User's Manual

IP Telephony Gateway

Model No.: SP5001C, SP5001D, SP5002A, SP5012A

World Wide Web: www.micronet.com.tw ; www.micronet.info

About this User's Manual

This User's Manual gives users basic steps on installation and operation. Please read this manual chapter by chapter.

Chapter 1. Introduction

Introduce the IP Telephony Gateway to users in terms of feature, appearance, and application.

Chapter 2. Startup

Help user complete basic configuration.

Chapter 3. Operation

Show user how to use the device to process phone call and FAX.

Chapter 4. Web Administration

Provide command reference of Web Interface for advanced setting.

Chapter 5. IVR/Keypad Management

Provide instructions on configuring the IP Telephony gateway via Keypad on the phone set.

Chapter 6. Specification

List the specification of the gateway in detail.

Online Upgrade

Please refer to http://www.micronet.info/ for additional support.

Table of Content

1. Introduction

Micronet SP5001C, SP5001D, SP5002A and SP5012A IP Telephony Gateway is designed to connect standard telephone devices to IP-based telephony networks, providing users with high-quality VoIP service.

SP5001C / SP5002A provides:

- 1/2 FXS port(s) for phone set, FAX machine, or PBX's trunk

SP5001D provides:

- 1 FXS port for phone set, FAX machine, or PBX's trunk
- 1 PSTN port for PSTN lifeline that tranceives PSTN calls as backup even if VoIP fails.

SP5012A provides:

- 1 FXS port for phone set, FAX machine, or PBX's trunk
- 1 FXO port for PSTN line or PBX's extension to make communication between PSTN and IP clients.

With built-in router function, they offer internet access sharing to co-located PCs. The simple operation and configuration features are the most suitable for residential and SOHO applications.

1.1 Key Features

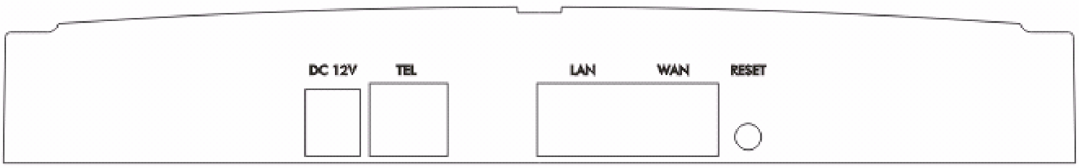
- Compliant with IETF SIP standards
- Provide 2 10/100M RJ-45 ports for WAN and LAN connection
- Support G.729a/b, G.711a/μ-law, and G.726 codecs
- Support up to 3 SIP service domains
- Support STUN and Outbound proxy for NAT traversal
- Support VAD, CNG, EC, and Adaptive Jitter Buffer
- Support FSK / DTMF caller ID display
- Support Call Hold / Call Waiting / Call Forward
- Support 3-way conference
- Provide phone address book and speed dialing function
- Transmit voice and FAX (T.38 and in-band)
- Support PPTP client for VPN
- Support IP ToS/DSCP, and 802.1q/p for QoS
- Easy management via WEB and IVR/keypad

1.2 Physical Description

SP5001C:

SP5001C Front Panel

LED	Status	Description
PWR	On/Green	Power On
STATUS	On/Amber	Line Registered
TEL	On/Amber	Phone set off-hook
LAN	On/Green	Link On
WAN	On/Green	Link On



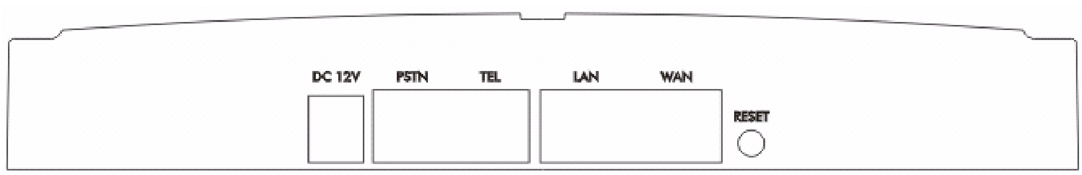
SP5001C Rear Panel

RESET	Factory default button. Press and hold for 5 seconds to reset
WAN	RJ-45 port of 10/100M for connecting to modem
LAN	RJ-45 port of 10/100M for connecting to PC or hub/switch that connects PCs
TEL	RJ-11 port for connecting to phone set or PBX trunk

SP5001D:

SP5001D Front Panel

LED	Status	Description
PWR	On/Green	Power On
PSTN	On/Amber	PSTN mode / VoIP Unregistered
TEL	On/Amber	Phone set off-hook
LAN	On/Green	Link On
WAN	On/Green	Link On



SP5001D Rear Panel

RESET Factory default button. Press and hold for 5 seconds to reset

WAN RJ-45 port of 10/100M for connecting to modem

LAN RJ-45 port of 10/100M for connecting to PC or hub/switch that connects PCs

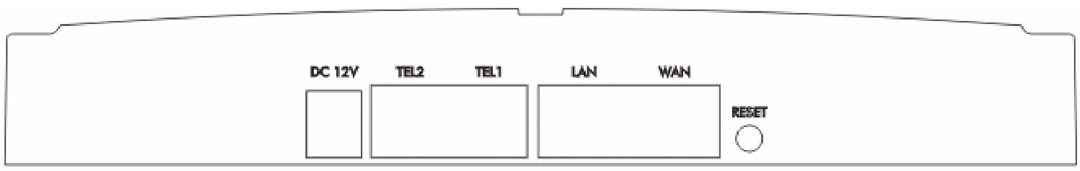
TEL RJ-11 port for connecting to phone set or PBX trunk

PSTN RJ-11 port for connecting to PSTN (lifeline)

SP5002A:

SP5002A Front Panel

LED	Status	Description
PWR	On/Green	Power On
TEL1	On/Amber	Line Registered
TEL2	On/Amber	Line Registered
LAN	On/Green	Link On
WAN	On/Green	Link On



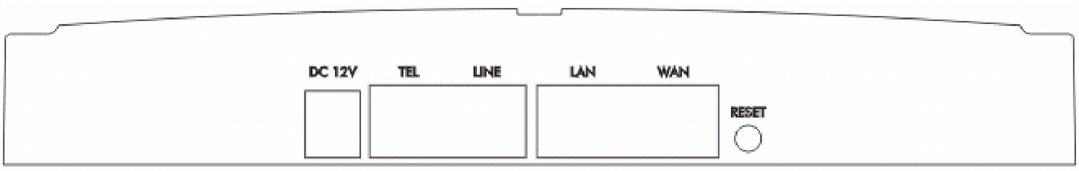
SP5002A Rear Panel

RESET	Factory default button. Press and hold for 5 seconds to reset
WAN	RJ-45 port of 10/100M for connecting to modem
LAN	RJ-45 port of 10/100M for connecting to PC or hub/switch that connects PCs
TEL1	RJ-11 port for connecting to phone set or PBX trunk
TEL2	RJ-11 port for connecting to phone set or PBX trunk

SP5012A:

SP5012A Front Panel

LED	Status	Description
PWR	On/Green	Power On
TEL	On/Amber	Line Registered
LINE	On/Amber	Line Registered
LAN	On/Green	Link On
WAN	On/Green	Link On

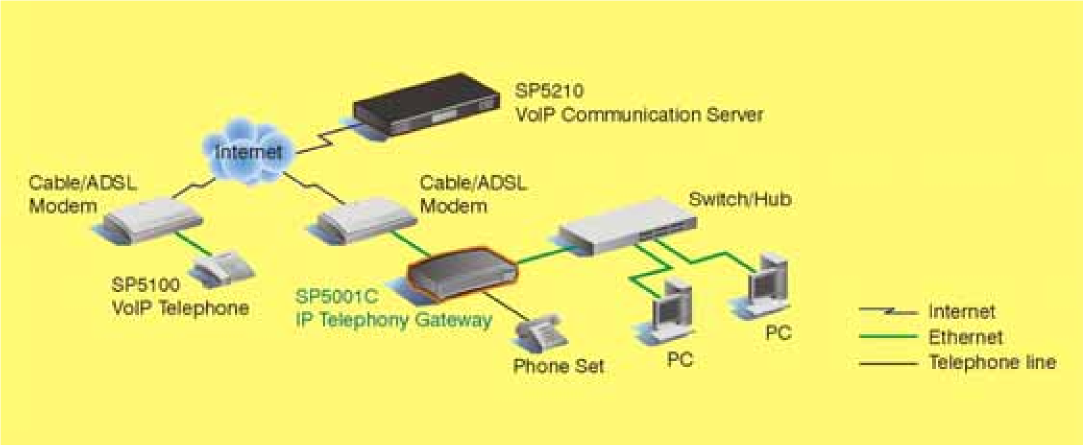


SP5012A Rear Panel

RESET	Factory default button. Press and hold for 5 seconds to reset
WAN	RJ-45 port of 10/100M for connecting to modem
LAN	RJ-45 port of 10/100M for connecting to PC or hub/switch that connects PCs
LINE	RJ-11 port for connecting to PSTN or PBX extension
TEL	RJ-11 port for connecting to phone set or PBX trunk

