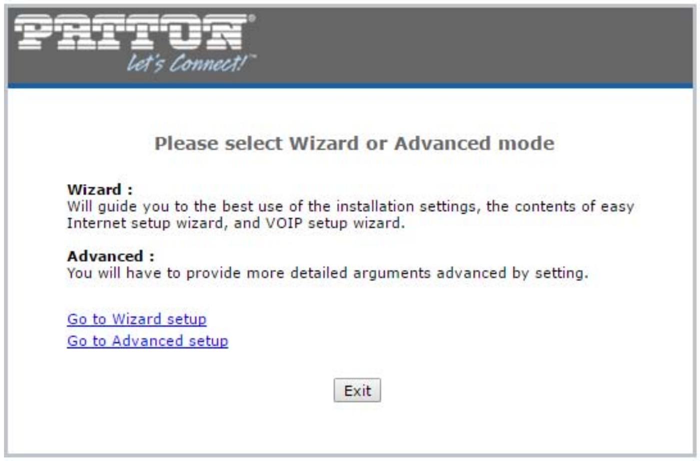



Figure 10. Please select Wizard or Advanced mode window

## Wizard Setup window

1. After finishing the authentication, the *Wizard Setup* window displays (see figure 11). Click *CONNECTION WIZARD* to begin. The *Welcome to the Connection Wizard* window displays (see figure 12 on page 32).

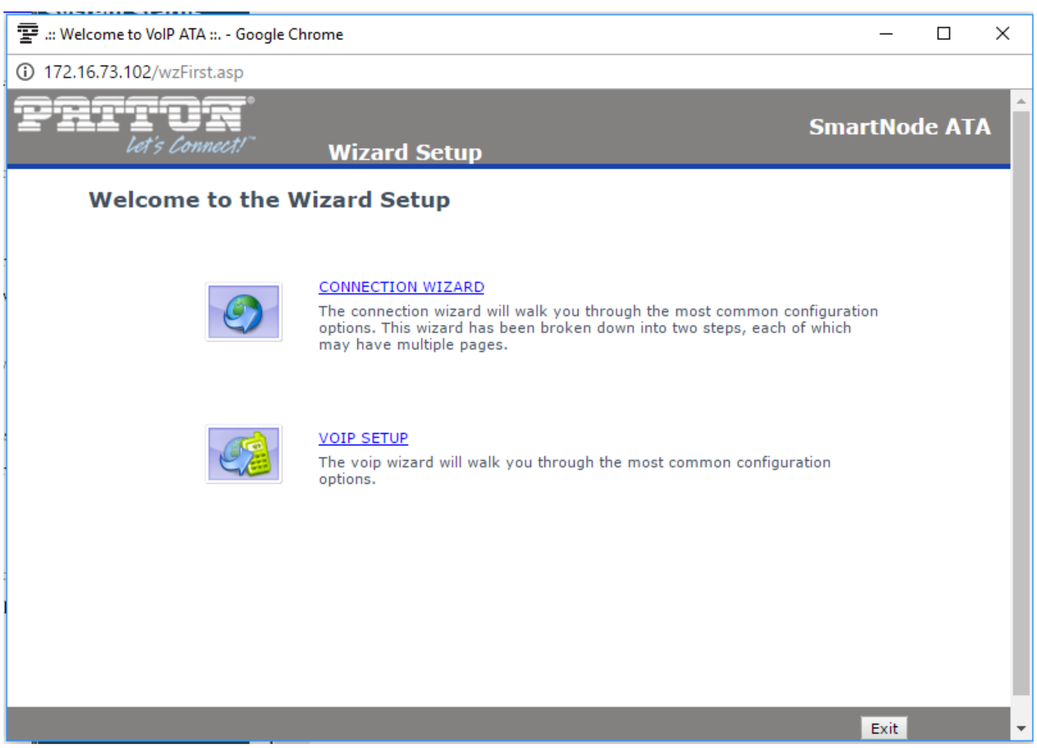



Figure 11. Wizard Setup window

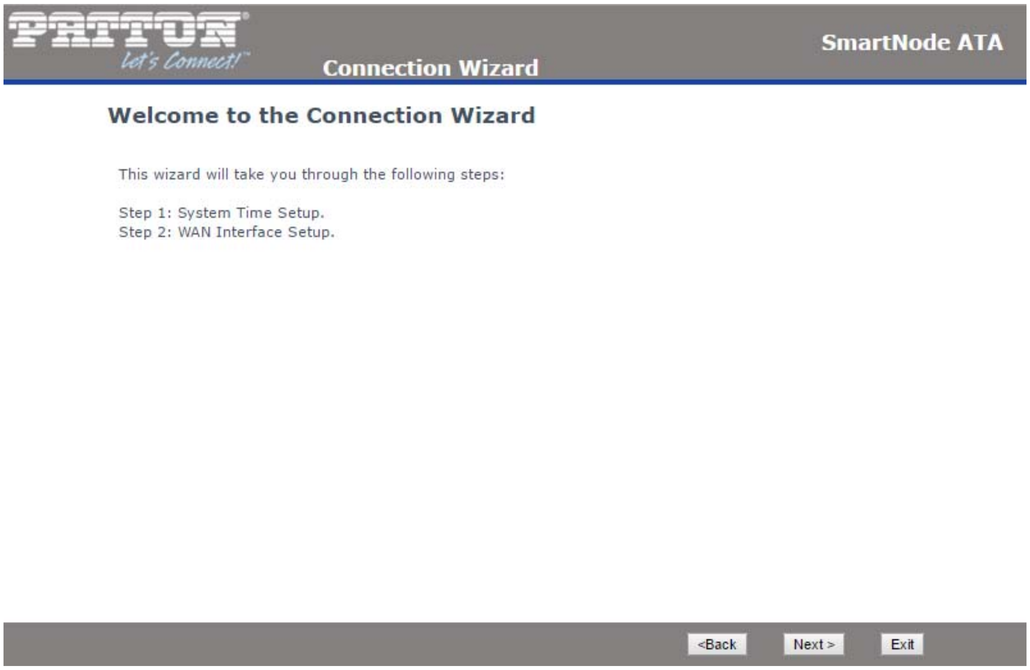



Figure 12. Welcome to the Connection Wizard window

**Note** To return to a previous window in the wizard, click the < *Back* button. To go to the next window in the wizard, click the *Next* > button. To cancel the setup process, click the *Exit* button.

IMPORTANT

Changes are only saved when you click the *Finish* button that is displayed at the end of each wizard, so if you click *Exit*, **all changes will be lost.**

2. Click the *Next>* button to begin system time setup. The *Time Zone Setting* window displays (see figure 13).

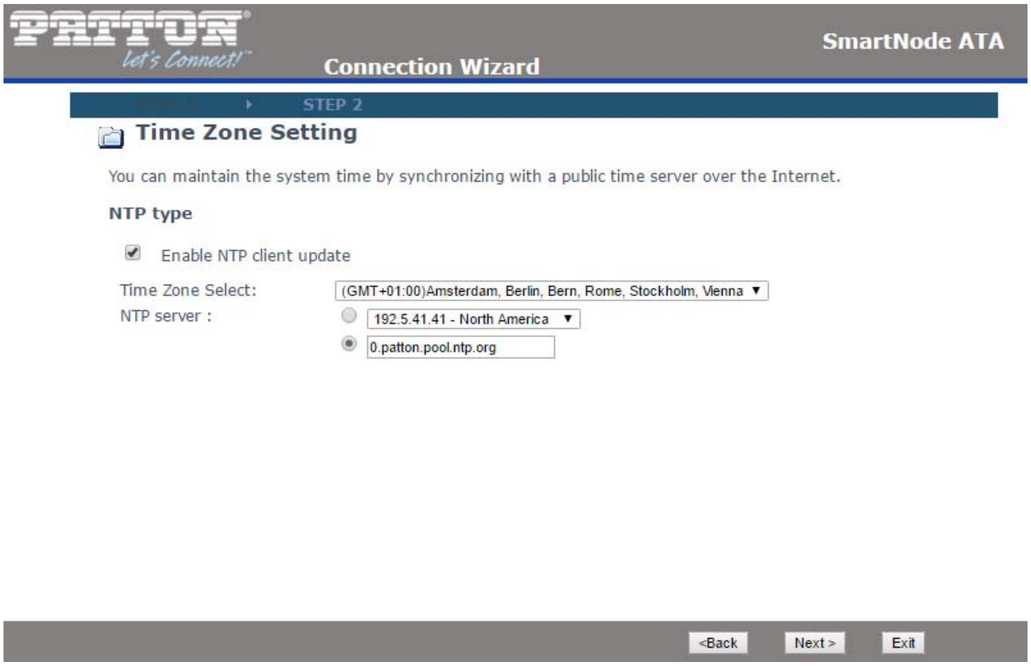



Figure 13. Time Zone Setting window

3. Click to enable *Enable NTP client update.*


4. Click the *Time Zone Select* drop-down menu, then select the appropriate time zone.


5. Click the *NTP server* drop-down menu, then select the appropriate NTP server.


6. Click the *Next>* button. The *WAN Interface Setting* window displays (see figure 14 on page 34).

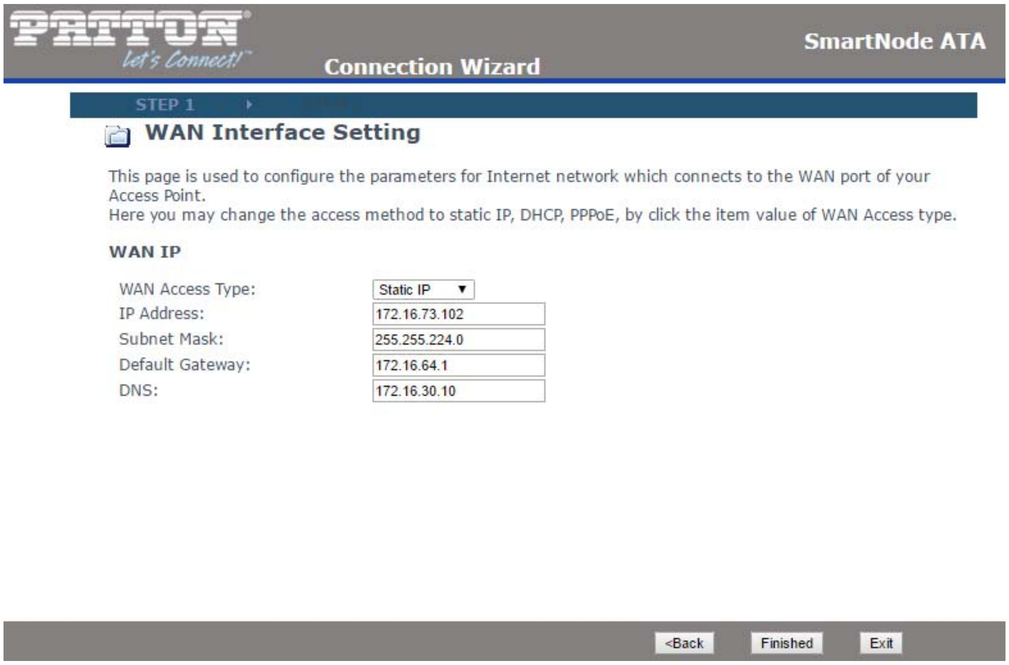



Figure 14. WAN Interface Setting window

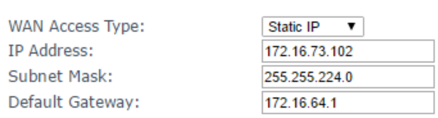
Go to section "WAN Port Type set up" to set up the WAN interface.

## WAN Port Type set up

For most users, Internet access is the primary application. The SN100 supports the WAN interface for Internet access and remote access.

The following sections explain details of WAN Port Internet access and broadband access setup:

1. Click the *WAN Access Type* drop-down menu, then select from the following options:




– **Static IP (Fixed IP User)**: If you are a leased line user with a fixed IP address, fill out the following items with the information provided by your ISP:
  - *IP Address*: check with your ISP provider
  - *Subnet Mask*: check with your ISP provider
  - *Default Gateway*: check with your ISP provider

– **Dynamic IP (DHCP Client)**: Obtain WAN IP address automatically

– **PPPoE (ADSL Dial-Up User)**: Some ISPs provide DSL-based service and use PPPoE to establish communication links with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to select this option.

- *User Name*: Enter User Name provided by your ISP
- *Password*: Enter Password provided by your ISP.

2. Click the *Finished>* button. The *Setting Internet interface* message window displays (see figure 15).


Figure 15. Setting Internet interface message window

When the Internet interface has been set up, the *Wizard Setup* window displays (see figure 11 on page 31).

## VoIP set up

This section describes configuring SN100 Series ATA VoIP settings. Chose the appropriate section from the following:

- "Configuring SN101/1JS/E (1 FXS) ATA VoIP settings" on page 36
- "Configuring SN102/2JS/E (2 FXS) ATA VoIP settings" on page 39
- "Configuring SN102/1JS1JO/E (1 FXS/1 FXO) ATA VoIP settings" on page 42

### Configuring SN101/1JS/E (1 FXS) ATA VoIP settings

1. Click *VOIP SETUP*. The *VoIP Configuration* window displays (see figure 16 on page 36). Do the following to configure Phone 1 SIP settings:

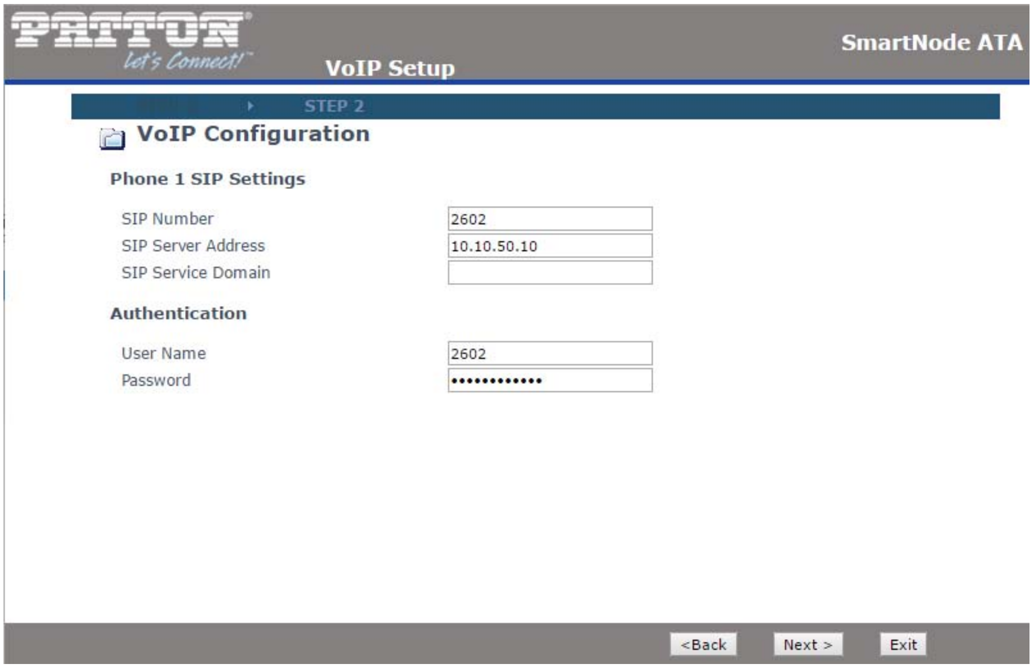



Figure 16. VoIP Configuration window

2. Type the SIP number that was provided by your ISP in the *SIP Number* box.


3. Type the proxy address that was provided by your ISP in the *SIP Server Address* box.


4. Type the SIP domain that was provided by your ISP in the *SIP Service Domain* box.


5. Type the user name that was provided by your ISP in the *Username* box.


6. Type the password that was provided by your ISP in the *Password* box.

7. Click the *Finish* button. The *Congratulations!* window displays (see figure 17). After a short delay, the *REG Status* table (see figure 18) should display *Reg Success* (registration successful).

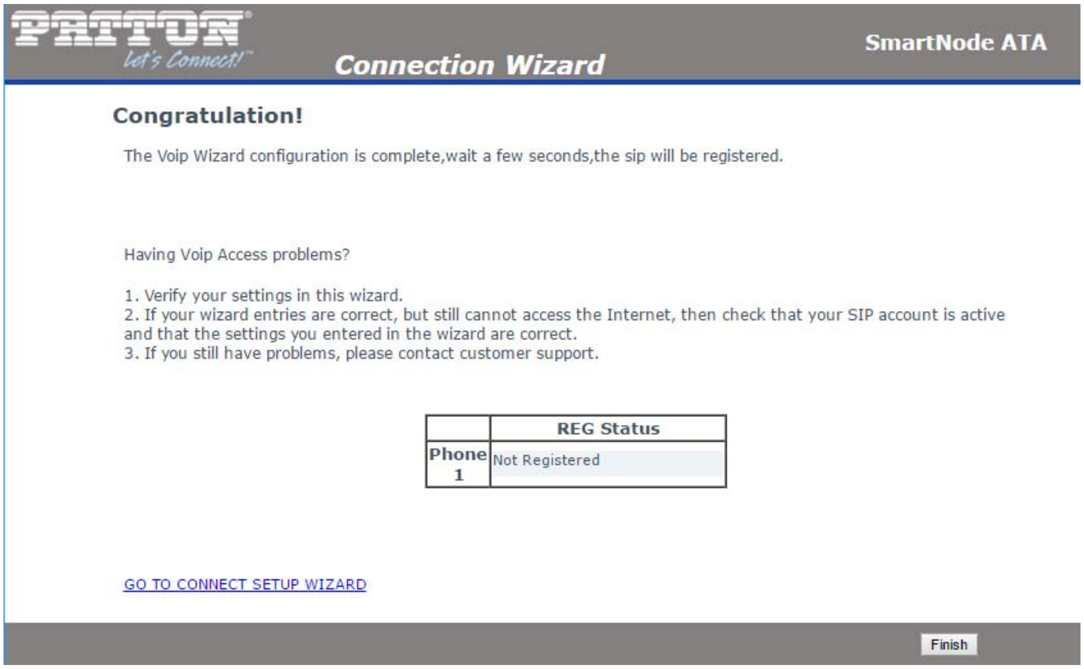



Figure 17. Congratulations! window

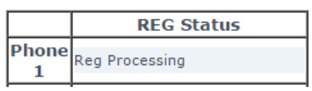



Figure 18. REG Status table

**Note** If the ATA is having a problem registering a phone or fax machine, do the following:

- Verify that the VoIP settings were entered correctly.
- If the VoIP entries are correct, but the ATA still cannot access the Internet, verify that your SIP account is active.
- If ATA is still having problems, please contact technical support (see Chapter 9, "Contacting Patton for assistance" on page 133 for details).

8. Click the *Finish* button. The SmartNode ATA home window (see figure 19 on page 38) displays.

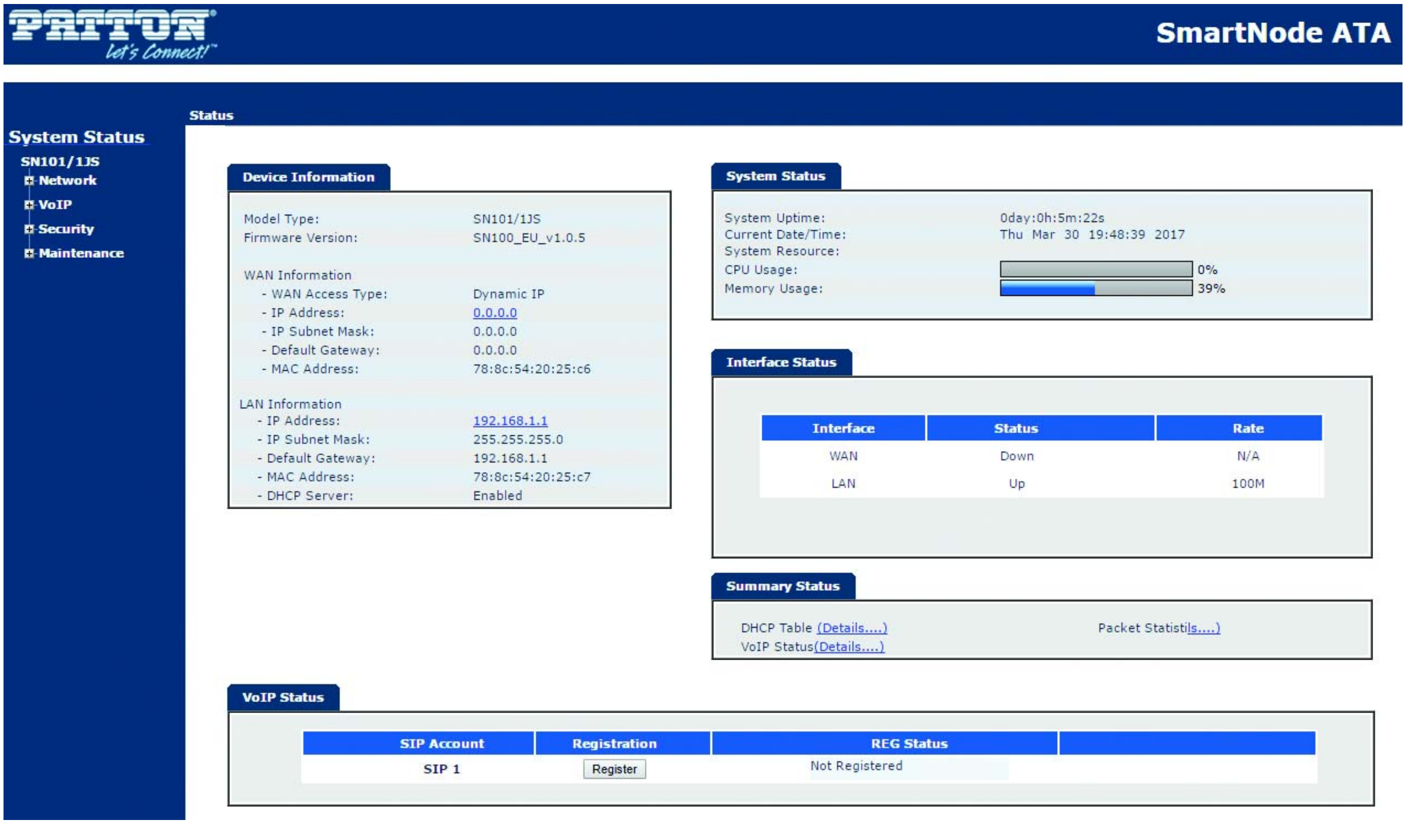



Figure 19. SN101/1JS/E ATA home window

Congratulations! The ATA is ready for use.

### Configuring SN102/2JS/E (2 FXS) ATA VoIP settings

1. Click *VOIP SETUP.* The *VoIP Configuration* window displays (see figure 20). Do the following to configure Phone 1 SIP settings:

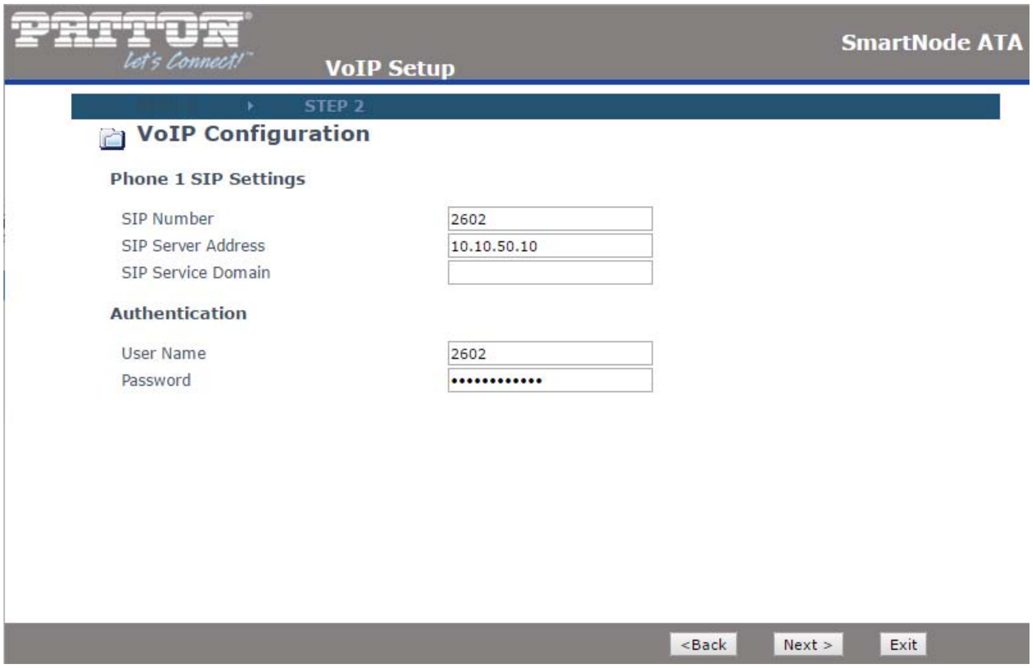



Figure 20. VoIP Configuration - Phone 1 window

2. Type the SIP number that was provided by your ISP in the *SIP Number* box.


3. Type the proxy address that was provided by your ISP in the *SIP Server Address* box.


4. Type the SIP domain that was provided by your ISP in the *SIP Service Domain* box.


5. Type the user name that was provided by your ISP in the *Username* box.


6. Type the password that was provided by your ISP in the *Password* box.

Click the *Next >* button. The *VoIP Configuration* window displays (see figure 21). Do the following to configure Phone 2 SIP settings:

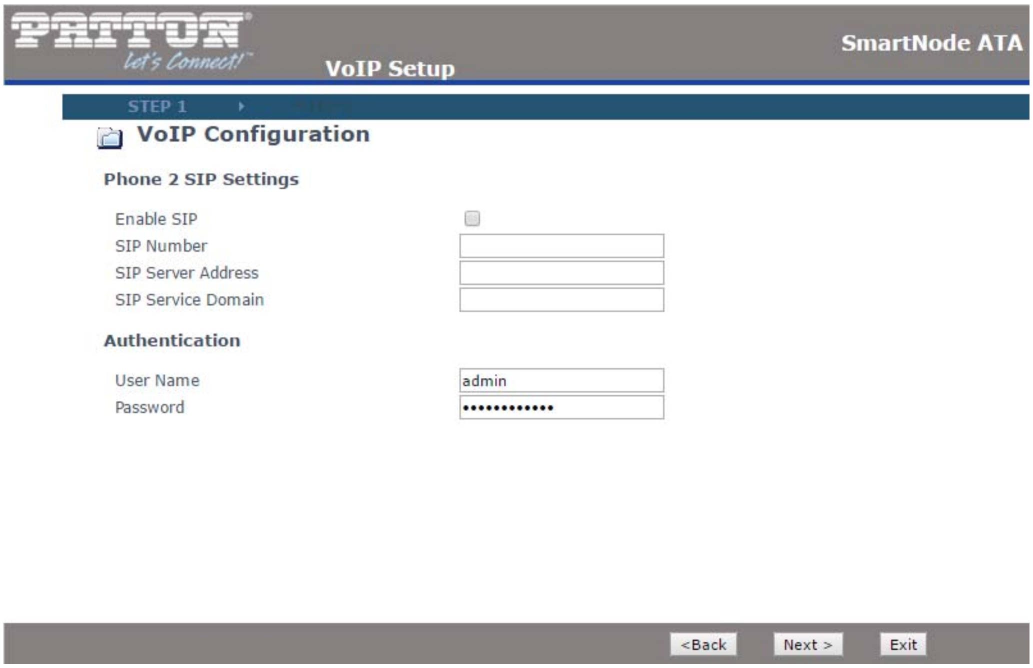



Figure 21. VoIP Configuration - Phone 2 window

7. Type the SIP number that was provided by your ISP in the *SIP Number* box.


8. Type the proxy address that was provided by your ISP in the *SIP Server Address* box.


9. Type the SIP domain that was provided by your ISP in the *SIP Service Domain* box.


10. Type the user name that was provided by your ISP in the *Username* box.


11. Type the password that was provided by your ISP in the *Password* box.

12. Click the *Finish* button. The *Congratulations!* window displays (see figure 22). After a short delay, the *REG Status* table (see figure 23) should display *Reg Success* (registration successful) for each phone you set up.

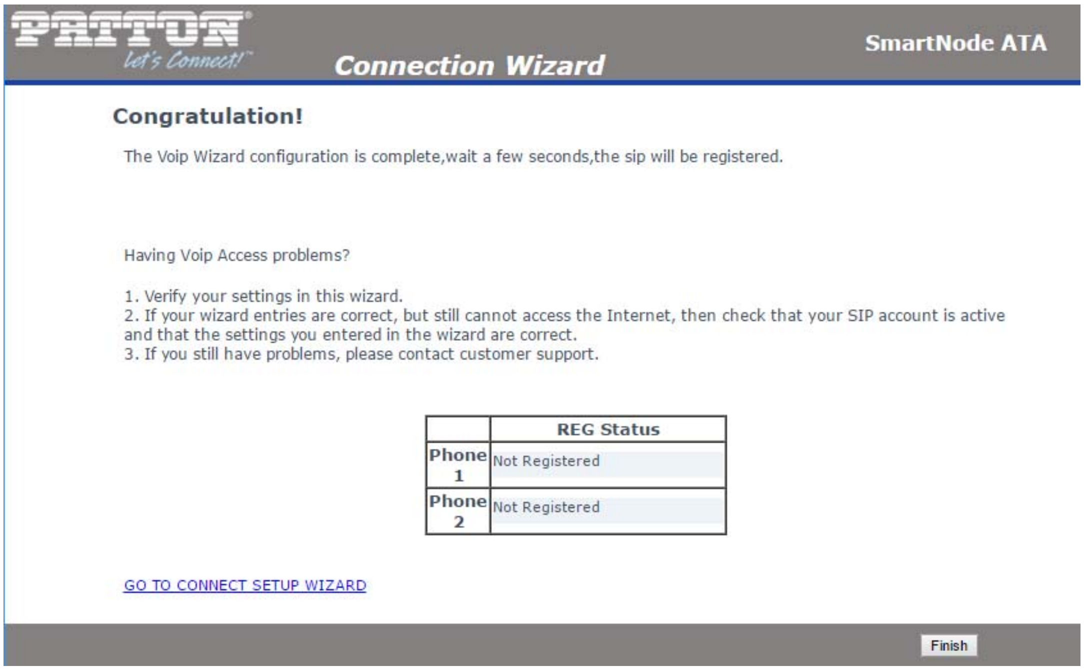



Figure 22. Congratulations! window

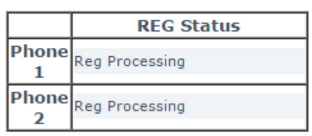



Figure 23. REG Status table

**Note** If the ATA is having a problem registering phones/fax machines, do the following:

- Verify that the VoIP settings were entered correctly.
- If the VoIP entries are correct, but the ATA still cannot access the Internet, verify that your SIP account is active.
- If ATA is still having problems, please contact technical support (see Chapter 9, "Contacting Patton for assistance" on page 133 for details).

