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DVG-4032S
VOIP Gateway

User Manual

Version 1.0

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Contents

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1. Introduction

Product Overview

The stand-alone VoIP Gateway carries both voice and facsimile over the IP network. It supports SIP industry standard call control protocol to be compatible with free registration services or VoIP service providers' systems. It works in two different modes: UA (User Agent) or Server. As a standard user agent, it is compatible to all well-known Soft Switches and SIP proxy servers. While running the optional server software, the gateway can be configured to establish a private VoIP network over the Internet without a 3rd party SIP Proxy Server.

The gateway can be seamlessly integrated to existing network by connecting to a phone set, PBX, key telephone system, fax machine or PSTN line. With only a broadband connection such as ADSL bridge/router, Cable Modem or leased line router, it allows you to gain access to voice and fax services over IP in order to reduce the cost of international and long distance calls.

With the support of DDNS, it makes the gateway reachable by its domain name where the ISP dynamically assigns the IP address. It helps users to host a web site or mail server in a PPPoE or DHCP network. By enabling the CDR function & setting up a simple server, administrators are allowed to log and view all call records such as call duration, time and date of calls and latency, etc.

The gateway can be assigned with fixed IP address or by DHCP, PPPoE. It adopts the G.711, G.726, G.729A or G.723.1 voice compression format to save the network bandwidth while providing real-time and toll quality voice.

Product Features

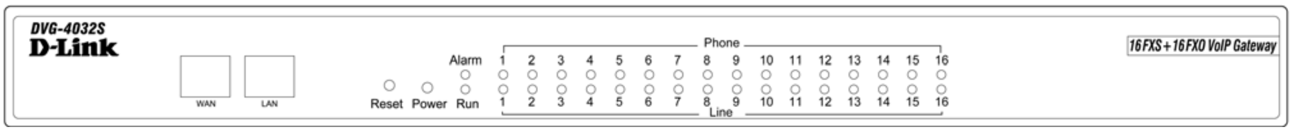
- ● SIP (RFC 3261) compliant
- ● Optional server enables small businesses to build up private VoIP network (SIP model)
- ● QoS support guarantees voice bandwidth in a busy network
- ● Supports IP TOS (Type Of Service)
- ● T.30 (G III) / Real Time T.38 / Secured T.38 Fax Relay
- ● Feasible for Fixed IP address or dynamic IP address network (PPPoE / DHCP client, support DDNS)
- ● Configurable Hot Line feature
- ● Supports IP-to-PSTN / PSTN-to-IP applications
- ● NAT traversal - STUN and UPnP (optional)
- ● Pass through NAT
- ● Call Detailed Records (CDR)
- ● Web-based firmware upgrade
- ● Caller ID Delivery
- ● Easy Configuration by IVR and Web-based GUI
- ● Greeting message
- ● Echo Cancellation: G.168 compliant
- ● Voice Activity Detection (VAD) & Comfort Noise Generation (CNG)
- ● Dialing: DTMF, PULSE (optional)
- ● Signaling Protocol: Loop Start
- ● Adaptive Jitter Buffer and Programmable Gain Control

CALL features (optional)

- ● Call hold
- ● Call waiting
- ● Call forward
  - Unconditional (follow me)
  - Busy forward
  - No answer forward
- ● Call transfer

Hardware Description

Front Panel

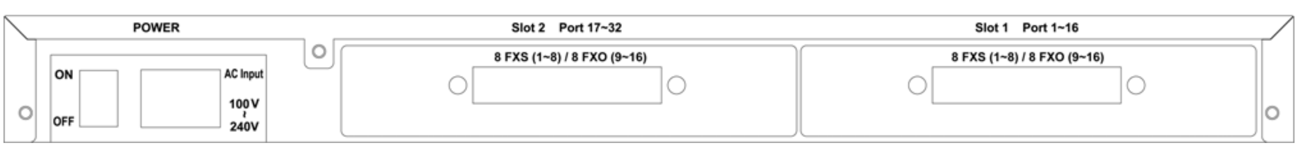


- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- Phone 1 – Phone 16 stands for Port 1 – Port 16: Connect to your analog telephone,
- Line 1 – Line 16 stands for Port 17 – Port 32: Connected to your original telephone line on the wall jack with RJ-11 cable.
- 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



NOTE: Do not connect Phone ports to each other. Also, do not connect any Phone ports directly to a PSTN line or internal PBX. If any of these are done, your DVG-4032S may be damaged.

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

(1) Pull off the power plug.
(2) Press reset (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.

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2. Installation and Applications

Network Interface

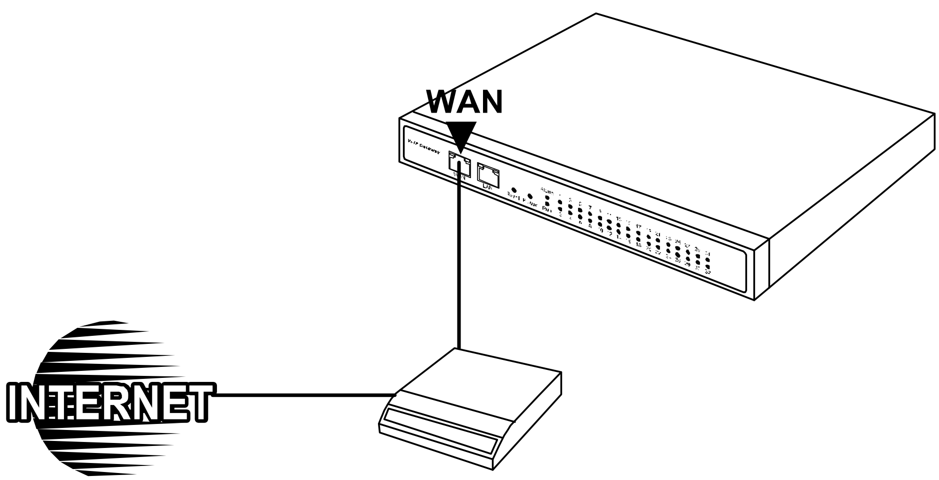
The network interface is divided into 3 basic modes as described below:

- Gateway can be assigned with a Public IP Address
- Gateway can be built under the existing NAT

Gateway Assigned with a Public IP Address

The gateway will have a Public IP address for Internet connection regardless of whether it is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dialup / ADSL).

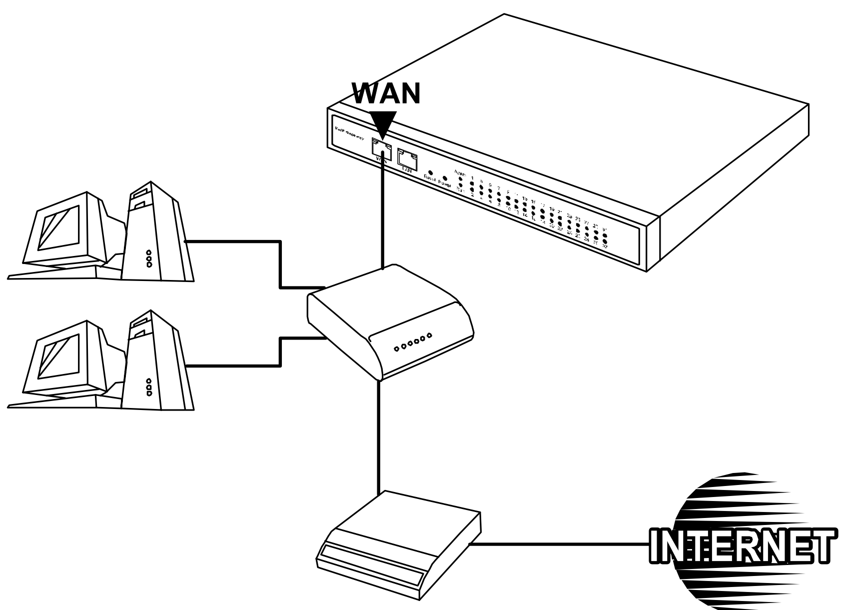
Gateway IP Settings	Need to set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	



Gateway in a NAT network

The gateway uses a virtual IP address and the IP sharing function of other systems to connect to the Internet.

