# DVG-6004S
# VoIP Gateway

# User's Manual

Version 1.0

(Sept. 2017)

**Warranty:** please contact your D-Link Authorized Reseller or the D-Link Branch Office nearest your place of purchase for information about the warranty offered on your D-Link product.

***Information in this document is subject to change without notice.***

## FCC Warning

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communication. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

## CE Mark Warning

This is a Class B product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

**Warnung!**

Dies ist ein Produkt der Klasse B. Im Wohnbereich kann dieses Produkt Funkstoerungen verursachen. In diesem Fall kann vom Benutzer verlangt werden, angemessene Massnahmen zu ergreifen.

**Precaución!**

Este es un producto de Clase B. En un entorno doméstico, puede causar interferencias de radio, en cuyo case, puede requerirse al usuario para que adopte las medidas adecuadas.

**Attention!**

Ceci est un produit de classe B. Dans un environnement domestique, ce produit pourrait causer des interférences radio, auquel cas l`utilisateur devrait prendre les mesures adéquates.

**Attenzione!**

Il presente prodotto appartiene alla classe B. Se utilizzato in ambiente domestico il prodotto può causare interferenze radio, nel cui caso è possibile che l`utente debba assumere provvedimenti adeguati.

# Contents

# 1. Introduction

## 1-1 Product Overview

The DVG-6004S are designed to carry both voice and facsimile over the IP network. It uses the industry standard SIP call control protocol so as to be compatible with free registration services or VoIP service providers' systems. As a standard user agent, it is compatible with all common Soft Switches and SIP proxy servers. While running optional server software, the VoIP Gateway can be configured to establish a private VoIP network over the Internet without a third-party SIP Proxy Server.

The DVG-6004S can be seamlessly integrated into an existing network by connecting to a phone set and fax machine. With only a broadband connection such as an ADSL bridge/router, a Cable Modem or a leased-line router, the VoIP Gateway allows you to use voice and fax services over IP in order to reduce the cost of all long distance calls.

The DVG-6004S can be configured a fixed IP address or it can have one dynamically assigned by DHCP or PPPoE. It adopts either the G.711, G.726, G.729A or G.723.1 voice compression format to save network bandwidth while providing real-time, toll quality voice transmission and reception.

# 1-2 Hardware Description

## Front Pannel


## Indicators

**Power:** Power LED. A steady light indicates a proper connection to a power source.

**Alm.:** DVG can't get IP from DHCP or PPPoE Server. Once the service connects, the LED will turn off. The LED will light solid red if the self-test or boot-up fails.

**Reg.:** The Reg. LED will turn on when the VoIP Gateway is connected to a VoIP service provider. The LED will blinking if not connected to a service provider.

**Prov.** A blinking light indicates the VoIP Gateway is attempting to connect with the Provisioning server

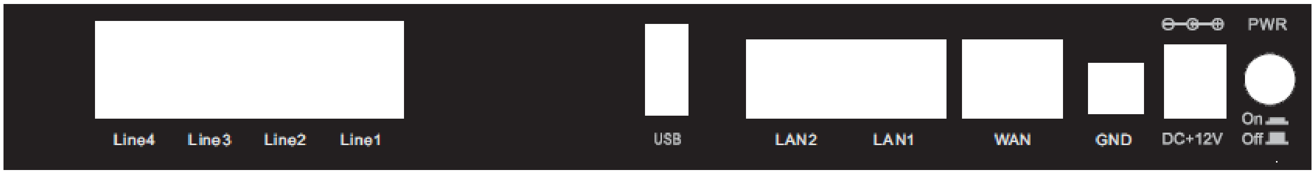
**WAN:** When a connection is established LED will light up solid. The LED will blink to indicate activity. If the LED does not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

**LAN:** When a connection is established the will light up solid on the appropriate port. The LEDs will blink to indicate activity. If the LED does not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

**Line:**

Green Blinking – FXO is alerting (ringing) for an inbound call.
Green Solid – The line is in use.

## Rear Panel




1. **Line Port (1-4):** Connect to your phones using standard phone cabling (RJ-11).
2. **USB:** Connect to a 3G/4G USB dongle or a printer, this feature is an option.
3. **LAN:** Connect to your Ethernet enabled computers using Ethernet cabling.
4. **WAN:** Connect to your broadband modem using an Ethernet cable.
5. **Ground:** A conducting connection with the earth. Connect with the ground so as to make the earth a part of an electrical circuit using metal wire.
6. **Power Receptor:** Receptor for the provided power adapter.
7. **Power Switch:** Press it down to turn on DVG.

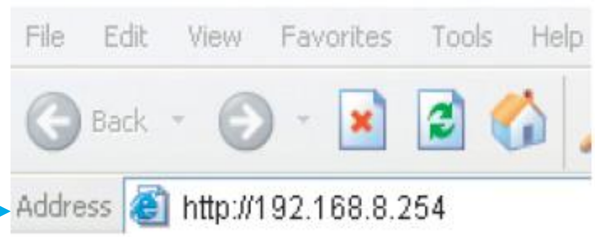

**Use Reset Button to restore factory default settings:**

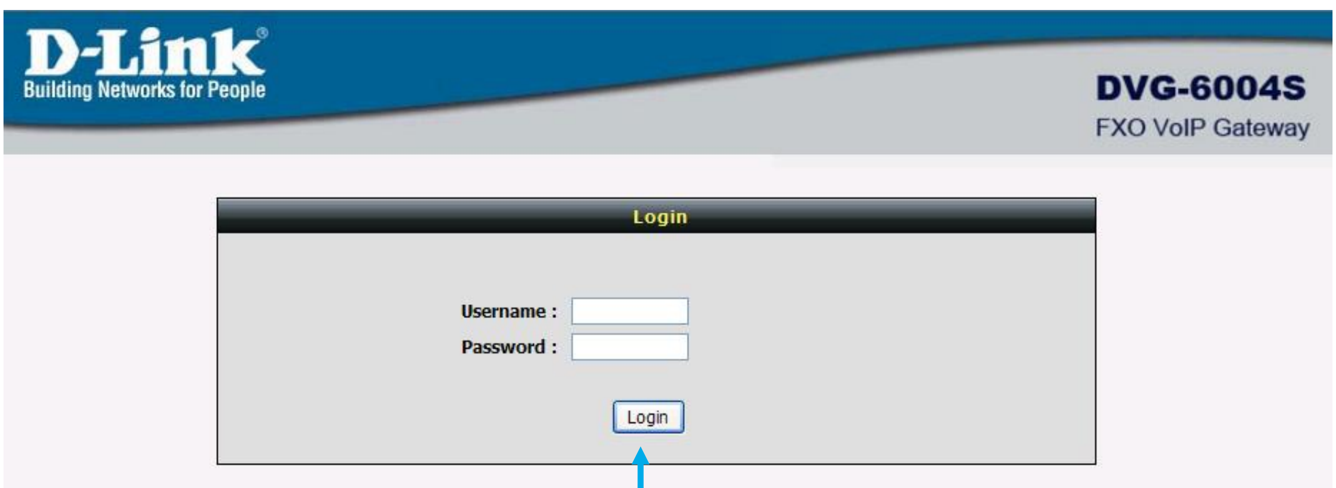
1. **Press and hold the reset button for 6 seconds and Alarm Indicator will be blinking fast**
2. **Release the reset button. Factory settings will be restored.**

# 2. Getting Started

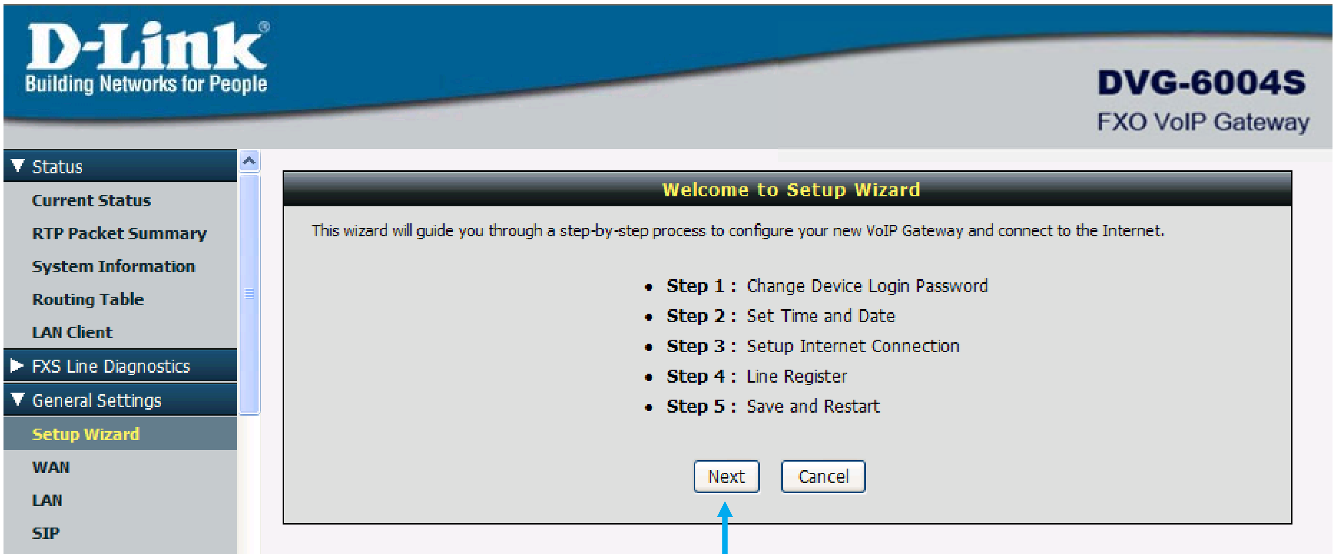
To access the web-based configuration utility, open a web browser such as Internet Explorer and enter the IP address of the DVG-6004S.

Open your Web browser and type http://192.168.8.254 into the URL address box. Press the Enter or Return Key.



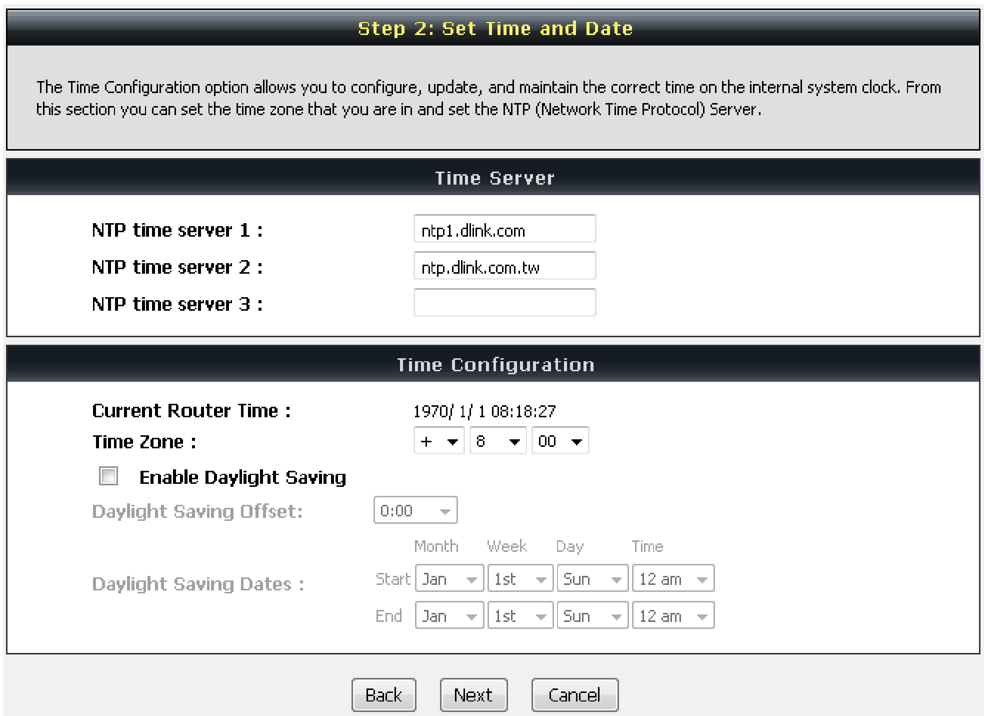

Click **Login** to enter Web Site.




Click **Setup Wizard** and **Next**.

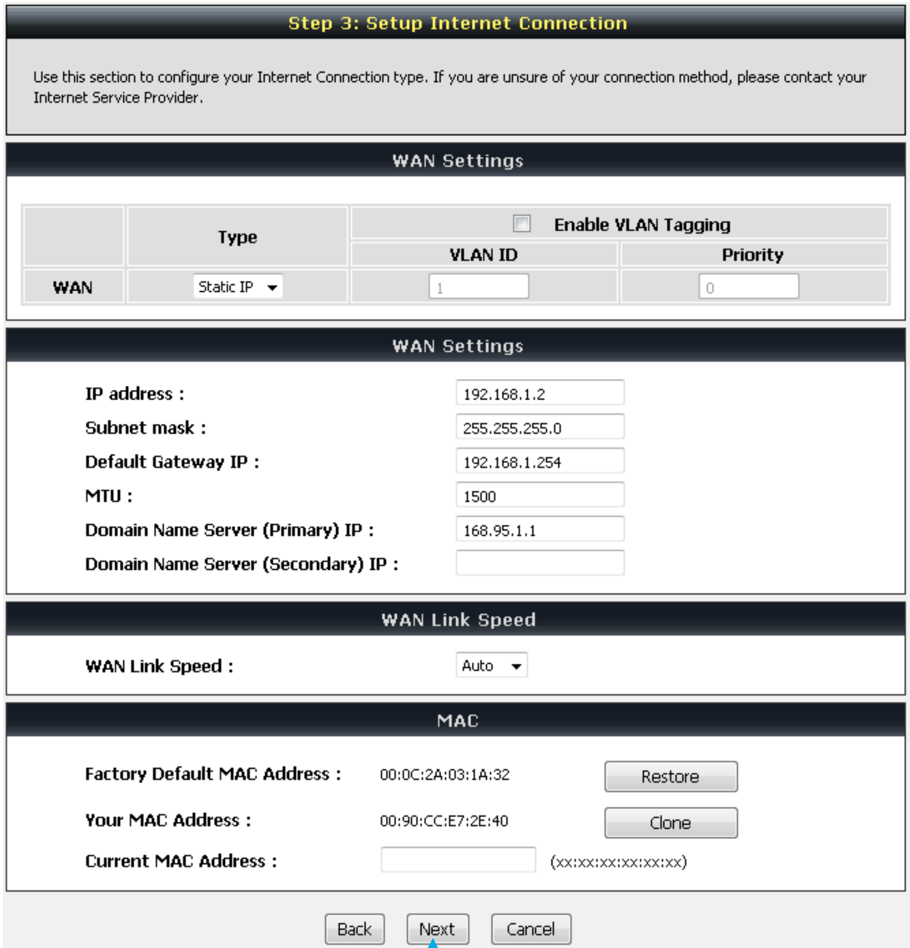

It is highly recommended to create a login ID and password to keep your gateway secure.

Click **Next**.




Enter a NTP server or use the default server.

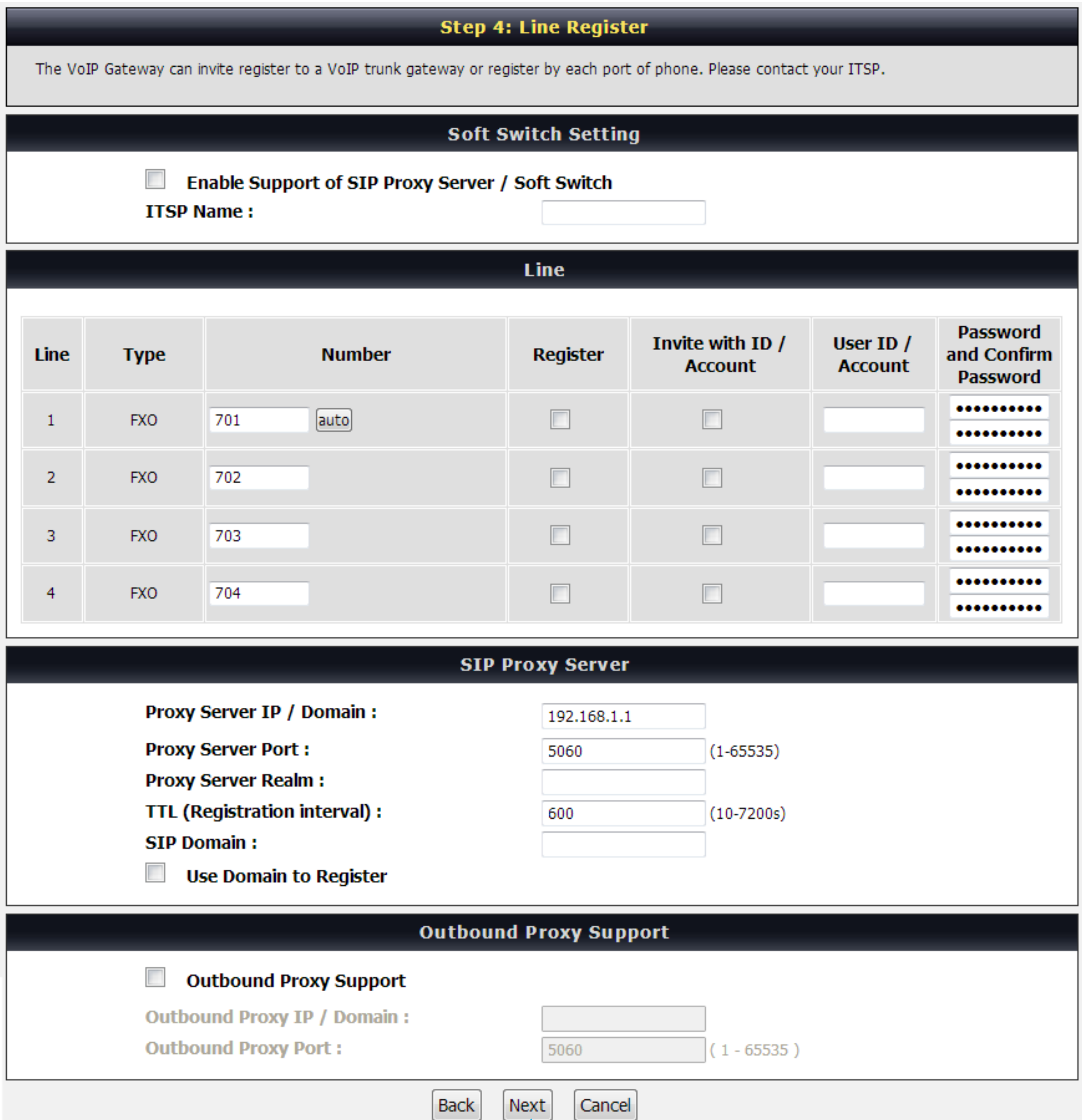
Click **Next**.




Select your Internet connection type:
**DHCP** – Most Cable ISPs or if you are connecting the DVG-6004S behind a router.
**Static IP** – Select if your ISP supplied you with your IP settings.
**PPPoE** – Most DSL ISPs.
**PPTP** – Select if required by your ISP.

Select **Manual** to manually enter IP address of DNS or select **Auto** if DNS is assigned by ISP.

Click **Next**.



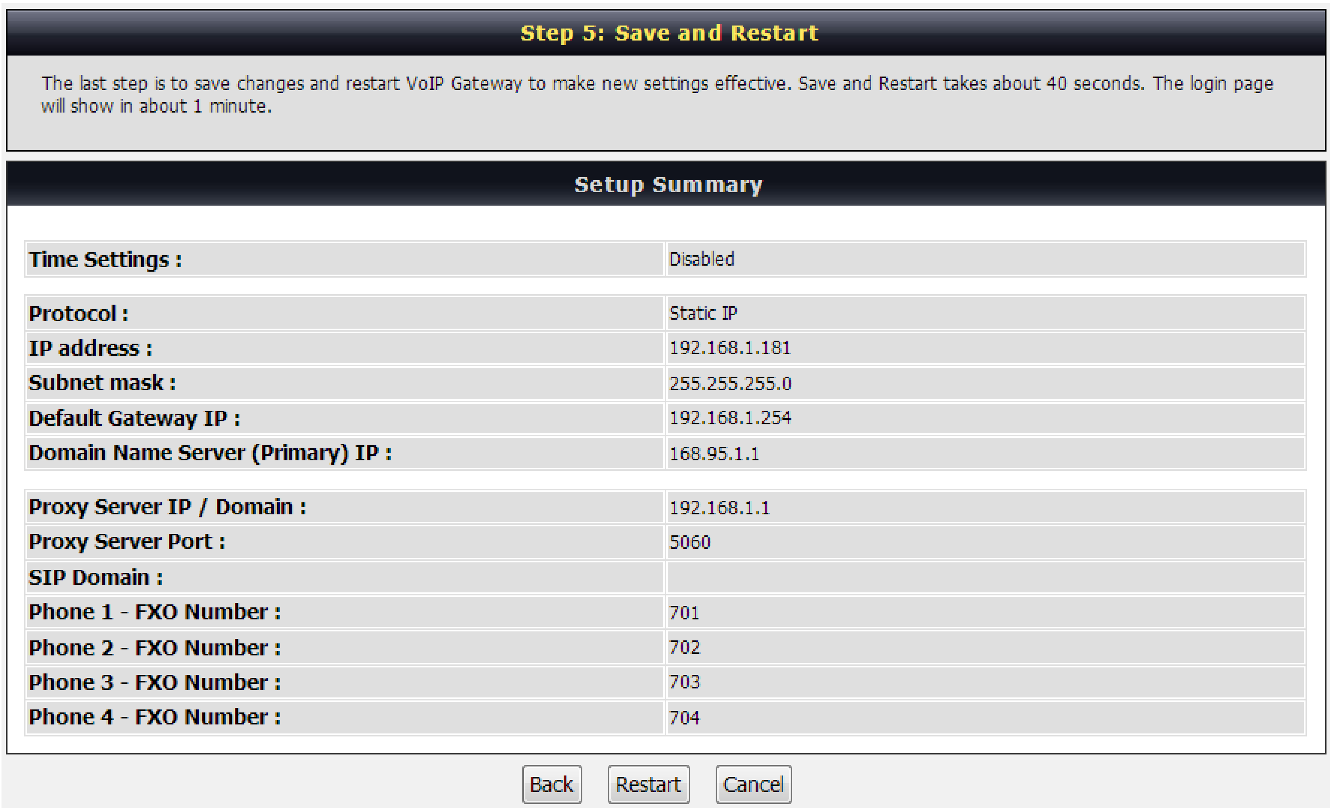

Register to the SIP Proxy Server by clicking **Enable support of SIP Proxy Server**. Enter **Proxy Server IP/Domain** and **Port**.

The **Outbound Proxy Support** is optional. To register, please click on the **Outbound Proxy Support** check box and enter **Outbound Proxy IP/Domain** and **Port** in it.

Registration by phone line: enter

**Number, User ID/Account** and **Password** supplied by your ITSP. Click on the **Register** check box to register to Proxy Server.

Click **Next**.


**Step 5: Save and Restart**

The last step is to save changes and restart VoIP Gateway to make new settings effective. Save and Restart takes about 40 seconds. The login page will show in about 1 minute.

**Setup Summary**

| | |
|---|---|
| **Time Settings :** | Disabled |
| **Protocol :** | Static IP |
| **IP address :** | 192.168.1.181 |
| **Subnet mask :** | 255.255.255.0 |
| **Default Gateway IP :** | 192.168.1.254 |
| **Domain Name Server (Primary) IP :** | 168.95.1.1 |
| **Proxy Server IP / Domain :** | 192.168.1.1 |
| **Proxy Server Port :** | 5060 |
| **SIP Domain :** | |
| **Phone 1 - FXO Number :** | 701 |
| **Phone 2 - FXO Number :** | 702 |
| **Phone 3 - FXO Number :** | 703 |
| **Phone 4 - FXO Number :** | 704 |

Back Restart Cancel

Setup is finished. Check the summary of your settings. To make new settings effective, you must click on the **Restart** button to reboot the DVG-6004S.

Click **Restart**.

# 3. VoIP Gateway Web Configuration

## 3-1 Status

### 3-1-1 Current Status

Status → Current Status

**Current Status**

Refresh Time (2 - 30s) : 5

**Port Status (D=Disabled, S=Successful, W=Waiting Reply, F=Failed)**

| Line | Type | Extension Number | Line Status | Calls | Number | Proxy Register |
|---|---|---|---|---|---|---|
| 1 | FXO | S1 701 | Disconnect | 0 | voip:<br>pstn: | D,D,D (00:07:50) |
| 2 | FXO | S1 702 | Disconnect | 0 | voip:<br>pstn: | D,D,D (00:07:50) |
| 3 | FXO | S1 703 | Disconnect | 0 | voip:<br>pstn: | D,D,D (00:07:50) |
| 4 | FXO | S1 704 | Disconnect | 0 | voip:<br>pstn: | D,D,D (00:07:50) |

SIP Proxy Hunting Number Registration : D,D,D (00:07:50)Trunk D,D,D (00:07:50)

**Server Registration Status**

| | |
|---|---|
| DDNS Registration : | Disabled (00:07:50) |
| STUN Registration : | Disabled (00:07:50) |

For Port Status, it includes if each port registers to Proxy successfully, the last dialed number, how many calls each port has made since the VoIP Gateway is start, etc.

