FXSO GATEWAY
In SIP
User Manual
(1FXS/1FXO or 2FXS/2FXO ports)

Steps in configuration

STEP 1 Start Up	To check out the peripheral equipments and understand the feature of this gateway. Please read this step very carefully before starting the configuring.
STEP 2 How to Setup and Connect basically	Connecting the gateway and computer to start configuring by WEB GUI.
	Setting the ip address for this gateway to make sure that it could connect with the internet.
	Setting the configurations of dialing, including the Peer-To-Peer, GK mode and how to set these tables to make calls by this gateway easily. The other configurations of make call will be discussed in this step.
STEP 3 Advanced	Advanced configurations and special functions of this gateway. Using the WEB GUI to show how to set this table and explain the meaning of these tables.
STEP 4 Command List	To explain the meaning of the command in the command line interface and example the usage of the command. To get more usages or configuration in this step and study about the command line configuration.

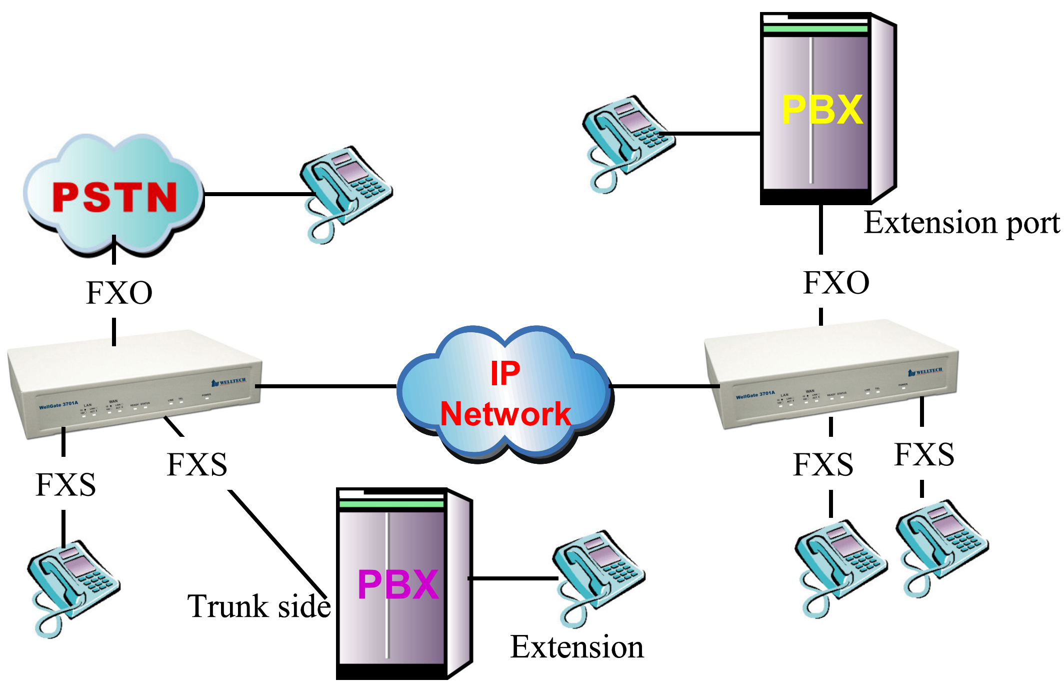
1. Start Up

1.1 Introduction

SIPFXSO 1S1O/2S2O is a one-port/two-port FXS+FXO gateway. It supports an innovative intelligent call rouging function that transparently routes calls to destination either through PSTN or Internet.

SIPFXSO 1S1O/2S2O provides voice over IP and FAX over IP services for ITSP/ISP Internet Telephony Services Provider and Office/SOHO IP-PBX application.

Application Architecture



- ***FXO ports can connect with PSTN Line or Extension Line of PBX***
- ***FXS ports can connect with Phone Set or Trunk Line of PBX***

1.2 Features and specification

Features

- IETF RFC 3261
- Automatically Dial Path Selection (IP or PSTN)
- PSTN Line switch to telephone set when power is failure
- PPPoE support
- Behind NAT router or IP sharing device
- DNS server inquiry
- Provide Peer-to-Peer Mode (Non SIP Proxy needed) selection
- E.164 Dial Plan
- TFTP/FTP software upgrade
- Remote configuration/ reset
- LED indication for system status
- Support Fix IP and DHCP

Audio feature

- Codec -- G.711 a/μ law, G.723.1 (6.3kbps), G.729, G.729A
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Completed voice band signaling support
- Provide In-band or RFC2833 DTMF generation/detection
- Provide call progress tone

Management Feature

- TELNET/Console port and Web Browser configuration

Certification

- UL, CE, FCC

FXS Features

- 2-wire loop start
- Programmable On-Hook voltage, Ring voltage/Cadence/Frequency, Loop current
- Line polarity reversal generation

FXO Features

- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)
- PSTN polarity reversal detection
- Disconnect tone detection
- Asking ping function with the incoming calls from PSTN side
- Record and analyze the Tone from PSTN side

Environmental

- Operation temp:0˚C to 40˚C
- Humidity: 10% to 90% (Non-condensing)

1.3 Accessories and equipment

- The voice gateway in 2 FXS and 2 FXO ports or 1 FXS port and 1 FXO port models and two RJ-45 connector (WAN and LAN).
- The AC adapter.
- The CD of user manual.
- The connection cable in RS-232 interface.