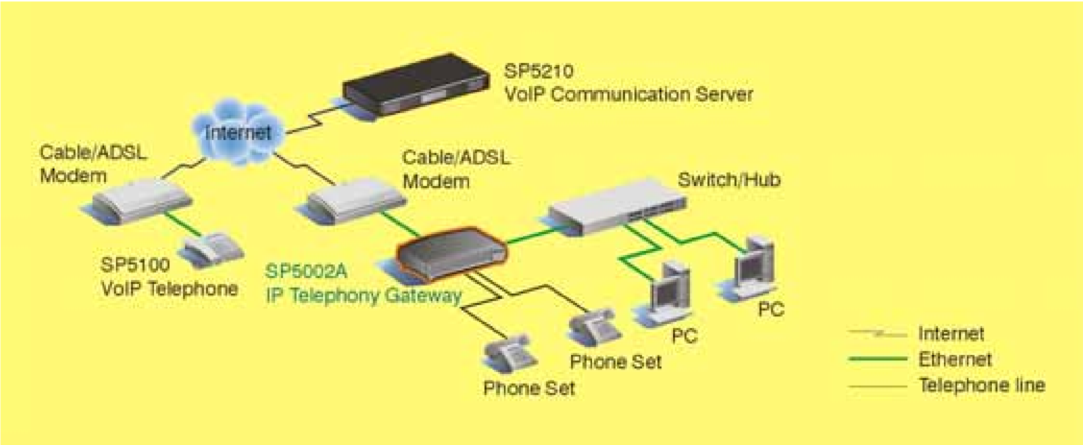
1.3 Application

SP5001C:



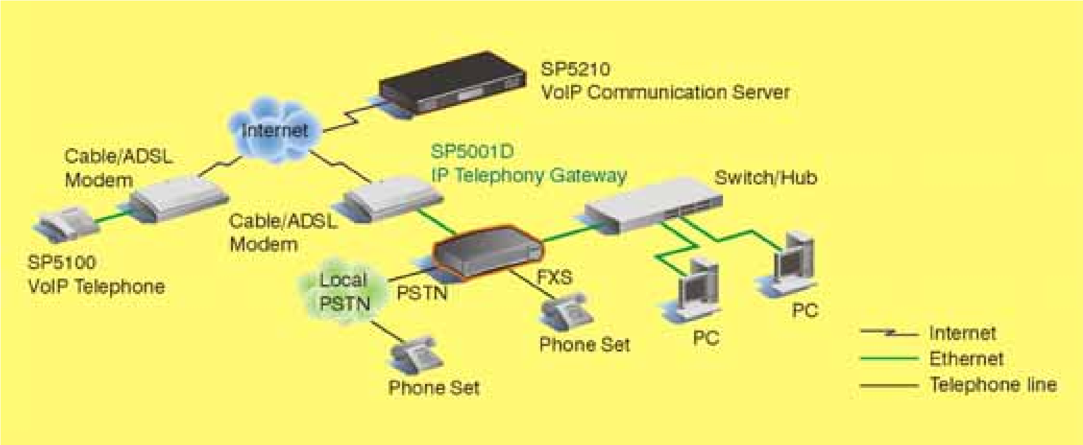
- 1 RJ-11 FXS port is provided for phone set or PBX's trunk line connection
- 2 RJ-45 ports of 10/100M are provided for WAN and LAN connection
- The IP telephony gateway can share Internet access with LAN clients

SP5002A:



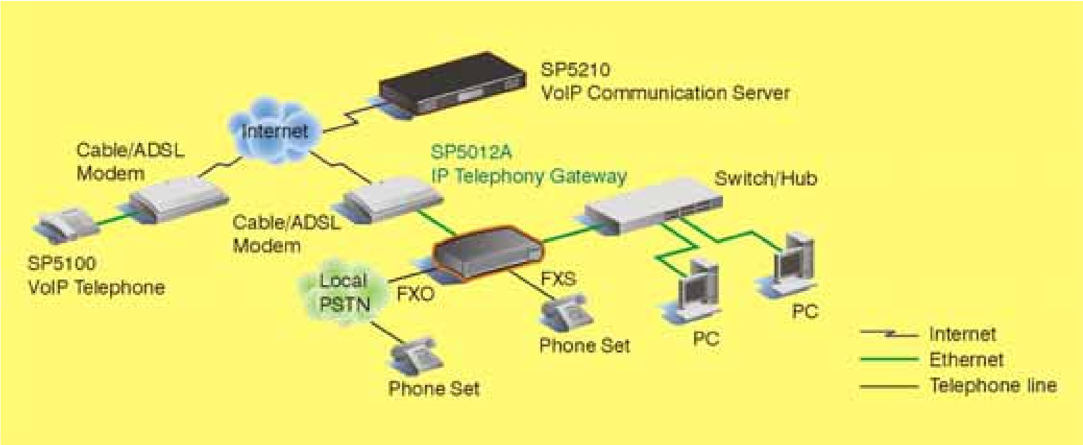
- 2 RJ-11 FXS ports are provided for phone set or PBX's trunk line connection
- 2 RJ-45 ports of 10/100M are provided for WAN and LAN connection
- The IP telephony gateway can share Internet access with LAN clients

SP5001D:



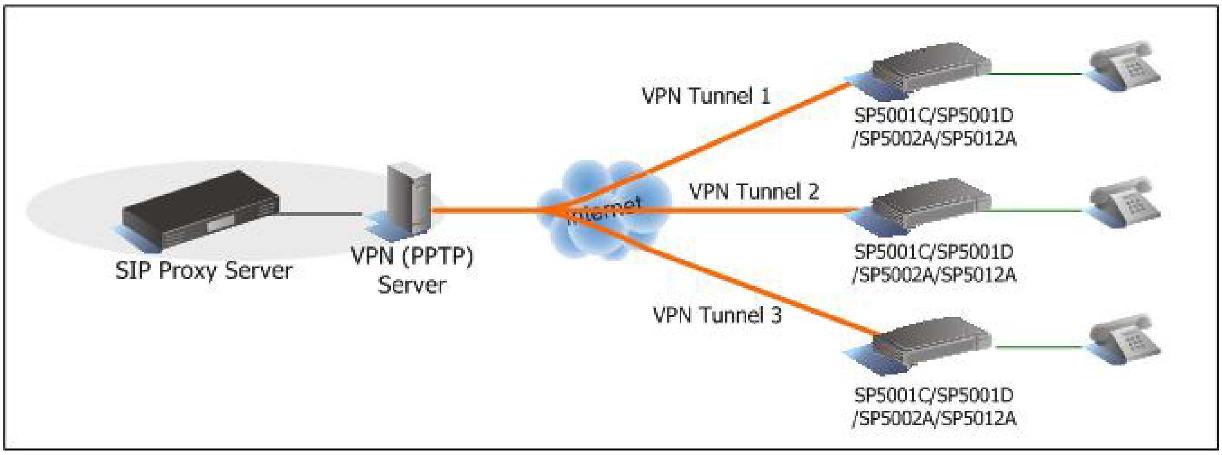
- 1 RJ-11 FXS port is provided for phone set or PBX's trunk line connection
- 1 RJ-11 PSTN port is provided to tranceive PSTN calls as backup even if VoIP fails
- 2 RJ-45 ports of 10/100M are provided for WAN and LAN connection
- The IP telephony gateway can share Internet access with LAN clients

SP5012A:



- 1 RJ-11 FXS port is provided for phone set or PBX's trunk line connection
- 1 RJ-11 FXO port is provided for PSTN or PBX's extension connection, and for PSTN client to communicate with IP client
- 2 RJ-45 ports of 10/100M are provided for WAN and LAN connection
- The IP telephony gateway can share Internet access with LAN clients

VPN (Virtual Private Network)



The IP telephony gateway series supports PPTP client for VPN function. It can establish VPN tunnel with PPTP server, and get access to the peer private network as if it is located in the same LAN.

For special condition that SIP proxy server is located in the private network of CO site, the gateway can register, and request a phone call via the VPN tunnel.

Please refer to the section *2.4 VPN Settings*.

2. Startup

2.1 Login into the System

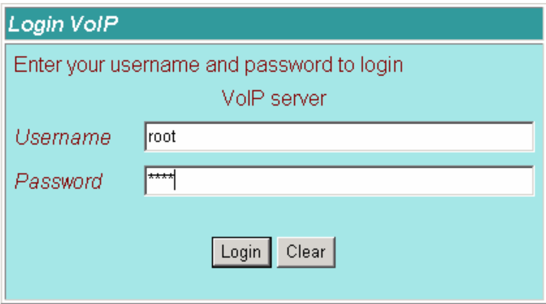
The embedded web configuration allows you to use a web browser to manage the IP Telephony gateway.

