

D-Link[®] CORPORATION

D-Link[®]

**DVG-2016S
VOIP Gateway**

User Manual

Version 1.0

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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

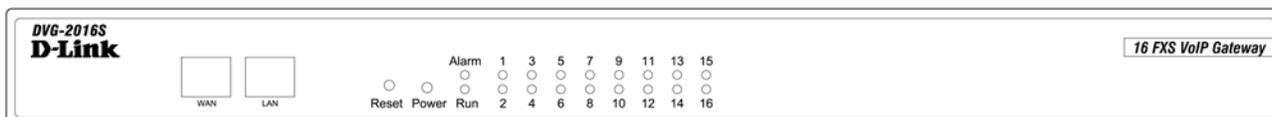
- SIP (RFC 3261) compliant
- Optional server enables small businesses to build up private VoIP network (SIP model)
- QoS support guarantees voice bandwidth in a busy network
- Supports IP TOS (Type Of Service)
- T.30 (G III) / Real Time T.38 / Secured T.38 Fax Relay
- Feasible for Fixed IP address or 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.
 - P1 – P16 stands for Port 1 – Port 16: Connect to your analog telephone.
 - 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-2016S 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.

2. Installation and Applications

Network Interface

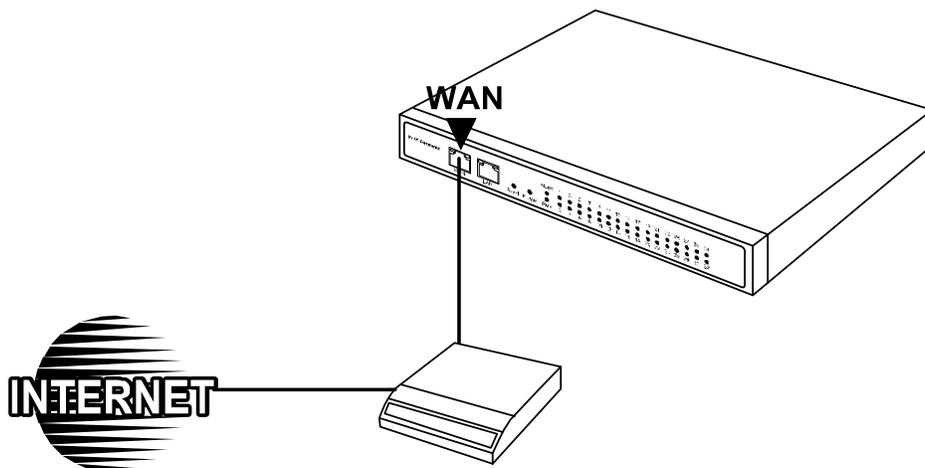
The network interface is divided into 3 basic modes as described below:

- 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).

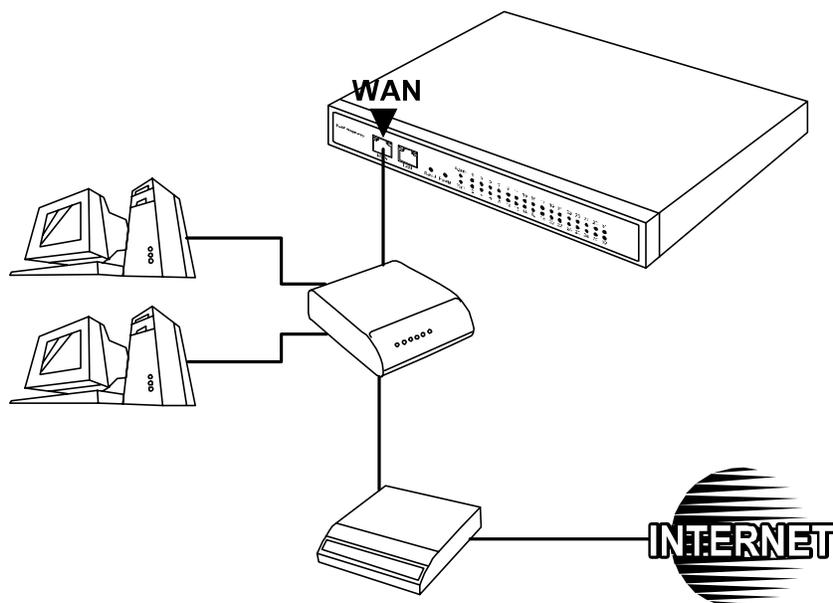
Gateway IP Settings	Need to set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	