13. Click the *Finish* button. The SmartNode ATA home window (see figure 24 on page 42) displays.

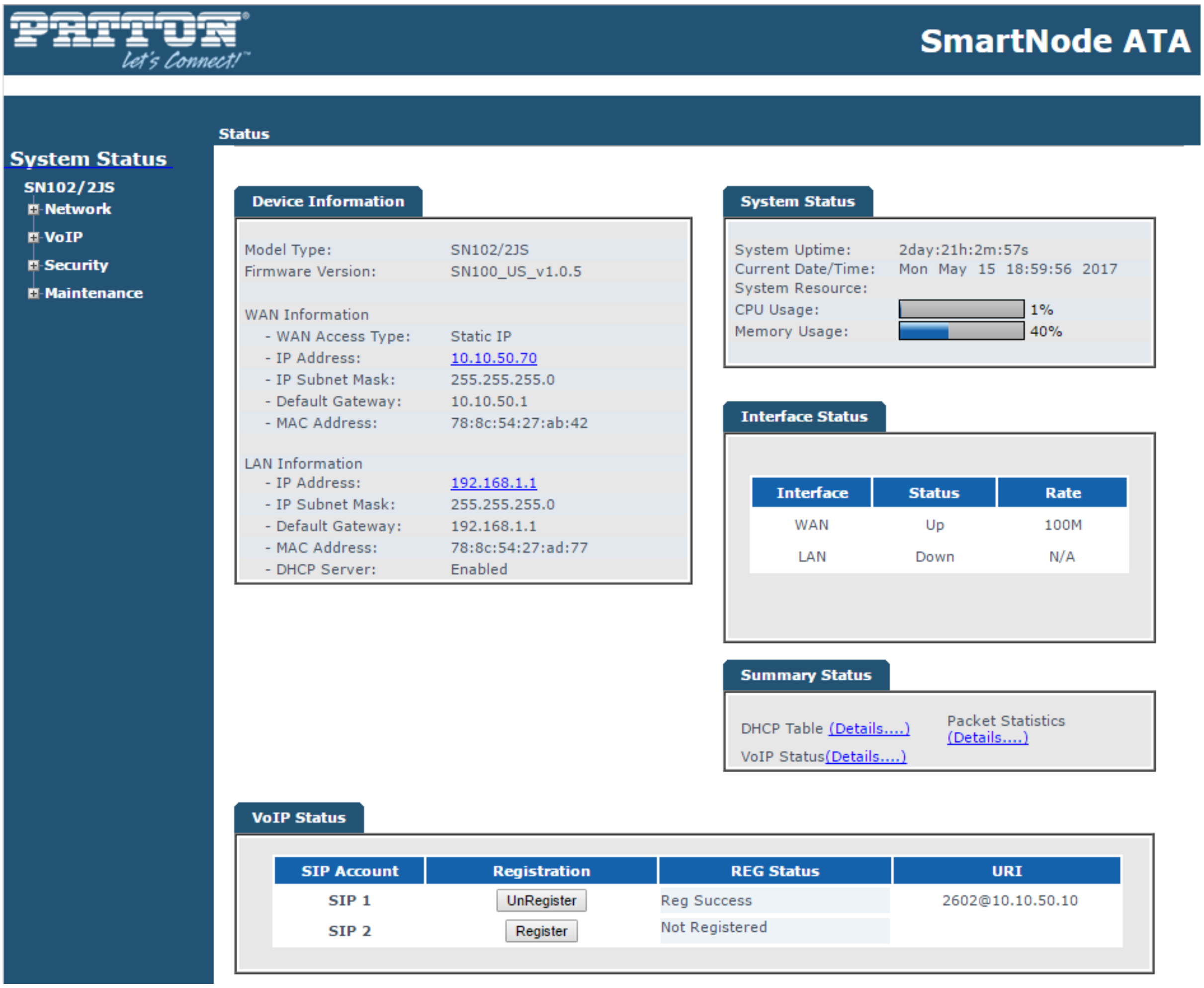



Figure 24. SN102/2JS/E ATA home window

Congratulations! The ATA is ready for use.

## Configuring SN102/1JS1JO/E (1 FXS/1 FXO) ATA VoIP settings

1. Click *VOIP SETUP*. The *VoIP Configuration* window displays (see figure 25 on page 43). Do the following to configure Phone 1 SIP settings:

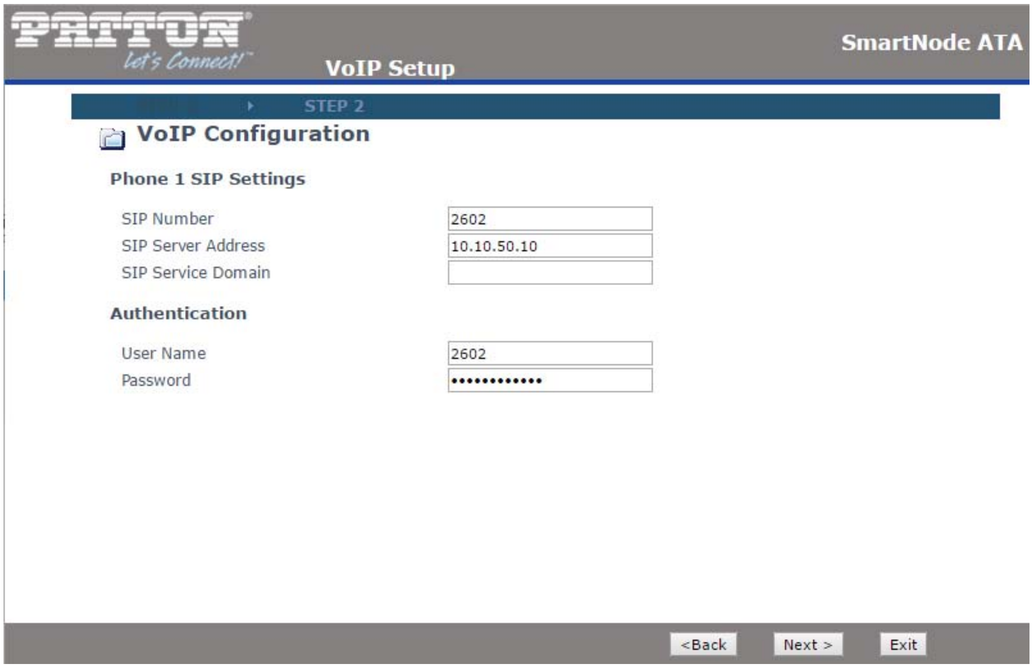



Figure 25. VoIP Configuration - Phone 1 window

2. Type the SIP number that was provided by your ISP in the *SIP Number* box.


3. Type the proxy address that was provided by your ISP in the *SIP Server Address* box.


4. Type the SIP domain that was provided by your ISP in the *SIP Service Domain* box.


5. Type the user name that was provided by your ISP in the *Username* box.


6. Type the password that was provided by your ISP in the *Password* box.

7. Click the *Finish* button. The *Congratulations!* window displays (see figure 26). After a short delay, the *REG Status* table (see figure 27) should display *Reg Success* (registration successful).

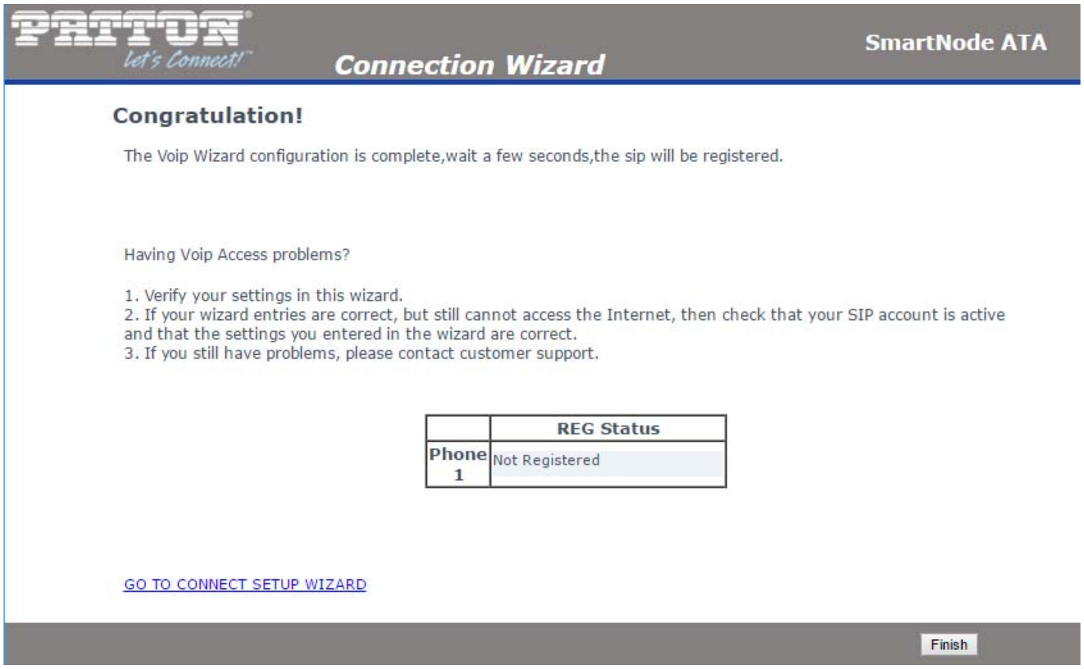



Figure 26. Congratulations! window

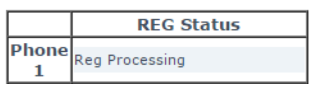



Figure 27. REG Status table

**Note** If the ATA is having a problem registering a phone or fax machine, do the following:

- Verify that the VoIP settings were entered correctly.
- If the VoIP entries are correct, but the ATA still cannot access the Internet, verify that your SIP account is active.
- If ATA is still having problems, please contact technical support (see Chapter 9, "Contacting Patton for assistance" on page 133 for details).

8. Click the *Finish* button. The SmartNode ATA home window (see figure 28 on page 45) displays.

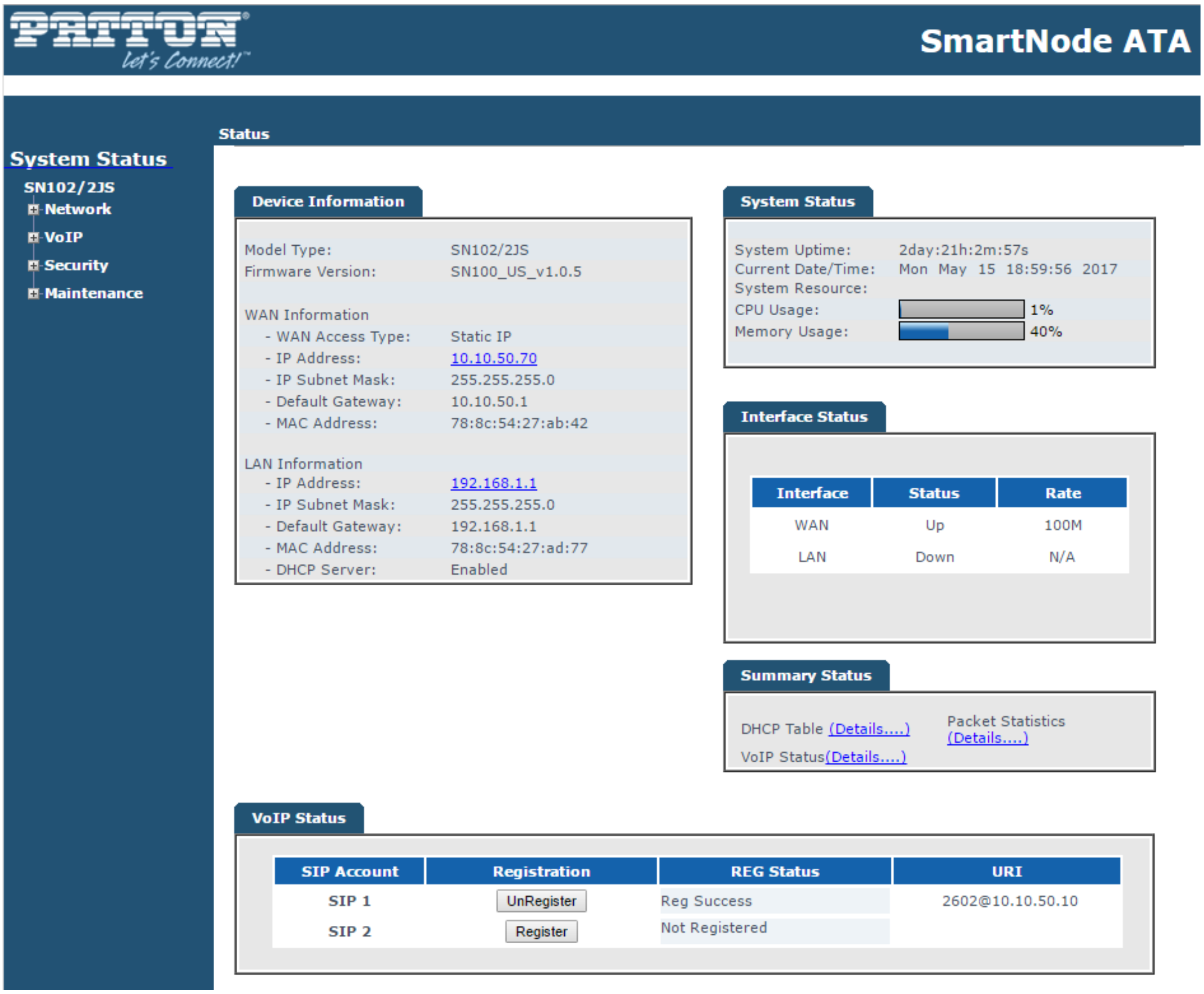



Figure 28. SN102/1JS1JO/E ATA home window

Congratulations! The ATA is ready for use.

IMPORTANT

For proper security we recommend that you change from the default credentials to a new administrator's user name and password as soon as possible (see Chapter 8, "Configuring Administrator Account's user name and password" on page 121).

# Chapter 4 SmartNode ATA Home

## Chapter contents

## Introduction

This chapter describes using the SmartNode ATA home window (see figure 29) to monitor ATA status or to use the advanced settings to configure the ATA.

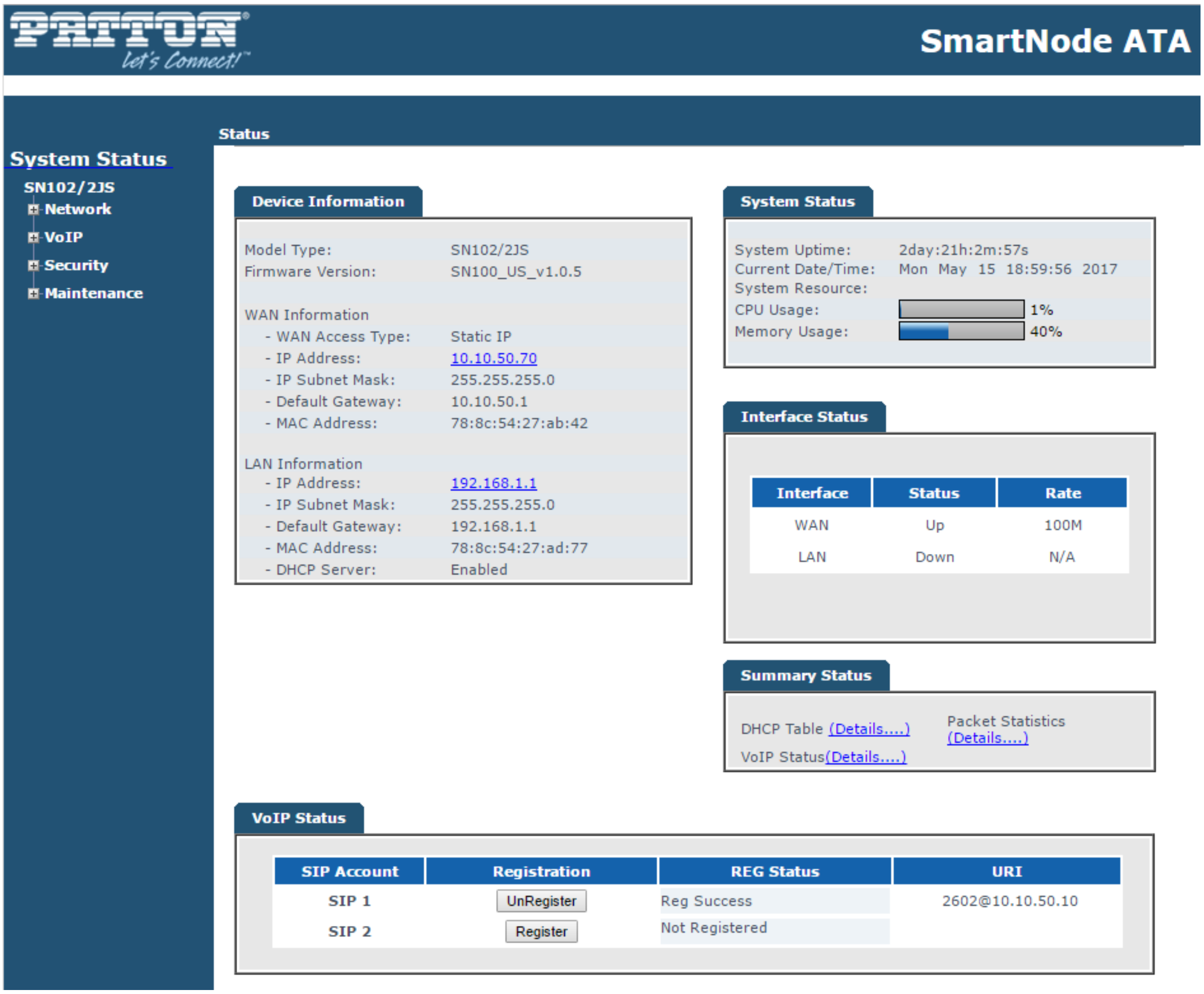



Figure 29. SmartNode ATA home window

## Accessing the home window

1. Using a Web browser (Microsoft Edge, Google Chrome, etc.) link to URL **http://192.168.1.1**. The login window displays (see figure 30 on page 48).

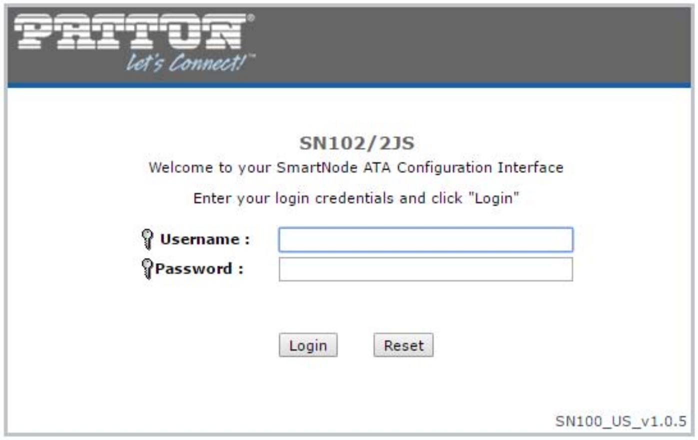



Figure 30. Login window

2. Type the default user name in the *Username* box: **admin**
3. Type the default password in the *Password* box: **admin**
4. Click the *Login* button. The *Please select Wizard or Advanced mode* window displays (see figure 31 on page 48).

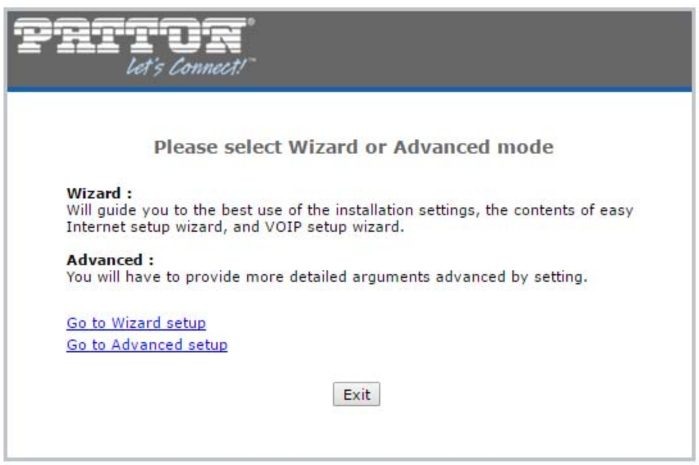



Figure 31. Please select Wizard or Advanced mode window

5. Click *Go to Advanced Setup*. The SmartNode ATA home window (see figure 29 on page 47) displays.

The home window consists of the following sections:

- “System Status pane” on page 49
- “Network, VoIP, Security, and Maintenance Menu” on page 52

## System Status pane

The *System Status* pane (see figure 32) consists of the following sections where you can view the current status and change basic settings of the ATA:

- "Device information" on page 50
- "System status" on page 50
- "Interface status" on page 51
- "Summary status" on page 51
- "VoIP status" on page 51

**Note** The *Status* pane is replaced when you click on items in the Network, VoIP, Security, and Maintenance menu. To return to the *Status* pane, click *System Status as shown in* figure 32.

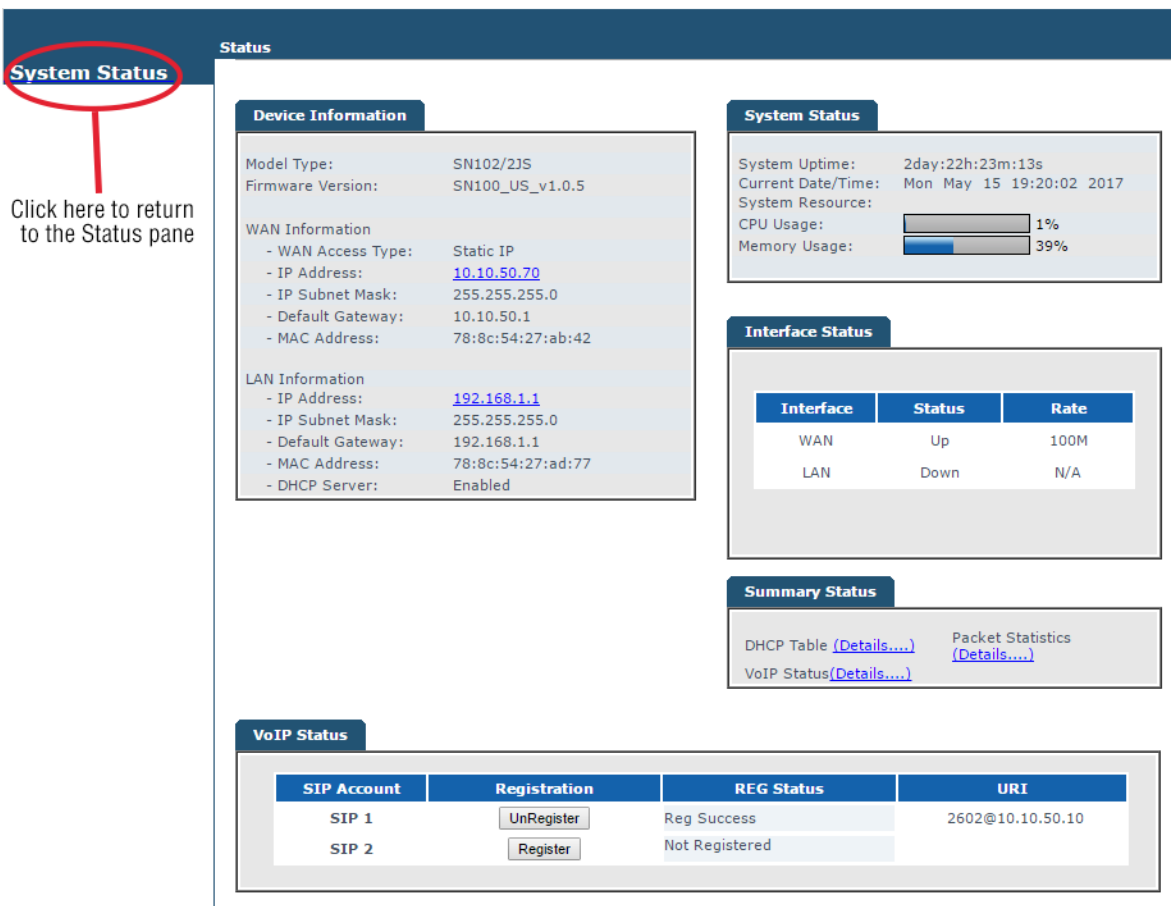



Figure 32. System Status pane

## Device information

The *Device Information* section of the *Status* pane displays model type, firmware version, and information about the WAN (WAN access type, IP address, IP subnet mask, default gateway address, and the MAC address) and information about the LAN (IP address, IP subnet mask, default gateway address, MAC address, and whether the DHCP server is enabled).

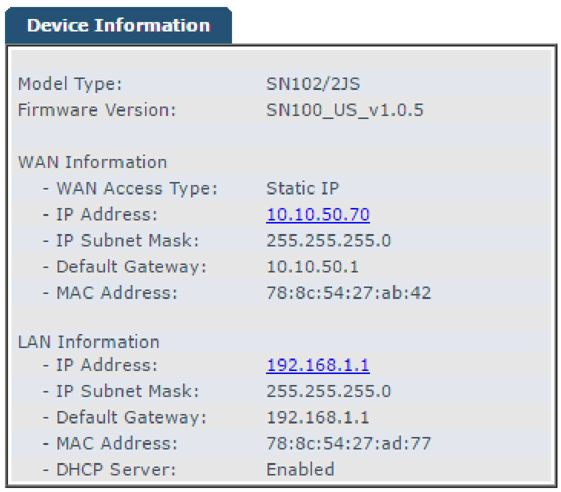



Figure 33. Device Information window

**Note** If you want to change WAN Internet connection settings, a shortcut is to click on the IP Address link under *WAN Information* to display the *Internet Connection* section of the *Network* > *WAN* > *Internet Connection* pane.

**Note** If you want to change LAN settings, a shortcut is to click on the IP Address link under *LAN Information* to display the *LAN Setup* section of the *Network* > *LAN* > *LAN Setup* pane.

## System status

The *System Status* section of the *Status* pane displays how long the system has been running since the last reset (uptime), the current date and time, the percent of CPU capacity in use, and the percent of memory capacity in use.

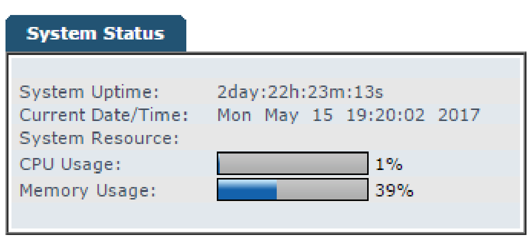



Figure 34. System Status window

### Interface status

The *Interface Status* section of the *Status* pane displays the status (up or down) of the WAN and LAN interfaces, and the rate that data is being transferred across each interface.

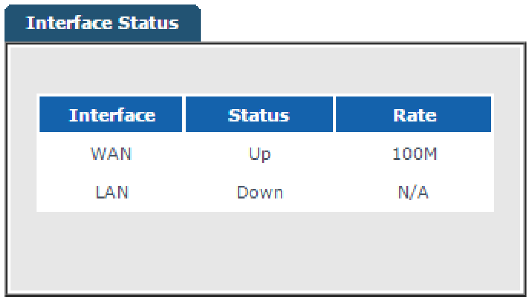



Figure 35. Interface Status window

### Summary status

The *Summary Status* section of the *Status* pane contains links to windows that display the DHCP table, VoIP SIP status and call statistics, and port packet statistics.

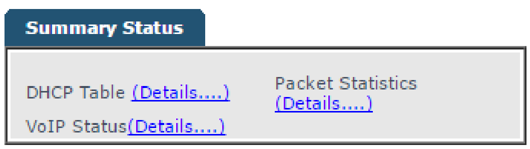



Figure 36. Summary Status window

### VoIP status

The *VoIP Status* section of the *Status* pane displays the SIP account registration status and URI value for each SIP account. It also includes (depending on the REG status of the SIP account) a *Register* button you can click to register a SIP account or an *UnRegister* button that will unregister a SIP account.

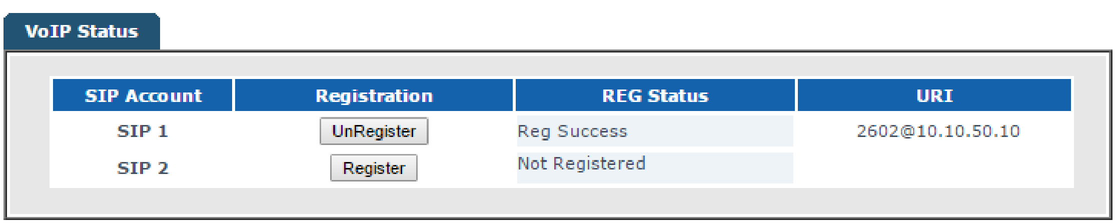



Figure 37. VoIP Status window

## Network, VoIP, Security, and Maintenance Menu

This menu (see figure 38) has the advanced settings for configuring the ATA. It is where you go to change the following:

- Network settings—WAN, LAN, and NAT (see Chapter 5, "Network Settings" on page 53)
- VoIP settings—SIP, phone, and phone book (see Chapter 6, "VoIP Settings" on page 61)
- Security settings—Firewall, content filter, and ACL (see Chapter 7, "Security Settings" on page 112)
- Maintenance settings—System, tools, and logs (see Chapter 8, "Maintenance Settings" on page 119)

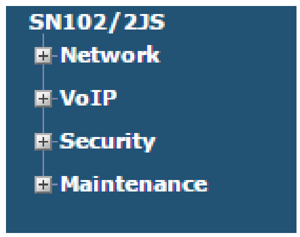



Figure 38. Network, VoIP, Security, and Maintenance Menu

**Note** As shown in figure 38, the ATA model type is displayed at the top of the menu (*SN102/2JS* in this example).

**Note** Click on the plus button next to each category to display sub-categories (see figure 39). The plus button will change to a minus while the sub-categories are displayed.

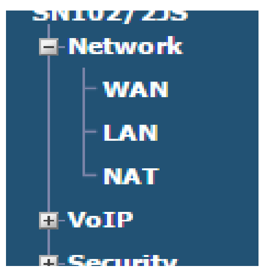



Figure 39. Displaying sub-categories

# Chapter 5 Network Settings

## Chapter contents

## Introduction

This chapter describes the following:

- Configuring the WAN interface (see section "WAN Interface")
- Configuring the LAN interface (see section "LAN Interface" on page 57)
- Configuring NAT (see section "NAT Port Forwarding" on page 59)

## WAN Interface

Do the following to display WAN settings:

1. Click on the plus button next to *Network* to display the sub-categories (see figure 40).

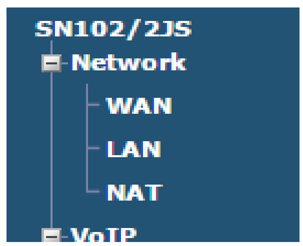



Figure 40. Network sub-categories

2. Click *WAN*. The *Internet Connection* window displays (see figure 41).

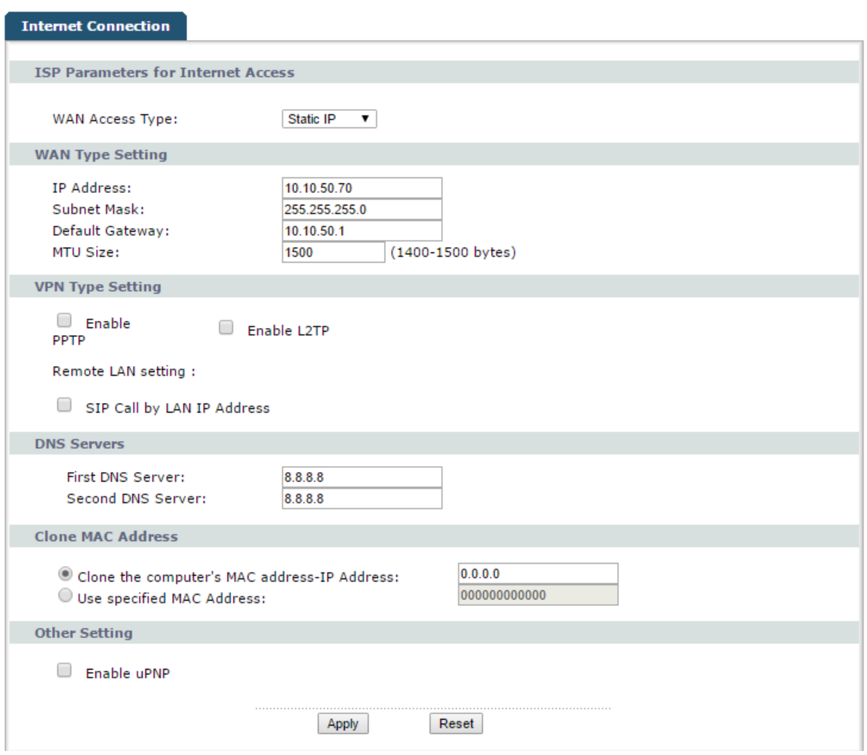



Figure 41. Internet Connection window

A wide area network (WAN) is a network connection connecting one or more local area networks (LANs) together over some distance. For example, two networked office buildings separated by several miles would be referred to as a WAN connection. The size of a WAN and the number of distinct LANs connected to a WAN is not limited by any definition. Therefore, the Internet can be called a WAN.

