S I P

VoIP Gateway

User's Manual

Version 1.0

(May 2011)

Contents

1. Introduction

1-1 Product Overview

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

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

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

1-2 Hardware Description

Front Pannel

16/32 FXS

16O/32 FXO

16FXS + 16 FXO

Indicators

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

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

Alarm: A blinking light indicates the VoIP Gateway is attempting to connect with the Provisioning server or VoIP Gateway can't get IP from DHCP or PPPoE Server. Once the service connects, the LED will turn off. The LED will light solid red if the self-test or boot-up fails.

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

WAN:

Green: Blinking indicates that WAN port is sending/receiving packets. As you connect WAN port to a 10 M hub, the orange LED will not be light on. If the LEDs do not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

Orange: Light on as WAN port connected to a 100M/1000M switch.

LAN:

Green: Blinking indicates that LAN port is sending/receiving packets. As you connect WAN port to a 10 M hub, the orange LED will not be light on. If the LEDs do not light up when a cable is connected, verify the cable connections and make sure your devices are powered on.

Orange: Light on as LAN port connected to a 100M/1000M switch.

Phone:

Green Blinking – FXS is alerting (ringing) for an inbound call.
Green Solid – The line is in use.
Red Solid – As users execute "Status-> FXS Line Diagnostic" and there is some error on the FXS port.

Line:

Green Blinking – VoIP Gateway detects an incoming alerting from FXO port.

Green Solid – The line is in use.

Connectors

WAN: Connect to your broadband modem using an Ethernet cable.
LAN: Connect to your Ethernet enabled computers using Ethernet cable.
Reset: Press and hold the reset button for 5 seconds, release button as Alarm is blinking. That VoIP Gateway will restore to factory default.

To restore factory default settings:

1. Press and hold the reset button for 5 seconds.
2. Release the reset button. Factory settings will be restored.

24 FXS

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Connectors

WAN: Connect to your broadband modem using an Ethernet cable.
LAN: Connect to your Ethernet enabled computers using Ethernet cabling.
Reset: Press and hold the reset button for 5 seconds., release button as Alarm is blinking. That VoIP Gateway will restore to factory default.
24 FXS Connector: Connect to phones with RJ-21 cable.

Rear Panel

FXS/FXO Connector: Connect to your phones or PBX line using attached RJ-21 cablie.
Ground: A conducting connection with the earth. Connect with the ground so as to make the earth a part of an electrical circuit using metal wire.
Power Receptor: Receptor for AC power 100V- 240V.

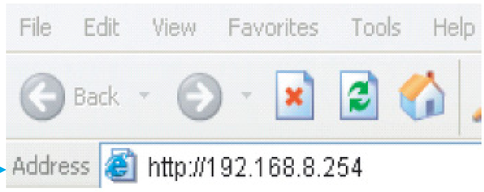
MODEL	SLOT 1	SLOT 2
32 FXS	M1700S(16 FXS)	M1600S (16 FXS)
32 FXO	M1700O (16 FXO)	M1600O (16 FXO)
16S16O	M1700S (16FXS)	M1700O (16 FXO)
	M1700SO (8FXS + 8FXO)	M1700SO (8FXS + 8FXO)

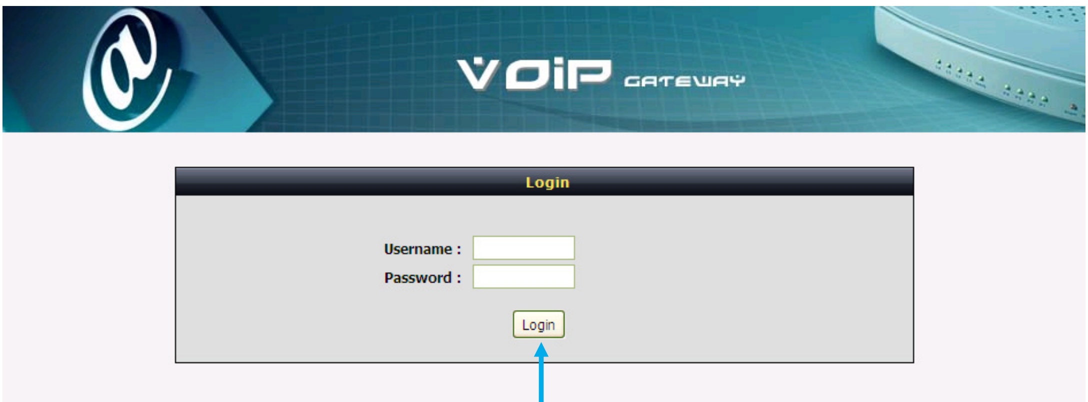
WARNING: DO NOT (1) connect the phone ports to each other (FXS to FXS) or (2) connect any phone port directly to a PSTN line (FXS to PSTN) or to an internal PBX line (FXS to PBX extension). Doing so may damage your VoIP gateway.

2. Getting Started

To access the web-based configuration utility, open a web browser such as Internet Explorer and enter the IP address of the VoIP Gateway. Default IP address of LAN port is 192.168.8.254 and default settings for **Login ID** and **Password** are blank (i.e., no login ID, no password).)

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





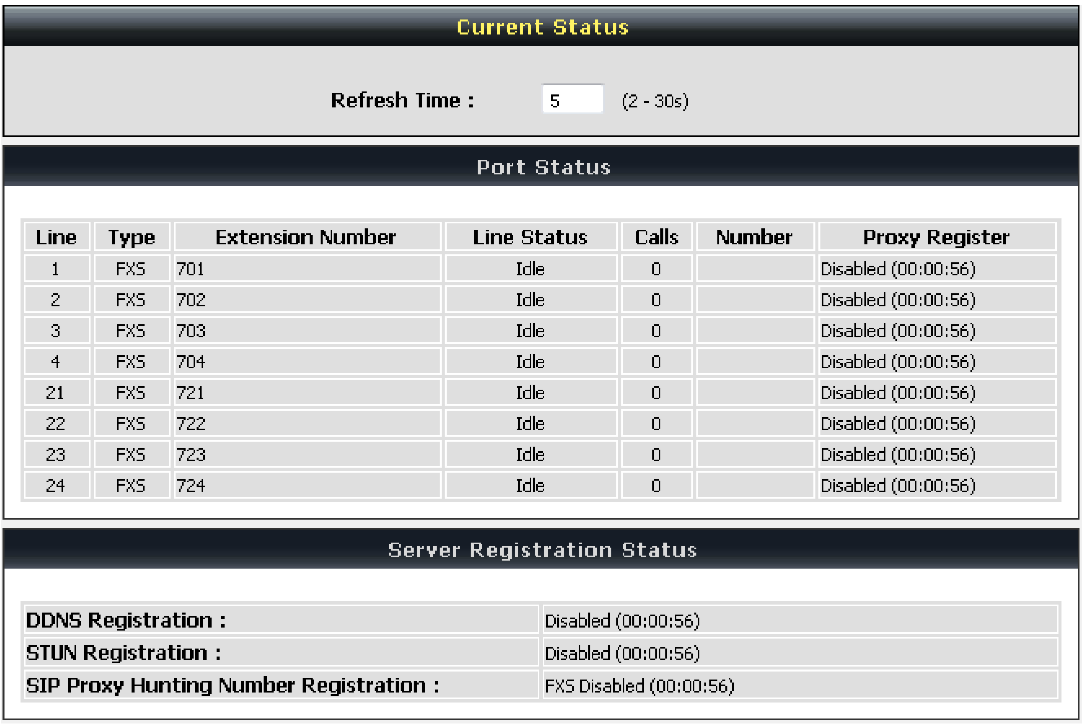
Click **Login** to enter Web Site.

3. VoIP Gateway Web Configuration

3-1 Status

3-1-1 Current Status

Status → Current Status



Current Status

Refresh Time : 5 (2 - 30s)

Port Status

Line	Type	Extension Number	Line Status	Calls	Number	Proxy Register
1	FXS	701	Idle	0		Disabled (00:00:56)
2	FXS	702	Idle	0		Disabled (00:00:56)
3	FXS	703	Idle	0		Disabled (00:00:56)
4	FXS	704	Idle	0		Disabled (00:00:56)
21	FXS	721	Idle	0		Disabled (00:00:56)
22	FXS	722	Idle	0		Disabled (00:00:56)
23	FXS	723	Idle	0		Disabled (00:00:56)
24	FXS	724	Idle	0		Disabled (00:00:56)

Server Registration Status

DDNS Registration :	Disabled (00:00:56)
STUN Registration :	Disabled (00:00:56)
SIP Proxy Hunting Number Registration :	FXS Disabled (00:00:56)

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

Number

FXS Ports: It displays calling number for inbound call or called number for outgoing call.

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

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

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

3-1-2 RTP Packet Summary

Status → RTP Packet Summary

RTP Packet Summary

Line

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

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

3-1-3 System Information

Status → System Information

System Information

System Information

Time and Date :	1969/12/31 12:03:48
Firmware Version :	1.02.38.57

WAN Port Information

Factory Default MAC Address :	000175100209
Net Link :	Disconnected
IP address :	192.168.1.181
Subnet mask :	255.255.255.0
Default Gateway :	192.168.1.254
Domain Name Server :	168.95.1.1

LAN Port Information

MAC Address :	00017510020A
IP address :	192.168.8.254
Subnet mask :	255.255.255.0

DHCP Server

DHCP Server :	Disabled

Hardware

Hardware Platform :	IXP435
Hardware :	A1-1.0
Driver :	1.2.2.1295.69/242

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

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

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

3-1-4 Routing Table

Status → Routing Table

It displays routing table of VoIP Gateway.

Routing Table

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

3-1-5 LAN Client

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

Status → LAN Client

DHCP CLIENTS

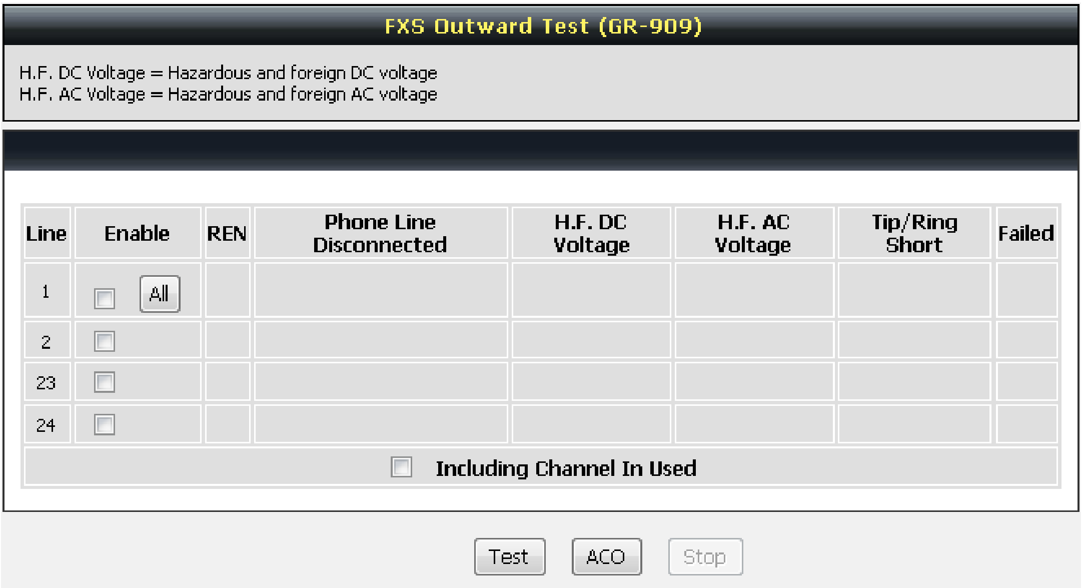
IP Address	MAC Address	Live Time
192.168.8.1	00:19:d2:35:45:60	2147448608

Refresh

3-2 FXS Line Diagnostics

3-2-1 FXS Outward Test

FXS Line Diagnostics → FXS Outward Test



This feature is available at FXS ports only. It allows operator to verify whether it is some problem on the cable between Phone Sets and FXS ports.

Enable: Select the lines you want to test.

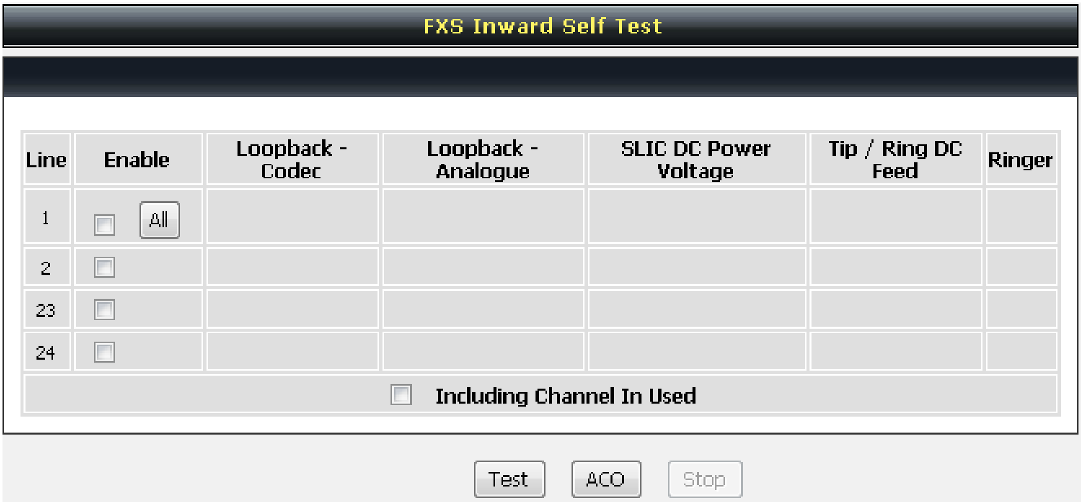
Including Channel In Used: Since the line test will interrupt a talking call, that VoIP Gateway will ignore the in used line. If you would like to test all the lines you select even it is in used, please tick this item.