For Server Registration Status, it shows the registration status of DDNS, STUN

## 3-1-2 RTP Packet Summary

Status → RTP Packet Summary

**RTP Packet Summary**

**Line**

| Line | Codec | The last packet's source IP | The last packet's source Port | Packet Sent | Packet Received | Packet Lost |
|---|---|---|---|---|---|---|
| 1 | G.711 u-law 64kbps | | 0 | 0 | 0 | 0 |
| 2 | G.711 u-law 64kbps | | 0 | 0 | 0 | 0 |
| 3 | G.711 u-law 64kbps | | 0 | 0 | 0 | 0 |
| 4 | G.711 u-law 64kbps | | 0 | 0 | 0 | 0 |

Refresh

Display the information of the last call made. Press **Refresh** button to get the latest RTP Packet Summary.

## 3-1-3 System Information

Status → System Information

**System Information**

| System Information | |
|---|---|
| Time and Date : | 2017/09/19 14:13:45 |
| Firmware Version : | 1.02.38.99 |
| Serial Number : | 000C2A07CC50 |

| WAN Port Information | |
|---|---|
| Factory Default MAC Address : | 000C2A07CC50 |
| Net Link : | Connected |
| IP address / Subnet mask : | 192.168.1.181 / 255.255.255.0 |
| Default Gateway : | 192.168.1.254 |
| Domain Name Server : | 168.95.1.1 |

| LAN Port Information | |
|---|---|
| MAC Address : | 000C2A07CC51 |
| IP address : | 192.168.8.254 |
| Subnet mask : | 255.255.255.0 |
| Net Link : | LAN1 Disconnected, LAN2 Connected |

| Hardware | |
|---|---|
| Hardware Platform : | SP5250O+ |
| Hardware : | C1 |
| Driver : | 1.4.2.252.274/3600 |

For WAN Port Information, it shows IP address, subnet mask, default gateway and DNS server. If you use PPPoE to obtain IP, you will know if the IP is obtained through this method. If IP address, subnet mask, default gateway is blank, it means that the VoIP Gateway does not obtain IP.

For LAN Port Information, it shows LAN port IP, subnet mask, and the status of DHCP server.

For Hardware, it shows the hardware platform and driver version.

## 3-1-4 Routing Table

Status → Routing Table

It displays routing table of DVG-6004S.

**Routing Table**

| Destination | Netmask | Gateway | Iface |
|---|---|---|---|
| 192.168.1.0 | 255.255.255.0 | 0.0.0.0 | WAN1 |
| 192.168.8.0 | 255.255.255.0 | 0.0.0.0 | LAN |
| default | 0.0.0.0 | 192.168.1.254 | WAN1 |

## 3-1-5 LAN Client

The **DHCP Clients** table displayed LAN device that has already been assigned an address from DVG-6004S. You can check if the DHCP client has obtain an IP address.

Status → LAN Client

**DHCP CLIENTS**

| IP Address | MAC Address | Live Time |
|---|---|---|
| 192.168.8.1 | 00:19:d2:35:45:60 | 2147448608 |

Refresh

# 3-2 General Settings

## 3-2-1 WAN

WAN (Wide Area Network) Settings are used to connect to your ISP (Internet Service Provider). The WAN settings are provided to you by your ISP and oftentimes referred to as "public settings". Please select the appropriate option for your specific ISP.

### IP Configuration (Setting WAN Port)

There are five methods of obtaining a WAN port IP address:

1. DHCP, which means a Dynamic IP (Cable Modem)
2. Static IP
3. PPPoE (dial-up ADSL)
4. PPTP

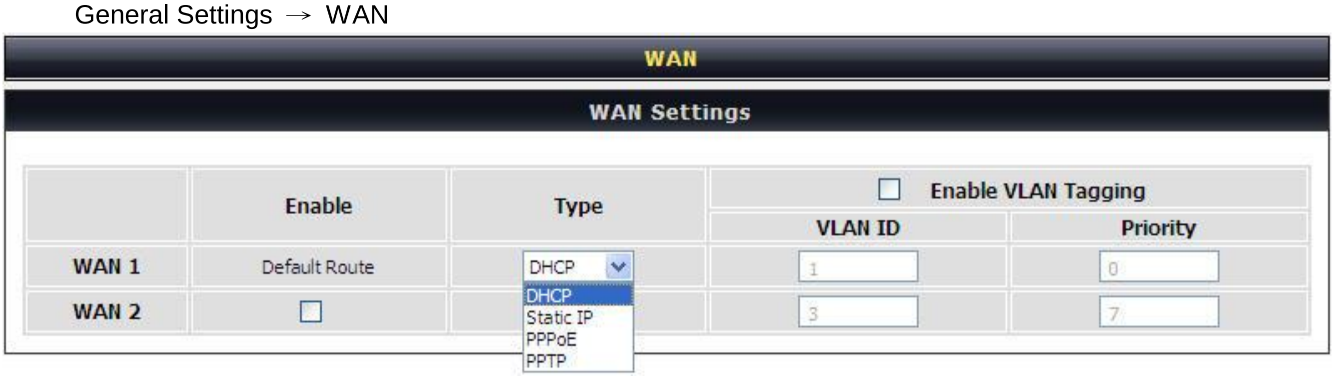
Methods for using DHCP and PPPoE for obtaining an IP address may vary. If you are not familiar with creating a network connection, please contact your local ISP.

After selecting the suitable option, click **Accept** at the bottom of the screen to save the settings.

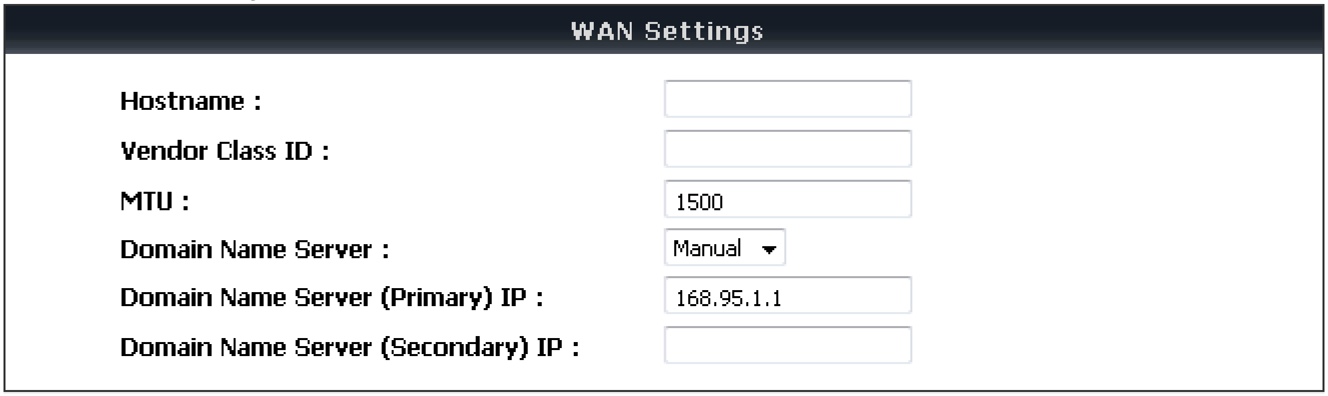
You need to save the changes and restart the VoIP Gateway to make the changes active. Saving the settings: Click **MAINTENANCE** and select **Save/Restart** in **System** from the left menu. Tick **Save Settings** and **Restart**, then click **Accept**. Wait for about 50 seconds before the VoIP Gateway obtaining an IP address by the method you selected.

**Note:** When the system has obtained a new IP address, and you are using a WAN port to enter the Web Configuration Screen, the new IP address has to be used before you can get connected to the VoIP Gateway. The same principle applies to the next two settings.

General Settings → WAN

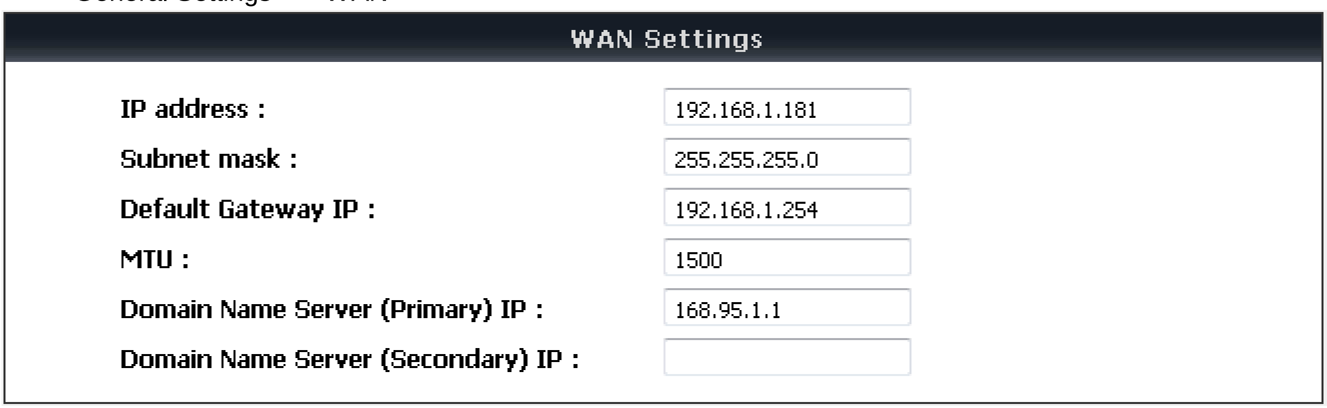


General Settings → WAN



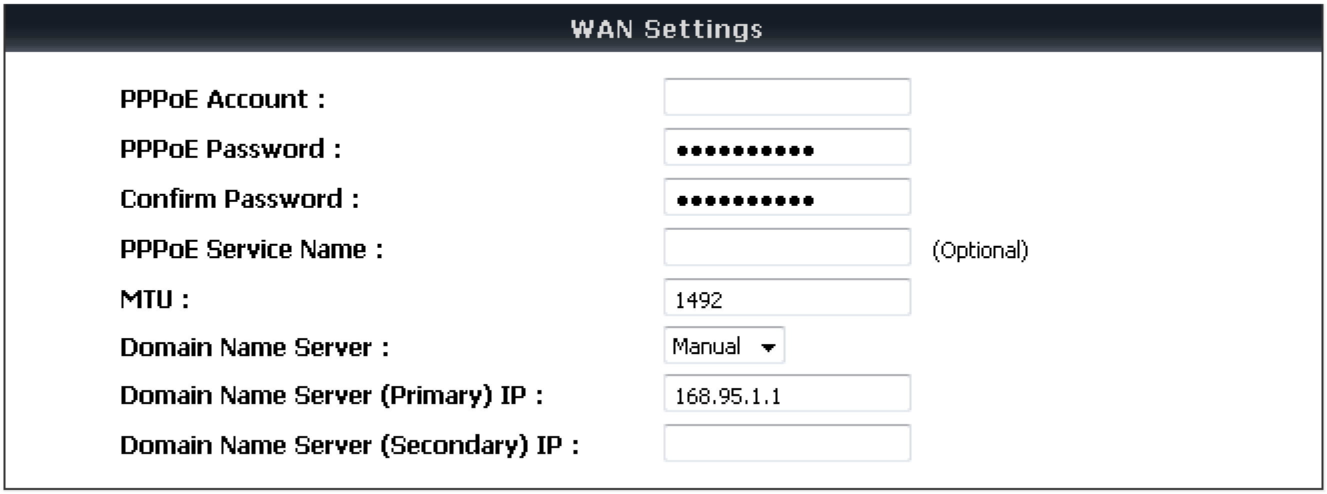

**DHCP:** Select this option if your ISP (Internet Service Provider) provides you an IP address automatically. Cable modem providers typically use dynamic assignment of IP Address. The Host Name field is optional but may be required by some Internet Service Providers.

General Settings → WAN



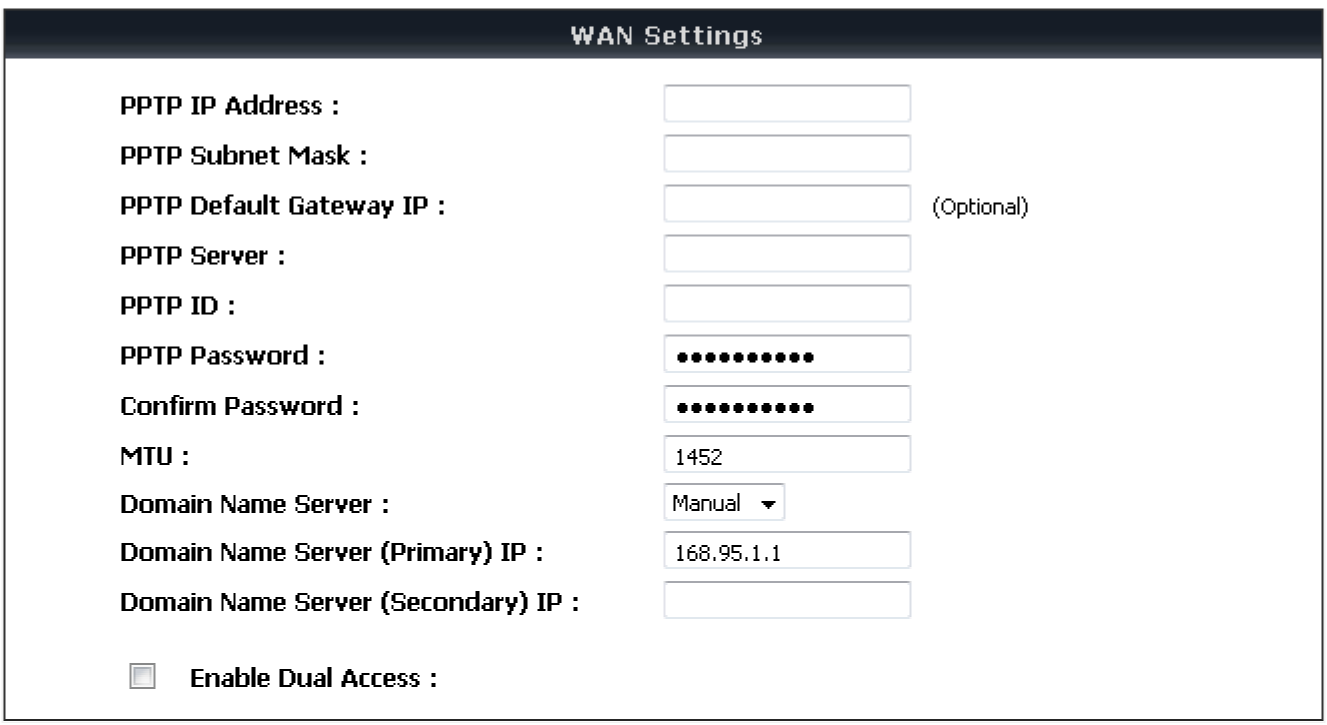

**Static IP:** Select this option if your ISP (Internet Service Provider) provides you a Static IP address. Enter the **IP address**, **Subnet Mask** and **Default Gateway IP**.

General Settings → WAN



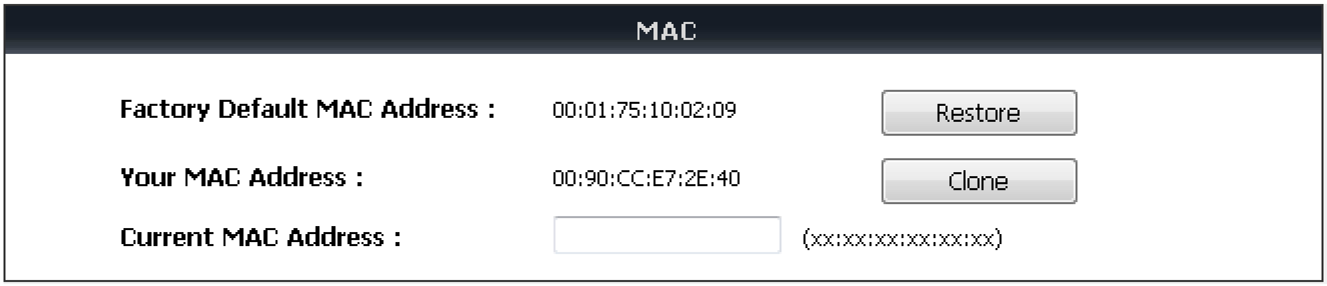

**PPPoE:** Select this option if your ISP requires you to use a PPPoE (Point-to-Point Protocol over Ethernet) connection. Enter the **PPPoE Account**, **PPPoE Password** and re-enter Password to confirm.

General Settings → WAN




**PPTP:** Point-to-Point Tunneling Protocol (PPTP) is a WAN connection. Enter the **IP Address**, **Subnet mask**, **PPTP Server**, **PPTP ID** and **Password**.

General Settings → WAN




**Factory Default MAC Address:** The original MAC address of the VoIP Gateway.

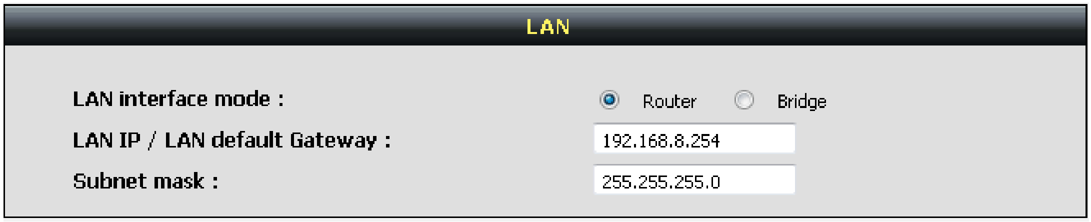
**Your MAC Address:** It is left blank as you log-in via the WAN port.

**Current MAC Address:** It shows the current MAC Address if you ever used the different MAC address from Factory Default MAC Address. You can click **Clone** to automatically copy the MAC address of the Ethernet Card installed in the computer used to configure the device.

**Note:** This is only necessary to fill the field if required by your ISP.

## 3-2-2 LAN

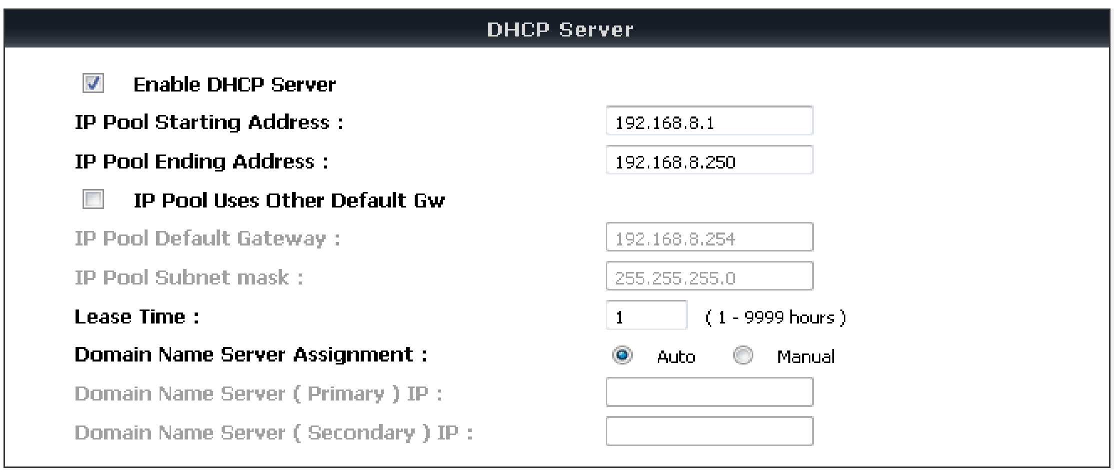
General Settings → LAN




**LAN Port Address:** Enter the LAN IP address of the VoIP Gateway. It is also the default gateway for DHCP clients.

**Subnet Make:** Enter the subnet mask for DHCP clients.

General Settings → LAN




**Enable DHCP Server:** This variable is to assign the IP address for the devices connected to LAN port of the VoIP Gateway.

**IP Pool Starting Address:** Enter the starting IP address for the DHCP server's IP assignment.

**IP Pool Ending Address:** Enter the ending IP address for the DHCP server's IP assignment.

**IP Pool Uses Other Default Gw:** Check the box to assign different default gateway for DHCP clients.

**IP Pool Default Gateway:** Enter the new default gateway that is different from LAN IP of the VoIP Gateway.

**IP Pool Subnet mask:** Enter the new subnet mask.

**Lease Time:** Enter the length of time for the IP lease.

**Domain Name Server Assignment:** Select **Auto** or **Manual** to get the IP address of Domain Name Server assigned by ISP or manually.

**Domain Name Server IP:** Enter the primary and secondary IP address of Domain Name Server if Domain Name Server Assignment is **Manual**. Otherwise, the VoIP Gateway will not be able to access hosts using hostnames instead of IPs.

General Settings → LAN

**LAN Port Control**

| Port | Enable Port | Incoming Rate Limit | Outgoing Rate Limit | Router/Bridge | VLAN ID |
|---|---|---|---|---|---|
| LAN Port 1 | ☑ | Full 0 kbps | Full 0 kbps | Router | 0 |
| LAN Port 2 | ☑ | Full 0 kbps | Full 0 kbps | Router | 0 |

**Enable Port:** Tick the box to enable LAN Port

**Incoming Rate Limit:** Use the drop-down menu to select the proper rate limit for the specific LAN port. The flow is from LAN to WAN, and the rate limit can not exceed the real upstream bandwidth.

**Outgoing Rate Limit:** Use the drop-down menu to select the proper rate limit for the specific LAN port. The flow is from WAN to LAN, and the rate limit can not exceed the real downstream bandwidth.

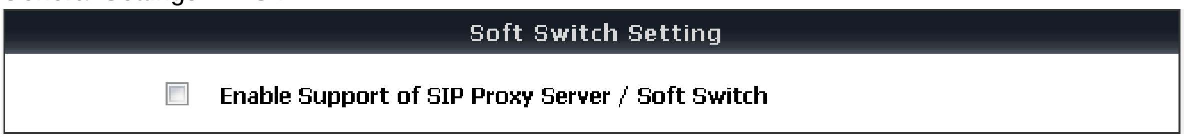
**Router / Bridge:** Select the VoIP GATEWAY serving as a **Router** with NAT or **Bridge** between WAN port and LAN port without NAT.

**VLAN ID:** This option is configurable after enable VLAN tagging at "WAN" page and set the LAN port to be bridge mode. The traffic at LAN port is un-tagged and will be tagged at WAN port.

## 3-2-3 SIP

As there are various Proxy Server providers, according to RFC standard, it has designed the gateway to be compatible with them. If any registration problem occurs, please consult your Internet telephony Server Provider.

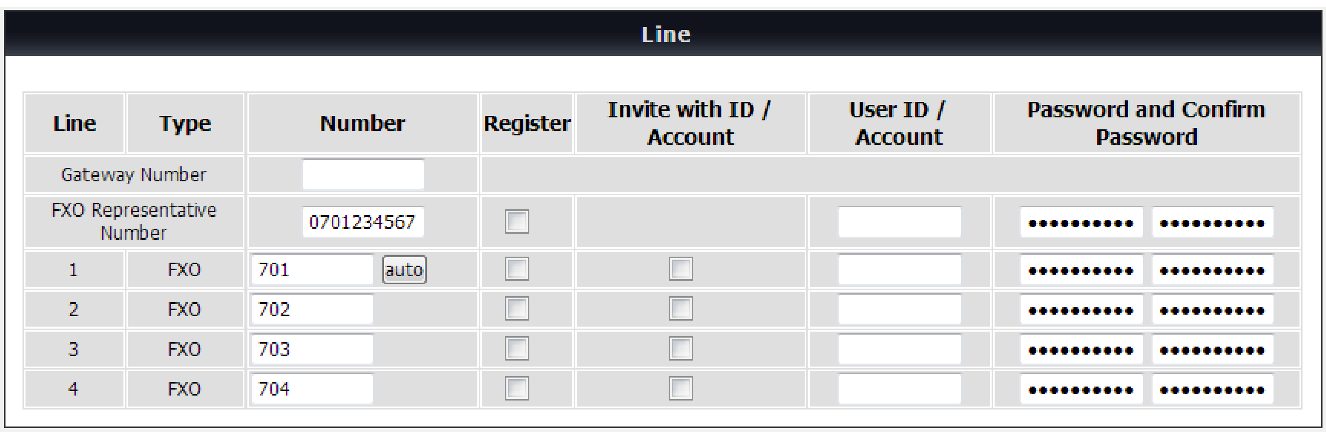
General Settings → SIP




**Enable Support of SIP Proxy Server / Soft Switch:** Check the box to register the VoIP Gateway with SIP proxy server or soft switch.

General Settings → SIP

**<u>FXO Representative Number registers to Proxy:</u>**



Line

| Line | Type | Number | Register | Invite with ID / Account | User ID / Account | Password and Confirm Password |
|---|---|---|---|---|---|---|
| Gateway Number | | | | | | |
| FXO Representative Number | | 0701234567 | | | | •••••••••• •••••••••• |
| 1 | FXO | 701 auto | | | | •••••••••• •••••••••• |
| 2 | FXO | 702 | | | | •••••••••• •••••••••• |
| 3 | FXO | 703 | | | | •••••••••• •••••••••• |
| 4 | FXO | 704 | | | | •••••••••• •••••••••• |

**Number:** Enter the representative number for Line 1-4. If the VoIP Gateway is configured to register with SIP proxy server, all the lines are using this number to call through SIP proxy server. It is the Caller ID for the called party when you make a VoIP call. If you register the VoIP Gateway to a SIP proxy server, then it should be the number that provided by SIP proxy server.

