



**User's Manual**

## **VoIP FXO Gateway**

**Model No.: SP5052A/S, SP5054A/S**

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# 1. Web configuration

The initial version for HTTPD web management interface provides user to configure easily rather than command operating method through RS-232 / TELNET.

The configuration function and step is similar with the way through command line. Basically this version is not the finalized version for web interface. Please refer to the manual for more information. Below provide a simple user guide for user to configure via web interface. Next version for HTTPD web management will not like the command format, but friendly interface.

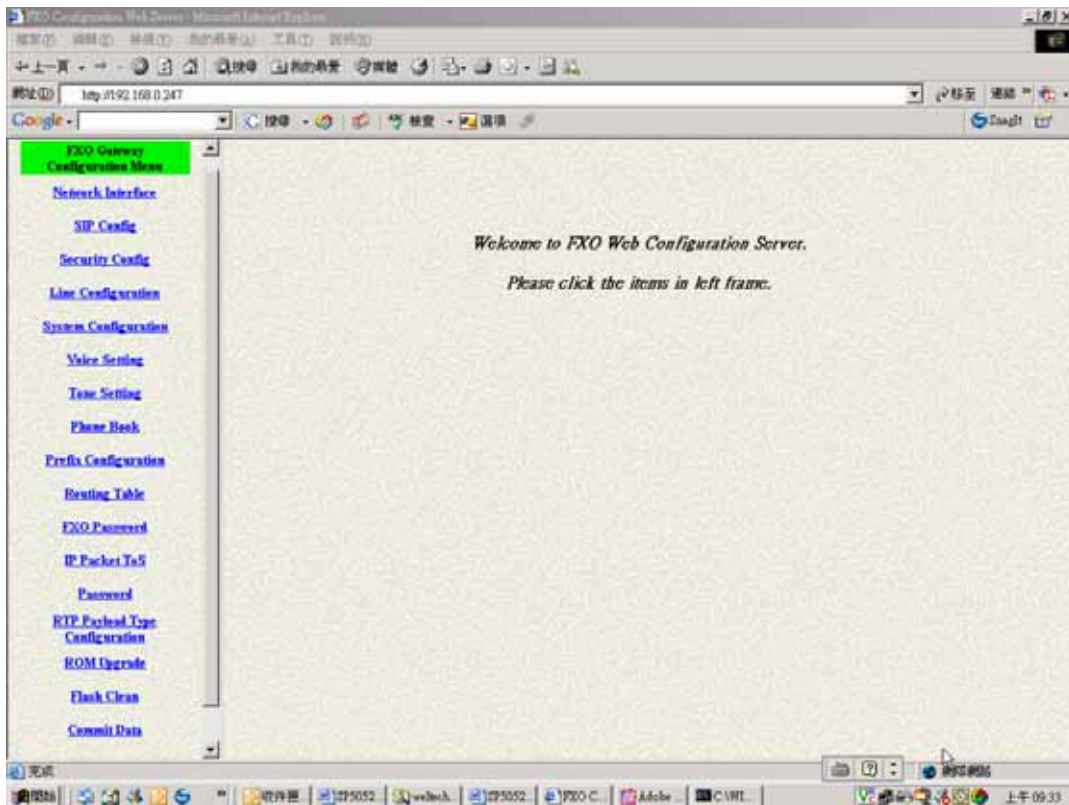
## Step 1. Use Web Browser to Connect Gateway



## Step 2. Input the login name and password

- Login name: root / administrator
- Password: None (just press Enter in default value)

The web interface main screen



### Step 3. Start to Configure

Most of all commands displayed in console / telnet are transfer to web interface. The most important commands are Network Interface, SIP Config and Reboot System. The method is as the same as command mode.

## 1.1 Network Interface

Users have to configure the Network configurations in this page. This gateway will be work while it is connecting with the internet network. Please get more info from the following descriptions. (See figure 3.1)

Network Interface				
IP Address:	192	.168	.1	.247
Subnet Mask:	255	.255	.255	.0
Default routing gateway:	192	.168	.1	.1
IP Mode:	<input checked="" type="radio"/> FIX IP <input type="radio"/> DHCP <input type="radio"/> PPPoE			
HTTP Port:	80			
DNS primary:	168	.95	.192	.1
DNS secondary:	168	.95	.1	.1
SNTP:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable			
SNTP Server Address:	168	.95	.195	.12
GMT:	+8			
IP Sharing:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable			
IP Sharing Server Address:	61	.219	.198	.205
OK				

#### Parameter Description:

- IP Address: Define the ip address for your networking if it is the fixed ip. Please get this info from your ISP.
- Subnet Mask: Define the mask address for your networking. Please get this info from your ISP.
- Default Routing Gateway: Define the default gateway for your networking. Please get this info from your ISP.
- IP Mode: To configure the fixed or dynamic ip address for this unit. Please configure to PPPoE if the ADSL is using the PPPoE type.
- HTTP Port: To configure the HTTP port for access this unit from the remote side.
- DNS primary: To configure the first ip address for the DNS server.
- DNS secondary: To configure the second ip address for the DNS server.
- SNTP: Enable the SNTP server registering function if user wants to get the correct time from the Command Line Interface.
- SNTP Server Address: Enter the correct ip address of the SNTP server or get the incorrect time from the Command Line Interface.
- GMT: Configuring the time area for the time display in the Command Line Interface.
- PPPoE User Name: To configure the user name for the PPPoE connection.
- PPPoE Password: To configure the password for the PPPoE

connection.

- PPPoE IP Address: In the PPPoE mode, this table will show the ip address that this unit gets from the ISP.
- PPPoE Destination: In the PPPoE mode, this table will show the default gateway address that this unit gets from the ISP.
- PPPoE DNS primary: In the PPPoE mode, this table will show the DNS ip address that this unit gets from the ISP.
- After Remote Host Disconnection: This unit will reboot and re-connect to the ISP

## 1.2 SIP Config

This WEB page will help user to configure the information about the dial mode, GK information and some TCP/IP port for the communication. Please get more info about this configuration from the below detail descriptions. (See figure 3.5)

SIP Configuration	
Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy <input type="radio"/> Gateway
Primary Proxy IP Address:	10.1.1.2
Primary Proxy port:	5060
Secondary Proxy IP Address:	null
Secondary Proxy port:	5060
Outbound Proxy:	null
Outbound Proxy port:	5060
Prefix String:	null
Line1 Number:	1001
Line2 Number:	1002
Line3 Number:	1003
Line4 Number:	1004
SIP port:	5060
RTP Port:	16384
Expire:	60
<input type="button" value="OK"/>	

Parameter Description:

- Mode – Pick up the calling mode for this gateway.

- Peer-2-Peer: It only supports the peer-to-peer mode and users have to define the phone book for this mode.
- Proxy: Users have to register on the Proxy if users picked up this option.
- Gateway: Only user line1 has to register on the Proxy, then the line1, line2, line3 and line4 join in the same hunting group automatically.
- Primary Proxy IP Address: Enter the proxy ip if users pick up the proxy mode.
- Primary Proxy port: Set Proxy port for SP5050 series to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Secondary Proxy IP Address: Set secondary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to Network Interface). When SP5050 series fail to register to primary Proxy, it will try to register to secondary Proxy, when it fails again, it will retry to register to Primary Proxy.
- Secondary Proxy port: Set the secondary proxy port for every SIP message, default value is 5060.
- Outbound Proxy: This version could support the outbound proxy. Users could define the ip address or domain name in this table.
- Outbound Proxy port: Set outbound Proxy port for SP5050 series to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Prefix String: Users could define this if the registration name was a phonetic alphabet not the numbers.
- Line 1 Number: The phone number of the Line 1.
- Line 2 Number: The phone number of the Line 2.
- Line 3 Number: The phone number of the Line 3.
- Line 4 Number: The phone number of the Line 4.
- SIP port: Users could change the sip port of this unit for the registration.
- RTP port: Users could change the beginning RTP ports in this table.
- Expire: Users could change the expire time for the register message sending.



## 1.3 Security Config

Some proxy will include the security policy. The endpoint may need the user account and password for the registration. If these are necessary, users could put the correct account and password in the correct table.

