



DVG-3016S/3032S
VoIP Gateway

User's Manual

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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.

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1. Introduction

1-1 Product Overview

The DVG-3016S/3032S VoIP Gateway carries 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 gateway can be configured to establish a private VoIP network over the Internet without a third-party SIP Proxy Server.

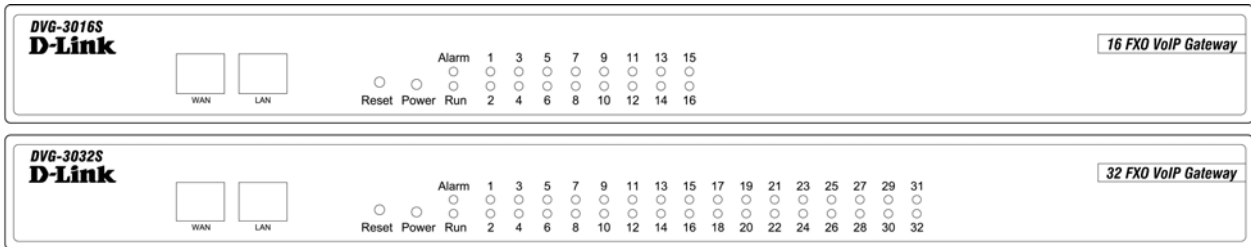
The gateway can be seamlessly integrated into an existing network by connecting to a phone set and fax machine. With only a broadband connection such as an ADSL bridge/router, a Cable Modem or a leased-line router, the gateway allows you to use voice and fax services over IP in order to reduce the cost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

DDNS support makes the gateway reachable via its domain name where an ISP dynamically assigns an IP address. By enabling the CDR function, administrators are allowed to log-in and view all call records, for example call duration, time and date of calls, and latency.

The gateway can be assigned a fixed IP address or it can have one dynamically assigned by DHCP over 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.

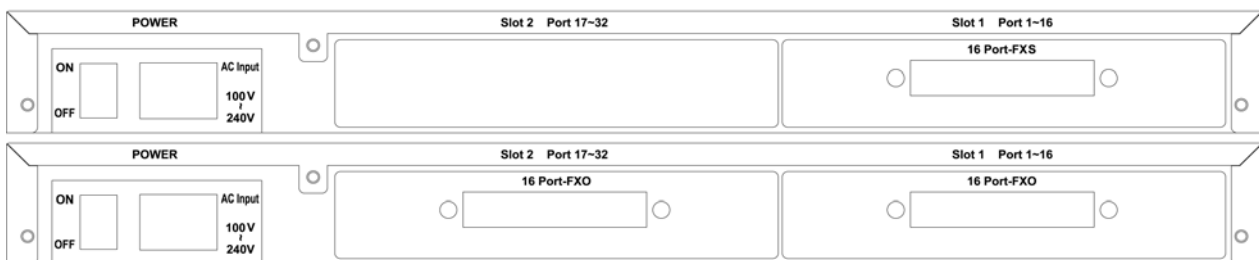
1-2 Hardware Description

Front Panel



- Power Indicator: Green light indicates a normal power supply.
 - Run Indicator: Blinking green light indicates normal operation.
 - Alarm Indicator: When the system starts up, the red light will blink. It also indicates abnormal gateway behavior.
 - 1 – 32 stands for Port1 – Port32 (Line1 – Line32).
 - WAN stands for the WAN Port Indicator.
 - LAN stands for the LAN Port Indicator.
- ✓ When starting up the system, the Alarm, Run, and Power indicators will light up. After about 40 seconds, the Alarm indicator will go off, the Run indicator will blink in green, and the Power indicator will stay green under normal operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with ISP and has yet to obtain an IP address.
 - ✓ When the WAN is connected, the WAN indicator will light up in green and if data is being transmitted over the Internet, the indicator blinks in green and orange.

Rear Panel



WARNING: DO NOT (1) connect the phone ports to each other (FXS to FXS) or (2) connect any phone port directly to a PSTN line (FXS to PSTN) or to an internal PBX line (FXS to PBX extension). Doing so may damage your VoIP gateway

Restore to factory default: (IP address, User's. Name and Password)

- (1) Pull off the power plug.
- (2) Press reset bottom (do not let go of the reset button)
- (3) Plug the plug back into the socket (do not let go of the reset button).
- (4) Let go of the reset button after 6 seconds. Factory settings will be restored.

2. Installation and Applications

2-1 Network Interface

The network interface is divided into three basic modes as described below:

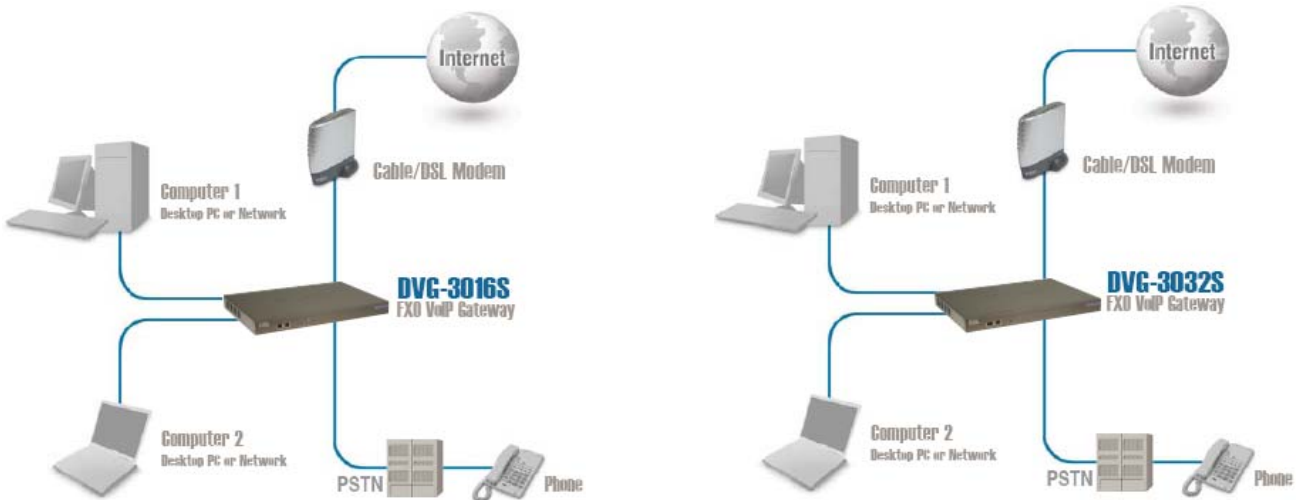
- Gateway can be assigned a Public IP Address
- Gateway can be built under the existing NAT
- Gateway can be assigned a Public IP address and serve as an IP-sharing router.

NOTE: The Network Interface function mode setup is accessed through a web UI.

Gateway Assigned a Public IP Address

The gateway will have a public IP address for Internet connection regardless of whether the IP is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dial-up / ADSL).

Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	

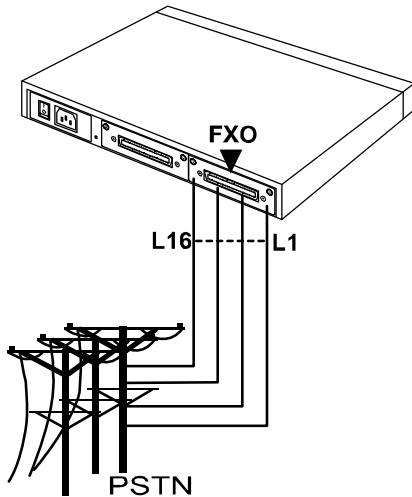


2-2 Telephone Interface Description

Example for DVG-3016S/3032S:

DVG-3016S/3032S connecting directly to the Telephone Line of a PSTN

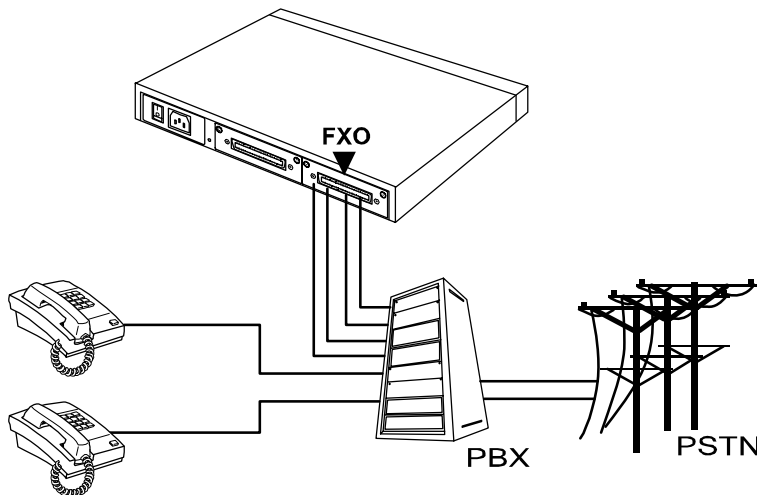
L1-L16 or L1-L32 is FXO interfaces and can all be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a user behind the Gateway.



Integrating the DVG-3016S/3032S with a PBX

DVG-3016S/3032S connecting directly to the Telephone Line of a PSTN

L1- L16 or L1-L32 is FXO interfaces, and some of them can be connected to PSTN for direct calls. Others can be connected to the PBX so other extension lines can make VoIP calls by calling extension phone number.



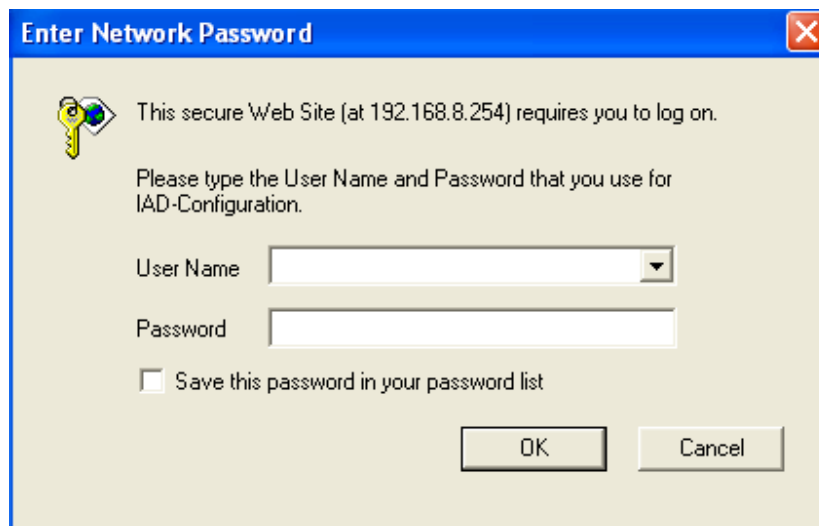
3. Gateway Configuration – Use Web Browser

The VoIP gateway allows users to configure its settings using a web interface (Web UI). You can access the Configuration Menu by opening a web-browser (e.g., Internet Explorer or Netscape Navigator) and entering the factory default LAN IP address: 192.168.8.254. The IP address of the Web UI is same as the default LAN IP noted elsewhere in this user's manual.

You can also use an ordinary telephone, connect it to the gateway, and dial "101" to inquire about the current WAN Port IP address and then use the WAN port to log-in.