**Register:** Check the box to register with SIP proxy server.

**User ID/Account**: User ID/Account are usually the same as Number from most SIP proxy severs.

**Password:** Enter password and re-enter to confirm.

**Note:** Please ensure if your VoIP Service Provider allows one account for multi-port using.

General Settings → SIP

**<u>Each line registers to Proxy independently:</u>**

| Line | | | | | | |
|---|---|---|---|---|---|---|
| **Line** | **Type** | **Number** | **Register** | **Invite with ID / Account** | **User ID / Account** | **Password and Confirm Password** |
| Gateway Number | | | | | | |
| FXO Representative Number | | | | | | •••••••••• •••••••••• |
| 1 | FXO | 0701234561 auto | | | | •••••••••• •••••••••• |
| 2 | FXO | 0701234562 | | | | •••••••••• •••••••••• |
| 3 | FXO | 0701234563 | | | | •••••••••• •••••••••• |
| 4 | FXO | 0701234564 | | | | •••••••••• •••••••••• |

**Number:** Enter the number, text or number and text in this field. It is the Caller ID for the called party when you make a VoIP call. If you register the VoIP Gateway to a SIP proxy server, then it should be the number that provided by SIP proxy server. Number and User ID/Account are usually the same from most SIP proxy severs. Each line has a number. And the number of each line is not reiteration.

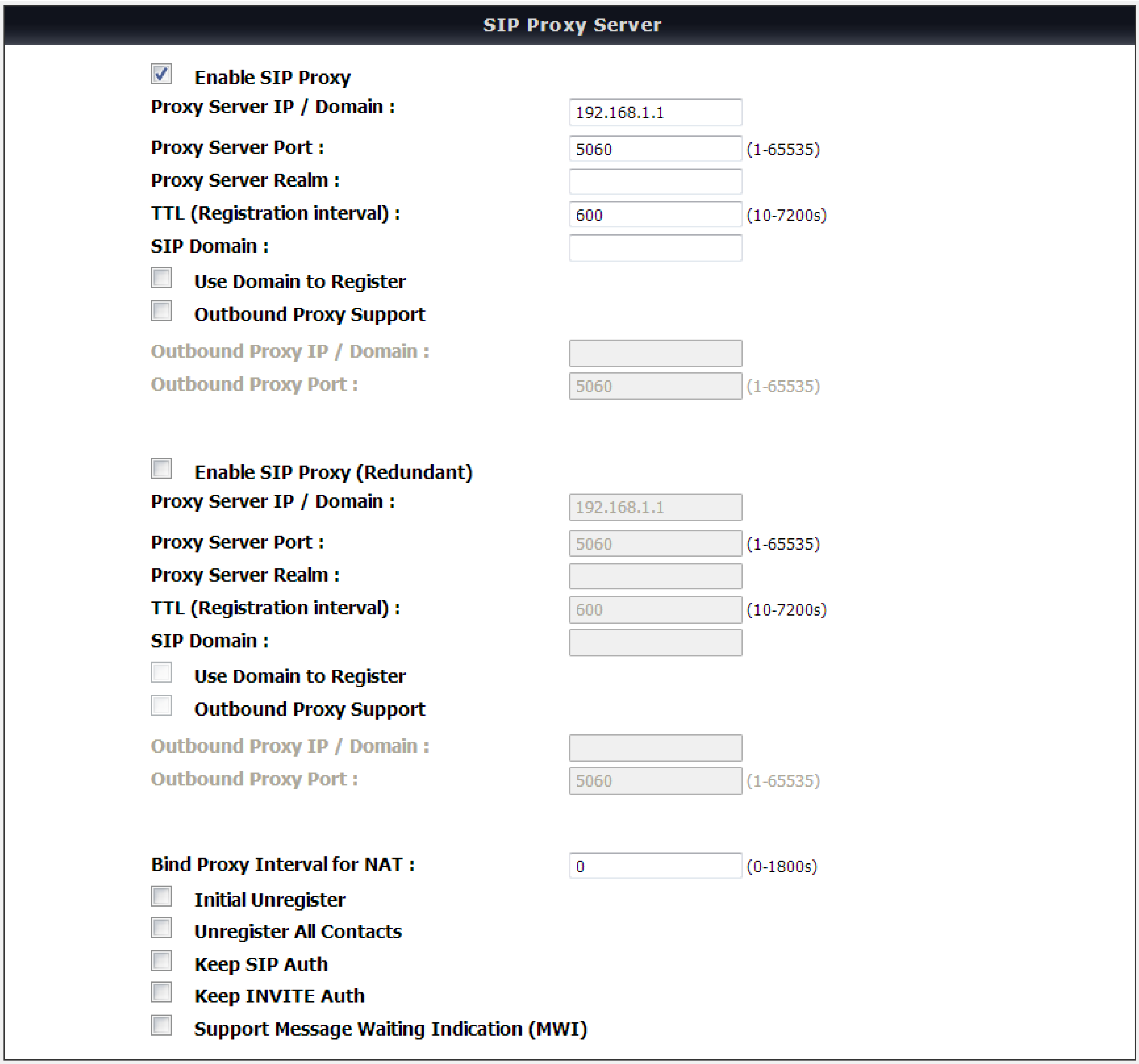
**Register:** Check the box to register with SIP proxy server.

**Invite with ID / Account:** Check the box to call through SIP proxy server without registration. It is always ticked when Register is also ticked. Most VoIP Service Providers will interdict the connection without registration.

**User ID/Account**: User ID/Account are usually the same as Number from most SIP proxy severs.

**Password:** Enter password and re-enter to confirm.

General Settings → SIP




**Proxy Server IP/Domain:** Enter the IP address or URL (Uniform Resource Locator) of SIP proxy server or soft switch.

**Proxy Server Port:** Enter the SIP proxy server's listening port for the SIP in this field. Leave this field to the default if your VoIP Service Provider did not give you a server port number for SIP.

**Proxy Server Realm:** Enter the realm for SIP proxy server. It is used for authentication in a SIP server. In most cases, the VoIP Gateway can automatically detect your SIP server realm. So you can leave this option blank. However, if your SIP server requires you to use a specific realm you can manually enter it in.

**TTL (Registration interval) [10-7200 s]:** The interval for VoIP Gateway re-report to SoftSwitch.

**SIP Domain:** Enter the SIP domain provided by your VoIP Service Provider. (Note some SIP proxy servers might not require this.) If you enable "Uses Domain to Register", the VoIP Gateway will register to the SIP proxy server with the domain name you filled in. Otherwise, the VoIP Gateway will register to a SIP proxy server with the IP it resolves.

**Use Domain to Register:** Check the box to use Domain to register with SIP proxy server. The VoIP Gateway is registered to the SIP proxy server with IP address if un-ticked.

**Note: Proxy Server Realm**, **SIP Domain** and **Use Domain to Register** are the parameters provided by VoIP Service Provider. If you fail to make a call, please contact your VoIP Service Provider.

**Outbound Proxy Support:** Check the box to send all SIP packets to the destined outbound proxy server. An outbound proxy server handles SIP call signaling as a standard SIP proxy server would do. Further, it receives and transmits phone conversation traffic (media) between two communication parties. This option tells the VoIP Gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote VoIP device. This helps VoIP calls to pass through any NAT protected network without additional settings or techniques. Please make sure your VoIP Service Provider supports outbound proxy services before you enable it.

**Outbound Proxy IP/Domain:** Enter the outbound proxy's IP address or URL.

**Outbound Proxy Port:** Enter the outbound proxy's listening port.

**Bind Proxy Interval for NAT:** Check the box to keep the binding exist by sending packets when the VoIP Gateway is behind a NAT and SIP proxy server is not able to keep the binding.

**Initial Unregister:** Check the box to send an unregistered message initially by the VoIP Gateway and then it will perform a general register process.

**Unregister All Contacts:** DVG-6004S sends un-register request to SoftSwitch which the contact field filled with a start sign(*) to un-register all FXO in this DVG-6004S.

**Keep SIP Auth:** VoIP GATEWAY keeps the last register SIP MD5 authentication information and re-use it for next register request.

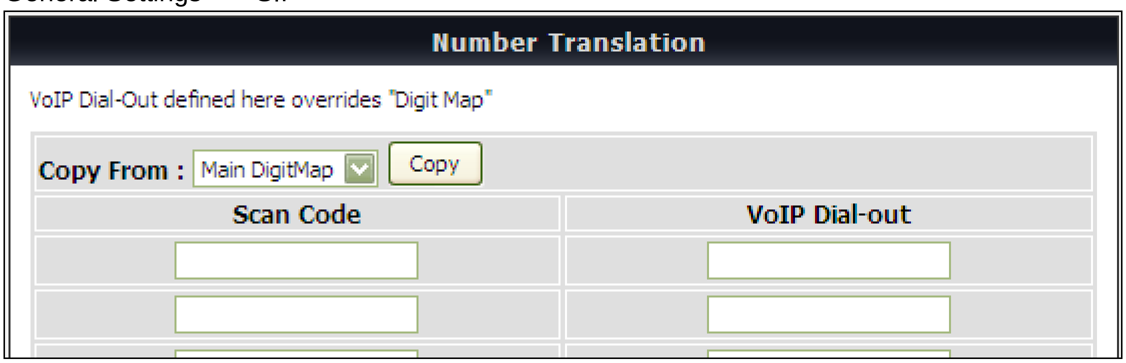
**Support Message Waiting Indication (MWI):** It is used to enable/disable Message Waiting Indication. It is available only when Voice Mail Service is available from the VoIP Service Provider.

General Settings → SIP


**Enable P-Assert:** Check the box to enable the caller ID protection.

**Privacy Type:** It is used to disguise the caller ID when queried via an ITSP/Third-Party Assertion. The Privacy Type includes ‘user’, ‘header’, ‘session’, ‘none’, ‘critical’, ‘id’ and ‘history’.

General Settings → SIP



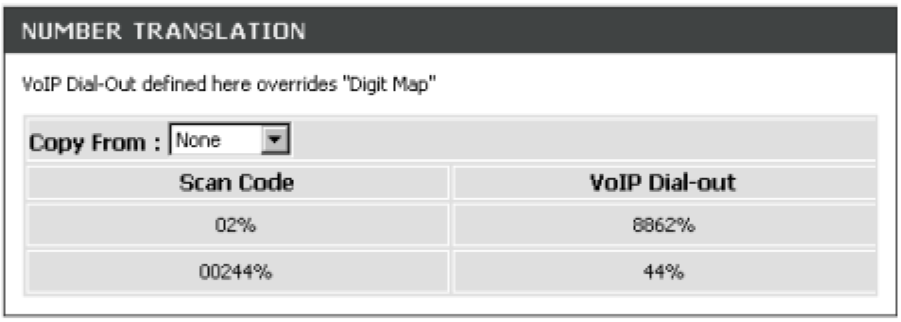

The rule of dialing of inviting to VoIP Service Providers may vary. That is, you have to configure different Digit Map for different VoIP Service Providers. In this filed, you can configure individual dialing plan for each VoIP Service Provider. The following examples introduce some cases. For general configuration, refer to **Digit Map** page. **Note: Press “Add” to add an entry. Don’t forget to press “Apply” which in the above of Number Translation.**

**For example** (Example in Taiwan)**,**

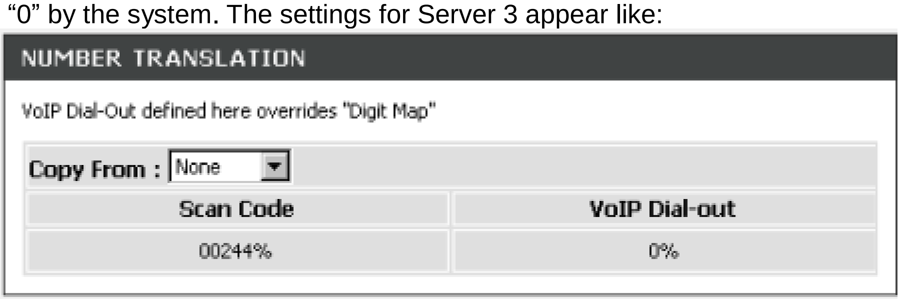
If Server 1 is local VoIP Service Provider you can refer to **Digit Map** page for general settings.

If Server 2 is global VoIP Service Provider (VoIP STUN, free to dial to some cities free charge) you can set individual dialing plan for VoIP STUN in **Number Translation** field. **Scan Code** can be your dialing custom, and **VoIP Dial-out** is the number on the basis of the dialing rule needed by VoIP STUN. Its dialing rule is Country code + Area Code + phone number. When you make calls to Taipei through VoIP STUN, you don't change the dialing custom, just dial 02xxxxxxxx, and the system will change the number from 02xxxxxxx to 8862xxxxxxxx. The same rule is for #2. When you make calls to UK via VoIP STUN, you'll dial 00244xxxxxx, and the system will change it to 44xxxxxx.

The settings for Server 2 appear like:




If Server 3 is a VoIP Service Provider in UK, you can set individual dialing plan in **Number Translation** field. As you make calls to UK through this VoIP Service Provider, "Country code" should be removed and plus "0" by the system. The settings for Server 3 appear like:



## 3-2-4 SIP Advanced

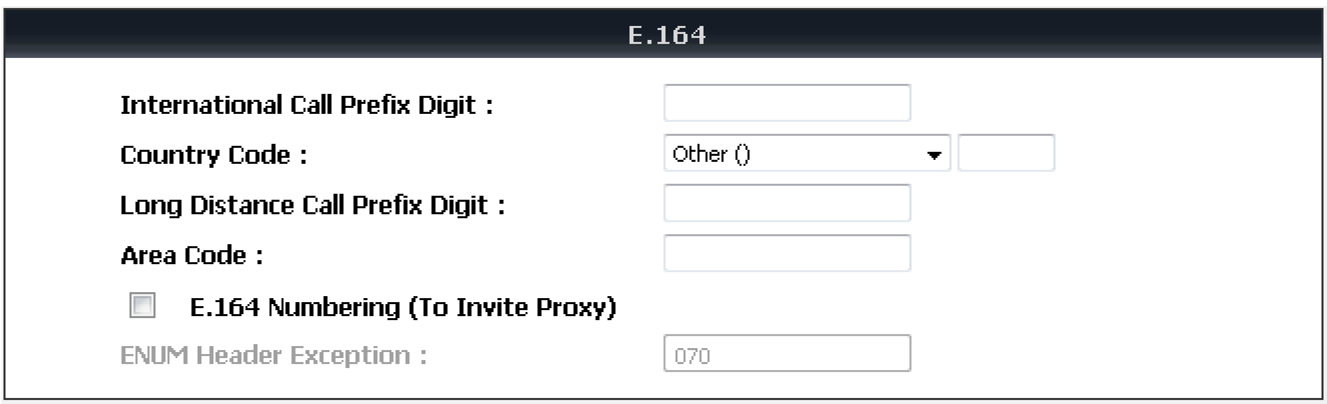
General Settings → SIP Advanced


**Listen Port UDP:** Enter the VoIP GATEWAY's listening port in this field. Leave it as default settings, unless it conflicts with ports used by other device in your network.

**RTP Starting Port UDP:** Enter the starting port number or transmitting voice data among VoIP devices. Each line requires 2 ports.

**SIP Transport Protocol:** UDP or TCP

General Settings → SIP Advanced




**International Call Prefix Digit:** Enter the International call prefix.

**Country Code:** Select the desired country code from the drop-down menu or enter the country code if **Other** is selected.

**Long Distance Call Prefix Digit:** Enter the long-distance prefix digit for making a long-distance call.

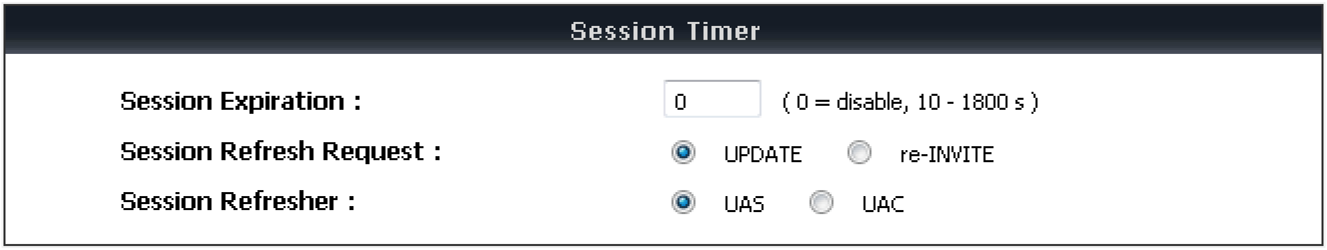
**Area Code:** Enter the area code.

**E.164 Numbering (To Invite Proxy):** This variable is followed the E.164 rule, but it depends on the SIP proxy server. Click the check box to send the number following the E.164 rule by the VoIP Gateway.

**ENUM Header Exception:** Enter the prefix number that the VoIP Gateway sends the number without followed the E.164 rule.

> **Note:** E.164 Numbering depends on the proxy. If you fail to make a call, please contact your VoIP Service Providers.

General Settings → SIP Advanced

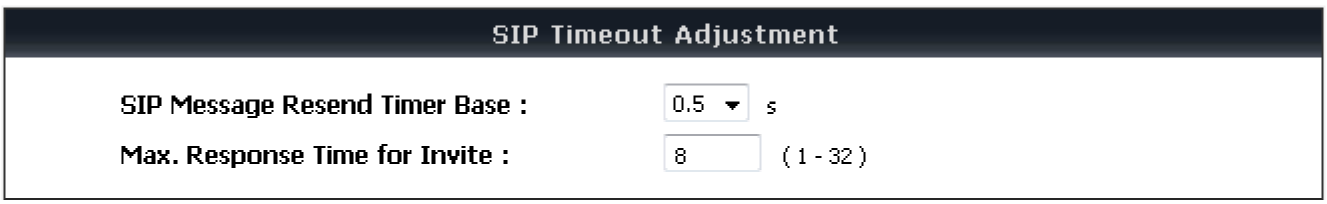



**Session Expiration:** This field will set the time that the VoIP Gateway will allow a SIP session to remain die (without traffic) before dropping it.

**Session Refresh Request:** Select **UPDATE** or **re-INVITE** to send refresh requests to the Server.

**Session Refresher:** This determines which side of an expired call session will initiate the session refresh. uac – specifies that the Caller side will initiate the session refresh. uas – specifies that the Call receiver (the "Callee") will initiate the session refresh.

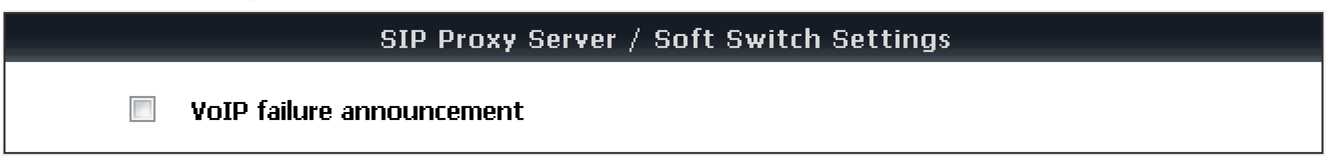
General Settings → SIP Advanced




**SIP Message Resend Timer Base:** Select the resend timer base from the drop-down menu if response is not received within the base time. The sequence of sending is like "base time" * 2, and send again at "base time" *2 *2. The maximum resend time is four seconds. Resend action will stop when the total resend time has reached 20 seconds.

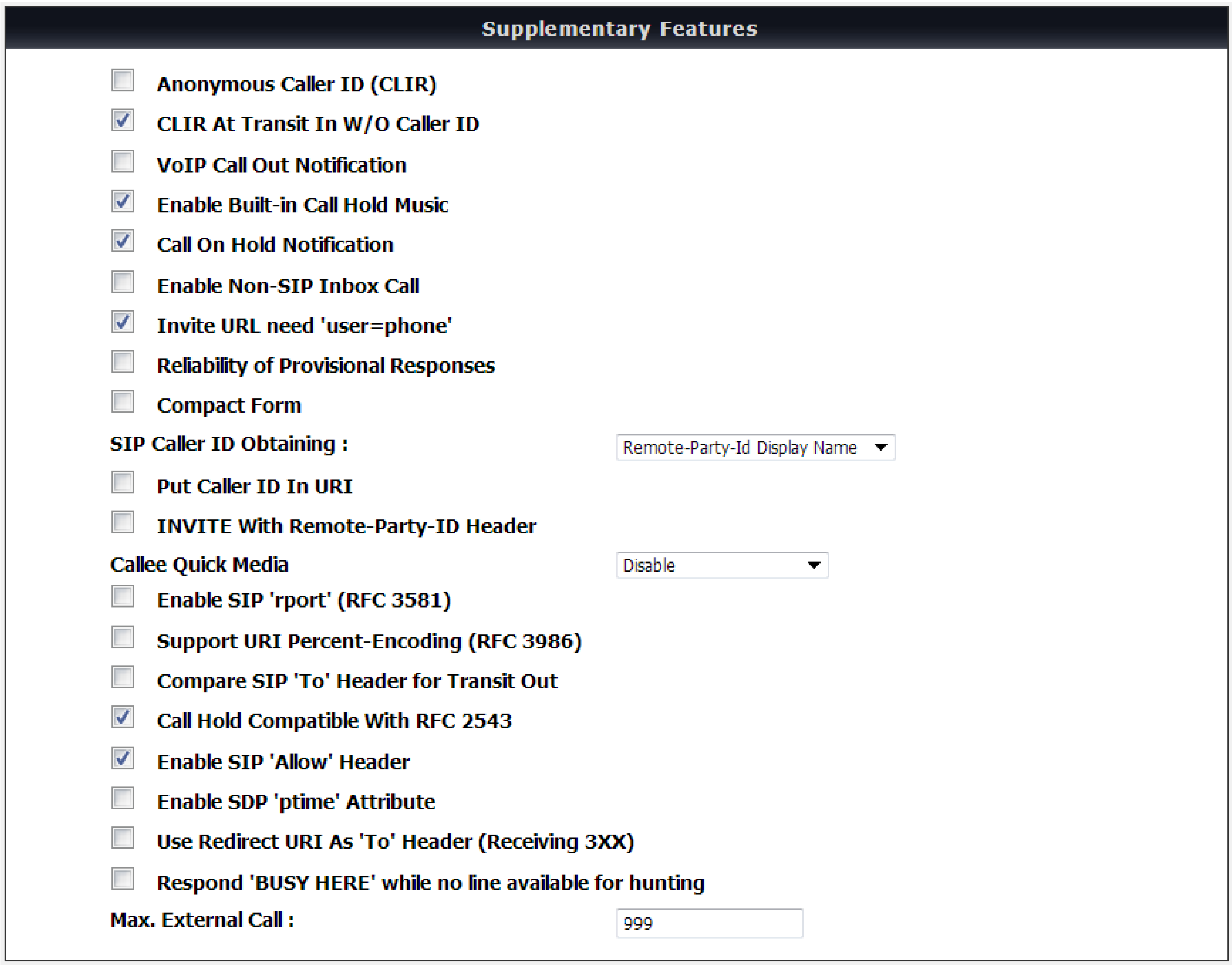
**Max. Response Time for Invite:** Enter the maximum response time for INVITE packet. When the destination does not reply within the set time, the call is failed.

General Settings → SIP Advanced




**VoIP failure announcement:** Check the box to play a voice announcement if the VoIP Gateway fails to register to the SIP proxy server while FXO is off-hook.

General Settings → SIP Advanced




**Anonymous Caller ID (CLIR):** Check the box to lock the delivery of the Caller ID to the called party.

**VoIP Call Out Notification:** Check the box to enable the function of playing a tone to notify user that the call is through VoIP.

**Enable Built-in Call Hold Music:** Check the box to enable the function of playing music when receiving Call Hold request.

**Call On Hold Notification:** FXO will send alert to phone set as users hang up if there is a call still held in another line.

**Enable Non-SIP Inbox Call:** Check the box to make local calls. Local Call here means the call does not go through the Internet and if the dialed number is the extension of other line. You can un-check it to configure as all calls go through the Internet.

**Invite URL need 'user=phone':** Check the box to add 'user=phone' as a hint that the part left to the '@' sign is actually a phone number.

**Reliability of Provisional Responses:** Check the box to send a PRACK request during the progress of the request processing. Reliability of Provisional Responses is to ACK at every SIP packet. With this method, SIP packet will act like TCP, ie. every packet sent will receive an ACK to make sure that packet sent has been received by other peer.

**Compact Form:** Check the box to represent common header field names in an abbreviated form. This may be useful when SIP message is too large to be carried on and recognized by the user agent.

**SIP Caller ID Obtaining:** Select the part of the SIP packet from the VoIP Gateway to obtain Caller ID. There are several places where the Caller ID is located.

**Remote-Party-ID Display Name -** It is located at SIP → Remote-Party-ID → Before [<sip:]
**Remote-Party-ID User Name -** It is located at SIP → Remote-Party-ID → After [<sip:], Before [@]
**From-Header Display Name -** The standard way is in SIP → Message Header → From → SIP Display info.

**From-Header User Name -** It locates at SIP -> Message Header -> From -> SIP from address before [@].

**Put Caller ID In URI:** This feature is to put Caller ID in URL. The Caller ID is located in SIP → Message Header → After [From:], Before [<sip:] by default settings. It will be located in SIP → Message Header → After [<sip:], Before [@]if ticked.