1.4 Appearance

Front panel: The LED lights provide related system messages of the gateway.

SIPFXSO 1S1O

SIPFXSO 2S2O

Power: Light on means Gateway is power on, and vice versa.
TEL: Light on means the line is in use (off-hook), and vice versa.
LINE: Light on means the line is in use (off-hook), and vice versa.
Status:

1. LED light on means Gateway has successfully registered to Proxy when it is in the Proxy Mode.
2. LED flash means Gateway is not registered to the Proxy when it is in the Proxy Mode.
3. Or when Gateway is in downloading mode, LED should be flash as well.
4. LED light off means Gateway is in Peer-to-Peer Mode.

Ready:

1. Light on and slow flash means Gateway is in normal mode.
2. Light on and fast flash means Gateway is in downloading mode.

WAN: Connected to Public Ethernet

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted / received.

LAN: Switch to another device, such as PC

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted received.

Back panel:

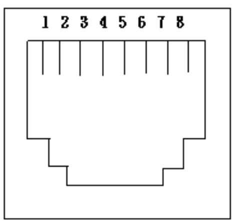
SIPFXSO 1S1O

SIPFXSO 2S2O

1. **Ethernet Port**

   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T.

   The pin-out is as following:



PIN 1, 2: Transmit

PIN 3, 6: Receive

2. **COM:**

   RS232 console port (DB-9pin **male** connector)

   Note: use straightforward cable to connect to your computer.

PINOUTS

Pin	Name	Dir	Description
2	RXD	←	Receive Data

3	TXD	→	Transmit Data
5	GND	—	System Ground

3. **TEL:**
   RJ-11 connector, FXS interface is for connecting the analog phone sets or trunk line of PABX.
4. **LINE:**
   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.
5. **12V DC:**
   Input AC 100V~120V;output DC12V.

4. Command List

4.1 Hyper Terminal Setting

A terminal emulator is needed when using RS-232 port to configure Gateway. There are kinds of terminal emulator software. Here, we use Microsoft HyperTerminal to depict how to set up terminal emulator:

1. Execute the *Hyper Terminal* program, and then the following windows will pop-up on the screen. (START – Program files – Accessories – Communication – Hyper Terminal)

