



**DVG-3032S/3016S**  
**VoIP Gateway**

**User's Manual**

Version 1.0  
(May 2011)

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### **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 Funkstörungen 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 caso, 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

<b>1. Introduction</b>	<b>4</b>
1-1 Product Overview	4
1-2 Hardware Description	5
<b>2. Getting Started</b>	<b>7</b>
<b>3. VoIP Gateway Web Configuration</b>	<b>13</b>
3-1 Status	13
3-1-1 Current Status	13
3-1-2 RTP Packet Summary	14
3-1-3 System Information	15
3-1-4 Routing Table	16
3-1-5 LAN Client	16
3-2 General Settings	17
3-2-1 WAN	17
3-2-2 LAN	21
3-2-3 SIP	22
3-2-4 SIP Advanced	26
3-2-5 Caller ID	30
3-2-6 Hot Line	31
3-2-7 Line settings	32
3-2-8 FAX	35
3-2-9 Calling Features	37
3-2-10 Phone Book	38
3-2-11 CDR Settings	39
Advanced Settings	40
Advanced Settings	40
3-2-12 Codec setting	40
3-2-13 Digit Map	41
3-2-14 DTMF & PULSE	44
3-2-15 CPT / Cadence	45
3-2-16 TR069	46
3-2-17 Caller Filter	48
3-2-18 Static Route	49
3-2-19 QoS Settings	50
3-2-20 DDNS	51
3-2-21 NAT Traversal	52
3-2-22 DoS Protection	53
3-3 SNMP	54
3-4 Tools	55
3-4-1 Ping Test	55
3-4-2 STUN Inquiry	56
3-5 System Settings	57
3-5-1 NTP	57
3-5-2 Login Account	58
3-5-3 Backup / Restore	59
3-5-4 System Log	60
3-5-5 Save / Restart	60
3-5-6 Software Upgrade	61
3-5-7 Logout	61
<b>4. Configuring the VoIP Gateway through IVR</b>	<b>62</b>
4-1 IVR (Interactive Voice Response)	62
4-2 IP Configuration Settings—Set the IP Configuration of the WAN Port	64
<b>5. Dialing Principles</b>	<b>66</b>
5-1 Dialing Options	66
5-2 Dialed Number Processing Flow	66
<b>Appendix</b>	<b>68</b>
Product Features	68

# 1. Introduction

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## 1-1 Product Overview

The DVG-3032S/3016S 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-3032S/3016S 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-3032S/3016S 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.

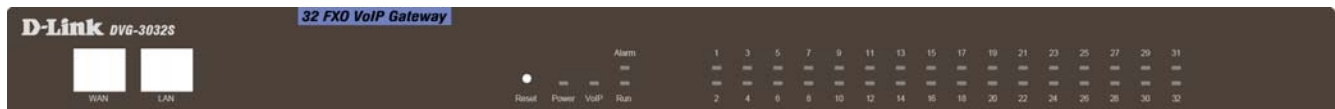
# 1-2 Hardware Description

## Front Panel

DVG-3016S



DVG-3032S



## Indicators

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

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

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

**Run:** The heartbeat, a blinking light indicates the VoIP Gateway keep running.

**WAN:** When a connection is established the 10 or 100 LED will light up solid. The LED will blink to indicate activity. If the 10 or 100 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 10 or 100 LED (bottom) will light up solid on the appropriate port. The LEDs will blink to indicate activity. If the 10 or 100 LED does not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

**Phone:**

Green Blinking – FXO is detecting incoming alerting.

Green Solid – The line is in use.

## Connectors

**WAN:** Connect to your broadband modem using an Ethernet cable.

**LAN:** Connect to your Ethernet enabled computers using Ethernet cabling.

**Reset:** Press and hold the reset button for 5 seconds., release button as Alarm is blinking. That DVG will restore to factory default.

**To restore factory default settings:**

1. Press and hold the reset button for 6 seconds.
2. Release the reset button. Factory settings will be restored.

## Rear Panel

DVG-3016S



DVG-3032S



**FXO Connector:** Connect to your phones using attached phone cabling.

**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.

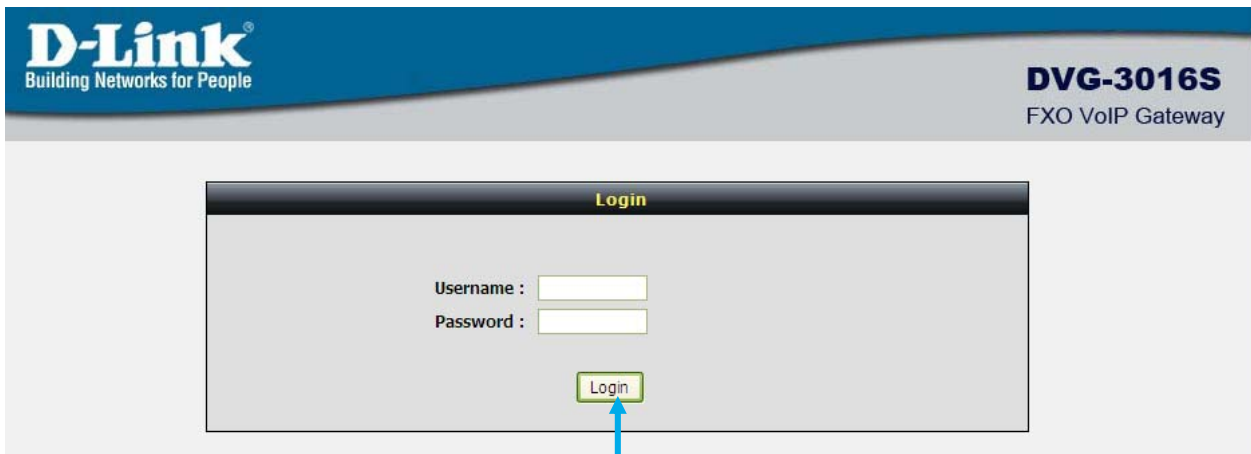
**Power Receptor:** Receptor for AC power 100V- 240V.

