



VoiceGATE GATEWAY
8E4137

PREFACE

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The following installation rules should be respected in order to have the best working order of the equipment and for the user's safety.

ENVIRONMENTAL CONDITIONS

ENVIRONMENTAL TEMPERATURE
from 0 to +45 °C

RELATIVE HUMIDITY from 20 to 80%
n.c.

Rapid changes of temperature or humidity should be avoided (0,03 °C/min).

This equipment, including cables, should be installed in an area free from:

- Dust, humidity, heat from direct sun light.
- Objects which irradiate heat. These could cause damage to the container or other problems.
- Objects which produce a strong electromagnetic field (loudspeakers, etc.)
- Liquids or chemical corrosive substances.

CLEANING THE TERMINAL

Use a clean and soft cloth. Wet the cloth with water or natural detergent if it is necessary to remove any stains. Never use chemical products such as petrol or solvents.

VIBRATIONS OR DROPPING

Caution against vibrations and dropping.


DECLARATION OF CONFORMITY

Digicom S.p.A. via Alessandro Volta 39 21010 Cardano al Campo-Varese-

This product satisfies the basic requirements of the below indicated Directive:

- **1999/5/CE**
 - **EN 55022**
 - **EN 61000-3-2**
 - **EN 55024**
 - **EN 60950**
- **EN 41003**

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1. Introduction

The VoiceGATE GATEWAY provides voice/fax service over IP network with H.323 v3 protocol. By connecting to your existing ADSL or cable modem service, which allows the use of a single, network for voice and fax services with consequent saving in network infrastructure and greatly reduced telephone charges. Ideal solution for providing low cost communications between headquarters and branch offices in the world, as well as for SOHO and office telephony applications.

VoiceGATE GATEWAY provides analog lines to connect local PSTN/PTT interface (FXO), and converts voice/fax signal onto IP network. The management feature is via RS-232C COM port and TELNET.

1.1 Features and specification

General Features

- ITU-T H.323 v3 compliance
- Automatically Gatekeeper Discovery
- Peer-to-Peer mode (non-Gatekeeper)
- Support auto-attendant (2nd dial Tone / Voice greeting)
- Dimensions : 221mm(W)*42mm(H)*217mm(L)
- Line hunting
- 4 RJ-11 FXO ports
- E.164 (Telephone Number Plan)
- DTMF dialing
- DTMF detection/generation
- TFTP software upgrade
- Remote configuration/reset via Telnet
- LED indication for system status
- LAN interface : One RJ-45 connector of 10Base-T
- Microsoft Netmeeting v3.0 compatible
- Support static IP and DHCP
- QoS by ToS (Type Of Service)
- SNTP (Simple Network Time Protocol)
- Security: Password setting

Audio feature

- Codec -- G.711 a/ μ law, G.723.1 (6.3K/bps), G.729A (Optional)
- VAD (Voice Activity Detection), CNG (Comfort Noise Generate)
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Bad Frame Interpolation
- Call Transfer (H.450.2)
- Call Forward (H.450.3)
- Call Hold (H.450.4)
- Gain Settings
- Provide Call Progress Tone: Dial tone, busy tone, call-holding tone and ring-back tone

Management Features:

Two easy ways for system configuration

- Console port: RS-232C port
- TELNET
- HTTP Browser (e.g. Internet Explorer)

1.2 Appearance

Front panel: The LED light provides system message of VoiceGATE GATEWAY.

Power : Light on means VoiceGATE GATEWAY is power on.

L1-L4 : Light on means the line is in use.

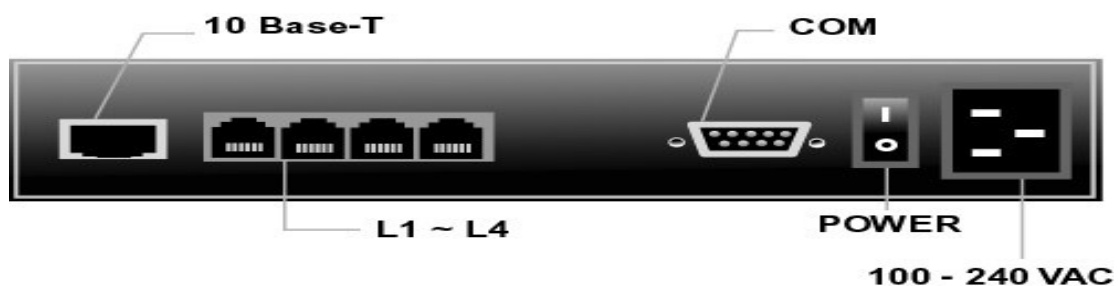
Link : Light on means VoiceGATE GATEWAY is connected to the network correctly.

Act : LED should be light on and in flash display when data is transmitting.

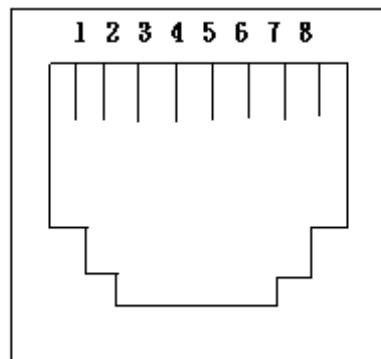
Ready : 1. Light on and in slow flash means VoiceGATE GATEWAY is in operation mode.

Status : 1. Light on means VoiceGATE GATEWAY successfully registered to Gatekeeper when it is set as Gatekeeper Mode.
2. LED flash means VoiceGATE GATEWAY is not registered to Gatekeeper when it is set as Gatekeeper Mode.
3. Or when VoiceGATE GATEWAY is in downloading mode, LED should be flash as well.
4. Light off means VoiceGATE GATEWAY is in Peer-to-Peer Mode.

Back panel:



10 Base-T: RJ-45 Modular Jack Female connector with 10 Mbps Ethernet.



PIN 1, 2: Transmit
PIN 3, 6: Receive

COM: RS232 console port (9-pin Male connector, as the same as the computer).

Male connector (as the same as the PC)



9 PIN D-SUB MALE at the VoiceGATE GATEWAY

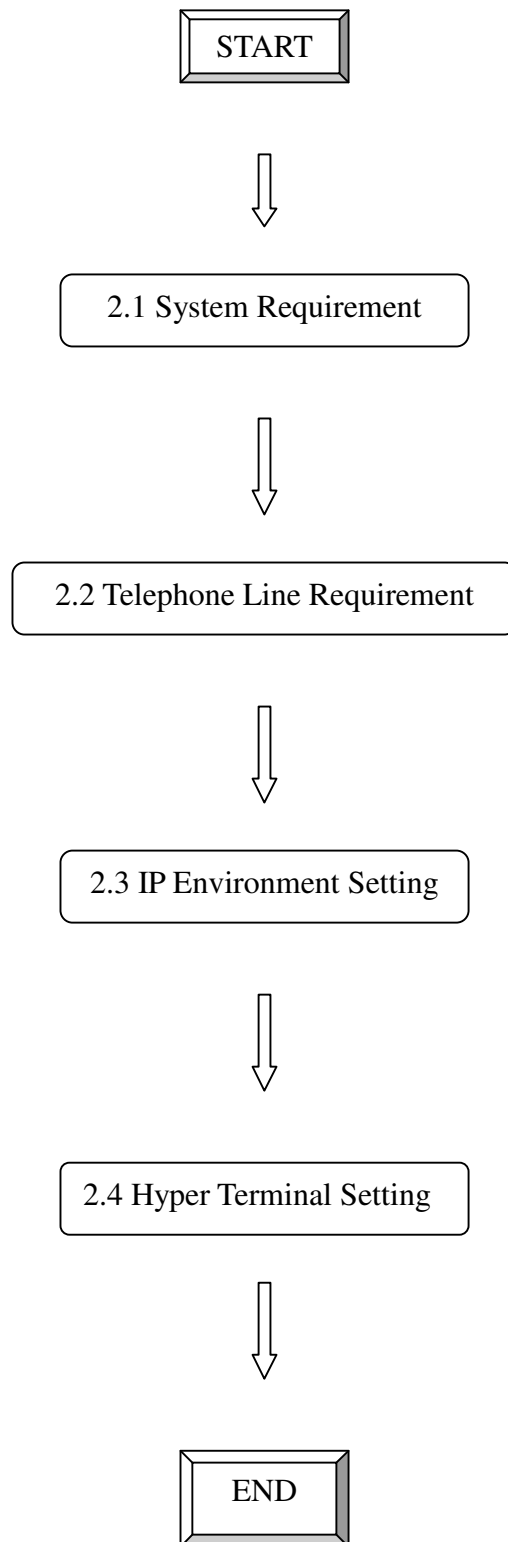
Pin	Name	Dir	Description
2	RXD	←	Receive Data
3	TXD	→	Transmit Data
5	GND	—	System Ground

L1 ~ L4: RJ-11 (PSTN or Extension Line of PBX)

On / Off: Power switch on/off.

100 - 240 VAC: AC Power supply.

2. System Operating Procedure



2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 MB DRAM, Windows 98 or above.
(b) Network card (RJ-45) & COM port
2. One standard RS-232 straight cable with **two female connectors** depended on the different model.
3. PSTN lines / PBX extension lines (up to 4 lines).
4. Software tools (a) Hyper terminal, telnet (Windows OS included); (b) Gatekeeper (optional)

2.2 Telephone Line Requirement

Two kinds of analog lines can be connected to RJ-11 of VoiceGATE GATEWAY.

1. PSTN (Public Switched Telephone Network, POTS) or
2. PABX (Private Automatic Branch Exchange) / PBX (Private Branch Exchange) extension line.

PSTN

1. It is necessary to provide PSTN/POTS telephone lines in order to plug into RJ-11 of VoiceGATE GATEWAY.
2. The maximum telephone lines are up to 4.

PABX / PBX

1. 4 PSTN lines can be replaced to the 4 extension lines of PBX.