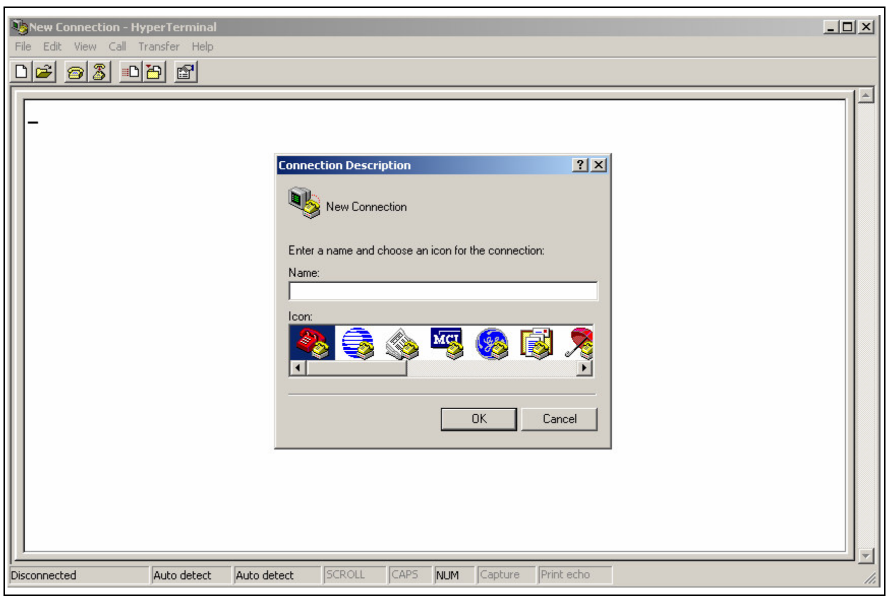


Figure 4.1: Hyper Terminal

2. Define a name such as ‘wg37’ for this new connection.

Figure 4.2: Edit the name of the connection

3. After pressing OK button, the next window appear, and then choose ***COM1/2 Port***, which you are going to use.

Figure 4.3: Pick up the right interface to use

4. Configure the COM Port Properties as following:
   - Bits per second: 9600
   - Flow control: None

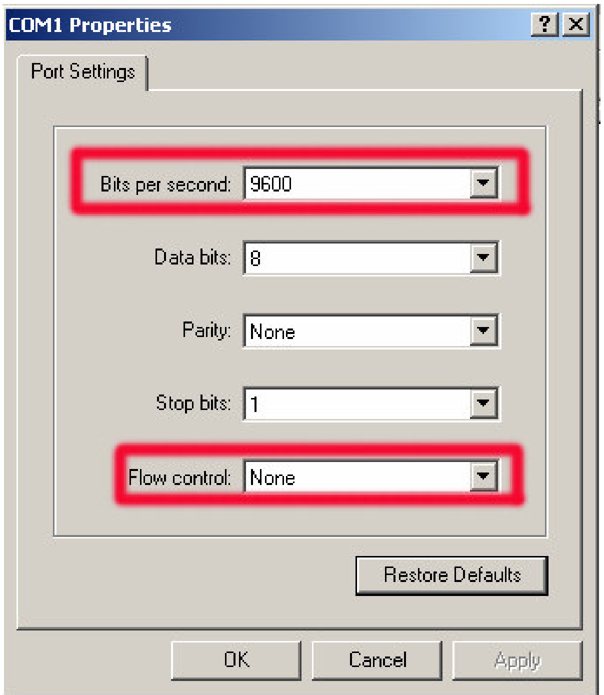


Figure 4.4: Configure the right Bps and control

5. Press ‘OK’ button, and then start to configure Gateway.

4.2 Command List

4.2.1 [help]

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

usr/config$ help

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*
prefix	*Prefix drop/insert information manipulation*
pause	*FXO Pause information manipulation*
pbook	*Phone book information manipulation*
voice	*Voice information manipulation*
phone	*Setup of call progress tones and ringing*
tone	*Setup of disconnect 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*

usage: help [command]

4.2.2 [quit]

Type **quit** will quit the Gateway configuration mode and turn back to login prompt (in console mode) or disconnect (in TELNET mode).

usr/config$ quit
Disconnecting...
login:

Note: It is recommended that type the "**quit**" command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

4.2.3 [debug]

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

usr/config$ debug -add fsm vp
usr/config$ debug -open

In this example, user open debug flags including fsm, vp.

Parameters Usage:

-status	Display the enabled debug flags.
-add	Add debug flag. *-- fsm: sip related information* *-- vp : voice related information*
-delete	Remove specified debug flag.
-open	Start to show debug messages.
-close	Stop showing debug messages.

4.2.4 [reboot]

After **commit** command, type **reboot** to reload Gateway in new configuration. The procedure is as below:

usr/config$ reboot

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

Hardware auto detect...
Hardware Type : 1FXS + 1FXO

HTTPD initialized...
VoicePacketizermain comming
WorkMode : PROXY_MODE
incoming InitCallArray....REAL_MAXCALL=4
SIP stack was constructed successfully. Version - 2.2.1.8
Start registering to Proxy server

AC4804[0] is ok
successful 1 4
Initialize OSS libraries...OK!
VP v1.44 stack open sucessfully.

login:

4.2.5 [flash]

This command will clean the configuration stored in the flash ROM and reboot Gateway in factory default setting.

Parameter Usage:

-clean	clean all the user defined values, and reboot Gateway in factory default mode.

Note: It is recommended that use "flash –clean" after application firmware id upgraded.

Warning:	Only user who login with **root** can execute this command. Configurations of IP address and accounts' passwords will be kept.

4.2.6 [commit]

Save changes after configuring Gateway.

---

usr/config$ commit

This may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$

---

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

4.2.7 [ifaddr]

Configure and display 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).*
-http	*Http port.*

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

usr/config$
```

Parameters Usage:

-print	print out current [ifaddr] settings and status
-ip	assign IP address for Gateway
-mask	assign internet subnet mask
-gate	assign IP default gateway
-mode	Switch the network type (0 = Static IP; 1 = DHCP mode 2 = PPPoE mode)
-sntp	Simple Network Time Protocol (1 = ON; 0 = OFF) When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:
-dns	configure the IP address for the DNS server
-timezone	set local time zone according to GMT
-id	To configure the pppoe connection account for the pppoe connection.
-pwd	To configure the pppoe connection password for the pppoe connection.
-reboot	If the connection disconnected by the ISP, the unit will reboot and get the ip again.
-echo	In the PPPoE mode, if the network connector or the ADSL modem was lost, after the connector and modem connected, it will reboot automatically for the re-connect with the PPPoE server.
-http	Change the http port. User can change default HTTP port (80) to another one for security or NAT application.

4.2.8 [time]

When SNTP function of 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 THU JAN 01 05:29:23 1970
```