## 2. Getting Started

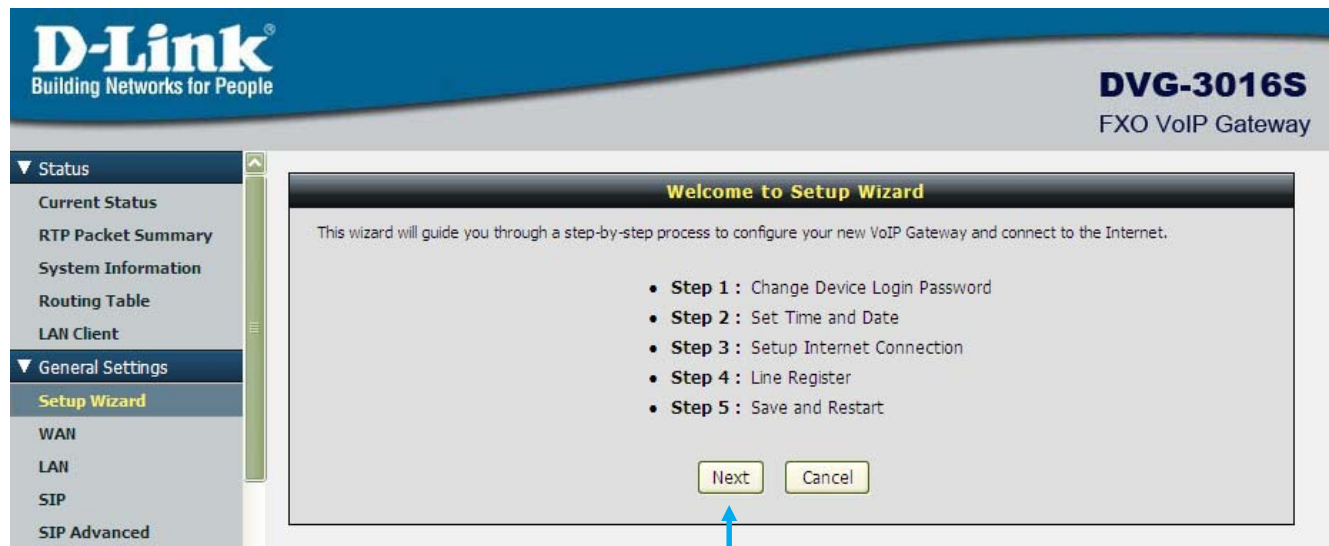
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To access the web-based configuration utility, open a web browser such as Internet Explorer and enter the IP address of the DVG-3032S/3016S.

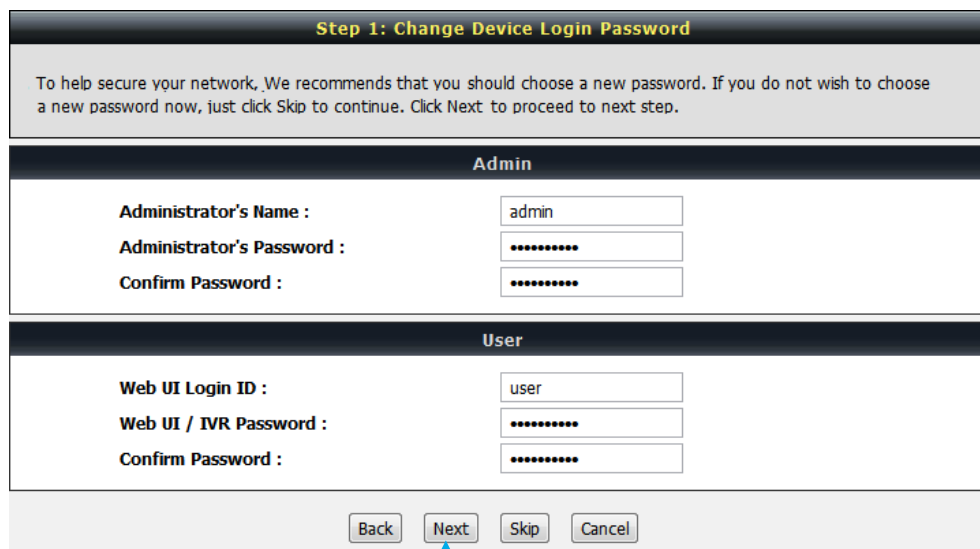
Open your Web browser and type <http://192.168.8.254> into the URL address box. Press the Enter or Return Key.



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**.



**Step 2: Set Time and Date**

The Time Configuration option allows you to configure, update, and maintain the correct time on the internal system clock. From this section you can set the time zone that you are in and set the NTP (Network Time Protocol) Server.

**Time Server**

NTP time server 1 :

NTP time server 2 :

NTP time server 3 :

**Time Configuration**

Current Router Time : 1970/ 1/ 1 08:18:27

Time Zone : + 8 00

Enable Daylight Saving

Daylight Saving Offset: 0:00

Daylight Saving Dates :

	Month	Week	Day	Time
Start	Jan	1st	Sun	12 am
End	Jan	1st	Sun	12 am

Back   Next   Cancel

Enter a NTP server or use the default server.  
Click **Next**.

**Step 3: Setup Internet Connection**

Use this section to configure your Internet Connection type. If you are unsure of your connection method, please contact your Internet Service Provider.

**WAN Settings**

	Type	<input type="checkbox"/> Enable VLAN Tagging	
		VLAN ID	Priority
WAN	Static IP	1	0

**WAN Settings**

IP address : 192.168.1.2  
Subnet mask : 255.255.255.0  
Default Gateway IP : 192.168.1.254  
MTU : 1500  
Domain Name Server (Primary) IP : 168.95.1.1  
Domain Name Server (Secondary) IP :

**WAN Link Speed**

WAN Link Speed : Auto

**MAC**

Factory Default MAC Address : 00:0C:2A:03:1A:32   
Your MAC Address : 00:90:CC:E7:2E:40   
Current MAC Address :  (xx:xx:xx:xx:xx:xx)

Select your Internet connection type:  
**DHCP** – Most Cable ISPs or if you are connecting the DVG-3032S/3016S 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**.

**Step 4: Line Register**

The VoIP Gateway can invite register to a VoIP trunk gateway or register by each port of phone. Please contact your ITSP.

**Soft Switch Setting**

Enable Support of SIP Proxy Server / Soft Switch

**Line**

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password
1	FXO	701 <input type="text" value="auto"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
2	FXO	702 <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
15	FXO	715 <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
16	FXO	716 <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>

**SIP Proxy Server**

Proxy Server IP / Domain :

Proxy Server Port :  (1-65535)

Proxy Server Realm :

TTL (Registration interval) :  (10-7200s)

SIP Domain :

Use Domain to Register

**Outbound Proxy Support**

Outbound Proxy Support

Outbound Proxy IP / Domain :

Outbound Proxy Port :  (1 - 65535)

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 :	192.168.1.2
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 15 - FXO Number :	715
Phone 16 - FXO Number :	716

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-3032S/3016S.

Click **Restart**.