<p>Note: Since the Line function feature starts from L1, please plug the telephone lines from L1.</p>
--

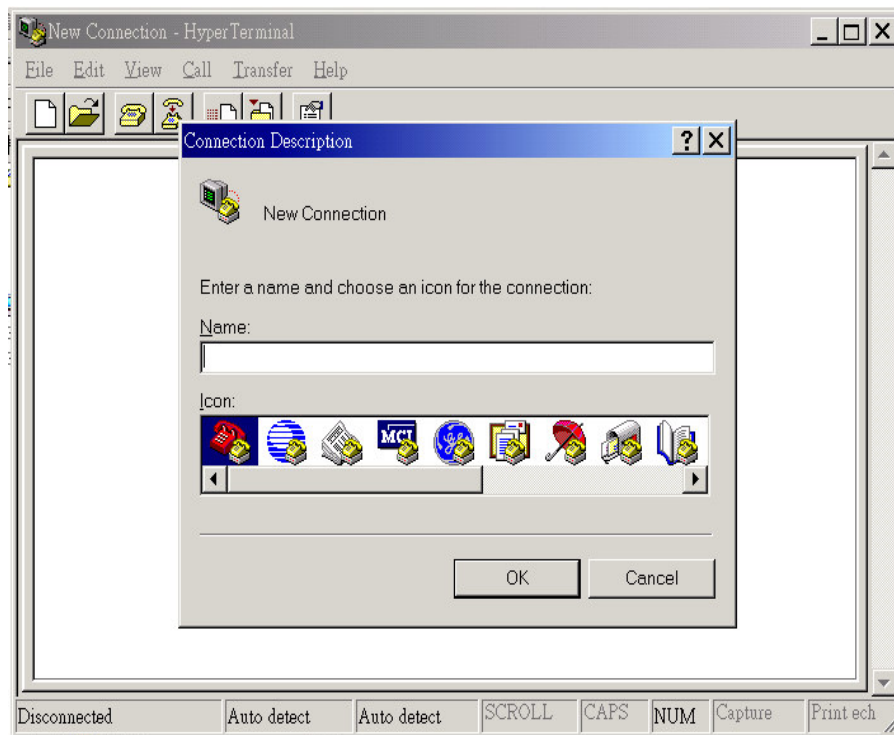
2.3 IP Environment Setting

User must prepare a valid IP address to be complied IP Network policy in order for VoiceGATE GATEWAY operating correctly.

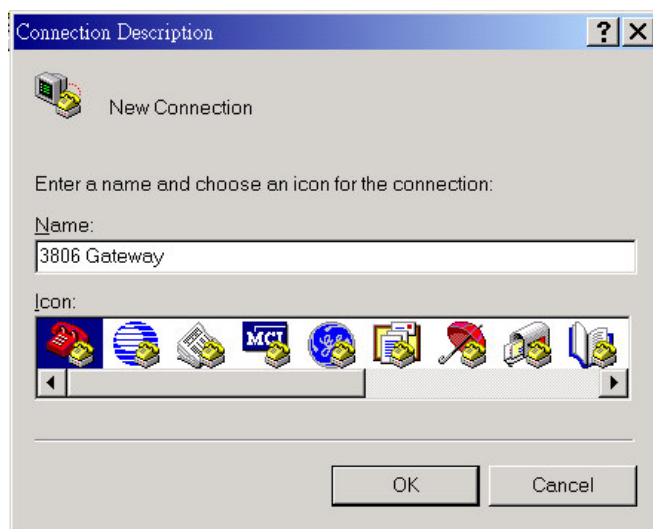
For example, if your company's IP address is 192.168.4.111, subnet mask is 255.255.0.0, default gateway is 192.168.1.254, you should prepare one IP for VoiceGATE GATEWAY, such as IP address is 192.168.4.99, and the same subnet mask and default gateway.

2.4 Hyper Terminal Setting

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



2. Define a name such as 'FXO Gateway' for this new connection.

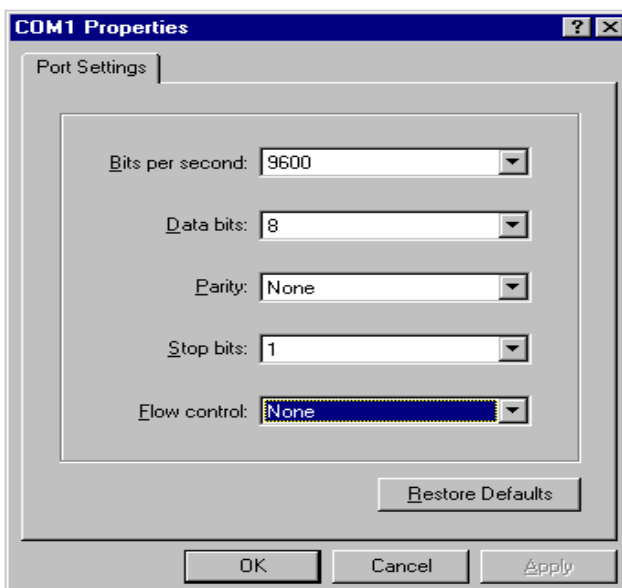


3. After pressing OK button, the next window popping up is necessary to connect choose COM Port.



Note: Some connection failed is derived the PC COM Port. If user cannot open the com port, for example com 1, please try another com port, ex.com port 2.

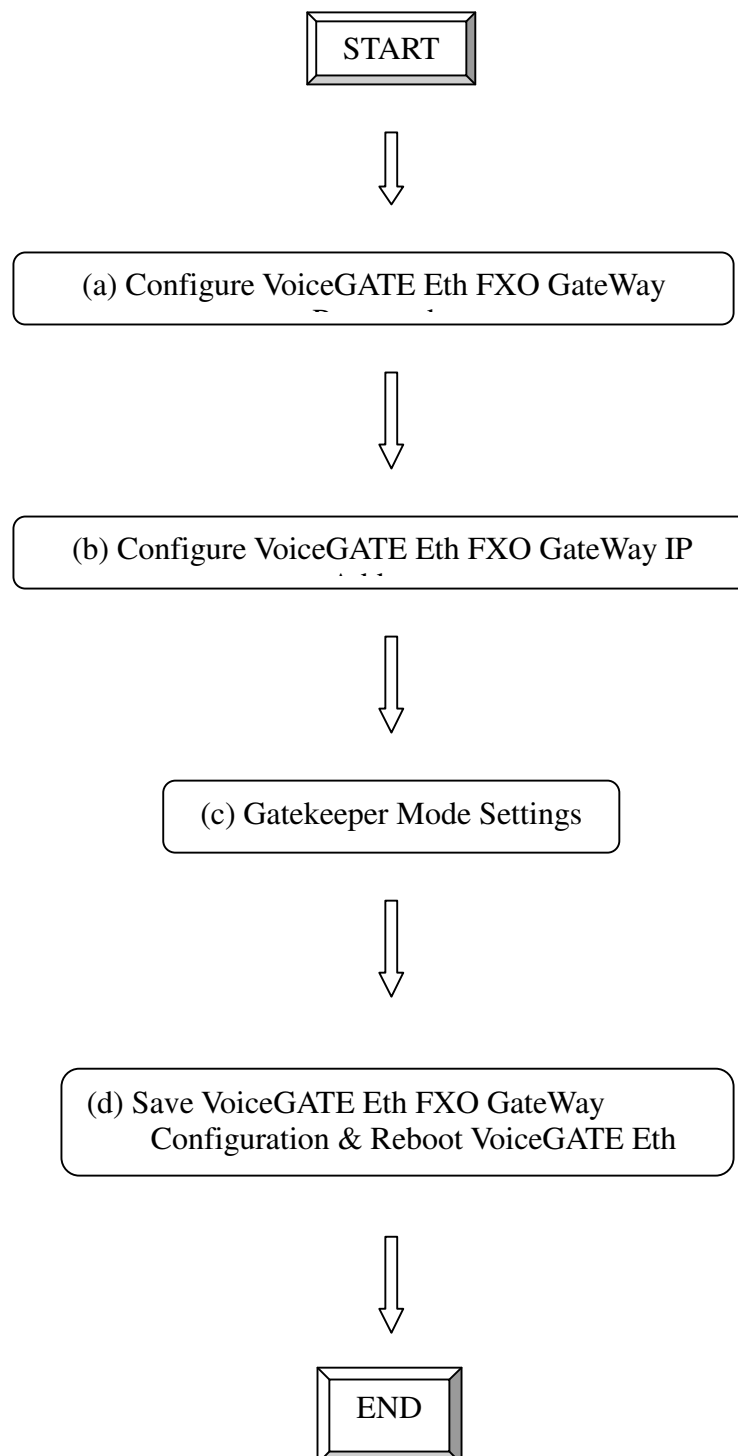
4. Configure the COM Port Properties as following:
 - (1) Bits per second : 9600
 - (2) Flow control : None



Press 'OK' button, and start to configure VoiceGATE GATEWAY.

Initializing VoiceGATE GATEWAY Setting

3.1 Gatekeeper Mode



(a) Configure VoiceGATE GATEWAY Password

It is important for the first time user to follow the operation procedure.

1. Power on the VoiceGATE GATEWAY and the sentence "Please wait while system is initializing.....S" is displayed.

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

Please wait while system is initializing S

2. Wait around 40 seconds, the login name and password are requested.

*Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done
AC4804[0] is OK
AC4804[1] is OK
AC4804[2] is OK
Successful*

*Initialize OSS libraries...OK!
open stack successful
cmInitialize succeed!
GK mode selected.*

login:

3. Login: when VoiceGATE GATEWAY is used for the first time, "root" is default login name without a password.
4. Password setting: type "passwd -set root ****" to define a password for "root" account. "****", in above description, stands for contents of the password. An example, to set **root**'s password as **good**, is demonstrated as following:

usr/config\$ passwd -set root good

*Setting
login: root
Password: good
OK*

(b) Configure VoiceGATE GATEWAY IP Address

Use “**ifaddr**” command to set up VoiceGATE GATEWAY’s IP address and related network information. An example is demonstrated below:

```
usr/config$ ifaddr -ip 10.1.1.1 -mask 255.255.255.0 -gate 10.1.1.254
```

Note: this is to assign VoiceGATE GATEWAY an IP address of “10.1.1.1”, subnet mask “255.255.255.0”, and default IP gateway “10.1.1.254”.