Instructions

- Open a Web-Browser (e.g., Explorer, Navigator, Opera, Firefox).
- Enter the LAN port IP address. The default LAN port IP address is: 192.168.8.254.
- The log-in screen below will appear after you connect. (The factory default settings for **Login ID** and **Password** are blank (i.e., no login ID, no password).)



Enter Network Password

This secure Web Site (at 192.168.8.254) requires you to log on.

Please type the User Name and Password that you use for IAD-Configuration.

User Name

Password

Save this password in your password list

OK Cancel

The gateway does not allow multiple people to configure the gateway simultaneously. Please remember to logout or restart the system if you are not using the web configuration function.

3-1 Network Settings (WAN)

The network settings are used to set the gateway's communication ports, IP configurations, and Phone Book Manager IP settings.

Current WAN IP Address	192.168.100.200
Listen Port UDP [1 - 65535]	5060
RTP Starting Port UDP [1 - 65500]	9000

- **Current WAN IP Address:** The IP address of the WAN port.
- **Listen Port UDP:** It is not necessary to change the protocol of the communication port used by the gateway, unless it conflicts with ports used by another device in your network.
- **RTP Starting Port UDP:** The initial value of the port number for transmitting voice data among gateway(s). Each line requires 2 ports. It is not necessary to change the setting, unless it conflicts with ports used by another network device.
For example, if the starting port is 9000, then Line 1 is using ports 9000 and 9001, and Line 2 is using ports 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

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

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

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.

Setting Dynamic IP (DHCP)

DHCP <input type="radio"/>	Hostname	<input type="text"/>
	Vendor Class ID	<input type="text"/>

Click "DHCP" to obtain a Dynamic IP address, enter the "Hostname" and "Vendor Class ID" if required by your local ISP and then click the "Accept" button at the bottom of the screen.

Saving the settings: Click System Operation to select "Save Settings", "Restart", and then click the "Accept" button. Wait for about 40 seconds, and the system will obtain the required IP value from the DHCP Server.

Setting Static IP

Static IP <input checked="" type="radio"/>	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254

Select "Static IP" and enter the IP address, Subnet Mask and Default Gateway values. Then click the "Accept" button at the bottom of the screen. Save the settings, and then restart the system. Wait for about 40 seconds for the system to restart.

ADSL PPPoE Settings

PPPoE <input checked="" type="radio"/>	PPPoE Account	<input type="text"/>
	PPPoE Password	<input type="text"/>
	Confirm Password	<input type="text"/>

Select "PPPoE" and enter the Account Number, Password and Reenter Password to confirm. Then click the "Accept" button at the bottom of the screen. Save the settings, and then restart the system. The system will take about 49 seconds to restart.

PPTP Settings

PPTP <input type="radio"/>	IP address	<input type="text"/>
	Subnet mask	<input type="text"/>
	PPTP Server	<input type="text"/>
	PPTP ID	<input type="text"/>
	PPTP Password	<input type="password"/>
	Confirm Password	<input type="password"/>

Click **PPTP** and enter **IP Address**, **Subnet mask**, **PPTP Server**, **PPTP ID** and **Password**. Then click the **Accept** button at the bottom. Save the settings, and then restart the system.

BigPond (for Australia only)

BigPond Cable <input checked="" type="radio"/>	User Name	<input type="text"/>
	BigPond Cable Password	<input type="text"/>
	Confirm Password	<input type="text"/>
	Login Server	<input type="text"/>

Click "BigPond Cable" and enter the User Name and Password. Then click the "Accept" button at the bottom of the screen.

(DNS) Settings

Domain Name Server Assignment	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Domain Name Server (Primary) IP	<input type="text" value="168.95.1.1"/>
Domain Name Server (Secondary) IP	<input type="text"/>

Domain Name Server (DNS): While a gateway is accessing another gateway or a computer with a hostname, it will look up the IP address from the DNS provided by your ISP. Normally, the ISP assigns DNS information while negotiating with PPPoE or DHCP. If the DNS is not assigned automatically or the WAN port is assigned a static IP address, the DNS settings must be assigned manually.

Auto: the gateway learns primary and secondary addresses from the ISP's DHCP server or PPPoE server.

Manual: enter the primary and secondary addresses manually. Please be sure that the IP addresses are correct otherwise the gateway will not be able to access hosts using hostnames instead of IPs.

WAN Link Speed

WAN Link Speed	<input type="text" value="Auto"/>
----------------	-----------------------------------

It is used to choose the WAN Ethernet link speed. The default is "Auto". Please choose the same speed with Router/Switch/Hub when the VoIP gateway connected to Router/Switch/Hub has the link problem.

Clone MAC

Factory Default MAC Address	<input type="text" value="00AABBCCDD00"/>	<input type="button" value="Restore"/>
Your MAC Address	<input type="text" value="000C295FC915"/>	<input type="button" value="Clone"/>
Current MAC Address	<input type="text"/>	

Some Internet Service Providers (ISPs) assign bandwidth via MAC (Media Access Control) addresses. You can click the "Clone" button to type in a MAC address which will be recognized by your ISP. It is only necessary to fill in the field if it is required by your ISP.

The "Your MAC Address" field will be blank as you log-in via the WAN port.

Using Phone Book Manager

Enable Phone Book Manager Server	<input type="checkbox"/> Clients List
Share Phone Book to Clients	<input type="checkbox"/>
TTL (Expire time: mins) [0 - 60]	<input type="text" value="1"/>
Register to Phone Book Manager	<input type="checkbox"/>
VoIP failure announcement	<input type="checkbox"/>
Gateway Name for Phone Book Manager	<input type="text"/>
Phone Book Manager Login Password	<input type="text"/>
Confirm Password	<input type="text"/>
Phone Book Manager IP/Domain	<input type="text" value="192.168.1.1"/>
Phone Book Manager Server Listen Port [1 - 65535]	<input type="text" value="1690"/>

- **Enable Phone Book Manager Server:** this allows other gateway users to register the IP address and telephone number in this Phone Book manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- **Share Phone Book to Clients:** While this option is enabled and the gateway is performing as a Phone Book Manager, this gateway will append its Phone Book entries to the Manager for other clients to lookup.
- **TTL (Time to Live):** If a gateway system that is controlled by the Phone Book Manager does not report back within the deadline set by TTL, the system will be excluded from the user's list. Each gateway should report to the Phone Book Manager once every 30 seconds.
- **Register to Phone Book Manager:** To register to the Phone Book Manager.
- **VoIP failure announcement:** As soon as the registration to Phone Book Manager is failed, the gateway will drive IVR system to play out failure announcements for the caller
- **Gateway Name for Phone Book Manager:** The alias registered with the Phone Book Manager.
- **Phone Book Manager Login Password:** Enter the registered password. If this system is serving as the Phone Book Manager, the set password is also the password used for registering other gateway systems.
- **Phone Book Manager IP/Domain:** Enter the IP address for the Phone Book Manager. (This variable also supports URL (Uniform Resource Locator).)
- **Phone Book Manager Server Listen Port [1 - 65535]:** The protocol communication port for transmitting signals between the Phone Book Manager and other gateway systems. Please confirm whether or not the setting is the same as that of the **Phone Book Manager**.
-

3-2 Network Settings (LAN)

Network Settings (LAN)	
LAN IP / LAN default Gateway	<input type="text" value="192.168.8.254"/>
Subnet mask	<input type="text" value="255.255.255.0"/>
DHCP Server	
Enable DHCP Server	<input type="checkbox"/>
IP Pool Starting Address	<input type="text" value="192.168.8.1"/>
IP Pool Ending Address	<input type="text" value="192.168.8.250"/>
IP Pool Uses Other Default Gw	<input type="checkbox"/>
IP Pool Default Gateway	<input type="text" value="192.168.8.254"/>
IP Pool Subnet mask	<input type="text" value="255.255.255.0"/>
Lease Time [1 - 9999 hours]	<input type="text" value="1"/>
Domain Name Server Assignment	<input checked="" type="radio"/> Auto <input type="radio"/> Manual
Domain Name Server (Primary) IP	<input type="text"/>
Domain Name Server (Secondary) IP	<input type="text"/>
Port of Web Access from WAN [0=disable, 1 - 65535]	<input type="text" value="80"/>
Enable Web UI	<input checked="" type="checkbox"/>
Enable Telnet Service	<input checked="" type="checkbox"/>

LAN IP/Subnet mask


Network Settings (LAN)	
LAN IP / LAN default Gateway	<input type="text" value="192.168.8.254"/>
Subnet mask	<input type="text" value="255.255.255.0"/>

Gateway LAN Port IP address and subnet mask settings.

DHCP Server	
Enable DHCP Server	<input checked="" type="checkbox"/>
IP Pool Starting Address	<input type="text" value="192.168.8.1"/>
IP Pool Ending Address	<input type="text" value="192.168.8.250"/>
IP Pool Uses Other Default Gw	<input type="checkbox"/>
IP Pool Default Gateway	<input type="text" value="192.168.8.254"/>
IP Pool Subnet mask	<input type="text" value="255.255.255.0"/>
Lease Time [1 - 9999 hours]	<input type="text" value="1"/>
Domain Name Server Assignment	<input checked="" type="radio"/> Auto <input type="radio"/> Manual
Domain Name Server (Primary) IP	<input type="text"/>
Domain Name Server (Secondary) IP	<input type="text"/>

- **Enable DHCP Server:** Enable or Disable DHCP server service of the gateway.
- **IP Pool Starting Address:** The first IP address to be assigned to DHCP clients.
- **IP Pool Ending Address:** The last IP address to be assigned to DHCP clients.
- **IP Pool Uses Other Default Gw:** Tick the check box to give DHCP client the other default gateway.
- **IP Pool Default Gateway:** Assign the default gateway and subnet mask to DHCP client.
- **IP Pool Subnet mask:** Assign the default gateway and subnet mask to DHCP client.
- **Lease Time:** The valid period of an assigned IP address.
- **Domain Name Server Assignment:** The DNS information to be assigned to DHCP clients.
Auto: the gateway learns primary and secondary addresses from the ISP's DHCP server or PPPoE server.
Manual: enter the primary and secondary addresses manually. Please be sure that the IP addresses are correct otherwise the gateway will not be able to access hosts using hostnames instead of IPs.

Port of Web Access from WAN [0=disable, 1 - 65535]	<input type="text" value="80"/>
Enable Web UI	<input checked="" type="checkbox"/>
Enable Telnet Service	<input checked="" type="checkbox"/>

- **Port of Web Access from WAN:** HTTP port for WAN. To change this setting, web configuration must be accessed via the gateway's LAN port. The gateway always uses port 80 for HTTP connection via the LAN port.
-  **NOTE:** Due to network security concerns, Web Access for WAN port is disabled by default (port number "0" in this option means disable web access). To enable it, simply enter a valid port number in this field.
- **Enable Web UI:** When "Enable Web UI" is un-ticked, you cannot access from Web.
 - **Enable Telnet Service:** It is to activate Telnet access from WAN or LAN when ticked. You can configure all settings on web user interface through telnet, such as IP configuration, SIP settings, Provision settings, reset to a default stat, etc.