# 3. VoIP Gateway Web Configuration

## 3-1 Status

### 3-1-1 Current Status

Status → Current Status

**Current Status**

Refresh Time (2 - 30s) :

**Port Status**

Line	Type	Extension Number	Line Status	Calls	Number	Proxy Register
1	FXO	701	Idle	0	voip: pstn:	Disabled (00:31:17)
2	FXO	702	Idle	0	voip: pstn:	Disabled (00:31:17)
15	FXO	715	Idle	0	voip: pstn:	Disabled (00:31:32)
16	FXO	716	Idle	0	voip: pstn:	Disabled (00:31:32)

**SIP Proxy Hunting Number Registration :** Disabled (00:31:32)Trunk Disabled (00:31:32)

**Server Registration Status**

**DDNS Registration :** Disabled (00:31:32)

**STUN Registration :** Disabled (00:31:32)

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.

Number-voip: It displays calling number for direction VoIP-> FXO or called number for direction FXO-> VoIP.

Number-pstn: It displays called number for VoIP-> FXO one-step directly call or calling number for direction FXO-> VoIP

For Server Registration Status, it shows the registration status of DDNS, STUN and FXS Represent Number.

### 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
21	G.711 u-law 64kbps		0	0	0	0
22	G.711 u-law 64kbps		0	0	0	0
23	G.711 u-law 64kbps		0	0	0	0
24	G.711 u-law 64kbps		0	0	0	0

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	
<b>System Information</b>	
Time and Date :	1969/12/31 12:03:48
Firmware Version :	1.02.38.57
WAN Port Information	
Factory Default MAC Address :	000175100209
Net Link :	Disconnected
IP address :	192.168.1.181
Subnet mask :	255.255.255.0
Default Gateway :	192.168.1.254
Domain Name Server :	168.95.1.1
LAN Port Information	
MAC Address :	00017510020A
IP address :	192.168.8.254
Subnet mask :	255.255.255.0
DHCP Server	
DHCP Server :	Disabled
Hardware	
Hardware Platform :	IXP435
Hardware :	A1-1.0
Driver :	1.2.2.1295.69/242

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-3032S/DVG-3016S.

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-3032S/3016S. 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 **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

WAN			
WAN Settings			
	Type	<input type="checkbox"/> Enable VLAN Tagging	
		VLAN ID	Priority
WAN	Static IP ▼	1	0
	DHCP		
	Static IP		
	PPPoE		
	PPTP		

General Settings → WAN

WAN Settings	
Hostname :	<input type="text"/>
Vendor Class ID :	<input type="text"/>
MTU :	<input type="text" value="1500"/>
Domain Name Server :	<input type="text" value="Manual"/>
Domain Name Server (Primary) IP :	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP :	<input type="text"/>

**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

WAN Settings	
IP address :	<input type="text" value="192.168.1.181"/>
Subnet mask :	<input type="text" value="255.255.255.0"/>
Default Gateway IP :	<input type="text" value="192.168.1.254"/>
MTU :	<input type="text" value="1500"/>
Domain Name Server (Primary) IP :	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP :	<input type="text"/>

**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

WAN Settings	
PPPoE Account :	<input type="text"/>
PPPoE Password :	<input type="password"/>
Confirm Password :	<input type="password"/>
PPPoE Service Name :	<input type="text"/> (Optional)
MTU :	<input type="text" value="1492"/>
Domain Name Server :	<input type="text" value="Manual"/>
Domain Name Server (Primary) IP :	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP :	<input type="text"/>

**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

WAN Settings	
PPTP IP Address :	<input type="text"/>
PPTP Subnet Mask :	<input type="text"/>
PPTP Default Gateway IP :	<input type="text"/> (Optional)
PPTP Server :	<input type="text"/>
PPTP ID :	<input type="text"/>
PPTP Password :	<input type="password"/>
Confirm Password :	<input type="password"/>
MTU :	<input type="text" value="1452"/>
Domain Name Server :	<input type="text" value="Manual"/>
Domain Name Server (Primary) IP :	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP :	<input type="text"/>
<input type="checkbox"/> Enable Dual Access :	

**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

MAC		
<b>Factory Default MAC Address :</b>	00:01:75:10:02:09	<input type="button" value="Restore"/>
<b>Your MAC Address :</b>	00:90:CC:E7:2E:40	<input type="button" value="Clone"/>
<b>Current MAC Address :</b>	<input type="text"/>	(xx:xx:xx:xx:xx:xx)

**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

**LAN**

LAN interface mode :  Router  Bridge

Enable NAT

LAN IP / LAN default Gateway :

Subnet mask :

**Enable NAT:** Tick it for NAT mode otherwise DVG will service as routing mode.

**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.

**DHCP Server**

Enable DHCP Server

IP Pool Starting Address :

IP Pool Ending Address :

IP Pool Uses Other Default Gw

IP Pool Default Gateway :

IP Pool Subnet mask :

Lease Time :  ( 1 - 9999 hours )

Domain Name Server Assignment :  Auto  Manual

Domain Name Server ( Primary ) IP :

Domain Name Server ( Secondary ) IP :

**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.

### 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

**Soft Switch Setting**

**Enable Support of SIP Proxy Server / Soft Switch**

**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

FXO Representative Number registers to Proxy:

Line						
Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password
Gateway Number		<input type="text"/>				
	FXO Representative Number	<input type="text" value="070223311"/>	<input checked="" type="checkbox"/>		<input type="text"/>	<input type="password" value="....."/> <input type="password" value="....."/>
1	FXO	<input type="text" value="701"/> <input type="button" value="auto"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password" value="....."/> <input type="password" value="....."/>
2	FXO	<input type="text" value="702"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password" value="....."/> <input type="password" value="....."/>
15	FXO	<input type="text" value="715"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password" value="....."/> <input type="password" value="....."/>
16	FXO	<input type="text" value="716"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	<input type="password" value="....."/> <input type="password" value="....."/>

Assuming that your registered ID and password are individual, the settings should be as above.

- **Gateway Number:** Register to a **Phone Book Manager** with this number.
- **FXO Representative Number:** Register all FXO ports as a hunting group.
- **Register:** Register to proxy if selected.
- **Invite with ID / Account:** The DVG-3016S/3032S can invite to a VoIP trunk gateway without registering to a proxy. Please contact your VoIP service provider.

**NOTE:** Please ensure that if Proxy Server allows one account for many ports using before using representative number to register.