(c) Gatekeeper Mode Settings

To assign a gatekeeper address for VoiceGATE GATEWAY, and define its own registered ID and phone number. For detail, please refer to *Chapter 5.14 [h323] command*.

Several important H323 parameters is listed below when setting gatekeeper mode:

“**-gk**”, “**-e164**”, and “**-alias**”.

An example is demonstrated below:

```
usr/config$ h323 -gk 10.2.2.2 -e164 -alias fxo
```

Note: This is to set gatekeeper IP address as “10.2.2.2”, e.164 number as “1”, and alias name (h323ID) as “fxo”.

(d) Save VoiceGATE GATEWAY Configuration & Reboot VoiceGATE GATEWAY

1. Confirming the values, type **commit** and press **enter** to save all the changes you have done.
2. Type **reboot** and press **enter** to reboot the VoiceGATE GATEWAY.
3. Wait for VoiceGATE GATEWAY initializing in gatekeeper mode.

3.2 Peer-to-Peer Mode

Peer-to-Peer Mode allows users to call other VoIP devices without using a gatekeeper. When in Peer-To-Peer mode, VoiceGATE GATEWAY will send SETUP message directly to the destination IP address once the dial is finished. Users have 2 methods of dial. One is IP dialing, and the other is Phonebook dial, which we will describe later. When using IP address as destination phone number, press “*” as “.” in IP address expression, and press “#” when dial is finished. When using Phonebook, users can dial predefined phone number, and press “#” (optional, to accelerate the dial) as end of dial.

To configure Peer-To-Peer Mode in VoiceGATE GATEWAY, follow the steps below:

1. Set Peer-To-Peer Mode, using “h323” command

```
usr/config$ h323 -mode 1
```

Note: mode 1 is for Peer-To-Peer (non-gk) mode, while mode 0 is for GK mode.

2. Configure Phonebook, using “pbook” command.

Users can refer to chapter 5.11 [pbook] command for more information.

```
usr/config$ pbook -add name TEST1 ip 10.1.1.1 e164 10
```

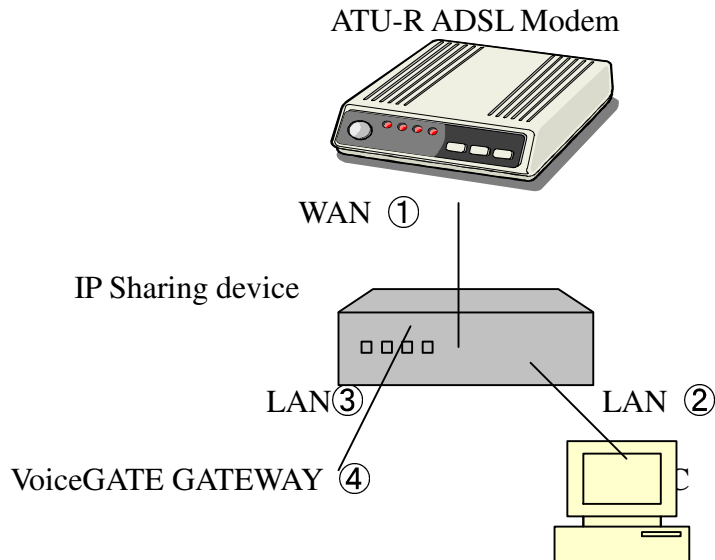
Note: the command is to add a record onto Phonebook. After the command completed, you can type “pbook -print” to see if the input record is correct. When adding a record to Phonebook, users do not have to reboot the machine, and the record will be effective immediately.

3.3 Behind IP-Sharing

IP Sharing function

The function is for user whose network environment is behind IP Sharing device. It is said VoiceGATE GATEWAY is connected to the IP Sharing device.

An example such as ADSL network is in the following.



- ① The WAN IP Address obtained from ADSL has two kinds of methods.
One is fixed IP Address, while user applies for one or more fixed IP Addresses.
Another is dynamic IP Address while user applies for dial-up connection way.
- ② The LAN IP Address of User's PC can be set as DHCP client in order to gain a valid one.
- ③ Another IP Address for VoiceGATE GATEWAY must be set as a fixed one in order for that IP Sharing device pass forwarding the relevant information from WAN to LAN. Besides, a valid IP Address which meets the IP Sharing device (LAN site) is the element.
- ④ VoiceGATE GATEWAY must enable the IP Sharing function for the fixed / dynamic WAN IP Address.
Fixed IP Address – `usr/config$ ifaddr -ipsharing 1 210.11.22.33`
Dynamic IP Address – `usr/config$ ifaddr -ipsharing 1`

=====

Please be noted

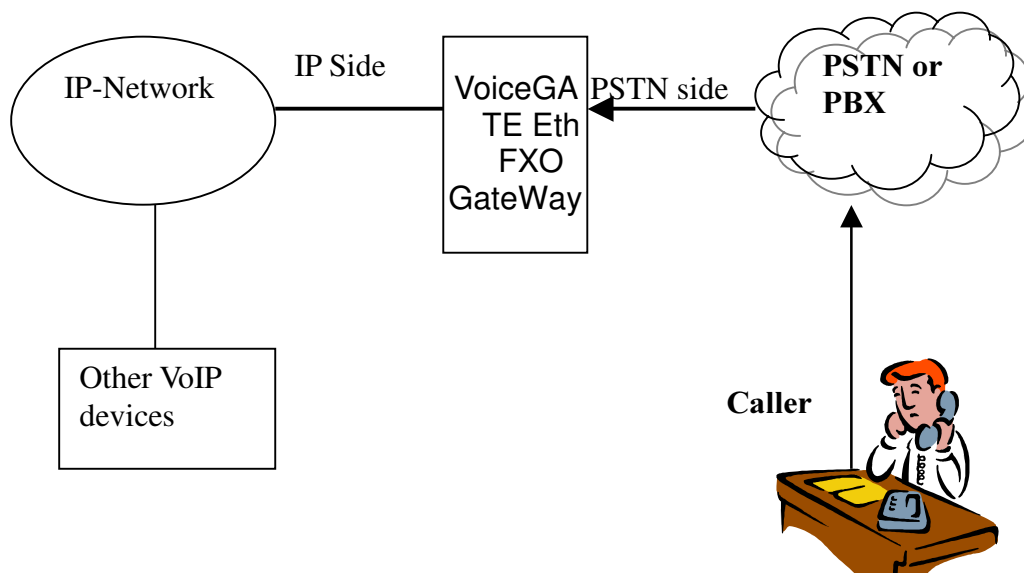
With Dynamic WAN IP Address, a valid Gatekeeper for VoiceGATE GATEWAY to get register on it is a must. *In other word, it is not workable in Peer-to-Peer mode while dynamic WAN IP Address.*

- ⑤ IP Sharing device must have a function to do IP/Port mapping. Some is named as DMZ, some is named as virtual server. The VoIP messages from WAN have to completely pass forward to the LAN. It is said if the VoiceGATE GATEWAY is assigned a virtual fixed IP Address such as 192.168.1.5, IP Sharing device must forward the VoIP messages to 192.168.1.5.

4. Disconnect Tone Configuration

This application note is going to describe the procedures of configuring the disconnect tone on VoiceGATE GATEWAY in order to release LINE ports of VoiceGATE GATEWAY after PSTN/PBX caller party is hung up.

4.1 What is Disconnect Tone



A caller make a telephone call to Gateway from PSTN side, VoiceGATE GATEWAY will answer the call automatically. If the IP side of other VoIP devices do not answer the call and the caller hang up the call, the PSTN/PBX will give Gateway a disconnect tone automatically. Or, both devices are installed with VoiceGATE GATEWAY and connect to local PSTN. If both parties are in talk mode, one side hang up the call the VoiceGATE GATEWAY has to recognize the disconnect tone from local PSTN. The VoiceGATE GATEWAY gateway will recognize this disconnect tone and release the LINE port with the pre-defined busy tone or reorder tone from VoiceGATE GATEWAY tone table.

If the other VoIP device of IP side hangs up the phone, the gateway will release the LINE port automatically without analyzing disconnect tone from PSTN/PBX.

There are three parameters received from PSTN/PBX.

- High level frequency and Low level frequency
- Tone Cadence (ON/OFF intervals)
- Tone level

These parameters have to be properly configured to VoiceGATE GATEWAY in order to recognize disconnect tone correctly. Each different PSTN/PBX have different parameters. So, VoiceGATE GATEWAY has to configure tone table when LINE port connect to different PSTN/PBX.

4.2 How to configure disconnect tone on VoiceGATE GATEWAY gateway

VoiceGATE GATEWAY has a default setting of disconnect tone (Busy tone 1, Busy tone 2, reorder tone 1 and reorder tone 2). If the disconnect tone was recognized correctly, the LINE port from PSTN/PBX will be released in two seconds. Otherwise it may be released after one minute or lock this LINE permanently.

The tone table parameters are shown as follows.

LowFreq 480:	Low frequency is 480 HZ
HighFreq 620:	High frequency is 620 HZ
LowFreqLevel 8:	Low frequency level received range from PSTN/PBX
HighFreqLevel 8:	High frequency level received range from PSTN/PBX
TON1 - 50:	Disconnect tone cadence ON time is 0.5 seconds
TOFF1 - 50:	Disconnect tone cadence OFF time is 0.5 seconds
(If this is continuous tone, the Toff has to set to 1023)	
TON2 - 1023:	Disconnect tone second cycle cadence ON time is OFF
TOFF2 - 1023:	Disconnect tone second cycle cadence OFF time is OFF
(If the tone cadence has only one cycle, the second cycle must set to 1023)	

(1) Examples how to configure Tone table

a. 480/620 frequency with ON/OFF time is 0.5 seconds
tone -busy1 480 620 8 8 50 50 1023 1023

b. 480 HZ single frequency with continuous tone
tone -reorder2 480 0 8 0 50 1023 1023 1023

(2) There are two ways to analyze the disconnect tone.

a. The first one is using command "greetrd" from VoiceGATE GATEWAY. Once you follow the instruction to analyze the disconnect tone, Gateway will configure the tone table (Busy tone 1, Busy tone 2, reorder tone 1 and reorder tone 2) with proper frequency and default tone level and cadence (Ton1/Toff1) automatically. Or you may read the analysis tone frequency from command line and configure to one of tone table manually.