For most users, Internet access is the primary application. The ATA supports the WAN interface for Internet access and remote access. The WAN settings on the *Internet Connection* window are used to connect to your Internet service provider (ISP). The WAN settings are provided by your ISP and are often referred to as "public settings".

The following sections will explain details of WAN port Internet access and broadband access set up.

- Static IP WAN access type. Click the *WAN Access Type* drop-down menu, then select *Static IP* (see section "Static IP" on page 55).


- DHCP client WAN access type. Click the *WAN Access Type* drop-down menu, then select *Dynamic IP* (see section "DHCP client" on page 56).


- PPPoE WAN access type. Click the *WAN Access Type* drop-down menu, then select *PPPoE* (see section "PPPoE" on page 56).

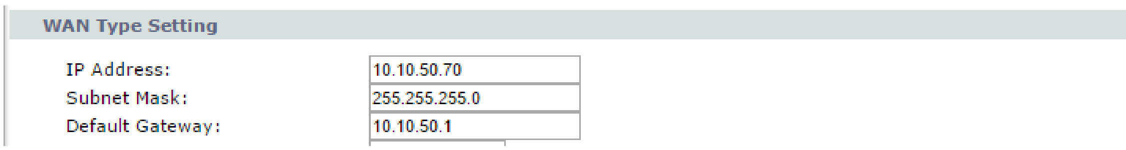

### *Static IP*

If you are a leased-line user with a fixed IP address, type the IP address, subnet mask, gateway address, and DNS (domain name server) address(es) provided to you by your ISP. Each IP address entered in the fields must be in the appropriate IP form, which are four octets separated by three dots (x.x.x.x). The ATA will not accept the IP address if it is not in this format.

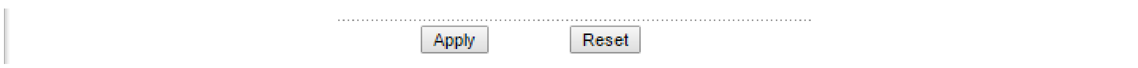
**Example**: 168.95.1.1

Do the following:

1. Type the IP address that was provided by your ISP.
2. Type the subnet mask that was provided by your ISP.
3. Type the default gateway address that was provided by your ISP.




4. Click the *Apply* button.




> **Note** If your ISP uses Layer 2 Tunneling Protocol (L2TP) to support virtual private networks (VPNs) or other services, go to section "L2TP with WAN" on page 57.

### DHCP client

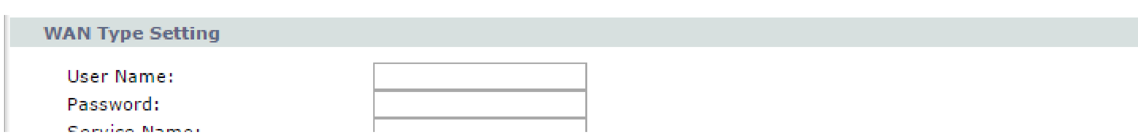
When Dynamic Host Configuration Protocol (DHCP) Internet Protocol (Dynamic IP) is selected, the WAN IP address, subnet mask, and default gateway IP address will be obtained automatically. If you are connected to the Internet through a cable modem line, dynamic IP will be assigned.

> **Note** If your ISP uses Layer 2 Tunneling Protocol (L2TP) to support virtual private networks (VPNs) or other services, go to section "L2TP with WAN" on page 57.

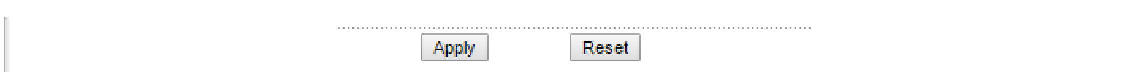
### PPPoE

Some ISPs provide DSL-based services and use Point-to-Point Protocol over Ethernet (PPPoE) to establish communication links with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to configure the following:

1. User name: Type the username provided by your ISP.
2. Password: Type the password provided by your ISP.




3. Click the *Apply* button.

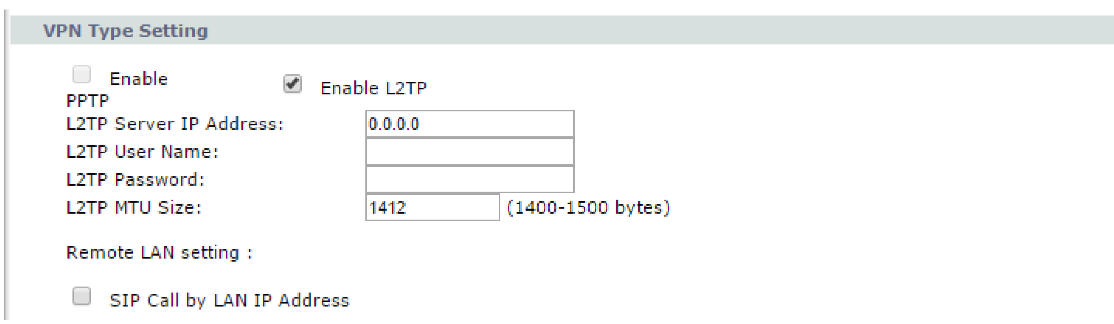


**Note** If your ISP uses Layer 2 Tunneling Protocol (L2TP) to support virtual private networks (VPNs) or other services, go to section "L2TP with WAN" on page 57.

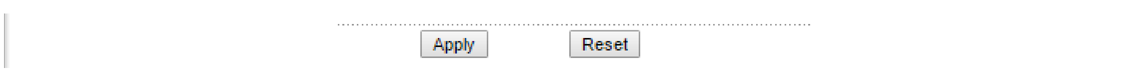
### *L2TP with WAN*

Some ISPs use L2TP with static, DHCP, or PPPoE to establish communication links with end-users.

1. Type the L2TP server IP address: that was provided by your VPN /PPTP server provider
2. Type the L2TP user name (L2TP dial-in account)
3. Type the L2TP password (L2TP dial-in password)
4. Type the L2TP MTU size
5. If appropriate, click the *Remote LAN setting* check box to *enable SIP Call by LAN IP Address*




6. If appropriate, click the *Remote LAN setting* check box to *enable SIP Call by LAN IP Address*
7. Click the *Apply* button.




## LAN Interface

Do the following to display LAN settings:

1. Click on the plus button next to *Network* to display the sub-categories (see figure 40 on page 54).
2. Click *LAN*. The *LAN Setup* window displays (see figure 41 on page 54).

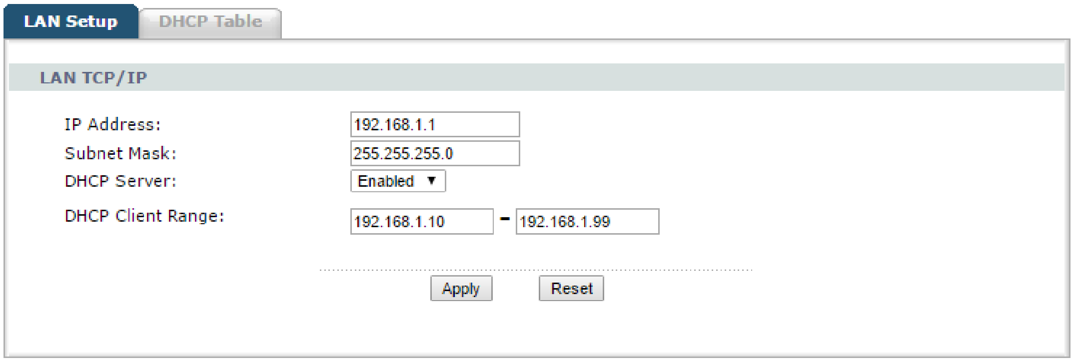


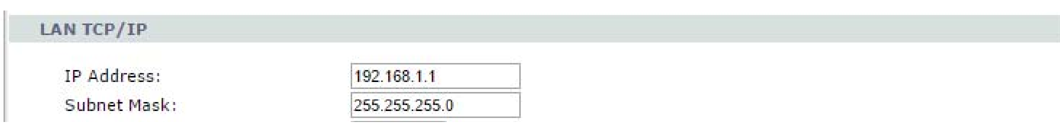

Figure 42. LAN setup window

A local area network (LAN) is a network of computers or other devices that are in relatively close range of each other. For example, devices in a home or office building would be considered part of a local area network.

The LAN Setup window contains the LAN TCP/IP interface settings for the ATA. These settings may be referred to as "private settings". As the LAN IP address is private to your internal network and cannot be seen on the Internet, you can change the LAN IP address if needed.

Do the following:

1. Type the IP address that was provided by your ISP. The default address is *192.168.1.1*.
2. Type the subnet mask that was provided by your ISP. The default is *255.255.255.0*.




3. If you will be using a DHCP Server, click on the drop-down menu, then select Enabled.

   DHCP stands for Dynamic Host Control Protocol. The DHCP server gives out IP addresses when a device is starting up and requests an IP address to be logged onto the network. The device must be set as a DHCP client to "Obtain the IP address automatically". By default, the DHCP Server is enabled in the unit. The DHCP address pool contains the range of the IP address that will automatically be assigned to the clients on the network.

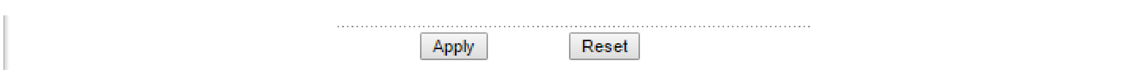
DHCP client computers connected to the unit will have their information displayed in the DHCP Client List table (click the *DHCP Table* tab to display the table). The table displays the type, host name, IP address, MAC address, description, and expired time of the DHCP lease for each client computer.


Figure 43. DHCP Table window

DHCP Server is a useful tool that automates the assignment of IP addresses to numbers of computers in your network. The server maintains a pool of IP addresses that you use to create scopes. (A DHCP scope is a collection of IP addresses and TCP/IP configuration parameters that are available for DHCP clients to lease.) Then, the server automatically allocates these IP addresses and related TCP/IP configuration settings to DHCP-enabled clients in the network. The DHCP Server leases the IP addresses to clients for a period that you specify when you create a scope. A lease becomes inactive when it expires. Through the DHCP Server, you can reserve specific IP addresses permanently for hardware devices that must have a static IP address (e.g., a DNS Server).

An advantage of using DHCP is that the service assigns addresses dynamically. The DHCP Server returns addresses that are no longer in use to the IP addresses pool so that the server can reallocate them to other machines in the network. If you disable this DHCP, you would have to manually configure IP for new computers, keep track of IP addresses so that you could reassign addresses that clients aren't using, and reconfigure computers that you move from one subnet to another. The DHCP Static MAP table lists all MAC and IP address which are active now.

4. Set the *Assigned DHCP IP Address* by typing the starting IP address for the DHCP server's IP assignment and the ending IP address for the DHCP server's IP assignment.
5. Set the *DHCP IP Lease Time* by assigning the length of time for the IP lease. The default setting is 86400 seconds.
6. Click the *Apply* button.




## NAT Port Forwarding

Port forwarding is an application of network address translation (NAT) that redirects a communication request from one address and port number combination to another while the packets are traversing a network gateway, such as a router or firewall. In other words, port forwarding is used when someone on an outside network (public network) wants to access a service running on a private network machine. An example where port forwarding is applicable is if you plan to run a web, email, or file server on your computer.

Do the following to display port forwarding settings:

1. Click on the plus button next to *Network* to display the sub-categories (see figure 40 on page 54).
2. Click *NAT*. The *Port Forwarding* window displays (see figure 44).

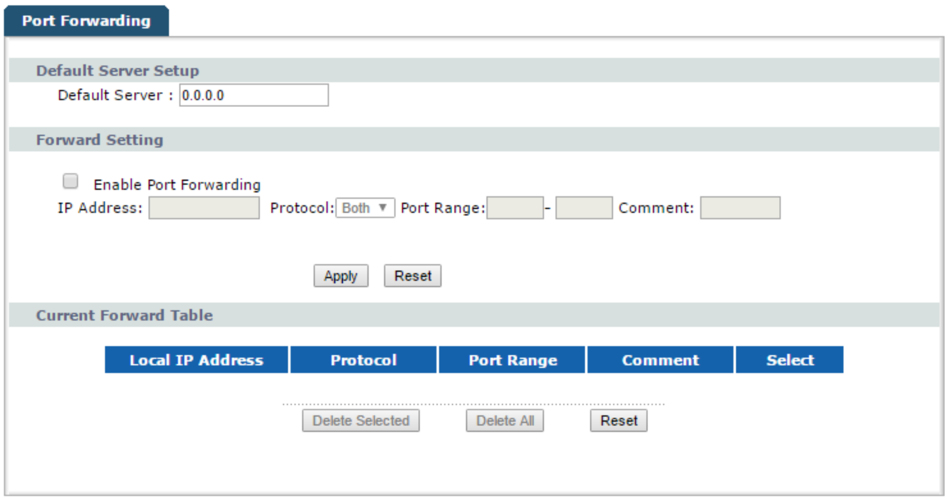



Figure 44. Port Forwarding window

Entries in this table enable you to automatically redirect common network services to a specific machine behind the NAT firewall. These settings are only necessary if you wish to host some sort of server like a web server or email server on the private local network behind your gateway's NAT firewall.

# Chapter 6 VoIP Settings

## Chapter contents

# Introduction

This chapter describes the following:

- Configuring session initiation protocol (SIP) and quality of service (QoS) settings (see section "SIP")
- Configuring analog phone settings (see section "Phone" on page 99)
- Configuring speed-dial settings (see section "Phone Book" on page 107)
- Configuring public switched telephone network (PSTN) line settings on the SN102/1JS1JO (FXS/FXO model) (see section "PSTN Line" on page 111)

# SIP

This section describes configuring SN100 Series ATA SIP and QoS settings. Chose the appropriate section from the following:

- "Configuring SN101/1JS/E (1 FXS) ATA SIP settings" on page 63
- "Configuring SN102/2JS/E (2 FXS) ATA SIP settings" on page 75
- "Configuring SN102/1JS1JO/E (1 FXS/1 FXO) ATA SIP settings" on page 87

## *Configuring SN101/1JS/E (1 FXS) ATA SIP settings*

Do the following to display SIP and QoS settings:

1. Click on the plus button next to *VoIP* to display the sub-categories (see figure 45).

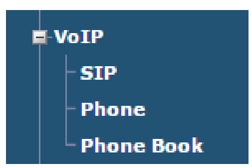



Figure 45. SN101/1JS/E VoIP sub-categories

2. Click *SIP*. The *SIP Settings* window displays (see figure 46 on page 64).

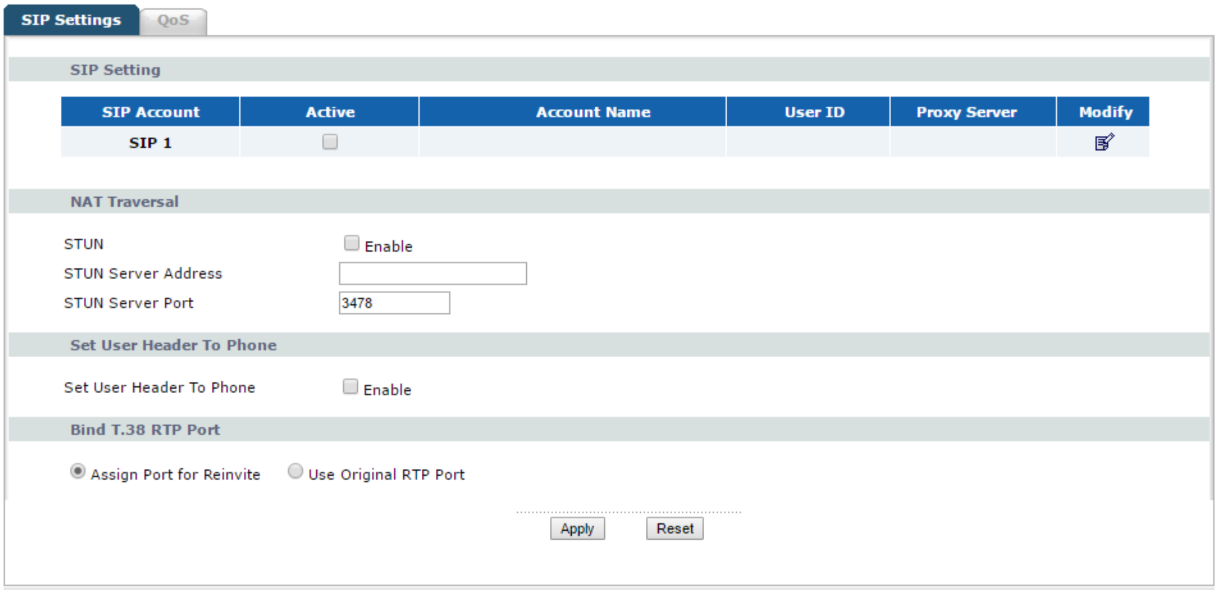



Figure 46. SN101/1JS/E SIP Settings window

3. Click the *QoS* tab to display the QoS window (see figure 47).


Figure 47. QoS window

Go to the appropriate section to modify SIP or QoS settings:

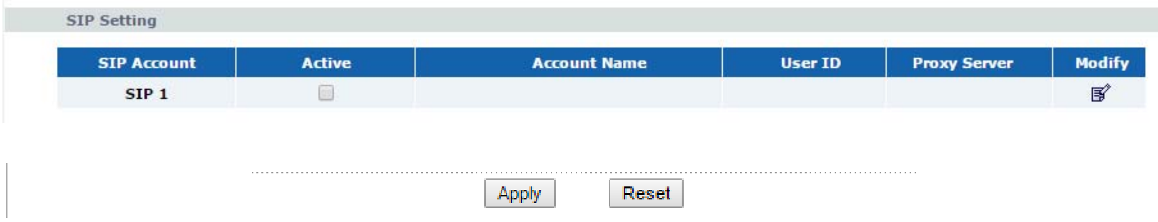
- "SIP Settings"
- "QoS" on page 74

### *SIP Settings*

SIP is a request-response protocol that deals with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media, and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

The *SIP Settings* window displays the following:

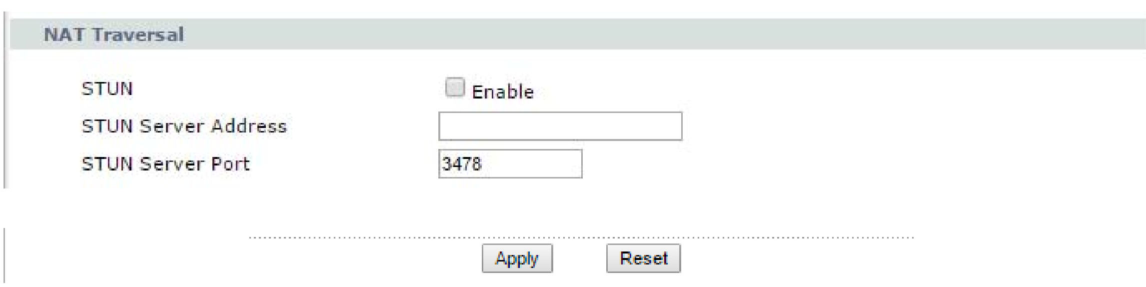
- SIP account information (whether the account is active, the account name, user ID, and proxy server address). SIP settings can be modified by clicking on the *Modify* button for the appropriate account (see section “Edit SIP Settings window” on page 66 for more information).




**Note** Click the *Active* check box to activate or deactivate a SIP account. Click the *Apply* button to save and apply the change.

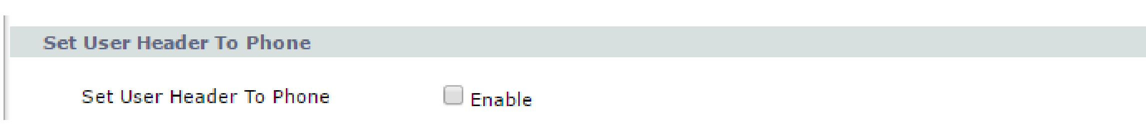
**Note** Once you are satisfied with the settings for a SIP account, if the ATA has another SIP account, you can click on the *CopyToSIP2* button to copy the settings to the other account. Click the *Apply* button to save and apply the change.

- NAT traversal information (whether STUN is enabled, the STUN server address, and the STUN server port identifier). STUN {Simple Traversal of UDP through NATs (Network Address Translation)} is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

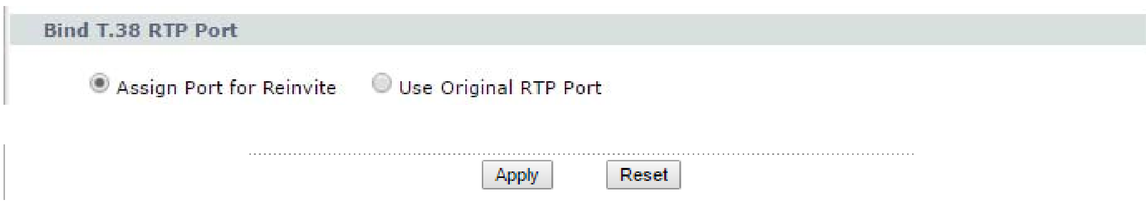



**Note** Click the *Enable* check box to activate or deactivate STUN. Click the *Apply* button to save and apply the change.

- *Set User Header to Phone* information. Click the check box to enable *Set User Header to Phone* to have the device add a “user=phone” parameter to the SIP URI before sending a SIP INVITE request. This parameter indicates the user part of the SIP URI (<user>@<sipserver>) the device is calling is a valid telephone number.(user name and password). Click the *Apply* button to save and apply the change




- *Bind T.38 RTP Port* information. If you need to send fax messages using T.38 and want to always use the same port number for transmitted and received T.38 packets, click the radio button to select *Use Original RTP Port*. Otherwise, select *Assign Port for Reinvite* to have the device chooses a port randomly for T.38 after the re-invite process. Click the *Apply* button to save and apply the change



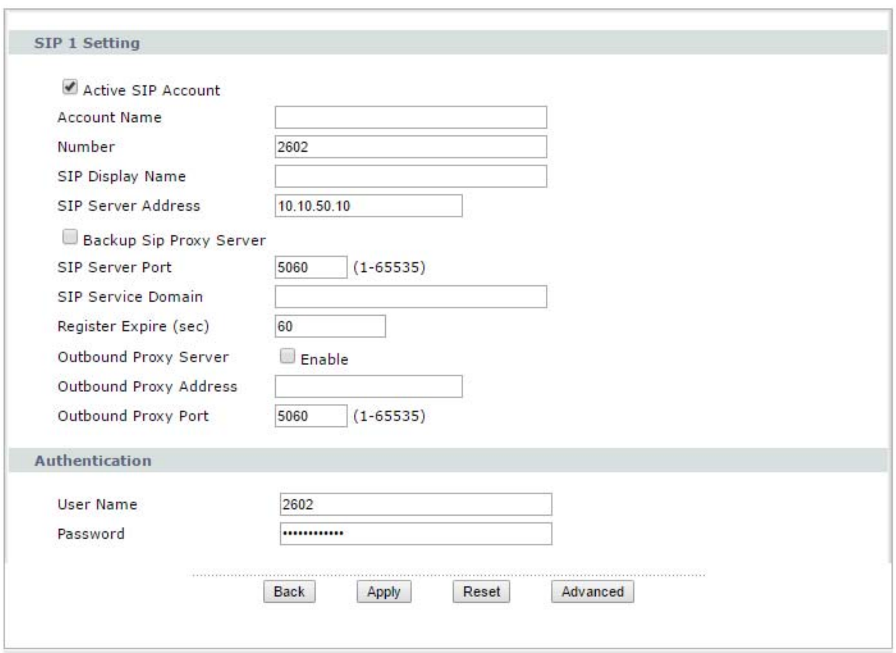



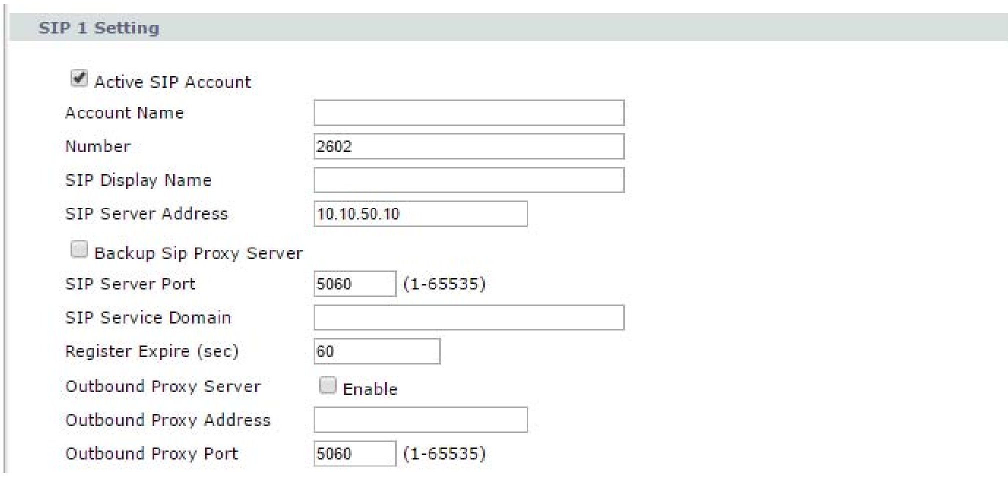

Figure 48. Edit SIP Settings window

Edit SIP Settings window. The Edit SIP Settings window (see figure 48) displays the following sections;

- SIP settings
- Authentication settings

**SIP Settings**. This section displays the following:

- Account Name: Type the name you want displayed.
- Number: Type the number provided by your ISP.




- SIP Display Name: Type the name you want displayed.
- SIP Server Address: Type the address.
- Backup SIP Proxy Server: Click to enable or disable the backup proxy server
- Backup1 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- Backup2 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- SIP Server Port: Type the Proxy port number provided by your ISP.
- SIP Server Domain: Type the SIP domain provided by your ISP.
- Register Expire (sec): SIP registration expired time (default is 600 and needed for calls through the satellite connection).
- Outbound Proxy Server: Click to enable or disable the outbound proxy server
- Outbound Proxy Address: Type the address provided by your ISP. If your ISP did not provide the information, you can skip this item.
- Outbound Proxy Port: Type the port number provided by your ISP.

**Authentication Settings**. This section displays the following:

- User Name: Type the Register Name provided by your ISP.
- Password: Type the Register Password provided by your ISP.


If you are satisfied with the settings, click the *Apply* button to save and apply the changes.

if you need to change the following:

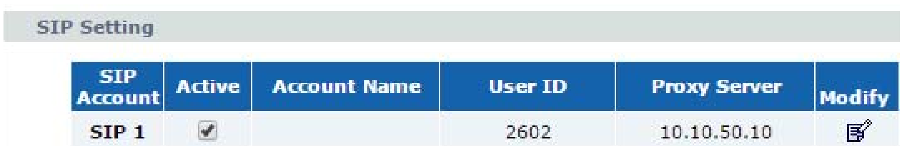
- Advanced SIP settings
- Forwarding settings
- Fax option settings
- P-Asserted-Identity setting
- Dial termination key setting
- Session timer settings
- MWI (message waiting indication) settings
- DND (do not disturb) settings
- Block anonymous call setting
- Codec settings

Go to section "Advanced Edit SIP Settings window" on page 68. Otherwise, SIP settings are complete.

### *Advanced Edit SIP Settings window*

To display the Advanced Edit SIP Settings window, do the following:

1. Click *SIP* under the *VoIP* submenu.
2. Click the *Modify* button in the *SIP Setting* section of the *SIP Settings* window.




3. Click the *Advanced* button to display the Advanced Edit SIP Settings window.


The Advanced Edit SIP Settings window (see figure 49 on page 69) displays the following;

- Advanced SIP settings (see section "SIP Advanced" on page 70)
- Forwarding settings (see section "Forward Setting" on page 71)
- Fax settings (see section "Fax Option" on page 71)
- P-Asserted-Identity setting (see section "P-Asserted-Identity" on page 71)
- Dial termination key setting (see section "Dial Termination Key" on page 72)
- Session timer settings (see section "Session Timer" on page 72)
- Message waiting indication (MWI) settings (see section "MWI (Message Waiting Indication)" on page 72)
- Do not disturb (DND) settings (see section "DND (Do Not Disturb)" on page 73)

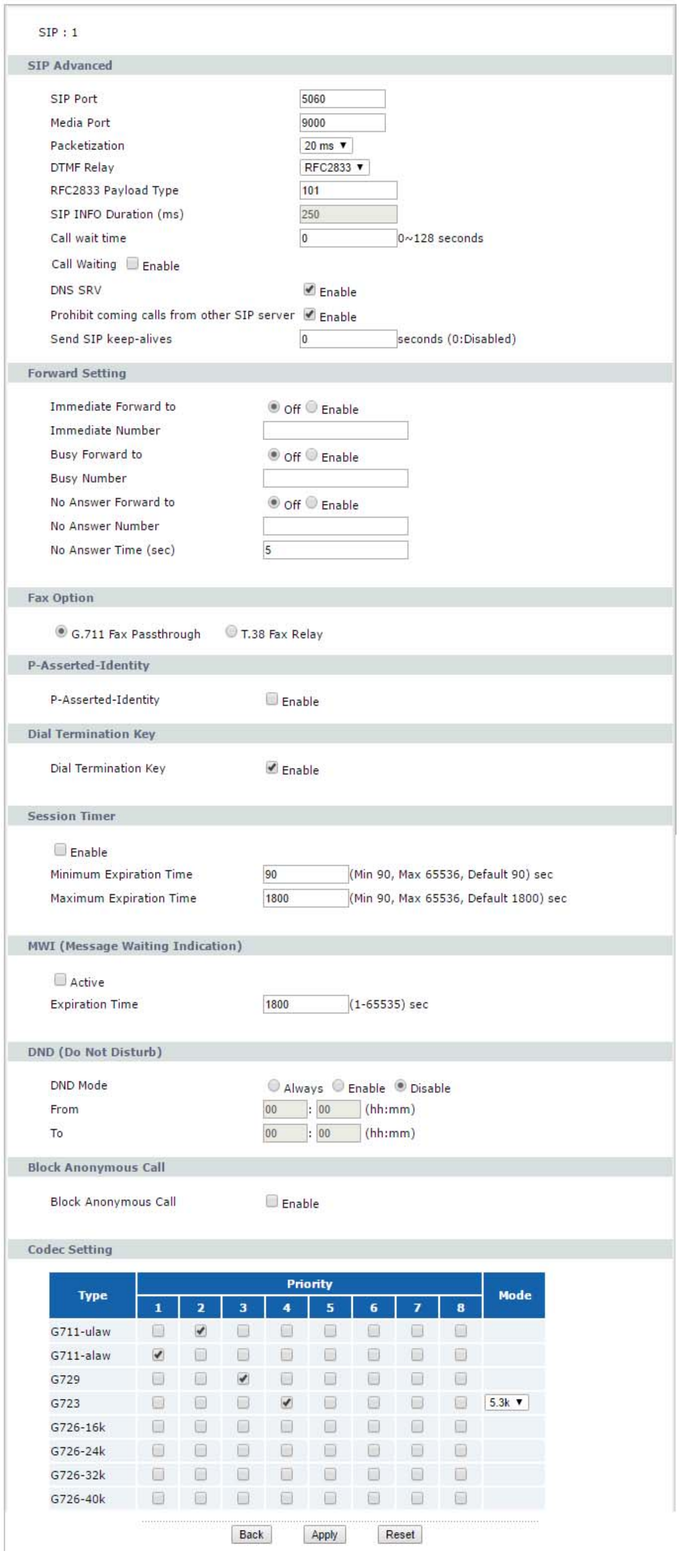
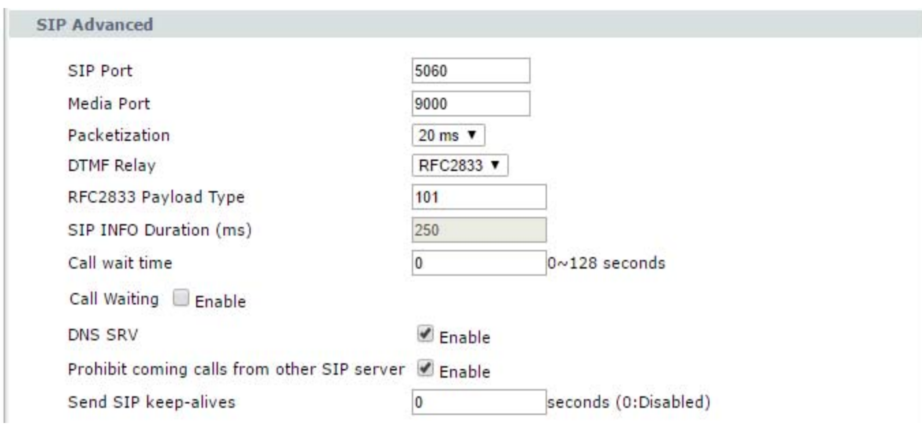



Figure 49. Advanced Edit SIP Settings window

- Block Anonymous Call setting (see section "Block Anonymous Call" on page 73)
- Codec settings (see section "Codec Setting" on page 73)

SIP Advanced. This section displays the following;

- SIP Port Number: Assign the SIP port number of the ATA. The default setting is 5060.
- Media Port: Port number for sending RTP packets. The default setting is 9000.
- Packetization: Select the voice codec packetization interval in milliseconds. The default is 30 ms (for terrestrial connection it is 20ms). This is used to minimize loss that happens during transmission of voice data over the network.
- DTMF Relay: Click the drop-down menu to select RFC2833, SIP INFO, or PCM as indicated by your ISP.
- RFC2833 Payload Type: Sending the DTMF tone as an RTP payload signal.
- SIP INFO Duration (ms): Modify SIP INFO duration time.