General Settings → SIP

**Each line registers to Proxy independently:**

Line						
Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password
Gateway Number		<input type="text"/>				
FXO Representative Number		<input type="text" value="FXO"/>	<input type="checkbox"/>		<input type="text"/>	<input type="password"/> <input type="password"/>
1	FXO	<input type="text" value="070123451"/> <input type="button" value="auto"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
2	FXO	<input type="text" value="70123452"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
15	FXO	<input type="text" value="70123465"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>
16	FXO	<input type="text" value="70123466"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="password"/> <input type="password"/>

**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 DVG 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 servers. 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 servers.

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

General Settings → SIP

SIP Proxy Server	
Proxy Server IP / Domain :	<input type="text" value="192.168.1.1"/>
Proxy Server Port :	<input type="text" value="5060"/> (1-65535)
Proxy Server Realm :	<input type="text"/>
TTL (Registration interval) :	<input type="text" value="600"/> (10-7200s)
SIP Domain :	<input type="text"/>
<input type="checkbox"/> Use Domain to Register	
Bind Proxy Interval for NAT :	<input type="text" value="0"/> (0-1800s)
<input type="checkbox"/> Initial Unregister	
<input type="checkbox"/> Unregister All Contacts	
<input type="checkbox"/> Support Message Waiting Indication (MWI)	
MWI Subscribe Interval :	<input type="text" value="7200"/> (0=disable, 60-86400s)

**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.

**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-3032S sends un-register request to SoftSwitch which the contact field filled with a start sign(\*) to un-register all ports in this DVG-3032S..

**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.

**MWI Subscribe Interval:** It is used to set the subscribe time for the VoIP Gateway to check the voice mail.



General Settings → SIP

Outbound Proxy Support	
<input type="checkbox"/> <b>Outbound Proxy Support</b>	
Outbound Proxy IP / Domain :	<input type="text"/>
Outbound Proxy Port :	<input type="text" value="5060"/> (1-65535)

**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.

General Settings → SIP

<input type="checkbox"/> <b>Enable P-Asserted</b>	
Privacy Type :	<input type="text" value="id"/>

**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'.

### 3-2-4 SIP Advanced

General Settings → SIP Advanced

SIP Advanced	
<b>Listen Port UDP :</b>	<input type="text" value="5060"/> (1-65535)
<b>RTP Starting Port UDP :</b>	<input type="text" value="9000"/> (1-65500)
<b>SIP Transport Protocol :</b>	<input type="text" value="UDP"/> ▼

**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.

**For example,** if the starting port is 9000, then Line 1 will take up ports 9000 and 9001, and Line 2 will take up ports 9002 and 9003, and so forth.

**SIP Transport Protocol:** DVG supports UDP and TCP for SIP signaling. Most of SIP Server support UDP, if you prefer TCP please make sure whether remote party supports TCP or not.

General Settings → SIP Advanced

E.164	
<b>International Call Prefix Digit :</b>	<input type="text"/>
<b>Country Code :</b>	<input type="text" value="Other ()"/> ▼ <input type="text"/>
<b>Long Distance Call Prefix Digit :</b>	<input type="text"/>
<b>Area Code :</b>	<input type="text"/>
<input type="checkbox"/> <b>E.164 Numbering (To Invite Proxy)</b>	
<b>ENUM Header Exception :</b>	<input type="text" value="070"/>

**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 Timer	
Session Expiration :	<input type="text" value="0"/> ( 0 = disable, 10 - 1800 s )
Session Refresh Request :	<input checked="" type="radio"/> UPDATE <input type="radio"/> re-INVITE
Session Refresher :	<input checked="" type="radio"/> UAS <input type="radio"/> UAC

**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 Timeout Adjustment	
SIP Message Resend Timer Base :	<input type="text" value="0.5"/> s
Max. Response Time for Invite :	<input type="text" value="8"/> ( 1 - 32 )

**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

SIP Proxy Server / Soft Switch Settings	
<input type="checkbox"/> <b>VoIP failure announcement</b>	

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

General Settings → SIP Advanced

**Supplementary Features**

- Anonymous Caller ID (CLIR)
- CLIR At Transit In W/O Caller ID
- VoIP Call Out Notification
- Enable Built-in Call Hold Music
- Call On Hold Notification
- Enable Non-SIP Inbox Call
- Invite URL need 'user=phone'
- Reliability of Provisional Responses
- Compact Form

**SIP Caller ID Obtaining :** Remote-Party-Id Display Name ▼

- Put Caller ID In URI
- INVITE With Remote-Party-ID Header

**Callee Quick Media** Disable ▼

- Enable SIP 'rport' (RFC 3581)
- Support URI Percent-Encoding (RFC 3986)
- Compare SIP 'To' Header for Transit Out
- Call Hold Compatible With RFC 2543
- Enable SIP 'Allow' Header
- Enable SDP 'ptime' Attribute
- Use Redirect URI As 'To' Header (Receiving 3XX)
- Respond 'BUSY HERE' while no line available for hunting

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

**CLIR At Transit In W/O Caller ID:** When not enabled, if the FXO can detect caller ID in a call from PSTN, DVG will use the detected caller ID as caller identification when it makes transit in calls; if FXO cannot detect caller ID in a call from PSTN, the gateway will use “anonymous” as caller identification for transit in calls. When it enabled, the gateway will always uses “anonymous” as caller identification for transit in calls.

**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:** FXS 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 detected 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-3032S will send RTP to remote party immediately as user answer an inbound call.

**Enable SIP 'rport':** Tick this option for DVG to take 'rport' in SIP message body.

**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.

**Compare SIP 'To' Header for Transit Out:** When FXO is called and the number of Request line and "To" is different, the system will use the number of "To" to dial out. Please consult your Proxy Server Provider or ITSP about the format of invite packet from Proxy.

**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

**Caller ID**

FXO Caller ID Detection

Detection Level :

FSK Caller ID Type :

**Transit In Caller ID Strip / Replace**

Scan code("?" = single digit ; "%" = wildcard)	Substitute
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

**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.

**Transit In Caller ID Strip/ Replace:** Used to replace Caller ID for transit-in call from FXO to VoIP.

### 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	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>
2	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>
15	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>
16	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>

**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 there is incoming call from PBX line to FXO port(Calls from PBX Line to FXO then direct forward to an assigned VoIP number).