Test: Click start to test.

ACO: Clear alarm indication of the last test result.

3-2-2 FXS Inward Self Test

FXS Line Diagnostics → FXS Inward Self Test



This feature is available at FXS ports only. It allows operator to verify if it is some problem on the FXS chip set.

Enable: Select the lines you want to test.

Including Channel In Used: Since the line test will interrupt a talking call, that VoIP Gateway will ignore the in used line. If you would like to test all the lines you select even it is in used, please tick this item.

Test: Click start to test.

ACO: Clear alarm indication of the last test result.

3-3 General Settings

3-3-1 WAN

WAN (Wide Area Network) Settings are used to connect to your ISP (Internet Service Provider). Please select the appropriate option for your specific ISP.

IP Configuration (Setting WAN Port)

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

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

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

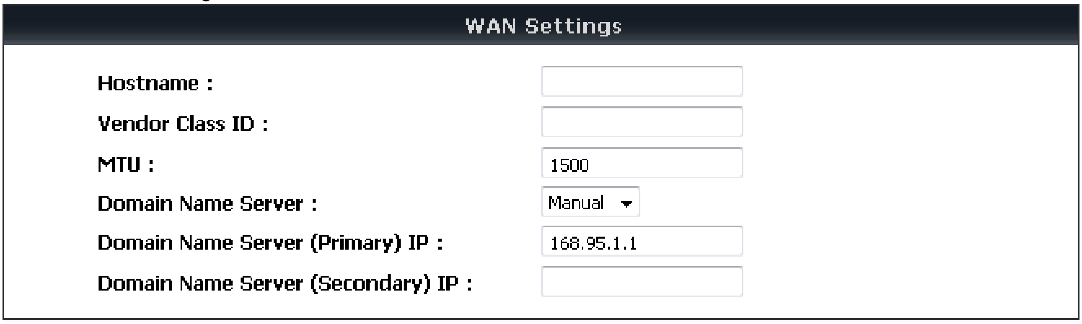
After selecting the suitable option, click **Accept** at the bottom of the screen to save the settings.

You need to save the changes and restart the VoIP Gateway to make the changes active. Saving the settings: Enter "Save/Restart" page. Tick **Save Settings** and **Restart** then click **Accept**. Wait for about 50 seconds before the VoIP Gateway obtaining an IP address by the method you selected.

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

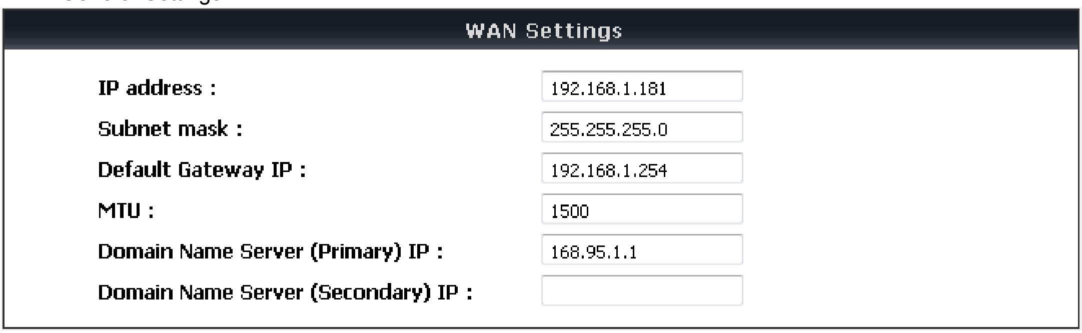
General Settings → WAN

General Settings → WAN



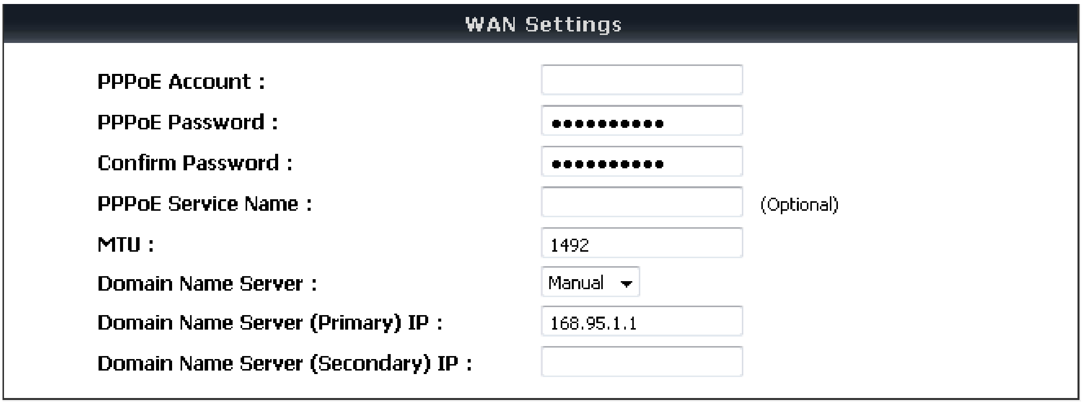
DHCP: Select this option if your ISP (Internet Service Provider) provides you an IP address automatically. Cable modem providers typically use dynamic assignment of IP Address. The Host Name field is optional but may be required by some Internet Service Providers.

General Settings → WAN



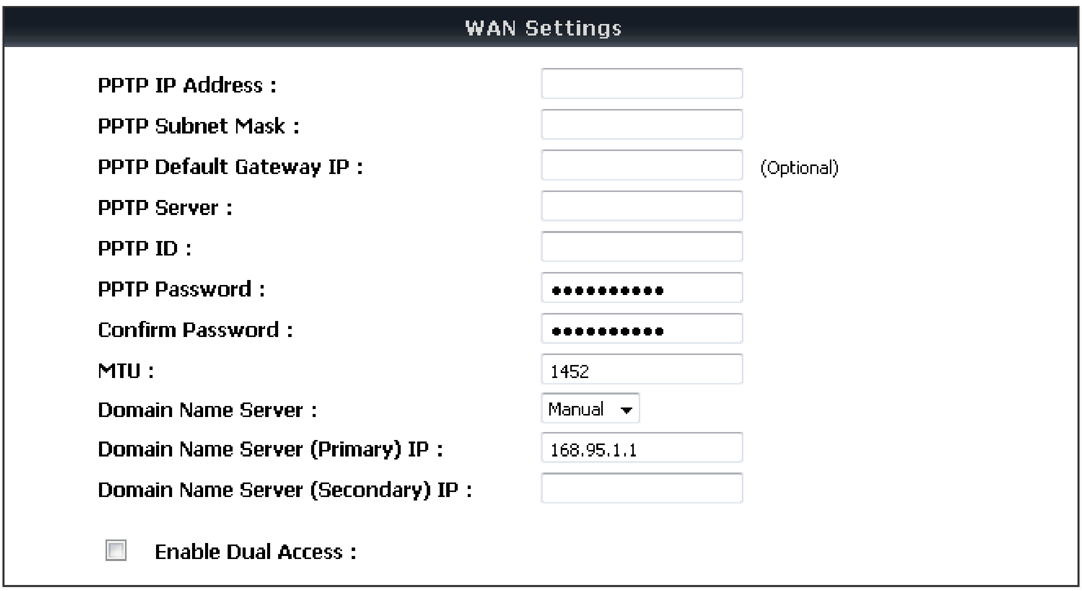
Static IP: Select this option if your ISP (Internet Service Provider) provides you a Static IP address. Enter the **IP address**, **Subnet Mask** and **Default Gateway IP**.

General Settings → WAN



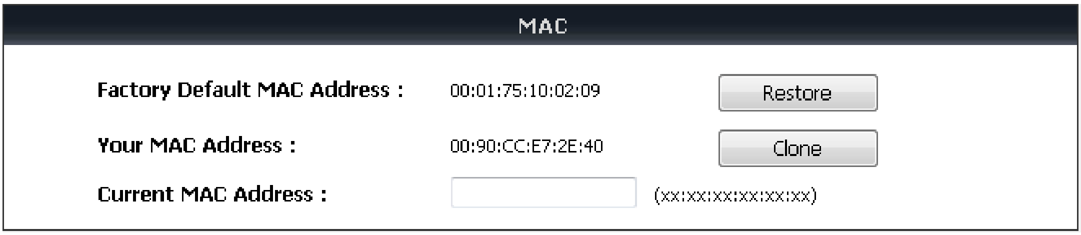
PPPoE: Select this option if your ISP requires you to use a PPPoE (Point-to-Point Protocol over Ethernet) connection. Enter the **PPPoE Account**, **PPPoE Password** and re-enter Password to confirm.

General Settings → WAN



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

General Settings → WAN



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

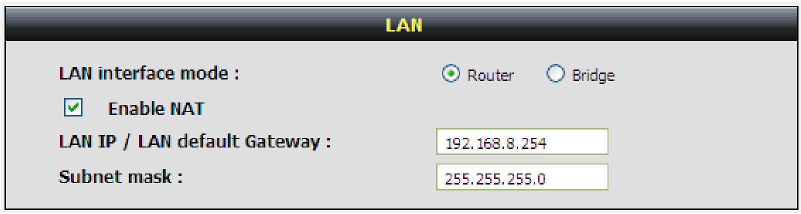
Your MAC Address: It is left blank as you log-in via the WAN port.

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

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

3-3-2 LAN

General Settings → LAN

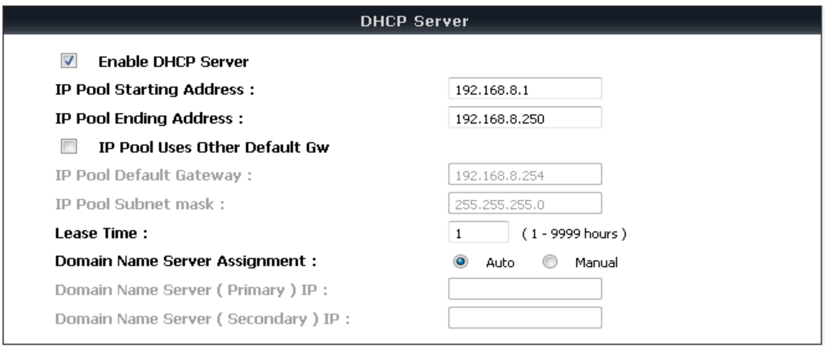


Enable NAT: Tick it for NAT mode otherwise VoIP Gateway will service as routing mode**.**

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

Subnet Make: Enter the subnet mask for DHCP clients.

General Settings → LAN



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

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

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

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

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

IP Pool Subnet mask: Enter the new subnet mask.

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

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

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

3-3-3 SIP

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

General Settings → SIP

Soft Switch Setting
☐ Enable Support of SIP Proxy Server / Soft Switch

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

General Settings → SIP

FXS Representative Number registers to Proxy:

Line

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password	FXS Group (0:Disable)
FXS Representative Number		0702332211	☑			•••••••••• ••••••••••	
1	FXS	701 auto	☐	☐		•••••••••• ••••••••••	1
2	FXS	702	☐	☐		•••••••••• ••••••••••	2
23	FXS	723	☐	☐		•••••••••• ••••••••••	23
24	FXS	724	☐	☐		•••••••••• ••••••••••	24

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

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

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

Password: Enter password and re-enter to confirm.

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

General Settings → SIP

Each line registers to Proxy independently:

Line

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password and Confirm Password	FXS Group (0:Disable)
FXS Representative Number			☐			•••••••••• ••••••••••	
1	FXS	070123451 auto	☑	☑		•••••••••• ••••••••••	1
2	FXS	70123452	☑	☑		•••••••••• ••••••••••	2
3	FXS	70123453	☑	☑		•••••••••• ••••••••••	3
4	FXS	70123454	☑	☑		•••••••••• ••••••••••	4

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

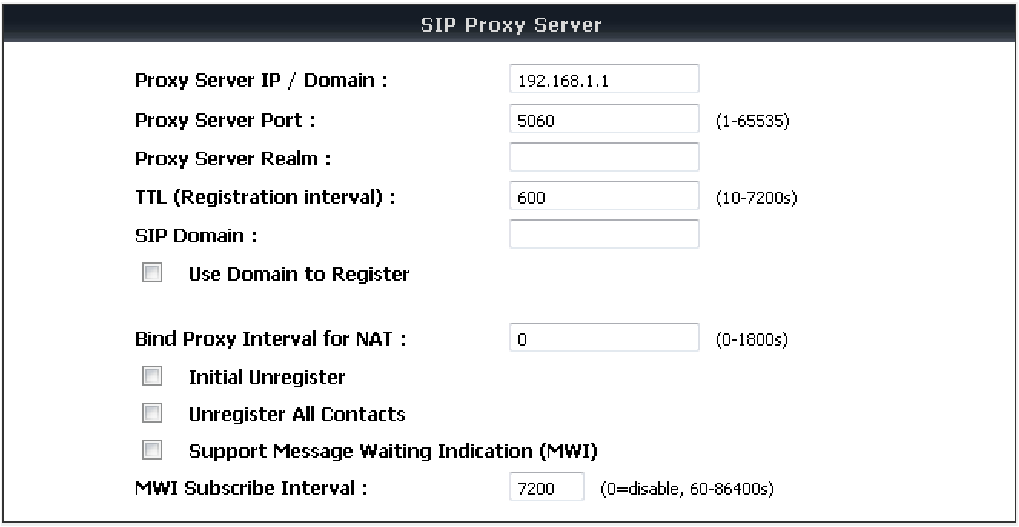
Register: Check the box to register with SIP proxy server.

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

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

Password: Enter password and re-enter to confirm.