The default tone level is set to 8. And the tone cadence (Ton1/Toff1) is set to four different values on tone table. They are 0.1 second, 0.25 seconds, 0.5 seconds and 0.75 seconds with parameters 10/10, 25/25, 50/50 and 75/75.

If the PBX/PSTN cadence is not the value as default shown as above, you need to use the following instruction to analyze ON/OFF intervals.

b. You may use your PC (START → Program Files → Accessories → Multimedia → Recorder) with Headset or Microphone to record the disconnect tone via a telephone set from PSTN/PBX and save to a voice file. Then you can use "CoolEdit Pro" software to analyze the frequency and ON/OFF time. Please visit <http://www.cooledit.com> to download demo version for analysis. You can use this program to analyze ON/OFF time and fill in to tone table.

4.3 Adjust Tone Table parameters manually

If the gateway still cannot release the LINE port in two seconds, try to adjust the frequency by 1 hz on tone table. For example, your analysis value is 620/480, take the following procedures.

```
620/479
620/480
620/481
621/479
621/480
621/481
619/479
619/480
619/481
```

If the LINE port of gateway was locked, please use “hangup 0” command to release LINE 1, “hanhup 1” to release LINE 2...etc.

4.4 Adjust Input Tone Level

Sometimes the disconnect tone level is too low to detect by VoiceGATE GATEWAY. You can increase input gain from the following command.

```
voice -volume input xx
commit
reboot
```

xy is the input gain parameters. 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.

5. Command lists

5.1 [help] command

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

<i>usr/config\$?</i>	
<i>help</i>	<i>help/man/? [command]</i>
<i>quit</i>	<i>quit/exit/close</i>
<i>debug</i>	<i>show debug message</i>
<i>reboot</i>	<i>reboot local machine</i>
<i>flash</i>	<i>clean configuration from flash rom</i>
<i>commit</i>	<i>commit flash rom data</i>
<i>ifaddr</i>	<i>internet address manipulation</i>
<i>time</i>	<i>show current time</i>
<i>ping</i>	<i>test that a remote host is reachable</i>
<i>greetr</i>	<i>Greeting voice and Disconnect tone Record mode</i>
<i>pbook</i>	<i>Phonebook information and configuration</i>
<i>sysconf</i>	<i>System information manipulation</i>
<i>h323</i>	<i>H.323 information manipulation</i>
<i>voice</i>	<i>Voice information manipulation</i>
<i>gk</i>	<i>H.323 gatekeeper manipulation</i>
<i>tos</i>	<i>IP Packet ToS (Type of Service) values</i>
<i>tone</i>	<i>Setup of call progress tones</i>
<i>support</i>	<i>Special Voice function support manipulation</i>
<i>group</i>	<i>Grouping setting information and configuration</i>
<i>bureau</i>	<i>Bureau line information manipulation</i>
<i>prefix</i>	<i>Prefix information manipulation</i>
<i>rom</i>	<i>ROM file update</i>
<i>passwd</i>	<i>Password setting information and configuration</i>

usage: help [command]

5.2 [quit] command

Type **quit** will quit the VoiceGATE 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, VoiceGATE GATEWAY will ask password again when next user connects to console port.

5.3 [debug] command

Open debug message will show up specific information while VoiceGATE 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 h323 vp  
usr/config$ debug -open
```

Parameters Usage:

- status Display the enabled debug flags.
- add Add debug flag.
 - h323 : h323 related information
 - vp : voice related information
- delete Remove specified debug flag.
- open Start to show debug messages.
- close Stop showing debug messages.

5.4 [reboot] command

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

```
usr/config$ reboot
Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done
AC4804[0] is OK
AC4804[1] is OK
AC4804[2] is OK
Successful

Initialize OSS libraries...OK!
open stack successful
cmInitialize succeed!
GK mode selected.
```

login:

5.5 [flash] command

This command will clean the configuration stored in the flash rom and reboot VoiceGATE GATEWAY in factory default setting.

Parameter Usage:

- clean clean all the user-defined value, and reboot VoiceGATE GATEWAY in factory default mode.

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

Warning: Once users execute **flash -clean**, all the configurations of VoiceGATE GATEWAY will be cleaned. This can only be executed by user who log in with **root**

5.6 [commit] command

Save changes after configuring the VoiceGATE 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.*

5.7 [ifaddr] command

Configure and display VoiceGATE GATEWAY network information.

usr/config\$ ifaddr

LAN information and configuration

Usage:

ifaddr [-print][/-dhcp used][/-sntp mode [server]]

ifaddr [-ipsharing used [deviceAddr]]

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

<i>-print</i>	<i>Display LAN information and configuration.</i>
<i>-ip</i>	<i>Specify VoiceGATE GATEWAY ip address.</i>
<i>-mask</i>	<i>Set Internet subnet mask.</i>
<i>-gate</i>	<i>Specify default gateway ip address</i>
<i>-dhcp</i>	<i>Set DHCP client service flag (On/Off).</i>
<i>-sntp</i>	<i>Set SNTP server mode and specify IP address.</i>
<i>-timezone</i>	<i>Set local timezone.</i>
<i>-cmcenter</i>	<i>Set Management Center IP Address.</i>
<i>-ipsharing</i>	<i>Specify usage of an IP sharing device and specify IP address.</i>

Note:

Range of ip address setting (0.0.0.0 ~ 255.255.255.255).

DHCP client setting value (On=1, Off=0). If DHCP set to 'On',

Obtain a set of Internet configuration from DHCP server assigned.

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 -dhcp 1

ifaddr -sntp 1 210.59.163.254

ifaddr -ipsharing 1 210.59.163.254

ifaddr -timezone 8

Parameters Usage:

-print	print current IP setting
-ip	assigned IP address for VoiceGATE GATEWAY
-mask	internet subnet mask
-gate	IP default gateway
-dhcp	Dynamic Host Configuration (1 = ON; 0 = OFF)
-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:

usr/config\$ ifaddr -sntp 1 10.1.1.1

while 10.1.1.1 stands for SNTP server's IP address.

-timezone	Set timezone for VoiceGATE GATEWAY. User can set different time zone according to the location VoiceGATE GATEWAY is. For example, in Taiwan the time zone should be set as 8, which means GMT+8.
-cmcenter	Set management center IP address. IF user specifies management center IP address, VoiceGATE GATEWAY will send information to management center, let user can easily configure via management center interface. (sysconf -cmcenter "IP address of management center")

Note: management center is optional software to help user can easily configure WellTech products, please contact your reseller to know more about it.

-ipsharing Specify usage of an IP sharing device and specify IP address. If VoiceGATE GATEWAY is behind a IP-sharing , user can enable IP sharing function and specify public IP address of IP-sharing.

5.8 [time] command

When SNTP function of VoiceGATE 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
```

5.9 [ping] command

Use **ping** to test whether a specific IP is reachable or not. For example: if 192.168.1.2 is not existing while 210.63.15.32 exists. Users will have the following results:

```
usr/config$ ping 210.54.23.129
PING 210.54.23.129: 56 data bytes
no answer from 210.54.23.129
usr/config$ ping 192.168.4.121
PING 192.168.4.121: 56 data bytes
64 bytes from 192.168.4.121: icmp_seq=0. time=5. ms
64 bytes from 192.168.4.121: icmp_seq=1. time=0. ms
64 bytes from 192.168.4.121: icmp_seq=2. time=0. ms
64 bytes from 192.168.4.121: icmp_seq=3. time=0. ms
----192.168.4.121 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
```

5.10 [greetrd] command

This command is for user to record their own greeting and analyze disconnect tone. If VoiceGATE GATEWAY can't hang up call and release line correctly, please use this function to analyze disconnect tone of PSTN side.

1. **Greeting Voice Record** : please follow instructions on screen ; first, call in line1 of VoiceGATE GATEWAY from PSTN side(now can't hear greeting) and press "enter" to start record .After finishing recording, please press "enter" again to stop recording. Then choose "y/n" to replay and save or not.

```
usr/config$ greetrd
```

```
=====
```

Welcome to Voice Record/Analysis Mode

```
-----
```

- 1.Greeting Voice Record.
- 2.Disconnect Tone Analysis.
- 3.exit.

```
-----
```

Please input function(1~3): 1

1.Greeting Voice Record.

Please Dial-in "Line 1" and press "Enter" to start record!!!

Press "Enter" to stop record!!!

Starting record...

Stoped record!!!

New Greeting Voice Infomation

```
-----
```

File size : 0 (K bytes)
Totally time: 8 (seconds)

Do not Hang up the phone!!

Please wait for Writing...block 0

Please wait for Writing...block 1

Please wait for Writing...block 2

Replay New Greeting Voice?(y/n):

-
- 2. Disconnect Tone Analysis :** please follow instructions on screen ; first call in line1 of VoiceGATE GATEWAY from PSTN side(now can't hear greeting), hang up the phone and press "enter" to start record disconnect tone. Finally, choose "y/n" to save data analyzed or not. Notice that system will save one set of frequency analyzed and 4 set different on/off time in "busytone1", "busytone2", "reordertone1", "reordertone2" (Please refer to tone command) .