**INVITE With Remote-Party-ID Header:** Check the box to comprise the information of Remote-Party-ID in the message header of INVITE. Different format of INVITE header might cause the call not to be connected. Please consult with your VoIP Service Provider before enabling it.

**Callee Quick Media:** DVG-6004S will send RTP to remote party immediately as user answer an inbound call.

**Support URI Percent-Encoding(RFC 3986):** Check the box to encode/decode the letters of the basic Latin alphabet, digits, and a few special characters which follow RFC 3986.

**Call Hold Compatible With RFC 2543:** It is used to set the procedure of Call Hold being compatible with RFC 2543.

**Enable SIP 'Allow' Header:** It is used to put "Allow" in SIP packets. The Allow header field lists the SIP requests supported by ITA when ticked.

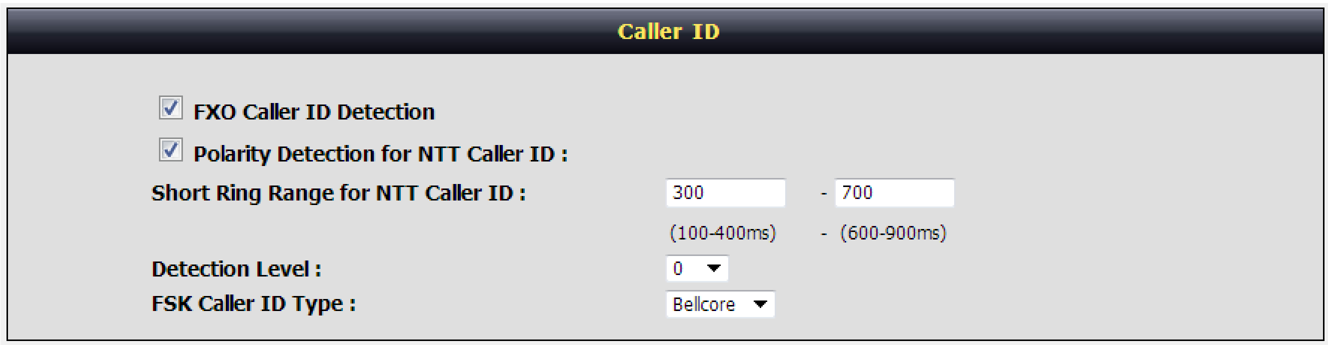
**Enable SDP 'ptime' Attribute:** It is used to put "ptime" in SDP packets when ticked.

**Use Redirect URI As 'To' Header (Receiving 3XX):** It is used to change the content of 'To' header field when receiving 3XX.

**Respond 'BUSY HERE' while no line available for hunting:** It is used to reply 'BUSY HERE' to the calling party while no line is available for hunting.

## 3-2-5 Caller ID

General Settings → Caller ID




**FXO Caller ID Detection:** Used to detect the Caller ID delivered from the PSTN to the FXO port. When enabled, the Caller ID detected on the FXO port will be sent to the SIP Proxy Server on transit in (dialing out) calls.

**Detection Level:** If FXO can't detect Caller ID, try to adjust it until DVG can caller ID.

**FSK Caller ID Type:** Either Bellcore, ETSI or NTT could be selected.

## 3-2-6 Hot Line

General Settings → Hot Line

**Line**

| Line | Enable | Type | Hot Line | Hot Line No. | Warm Line (Hot Line Delay) [0=disable,0-60s] | Dial-Out Prefix | FXO Line Default Dial-Out |
|---|---|---|---|---|---|---|---|
| 1 | ☑ | FXO | ☐ | | 0 | | |
| 2 | ☑ | FXO | ☐ | | 0 | | |
| 3 | ☑ | FXO | ☐ | | 0 | | |
| 4 | ☑ | FXO | ☐ | | 0 | | |

**Enable:** Tick the check box to enable a line. If some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to the line.

**Hot Line:** Check to direct the call automatically to a pre-configured destination without any action when the FXO is off-hook. (ie. as the user picks up the phone). When the FXO is under Hot Line mode, no other phone numbers can be dialed.

**Hot Line No.:** Enter the number for pre-defined destination.

**Warm Line:** Enter the time for the call to start with a pause, so the user can dial another number. The call will be automatically directed to the pre-configured destination within timeout period.

**Dial-out Prefix**: It is the number dialed for prefix automatically by FXO when the DVG transit-out calls from VoIP to FXO.

**FXO Line Default Dial-Out:** The number FXO will dial out when it receive an incoming call from VoIP.

## 3-2-7 Line settings

General Settings→ Line settings

**Line Settings**

**Line**

| Line | Type | Listening Volume (3dB per step) | Speaking Volume (3dB per step) | Tone Volume |
|---|---|---|---|---|
| 1 | FXO | 0 All | 0 All | 5 All |
| 2 | FXO | 0 | 0 | 5 |
| 3 | FXO | 0 | 0 | 5 |
| 4 | FXO | 0 | 0 | 5 |

| Line | Type | Flash Time [50-900ms] | Polarity Reversal | PSTN Answer Detection | PSTN Ring OFF Length [1000 - 20000ms] |
|---|---|---|---|---|---|
| 1 | FXO | 600 All | ☐ All | Disable All | 4000 All |
| 2 | FXO | 600 | ☐ | Disable | 4000 |
| 3 | FXO | 600 | ☐ | Disable | 4000 |
| 4 | FXO | 600 | ☐ | Disable | 4000 |

**Listening Volume:** Use the drop-down menu to adjust the hearing (listening) volume.

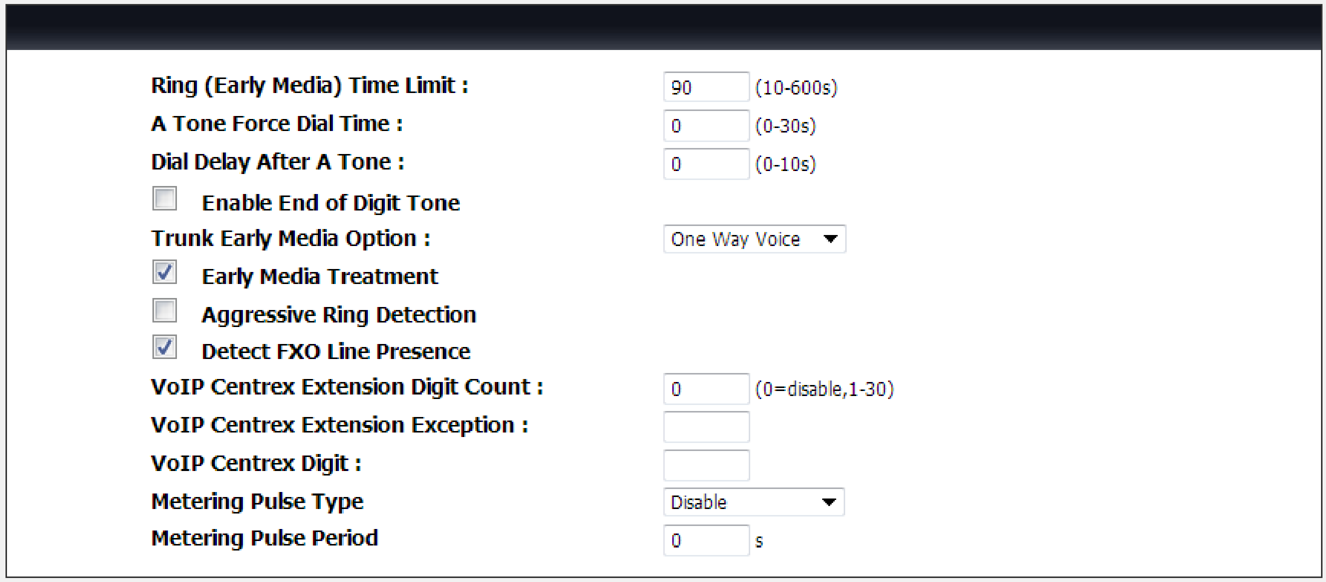
**Speaking Volume:** Use the drop-down menu to adjust the speaking volume.

**Tone Volume:** Use the drop-down menu to adjust the tone volume. It will apply to all tones generated by the VoIP Gateway including Dial Tone, Ring Back Tone and Busy Tone.

**Flash Time:** Enter the minimum flash and maximum flash time for FXO detecting. When the flash signal generated by the phone set is shorter than Min. FXO Hook Flash Time, FXO port will ignore it. As the flash signal generated by the phone set is longer than the Flash Time, FXO port will be on-hook.

**Polarity Reversal:** Check the box to activate the generation of polarity reversal from FXO.

General Settings→ Line settings




**Ring (Early Media) Time Limit[10 - 600secs]:** Enter the timeout to cancel a call if no one answers the phone.

**A Tone Force Dial Time:** A tone detection is used for DVG transit from VoIP to FXO, DVG will dial PSTN called number after it detect DTMF "A". In case DVG doesn't detect A tone, it could also dial after setting time.

**Enable End of Digit Tone:** Check the box to activate the function of playing a "Beep-Beep" tone to notify the user that the call is in progress.

**Trunk Early Media Option:** Early Media refers is used for calling party to hear Color Ring or In-band Ring Back Tone. As DVG transit calls from VoIP to FXO in one stage method, it allows DVG response 183 or 180 before PSTN called part answer calls. This option must work with "**Polarity Reversal"** and **"PSTN Answer Detection"**

> **Both Way Voice:** Use bi-directional early media to obtain information between caller and callee prior to the connection of a call.
> **One Way Voice:** Only the caller can hear early media from the callee prior to the connection of a call.
> **Ring Back:** Playing ring back tone for the caller, indicating that the callee is being alerted prior to the connection of a call.

**Early Media Treatment:** Check the box to send the one-way RTP immediately when a connection with a VoIP service provider has been set up.

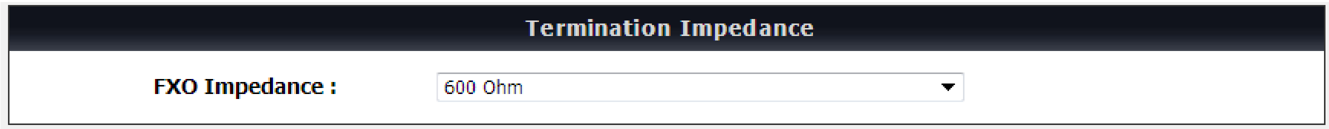
**Aggressive Ring Detection:** Try to enable this option in case DVG can't detect transit-in call from FXO to VoIP.

**Detect FXO Line Presence:** Tick the check box to detect the line presence that FXO port is connected to PBX or a PSTN line. Untick the check box to disable this function if it mis-detects line presence on FXO port while ringing.

**VoIP Centrex Extension Digit Count:** This feature is to enable and set the digit count of VoIP Centrex. The setting "0" zero is to disable this function.

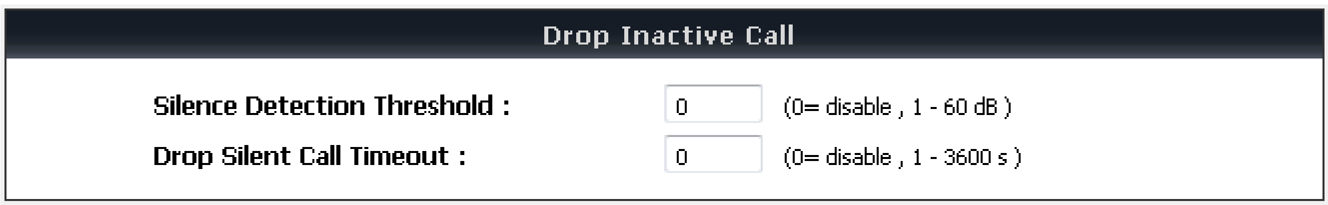
**VoIP Centrex Digit:** Enter the digit for VoIP call. If you dial VoIP Centrex Digit first, the dialing plan is according to the Digit Map; otherwise the VoIP Gateway will send the number which digit count is the same as VoIP Centrex Extension Digit Count.

General Settings→ Line settings




**FXO Impedance:** Select different impedance from the drop-down menu.

General Settings→ Line settings



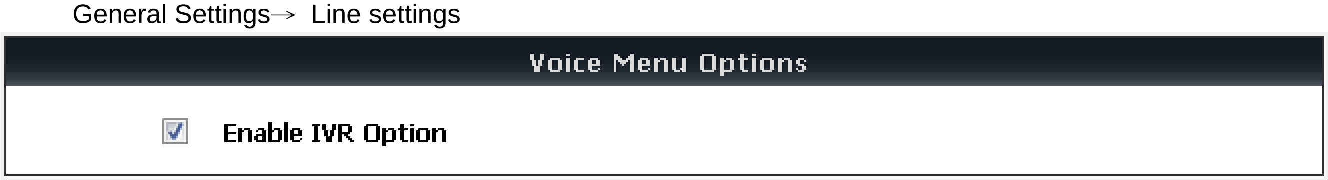

This feature is a call drop standard for a VoIP Gateway to determine whether or not to hang up the phone. The VoIP Gateway will disconnect the call automatically to avoid keeping the line engaged if the detected volume is below the **Silence Detection Threshold** or the time exceeds the **Drop Silent Call Timeout**.

**Silence Detection Threshold:** Enter the threshold (dB) to detect if there is voice coming from RJ-11 interface.

**Drop Silent Call Timeout:** Enter the duration (second) for detecting if there are RTP packets receiving from IP network.

**Note:** Improper values for above settings might cause unexpected automatic disconnection of a call. Default values are recommended.

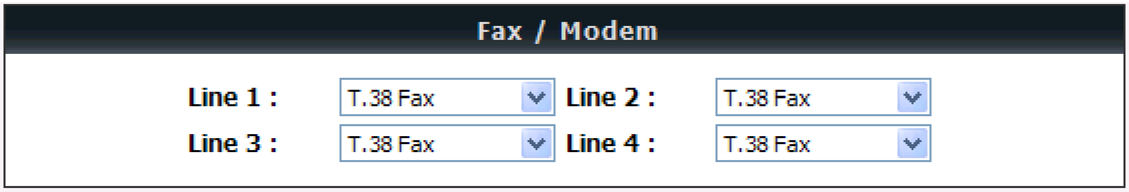
General Settings→ Line settings




**Enable IVR Option:** Check the box to enable IVR function.

## 3-2-8 FAX

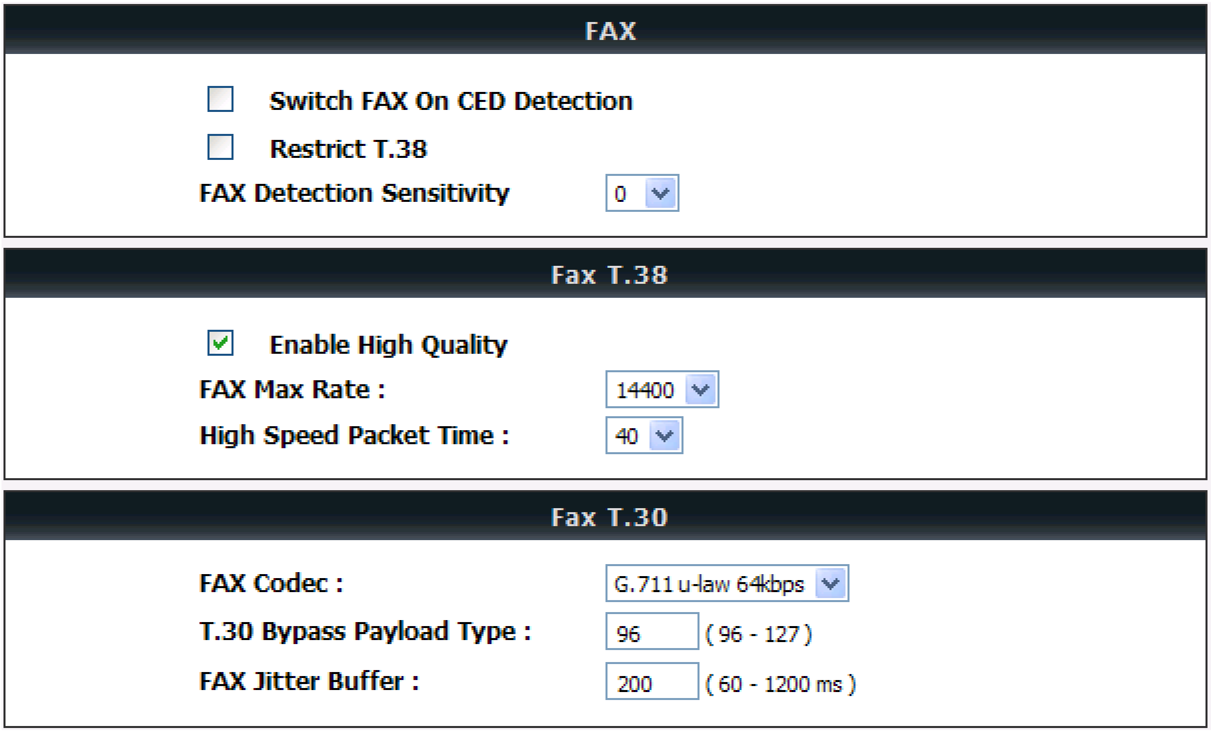
General Settings → FAX




**Disable -** Select it if you are not sending fax, but it is still accepted fax by the VoIP Gateway.
**T.38 Fax -** Select it if you are using T.38 as the protocol for fax transmission. T.38 is used for reliable and efficient facsimile transmission over network. It transmits and receives FAX waveform (relaying) over the codec negotiated during call setup this bandwidth consumed is lowered. T.38 protocol also supports redundancy to get better FAX quality.
**T.30 Fax -** Select it if you are using T.30 as the protocol for fax transmission. It transmit FAX signal as voice thus uncompressed G.711 would be the choice. (G.726 also works but not recommended). Due to this nature, T.30 always requires a SDP change (change of codec within a session, SIP Re-Invite required) after FAX tone detected by the callee. It will consume more network resources and will affect transmission quality. The VoIP Gateway is still able to change the protocol from T.38 to T.30 if the called party uses T.38 for fax transmission.
**T.30 Fax/Modem -** Select it if you use it as the protocol for transmission of fax/modem over IP network.
**T.30 Only -** Select it if you are using G.711 a-law or G.711 u-law for fax transmission. The VoIP Gateway won't accept T.38 for fax transmission.
**T.38 Native -** Select it if you are only using T.38 for fax transmission.
**T.30 V.152 –** As GW detect FAX tone, it will change RTP codec to be T.30 codec directly without sending Re-Invite to change codec.

**Note:** When a fax tone is detected from the call, the VoIP Gateway will automatically switch from voice mode to fax mode. Hence, the fax settings will be temporarily applied to a specific port which detects the fax tones, instead of its default voice settings.

General Settings → FAX




**Switch FAX On CED Detection:** DVG will send FAX Re-Invite immediately as it detect FAX CED tone, that will save handshaking time between FAX machines.

**Restrict T.38:** DVG will reject T.38 Re-invite in case the FAX type contains without T.38.

**FAX Detection Sensitivity:** To set higher value to make DVG to be more sensitive.

**Enable High Quality:** Check the box to increase approximately two times the bandwidth in order to compensate possible loss of packet during transmission and offers a better and reliable fax quality.

**FAX Codec:** Select **G.711 a-law**, **G.711 u-law**, or **G.726** for T.30 from the drop-down menu.

**T.30 Bypass Payload Type:** Fill correct payload type of T.30 bypass method.

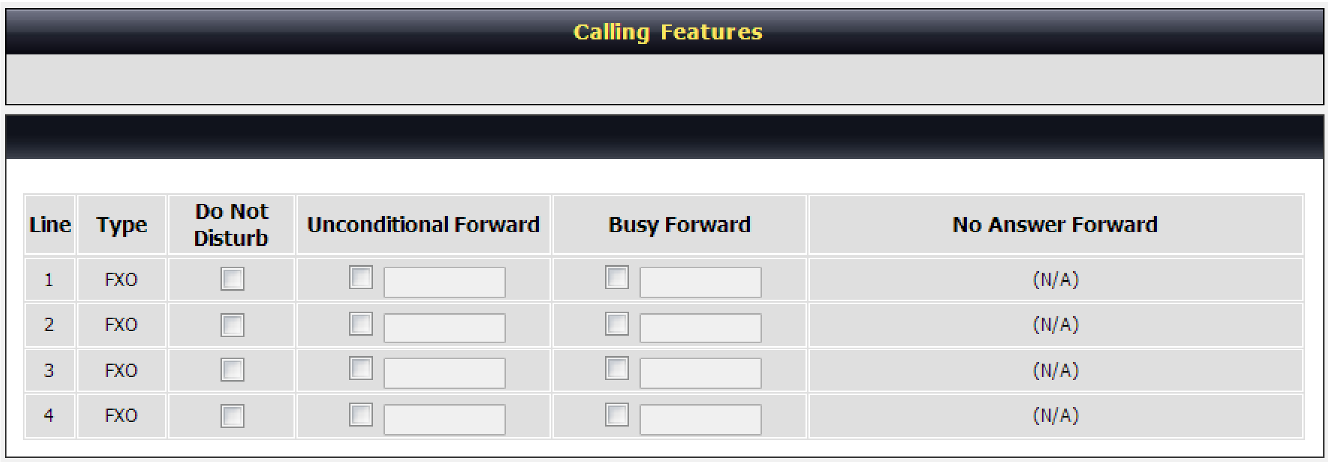
**FAX Jitter Buffer:** Enter the buffer or jitter when receiving packets.

> **Note:** When you send a fax over an IP network, the IP network needs to support fax over IP functionality (either T.38 or T.30). Please consult your VoIP Service Provider for this setting.

| Function | Fax Detection | Content of SDP of re-INVITE | Receive re-INVITE with T.38 |
|---|---|---|---|
| Disable | No | N/A | Accept and change RTP to T.38 |
| T.38 Fax | Yes | re-INVITE with T.38 and T.30 | Accept and change RTP to T.38 |
| T.30 Fax | Yes | re-INVITE with T.30 | Accept and change RTP to T.38 |
| T.30 Fax/Modem | Detect CED only | re-INVITE with T.30 | Accept and change RTP to T.38 |
| T.30 Only | No | N/A | Accept and change RTP to T.38 |
| T.38 Native | Yes | re-INVITE with T.38 | Accept and change RTP to T.38 |
| T.30 V.152 | Yes | There is no Re-Invite for T.30 Bypass mode | Accept and change RTP to T.38 |

## 3-2-9 Calling Features

General Settings → Calling Features


Calling Features

| Line | Type | Do Not Disturb | Unconditional Forward | Busy Forward | No Answer Forward |
|---|---|---|---|---|---|
| 1 | FXO | | | | (N/A) |
| 2 | FXO | | | | (N/A) |
| 3 | FXO | | | | (N/A) |
| 4 | FXO | | | | (N/A) |

**Do Not Disturb:** Check the box to reject (busy tone played) incoming calls.

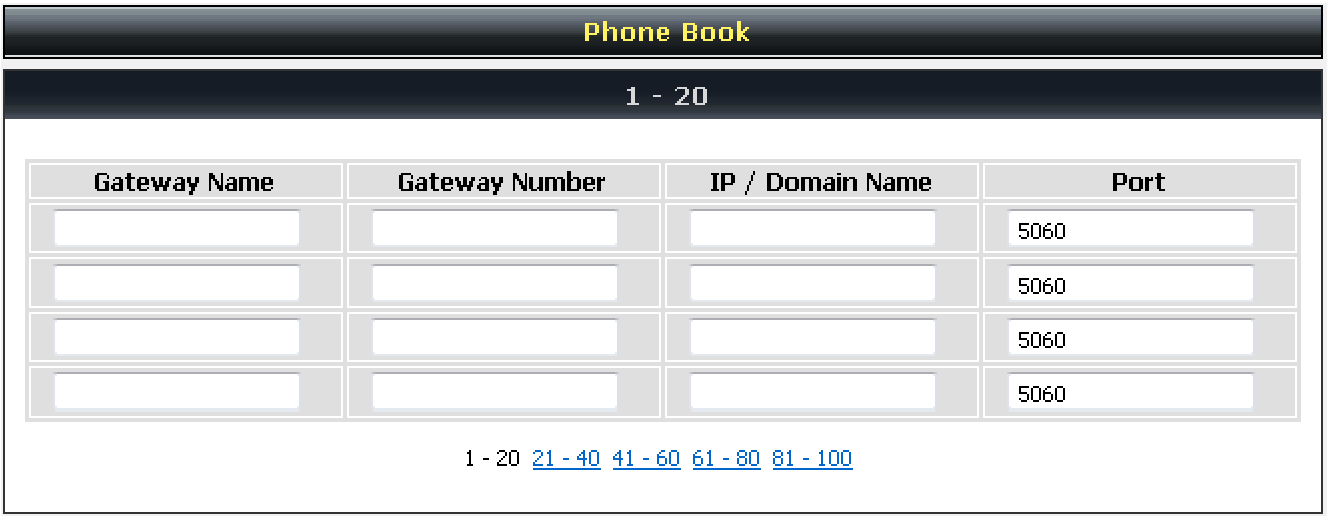
**Unconditional Forward:** Check the box to forward incoming calls to the assigned "Forwarding Number" automatically.