LAN IP address of IP sharing	Please avoid IP address 192.168.0.1-192.168.8.254 (You may need to change the settings of IP sharing or change SIP series Gateway LAN Port IP address)	
Gateway IP Settings	Set as static IP address, and assign the LAN IP address of the IP sharing to the Default Gateway.	
NAT /STUN Settings	Enable	If the WAN of the IP sharing device has static IP address, then the NAT IP address is set as the Public IP address of the IP sharing.
		If the WAN of the IP sharing device uses a dynamic IP address, then it has to comply with the DDNS settings. When suing NAT, you must enter the URL (Uniform Resource Locator) that is registered to the DDNS server.
DDNS Settings	The WAN of the IP sharing device has a static IP address.	Disabled
	The WAN of the IP sharing device has a dynamic IP address.	Enabled: enter the registered URL (Uniform Resource Locator) into the network settings -> under NAT

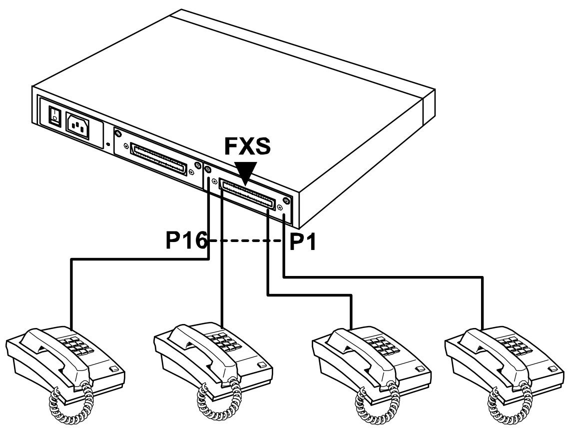


Telephone Interface Description

Example for Phone Ports:

DVG-4032S connecting directly to phone sets

After connecting telephone sets to P1-P8, users can make direct calls, (P1-P8 are FXS interfaces). Each set acts as an independent extension line.



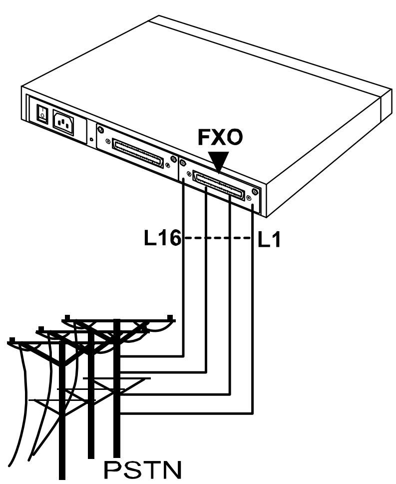
Integrating the DVG-4032S with PBX

P1-P16 is FXS interfaces, and some of them can be connected to telephone sets for direct calls. Others can be connected to the PBX so other extension lines can make VoIP calls.

Example for Line Ports:

DVG-4032S connecting directly to the Telephone Line of a PSTN

P9-P16 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.



3. Setting the Gateway through IVR

VoIP transmits voice data (packet) via the Internet to achieve telecommunications. This means that the telecommunication quality is closely related to the whole network environment. If any one of the telecommunicating parties has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, an excellent telecommunication can only be created when Gateway is connected to the Internet and when 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 desired 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 using dialup ADSL (PPPoE) for network connection, confirm the dialup account number and password.
- If users wish to build Gateway under the NAT, Gateway WAN Port IP address and LAN Port should not use the same range. This is to avoid phone failures.

Basic Settings of a Gateway

- IP Settings— Connecting Gateway to the Internet.

Gateway provides two setting modes:

1. Telephone IVR Setting Mode
2. Browser Setting Mode

The IVR provides basic query and setting functions, while the browser provides a full setting function.

IVR (Interactive Voice Response)

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

NOTE: After finishing the settings, make sure the new settings are saved. This is so that the new settings will take effect after the system is restarted.

Instructions

- FXS Port: Connected to telephones. To enter IVR mode, enter “ * * password #” after hearing the dial tone. When you hear a second dial tone, the system is in IVR mode, enter the function code. (Please refer to the Advanced Settings on page 35 for these codes)

- FXO Port: to use IVR functions, dial the phone number of FXO Port using an external line. You will hear the instruction “enter value”, and then enter a PIN number. The factory default code is blank. Enter “**#” as above. You are now in IVR setting mode.

Example: The factory default code is blank. Enter **“**#”**. You are now in IVR setting mode, enter the desired code. E.g.: if the code is 1234, then enter **“**1234#”**. Or If your password is **abc123** then you access IVR by pressing *****414243010203#**

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

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

Save Settings

After entering IVR mode, dial 509 (Save Settings). Wait for about 3 seconds and after hearing a confirmation tone “1”, hang up the phone. Please reboot the Gateway to enable the new settings.

To inquire about current Gateway’s WAN Port IP address

After entering IVR mode, dial 101. The system will repeat the current WAN Port IP address. If the system does not repeat the IP address, it indicates that the Gateway is not currently connected to the Internet. Please check if the cable connection, account number, and password are correct.

Recorded Voice File

- The gateway allows users to record their incoming call greeting messages, when calling via FXO.
- After entering IVR mode, dial 132. After hearing “Enter value”, record the incoming call greeting message. To end recording, simply hang up.
- After recording, to listen to the recorded message, press 131. Press 133 to save the message.

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IVR Functions Table:

Function Code	Description	Example
111/101	WAN Port IP address Set/Query	Use in conjunction with function code **114**, select 1 for a Static IP function.
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 books manager IP address Set/Query	Must use 116/106, 117/107 in conjunction with each other.
117/107	Set/Query whether or not to use Public Telephone Book (0: Disable 1:Enable)	
199/099	Set/Query whether or not this Gateway acts as the phone books manager (0: Disable 1: Enable)	
066	Querying the connection to Phone books manager	
118	Restart	
121	Setting PPPoE Account	Use in conjunction with function code **114,** select 3 for a PPPoE function
122	Setting PPPoE Password	
123	Setting NAT IP address	Must use 123 and 124 in conjunction with each other.
124	Uses NAT (0: Disable 1: Enable)	
151/141	Register to Proxy Server Set/Query (0: Disable 1: Enable)	
152/142	Proxy Server IP address Set/Query	
153/143	Proxy Server Port Set/Query	
125	Set Proxy Server account	
126	Set Proxy Server password	
154/144	Uses STUN Set/Query (0: Disable 1: Enable)	
155/145	STUN IP address Set/Query	
156/146	STUN Port Set/Query	
311/301		
312/302		
131/132		
133	Saving greeting message	
211/201	Set/Query International Prefix code	Prefix dialed before making an international call e.g. 002 and 005.
212/202	Set/Query Country Code	Setting country code, e.g. 886
213/203	Set/Query Area Prefix Code (Long-Distance Prefix Code)	Prefix dialed before making a long-distance call e.g. 0.
214/204	Set/Query Area Code	eg. “2” for Node B area.

Function Code	Description	Example
215/205	Set/Query Gateway Telephone Number (Representative Number)	
216/206	Set/Query the extension number of Line 1.	
217/207		
109	Restoring 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	Restoring factory default settings	
509	Save settings	
900	Setting IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Soft Upgrade	

IP Configuration Settings—Setting IP Configuration of WAN Port

Static IP Settings

NOTE: Complete static IP settings should include a static IP (Option 1 under114), IP address (111), Subnet Mask (112), and Default Gateway (113). Please contact your local Internet Service Provider (ISP) if you have any questions.