Step 1. Connect LAN port to your managing PC. Or, connect the gateway with PC by hub/switch.

Step 2. Launch your web browser with http://192.168.123.1:9999/. Please configure IP address of PC with 192.168.123.x.

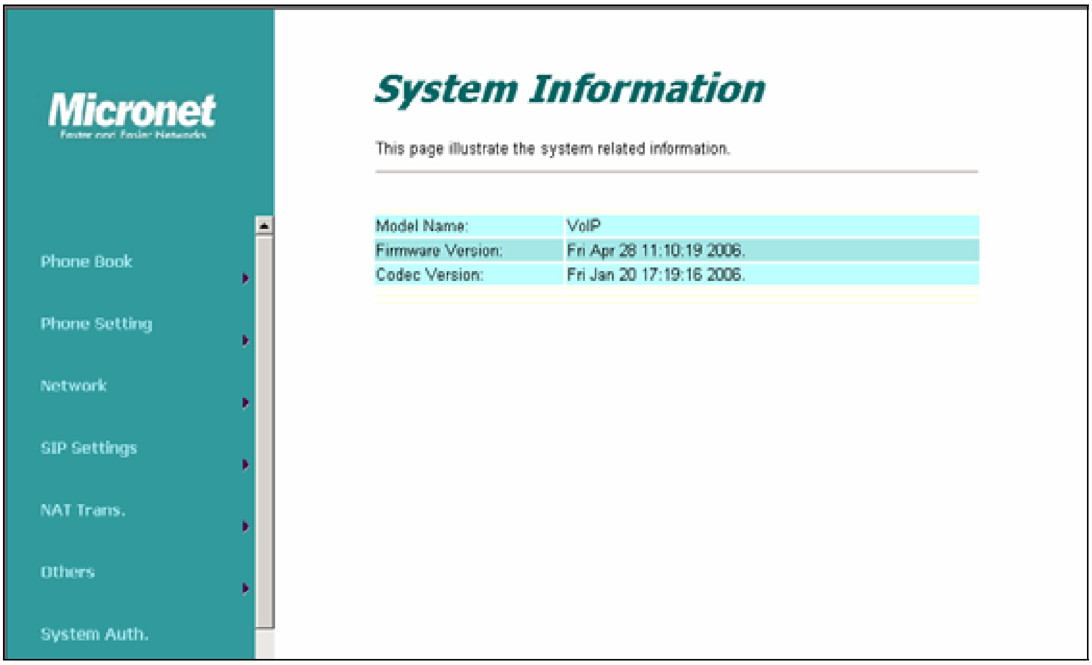
Step 3. The Password screen now appears. Type "***root***" in the user name field and your password (none by default) in the password field.

Step 4. Click on $\boxed{\text{Login}}$.

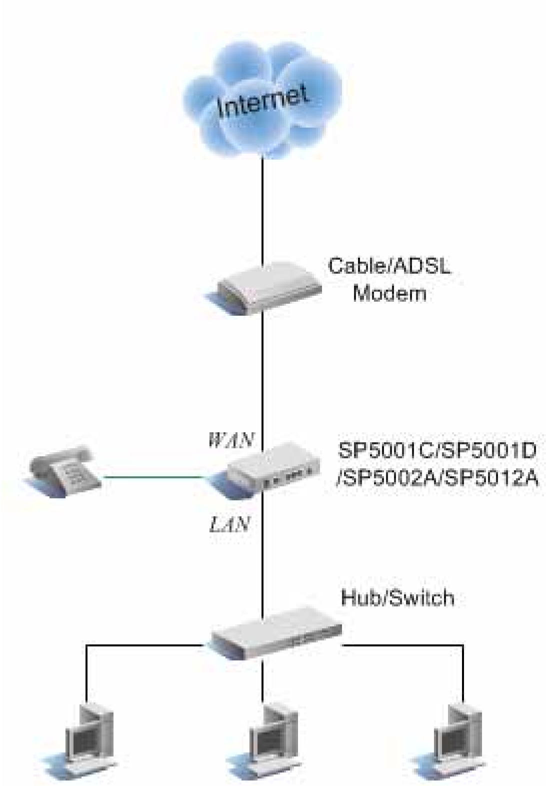


Login Screen

Step 5. After a successful login, you will see the screen *System Information* as shown below.

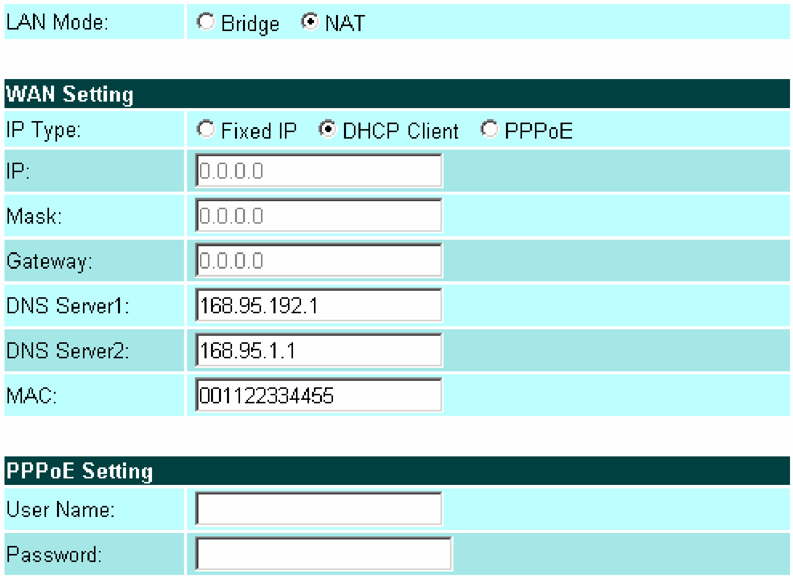

Micronet
System Information
This page illustrate the system related information.
Model Name: VoIP
Firmware Version: Fri Apr 28 11:10:19 2006.
Codec Version: Fri Jan 20 17:19:16 2006.
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans.
Others
System Auth.

2.2 Network Configuration



By default, the gateway is in NAT mode (router mode) and can share Internet access with PCs. Go to [*Network / WAN Settings*], and configure WAN setting according to actual condition. In default IP type of DHCP client, it requests necessary IP information from your ISP automatically.

***Note:** Different ISPs require different methods of connecting to the Internet. Please consult your ISP to select right IP type (Fixed IP, PPPoE) of WAN.*



Parameter	***Description***
LAN Mode	Bridge: pure VoIP gateway
	NAT: VoIP router
IP Type	Select Fixed IP, DHCP (default), or PPPoE
IP	IP address provided by ISP
Mask	Subnet mask provided by ISP
Gateway	ISP's IP address gateway
DNS Server1/2	IP address of primary/secondary DNS server
MAC	MAC address
PPPoE	
Username	User Name provided by ISP for the PPPoE connection
Password	Password provided by your ISP for the PPPoE connection

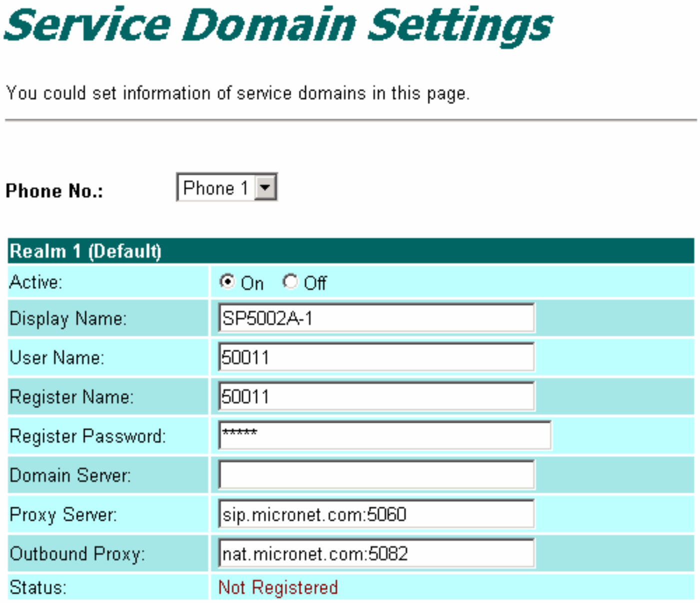
When Router mode is disabled, the unit is just a pure VoIP gateway. In LAN mode, select ***Bridge*** to disable router mode.