If VoiceGATE GATEWAY still can't hang up call correctly, it could be tone cadence issue (on/off time). Please count on/off time and configure it into tone command.

```
-----
```

```
usr/config$ greetrd
```

```
=====
```

Welcome to Voice Record/Analysis Mode

```
-----
```

- 1.Greeting Voice Record.
- 2.Disconnect Tone Analysis.
- 3.exit.

```
-----
```

Please input function(1~3): 2

2. Disconnect Tone Analysis.

*Please Dial-in "Line 1" and then Hang up the phone!!!
Press "Enter" to start record!!!*

*Waiting for Disconnect Tone from PSTN....
Disconnect Tone Detected....
Starting Record...*

Set parameters to flash? (Y/N)

3. **exit** : exit this command

```
usr/config$ greetrd
```

```
=====
Welcome to Voice Record/Analysis Mode
-----
```

```
1. Greeting Voice Record.
2. Disconnect Tone Analysis.
3. exit.
```

```
-----
Please input function(1~3): 3
```

```
Are you sure to EXIT?!(y/n): y
```

```
usr/config$
```

5.11 [pbook] command

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
```

Phonebook information and configuration

Usage:

```
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber]]
    -print      Display Phonebook data.
    -add        Add an record to Phonebook.
    -search     Search an record in Phonebook.
    -delete     Delete an record from Phonebook.
    -insert     Insert an record to Phonebook in specified position.
    -modify     Modify an exist record.
```

Note:

If parameter 'end_record' is omitted, only record 'start_record' will be display.

If both parameters 'end_record' and 'start_record' are omitted, all records will be display.
 Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
 Range of index setting value (1~100),

Example:

```
pbook -print 1 10
pbook -print 1
pbook -print
pbook -add name Test ip 210.59.163.202 e164 1001
pbook -insert 3 name Test ip 210.59.163.202 e164 1001
pbook -delete 3
pbook -search ip 192.168.4.99
pbook -modify 3 name Test ip 210.59.163.202 e164 1001
```

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, VoiceGATE 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 record.
- 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
- 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 → VoiceGATE GATEWAY

In Phonebook, there's a record :

Index	Name	IP	E164
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 VoiceGATE GATEWAY, and then dial "082265699". After receiving the complete dialed number, VoiceGATE 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". Pleased be noted that "0" is eliminated from VoiceGATE 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, VoiceGATE 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.

5.12 [pppoe]

Display PPPoE related information.

PPPoE device information and configuration

Usage:

pppoe [-print][[-open]][[-close]]

pppoe [-dev on/off][[-id username]][-pwd password]

-print	Display PPPoE device information.
-dev	Enable(=1) or Disable(=0) device.
-open	Open PPPoE connection.
-close	Disconnect PPPoE connection.
-id	Connection user name.
-pwd	Connection password.
-reboot	Reboot after remote host disconnection.

Parameter Usage:

-print	print PPPoE status.
-dev	Enable PPPoE Dial-up function
-open	Open the connection
-close	Close the connection
-id	Input the User name ID provided by ISP
-pwd	Input the User name password provided by ISP
-reboot	Reboot the PPPoE connection.

5.13 [sysconf] command

This command displays the system information and configuration.

usr/config\$ sysconf

System information and configuration

Usage:

*sysconf [-service type] [-plan digits] [-2nddial flag]
 [-keypad dtmf] [-ringdet method] [-callalive flag]
 [-port s1 s2 s3 s4]
 [-seizure mode] [-2nddial switch]*

```

[-drule [in_filter str1] [in_drop str2] [in_insert str3]
[out_filter str4] [out_drop str5] [out_insert str6]]
[-askpin f] [-pincode [set1 pin1] [set2 pin2] [set3 pin3] [set4 pin4]]
sysconf -print

```

```

-print      Display system overall information and configuration.
-service    Specify gateway service type.
            (0: Dial in service, 2: HotLine/LineToLine service.)
-ringdet    Specify gateway ring detect method. (0: For 1st hardware version,
            1: For 2nd hardware version.)
-plan       Number of digits for dial plan. (any positive
            number.)
-port       Enable/Disable individual port.
-idto       The duration of two pressed digits in dial mode
-eod        Digit type of end of dialing. ( 0: No end of dialing, 1: [*] button, 2: [#] button )
-seizure    Choose line seizure mode (None/UCD).
-2nddial    Config GW to accept 2nd dtmf set. In this mode, device
            from IP side needs to dial GW's E164, wait for PSTN
            dialtone, and then dial out.
-drule      Specify digits to be filtered/dropped/inserted before
            making an outgoing IP call or after receiving an incoming
            IP call.
-askpin     PIN code prompt before greeting.
            0: Disable 1: Per Unit 2: Per Channel.
-ring       Ring number before answer.
            0: Disable, other is number of ring ( 1 ~ 5 ).
-callalive  Enable or disable auto-disconnection after 10 seconds
-keypad     DTMF setting: 0=In-band, 1=H.245 Alphanumeric,
            2=H.245 SignalType, 3=Q.931 UserInfo, 4=RFC2833.
-pincode    Specify 6 of PIN codes.
-sendxcode  Send access code after connection.
            0: Disable 1: Enable.
-access     Specify access codes.

```

Note:

Use character 'x' to delete the drule parameter.

For line seizure 0: None, 1: UCD.

For askpin: f=0: No, f=1: Yes.

Direct in line feature should be used together with:

```
$sysconf -2nddial 0 (2nddila off)
```

```
$h323 -mode 0 (Gatekeeper mode)
```

```
$bureau -print for Direct in line table configuration
```

Hotline feature should be used together with:

```
$sysconf -2nddial 0 (2nddial off)
```

```
$h323 -mode 1 (peer-to-peer mode)
```

```
$bureau -print for Hotline/LineToLine table configuration.
```

LineToLine feature should be used together with:

```
$sysconf -2nddial 1 (2nddial on)
```

```
$h323 -mode 1 (peer-to-peer mode)
```

```
$bureau -print for Hotline/LineToLine table configuration.
```

Example:

```
sysconf -service 0 -plan 4 -port 1 1 1 1 0 0
```

```
sysconf -callalive 0 -keypad 0
```

```
sysconf -2nddial 0 -drule out_filter 002 in_insert x in_drop 1
```

```
sysconf -askpin 1 -pincode set1 12345
```

```
sysconf -sendxcode 1 -access set1 12345#
```

- service:

0 → Dial In Service

in Dial In Service, VoiceGATE GATEWAY will pick up incoming calls from PSTN, play pre-recorded voice greeting or, and then have users

to make a 2nd dial to destination.

- 1 → Direct In Line Service (this feature must be implemented in a pair of FXO products in Gatekeeper mode and set bureau –table command)

In Direct In Line Service VoiceGATE GATEWAY will connected via gatekeeper to pre-defined E.164 number.

For example:

\$bureau –table 1 192.168.4.184 123

(please refer to 5.21 bureau command)

If L1 of VoiceGATE GATEWAY is assigned to pre-defined E.164 number 123 in Direct in line mode. When users from PSTN make a call to L1 of VoiceGATE GATEWAY, it will sent out the number 123 to GK, and GK will route this number to the endpoint which registered E.164 is 123 without 2nd dial.

Note: In Direct In Line service, must set VoiceGATE GATEWAY sysconf –2nddial 0

- 2 → HotLine/ LineToLine Service (this feature must be implemented in a pair of FXO products in P2P mode and set bureau –table command)

HotLine Service provides Hot Line function, which connects directly to pre-defined destination. For ex: if L1 of VoiceGATE GATEWAY is assigned to destination address 192.168.1.12 in Hot Line Mode. When users from PSTN make a call to L1 of VoiceGATE GATEWAY, it will directly connect to 192.168.1.12 without a 2nd dial.

Note: In hotline service, must set VoiceGATE GATEWAY sysconf –2nddial 0 .

LineToLine Service is like HotLine Service, but ask for a specific line number. For ex: if L1 of VoiceGATE GATEWAY is assigned to destination address 192.168.1.12 /Line4 in LineToLine Mode. When users from PSTN make a call to L1 of VoiceGATE GATEWAY, it will directly connect to 192.168.1.12 and choose Line4 to call out to PSTN. This is mostly applied to ITSP, who provides international VoIP solution.

Note: In LineToLine service, must set VoiceGATE GATEWAY sysconf –2nddial 1 .

- ringdet: to define ring detection method. (0 is for old hardware version; 1 for new hardware version)
- plan: It is for setting dial-numbering plan. While e164 number is three digits, the plan should be set as 3 or 0. The plan 0 is for any positive digits
- port: This command can enable or disable individual port. The default value is set to enable all ports.
- idto: The duration of two pressed digits in dial mode
- eod: Digit type of end of dialing. (0:No end of dialing, 1:[*] button, 2:[#]button)

- seizure: line seizure mode.

None (0) → when calling from IP side, choose L1 every time if it is available.

UCD (1) → when calls from IP side, choose L1 for the first time, and L2 for the 2nd time, (cyclic)

Note: Do not enable this function together with **group** (please refer to 5.18).

- 2nddial: This command is necessary for setting one time dial method use. While user would like to skip 2nddial process, VoiceGATE GATEWAY must close 2nddial and set as 0 (2nddial off). The default value is set as 1 (2nddial on).

- drule: This command only works while 2nddial is off. When user would like to make an outgoing call or receive an incoming call shortly, it is necessary to set the following three commands belonged to drule.

- drop: drop the dial digit.
- insert: insert the dial digit
- filter: filter the dial digit.

Note:

1. out: Through VoiceGATE GATEWAY to dial out to another Gateway's e164 number. When making an outgoing call, it is necessary to set three commands in order, filter, insert then drop.

Example: sysconf -drule out_filter 002886 out_insert 0 out_drop 02

2. in: Through pass VoiceGATE GATEWAY in order to connect with PSTN / PBX side. When making an incoming call from other Gateway, the three commands is necessary to be set in order, drop, insert, then filter.

Example: sysconf -drule in_drop 002886 in_insert 0 in_filter 02

3. 3. While the specified digit would like to be deleted, use the character x instead of any digits have configured.

-askpin:

0 → disables ASKPIN function

1 → enables ASKPIN function, and apply to the whole unit. Every channel uses the same PINCODE.

2. → enables ASKPIN function, and apply to each channel respectively. Every channel uses a different pincode.

-ring: To set when dial in VoiceGATE GATEWAY from PSTN side, VoiceGATE GATEWAY will pick the call immediately or rings for specific times before picks up.

0 →disable: pick up immediately

1-5 →times of ring before VoiceGATE GATEWAY picks up.

- callalive: Call Alive function (1 = ON; 0 = OFF). The function is used to check if the opposite party is alive when connection is established. When CallAlive is activated, VoiceGATE GATEWAY will send H.245 RoundTripDelay message to other party, and wait for response. If the other party cannot respond the message in 10 seconds, VoiceGATE GATEWAY will regard the opposite party as IDLE state and disconnect the call. When CallAlive is deactivated, RoundTripDelay message will not be sent during connection.

- keypad: keypad type when relay DTMF signal.
 - 0 → In-Band
 - 1 → h.245 alphanumeric
 - 2 → h.245 signal type
 - 3 → q.931 user info
- pincode: to specify 2 sets of pincode.
- sendxcode: send access code after connection (1 = ON; 0 = OFF)
- access: specify access codes (per port basis) .

Note:

1. This feature can only implement with LineToLine service. Please refer to –service above.
2. This function can help users to restrict callers to dial particular numbers from IP side to PSTN side. For example, if user set sysconf –access set1 1111, when callers call from IP side and enter VoiceGATE GATEWAY port 1, if user dial 234 after hearing dial tone, VoiceGATE GATEWAY will dial out 1111234.