Function	Command
Select a Static IP	• After entering IVR mode, dial 114. • After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	• After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address, followed by "#". Example: If the IP address is 192.168.1.200, dial 192*168*1*200#.
Subnet Mask Settings	• After entering IVR mode, dial 112. After hearing "Enter value", enter your subnet mask, followed by "#". Example: If the mask value is 255.255.255.0, dial 255*255*255*0#.
Default Gateway Settings	• After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address, followed by "#". Example: If the Default Gateway is 192.168.1.254, dial 192*168*1*254#.
Save Settings and Restart	• To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are 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 if the IP address is retained.
If the system does not repeat the IP address, it indicates Gateway has not been properly connected to the Internet. Please check the cable connection.

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 #.

Please note that it is necessary to enter two digits for each character/number; for example, enter “01” for “1” and “11” for “A”.

PPPoE Password Setting

- After entering IVR mode, dial 122 after hearing “enter value” 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, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are correct.

PPPoE Character Conversion Table

Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	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

4. Setting a Gateway with WEB Browser

The gateway allows users to make settings using a web browser. After opening a browser, enter Gateway's IP address as the website address in order to enter the Web configuration screen as shown in the following diagram.
The factory default WAN IP address for Gateway is **192. 168. 1. 2**.You can also enter "101" from the handset to inquire about the current WAN Port IP address. The factory default LAN Port IP address is 192.168.8.254.

Instructions

- Open an Internet browser.
- Enter gateway's WAN Port IP address in the website address area (If the PC is connected to the LAN Port, enter the LAN Port IP address. The default is 192.168.8.254)
- The following registration screen will appear (The factory default settings for **Login ID and Password are set to blank**).
- After completing and confirming the settings, some of the settings will take effect immediately. But network related settings would take effect after the gateway is restarted. Please go to **System Operation** to save the settings before restarting the system.

To avoid several people simultaneously configuring the web and causing problems to users, please enter the correct Login ID and password. If a user logs into the system, other users from different IP addresses cannot login at the same time. Please remember to logout or restart the system if not using the web configuration function.

Network Settings

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

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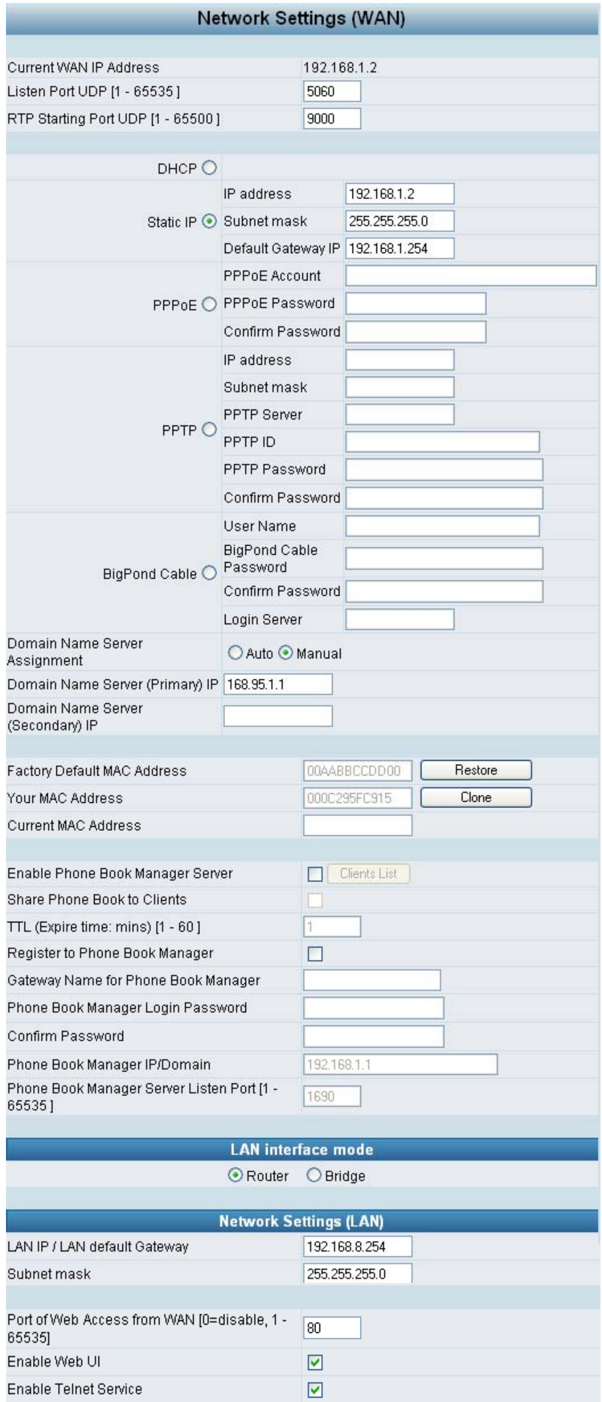


- Current WAN IP Address: The IP address of WAN port.
- Listen Port UDP: It is not necessary to change the protocol of the communication port used by DVG-3016S.
- RTP Starting Port UDP: The initial value of port number for transmitting voice data among Gateway(s). Each line requires 2 ports. It is not necessary to change these.

For example, If the starting port is 9000, then Line 1 is using 9000 and 9001, and Line 2 is using 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

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

1. Static IP
2. DHCP, means a Dynamic IP (Cable Modem)
3. PPPoE (Dialup ADSL)
4. PPTP.

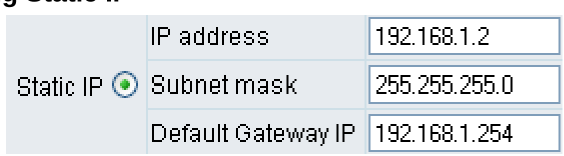
Using the DHCP and PPPoE for obtaining an IP address may vary. If not familiar with the network connection, please contact your local ISP.

Setting Dynamic IP (DHCP)

Click “DHCP” to obtain a Dynamic IP address, then click the “Accept” button at the bottom of the screen.
Save the settings: Click **System Operation** to select “Save Settings”, ”Restart”, and then click the ”Accept” button. Wait for a while (about 40 seconds), and the system will obtain the related IP value from the DHCP Server.

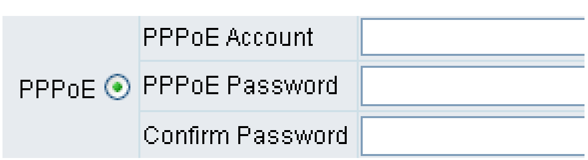
NOTE: After the system has obtained a new IP address, if using WAN Port to enter the Web Configuration Screen, a new IP address has to be used. The same applies to the following two settings.

Setting Static IP



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



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

PPTP※

PPTP ◉	IP address	
	Subnet mask	
	PPTP Server	
	PPTP ID	
	PPTP Password	
	Confirm Password	

Select "PPTP" and enter the 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. The system will take about 40 seconds to restart.
If not familiar with the network connection, please contact your local ISP.Domain Name Server.

BigPond (for Australia use only)

BigPond Cable ◉	User Name	
	BigPond Cable Password	
	Confirm Password	
	Login Server	

Click "BigPond Cable" Enter User Name and Password. Login Server is optional. Then click the "Accept" button at the bottom.

(DNS) Settings

Domain Name Server Assignment	○ Auto ◉ Manual
Domain Name Server (Primary) IP	168.95.1.1
Domain Name Server (Secondary) IP	

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

Auto : Gateway learns primary & secondary addresses from ISP's DHCP server or PPPoE server.

Manual : Enter the primary & secondary addresses manually. Please be sure the IP addresses are correct otherwise the gateway will not be able to access hosts with its hostname.