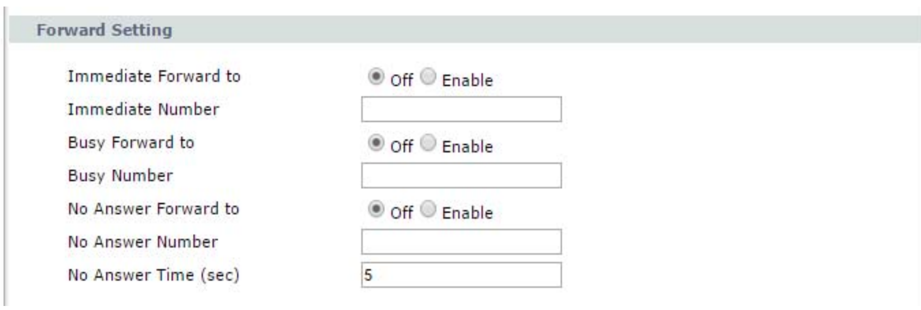

- Call wait time (seconds)
- Call Waiting: Click the check box to enable call waiting if the user wants to be informed when there is a new incoming call while he or she is on another call. Disable the feature if the user doesn't want to be informed of a new incoming call.
- DNS SRV: Click the check box to enable having the ATA query your ISP's DNS server for a list of available SIP servers that it maintains.
- Prohibit coming calls from other SIP server: Enable or disable prohibiting incoming calls from other SIP servers.
- Send SIP keep-alives (seconds)

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

**Forward Setting.** You can set up the phone number you want to forward in this section. There are three forwarding modes:

- Immediate Forward: All incoming call will forward to the number you entered. You can input the phone number. If you select this function, then all the incoming call will direct forward to the speed dial number you entered.
- Busy Forward: If you are on the phone, the new incoming call will forward to the number you entered.
- No Answer Forward: If nobody picks up the phone, the incoming call will forward to the number you entered.

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.





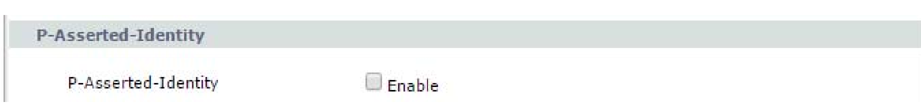
**Fax Option.** G.711 Fax Pass through: Select this if the SN100 should use G.711 to send fax messages. The peer devices must also use G.711.

T.38 Fax Relay: Select this if the SN100 should send fax messages as UDP packets through IP networks. This provides better quality, but it may have interoperability problems. The peer devices must also use T.38.


When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


**P-Asserted-Identity.** The SIP server authenticates the user that sent the SIP request and then uses the identity from the result of authentication to generate a P-Asserted-Identity header field.

Select enable to have the device look at the P-Asserted-Identity header field which carries the identity of the caller in SIP message, and display the caller's location if available.



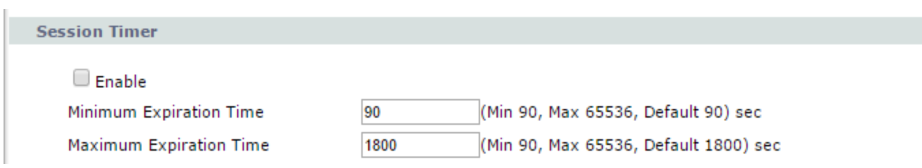
When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**Dial Termination Key.** Select enable if you want to use the pound (#) key to tell device to make the phone call immediately.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**Session Timer.** Select Enable if you want to define how long the ATA waits to receive a session-alive packet for a voice session from the SIP server.

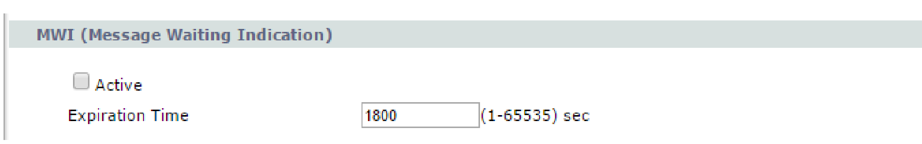
Minimum Expiration Time: Enter the minimum time the ATA waits for a session-alive packet (90-65536 seconds). If a session-alive packet is not received during this time, the voice session is terminated.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**MWI (Message Waiting Indication).** Active: Select this if you want to hear a waiting (beeping) dial tone on your phone when you have at least one voice message. Your SIP service provider must support this feature.

Expiration Time: Keep the default value, unless your SIP service provider tells you to change it. Enter the number of seconds the SIP server should provide the message waiting service each time the ATA subscribes to the service. Before this time passes, the ATA automatically subscribes again.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

DND (Do Not Disturb). You can set up the DND setting to keep the device silence. You can choose from the following:

- Always: All incoming call will be blocked until this feature is disabled by clicking on *Disable*.
- Enable: The ATA will be blocked during the time period. If the “From” time is large than the “To” time, the block time will from 00:00 to 23:59

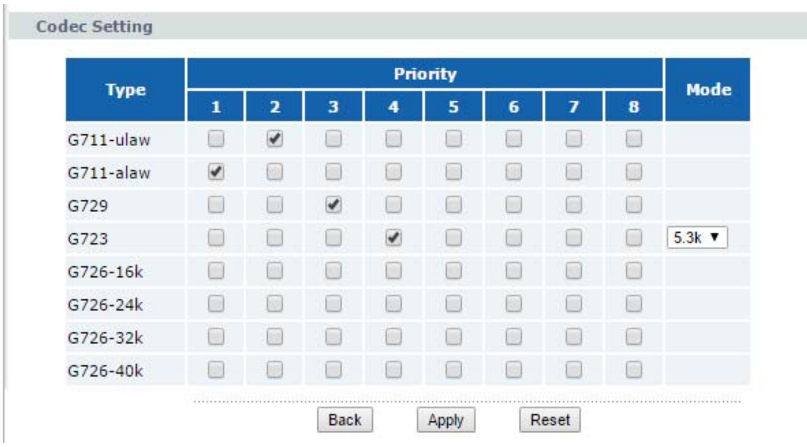

When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Block Anonymous Call. If this feature is enabled, calls that have caller ID information appearing as "private" or "anonymous" will be blocked.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Codec Setting. A codec is an algorithm for taking voice or video and compressing the information. This type of codec combines analog-to-digital conversion and digital-to-analog conversion functions in a single chip. The codec is used to compress the voice signal into data packets. Each codec has different bandwidth requirement. There are the following types of codecs:

- G.711/Ulaw
- G.711/Alaw
- G.729 (default and standard for calls through a satellite connection)
- G.723 (5.3k/6.3k bps)
- G.726(16K bps)
- G.726(24K bps)
- G.726(32K bps)
- G.726(40K bps)

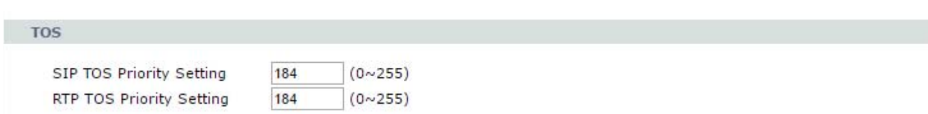



When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

### *QoS*

Quality of service (QoS) refers to traffic prioritization and resource reservation control mechanisms; it is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

**ToS.** ToS ensures that when voice and data are being transmitted at the same time, the data will not degrade the voice signal quality. When type of service (ToS) bits are enabled, voice will have the first priority pass through ToS enabled devices.




**VLAN Tagging.** You can set the VLAN setting in this section.


VLAN tags: if you enable *VOICE VLAN Tags*, and set the *Voice VLAN ID* and *Priority*, then all the incoming packets will be tagged with the IP address and the VLAN ID (VID):

- Voice VLAN ID: Use the voice VLAN ID provided by your ISP to set the *Voice VLAN ID*.
- Data VLAN ID: Use the data VLAN ID provided by your ISP to set the *Data VLAN ID*.
- Priority: Defines user priority, giving eight ($2^3$) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be provided by your ISP.

### Configuring SN102/2JS/E (2 FXS) ATA SIP settings

Do the following to display SIP and QoS settings:

1. Click on the plus button next to *VoIP* to display the sub-categories (see figure 50).

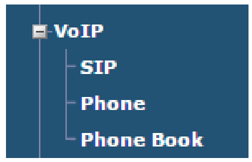



Figure 50. SN102/2JS/E VoIP sub-categories

2. Click *SIP*. The *SIP Settings* window displays (see figure 51).

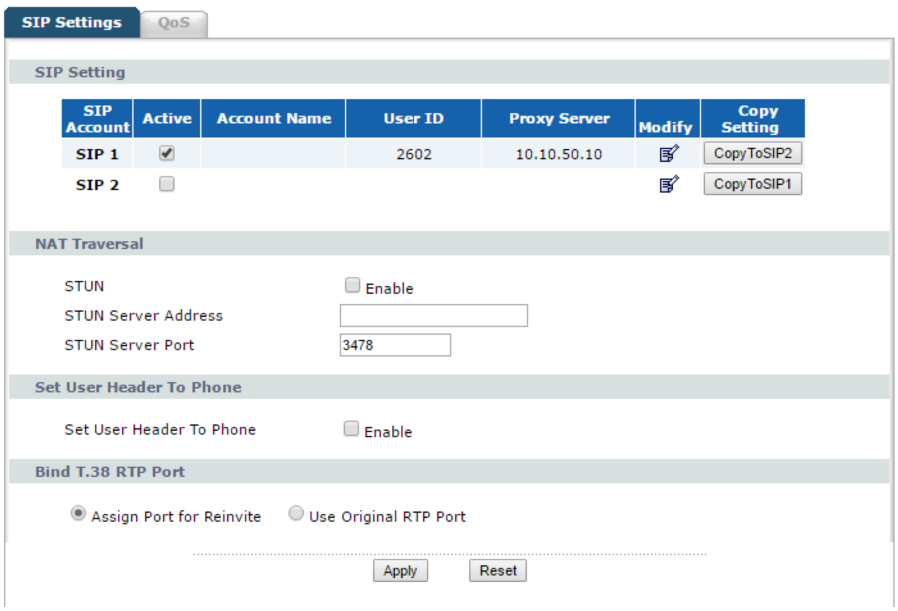



Figure 51. SN102/2JS/E SIP Settings window

3. Click the *QoS* tab to display the QoS window (see figure 52 on page 76).

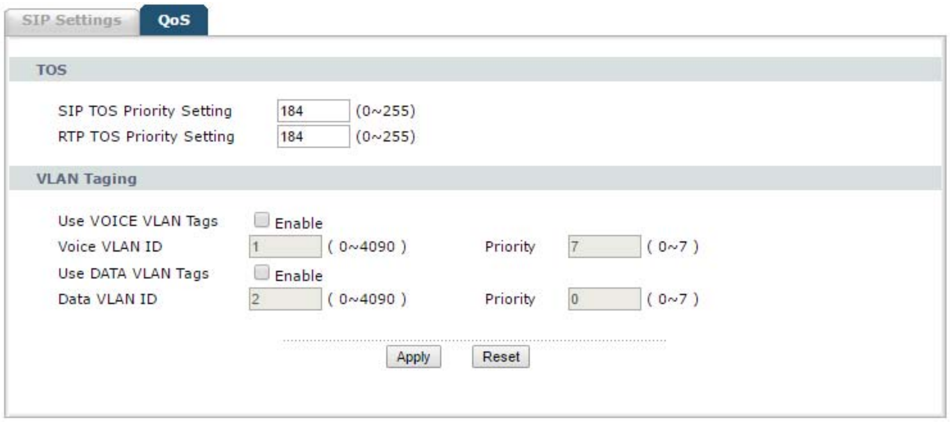



Figure 52. QoS window

Go to the appropriate section to modify SIP or QoS settings:

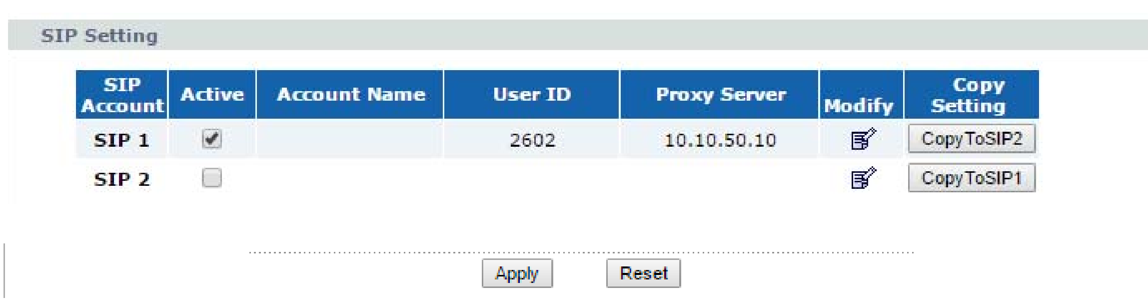
- “SIP Settings”
- “QoS” on page 86

### *SIP Settings*

SIP is a request-response protocol that deals with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media, and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

The *SIP Settings* window displays the following:

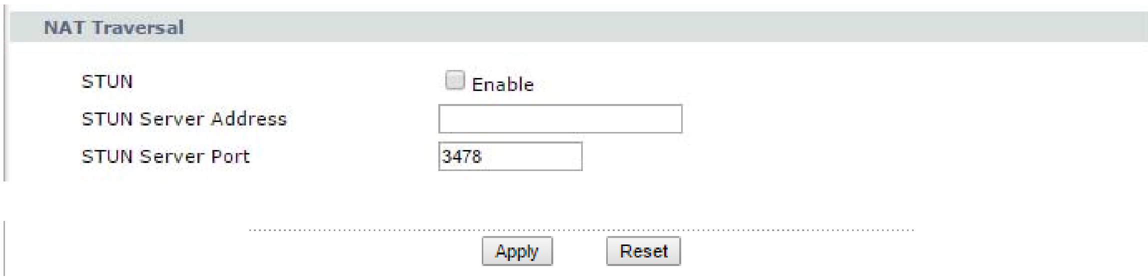
- SIP account information (whether the account is active, the account name, user ID, and proxy server address). SIP settings can be modified by clicking on the *Modify* button for the appropriate account (see section “Edit SIP Settings window” on page 66 for more information).




**Note** Click the *Active* check box to activate or deactivate a SIP account. Click the *Apply* button to save and apply the change.

**Note** Once you are satisfied with the settings for a SIP account, if the ATA has another SIP account, you can click on the *CopyToSIP2* button to copy the settings to the other account. Click the *Apply* button to save and apply the change.

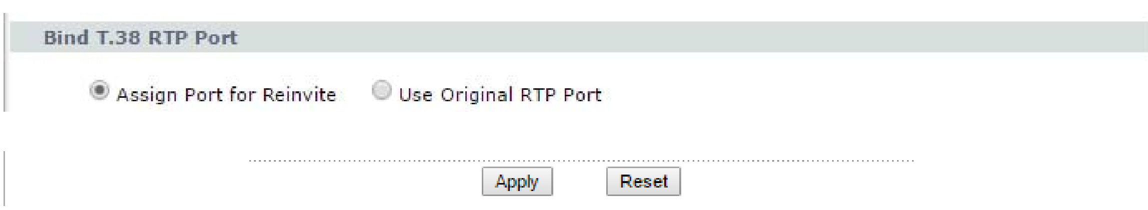
- NAT traversal information (whether STUN is enabled, the STUN server address, and the STUN server port identifier). STUN {Simple Traversal of UDP through NATs (Network Address Translation)} is a protocol for assisting devices behind a NAT firewall or router with their packet routing.




**Note** Click the *Enable* check box to activate or deactivate STUN. Click the *Apply* button to save and apply the change.

- *Set User Header to Phone* information. Click the check box to enable *Set User Header to Phone* to have the device add a “user=phone” parameter to the SIP URI before sending a SIP INVITE request. This parameter indicates the user part of the SIP URI (<user>@<sipserver>) the device is calling is a valid telephone number.(user name and password). Click the *Apply* button to save and apply the change


- *Bind T.38 RTP Port* information. If you need to send fax messages using T.38 and want to always use the same port number for transmitted and received T.38 packets, click the radio button to select *Use Original RTP Port*. Otherwise, select *Assign Port for Reinvite* to have the device chooses a port randomly for T.38 after the re-invite process. Click the *Apply* button to save and apply the change



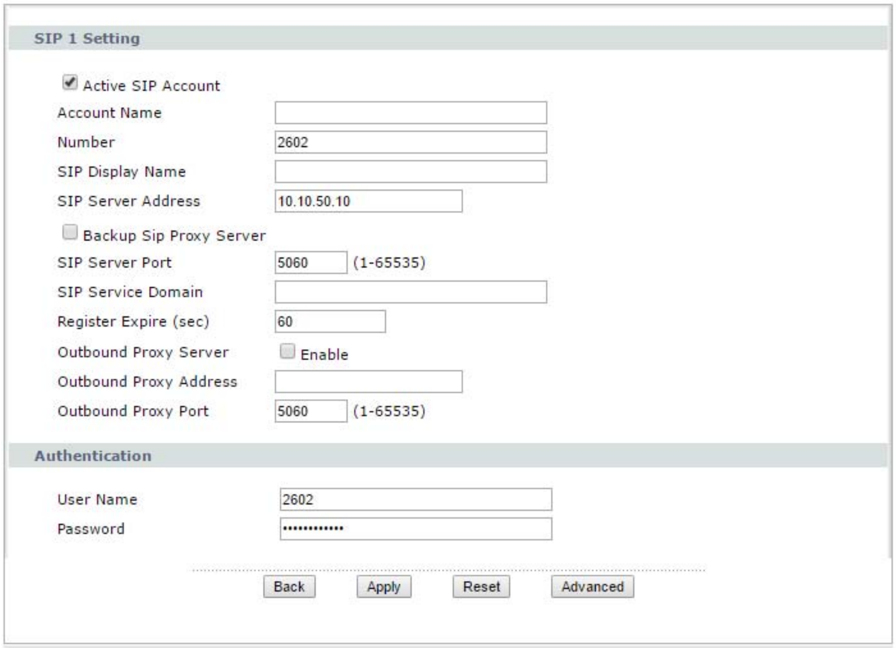


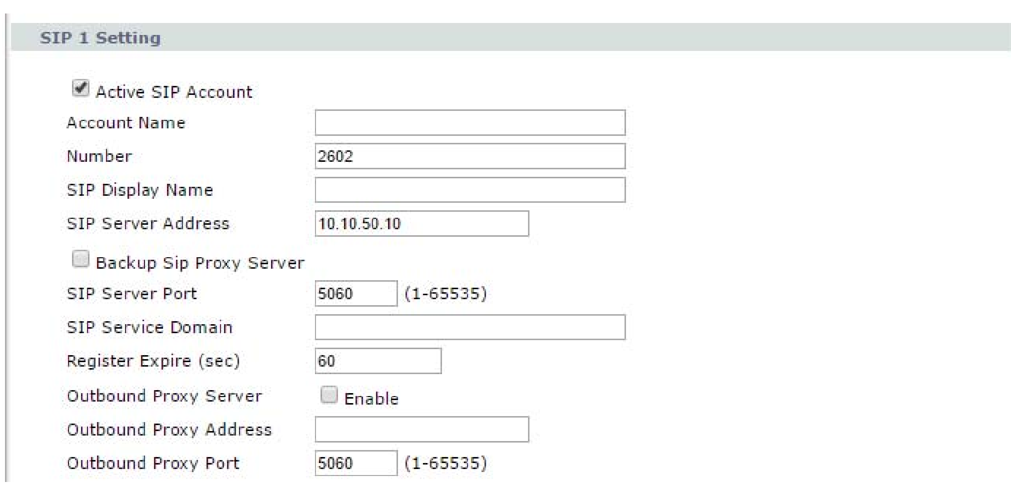

Figure 53. Edit SIP Settings window

Edit SIP Settings window. The Edit SIP Settings window (see figure 53) displays the following sections;

- SIP settings
- Authentication settings

**SIP Settings**. This section displays the following:

- Account Name: Type can the name you want displayed.
- Number: Type the number provided by your ISP.




- SIP Display Name: Type the name you want displayed.
- SIP Server Address: Type the address.

- Backup SIP Proxy Server: Click to enable or disable the backup proxy server
- Backup1 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- Backup2 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- SIP Server Port: Type the Proxy port number provided by your ISP.
- SIP Server Domain: Type the SIP domain provided by your ISP.
- Register Expire (sec): SIP registration expired time (default is 600 and needed for calls through the satellite connection).
- Outbound Proxy Server: Click to enable or disable the outbound proxy server
- Outbound Proxy Address: Type the address provided by your ISP. If your ISP did not provide the information, you can skip this item.
- Outbound Proxy Port: Type the port number provided by your ISP.

**Authentication Settings**. This section displays the following:

- User Name: Type the Register Name provided by your ISP.
- Password: Type the Register Password provided by your ISP.


If you are satisfied with the settings, click the *Apply* button to save and apply the changes.

if you need to change the following:

- Advanced SIP settings
- Forwarding settings
- Fax option settings
- P-Asserted-Identity setting
- Dial termination key setting
- Session timer settings
- MWI (message waiting indication) settings
- DND (do not disturb) settings
- Block anonymous call setting
- Codec settings

Go to section "Advanced Edit SIP Settings window" on page 68. Otherwise, SIP settings are complete.

### *Advanced Edit SIP Settings window*

To display the Advanced Edit SIP Settings window, do the following:

1. Click *SIP* under the *VoIP* submenu.
2. Click the *Modify* button in the *SIP Setting* section of the *SIP Settings* window.


3. Click the *Advanced* button to display the Advanced Edit SIP Settings window.


The Advanced Edit SIP Settings window (see figure 54 on page 81) displays the following;

- Advanced SIP settings (see section "SIP Advanced" on page 82)
- Forwarding settings (see section "Forward Setting" on page 83)
- Fax settings (see section "Fax Option" on page 83)
- P-Asserted-Identity setting (see section "P-Asserted-Identity" on page 83)
- Dial termination key setting (see section "Dial Termination Key" on page 84)
- Session timer settings (see section "Session Timer" on page 84)
- Message waiting indication (MWI) settings (see section "MWI (Message Waiting Indication)" on page 84)
- Do not disturb (DND) settings (see section "DND (Do Not Disturb)" on page 85)

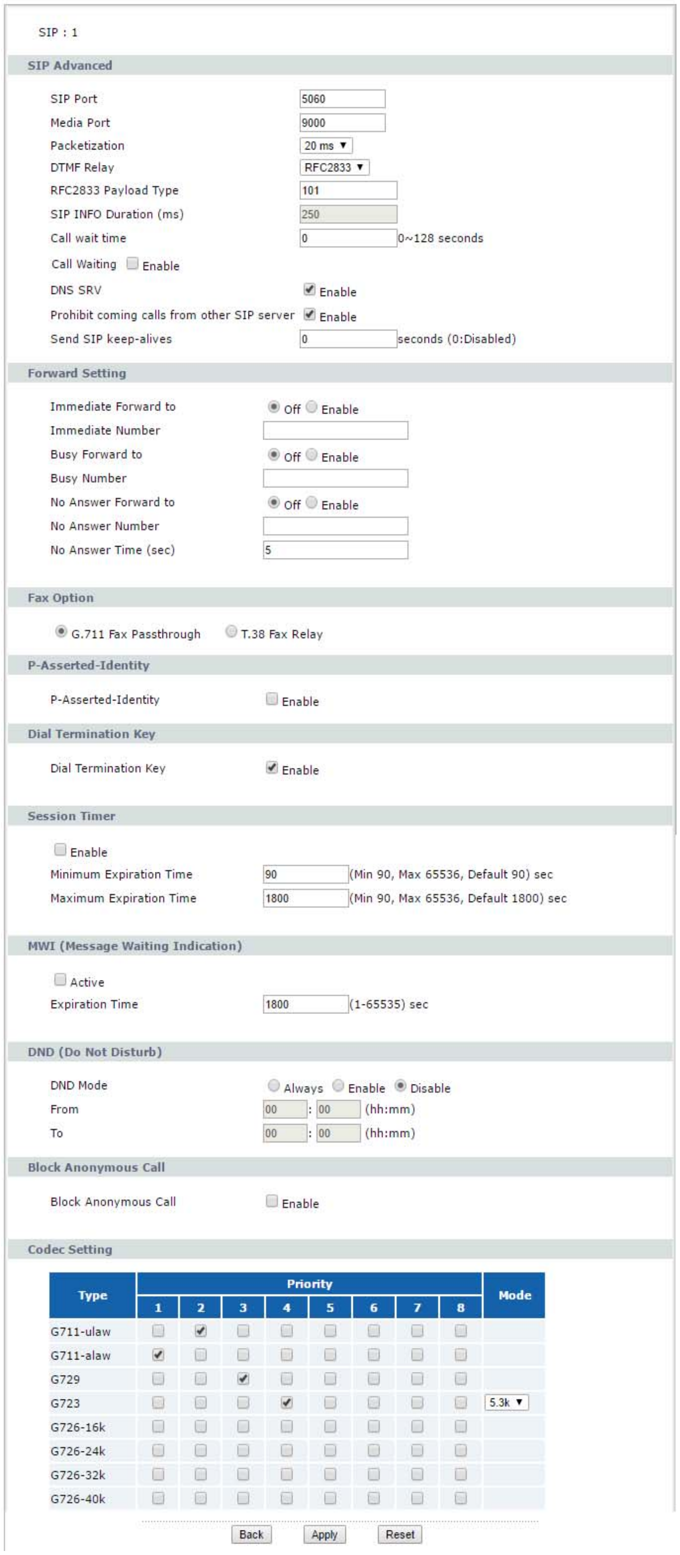
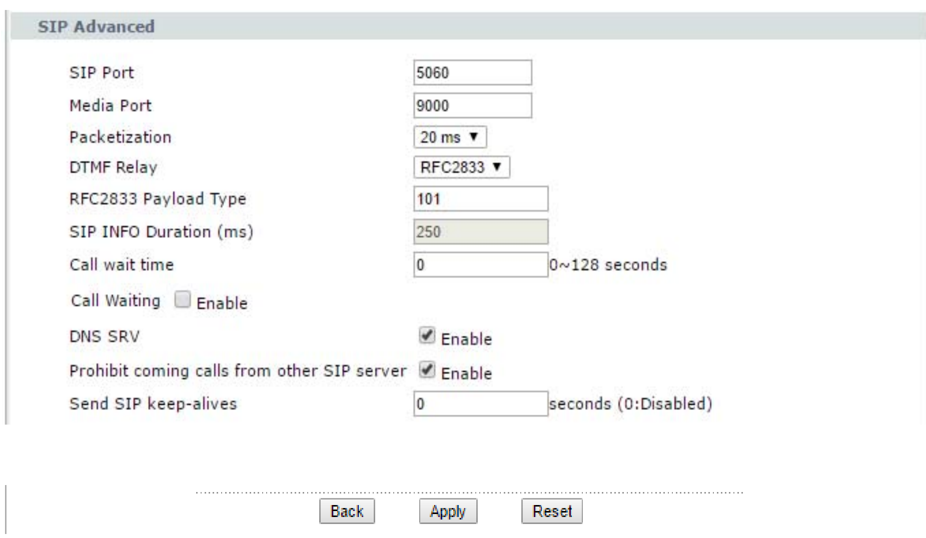



Figure 54. Advanced Edit SIP Settings window

- Block Anonymous Call setting (see section “Block Anonymous Call” on page 85)
- Codec settings (see section “Codec Setting” on page 85)

SIP Advanced. This section displays the following;

- SIP Port Number: Assign the SIP port number of the ATA. The default setting is 5060.
- Media Port: Port number for sending RTP packets. The default setting is 9000.
- Packetization: Select the voice codec packetization interval in milliseconds. The default is 30 ms (for terrestrial connection it is 20ms). This is used to minimize loss that happens during transmission of voice data over the network.
- DTMF Relay: Click the drop-down menu to select RFC2833, SIP INFO, or PCM as indicated by your ISP.
- RFC2833 Payload Type: Sending the DTMF tone as an RTP payload signal.
- SIP INFO Duration (ms): Modify SIP INFO duration time.



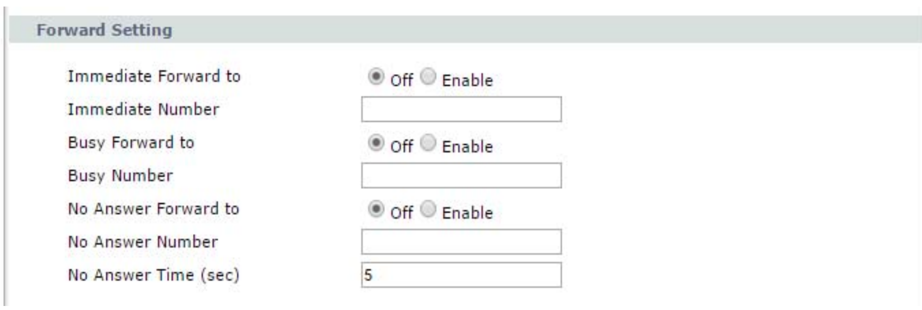

- Call wait time (seconds)
- Call Waiting: Click the check box to enable call waiting if the user wants to be informed when there is a new incoming call while he or she is on another call. Disable the feature if the user doesn’t want to be informed of a new incoming call.
- DNS SRV: Click the check box to enable having the ATA query your ISP’s DNS server for a list of available SIP servers that it maintains.
- Prohibit coming calls from other SIP server: Enable or disable prohibiting incoming calls from other SIP servers.
- Send SIP keep-alives (seconds)

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

Forward Setting. You can set up the phone number you want to forward in this section. There are three forwarding modes:

- Immediate Forward: All incoming call will forward to the number you entered. You can input the phone number. If you select this function, then all the incoming call will direct forward to the speed dial number you entered.
- Busy Forward: If you are on the phone, the new incoming call will forward to the number you entered.
- No Answer Forward: If nobody picks up the phone, the incoming call will forward to the number you entered.

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.





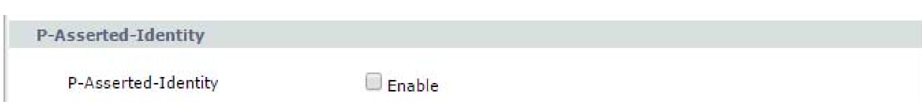
Fax Option. G.711 Fax Pass through: Select this if the SN100 should use G.711 to send fax messages. The peer devices must also use G.711.