**Busy Forward:** Check the box to forward incoming calls to the "Forward incoming Number" when the line is busy.

## 3-2-10 Phone Book

**Phone Book:** It is used for peer-to-peer communication. Some peer information needs to be added to this section prior to making peer-to-peer calls. You need to enter the phone number and the IP address of the remote peer.

General Settings → Phone Book




**Gateway Name:** Enter the alias of the remote peer.

**Gateway Number:** Enter the phone number of the remote peer.

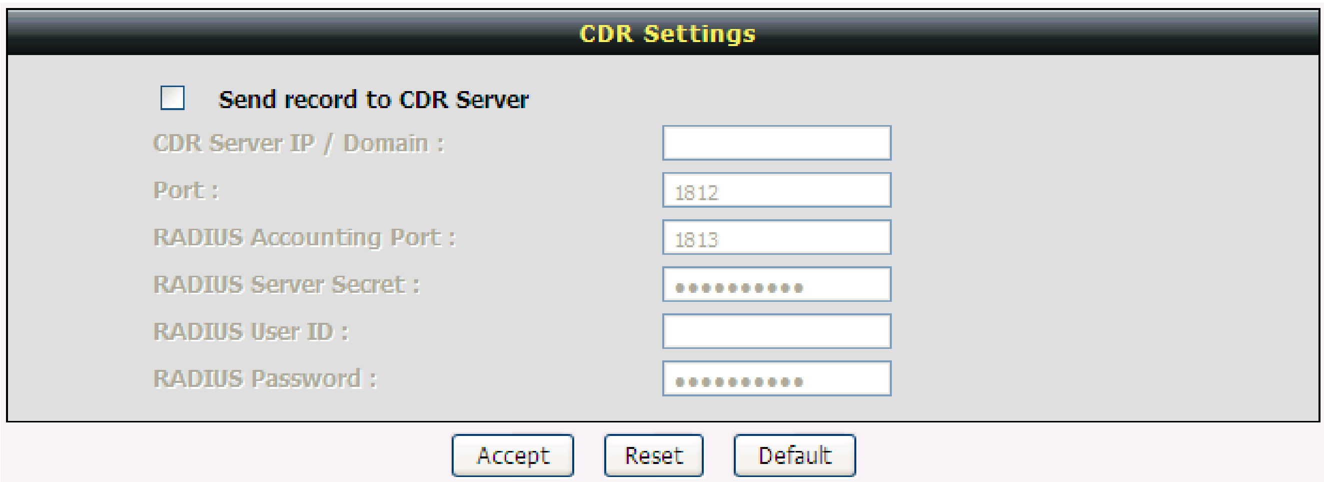
**IP / Domain Name:** Enter the IP address or URL (Uniform Resource Locator) of the remote peer.

**Port:** Enter the listen port of the remote peer.

## 3-2-11 CDR Settings

The user can set up a CDR Server to record call details for every phone call.

General Settings → CDR

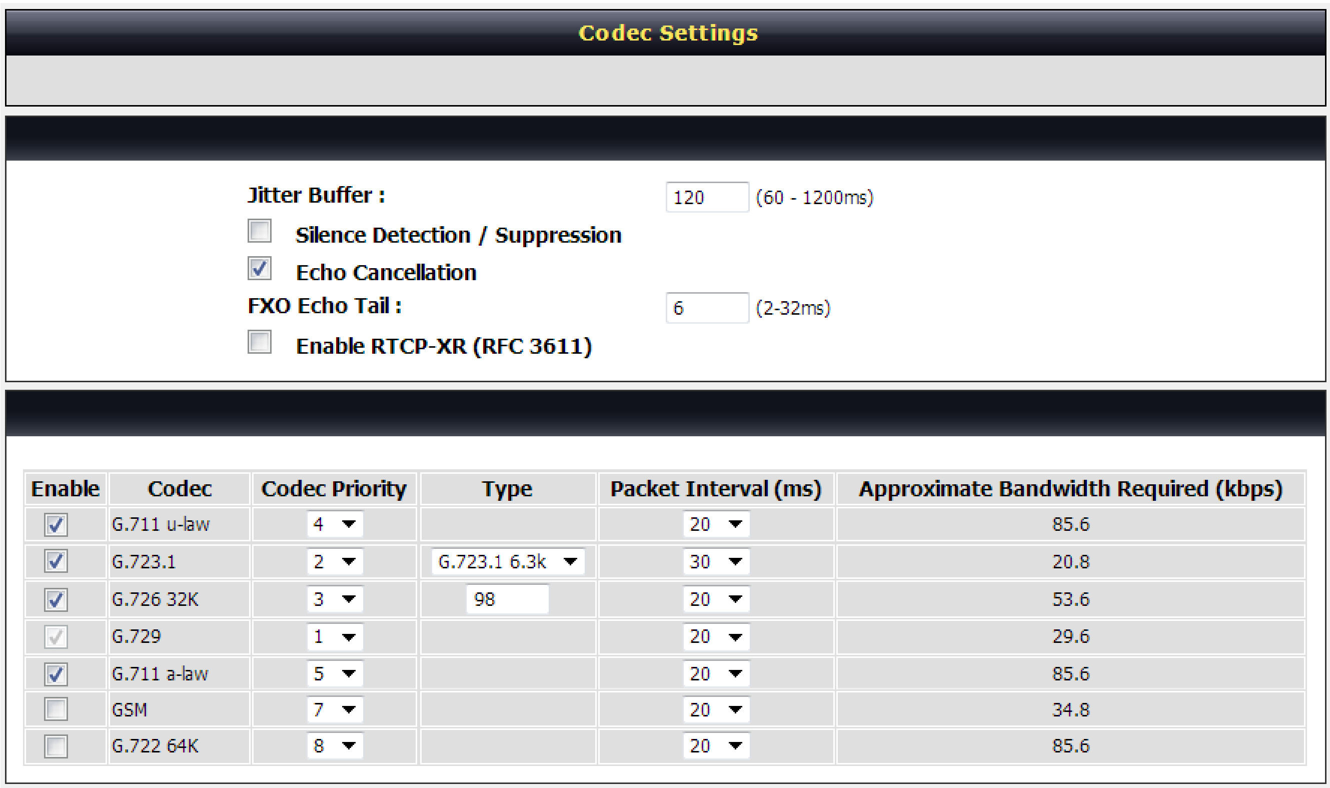



**Send record to CDR Server:** Tick the check box to enable the call detail recording.
**CDR Server IP / Domain:** Enter the IP address of the CDR server.
**Port:** Enter the listen port of the CDR server.
**RADIUS:** Tick the checkbox to enable RADIUS as database and enter the information of RADIUS needed. It includes RADIUS Accounting Port, RADIUS Server Secret, RADIUS User ID and RADIUS Password.

# 3-3 Advanced Settings

## 3-3-1 Codec setting

Advanced Settings→ Codec settings



**Codec Settings**

Jitter Buffer : 120 (60 - 1200ms)

☐ Silence Detection / Suppression

☑ Echo Cancellation

FXO Echo Tail : 6 (2-32ms)

☐ Enable RTCP-XR (RFC 3611)

| Enable | Codec | Codec Priority | Type | Packet Interval (ms) | Approximate Bandwidth Required (kbps) |
|---|---|---|---|---|---|
| ☑ | G.711 u-law | 4 | | 20 | 85.6 |
| ☑ | G.723.1 | 2 | G.723.1 6.3k | 30 | 20.8 |
| ☑ | G.726 32K | 3 | 98 | 20 | 53.6 |
| ☑ | G.729 | 1 | | 20 | 29.6 |
| ☑ | G.711 a-law | 5 | | 20 | 85.6 |
| ☐ | GSM | 7 | | 20 | 34.8 |
| ☐ | G.722 64K | 8 | | 20 | 85.6 |

**Jitter Buffer:** Enter the jitter of receiving packets.

**Silence Detection / Suppression:** Check the box to enable the silence packets and send less voice data (package) during the silent period while talking.

**Echo Canceling:** Check the box to remove echo and improve voice quality during conversation.

**Codec:** Check the box to codec for the VoIP Gateway to support. All codecs are selected and supported by default. You can un-check the box that is not used.

**Codec Priority:** The priority of code for communication.

**Type:** To set dynamic payload types of codec.

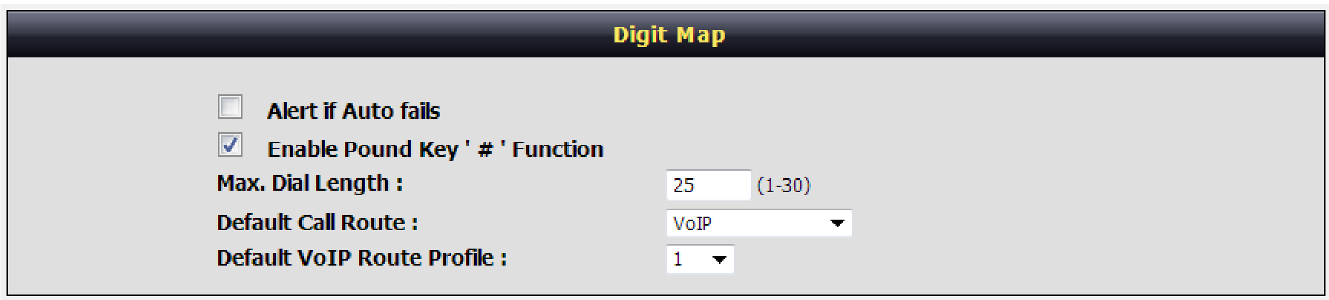
**Packet Interval:** Select the frame size of voice package from different codec. It defines the time interval for the VoIP Gateway to send a RTP packet or voice packet to the receiving side. The smaller the value, the greater the bandwidth takes, and larger values might cause voice delay.

**Approximate Bandwidth Required:** It shows the bandwidth required from different codec and packet interval.

## 3-3-2 Digit Map

Digit Map supports multiple dial plans which help users to arrange least cost route. Each Proxy Server has individual dial plan which combines the original feature of Digit Map and Speed Dial. You can use “?” or “%” in the column of Scan Code and VoIP Dial-out. “?” represents a single digit, and “%” represents a wildcard. The function of the signs is to mapping the numbers between the number received from user and the replaced or modified number for actual dial out. With this function, users can easily add certain leading digits to replace a full set of numbers. There are 50 sets of leading digit entries to choose voice routing interface.

Advanced Settings → Digit Map

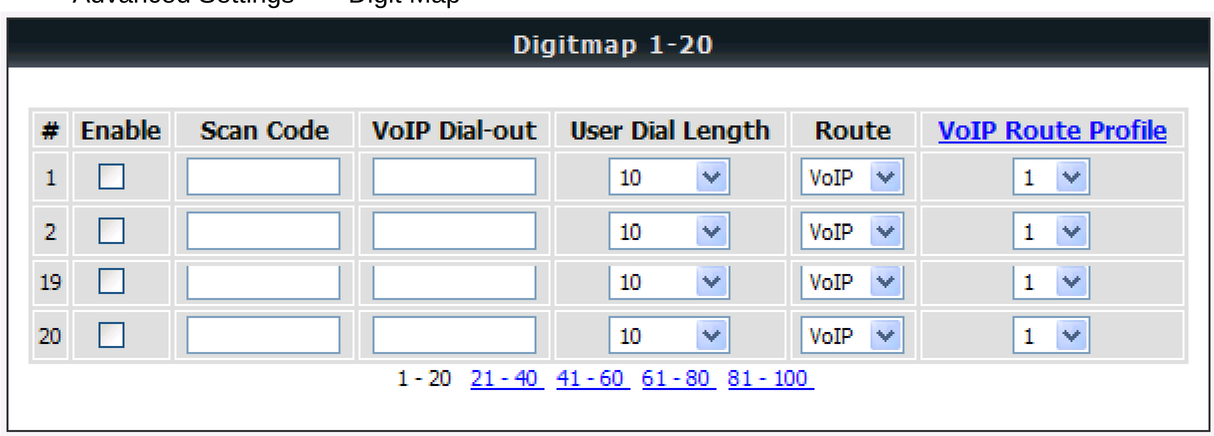



**Enable Pound Key ' # ' Function:** Check the box to treat ‘ # ‘ as a digit and send out with other numbers when dialing. If you un-check the box and ‘ # ‘ is pressed after dialing, it will speed up the phone number detection of the VoIP Gateway.

**Default Call Route:** Select **VoIP** or **Deny** as the default call route for the calls.

**Default VoIP Route Profile:** Enter the Profile ID (ranging from 1-10) for the Default VoIP routing.

Advanced Settings → Digit Map




**Scan Code:** Enter the digits for the VoIP Gateway to scan while user is dialing.

**VoIP Dial-out:** Enter the actual dialing number rule for the VoIP Gateway to call through the Internet.

**User Dial Length:** Enter the total number of digits that user dialed.

**Route:** Select **VoIP** or **Deny** for this entry.

**Methods of Digit Map:**

**Method 1- Single mapping:** Fill a short code into the Scan Code column, and enter the desired phone number into the VoIP Dial-out column.

**For example**,
Scan Code: 09

VoIP Dial-out: 0911888997

User Dial Length: 2

Route: VoIP

VoIP Route Profile: Route # 1

**Digitmap**

| Enable | Scan Code | VoIP Dial-out | User Dial Length | Route |
|---|---|---|---|---|
| ☑ | 09 | 0911888997 | 2 | VoIP |
| ☐ | | | 10 | VoIP |

Pick up the handset and dial 09, the VoIP Gateway will dial 0911888997 and follow Route # 1.

**Method 2- Multi mapping:** Fill the prefix code into the Scan Code column and the format to transfer into the VoIP Dial-out column.

**For example**,
Scan Code: 2???

VoIP Dial-out: 35106???

User Dial Length: 4

Route: VoIP

VoIP Route Profile: Route # 2

**Digitmap**

| Enable | Scan Code | VoIP Dial-out | User Dial Length | Route |
|---|---|---|---|---|
| ☑ | 09 | 0911888997 | 2 | VoIP |
| ☑ | 2??? | 35106??? | 4 | VoIP |
| ☐ | | | 10 | VoIP |

Pick up the handset and dial 2301. The VoIP Gateway will dial 35106301 and follow Route # 2.

**For example**,
Scan Code: 0%
VoIP Dial-out: 1805%
User Dial Length: Disable
Route: VoIP
VoIP Route Profile: Route # 3

**Digitmap**

| Enable | Scan Code | VoIP Dial-out | User Dial Length | Route |
|---|---|---|---|---|
| ☑ | 09 | 0911888997 | 2 | VoIP |
| ☑ | 2??? | 35106??? | 4 | VoIP |
| ☑ | 0% | 1805% | Disable | VoIP |
| ☐ | | | 10 | VoIP |

Pick up the handset and dial 0423456789. The VoIP Gateway will dial 1805423456789 and go through Internet first and follow Route # 3.

**Method 3- Substitution:** It helps you dial to destination that you can not dial by phone. Destination like: test@1.1.1.1. Fill in the number into the **Scan Code** column and enter the desired name into the **VoIP Dial-out** column.

**For example**,
Scan Code: 11
VoIP Dial-out: test
User Dial Length: 2
Route: VoIP
VoIP Route Profile: Route # 1.

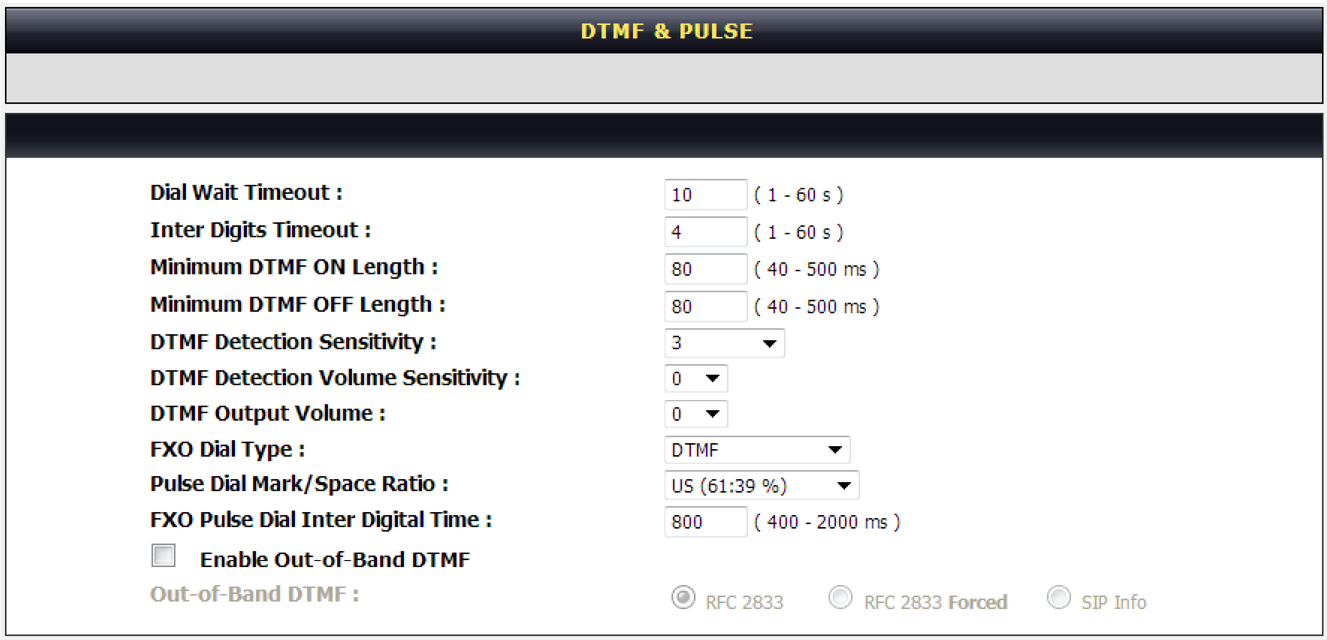
**Digitmap**

| Enable | Scan Code | VoIP Dial-out | User Dial Length | Route |
|---|---|---|---|---|
| ☑ | 11 | test | 2 | VoIP |
| ☐ | | | 10 | VoIP |

Pick up the handset and dial 11. The VoIP Gateway will dial "test" and go through Internet and follow Route # 1.

## 3-3-3 DTMF & PULSE

Advanced Settings → DTMF & PULSE




**Dial Wait Timeout:** Enter the timeout duration after the user picks up the phone set.

**Inter Digits Timeout:** Enter the timeout duration between the intervals of each key pressed. When exceeding the set timeout duration without entering further digits, the numbers entered will be dialed out.

**Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones):** This variable is to set the length of DTMF playback.

**DTMF Detection Sensitivity:** Adjust DTMF detect threshold of duration.

**DTMF Detection Volume Sensitivity:** Adjust DTMF detect threshold of DTMF volume

**DTMF Output Volume:** Adjust the Tx volume of FXO port for DTMF Caller ID or Out of Band DTMF.

**Enable Out-of-Band DTMF:** This variable is to set the method of DTMF transmission. RFC2833 or SIP Info.

> **Note:** Out-of-Band DTMF transport method varies from VoIP networks, please contact your VoIP provider for the preferred method.

**Enable Hook Flash Event:** Select **Auto**, **RFC2833**, or **SIP info** for the signaling method of Hook Flash Event.

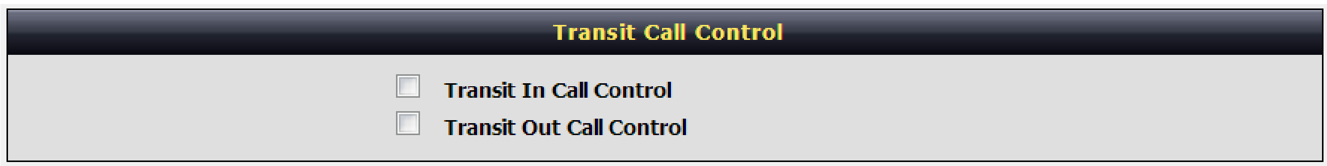
**Payload Type:** payload type of RFC2833.

**Volume:** Select the volume of RFC 2833 from the drop-down menu.

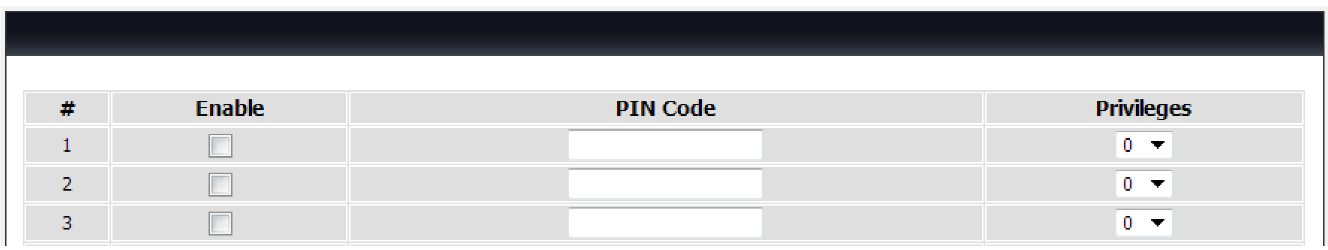
## 3-3-4 Transit Call Control

This is to control outgoing call and incoming call through FXO. Transit Call Control is effective when it cooperates with Long-Distance Control Table. Long-Distance Exception Table is for an exception and it will not be restricted by Transit Call Control and Long-Distance Control Table. You have to enable both of **Inbound/Outbound Call Control** and **PIN Code**.

**NOTE: Transit Call Control is active in one-stage dial.**




- Inbound Call Control: It is the inbound PIN code to check the calls from a PSTN network to FXO and then using a VoIP when ticked — only effective for incoming calls calling from PSTN network.
- Outbound Call Control: It is the outbound PIN code to check the calls from FXO to divert to PSTN network when ticked — only effective for outgoing calls being diverted to PSTN network.



| # | Enable | PIN Code | Privileges |
|---|---|---|---|
| 1 | ☐ | | 0 |
| 2 | ☐ | | 0 |
| 3 | ☐ | | 0 |

- PIN Code: Enter the PIN code (4-6 digits or leave blank. A blank indicates no PIN code is required at this level. Generally, the PIN at level 5 can remain blank to simplify the phone number.)
- Enable: It is to activate the PIN code at each level when ticked.
- Privileges: The level is divided into 0~5 (The levels are in descending order; 0 stands for the highest authority and 5 stands for the lowest.)

**The dialing principle to PIN Code is below:**

<u>* inbound call control PIN code * phone number</u>
OR
<u>* outbound call control PIN code * phone number</u>

**Using * to separate PIN code and the phone number is based on actual settings.**

## 3-3-5 Long-Distance Control Table

This table controls the level of authority of an outgoing (transit out) call that is dialed through FXO and diverted to PSTN network, as below

**Long Distance Control Table**

**Table**

| # | Level 0 | Level 1 | Level 2 | Level 3 | Level 4 | Level 5 |
|---|---|---|---|---|---|---|
| 1 | | | | | | |
| 2 | | | | | | |
| 3 | | | | | | |

**Descriptions:**

- Digit strings in this table are prefixes that the gateway will check on dialed numbers in transit out calls.
- This table is used to prohibit dialing any numbers started with specified prefixes.
- If Level 0 (the highest level) is set to prohibit dialing any number started with prefix 0204, then any level below 0 (including Levels 1 to 5) is also prohibited.
- If Level 1 is set to prohibit dialing any number with prefix 0, then any level below 1 (including Levels 2 to 5) is also prohibited. Since Level 0 is not restricted to any prefix, therefore at level 0 users can dial a number with the prefix 0.

**NOTE: Downward Restriction — If the users at a higher level cannot dial a number with a certain prefix, then users at lowers level also cannot dial a number with the same prefix.**

## 3-3-6 Long Distance Exception Table

This table handles any exceptions to the long-distance call table.

According to the Long Distance Control Table, users at Level 0 are prohibited from dialing a number with the prefix 0204. But, if the number 020488988 is set in the Exception Table as above, then users could then dial this number.

**Long Distance Exception Table**

**Table**

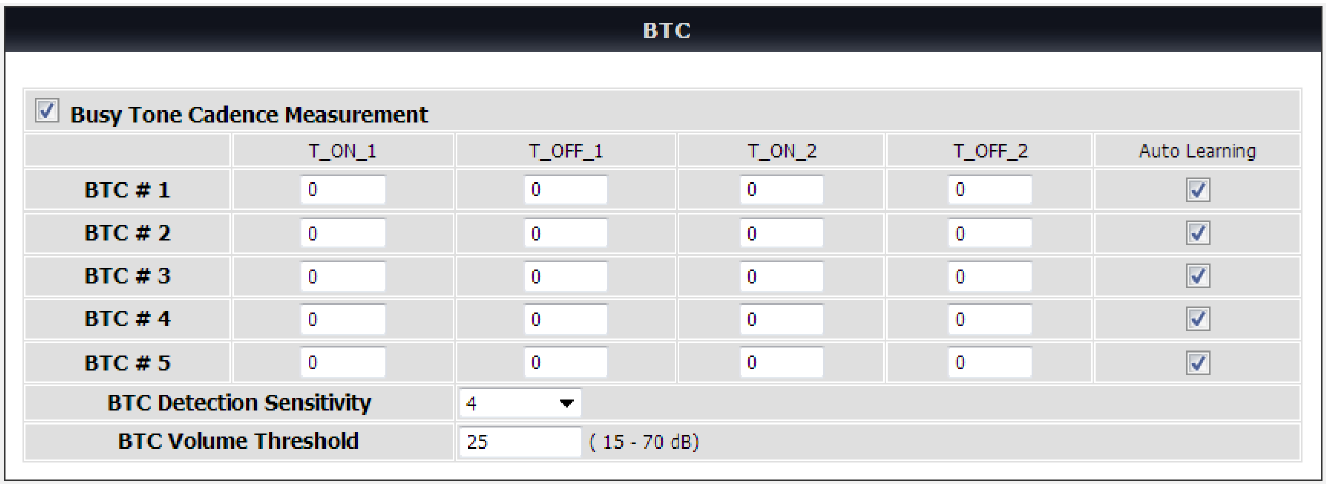
| # | Level 0 | Level 1 | Level 2 | Level 3 | Level 4 | Level 5 |
|---|---|---|---|---|---|---|
| 1 | | | | | | |
| 2 | | | | | | |
| 3 | | | | | | |

**NOTE: Upward Opening —If the users at a lower level can dial a number with a certain prefix, then the users at higher levels can also dial a number with the same prefix.**

## 3-3-7 CPT / Cadence

CPT/Cadence parameters serve as the basis of an FXO interface to determine whether or not a PSTN-call receiving party has hung up the phone. If the following parameters differ from the parameters of the actual assigned lines, it could cause the FXO to continue to engage a line.