Clone MAC

Factory Default MAC Address	00AABBCCDD00	Restore
Your MAC Address	000C295FC915	Clone
Current MAC Address		

Some Internet Service Providers (ISP) assigns the bandwidth via the MAC (Media Access Control) Address. You can click the " Clone" button to copy the MAC address of the Ethernet Card installed in the computer used to configure the device. It is only necessary to fill in the field if required by your ISP.
The "Your MAC Address" will be blank as you log in through WAN port.

Using Phone Books Manager

Enable Phone Book Manager Server	☐ Clients List
Share Phone Book to Clients	☐
TTL (Expire time: mins) [1 - 60]	1
Register to Phone Book Manager	☐
Gateway Name for Phone Book Manager	
Phone Book Manager Login Password	
Confirm Password	
Phone Book Manager IP/Domain	192.168.1.1
Phone Book Manager Server Listen Port [1 - 65535]	1690

- Enable Phone Books Manager Server: It allows other Gateway users to register the IP address and telephone number in this Phone books manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- Share Local Phone Book: While this option is enabled and the gateway is performing as a Phone Books Manager, this gateway will append its local 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 Books 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 Books Manager once every 30 seconds.
- Register to Phone Books Manager: To register to the Phone Books Manager.
- Gateway Name for Phone Book Manager: The alias registered with the Phone Books Manager.
- Phone Books Manager Login Password: Enter the registered password. If this system is serving as the Phone Books Manager, the set password is also the password used for registering other Gateway systems.
- Phone Books Manager IP/Domain: Enter the IP address for the Phone Books Manager. It supports URL (Uniform Resource Locator).
- Phone Books Manager Port: The protocol communication port for transmitting signals between the Phone Books Manager and other Gateway systems. Please confirm whether the setting is the same as that of the **Phone Books Manager.**

Network Settings (LAN)

Network Settings (LAN)	
LAN IP / LAN default Gateway	192.168.8.254
Subnet mask	255.255.255.0

- Network Settings (LAN): Gateway LAN Port IP address and the subnet mask value. Please note that Gateway is built under NAT: Gateway LAN Port IP address cannot be in the same section as the NAT LAN Port IP address, or else it is unable to make or receive calls. For example, if the NAT LAN Port IP address is 192.168.8.1, then Gateway LAN Port cannot be ranged between 192.168.8.1 ~ 192.168.8.254. If so, please change the LAN port IP address, (e.g. setting the IP address to 192.168.99.254.)

Port of Web Access from WAN [0=disable, 1 - 65535]	80
Enable Web UI	☑
Enable Telnet Service	☑

- Port of Web Access from WAN: Http port for WAN. To make this setting, the LAN Port must be used. Settings cannot be made using the WAN Port. Always use port 80 when connecting to LAN port.

QoS Settings

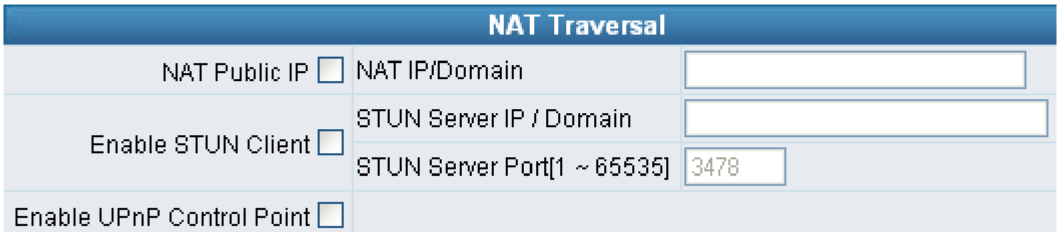
WAN QoS		
☐ QoS	Downstream Bandwidth	Full
	Upstream Bandwidth	Full
ToS / DiffServ Settings		
ToS IP Precedence ◉	Signaling Precedence	3 (Flash)
	Voice Data Precedence	5 (CRITIC / ECP)
DiffServ (DSCP) ○	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31)
	Voice Data Value	46 (Expedited Forwarding, EF)

- QoS (Quality of Service): Sets an external bandwidth to ensure sound quality during transmission (When this function is enabled, the voice packet has the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission).

- ToS/DiffServ (Type of Service/DSCP): The voice packet has the highest priority to ensure telecommunication quality; the larger the value you set, the higher priority you will get.

NAT/DDNS

NAT Traversal



If a Gateway is set up under an IP sharing you can select NAT or STUN protocol.

- NAT: The IP address used by Gateway should be a virtual address. Further, users must set the Virtual Server Mapping in the NAT Server (A virtual server is defined as a Service Port，and all requests to this port will be redirected to this specified the server IP address).
  The default port is listed below:
  Listen Port (UDP): 5060
  RTP Port Base (UDP): 9000~9031 (Channel used for telephone communication).
  Http Port (TCP): 80 (The port can remain disable if users are not going to login from a remote end for the setting).
- NAT IP/Domain: Enter the NAT Server IP address (Real External IP address of NAT Server). If using DDNS (Please refer to the next setting item), then fill in the URL (Uniform Resource Locator).
- STUN: Use STUN protocol prevents problems setting IP sharing, but some NAT do not support this protocol.
- STUN Server and Port: Enter the STUN server IP address and Listen Port number.
- Uses UPnP: Add a new function to enable the gateway's IP traffic to pass through a NAT server. This function only works when the NAT server support UPnP and has it enabled.

Setting up the DDNS (Dynamic Domain Name Service) to solve the problem of general Gateway, which cannot be set up under NAT (which uses a dynamic IP address).

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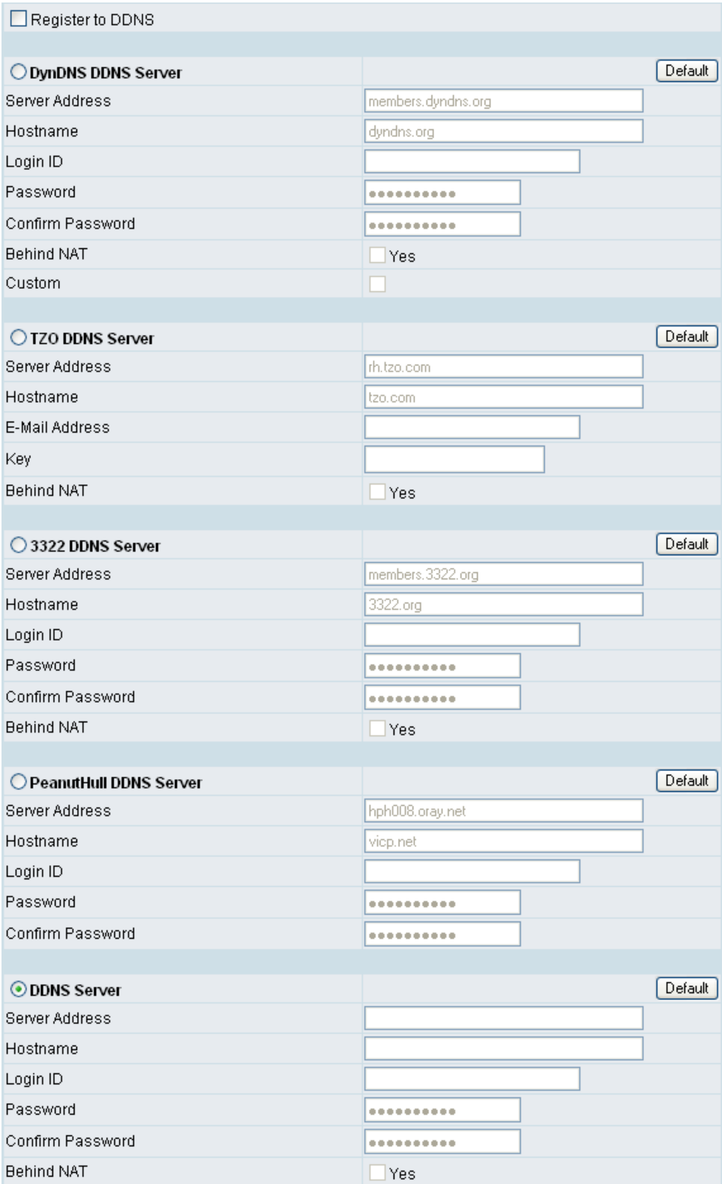

These settings are only necessary when Gateway is set up under a NAT, which not only uses a dynamic IP address but also does not support DDNS.
Choose a DDNS Server: The current system allows users to choose either DynDNS、TZO、3322.org、PeanutHull or a private server. Please apply for a user account before choosing a service provider.