T.38 Fax Relay: Select this if the SN100 should send fax messages as UDP packets through IP networks. This provides better quality, but it may have interoperability problems. The peer devices must also use T.38.


When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


P-Asserted-Identity. The SIP server authenticates the user that sent the SIP request and then uses the identity from the result of authentication to generate a P-Asserted-Identity header field.

Select enable to have the device look at the P-Asserted-Identity header field which carries the identity of the caller in SIP message, and display the caller's location if available.



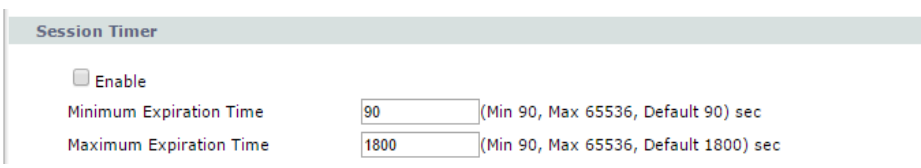
When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Dial Termination Key. Select enable if you want to use the pound (#) key to tell device to make the phone call immediately.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Session Timer. Select Enable if you want to define how long the ATA waits to receive a session-alive packet for a voice session from the SIP server.

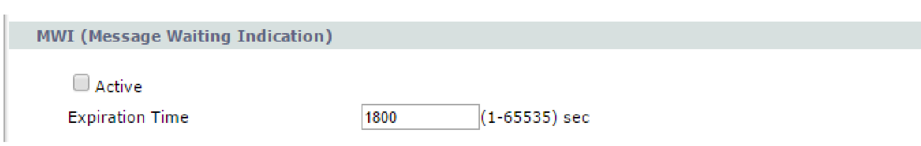
Minimum Expiration Time: Enter the minimum time the ATA waits for a session-alive packet (90-65536 seconds). If a session-alive packet is not received during this time, the voice session is terminated.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


MWI (Message Waiting Indication). Active: Select this if you want to hear a waiting (beeping) dial tone on your phone when you have at least one voice message. Your SIP service provider must support this feature.

Expiration Time: Keep the default value, unless your SIP service provider tells you to change it. Enter the number of seconds the SIP server should provide the message waiting service each time the ATA subscribes to the service. Before this time passes, the ATA automatically subscribes again.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

DND (Do Not Disturb). You can set up the DND setting to keep the device silence. You can choose from the following:

- Always: All incoming call will be blocked until this feature is disabled by clicking on *Disable*.
- Enable: The ATA will be blocked during the time period. If the "From" time is large than the "To" time, the block time will from 00:00 to 23:59

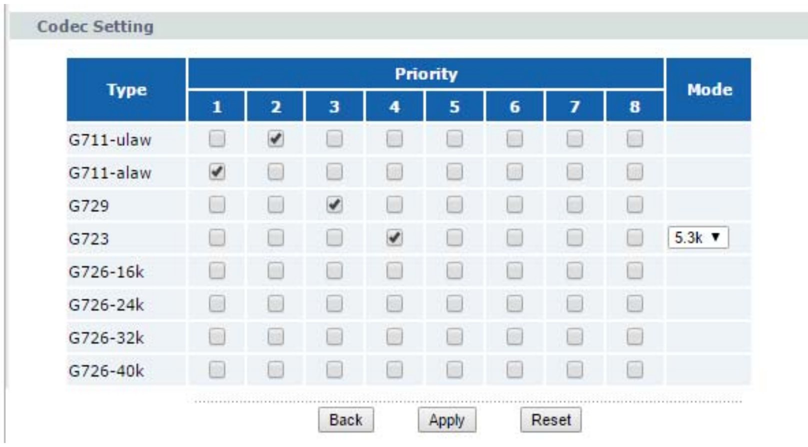

When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Block Anonymous Call. If this feature is enabled, calls that have caller ID information appearing as "private" or "anonymous" will be blocked.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Codec Setting. A codec is an algorithm for taking voice or video and compressing the information. This type of codec combines analog-to-digital conversion and digital-to-analog conversion functions in a single chip. The codec is used to compress the voice signal into data packets. Each codec has different bandwidth requirement. There are the following types of codecs:

- G.711/Ulaw
- G.711/Alaw
- G.729 (default and standard for calls through a satellite connection)
- G.723 (5.3k/6.3k bps)
- G.726(16K bps)
- G.726(24K bps)
- G.726(32K bps)
- G.726(40K bps)

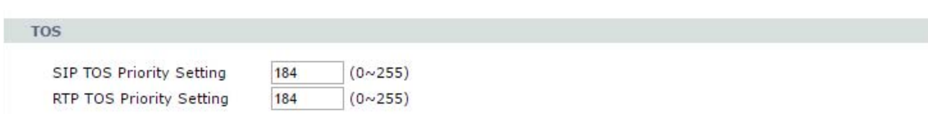



When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

### *QoS*

Quality of service (QoS) refers to traffic prioritization and resource reservation control mechanisms; it is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

ToS. ToS ensures that when voice and data are being transmitted at the same time, the data will not degrade the voice signal quality. When type of service (ToS) bits are enabled, voice will have the first priority pass through ToS enabled devices.




VLAN Tagging. You can set the VLAN setting in this section.


VLAN tags: if you enable *VOICE VLAN Tags*, and set the *Voice VLAN ID* and *Priority*, then all the incoming packets will be tagged with the IP address and the VLAN ID (VID):

- Voice VLAN ID: Use the voice VLAN ID provided by your ISP to set the *Voice VLAN ID*.
- Data VLAN ID: Use the data VLAN ID provided by your ISP to set the *Data VLAN ID*.
- Priority: Defines user priority, giving eight ($2^3$) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be provided by your ISP.

## Configuring SN102/1JS1JO/E (1 FXS/1 FXO) ATA SIP settings

Do the following to display SIP and QoS settings:

1. Click on the plus button next to *VoIP* to display the sub-categories (see figure 55).

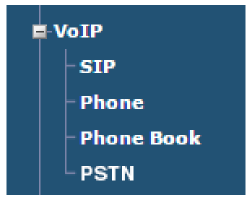



Figure 55. SN102/1JS1JO/E VoIP sub-categories

2. Click *SIP*. The *SIP Settings* window displays (see figure 56).

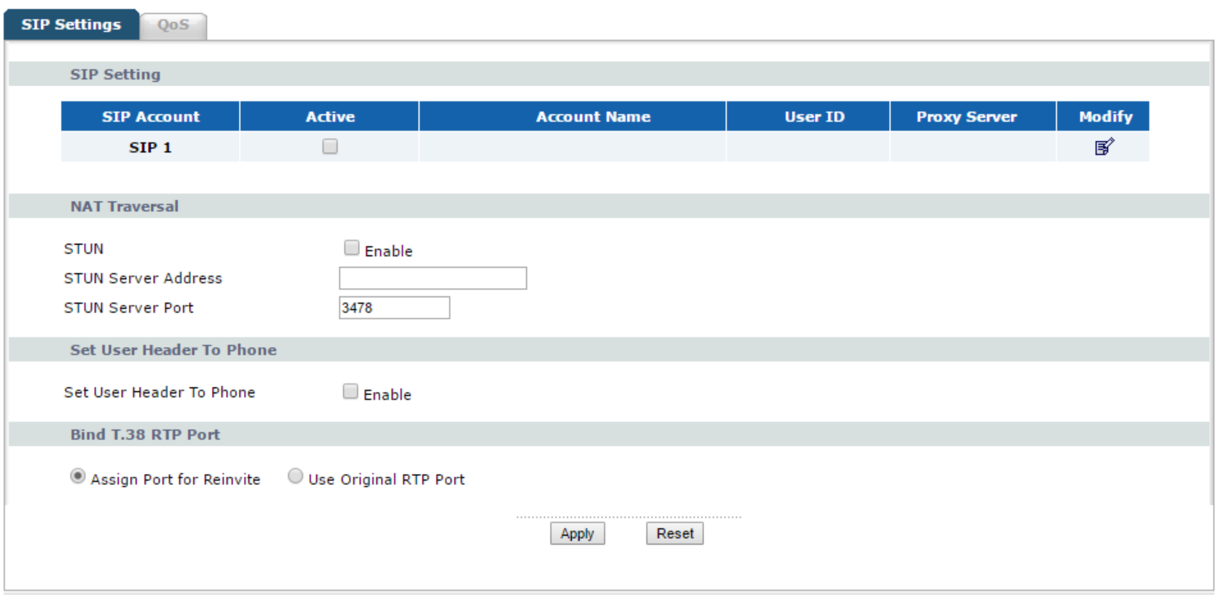



Figure 56. SN102/1JS1JO/E SIP Settings window

3. Click the *QoS* tab to display the QoS window (see figure 57 on page 88).

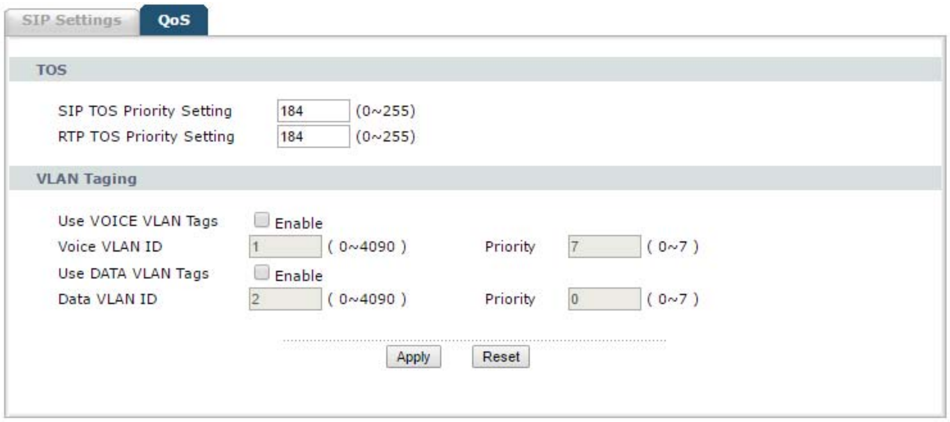



Figure 57. QoS window

Go to the appropriate section to modify SIP or QoS settings:

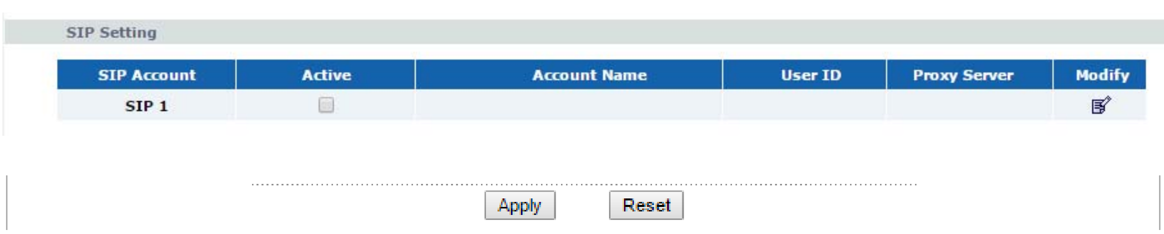
- “SIP Settings”
- “QoS” on page 98

### *SIP Settings*

SIP is a request-response protocol that deals with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media, and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

The *SIP Settings* window displays the following:

- SIP account information (whether the account is active, the account name, user ID, and proxy server address). SIP settings can be modified by clicking on the *Modify* button for the appropriate account (see section “Edit SIP Settings window” on page 90 for more information).



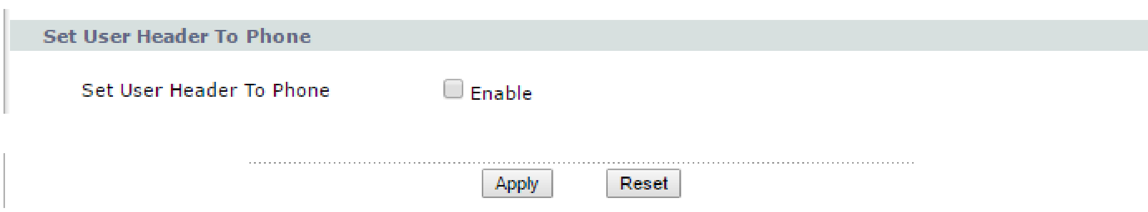

**Note** Click the *Active* check box to activate or deactivate a SIP account. Click the *Apply* button to save and apply the change.

**Note** Once you are satisfied with the settings for a SIP account, if the ATA has another SIP account, you can click on the *CopyToSIP2* button to copy the settings to the other account. Click the *Apply* button to save and apply the change.

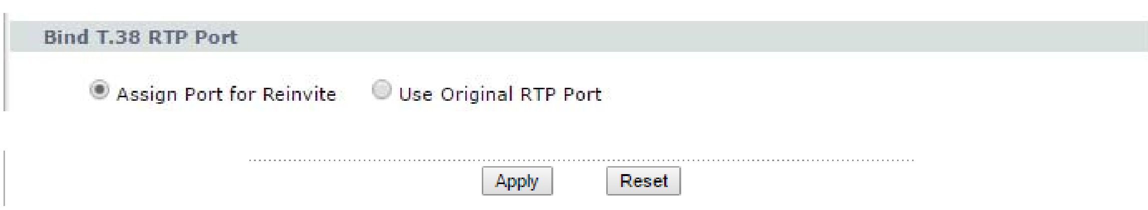
- NAT traversal information (whether STUN is enabled, the STUN server address, and the STUN server port identifier). STUN {Simple Traversal of UDP through NATs (Network Address Translation)} is a protocol for assisting devices behind a NAT firewall or router with their packet routing.


**Note** Click the *Enable* check box to activate or deactivate STUN. Click the *Apply* button to save and apply the change.

- *Set User Header to Phone* information. Click the check box to enable *Set User Header to Phone* to have the device add a “user=phone” parameter to the SIP URI before sending a SIP INVITE request. This parameter indicates the user part of the SIP URI (<user>@<sipserver>) the device is calling is a valid telephone number.(user name and password). Click the *Apply* button to save and apply the change




- *Bind T.38 RTP Port* information. If you need to send fax messages using T.38 and want to always use the same port number for transmitted and received T.38 packets, click the radio button to select *Use Original RTP Port*. Otherwise, select *Assign Port for Reinvite* to have the device chooses a port randomly for T.38 after the re-invite process. Click the *Apply* button to save and apply the change



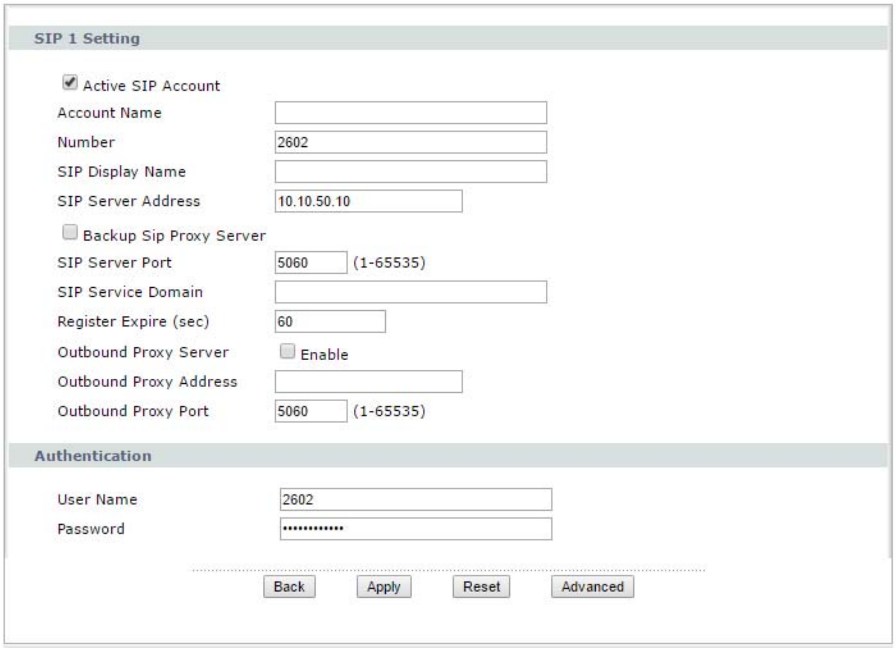


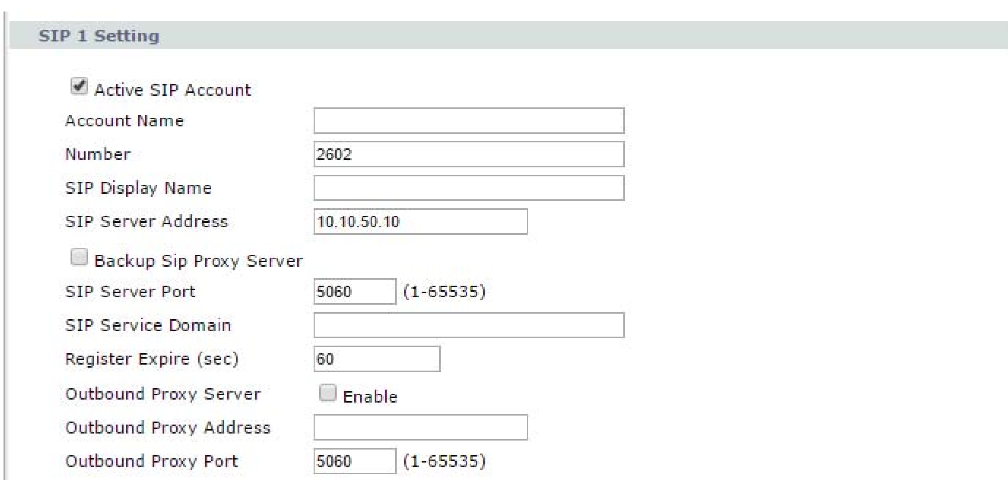

Figure 58. Edit SIP Settings window

Edit SIP Settings window. The Edit SIP Settings window (see figure 58) displays the following sections;

- SIP settings
- Authentication settings

**SIP Settings**. This section displays the following:

- Account Name: Type can the name you want displayed.
- Number: Type the number provided by your ISP.




- SIP Display Name: Type the name you want displayed.
- SIP Server Address: Type the address.

- Backup SIP Proxy Server: Click to enable or disable the backup proxy server
- Backup1 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- Backup2 Server Address: This setting appears if Backup SIP Proxy Server is enabled. Type the address.
- SIP Server Port: Type the Proxy port number provided by your ISP.
- SIP Server Domain: Type the SIP domain provided by your ISP.
- Register Expire (sec): SIP registration expired time (default is 600 and needed for calls through the satellite connection).
- Outbound Proxy Server: Click to enable or disable the outbound proxy server
- Outbound Proxy Address: Type the address provided by your ISP. If your ISP did not provide the information, you can skip this item.
- Outbound Proxy Port: Type the port number provided by your ISP.

**Authentication Settings**. This section displays the following:

- User Name: Type the Register Name provided by your ISP.
- Password: Type the Register Password provided by your ISP.


If you are satisfied with the settings, click the *Apply* button to save and apply the changes.

if you need to change the following:

- Advanced SIP settings
- Forwarding settings
- Fax option settings
- P-Asserted-Identity setting
- Dial termination key setting
- Session timer settings
- MWI (message waiting indication) settings
- DND (do not disturb) settings
- Block anonymous call setting
- Codec settings

Go to section "Advanced Edit SIP Settings window" on page 92. Otherwise, SIP settings are complete.

### *Advanced Edit SIP Settings window*

To display the Advanced Edit SIP Settings window, do the following:

1. Click *SIP* under the *VoIP* submenu.
2. Click the *Modify* button in the *SIP Setting* section of the *SIP Settings* window.


3. Click the *Advanced* button to display the Advanced Edit SIP Settings window.


The Advanced Edit SIP Settings window (see figure 49 on page 69) displays the following;

- Advanced SIP settings (see section "SIP Advanced" on page 94)
- Forwarding settings (see section "Forward Setting" on page 94)
- Fax settings (see section "Fax Option" on page 95)
- P-Asserted-Identity setting (see section "P-Asserted-Identity" on page 95)
- Dial termination key setting (see section "Dial Termination Key" on page 95)
- Session timer settings (see section "Session Timer" on page 96)
- Message waiting indication (MWI) settings (see section "MWI (Message Waiting Indication)" on page 96)
- Do not disturb (DND) settings (see section "DND (Do Not Disturb)" on page 96)
- Block Anonymous Call setting (see section "Block Anonymous Call" on page 97)
- Codec settings (see section "Codec Setting" on page 97)

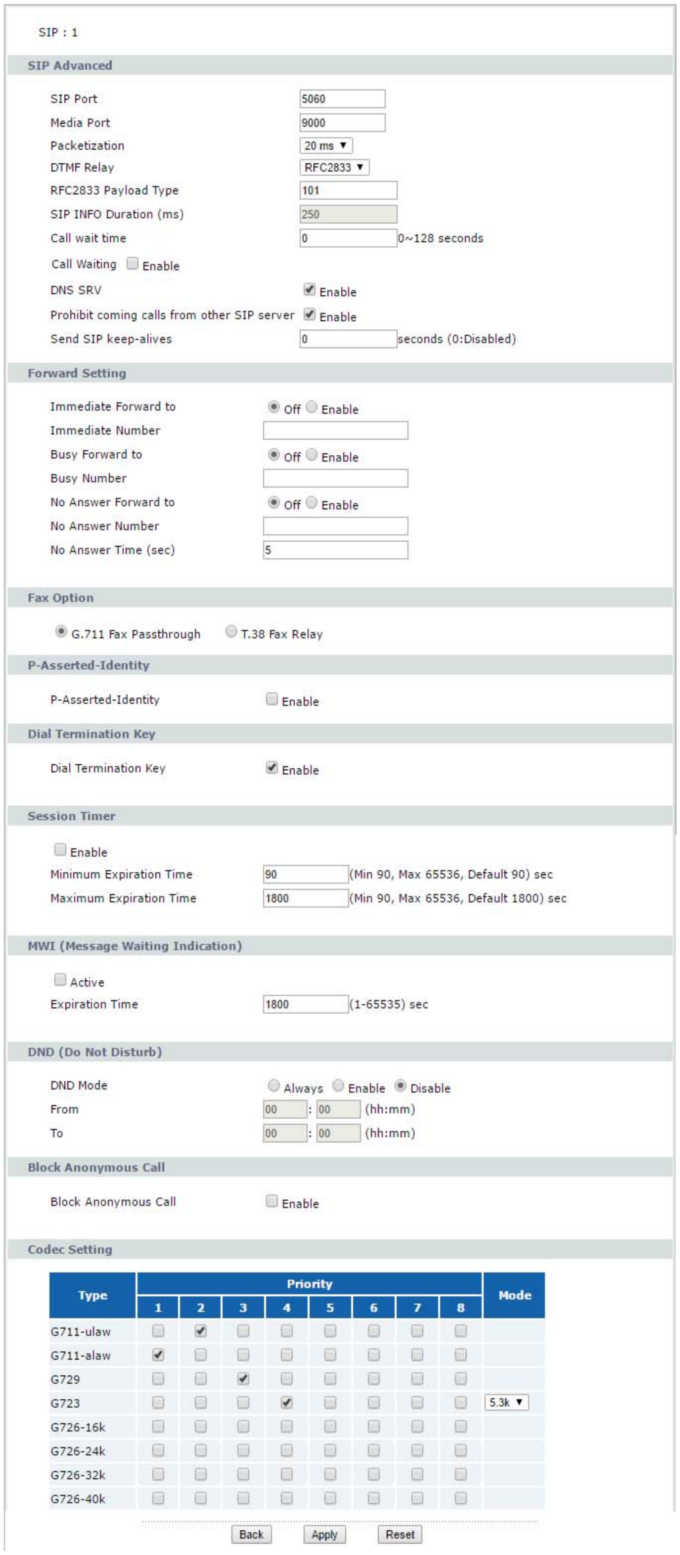
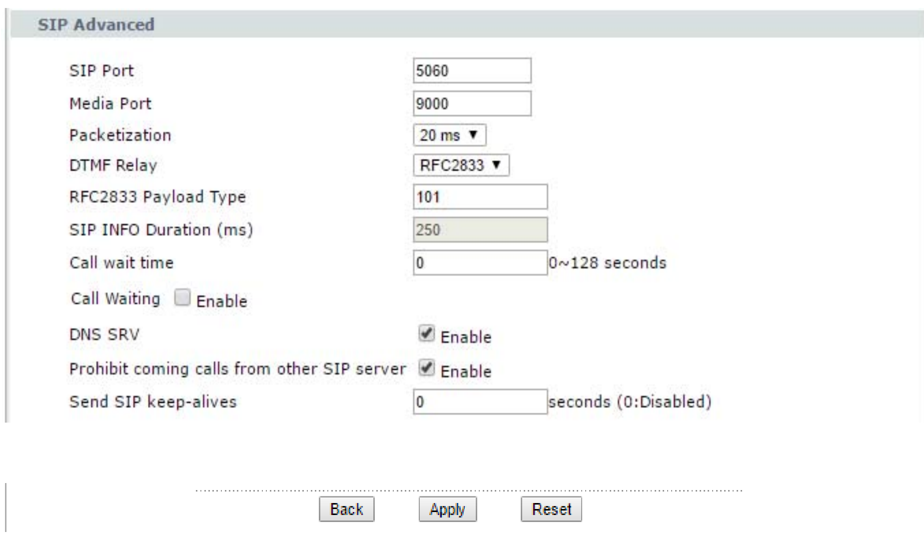



Figure 59. Advanced Edit SIP Settings window

SIP Advanced. This section displays the following;

- SIP Port Number: Assign the SIP port number of the ATA. The default setting is 5060.
- Media Port: Port number for sending RTP packets. The default setting is 9000.
- Packetization: Select the voice codec packetization interval in milliseconds. The default is 30 ms (for terrestrial connection it is 20ms). This is used to minimize loss that happens during transmission of voice data over the network.
- DTMF Relay: Click the drop-down menu to select RFC2833, SIP INFO, or PCM as indicated by your ISP.
- RFC2833 Payload Type: Sending the DTMF tone as an RTP payload signal.
- SIP INFO Duration (ms): Modify SIP INFO duration time.




- Call wait time (seconds)
- Call Waiting: Click the check box to enable call waiting if the user wants to be informed when there is a new incoming call while he or she is on another call. Disable the feature if the user doesn't want to be informed of a new incoming call.
- DNS SRV: Click the check box to enable having the ATA query your ISP's DNS server for a list of available SIP servers that it maintains.
- Prohibit coming calls from other SIP server: Enable or disable prohibiting incoming calls from other SIP servers.
- Send SIP keep-alives (seconds)

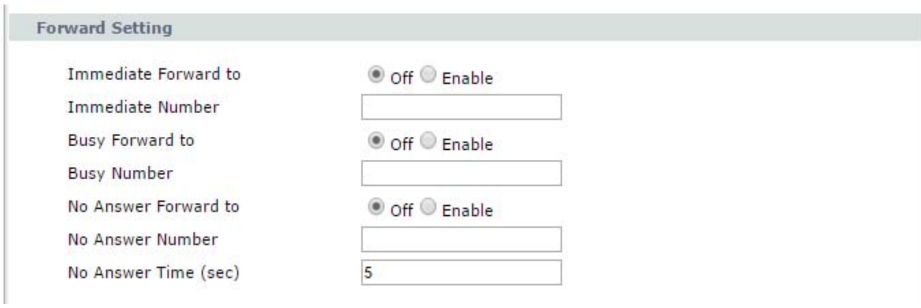
When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

Forward Setting. You can set up the phone number you want to forward in this section. There are three forwarding modes:

- Immediate Forward: All incoming call will forward to the number you entered. You can input the phone number. If you select this function, then all the incoming call will direct forward to the speed dial number you entered.

- Busy Forward: If you are on the phone, the new incoming call will forward to the number you entered.
- No Answer Forward: If nobody picks up the phone, the incoming call will forward to the number you entered.

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.





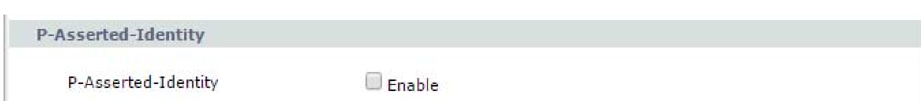
**Fax Option.** G.711 Fax Pass through: Select this if the SN100 should use G.711 to send fax messages. The peer devices must also use G.711.

T.38 Fax Relay: Select this if the SN100 should send fax messages as UDP packets through IP networks. This provides better quality, but it may have interoperability problems. The peer devices must also use T.38.


When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


**P-Asserted-Identity.** The SIP server authenticates the user that sent the SIP request and then uses the identity from the result of authentication to generate a P-Asserted-Identity header field.

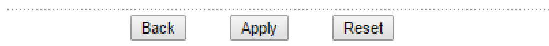
Select enable to have the device look at the P-Asserted-Identity header field which carries the identity of the caller in SIP message, and display the caller's location if available.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

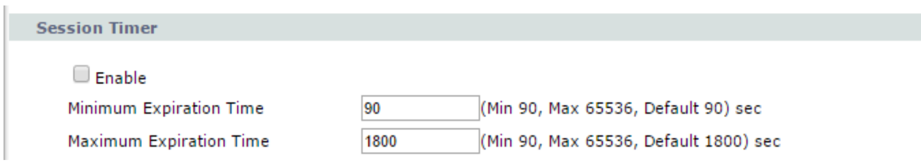

**Dial Termination Key.** Select enable if you want to use the pound (#) key to tell device to make the phone call immediately.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.




**Session Timer.** Select Enable if you want to define how long the ATA waits to receive a session-alive packet for a voice session from the SIP server.

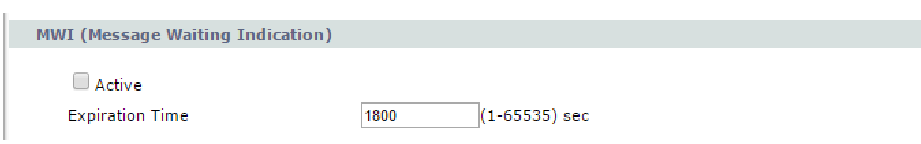
Minimum Expiration Time: Enter the minimum time the ATA waits for a session-alive packet (90-65536 seconds). If a session-alive packet is not received during this time, the voice session is terminated.



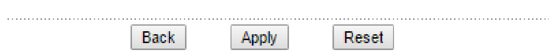

When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**MWI (Message Waiting Indication).** Active: Select this if you want to hear a waiting (beeping) dial tone on your phone when you have at least one voice message. Your SIP service provider must support this feature.