3-3 QoS Settings

WAN QoS

WAN QoS		
ToS / DiffServ Settings		
ToS IP Precedence <input checked="" type="radio"/>	Signaling Precedence	3 (Flash)
	Voice Data Precedence	5 (CRITIC / ECP)
DiffServ (DSCP) <input type="radio"/>	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31)
	Voice Data Value	46 (Expedited Forwarding, EF)

- **ToS/DiffServ** (Type of Service/DSCP): Voice packets have the highest priority to ensure telecommunication quality; the larger the value you set, the higher the priority.



NOTE: Please contact your ISP when you configure these values.

* **VLAN** – This feature is only supported for the special hardware.

VLAN		
Enable VLAN Tagging	<input type="checkbox"/>	
	VLAN ID [1 - 4094]	Priority [0 - 7]
Voice Traffic	3	7
LAN Traffic	1	0

VLAN is optional. It works with the Router or Switch that supports VLAN. There are 2 VLAN groups: Voice and data (LAN). This improves efficiency of network performance and security.

Enable VLAN Tagging: It is to tag the packets for VLAN Router or Switch identifying.

VLAN ID: It is to assign uniquely a user-defined ID to each packet.

Priority: It is the proprietary to VLAN Router or Switch.



Note: Please do not change anything here unless requested by your ISP.

3-4 NAT/DDNS

NAT Traversal

NAT Traversal		
NAT Public IP <input type="checkbox"/>	NAT IP/Domain	<input type="text"/>
Enable STUN Client <input type="checkbox"/>	STUN Server IP / Domain	<input type="text"/>
	STUN Server Port[1 ~ 65535]	<input type="text" value="3478"/>
Enable UPnP Control Point <input type="checkbox"/>		

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

- **NAT Public IP:** The IP address used by the gateway should be a private address. Furthermore, users must set Virtual Server Mapping in the Internet sharing device. (For example, a virtual server is usually defined as a Service Port, and all requests to this port will be redirected to this specified server's private IP address).

The default ports of the gateway are listed below:

Listen Port (UDP): 5060

RTP Starting Port (UDP): Listen Port used for telephone communication.

DVG-3016: 9000~9095

DVG-3032: 9000~9191

- **NAT IP/Domain:** Enter the NAT Server IP address (real public IP address of the Internet sharing device); 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 whether your Internet connection has static public IP addresses.

- **Enable STUN Client:** Using the STUN protocol prevents problems with setting the IP sharing function, but some NATs do not support this protocol.



NOTE: If the NAT type is "Symmetric NAT," then the gateway is not able to traverse the NAT. It is not a flaw of the gateway design, but rather a limitation of the STUN protocol.

- **STUN Server IP/Domain and Port:** Enter the STUN server IP address and Listen Port number. You can set two STUN server IPs separated by a semicolon.
- **Enable UPnP Control Point:** This variable will enable the gateway's IP traffic to pass through an Internet sharing device. This function only works when the Internet sharing device supports UPnP and has it enabled.

DDNS

Register to DDNS

DynDNS DDNS Server Default

Server Address:

Hostname:

Login ID:

Password:

Confirm Password:

Behind NAT: Yes

Custom:

TZO DDNS Server Default

Server Address:

Hostname:

E-Mail Address:

Key:

Behind NAT: Yes

3322 DDNS Server Default

Server Address:

Hostname:

Login ID:

Password:

Confirm Password:

Behind NAT: Yes

PeanutHull DDNS Server Default

Server Address:

Hostname:

Login ID:

Password:

Confirm Password:

DDNS Server Default

Server Address:

Hostname:

Login ID:

Password:


Confirm Password:

Behind NAT: Yes

These settings are only necessary when the gateway is set up behind an Internet sharing device that uses a dynamic IP address and does not support DDNS.

Choose a DDNS Server: The current system allows users to choose either DynDNS · TZO · 3322.org · PeanutHull or a private server. You will need to apply for an account with DynDNS · TZO · 3322.org · PeanutHull or a private server before you type in the following information.

- **Server address:** the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- **Hostname:** the URL of the system (or NAT) – applied from domain name registration providers (e.g. www.dyndns.org).
- **Login ID and Password:** The ID and password used to log-in to the DDNS server.
- **Behind NAT:** Select this only when the system is set up behind a NAT device.

 **NOTE:** If the 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.

Example:

NAT

NAT Traversal	
NAT Public IP <input checked="" type="checkbox"/>	NAT IP/Domain <input type="text" value="hostname.ddnsserv.com"/>

DDNS

<input checked="" type="checkbox"/> Register to DDNS	
<input checked="" type="radio"/> DynDNS DDNS Server	<input type="button" value="Default"/>
Hostname	<input type="text" value="hostname.ddnsserv.com"/>

3-5 Caller ID

FXO Caller ID Detection	<input checked="" type="checkbox"/>
Detection Level	0 ▾
FSK Caller ID Type	<input checked="" type="radio"/> Bellcore <input type="radio"/> ETSI
Anonymous Caller ID (CLIR)	<input type="checkbox"/>
CLIR At Transit In W/O Caller ID	<input type="checkbox"/>

- **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 it can.



NOTE: If you register the gateway to a Proxy, you may be unable to make a call because the gateway will not send the number for authorization.

- **FSK Caller ID Type:** Bellcore is used in Australia caller ID standards.
- **Anonymous Caller ID (CLIR):** When enabled, anyone receiving a call from you will not display your number if they have caller ID.



NOTE: If you register the gateway to a Proxy and you check this option, you may be unable to make a call. This is due to the fact that the VoIP gateway doesn't send the number for authorization.

- **CLIR At Transit In W/O Caller ID:** When not enabled, if the FXO can detect caller ID in a call from PSTN, the gateway 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 use "anonymous" as caller identification for transit in calls.

Transit In Caller ID Strip / Replace

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"/>
<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"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

- **Scan code:** Defines the rule of the Caller IDs detected by FXO. It can be a prefix or a full number.
- **Substitute:** Defines the changed display info. of the calling party while making calls to Internet by FXO.

3-6 Telephony Settings

Prefix Number Rules

Trunk Dial Out Verify	01:00
Trunk Dial Out Replace	1906001;190200
Trunk Dial Out Deny	

- **Trunk Dial Out Verify/ Trunk Dial Out Replace:** VoIP gateway will check (verify) the dial out prefix from dial out numbers and change (replace) the prefix to transit out through FXO port.

Example:

If you transit out with 01907123456, the system will transfer to 190601 907123456. If you transit out with 008621123456 the system will replace it with 190200 8621123456. The maximum digit is 40. In the example is 13 digits.

- **Trunk Dial Out Deny:** The system will deny the call with the leading number filled in this column.

Trunk Incoming Prompt Voice	<input checked="" type="radio"/> Default Greeting <input type="radio"/> Custom Greeting <input type="radio"/> Dial Tone				
Custom Greeting Upload / Backup	<input type="text"/>	<input type="button" value="Browse..."/>	<input type="button" value="Upload"/>	<input type="button" value="Backup"/>	<input type="button" value="Clear Greeting"/>
FXO Hunting VoIP call in option	Caller Indicate Dial-Out ▼				
FXO Hunting Default Dial-Out	<input type="text"/>				
FXO Line VoIP call in option	Caller Indicate Dial-Out ▼				

- **Trunk Incoming Prompt Voice:** Select the greeting (must use the IVR 132 function to record a voice file) when FXO receives an inbound call (transit in).
- **FXO Hunting VoIP call in option:** To set FXO transit out to POTS mode. By using a fixed number assigned in “**FXO Hunting Default Dial-Out** “. Or transit out with the number sent from remote SIP UA when the VoIP call calls FXO hunting number.
- **FXO Hunting Default Dial-Out:** To set FXO default dial-out number.
This will take effect as FXO Line VoIP call in option is set to Default Dial-Out. When some one makes a call to this FXO port from Internet, it will dial to PSTN with that default number.
- **FXO Line VoIP call in options:** To set FXO dial-out mode when the VoIP call indicates the FXO extension number.
- **Caller Indicate Dial-Out:** When someone makes a call to this FXO port from Internet, it will dial to PSTN with the number assigned in SIP packet.
- **Default Dial-Out:** When some one makes a call to this FXO port from Internet, it will dial to PSTN with the number filled in “FXO Line Default Dial-Out”.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	Dial-Out Prefix	FXO Line Default Dial-Out	Fax / Modem
1	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>	T.38 Fax <input type="button" value="v"/>
2	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>	T.38 Fax <input type="button" value="v"/>
3	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>	T.38 Fax <input type="button" value="v"/>
4	<input checked="" type="checkbox"/>	FXO	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>	T.38 Fax <input type="button" value="v"/>

- **Enable:** Enable a line; if some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to this line.
- **Hot Line:** When receiving a call from an outside line, the gateway will divert the call to the assigned hotline number.
- **Hot Line No.:** Enter the hot line number for an automatic dialing function.
- **Warm Line:** When the Warm Line function is in use, user can dial a number. Otherwise the system will divert incoming calls from an outside line to the Hot Line Number after a set wait time.

Example:

Assume the assigned hotline for DVG-3016S/3032S line 1 is 701 and the Warm Line (Hot Line Delay) is 5 seconds. If no extension number is dialed within 5 seconds, the call will be automatically diverted to the assigned hotline (ext 701). The system allows users to record a voice prompt (e.g. "please enter an extension number or wait for the operator to connect you") to use in this situation.

Assume the assigned hotline for DVG-3016S/3032S line 2 is 702 and the wait time is 0 second. When Port2 receives a call from an outside line, it will be automatically diverted to extension 702.

- **Dial-out Prefix:** It is the number dialed automatically by the system when the FXO interface diverts a call to the PSTN by VoIP.
- **FXO Line Default Dial-Out:** Default number that FXO will dial out when it receive an incoming call from VoIP.

Example:

If PBX extension needs to dial "0" to make a PSTN call, and the FXO ports are connected to PBX extension. In this case, the Dial-out prefix should be set to "0". If the PBX requires some delay time before capturing a line, then the trunk prefix should be set as "0," so that after dialing a 0, it will pause for 1 second before dialing the destination number. Each comma represents a 1 second delay. If more delay time is required, simply add more commas. **Please note that if a Dial-out prefix is set, the line won't be able to dial any PBX extension line**

- **FAX/Modem:** Enable this line to detect if there is a fax tone and transfer the call to a fax line.

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(All the calls use T.30 codec of RTP stream, not need to change codec.)	Accept and change RTP to T.38
T.38 Native	Yes	re-INVITE with T.38	Accept and change RTP to T.38

NOTE: When a fax tone is detected from the call, the 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.

Trunk Hunting Order	First Idle <input type="button" value="v"/>
Enable FXO / Trunk Extension Number	<input checked="" type="checkbox"/>
Pick up Line by Dialing Extension Number	<input checked="" type="checkbox"/>
Wait for Caller ID before FXO / Trunk pick up	<input checked="" type="checkbox"/>
Transit In Busy Tone Limit [0 - 60 s]	<input type="text" value="3"/>
Detect FXO Line Presence	<input checked="" type="checkbox"/>
Ring (Early Media) Time Limit [10 - 600 s]	<input type="text" value="90"/>
Enable End of Digit Tone	<input type="checkbox"/>
VoIP Call Out Notification	<input type="checkbox"/>
Enable Built-in Call Hold Music	<input checked="" type="checkbox"/>
Force Calling Thru PSTN Code	<input type="text"/>
Trunk Early Media Option	One Way Voice <input type="button" value="v"/>
Early Media Treatment	<input checked="" type="checkbox"/>
Max. External Call	<input type="text" value="999"/>

- **Trunk Hunting Order:** To set FXO dial-out mode when there is an incoming call dialed FXO representative number.