Advanced Settings → CPT / Cadence




- Busy Tone Cadence Measurement: Provide a solution of FXO integrated with PSTN or PBX. FXO will learn the busy tone automatically.
- BTC Detection Sensitivity: The more sensitivity, the more quickly the gateway will cut off the call. If the gateway often cut off an un-finished call, select less sensitivity.
- BTC Volume Threshold: It is the threshold of busy tone detection. The gateway will learn BTC succesfully if the volume threshold is in the range of busy tone cadence.

**CPT # 1** Default

| Tone Type | Low Frequency | High Frequency | T_ON_1 | T_OFF_1 | T_ON_2 | T_OFF_2 | T_ON_3 | T_OFF_3 |
|---|---|---|---|---|---|---|---|---|
| Dial Tone | 350 | 440 | 3000 | 0 | 0 | 0 | 0 | 0 |
| Congestion Tone | 480 | 620 | 250 | 250 | 0 | 0 | 0 | 0 |
| Busy Tone | 480 | 620 | 500 | 500 | 0 | 0 | 0 | 0 |
| Ring-Back Tone | 440 | 480 | 1000 | 2000 | 0 | 0 | 0 | 0 |

☑ **CPT # 2** Default

| Tone Type | Low Frequency | High Frequency | T_ON_1 | T_OFF_1 | T_ON_2 | T_OFF_2 | T_ON_3 | T_OFF_3 |
|---|---|---|---|---|---|---|---|---|
| Dial Tone | 400 | 0 | 300 | 100 | 3500 | 100 | 0 | 0 |
| Congestion Tone | 400 | 0 | 250 | 250 | 0 | 0 | 0 | 0 |
| Busy Tone | 400 | 0 | 500 | 500 | 0 | 0 | 0 | 0 |
| Ring-Back Tone | 400 | 0 | 500 | 100 | 500 | 2000 | 0 | 0 |

Default

| Tone Type | Low Frequency | High Frequency | T_ON_1 | T_OFF_1 | T_ON_2 | T_OFF_2 | T_ON_3 | T_OFF_3 |
|---|---|---|---|---|---|---|---|---|
| FWD/DND Dial Tone | 400 | 0 | 800 | 80 | 0 | 0 | 0 | 0 |

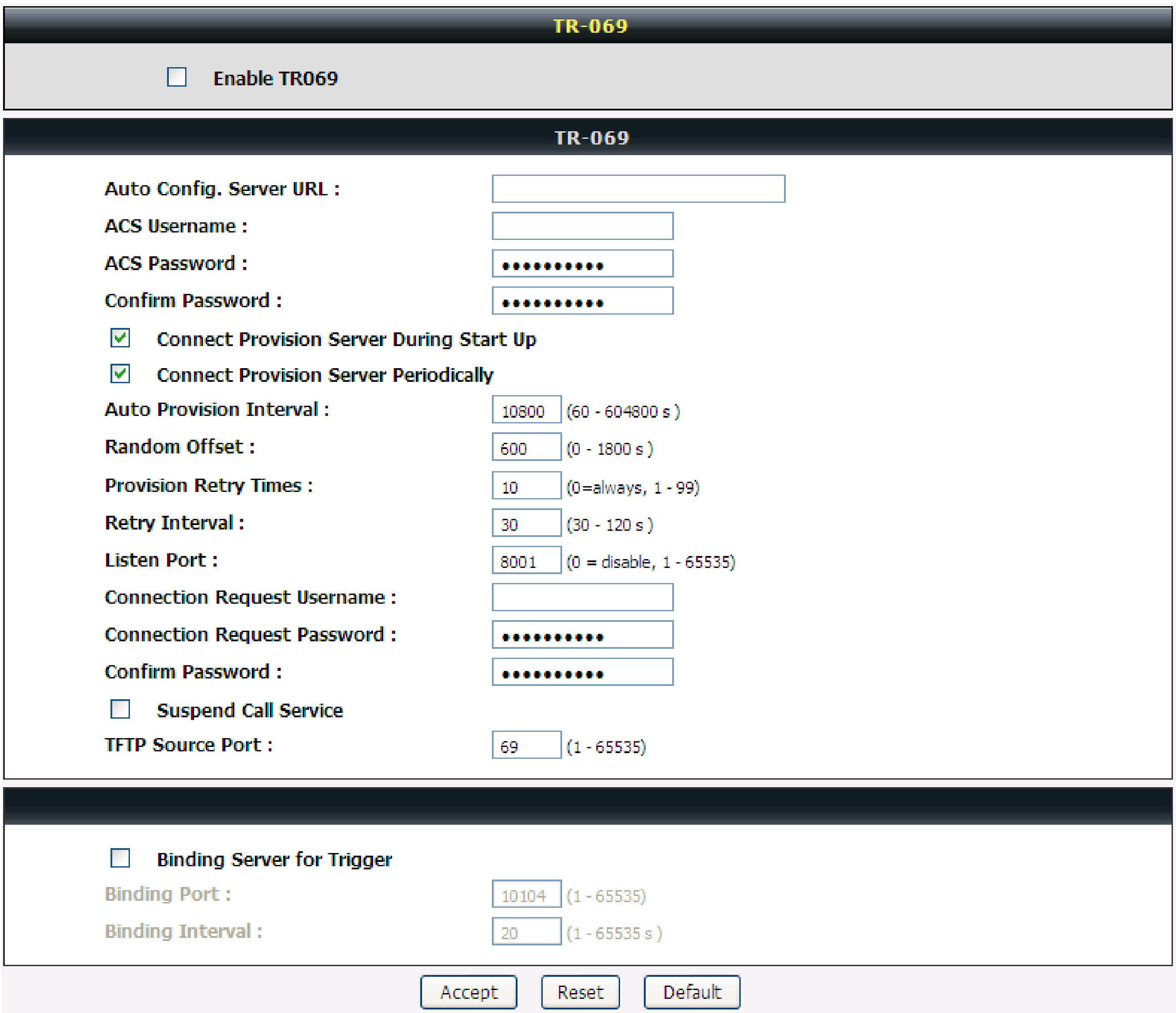
**CPT # 1 Enable Setting 1:** The CPT has a set of parameter table. Please adjust the CPT based on the local PSTN or PBX settings and requirements.

## 3-3-8 TR069

TR069 allows operator to manage DVG with a TR069 standard protocol.

**Note:** Fill in the parameters needed by your VoIP Service Provider. Please check with your VoIP Service Provider about the availability of these services.

Advanced Settings → TR069




**Enable TR069:** Check the box to start TR069 service

**Auto Config. Server URL:** Enter the Provisioning Server's URL required by your VoIP Service Provider.

**ACS Username:** Enter an available user name

**ACS Password:** Enter the password.

**Connect Provision Server During Start Up:** It allows DVG connect to a TR069 server as it starts up.

**Connect Provision Server Periodically:** It allows DVG connect to a TR069 server periodically according to following parameters.

**Auto Provision Interval:** Enter the time for auto provisioning.

**Random Offset:** Enter the offset of the time for auto provisioning.

**Provision Retry Times:** Enter the retry time if a provisioning attempt fails.

**Retry Interval:** Enter the interval for retrying.

**Listen Port:** TR069 listen port for remote trigger.

**Connection Request Username:** Enter username for remote trigger.

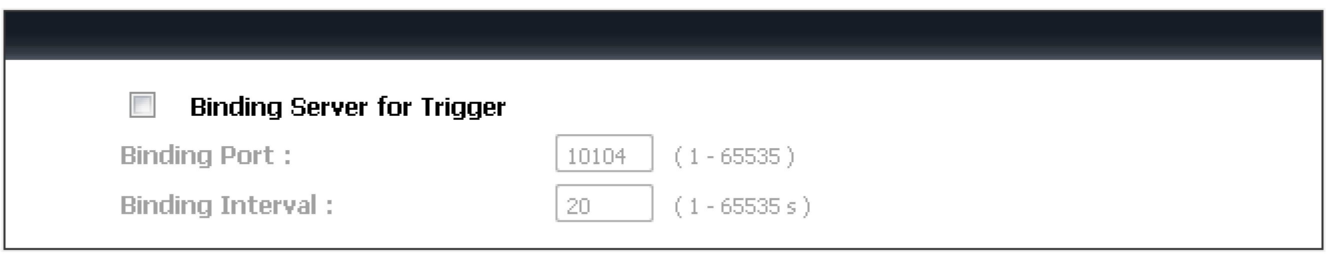
**Connection Request Password:** Enter password for remote trigger**.**

**Suspend Service:** Check the box to stop VoIP call service.

**TFTP Source Port:** Assign TFTP sourct port for TFTP download.

**Note:** Contact your server provider if necessary.

MAINTENANCE → TR069




**Binding Server for Trigger:** Check the box to trigger a connection between Provisioning Server and the VoIP Gateway. Provisioning Server will bind a port for the VoIP Gateway to send provision request.

**Binding Port:** Enter the port number of Provisioning Server is used for binding.

**Binding Interval:** Enter the interval at which the VoIP Gateway will keep the binding.

## 3-3-9 Caller Filter

This function allows you to accept or reject any incoming call from the IP address listed in the filter rule. The call from the IP address of SIP proxy server is always accepted, despite Deny is selected or the IP address of SIP proxy server is not in the filter rule of Allow.

Advanced Setting → Caller Filter


**Caller Filter:** It is to allow or deny the filter rule.

**Status:** It is to show the status of enable or disable.

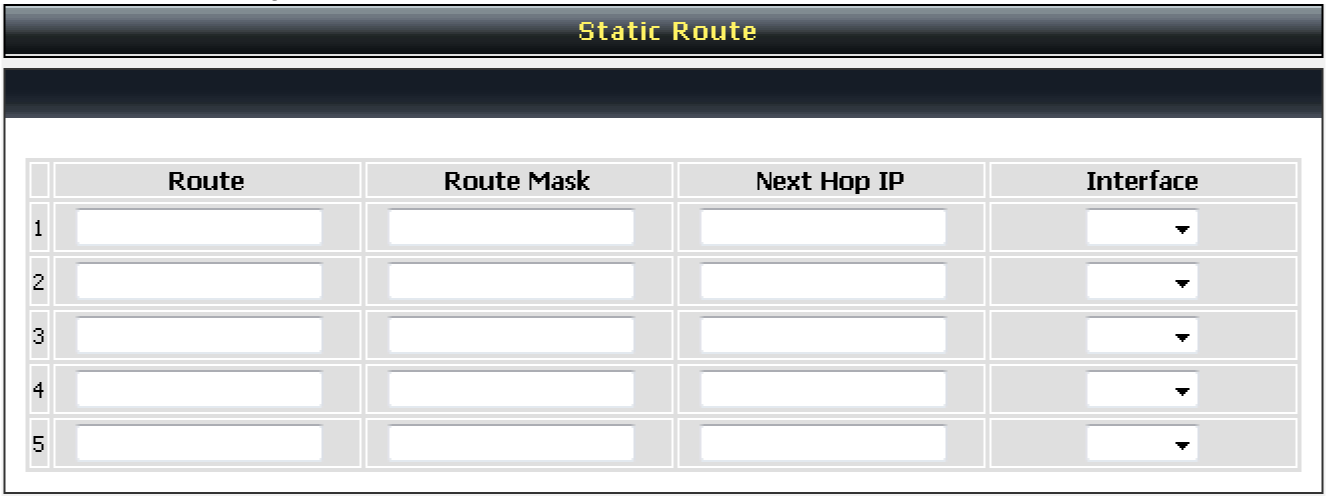
**Filter IP Address:** Enter the start IP address which you would like to Allow or Deny.

**Subnet mask:** Enter the subnet mask you would like to Allow or Deny.

## 3-3-10 Static Route

Build static routes within an internal network. These routes will not apply to the Internet.

Advanced Settings → Static Route




**Route:** Destination network of the route.

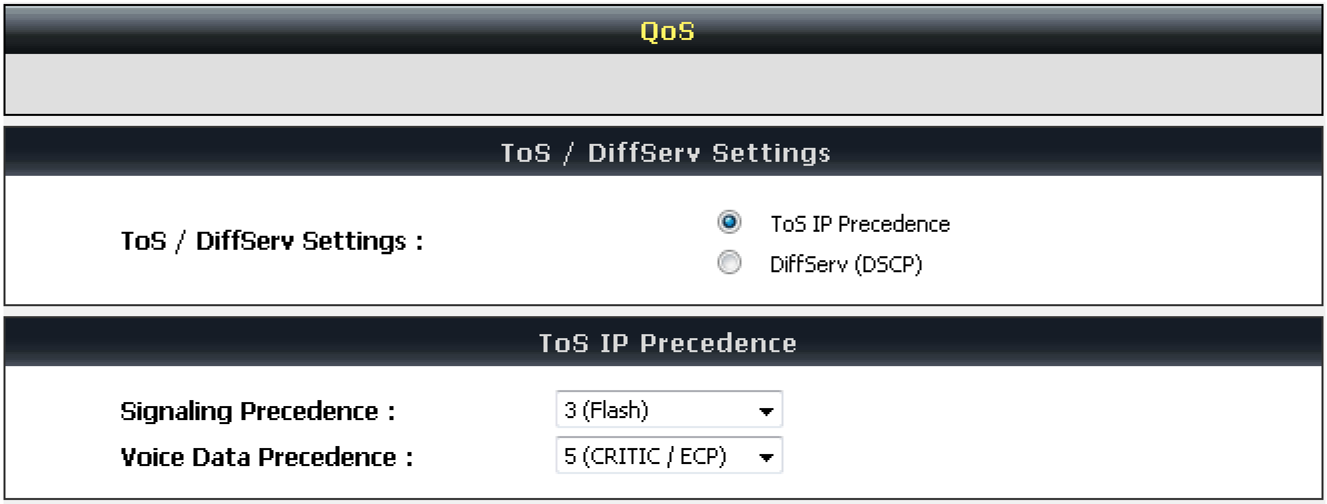
**Route Mask:** Subnet mask to apply on destination network.

**Next Hop IP:** The next hop IP address to the specified network.

**Interface:** The interface attached to this route.

## 3-3-11 QoS Settigs

Advanced Settings → QoS



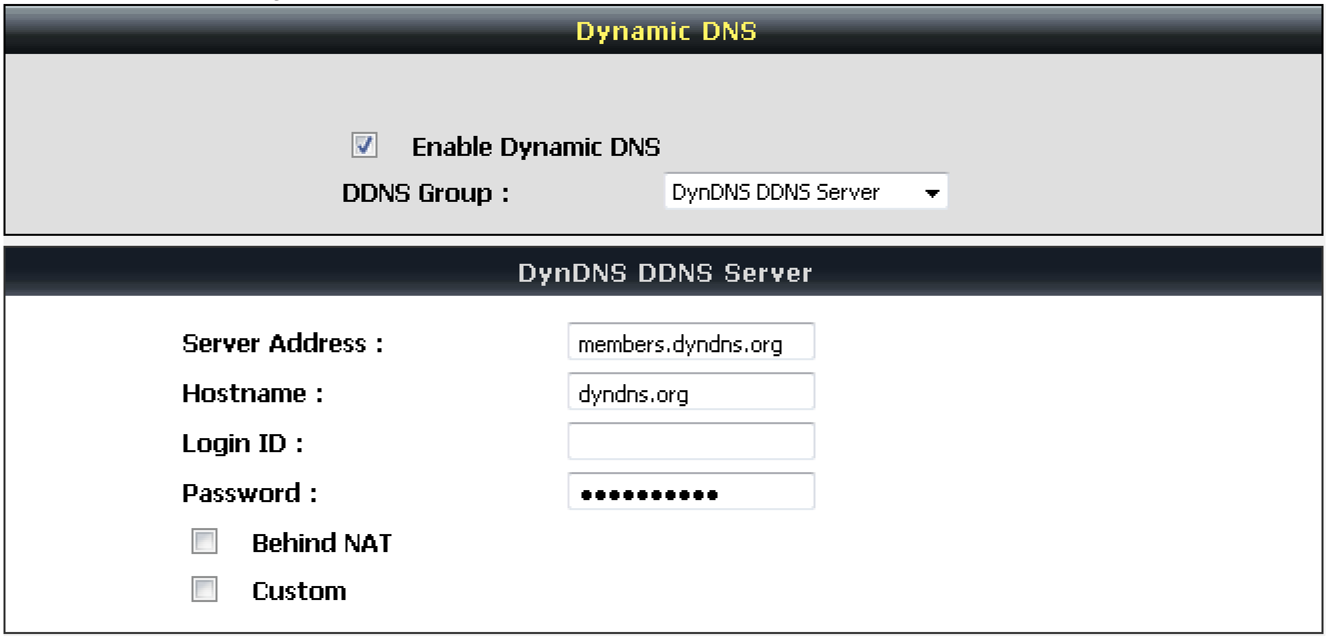

**ToS IP Precedence:** Select the precedence for signaling (data) and voice (voice data).

**DiffServ (DSCP):** Select the number of signaling (data) and voice (voice data) values.

**Note:** For the VoIP Gateway, ToS IP Precedence and DiffServ are the same function. You only select one for priority marking.

## 3-3-12 DDNS

Advanced Settings → DDNS




**Enable Dynamic DNS:** Check the box to enable DDNS function. It is only necessary when the VoIP Gateway is set up behind an Internet sharing device that uses a dynamic IP address and does not support DDNS.

**Server address:** Accept the default setting or fill a correct DDNS Service FQDN.

**Hostname:** Enter the URL of the system (or NAT) – applied from domain name registration providers (e.g. DVG01.dyndns.org).

**Username or Key/Password or Key:** Enter the Login ID and password used to log-in to the DDNS server.

> **Note:** If the VoIP Gateway is set up under NAT, then enter the hostname in the NAT IP/Domain that is the same as the Hostname of the DDNS.

## 3-3-13 NAT Traversal

If your VoIP Gateway is set up behind an Internet sharing device, you can select either the NAT or STUN protocol.

Advanced Settings → NAT Traversal


**Enable(NAT Public IP):** Check the box to use the IP address of the Internet sharing device if the VoIP Gateway is set up behind an Internet sharing device. Also the VoIP Gateway will use the IP address of the Internet sharing device as the public IP when it connects to Internet. Furthermore, some of the Internet sharing device's type is symmetric NAT. You need to set Virtual Server or Port Mapping (Forwarding) from the Internet sharing device for the listen port and communication ports (RTP ports) of the VoIP Gateway.

**NAT IP/Domain:** Enter the real public IP address of the IP sharing device or the router; or enter a true URL (Uniform Resource Locator) when DDNS is used. Please refer to the DDNS settings.

> **Note:** If you are setting a public IP in this field, it has to be a static public IP, otherwise VoIP communication may not be established properly. Please contact your ISP to check if your Internet connection has static public IP addresses.

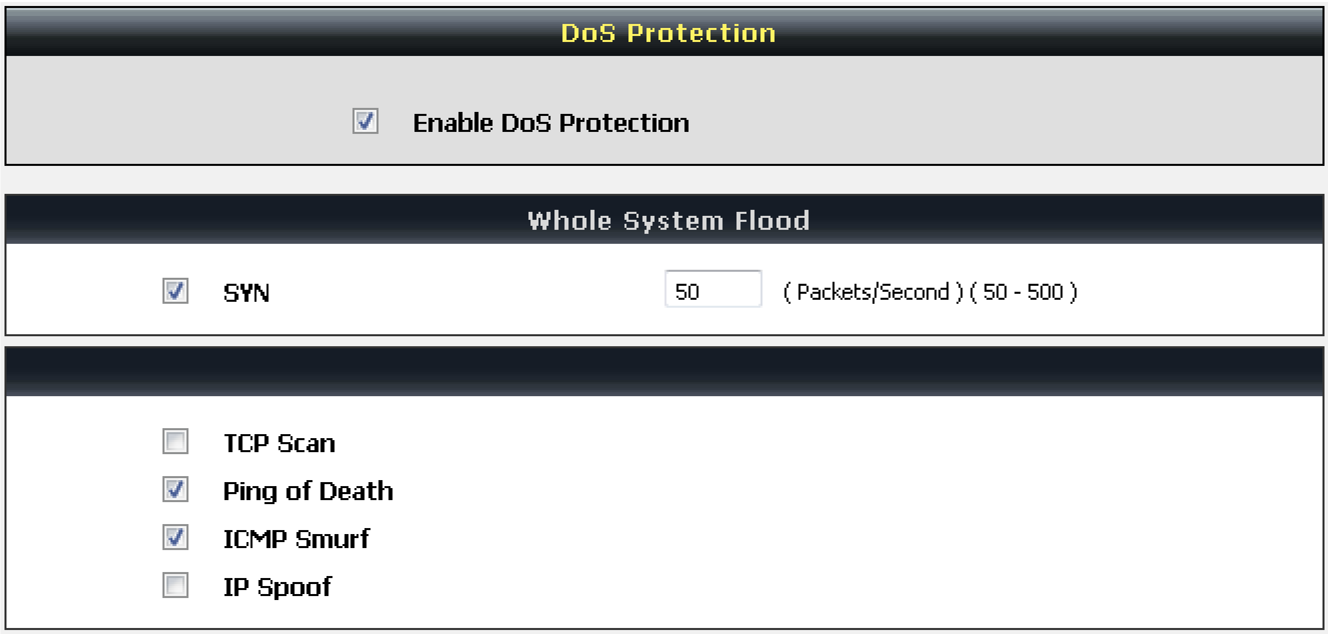
**Enable STUN Client:** Check the box to use the STUN protocol prevents problems from setting the IP sharing function. (Some NATs do not support this protocol.)

> **Note:** You can use the "Status → STUN Inquiry" page to detect the NAT type of your Internet sharing device. If the NAT type is "Symmetric NAT," then the VoIP Gateway is not able to traverse the NAT. It is not a flaw of the VoIP Gateway design, but rather a limitation of the STUN protocol.

**STUN Server IP/Domain and Port:** Enter the IP address and listen port of the STUN server. You can set two STUN server IPs separated by a semicolon.

## 3-3-14 DoS Protection

Advanced Settings → DoS Protection




**Enable DoS Prevention:** Check the box to prevent DoS attacks from WAN. There are various types of DoS attacking. Leave settings in this field to the default if you are not familiar with it.

## 3-3-15 DMZ / ALG

DMZ (Demilitarized Zone) allows the server on the LAN site to be directly exposed to the Internet for accessing data and to forward all incoming ports to the DMZ Host. Adding a client to the DMZ may expose that computer to a variety of security risks; so only use this option as a last resort.

Advanced Settings → DMZ /ALG


**Enable DMZ:** Check the box to enable DMZ feature.

**DMZ Host IP Address:** Enter the IP address of that computer as a DMZ Host with unrestricted Internet access.

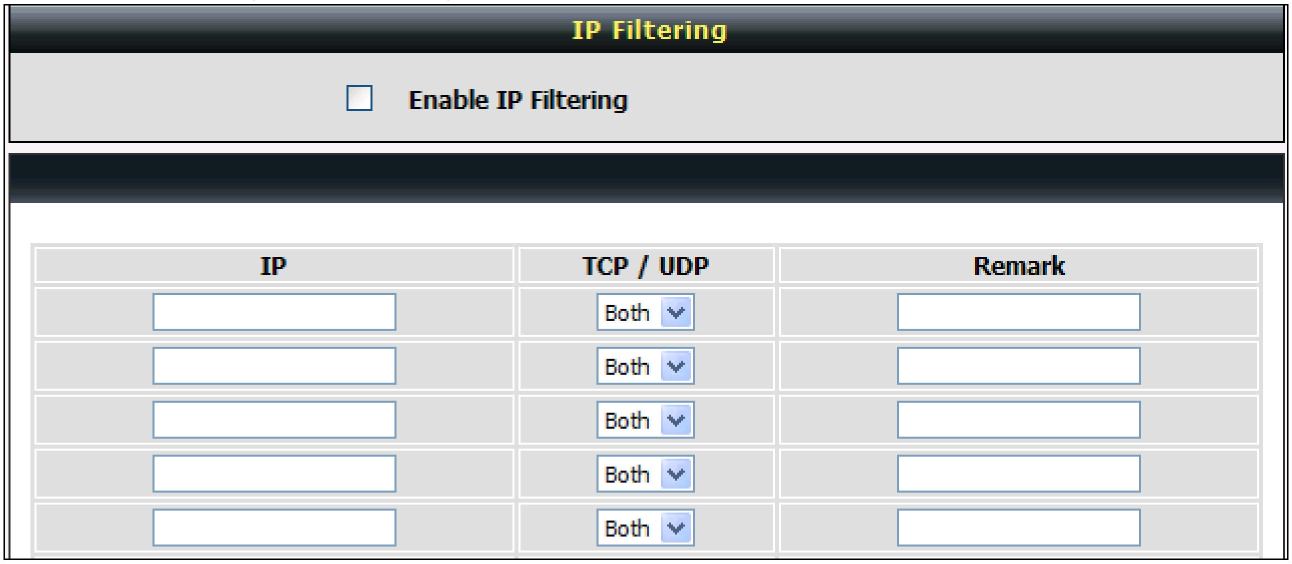
**Note:** Either this function or virtual server can be selected for use in accessing external services.