General Settings → SIP



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

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

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

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

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

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

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

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

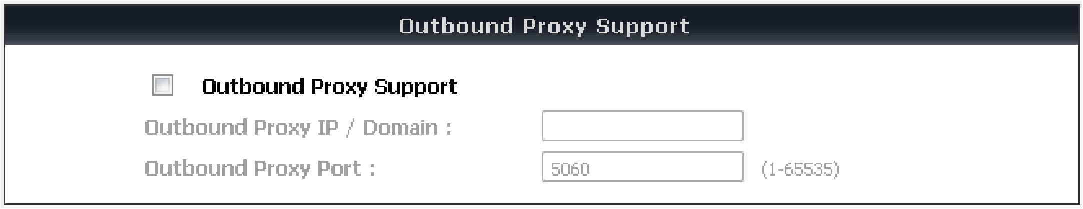
Initial Unregister: Check the box to send an unregistered message initially by the VoIP Gateway and then it will perform a general register process.

Unregister All Contacts: VoIP Gateway sends un-register request to SoftSwitch which the contact field filled with a start sign(*) to un-register all lines..

Keep SIP AUTH.: VoIP Gateway will not send a new Register Request to SoftSwitch, but send the last Register Request which contain authentication information.

Support Message Waiting Indication (MWI): It is used to enable/disable Message Waiting Indication. It is available only when Voice Mail Service is available from the VoIP Service Provider.
MWI Subscribe Interval: It is used to set the subscribe time for the VoIP Gateway to check the voice mail.

General Settings → SIP



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

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

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

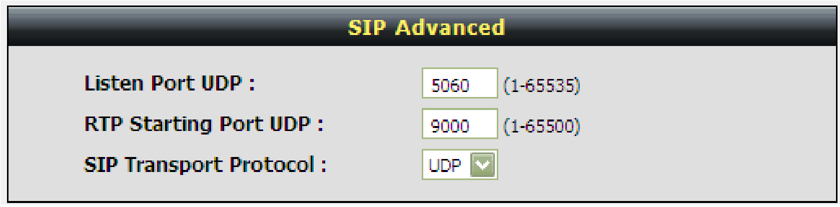
General Settings → SIP

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

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

3-3-4 SIP Advanced

General Settings → SIP Advanced



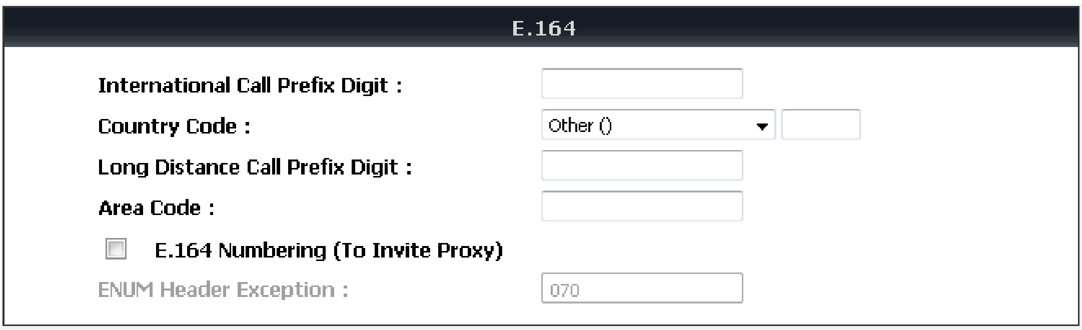
Listen Port UDP: Enter the VoIP Gateway's listening port in this field. Leave it as default settings, unless it conflicts with ports used by other device in your network.

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

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

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

General Settings → SIP Advanced



International Call Prefix Digit: Enter the International call prefix.

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

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

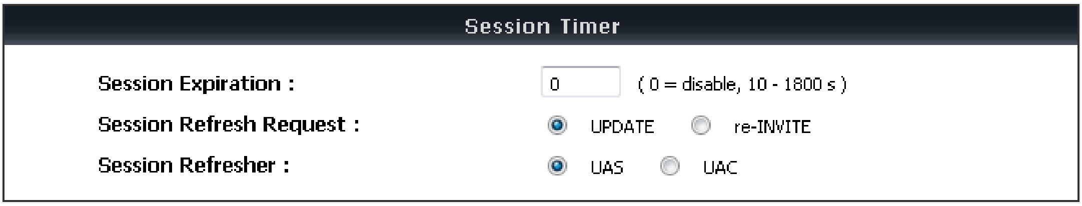
Area Code: Enter the area code.

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

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

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

General Settings → SIP Advanced

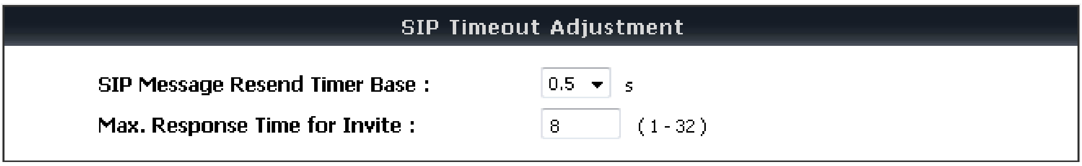


Session Expiration: This field will set the time that the VoIP Gateway will allow a SIP session to remain die (without traffic) before dropping it.

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

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

General Settings → SIP Advanced



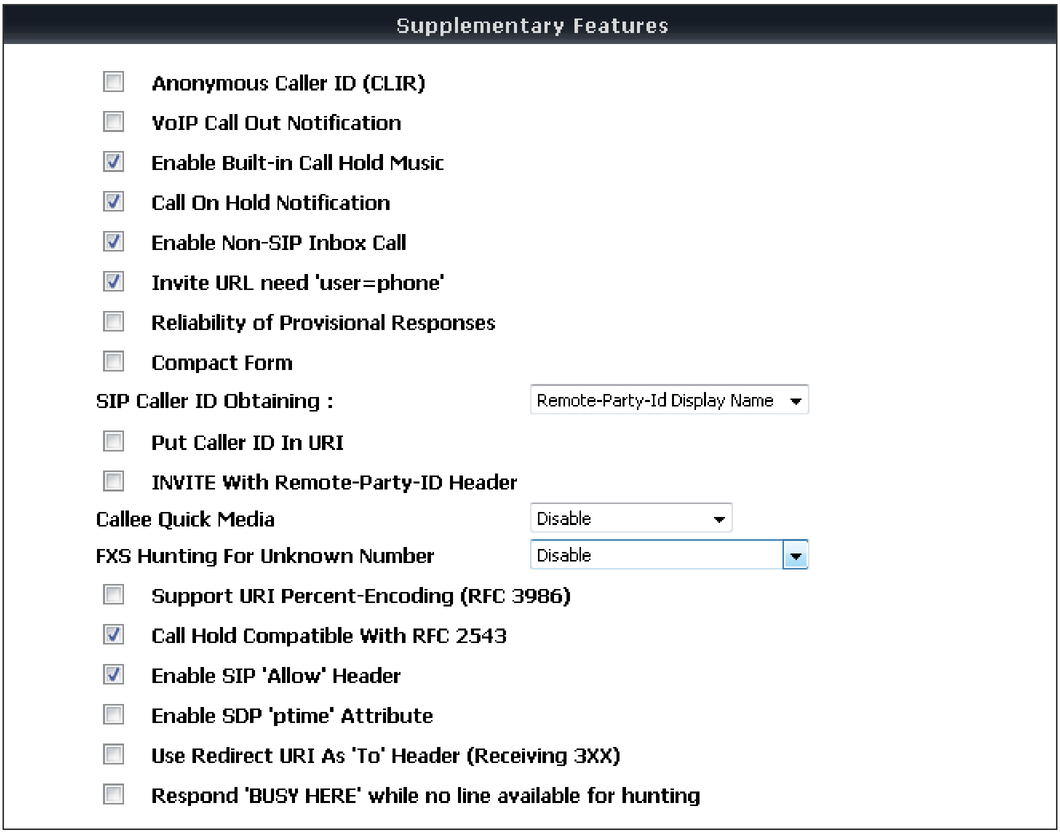
SIP Message Resend Timer Base: Select the resend timer base from the drop-down menu if response is not received within the base time. The sequence of sending is like "base time" * 2, and send again at "base time" *2 *2. The maximum resend time is four seconds. Resend action will stop when the total resend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

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

General Settings → SIP Advanced

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

General Settings → SIP Advanced



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

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

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

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

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

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

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

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

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

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

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

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

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

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

Callee Quick Media: VoIP Gateway will send RTP to remote party immediately as user answer an inbound call.

FXS Hunting For Unknown Number: Select the response for an incoming call which the called number is not exist in on this VoIP Gateway.

Disable – VoIP Gateway responses 404 not fornd.
Hunt and Transit Dial – VoIP Gateway sends alert to an available FXS port and dial the number to PBX as the FXS port picked up by PBX. It works with SoftSwitch or IPPBX to allow a remote client reach the PBX extension for one step dial. (For virtual extension)
FXS Group Hunting/ Ring Type – VoiP Gateway sends alert to an available FXS port for hunting group.

Enable SIP 'rport'(RFC 3581): Tick this option for VoIP Gateway to take 'rport' in SIP message body.

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

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

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

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

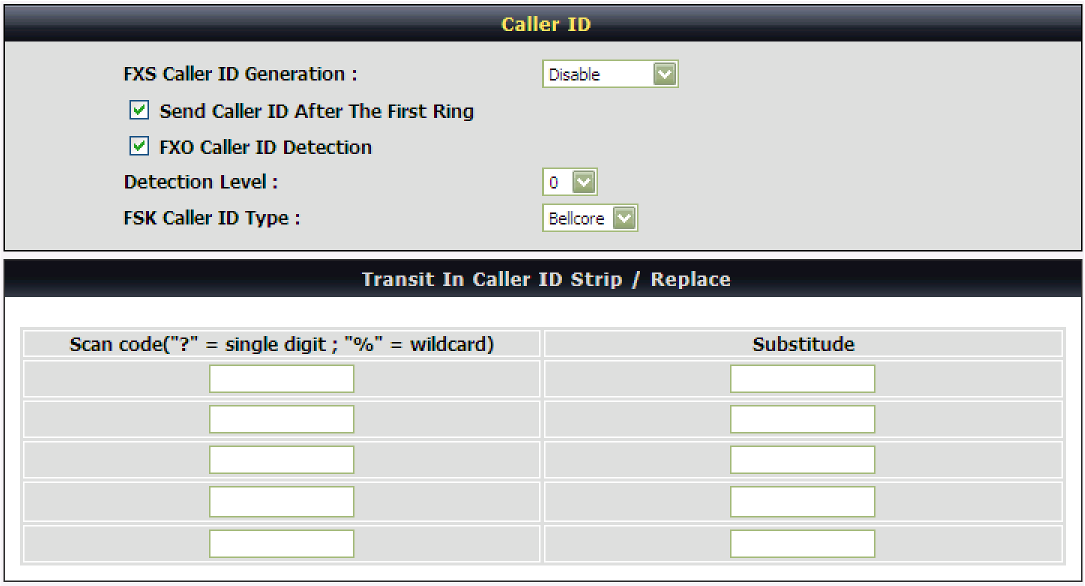
Enable SDP 'ptime' Attribute: It is used to put "ptime" in SDP packets when ticked.

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

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

3-3-5 Caller ID

General Settings → Caller ID



FXS Caller ID Generation:

DTMF – Sending Caller ID in DTMF signaling.

FSK – Sending Caller ID in FSK signaling.

FSK + TypeII – Send Caller ID in FSK signaling. As the phone set supports Call Waiting Caller ID that FXS will send third party's number.

Send Caller ID After the First Ring:

Un-Ticked – FXS sends Caller ID before the first ring. Usually it is used in DTMF mode.

Ticked – FXS sends Caller ID between the first and second ring. Usually it is used in FSK mode.

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

Detection Level: If FXO can't detect Caller ID, try to adjust it until VoIP Gateway can get caller ID.

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

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

3-3-6 Hot Line

General Settings → Hot Line

Hot Line

Line

Line	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay) [0=disable,0-60s]	Dial-Out Prefix	FXO Line Default Dial-Out	FXS Group (0:Disable)
1	☑	FXS	☐		0			1
2	☑	FXS	☐		0			2
3	☑	FXS	☐		0			3
4	☑	FXS	☐		0			4
29	☑	FXO	☐		0			
30	☑	FXO	☐		0			
31	☑	FXO	☐		0			
32	☑	FXO	☐		0			

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

Hot Line:

> **FXS:** Check to direct the call automatically to a pre-configured destination without any action when the FXS is off-hook. (ie. as the user picks up the phone). When the FXS is under Hot Line mode, no other phone numbers can be dialed.
>
> **FXO:** Check to direct the call automatically to a pre-configured destination without any action when there is incoming call from PBX line to FXO port(Calls from PBX Line to FXO then direct forward to an assigned VoIP number)

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

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

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

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

FXS Group: It is alerting sequence of incoming call for FXS represent. Set it to be "0" to disable hunt group for a FXS port.

3-3-7 Line settings

General Settings→ Line settings

Line

Line	Type	Listening Volume (3dB per step)	Speaking Volume (3dB per step)	Tone Volume	FXS Current (18-48mA)
1	FXS	0 ▼ [All]	0 ▼ [All]	5 ▼ [All]	0 [All]
2	FXS	0 ▼	0 ▼	5 ▼	0
31	FXO	0 ▼	0 ▼	5 ▼	
32	FXO	0 ▼	0 ▼	5 ▼	

Line	Type	Flash Time [50-900ms]		Polarity Reversal	PSTN Answer Detection	PSTN Ring OFF Length [1000-20000ms]	FXS Chip Option 1
1	FXS	90 [All]	600 [All]	☐ [All]			☑ [All]
2	FXS	90	600	☐			☑
31	FXO		600	☐	Disable ▼		
32	FXO		600	☐	Disable ▼		

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

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

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

FXS Current: Set the output D.C. current of FXS port.

Flash Time:

FXS: Enter the minimum and maximum flash time for FXS detecting. When the duration of flash signal generated by the phone is in the range, FXS will detect as a hook flash event. If the duration is shorter than the low value, FXS will ignore it. If the duration is longer than the high value, FXS will detect an FXS hang up event.