First Idle: The gateway will assign each unassigned call from first FXO line.

Sequential: The gateway will automatically assign the first unassigned call to the first FXO line. The second FXO line will dial the second unassigned call out. Each line will be used.

- **Enable FXO/Trunk Extension Number:** Select this function only when FXO receives 2 or more different PBX or PSTN, or under special circumstances. Users are free to call out from a desired channel, if assigned. If you register to a Proxy it MUST be checked.
- **Pick up Line by Dialing Extension Number:** Allows user to dial just the FXO extension – 717 - to use when the PSTN line is connected on the FXO port. If you are registered to a Proxy, it MUST be checked.
- **Wait for Caller ID before FXO / Trunk pick up:** To detect caller ID from FXO port.
- **Transit in Busy Tone Limit:** The duration gateway plays a busy tone before FXO hook-on. To notify the caller from PSTN that this call is finished.
- **Detect FXO Line Presence:** Enable this function 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-detect line presence on FXO port while ringing.
- **Ring (Early Media) Time Limit[10 - 600secs] :** The timeout to cancel a call when no one answers.
- **Enable End of Digit Tone :** The gateway will play a “Beep-Beep” tone to notify that the call is in progress.
- **VoIP Calling Notification:** The gateway will play a tone to notify that the call is via VoIP.
- **Enable Built-in Call Hold Music:** The default setting is that when receiving a call hold request, the gateway will play music on hold. Untick the check box to disable this function while necessary.
- **Force Calling Thru PSTN code:** Dial the code to get a PSTN line for dialing out. For example: If you specify “33” in this option and would like dial “23456789” via a PSTN line, dial “33 23456789”
- **Trunk Early Media Option:** Early Media refers to media that is generated prior to connection or answer of a call is established by the called party. It may be unidirectional or bidirectional, and can be generated by the caller, the callee, or both. The gateway supports three early media mechanisms. These mechanisms occur from the moment “200 OK” being sent in response to an “INVITE” message.
 - **Both Way Voice:** Use bidirectional 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:** If this variable is disabled, the system will send RTP immediately after a connection with a proxy is set up. The default setting is enabled. If communicating with other gateways encounters problems, please disable this function.
- **Max. External Call:** To control network voice quality according to bandwidth, defines the maximum concurrent Internet call is allowed by the gateway.

3-7 SIP

FXO Representative Number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
Gateway Number		10390376					
FXO Representative Number		FXO	<input checked="" type="checkbox"/>		account
1	FXO	701 <input type="button" value="Auto"/>	<input type="checkbox"/>	<input type="checkbox"/>	
2	FXO	702	<input type="checkbox"/>	<input type="checkbox"/>	
3	FXO	703	<input type="checkbox"/>	<input type="checkbox"/>	
4	FXO	704	<input type="checkbox"/>	<input type="checkbox"/>	

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.

Each line registers to Proxy independently:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
Gateway Number		10390376					
FXO Representative Number		FXO	<input type="checkbox"/>		
1	FXO	701 <input type="button" value="Auto"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	account
2	FXO	702	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	account
3	FXO	703	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	account
4	FXO	704	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	account

- **Number:** VoIP phone number
- **User ID/Account:** VoIP account Authentication ID or account name
- **Password:** password for VoIP account authentication
- **Invite with ID / Account:** VoIP gateway can invite to a VoIP trunk gateway without registering with a proxy. Please contact your VoIP service provider.

As there are various Proxy Server providers, our company has designed the gateway to be compatible with them, and according to RFC standards. If any registration problem occurs, please consult your Proxy Server provider.

 **NOTE:** When you register with a Proxy Server, dialing principles may vary with different Proxy Servers, especially when dialing through a remote end FXO port. Please consult your Proxy Server Provider.

DNS SRV Settings

Use DNS SRV	<input type="checkbox"/>
DNS SRV Auto Prefix	<input checked="" type="checkbox"/>
Proxy Fallback Interval [0 - 10800 s]	1800

- **Use DNS SRV:** The gateway asks for the related IP address of SIP Server from the records of DNS SRV. DNS SRV uses several servers for a single domain for SIP proxy, to move services from host to host and design some hosts as primary servers (the highest priority) for a service and others as backups. If the primary server is not reachable, the gateway will go for backup server, and so forth...
- **DNS SRV Auto Prefix:** This option tells the gateway to send packet with service type when using DNS SRV.
- **Proxy Fallback Interval:** Set the preferred Proxy Fallback Interval. After the time expires, the gateway gets back for registration with the primary server.

Enable Support of SIP Proxy Server / Soft Switch

<input type="checkbox"/> Enable Support of SIP Proxy Server / Soft Switch	
<input checked="" type="checkbox"/> Enable SIP Proxy 1	
Proxy Server IP / Domain	192.168.1.1
Proxy Server Port [1 - 65535]	5060
Proxy Server Realm	
TTL (Registration interval) [10 - 7200 s]	600
SIP Domain	
Use Domain to Register	<input type="checkbox"/>
<input type="checkbox"/> Enable SIP Proxy 2	
Proxy Server IP / Domain	192.168.1.1
Proxy Server Port [1 - 65535]	5060
Proxy Server Realm	
TTL (Registration interval) [10 - 7200 s]	600
SIP Domain	
Use Domain to Register	<input type="checkbox"/>
VoIP failure announcement	<input type="checkbox"/>
Bind Proxy Interval for NAT [0 - 180 s]	0
Initial Unregister	<input type="checkbox"/>
Support Message Waiting Indication (MWI)	<input type="checkbox"/>
MWI Subscribe Interval [0=disable, 60 - 86400 s]	7200

- **Enable Support of SIP Proxy Server / Soft Switch:** Enable the functions to inter-work with Proxy Server / Soft Switch. When SIP Proxy 1 and 2 are enabled, the system will register to SIP Proxy 2 after all lines have failed to register to SIP Proxy 1. SIP Proxy 2 is a backup system..

- **Proxy Server IP/Domain:** Enter the Proxy Server IP address or URL (Uniform Resource Locator).
- **Proxy Server Port:** Enter the Proxy Server **listen** port number. (The default value is 5060)
- **Proxy Server Realm:** This variable is used for gateway SIP account authentication in a SIP server. In most cases, the 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 here. **If you fail to make a call, please contact your VoIP service provider.**
- **TTL (Registration interval) [10-7200 s]:** Enter the desired time interval at which the gateway will report to your Proxy Server.
- **SIP Domain/Use Domain to Register:** Enter the correct SIP domain to avoid registration failure (it is not necessary to set this with some Proxy Servers). If you enable "Uses Domain to Register" the VoIP gateway will register to the proxy with the domain name you filed. Otherwise the VoIP Gateway will register to a Proxy with the IP it resolves. **If you fail to make a call, please contact your VoIP service provider.**
- **VoIP failure announcement:** As soon as the registration to proxy server is failed, the gateway will drive IVR system to play out failure announcements for the caller.
- **Bind Proxy Interval for NAT:** This function is able to keep the binding that exists when the VoIP gateway is behind a NAT and SIP Proxy is not able to keep the binding.
- **Initial Unregister:** After rebooting, the gateway is initially unregistered and then it will perform a general register process.
- **Support Message Waiting Indication:** The system will play a tone to remind users that there are messages in the SIP Server.
- **MWI Subscribe Interval:** The subscribe interval is for the gateway check of the voice mail.

Outbound Proxy Support	<input type="checkbox"/>
Outbound Proxy IP / Domain	<input type="text"/>
Outbound Proxy Port [1 - 65535]	<input type="text" value="5060"/>

- **Outbound Proxy Support :** An outbound proxy server handles SIP call signaling as a standard SIP proxy server would. Further, it receives and transmits phone conversation traffic (media) between two communication parties. This option tells the gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote gateway. 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.

Session Expiration [0=disable, 10 - 1800]	<input type="text" value="0"/>
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 variable is used to avoid billing for abnormally dropped calls due to Internet problems. The default is disabled.
- **Session Refresh Request:** Used to resend UPDATE or re-INVITE requests to the Server.
- **Session Refresher:** Selects which user agent is the session refresher. UAS (User Agent Server) is an originator, and UAC (User Agent Client) is a replier.

Enable P-Assert	<input type="checkbox"/>
Privacy Type	id

- **Enable P-Assert:** This variable is for caller ID protection.
- **Privacy Type:** Privacy type is used to disguise the caller ID when queried via an ITSP/Third-Party Assertion.

SIP Message Resend Timer Base [s]	0.5
Max. Response Time for Invite [1 - 32]	8
Invite URL need 'user=phone'	<input checked="" type="checkbox"/>
Reliability of Provisional Responses	<input type="checkbox"/>
Compact Form	<input type="checkbox"/>
SIP Caller ID Obtaining	Remote-Party-Id Display Name
Put Caller ID In URI	<input type="checkbox"/>
INVITE With Remote-Party-ID Header	<input type="checkbox"/>
Retry-After Value for Temporarily Unavailable	0
Support URI Percent-Encoding (RFC 3986)	<input type="checkbox"/>
Compare SIP 'To' Header for Transit Out	<input type="checkbox"/>
Enable SIP 'Allow' Header	<input checked="" type="checkbox"/>
Call Hold Compatible With RFC 2543	<input checked="" type="checkbox"/>

- **SIP Message Resend Timer Base:** SIP packet will resend if response did not arrive in the base time set in this column. It will send again at "base time" * 2, and send again at "base time" * 2 * 2. The maximum resend time is four seconds. Resend will stop and restart when the total resend time has reached 20 seconds.
- **Max. Response Time for Invite:** If the destination does not reply in the set time, the call is failed.
- **Invite URL need 'user=phone':** There is 'user=phone' in invite packet.
- **Reliability of Provisional Responses:** Provide information on the progress of the request processing if selected.
- **Compact Form:** It decreases the size of SIP header if selected.
- **SIP CallerId Obtaining:** Defines from which part of the SIP packet will the gateway obtain caller ID. There are several places where you can put your caller ID.
Remote-Party-Id Display Name: It is locate at SIP → Remote-Party-ID → Before [<sip:]
Remote-Party-Id User Name: It is locate at SIP → Remote-Party-ID → After [<sip:], Before [@]
From-Header Display Name: The standard way is in SIP → Message Header → From → SIP Display info.
- **Retry-After Value for Temporarily Unavailable:** To set the time interval . When ISDN PRI Gateway is full loading, ISDN PRI Gateway will inform VSP to stop sending INVITE packets and start to send INVITE packets after Retry-After Value.
- **Support URI Percent-Encoding(RFC 3986):** It follows RFC 3986 to encode/decode the letters of the basic Latin alphabet, digits, and a few special characters.
- **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.
- **Enable SIP 'Allow' Header:** Check the box to comprise the information of "Allow" in the message header of INVITE. Disable it if the call can't be connected.
- **Call Hold Compatible With RFC 2543:** Check the box to comprise c=0.0.0.0 in SDP message to be compatible with RFC2543.