- Server: Set up the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- Hostname: The URL (Uniform Resource Locator) of the system (or NAT) – apply for from a domain name registration providers.
- ID and User’s Password: The ID and password are used to login the DDNS server.
- IP Auto Detect: Select only when the system is set up under NAT.

NOTE: If the Gateway is set up under NAT, then enter the hostname into the NAT IP/Domain.

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Telephony Settings

Prefix Number Rules

Prefix Number Rules	
Trunk Dial Out Verify	
Trunk Dial Out Replace	
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.

For example:
If you transit out with 01907123456, the system will trans to 190601 907123456. If you transit out with 008621123456 the system will replace it with 190200 8621123456.

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

FXS Caller ID Generation	(•) Disable ( ) DTMF ( ) FSK
FXO Caller ID Detection	[x]
Detection Level	0
FSK Caller ID Type	(•) Bellcore ( ) ETSI
Anonymous Caller ID (CLIR)	[ ]
Anonymous Transit in W/O Caller ID	[ ]
Trunk Incoming Prompt Voice	(•) Default Greeting ( ) Recorded voice file ( ) Dial Tone
Upload Greeting	Browse... Upload Backup
FXO Hunting VoIP call in option	Caller Indicate Dial-Out
FXO Hunting Default Dial-Out	
FXO Line VoIP call in option	Caller Indicate Dial-Out

- FXS Caller ID Generation: Select this option to enable the caller ID display function. When enabled, the caller's phone number will be displayed on your phone set when the call comes through. FSK is preferred in some countries.
- FXO Caller ID Detection: To detect the Caller ID delivered from PSTN to the FXO port. While enabled, the Caller ID detected on the FXO port will be send to the SIP Proxy Server on dialing out calls.
- Detection Level: The gain volume occurs on Caller ID detection.

Note: You have to enable "Wait for Caller ID before FXO / Trunk pick up" to ensure Caller ID is detected correctly.

- FSK Caller ID Type: Select FSK type. In most cases, Bellcore is preferred in North America and ETSI in Europe.

- Anonymous Caller ID (CLIR): When enabled, the caller's phone set will not display your number.

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

- Anonymous Transit in W/O Caller ID: FXO won't detect caller ID, and the gateway will dial out with anonymous caller identification. If the call needs caller ID to be identified for Proxy, Proxy will reject this call without caller id.
- Trunk Incoming Prompt Voice: Select the greeting (must use the IVR 132 function to record a voice file) when FXO receives an inbound call.
- Upload Greeting: It is able to upload the recorded voice file. The format must be G.723.
- FXO Hunting VoIP call in option: To set FXO dial-out mode by using the default setting or waiting caller to dial out when the VoIP call calls FXO hunting number.
- FXO Hunting Default Dial-Out: To set FXO default dial-out number.
- FXO Line VoIP call in options: To set FXO dial-out mode when the VoIP call indicates the FXO extension number.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	Dial-Out Prefix	FXO Line Default Dial-Out	FXS Group	Enable FAX
1	☑	FXS	☐		0			☑	☑
2	☑	FXS	☐		0			☑	☑
3	☑	FXO	☐		0				☑
4	☑	FXO	☐		0				☑

- 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.

Hotline Functions

- FXS port: When the user picks up the phone, the gateway automatically dials your assigned hotline number. When in hotline mode, other lines cannot be used.
- FXO port: 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:

1. The assigned hotline for Line 1(Port 9) 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.
2. The assigned hotline for Line 2(Port 10) is 702 and the wait time is 0 second. When Line 2 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: To set the dial-out number when this FXO line is indicated.

Example:

If PBX extension needs to dial "0" to make a PSTN call, and the FXO 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 to any PBX extension line (FXS interface does not have a trunk prefix function). Please refer to Section 6 if required to dial to a PBX extension line and PSTN concurrently.

- Enable FAX: Enable this line to detect if there is a FAX tone to transfer the Codec.

Trunk Hunting Order	First Idle
Enable FXO / Trunk Extension Number	☑
Pick up Line by Dialing Extension Number	☑
Wait for Caller ID before FXO / Trunk pick up	☑
Transit in Busy Tone Limit [0 - 60 s]	3
Ring (Early Media) Time Limit [10 - 600 s]	90
Enable End of Digit Tone	☐
VoIP Call Out Notification	☐
Enable Built-in Call Hold Music	☑
Force Calling Thru PSTN Code	
Trunk Early Media Option	One Way Voice
Early Media Treatment	☑
Compare SIP 'To' Header for Transit Out	☐
Max. External Call	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.

Description:

If the user at Phone 1 (Port 1) of this system wants to assigns Line 8 (FXO) to make a call, he/she can dial 708 22520199.
If this item isn't checked, the gateway will select a line automatically to call out from. For example, dial 22520199 without adding the extension number of the FXO port.

- Pick up Line by Dialing Extension Number: Allows user to dial just the FXO extension – 708 - 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.
- Ring 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 the call is in progress.
- VoIP Calling Notification: The gateway will play a tone to notify the call is through VoIP.
- Force Calling Thru PSTN code: Dial the code to get a PSTN line for dial out. For example: If you would like dial "23456789" through PSTN and Force Calling Thru PSTN code is *33, just dial "*33 23456789"
- Early Media Treatment: If it is disabled, the system will send RTP immediately when the connection with Proxy is set up. The default is enabled. If communicating with other Gateway has problem, please disable this function.
- Compare SIP 'To' Header for Transit Out: When FXO is callee 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.

SIP

All Call through OutBound Proxy	☐
OutBound Proxy IP / Domain	
OutBound Proxy Port [1 - 65535]	5060

- All Call through OutBound Proxy：An outbound proxy server handles SIP call signaling as a standard SIP proxy server would. Furthermore, it receives and transmits phone conversation traffic (media) in between two talking gateways. 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 enable it.

Session Expiration [0=disable, 10 - 1800]	0
Session Refresh Request	◉ UPDATE ○ re-INVITE
Session Refresher	◉ UAS ○ UAC

- Session Expiration: It is to avoid the billing of abnormal dropping the call because of

Internet. The default is disabled.

- Session Refresh Request: to send the packet of UPDATE or re-INVITE to
- Session Refresher: It is the gateway's role in Session Timer. UAS is an originator, and UAC is a replier.

Enable P-Assert	☐
Privacy Type	id

- Enable P-Assert: It is for caller id protect.
- Privacy Type: Privacy requested for Third-Party Asserted.

SIP Message Resend Timer Base [s]	0.5
Max. Response Time for Invite [1 - 20]	8
Invite URL need 'user=phone'	☑
Reliability of Provisional Responses	☐
Compact Form	☑

- SIP Message Resend Timer Base: SIP packet will resend if response didn't 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 max of resend time is 4 sec. Resend will stop/restart when total resend 20sec has reached.
- Max. Response Time for Invite: If the destination does not reply in the set time, this 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 ticked.
- Compact Form: It decreases the size of SIP header if ticked.