**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					
Line	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	
1	FXO	0 <input type="button" value="v"/> All	0 <input type="button" value="v"/> All	5 <input type="button" value="v"/> All	
2	FXO	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	
15	FXO	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	
16	FXO	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	

Line	Type	Flash Time [50-900ms]	Polarity Reversal	PSTN Answer Detection	PSTN Ring OFF Length [1000-20000ms]
1	FXO	600 <input type="button" value="All"/>	<input type="checkbox"/> <input type="button" value="All"/>	Disable <input type="button" value="v"/> All	4000 <input type="button" value="All"/>
2	FXO	600	<input type="checkbox"/>	Disable <input type="button" value="v"/>	4000
15	FXO	600	<input type="checkbox"/>	Disable <input type="button" value="v"/>	4000
16	FXO	600	<input type="checkbox"/>	Disable <input type="button" value="v"/>	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 flash time for DVG generator hook-flash at FXO ports.

**Polarity Reversal:** As a remote site answers calls from this extension the FXO port will reverse polarity.

**PSTN Answer Detection:** It works as Polarity Reversal is enabled and select this field as Polarity Reversal. As there is one step dial outgoing call from VoIP to FXO port (Calls from VoIP transit out via FXO to PBX line), GW will response 183 after FXO dial out. Then GW response 200 ok as FXO detect polarity reversal (usually caused by PBX called party answer the call)

**PSTN Ring OFF Length:** FXO/PSTN make out if the call from PSTN hangs up before VoIP remote party answer the call.



General Settings → Line settings

<b>Ring (Early Media) Time Limit :</b>	<input type="text" value="90"/> (10-600s)
<b>A Tone Force Dial Time :</b>	<input type="text" value="0"/> (0-30s)
<input type="checkbox"/> <b>Enable End of Digit Tone</b>	
<b>Force Calling Thru PSTN Code :</b>	<input type="text"/>
<b>Trunk Early Media Option :</b>	One Way Voice <input type="button" value="v"/>
<input checked="" type="checkbox"/> <b>Early Media Treatment</b>	
<input type="checkbox"/> <b>Aggressive Ring Detection</b>	
<input checked="" type="checkbox"/> <b>Detect FXO Line Presence</b>	
<b>VoIP Centrex Extension Digit Count :</b>	<input type="text" value="0"/> (0=disable, 1-30)
<b>VoIP Centrex Digit :</b>	<input type="text"/>
<b>Metering Pulse Type</b>	Disable <input type="button" value="v"/>
<b>Metering Pulse Period</b>	<input type="text" value="0"/> s

**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

Termination Impedance	
<b>FXO Impedance :</b>	<input type="text" value="600 Ohm"/> 

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

General Settings → Line settings

Drop Inactive Call	
<b>Silence Detection Threshold :</b>	<input type="text" value="0"/> (0= disable , 1 - 60 dB )
<b>Drop Silent Call Timeout :</b>	<input type="text" value="0"/> (0= disable , 1 - 3600 s )

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

Voice Menu Options	
<input checked="" type="checkbox"/> <b>Enable IVR Option</b>	

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

## 3-2-8 FAX

General Settings → FAX

Fax / Modem			
Line 1 :	T.38 Fax	Line 2 :	T.38 Fax
Line 3 :	T.38 Fax	Line 4 :	T.38 Fax
Line 13 :	T.38 Fax	Line 14 :	T.38 Fax
Line 15 :	T.38 Fax	Line 16 :	T.38 Fax

FAX	
<input type="checkbox"/>	Switch FAX On CED Detection
<input type="checkbox"/>	Restrict T.38

- **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.

**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.

General Settings → FAX

Fax T.38	
High Speed Redundancy :	1
Low Speed Redundancy :	1
FAX Max Rate :	14400
High Speed Packet Time :	40

Fax T.30	
FAX Codec :	G.711 u-law 64kbps
T.30 V.152 Payload Type :	96 (96-127)
FAX Jitter Buffer :	200 (60-1200 ms)

**High Speed Redundancy:** Set redundancy packets for FAX image. It could repair FAX image for non-continuous packets lost. The higher redundancy the higher bandwidth required.

**Low Speed Redundancy:** Set redundancy packets for FAX handshaking signaling.

**FAX Codec:** Select **G.711 a-law** or **G.711 u-law** 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

Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward
	FXO Representative number		<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	
1	FXO	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	(N/A)
2	FXO	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	(N/A)
15	FXO	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	(N/A)
16	FXO	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	(N/A)

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

**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

Phone Book			
1 - 20			
Gateway Name	Gateway Number	IP / Domain Name	Port
<input type="text"/>	<input type="text"/>	<input type="text"/>	5060
<input type="text"/>	<input type="text"/>	<input type="text"/>	5060
<input type="text"/>	<input type="text"/>	<input type="text"/>	5060
<input type="text"/>	<input type="text"/>	<input type="text"/>	5060

1 - 20 [21 - 40](#) [41 - 60](#) [61 - 80](#) [81 - 100](#)

**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

**CDR Settings**

**Send record to CDR Server**

CDR Server IP / Domain :

Port :

RADIUS Accounting Port :

RADIUS Server Secret :

RADIUS User ID :

RADIUS Password :

Accept    Reset    Default

**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.

## Advanced Settings

### 3-2-12 Codec setting

Advanced Settings → Codec settings

Codec

**Jitter Buffer :**  (60 - 1200ms)

**Silence Detection / Suppression**

**Echo Cancellation**

Enable	Codec	Codec Priority	Type	Packet Interval (ms)	Approximate Bandwidth Required (kbps)
<input type="checkbox"/>	G.711 u-law	1 ▼		10 ▼	107.2
<input checked="" type="checkbox"/>	G.723.1	5 ▼	G.723.1 6.3k ▼	30 ▼	20.8
<input checked="" type="checkbox"/>	G.726 32K	3 ▼		20 ▼	53.6
<input checked="" type="checkbox"/>	G.729	4 ▼		20 ▼	29.6
<input checked="" type="checkbox"/>	G.711 a-law	2 ▼		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.