E.164

International Call Prefix Digit	<input type="text"/>
Country Code	(Other) <input type="text"/> <input type="button" value="v"/>
Long Distance Call Prefix Digit	<input type="text"/>
Area Code	<input type="text"/>
<hr/>	
E.164 Numbering	To Invite Proxy <input type="checkbox"/>
	Transform to Transit Out <input type="checkbox"/>
ENUM Header Exception	<input type="text" value="070"/>

- **International Call Prefix Digit:** Enter the International call prefix.
- **Country Code:** Users please select the desired country code.
- **Long Distance Call Prefix Digit:** The long-distance prefix digit for making a long-distance call.
- **Area Code:** Please enter the area code.
- **E.164 Numbering:** This variable invites the proxy to follow the E.164 rule, but it depends on the proxy. If you fail to make a call, please contact your ITSP.



NOTE: All settings in this section are specific to your VoIP network. Please ask your VoIP service provider whether or not they require these settings.

3-8 Calling Features

Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID
FXO Representative Number			<input type="checkbox"/>	<input type="checkbox"/>	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
1	FXO	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
2	FXO	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
3	FXO	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
4	FXO	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	(N/A)	(N/A)	(N/A)	(N/A)	(N/A)

- **Do Not Disturb:** The particular line will only be able to call out when this variable is enabled. All incoming calls will be rejected.
- **Unconditional Forward:** All incoming calls will be forwarded to the “Forwarding Number” automatically.
- **Busy Forward:** Forward incoming calls to the “Forwarding Number” when the port is busy.
- **No Answer Forward:** Forward incoming calls to the “Forwarding Number” after ring timeout expires without answer.

3-9 Advanced Options

Administrator's Name	<input type="text"/>
Administrator's Password	<input type="password"/>
Confirm Password	<input type="password"/>
Web UI Login ID	<input type="text"/>
Web UI / IVR Password	<input type="password"/>
Confirm Password	<input type="password"/>
Web UI auto logout [30 - 300 s]	<input type="text" value="60"/>

NOTE: There are two operating levels when entering the Web UI. Logging-in as the Administrator allows you to change all settings. A Web UI user only has access to some settings.

- **Administrator's Name and Password:** Enter the administrator name and password, which has the highest level of control of the gateway.
- **Web UI Login ID and Web UI/IVR Password:** Enter log-in ID and password when you log-in to the Web interface/IVR of the gateway as a normal user.
- **Web UI auto logout:** If a user does not act within the effective time range when logging into the web interface, the user will be disconnected from the web page to allow others to log-in.


Dial Wait Timeout [1 - 60 s]	<input type="text" value="10"/>
Inter Digits Timeout [1 - 60 s]	<input type="text" value="4"/>
Minimum DTMF ON Length [40 - 500 ms]	<input type="text" value="80"/>
Minimum DTMF OFF Length [40 - 500 ms]	<input type="text" value="80"/>
DTMF Detection Sensitivity	(less) <input type="radio"/> 1 <input type="radio"/> 2 <input checked="" type="radio"/> 3 <input type="radio"/> 4 <input type="radio"/> 5 (more)

- **Dial Wait Timeout:** Use this variable to set the wait time for the user's first key pressing when dialing a number. The user will hear a busy tone if he or she does not press the first key within the set time frame.
- **Inter Digits Timeout:** Set the waiting time between each key press. The numbers input up to that point will be dialed after the timeout.
- **Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones):** Used to set the dial tone when a call is being diverted to another extension.
- **DTMF Detection Sensitivity:** Used to adjust the sensitivity of the telephone keys.

FXO Dial Type	<input type="text" value="DTMF"/>
Pulse Dial Mark/Space Ratio	<input style="border: none; background-color: #e0e0e0; width: 100%;" type="text" value="US (61:39 %)"/>
FXO Impedance	<input style="border: none; background-color: #e0e0e0; width: 100%;" type="text" value="Taiwan 600 Ohm"/>

- **FXO Dial Type:** Choose dialing type of FXO. There are DTMF and Pulse.
- **Pulse Dial Mark/Space Ratio:** Duration and break of pulse dial ration.
- **FXO Impedance:** Choose the correct impedance in your country or area.

Enable Out-of-Band DTMF <input type="checkbox"/>	<input checked="" type="radio"/> RFC 2833 Payload Type <input type="text" value="101"/> Volume <input type="text" value="0 dB"/>
Enable Hook Flash Event	<input type="radio"/> SIP_INFO <input type="text" value="Disable"/>
Use Second CPT after SIP registered	<input type="checkbox"/>
Enable Non-SIP Inbox Call	<input checked="" type="checkbox"/>

- **Enable Out-of-Band DTMF:** To send DTMF keys (0~9, *, #,) follow the RFC2833 rules or via SIP Info.
-  **NOTE:** Out-of-Band DTMF transport method may vary with different VoIP networks, please contact your VoIP provider for their preferred method.
- **Payload Type:** payload type of RFC2833.
- **Volume:** Defines the volume of RFC2833.
- **Uses Second CPT for VoIP Call:** This function is usually applied when the user selects VoIP as the primary path for outgoing calls and PSTN as the backup. By enabling this function, the gateway will generate a different set of tones to inform the user that VoIP is in service. Should VoIP fail and fallback to PSTN, the user will hear PSTN tones instead of the second set CPT. (for CPT related settings, please refer to Trunk Management -> CPT/Cadence Settings).
- **Enable Non-SIP Inbox Call:** Untick on the check box to disable Non-SIP inbox call if all calls need to go through ITSP.

Line Settings

Line	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time FXO [30-900 ms]	Enable Polarity Reversal	PSTN Answer Detection	PSTN Ring OFF Length [1000 - 20000 ms]
1	FXO	<input type="text" value="0"/> <input type="text" value="All"/>	<input type="text" value="0"/> <input type="text" value="All"/>	<input type="text" value="5"/> <input type="text" value="All"/>	<input type="text" value="200"/> <input type="text" value="All"/>	<input type="checkbox"/>	<input type="text" value="Disable"/>	<input type="text" value="4000"/>
2	FXO	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="5"/>	<input type="text" value="200"/>	<input type="checkbox"/>	<input type="text" value="Disable"/>	<input type="text" value="4000"/>
3	FXO	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="5"/>	<input type="text" value="200"/>	<input type="checkbox"/>	<input type="text" value="Disable"/>	<input type="text" value="4000"/>
4	FXO	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="5"/>	<input type="text" value="200"/>	<input type="checkbox"/>	<input type="text" value="Disable"/>	<input type="text" value="4000"/>

- **Listening Volume:** Adjusts the earpiece or speaker volume.
- **Speaking Volume:** Adjusts the microphone volume.
- **Tone Volume:** Adds a new option to make tone volume adjustable. This setting will be applied to all tones generated by the gateway including dial tone and busy tone.
- **Flash Time:** Used to set the time frame that FXO generates a FLASH signal.
- **Enable Polarity Reversal:** As a remote site answers calls from this extension the FXO port will reverse polarity.
- **PSTN Answer Detection:** This is used only by ITSP.
- **PSTN Ring OFF Length:** FXO/PSTN make out if the call from PSTN hangs up before FXS answer the call.

Codec Settings

Codec Settings					
Preferred Codec Type	G.723.1 6.3kbps				
Jitter Buffer [60 - 1200 ms]	120				
Silence Detection / Suppression	<input type="checkbox"/>				
Echo Cancellation	<input checked="" type="checkbox"/>				
Codec	<input checked="" type="checkbox"/> G.711 u-law	<input checked="" type="checkbox"/> G.723.1	<input checked="" type="checkbox"/> G.726	<input checked="" type="checkbox"/> G.729A	<input checked="" type="checkbox"/> G.711 a-law
Packet Interval (ms)	20	30	20	20	20
Approximate Bandwidth Required (kbps)	85.6	20.8	53.6	29.6	85.6

- **Preferred Codec Type:** Since different voice codecs have different compression ratios, the sound quality and occupied bandwidths are also different. It is recommended that you use the default provided (G.723.1) because it occupies less bandwidth and will provide better sound quality.
- **Jitter Buffer:** Adjusts the jitter to receive a packet. If the jitter range is too wide, it will delay voice transmission.
- **Silence Suppression:** If one side of a connection is not speaking, the system will stop sending voice data (package) to decrease bandwidth usage.
- **Echo Canceling:** Prevents poor telecommunication quality caused by echo interference.
- **Codec:** Choose the codec that you needs.
- **Packet Time:** Defines how long the gateway sends a RTP packet or voice packet to the receiving side. The smaller the value, the greater the bandwidth usage, but larger values increase voice delay.
- **Approximate Bandwidth Require:** The bandwidth required varies with codec format and packet time.

Fax Settings

T.38	<input checked="" type="checkbox"/> Enable High Quality
T.30	FAX Codec: G.711 u-law 64kbps
T.30	FAX Jitter Buffer [60 - 1200 ms]: 200


- ◆ **NOTE:** When a fax tone is detected in a call, the gateway will automatically switch from voice mode to fax mode. So fax settings will be temporarily applied to a specific port which detects fax tones, instead of its default voice settings.
- **Enable High Quality:** The system sends the same fax frame twice to get a high quality fax transmission. Enabling this variable increases bandwidth requirements.
- **T.30:** The system uses T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the T.30 protocol will consume more network resources and will affect transmission quality.
- ◆ **NOTE:** When you send fax over an IP network it needs your network to support fax over IP functionality (either T.38 or T.30). Please consult your VoIP service provider for this setting.

Drop Inactive Call

Drop Inactive Call	
Silence Detection Threshold [0=disable, 1 - 60 dB]	<input type="text" value="0"/>
Drop Silent Call Timeout [0=disable, 1 - 3600 s]	<input type="text" value="120"/>

This is used as a standard to determine whether or not to hang up the phone. The system will hang up the phone automatically to avoid keeping the line engaged if the detected volume is below the Silence Detection Threshold and the time exceeds the Drop Silent Call Timeout.

- **Silence Detection Threshold:** The volume below the threshold is used as a standard to determine whether or not to hang up the phone.
- **Drop Silent Call Timeout:** If the detected volume is below the threshold and the time exceeds the silence detection interval, the system will hang up the phone automatically to avoid keeping the line engaged.

 **NOTE:** Please be careful with these settings. Improper values might cause unexpected automatic disconnection of a call. Default values are recommended.

Voice Menu Options

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

- **Voice Menu Options:** Tick the check box to enable or disable IVR function.

 **NOTE:** When disabled, call pickup, Automatic Redial and unattend transfer will be disabled.

3-10 Digit Map

Digit Map now is combined the original feature of Digit Map and Speed Dial. You can use “?” or “%” in the column of Scan Code, VoIP Dial-out and PSTN Dial-out. “?” is a single digit, and “%” is wildcard. It provides a mapping between the number received from user and the replaced or modified number for real dial out. With this function, user can easily add certain leading digits to replace full number.