4.2.9 [ping]

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

For example: if 192.168.1.2 is not existing while 192.168.1.254 exists. Users will have the following results:

```
usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping 192.168.1.254

PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)    min/avg/max = 0/1/5
usr/config$
```

4.2.10 [sysconf]

This command displays system information and configurations.

```
usr/config$ sysconf

System information and configuration
Usage:
 sysconf [-idtime digit][-keypad dtmf]
         [-rba digit][-eod digit][-billing digit]
         [-localrbt digit]
         [-ring on_time off_time]
         [-hwtype digit]          [-forwardtime digit][-callerid digit]
 sysconf -print

 -print          Display system overall information and configuration.
 -idtime         Inter-Digits time.(1~10 sec)
 -forwardtime    Forward time for FXS line if no answer.(5~65535 sec)
```

-keypad Select DTMF type: 0=In-band,
1=RFC2833.
2=INFO.
-hwtype Hardware type.(Auto:0 / 1FXS+1FXO:1 / 2FXS+2FXO:2)
-callerid CallerId Type .(Caller type, 0: none, 1: FSK, 2: ETSI, 3: DTMF)
-dtmfstart DTMF CallerID Start Symbol.
-dtmfend 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)
-billing Billing.(0=none, 1=reverse, 2=billing tone)
-silence Silence Detection.(0=Disable, 1=Enable)
-connect Auto connect time.(0=Disable, for 1~65535 sec)
-onhook Auto ON-HOOK if detect reverse.(Enable:1 / Disable:0)
-ivr General IVR in FXO.(Enable:1 / Disable:0)
-sipping Enable the SIP-Ping function(0: OFF, 1: ON)

Example: sysconf -ring 500
usr/config$

Parameters Usage:

-print print out all current settings

-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)

-forwardtime set forward time(5-65535 seconds)for FXS Line. If called party hasn't answered the call in this time, call will be forward to assigned number in [line] command. (please refer to ***[line]*** command for forward setting)

-keypad DTMF replay type. When value is "0", Gateway will transfer DTMF signal via In-Band type, "1" via RFC2833 type. "2" is SIP info method.Users can adjust the value according to various applications.

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".*

-hwtype application rom file of 37 series are the same no matter how many ports is the module, so after user downloads the application rom file, user can select which hardware type is . "0" means gateway will automatically detect the hardware type, "1" means the hardware type is 1FXS+1FXO, "2" means the hardware type is 2FXS+2FXO.

Note:

The default value is to auto detect hardware type. Usually it is not necessary to change this setting. Please make sure about your Hardware Type, Gateway may be not functional if set wrong hardware type.

-callerid Caller ID function disables or enable.

The FXS can support FSK; ETSI and DTMF callerid type and user can set it via this command. And the FXO can display the PSTN CallerID,too. But it must match two conditions, the PSTN CallerID must be a FSK or ETSI callerid and the FXO must under hotline mode.

The FXO only support FSK or ETSI callerid or PSTN, it does not support DTMF type till now, so please check it first before you test this case.

-dtmfstart DTMF CallerID Start Symbol.

-dtmfstart DTMF CallerID End Symbol.

-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 that 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 extend the time waiting for dial tone*

.-rba When the calls from the PSTN side, FXO port will off hook if the ring time is matched with this number.

-eod It will transfer the DTMF in “#” if users disable the end of dial function. Users have to press the key pad in “#” if the end of dial function is enabled.

-billing The billing methods are actually based on the ISP's billing server. Now our FXO gateway can support two methods for billing. The "Reverse" means that the "circuit" and "voltage" will change when the call is setup, and your billing server will start to bill according to the "voltage". The "Tone" means that the FXO will send a special tone to your server, named C tone and D tone, your server will recognize these tones, start or stop billing.

-silence Enable or Disable silence detect function.

-connect The unit will send the connect message to the IP side automatically. This function just supports the one-stage-dialing function.

-onhook Enable or Disable FXO reverse detect function.
Welltech FXO has two ways to release FXO port, one is disconnect tone detect, another is PSTN reverse. This command can let user enable or disable FXO reverse detect function.

-ivr User can disable FXO default greeting by this command, using dial tone insteaded.

-sipping This function is for some special platform, such as Nortel proxy server. The FXO will send SIP Ping message to server before registering or making call.

4.2.11 [sip]

This command is for sip configuration 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.

-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 S Call and E.164 number at TEL 1.

-line2 Line 2 is O Call and E.164 number at LINE 1.

