



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 <u>http://www.micronet.info/</u> for additional support.

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# 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:

PWR STATUS TEL LAN WAN

SP5001C IP Telephony Gateway

#### 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		

DC 12V	TEL	LAN	WAN	RESET
				$\bigcirc$

#### 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:

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PWR PSTN TEL LAN WAN

SP5001D IP Telephony Gateway

#### 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

		_			~
DC 12V	P\$TN	TEL	LAN	WAN	7
					RESET
					0

#### SP5001D Rear Panel

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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:

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PWR TEL2 TEL1 LAN WAN

\_\_\_

SP5002A IP Telephony Gateway

#### 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

DC 12Y TEL2 TEL1 LAN WAN RESET						
RESET	DC 12V	TEL2	TEL1	LAN	WAN	
RESET						]
						RESET

#### 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

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SP5012A IP Telephony Gateway
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		

DC 12Y TEL LINE LAN WAN	_						
RESET	DC	127	TEL	LINE	LÂN	WAN	
RESET		$\neg \Gamma$					
							RESET
		11					

#### 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 <u>http://192.168.123.1:9999/</u>. 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 Login.

Login VolP	
Enter your us	ername and password to login
	VoIP server
Username	root
Password	****
	Login Clear

Login Screen

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

Micronet		<b>System</b> . This page illustrate the	Information e system related information.
		Model Name:	VolP
Phone Book		Firmware Version:	Fri Apr 28 11:10:19 2006.
		Codec Version:	Fri Jan 20 17:19:16 2006.
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 <u>DHCP client</u>, 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.

LAN Mode:	C Bridge
WAN Setting	
IP Type:	C Fixed IP Client C PPPoE
IP:	0.0.0.0
Mask:	0.0.0.0
Gateway:	0.0.0
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	001122334455

PPPoE Setting	
User Name:	
Password:	

Parameter	Description
LAN Mode	Bridge: pure VoIP gateway
	NAT: VoIP router
ІР Туре	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.

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# 2.3 SIP Configuration

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

# Service Domain Settings

You could set information of service domains in this page.

Phone No.: Phone 1 💌				
Realm 1 (Default)				
Active:	● On C Off			
Display Name:	SP5002A-1			
User Name:	50011			
Register Name:	50011			
Register Password:	****			
Domain Server:				
Proxy Server:	sip.micronet.com:5060			
Outbound Proxy:	nat.micronet.com:5082			
Status:	Not Registered			

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></sip_proxy_port>		
Outbound Proxy	Outbound proxy IP and Port		
<outbou< td=""><td>nd_proxy_ip&gt;:&lt; outbound_proxy_port&gt;</td></outbou<>	nd_proxy_ip>:< outbound_proxy_port>		
Status Registered or not.			

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



\_\_\_\_\_

You could set the IP of STUN server in this page.

\_\_\_\_\_

STUN of Phone1:	⊙On ○Off
STUN of Phone2:	⊙On OOff
STUN Server:	66.7.238.210
STUN Port:	3478 (1024~65535)
	Submit Reset

**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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# 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).

	PPTP Settings					
	You could set the F	PTP s	erver in this page.			
	PPTP:		⊙ On ◯ Off			
	PPTP Server:		220.132.174.242			
	PPTP Username:		admin			
	PPTP Password:		••••			
			Submit Reset			
Parame	ter	Des	cription			
PPTP S	Server	Set	IP address of PPTP server			
ΡΡΤΡ	<sup>o</sup> TP Username Set username for authentication to set up VPN tunne		username for authentication to set up VPN tunnel			
PPTP F	PTP 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	
Туре:	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	
Туре:	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	
Туре:	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.

guration of DDNS in this page.
©On COff
micronet01.dyndns.org
micronet01
*****
members. dyndns. org 💌
dyndns 💌
on 💌
COn ©Off
C On C Off

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

# 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> + #</telephone_number>	
No	ote:		
•	Once the first registrati	ion fails, the second realm will be activated	d.
•	The gateway can alwa	ys receive incoming call from the client of	either registered
•	The gateway is access DDNS setting is neces	ible for P2P client to dial the configured tel sary when WAN IP address is dynamic.	lephone numbers.