Security Configuration	
Line1 Account:	<input type="text" value="1001"/>
Line1 Password:	<input type="password" value="****"/>
Line2 Account:	<input type="text" value="1002"/>
Line2 Password:	<input type="password" value="****"/>
Line3 Account:	<input type="text" value="1003"/>
Line3 Password:	<input type="password" value="****"/>
Line4 Account:	<input type="text" value="1004"/>
Line4 Password:	<input type="password" value="****"/>
<input type="button" value="OK"/>	

Parameter Description:

- Line 1 Account: The user name for the line 1 account.
- Line 1 Password: The password for the line 1 account.
- Line 2 Account: The user name for the line 2 account.
- Line 2 Password: The password for the line 2 account.
- Line 3 Account: The user name for the line 3 account.
- Line 3 Password: The password for the line 3 account.
- Line 4 Account: The user name for the line 4 account.
- Line 4 Password: The password for the line 4 account.

**Note:** These configuration settings are provided by Service Provider.

## 1.4 Line Configuration

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table.

Line Configuration							
Line1 (LINE):	Type: FXO	Hunting Group: 1	Hot Line: x	Fwd. Type: Disable	Fwd. Number: x	Registration: Not Registered	Status: Ready
Line2 (LINE):	Type: FXO	Hunting Group: 2	Hot Line: x	Fwd. Type: Disable	Fwd. Number: x	Registration: Not Registered	Status: Ready
Line3 (LINE):	Type: FXO	Hunting Group: 3	Hot Line: x	Fwd. Type: Disable	Fwd. Number: x	Registration: Not Registered	Status: Ready
Line4 (LINE):	Type: FXO	Hunting Group: 4	Hot Line: x	Fwd. Type: Disable	Fwd. Number: x	Registration: Not Registered	Status: Ready
OK							

#### Parameter Description:

- Type: Show the type of this port. There are only one types of this gateway. (SP5052A/S / SP5054A/S will show the FXS and FXO type. It can not be changed.)
- Hunting Group: Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group.
- Hotline: Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- Fwd. Type: Forward the call to IP. Forward type: 0: disable, 1: uncondition, 2:busy
- Fwd. Number: destination number to which the call is forwarded
- Registration: To show the gateway registered on the Proxy Server or not.
- Status: To show the port is busy or ready.

## 1.5 System Configuration

There are some parameters in the system configurations, please get more detail as following.

System Configuration	
Keypad Type:	<input type="radio"/> In-Band <input checked="" type="radio"/> RFC2833 <input type="radio"/> INFO
Inter Digit Time:	<input type="text" value="3"/>
Ring Time:	<input type="text" value="200"/> ms
Ring Before Answer:	<input type="text" value="1"/>
Flash Time( 1~200, 1.>10ms):	<input type="text" value="30"/>
End of Dial:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
<input type="button" value="OK"/>	

#### Parameter Description:

- Keypad Type: Select different DTMF Keypad Type
- In-Band: The DTMF signal sending by RTP.
- RFC2833: RTP Payload for DTMF Digits
- INFO: The DTMF signal sending by SIP INFO.
- Inter Digit Time: Set the DTMF inter-digit time (second). Parameter defines the maximum amount of time between digits. If a digit is not followed by another within this time limit, then digit collection is terminated.
- Ring Time: It for the ring detection from the PSTN. The ring detection will be failed if users configure it too long.
- Ring Before Answer: Decide how many rings the gateway will pick up the call from FXO ports.
- Flash time: Set the detective flash range in ms.
- End of Dial: Press # key after numbers dialed.

## 1.6 Voice Setting (For Advanced User)

Users could configure the voice codec or gain level in this web page. Please get more detail info from the following description.

Voice Configuration							
Codec Priority	1st G.729	2nd G.729a	3rd G.729b	4th G.729ab	5th G.723.1	6th G.711mu-Law	7th G.711A-Law
Frame Size	G.723 60	G.729 60	G.729a 60	G.711u 40	G.711a 40	G.729b 60	G.729ab 60
G.723 Silence Suppression:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable						
Line1(LINE) Volume:	Voice 30	Input 32	DTMF 23				
Line2(LINE) Volume:	Voice 30	Input 32	DTMF 23				
Line3(LINE) Volume:	Voice 30	Input 32	DTMF 23				
Line4(LINE) Volume:	Voice 30	Input 32	DTMF 23				
Echo Canceller:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable						
Jitter Buffer:	Min. Delay 60			Max. Delay 120			
OK							

#### Parameter Description:

- Codec Priority: This could help users configure the codec priority for using.
- Frame Size: To configure the packet size for the codec that users want.
- G.723 Silence Suppression: To enable the VAD and CNG function for the G.723 codec.
- Volume: Adjust the volume in “Voice” (sending out); “Input” (receiving); “DTMF” (DTMF sending out). Please Note the value is limited.
- Echo Cancel: To enable or disable the echo cancellation function.
- Jitter Buffer: TO configure the Min or Max delay for the Jitter Buffer. The min is from 0ms and the max is 150ms.

**Note:** A large jitter buffer causes increase in the delay and decreases the packet loss. A small jitter buffer decreases the delay but increases the packet loss. The size of the jitter buffer depends on the condition of the network, which varies with time. Typically the packet loss should be less than 10% for a good quality of speech.

## 1.7 Tone Setting

The Tone Setting is for the Tone detecting. The call will be dropped if the pattern of the tone from PSTN side is as same as the pattern in the disconnect tone table. The same result for the Ring Back Tone. User could

define the pattern of the disconnect tone if the disconnect tone from PSTN side is not the standard tone.

Tone Configuration								
Disconnect Tone 1:	High(freq) 620	Low(freq) 480	High(lev) 8	Low(lev) 8	On1 25	Off1 25	On2 1023	Off2 1023
Disconnect Tone 2:	High(freq) 450	Low(freq) 0	High(lev) 8	Low(lev) 0	On1 35	Off1 35	On2 1023	Off2 1023
Disconnect Tone 3:	High(freq) 620	Low(freq) 480	High(lev) 8	Low(lev) 8	On1 50	Off1 50	On2 1023	Off2 1023
Disconnect Tone 4:	High(freq) 620	Low(freq) 480	High(lev) 8	Low(lev) 8	On1 50	Off1 50	On2 1023	Off2 1023
Remote Ring Back Tone 1:	High(freq) 480	Low(freq) 440	High(lev) 13	Low(lev) 13	On1 100	Off1 200	On2 1023	Off2 1023
Remote Ring Back Tone 2:	High(freq) 480	Low(freq) 440	High(lev) 13	Low(lev) 13	On1 100	Off1 300	On2 1023	Off2 1023
Remote Ring Back Tone 3:	High(freq) 480	Low(freq) 440	High(lev) 13	Low(lev) 13	On1 100	Off1 400	On2 1023	Off2 1023
Remote Ring Back Tone 4:	High(freq) 480	Low(freq) 440	High(lev) 13	Low(lev) 13	On1 100	Off1 200	On2 1023	Off2 1023
OK								

Parameter Description:

- Disconnect Tone 1 – 4: To configure the frequency, level gain and on/off time for the busy tone from PSTN or PABX side. The busy tone supports 4 tables.
- Remote Ring Back Tone: To configure the frequency, level gain and on/off time for the ring tone from PSTN side. The gateway won't connect the calls if the ring tone value is incorrect.

## 1.8 Phone Book

The Phone Book configuration is only support the gateway in Peer-to-Peer mode. Please refer the chapter 2 about the Peer-to-Peer mode.

Phone Book					
Index	Name	E164	IP Address	Drop	Insert

New Record					
Index	Name	E164	IP Address	Drop Prefix	Insert Prefix
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input checked="" type="radio"/> Disable <input type="radio"/> Enable	<input type="text"/>
<input type="button" value="Add Data"/>		<input type="button" value="Delete Data"/>			

Parameter Description:

- Index: The list number of the Phone Book.
- Name: The name for this contact number.
- E164: The dialing number for the calling side.
- IP Address: The destination IP address for this phone number.
- Port: The call signal port of the destination.
- Drop: Support the drop function. Enable is for enable this drop function; Disable is for disable this drop function. The Drop Prefix will drop the E164 number, which you had configured in the E164 table.
- Insert: Support the insert digits function.

**Note:**

- It will be the drop function if user enable the Drop Prefix function and put nothing into the Insert Prefix table.
- It will be the insert function if user disable the Drop Prefix function and put the digits into the Insert Prefix table.
- It will be the replace function if user enable the Drop Prefix function and put the digits into the Insert Prefix table.