FXO: Enter the flash time for VoIP Gateway generator hook-flash at FXO ports.

Polarity Reversal:

FXS: It is used to activate the generation of polarity reversal at FXS as VoIP remote party answer or terminate a call.

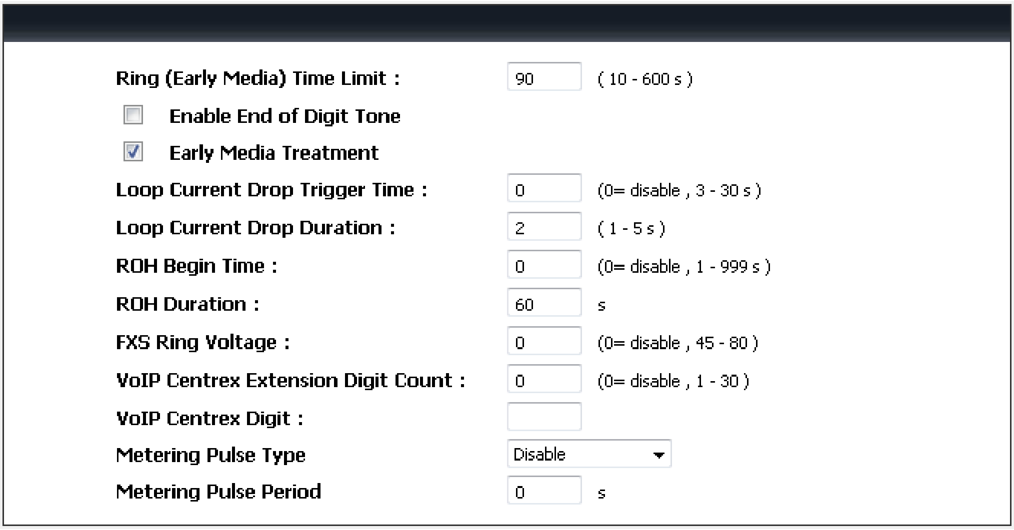
FXO: It is used to activate the detection of polarity reversal from FXO. As FXO port detect polarity reversal, VoIP Gateway will send "Bye" to VoIP remote party to release a call.

Note: The line which connect to FXO port must support Polarity Reversal Generate, or FXO port can't detect polarity reversal event.

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

FXS Chip Option 1: Check the box to avoid mis-detecting the loop state of a subscriber line or PBX user loop from FXS interface. In some cases, the off-hook voltage might cause the FXS interface mis-detect the idle and the active state, in order to avoid this situation, un-check this feature.

General Settings→ Line settings



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

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

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

Force Calling Thru PSTN Code: It is the number that switches the route to FXO port (i.e. if you dial the number first, all calls will go through FXO port to PSTN).

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

Loop Current Drop Trigger Time: Enter the time to avoid the line being engaged when FXS port is connected to PBX. It stops the loop current from FXS port when FXS port is playing busy tone. The setting "0" zero is to disable this function.

Loop Current Drop Duration: Enter the drop duration for loop current.

ROH Begin Time: As users forget hang up phone set it makes FXS play loud Howler Tone to notify users put hand set correctly. If this timer is set to be 20 seconds, that FXS play busy tone for 20 seconds then play ROH.

ROH Duration: It is the maximum time for FXS play ROH, then FXS will stop play ROH and keep silence.

FXS Ring Voltage: It is to set the Ring Voltage of FXS.

FXS Onhook Voltage: It is to set idle voltage of FXS.

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

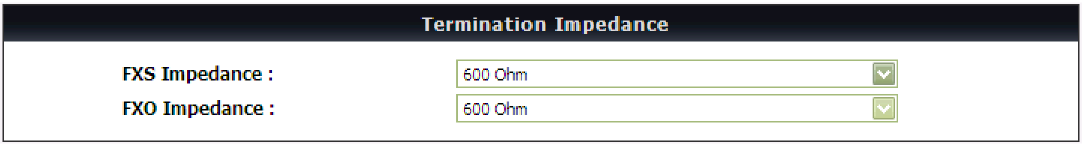
Detect FXO Line Presence: Tick the check box to detect the line presence that FXO port is connected to PBX or a PSTN line. Untick the check box to disable this function if it mis-detects line presence on FXO port while ringing.

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

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

Metering Pulse Type/ Metering Pulse Period: It is used for telephony device which connected to FXS port for billing purpose. **VoIP Gateway provide 14k Hz and 16k Hz metering capacity. The fully support for detail Metering Pulse Period is not free charge, please contact with your vendor.**

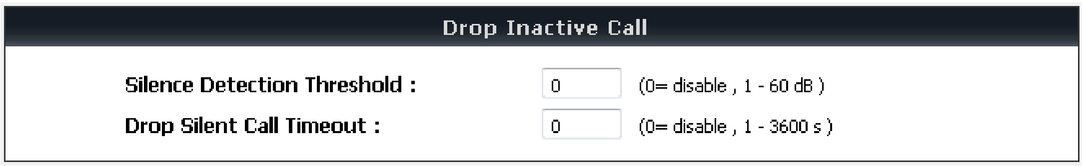
General Settings→ Line settings



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

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

General Settings→ Line settings



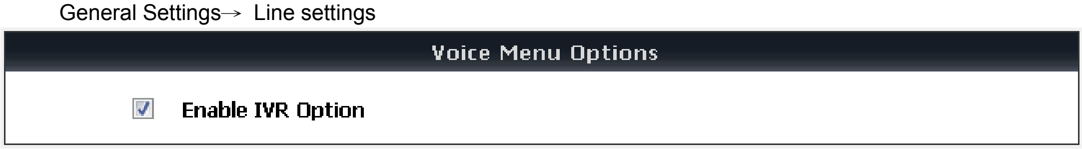
This feature is a call drop standard for a VoIP Gateway to determine whether or not to hang up the phone. The VoIP Gateway will disconnect the call automatically to avoid keeping the line engaged if the detected volume is below the **Silence Detection Threshold** or the time exceeds the **Drop Silent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

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

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

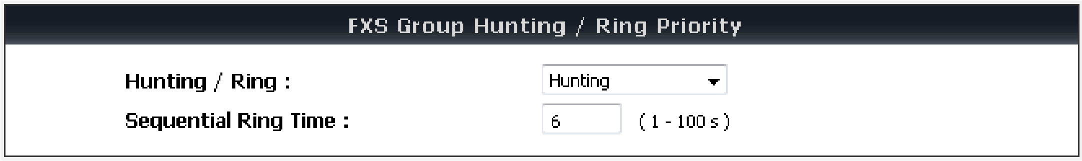
Note: Improper values for above settings might cause unexpected automatic disconnection of a call. Default values are recommended.

General Settings→ Line settings



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

General Settings → Line settings

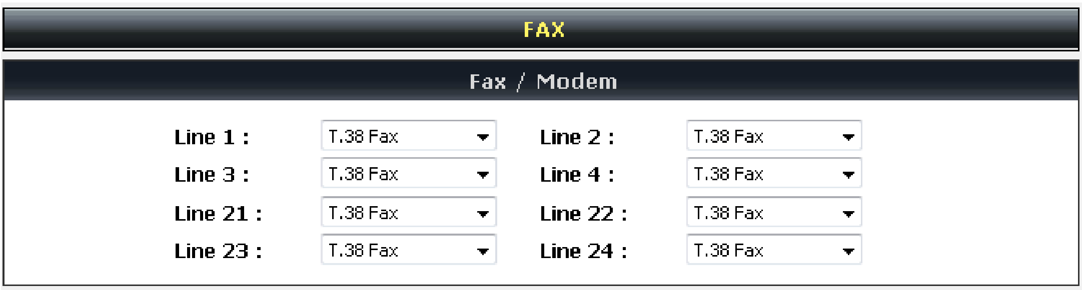


Hunting/Ring: It is used to set FXS group hunting mode. There are **Hunting**, **Simultaneous Ring** and **Sequential Ring**.

Hunting: When someone calls in by dialing FXS representative number, the system will alaways assign the call to the first line according the Ring Priority. You can use up and down arrows to adjust the hunting priority.
Simultaneous Ring: When someone calls in by dialing FXS representative number, all FXS ports will ring at the same time.
Sequential Ring: When someone calls in by dialing FXS representative number, the system will assign the call to each FXS ports in order according **Sequential Ring Time**. You can adjust **Sequential Ring Time** for the ring time of each port.

3-3-8 FAX

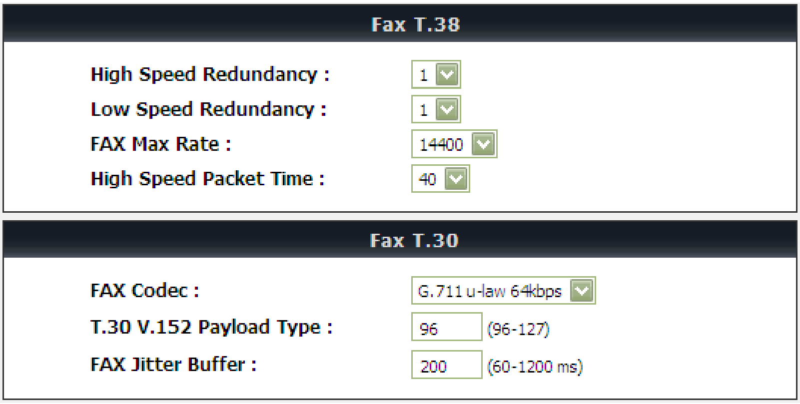
General Settings → FAX



Disable - Select it if you are not sending fax, but it is still accepted fax by the VoIP Gateway.
T.38 Fax - Select it if you are using T.38 as the protocol for fax transmission. T.38 is used for reliable and efficient facsimile transmission over network. It transmits and receives FAX waveform (relaying) over the codec negotiated during call setup this bandwidth consumed is lowered. T.38 protocol also supports redundancy to get better FAX quality.
T.30 Fax - Select it if you are using T.30 as the protocol for fax transmission. It transmit FAX signal as voice thus uncompressed G.711 would be the choice. (G.726 also works but not recommended). Due to this nature, T.30 always requires a SDP change (change of codec within a session, SIP Re-Invite required) after FAX tone detected by the callee. It will consume more network resources and will affect transmission quality. The VoIP Gateway is still able to change the protocol from T.38 to T.30 if the called party uses T.38 for fax transmission.
T.30 Fax/Modem - Select it if you use it as the protocol for transmission of fax/modem over IP network.
T.30 Only - Select it if you are using G.711 a-law or G.711 u-law for fax transmission. The VoIP Gateway won't accept T.38 for fax transmission.
T.38 Native - Select it if you are only using T.38 for fax transmission.
T.30 V.152 – As GW detects FAX tone, it will change RTP codec to be T.30 codec directly without sending Re-Invite to change codec.
T.30 V.25 – VoIP Gateway notify/receive FAX code change via RFC 2833 protocol.
Note: RFC 2833 must be enabled to provide this feature.

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

General Settings → FAX



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

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

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

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

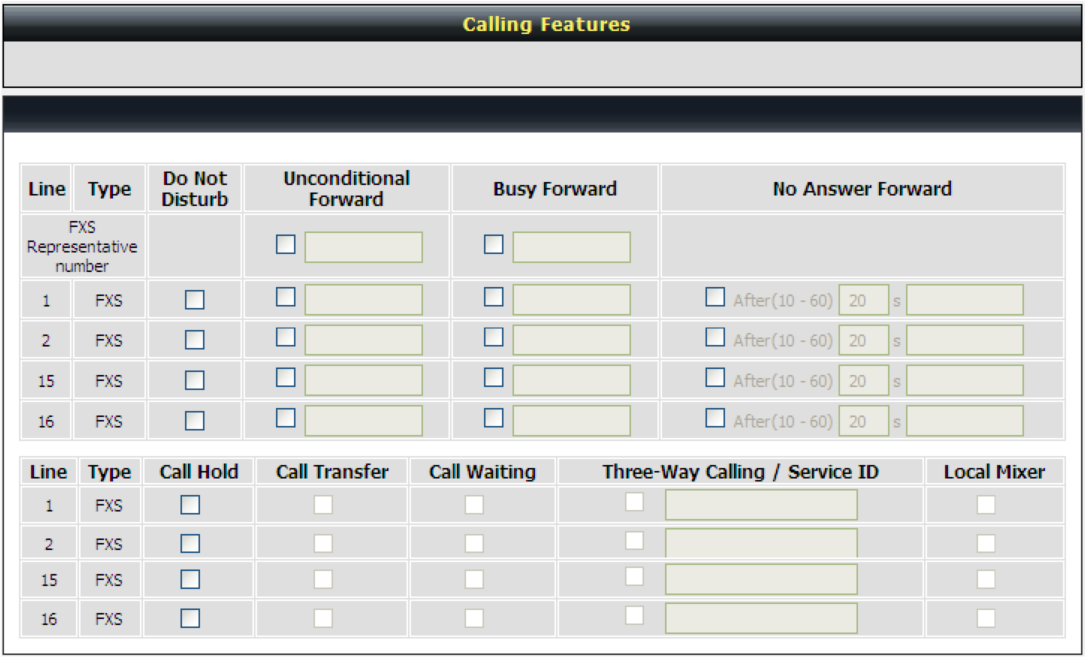
FAX Jitter Buffer: Enter the buffer or jitter when receiving packets.