Alert if Auto fails	<input type="checkbox"/>
Enable Pound Key '#' Function	<input checked="" type="checkbox"/>
Default Call Route	Auto (VoIP first) ▼

- **Alert if Auto fails:** Tick the check box to play a voice announcement before calling out. It reminds user that this call is through PSTN.
- **Enable Pound Key '#' Function:** It is to speed up the connection of a call by entering '#' after a complete phone number is dialed.
- **Default Call Route:** The default call route can be Auto, VoIP or Deny.
Auto (VoIP first): The call route is VoIP first, and the next is PSTN.
VoIP: The call route is VoIP only.
PSTN: The call route is PSTN only.
Deny: The call will be denied.

Digit Map Testing

Digit Map Testing	
Test Dial No.	<input type="text"/> <input type="button" value="Run"/>
Result	<input type="text"/>

- **Test Dial No.:** You have to set some rules in Digit Map Setting first and enter the number for test.
- **Result:** The gateway will show the number for VoIP Dial-out and PSTN Dial-out according to the Digit Map Setting as below.

Digit Map Rule

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	10	Auto (VoIP first) ▼
2	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	10	Auto (VoIP first) ▼
3	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	10	Auto (VoIP first) ▼
4	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	10	Auto (VoIP first) ▼
5	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	10	Auto (VoIP first) ▼

- **Enable:** Enable detection of this entry.
- **Scan Code:** Defines the digits for the gateway to scan while user is dialing.
- **VoIP Dial-out:** Defines the dialed number rule for the gateway to call through Internet.
- **PSTN Dial-out:** Defines the dialed number rule for the gateway to call through PSTN.

- **User Dial Length:** Defines total number of digits that user dialed. A setting of zero tells the gateway scans digits only and disregards the total digit count.
- **Route:** Determine the interface calls should go through if above conditions satisfied.

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 or PSTN Dial-out column.

For example,

Scan Code: 55

VoIP Dial-out: 07021234567

User Dial Length: 2

Route: VoIP

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	<input checked="" type="checkbox"/>	55	07021234567		2	VoIP
2	<input type="checkbox"/>				10	Auto (VoIP first)
3	<input type="checkbox"/>				10	Auto (VoIP first)

As an incoming call from PSTN, the user dials 55 and the system will dial 07021234567. You also can use Digit Map Testing to know that the system will dial 07021234567 and go through Internet.

Digit Map Testing	
Test Dial No.	55 <input type="button" value="Run"/>
Result	#1, Seq: VoIP=07021234567

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

For example,

Scan Code: 2???

PSTN Dial-out: 35106???

User Dial Length: 4

Route: PSTN

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	<input checked="" type="checkbox"/>	55	07021234567		2	VoIP
2	<input checked="" type="checkbox"/>	2???		35106???	4	PSTN
3	<input type="checkbox"/>				10	Auto (VoIP first)

As an incoming call from PSTN, the user dials 2301. The system will dial 351006301 and go through PSTN/FXO. You also can use Digit Map Testing to know that the system will dial 351006301 and go through PSTN/FXO.

Digit Map Testing	
Test Dial No.	2301 <input type="button" value="Run"/>
Result	#2, Seq: PSTN=35106301

For example,
 Scan Code: 0%
 VoIP Dial-out: 0%
 PSTN Dial-out: 1805%
 User Dial Length: 0
 Route: Auto

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	<input checked="" type="checkbox"/>	55	07021234567		2	VoIP
2	<input checked="" type="checkbox"/>	2???		35106???	4	PSTN
3	<input checked="" type="checkbox"/>	0%	0%	1805%	0	Auto (VoIP first)

As an incoming call from PSTN, the user dials 0423456789. The system will dial 0423456789 and go through Internet first. If the call is fail to Internet, the system will dial 1805423456789 and go through PSTN/FXO. You also can use Digit Map Testing to know that the system will dial 0423456789 to Internet and 1805423456789 to PSTN/FXO.

Digit Map Testing	
Test Dial No.	0423456789 <input type="button" value="Run"/>
Result	#3, Seq: VoIP=0423456789 PSTN=1805423456789

Method 3- Substitution: It helps you dial to destination that you can not dial by phone. Destination like: test@1.1.1.1. Fill 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: Auto

#	Enable	Scan Code	VoIP Dial-out	PSTN Dial-out	User Dial Length [0=disable, 1 - 25]	Route
1	<input checked="" type="checkbox"/>	11	test		2	Auto (VoIP first) ▼
2	<input type="checkbox"/>				10	Auto (VoIP first) ▼
3	<input type="checkbox"/>				10	Auto (VoIP first) ▼

As an incoming call from PSTN, the user dials 11. the system will dial "test" and go through Internet. You also can use Digit Map Testing to know the dialing result.

Digit Map Testing	
Test Dial No.	11 <input type="button" value="Run"/>
Result	#1, Seq: VoIP=test PSTN=11

! **Note:** In the example of Method 3, the result also shows that the system will dial 11 and go through PSTN. That means the system will dial 11 to PSTN if the call is fail to Internet. Please select the route is VoIP in this rule if the route is only able to Internet.

3-11 Phone Book

This system can set up and store 100 phone numbers into a phone book and provides an IP address query when calling to other gateway(s). If no Phone Book manager is set within a gateway group, then all gateway systems have to set up phone data to allow each gateway to communicate with the others.

#	Gateway Name	Gateway Number	IP / Domain Name	Port
1				5060
2				5060
3				5060
4				5060
5				5060

- **Gateway Name:** Enter another gateway's code or an easy-to-remember name.
- **Gateway Number:** Enter the desired number of another gateway.
- **IP / Domain Name:** Enter the IP address or URL (Uniform Resource Locator) of another gateway.
- **Port:** Enter another gateway's listen port setting.

3-12 Caller Filter

This function is used to allow or deny SIP invitations from the proxy list ONLY.

<input checked="" type="radio"/> Allow <input type="radio"/> Deny		
Enable	Filter IP address	Subnet mask
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text"/>

- **Filter IP Address:** Enter the start IP you would like to allow or deny.
- **Subnet mask:** Enter the subnet mask you would like to allow or deny.

3-13 CDR Settings

The user can set up a CDR Server to record call details for every phone call. CDR provides call detail recording in a text file and which can be imported to prepare an analysis report.

CDR Settings	
<input type="checkbox"/> Send record to CDR Server	
CDR Server IP / Domain	<input type="text"/>
Port [1 - 65535]	<input type="text" value="1812"/>
Support RADIUS	<input type="checkbox"/>
RADIUS Accounting Port [1 - 65535]	<input type="text" value="1813"/>
RADIUS Server Secret	<input type="password" value="....."/>
RADIUS User ID	<input type="text"/>
RADIUS Password	<input type="password" value="....."/>

- **Send record to CDR Server:** Enables call detail recording.
- **CDR Server IP:** Enter the IP address of the CDR server.
- **Port:** Enter the listen port of the CDR server.
- **Support RADIUS:** Enable RADIUS for CDR database.
- **RADIUS Server Secret:** Enter the secret.
- **RADIUS User ID/Password:** Enter the User ID and password.

3-14 Language

The system provides English, Traditional Chinese, and Simplified Chinese for displaying text on web pages. Changing the language setting also changes the language for IVR (Interactive Voice Response).

Web UI / IVR Language	English	▼
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3-15 Transit Call Control

If you wish to restrict a general user (one who is not required to enter the PIN code) to local calls only and prohibit him/her from making long-distance calls started with a prefix "0", do the following steps:

1. Enable the Outbound Call Control function,
2. Set the PIN code for Outbound Level 5 to blank,
3. Set the Long-Distance Control Table to correspond with the Outbound Level 5 to prohibit making any call with the prefix "0" (as shown above).



Note: Transit Call Control is effective when it cooperate with Long-Distance Control Table.

Inbound Call Control	<input type="checkbox"/>
Outbound Call Control	<input type="checkbox"/>

- **Inbound Call Control:** To determine when users make a phone call from a PSTN to Gateway FXO whether or not they check the inbound PIN code while using a VoIP — only effective for incoming calls calling from a PSTN trunk.
- **Outbound Call Control:** To determine when users utilize Gateway FXO interface to divert to a PSTN whether or not they check the outbound PIN code — only effective for outgoing calls being diverted to a PSTN Trunk.

#	PIN Code	Enable	Privileges
1	<input type="text"/>	<input type="checkbox"/>	0 ▼
2	<input type="text"/>	<input type="checkbox"/>	0 ▼
3	<input type="text"/>	<input type="checkbox"/>	0 ▼

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

3-16 Long-Distance Control Table

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

#	0	<u>1</u>	<u>2</u>	<u>3</u>	<u>4</u>	<u>5</u>
1	<input type="text" value="0204"/>					
2	<input type="text"/>					
3	<input type="text"/>					

Descriptions:

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

Principle: Downward Restriction —If users at a higher level cannot dial a number with a certain prefix, then users at lower levels also cannot dial a number with the same prefix.

3-17 Long Distance Exception Table

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

#	0	<u>1</u>	<u>2</u>	<u>3</u>	<u>4</u>	<u>5</u>
1	020488988					
2						
3						

Instruction:

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

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

3-18 CPT/Cadence Settings

The CPT has 3 sets of parameter tables. Please adjust the CPT based on the local PSTN or PBX settings and requirements.

# 1 Enable		Setting 1					Default
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	
Dial Tone	350	440	3000	0	0	0	
Congestion Tone	480	620	250	250	0	0	
Busy Tone	480	620	500	500	0	0	
Ring-Back Tone	440	480	1000	2000	0	0	

Busy Tone Cadence Measurement

Enable Busy Tone Cadence Measurement		T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning
BTC # 1		0	0	0	0	Yes
BTC # 2		0	0	0	0	Yes
BTC # 3		0	0	0	0	Yes
BTC # 4		0	0	0	0	Yes
BTC # 5		0	0	0	0	Yes

BTC Detection Sensitivity (less) 1 2 3 4 5 (more)

- **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.

The CPT has 2 sets of parameter tables. Please adjust the CPT based on local PSTN or PBX.

# 1 Enable		Setting 1					Default
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	
Dial Tone	350	440	3000	0	0	0	
Congestion Tone	480	620	250	250	0	0	
Busy Tone	480	620	500	500	0	0	
Ring-Back Tone	440	480	1000	2000	0	0	

UDT Detection

If the CPT auto detect function is not able to determine whether or not a PSTN-call receiving party has hung up the phone, then the UDT detection function can serve as a back up. To do this, enter the high/low frequency parameters from the CPT table into the UDP table.

Disable UDT	UDT Setting	# 1 Disable	# 2 Disable	# 3 Disable	# 4 Disable
Congestion Tone / Busy Tone	Low Frequency	0	0	0	0
	High Frequency	0	0	0	0

 **NOTE: If FXO cut off calls un-normally, please disable this function.**

3-19 Current Status

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

Server Registration Status: It shows the registration status of DDNS, Phone Book Manager, STUN and UPnP.

DVG-3016S							
Port Status							
No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXO	701	Idle	0		Disabled	
2	FXO	702	Idle	0		Disabled	
3	FXO	703	Idle	0		Disabled	
16	FXO	716	Idle	0		Disabled	
SIP Proxy Hunting Number Registration					Trunk Disabled (02:14:19)		
Server Registration Status							
DDNS Registration					Disabled (02:14:19)		
Phone Book Manager Registration					Disabled (02:14:19)		
STUN Registration					Disabled (02:14:19)		
UPnP Negotiation					Disabled (02:14:19)		

3-20 RTP Packet Summary

Display the information of the last completed call. This report contains peer IP, peer port, packets sent, packet received, and packet loss information. Press the **Refresh** button to get the latest RTP Packet Summary.