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 using 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

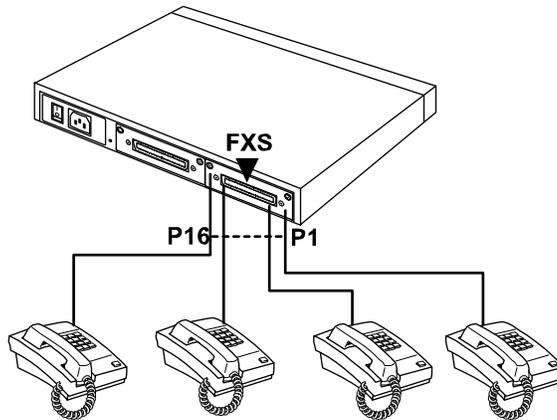


Telephone Interface Description

Example for DVG2016S:

DVG2016S connecting directly to phone sets

After connecting telephone sets to P1-P16, users can make direct calls, (P1-P16 are FXS interfaces). Each set acts as an independent extension line.



Integrating the DVG2016S 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.

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)

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.

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 under 114), 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	<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 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 settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter <u>101</u> 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 080409040301020271484954456072544560#.

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.

1. 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.

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 left blank**).
- Change the default settings of Administrator's Name, Password and Web UI Login ID, Password in **Advanced Options**.
- 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.



The gateway doesn't allow multiple people to configure the gateway at same time. 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 (WAN)

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

Network Settings (WAN)	
Current WAN IP Address	192.168.1.2
Listen Port UDP [1 - 65535]	5060
RTP Starting Port UDP [1 - 65500]	9000
DHCP <input type="radio"/>	
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
PPPoE Account	
PPPoE <input type="radio"/>	PPPoE Password
	Confirm Password
PPTP <input type="radio"/>	
	IP address
	Subnet mask
	PPTP Server
	PPTP ID
	PPTP Password
	Confirm Password
BigPond Cable <input type="radio"/>	
	User Name
	BigPond Cable Password
	Confirm Password
	Login Server
Domain Name Server Assignment	<input type="radio"/> Auto <input checked="" type="radio"/> Manual
Domain Name Server (Primary) IP	168.95.1.1
Domain Name Server (Secondary) IP	
Factory Default MAC Address	00AA8BCCDD00 <input type="button" value="Restore"/>
Your MAC Address	000C295FC915 <input type="button" value="Clone"/>
Current MAC Address	
Enable Phone Book Manager Server	<input type="checkbox"/> <input type="button" value="Clients List"/>
Share Phone Book to Clients	<input type="checkbox"/>
TTL (Expire time: mins) [1 - 60]	1
Register to Phone Book Manager	<input type="checkbox"/>
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
LAN interface mode	
<input checked="" type="radio"/> Router <input type="radio"/> Bridge	
Network Settings (LAN)	
LAN IP / LAN default Gateway	192.168.0.254
Subnet mask	255.255.255.0
DHCP Server	
Enable DHCP Server	<input checked="" type="checkbox"/>
IP Pool Starting Address	192.168.0.1
IP Pool Ending Address	192.168.0.250
Lease Time [1 - 9999 hours]	1
Domain Name Server Assignment	<input checked="" type="radio"/> Auto <input type="radio"/> Manual
Domain Name Server (Primary) IP	
Domain Name Server (Secondary) IP	
Port of Web Access from WAN [0=disable, 1 - 65535]	80
Enable Web UI	<input checked="" type="checkbox"/>
Enable Telnet Service	<input checked="" type="checkbox"/>

- Listen Port UDP: It is not necessary to change the protocol of the communication port used by the gateway.
- RTP Starting Port UDP: The initial value of port number for transmitting voice data among Gateway(s). Each line requires 2 ports (RTP/RTCP). It is not necessary to change these.