## 1.9 Prefix Configuration

The Prefix function is using the drop and inserts function.

Prefix Drop/Insert Configuration			
Index	Prefix	Drop	Insert

New Prefix			
Index	Prefix	Drop	Insert
<input type="text"/>	<input type="text"/>	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	<input type="text"/>
<input type="button" value="Add Data"/>		<input type="button" value="Delete Data"/>	

Parameter Description:

- Index: The list number of the Phone Book
- Prefix: The prefix number of the whole numbers that could be into this gateway
- Drop: The drop function. Enable this function by the Enable button; Disable this function by the Disable button
- Insert: The insert function. Users could enter the digits that you want to insert in this number
- Add Data: Press this button if users fill the entire information table above
- Delete Date: If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted

This function is just like the Phone Book configuration. But it will make the drop and insert function in the GK routed mode. All the numbers into this gateway will check out the prefix table first and find out the destination in the Routing Table.

There is an example about the configuration, please follow up these steps.

1. Press the Prefix Configuration button to enter the configuration table.
2. Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert.

The screenshot shows the 'FXO Gateway Configuration Menu' on the left with 'Prefix Configuration' selected. The main area displays two tables:

Prefix Drop/Insert Configuration			
Index	Prefix	Drop	Insert

New Prefix			
Index	Prefix	Drop	Insert
1	0	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	886

Buttons: Add Data, Delete Data

The usage is as same as the drop, insert function of the Phone Book.

Input (Prefix)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

3. Press the Prefix Configuration button to reload the configuration table



**FXO Gateway Configuration Menu**

- Network Interface
- SIP Config
- Security Config
- Line Configuration
- System Configuration
- Voice Setting
- Tone Setting
- Phone Book
- Prefix Configuration
- Routing Table
- FXO Password
- IP Packet ToS
- Password
- ROM Upgrade
- Flash Clean
- Commit Data
- Reboot System

### Prefix Drop/Insert Configuration

Index	Prefix	Drop	Insert
1	0	Disable	886

### New Prefix

Index	Prefix	Drop	Insert
<input type="text"/>	<input type="text"/>	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	<input type="text"/>

4. Please Commit it and Reboot the system if the configuration is finished.

## 1.10 Routing Table

### Routing Table Configuration

Index	Prefix	Destination	E.164	Min Digits	Max Digits	Hunt Method
IP Default		FXO	x			
FXO Default		IP	x			

### New Route

Index	Default	Prefix	Destination	E.164	Min Digits	Max Digits	Hunt Method
<input type="text"/>	<input checked="" type="radio"/> FXO <input type="radio"/> IP	<input type="text"/>	<input checked="" type="radio"/> FXO <input type="radio"/> IP	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input checked="" type="radio"/> NONE <input type="radio"/> GROUP <input type="radio"/> ALL

Parameter Description:

- Index: Define the number of this data.
- Prefix: Define the number you dial. You could just define the first

digit of the numbers

- Destination: Define the destination of this rule. There are three directions of the destination.
- E164: Define a right E164 number of the destination you want.
- Min Digits: The minima digits you dial.
- Max Digits: The maxima digits you dial.
- Hunt Method – Enable the Hunt Group function and pick up the hunt type.
  - None – Disable this function
  - Group – The call will search other ports to be the destination with the same group if the origin destination is busy.
  - All – The call will search other ports to be the destination with the same type if the origin destination is busy.

The min and max digits are the range for the number you dial. For example: The min number of digits is 1 and max number of digits is 10. The call will follow this routing if the number I dial is between 1 and 10 digits. If I dial over 10 digits, this call will follow the default routing.

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table.

In fact, there are two directions of the incoming calls (from IP or FXO side). The explanation of the default routing is as below:

Incoming calls location	Destination	Explanation
IP (Default)	FXO	The destination will be the FXO port when the calls from the IP side without any define rules.
FXO (Default)	IP	The destination will be the IP side when the calls from the FXO port without any define rules.

The most important usage is for the one-stage-dialing function. For the one-stage-dialing function under the Proxy mode, users have to make sure about that the Proxy could support some kind of the function just like the routing.

## 1.11 FXO Password

FXO Password Configuration	
Index	Password

New Password	
Index	Password
<input type="text"/>	<input type="text"/>
<input type="button" value="Add Data"/> <input type="button" value="Delete Data"/>	

You will get the IVR if you make calls from PSTN side. The IVR will ask you the password you set, and you could make other calls to IP side if the password you type is correct. Please press the FXO Password button to configure the password

### Note:

This function is only for the calls from the PSTN side. It's not ready for the IP side as so far.

## 1.12 IP Packet ToS

TOS Configuration	
Signalling Packet DSCP Code:	<input type="text" value="7"/>
Media Packet DSCP Code:	<input type="text" value="7"/>
<input type="button" value="OK"/>	

The Type of Service needs worked with the network router supported. The router will check all the packets if it supports the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure

these values to make the packets with the correct values for the TOS service from the gateway.

According to the RFC 1349 document, the TOS value as following:

- 1000 – minimize delay
- 0100 – maximize throughput
- 0010 – maximize reliability
- 0001 – minimize monetary cost
- 0000 – normal service

These values are the Binary format. Please change to the Decimal and put these values in to the correct table.

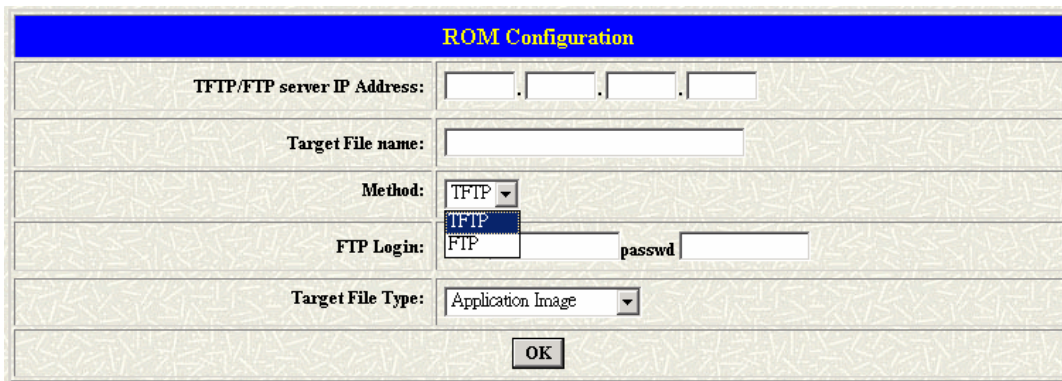
## 1.13 Password



Password Configuration	
root ▼	Current Password: <input type="text"/>
	New Password: <input type="text"/>
	Confirm New Password: <input type="text"/>
<input type="button" value="CHANGE"/> <input type="button" value="ABORT"/>	

There are two accounts for login to access or change the configurations. One is “root”, another is “administrator”. Users could define the password for these two login account. The account “root” could make all the configurations back to the default setting, but the account “administrator” couldn’t. This is the difference between these two accounts. Users could define the password for the accounts in this page.

## 1.14 ROM Upgrade



The image shows a 'ROM Configuration' dialog box with a blue title bar. It contains several input fields and a dropdown menu. The fields are: 'TFTP/FTP server IP Address' (four separate input boxes for IP address), 'Target File name' (a single text input box), 'Method' (a dropdown menu with 'TFTP' selected), 'FTP Login' (two text input boxes for login name and password), and 'Target File Type' (a dropdown menu with 'Application Image' selected). An 'OK' button is located at the bottom center of the dialog.

User could update the firmware just by the web configuration interface. There are two types for the upgrading procedure. One is using the TFTP server; another is using the FTP server.

Parameter Description:

- TFTP Server IP Address: Set TFTP server IP address
- Target File name: Set file name prepared to upgrade
- Method: Select download method as TFTP or FTP
- FTP Server IP Address: Set FTP server IP address
- FTP Login: Set FTP login name and password
- Target File Type: Select which sector of Gateways to upgrade

## 1.15 Flash Clean



The image shows a 'Flash Clean' dialog box with a blue title bar. It contains a red warning message: 'FXO Gateway will be reset to factory default values.' Below the message is a 'CLEAN' button.

Press CLEAN will clean all configurations of Gateways and reset to factory default value.