-line3 Line 3 is S Call and E.164 number at TEL 2.

-line4 Line 4 is O Call and E.164 number at LINE 2.

-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*

Example:

sip -px 210.59.163.171 -line1 70 -line2 71

usr/config$

Parameters Usage:

-print	print current h323 related settings
-mode	alternatives for proxy or peer-to-peer mode (1=proxy mode; 0=peer-to-peer mode). If users select proxy mode, a valid proxy is needed when Gateway is in operation.
-px	to assign the ip address of the second proxy when Gateway is in proxy mode.
-px2	to assign the ip address of the proxy when Gateway is in proxy mode.
-pxport	define the proxy port for the registration or call.
-px2port	define the second proxy port for the registration or call.
-outpx	define the out bound proxy for the endpoints.
-prefix	this will be prefix the alphabets before the sip line number.
-line1	assign FXS TEL1 number.
-line2	assign FXO Line1 number.
-line3	assign FXS TEL2 number.
-line4	assign FXO Line2 number.

Note:

User can also set "x" in line number to disable the port. If the port is disabled, it can only receive calls but not calling out.

Note:

1. This is for FXSO model only, there are only line1 and line2 command.
2. No matter in Proxy or P2P mode, user only needs to dial line number to reach local port. For example, in P2P mode, user wants to dial from FXS TEL1 to FXO Line1, only need to dial number of line2.

-pbsearch	Enable the pbook function in Proxy mode
-expire	It just like the TTL function in H323, the gateway will make sure the registration is success or not for a period times.
-port	Define the local sip port for this gateway.

-rtp To allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy. (each port of Gateway use 2 RTP ports)

4.2.12 [security]

This is the authentication for the SIP account.

```
usr/config$ security

Secuirty 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
```

Parameter Usages:

-print	print out all current settings of security.
-line	the line number, which you want to define the security info
-name	the name is as same as the SIP number.
-password	the password for the authentication if it is the necessary for the proxy.

4.2.13 [line]

This command is for configure each line parameters of Gateway.

```
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.
```

Example:

usr/config$

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 in limited two channels. User just picks up the phone set of one FXS TEL or calls in one FXO line, and 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.

Note: This function can both work in Proxy or P2P mode.

(1) Call out from FXS Line

Proxy Mode Usage:

Set gateway under proxy mode.

Create a Hotline table with "***line*** " command.

----

usr/config$ line –config 1(3) hotline 1001

In this example means: if user picks up phone set of FXS 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 sipfxso ip 10.1.1.1 e164 1001
usr/config$ line –config 1(3) hotline 1001

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

(2) Call out from FXO Line

Proxy Mode Usage:

Set gateway under proxy mode.
Create a Hotline table with "***line*** " command.

----

usr/config$ line –config 2(4) hotline 1001

In this example means: if user calls in 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 sipfxso ip 10.1.1.1 e164 1001
usr/config$ line –config 2(4) hotline 1001

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

-fwdtype This version could support all the forward function, including the unconditional, busy and no answer forward function. All the forward function will make all the incoming calls routed to other number which users define in the forward table. The busy won't be worked if the hunt group function had been enabled. The no answer forward function could support the IP and local forward at the same time. Please define the special route table for the local no answer forward function. The forward could support the FXS port only.

-forward Users could define the forward number for the forward function.

----

usr/config$ line –config 1 forward 1002

In this example means: if user define the forward type for this line, the call will be forward to the destination of this phone number.

4.2.14 [route]

This command is to set routing table for Gateway.

```
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][SIP number]
route -fxs [dst number][SIP number]
route -fxo [dst number][SIP number]
route -print     Routing table information.
        prefix      The prefix of dialed number.
        dst         Destination port(FXS:0/FXO:1/IP:2).
        e164        Destination e164 number(when destination is FXS or
                    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 0 e164 1001 min 1 max 3 hunt 1
route -ip dst 0 e164 1001
route -fxs dst 2
route -fxo dst 2 e164 x
route -modify 1 prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1
route -delete 1
usr/config$

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 "SIP 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 "SIP 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 "SIP number of port"***)
-fxs	create routing table for incoming call from FXS TELs. (***route –fxs dst "destination port type" e164 "SIP number of port"***)
-fxo	create routing table for incoming call from FXO Lines. (***route –fxo dst "destination port type" e164 "SIPnumber of port"***)
prefix	prefix of dialed number
dst	destination port, 0 means FXS TELs, 1 means FXO Lines, 2 means IP side, x means no determinate number.
e164	destination SIP number. This only need to be set when routed port is FXS TELs or FXO Lines to determine which port will this call be routed to.
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, for example, destination port is FXS TEL will hunt in all FXS TELs, destination port is FXO Lines will hunt in all FXO Lines.

Usage Example:

1. ***route –add prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1***
   This command means if gateway has incoming call's prefix number is 100, and total digits is between 1 to 3, this call will be routed to FXS TEL 1001, and if TEL 1001 is busy, call will be routed to another FXS TEL.
2. ***route –ip dst 1 e164 1002***
   This command means incoming call from IP side will be routed to FXO Line of number 1002.
3. ***route –fxs dst 1 e164 1002***
   This command means incoming call from FXS TELs will be routed to FXO Line of number 1002.
4. ***route –fxo dst 2***
   This command means incoming call from FXO Lines will be routed to IP side.

Note:

(1) When destination is IP side, SIP number doesn't need to determine. (Ex. route –fxs dst 2)

(2) If user doesn't want to determine a specific port to route, SIP number must set as "x". (Ex. route –ip dst 1 e164 x)

(3) Default value: Incoming call from FXS and FXO ports will be forward to IP side directly.

4.2.15 [prefix]

This command is for make rules for drop or inserts prefix 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

usr/config$

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.

4.2.16 [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

usr/config$
```