For example: If the starting port is 9000, then Line 1 is using 9000(RTP) and 9001(RTCP),

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

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" 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 <input checked="" 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="text"/>
	Confirm Password	<input type="text"/>

Select "PPTP" and enter the IP Address, Subnet mask, PPTP Server, PPTP ID and Password. Then click the "Accept" button at the bottom.

BigPond (for Australia use 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" Enter User Name and Password. Login Server is optional. Then click the "Accept" button at the bottom.

(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 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	<input type="checkbox"/> Clients List
Share Phone Book to Clients	<input type="checkbox"/>
TTL (Expire time: mins) [1 - 60]	1
Register to Phone Book Manager	<input type="checkbox"/>
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 Phone Book to Clients:** While this option is enabled and the gateway is performing as a Phone Books 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 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 Server Listen 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)

LAN interface mode	
<input checked="" type="radio"/> Router <input type="radio"/> Bridge	
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"/>
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 interface mode※

LAN interface mode
<input checked="" type="radio"/> Router <input type="radio"/> Bridge

- Router: The system serves as a router with NAT.
- Bridge: The system serves as a bridge between WAN port and LAN port without NAT. (LAN default gateway will still be accessible for configuration).

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 the subnet mask value.

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 make this setting, the LAN Port must be used. **WAN Port cannot make this setting.** Always use port 80 when connecting to LAN port. 0 is to disable http port for WAN.
- Enable Web UI: It disables any access from WAN and LAN if ticked.
- Enable Telnet Service: It allows user to make settings from Telnet.

QoS Settings

WAN QoS

WAN QoS		
<input type="checkbox"/> QoS	Downstream Bandwidth	Full
	Upstream Bandwidth	Full
ToS / DiffServ Settings		
Precedence <input checked="" type="radio"/>	ToS IP 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)

- 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). Some models of the gateway without this function can adjust the bandwidth automatically.
- 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

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 a Gateway is set up under an IP sharing setting, you can select either the NAT or STUN protocol.

- NAT Public IP: The IP address used by the gateway should be a virtual address. Further more, 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 Starting Port (UDP): 9000~9031 (Listen Port used for telephone communication).

Http Port (TCP): 80

- NAT IP/Domain: Enter the NAT Server IP address (Real External IP address of NAT Server) then fill in the URL (Uniform Resource Locator). If using DDNS, please refer to the next setting item
- Enable STUN Client: Using STUN protocol prevents problems with setting the IP sharing function, but some NAT do not support this protocol.
- STUN Server IP/Domain and Port: Enter the STUN server IP address and Listen Port number. You can set 2 STUN server spread by semicolon.
- Enable UPnP Control Point: To enable the gateway's IP traffic to pass through a NAT server. This function only works when the NAT server supports UPnP and has it enabled.