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#### 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.

	VoIP Unregistered:
	<ul> <li>Press <pstn_number></pstn_number></li> </ul>
Make PSTN Call	VoIP Registered:
	• Press <b>0</b> * to switch to PSTN mode
	<ul> <li>press <pstn_number></pstn_number></li> </ul>

# 3.2 Speed Dial / P2P call

Speed dial Press <index> + #</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			Γ
5			Γ
6			Γ
7			
8			Γ
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></peer_device_ip_address>
<	pstn_number> @ <peer_fxo_gateway_ip_address></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.

All Forward:	O Off	🖸 On			
Busy Forward:	<ul> <li>Off</li> </ul>	🔿 On			
No Answer Forward:	<ul> <li>Off</li> </ul>	O On			
	1	lame		URL	
All Fwd No.:	sp5100		51001		
Busy Fwd No.:					
No Answer Fwd No.:					

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

Allas of the forwarding humber

- 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	• Establish a call
Conference	• 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 <u>4.2.7 Flash Time Setting</u>.

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Call waiting function has to be enabled. Please refer to the section <u>4.2.8 Call</u>
 <u>Waiting Setting</u>.

# 3.5 FAX

Send FAX	Press <fax_number> to connect fax ma</fax_number>	achine
	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.

T.38 (FAX) Setting		
You could enable/disable the FAX function in this page.		
T.38 (FAX):	🖲 On 💭 Off	
T.38 Port of Phone1:	61000 (1024~65533)	
T.38 Port of Phone2:	61100 (1024~65533)	

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

-----

**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 <u>4.4.3 Codec Settings</u>.

------

# 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.

# SNTP Settings

You could set the SNTP servers in this page.	
SNTP:	ତOn COff
Primary Server:	time.windows.com
Secondary Server:	208.184.49.9
Time Zone:	GMT + 🕶 08 💌 : 00 💌 (hh:mm)
Sync. Time:	1 : 0 : 0 (dd:hh:mm)

#### 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.

Volume Setting		
You could set the volume of your phone in this page.		
Handset Volume:	10 (0~12)	
Ringer Volume:	6 (0~10)	
Handset Gain:	10 (0~15)	

#### 4.2.4 Block Settings

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

# **Block Setting**

You could set the block period of your phone in this page.

Always Block:
C On C Off
Block Period:
C On C Off
From:
12:00 (hh:mm)
To:
13:30 (hh:mm)

#### 4.2.5 Caller ID

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

# Caller ID Setting You could enable/disable the caller ID setting in this page. Caller ID: Caller ID: Caller ID: Organized Content of the setting of th

#### 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  $\rightarrow$  005

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

#### Dial Plan: \*xx + #xx + 11x + xxxxxxx

Rule	Allowable input number
*xx	*01, *02, *15, *89,
#xx	#11, #28, #96
11x	111, 112, 113,, 119, 110
xxxxxxx	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.

# Flash Time Setting

You could set the flash time in this page.		
Max Flash Time:	60 (4~255, 1->10ms)	

#### 4.2.8 Call Waiting Setting

User can enable Call Waiting function.



You could enable/disable the call waiting setting in this page.

Call Waiting: On O Off

# 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	
Туре:	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	
Туре:	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 C 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.



You could set the VLAN settings in this page.		
VLAN Packets:	O Off	• On
VID:	136	(2 ~ 4094)
User Priority:	5	(0 ~ 7)
CFI:	0	(0 ~ 1)
NAT VLAN Setting		
VLAN Packets:	🖲 Off	O On
VID1:	4	(2 ~ 4094), 0->Off
VID2:	5	(2 ~ 4094), 0->Off
VID3:	6	(2 ~ 4094), 0->Off
VID4:	7	(2 ~ 4094), 0->Off

#### 4.3.6 PPTP Settings

Please refer to the section 2.4 VPN Configuration.