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.

4.2.17 [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 also 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]
          [port number][drop digit][insert number]]
          [-modify number [name string][e164 number][ip address]
```

```
          [port number][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 adress.
                     port     : 1024 ~ 65535.
                     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

usr/config$
```

Parameter Usages:

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

Note: <index number> means the sequence number in phone book. If users do request a specific index number in phone book, 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.

name — name to represent callee.

e164 — The SIP number for mapping with IP address of called

ip	ip address of called
port	user could define the sip port for the peer to peer mode calling mode
drop	drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
insert	insert digits.(1~10 digits)
-delete	delete a specific record. "pbook –delete 3" means delete **index 3** record.
-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:

The SIP number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this number to reach the destination, destination will receive the number and find out if it is matched to itself, including Line number in some particular device.

4.2.18 [voice]

The voice command is associated with the audio setting information.

There are four voice codecs supported by Gateway.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711A ms] [G711U ms] [G729 ms] [G729A ms]
[G729B ms] [G729AB ms] ]
        [-volume [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] [G711A] [G711U] [G729] [G729A] [G729B]
[G729AB]

    -print     Display voice codec information and configuration.
    -send      Specify sending packet size.
```

G.723 (30/60 ms)
G.711A (20/40/60 ms)
G.711U (20/40/60 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)

-priority Priority preference of installed codecs.
G.723
G.711A
G.711U
G.729
G.729A
G.729B
G.729AB

-volume Specify the following levels:
voice volume (0~63, default: 29,28),
input gain (0~63, default: 26),
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 g711a 60 g711u 60 g729 60 g729a 60 g729b 60 g729ab 60
voice -volume voice 20 input 32 dtmf 27
voice -echo 1 1
usr/config$

Parameters Usage:

-print print current voice information and configurations.

-send 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 delay time. 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.729/G.729A/G.729B/G.729AB codec, while 30/60ms is applicable to G.723.1 codec.

-priority codec priority while negotiating with other h323 device. This parameter determines the listed sequence in h.245 TCS message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the particular codec without others.

---

usr/config$ voice –priority g723 (only select this codec)
usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and g723 is the first choice)

Note:
(1) For 2S2O there are 2 versions of Application rom, please check out the version of Application rom (rom –print). If the version is 2sipfxso729.102, 2S2O doesn't have the codec G.723.1. If the version is 2sipfxso723.102, 2S2O doesn't have the codec G.729 series.
(2) For 1S1O, the Application rom has the only one version which is named sipfxso.102 provide all codec.

-volume There are three adjustable value. **voice volume** stands for volume, which can be heard from Gateway side; **input gain** stands for volume, which the opposite party hears; **dtmf** volume stands for DTMF volume/level, which sends to its own Line.

Note: level of volume is too high or too low may be result in bad performance while connecting to each other.

-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

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

---

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

4.2.19 [phone]

Gateway's progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone specification or users-defined tone specification.

```
usr/config$ phone

Phone ringing , ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq    ][ringON    ][ringOFF ][ringLevel]]
      [-rbt   [freqHi ][freqLo    ][freqHiLev][freqLoLev]
               [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
      [-bt    [freqHi ][freqLo    ][freqHiLev][freqLoLev]
               [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
      [-dt    [freqHi ][freqLo    ][freqHiLev][freqLoLev]
               [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
      [-flash [freqLo ][freqHi ]]
phone [-print [ring]\[rbt]\[bt]\[dt]\[flash]]

     -print   Display phone ringing/tone configuration.
                ring :   ringing
                rbt  :   ringback tone
                bt   :   busy tone
                dt   :   dial tone
```

flash: flash tone

-ring ringing configuration set .

-rbt ringback tone configuration set .