**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.



### 3-2-13 Digit Map

**Digit Map is dialing rules for calls from FXO to VoIP.** It 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

**Digit Map**

**Enable Pound Key ' #' Function**

**Max. Dial Length :**  ( 1 - 30 )

**Default Call Route :**

**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

**Digitmap**

Enable	Scan Code	VoIP Dial-out	User Dial Length	Route
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	10 ▼	VoIP ▼
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	10 ▼	VoIP ▼
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	10 ▼	VoIP ▼
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	10 ▼	VoIP ▼

1 - 20 [21 - 40](#) [41 - 60](#) [61 - 80](#) [81 - 100](#)

**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
<input checked="" type="checkbox"/>	09	0911888997	2	VoIP
<input type="checkbox"/>			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
<input checked="" type="checkbox"/>	09	0911888997	2	VoIP
<input checked="" type="checkbox"/>	2???	35106???	4	VoIP
<input type="checkbox"/>			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
<input checked="" type="checkbox"/>	09	0911888997	2	VoIP
<input checked="" type="checkbox"/>	2???	35106???	4	VoIP
<input checked="" type="checkbox"/>	0%	1805%	Disable	VoIP
<input type="checkbox"/>			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
<input checked="" type="checkbox"/>	11	test	2	VoIP
<input type="checkbox"/>			10	VoIP

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

## 3-2-14 DTMF & PULSE

Advanced Settings → DTMF & PULSE

<b>Dial Wait Timeout :</b>	<input type="text" value="10"/> ( 1 - 60 s )
<b>Inter Digits Timeout :</b>	<input type="text" value="4"/> ( 1 - 60 s )
<b>Minimum DTMF ON Length :</b>	<input type="text" value="80"/> ( 40 - 500 ms )
<b>Minimum DTMF OFF Length :</b>	<input type="text" value="80"/> ( 40 - 500 ms )
<b>DTMF Detection Sensitivity :</b>	<input type="text" value="3"/> ▼
<b>DTMF Output Volume :</b>	<input type="text" value="0"/> ▼
<b>FXO Dial Type :</b>	<input type="text" value="DTMF"/> ▼
<b>Pulse Dial Mark/Space Ratio :</b>	<input type="text" value="US (61:39 % )"/> ▼
<input type="checkbox"/> <b>Enable Out-of-Band DTMF</b>	
<b>Out-of-Band DTMF :</b>	<input checked="" type="radio"/> RFC 2833 <input type="radio"/> SIP Info

**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:** This variable is to set the sensitivity of the telephone keys for the VoIP Gateway to detect the DTMF.

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

**FXO Dial Type:** Choose dialing type of FXO. There are DTMF and Pulse.

**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.

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

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

### 3-2-15 CPT / Cadence

The parameters of Busy Tone Cadence Measurement 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

BTC					
<input checked="" type="checkbox"/> <b>Busy Tone Cadence Measurement</b>					
	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning
BTC # 1	0	0	0	0	<input checked="" type="checkbox"/>
BTC # 2	0	0	0	0	<input checked="" type="checkbox"/>
BTC # 3	0	0	0	0	<input checked="" type="checkbox"/>
BTC # 4	0	0	0	0	<input checked="" type="checkbox"/>
BTC # 5	0	0	0	0	<input checked="" type="checkbox"/>
BTC Detection Sensitivity	4				
BTC Volume Threshold	20 (20 - 70 dB)				

**Busy Tone Cadence Measurement and auto learning:** Provides 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 system will cut off the call. If the system often cuts off an un-finished call, select less sensitivity.

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 # 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.

### 3-2-16 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 → Provision Settings

**TR-069**

**Enable TR069**

**TR-069**

**Auto Config. Server URL :**

**ACS Username :**

**ACS Password :**

**Confirm Password :**

**Connect Provision Server During Start Up**

**Connect Provision Server Periodically**

**Auto Provision Interval :**  (60 - 604800 s)

**Random Offset :**  (0 - 1800 s)

**Provision Retry Times :**  (0=always, 1 - 99)

**Retry Interval :**  (30 - 120 s)

**Listen Port :**  (0 = disable, 1 - 65535)

**Connection Request Username :**

**Connection Request Password :**

**Confirm Password :**

**Suspend Call Service**

**TFTP Source Port :**  (1 - 65535)

**Binding Server for Trigger**

**Binding Port :**  (1 - 65535)

**Binding Interval :**  (1 - 65535 s)

**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 source port for TFTP download.

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

MAINTENANCE → Provision

<input type="checkbox"/> <b>Binding Server for Trigger</b>	
Binding Port :	<input type="text" value="10104"/> ( 1 - 65535 )
Binding Interval :	<input type="text" value="20"/> ( 1 - 65535 s )

**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.

### 3-2-17 Caller Filter

This function allows you to accept or reject any incoming call from the IP address listed in the filter rule for higher security. 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**

**Caller Filter :** Allow ▼

Enable	Filter IP address	Subnet mask
<input checked="" type="checkbox"/>	<input type="text" value="192.168.8.21"/>	<input type="text" value="255.255.255.0"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>

**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.



### 3-2-18 Static Route

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

Advanced Settings → Static Route

Static Route				
	Route	Route Mask	Next Hop IP	Interface
1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

**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.

### 3-2-19 QoS Settings

Advanced Settings → QoS

QoS	
ToS / DiffServ Settings	
ToS / DiffServ Settings :	<input checked="" type="radio"/> ToS IP Precedence <input type="radio"/> DiffServ (DSCP)
ToS IP Precedence	
Signaling Precedence :	3 (Flash) ▼
Voice Data Precedence :	5 (CRITIC / ECP) ▼

**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.

### 3-2-20 DDNS

Advanced Settings → DDNS

Dynamic DNS

**Enable Dynamic DNS**

**DDNS Group :** DynDNS DDNS Server ▼

---

DynDNS DDNS Server

**Server Address :** members.dyndns.org

**Hostname :** dyndns.org

**Login ID :**

**Password :** ●●●●●●●●

**Behind NAT**

**Custom**

**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.

### 3-2-21 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

NAT Traversal	
<b>NAT Public IP</b>	
<input type="checkbox"/> <b>Enable</b>	
NAT IP/Domain :	<input type="text"/>
<b>STUN Client</b>	
<input type="checkbox"/> <b>Enable STUN Client</b>	
STUN Server IP / Domain :	<input type="text"/>
STUN Server Port :	<input type="text" value="3478"/> ( 1 - 65535 )

**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.

### 3-2-22 DoS Protection

Advanced Settings → DoS Protection

DoS Protection	
<input checked="" type="checkbox"/> Enable DoS Protection	
Whole System Flood	
<input checked="" type="checkbox"/> SYN	<input type="text" value="50"/> (Packets/Second) ( 50 - 500 )
<input type="checkbox"/> TCP Scan	
<input checked="" type="checkbox"/> Ping of Death	
<input checked="" type="checkbox"/> ICMP Smurf	
<input type="checkbox"/> IP Spoof	