RTP Packet Summary				
Line 1	G.723.1 6.3kbps	Packet Sent 61	Packet Received 43	Packet Lost N/A
	The last packet's source IP 192.168.100.199		The last packet's source Port 9000	
Line 2	G.729A 8kbps	Packet Sent 6	Packet Received 155	Packet Lost N/A
	The last packet's source IP 192.168.100.181		The last packet's source Port 9008	
Line 3	G.729A 8kbps	Packet Sent 12	Packet Received 168	Packet Lost N/A
	The last packet's source IP 192.168.100.181		The last packet's source Port 9004	
Line 4	G.729A 8kbps	Packet Sent 5	Packet Received 153	Packet Lost N/A
	The last packet's source IP 192.168.100.181		The last packet's source Port 9006	

3-21 System Information

These two pages show the status of the VoIP gateway. The categories are WAN Port Information, LAN Port Information and Model Information.

WAN Port Information	
Factory Default MAC Address	000C2A05CA3C
Net Link	Disconnected
IP Address	192.168.100.200
Subnet Mask	255.255.255.0
Default Gateway	192.168.100.254
DNS	168.95.1.1
LAN Port Information	
MAC Address	000C2A05CA3D
IP Address	192.168.8.254
Subnet Mask	255.255.255.0
DHCP Server	
DHCP Server	Disabled
Model Information	
Hardware Platform	PNP1632-32
Hardware	3.3.0
Software Version	1.02.37.61-b8259
Driver Version	0.9.5.1.735 29/Nov/2007 10:30:08
DSP	482-128K

- **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.
- **LAN Port Information:** It shows LAN port IP, subnet mask, and the status of DHCP server.
- **Model Information:** It shows the hardware platform, software and hardware version.

3-22 Routing Table

The Routing Table stores the information for particular network destination around the VoIP gateway. Press the **Refresh** button to generate the details.

Destination	Netmask	Gateway	Interface
192.168.100.0	255.255.255.0	0.0.0.0	WAN
192.168.8.0	255.255.255.0	0.0.0.0	LAN
default	0.0.0.0	192.168.100.254	WAN

3-23 STUN Inquiry

Use "STUN Inquiry" to detect your IP sharing device's NAT type and communication between a STUN server and client (built-in to the DVG-2016S/2032S gateway).

NAT Type	Unknown
STUN Server IP / Domain	<input type="text"/>
STUN Server Port [1 - 65535]	<input type="text" value="3478"/>

3-24 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.

Ping Destination	<input type="text"/>
Number of Ping [1 - 100]	<input type="text" value="4"/>
Ping Packet Size [56 - 5600 bytes]	<input type="text" value="56"/>
<input type="button" value="Test"/> <input type="button" value="Stop"/>	

3-25 NTP (Network Time Protocol)

	Year	Month	Day	Hour	Minute	Second
Gateway Time	2000	1	1	8	1	5
Time Zone	+ 8 :00					
#	Time Server					
1	ntp1.dlink.com					
2	ntp.dlink.com.tw					
3						

- **Time Zone:** Set the Time Zone where the gateway resides.
- **Time Server #1~#3:** Set the Time Server where the gateway should sync up during start up.

3-26 SNMP

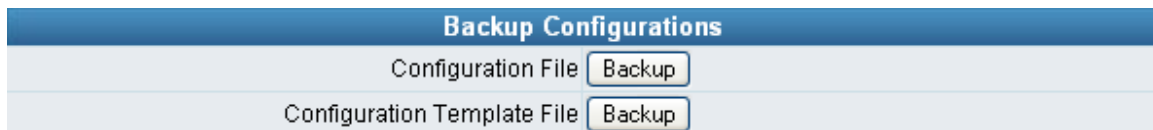
Enable SNMP Agent	<input type="checkbox"/>
Get Community	public
Set Community	private
Trap Community	public
Trap Host	

- **Enable SNMP Agent:** Enable SNMP if selected.
- **Get/Set/Trap Community:** Enter Community name to Read, Write and Trap.
- **Trap Host:** Enter the IP of the Trap Host.

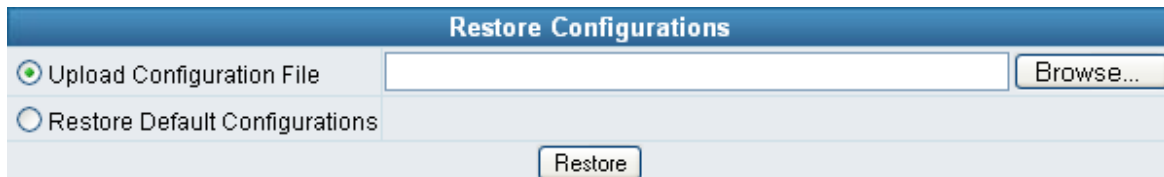
3-27 Backup/Restore

You can backup settings to a file and restore settings from that file. You also can restore all settings back to default by selecting **Restore Default Configurations** and click **Restore**.

◆ **NOTE:** the gateway needs for you to Save Settings and Restart so that all settings will be backed up.



- **Configuration File:** Backup all settings.
- **Configuration Template File:** Backup the settings as a template file for editing.



- **Upload Configuration File:** Upload the configuration file from somewhere to the device.
- **Restore Default Configurations:** Reset the device back to the factory default settings.

3-28 System Log


It will send all logs to the server. Please contact with your provider.

Enable	<input type="checkbox"/>
Server Address	<input type="text"/>
Port [1 - 65535]	<input type="text" value="514"/>

3-29 Provision Settings

Options in this section are only required for VoIP networks in which a provisioning system has been implemented. Fill in the parameters needed by the Provision Server from your service provider.

Enable Auto Provisioning	<input type="checkbox"/>
Provision Server Address	<input type="text"/>
Port [1 - 65535]	<input type="text" value="10101"/>
Packet Format	Proprietary <input type="button" value="v"/>
Connect Provision Server During Start Up	<input checked="" type="checkbox"/>
Connect Provision Server Periodically	<input checked="" type="checkbox"/>
Auto Provision Interval [60 - 604800 s]	<input type="text" value="10800"/> Random Offset [1 - 1800 s] <input type="text" value="600"/>
Provision Retry Times [0=always, 1 - 99] [0 - 99]	<input type="text" value="10"/>
Retry Interval [30 - 120 s]	<input type="text" value="30"/>
Suspend Service	<input type="checkbox"/>

 **NOTE:** The availability of the above features also depends on your VoIP network. Please check with your service provider about the availability of these services.

3-30 System Operations (Save Settings)

Some settings are effective only after **Restart**. Remember to save all settings using **Save Settings** before you restart.

<input type="checkbox"/> Save Settings	Save all configurations.
Be sure to save all settings before restart.	
<input type="checkbox"/> Restart	Restart the Gateway right away. All calls will be DROPPED when Restart.

- **Save Settings:** Save settings after completing changes. The new settings will take effect after the system is restarted. Please select "Save Settings".
- **Restart:** If it is necessary to restart the system, please select "Restart" and click the "Accept" button.

3-31 Software Upgrade

The gateway provides a software upgrade function from a remote source. Please consult your service provider for information about the following details.

Software Upgrade

To Save Current Settings, [Save Settings](#)

Current Software Version No. [1.02.36.7]

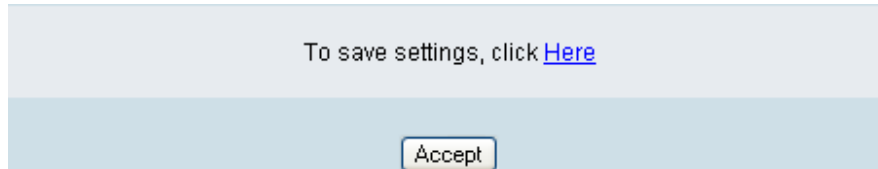
Upgrade Server	<input checked="" type="radio"/> TFTP <input type="radio"/> FTP <input type="radio"/> HTTP
Server IP Address	<input style="width: 100%;" type="text"/>
Server Port [1 - 65535]	<input style="width: 50%;" type="text" value="69"/>
User Name	<input style="width: 80%;" type="text"/>
Password	<input style="width: 80%;" type="password"/>
Directory	<input style="width: 100%;" type="text"/>

All calls will be DROPPED during upgrade.

- **Upgrade Server:** Choose the server type given by your provider.
- **Software Upgrade Server IP:** Enter the software server IP address.
- **Software Upgrade Server Port:** Enter the port that the server uses. TFTP is 69, and FTP is 21.
- **User Name/ Password:** The account information to access an FTP server.
- **Directory:** The directory path of the upgrade files for TFTP or FTP.

3-32 Logout

The gateway only allows one user at a time to log-in, so whenever a change is made, please save the settings and restart the system, or logout to avoid a situation where other users cannot log-in to change settings.



4. Configuring the Gateway through IVR

VoIP transmits voice data (packets) via the Internet. One effect of this is that telecommunications quality is closely related to the condition and status of the network environment. If any of the parties involved in VoIP communications has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, excellent telecommunication can only happen when the gateway is connected to the Internet and when the network environment is stable.

Preparation

- Install the gateway according to instructions. Connect the power supply, telephone set, telephone cable, and network cable properly as described in Chapter 2.
- If a static IP is used, 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 questions.
- If you are using ADSL (PPPoE) for your network connection, confirm the account number and password.
- If you intend to operate the gateway under a NAT, the Gateway WAN Port IP Address and LAN Port should not use the same range in order to avoid phone failures.

Basic Setup

The gateway provides two setup modes:

1. Telephone IVR Configuration Mode
2. Browser Configuration Mode

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

4-1 IVR (Interactive Voice Response)

The gateway provides convenient IVR functions. Users only need to pick up a handset and enter the function code for query and setup without using a PC.



NOTE: After 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 “admin”. Enter “***4144534954#” as above. You are now in IVR setting mode. (Please refer to page 45 for function codes).
- Once the first setting or query has been completed, you will hear the tone indication again. Then use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

Example: If your password is “1234”, enter “***1234#” so that you are now in IVR setup mode. Next enter a function code to check or configure the gateway. If your password is “admin”, enter “***4144534954#”. Please refer to the IVR Functions Table (page 45) for available functions and codes.

Example: enter “***#” (you are now in IVR mode) → enter 101 (to query about 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

After completing all of your settings, dial 509 (Save Settings). Wait for about three seconds; you should hear a confirmation tone "1." You can now hang up the phone. Please reboot the gateway to enable the new settings.

To inquire about the current gateway WAN Port IP address setting


After completing all your settings, dial 101. The system will repeat the current WAN Port IP address. If the system does not repeat the IP address, this indicates that the gateway is not currently connected to the Internet. Please check to be certain that the cable connection, account number, and password are all correct.