E.164

International Call Prefix Digit		
Country Code	(Other)	
Long Distance Call Prefix Digit		
Area Code		
E.164 Numbering	To Invite Proxy	☐
	Transform to Transit Out	☐
ENUM Header Exception	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: To invite Proxy to follow the E.164 rule. It depends on the Proxy. **If you fail to make a call, please contact your ITSP**.

[ ] Enable Support of SIP Proxy Server / Soft Switch	
[x] **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	[ ]
[ ] **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	[ ]
Bind Proxy Interval for NAT [0 - 180 s]	0
Initial Unregister	[ ]
Enable Message Waiting Indication (MWI)	[ ]
Proxy-Require	

- 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 are 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). You can set 3 redundant Proxy spread by semicolon.
  EX: 61.123.231.1;12.34.56.78;proxy.sip.sip
- Proxy Server Port: Enter the Proxy Server **listen** port number. (The factory default value is 5060)
- Proxy Server Realm: Enter the correct registered Proxy Server Realm name to avoid registration failure. **If you fail to make a call, please contact your ITSP.**
- TTL: Enter the desired time interval at which the gateway will report to you Proxy Server.
- SIP Domain/Use Domain to Register: Enter the correct SIP domain to avoid registration failure (it is not necessary to set with some Proxy Servers). If you enable "Uses Domain to Register" the gateway will register to Proxy with the domain name you filed. Else, the gateway will register to a Proxy with the IP it resolves. **If you fail to make a call, please**

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contact your ITSP.

- Bind Proxy Interval for NAT: This function is able to keep the binding is existed when the gateway is behind NAT and SIP Proxy is not able to keep the binding.
- Initial Unregister: After rebooting, it is unregistered first and then do the general registry process.
- Enable Message Waiting Indication: The system will play a tone to remind users that there are messages in SIP Server.
- Proxy-Require: Some SIP Sever need SIP UA to add this header to it's sip message.

FXS/ FXO Representative number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
FXS Representative Number		19007241	☑		nekol1	●●●●●●●●●●	●●●●●●●●●●
FXO Representative Number		0144	☑		nekol2	●●●●●●●●●●	●●●●●●●●●●
FXO Representative Number		0144	☑		nekol2	●●●●●●●●●●	●●●●●●●●●●

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

- FXS Representative Number: Register all FXS ports as a hunting group.
- FXO Representative Number: Register all FXO ports as a hunting group. All the grouped FXO ports will be hunted automatically. It is available when FXO registers to Proxy.
- Register: Register to Proxy if ticked.
- Invite with ID / Account: DVG-4032S can be invited to a VoIP trunk gateway w/o register to a Proxy. Please contact your ITSP

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
1	FXS	701 Auto	☑	☑	nekol1	●●●●●●●●●●	●●●●●●●●●●
2	FXS	702	☑	☑	nekol2	●●●●●●●●●●	●●●●●●●●●●
3	FXO	703	☑	☑	nekol3	●●●●●●●●●●	●●●●●●●●●●
4	FXO	704	☑	☑	nekol4	●●●●●●●●●●	●●●●●●●●●●

- Invite with ID / Account: The gateway can be invited to a VoIP trunk gateway w/o register to a Proxy. Please contact your ITSP

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.

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
FXS Representative Number					(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
FXO Representative Number					(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
1	FXS				After[10 - 60] 20 s				
2	FXS				After[10 - 60] 20 s				
3	FXO				(N/A)	(N/A)	(N/A)	(N/A)	(N/A)
4	FXO				(N/A)	(N/A)	(N/A)	(N/A)	(N/A)

- Unconditional Forward: All incoming calls will be forwarded to the “Forwarding Number” automatically.
- Busy Forward: Forward the incoming call to “Forwarding Number” when the port is busy.
- No Answer Forward: Forward the incoming call to “Forwarding Number” after ring timeout expires without answer.
- Call Hold: Enable the call hold feature on the specific FXS port.
- Call Transfer: Enable the call transfer feature on the specific FXS port.
- Call Waiting: Enable the call waiting feature on the specific FXS port.
- Three-Way Calling / Service ID: Feature code of conference call defined on Nortel Soft Switch

Calling Feature Instructions:

- Call Hold: Ongoing call will be put on hold after FLASH button pressed on the phone set. The gateway will play a repeating music to the remote end.
- Call Transfer: Ongoing call will be put on hold after FLASH button pressed on local phone set (gateway plays a repeating music to the remote end). Meanwhile, local user can dial out to another number after dial tone observed. After the handset is back on the hook, the call on hold will then be transferred to the new call regardless of the status of the new cal. If wrong number is dialed for the new call, just press the FLASH button to get back the call on hold. In another case, if the local user doesn’t hang up the phone after new call sets up, press FLASH button to switch between the first call and the new call. Please be informed that PBX between phone sets and the gateway must support FLASH features to make this function work correctly. If a phone set is connecting directly

to the FXS port of the gateway and not functioning to FLASH, please adjust the settings in “Flash Detect Time” in category “Advanced Options”.

- Example of a Three-Way calling:

1. Alex dials to Bob, Bob answers that call.
2. Alex presses Flash and call to Coral (Bob is on hold), Coral answers that call.
3. Alex dials *61 then presses Flash, thus conference call is created.

Or

1. Alex dials to Bob, Bob answers that call.
2. Coral dials to Alex (Call Waiting), Alex presses Flash to pick the second call and talk to Coral.
3. Alex dials *61 then presses Flash, thus conference call is created.

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Advanced Options

Administrator's Name	
Administrator's Password	●●●●●●●●●●●
Confirm Password	●●●●●●●●●●●
Web UI Login ID	
Web UI / IVR Password	●●●●●●●●●●●
Confirm Password	●●●●●●●●●●●
Web UI auto logout [30 - 300 s]	60

- There are two levels to enter Web. Administrator is able to change all settings. Web UI only changes some settings.
- Web UI auto log out: When logging in a web page, if a user does not act within the effective time range, the user will be disconnected from the web page to allow others to login.

Dial Wait Timeout [1 - 60 s]	10
Inter Digits Timeout [1 - 60 s]	4
Minimum DTMF ON Length [40 - 500 ms]	80
Minimum DTMF OFF Length [40 - 500 ms]	80
DTMF Detection Sensitivity	(less) ○ 1 ○ 2 ● 3 ○ 4 ○ 5 (more)

- Dial Wait Timeout: Use it to set the waiting time for the user's first key pressing when dialing a number. The user will hear a busy tone if he/she does not press the first key within the set time frame.
- Inter Digits Timeout: Set the waiting time between each key pressing. The inputted numbers will be dialed after the timeout.
- Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones): Used to set 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	DTMF
Pulse Dial Mark/Space Ratio	US (61:39 %)
FXO Impedance	Taiwan 600 Ohm
FXS Impedance	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/FXS Impedance: Choose correct impedance in your country/area. ※

Enable Out-of-Band DTMF ☐	☐ Enable Hook Flash Event
	◉ RFC 2833 Payload Type 101
	○ SIP Info
Use Second CPT after SIP registered	☐

- Enable Out-of-Band DTMF: To send DTMF keys (0~9, *, #,) follow the RFC2833 rules or via SIP Info.
- Enable Hook Flash Event: The gateway will deliver the flash signal to remote party via RFC2833 or SIP Info.
- Payload Type：Payload type 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 fails 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 Settings)

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Line Settings

Line	Extension Number	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time	Enable Polarity Reversal	PSTN Answer Detection	PSTN Ring OFF Length [1000 - 20000 ms]
1	701	FXS	0 All	0 All	5 All	0.6 All	☐		
2	702	FXS	0	0	5	0.6	☐		
3	703	FXO	0	0	5	0.6	☐	Disable	2000
4	704	FXO	0	0	5	0.6	☐	Disable	2000

- Listening Volume: Adjusts the hearing volume.
- Speaking Volume: Adjusts the speaking 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, Busy Tone, and so on.
- Flash Time:

  FXS: Used to adjust the detecting period of flash signal from the phone set connected to the FXS port. For example, if pressing the HOLD key will disconnect a call, increase the “Flash Detect Time” should fix this issue.