DDNS

<input type="checkbox"/> Register to DDNS	
<hr/>	
<input type="radio"/> DynDNS DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="members.dyndns.org"/>
Hostname	<input type="text" value="dyndns.org"/>
Login ID	<input type="text"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
Behind NAT	<input type="checkbox"/> Yes
Custom	<input type="checkbox"/>
<hr/>	
<input type="radio"/> TZO DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="rh.tzo.com"/>
Hostname	<input type="text" value="tzo.com"/>
E-Mail Address	<input type="text"/>
Key	<input type="text"/>
Behind NAT	<input type="checkbox"/> Yes
<hr/>	
<input type="radio"/> 3322 DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="members.3322.org"/>
Hostname	<input type="text" value="3322.org"/>
Login ID	<input type="text"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
Behind NAT	<input type="checkbox"/> Yes
<hr/>	
<input type="radio"/> PeanutHull DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text" value="hph008.oray.net"/>
Hostname	<input type="text" value="vicp.net"/>
Login ID	<input type="text"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
<hr/>	
<input checked="" type="radio"/> DDNS Server	<input type="button" value="Default"/>
Server Address	<input type="text"/>
Hostname	<input type="text"/>
Login ID	<input type="text"/>
Password	<input type="password" value="....."/>
Confirm Password	<input type="password" value="....."/>
Behind NAT	<input type="checkbox"/> Yes

These settings are only necessary when the gateway is set up under a NAT that uses a dynamic IP address and do 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 address: Sets up the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- Hostname: The URL of the system (or NAT) – apply from a domain name registration providers.
- Login ID and Password: The ID and password are used to log into the DDNS server.
- Behind NAT: 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 that is the same with Hostname of DDNS.

Example:

NAT

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

DDNS

Register to DDNS

DynDNS DDNS Server Default

Hostname

Telephony Settings

FXS Caller ID Generation	<input checked="" type="radio"/> Disable <input type="radio"/> DTMF <input type="radio"/> FSK
FSK Caller ID Type	<input checked="" type="radio"/> Bellcore <input type="radio"/> ETSI
Anonymous Caller ID (CLIR)	<input type="checkbox"/>

- 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.
- 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.

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0 - 60 s]	FXS Group	Enable FAX
1	<input checked="" type="checkbox"/>	FXS	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
2	<input checked="" type="checkbox"/>	FXS	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

- 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.
- 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.
- FXS Group: Select group hunting when there is an incoming call, the gateway will automatically assign an unassigned call according to Hunting Priority. If Line 2 does not want to be set as an assigned line to receive any inbound calls, the function can be disabled. Users can also use the Up or Down key to adjust hunting priority.
- Enable FAX: Enable this line to detect if there is a FAX tone to transfer the Codec.

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"/>
Early Media Treatment	<input checked="" type="checkbox"/>
Max. External Call	<input type="text" value="999"/>

- 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.

FXS Group Hunting / Ring Priority			
Hunting / Ring	Hunting		
Sequential Ring Time [1 - 100 s]	6		
1	Line 1		▼
2	Line 2	▲	▼
3	Line 3	▲	▼
4	Line 4	▲	

- Hunting/Ring: It is able to set FXS group hunting using simultaneous ring or sequential ring.
- Sequential Ring Time: To set the ring time of each port, when sequential ring is chosen.

SIP

All Call through OutBound Proxy	<input type="checkbox"/>
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	<input checked="" type="radio"/> UPDATE <input type="radio"/> re-INVITE
Session Refresher	<input checked="" type="radio"/> UAS <input type="radio"/> 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	<input type="checkbox"/>
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'	<input checked="" type="checkbox"/>
Reliability of Provisional Responses	<input type="checkbox"/>
Compact Form	<input checked="" type="checkbox"/>

- 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	<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"/>
E.164 Numbering	To Invite Proxy <input type="checkbox"/>
	Transform to Transit Out <input type="checkbox"/>
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.**

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

- 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 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 Representative number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password	Confirm Password
	FXS Representative Number	198	<input checked="" type="checkbox"/>		198

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.
- Register: Register to Proxy if ticked.
- Invite with ID / Account: DVG-2016S 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 <input type="button" value="Auto"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	701
2	FXS	702	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	702
3	FXS	703	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	703
4	FXS	704	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	704

- 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.