The password of the account and the networking configuration couldn't be back to the default setting by this command.

## 1.16 Commit Data



Press **COMMIT** when any configuration has been changed before reboot

## 1.17 Reboot System



After **commit** configuration, user has to **REBOOT** device to be able run the configurations properly.

## 2. Command list

Command Line user interface is provided via out-of-band console access and telnet.

### Console Access

- Launch the HyperTerminal program, then the following windows pop-up on the screen. (START – Program files – Accessories – Communication – HyperTerminal)



- Define a name such as 'SP5052A' for this new connection.



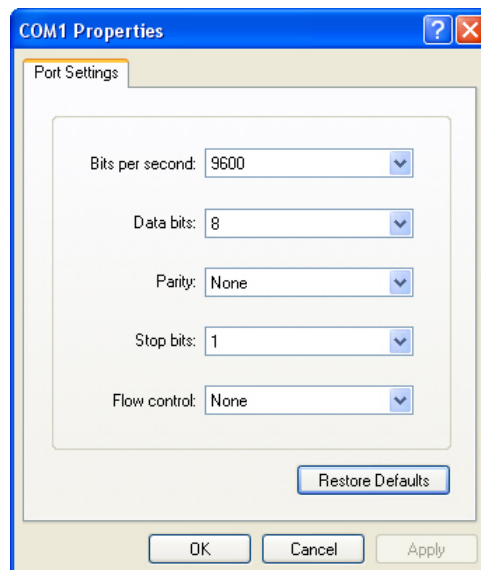
- After pressing OK button, the next window popping up is necessary to connect choose COM Port.



***Note:***

**Make sure the gateway serial cable is connected to correct COM port as the terminal selected, try the different port if failed.**

- Configure the COM Port Properties as following:
  - (1) Bits per second: 9600
  - (2) Flow control: None



Press 'OK' button, and start configuring Gateway.

- Power on the gateway, the message shows while booting. About 40 seconds for complete boot-up.



```
SP5050S - Hyper Terminal
File Edit View Call Transfer Help

Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

Please wait while system is initializing ..... SAttached TCP/IP interface t
o cpm unit 0
Attaching interface lo0...done

HTTPD initialized...
VoicePacketizermain comming

AC4804[0] is ok
AC4804[1] is ok
AC4804[2] is ok
successful 3 2
Initialize OSS libraries... WorkMode : PROXY_MODE
incoming InitCallArray....REAL_MAXCALL=6
SIP stack was constructed successfully. Version - 2.2.1.8
Start registering to Proxy server
0x7ac010 (tNetTask): arp info overwritten for c0a80001 by 00:08:54:d0:02:40
OK!
YP v1.44 stack open sucessfully.

login:
```

## Telnet

Please set your PC with IP address (10.1.1.x), mask (255.255.0.0), and default gateway (10.1.1.254). Click on **START** → **RUN** → **telnet 10.1.1.3**.

```
C:\WINNT\system32\telnet.exe

login: _
```

Then, login with username (**root**) and password (**no password**).

## 2.1 [help]

Type help or man or ? to list all the available command.

---

usr/config\$ ?

help	help/man/? [command]
quit	quit/exit/close
debug	show debug message
reboot	reboot local machine
flash	clean configuration from flash rom
commit	commit flash rom data
ifaddr	internet address manipulation
time	show current time
ping	test that a remote host is reachable
sysconf	System information manipulation
sip	sip information manipulation
security	Security information manipulation
line	Line information manipulation
route	Routing information manipulation
pause	FXO Pause information manipulation
prefix	Prefix drop/insert information manipulation
pbook	Phone book information manipulation
voice	Voice information manipulation
tone	Setup of disconnect tone and remote ring back tone
fxopwd	Setup of FXO password
record	Record voice for greeting and ask pin code
tos	IP Packet ToS (Type of Service)values
pt	DSP payload type configuration and information
rom	ROM file update
passwd	Password setting information and configuration
auth	Set configuration items for "administrator" user.

usage: help [command]

---

## 2.2 [quit]

Type quit will quit the Micronet SP5052A/S / SP5054A/S Gateway configuration mode. And turn back to login prompt.

```
-----  
usr/config$ quit  
Disconnecting...  
login:  
-----
```

Note: It is recommended that type the “quit” command before you leave the console. If so, Micronet SP5052A/S / SP5054A/S Gateway will ask password again when next user connects to console port.

## 2.3 [debug]

Open debug message will show up specific information while Micronet SP5052A/S / SP5054A/S Gateway is in operation. After executing the debug command, it should execute command debug -open as well. One example is demonstrated below.

Debug message information and configuration

```
-----  
usr/config$ debug
```

Debug message information and configuration

Usage:

```
debug [-add type1 [[type2]...]] | -open | -close | -status
```

-status	Display the enabled debug flags.
-add	Add debug flag.
-delete	Remove specified debug flag.
-open	Start to show debug messages.
-close	Stop showing debug messages.

Example:

```
debug -add sip fsm  
debug -add vp  
debug -open  
-----
```

### Parameters Usage:

-status	Display the enabled debug flags.
-add	Add debug flag.

- sip: sip related information
- vp : voice related information
- delete Remove specified debug flag.
- open Start to show debug messages.
- close Stop showing debug messages.

## 2.4 [reboot]

After commit command, type **reboot** to reload Micronet SP5052A/S / SP5054A/S Gateway in new configuration.

## 2.5 [flash]

This command will clean the configuration stored in the flash rom and reboot Micronet SP5052A/S / SP5054A/S Gateway in factory default setting.

---

Flash memory information and configuration

Usage:

flash -clean

Note:

This command will clean the configuration stored in the flash and reboot it.

---

**Warning:** Once users execute **flash -clean**, all the configurations of Micronet SP5052A/S / SP5054A/S Gateway will be cleaned. This can only be executed by user who log in with root.

## 2.6 [commit]

Save changes after configuring the Micronet SP5052A/S / SP5054A/S Gateway.

```
-----  
usr/config$ commit
```

This may take a few seconds, please wait....

Commit to flash memory ok!

```
usr/config$  
-----
```

**Note:** Users should use commit to save modified value, or they will not be activated after system reboot.

## 2.7 [ifaddr]

Configure and display Micronet SP5052A/S / SP5054A/S Gateway network information.

```
-----  
usr/config$ ifaddr
```

LAN information and configuration

Usage:

```
ifaddr [-print][[-mode used]][-sntp mode [server]][-cmcenter ipaddress]]
```

```
ifaddr [-ip ipaddress][[-mask subnetmask]][-gate defaultgateway]
```

```
ifaddr [-dns index [dns server address]][-reboot on/off]
```

```
ifaddr [-id username][[-pwd password]][-http http port][[-autodns used]]
```

- print     Display LAN information and configuration.
- ip        Specify ip address.
- mask      Set Internet subnet mask.
- gate      Specify default gateway ip address
- mode      Set ip client service(0=FIX IP, 1=DHCP, 2=PPPoE).
- sntp      Set SNTP server mode and specify IP address.
- autodns   Specify the way to obtain DNS Server (0:Manual/1:Auto).
- dns       Specify IP address of DNS Server.
- timezone  Set local timezone.
- ipsharing  Specify usage of an IP sharing device and specify IP address.
- id        Connection user name for PPPoE.
- pwd       Connection password for PPPoE.
- reboot    Reboot after remote host disconnection.
- echo      PPPoE Echo Request (0=disable, 1=enable).

Note:

SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).

Example:

```
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -mode 1
ifaddr -sntp 1 210.59.163.254
ifaddr -autodns 1
ifaddr -dns 1 168.95.1.1
ifaddr -ipsharing 1 210.59.163.254
```

---

## 2.8 [time]

When SNTP function of Micronet SP5052A/S / SP5054A/S Gateway is enabled and SNTP server can be found as well, type **time** command to show current network time.

---

```
usr/config$ time
```

```
Current time is WED JUL 14 16:00:00 2004
```

---

## 2.9 [ping]

Use **ping** to test whether a specific IP is reachable or not.