#### 4.3.7 DMZ \*

Open another page <u>http://192.168.123.1:9999/dmzset.htm</u>. 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.

DMZ Setting		
You could configure your demilitarized zone setting in this page.		
DMZ:	©On COff	
DMZ Host IP:	192.168.123.8	
	Submit Reset	

#### 4.3.8 Virtual Server \*

Open another page <u>http://192.168.123.1:9999/vsset.htm</u>. 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.



You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].

Virtual Server Page: page 1 💌

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						Γ
4						Γ
5						Γ
6						
7	Γ					Γ

# 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.

# SIP Port of Phone1: 5060 (10~65533) RTP Port of Phone1: 60000 (10~65533) SIP Port of Phone2: 5062 (10~65533) RTP Port of Phone2: 5062 (10~65533)

#### 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.



You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.711 u-law 💌
Codec Priority 2:	G.711 a-law 💌
Codec Priority 3:	G.729 💽
Codec Priority 4:	G.723 💌
Codec Priority 5:	G.726 - 16 💌
Codec Priority 6:	G.726 - 24 💌
Codec Priority 7:	G.726 - 32 💌
Codec Priority 8:	G.726 - 40 💌

RTP Packet Length	
G.711 & G.729:	20 ms 💌
G.723:	30 ms 💌
G.723 5.3K	
G.723 5.3K:	OOn ⊙Off
Voice VAD	
Voice VAD:	OOn ⊙Off

#### 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.

# Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	☑ 23
G726-24 ID:	22 (95~255)	22
G726-32 ID:	2 (95~255)	2
G726-40 ID:	21 (95~255)	☑ 21
RFC 2833 ID:	101 (95~255)	☑ 101

#### 4.4.5 DTMF Setting

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

# DTMF Setting

You could set the DTMF setting in this page.	
• 2833	
C Inband DTMF	
C Send DTMF 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.

RPort Setting		
You could enable/disable the RPort setting in this page.		
RPort of Phone1:	COn . Off	
RPort of Phone2:	COn . COff	

#### 4.4.7 Other Settings

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

# **Other Settings**

You could set other settings i	n this page.
Hold by RFC of Phone1:	COn .€Off
Hold by RFC of Phone2:	On €Off
Voice QoS:	40 (0~63)
SIP QoS:	40 (0~63)
SIP Expire Time:	300 (30~86400 sec)

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.

<b>Auto Configuration Setting</b> You could enable/disable the auto configuration setting in this page.					
TFTP Server:	192.168.1.10				
FTP Server:	0.0.0.0				
FTP Username:					
FTP Password:					
File Path:					

#### 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.

# **ICMP Setting**

You could enable/disable the ICMP setting in this page.				
ICMP Not Echo				
Total Hot Lond.				

#### 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.

PTT Se	etting			
You could select the PTT setting for different country in this page.				
SLIC-PTT:	USA			

# 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.

Update Firmware			
You could updat	e the newest firmware.		
Method:	⊙HTTP ○TFTP		
НТТР			
Code Type:	Risc 👻		
File Location:	D:\voip\firmware\SIP\fx( Browse		
TFTP			
TFTP Server:	192.168.1.10		
	Update Reset		

#### 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.

# Reboot System

You could press the reboot button to restart the system.

Reboot system: Reboot

# 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	Parameter(s)	Notes	
	Choice			
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*x xx*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*x xx*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*x xx*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*x xx*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 (RF	C3261)		
Telephone Port	1 FXS	1 FXS	2 FXS	1 FXS	
		1 PSTN (lifeline)		1 FXO	
Ethernet Port	2 10/100M ports	for WAN and LAN co	nnection		
Voice	Codec:				
	• G.711: 64k k	oit/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.