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	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	(N/A) <input type="text"/>	(N/A)	(N/A)	(N/A)	(N/A)
1	FXS	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> After[10 - 60] [20] s <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>
2	FXS	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> After[10 - 60] [20] s <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>
3	FXS	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> After[10 - 60] [20] s <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>
4	FXS	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> <input type="text"/>	<input type="checkbox"/> After[10 - 60] [20] s <input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/> <input type="text"/>

- Do Not Disturb: It will only be able to call out when it is enabled.
- Unconditional Forward: All incoming calls will be forwarded to the “Forwarding Number” automatically. If it forwards to FXO, it only make FXO hook off, not make FXO dial out.
- 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.

NOTE: Call Hold must be checked; Call Transfer or Call Waiting is active.

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

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"/>

- There are two levels to enter Web. Administrator is able to change all settings. Web UI only changes some settings.

NOTE: Enter new Login ID and password for two levels.

- 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]	<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 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.

FXS Impedance	<input type="text" value="Taiwan 600 Ohm"/>
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- FXS Impedance: Choose correct impedance in your country/area. ✖

Enable Out-of-Band DTMF <input type="checkbox"/>	<input type="checkbox"/> Enable Hook Flash Event
	<input checked="" type="radio"/> RFC 2833 Payload Type <input type="text" value="101"/>
	<input type="radio"/> SIP Info
Use Second CPT after SIP registered	<input type="checkbox"/>

- 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)

Line Settings

Line	Extension Number	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time	Enable Polarity Reversal
1	701	FXS	0 <input type="button" value="v"/> All	0 <input type="button" value="v"/> All	5 <input type="button" value="v"/> All	0.6 <input type="button" value="v"/> All	<input type="checkbox"/>
2	702	FXS	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	0.6 <input type="button" value="v"/>	<input type="checkbox"/>
3	703	FXS	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	0.6 <input type="button" value="v"/>	<input type="checkbox"/>
4	704	FXS	0 <input type="button" value="v"/>	0 <input type="button" value="v"/>	5 <input type="button" value="v"/>	0.6 <input type="button" value="v"/>	<input type="checkbox"/>

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

Codec

Codec Settings					
Preferred Codec Type	G.723.1 6.3kbps <input type="button" value="v"/>				
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 <input type="button" value="v"/>	30 <input type="button" value="v"/>	20 <input type="button" value="v"/>	20 <input type="button" value="v"/>	20 <input type="button" value="v"/>
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.
- Codec: Choose the codec that you needs.
- Packet Time: Defines how long the gateway sends 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 Settings

FAX Settings	
T.38 <input checked="" type="radio"/>	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input checked="" type="checkbox"/> Enable High Quality
T.30 <input type="radio"/>	FAX Codec: G.726 32kbps FAX Jitter Buffer [60 - 1200 ms]: 360
FAX Tone Detection Sensitivity	(less) <input type="radio"/> 1 <input checked="" type="radio"/> 2 <input type="radio"/> 3 <input type="radio"/> 4 <input type="radio"/> 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: The system sends the same FAX frame twice to get a high quality of the FAX. It requires more bandwidth.
- Enable Secure T.38: This allows gateways to send faxes on both sides.
- 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 fax function will consume more network resources and will affect transmission quality.
- FAX Detect sensitivity : Used to adjust the sensitivity of detection as to whether a phone call is a 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 ▼

- Default Call Route: The default call route can be Auto, VoIP and Deny.
Auto (VoIP first): The call route is VoIP first, and the next is Deny.
VoIP: The call route is VoIP 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	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="10"/>	Auto (VoIP first) ▼
2	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="10"/>	Auto (VoIP first) ▼
3	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="10"/>	Auto (VoIP first) ▼

- Enable: Enable detection of this entry.
- Leading Digits: The leading digits for the gateway to scan while user is dialing.
- Total Digit Count: Total number of digits that the gateway should accept. 0 is that the gateway scans leading digits only and disregards total digit count.
- Route: The interface calls should go through if above conditions satisfied.