### Responded

---

```
usr/config$ ping www.yahoo.com
```

```
PING www.yahoo.com (66.94.230.46): 56 data bytes
64 bytes from 66.94.230.46: icmp_seq=0. time=195. ms
64 bytes from 66.94.230.46: icmp_seq=1. time=195. ms
64 bytes from 66.94.230.46: icmp_seq=2. time=195. ms
64 bytes from 66.94.230.46: icmp_seq=3. time=195. ms
----www.yahoo.com PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 195/195/195
```

---

## No Responded

---

```
usr/config$ ping 192.168.0.20
```

```
PING 192.168.0.20: 56 data bytes  
no answer from 192.168.0.20
```

---

## 2.10 [sysconf]

This command displays the system information and configuration.

---

```
usr/config$ sysconf
```

```
sysconf [-idtime digit][-keypad dtmf]  
        [-rba digit][-eod digit][-bl digit]  
        [-localrbt digit]  
        [-ring on_time off_time]
```

```
sysconf -print
```

-print	Display system overall information and configuration.
-idtime	Inter-Digits time.(1~10 sec)
-keypad	Select DTMF type: 0=In-band, 1=RFC2833.
-callerid	CallerId Type .(Caller type, 0: none, 1: FSK, 2: ETSI)
- dtmfstart	DTMF CallerID Start Symbol
-dtmaend	DTMF CallerID End Symbol
-ring	The ring time for ring detection.(Uint:ms)
-delay	The FXO dial DTMF delay.(1~9)(Uint:s)
-rba	the number of ring times before answer.(1~5)
-eod	End of dial.(Enable:1 / Disable:0)
-connect	Auto connect time.(0=Disable, for 1~65535 sec)
-onhook	Auto ON-HOOK if detect reverse.(Enable:1 / Disable:0)
-billing	Billing.(0=none, 1=reverse, 2=billing tone)
-Int	General IVR in FXO.(Enable:1 / Disable:0)
-silence	Silence Detection.(0=Disable, 1=Enable)
-ivr	General IVR in FXO.(Enable:1 / Disable:0)

-flashtime      Flash time setting.(1-200, 1->10ms)

Example: sysconf -ring 500

---

**Parameter Usages:**

-idtime      set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed. (1-10 seconds)

-keypad      keypad type when relay DTMF signal.

0 → In-Band

1 → RFC2833.

1. *number (instead of Line number of FXO Line)+ PSTN number to make a call to PSTN side connected with FXO Line.*
2. *After gateway-prefix-drop function is enabled, user must remember to re-configure line number of FXS Line, because line number of FXS Line must remove prefix number. For example, origin line number of FXS line is 1001, prefix is 100, since prefix number will be drop, once gateway has incoming call 1001, after drop gateway prefix 100, it will search line number "1". So line number must be set as "1".*

-callerid      Support FSK and ETSI caller ID function. After the first ring destination site, device will send line number as caller ID to called site. It only supports on Hotline mode.

-ring      ring time for ring detection (in ms). When Gateway has incoming call from PSTN side to FXO port, Gateway will determine it is a ring but not noise only if it is longer than this ring time.

**Note:** *In Taiwan, the ring time of PSTN usually is 1000ms, so if user set ring time longer than 1000ms, FXO port may not be able to pick up the call from PSTN side.*

-delay      When FXO port has an incoming call from IP side and signal connection is established, it will wait the dial tone from PSTN or PBX. But sometimes the dial tone from PBX is too late so some errors will occur. Now user can use this command to



change the time waiting for dial tone.

- rba            set how many rings the gateway will answer the call  
(1 ~ 5 rings)
- eod           End of dial.(Enable:1 / Disable:0)
- connect       The unit will send the connect message to the IP side  
automatically. This function just supports the  
one-stage-dialing function.

## 2.11    [sip]

This command is to configure SIP related parameters.

-----  
usr/config\$ sip

SIP stack information and configuration

Usage:

sip [-mode pxmode]

sip [-px address] [-px2 address] [-outpx address]

sip [-pxport number] [-px2port number] [-outpxport number]

    [-expire t1] [-prefix prefixstring] [-line number]

sip -print

- print        Display SIP stack information and configuration.
- mode         Configure as Peer-to-Peer mode:0/Proxy mode:1/Gateway:2.
- px           Primary Proxy server address. (IPv4 address or dns  
              name)
- px2          Secondary Proxy server address. (IPv4 address or dns  
              name)
- pxport       Proxy server port.        (the port of proxy)
- px2port      Secondary Proxy server port. (the port of Secondary  
              proxy)
- outpx        OutBound Proxy server address. (IPv4 address or dns  
              name)
- outpxport    OutBound Proxy server port. (the port of OutBound  
              proxy)
- prefix       Specify prefix string, use it when UserID contains  
              Alphabets (if UserID uses numerals, specify as null)
- line1        Line 1 is E.164 number of L1.

-line2 Line 2 is E.164 number of L2.  
 -line3 Line 3 is E.164 number of L3.  
 -line4 Line 4 is E.164 number of L4.

-pbsearch Search phone book 0: off/1: on.  
 -expire The relative time after which the message expires (0 ~ (2^31-1))  
 -port SIP local UDP port number (1~65534), Default: 5060  
 -rtp RTP port number (1~65534), Default: 16384  
 -lmedia Late Media 0:off/1:on.  
 -cidchange CID Change in From Header 0:off/1:on.  
 -busy busy response message, Default: 486

Example:

```
sip -px 210.59.163.171 -line1 70 -line2 71
```

---

## 2.12 [security]

This command is to configure proxy related parameters. Proxy provides a simple security function.

---

```
usr/config$ security
```

Security information and configuration

Usage:

```
security [-name username] [-password password]
```

```
security -print
```

-print Display system account information and configuration.  
 -line Specify which line number you want to set the account.  
 -name Specify user name.  
 -password Specify password.

Example:

```
security -line 1 -name kkk -password 12345
```

---

### Parameters Usage:

-print print current proxy related information and configurations.  
 -line specify which line to be configured.  
 -name to specify proxy user name.  
 -password to specify proxy user password.

## 2.13 [line]

Configure the Hotline, Hunt and No Answer Forward

-----  
usr/config\$ line

Gateway line information and configuration

Usage:

line -config number [hunt number][hotline number]

line -print            Gateway line information.

    hunt            Hunting group.

    hotline        Hot line configuration.

    fwdtype        forward type.

                  To IP:

                          0:disable

                          1:un condition

                          2:busy

    forward        forward.

Example:

    line -config 1 hunt 1 hotline 1003 forward1002

-----  
**Parameter Usages:**

-print            print out all current settings of line

-config          determine which line to configure

-hunt            set hunting group flag of each line.

User can assign different hunt group number represent different hunt group. For example, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 2, they will be determined as 2 different groups. On the other hand, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 1 too, when having incoming call to FXS TEL1, which is busy, this call will be route to FXS Line2. Note: FXO Lines and FXS TELs are treated as 2 different groups, so even they are in the same hunt group, call will only be routed to the same FXS or FXO Lines.

-hotline        set hotline table.

The Hotline Mode is applied to the gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device.

## Example : Call out from FXO Line

**Proxy Mode Usage:** Set gateway under proxy mode. Create a Hotline table with "line " command. -----

```
usr/config$ line - config 1 hotline 1001
```

-----  
In this example means: if user picks up phone set of FXO Line1, gateway will automatically dial out "1001".

**P2P Mode Usage:** Set gateway under P2P mode. Create phone book table with "pbook " command. Create a Hotline table with "line " command.

```
usr/config$ pbook - add name micronet ip 10.1.1.1 e164 1001
usr/config$ line - config 1 hotline 1001
```

-----  
In this example means: if user dials into FXO Line1, gateway will automatically dial out IP address of "1001".

## 2.14 [route]

Routing table for your calls

```
usr/config$ route
```

Routing table information and configuration

Usage:

```
route -add [prefix number][dst number][e164 number]
           [min number][max number][hunt number]
```

```
route -delete index
```

```
route -modify index [prefix number][dst number][e164 number]
                    [min number][max number][hunt number]
```

```
route -ip [dst number][e164 number]
```

```
route -fxo [dst number][e164 number]
```

```
route -print Routing table information.
```

prefix	The prefix of dialed number.
dst	Destination port(FXO:1/IP:2).
e164	Destination e164 number(when destination is FXO).
min	Min digits.(0 ~ 255)
max	Max digits.(0 ~ 255)
hunt	Hunt method for busy forward

(NONE:0/GROUP:1/ALL:2).