```
usr/config$ sysconf –sendxcode 1 –access set1 1111
```

5.14 [h323] command

This command is to configure H.323 related parameters.

```
usr/config$ h323
```

H.323 stack information and configuration

Usage:

h323

h323 [-gk ipaddress] [-multicast used] [-e164 number] [-alias h323id]
 [-rtp port] [-h245 port] [-ttl time] [-gkfind port] [-ras port]
 [-range [start num1] [end num2]]

h323 -print

-print	Display H.323 stack information and configuration.
-mode	Configure as Gatekeeper mode or Peer-to-Peer mode.
-gk	Gatekeeper ip address. (0.0.0.0 ~ 255.255.255.255)
-gkname	Gatekeeper ID
-dfgw	Default Gateway ip address. (0.0.0.0 ~ 255.255.255.255)
-e164	IP side registered number (phone number).
-alias	IP side registered H.323 alias (account name).
-gkdis	Gatekeeper auto discovery (On=1, Off=0).
-rtp	RTP port number (1024~65532).
-h245	H.245 port number (N/A).
-ttl	RAS TTL time (0~3600 second).
-gkfind	Gatekeeper finding port (1024~65535).
-gwtype	Register as Gateway (1) or Terminal (0) type
-ras	Gatekeeper RAS port (1024~65535).
-range	Dynamically allocated port range (1500~65535).
-respto	Max waiting time for 1st response to a new call (1~200).
-connto	Max waiting time for call establishment after receiving 1st response of a new call (1~20000).

Note:

H.245 port configuration is not available now.

Options -gk -e164 -alias -multi -ttl -gkfind -ras are ignored when

RAS mode is configured as Peer-to-Peer mode.

Example:

h323 -gk 210.59.163.171 -e164 0 -alias fxo

h323 -mode 1

Parameters Usage:

-print	print current h323 related settings
-mode	alternatives for gatekeeper or peer-to-peer mode (0=gatekeeper mode; 1=peer-to-peer mode). If users select gatekeeper mode, a extra gatekeeper is need when VoiceGATE GATEWAY is in
-gk	to assign gatekeeper's IP address when VoiceGATE GATEWAY is in gatekeeper mode.
-gkname	to assign Gatekeeper ID when VoiceGATE GATEWAY is in gatekeeper mode.
-dfgw	to set IP address of default gateway, this function is the same as Microsoft NetMeeting. A.To implement this feature both endpoints must be under peer-to-peer mode. B.If the other endpoint is FXO products, such as VoiceGATE GATEWAY , which have to set as sysconf -2nddial 0 to make one-stage dialing. ● From PSTN side dial in VoiceGATE GATEWAY, when hearing greeting user can dial remote PSTN number under default gateway, VoiceGATE GATEWAY will automatically dial to default gateway, then default gateway will dial this number to PSTN side. ● For example, user wants to dial from VoiceGATE GATEWAY A to ext.888 under VoiceGATE GATEWAY B, user only have to dial 888 after hearing greeting of VoiceGATE GATEWAY . C.If the other endpoint is FXS products such as VoiceGATE FXS : From PSTN side dial in VoiceGATE GATEWAY, when hearing greeting user can dial line number of VoiceGATE FXS.
-e164	e164 number, which is registered as phone number in
-alias	gatekeeper identification in h323 world for other parties' recognition. The field might be used as a key of authorization or accounting in some VoIP application. It is recommended to assign a special name, or it might conflict with other devices.
-gkdis	Switch ON or OFF gatekeeper discovery function (1 = ON; 0 = OFF). When it's ON, VoiceGATE GATEWAY will send GRQ with GK ID to default gatekeeper. If the GK ID didn't matched, GW will send GRQ with GK ID in multicast.
-rtp	to allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy.
-h245	to assign h.245 port number, NOT AVAILABLE for the moment.
-ttl	to set timer for TTL(Time To Live). VoiceGATE GATEWAY would send RRQ, with keepAlive, to gatekeeper periodically according to TTL timer.
-gkfind	gatekeeper finding port. Port number, which VoiceGATE GATEWAY uses it to discover a gatekeeper. Default value is 1718.

- gwtype to set VoiceGATE GATEWAY register mode as terminal or gateway, 0 as terminal 1 as gateway. Please notice that if set VoiceGATE GATEWAY as terminal mode, must set sysconf -2nddial 1 (refer to 5.12).
- ras to set default gatekeeper RAS port number. Default value, 1719, is well-known port for RAS communication.
- range to allocate dynamic port range, which VoiceGATE GATEWAY might be using.
- respto response timeout. Max waiting time for 1st response to a new call (1~200).
- connto connection timeout. Max waiting time for call establishment after receiving 1st response of a new call (1~20000).

5.15 [gk] command

This command is to configure embedded simple gatekeeper related parameters. If user doesn't have a gatekeeper or WellTech Call Manager, VoiceGATE GATEWAY provides a simple embedded gatekeeper for up to 10 endpoints.

```
usr/config$ gk
```

Gatekeeper information and configuration

Usage:

```
gk [-add type1 [[type2]...]] [-delete h323] [-ttl value]
   [-enable 0/1] [-security enable/disable]
```

- print *Display the enabled debug flags.*
- enable *Enable simple gatekeeper*
- ttl *Set TTL value*
- add *Add dynamic endpoint*
 (h323 ID, E164, IP, port, type)
- delete *Delete dynamic endpoint*
- security enable *Enable security check*
- security disable *Disable security check*
- security add *Add security record*
- security delete *Delete security record*

Example:

```
gk -add h323 256 192.168.1.1 1720 0
gk -delete h323
gk -security delete h323
gk -security add h323
```

Parameters Usage:

- print print current embedded gatekeeper information and configurations.
- enable to enable gatekeeper feature(gk -enable 0/1)
- ttl to set timer for TTL(Time To Live).In this period of time if endpoint doesn't send RRQ to GK,GK will determine this endpoint as not exist anymore and delete it from registered
- add to add a dynamic endpoint that doesn't send RRQ to GK. User can predefine an endpoint in GK, and GK will determine this endpoint has already registered to GK, though it doesn't send register request to GK.
(gk -add "H.323 ID" "e164" "IP address" "signaling port")

“gateway type,0=terminal,1=gateway” ; ex. gk –add test 123 10.1.1.1 1720 0)

Note:After you reboot the machine, the register information will disappear.

- delete To delete dynamic endpoint which user added formerly.
(gk –delete “H.323 ID”)
- security enable To enable security check. If this function is enabled, GK will only accept registration request from endpoints, which are added with gk –security add command.
- security disable To disable security check.
- security add To add endpoints to register to GK which enable security check.(gk –security add “H.323 ID”)
- security delete To delete endpoints that added formerly in security check list.(gk –security delete “H.323 ID”)

5.16 [voice] command

The voice command is associated with the audio setting information. There are four voice codecs (g.729a optional) supported by VoiceGATE GATEWAY.

```
usr/config$ voice
```

Voice codec setting information and configuration

Usage:

```
voice [-send [G723 ms] [G711A ms] [G711U ms] [G729A ms] ]
      [-volume [voice level] [input level] [dtmf level]] [-nscng G723 used]
      [-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
```

```
voice -print
```

```
voice -priority [G723] [G711A] [G711U] [G729A]
```

```
-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.729A (20/40/60 ms)
```

```
-priority Priority preference of installed codecs.
```

```
          G.723
```

```
          G.711A
```

```
          G.711U
```

```
          G.729A
```

```
-volume   Specify the following levels:
```

```
          voice volume (0~63, default: 28),
```

```
          input gain (0~63, default: 28),
```

```
          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: 100).
```

```
-maxdelay Setting of jitter buffer max delay. (0~150, default: 150).
```

Example:

```
voice -send g723 60 g711a 60 g711u 60 g729a 60
```

```
voice -volume 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 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.729a 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.
- volume There are three adjustable value. **voice volume** stands for volume, which can be heard from VoiceGATE GATEWAY side; **input gain** stands for volume, which the opposite party hears. **dtmf** volume stands for DTMF volume/level, which sends to its sender or receiver.
- 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.*

5.17 [tos] command

TOS service allows users to achieve QoS on IP network.

```
usr/config$ tos
```

IP Packet ToS(type of Service)information and configuration

Usage:

```
tos [-rtptype precedence]
```

```
    [-rtpdelay mode]
```

```
    [-rtpthruput mode]
```

```
    [-rtpreliab mode]
```

```
tos -print
```

```
    [-sigtype][[-rtptype]][[-rtcptype]
```

```
    0 routine.
```

```
    1 priority.
```

```
    2 immediate.
```

```
    3 flash.
```

```
    4 flash override.
```

```
    5 critic.
```

```
    6 internet control.
```

```
    7 network control.
```

```
    [-sigdelay][[-rtpdelay]][[-rtcpcdelay]
```

```
    0 normal delay.
```

<i>[-sigthruput]\[-rtpthruput]\[-rtcpthruput]</i>	<i>1 low delay.</i>
	<i>0 normal throughput.</i>
<i>[-sigreliab]\[-rtpreliab]\[-rtcpreliab]</i>	<i>1 high throughput.</i>
	<i>0 normal reliability.</i>
	<i>1 high reliability.</i>

Example:

tos -rtptype 7 -rtpdelay 0 -rtpthruput 0 -rtpreliab 0

Parameter Usages:

-print : display current TOS values configurations.