**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 SNMP

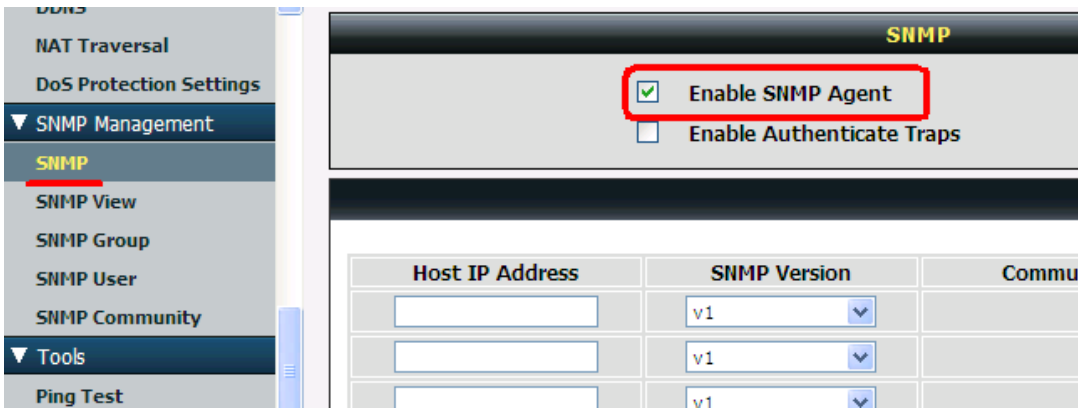
DVG support SNMP V1, V2 and V3. Please enter required parameter for SNMP V3.

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

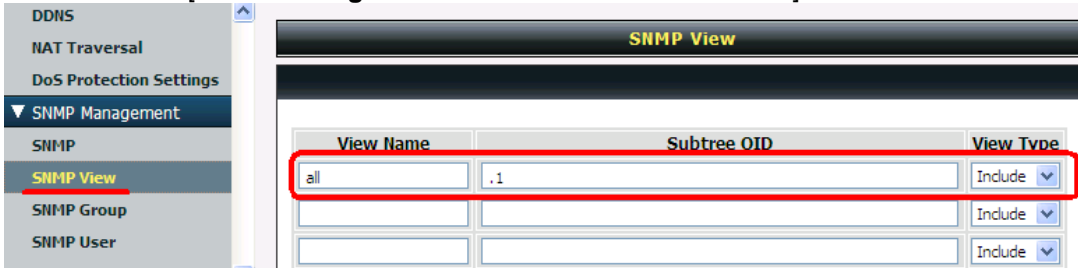
**Example for SNMP V2 configuration:**

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-> Subtree OID] as ".1"



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



### 3-4 Tools

#### 3-4-1 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

Tool → Diagnostics

**Ping Test**

<b>Ping Destination :</b>	<input type="text" value="192.168.8.254"/>	
<b>Number of Ping :</b>	<input type="text" value="4"/>	( 1 - 100 )
<b>Ping Packet Size :</b>	<input type="text" value="56"/>	( 56 - 5600 bytes )

**Result**

```
PING 192.168.8.254 (192.168.8.254): 56 data bytes
64 bytes from 192.168.8.254: seq=0 ttl=64 time=0.4 ms
64 bytes from 192.168.8.254: seq=1 ttl=64 time=0.4 ms time=0.3 ms
64 bytes from 192.168.8.254: seq=2 ttl=64 time=0.3 ms time=0.3 ms
64 bytes from 192.168.8.254: seq=3 ttl=64 time=0.3 ms time=0.3 ms

--- 192.168.8.254 ping statistics ---
4 packets transmitted, 4 packets received, 0% packet loss
round-trip min/avg/max = 0.3/0.3/0.4 ms
```

### 3-4-2 STUN Inquiry

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

Tool → STUN Inquiry

STUN Inquiry	
NAT Type :	Unknown
STUN Server IP / Domain :	<input type="text"/>
STUN Server Port :	<input type="text" value="3478"/> ( 1 - 65535 )

**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.



## 3-5 System Settings

### 3-5-1 NTP

System settings → NTP

NTP

Time Server

**Enable**

**NTP time server 1 :**

**NTP time server 2 :**

**NTP time server 3 :**

Time Configuration

**Current Router Time :** 1970/ 1/ 1 10:16:46

**Time Zone :** +

**Enable Daylight Saving**

**Daylight Saving Offset :**

	Month	Week	Day	Time
<b>Daylight Saving Dates :</b> Start	<input type="text" value="Jan"/>	<input type="text" value="1st"/>	<input type="text" value="Sun"/>	<input type="text" value="12 am"/>
End	<input type="text" value="Jan"/>	<input type="text" value="1st"/>	<input type="text" value="Sun"/>	<input type="text" value="12 am"/>

**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.

### 3-5-2 Login Account

System settings → Login Account

Login Account	
<b>Admin</b>	
Administrator's Name :	<input type="text" value="admin"/>
Administrator's Password :	<input type="password" value="••••••••"/>
Confirm Password :	<input type="password" value="••••••••"/>
<b>User</b>	
Web UI Login ID :	<input type="text" value="user"/>
Web UI / IVR Password :	<input type="password" value="••••••••"/>
Confirm Password :	<input type="password" value="••••~•••"/>

**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 :	<input type="text" value="80"/>
Web UI auto logout :	<input type="text" value="60"/> ( 30 - 300 s )
<input checked="" type="checkbox"/> Enable Web UI	
<input checked="" type="checkbox"/> Enable Telnet Service	

**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.

### 3-5-3 Backup / Restore

#### Backup Configurations File

System settings → Backup and Restore

Backup Configurations	
Configuration File :	<input type="button" value="Backup"/>
Configuration Template File :	<input type="button" value="Backup"/>

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

Restore Configurations		
Upload Configuration File :	<input type="text"/>	<input type="button" value="瀏覽..."/> <input type="button" value="Restore"/>
Restore Default Configurations :	<input type="button" value="Restore"/>	

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

### 3-5-4 System Log

System settings → System log

**System Log**

**Enable**

Server Address :

Port :  ( 1 - 65535 )

Facility:

- General
- CDR
- SIP And Provisioning

**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.

### 3-5-5 Save / Restart

#### Save and Reboot

System settings → Backup and Restore

**Save / Restart**

**Save Settings**

**Restart**

**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.

### 3-5-6 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

Software Upgrade	
Current Version :	1.02.38.57
Upgrade Server :	TFTP ▾
Server IP Address :	<input type="text"/>
Server Port :	<input type="text" value="69"/> ( 1 - 65535 )
User Name :	<input type="text"/>
Password :	<input type="text"/>
Directory :	<input type="text"/>

**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.

### 3-5-7 Logout

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

Logout

Logout
Logout
<input type="button" value="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:** to use IVR functions, dial the phone number of the FXO Port using an external line. You will hear the instruction “enter value”, and then enter IVR password. The factory default code is blank. Enter “\*\*#”. You are now in IVR setting mode..