Example:

```
route -add prefix 100 dst 1 e164 1001 min 1 max 3 hunt 1
route -ip dst 1 e164 1001
route -fxo dst 2 e164 x
route -modify 1 prefix 100 dst 2 e164 1001 min 1 max 3 hunt 1
route -delete 1
```

---

### Parameter Usages:

-print	print out all routing table information
-add	add a routing rule in routing table. User can add less than 50 rules. (route –add prefix "prefix number "dst" destination port type "e164" e.164 number of port "min" minimum digits needed "max" maximum digits can't be exceeded")
-delete	delete a routing rule in routing table (route –delete "index of routing rule")
-modify	modify a routing rule in routing table. (route –modify "index of routing rule" prefix "prefix number "dst"destination port type "e164" e.164 number of port "min" minimum digits needed "max" maximum digits can't be exceeded")
-ip	create routing table for incoming call from IP side. (route –ip dst "destination port type" e164 "e.164 number of port")
-fxo	create routing table for incoming call from FXO Lines. (route –fxo dst "destination port type" e164 "e.164 number of port")
prefix	prefix of dialed number
dst	destination port, 1 means FXO Lines 2 means IP side x means no determinate number.
e164	destination e.164 number.
min	minimum digits needed.
max	maximum digits needed.
hunt	set hunt method for busy forward. 0 means no hunting, 1 means hunting method follows the rule of [line], 2 means hunting method is to hunt between all ports in the same type.

- Note:**
1. When destination is IP side, e.164 number doesn't need to determine.
  2. If user doesn't want to determine a specific port to route, e.164 number must set as "x".

## 2.15 [prefix]

For drop or insert digits

```
usr/config$ prefix
Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print      Prefix drop/insert information.
                  prefix      The prefix of dialed number.
                  drop        Drop prefix(Enable:1/Disable:0).
                  insert      Insert digits.
```

Example:

```
prefix -add prefix 100 drop 1 insert 2000
prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -delete 1
prefix -modify 1 prefix 100 drop 0 insert 300
```

### Parameter Usages:

- add      add a rule to drop or insert prefix digits of incoming call.(prefix –add prefix “prefix number” drop 0/1 insert “insert number”)
  - delete   delete a rule to drop or insert prefix digits of incoming call. (prefix –delete prefix “prefix number”)
  - modify   modify a rule to drop or insert prefix digits of incoming call. (prefix –modify prefix “prefix number” drop 0/1 insert “insert number”)
- 
- prefix    set which prefix number to implement prefix rule.
  - drop      enable or disable drop function. If this function is enabled,

Gateway will drop prefix number on incoming call.  
insert set which digit to insert on incoming call.

## 2.16 [pause]

Pause function allows users define a prefix for FXO, it usually apply to one-stage-dialing. For example, the FXO port is connect to a PBX, when an incoming call from IP side, users will hear a dial tone from PBX. If they want to dial to a PSTN, they must press a special code and wait 1~2 seconds for the PSTN dial tone. But in one-stage-dialing application, the FXO will not wait for the dial tone and it will dial immediately. Now user can define a special prefix, so if FXO detect the prefix, it will wait a moment then keep dialing.

---

usr/config\$ pause

Prefix drop/insert information and configuration

Usage:

pause -add [prefix number][delay number]

pause -delete index

pause -modify index [prefix number][drop number][insert number]

pause -print Prefix drop/insert information.

prefix The prefix of dialed number.

delay delay time(second).

Example:

pause -add prefix 100 delay 1

pause -delete 1

pause -modify 1 prefix 101 delay 0

---

### Parameter Usages:

- add add a new record to pause function. When adding a record, users have to specify **prefix** and **delay** seconds to complete the command.
- delete delete a record to pause function.
- print print out current contents of Pause function.

## 2.17 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users do not have to reboot the machine, and the record will be effective immediately.

---

```
usr/config$ pbook
```

Phone book information and configuration

Usage:

```
pbook [-add [name string][e164 number][ip address]
      [drop digit][insert number]]
      [-modify number [name string][e164 number][ip address]
      [drop digit][insert number]]
      [-delete number]

pbook -print
```

```
-print      Display phone book information and configuration.
-add        Add new phone book record)
-delete     Delete phone book record
-modify     Modify phone book record.
            name      : 1 ~ 10 characters.
            e164      : 1 ~ 10 digits.
            ip        : IP address.
            drop      : 0:Disable/1:Enable.
            insert    : 1 ~ 10 digits.
```

Example:

```
pbook -add name test e164 1234 ip 192.168.1.10 drop 1 insert 5678
pbook -delete 1
pbook -modify 1 name test e164 5678 ip 192.168.1.10 drop 0
```

---

### Parameter Usages:

-print     print out current contents of Phone Book. Users can also add *index number*, from 1 to 100, to the parameter to show specific phone number.

**Note:** <index number> means the sequence number in phone book. If users do request a specific index number in phone book, Micronet SP5052A/S / SP5054A/S Gateway will give each record a automatic sequence number as index.



- add      add a new record to phone book. When adding a record, users have to specify **name**, **ip**, and **e164** number to complete the command.
- search   search a record in phone book. The searching criteria can be **name**, **ip**, or **e164**.
- delete   delete a specific record. “pbook –delete 3” means delete **index 3** record.
- insert   add a new record and force to assign a specific index number for it.
- modify   modify an existing record. When using this command, users have to specify the record’s index number, and then make the change.

**Phonebook Rules:**

To meet the requirements of communicating with trunk Gateway or other applications, Phonebook has following characteristics to be noticed.

When the destination side is a terminal, for ex: IP Phone or soft phone, e164 number stands for exact destination phone number.

When the destination side is a Gateway, for ex: T1/E1 Gateway, e164 phone number stands only for Gateway prefix. That is to say, users have to continue to dial destination number, following the prefix number. A example is as below:

A → Micronet FXO Gateway

In Phonebook, there’s a record:

<i>Index</i>	<i>Name</i>	<i>IP</i>	<i>E164</i>
1	B_Gateway	192.168.1.2	0

B → E1 trunk Gateway, which connects to PSTN with E1 PRI.

If users want to make a call to PSTN number “82265699”, they have to pickup one of the phone connected to Micronet FXO Gateway, and then dial “082265699”. After receiving the complete dialed number, Micronet FXO Gateway will search for its Phone Book, find “0” as matched prefix, and then dial out to B’s IP address directly with destination e.164 (phone number)

“82265699”. Please be noted that “0” is eliminated from Micronet FXO Gateway itself.

**Note:**

1. Because of above characteristics, users have to take care of the number plan very well to avoid the numbering conflict. If users already defined “0” for specific trunk Gateway, other terminal started with “0” shall be avoided, or the number will be routed to the trunk Gateway defined “0”.
2. If user wants to set 2 sets of similar e164 such as 123 and 1234, please be careful configure 123 first, or it may cause problem when user dial 1234, Micronet FXO Gateway may dials out IP address of 123.
3.
  - (1) If called party is FXO product, please set e164 of pbook as e.164 of called party, and remember to set sysconf –drule in\_drop “e.164”(refer to 5.12.)in called party.
  - (2) If called party is FXS product, please set e164 of pbook as prefix of called party, when dialing to different line of FXS product, please dial line number.

## 2.18 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Micronet SP5052A/S / SP5054A/S Gateway.

-----  
[usr/config\\$ voice](#)

[Voice codec setting information and configuration](#)

[Usage:](#)

[voice \[-send \[G723 ms\] \[G729 ms\] \[G729A ms\] \[G729B ms\] \[G729AB ms\] \[G711U ms\] \[G711A ms\] \]](#)

[\[-volume line \[voice level\] \[input level\] \[dtmf level\]\]](#)

```

[-nscng [G711U used1] [G711A used2] [G723 used3]]
[-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G729] [G729A] [G729B] [G729AB] [G711U] [G711A]

```

```

-print      Display voice codec information and configuration.
-send      Specify sending packet size.
           G.723   (30/60/90 ms)
           G.729   (20/40/60 ms)
           G.729A (20/40/60 ms)
           G.729B  (20/40/60 ms)
           G.729AB (20/40/60 ms)
           G.711U (20/40/60 ms)
           G.711A (20/40/60 ms)
-priority  Priority preference of installed codecs.
           G.723
           G.729
           G.729A
           G.729B
           G.729AB
           G.711U
           G.711A
-volume    Specify the following levels:
           voice volume (0~63, default: 30),
           input gain (0~63, default: 30),
           dtmf volume (0~31, default: 23),
-nscng     No sound compression and CNG. (G.723.1 only, On=1, Off=0).
-echo      Setting of echo canceller. (On=1, Off=0, per port basis).
-mindelay  Setting of jitter buffer min delay. (0~150, default: 90).
-maxdelay  Setting of jitter buffer max delay. (0~150, default: 150).