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

Function	Fax Detection	Content of SDP of re-INVITE	Receive re-INVITE with T.38
Disable	No	N/A	Accept and change RTP to T.38
T.38 Fax	Yes	re-INVITE with T.38 and T.30	Accept and change RTP to T.38
T.30 Fax	Yes	re-INVITE with T.30	Accept and change RTP to T.38
T.30 Fax/Modem	Detect CED only	re-INVITE with T.30	Accept and change RTP to T.38
T.30 Only	No	N/A	Accept and change RTP to T.38
T.38 Native	Yes	re-INVITE with T.38	Accept and change RTP to T.38
T.30 V.152	Yes	There is no Re-Invite for T.30 Bypass mode	Accept and change RTP to T.38
T.30 V.25	Yes	N/A VoIP Gateway notifies FAX codec change via RFC 2833.	Accept and change RTP to T.38

3-3-9 Calling Features

General Settings → Calling Features



Calling Features

Line	Type	Do Not Disturb	Unconditional Forward	Busy Forward	No Answer Forward
FXS Representative number					
1	FXS				After(10 - 60) 20 s
2	FXS				After(10 - 60) 20 s
15	FXS				After(10 - 60) 20 s
16	FXS				After(10 - 60) 20 s

Line	Type	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID	Local Mixer
1	FXS					
2	FXS					
15	FXS					
16	FXS					

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

Unconditional Forward: Check the box to forward incoming calls to the assigned “Forwarding Number” automatically.

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

No Answer Forward: Check the box to forward incoming calls to the “Forward incoming Number” after ringing timeout (configurable from 10 to 60 seconds) expires.

Call Hold: Check the box to hold the call on the specific FXS port.

Note: Call Transfer or Call Waiting can only be activated when Call Hold is checked..

Call Transfer: Check the box to transfer the call to another destination.

Call Waiting: Check the box to accept incoming call while talking.

Three-Way Calling /Service ID: It is for conference all based on Nortel Soft Switch and must work with Proxy Server that supports Three-Way Calling service.

Local Mixer: It is used to setup the conference call when your Proxy Server did not support Three-Way Calling service.

Enable Call Feature Code

Call Feature Code

☑ **Enable Call Feature Code**

	Enable	Disable
Unconditional Forward (FXS Representative Number)	*78	#78
Warm Line (Hot Line Delay)		
Do Not Disturb	*74	#74
Unconditional Forward	*77	#77
Busy Forward	*76	#76
No Answer Forward	*75	#75
Call Hold	*70	#70
Call Transfer	*71	#71
Call Waiting	*72	#72
Local Mixer	*73	#73
Call Pickup	*40	
Call Back on Busy	*41	#41
Blind Transfer	*50	

Enable Call Feature Code: Check the box to enable the advanced function for Call Features, such as Call Pickup, Automatic Redial and Unattended transfer.

Calling Feature Instructions:

Call Hold: The call will be held after the FLASH button is pressed on the phone set. The VoIP Gateway will play music on hold (provided by your ITSP or VSP) to the remote end.

Call Transfer: The call will be held after FLASH button is pressed on local phone set (the VoIP Gateway plays on-hold music to the remote end). Meanwhile, the local user can dial out another number after the dial tone is heard. After the handset is on-hooked, the call originally on hold will then be transferred to the new number regardless the status of the new call. If wrong number is dialed for the new call, press the FLASH button will switch back to the call on hold. Also, if the local user doesn't hang up the phone after the new call is set up, press the FLASH button will switch between the original call and the new call. Please note that the PBX between phone sets and the VoIP Gateway must support FLASH features in order to use this function. If a phone set is connecting directly to the FXS port of the VoIP Gateway and the FLASH button does not function, please adjust the settings in "Flash Detect Time" from "Advanced Options" section.

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

Examples of establishing a Three-Way call:

1. Phone1 dials to Phone2, Phone2 answers the call.
2. Phone1 presses Flash then calls Phone3 (Phone2 is on hold) and Phone3 answers the call.
3. Phone1 presses Flash to start the conference call.
   Or
4. Phone1 dials to Phone2, Phone2 answers the call.
5. Phone1 presses Flash then calls Phone3 (Phone2 is on hold) and Phone3 answers the call.
6. Phone1 presses Flash and dial 3 to start the conference call.

Note: The availability of a Three-Way call also depends on your VoIP network. Please also check with your service provider for these services.

Local Mixer Rule:

- It will separate VoIP Gateway to 4 groups- group #1: FXS #1- #8, group #2: FXS #9- #16 ,group #3: FXS #17- #24 and group #4: FXS #25- #32
- VoIP Gateway will take a free DSP channel of a FXS port in the group member to provide mixer. And VoIP Gateway could arrange dynamic DSP channel for following calls by other FXS ports.

For example:

As FXS #1 need DSP resource to establish 3-Way call, VoIP Gateway will arrange a free DSP mixer(It may take DSP channel #8) .

And a user make call from FXS #8, VOIP GATEWAY will arrange another free DSP channel for FXS #8.

- **Total current calls of group #1 will be (8 - mixer calls). If there are 2 3-way calls, VoIP Gateway provides total 6(=8-2) current calls.**

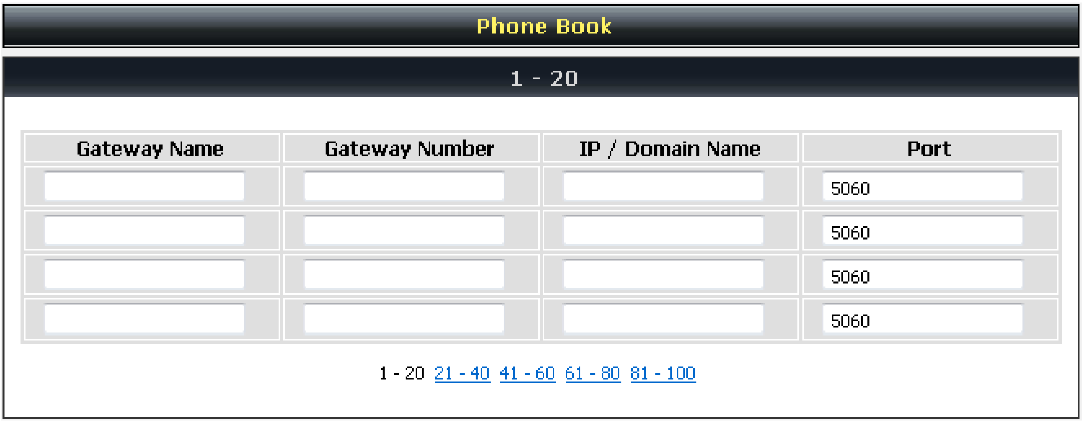
  If FXS #1- FXS #6 are talking and FXS #1- FXS #2 in 3-Way Conference mode(it takes 8 DSP channel at present), VoIP Gateway can't arrange any DSP resource for FXS #7 nor FXS # 8 at present.

  That FXS #7 and FXS # can't make a call nor receive incoming call until one of FXS 31 – FXS #6 back to idle.

3-3-10 Phone Book

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

General Settings → Phone Book



Gateway Name: Enter the alias of the remote peer.

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

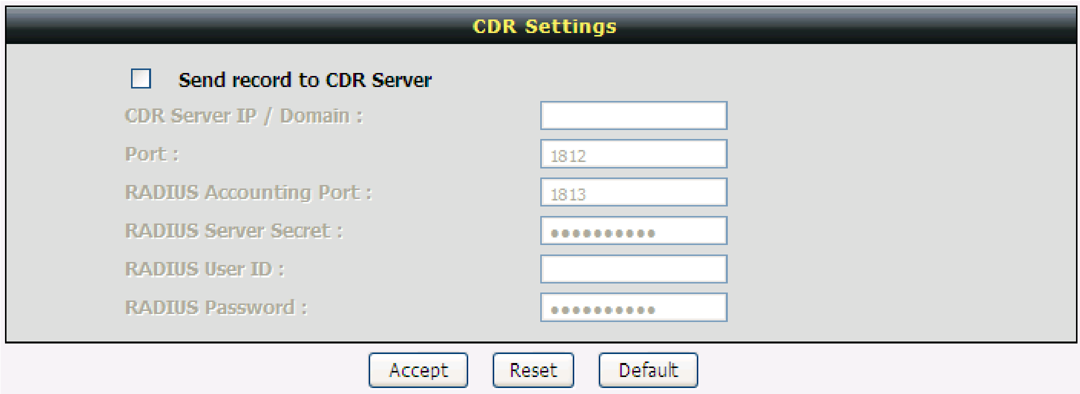
IP / Domain Name: Enter the IP address or URL (Uniform Resource Locator) of the remote peer.

Port: Enter the listen port of the remote peer.

3-3-11 CDR Settings

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

General Settings → CDR



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

3-4 Advanced Settings

3-4-1 Codec setting

Advanced Settings→ Codec settings

Codec

Jitter Buffer : 120 (60 - 1200ms)

☐ Silence Detection / Suppression

☑ Echo Cancellation

Enable	Codec	Codec Priority	Type	Packet Interval (ms)	Approximate Bandwidth Required (kbps)
☑	G.711 u-law	1		10	107.2
☑	G.723.1	5	G.723.1 6.3k	30	20.8
☑	G.726 32K	3		20	53.6
☑	G.729	4		20	29.6
☑	G.711 a-law	2		20	85.6

Jitter Buffer: Enter the jitter of receiving packets.

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

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

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

Codec Priority: The priority of code for communication.

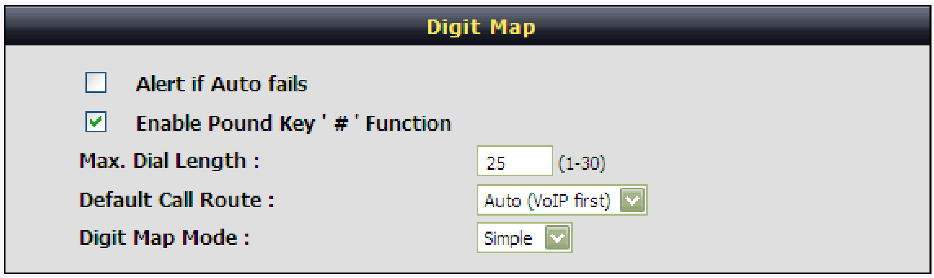
Packet Interval: Select the frame size of voice package from different codec. It defines the time interval for the VoIP Gateway to send a RTP packet or voice packet to the receiving side. The smaller the value, the greater the bandwidth takes, and larger values might cause voice delay.

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

3-4-2 Digit Map

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

Advanced Settings → Digit Map



Alert if Auto Fails: FXS play notification in case VoIP Gateway transit calls to FXO port when VoIP route is not available.

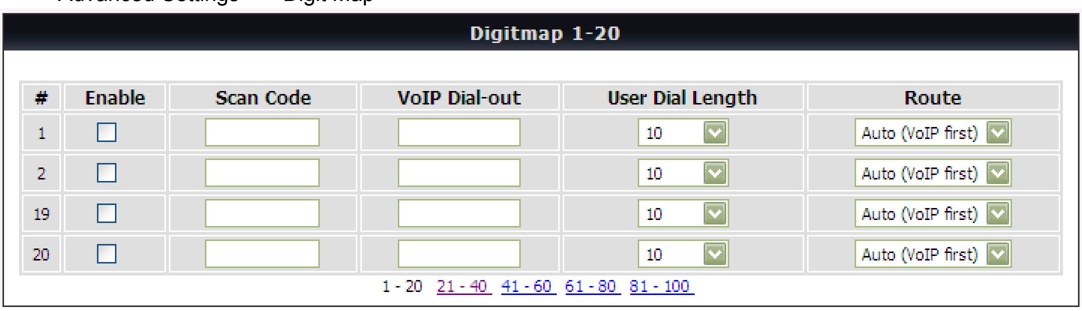
Enable Pound Key ' # ' Function: Check the box to treat ' # ' as a digit and send out with other numbers when dialing. If you un-check the box and ' # ' is pressed after dialing, it will speed up the phone number detection of the VoIP Gateway.

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

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

Digit Map Mode: Select Expert mode to configure dialing plan to follow RFC Digit Map rule.

Advanced Settings → Digit Map



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

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

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

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

Methods of Digit Map:

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

For example,
Scan Code: 09

VoIP Dial-out: 0911888997

User Dial Length: 2

Route: VoIP

VoIP Route Profile: Route # 1

Digitmap

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

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

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

For example,
Scan Code: 2???

VoIP Dial-out: 35106???

User Dial Length: 4

Route: VoIP

VoIP Route Profile: Route # 2

Digitmap

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

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

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

Digitmap

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

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

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

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

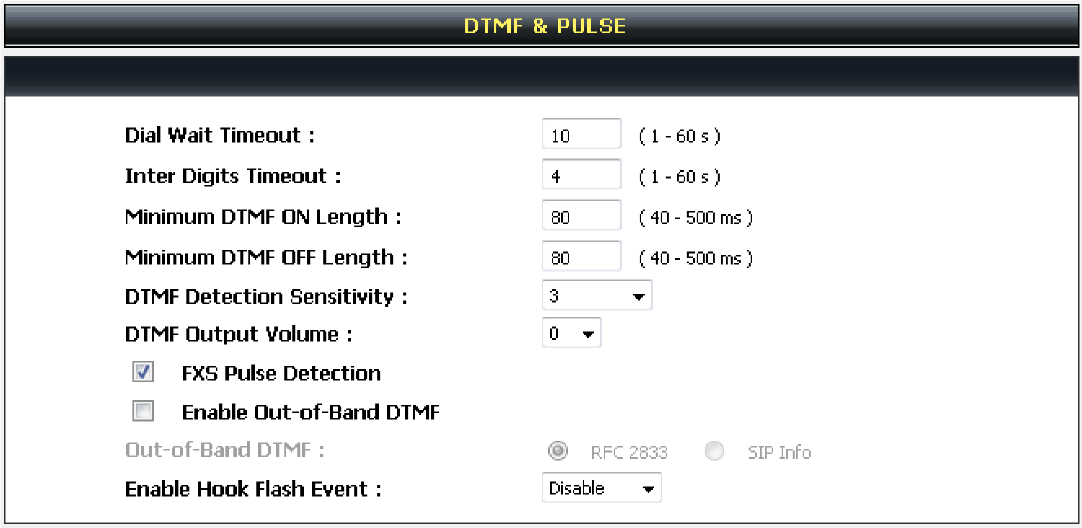
Digitmap

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

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

3-4-3 DTMF & PULSE

Advanced Settings → DTMF & PULSE



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

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

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

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

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

FXO Dial Type: It is the dialing type of FXO. **DTMF** and **Pulse** can be selected.