-bt busy tone configuration set .

-dt dial tone configuration set .

-2dt second dial tone configuration set .

-flash flash configuration set .

Note:

ringing frequency : 15 ~ 100 (Unit : Hz)

ringing ring ON/OFF : 0 ~ 8000 (Unit : ms)

ringing level : 0 ~ 94 (Unit : V)

tone frequency : 0 ~ 65535 (Unit : Hz)

tone freqLevel : 0 ~ 65535 (Unit : mVrms)

tone Tone ON/OFF : 0 ~ 8000 (Unit : ms)

Example:

phone -print rbt

phone -ring 20 2000 4000 94

phone -rbt 480 440 8 8 2000 4000 2000 4000

phone -bt 620 480 8 8 500 500 500 500

phone -dt 440 350 8 8 500 1023 1023 1023

phone -flash 100 300

usr/config$

Parameters Usage:

-print print current call progress tone configurations (**ring** – ring tone, **rbt** – ring back tone, **bt** – busy tone, **dt** – dial tone, flash – flash). This parameter should be accompanied with tone type. For example:

----

usr/config$ phone -print rbt

Phone ring back tone paramter

Ringback Tone frequency high : 480

Ringback Tone frequency low : 440

Ringback Tone frequency high level : 13
Ringback Tone frequency low level : 13
Ringback Tone tone1 on : 100
Ringback Tone tone1 off : 200
Ringback Tone tone2 on : 1023
Ringback Tone tone2 off : 1023
usr/config$

Note:
For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

-ring to set RING Tone value.
The played tone type, when Gateway is receiving a call.
-rbt to set Ring Back Tone value
The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.
-bt to set Busy Tone value.
The played tone type, when destination is busy.
-dt to set Dial Tone value.
The played tone type, when off hook a phone set of workable Gateway.
-2dt to set the 2nd dial tone value.
The played tone type, when the unit accept the call hold function, it will generate the 2nd dial tone.
-flash set the detective flash range in ms, for example, 300-500 ms.

4.2.20 [tone]

This command is basically for FXO ports.

---

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 disconnect 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

usr/config$

Parameter Usages:

-print	show all tone configuration
[num]	tone index. 1~4 is disconnect tone, 5~8 is remote ring back tone. For FXO ports Gateway must detect disconnect tone to determine when to disconnect the call, so user must set disconnect tone of PBX or PSTN network connected to FXO ports. When making a call from FXO ports, there are 2 ways to detect callee has already picked up the call, one is to detect reverse signal, the other is to detect the termination of ring back tone, so user must set ring back tone of PBX or PSTN network. (If user doesn't know about the frequency of disconnect tone or ring back tone, please refer to ***[record]*** command to detect frequency.) For each tone may has 1 set or 2 sets (high and low) of frequencies. If user wants to set 0 in on/off time, please set "1023" represent "0". (ex. ***tone 1 620 480 8 8 50 50 1023 1023***) (***tone "index of tone" "frequency of high" "frequency of low" "level of high" "level of low" "on time of high" "off time of high" "on time of low" "off time of low"***)

4.2.21 [fxopwd]

This command is for FXO ports.

usr/config$ fxopwd

FXO password information and configuration

Usage:

fxopwd -add [passwd number][direction number]

fxopwd -delete index

fxopwd -modify index [passwd number][direction number]

fxopwd -print FXO password information.

passwd The password.

Example:

fxopwd -add passwd 1234

fxopwd -delete 1

fxopwd -modify 1 passwd 1234

usr/config$

Parameter Usages:

-print	show all FXO password configuration
-add	add 1 set of FXO password
-delete	delete 1 specific set of FXO password
-modify	modify 1 specific set of FXO password
passwd	password

4.2.22 [record]

User can record greeting and askpin file and analyze tone frequency by calling in FXO line of Gateway.

---

usr/config$ record

Recoed greeting voice and ask pin code voice, tone analize.

Usage:

record -greeting filename

-askpin filename

-tone

Example:

record -greeting greeting.100

record -askpin askpin.100

record -tone

---

usr/config$

Parameter Usages:

-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 test.100

Please off hook TEL 1 and press (N) for 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 : sdboot.200
Application Rom : fxso.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300

DSP Test Code : 483cbit.bin
Greetings : test.100
Ask Pin : askpin.100
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 askpintest

Please off hook TEL 1 and press (N) for 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 : sdboot.200
Application Rom : fxso.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : greeting.100
Ask Pin : askpintest
q
usr/config$

Note: Remember to press enter after press any command.