```

Example:

```

voice -send g723 60 g729 60 g729a 60 g729b 60 g729ab 60 g711u 60 g711a 60
voice -volume 1 voice 20 input 32 dtmf 27
voice -echo 1 1

```

---

### Parameters Usage:

```

-print      print current voice information and configurations.
-send      to define packet size for each codec. 20/40/60ms means to
           send a voice packet per 20/40/60 milliseconds. The smaller
           the packet size, the shorter the time delay. If network is in

```

good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729a codec, while 30/60ms is applicable to G.723.1 codec.

**-priority** codec priority while negotiating with other sip device. This parameter determines the listed sequence in TCS message. The codec listed in left side has the highest priority when both parties determining final codec.

**-volume** There are three adjustable value.

- **voice volume** stands for volume, which can be heard from Micronet SP5052A/S / SP5054A/S Gateway side;
- **input gain** stands for volume, which the opposite party hears. **The maximum number is 35. If the number is over 35, the echo may be happened. Once you increase input gain, the voice volume from PSTN to IP side is increased too.**
- **dtmf** volume stands for DTMF volume/level, which sends to its own Line1 or Line2.

**-nscng** silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only. An example is demonstrated below:

```
-----  
usr/config$ voice -nscng g723 1  
-----
```

**-mindelay** the minimum jitter buffer size. (Default value= 90 ms)  
**-maxdelay** the minimum jitter buffer size. (Default value= 150 ms)

```
-----  
usr/config$ voice -mindelay 90 -maxdelay 150 -optfacor 7  
-----
```

**-echo** activate each canceller (1 = ON; 0 = OFF).

**Note:** be sure to know well the application before you change **voice** parameters because this might cause incompatibility.

## 2.19 [tone]

Disconnecting and Ring Back tone configuration

---

```
usr/config$ tone
```

Disconnect tone and remote ring back tone configuration

Usage:

```
tone [num][freqHi ][freqLo ][freqHiLev][freqLoLev]
      [Tone1ON][Tone1OFF][Tone2ON ][Tone2OFF ]
tone -print Display tone configuration.
      [num] Tone index(1~4:Disconnect tone / 5~8:Remote ring back
           tone).
```

Example:

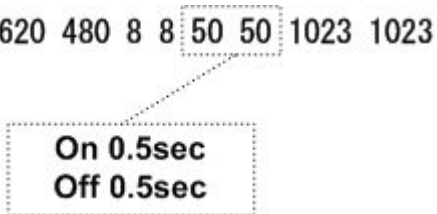
```
tone -print
tone 1 620 480 8 8 50 50 1023 1023
```

---

### Parameters Usage :

tone x 1 - 4 is disconnect tone, 5 - 8 is remote ring back tone.

```
tone 1 620 480 8 8 50 50 1023 1023
```



On 0.5sec  
Off 0.5sec

## 2.20 [fxopwd]

Call restriction from FXO port

---

```
usr/config$ fxopwd
```

FXO password information and configuration

Usage:

```
fxopwd -add [passwd number]
fxopwd -delete index
fxopwd -modify index [passwd number]
```

fxopwd -print      FXO password information.  
          passwd      The password.

Example:

```
fxopwd -add passwd 1234
fxopwd -delete 1
fxopwd -modify 1 passwd 1234
```

---

Caller will be asked PIN number to pass the call through the gateway from FXO port to IP side when **fxopwd** set

## 2.21 [record]

Record your own Greeting, Asking PIN and analyze Disconnect tone

---

```
usr/config$ record
```

Record greeting voice and ask pin code voice, tone analyze.

Usage:

```
record -greeting filename
          -askpin filename
          -tone
```

Example:

```
record -greeting greeting.100
record -askpin askpin.100
record -tone
```

---

### Parameters Usage :

-greeting    record greeting file. User must assign a file name for greeting, once record is finished, file recorded will be display in rom -print.

---

--

```
usr/config$ record -greeting greeting.test
```

*Please off hook the LINE 1 and press (N) for the next step...*

*n*

*Press (R) to start record...*

*r*

---

*Press (S) to stop record...*

.....  
.....  
.....S.....

*Press (P) to play the voice or (W) to write to flash or (Q) to quit...*

*p*  
*w*

*Please wait a moment...*

*Write flash ok...*

*Boot Rom : boot.104*  
*Application Rom : 4fxo.118a*  
*DSP App : 48302ce3.140*  
*DSP Kernel : 48302ck.140*  
*DSP Test Code : 483cbit.bin*  
*Greetings : greeting.test*  
*Ask Pin : askpin.102*

*q*  
*usr/config\$*

---

**-askpin** record askpin file. User must assign a file name for askpin file, once record is finished, file recorded will be display in rom –print.

-----  
--

*usr/config\$ record -askpin askpin.test*

*Please off hook the LINE 1 and press (N) for the next step...*

*n*

*Press (R) to start record...*

*r*

Press (S) to stop record...

.....  
.....  
.....S.....  
.....  
.....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...

p  
w

Please wait a moment...

Write flash ok...

Boot Rom : boot.104  
Application Rom : 4fxo.118a  
DSP App : 48302ce3.140  
DSP Kernel : 48302ck.140  
DSP Test Code : 483cbit.bin  
Greetings : test.100  
Ask Pin : askpin.test

q  
usr/config\$

---

-tone analyze tone frequency. Gateway can analyze tone frequency as users provide tone in FXO Line1.

-----  
--

usr/config\$ record -tone

usr/config\$ record -tone

Press (R) to start record...

r  
.....  
.....  
.....



.....  
.....  
*Analyzing!! Please wait a moment...*  
*Frequency 1 : 620*  
*Frequency 2 : 474*  
*0.25sec on 0.25sec off*  
*usr/config\$*

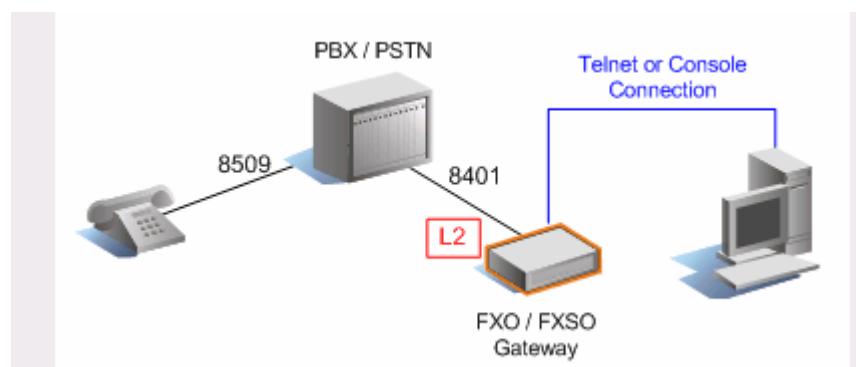
---

- greeting      record the greeting file
- askpin        record the askpin file
- tone          analyze the tone from PSTN or PABX

## The Procedures of recording the disconnect tone

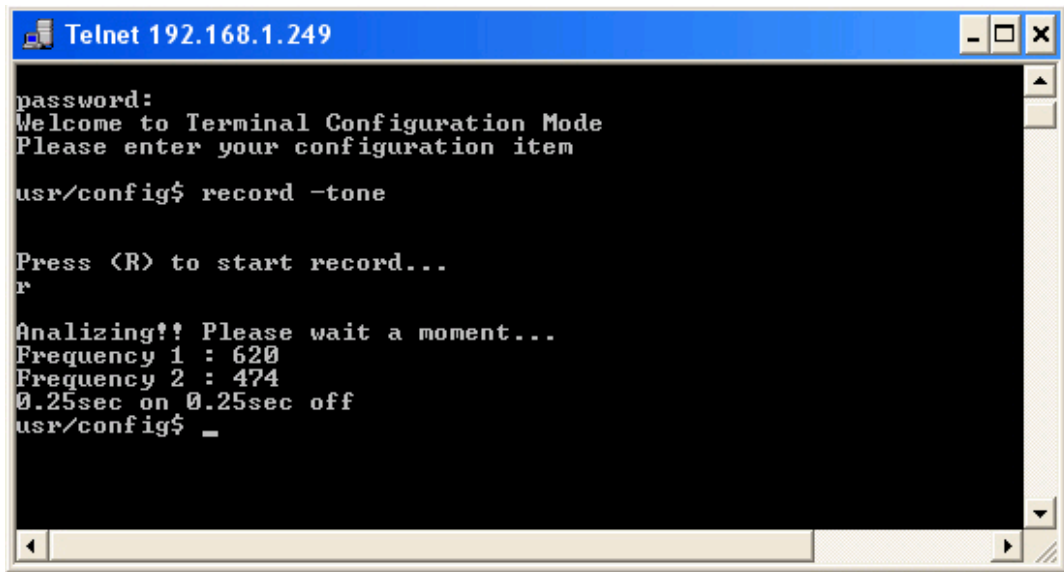
Micronet FXO/FXSO Gateway in the latest version use the "record -tone" command to analyze the disconnect tone. Please prepare connection as the figure shown.