**RTSP ALG:** Enable ALG for RTSP multimedia stream.

## 3-3-16 IP Filtering

Use IP Filters to deny particular LAN IP addresses from accessing the Internet. You can deny specific port numbers or all ports for a specific IP address. The screen will display well-known ports that are defined. To use them, click on the edit icon. You will only need to input the LAN IP address(es) of the computer(s) that will be denied Internet access.

Advanced Settings → IP Filtering




**Enable IP Filtering:** Check the box to deny particular LAN IP addresses from accessing the Internet.

**IP:** Enter the IP address that you want to deny in this filed.

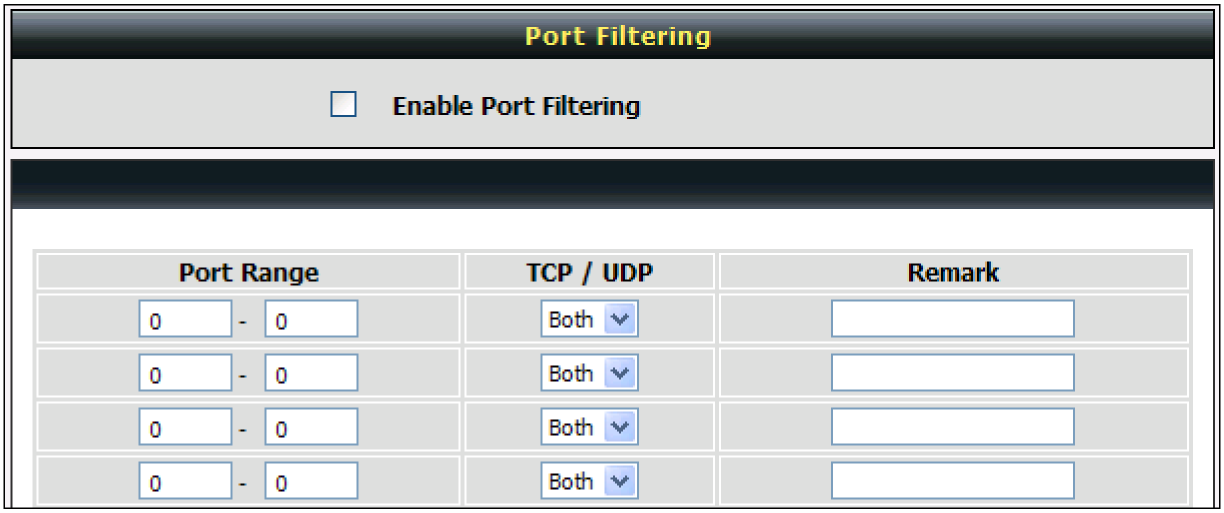
**TCP/UDP:** Select **TCP**, **UDP** or **Both** that will be used with the IP address that will be blocked.

**Remark:** Enter comments.

## 3-3-17 Port Filtering

Port filtering enables you to control all data that can be transmitted over routers. When the port used at the source end is within the defined scope, it will be filtered without transmission.

Advanced Settings → Port Filtering




**Enable Port Filtering:** This variable is to restrict certain types of data packets by port.

**Port Range:** Enter the port range that will be denied access to the Internet.

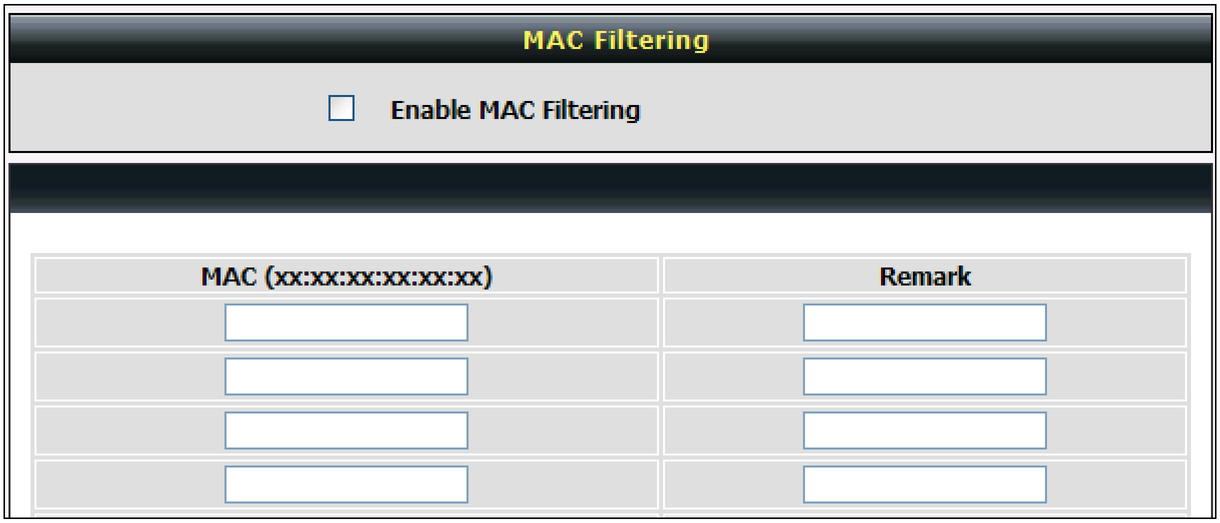
**TCP/UDP:** Select **TCP**, **UDP** or **Both** that will be used with the port that will be blocked.

**Remark:** Enter comments.

## 3-3-18 MAC Filtering

Use MAC Filters to deny computers within the local area network from accessing the Internet. You can either manually add a MAC address that are connected to the VoIP Router.

Advanced Settings → MAC Filtering




**Enable MAC Filtering:** Check the box to deny from accessing Internet.

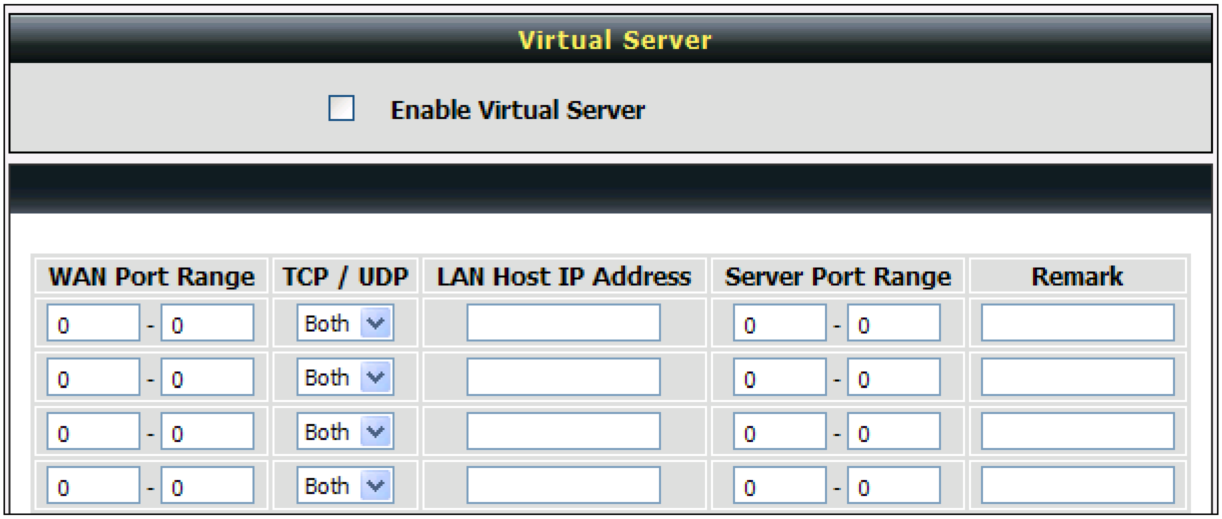
**MAC:** Enter the MAC of the computer in the LAN (Local Area Network) to be used in the MAC filter table.

**Remark:** Enter comments.

## 3-3-19 Virtual Server

Enable users on Internet to access the WWW, FTP and other services from your NAT. It is also known as port forwarding. When remote users are accessing Web or FTP servers through WAN IP address, it will be routed to the server with LAN IP address.

Advanced Settings → Virtual Server




**Enable Virtual Server:** Check the box to enable port forwarding.

**WAN Port Range:** Enter the port range for the WAN side.

**TCP/UDP:** Select the communication protocols used by the server, **TCP**, **UDP** or **Both**.

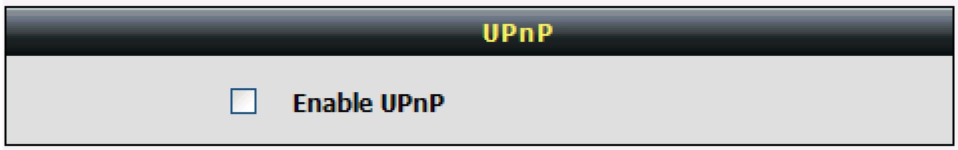
**LAN Host IP Address:** Enter the IP address of the device that provides various services.

**Server Port Range:** Enter comments.

**Remark: Enter comments.**

## 3-3-20 UPnP

Advanced Settings → UPnP




**Enable UPnP:** Check the box to enable the VoIP Router’s IP traffic to pass through an Internet sharing device.

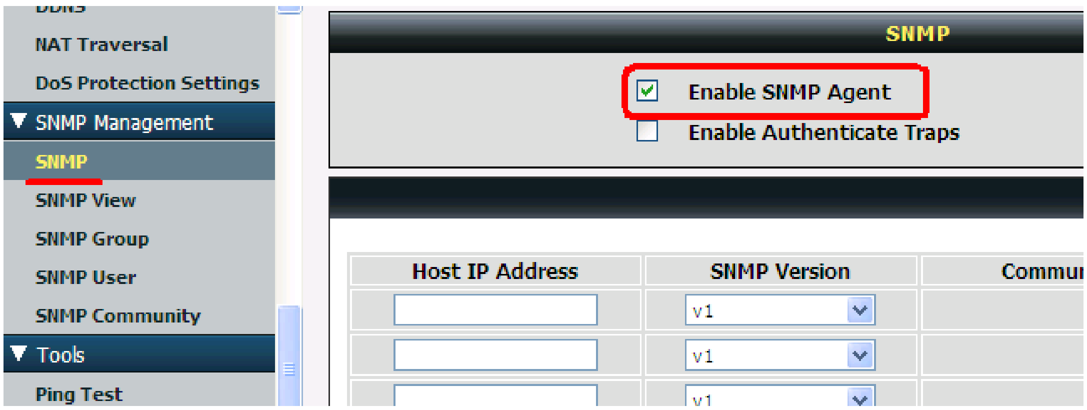
# 3-4 SNMP management

DVG support SNMP V1, V2 and V3. Please enter required parameter for SNMP V3.

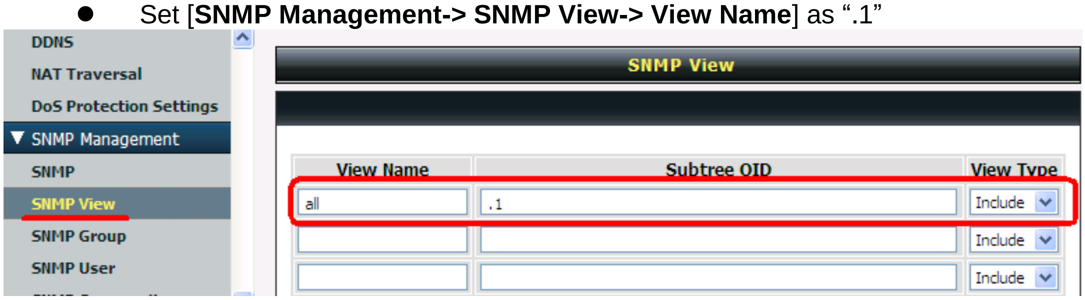
If the MIB browser supports SNMP V1 and V2 only, please refer to following configuration:

**Example for SNMP V2 configuration**:
Some keys configured on MIB browser:
Get Community: public
Set Community: private
Trap Community: public

- Enable [**SNMP Management-> SNMP-> Enable SNMP Agent**]




- Set [**SNMP Management-> SNMP View-> View Name**] as "all".
- Set [**SNMP Management-> SNMP View-> View Name**] as ".1"



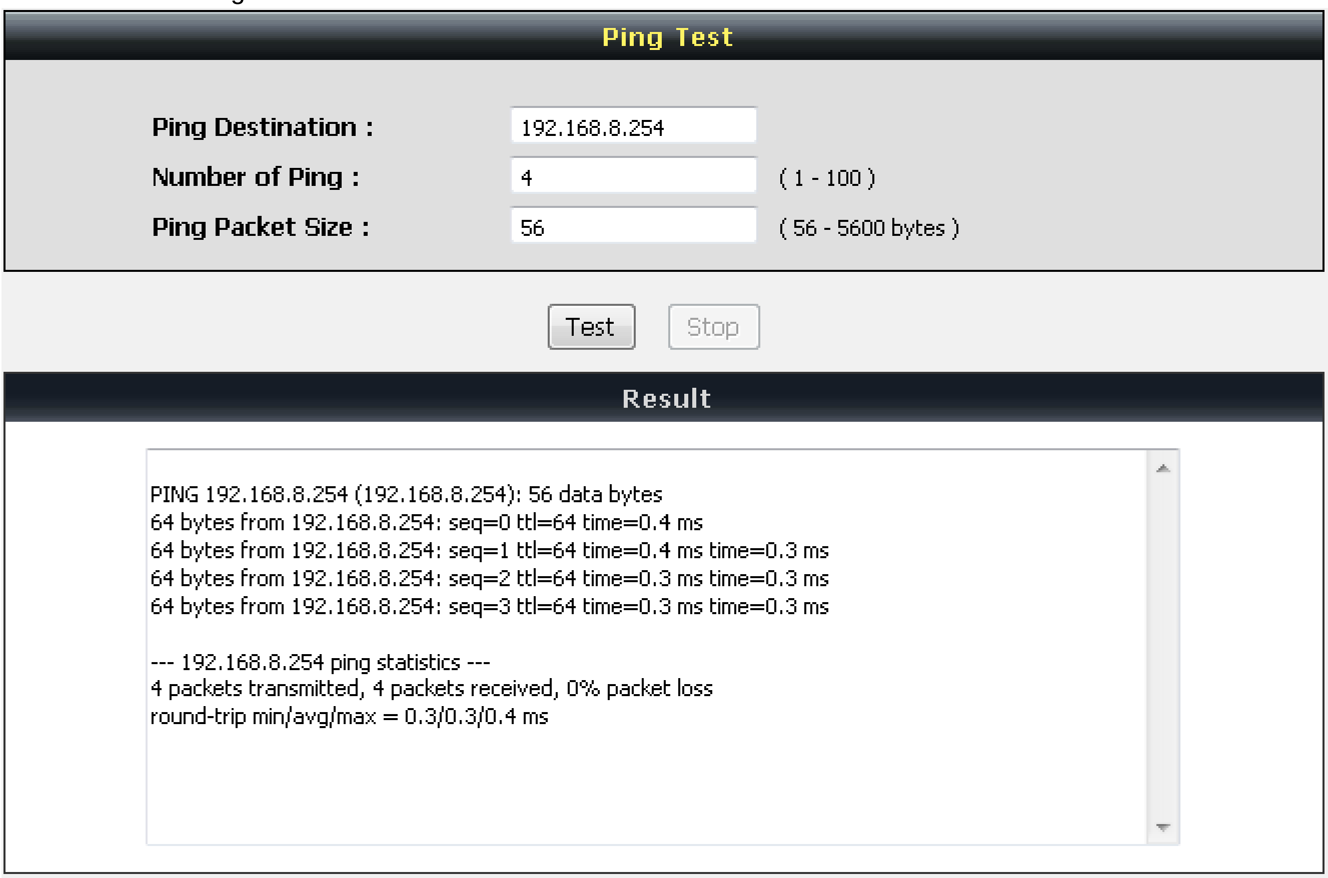

- Set [SNMP Management-> SNMP Community-> Community Name] for public and private.
- Select a configured View Name.

# 3-5 Tools

## 3-5-1 Ping Test

Use "Ping" to verify if a remote peer is reachable. Enter a remote IP address and click "Test" to ping the remote host. The result would be shown on **Result** Table

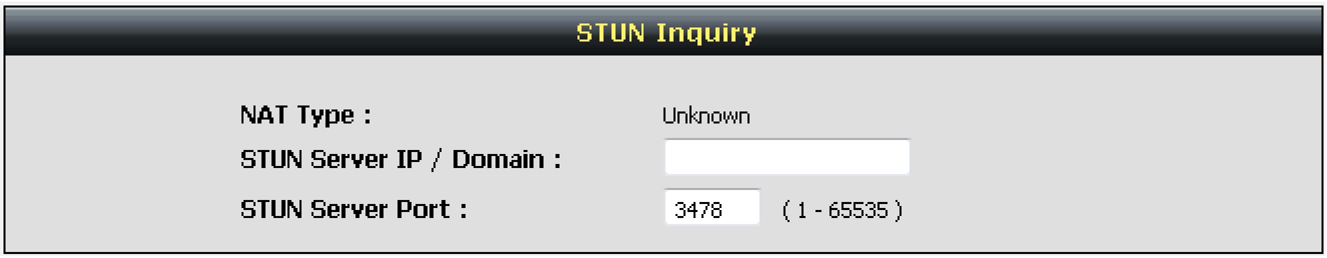
Tools → Ping Test



## 3-5-2 STUN Inquiry

Use "STUN Inquiry" to detect your IP sharing device's NAT type and communication between a STUN server and client.

Tools → STUN Inquiry




**NAT Type:** It shows the NAT type of your router.

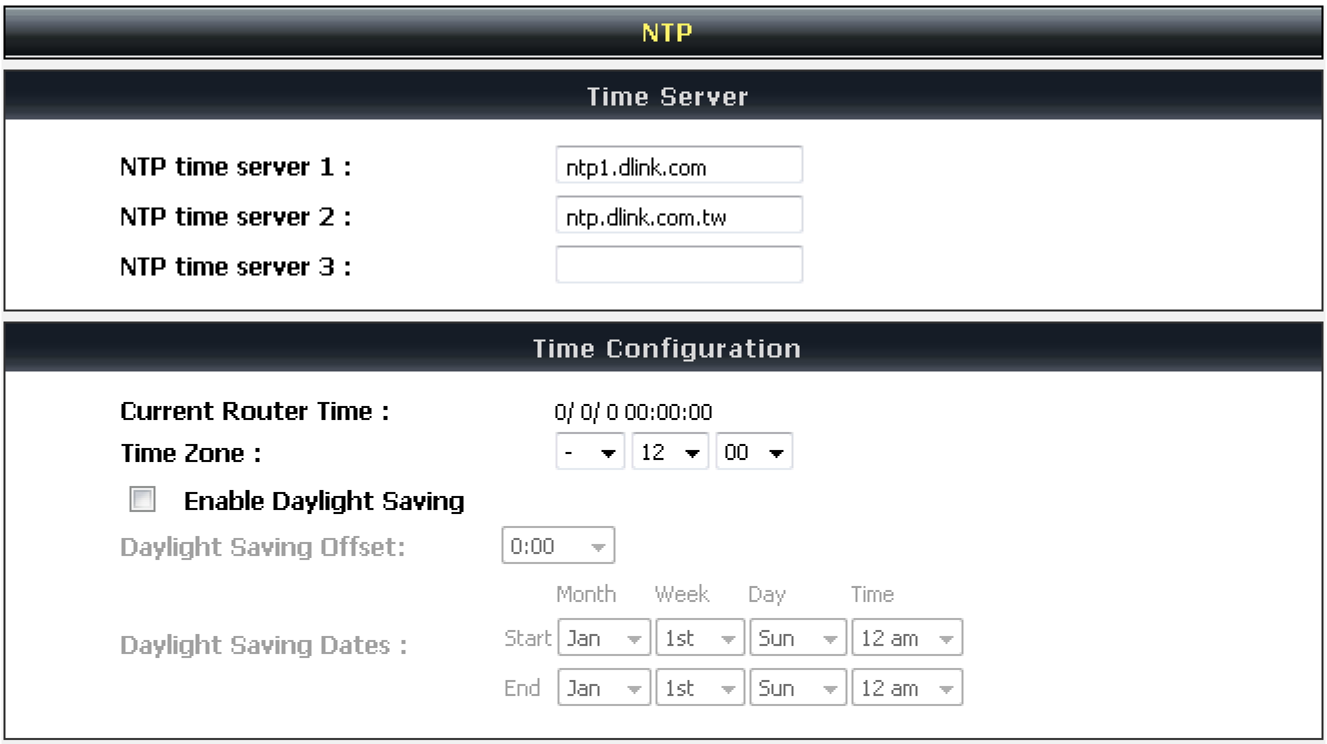
**STUN Server IP/Domain:** Enter the IP address or URL of the STUN server for query.

**STUN Server Port:** Enter the STUN Server's listening port.

# 3-6 System Settings

## 3-6-1 NTP

System settings → NTP




**Automatically synchronize with Internet time servers:** The VoIP Gateway should automatically sync up with time servers.

**First NTP time server:** Select the desired domain name of a NTP server as first priority.

**Second NTP time server:** Select the domain name of a NTP server as second priority.

**Current Router Time:** It shows the current time of the VoIP Gateway.

**Time Zone:** Select your time zone from the drop-down menu.

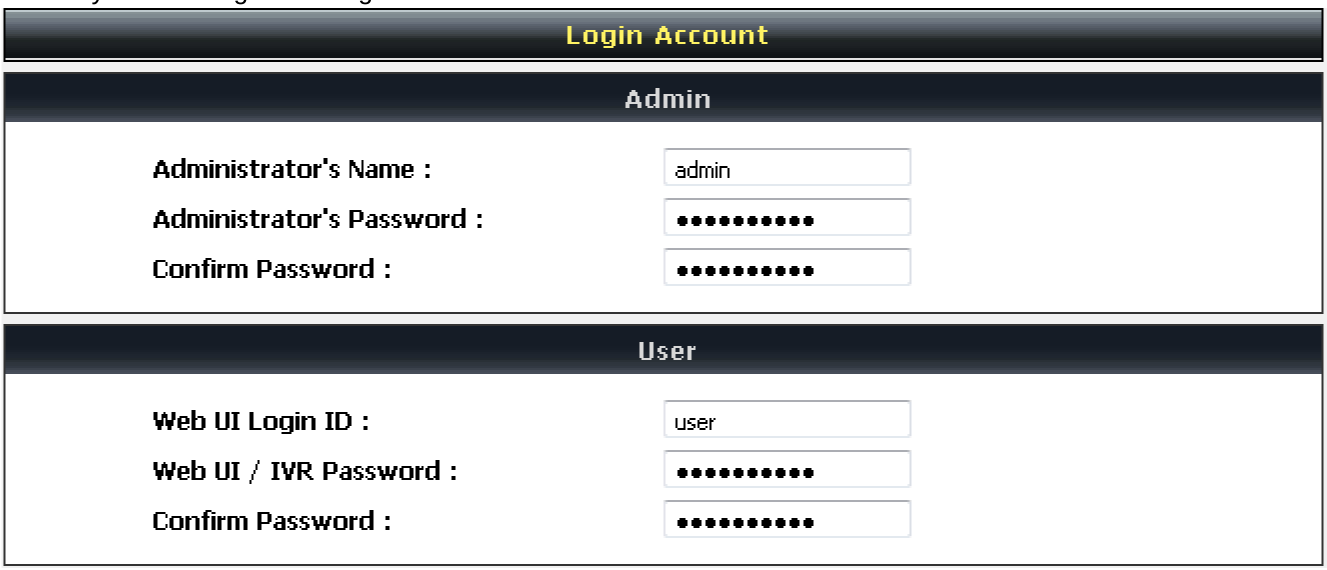
**Enable Daylight Saving:** To enable/disable daylight saving time.