Pulse Dial Mark/Space Ratio: It is to set duration and break of pulse dial ration.

FXS PULSE Detect: It allows to enable/disable pulse dial detect at FXS port.

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

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

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

Payload Type: payload type of RFC2833.

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

3-4-4 CPT / Cadence

Advanced Settings → CPT / Cadence

BTC

☑ Busy Tone Cadence Measurement

	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning
BTC # 1	0	0	0	0	☑
BTC # 2	0	0	0	0	☑
BTC # 3	0	0	0	0	☑
BTC # 4	0	0	0	0	☑
BTC # 5	0	0	0	0	☑
BTC Detection Sensitivity	4				
BTC Volume Threshold	20	(20 - 70 dB)			

Busy Tone Cadence Measurement and auto learning: Provides a solution of FXO integrated with PSTN or PBX. FXO will learn the busy tone automatically.

BTC Detection Sensitivity: The more sensitivity, the more quickly the system will cut off the call. If the system often cuts off an un-finished call, select less sensitivity.

Advanced Settings → CPT / Cadence

CPT # 1 — Default

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

☑ CPT # 2 — Default

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

Default

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

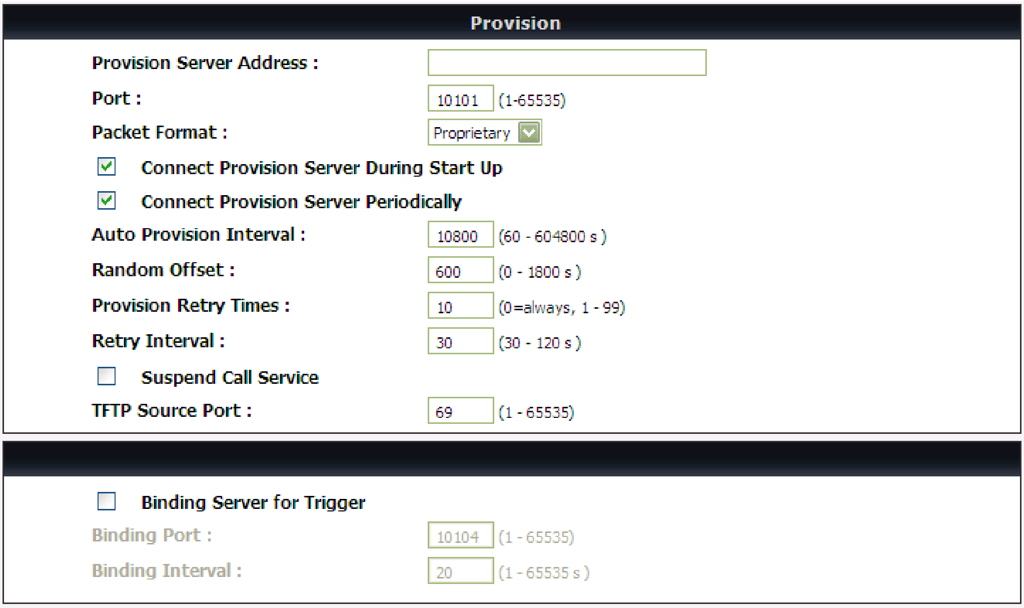
CPT # 1 Enable Setting 1: The CPT has a set of parameter table. Please adjust the CPT based on the local PSTN or PBX settings and requirements.

Advanced Settings → CPT / Cadence

FXS Ring Cadence Settings						Default
Range	ON_1 [250-8000ms]	OFF_1 [200-8000ms]	ON_2 [0,250-8000ms]	OFF_2 [0,200-8000ms]	ON_3 [0,250-8000ms]	OFF_3 [0,200-8000ms]
1	1000	2000	0	0	0	0
2	500	500	500	1500	0	0
3	500	500	500	1500	0	0

FXS Ring Cadence Settings: Specify the ring cadence for the FXS port. In this field, you specify the on and off pulses for the ring. The ring cadence that should be configured differs depending on local PSTN or PBX settings and requirements.

3-4-5 Provision Settings



Enable Auto Provisioning: It is to start provisioning if ticked.

Provision Server Address: Enter the IP address or URL of the Provisioning Server required by your provider in this field.

Prot: Enter the port number of the Provisioning Server used.

Packet Format: Select the packet transmitting format required by provision server.

Connect Provision Server During Start Up: The gateway will connect to Provision Server when it is powered on or rebooted.

Connect Provision Server Periodically: It is to adjust the parameter for the gateway to connect to provision server periodically.

Auto Provision Interval/Random Offset: Adjust the parameters for the gateway to do auto provision task.

Provision Retry Times/Retry Interval: Adjust Retry times or interval.

Suspend Service: When ticked, indicating the server has stopped providing provision and VoIP call service. Every FXO/FXS port is not able to make any call.

TFTP Source Port: Change TFTP source for firmware upgrade or configuration profile download request.

Binding Server for Trigger: It is used to trigger a connection between server and the gateway if ticked. The server will bind a port for the gateway to send provision request.

Binding Port: The binding port number of the server is used for telling the gateway the path of binding server.

Binding Interval: Set the desired Interval at which the gateway will keep the binding.

3-4-6 Caller Filter

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

Advanced Setting → Caller Filter

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

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

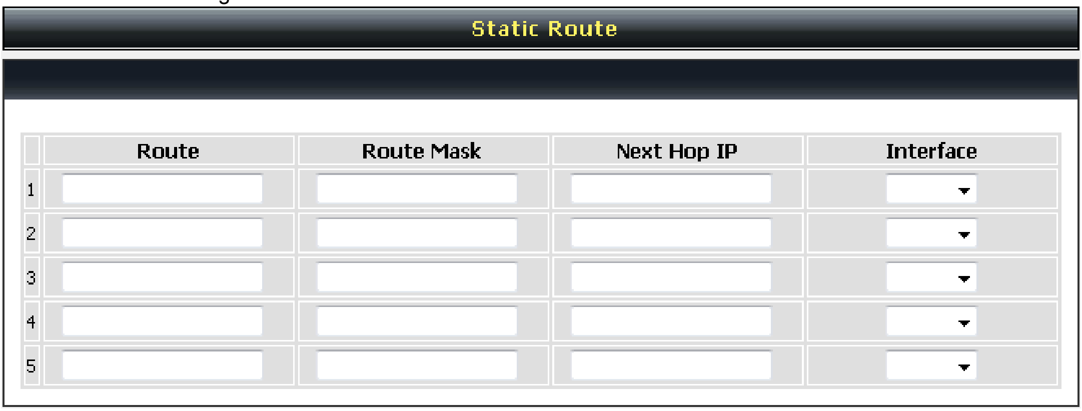
Filter IP Address: Enter the start IP address which you would like to Allow or Deny.

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

3-4-7 Static Route

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

Advanced Settings → Static Route



Route: Destination network of the route.

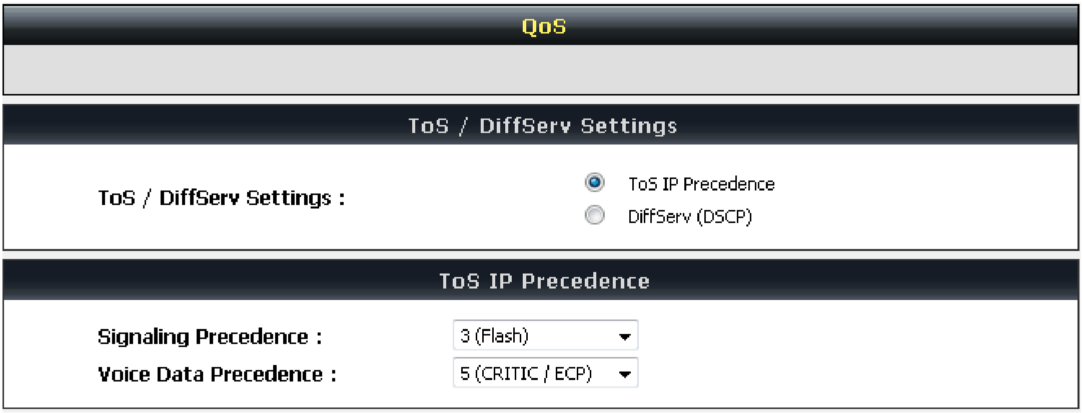
Route Mask: Subnet mask to apply on destination network.

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

Interface: The interface attached to this route.

3-4-8 QoS Settigs

Advanced Settings → QoS



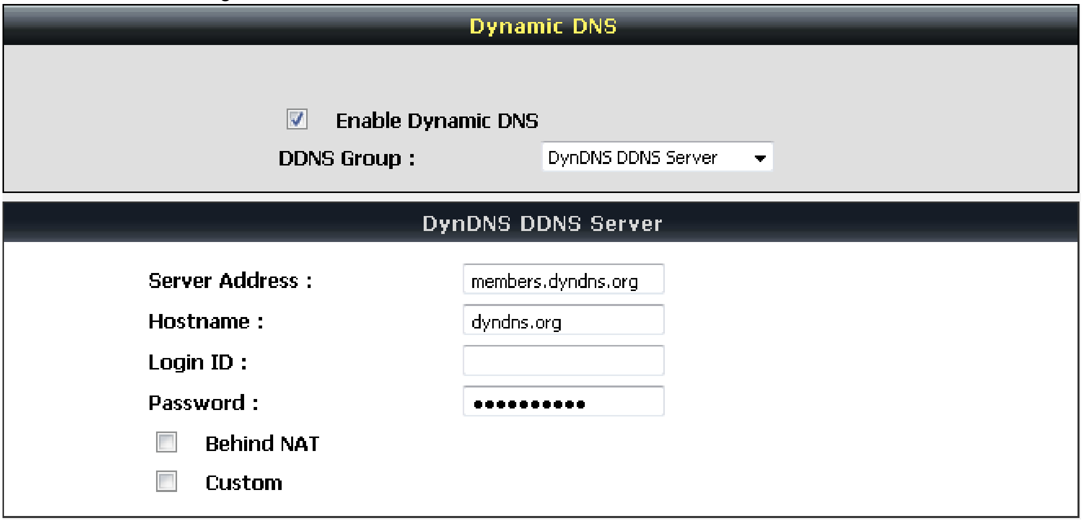
ToS IP Precedence: Select the precedence for signaling (data) and voice (voice data).

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

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

3-4-9 DDNS

Advanced Settings → DDNS



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

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

Hostname: Enter the URL of the system (or NAT) – applied from domain name registration providers (e.g. GW01.dyndns.org).

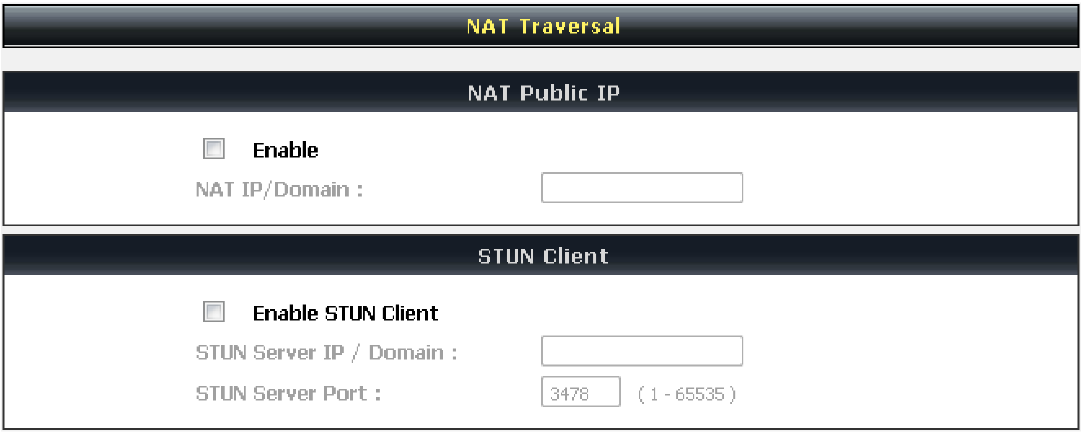
Username or Key/Password or Key: Enter the Login ID and password used to log-in to the DDNS server.

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

3-4-10 NAT Traversal

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

Advanced Settings → NAT Traversal



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

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

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

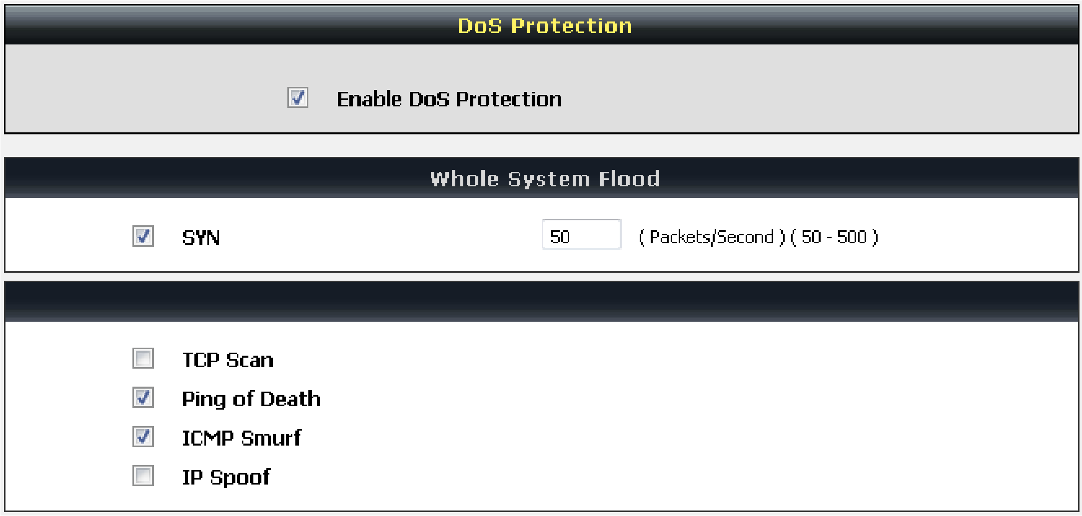
Enable STUN Client: Check the box to use the STUN protocol prevents problems from setting the IP sharing function. (Some NATs do not support this protocol.)