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

----

usr/config$ record -tone

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

Analyzing!! Please wait a moment……..
Frequency 1 : 480
Frequency 2 : 620
Frequency 3 (2623) is more than 1000, please ignore it.
usr/config$

Note:

1. ***Record ring back tone: user can use FXS Line1 to call FXO Line1, after hearing ring back tone, use this command to detect frequency of ring back tone.***

2. ***Record disconnect tone: Please read the procedure of recording disconnect tone file from the web site in application.***
3. ***The value of disconnect tone and ring back tone will not write in flash automatically. Please use the command in "tone" to write in the tone table.***

The Procedures of recording the disconnect tone

Before you start ：

A PSTN line which connect with the Line 1 port.
A analog phone connect with the Tel 1 port.
Configure Peer-to-Peer mode.

Please record the disconnect tone just follow the stage as below ：

1. Please enter the command before you record the disconnect tone：
   record –tone
2. Make a call from PSTN side into Line 1 port.
3. You will get a greeting when the Line 1 port got a PSTN incoming call.
4. Pease dial the number of the Tel 1 port.
5. The phone will ring if the number you dial is correct.
6. Pick up the phone and make sure the call is connect.
7. Hang up the phone which is from PSTN side and Tel 1 port will get the disconnect tone.
8. When you get the disconnect tone from the phone set of the Tel 1 port, press <**R**> and <**ENTER**> buttons to start recording the disconnect tone.
9. Please hang up the phone Tel 1 port if you get the message as below：
   Analizing!! Please wait a moment…
10. There are three values you will get after analyzing. Please ignore the value which is over 1000 Hz, this is not the frequency of disconnect tone.
11. Please put the frequency in the tone table just follow the command：
    tone 4 420 680 8 8 25 25 50 50

【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 2 (2623) is more than 1000, please ignore it.
Frequency 3 : 621

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

Frequency 1 : 473
Frequency 2 (2623) is more than 1000, please ignore it.
Frequency 3 (1856) is more than 1000, please ignore it.

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 : 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)

4.2.23[support]

Special Voice function support manipulation

Special Voice function support manipulation

Usage:

support [-t38 enable][-t38ecm enable][-faxrdd digits]

support -print

-t38 T.38(FAX) enabled/disabled.

-t38ecm T.38(FAX) ECM enabled/disabled.

-faxrdd FAX redundancy depth(0 ~ 2).

Example:

support -t38 1

support -t38ecm 1

support -faxrdd 1

usr/config$

Parameter Usages:

-t38 Enable or disable T.38 fax ability. The function will automatically switch codec (G.723 or G.729a) to T.38 when FAX signal is detected.

-t38ecm Enable or disable t38ecm function. The function is support the error correction in the high-speed fax mode.

-faxrdd Set Fax redundancy depth. User can increase FAX redundancy depth when network traffic is heavy. For example, if user set fax redundancy as 2, Gateway will resend fax packets every 2 packets.

4.2.24 [tos]

IP Packet ToS(type of Service)/Differentiated Service configuration.

usr/configtos

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

Parameter Usages:

-rtptype the packages of voice

-sigtype the package of call signal

Note:

The value of rtptype and sigtype is from 0 to 63.

It's working if it supported by your network.

4.2.25 [pt]

RTP payload type configuration and information

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 Configute the DTMF payload type
-fax Configure the FAX payload type
-faxbypass Configure the FAX ByPass payload type
-modembypass Configure the MODEM ByPass payload type
-redundancy Configure the Redundancy payload type
-modemrelay Configure the MODEM Relay payload type

Example:
pt -rfc2833 96 -fax 101

usr/config$

4.2.26 [rom]

ROM file information and firmware upgrade function.

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

usr/config$

Parameter Usages:

-print	show versions of all rom files
-app, boot, boot2m, dsptest, dspcore, dspapp, greeting, askpin	to update main Application program code, Boot code, DSP testing code, DSP kernel code, DSP application code, greeting file, askpin file.
-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" stands for TFTP, while "1" stands for FTP.
-ftp	if users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

4.2.27 [passwd]

For security concern, users have to input the password before entering configuration mode. "**passwd**" command is for password setting purpose.

---

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

usr/config$

Parameter Usages:

-set

(***passwd –set "login name" "password"***)

Note : "login name" can be "***root***" or "***administrator***" only. "root" and "administrator" have the same authorization, except some commands that can be executed by "root" only – "***passwd –clean***", "***rom –boot***", "***rom –bot2m***" and "***flash –clean***".

4.2.28 [auth]

For security concern, the "root" user can customize some configurable items for "administrator" user.

---

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

usr/config$

Parameter Usages:

-print Display the configurable items for "administrator" user.

-"item name" Assign the configurable item for "administrator" user.

Note : For example, you can set below authorization for root user.

```
usr/config$ auth -ifaddr 1
usr/config$ auth -sip 1
usr/config$ auth -voice 1
```

Now the Administrator can use these command which Root user assign to them.