**Note: Please connect extension (or PSTN) line to the second FXO port (L2) of FXO gateway that takes charge of tone detection. For FXS/FXO gateway, however, the first FXO port (LINE1) is in charge.**



Procedure:

- (1). Prepare Telnet or Console connection, and login the system.
- (2). Place command "record -tone", and press "r"...
- (3). Make a call from handset (Ext.8509) to the Ext.8401 into the FXO port.
- (4). After greeting tone, hang up the call from Ext.8509
- (5). Press R key and Enter to start analysis



6). According to the result, enter the Freq. into Tone table.

command:

```
usr/config$ tone 1 480 620 1 1 25 25 1023 1023
```

### 【Example-1】

(Make a call from PSTN to FXO port)

```
usr/config$ record -tone
```

Press (R) to start record...

(Please make sure that you are already finish the steps 2 ~ 7)

r (Press "Enter" button after you key in "R")

.....  
 .....  
 .....  
 .....

Analizing!! Please wait a moment...

(You coule hang up the call from PSTN if you get this message)

Frequency 1 : 481

Frequency 3 : 621

0.25sec on 0.25sec off

```
tone 4 481 621 8 8 25 25 1023 1023
```

(Put this value in to the tone table)

tone -print

Disconnect tone 1 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 2 paramter

Frequency high	: 450
frequency low	: 0
frequency high level	: 8
frequency low level	: 0
Tone1 on	: 35
Tone1 off	: 35
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 3 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 50
Tone1 off	: 50
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 4 paramter

Frequency high	: 621
frequency low	: 481
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 50

Tone2 off : 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

**【Example-2】**

(Make a call into FXO port)

usr/config\$ record -tone

Press (R) to start record...

(Please make sure that you are already finish the steps 2 ~ 7)

r (Press "Enter" button after you key in "R")

.....  
.....  
.....  
.....

Analizing!! Please wait a moment...

(You could hang up the call from PSTN if you get this message)

Frequency 1 : 473

Frequency 2 (2623) is more than 1000, please ignore it.

0. 25sec on 0.25sec off

tone 4 473 473 8 8 25 25 1023 1023

(Please configure the high and low frequency as the same value if you just get a singal frequency)

tone -print

Disconnect tone 1 paramter

Frequency high : 620

frequency low : 480

frequency high level : 8

frequency low level : 8  
Tone1 on : 25  
Tone1 off : 25  
Tone2 on : 1023  
Tone2 off : 1023

Disconnect tone 2 paramter

Frequency high : 450  
frequency low : 0  
frequency high level : 8  
frequency low level : 0  
Tone1 on : 35  
Tone1 off : 35  
Tone2 on : 1023  
Tone2 off : 1023

Disconnect tone 3 paramter

Frequency high : 620  
frequency low : 480  
frequency high level : 8  
frequency low level : 8  
Tone1 on : 50  
Tone1 off : 50  
Tone2 on : 1023  
Tone2 off : 1023

Disconnect tone 4 paramter

Frequency high : 473  
frequency low : 473  
frequency high level : 8  
frequency low level : 8  
Tone1 on : 25  
Tone1 off : 25  
Tone2 on : 50  
Tone2 off : 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

## 2.22 [tos]

TOS service allows users to achieve QoS on IP network.

---

```
usr/config$ tos
IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
    [-rtpreliab mode]
tos -print
```

Example:

```
tos -rtptype 7 -sigtype 0
```

---

In QoS, a modification of the type of service byte. Six bits of this byte are being reallocated for use as the DSCP field, where each DSCP specifies a particular per-hop behavior that is applied to a packet. Support for DSCP still is lacking in some network equipment, please refer to RFC 2474

DSCP : Differentiated Services Code Point

```
00 01 02 03 04 05 06 07 08 09 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31
Ver  P  X    CC    M      PT                Sequence Number
                                Timestamp
                                SSRC
                                CSRC [0..15] ...
```

**Note:**

1. The value of rtptype and sigtype is from 0 to 63.
2. Users should be aware that TOS is effective only when network devices (for ex: router, switch.. etc.) support TOS.

## 2.23 [pt]

RTP Payload Type Configuration

---

usr/config\$ pt

RTP payload type configuration and information

Usage:

pt -print	Display the RTP payload type information
-rfc2833	Configure the DTMF RFC2833 payload type
-dtmf	Configure the DTMF payload type
-fax	Configure the FAX payload type

Example:

```
pt -rfc2833 96 -fax 101
```

---

The RTP (Real-time Transport Protocol) Payload Type

Idea for use the Payload Type:

- **Selective Retransmission :**  
Priority bit indicates the importance of the payload  
Only important packets are retransmitted
- **Multiple Retransmission Attempts :**  
New SNHP value assigned for each transmission of a high priority packet. Client is able to detect a failure in a high priority (re)transmission

For more informations, please refer to your network administrator

## 2.24 [rom]

Firmware Information and Upgrade command

---

usr/config\$ rom

ROM files updating commands

Usage:

```
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp][-greet][-askpin]  
-s TFTP/FTP server ip -f filename
```

rom -print

-print	show versions of rom files. (optional)
-app	update main application code(optional)
-boot	update main boot code(optional)
-boot2m	update 2M code(optional)
-dsptest	update DSP testing code(optional)

- dspcore    update DSP kernel code(optional)
- dspapp    update DSP application code(optional)
- greeting    update greeting voice file(optional)
- askpin    update ask pin code voice file(optional)
- s            IP address of TFTP/FTP server (mandatory)
- f            file name(mandatory)
- method    download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
- ftp        specify username and password for FTP

Note:

This command can run select one option in 'app', 'boot',  
, 'dsptest', 'dspcore', and 'dspapp'.

Example:

```
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
```

---

### Parameter Usages:

- print        show versions of all rom files
- app         application ROM file
- boot        boot ROM file
- boot2m     contains application and boot ROM files together
- s to        specify TFTP server's IP address when upgrading ROM files.
- f to        specify the target file name, which will replace the old one.
- method     to decide using TFTP or FTP as file transfer server.  
              "0" is for TFTP and "1" is for FTP.
- ftp        if users choose FTP in above item, it is necessary to specify  
              pre-defined username and password when upgrading files.

## 2.25 [passwd]

---

```
usr/config$ passwd
```

Password setting information and configuration

Usage:

```
passwd -set Loginname Password
passwd -clean
```

Note:



1. Loginname can be only 'root' or 'administrator'
2. passwd -clean will clear all passwd stored in flash, please use it with care.

Example:

```
passwd -set root Your_Passwd_Setting
```

## 2.26 [auth]

---

```
usr/config$ auth
```

Root control what command administrator can use.

Usage:

auth -print Display auth switch configuration.

Use item name to do config name (0=Disable, 1=Enabled).

Example: auth -ifaddr 1

## 2.27 [setmac]

To set mac address please key in command setmac:(when key in MAC address ,press enter each time after key in two characters). Please get the MAC address from the bottom of your FXO VOIP units.

---

```
usr/config$ setmac
```

- enter mac address (xxxxxxxxxxx):

```
0001a8002baa
```

- the mac address is 00 01 a8 00 2b aa

- if mac address is correct,please press 'y' to  
setup configuration,else press 'n' to continue

```
y
```

---