**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 <b>114</b> 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)	
118	Restart	
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	<ul style="list-style-type: none"> <li>After entering IVR mode, dial 114.</li> <li>When voice prompt plays “Enter value”, dial 1 (to select static IP)</li> </ul>
IP address Settings	<ul style="list-style-type: none"> <li>After entering IVR mode, dial 111. When voice prompt plays “Enter value”, enter your IP address followed by “#”.</li> </ul> <p><b>Example:</b> If the IP address is 192.168.1.200, dial 192*168*1*200#.</p>
Subnet Mask Settings	<ul style="list-style-type: none"> <li>After entering IVR mode, dial 112. When voice prompt plays “Enter value”, enter your subnet mask followed by “#”.</li> </ul> <p><b>Example:</b> If the subnet mask value is 255.255.255.0, dial 255*255*255*0#.</p>
Default Gateway Settings	<ul style="list-style-type: none"> <li>After entering IVR mode, dial 113. When voice prompt plays “Enter value”, enter your default gateway’s IP address followed by “#”.</li> </ul> <p><b>Example:</b> If the default gateway is 192.168.1.254, dial 192*168*1*254#.</p>
Save Settings and Restart	<ul style="list-style-type: none"> <li>To save settings, dial <a href="#">509</a> (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 <a href="#">101</a> to check whether the IP address was retained. If the system does not play back the IP address after dialing <a href="#">101</a>, 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.</li> </ul>

### 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

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### 5-1 Dialing Options

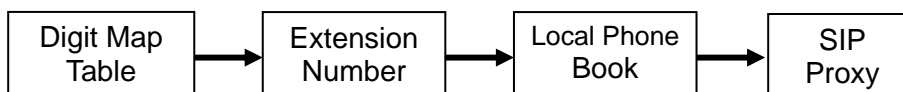
Dial the phone number which you want to call and press # to call out immediately. Note that if the “# (pound)” not dialed, the number will be called out after 4 seconds by default. The period between number dialed and call out is named “Inter Digits Timeout”. (Configurable from “DTMF and PULSE”, default=4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

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

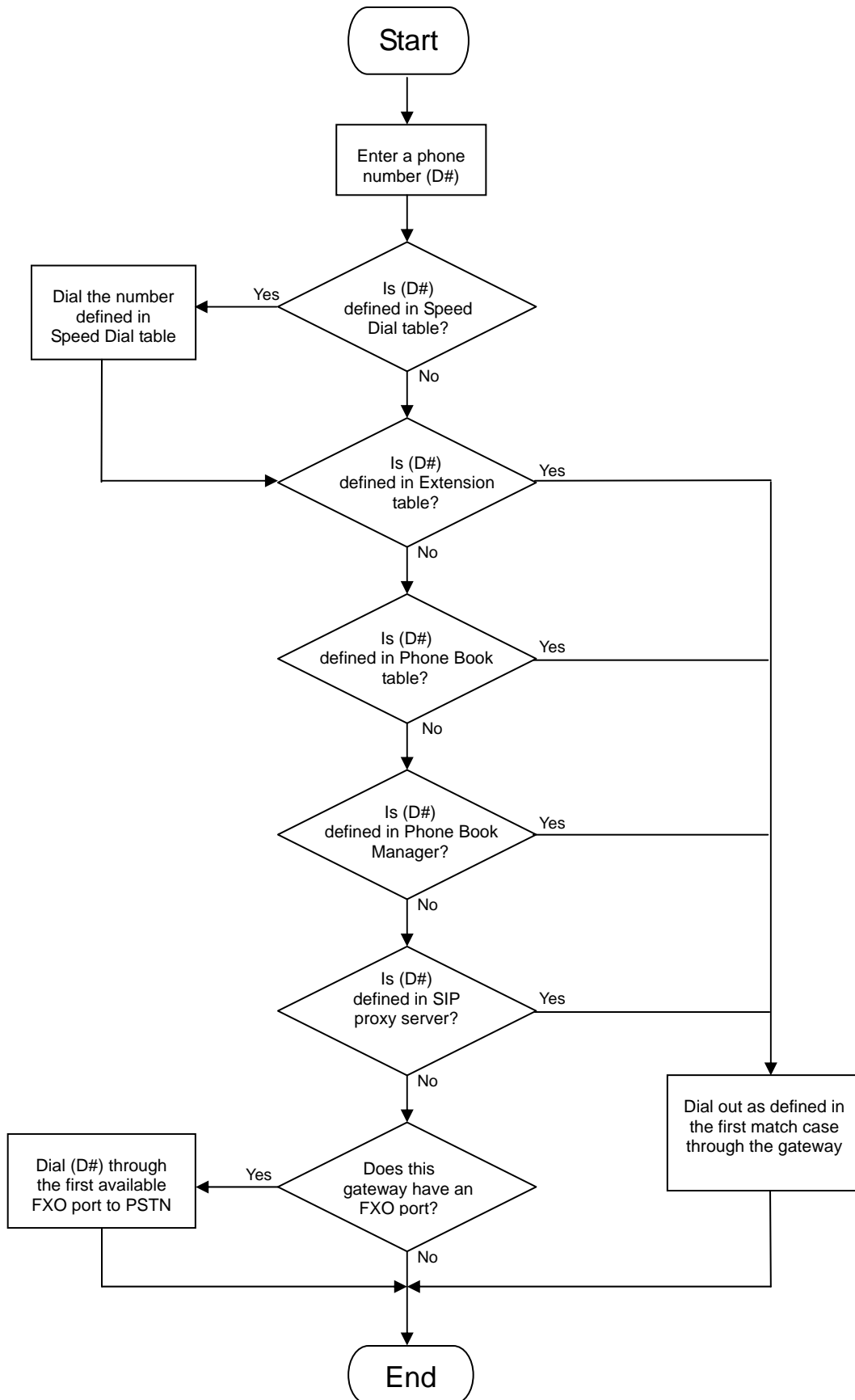
The phone number should contain at least 2 digits (not including \* and #).

### 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.



# Appendix

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## Product Features

### ***WAN***

- One 10/100Mbps 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***

- One 10/100Mbps auto-negotiation, auto-crossover RJ 45 Ethernet port
- 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.111) / Real time T.38 / Secured T.38 FAX relay
- DTMF, FSK (Bellcore ,ETSI and NTT) Caller ID generation.
- 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-21
  - Signaling protocol : Loop Start

**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, Run, Alarm, VoIP, WAN, LAN and FXO port