Expiration Time: Keep the default value, unless your SIP service provider tells you to change it. Enter the number of seconds the SIP server should provide the message waiting service each time the ATA subscribes to the service. Before this time passes, the ATA automatically subscribes again.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

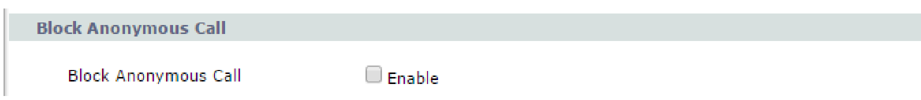



**DND (Do Not Disturb).** You can set up the DND setting to keep the device silence. You can choose from the following:

- Always: All incoming call will be blocked until this feature is disabled by clicking on *Disable*.
- Enable: The ATA will be blocked during the time period. If the “From” time is large than the “To” time, the block time will from 00:00 to 23:59


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

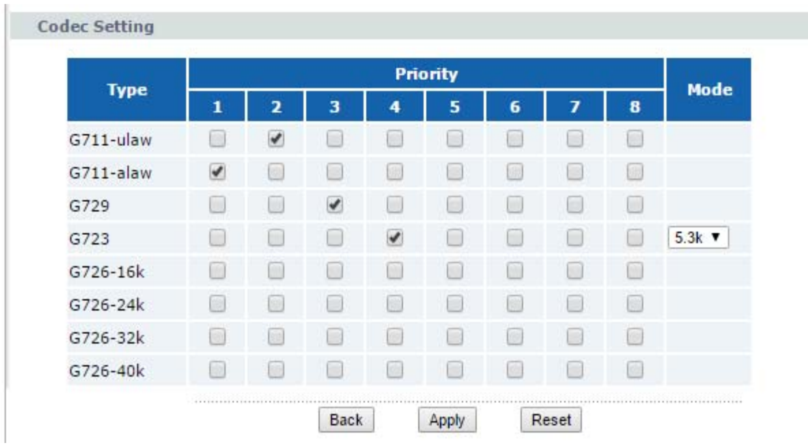

Block Anonymous Call. If this feature is enabled, calls that have caller ID information appearing as "private" or "anonymous" will be blocked.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


Codec Setting. A codec is an algorithm for taking voice or video and compressing the information. This type of codec combines analog-to-digital conversion and digital-to-analog conversion functions in a single chip. The codec is used to compress the voice signal into data packets. Each codec has different bandwidth requirement. There are the following types of codecs:

- G.711/Ulaw
- G.711/Alaw
- G.729 (default and standard for calls through a satellite connection)
- G.723 (5.3k/6.3k bps)
- G.726(16K bps)
- G.726(24K bps)
- G.726(32K bps)
- G.726(40K bps)

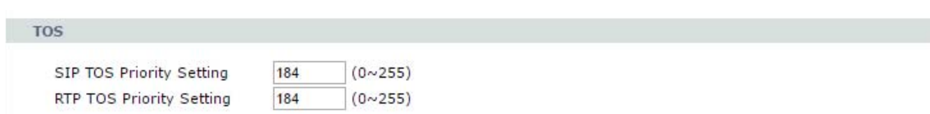



When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

### *QoS*

Quality of service (QoS) refers to traffic prioritization and resource reservation control mechanisms; it is the ability to provide different priority to different applications, users, or data flows, or to guarantee a certain level of performance to a data flow.

**ToS.** ToS ensures that when voice and data are being transmitted at the same time, the data will not degrade the voice signal quality. When type of service (ToS) bits are enabled, voice will have the first priority pass through ToS enabled devices.




**VLAN Tagging.** You can set the VLAN setting in this section.


VLAN tags: if you enable *VOICE VLAN Tags*, and set the *Voice VLAN ID* and *Priority*, then all the incoming packets will be tagged with the IP address and the VLAN ID (VID):

- Voice VLAN ID: Use the voice VLAN ID provided by your ISP to set the *Voice VLAN ID*.
- Data VLAN ID: Use the data VLAN ID provided by your ISP to set the *Data VLAN ID*.
- Priority: Defines user priority, giving eight ($2^3$) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be provided by your ISP.

## Phone

Do the following to display the *Analog Phone* window:

1. Click on the plus button next to *VoIP* to display the *Phone* sub-category (see figure 60).

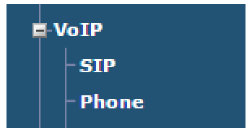



Figure 60. Phone sub-category

2. Click *Phone*. The *Analog Phone* window displays (see figure 61).

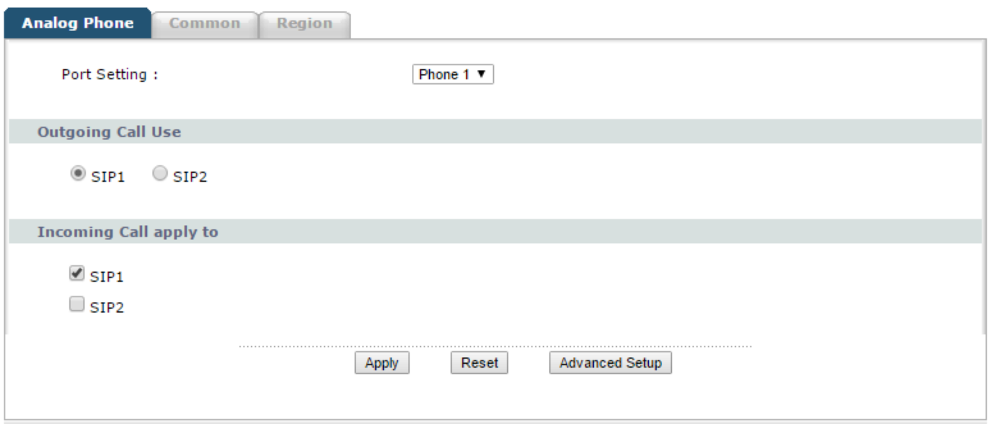



Figure 61. Analog Phone window

3. Click the *Common* tab to display the *Common* window (see figure 62).

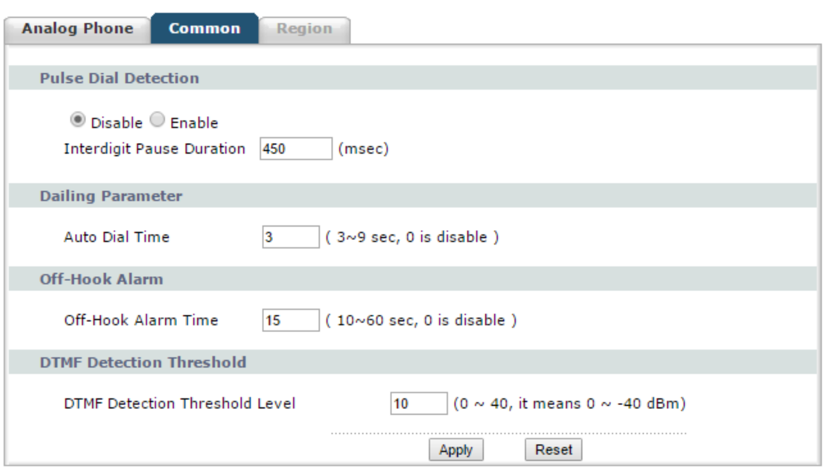



Figure 62. Common window

4. Click the *Region* tab to display the *Region* window (see figure 63 on page 100).

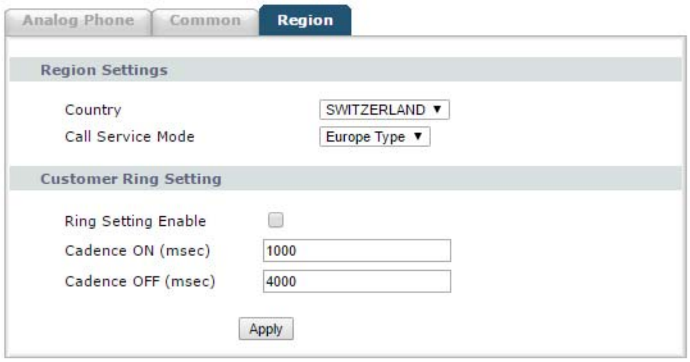



Figure 63. Region window

Go to the appropriate section to modify analog phone, common, or region settings:

- ““Region window” on page 106”
- “Common window” on page 105
- “Region window” on page 106

## *Analog Phone window*

1. If your SN100 ATA has only one FXS port, go to step 2. Otherwise, select *Phone 1* or *Phone 2* from the Port Setting drop-down menu.


2. Click the *Advance Setup* button. The Advanced Setup window appears (see figure 64 on page 101).

### *Phone Advanced Setup window*

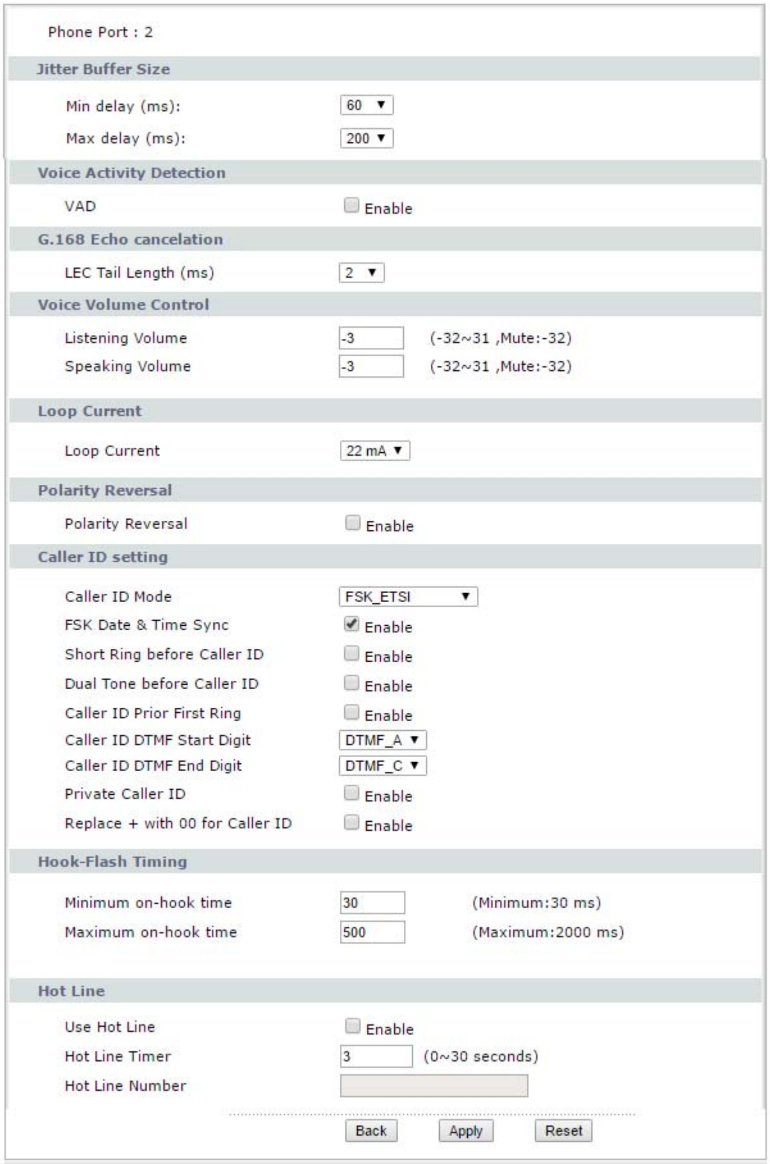



Figure 64. Phone Advanced Setup window

The Phone Advanced Setup window displays the following;

- Jitter Buffer Size settings (see section “Jitter Buffer Control settings” on page 102)
- Voice Activity Detection settings (see section “Voice Activity Detection settings” on page 102)
- G.168 Echo cancellation settings (see section “G.168 Echo cancellation settings” on page 102)
- Voice Volume Control settings (see section “Voice Volume Control settings” on page 103)
- Loop Current settings (see section “Loop Current settings” on page 103)

- Polarity Reversal settings (see section “Polarity Reversal settings” on page 103)
- Caller ID settings (see section “Caller ID settings” on page 104)
- Hook-Flash Timing settings (see section “Hook-Flash Timing settings” on page 104)
- Hot Line settings (see section “Hot Line settings” on page 105)

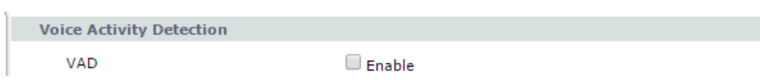
**Jitter Buffer Control settings.** Changes in packets arrival time, called jitter, can occur because of network congestion, timing drift, or route changes. The jitter buffer control is a shared data area where voice packets can be stored, collected, and sent to the voice shared buffer in evenly spaced intervals.

- Min delay (ms): Select minimum delay buffer time from the drop-down menu
- Max delay (ms): Select maximum delay buffer time from the drop-down menu


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**Voice Activity Detection settings.** If this function is enabled, if silence occurs for a period of time, no data will be sent across the network during that period in order to save bandwidth.

Click the check box to enable voice activity detection (VAD).




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

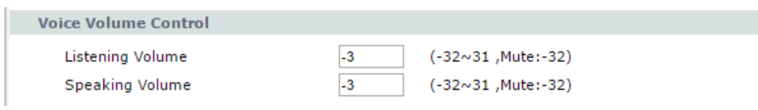

**G.168 Echo cancellation settings.** Select the LEC tail length (ms) from the drop-down menu.


When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

**Voice Volume Control settings.** Voice volume control sets the intensity in dB for receiving sound (listening), and for transmitting sound (speaking):

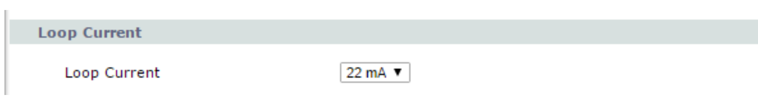
- Listening Volume: Sets a specific sound intensity for receiving sound. Type a value from -32 to 31 dB
- Speaking Volume: Sets a specific sound intensity for transmitting sound. Type a value from -32 to 31 dB

**Note** A value of -32 will mute the volume.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.

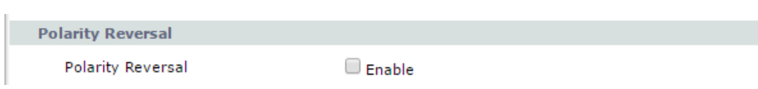

**Loop Current settings.** Select the loop current from the drop-down menu.



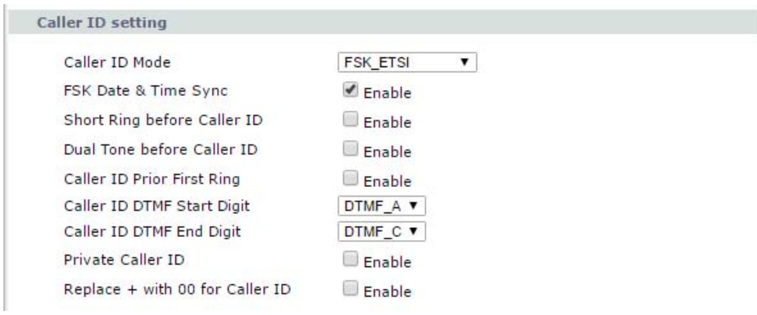

When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


**Polarity Reversal settings.** A function that changes the polarity on the current on a FXS phone line to notify billing software or other applications of events like such as a call being answered.

Click the check box to enable polarity reversal.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.




Caller ID settings. The following settings are available:

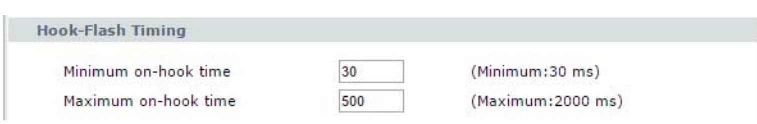
- Caller ID Mode: Click the drop-down menu to select the caller ID mode.
- FSK Date & Time Sync: Click the check box to enable sending FSK date and time to the display device.
- Reverse Polarity before Caller ID: Click the check box to enable sending reverse polarity before caller ID
- Short Ring before Caller ID: Click the check box to enable sending a short ring before caller ID.
- Dual Tone before Caller ID: Click the check box to enable sending dual tone before caller ID.
- Caller ID Prior First Ring: Click the check box to enable sending caller ID before first ring.
- Caller ID DTMF Start Digit: Click the drop-down menu to set the caller ID DTMF start digit.
- Caller ID DTMF END Digit: Click the drop-down menu to set the caller ID DTMF end digit.
- Private Caller ID: A private caller has hidden or otherwise obfuscated their phone number. Often their number displays as "Private Caller", "Restricted", "Unknown Caller ID" or "No Caller ID". Click the check box to enable private caller ID.

When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

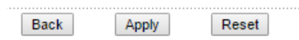

Hook-Flash Timing settings. Hook-flash is a button on a telephone that simulates quickly hanging up then picking up again (a quick off-hook/on-hook/off-hook cycle). This action can signal the telephone exchange to do something. A common use of hook flash is to switch to another incoming call with the call waiting service. Another use is to indicate a request for voice conferencing, for example, a user may use a procedure like the following to initiate three-way calling.

The following settings are available:

- Minimum on-hook time: Type the value (default is 30 ms)
- Maximum on-hook time: Type the value (default is 2000 ms)



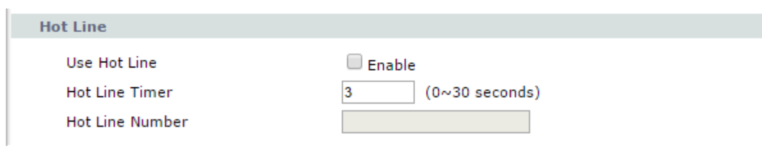
When you are satisfied with the setting, click the *Apply* button to save and apply the changes.




Hot Line settings. This service enables you to call a pre-programmed number simply by lifting the handset.

The following settings are available:

- Use Hot Line: Click the check box to enable the hot line service.
- Hot Line Timer: This feature is typically used for setting up a hot line or warm line, with a 0-second timer delay for the hot line and a non-zero delay for a warm line. Type the value in seconds.
- Hot Line Number: Input the number for the hot line service—This number will be called immediately when the telephone goes off-hook.




When you are satisfied with the setting, click the *Apply* button to save and apply the changes.


## Common window

The Common window displays the following;

- Pulse Dial Detection settings (see section “Pulse Dial Detection settings” on page 105)
- Off-Hook Alarm settings (see section “Off-Hook Alarm settings” on page 106)
- DTMF Detection Threshold settings (see section “DTMF Detection Threshold settings” on page 106)

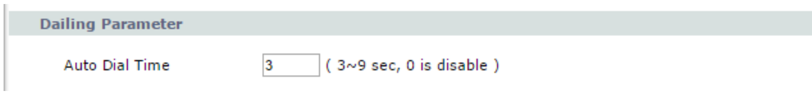
### *Pulse Dial Detection settings*

1. Click to enable pulse dial detection if you are connecting a pulse (rotary) dialing telephone to the SN100.


2. Type the Interdigit Pause Duration value.
3. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

### *Dialing Parameter settings*

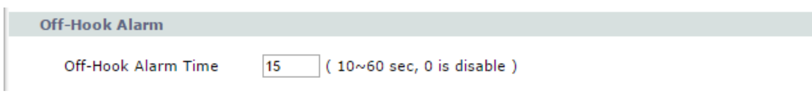
If no other number is being dialed within this interval, the SN100 will terminate the call. Assign the time interval from 1 to 9 seconds.




When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### *Off-Hook Alarm settings*

The amount of time the phone can be off-hook before an alarm sounds.




When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### *DTMF Detection Threshold settings*

Type the DTMF Detection Threshold Level value.


When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## *Region window*

The Region window displays the following;

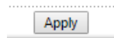
- Region settings (see section "Region settings" on page 106)
- Customer Ring settings (see section "Customer Ring settings" on page 107)

### *Region settings*

Adjust the tone frequency according to each country

1. Select the country from the drop-down menu

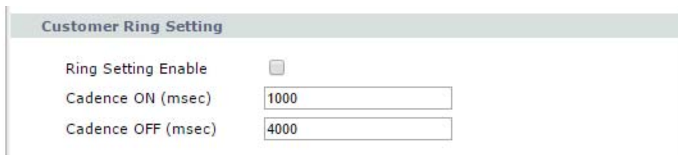
2. Select the call service mode from the drop-down menu
3. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.



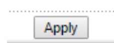

### *Customer Ring settings*

Create custom ring cadences.

1. Click to enable ring setting.




2. Type the Cadence ON (length of time the tone will sound) value.
3. Type the Cadence OFF (length of time the tone is silent) value.
4. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.




## Phone Book

Do the following to display the *Speed Dial* window:

1. Click on the plus button next to *VoIP* to display the *Phone* sub-category (see figure 65).

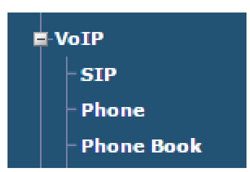



Figure 65. Phone Book sub-category

2. Click *Phone Book*. The *Speed Dial* window displays (see figure 66 on page 108).

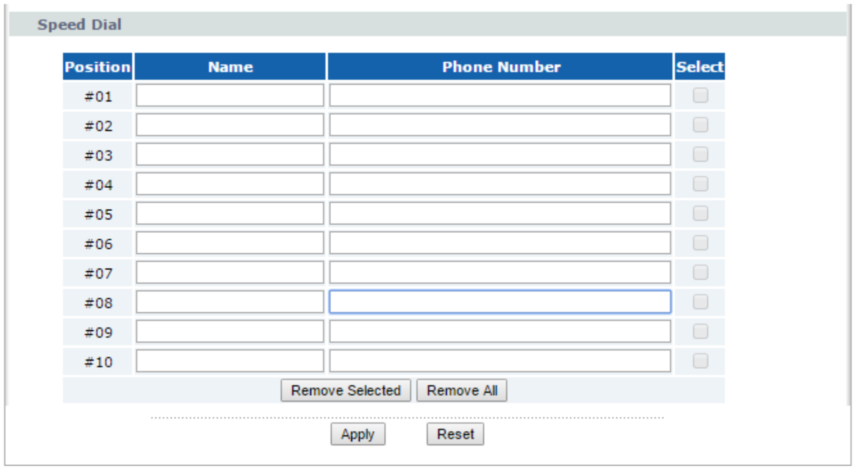



Figure 66. Speed Dial window

3. Click the *Dial Plan with URL* tab to display the *Dial Plan with URL* window (see figure 67).

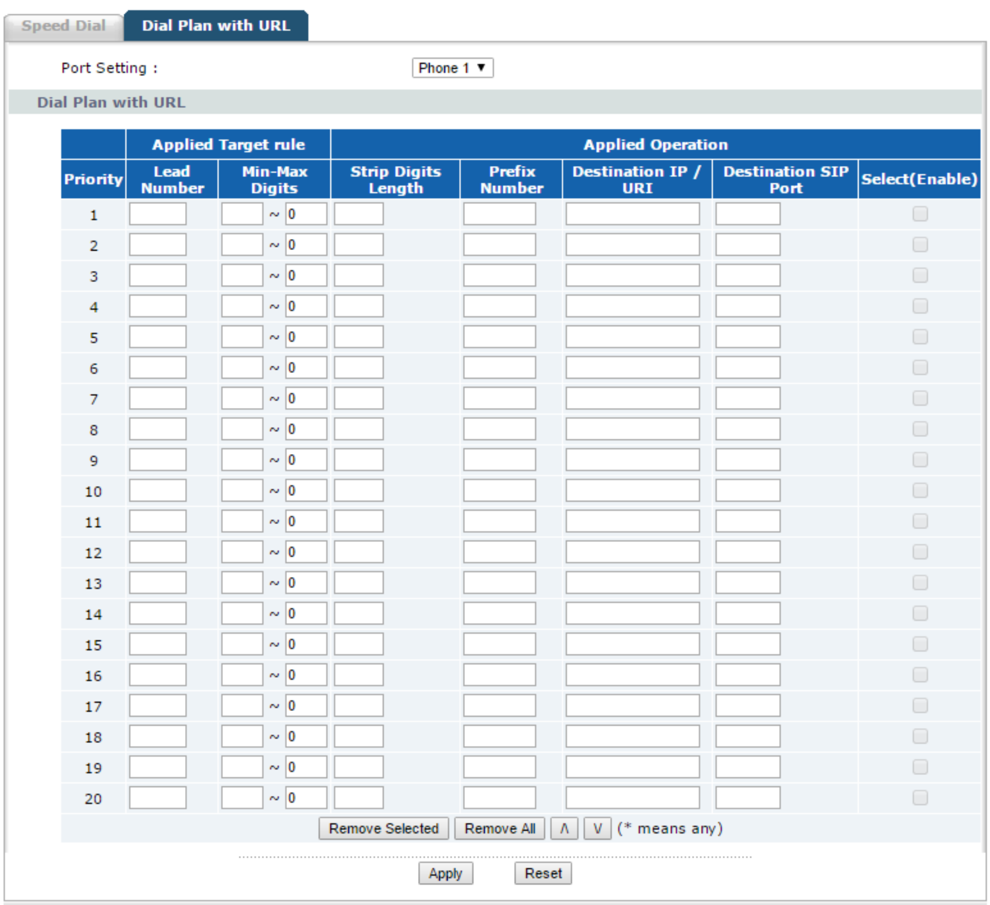



Figure 67. Dial Plan with URL window

Go to the appropriate section to modify analog phone, common, or region settings:

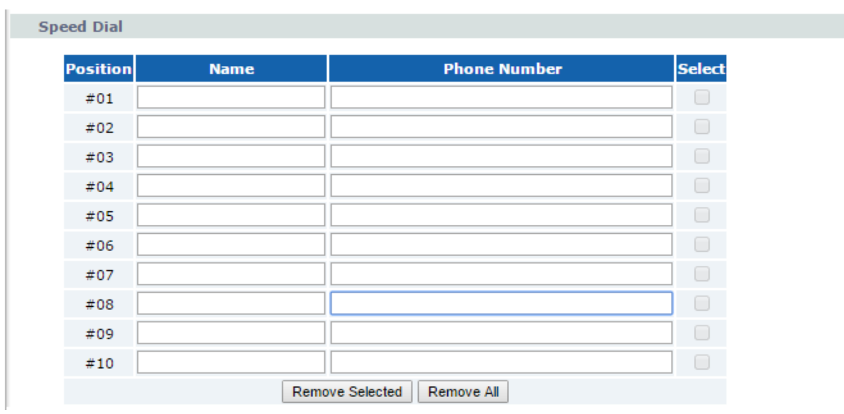
- "Speed Dial window"
- "Dial Plan with URL window" on page 109

### Speed Dial window

Speed Dial lets you define a button or a set of buttons to link to numbers defined in the Speed Dial list.

1. If your SN100 ATA has only one FXS port, go to step 2. Otherwise, select *Phone 1* or *Phone 2* from the Port Setting drop-down menu.


2. Position: Decide which speed dial shortcut to use from 01 to 10.

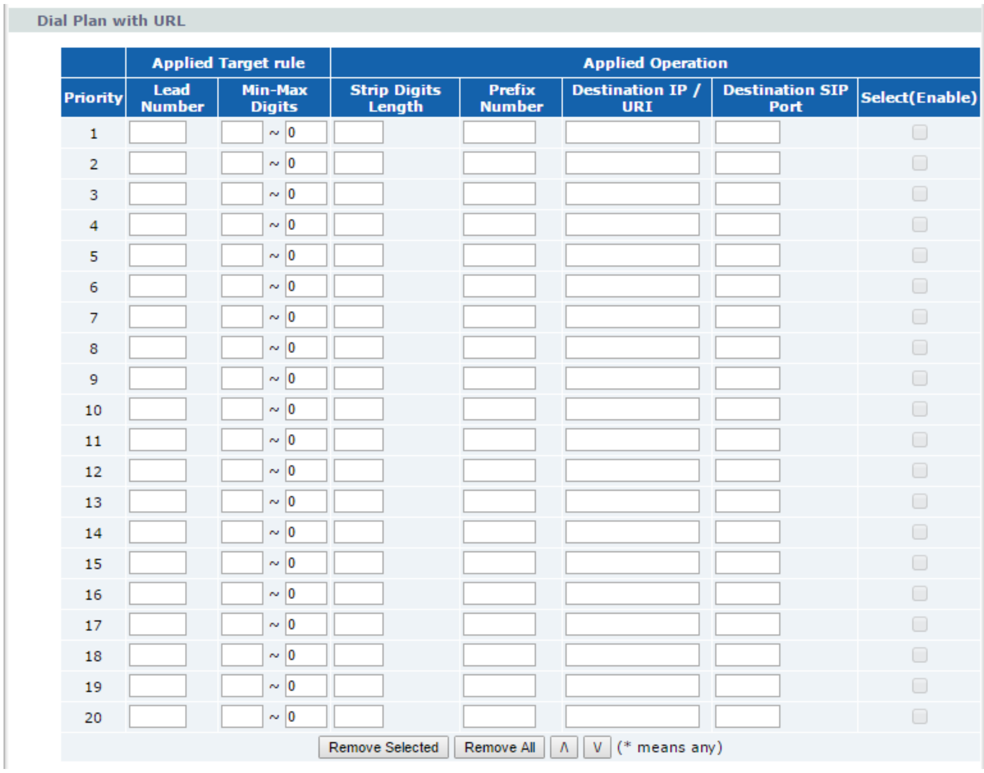



3. Name: Type a short name for the phone number shortcut.
4. Phone Number: Enter the desired phone number in the corresponding format that is required, using the ISP of your preference.
5. Repeat steps 2 through 4 to create additional speed dialing shortcuts.
6. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### Dial Plan with URL window

Use this window to create dial plan with URL entries that enable users to make peer-to-peer calls. In peer-to-peer calls, a user calls another VoIP device directly without going through a VoIP service provider's SIP server. Enter the callee's IP address or domain name. The SN100 sends SIP INVITE requests to the specified peer VoIP device when you dial the SIP number configured in this screen.

1. If your SN100 ATA has only one FXS port, go to step 2. Otherwise, select *Phone 1* or *Phone 2* from the Port Setting drop-down menu.




2. Make the following settings:

> **Note** When a user dials a number *xxxxxxx*, the ATA will check it against this dial plan, first by *Priority* order, and then execute.

– *Lead Number* is the first digit of the call-out dialing number.

– *Min-Max Digits* has two text fields that must be completed: *Min Length* and *Max Length* are the min/max allowed length of the number you can dial.

– *Strip Digits Length* is the number of digits that will be stripped from beginning of the dialed number.

– *Prefix Number* is the digits that will be added to the beginning of the dialed number.

– *Destination IP/ URI* is the IP address/domain name of the destination SN100 that owns this phone number. This for peer-to-peer IP calls.

– *Destination SIP Port* is used to overwrite the UDP port for the destination configured of this rule. Default port is 5060 as per the SIP server settings in Chapter 6.

3. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

## PSTN Line

Do the following to display the PSTN Line *General* window:

1. Click on the plus button next to *VoIP* to display the *PSTN LINE* sub-category (see figure 68 on page 111).

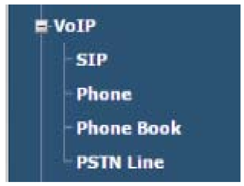



Figure 68. PSTN sub-category

2. Click *PSTN Line*. The *General* window displays (see figure 69).

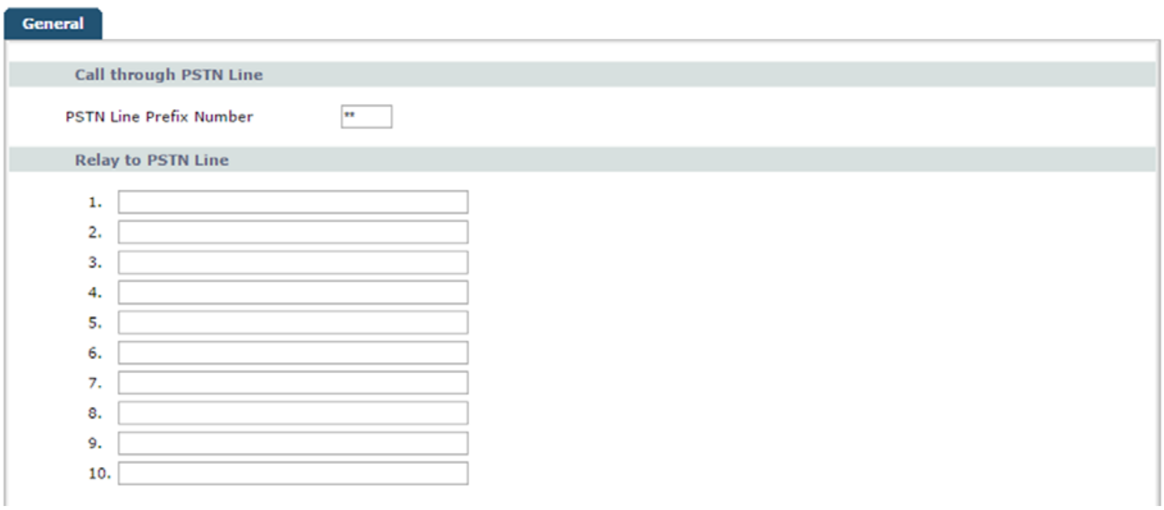



Figure 69. PSTN Line General window

3. Make the following settings:
   - –*Call through PSTN line:* Set up the prefix digit function key. For example, if the PSTN Line Prefix Number is **, it means that the analog phone connected to the FXS port can press **, then get a dial tone of PSTN port to dial destination number via PSTN port.
   - *Relay to PSTN Line*: Up to 10 pass through numbers can be configured. The phone connected to the FXS port, dialing one of these numbers, will cause the call to directly go out on the PSTN line, without routing it to the SIP account. This function may be used for emergency calls such as 911 for USA or 112 for Europe.
4. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

# Chapter 7 **Security Settings**

## *Chapter contents*

## Introduction

This chapter describes the following:

- Configuring the Firewall settings (see section "Firewall")
- Configuring the Content Filter settings (see section "Content Filter" on page 116)
- Configuring ACL settings (see section "ACL Setting" on page 117)

## Firewall

This section describes configuring SN100 Series ATA IP filtering, MAC filtering, and port filtering settings. Chose the appropriate section from the following:

- "Configuring IP Filtering settings"
- "Configuring MAC Filtering settings" on page 114
- "Configuring Port Filtering settings" on page 115

### *Configuring IP Filtering settings*

Entries in the table (see figure 71 on page 114) are used to restrict IP addresses from the LAN accessing the Internet. Use of such filters can be helpful in securing or restricting your local network.