Note:

1. *Please save and reboot the system to take effect. Go to [Save Change] to save configuration, and the system will reboot automatically.*
2. *Please unplug LAN port when LAN mode is **Bridge**. Just keep WAN port plugged.*

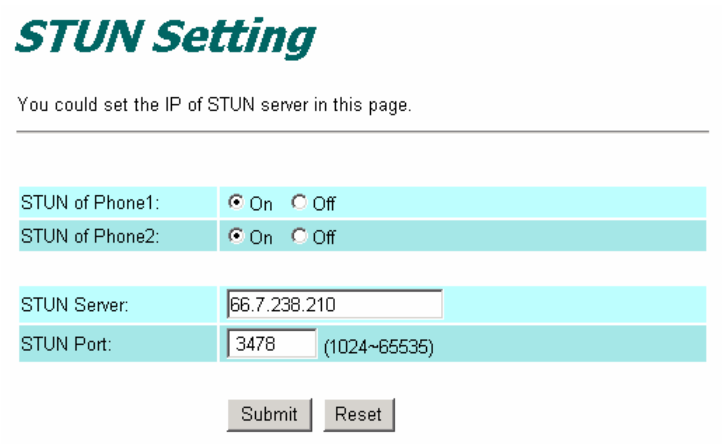
2.3 SIP Configuration

Go to [*SIP Settings / Service Domain*]. Each port can be configured to register 3 different service domains.



Parameter	***Description***
Phone No.	Select specific port to configure
Active	Select "On" to activate
User Name	IP telephony number of the line
Register Name	User's ID
Register Password	Password
Domain Server	Domain server IP
Proxy Server	SIP proxy IP and Port (default: 5060) <sip_proxy_ip>:<sip_proxy_port>
Outbound Proxy	Outbound proxy IP and Port <outbound_proxy_ip>:< outbound_proxy_port>
Status	Registered or not.

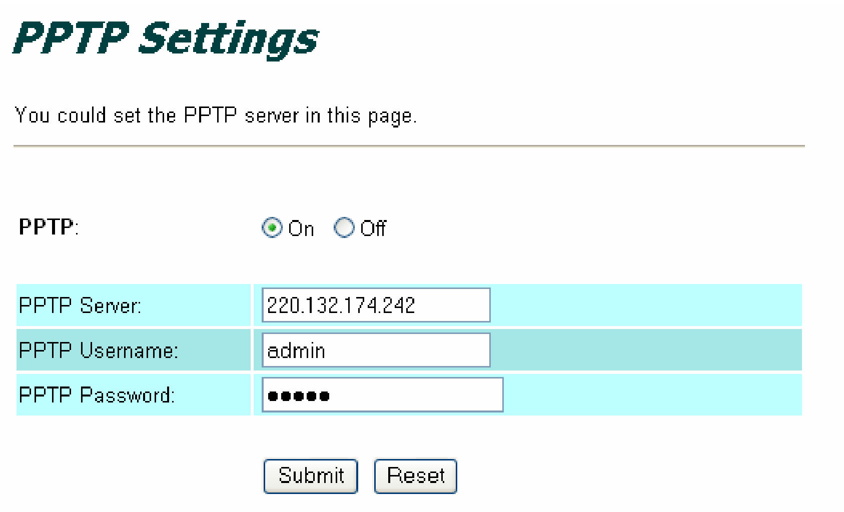
Go to [*NAT Trans. / STUN Setting*] to set up ***STUN On***, if necessary.



Note: *Please save and reboot the system to take effect. Go to [Save Change] to save configuration, and the system will reboot automatically.*

2.4 VPN Configuration

IP telephony gateway supports PPTP client for VPN. It can establish VPN tunnel with PPTP server. Go to [*Network / PPTP Settings*], and set PPTP server address, and authentication information (username, password).



Parameter	***Description***
PPTP Server	Set IP address of PPTP server
PPTP Username	Set username for authentication to set up VPN tunnel
PPTP Password	Set password for authentication to set up VPN tunnel

Click on Submit, and save to take effect.

After tunnel is established, the gateway gets one private IP address (**Interface 2**) from PPTP server as shown in [*Network / Status*].

Interface 0	
Type:	PPPoE Client
IP:	61.229.27.228
Mask:	255.0.0.0
Gateway:	61.229.24.254
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

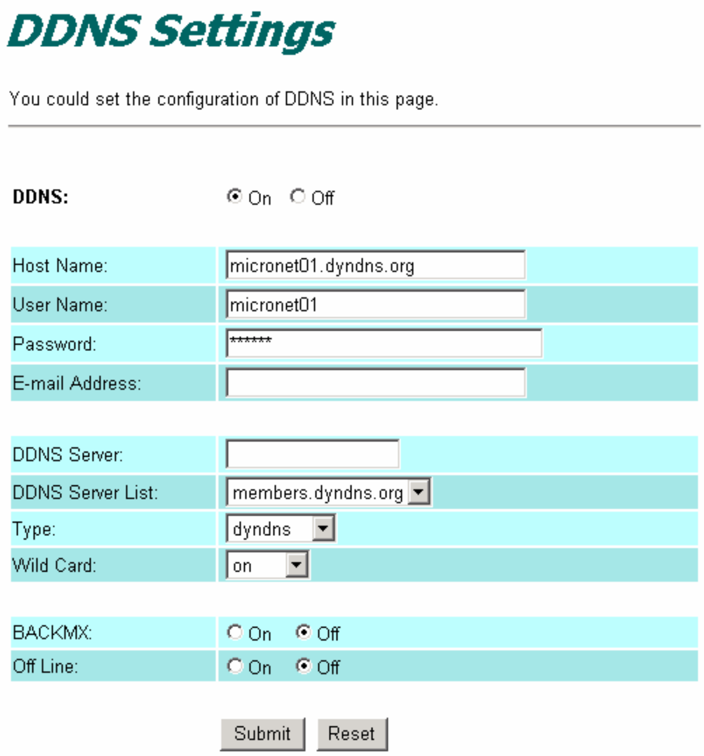
Interface 2	
Type:	Fixed IP Client PPPoE
IP:	192.168.1.101
Mask:	255.255.255.0
Gateway:	192.93.117.2
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

2.5 DDNS

DDNS allows you to map the static domain name to a dynamic IP address. You must get an account, password and your static domain name from the DDNS service providers.

- DDNS makes the gateway accessible for other client to call in P2P (peer-to-peer) mode, when the IP address is dynamic.
- DDNS setting is not necessary when IP call is tranceived via service domain only.

Go to [*Network / DDNS Settings*] and set up DDNS.



Note: *Please save and reboot the system to take effect. Go to [Save Change] to save configuration, and the system will reboot automatically.*

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3. Operation

3.1 Make a Call

By default, call is sent via the first registrar server only. The telephone number of clients in the second/third service domain will be not accessible.

Make a Call	*Press **<telephone_number>*** + #

Note:

- *Once the first registration fails, the second realm will be activated.*
- *The gateway can always receive incoming call from the client of either registered service domain.*
- *The gateway is accessible for P2P client to dial the configured telephone numbers. DDNS setting is necessary when WAN IP address is dynamic.*

3.1.1 Make PSTN Call (SP5001D only)

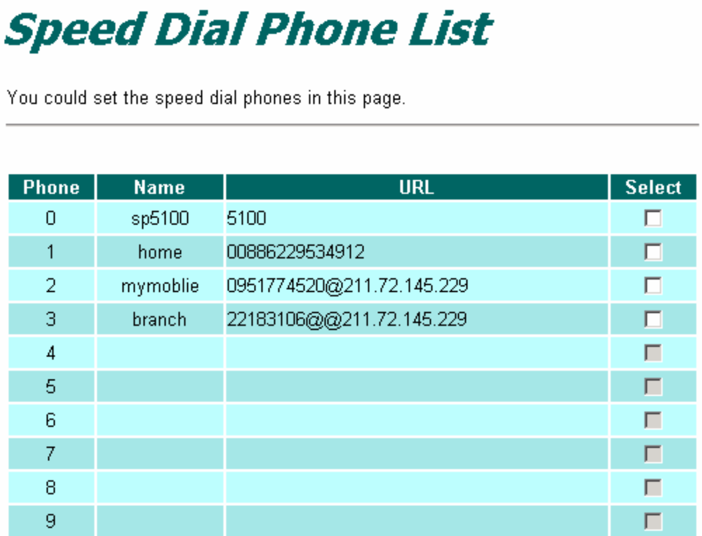
SP5001D provides 1 RJ-11 port for PSTN lifeline, and can tranceive PSTN calls even if VoIP fails.

Make PSTN Call	*VoIP Unregistered:* • *Press **<pstn_number>*** *VoIP Registered:* • *Press **0** ⁎ to switch to PSTN mode* • *press <pstn_number>*

3.2 Speed Dial / P2P call

Speed dial	*Press **<index>** + #*

Go to [*Phone Book / Speed Dial Setting*]. User can create 10 entries (0~9) in ***Speed Dial Phone List***.



Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	URL	Select
0	sp5100	5100	☐
1	home	00886229534912	☐
2	mymoblie	0951774520@211.72.145.229	☐
3	branch	22183106@@211.72.145.229	☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

Parameter	***Description***
Position	Speed dial index, 0~9
Name	Alias of the specific entry
URL	It can be…

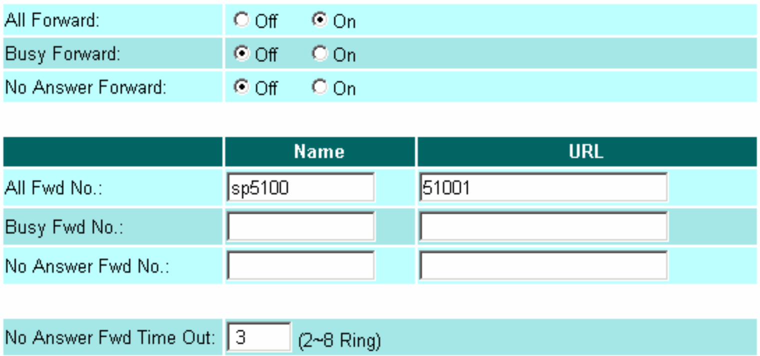
- Telephone number of registered client
- PSTN number (if server provides off-net call)
- Address-like URL (peer-to-peer call)

<telephone_number> @ <peer_device_ip_address>

<pstn_number> @ <peer_fxo_gateway_ip_address>

3.3 Call Forward

Go to [*Phone Setting / Call Forward*]. There are 3 selections in Forward type. User must select the condition under which to forward calls.



Parameter	***Description***
All Forward	Forward the call in any conditions

- Off: call forward disabled.
- On/IP: Call forward to IP. It disables other 2 forwarding types.
- PSTN: Call forward to PSTN. It disables other 2 forwarding types.

Busy Forward	When the phone is in busy status, forward the call

- Off: call forward disabled.
- On: IP Call forward to IP.

No Answer Forward	When the phone is not picked up for a period of time, forward the call

- Off: call forward disabled.
- On/IP: Call forward to IP.
- PSTN: Call forward to PSTN.

Name	Alias of the forwarding number
URL	Destination number (that call is forwarded to) or address-like URL. It can be…

- Telephone number of registered client
- PSTN number (if server provides off-net call)
- Address-like URL (Calling party has to support P2P call.)

3.4 Call Hold / Call Waiting / Conference

The IP telephony gateway provides telephony features, as call hold, call waiting, and 3-way conference.

- Call Hold: Hold a existing call
- Call waiting: Hold a existing call, and answer a new incoming call
- 3-way conference: Talk with other 2 party in the same session.

Call Hold	● *Press **Hook** or **Flash** to hold* ● *Press **Hook** or **Flash** again to resume*
Call Waiting	● *Voice "do-do" informs of a new call when a talk is in progress* ● *Press **Hook** or **Flash** to hold the existing call, and proceed to the new call* ● *Press **Hook** or **Flash** to switch between two calls*
3-way Conference	● *Establish a call* ● *Press **Hook** or **Flash** to hold the existing call* ● *Hear a dial tone and dial to establish another* ● *Press **Hook** or **Flash** again*

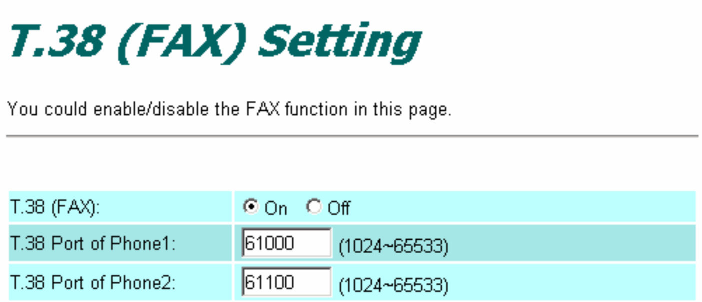
Note:

- *Adjust "Flash Time" to make **Hook** key work, if necessary. Set up the time to suit the connected phone set. Please refer to the section **4.2.7 Flash Time Setting**.*
- *Call waiting function has to be enabled. Please refer to the section **4.2.8 Call Waiting Setting**.*

3.5 FAX

Send FAX	• *Press **<fax_number>** to connect fax machine* • *Start to send FAX*

T.38 FAX: Go to [*Phone Setting / T.38 (FAX) Setting*]. Click on “On” and enable T.38 to tranceive FAX over IP.



In-band FAX: disable T.38 and choose G.711 codec as top priority. Please refer to the section ***4.4.3 Codec Settings***.

Note: *When sending in-band FAX (in G.711), please disable T.38 and choose G.711 codec as top priority. Please refer to the section **4.4.3 Codec Settings**.*

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4. Web Administration

4.1 Phone Book

Please refer to the section ***3.2 Speed Dial / P2P call***.

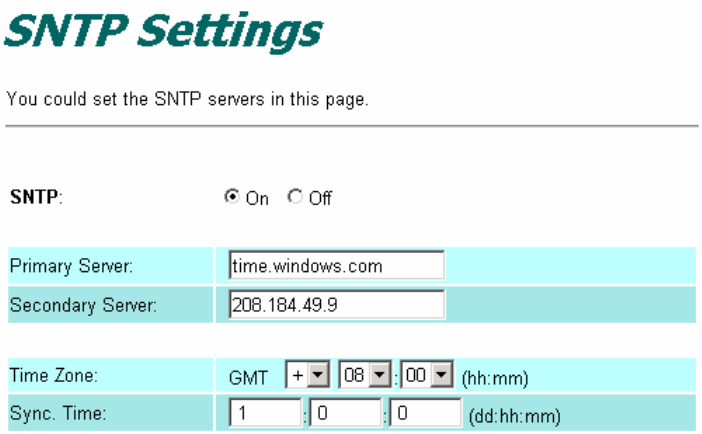
4.2 Phone Setting

4.2.1 Call Forward

Please refer to the section ***3.3 Call Forward***.

4.2.2 SNTP Settings

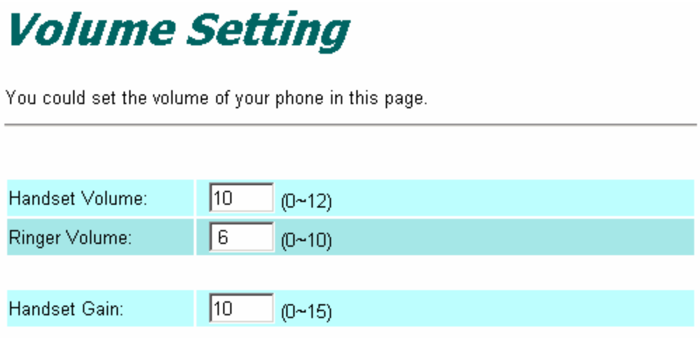
User can set up the primary and second SNTP Server IP Address, to get the date/time information.



4.2.3 Volume Settings

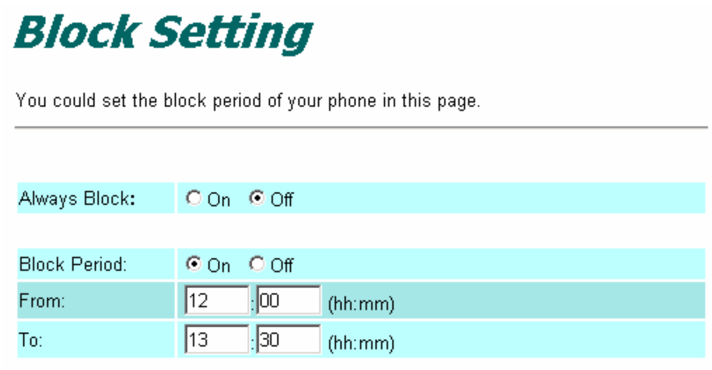
User can set up the Handset Volume, Ringer Volume, and the Handset Gain.

- Handset Volume: adjust the volume that you hear from the handset.
- Ringer Volume: adjust the ringer volume of phone set
- Handset Gain: adjust the volume that the gateway sends out to the other side.



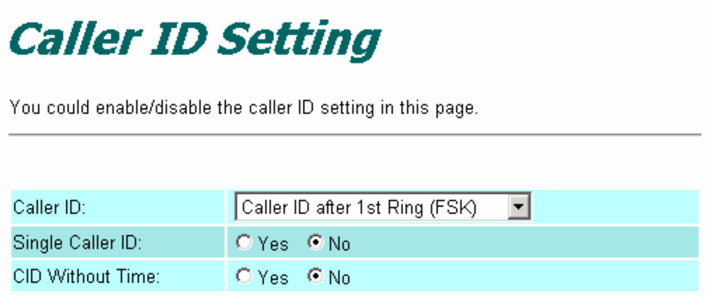
4.2.4 Block Settings

User can set up the gateway to block incoming calls and the period.