**Daylight Saving Offset:** Set the current time zone offset for your location.

**Daylight Saving Dates:** Set the start and end dates for daylight saving time.

## 3-6-2 Login Account

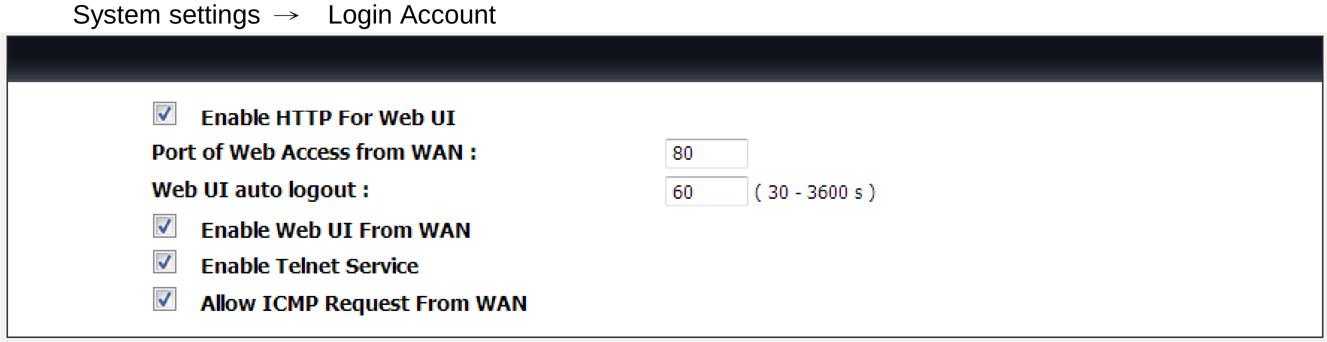
System settings → Login Account




**Note:** There are two operating levels when entering the Web UI. Logging-in as the ADMIN allows you to change all settings. A Web UI USER only has access to some settings.

**Password:** It is highly recommended that you create a password to keep your DVG secure.

System settings → Login Account




**Port of Web Access from WAN:** Enter the port number when accessing the web-based configuration utility from the WAN port.

**Web Idle Time Out:** Enter the range of effective time when log-in the web interface. The user will be disconnected from the web page to allow others to log-in.

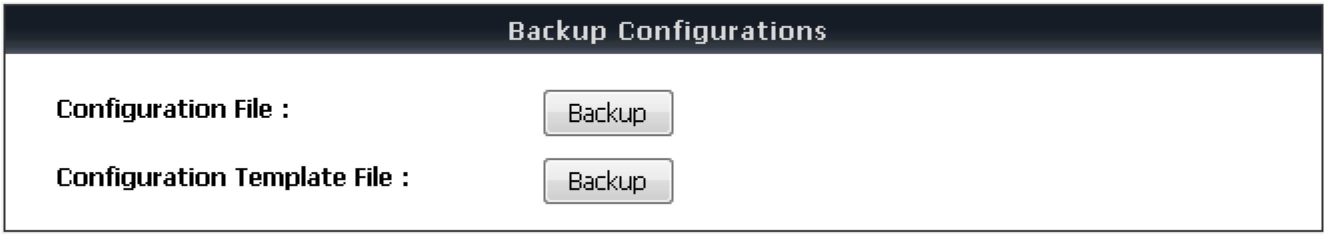
**Enable Web UI:** Check the box to enable WEB access from WAN or LAN.

**Enable Telnet Service:** Check the box to enable Telnet access from WAN or LAN.

# 3-6-3 Backup / Restore

## Backup Configurations File

System settings → Backup and Restore




The current system settings can be saved as a file onto the local hard drive. Click the **Backup Settings** button to save your current settings to a file.

Click the **Backup Settings** button to save your current settings to a template file for editing.

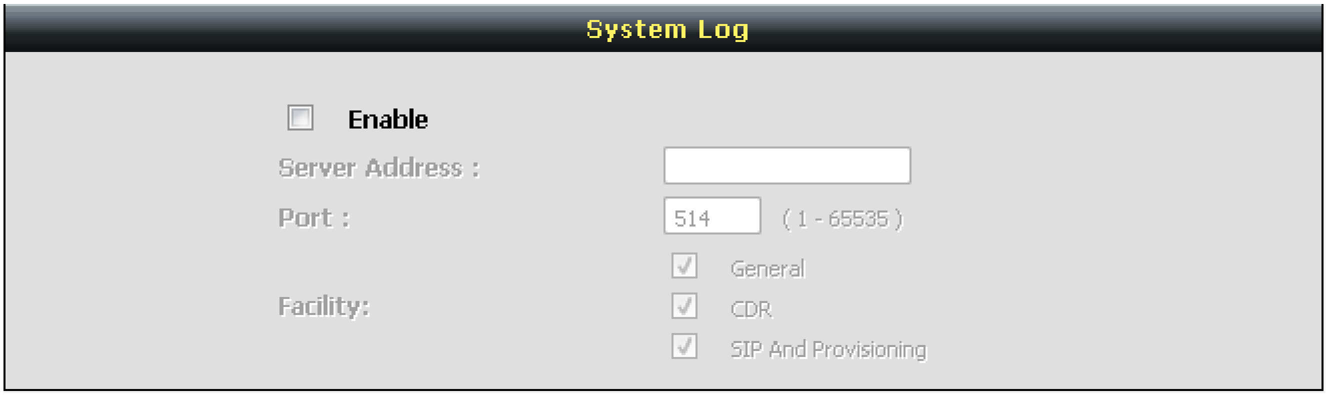
## Restore Default Settings

System settings → Backup and Restore


Select **Restore Default Settings** to reset the VoIP Gateway’s settings back to the factory default settings.

## 3-6-4 System Log

System settings → System log




**Enable:** Check the box to send event notification messages across IP networks to the Server.

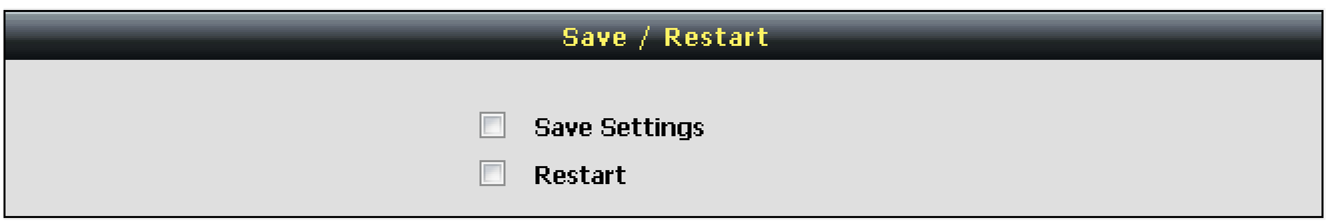
**Server Address:** Enter the System Log Server's IP address.

**Port:** Enter the System Log Server's listening port. Leave this field to the default if your VoIP Service Provider did not provide you a server port number for System Log Server.

## 3-6-5 Save / Restart

### Save and Reboot

System settings → Backup and Restore

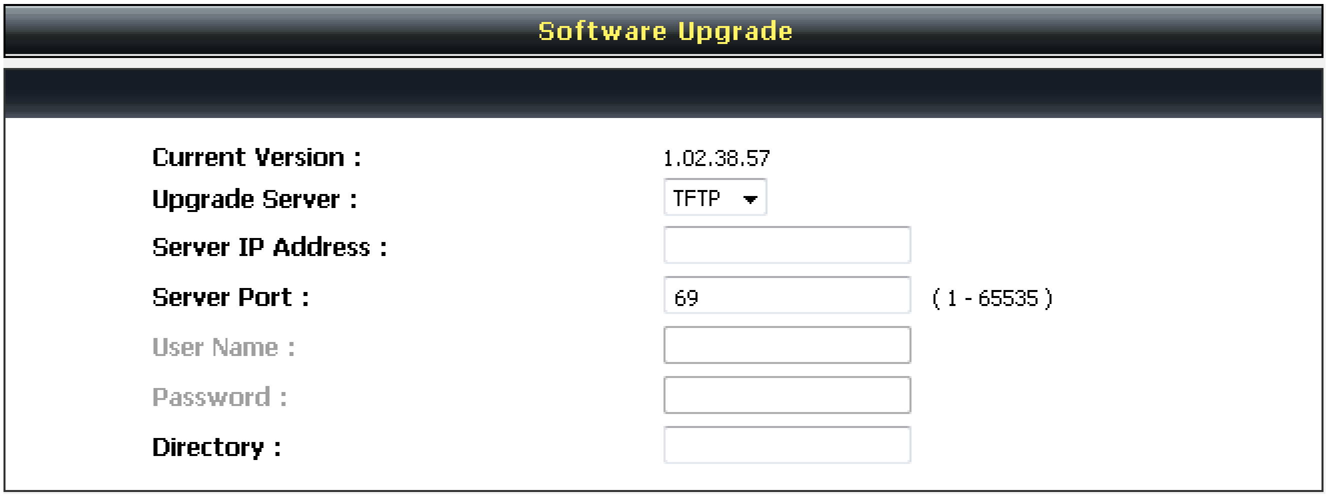



**Save All Settings:** Click the **Save All Settings** check box and reboot the system after completing changes. The new settings will take effect after the VoIP Gateway is restarted.
**Restart:** Click the **Reboot** button to reboot the system.

## 3-6-6 Software Upgrade

The VoIP Gateway supports a software upgrade function from a remote server. Please consult your VoIP Service Provider for information about the following details.

System settings → Software upgrade




**Upgrade Server:** Select the upgrade type: **TFTP**, **FTP**, or **HTTP**.

**Server IP Address:** Enter the server's IP address.

**Server Port:** Enter the server's listen port.

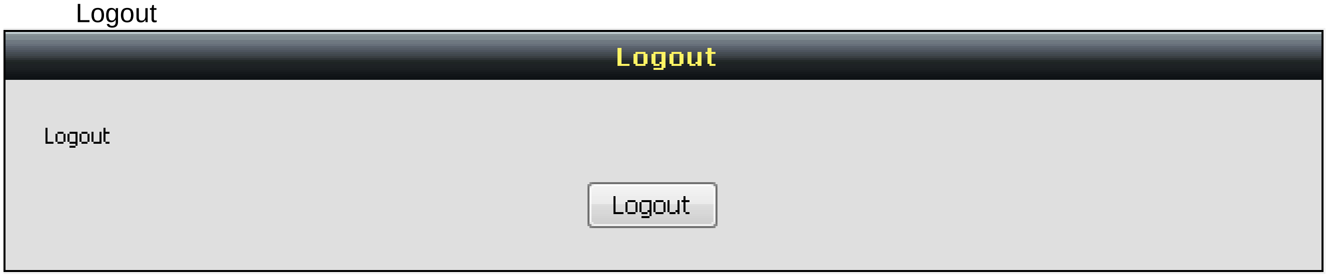
**User Name/ Password:** Enter the account information for accessing the server if needed.

**Directory:** Enter the location of the firmware file.

## 3-6-7 Logout

If setting or parameter has been changed, remember to save the changes before you logout the configuration menu.

Logout



# 4. Configuring the VoIP Gateway through IVR

**Preparation**

1. Connect the power supply, telephone set, telephone cable, and network cable properly.
2. If a static IP is provided, confirm the correct IP settings of the WAN Port (IP address, Subnet Mask, and Default gateway). Please contact your local Internet Service Provider (ISP) if you have any question.
3. If you intend to operate the VoIP Gateway under NAT, the IP range of VoIP Gateway WAN Port and LAN Port IP Address should not be the same in order to avoid phone failures.

IVR configuration provides basic query and setup functions, while browser configuration provides full setup functions.

## 4-1 IVR (Interactive Voice Response)

The VoIP Gateway provides convenient IVR functions. Users are able to get query and setup the VoIP Gateway with a phone-set and function-codes without turning on the PC.

**Note:** When finishing the setup, make sure the new settings are saved. This will enable the new settings to take effect after the system is restarted.

**Instructions**

**FXO Port:** Connect to telephones. To access IVR mode, passwords should be entered, "* * password #". Alphabets to digits conversion information is provided in the PPPoE Character Conversion Table. When correct IVR passwords are entered and accepted, an indication tone can be heard indicates the system is in IVR setup mode. Enter function codes to check or configure the VoIP Gateway.

**Example:** If your password is "1234", enter * (star) * (star) 1 2 3 4 # (pound)**,** and now you are entering IVR setup mode. Next, enter a function code to check or configure the VoIP Gateway. If your password is "admin", enter * (star) * (star) * (star) 41 44 53 49 54 # (pound). Please refer to the IVR Functions Table (page 68) for available functions and codes.

Once the setting or query has been completed, you can hear a dial tone. Use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

**Example:** enter **"**#"** (you are now in IVR mode)→ enter **101** (to query the current IP address) → the system responds with an IP address. You can continue with more settings or queries: enter **111** (to set a new IP address) →enter **192*168*1*2** (new IP address).

**Save Settings**

When all setting procedures are completed, dial 509 (Save Settings) from phone keypad. Wait for about three seconds, you should hear a voice prompt "1 (one)." You can now hang up the phone and please reboot the VoIP Gateway to enable the new settings.

**To inquire about the current VoIP Gateway WAN Port IP address setting**

After completing all your settings, dial 101 from the keypad, then you can hear the system play back the current WAN Port IP address. If the system does not play back the IP address after dialing 101, this indicates that the VoIP Gateway currently is not connected to the Internet. Please check and make sure the cable connections, account numbers, and passwords are correct.

## 4-1-1 IVR Functions Table:

| Function Code | Description | Example / Notes |
|---|---|---|
| 111/101 | WAN Port IP address Set/Query | Dial function code **114** and then dial 1 for a Static IP connection then setup the IP address. |
| 112/102 | WAN Port Subnet Mask Set/Query | |
| 113/103 | WAN Port Default Gateway Set/Query | |
| 114/104 | Current Network IP Access Set/Query (1: Static IP, 2: DHCP, 3: PPPoE) | |
| 115/105 | DNS IP address Set/Query | |
| 118 | Restart | |
| 311/301 | LAN Port IP Set/Query | |
| 312/302 | LAN Port Subnet Mask Set/Query | |
| 109 | Restore factory default IP address configuration | A static IP address for WAN Port<br>IP：192.168.1.2<br>Mask：255.255.255.0<br>Gateway：192.168.1.254 |
| 409 | Restore factory default settings | |
| 509 | Save settings | |

# 4-2 IP Configuration Settings—Set the IP Configuration of the WAN Port

### Static IP Settings

**Note:** Complete static IP settings should include a static IP (option 1 under 114), IP address (111), Subnet Mask (112), and Default Gateway (113). Please contact your Internet Service Provider (ISP) if you have any question.

| Function | Command |
|---|---|
| **Select a Static IP** | • After entering IVR mode, dial 114.<br>• When voice prompt plays "Enter value", dial 1 (to select static IP) |
| **IP address Settings** | • After entering IVR mode, dial 111. When voice prompt plays "Enter value", enter your IP address followed by "#".<br>**Example**: If the IP address is 192.168.1.200, dial 192*168*1*200#. |
| **Subnet Mask Settings** | • After entering IVR mode, dial 112. When voice prompt plays "Enter value", enter your subnet mask followed by "#".<br>**Example**: If the subnet mask value is 255.255.255.0, dial 255*255*255*0#. |
| **Default Gateway Settings** | • After entering IVR mode, dial 113. When voice prompt plays "Enter value", enter your default gateway's IP address followed by "#".<br>**Example**: If the default gateway is 192.168.1.254, dial 192*168*1*254#. |
| **Save Settings and Restart** | • To save settings, dial **509** (Save Settings). The system will save the current settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check whether the IP address was retained. If the system does not play back the IP address after dialing 101, this indicates that the VoIP Gateway currently is not connected to the Internet. Please check and make sure the cable connections, account numbers, and passwords are correct. |

### Dynamic IP (DHCP) Settings

After entering IVR mode, dial 114.

When voice prompt plays "Enter value", dial 2 (to select DHCP).

Saving settings –press 509 (Save Settings). Please restart the system. After the system is restarted, press 101 to check whether or not the IP address was retained.

**Note:** If the system does not play back the IP address, this indicates that the VoIP Gateway failed to communicate with a DHCP server. Please check with your DHCP server or ISP.

### Save Settings and Restart

To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, then enter **101** to check whether the IP address was retained. If the system does not play back the IP address after dialing **101**, this indicates that the VoIP Gateway currently is not connected to the Internet. Please check and make sure the cable connections, account numbers, and passwords are correct.

## 4-2-1 Character Conversion Table:

The table below provides a list of conversion codes. The first row (high-lighted) of each pair of the column lists the numbers, alphabets or symbols and the second row (high-lighted) of each pair of the column ("Input Key") represents the codes to be entered for the corresponding numbers, alphabets or symbols. For example, to enter "D-Link" according to the table below, enter: 148322495451

| Numbers | Input Key | Upper Case Letters | Input Key | Lower Case Letters | Input Key | Symbols | Input Key |
|---|---|---|---|---|---|---|---|
| 0 | 00 | A | 11 | a | 41 | @ | 71 |
| 1 | 01 | B | 12 | b | 42 | • | 72 |
| 2 | 02 | C | 13 | c | 43 | ! | 73 |
| 3 | 03 | D | 14 | d | 44 | " | 74 |
| 4 | 04 | E | 15 | e | 45 | $ | 75 |
| 5 | 05 | F | 16 | f | 46 | % | 76 |
| 6 | 06 | G | 17 | g | 47 | & | 77 |
| 7 | 07 | H | 18 | h | 48 | ' | 78 |
| 8 | 08 | I | 19 | i | 49 | ( | 79 |
| 9 | 09 | J | 20 | j | 50 | ) | 80 |
| | | K | 21 | k | 51 | + | 81 |
| | | L | 22 | l | 52 | , | 82 |
| | | M | 23 | m | 53 | - | 83 |
| | | N | 24 | n | 54 | / | 84 |
| | | O | 25 | o | 55 | : | 85 |
| | | P | 26 | p | 56 | ; | 86 |
| | | Q | 27 | q | 57 | < | 87 |
| | | R | 28 | r | 58 | = | 88 |
| | | S | 29 | s | 59 | > | 89 |
| | | T | 30 | t | 60 | ? | 90 |
| | | U | 31 | u | 61 | [ | 91 |
| | | V | 32 | v | 62 | \ | 92 |
| | | W | 33 | w | 63 | ] | 93 |
| | | X | 34 | x | 64 | ^ | 94 |
| | | Y | 35 | y | 65 | _ | 95 |
| | | Z | 36 | z | 66 | { | 96 |
| | | | | | | \| | 97 |
| | | | | | | } | 98 |

# 5. Dialing Principles

## 5-1 Dialing Options

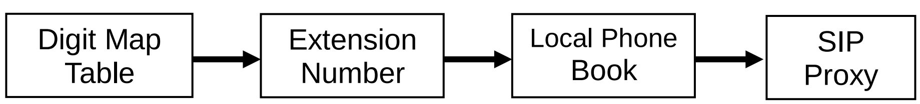
Dial the phone number which you want to call and press # to call out immediately. Note that if the "# (pound)" not dialed, the number will be called out after 4 seconds by default. The period between number dialed and call out is named "Inter Digits Timeout". (Configurable from "DTMF and PULSE", default=4 seconds).

If the phone number matches the setting of the Digit Map, the phone number will be dialed out through the assigned interface automatically.

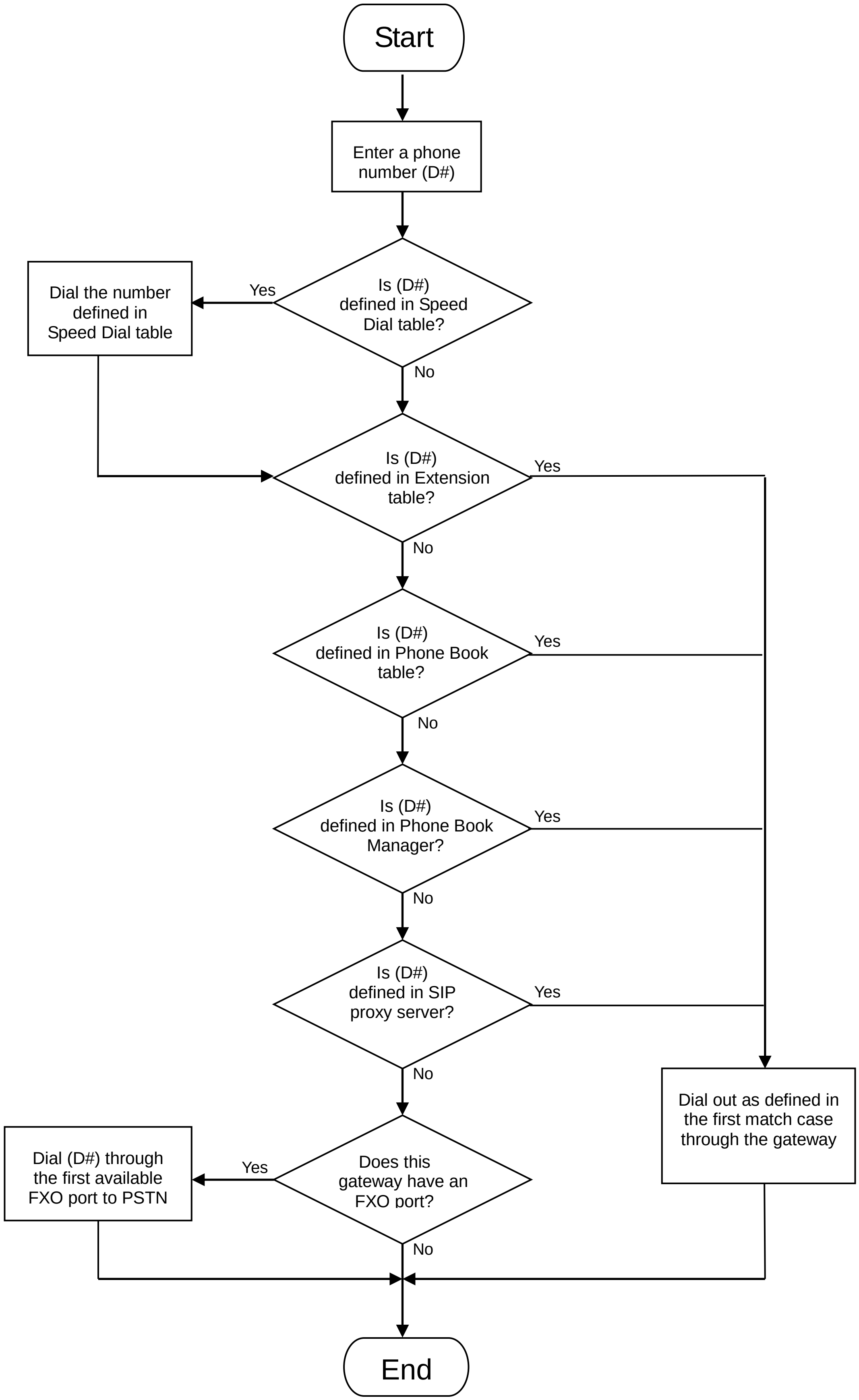
The phone number should contain at least 2 digits (not including * and #).

## 5-2 Dialed Number Processing Flow

To achieve maximum flexibility, the number dialed will be looked up in several tables defined by the VoIP Gateway. If no match is found from Digit Map Table, it will then look up the number from another table and to the registered SIP Proxy Server. The number look up flow is shown below:




A complete flow chart is shown on the next page.


Start
Enter a phone number (D#)
Is (D#) defined in Speed Dial table?
Yes
Dial the number defined in Speed Dial table
No
Is (D#) defined in Extension table?
Yes
No
Is (D#) defined in Phone Book table?
Yes
No
Is (D#) defined in Phone Book Manager?
Yes
No
Is (D#) defined in SIP proxy server?
Yes
No
Dial out as defined in the first match case through the gateway
Does this gateway have an FXO port?
Yes
Dial (D#) through the first available FXO port to PSTN
No
End

# Appendix

## Product Features

***WAN***

- One 10/100/1000Mbps auto-negotiation, auto-crossover RJ-45 Ethernet port
- Support static IP, PPPoE and DHCP address assignment and dynamic DNS (DDNS)
- QoS: IP TOS (Type of Services) and DiffServ (Differentiated Services) for both SIP signaling and RTP
- NAT Traversal : STUN and Outbound Proxy
- NTP: (Network Time Protocol RFC 1305)
- Time Zone Support
- MAC Address Clone
- RTP Packet Summary: packet sent, packet received, packet loss for voice quality analysis

***LAN***

- Two 10/100/1000 Mbps auto-negotiation, auto-crossover RJ 45 Ethernet ports
- Supports router and bridge mode
- DHCP server

***Voice Features***

- SIP (RFC3261) compatible
- Voice codecs : G.711 a /u law, G.726, G.729A, G.723.1
- CNG (Comfort Noise Generation)
- VAD (Voice Activity Detection)
- G.165/G.168 echo cancellation
- Adjustable Jitter Buffer and programmable Gain Control
- In-Band DTMF, Out-Of-Band DTMF relay (RFC2833, SIP INFO)
- Multiple SIP Proxy server entries with failover mechanism
- Polarity reversal generation
- T.30 (G.III) / Real time T.38 / Secured T.38 FAX relay
- DTMF, FSK (Bellcore ,ETSI and NTT) Caller ID Detection.
- Support Caller ID Restriction (CLIR)
- Digit Map for dial plan
- Speed Dial
- Local phone book for peer-to-peer calling
- E.164 Numbering & ENUM support
- Hot-Line, Warm-Line support
- Single Number / Account (reprehensive number) for multiple ports
- Call features:
    - Call Forward - Unconditional, Busy
- Analogue interface
    - Connector: RJ-11

**Configuration & Maintenance**

- Configuration methods:
  - Web
  - IVR
  - Telnet
- Status reports:
  - Port status
  - Registration status
  - Ping tests
  - Hardware / software information
- Firmware Upgrade through TFTP, FTP and HTTP server
- Configuration Backup/Restore
- Reset button (with restore factory default function)
- Front Panel LED : Power, Alarm, Register, WAN, LAN and FXO port