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 the 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

It 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.

<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: 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.

<input type="checkbox"/> Enable ACL for Management		
Filter IP address	Subnet mask	Web
<input type="text"/>	<input type="text"/>	<input type="checkbox"/>
<input type="text"/>	<input type="text"/>	<input type="checkbox"/>
<input type="text"/>	<input type="text"/>	<input type="checkbox"/>

- 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 details for every phone call. The present CDR provides the call detail recording in a text file and if needed, it can be imported to prepare for an analysis report.

<input type="checkbox"/> Send record to CDR Server
CDR Server IP <input type="text"/>
Port [1 - 65535] <input type="text" value="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	<input type="text" value="English"/>
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CPT/Cadence Settings

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	<input type="text" value="350"/>	<input type="text" value="440"/>	<input type="text" value="3000"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	
Congestion Tone	<input type="text" value="480"/>	<input type="text" value="620"/>	<input type="text" value="250"/>	<input type="text" value="250"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	
Busy Tone	<input type="text" value="480"/>	<input type="text" value="620"/>	<input type="text" value="500"/>	<input type="text" value="500"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	
Ring-Back Tone	<input type="text" value="440"/>	<input type="text" value="480"/>	<input type="text" value="1000"/>	<input type="text" value="2000"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	

System Information

This page shows that the status of VoIP Gateway. There are Port Status, Server Registration Status, WAN Port Information, LAN Port Information and Hardware.

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	FXO	703	Idle	0		Disabled	
4	FXO	704	Idle	0		Disabled	
SIP Proxy Hunting Number Registration				FXS Disabled (00:00:19), FXO Disabled (00:00:19)			
Server Registration Status							
DDNS Registration				Disabled (00:00:19)			
Phone Book Manager Registration				Disabled (00:00:19)			
STUN Registration				Disabled (00:00:19)			
UPnP Negotiation				Disabled (00:00:19)			
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			
Subnet Mask				255.255.255.0			
DHCP Server							
DHCP Server				Enabled			
IP Pool Range				192.168.8.1 - 192.168.8.250			
Lease Time				1 hour(s)			
DNS				168.95.1.1			
Hardware							

- Port Status: It includes if each port registers to Proxy successfully, the lasted dialed number, how many calls each port had since the system is start, etc.
- Server Registration Status: It shows the registration status of DDNS, Phone Book Manager, STUN and UPnP.
- WAN Port Information: It shows IP address, subnet mask, default gateway and DNS server. If you use PPPoE to obtain IP, you can know if the IP is obtained through this.
- LAN Port Information: It shows LAN port IP, subnet mask, and the status of DHCP server.
- Hardware: It shows the hardware platform.

RTP Packet Summary

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

STUN Inquiry

Use STUN Inquiry to know what NAT type of the router.

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

SNMP

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

- 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

This is the time setting. After the gateway is on the Internet, it will set its watch with Time Server.

	Year	Month	Day	Hour	Minute	Second
Gateway Time	<input type="text" value="2000"/>	<input type="text" value="1"/>	<input type="text" value="1"/>	<input type="text" value="9"/>	<input type="text" value="18"/>	<input type="text" value="36"/>
Time Zone	+ <input type="text" value="8"/> : <input type="text" value="00"/>					
#	Time Server					
1	<input type="text" value="ntp.ucsd.edu"/>					
2	<input type="text" value="ntp.univ-lyon1.fr"/>					
3	<input type="text" value="time.nuri.net"/>					

- 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. (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	<input type="button" value="Backup"/>
Configuration Template File	<input type="button" value="Backup"/>

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

Restore Configurations	
<input checked="" type="radio"/> Upload Configuration File	<input type="text"/> <input type="button" value="Browse..."/>
<input type="radio"/> Restore Default Configurations	
<input type="button" value="Restore"/>	

Provision Settings

Fill in the parameters needed of Provision Server from your 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"/>

System Operations (Save Settings)

Some settings are effective by **Restart**. Remember to save all settings by **Save Settings** before to 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. 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.
Your provider gives all parameters.