  FXO: Used to set the time frame that FXO generates a FLASH signal.
- Enable Polarity Reversal:

  FXS: As the remote site answer this call or hook on the FXS port will reverse the polarity.

  FXO: This option forces the gateway to detect the reversal of polarity on FXO port as the primary signal to drop a call. Some telephone switches or PBX reverse the line polarity to inform the remote end to drop an ongoing call. Please consult with the telephone service provider for availability of this feature.
- PSTN Answer Detection: This is only used with ITSP.
- PSTN Ring OFF Length: FXO/PSTN make out if the call from PSTN hangs up before FXS answer the call. ※

Voice

Codec Settings					
Preferred Codec Type	G.723.1 6.3kbps				
Jitter Buffer [60 - 1200 ms]	120				
Silence Detection / Suppression	☐				
Echo Cancellation	☑				
Codec	☑ G.711 u-law	☑ G.723.1	☑ G.726	☑ G.729A	☑ 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 codec have different compression ratios, so the sound quality and occupied bandwidths are also different. It is recommended to 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.
- Packet Time: Defines how long the DVG-3016S send a RTP packet-voice packet- to the other side. The smaller the value, the more bandwidth usage. The larger the value, the more voice delay.
- Approximate Bandwidth Require: The bandwidth required varies with Codec format and packet time.

FAX

FAX Settings		
T.38 ◉	◉ UDP ○ TCP	
	☑ Enable High Quality	
T.30 ○	FAX Codec	G.726 32kbps
	FAX Jitter Buffer [60 - 1200 ms]	360
FAX Tone Detection Sensitivity	(less) ○ 1 ◉ 2 ○ 3 ○ 4 ○ 5 (more)	

- T.38: The T.38 protocol is used for better and faster facsimile transmission. When this function is enabled, the following fax and voice parameter settings will be disabled, so it is recommended to enable this function to gain better fax quality. When this function is enabled, please select UDP, TCP, or AUTO. If selecting TCP and some routers cannot use the Fax function, please select UDP instead.
- Enable High Quality: To ensure better quality.
- If T.38 is disabled, then the system will use T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the fax function will consume more network resources and will affect transmission quality.
- FAX Detect sensitivity：used to adjust the sensitivity of detect a phone call whether be FAX or not.

Drop Inactive Call		
Silence Detection Threshold [0 - 60 db]	0	(0 : Disable)
Drop Silent Call Timeout [30 - 3600 s]	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: Set the volume as a standard.
- Drop Silent Call Timeout: Set the time to hang up the phone.

Digit Map

There are 50 sets of leading digit entries to choose voice routing interface – Auto select, PSTN or VoIP.

Default Call Route	Auto (VoIP first)

- Default Call Route: The default call route can be Auto, VoIP, PSTN and 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 deny if the dial-out number is not in the table.

#	Enable	Leading Digits	Total Digit Count [0=disable, 1- 40]	Route
1	☐		10	Auto (VoIP first)
2	☐		10	Auto (VoIP first)
3	☐		10	Auto (VoIP first)

- Enable: Enable detection of this entry.
- Leading Digits: The leading digits for DVG-3016S to scan while user is dialing.
- Total Digit Count: Total number of digits that DVG-3016S should accept. 0 disables this feature.
- Route: The interface calls should go through if above conditions satisfied.

Local 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 books manager is set within a Gateway group, then all Gateway systems have to set up phone data for each VoIP Gateway to communicate with each other.

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

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

Speed Dial

This system can set up 100 numbers for speed dialing. Setting methods are as follows:

#	Speed Dial Code("?" = single digit ; "%" = wildcard)	Number To Dial
1	55	32568791
2	3??	5213??
3	00%	856%

Method 1- Single mapping: Fill a short code into the "Speed Dial Code" column, and enter the desired phone number into the "Number To Dial" column.
For example, pick up the handset and dial **55**＃ and the system will dial 32568791.

Method 2- Multi mapping; Fill the prefix code into the " Speed Dial Code" column and the format to transfer into the "Number To Dial" column.
For example, pick up the handset and dial **301**＃, and the system will dial 521301.
If the user dial 00 1657987456321, the system will DIAL 856 1657987456321

Caller Filter

This function is used at allow or deny SIP Invite from the Proxy list ONLY.

(●) Allow ( ) Deny

Enable	Filter IP address	Subnet mask
☐		
☐		
☐		

- Filter IP Address: Fill up with the start IP you would like to allow/deny.
- Subnet mask: Fill up with the subnet mask you would like to allow/deny.

ACL for Management

You can use ACL to allow the user that is from some IP address to enter Web.

☐ Enable ACL for Management

Filter IP address	Subnet mask	Web
		☐
		☐
		☐

- Enable ACL for Management: Enable ACL for Management if ticked.
- Filter IP Address: Fill up with the start IP you would like to allow.
- Subnet mask: Fill up with the subnet mask you would like to allow.

- Web: Enable management from Web if ticked.

CDR Settings

The user can set up a CDR Server to record call detail for every phone call.
The present CDR provides the call detail recording in a text file and imports the text file to prepare for an analysis report, if needed.

☐ Send record to CDR Server	
CDR Server IP	
Port [1 - 65535]	8080

- Send record to CDR Server: Enables the call detail recording function.
- CDR Server IP: Enter the IP address of the CDR server.
- Port: Enter the listen port of the CDR server.

Language

The system provides English, Traditional Chinese, and Simplified Chinese to display text on Web pages. Meanwhile, it will change the language for IVR (Interactive Voice Response).

Web UI / IVR Language	English

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	☐
Outbound Call Control	☐

- 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		☐	0
2		☐	0
3		☐	0

- PIN Code: Enter the PIN code (4-6 digits or leave blank. A blank indicates no PIN code is required at this level. Generally, the PIN at level 5 can remain blank to simplify the phone number.)
- Enable: 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.)

Long-Distance Control Table

This table controls the level of authority of an outgoing call through FXO.

#	0	1	2	3	4	5
1	0204					
2						
3						

If Level 0 (the highest level) is set to prohibit dialing any number started with prefix 0204, then any level below 0 (including Levels 1 to 5) is also prohibited.

If Level 1 is set to prohibit dialing any number with prefix 0, then any level below 1 (including Levels 2 to 5) is also prohibited. Since Level 0 is not restricted to any prefix, therefore at level 0 users can dial a number with the prefix 0.

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

Long Distance Exception Table

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

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

#	0	1	2	3	4	5
1	020488988					
2						
3						

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

CPT/Cadence Settings

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

Busy Tone Cadence Measurement

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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: Provide a best 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 cut off an un-finished call, select less sensitivity.

CPT parameters Table

The CPT has 3 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

System Information

Port Status							
No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	701	Idle	0		Disabled	
2	FXS	702	Idle	0		Disabled	
3	FXS	703	Idle	0		Disabled	
4	FXS	704	Idle	0		Disabled	
SIP Proxy Hunting Number Registration					FXS Disabled (01:08:44)		

Server Registration Status	
DDNS Registration	Disabled (01:08:44)
Phone Book Manager Registration	Disabled (01:08:44)
STUN Registration	Disabled (01:08:44)
UPnP Negotiation	Disabled (01:08:44)

WAN Port Information	
Factory Default MAC Address	00 AA BB CC DD 00
IP Address	192.168.1.2
Subnet Mask	255.255.255.0
Default Gateway	192.168.1.254
DNS	168.95.1.1

LAN Port Information	
MAC Address	00 AA BB CC DD 01
IP Address	192.168.8.254

RTP Packet Summary

Displays the information of the last finished call. It contains peer IP, peer port, packets sent, packet received and packet lost.