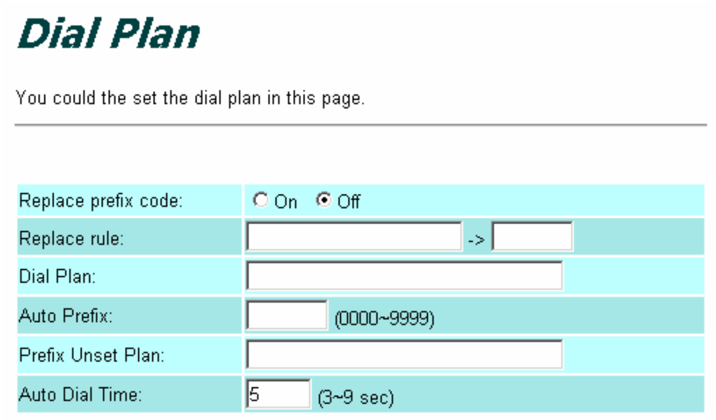
4.2.5 Caller ID

User can set the device to show Caller ID in your PSTN Phone. The gateway supports FSK and DTMF.



4.2.6 Dial Plan Setting

User can set dialing plan and timeout to send a phone call after dialing number is input.



Parameter	***Description***
Replace Prefix code	On: enable "Replace rule" Off: disable "Replace rule"
Replace rule	Replace matched prefix with another
Dial Plan	If dialed numbers match the rule, numbers is sent out. If not, numbers would not be sent out. Plan with prefix "0" is invalid
Auto Prefix	Prepend the prefix before the dialed number is sent out
Prefix Unset Plan	Auto prefix is ignored when matching unset prefix
Auto Dial Time	Set up timeout to send a phone call after dialing number is input without pressing "#"

Example:

Replace Rule: 001 + 006 + 009 → 005

Input	Sent out
001-6601	005-6601
006-5211	005-5211
009-4644	005-4644

Dial Plan: *xx + #xx + 11x + xxxxxxxx

Rule	Allowable input number
*xx	∗01, ∗02, ∗15, ∗89, …
#xx	#11, #28, #96
11x	111, 112, 113, ..., 119, 110
xxxxxxxx	Any 8-digit number, 12345687, 97856412, …

* means: keypad ∗ on the phone
x means: digit 0, 1, 2~9
means: keypad # on the phone
+ means: or

Auto prefix: 02 (0000~9999)

Input	Sent out
22183656	02-22183656
82265630	02-82265630

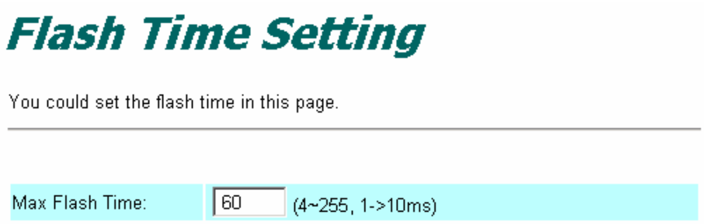
Prefix Unset Plan: 1 + 0 + xxxxxx

- With prefix “0”, auto prefix “02” is not prepended to dialing number
- With prefix “1”, auto prefix “02” is not prepended to dialing number
- With 6-digit number, auto prefix “02” is not prepended to dialing number

Auto Prefix	Input	Sent out
02	0075	0075
02	1075	1075
02	2075	02-2075
02	22075	02-22075
02	222075	222075

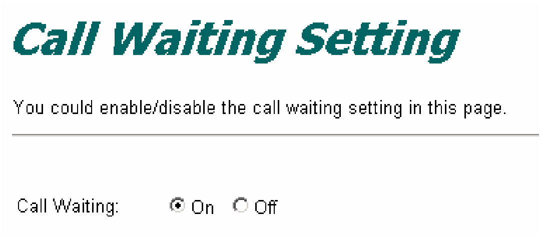
4.2.7 Flash Time Setting

User can set up “Flash Time” to detect flash signal initiated by pressing **Hook** or **Flash** key.



4.2.8 Call Waiting Setting

User can enable Call Waiting function.



4.2.9 T.38 (FAX) Setting

Please refer to the section ***3.5 FAX***.

4.3 Network

4.3.1 Status

User can check the current Network setting.

Network Status

This page shows current status of network interfaces of the system.

Interface 0	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.2.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Client
IP:	0.0.0.0
Mask:	0.0.0.0
Gateway:	0.0.0.0
DNS Server 1:	0.0.0.0
DNS Server 2:	0.0.0.0

4.3.2 WAN Settings

Please refer to the section ***2.2 Network Configuration***.

4.3.3 LAN Settings

User can configure for LAN clients.

LAN Setting	
IP:	192.168.123.1
Mask:	255.255.255.0
MAC:	001122334455

DHCP Server	
DHCP Server:	On Off
Start IP:	150
End IP:	200
Lease Time:	1 : 0 (dd:hh)

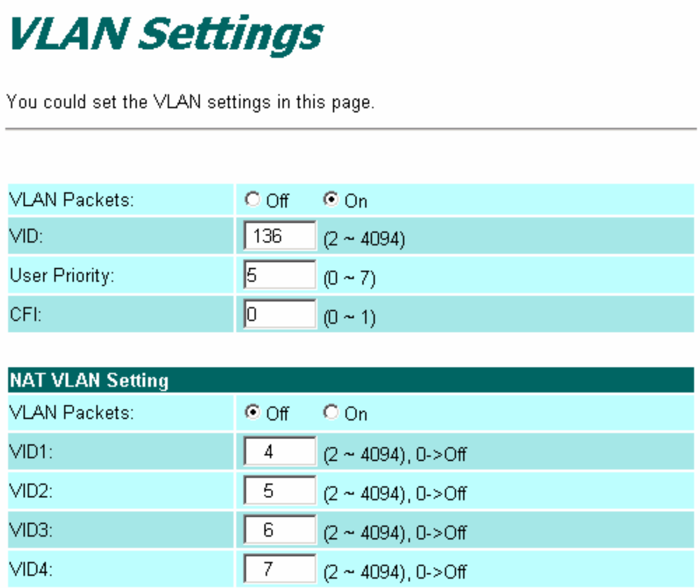
Parameter	Description
IP	IP address of LAN port. Default gateway IP of LAN clients in the local network
Mask	Subnet mask
MAC	MAC address
DHCP Server	"On" means DCHP server enabled. By enabling the DHCP server, the router will automatically give your LAN clients an IP address
Start/End IP	The IP range is released by DHCP
Lease Time	Time period that the DHCP lends an IP address to your LAN clients

4.3.4 DDNS Settings

Please refer to the section ***2.5 DDNS***.

4.3.5 VLAN Settings

User can configure VLAN and prioritization respectively for voice and data packets.



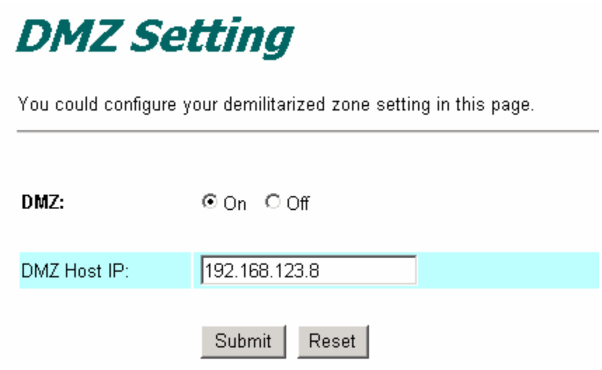
4.3.6 PPTP Settings

Please refer to the section ***2.4 VPN Configuration***.

4.3.7 DMZ *

Open another page http://192.168.123.1:9999/dmzset.htm. With the function enabled, the gateway will re-direct all packets going to your WAN port IP address to a particular IP address in your LAN.

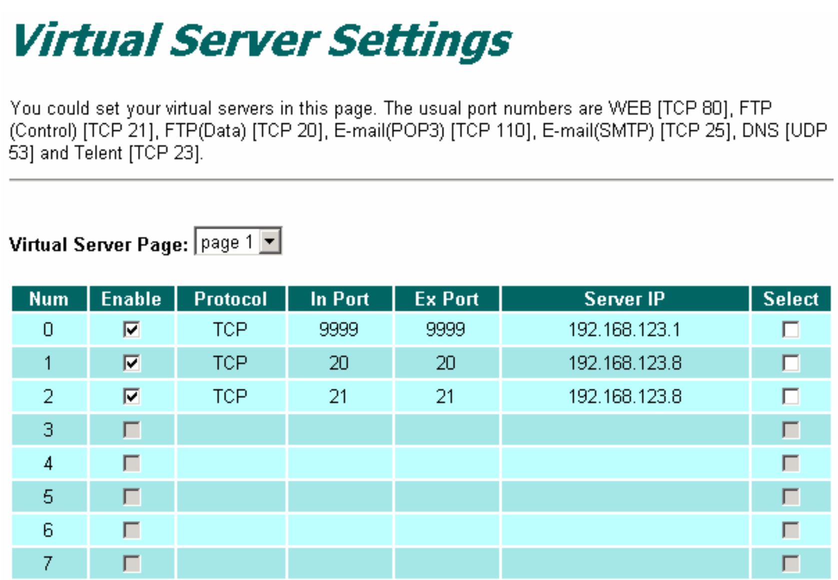
* *will be available in the later version.*



4.3.8 Virtual Server *

Open another page http://192.168.123.1:9999/vsset.htm. With the function enabled, it allows you to re-direct a particular service port number (from the Internet/WAN Port) to a particular LAN IP address and its service port number.