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

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

3-4-11 DoS Protection

Advanced Settings → DoS Protection



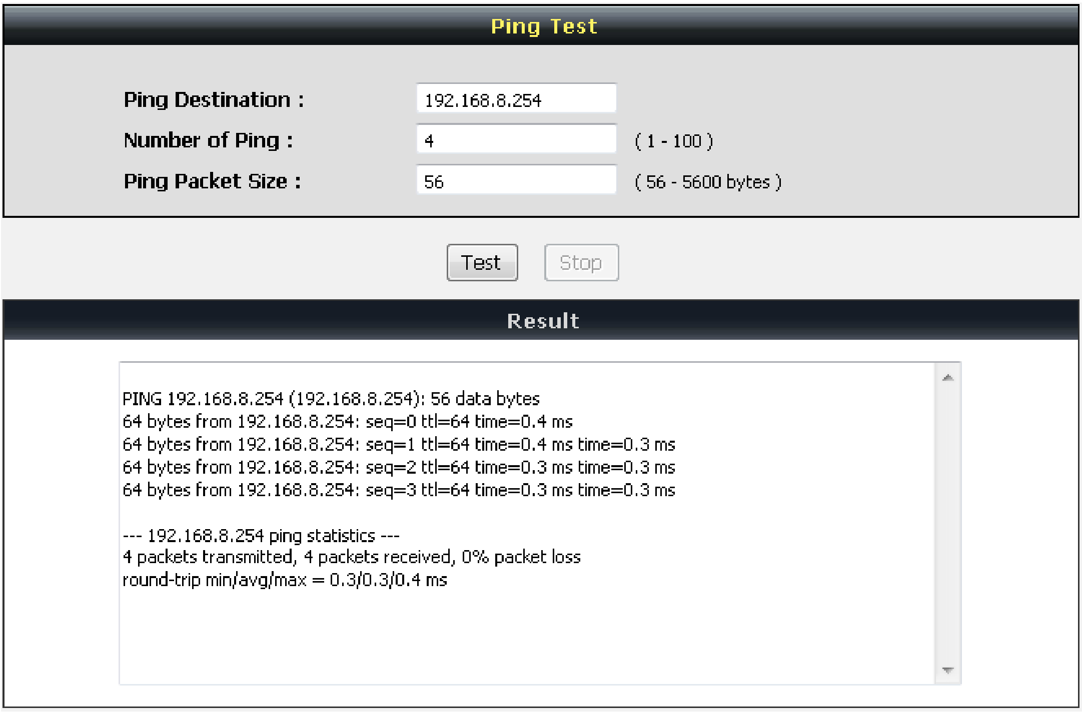
Enable DoS Prevention: Check the box to prevent DoS attacks from WAN. There are various types of DoS attacking. Leave settings in this field to the default if you are not familiar with it.

3-5 Tools

3-5-1 Ping Test

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

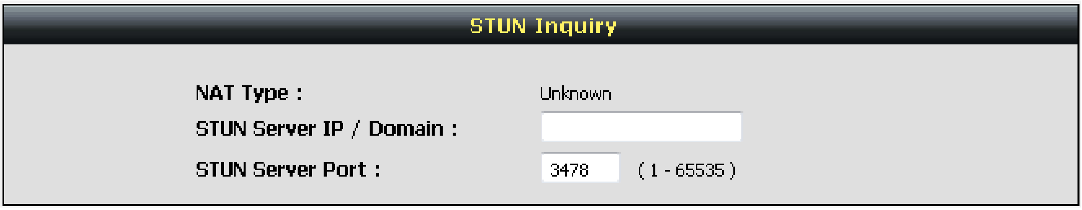
Tool → Diagnostics



3-5-2 STUN Inquiry

Use “STUN Inquiry” to detect your IP sharing device’s NAT type and communication between a STUN server and client.

Tool → STUN Inquiry



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

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

STUN Server Port: Enter the STUN Server’s listening port.

3-6 System Settings

3-6-1 NTP

System settings → NTP

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

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

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

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

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

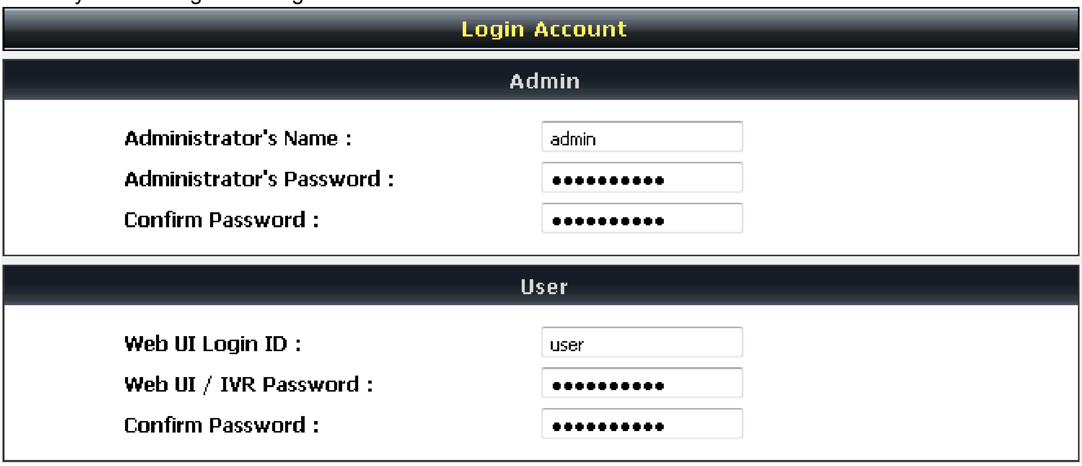
Enable Daylight Saving: To enable/disable daylight saving time.

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

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

3-6-2 Login Account

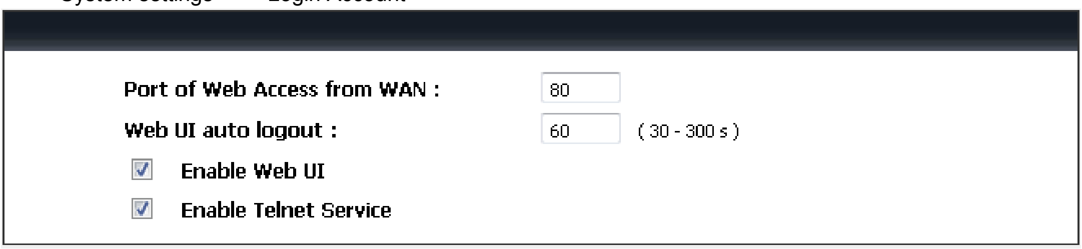
System settings → Login Account



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

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

System settings → Login Account



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

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

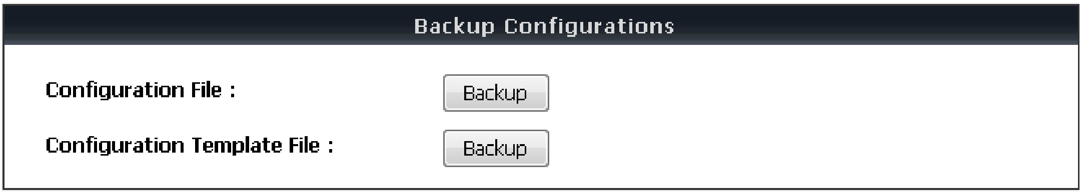
Enable Web UI: Check the box to enable WEB access from WAN or LAN.

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

3-6-3 Backup / Restore

Backup Configurations File

System settings → Backup and Restore



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

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

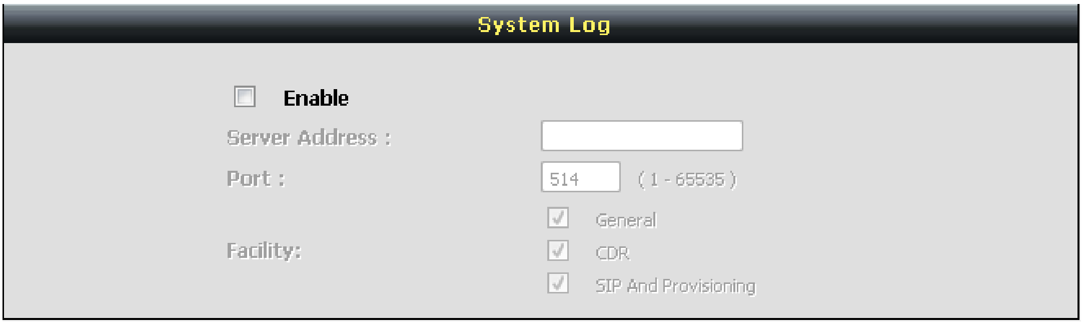
Restore Default Settings

System settings → Backup and Restore

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

3-6-4 System Log

System settings → System log



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

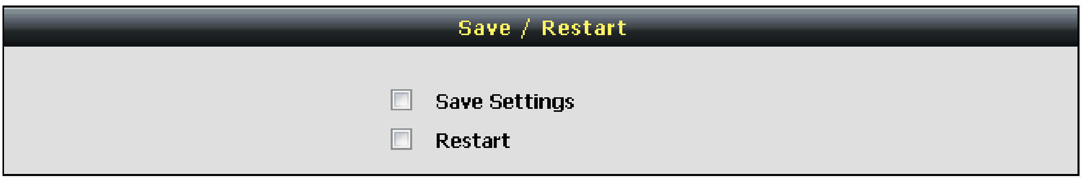
Server Address: Enter the System Log Server's IP address.

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

3-6-5 Save / Restart

Save and Reboot

System settings → Backup and Restore



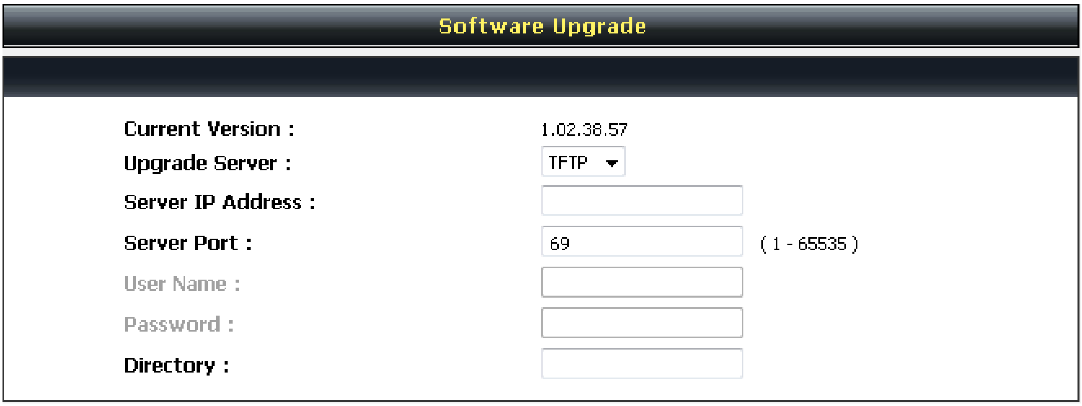
Save All Settings: Click the **Save All Settings** check box and reboot the system after completing changes. The new settings will take effect after the VoIP Gateway is restarted.

Restart: Click the **Reboot** button to reboot the system.

3-6-6 Software Upgrade

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

System settings → Software upgrade



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

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

Server Port: Enter the server's listen port.

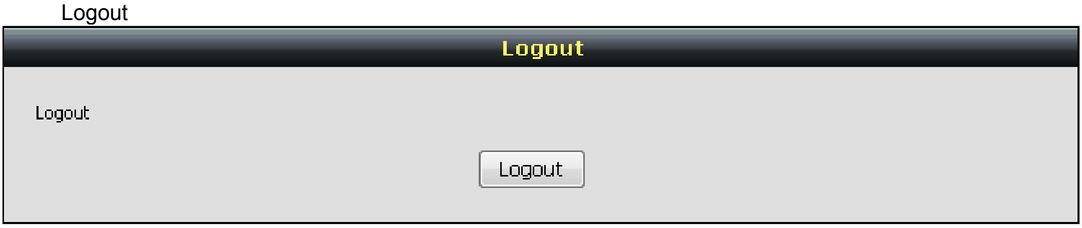
User Name/ Password: Enter the account information for accessing the server if needed.

Directory: Enter the location of the firmware file.

3-6-7 Logout

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

Logout



4. Configuring the VoIP Gateway through IVR

Preparation

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

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

4-1 IVR (Interactive Voice Response)

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

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

Instructions

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

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

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

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

Save Settings

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

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

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

4-1-1 IVR Functions Table:

Function Code	Description	Example / Notes
111/101	WAN Port IP address Set/Query	Dial function code **114** and then dial 1 for a Static IP connection then setup the IP address.
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2: DHCP, 3: PPPoE)	
118	Restart	
409	Restore factory default settings	
509	Save settings	

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

Static IP Settings

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

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

Dynamic IP (DHCP) Settings

After entering IVR mode, dial 114.