Do the following to display IP Filtering settings:

1. Click on the plus button next to *Security* to display the sub-categories (see figure 70).

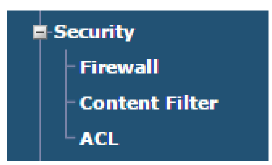



Figure 70. Security sub-categories

2. Click *Firewall*. The *IP Filtering* window displays (see figure 71 on page 114).

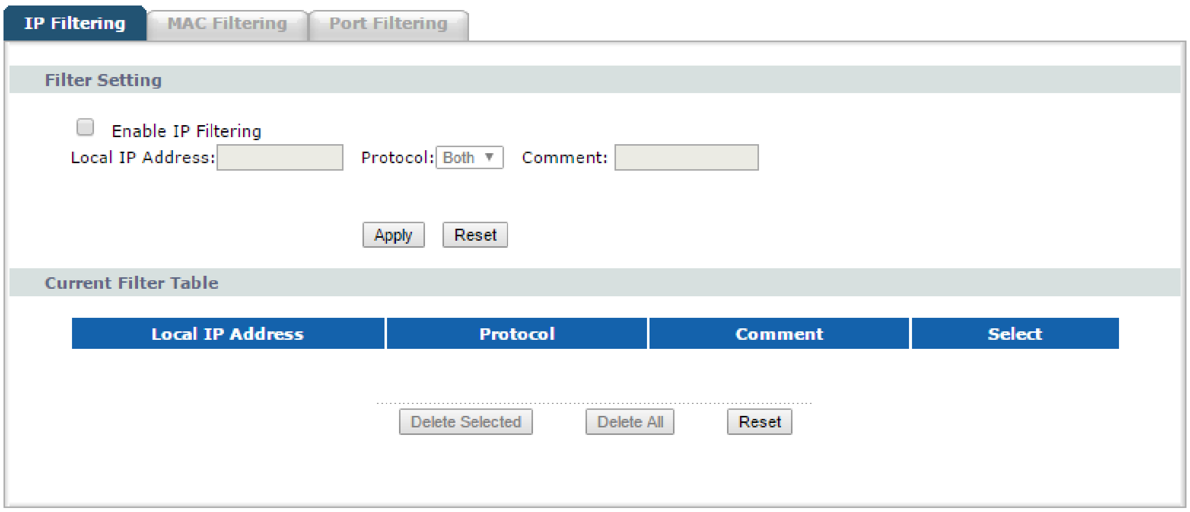



Figure 71. IP Filtering window

3. *Enable/Disable IP Filtering*: The IP address filter function may be enabled to restrict certain IPs from the LAN accessing the Internet for a specific protocol.
4. The comment field may be used for reference purposes such as “Storage Room PC”
5. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### Configuring MAC Filtering settings

Entries in the table (see figure 72 on page 115) are used to restrict MAC addresses from the LAN accessing the Internet.

Do the following to display MAC Filtering settings:

1. Click on the plus button next to *Security* to display the sub-categories (see figure 70 on page 113).
2. Click *Firewall*. The *IP Filtering* window displays (see figure 71 on page 114).
3. Click the *MAC Filtering* tab to bring it to the front (see figure 72 on page 115).

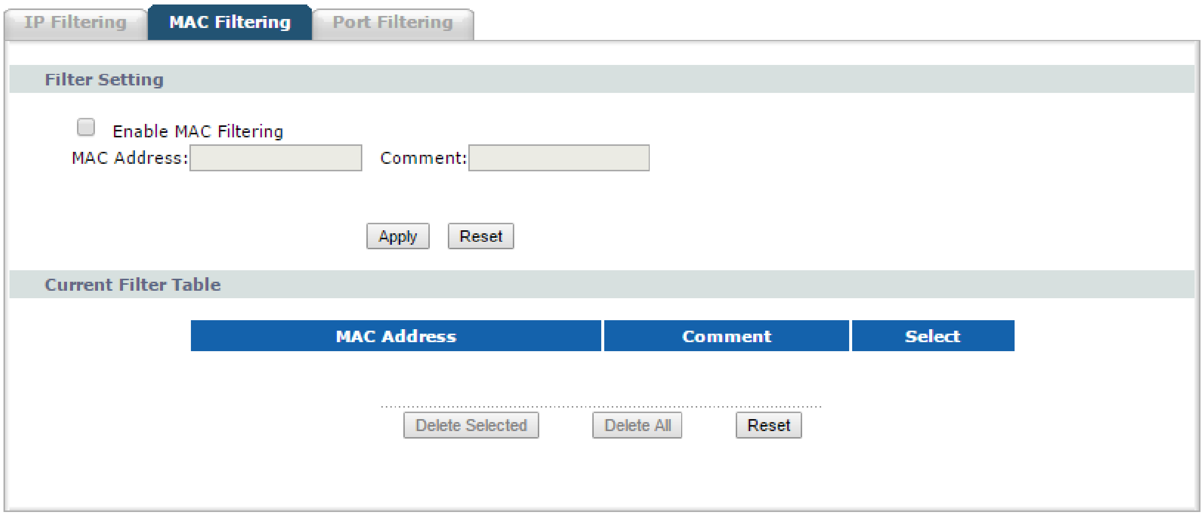



Figure 72. MAC filtering window

4. Enable/Disable MAC Filtering: Enable/Disable MAC filtering: Specify which MAC addresses are not allowed to access the Internet. The comment field may be used for reference purposes such as “Storage Room PC”.
5. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## Configuring Port Filtering settings

Entries in the table (see figure 73 on page 116) are used to restrict traffic based on certain ports, originating from hosts in the LAN destined to the Internet.

Do the following to display Port Filtering settings:

1. Click on the plus button next to *Security* to display the sub-categories (see figure 70 on page 113).
2. Click *Firewall*. The *IP Filtering* window displays (see figure 71 on page 114).
3. Click the *Port Filtering* tab to bring it to the front (see figure 73 on page 116).

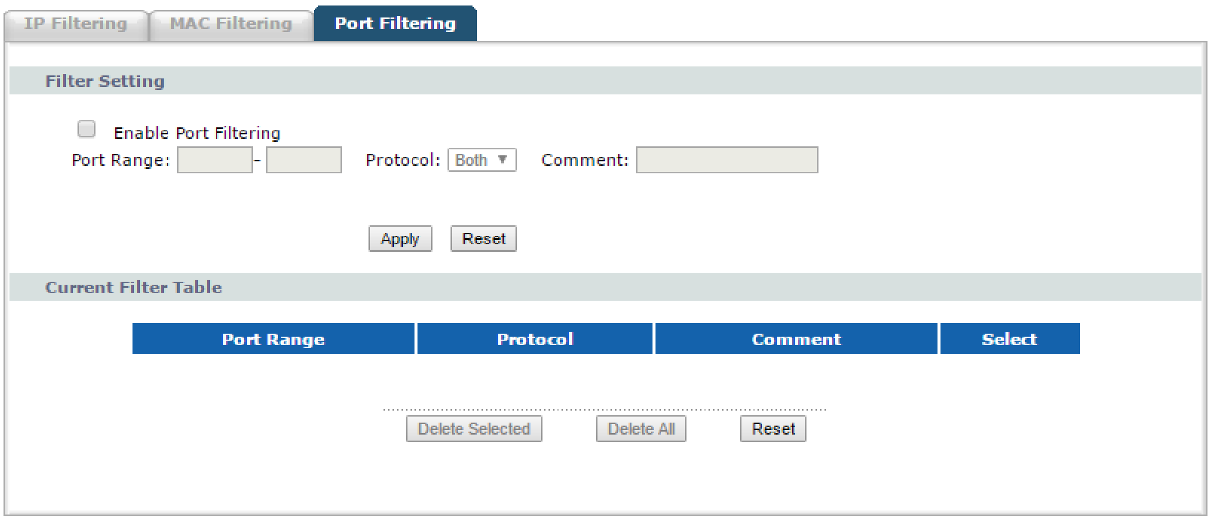



Figure 73. Port filtering window

4. Enable/Disable Port Filtering: Restrict hosts from LAN accessing the Internet, by specific ports.
5. The comment field may be used for reference purposes such as "Storage Room PC".
6. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## Content Filter

URL filter is used to deny LAN users from accessing the internet. Block those URLs which contain keywords listed below.

Do the following to display Content Filter settings:

1. Click on the plus button next to *Security* to display the sub-categories (see figure 70 on page 113).
2. Click *Content Filter*. The *Content Filter* window displays (see figure 74 on page 117).

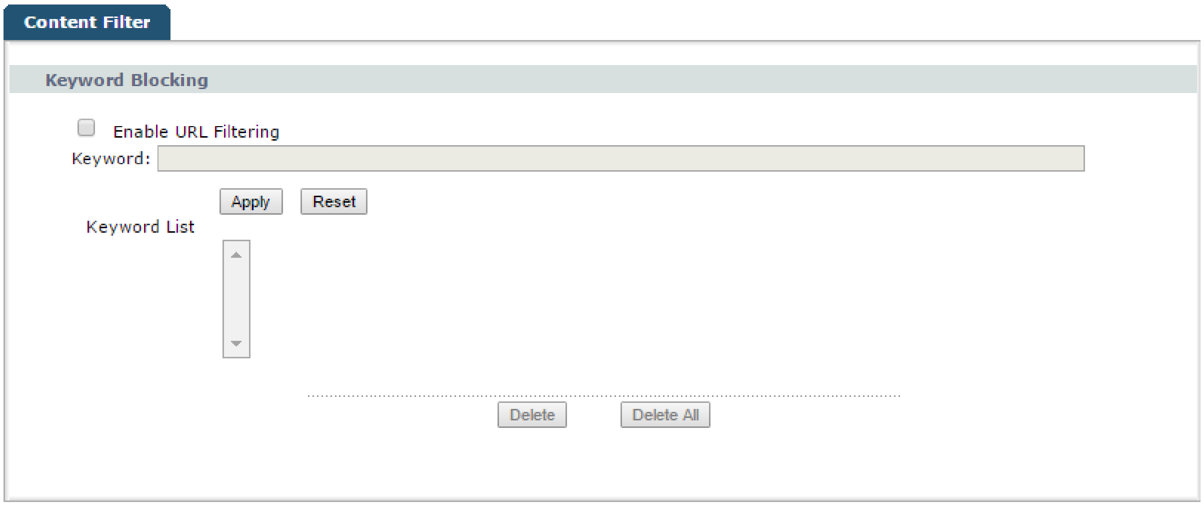



Figure 74. Content Filter Window

3. Make the following settings:
   – Enable/Disable the URL filter function (default setting is disable).
   – Enter the keywords you would like to be blocked
4. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## ACL Setting

The ACL is being used to control which services of the SmartNode 100 ATA can be accessed from WAN or LAN. Do the following to display the ACL Setting window:

1. Click on the plus button next to *Security* to display the sub-categories (see figure 70 on page 113).
2. Click *ACL Setting*. The *ACL Setting* window displays (see figure 75 on page 118).

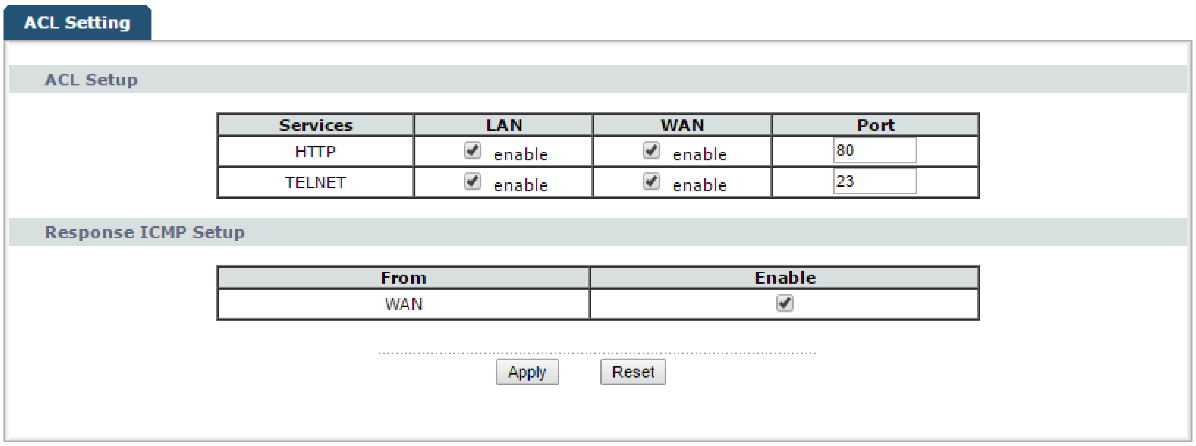



Figure 75. ACL Setting window

3. Make the following settings in the *ACL Setup* section:

   – Enable/Disable the LAN and WAN HTTP service as needed (default setting is enable).

   – Enable/Disable the LAN and WAN TELNET service as needed (default setting is enable).

4. Enable/Disable the From LAN setting (default setting is enable).

5. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

# Chapter 8 Maintenance Settings

## Chapter contents

## Introduction

This chapter describes the following:

- Configuring the System settings (see section "System")
- Configuring the Tools settings (see section "Tools" on page 127)
- Configuring Logs settings (see section "Logs" on page 131)

## System

This section describes the following:

- Configuring general system settings (see "Configuring General System settings")
- Configuring Administrators Account's user name and password (see "Configuring Administrator Account's user name and password" on page 121)
- Configuring User Account's user name and password (see "Configuring User Account's user name and password" on page 122)
- Configuring time settings (see "Configuring Time settings" on page 122)
- Configuring auto-provisioning settings (see "Configuring Auto Provision settings" on page 123)
- Configuring dynamic DNS settings (see "Configuring Dynamic DNS settings" on page 124)
- Configuring TR069 settings (see "Configuring TR069 settings" on page 125).

### *Configuring General System settings*

This window is where you set router or bridge mode. In bridge mode, firewall and NAT functions are disabled, which also means that there is no longer LAN IP addresses being assigned to devices connected to it. Accessing the SN100 when in bridge mode, requires a static IP to be set on the device which is to be specified under Manager IP Address as shown in figure 77 on page 121.

Do the following to display General settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76).

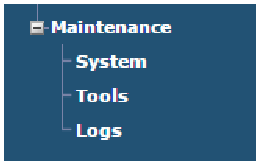



Figure 76. Maintenance sub-categories

2. Click *Maintenance*. The *Gemeral* window displays (see figure 77 on page 121).

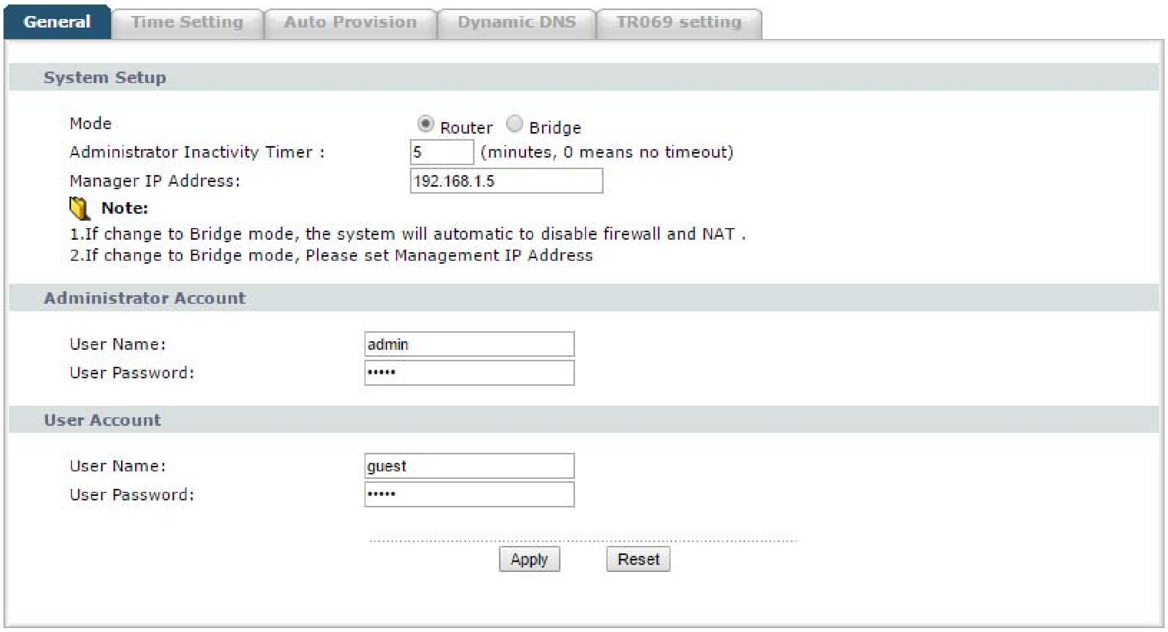



Figure 77. General System settings window

3. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## Configuring Administrator Account's user name and password

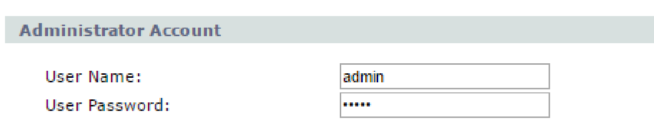
This section describes setting the administrator's user name and password.

IMPORTANT

For proper security we recommend that you change from the default credentials to a new administrator's user name and password as soon as possible.

Do the following to display General settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76).
2. Click *Maintenance*. The *Gemeral* window displays (see figure 77).
3. In the *Administrator Account* section, click the *User Name* box, then type a new user name.



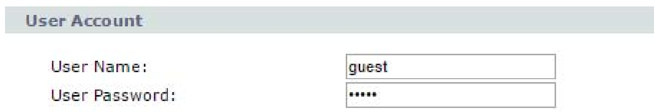

4. Click the *User Password* box, then type a new password.
5. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

## Configuring User Account's user name and password

This section describes setting the user account's user name and password. This account has limited capabilities in that it only allows changing network settings.

Do the following to display General settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76).
2. Click *Maintenance*. The *Gemeral* window displays (see figure 77).
3. In the *User Account* section, click the *User Name* box, then type a new user name.




4. Click the *User Password* box, then type a new password.
5. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.

## Configuring Time settings

You can maintain the system time by synchronizing with a public time server over the Internet.

Do the following to display Time settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *System*. The *General* window displays (see figure 77 on page 121).
3. Click the *Time Setting* tab to bring it to the front (see figure 78 on page 122).

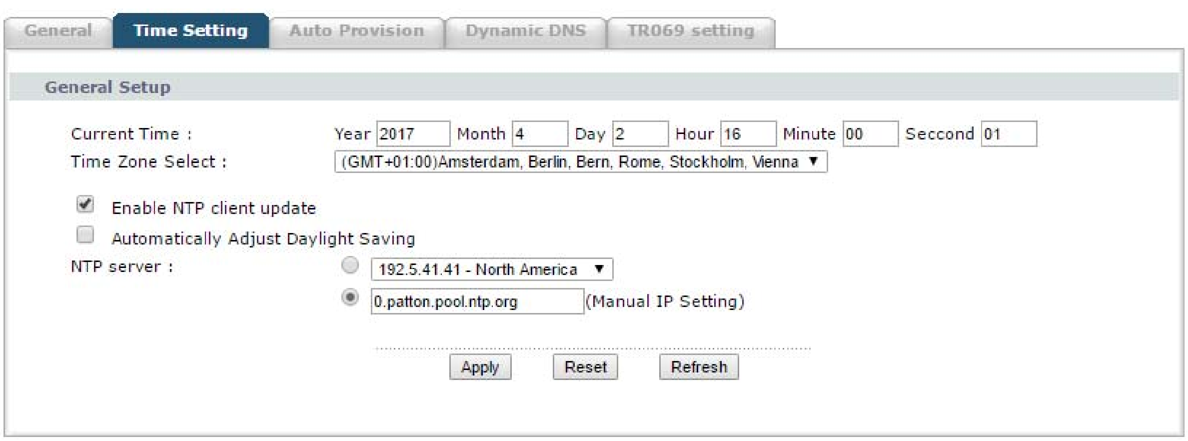



Figure 78. Time Setting window

4. Choose the desired time zone from the *Time Zone Select* drop down-menu
5. Enabling NTP client update let's the SN100 synchronize its clock automatically to the network time server.
6. Click the check box to enable the *Automatically Adjust Daylight Saving* setting.

7. If you did not enable NTP client update, go to section “Setting the time manually”. Otherwise click the button next to the NTP server drop-down menu, then choose the desired NTP server from the menu. If you want to manually enter the NTP server address, click the button next to the *Manual IP Setting* box, then type the address (for example: *0.patton.pool.ntp.org*).
8. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### Setting the time manually

If you wish to set the local time of the SN100 manually, in case there is no NTP server accessible from the network where the device is being installed, do the following.

1. Uncheck the box for *Enable NTP client update* (see figure 78 on page 122).
2. Enter the date and time.
3. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### Configuring Auto Provision settings

The provisioning function may be used for an automated, mass configuration of SN100 CPEs in the field. The provisioned configuration file, is loaded in to the system startup config once it has been downloaded. After the download has been done, the device will reboot to apply the changes as per the new config file.

Do the following to set up auto-provisioning:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *System*. The *General* window displays (see figure 77 on page 121).
3. Click the *Auto Provision* tab to bring it to the front (see figure 79 on page 123).

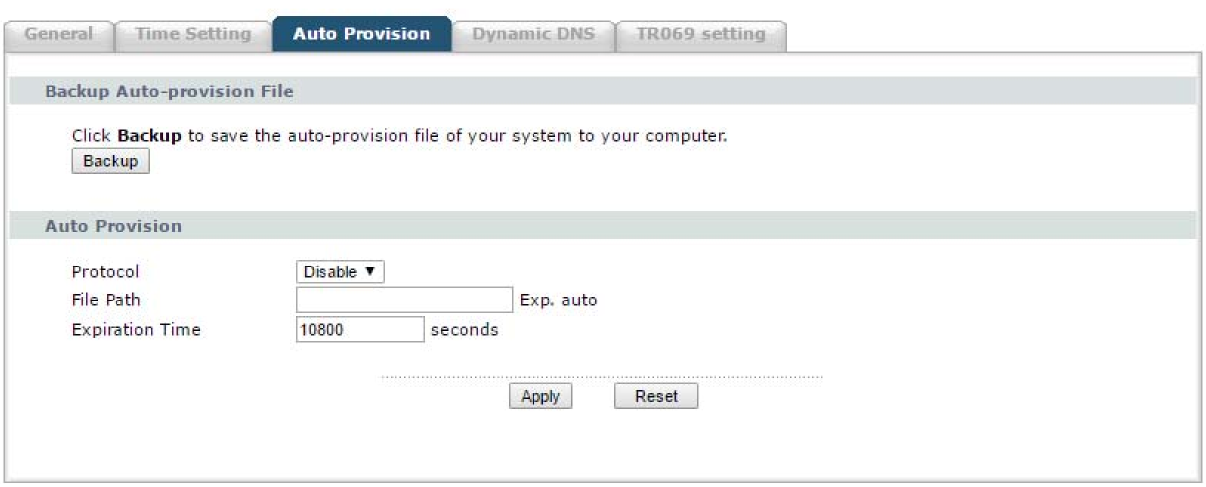



Figure 79. Auto Provision window

4. Select the provisioning protocol to match your environment. Choose between FTP, HTTP or TFTP
5. Enter the address path to access the config file which is to be provisioned
6. Optionally an expiration time may be set for the SN100 to periodically check if there is a new file available to be provisioned.
7. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


### Configuring Dynamic DNS settings

End users of Internet access receive an allocation of IP addresses, often only a single address, by their Internet service provider. The assigned addresses may be fixed (i.e. *static*), or may change from time to time, a situation called *dynamic*. Dynamic addresses are generally given only to residential customers and small businesses, as most enterprises specifically require static addresses.

Dynamic IP addresses present a problem if the customer wants to provide a service to other users on the Internet, such as a web service. As the IP address may change frequently, corresponding domain names must be quickly re-mapped in the DNS, to maintain accessibility using a well-known URL.

Dynamic DNS service addresses this scenario. The automatic reconfiguration is generally implemented in the user's router or computer, which runs software to update the DDNS service. The communication between the user's equipment and the provider is not standardized, although a few standard web-based methods of updating have emerged over time.Dynamic DNS is a service, that provides you with an unchanging Internet domain name (URL) to go with that (possibly ever changing) IP address.

Do the following to display Dynamic DNS settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *System*. The *General* window displays (see figure 77 on page 121).
3. Click the *Dynamic DNS* tab to bring it to the front (see figure 80 on page 125).

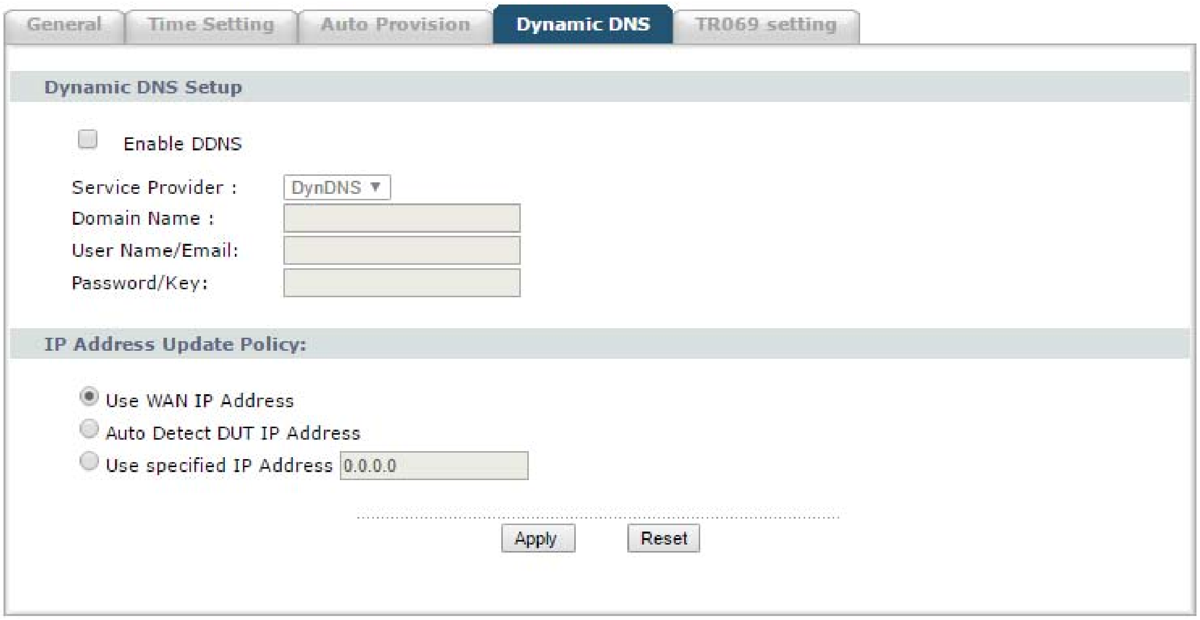



Figure 80. Dynamic DNS window

4. Click the check-box to enable or disable the DDNS service (default setting is disable).
5. Click the *Service Provider* drop-down menu to select the desired DDNS service provider.
6. Type the hostname you registered at the DynDns.org or TZO.net website in the *Domain Name* box.
7. Type username you registered at the DynDns.org or TZO.net website in the *User Name/Email* box.
8. Type the password you registered at the DynDns.org or TZO.net website in the *Password/Key* box.
9. In case your service at DynDNs.org or TZO.net includes a fixed IP address, you may enter this one in to the corresponding field. When doing so, make sure you set the radio button for that field as well.
10. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## Configuring TR069 settings

TR-069 is a DSL Forum specification for CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices. As a bidirectional SOAP/HTTP-based protocol, it provides the communication between customer-premises equipment (CPE) and Auto Configuration Servers (ACS).

Do the following to display TR069 settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *System*. The *General* window displays (see figure 77 on page 121).
3. Click the *TR069 Setting* tab to bring it to the front (see figure 81 on page 126).

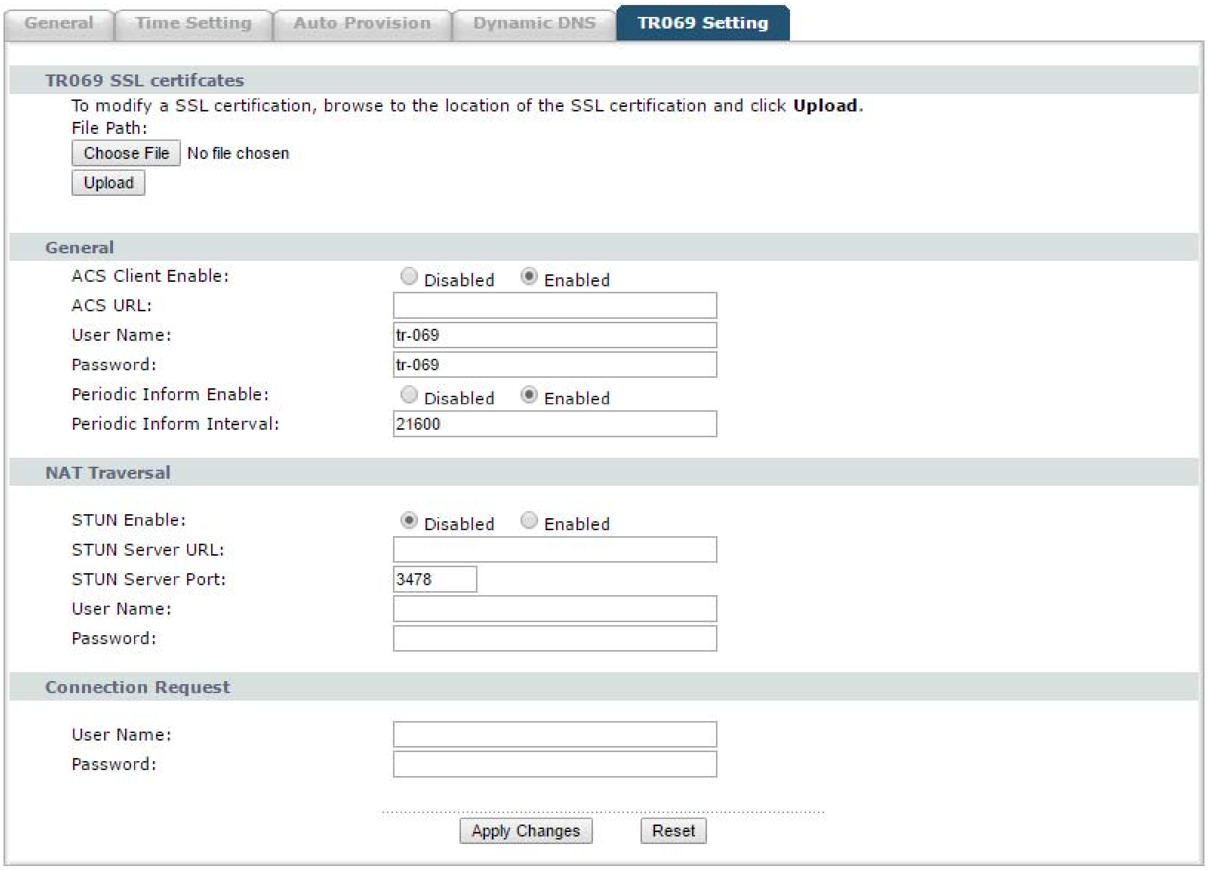
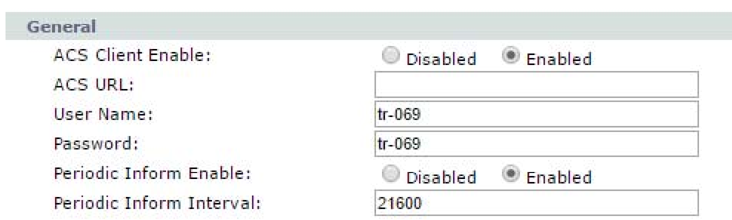
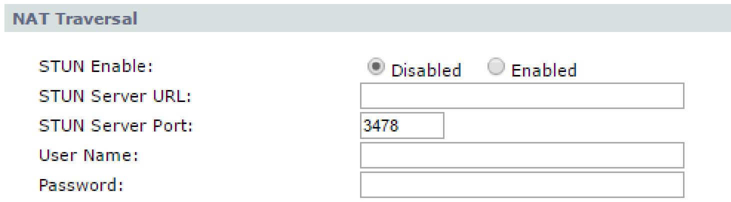



Figure 81. TR069 window

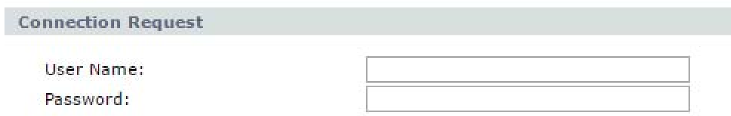
4. Set *ACS Client Enable* to *Enabled*.