* *will be available in the later version.*



Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	☑	TCP	9999	9999	192.168.123.1	☐
1	☑	TCP	20	20	192.168.123.8	☐
2	☑	TCP	21	21	192.168.123.8	☐
3	☐					☐
4	☐					☐
5	☐					☐
6	☐					☐
7	☐					☐

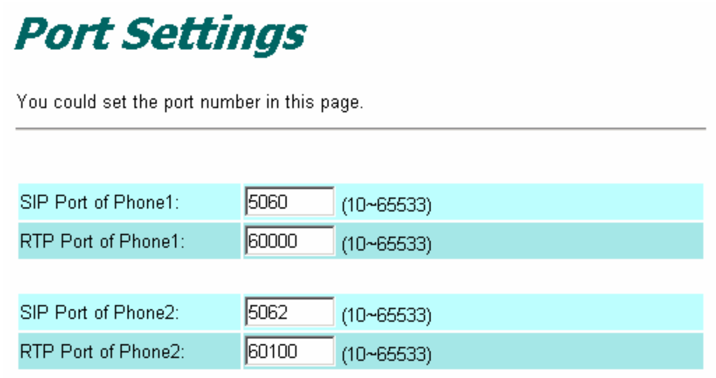
4.4 SIP Settings

4.4.1 Service Domain

Please refer to the section ***2.3 SIP Configuration***.

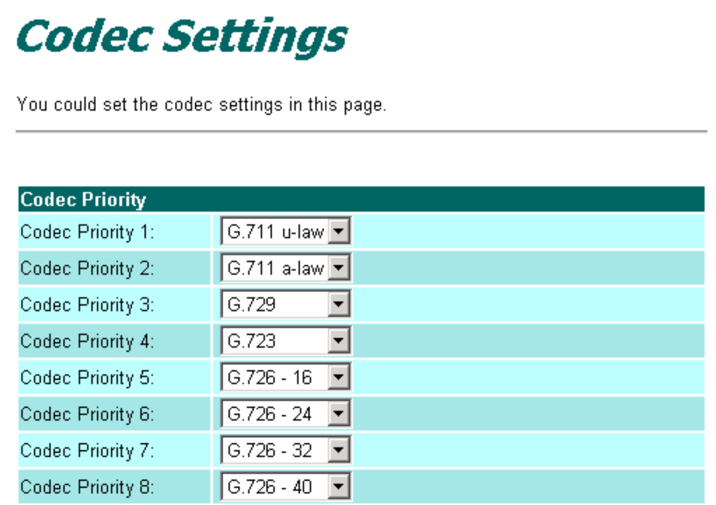
4.4.2 Port Settings

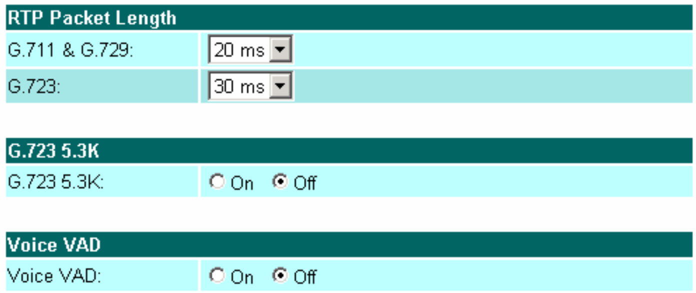
User can set up SIP and RTP ports.



4.4.3 Codec Settings

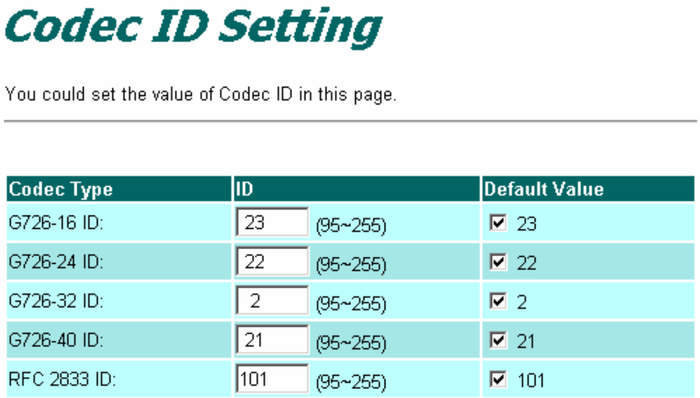
User can set up the Codec priority, RTP packet length, and VAD function. Please follow service provider’s suggestion to configure.





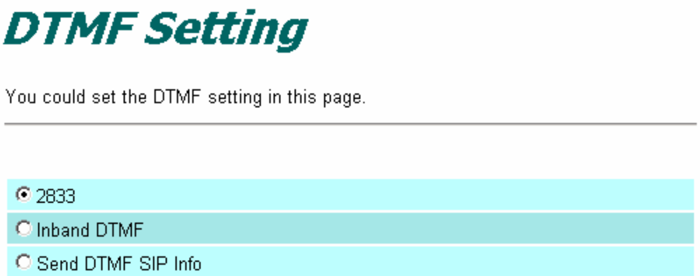
4.4.4 Codec ID Settings

User can set up codec ID for different codec. This ID represents the codec used to encode data in the Track.



4.4.5 DTMF Setting

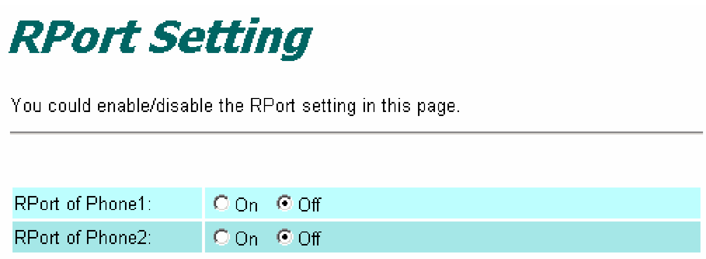
User can set up the method of DTMF transmission: In-band, RFC2833, or SIP Info.



4.4.6 RPort Setting

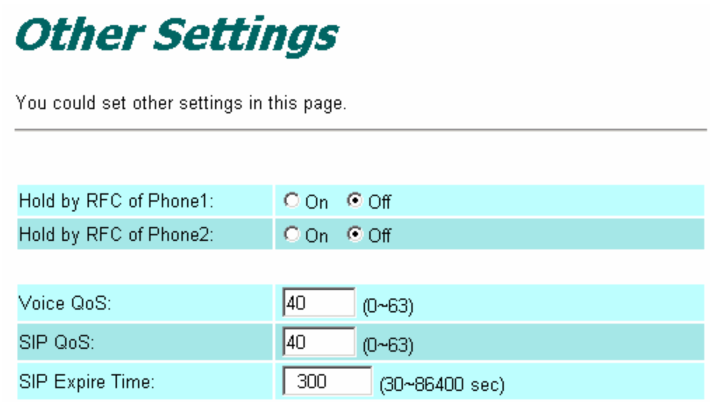
User can set up the RPort Enable/Disable. “RPort” is an extension to SIP for

Symmetric Response Routing. This behavior is not desirable in many cases. Please configure it according to your service provider.



4.4.7 Other Settings

User can set up the Hold by RFC, Voice/SIP QoS and SIP expire time in this page.



Parameter	***Description***
Hold by RFC	Enable or disable
Voice QoS (1-63)	Assign a specific value for the first 6 bits of the IP ToS/DS (DiffServ) field in the Voice (RTP) packet header.
SIP QoS (1-63)	Assign a specific value for the first 6 bits of the IP ToS/DS (DiffServ) field in the Voice (RTP) packet header.

NOTE:

- *The function can discriminate the IP DSCP of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority.*
- *Junction devices (switch or router), within a converged network of voice and data, should support TOS / DiffServ to identify and prioritize voice traffic (QoS higher than 0) over others (QoS= 0).*
- *Setting Voice/SIP QoS as 46, voice is transmitted to Internet in the service class of*

EF (Expedited Forwarding).

SIP expire time	the time used to inform proxy server of the valid duration of registration information.