-sigtype configure TOS type of signaling packets from 0 to 7

-rtptype configure TOS type of RTP packets from 0 to 7

-rtcpctype configure TOS type of RTCP packets from 0 to 7

-sigdelay configure signaling packets as normal delay or low delay

-rtpdelay configure RTP packets as normal delay or low delay

-rtcpdelay configure RTCP packets as normal delay or low delay

-sigthruput configure signaling packets as normal throughput or high throughput

-rtpthruput configure RTP packets as normal throughput or high throughput

-rtcpthruput configure RTCP packets as normal throughput or high throughput

-sigreliab configure signaling packets as normal reliability or high reliability

-rtpreliab configure RTP packets as normal reliability or high reliability

-rtcpreliab configure RTCP packets as normal reliability or high reliability

Note: Users should be aware that TOS is effective only when network devices (for ex: router, switch.. etc.) support TOS.

5.18 [tone] command

Tone detection of VoiceGATE GATEWAY is configurable if the bureau line is connected to PABX or PSTN. Users can refer to “**greetrd**” command for tone recording and analysis. Sometimes the frequencies might shift from standard level. In such a situation, users have to adjust the tone value manually using this command.

usr/config\$ tone

Setup of call progress tones

Usage:

tone -toneX LowFreq HighFreq LowFreqLevel HighFreqLevel TOn1 TOff1 TOn2 TOff2

tone -print

Note:

toneX has the following possibility:

busy1 busy2 reorder1 reorder2 ringtone1 ringtone2 dialtone

Example:

tone -busy1 400 0 8 0 50 50 0 0

tone -dialtone 400 0 19 0 25 25 0 0

5.19 [support] command

This command provides some extra functions that might be needed by users.

usr/config\$ support

Special Voice function support manipulation

Usage:

support[-tunnel enable]

support -print

-t38 T.38(FAX) enabled/disabled.
-fstart Fast start enabled/disabled.
-tunnel H245 Tunneling enabled/disabled.
-h245fs H245 separate channel after faststart.

Example:

support -fstart 1
support -tunnel 0
support -h245fs 1

Parameter Usages:

-print print current setting in **support** command.

-t38 to switch ON/OFF (1 = ON; 0 = OFF) T.38 function. T.38 function is for FAX. If user will use FAX machines, please switch on T.38 function.

-fstart to switch ON/OFF (1 = ON; 0 = OFF) FastStart function. Fast Start function can shorten the connection time if the opposite party also support FastStart.

-tunnel to switch ON/OFF (1 = ON; 0 = OFF) H.245 tunneling function. If the function is ON, VoiceGATE GATEWAY will send H.245 (Call Control messages) via H.225's (Call Signal messages) link. The function is effective only when both side support h245 tunnel.

-h245fs to switch ON/OFF (1 = ON; 0 = OFF) H.245 separate channel after fast start or not. (1 = ON; 0 = OFF)

Note:

1. *it is not recommended to change the value in this command, only if users do know well the application. This might cause incompatibility with other devices.*
2. *If user wants to use T.38 fax under fast start mode, please make sure "h245fs" function is enabled, or fax can't work normally.*

5.20 [group] command

This command is for grouping 4 ports of VoiceGATE GATEWAY. If users need to register at least 2 numbers separately to gatekeeper, then this command is

needed for such an application.

```
usr/config$ group
```

PSTN side grouping information and configuration

Usage:

```
group -print | -enable | -disable |
      -number group_number -pattern pattern_numbers -e164 e164_numbers |
      -pattern pattern_numbers -e164 e164_numbers |
      -e164 e164_numbers
```

Comment:

```
-print      : Print current group configuration
-enable     : Enable PSTN Grouping
-disable    : Disable PSTN Grouping
-number     : Set number of divided groups
-pattern    : Set number of members in each group
-e164       : Set E.164 number for each group
```

Example:

```
group -print
group -enable
group -disable
group -number 2 -pattern 3 3 -e164 01 02
group -pattern 3 1 -e164 100 200
group -e164 11 22
```

Parameter Usages:

- print : display current grouping information
- enable : enable grouping function
- disable : disable grouping function
- number : set how many groups will be divided
- pattern : set how many members in each group
- e164 : set e164 of each group

For ex: if users need to divide VoiceGATE GATEWAY into L1 in the 1st group, and L2 in the 2nd group), and have them register to gatekeeper separately (e164=100 for 1st group; e164=200 for 2nd group). They have to use the following command:

```
usr/config$ group -pattern 1 3 -e164 100 200
```

Note: GROUP function is effective only in gatekeeper mode.

5.21 [bureau] command

Type **bureau** to display the command usage.

```
usr/config$ bureau
```

Bureau line setting information and configuration

Usage:

```
bureau [-pstn number] [-hold used] [-table [Port DestIP TELnum]]
```

bureau -print

- print *Display Bureau line informatio and configuration.*
- pstn *PSTN number (per port basis). This number is used to display as a caller ID when the caller ID is not available. The maximum digit length is 32.*
- hold *Specify the hold tone generation (using PCM file). (On/Off) Setting value (On=1, Off=0).*
- table *Set Hot line/Line To Line information. (Port range: 1~2)*

Note:

Hotline feature should be used together with:

- \$sysconf -service 2 (HotLine service)*
- \$sysconf -2nddial 0 (2nddial off)*
- \$h323 -mode 1 (peer-to-peer mode)*

Line To Line feature should be used together with:

- \$sysconf -service 2 (HotLine service/Line To Line)*
- \$sysconf -2nddial 1 (2nddial on)*
- \$h323 -mode 1 (peer-to-peer mode)*

Example:

```
bureau -pstn 2011 2012
bureau -table 1 192.168.4.69 628 2 192.168.4.200 9992
```

Parameter Usages:

- print: display bureau line information and configuration.

```
usr/config$ bureau -print
```

Bureau line setting relate information

```
PSTN number           : 2011 2012 2013 2014 2015 2016
Hold tone generation : On
Hot line / Line to Line table
```

<i>Port</i>	<i>Destination Address</i>	<i>Remote TEL/CHANNEL</i>
<i>1</i>	<i>192.168.4.69</i>	<i>628</i>
<i>2</i>	<i>192.168.4.69</i>	<i>628</i>

- pstn: PSTN number (per port basis). This number is used to display as a caller ID when the caller ID is not available. The maximum digit length is 32.
- hold: while the terminals support H.450 **hold** function, the VoiceGATE GATEWAY will play the hold tone to PSTN side.
- table: Set Hot line/LineToLine destination IP and e164 numbers information.

Note:

- 1. HotLine and LineToLine functions are using the same table.*
- 2. In HotLine service, user have to set No. prepared to dial out; in LineToLine service ,user have to set port No.*
For example, if user set bureau -table 1 192.168.4.69 628 in hotline service, after user dial in VoiceGATE GATEWAY port 1, VoiceGATE GATEWAY will direct dial to 192.168.4.69 and dial 628 to PSTN side, then Phone 628 will ring, user will hear ring back tone.
If user set bureau -table 1 192.168.4.69 1 in LineToLine service, after user dial in VoiceGATE GATEWAY port 1 , VoiceGATE GATEWAY will direct dial

to 192.168.4.69 port 1, user will hear dial tone, then user can dial out No. to PSTN side.

5.22 [prefix] command

This function can do digits replacement of incoming call from IP side or PSTN side.

```
usr/config$ prefix
```

Prefix setting information and configuration

Usage:

prefix [-pstnrule index oldnumber newnumber (index = 1 ~ 6)]

[-iprule index oldnumber newnumber (index = 1 ~ 6)]

prefix -print

-print Display prefix information and configuration.

-pstnrule Set PSTN incoming prefix rule information.

-iprule Set IP incoming prefix rule information.

Example:

prefix -pstnrule 1 2 8862 : prefix 2 will be replaced with 8862

Parameter Usages:

-print print current setting in **prefix** command.

-pstnrule to do digit replacement of incoming call from PSTN side. Ex, to set **prefix -pstnrule 1 123 456**, which means the first set of PSTN side rule is: IF user press 123888 after dialing in VoiceGATE GATEWAY from PSTN side, the real number dialed out will be replaced with 456888.

-iprule to do digit replacement of incoming call from IP side. Ex, to set **prefix -iprule 1 456 789**, which means the first set of IP side rule is: IF user press 456000 after dialing in VoiceGATE GATEWAY from IP side, the real number dialed out will become 789000.