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

Current Software Version No. [1.2.33.7]

Upgrade Server	<input checked="" type="radio"/> TFTP <input type="radio"/> FTP <input type="radio"/> Image Server
Software Upgrade Server IP	<input type="text"/>
Software Upgrade Server Port [1 - 65535]	<input type="text" value="69"/>
User Name	<input type="text"/>
Password	<input type="text"/>
Directory	<input type="text"/>

All calls will be DROPPED during upgrade.

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

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](#)

2. IP Sharing Functions

All Gateway series have a built-in IP sharing function. The settings and instructions at a PC end are described below:

Current Intranet only supports static IP mode, and the settings at the PC end are as follow:
Available IP address Range : 192.168.8.1 – 192.168.8.253 (default address of Gateway is 192.168.8.254)

Subnet Mask : 255.255.255.0
Default Gateway : 192.168.8.254

The above values vary with different LAN Port Settings.

Assume Gateway's LAN settings are,

IP address : 192.168.3.1

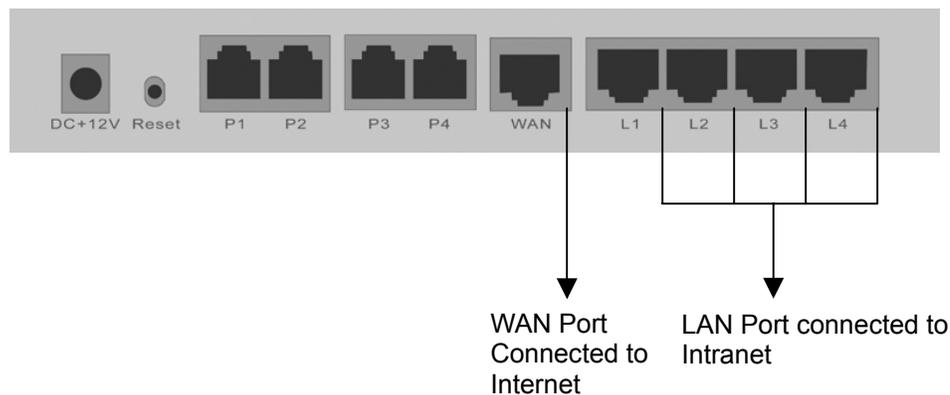
Subnet Mask : 255.255.255.0

Then, the settings at PC end should be as follows:

Valid IP address range : 192.168.3.2 – 192.168.3.254

Subnet Mask : 255.255.255.0

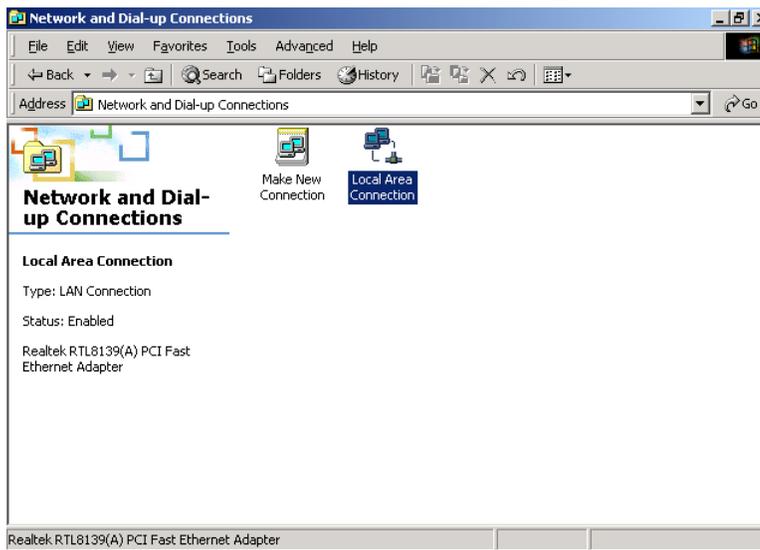
Default Gateway : 192.168.3.1



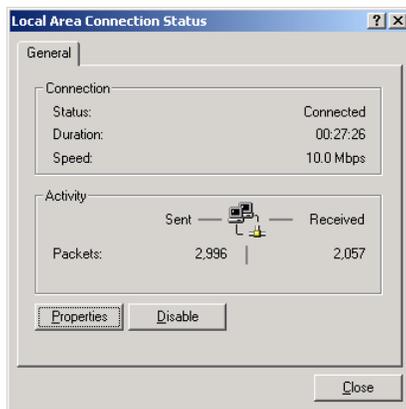
The IP settings on PC are as follows (using Windows 2000 for example)
Open Start->Settings->**Control Panel**
Open **Network and Dial-up Connection**



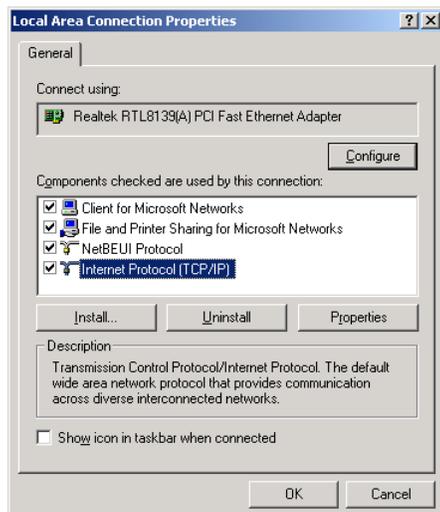
Open **Local Area Connection**



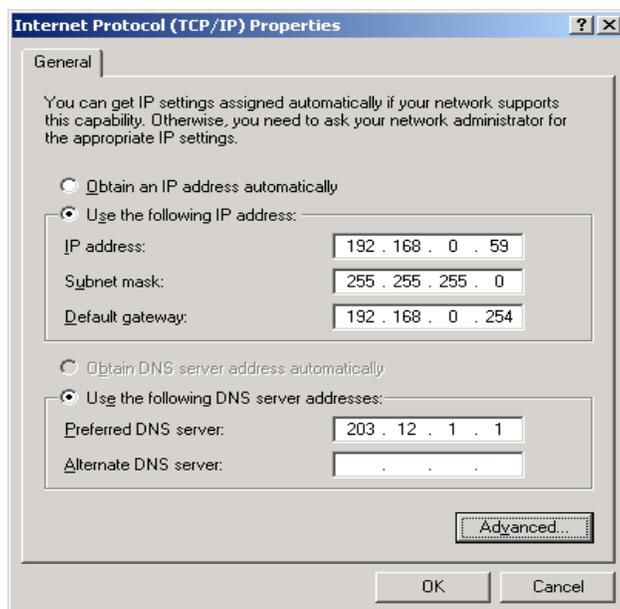
Click **Properties**



Select **TCP/IP**, and then click **Properties**.



Select **“Use the following IP Address ”** and enter IP address, Subnet Mask, and Default Gateway. Please note that an IP address in the same domain cannot be reused. Then, enter the DNS server IP address (varies in different networks. consult your ISP’s service for information). Click the **“OK”** button and after completing the settings, users can use both the VoIP and network services concurrently.



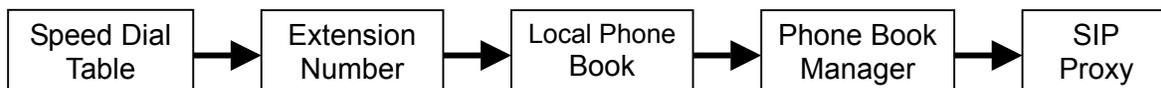
3. Coding Principle

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.