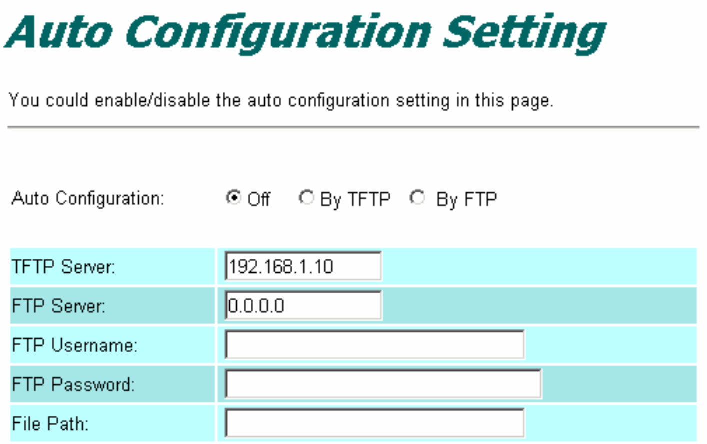
4.5 NAT Trans. / STUN

Please refer to the section ***2.3 SIP Configuration***.

4.6 Others

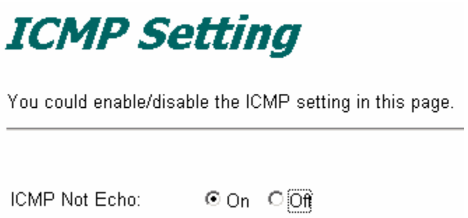
4.6.1 Auto Config

User can disable Auto Configuration or enable the function by TFTP/FTP. Please contact with your service provider for necessary information.



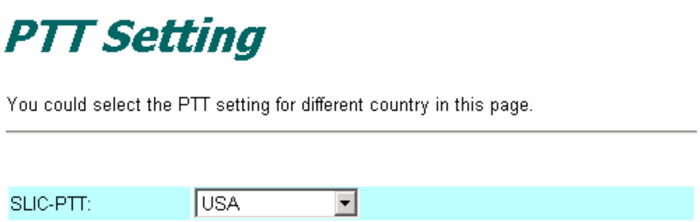
4.6.2 ICMP Setting

User can set the gateway to reply ICMP echo request or not. Setting this function to “ON”, you will get reply when you PING this gateway. Setting this function to “Off”, you get no reply when you PING this gateway.



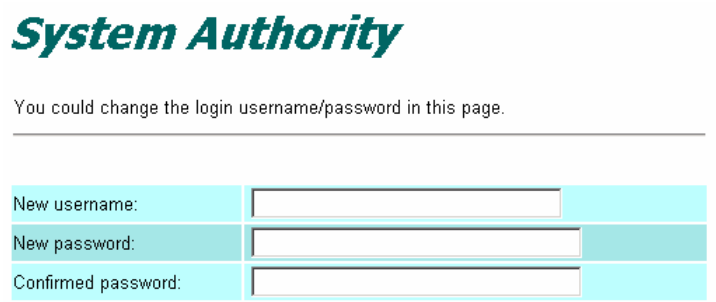
4.6.3 PTT Setting

Select the PTT setting for FXS interface by different country. When you finished the setting, please click on the Submit button.



4.7 System Auth.

Change system login name and password.



4.8 Save Change

Click on the Save button. The system will automatically restart and the new setting will take effect.

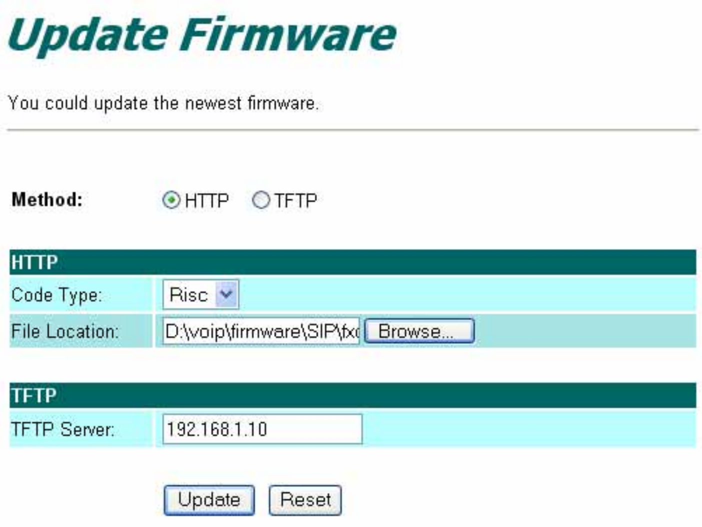
4.9 Update

4.9.1 New Firmware

User can upgrade the system via TFTP or HTTP in this page. Please upgrade the firmware by the following steps:

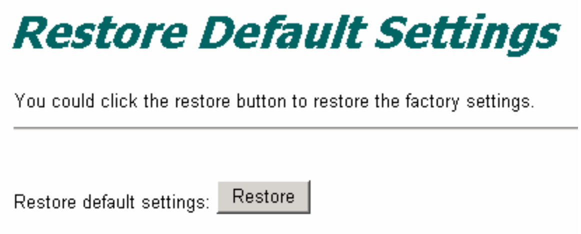
Parameter	***Description***
Method	Upgrade via HTTP or TFTP
Code Type	Select the firmware code type, Risc or DSP code.
File Location	Click on the Browse button to locate the firmware file. Or you can type the correct path and the filename in the field.
TFTP Server	IP address of TFTP server

Click on the Update button to start upgrading.



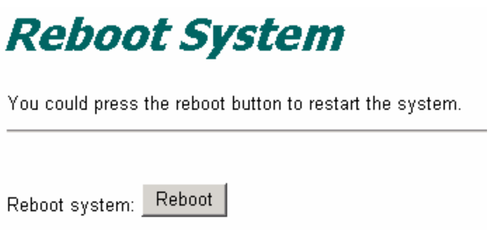
4.9.2 Default Setting

Click on the Restore button. Then, the system will restore factory default setting and automatically restart again. Changed network and SIP setting will be removed.



4.10 Reboot

Press the reboot button. The system will restart automatically.



5. IVR / Keypad Management

You can use the PSTN phone keypad to operate the IP Telephony gateway. Please follow the instruction to configure.

IVR Action	IVR Menu Choice	Parameter(s)	Notes
Dial out from PSTN Line	**0#**	None	Press 0# can pass relay to PSTN Line, user can dial out from PSTN Line. (SP5001D only)
Unlock keypad setting	**#190#**	None	After you unlock keypad setting, then you may configure the ATA.
Reboot	**#195#**	None	After you hear "Option Successful," hang-up. The system will reboot automatically.
Factory Reset	**#198#**	None	System will automatically Reboot. WARNING!! ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Check WAN IP Address	**#126#**	None	IVR will announce the current WAN IP address of the gateway.
Check LAN IP Address	**#120#**	None	IVR will announce the current LAN IP address of the gateway.
Check IP Type	**#121#**	None	IVR will announce if DHCP in enabled or disabled.
Check the Phone Number	**#122#**	None	IVR will announce current in use VoIP number.
Check Network Mask	**#123#**	None	IVR will announce the current network mask of the gateway.
Check Gateway IP Address	**#124#**	None	IVR will announce the current gateway IP address of the gateway.
Check Primary DNS Server Setting	**#125#**	None	IVR will announce the current setting in the Primary DNS field.
Check Firmware Version	**#128#**	None	IVR will announce the version of the firmware running on the gateway.
Set DHCP client	**#111#**	None	The system will change to DHCP Client type
Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad.	DHCP will be disabled and system will change to the Static IP type.

		Use the * (star) key when entering a decimal point.	
Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Enter value-using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Set Handset Gain	**#131+[00~15]#**	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6.
Set Handset Volume	**#132+[00~12]#**	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10.

6. Specification

Model	SP5001C	SP5001D	SP5002A	SP5012A
Standard	IETF SIP (RFC3261)			
Telephone Port	1 FXS	1 FXS 1 PSTN (lifeline)	2 FXS	1 FXS 1 FXO
Ethernet Port	2 10/100M ports for WAN and LAN connection			
Voice	Codec: ● G.711: 64k bit/s (PCM) ● G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) ● G.729A: 8k bit/s (CS-ACELP) ● G.729B: adds VAD & CNG to G.729 CNG, EC (G.168), VAD Adaptive Jitter Buffer Gain (Voice Volume) Settings Provide Call Progress Tone			
DTMF	In-band, SIP Info, RFC2833			
Telephony	Speed dial (10 sets) Call Forward / Call Hold / Call Waiting 3-way Conference Caller ID Display (DTMF / FSK) Call Block (Do Not Disturb)			
SIP Server	Registrar Server (3 SIP accounts) Outbound proxy			
QoS	IEEE802.1q/p, VLAN & Port prioritization IP ToS / DSCP			
Router	NAT, VPN (PPTP client) DMZ*, Virtual Server*			
NAT Traversal	STUN			
Networking	Static assign, PPPoE, DHCP			
Management	Web / Keypad (IVR)			
Environment	Temperature: 0 - 40 degree C Humidity: 10% to 90%			
Power Supply	12VDC, 1A			
Emission	CE			

* will be available in the later version.

Date: 2006 / 05 / 10