5. Type the ACS address in the *ACS URL* field. This is the address to which the SN100 device should send connection requests.
6. Specify *User Name* and *Password* for the TR-069 secure connection.
7. When enabled, *Periodic Inform Enable* will automatically contact the ACS for new configurations according to the time interface specified. Click the *Enabled* button to activate the feature.




8. Optionally STUN may be configured, if the SN100 device is installed behind NAT.
9. Enter the STUN Server address
10. Specify the port for STUN. Default setting: 3478.
11. There might be a username and password required for STUN that is different from the ACS settings. If so, type the STUN *User Name* and *Password* in the corresponding fields




12. For TR-069 connection requests, User Name and Password is to be configured as per figure 81
13. When you are satisfied with the settings, click the *Apply* button to save and apply the changes.


## Tools

This section describes the following:

- How to back up the SN100 configuration, restore a back up configuration, and reset the SN100 configuration to factory default settings (see “Configuration settings” on page 127).
- How to upgrade SN100 firmware (see “Reset the SN100 to factory default settings” on page 129)
- How to run a ping test to troubleshoot problems network problems (see “Running a Ping Test” on page 130)
- How to make the SN100 perform a software restart (see “Performing an SN100 software restart” on page 130)

### *Configuration settings*

This window enables you to do the following:

- Back up current SN100 settings to a file
- Restore settings from a previous backup file.
- Reset the current configuration to factory default settings

Do the following to display tools configuration settings:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *Tools*. The *Configuration* window displays (see figure 82).

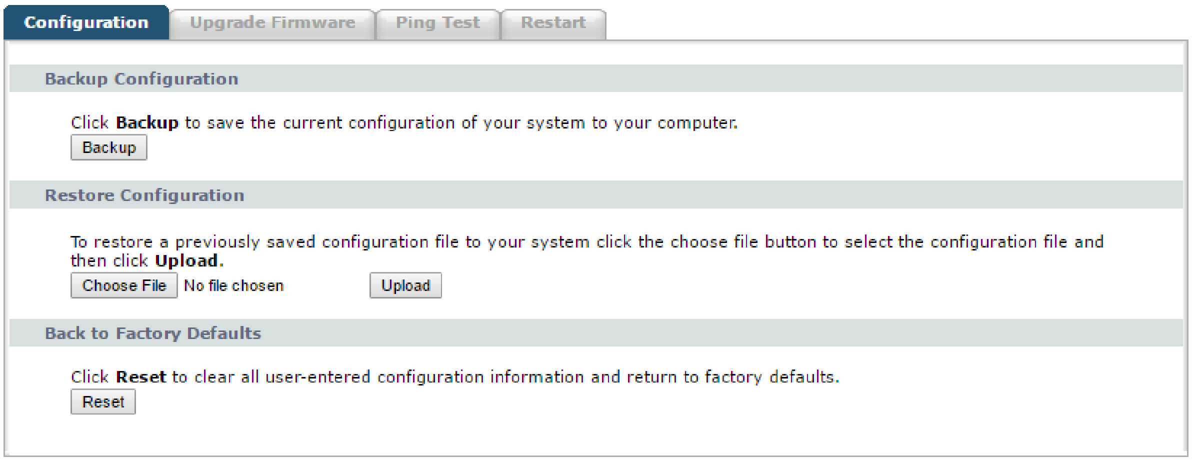



Figure 82. Tools Configuration window

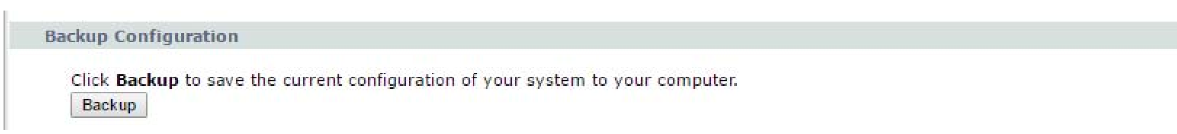
Choose from the following:

- Back up the SN100 configuration (see "Backup Configuration")
- Restore SN100 configuration from a previous back up (see "Restore Configuration" on page 128)
- Reset the SN100 to factory default settings (see "Reset the SN100 to factory default settings" on page 129)

### *Backup Configuration*

Do the following to back up the current SN100 configuration:

1. Click the *Backup* button to save a copy of SN100 configuration.



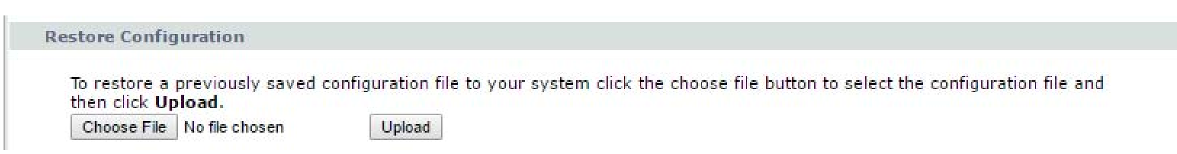

2. In the popup window select the location of where the backup configuration file should be stored.
3. Specify the filename of the backup config.
4. Click save for configuration the to be stored.

Configuration backup is completed.

### *Restore Configuration*

Do the following to restore a previously saved configuration file:

1. Click the *Choose File* button.

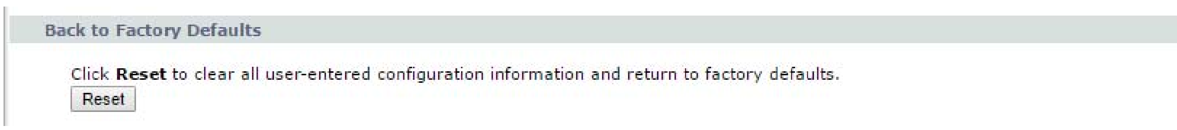


2. Select the configuration file, then click the *Upload* button.

The configuration has been restored.

### *Reset the SN100 to factory default settings*

Click the Reset button to clear all user-entered configuration information and return the SN100 to factory defaults




The SN100 has been reset to factory defaults.

## *Upgrading SN100 firmware*

The latest software builds are located at **https://www.patton.com/support/upgrades/**

Do the following to perform a firmware upgrade:

IMPORTANT

Do not power off the device during the upload. Doing so may crash the system.

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *Tools*. The *Configuration* window displays (see figure 82 on page 128).
3. Click the *Upgrade Firmware* tab to bring it to the front (see figure 83).

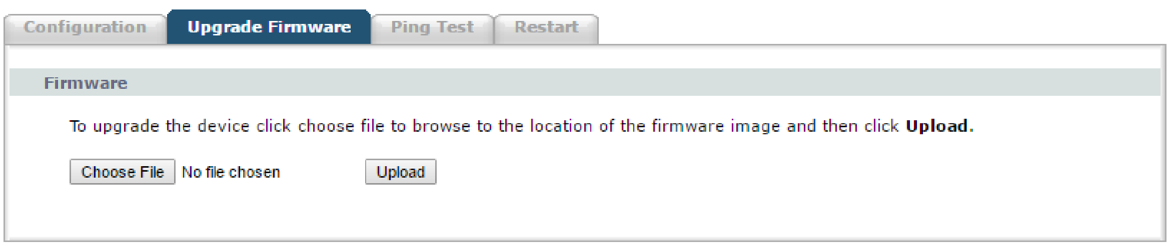
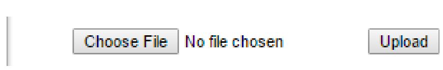



Figure 83. Upgrade Firmware window

4. Click the *Choose File* button (see figure 83 on page 129), then browse to the location of the firmware image.




5. Select the firmware image, then click the *Upload* button.

The firmware has been upgraded.

### Running a Ping Test

The ping test checks the reaction time of your connection—how fast you get a response after you've sent out a request. A fast ping—measured in milliseconds (ms)—means a more responsive connection.

Do the following to run the test:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *Tools*. The *Configuration* window displays (see figure 82 on page 128).
3. Click the *Ping Test* tab to bring it to the front (see figure 84).

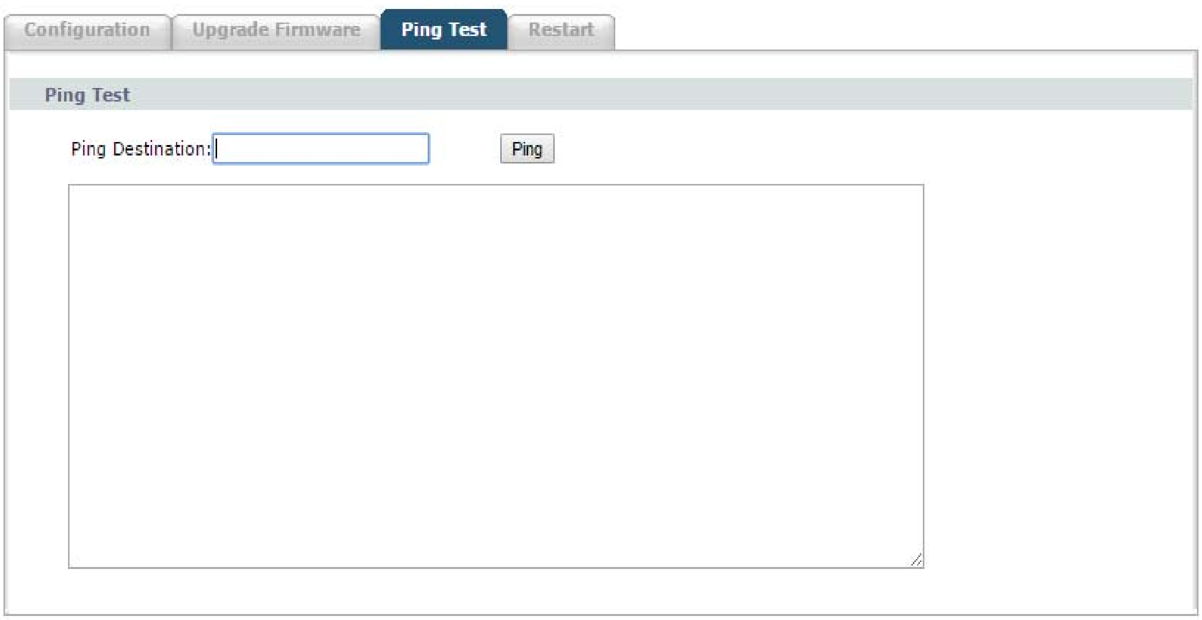



Figure 84. Ping Test window

4. Type the IP address or domain name of the destination computer in the *Ping Destination* box.
5. Click the *Ping* button to start the test. The results displayed will resemble the following example in which the IP address *169.95.1.1* is being pinged;

```
PING 168.95.1.1 (168.95.1.1): 56 data bytes
64 bytes from 168.95.1.1: icmp_seq=0 ttl=247 time=80.0 ms
64 bytes from 168.95.1.1: icmp_seq=1 ttl=247 time=100.0 ms
64 bytes from 168.95.1.1: icmp_seq=2 ttl=247 time=240.0 ms
64 bytes from 168.95.1.1: icmp_seq=3 ttl=247 time=30.0 ms
--- 168.95.1.1 ping statistics ---
5 packets transmitted, 4 packets received, 20% packet loss
round-trip min/avg/max = 30.0/112.5/240.0 ms
```

### Performing an SN100 software restart

If the SN100 is not responding correctly, you may decide to restart the system software. If so, do the following:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).

2. Click *Tools*. The *Configuration* window displays (see figure 82 on page 128).
3. Click the *Restart* tab to bring it to the front (see figure 85).

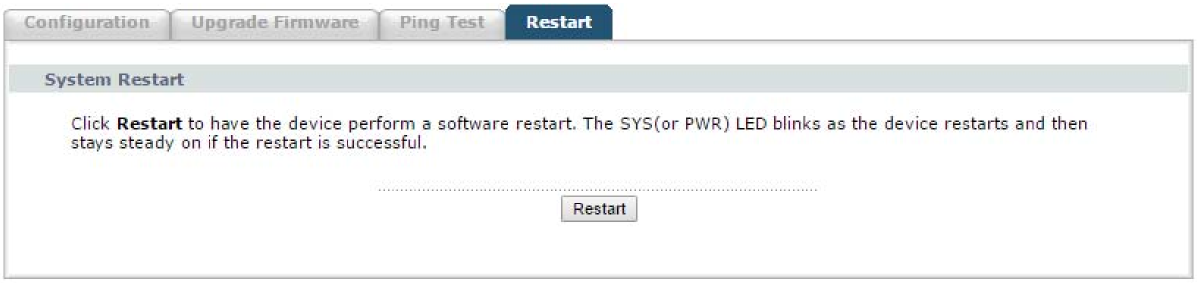



Figure 85. Restart window

4. Click the *Restart* button to have the device perform a software restart. Verify that the Power LED blinks as the device restarts and then stays on once the restart is completed.
5. Restart is successful.

## Logs

To keep a log of SN100 activities, do the following:

1. Click on the plus button next to *Maintenance* to display the sub-categories (see figure 76 on page 120).
2. Click *Tools*. The *Configuration* window displays (see figure 82).
3. Click the *Logs* tab to bring it to the front. The System Log window displays. (see figure 86 on page 132).

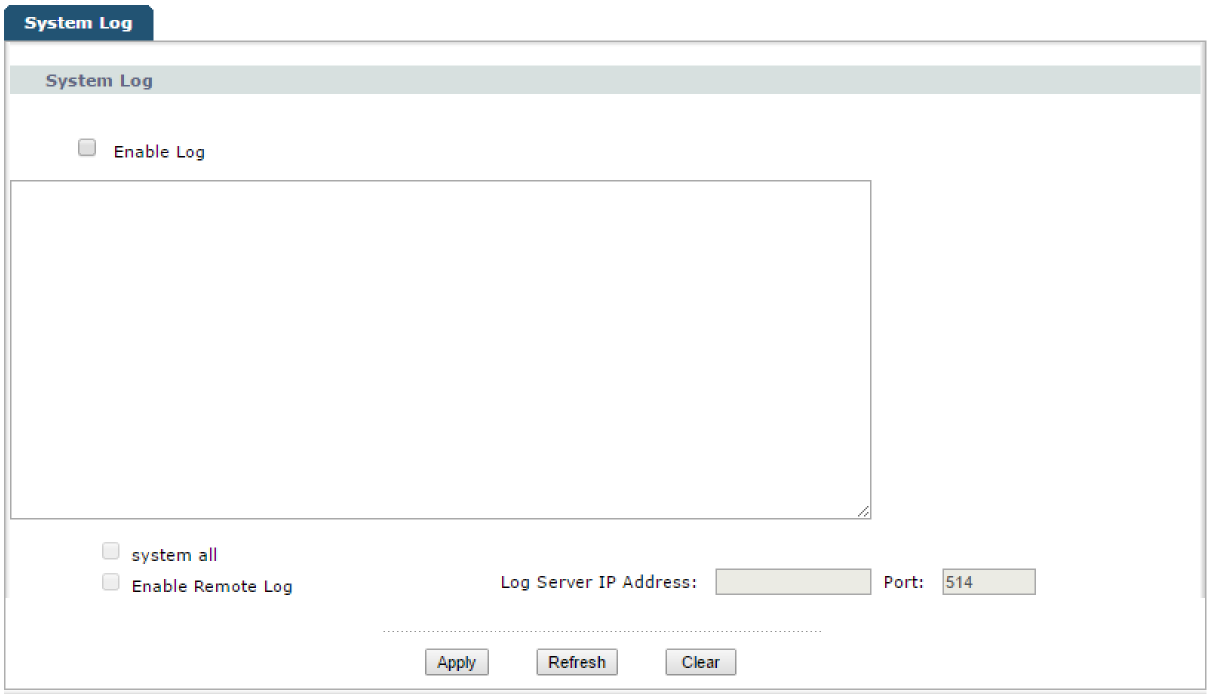



Figure 86. System Log window

4. Click the check box to enable or disable the log.
5. Specify if all system logs should be part of the logging or just the major ones which are enabled by default.
6. Optionally, the logs generated by the SN100 can be sent to a centralized Syslog server. To do so, specify the Syslog server IP address and port. (Default port is 514).

# Chapter 9 Contacting Patton for assistance

## *Chapter contents*

## Introduction

This chapter contains the following information:

- "Contact information"—describes how to contact Patton technical support for assistance.
- "Warranty Service and Returned Merchandise Authorizations (RMAs)"—contains information about obtaining a return merchandise authorization (RMA).

## Contact information

Patton Electronics offers a wide array of free technical services. If you have questions about any of our other products we recommend you begin your search for answers by using our technical knowledge base. Here, we have gathered together many of the more commonly asked questions and compiled them into a searchable database to help you quickly solve your problems:

- Online support—available at **http://www.patton.com/returns/**
- E-mail support—e-mail sent to **support@patton.com** will be answered within 1 business day
- Telephone support—standard telephone support is available five days a week—from **8:00 am** to **5:00 pm EST** (**1300** to **2200 UTC**)—by calling **+1 (301) 975-1007**

## Warranty Service and Returned Merchandise Authorizations (RMAs)

Patton Electronics is an ISO-9001 certified manufacturer and our products are carefully tested before shipment. All of our products are backed by a comprehensive warranty program.

> **Note** If you purchased your equipment from a Patton Electronics reseller, ask your reseller how you should proceed with warranty service. It is often more convenient for you to work with your local reseller to obtain a replacement. Patton services our products no matter how you acquired them.

### *Warranty Coverage*

Our products are under warranty to be free from defects, and we will, at our option, repair or replace the product should it fail within one year from the first date of shipment. Our warranty is limited to defects in workmanship or materials, and does not cover customer damage, lightning or power surge damage, abuse, or unauthorized modification.

### *Out-of-Warranty Service*

Patton services what we sell, no matter how you acquired it, including malfunctioning products that are no longer under warranty. Our products have a flat fee for repairs. Units damaged by lightning or other catastrophes may require replacement.

### *Returns for Credit*

Customer satisfaction is important to us, therefore any product may be returned with authorization within 30 days from the shipment date for a full credit of the purchase price. If you have ordered the wrong equipment or you are dissatisfied in any way, please contact us to request an RMA number to accept your return. Patton is not responsible for equipment returned without a Return Authorization.

### *Return-for-Credit Policy*

- Less than 30 days: No Charge. Your credit will be issued upon receipt and inspection of the equipment.
- 30 to 60 days: We will add a 20% restocking charge (crediting your account with 80% of the purchase price).
- Over 60 days: Products will be accepted for repairs only.

## *RMA Numbers*

RMA numbers are required for all product returns. You can obtain an RMA by doing one of the following:

- Completing a request on the RMA Request page in the *Support* section at **http://www.patton.com/returns/**
- By calling **+1 (301) 975-1007** and speaking to a Technical Support Engineer
- By sending an e-mail to **returns@patton.com**

All returned units must have the RMA number clearly visible on the outside of the shipping container. Please use the original packing material that the device came in or pack the unit securely to avoid damage during shipping.

## *Shipping Instructions*

The RMA number should be clearly visible on the address label. Our shipping address is as follows:

**Patton Electronics Company**
RMA#: xxxx
7622 Rickenbacker Dr.
Gaithersburg, MD 20879-4773 USA

Patton will ship the equipment back to you in the same manner you ship it to us. Patton will pay the return shipping costs.

# Appendix A Compliance Information

## Chapter contents

## Regulatory Information

### EMC Directive

- FCC Part 15, Class B
- EN 55032 Class B
- EN55024
- EN50581

### Low-Voltage Directive (Safety)

- IEC/EN60950-1, $2^{nd}$ Edition

### PSTN

This device is not intended nor approved for connection to the PSTN

## Radio and TV Interference (FCC Part 15)

This device generates and uses radio frequency energy, and if not installed and used properly-that is, in strict accordance with the manufacturer's instructions-may cause interference to radio and television reception. The device has been tested and found to comply with the limits for a Class B computing device in accordance with specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the device does cause interference to radio or television reception, which can be determined by disconnecting the unit, the user is encouraged to try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

## CE Declaration of Conformity

This device is in compliance with the essential requirements and other relevant provisions of Directive 2004/108/EC relating to electromagnetic compatibility and Directive 2006/95/EC relating to electrical equipment designed for use within certain voltage limits. The Declaration of Conformity may be obtained from Patton Electronics, Inc at www.patton.com/certifications.

The safety advice in the documentation accompanying this device shall be obeyed. The conformity to the above directive is indicated by CE mark on the device.

## Authorized European Representative

Martin Green
**European Compliance Services Limited**
Milestone house
Longcot Road
Shrivenham
SN6 8AL, UK

## Service

All warranty and non-warranty repairs must be returned freight prepaid and insured to Patton Electronics. All returns must have a Return Materials Authorization number on the outside of the shipping container. This number may be obtained from Patton Electronics Technical Services at:

- Tel: +1 (301) 975-1007
- Email: **support@patton.com**
- URL: **http://www.patton.com**

**Packages received without an RMA number will not be accepted.**

# Appendix B Specifications

## Chapter contents

## Telephone features

- G.711-ulaw
- G711-alaw
- G729
- G723
- G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)
- Call Waiting
- Call Transfer
- Call Forward as Busy forward; Non-Answer forward; unconditional forward
- Do-not-disturb (DND) support
- 3-way conferencing

## DTMF Function

- RFC2833
- In-Band DTMF
- SIP Info

## Fax Modes

- T.38 relay support
- G.711 pass-through

## IP Specifications

- SIP (RFC 3261)
- SDP (RFC 2327)
- STUN (RFC3489)
- RTP Payload for DTMF Digits (RFC2833)
- SIP Session Timers (RFC 4028)
- DNS SRV (RFC 2782)
- Outbound Proxy Support
- SIP REFER method (RFC 3515)
- Media/Ringing Tone Generation (RFC 3960)
- Message Waiting Indicator (RFC 3842)

## Call Features

- Adjustable Volume
- VAD/Dynamic Jitter Buffer
- Caller ID Generation: DTMF/FSK/NTT CID
- Peer-to-Peer Call by Dial Plan
- Different Country Tone Table Configuration

## Network and Security

- WAN: PPPoE client, DHCP client, Fixed IP Address, PPTP
- QoS: DSCP/VLAN Tag
- PPTP/L2TP VPN Client
- NAT
- Port/IP/MAC Filter

## Configuration/Management

- Web configuration interface; password protected
- Firmware upgrades through HTTP
- SNMP v1/v2 (Customization Project)
- VR
- TR069 support with STUN
- Auto-Provision by HTTP/TFTP/FTP

## Interfaces

- 1 x WAN 10/100BASE-TX
- 1 x LAN 10/100BASE-TX
- 1 x FXS (SN101)
- 2 x FXS (SN102)
- 1 x FXS/1 x FXO (SN102)
- 1 x Reset switch

## Operating Temperature

- Operating Temperature: 0~40°C
- Storage Humidity: 10~90% relative humidity

## Power

- AC: 100–240 VAC, 50/60 Hz
- DC:?12 VDC/1A

## Dimensions and Weight

- 98mm x 86mm x 29mm
- 100g

## Certifications

- CE, FCC, RoHS

# Appendix C Example Applications

## Chapter contents

## ATA with SIP Server Application

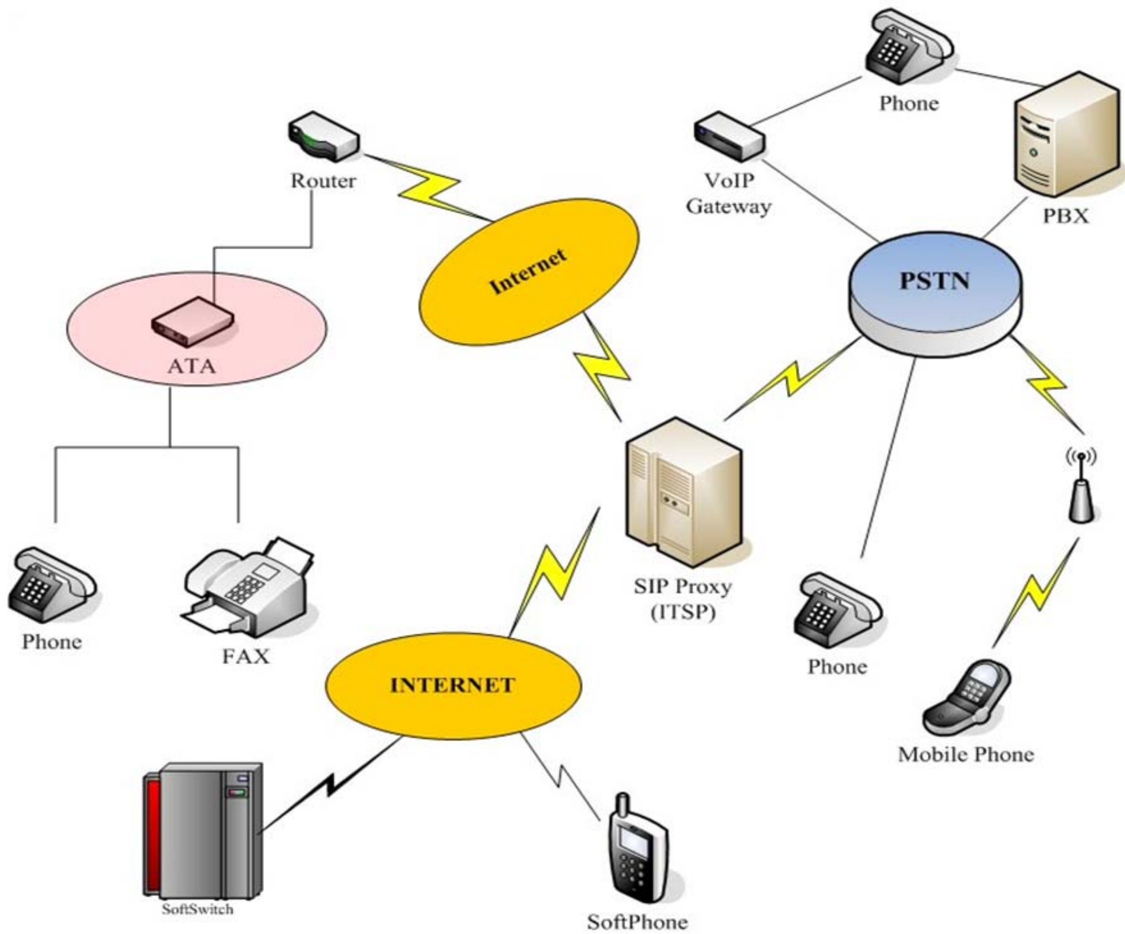



Figure 87. ATA with SIP Server Application

## ATA Peer to Peer Call (IP Domain Call) Application

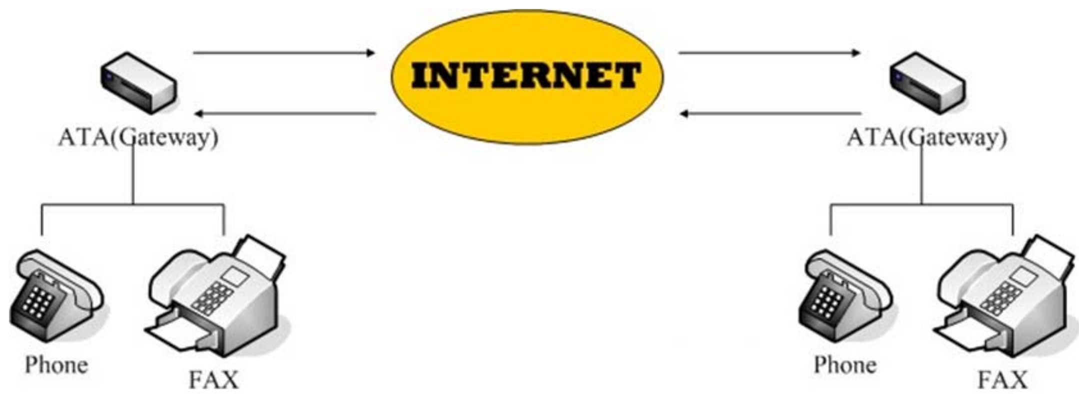



Figure 88. ATA Peer to Peer Call (IP Domain Call)

# Appendix D FAQ List

## Chapter contents

## Q: What is the default administrator username and password to log in to the SN100?

**A**: The default username is **admin**, and the default password is also **admin** to log in to the ATA. For security, you should change the password to protect your ATA against hacker attacks.

Default Wan Port Access type is **DHCP Client**

LAN Port IP Address is **192.168.1.1**.

Logging Web User Interface, open the Web browser (Microsoft Edge, Google Chrome, etc.), then input IP address.

## Q: I forgot the administrator password. What should I do?

**A**: Press the reset button on the rear panel for at least 5 seconds to return all settings to default factory values. Then you can use the default username: **admin**, and password: **admin** to log in.

## Q: Why is it that I can ping outside hosts, but not access Internet Web sites?

**A**: Verify the DNS server settings on your PC. You should get the DNS servers settings from your ISP. If your PC is running a DHCP client, remove any DNS IP address setting. As the router will assign the DNS settings to the DHCP-client-enabled PC.

## Q: What is the maximum number of IP addresses that the DHCP server of the gateway can assign to local PCs?

**A**: The built-in DHCP server can support 253 IP addresses for local network usage.

## Q: Why can’t I call out from the SN100?

**A**: Verify that the SN100 is registered SIP Proxy Server (ITSP), and verify that your Internet works fine. The SN100 can't make a call without Internet or a SIP account provided by ITSP. You must have a SIP account or know the other ATA/gateway IP/domain name, before you can make a VoIP call.

## Q: Why can't I use the Web Interface to configure the SN100?

**A**: Verify that your PC connects the SN100 LAN port, or PC and SN100 with the same subnet. If your PC isn’t on the same subnet, you will not be able to log into the SN100 Web interface.

## Q: Why does one-way talk happen?

**A**: Usually, one-way talk occurs when different codecs are used between VoIP devices making the call. Make sure both calling parties are using the same codec.

## Q: Why can’t I call out when the SN100 under the NAT?

**A**: Most VoIP products have problem in NAT Pass through. By SIP, there are many NAT Pass through Function can solve 80% NAT Problem. You can choose STUN/Outbound Proxy/ Symmetric RTP to Pass through NAT, you don't set any other setting (DMZ/Virtual Server) by router side. If you use STUN/Outbound Proxy, you must have a STUN/Outbound Proxy Server to support. If they can't pass NAT, please open the DMZ/Virtual Server by Router/NAT/Firewall.