IVR Functions Table:

Function Code	Description	Example / Notes
111/101	WAN Port IP address Set/Query	First, use function code 114 to select 1 for a Static IP connection then setup the IP address.
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2: DHCP, 3: PPPoE)	
115/105	DNS IP address Set/Query	
116/106	Phone Book manager IP address Set/Query	
117/107	Set/Query whether or not to use a Public Telephone Book (0: Disable 1: Enable)	
199/099	Set/Query whether or not this gateway acts as the Phone Book manager (0: Disable 1: Enable)	
066	Querying the connection to Phone Book manager	
118	Restart	
121	Setup PPPoE Account	Use function code 114 to select 3 for a PPPoE connection.
122	Set PPPoE Password	
123	Set NAT IP address	
124	Uses NAT (0: Disable 1: Enable)	
311/301	LAN Port IP Set/Query	
312/302	LAN Port Subnet Mask Set/Query	
131/132	Play/Record greeting message	
133	Save greeting message	
109	Restore factory default IP address configuration	A static IP address for WAN Port IP : 192.168.1.2 Mask : 255.255.255.0 Gateway : 192.168.1.254
409	Restore factory default settings	
509	Save settings	
900	Set the IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Software Upgrade	

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 questions.

Function	Command
Select a Static IP	<ul style="list-style-type: none"> After entering IVR mode, dial 114. After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	<ul style="list-style-type: none"> After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address followed by "#". <p>Example: If the IP address is 192.168.1.200, dial 192*168*1*200#.</p>
Subnet Mask Settings	<ul style="list-style-type: none"> After entering IVR mode, dial 112. After hearing "Enter value", enter your subnet mask followed by "#". <p>Example: 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"> After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address followed by "#". <p>Example: 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"> To save settings, dial <u>509</u> (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 <u>101</u> to check whether or not the IP address was retained. If the IP address is not repeated, this indicates that the gateway is not properly connected. Please check to be certain that the cable connection, account, and password are all correct.

Dynamic IP (DHCP) Settings

After entering IVR mode, dial 114.

After hearing "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 repeat the IP address, this indicates that the gateway failed to communicate with a DHCP server. Please check your DHCP server or ISP.

ADSL PPPoE Settings

 **NOTE:** Complete PPPoE settings should include: Select PPPoE (option 3 of 114), PPPoE account (121) and PPPoE password (122).

Please contact your local Internet Service Provider (ISP) if you have any questions.

Select a PPPoE

- After entering IVR mode, dial 114.
- After hearing "Enter value," dial 3 (to select PPPoE).

PPPoE Account Settings

- After entering IVR mode, dial 121.
- After hearing "Enter value", enter the account number followed by "#".
Example: If the account is "84943122@hinet.net," please enter 08 04 09 04 03 01 02 02 71 48 49 54 45 60 72 54 45 60#.



NOTE: it is necessary to enter two digits for each character/number; for example, you must enter "01" for "1" and "11" for "A". It is recommended that you use the web Interface to configure your PPPoE account details. Refer to the PPPoE Character Conversion Table on the next page for key mappings if you choose to use IVR setup.

PPPoE Password Setting

- After entering IVR mode, dial 122.
- After hearing "Enter value," enter the new password followed by "#".

Example: If the password is "3ttixike", please enter "03 60 60 49 64 49 51 45#".

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 or not the IP address was retained. If the IP address is not repeated, this indicates that the gateway is not properly connected. Please check to be certain that the cable connection, account, and password are all correct.

PPPoE Character Conversion Table:

The table below provides a list of PPPoE conversion codes. The first column in each pair of columns lists the number, letter or symbol that you want to enter. The second column in each pair ("Input Key") tells you what code to enter for the corresponding number, letter or symbol. For example, to enter "D-Link" using the codes below, enter: 148322495451

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

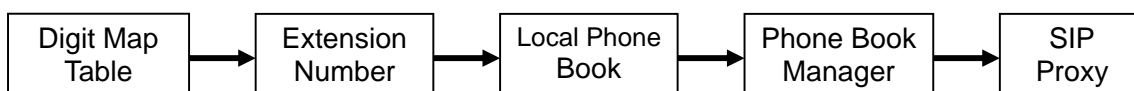
5. Dialing Principles

5-1 Dialing Options

1. Dial the phone number which you want to call and press # to call out immediately, or wait until the "Inter Digits Timeout" expires (defined in "Advanced Options", default=4 seconds, see page 25).
2. If the phone number fits the setting of the Digit Map, the gateway dials out the phone number through the assigned interface automatically.
3. The phone number should have at least 2 digits (not including * and #).

5-2 Dialed Number Processing Flow

To maintain maximum flexibility, the number dialed will be looked up in several tables defined by the gateway. If no match is found, it will look up the number from the registered SIP Proxy Server. The number look up flow is shown below:

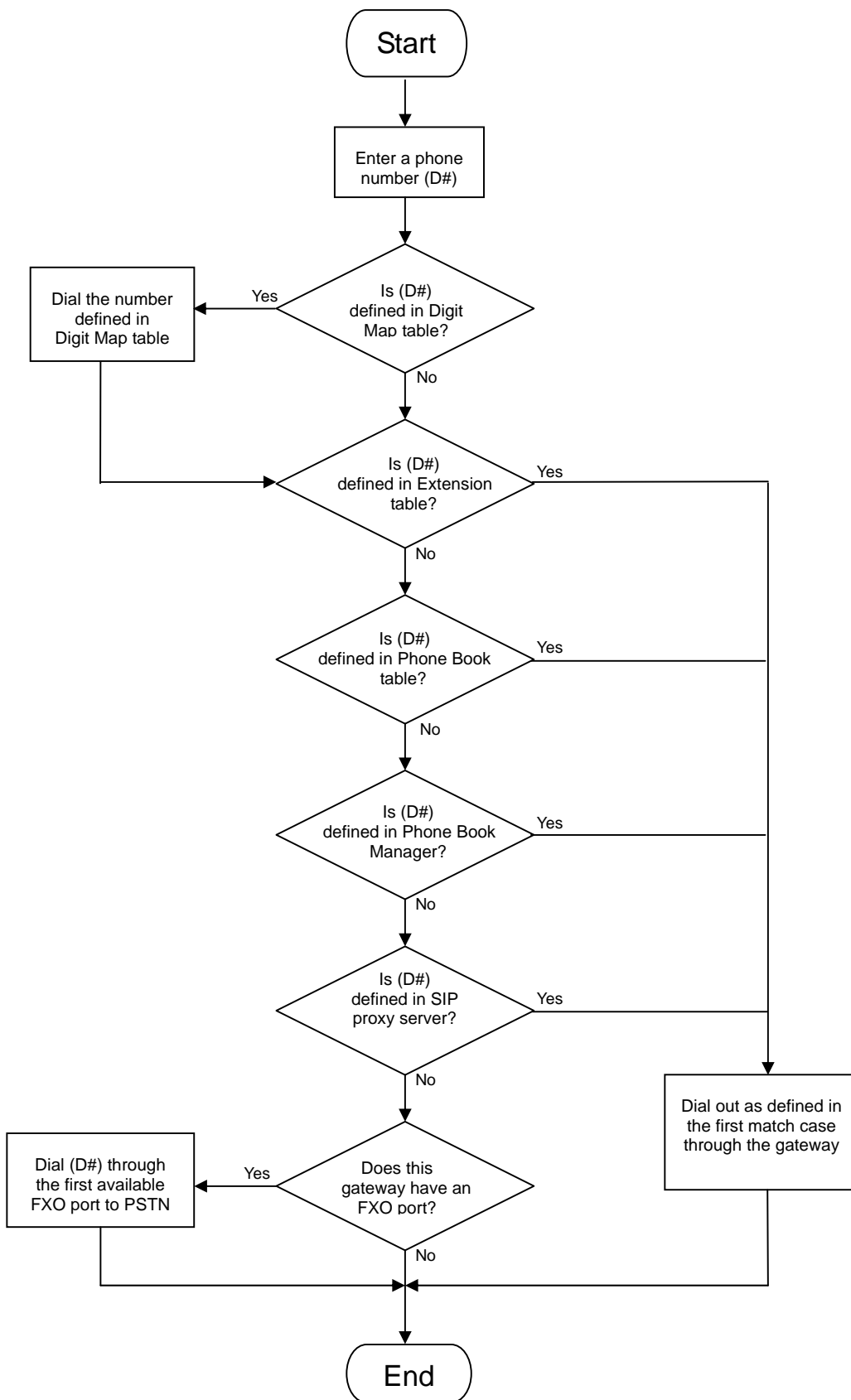


A complete flow chart is shown on the next page.

5-3 Example for Match phone numbers invited by callers

The table below is provided as a general reference expresses phone numbers dialed by the gateway instead of real phone numbers that callers dial.

Match	Scheme	Description
The same as "FXS Representative Number"	Ring FXS according to "FXS Group Hunting / Ring Priority" settings	
The same as "FXO Representative Number"	Off hook a FXO	It is not applied to registration with SIP Proxy
The same as "FXS Representative Number + Extension Number"	Ring or off hook the Extension	If Extension line is FXS, it should ring. If Extension line is FXO, it should off hook It is not applied to registration with SIP Proxy
The same as FXO Extension Number	Off hook the FXO	
A Prefix is the same as "FXS Representative Number +FXO Extension Number"	Eliminate a Prefix and use remaining digits to route calls via FXO	FXS Representative Number is 2252 and one of FXO Extension is 070123456 If callers dial 2252070123456 6371, the gateway dial 6371 via FXO Extension
A Prefix is the same as FXO Extension Number	Eliminate a Prefix and use remaining digits to route calls via FXO	One of FXO Extension is 070123456 If callers dial 070123456 6371, the gateway dial 6371 via FXO Extension
Differ from FXS/FXO numbers	Use these digits to route calls via FXO	If callers dial 6371, the gateway dial 6371 via one of FXO line



Appendix

WAN

- One 10/100Mbps auto-negotiation, auto-crossover RJ-45 Ethernet port
- Support static IP, PPPoE, BigPond Cable 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 : Port Forwarding, STUN, UPnP and Outbound Proxy
- NTP: (Network Time Protocol RFC 1305), Accepts up to 3 Time Server
- 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 ports
- DHCP server

Voice Features

- 16/32 combination of FXO (Foreign eXchange Office)
- ports: DVG-3016S(16FXO), DVG-3032S (32FXS).
- SIP (RFC3261) compatible
- Voice codecs : G.711 a /ulaw, 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)
- Polarity reversal generation (FXS)
- T.30 (G.111) / Real time T.38 / Secured T.38 FAX relay
- DTMF, FSK (Bellcore & ETSI) Caller ID detection.
- Support Caller ID Restriction (CLIR)
- Digit Map for dial plan
- 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
- Recordable greeting message
- Call features:
 - Call Forward - Unconditional, Busy
 - Three Way Calling (Media Server required)
- Analogue interface
 - Connector : RJ-11
 - Signaling protocol : Loop Start

Configuration & Maintenance

- Configuration methods:
 - Web
 - IVR
 - Telnet
- Status reports:
 - Port status
 - Registration status
 - Ping tests
 - STUN/UPnP status
 - Hardware / software information
- Firmware Upgrade through TFTP, FTP and HTTP
- Configuration Backup/Restore
- Reset button (with restore factory default function)
- Front Panel LED : voice ports, WAN, LAN, Run, Power, Alarm
- Optional Auto Provisioning Server (APS) for mass deployment