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

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

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

Save Settings and Restart

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

4-2-1 Character Conversion Table:

The table below provides a list of conversion codes. The first row (high-lighted) of each pair of the column lists the numbers, alphabets or symbols and the second row (high-lighted) of each pair of the column ("Input Key") represents the codes to be entered for the corresponding numbers, alphabets or symbols. For example, to enter "VoIP" according to the table below, enter: 32551926

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

5. Dialing Principles

5-1 Dialing Options

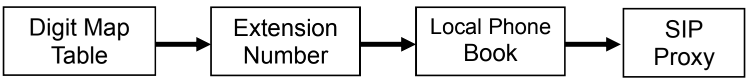
Dial the phone number which you want to call and press # to call out immediately. Note that if the "# (pound)" not dialed, the number will be called out after 4 seconds by default. The period between number dialed and call out is named "Inter Digits Timeout". (Configurable from "DTMF and PULSE", default=4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

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

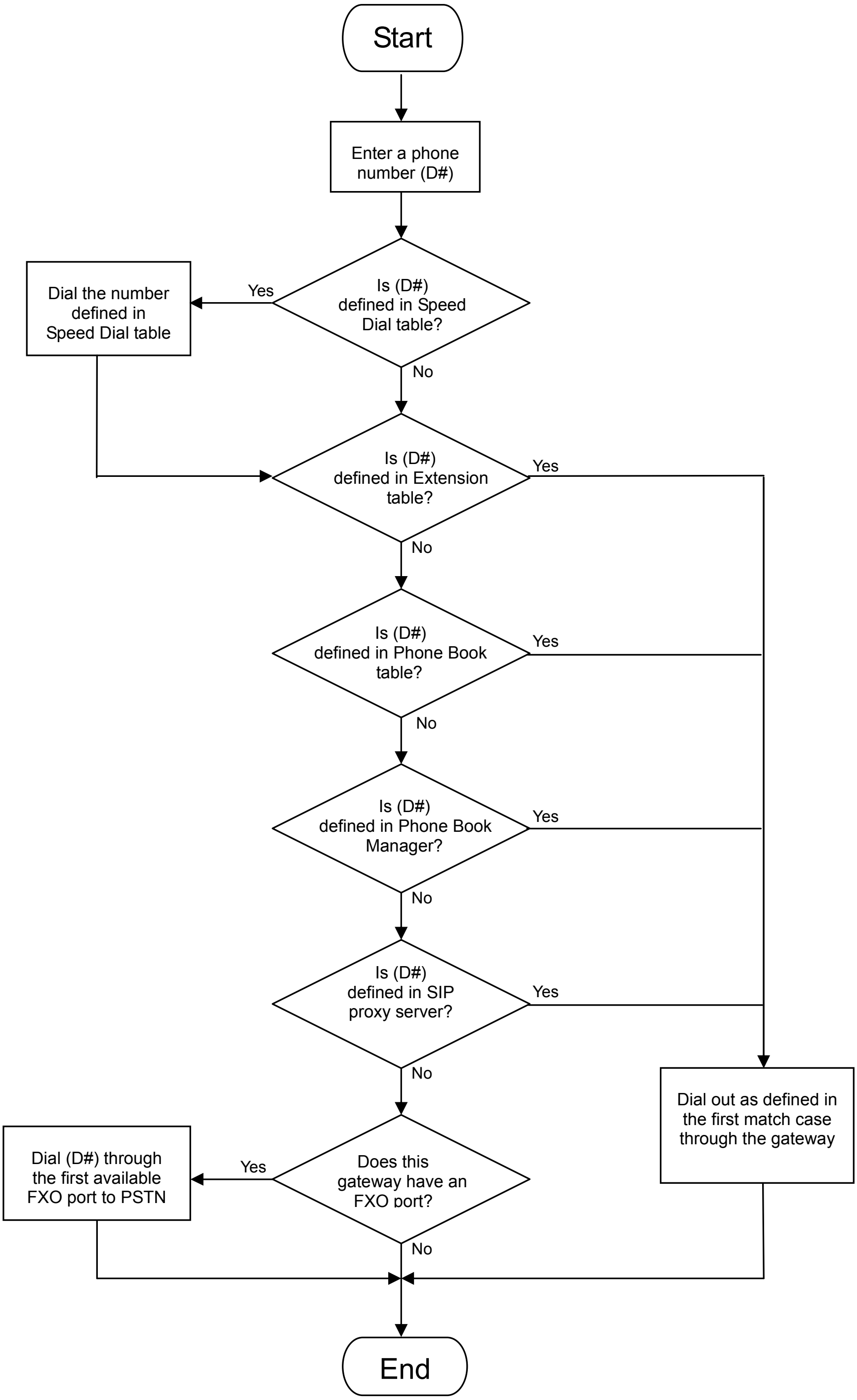
The phone number should contain at least 2 digits (not including * and #).

5-2 Dialed Number Processing Flow

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



A complete flow chart is shown on the next page.


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

5-3 Example for Match phone numbers invited by callers

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

Match	Scheme	Description
The same as "FXS Representative Number"	Ring FXS according to "FXS Group Hunting / Ring Priority" settings	
The same as "FXO Representative Number"	Off hook a FXO	It is not applied to registration with SIP Proxy
The same as "FXS Representative Number + Extension Number"	Ring or off hook the Extension	If Extension line is FXS, it should ring. If Extension line is FXO, it should off hook It is not applied to registration with SIP Proxy
The same as FXO Extension Number	Off hook the FXO	
A Prefix is the same as "FXS Representative Number +FXO Extension Number"	Eliminate a Prefix and use remaining digits to route calls via FXO	FXS Representative Number is 88 and one of FXO Extension is 070123456 If callers dial 88070123456 6371, the gateway dial 6371 via FXO Extension
A Prefix is the same as FXO Extension Number	Eliminate a Prefix and use remaining digits to route calls via FXO	One of FXO Extension is 070123456 If callers dial 070123456 6371, the gateway dial 6371 via FXO Extension
Differ from FXS/FXO numbers	Use these digits to route calls via FXO	If callers dial 6371, the gateway dial 6371 via one of FXO line

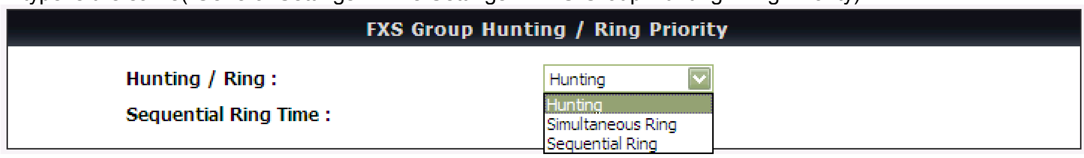
6. Multi Group Application

VoIP Gateway supports multi group hunt for incoming and outgoing calls. It make a GW could be distributed to more than one hunting group for different departments or different company.

FXS

Note： Those Extension Numbers can't be duplicate.

Note： In a GW it supports only a ringing type for group hunting. If there are three hunt groups, the ringing type is the same(General Settings-> Line Settings-> FXS Group Hunting/ Ring Priority).



[Rule 1] Make out bound call take Hunting number as the calling number.

Line	Type	Number	Hunt Group Port	Register	Invite with ID / Account
FXS Representative Number		FXS		☐	
1	FXS	0421234567 auto	1	☑	☑
2	FXS	602	1	☐	☐
3	FXS	603	5	☐	☐
4	FXS	604	1	☐	☐
5	FXS	0421234568	5	☑	☑
6	FXS	606	5	☐	☐
7	FXS	0421345679	No Group	☑	☑
8	FXS	0421345670	No Group	☑	☑

Members of group 1: #1, #2 and #4
Members of group 2: #3, #5 and #6
Hunting number of Group 1: 0421234567
Hunting number of Group 2: 0421234568
#7 and #8 are assigned as an independent number from all the groups.

- Hunt Group: To assign the master extension of the group. If you assign Extension #1, #2 and #4 in the same group. And Extension to be the represent number, please set Hunt Group of these extension to be "1". The "1" means extension "1" is the master line of this group.
- Register: The master line of the hunt group must register to a SoftSwitch or IPPBX.

According to above settings:

- As a user make an outgoing call with extension #1, #2 or #4, the calling number will be "0421234567".
- As a user make an outgoing call with extension #3, #5 or #6, the calling number will be "0421234568".
- As a user make an outgoing call with extension #7, the calling number will be "0421234569".
- As a user make an outgoing call with extension #8, the calling number will be "0421234560".
- As there is an incoming call to "0421234567", that extension #1, #2 or #4 will be alerting. It depends on the "Hunting/ Ring" type at Telephony Settings page.
- As there is an incoming call to "0421234568", that extension #3, #5 or #6 will be alerting. It depends on the "Hunting/ Ring" type at Telephony Settings page.
- As there is an incoming call to "0421345679", only extension #7 will be alerting.
- As there is an incoming call to "0421345670", only extension #8 will be alerting.

[Rule 2] Make out bound call take hunting number as the calling number.

Line	Type	Number	Hunt Group Port	Register	Invite with ID / Account	User I Accou
FXS Representative Number		FXS		☐		
1	FXS	0321234561 auto	1	☑	☑	
2	FXS	321234562	1	☑	☐	
3	FXS	321234563	5	☑	☐	
4	FXS	321234564	1	☑	☐	
5	FXS	321234565	5	☑	☑	
6	FXS	321234566	5	☑	☐	
7	FXS	321234567	No Group	☑	☑	
8	FXS	321234568	No Group	☑	☑	

Members of group 1: #1, #2 and #4
Members of group 2: #3, #5 and #6
Hunting number of Group 1: 0321234561
Hunting number of Group 2: 0321234565
#7 and #8 are assigned as an independent number from all the groups.
And each extension

- Hunt Group: To assign the master extension of the group. If you assign Extension #1, #2 and #4 in the same group. And Extension to be the represent number, please set Hunt Group of these extension to be "1". The "1" means extension "1" is the master line of this group.
- Register: The master line of the hunt group must register to a SoftSwitch or IPPBX.

According to above settings:

- As a user make an outgoing call with extension #1, #2 or #4, the calling number will be "0321234561".
- As a user make an outgoing call with extension #3, #5 or #6, the calling number will be "0321234565".
- As a user make an outgoing call with extension #7, the calling number will be "0321234567".
- As a user make an outgoing call with extension #8, the calling number will be "0321234568".
- As there is an incoming call to "0321234561", that extension #1, #2 or #4 will be alerting. It depends on the "Hunting/ Ring" type at Telephony Settings page.
- As there is an incoming call to "0321234562", only extension #2 will be alerting.
- As there is an incoming call to "0321234564", only extension #4 will be alerting.
- As there is an incoming call to "0321234565", that extension #3, #5 or #6 will be alerting. It depends on the "Hunting/ Ring" type at Telephony Settings page.
- As there is an incoming call to "0321234563", only extension #3 will be alerting.
- As there is an incoming call to "0321234566", only extension #6 will be alerting.
- As there is an incoming call to "0321234567", only extension #7 will be alerting.
- As there is an incoming call to "0321234568", only extension #8 will be alerting.

FXO

Hunt Group Port rule:

Make out bound from PBX via FXO port to VoIP take Hunting number as the calling number.

FXO Representative Number		FXO		☐		
1	FXO	0421234567 auto	1	☑	☑	
2	FXO	602	1	☐	☐	
3	FXO	603	5	☐	☐	
4	FXO	604	1	☐	☐	
5	FXO	0421234568	5	☑	☑	
6	FXO	606	5	☐	☐	
7	FXO	0421234569	No Group	☑	☑	
8	FXO	0421234570	No Group	☑	☑	

Members of group 1: #1, #2 and #4
Members of group 2: #3, #5 and #6
Hunting number of Group 1: 0421234567
Hunting number of Group 2: 0421234568
#7 and #8 are assigned as an independent number without group.

- Hunt Group: To assign the master line of the group. If you assign Line #1, #2 and #4 in the same group. Please set Hunt Group of these ports to be "1". The "1" means Line(port) "1" is the master line of this group.
- Register: The master line of the hunt group must register to a SoftSwitch or IPPBX.

According to above settings:

- As a user make an outgoing call from PBX extension via FXO port to VoIP with line #1, #2 or #4, the calling number will be "0421234567".
- As a user make an outgoing call from PBX extension via FXO port to VoIP with line #3, #5 or #6, the calling number will be "0421234568".
- As a user make an outgoing call with line #7, the calling number will be "0421234569".
- As a user make an outgoing call with line #8, the calling number will be "0421234570".
- As there is an incoming call from VoIP with SIP called number "0421234567", that line #1, #2 or #4 will be transit-out hunt group. VoIP Gateway will select the first free port to transit-out.
- As there is an incoming call from VoIP with SIP called number "0421234568", that line #3, #5 or #6 will be transit-out hunt group. VoIP Gateway will select the first free port to transit-out.
- As there is an incoming call from VoIP with SIP called number "0421234569", that VoIP Gateway will select the line #7 to transit-out.
- As there is an incoming call from VoIP with SIP called number "0421234570", that VoIP Gateway will select the line #8 to transit-out.

Appendix

Product Features

WAN

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

LAN

- One 10/100Mbps auto-negotiation, auto-crossover RJ 45 Ethernet port
- Supports router and bridge mode
- DHCP server

Voice Features

- SIP (RFC3261) compatible
- Voice codecs : G.711 a /u law, G.726, G.729A, G.723.1
- CNG (Comfort Noise Generation)
- VAD (Voice Activity Detection)
- G.165/G.168 echo cancellation
- Adjustable Jitter Buffer and programmable Gain Control
- In-Band DTMF, Out-Of-Band DTMF relay (RFC2833, SIP INFO)
- Multiple SIP Proxy server entries with failover mechanism
- Polarity reversal generation
- T.30 (G.III) / Real time T.38 / Secured T.38 FAX relay
- DTMF, FSK (Bellcore ,ETSI and NTT) Caller ID generation.
- Support Caller ID Restriction (CLIR)
- Digit Map for dial plan
- Speed Dial
- Local phone book for peer-to-peer calling
- E.164 Numbering & ENUM support
- Hot-Line, Warm-Line support
- Single Number / Account (reprehensive number) for multiple ports
- Call features:
    - Call Hold, Call Waiting, Call Pickup
    - Call Forward - Unconditional, Busy, No Answer
    - Call Transfer - Unattended, Attended
    - Three Way Calling (Media Server required)
- Analogue interface
    - Connector: RJ-21
    - Signaling protocol : Loop Start

Configuration & Maintenance

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