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	

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STUN Inquiry

NAT Type	Unknown
STUN Server IP / Domain	
STUN Server Port [1 - 65535]	3478

Ping Test

Use "ping" to identify if the remote peer is reachable. Fill in remote IP address and click "Test" will start the test.

Ping Destination	
Number of Ping [1 - 100]	4
Ping Packet Size [56 - 5600 bytes]	56
	Test Stop

SNMP

Enable SNMP Agent	☐
Get Community	
Set Community	
Trap Community	
Trap Host	

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

NTP

	Year	Month	Day	Hour	Minute	Second
Gateway Time	2000	1	1	9	18	36
Time Zone	+ 8 : 00					
#	Time Server					
1	ntp.ucsd.edu					
2	ntp.univ-lyon1.fr					
3	time.nuri.net					

- Time Zone: Set the Time Zone where DVG-3016S resides.
- Time Server #1~#3: Set the Time Server where DVG-3016S should sync up during start up. (NTP protocol)

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: It needs to Save Settings and Restart, and all settings will back up default settings or have new setting that you upload.

Backup Configurations	
Configuration File	Backup
Configuration Template File	Backup

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

Restore Configurations	
◉ Upload Configuration File	Browse...
○ Restore Default Configurations	
Restore	

System Operations

☐ Save Settings	Save all configurations.
Be sure to save all settings before restart.	
☐ Restart	Restart the Gateway right away. All calls will be DROPPED when Restart.

- Save Settings: Save settings after completing. 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.

Software Upgrade

Gateway provides software upgrade function for a remote end.

To Save Current Settings, Save Settings	
Current Software Version No. [1.2.33.7]	
Upgrade Server	◉ TFTP ○ FTP ○ Image Server
Software Upgrade Server IP	
Software Upgrade Server Port [1 - 65535]	69
User Name	
Password	
Directory	
All calls will be DROPPED during upgrade.	
Accept	

- Software Upgrade Server IP address: Please enter the software server IP address.
- Software Upgrade Server Port: The default setting is 6001(Do not change this setting).

Logout

Gateway only allows one user to login at a time, so whenever a change is made, please save the settings, restart the system, or logout to avoid the situation where other users cannot login to change settings.

To save settings, click Here
Accept

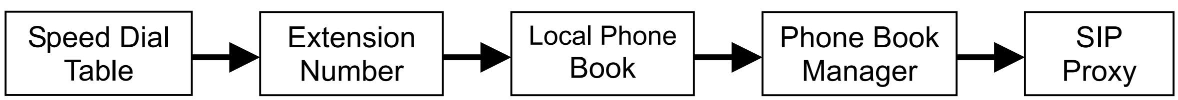
5. Coding Principle

Instruction

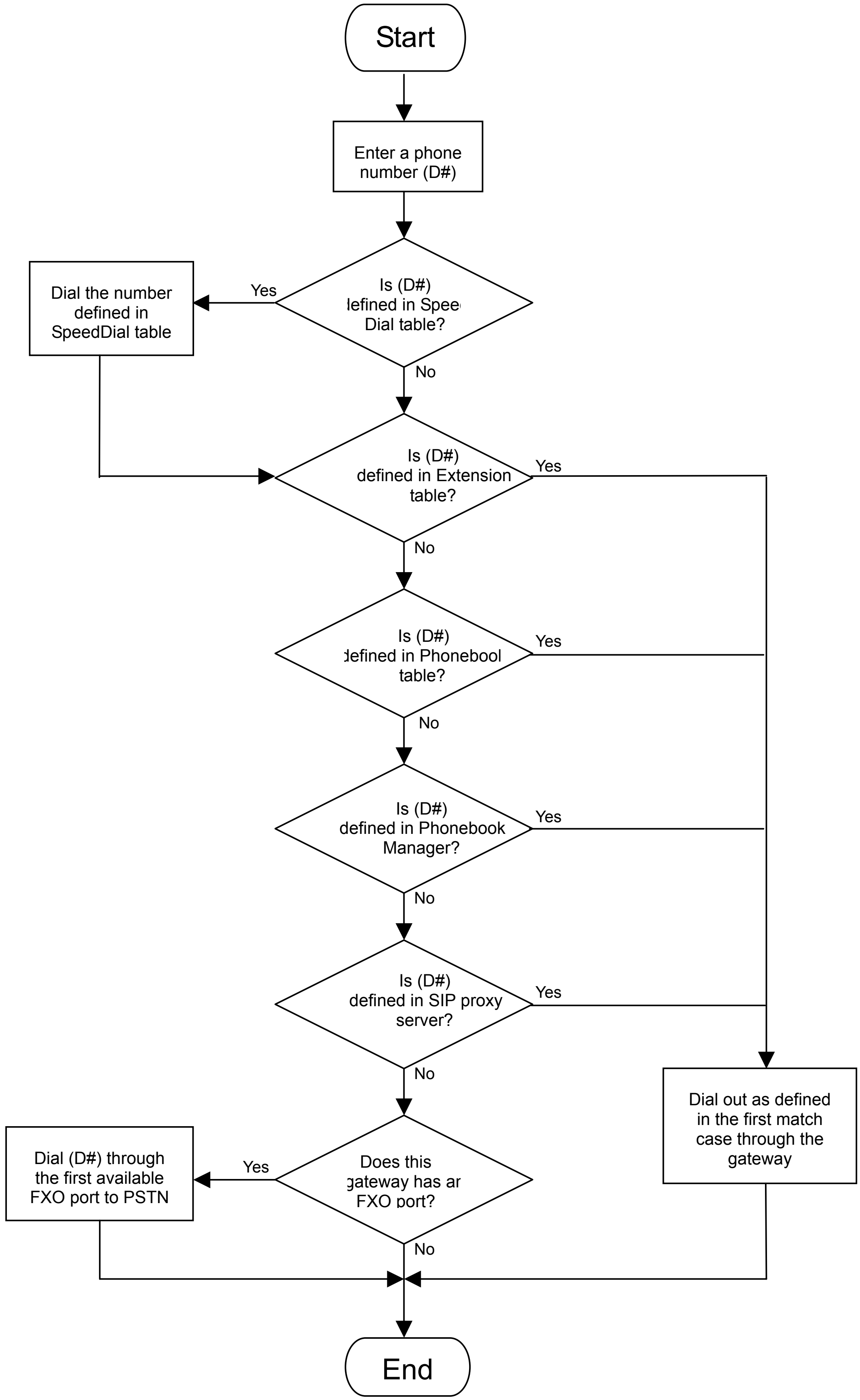
- After a phone number is entered, dial # to call out immediately or, wait until the "Inter DTMF Timeout" expires (defined in "Advanced Options", default=4 seconds).
- If the phone number fits the setting of Digit Map, the gateway dials out the phone number through the assigned interface automatically.
- The phone number should have at least 2 digits (not including * and #).

Dialed Number Processing Flow

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



A complete flow chart is on the next page.


Start
Enter a phone number (D#)
Is (D#) lefined in Spee Dial table?
Yes
Dial the number defined in SpeedDial table
No
Is (D#) defined in Extension table?
Yes
No
Is (D#) defined in Phonebool table?
Yes
No
Is (D#) defined in Phonebook Manager?
Yes
No
Is (D#) defined in SIP proxy server?
Yes
No
Dial out as defined in the first match case through the gateway
Does this gateway has ar FXO port?
Yes
Dial (D#) through the first available FXO port to PSTN
No
End