5.23 [rom] command

ROM file information and firmware upgrade function.

```
usr/config$ rom
```

ROM files updating commands

Usage:

rom [-app] [-dsptest] [-dspcore] [-dspapp] [-rbpcm] [-htpcm]

[-greeting] -s TFTP/FTPserver ip -f filename

rom [-method mode] [-ftp username password]

rom -print

-print show versions of rom files. (optional)

5.24 [passwd] command

For security concern, users have to input the password before entering configuration mode.

usr/config\$ passwd

Password setting information and configuration

Usage:

passwd -set Loginname Password

Note:

Loginname can be only 'root' or 'administrator'

Example:

passwd -set root 2fxo

Parameter Usages:

-passwd <login name> <password>

Note: <login name> can be "root" or "administrator" only. "root" and "administrator" have the same authorization, except 3 commands that can be executed by "root" only – "passwd –set root", "rom –boot", and "flash –clean"

Appendix: Web configuration

Web management simple user guide

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

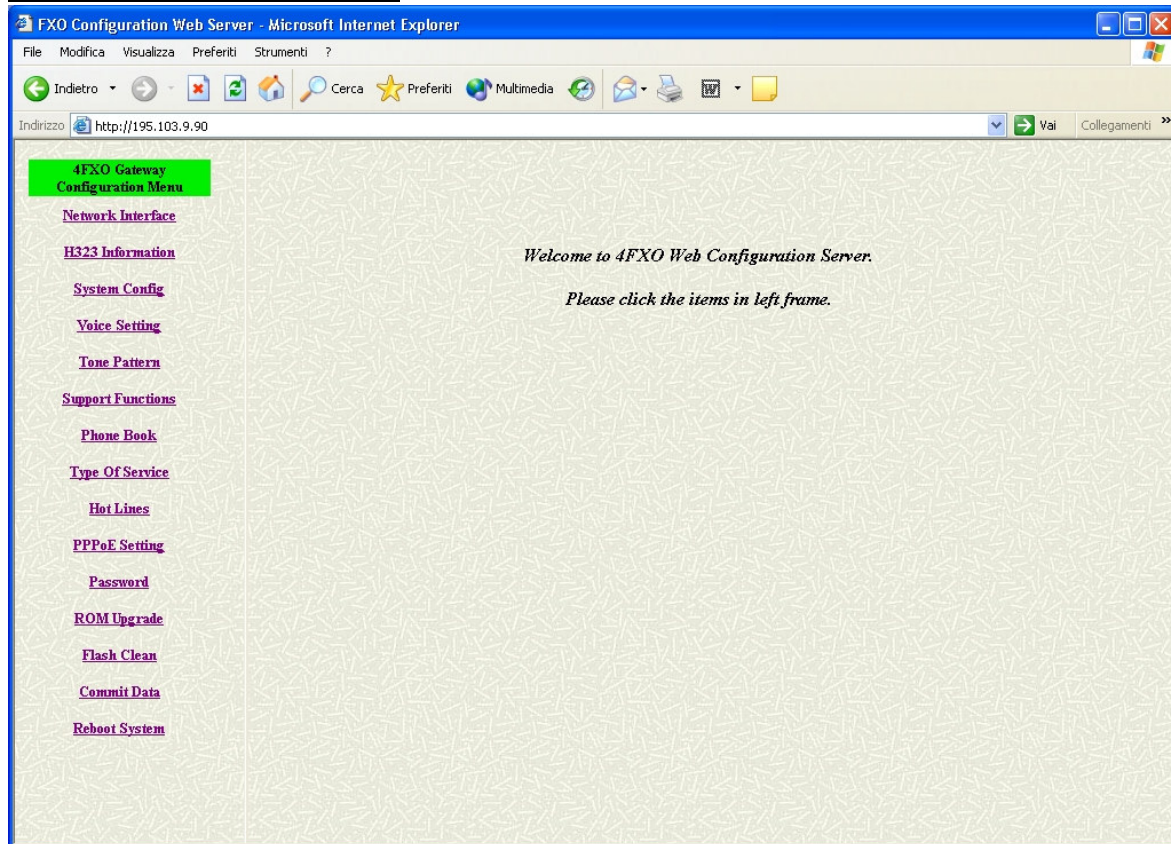
The configuration function and step is similar with the way through command line. Basically this version is not the finalized version for web interface. Initially user please refer to the manual for more information. Below provide a simple user guide for user to configure via web interface. Next version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

Step 1. Browse the IP Address which has predefined via RS-232

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 configure

Most of all commands displayed in console / telnet are transfer to web interface. The most important commands are [Network Interface](#), [H323 Information](#), [Commit Data](#) and [Reboot System](#). The method is as the same as command mode.

1.1 Network Interface

- [VoiceGATE GATEWAY IP Address](#): Set IP Address
- [Subnet Mask](#): Set the Subnet Mask
- [Default routing gateway](#): Set Default routing gateway
- [DHCP](#): Enable / Disable to DHCP mode
- [SNTP](#): Enable / Disable the Simple Network Time Protocol
- [SNTP Server Address](#): Set SNTP Server Address
- [GMT](#): Set time zone for SNTP Server time
- [IP Sharing](#): Enable it if behind IP Sharing router
- [IP Sharing Server Address](#): Set WAN IP Address of IP Sharing Server router if it is a fixed one.

Please be noted:

If the WAN IP Address of IP Sharing Server router is not a fixed one, it is not necessary to input any values.

If behind the dynamic WAN IP Address situation please configure as GK mode and select Welltech Call Manager as proxy server.

1.2 H323 Information

- **Mode:** Select GK mode or Peer-to-Peer mode
- **Gatekeeper IP Address:** Set Gatekeeper IP Address
- **Gateway Type:** Set Register Type to GK (Gateway / Terminal)
Registered Prefix: Set Prefix Number as E.164 number
- **Registered Alias:** Set Registered Alias as H323 ID
- **Gatekeeper Discovery** **RTP Port** **Time to Live (TTL)**
Gatekeeper finding port **RAS Port** **Response Timeout**
- **Connection Timeout:** For Advanced User Only

The screenshot shows the 'FXO Configuration Web Server - Microsoft Internet Explorer' window. The address bar shows 'http://195.103.9.90'. The left sidebar contains a '4FXO Gateway Configuration Menu' with links: Network Interface, H323 Information (selected), System Config, Voice Setting, Tone Pattern, Support Functions, Phone Book, Type Of Service, Hot Lines, PPPoE Setting, Password, ROM Upgrade, Flash Clean, Commit Data, and Reboot System. The main area is titled 'H323 Configuration' and contains the following fields:

Mode:	<input checked="" type="radio"/> GK routed <input type="radio"/> Direct
Gatekeeper IP Address:	217 . 57 . 153 . 140
Gateway Type:	<input checked="" type="radio"/> Gateway <input type="radio"/> Terminal
Registered Prefix:	0
Registered Alias:	4FXO-0017a5
Gatekeeper Discovery:	<input type="radio"/> enable <input checked="" type="radio"/> disable
Gatekeeper ID:	GK2
RTP Port:	16384
Time To Live (TTL):	60
Gatekeeper finding port:	1718
RAS Port:	1719
Response Timeout:	5
Connection Timeout:	200

An 'OK' button is located at the bottom right of the configuration area.

1.3 System Config

- **Keypad Type:** Select different DTMF Keypad Type (For Advanced User)
- **Dial Plan:** Set DTMF digit limitation (0 is for any digits)
- **Inter Digit Time:** Set the DTMF inter digit time (second)
- **End of Dial:** Digit type of end of dialing. (0:No end of dialing, 1:[*] button, 2:[#] button)
- **2nddial:** This command is necessary for setting one time dial method use. While user would like to skip 2nddial process, VoiceGATE GATEWAY must close 2nddial and set as 0 (2nddial off). The default value is set as 1 (2nddial on).
- **Call Alive:** Enable the function to check connection (Both side must support)
- **Line Seizure:** Choose line seizure mode (None/UCD)
- **Gateway Service:** Specify gateway service type.
(0: Dial in service,1: Direct in line service,2: HotLine/LineToLine service.)

System Configuration	
Keypad Type:	<input type="radio"/> In-Band <input type="radio"/> H.245(Alpha) <input checked="" type="radio"/> H.245(Sig) <input type="radio"/> Q.931 <input type="radio"/> RFC2833
Dial Plan:	<input type="text" value="0"/>
Inter Digit Time:	<input type="text" value="5"/>
End of Dial:	<input type="radio"/> No Eod <input type="radio"/> * <input checked="" type="radio"/> #
2nd Dial:	<input checked="" type="radio"/> ON <input type="radio"/> OFF
Call Alive:	<input type="radio"/> ON <input checked="" type="radio"/> OFF
Line Seizure:	<input type="radio"/> UCD <input checked="" type="radio"/> None
Gateway Service:	<input type="text" value="0"/>
<input type="button" value="OK"/>	

1.4 Voice Setting (For Advanced User)

- **Frame Size:** It got wrong order with “Codec Priority”. Select the Codec Priority. (For Advanced User)
- **Codec Priority:** It got wrong order with “Frame Size”. Select the packet size in sending process. (For Advanced User)
- **G.723 Silence Suppression:** Enable / Disable (For Advanced User)
- **Volume:** Adjust the volume in “Voice” (sending out); “Input” (receiving); “DTMF” (DTMF sending out) **Please Noted the value is limited.**
- **Echo Cancel:** Enable / Disable (suggested always Enable)
- **Jitter Buffer:** Min. Delay and Max. Delay (For Advanced User)
- **Optimized Factor (Jitter):** (For Advanced User)

Voice Configuration				
Frame Size	1st <input type="text" value="G.723.1"/>	2nd <input type="text" value="G.711A-Law"/>	3rd <input type="text" value="G.711mu-Law"/>	4th <input type="text" value="G.729a"/>
Codec Priority	G.723.1 <input type="text" value="30ms"/>	G.729a <input type="text" value="20ms"/>	G.711mu <input type="text" value="20ms"/>	G.711A <input type="text" value="20ms"/>
G.723 Silence Suppression:	<input type="radio"/> enable <input checked="" type="radio"/> disable			
Volume:	voice <input type="text" value="32"/>	input <input type="text" value="32"/>	DTMF <input type="text" value="23"/>	
Echo Cancelor:	<input checked="" type="radio"/> enable <input type="radio"/> disable			
Jitter Buffer:	Min. Delay <input type="text" value="90"/>		Max. Delay <input type="text" value="150"/>	
Optimized Factor (Jitter):	<input type="text" value="9"/>			
<input type="button" value="OK"/>				

1.5 Phone Pattern (For Advanced User)

- [Busy Tone:](#)
- [Reorder Tone:](#)
- [Ring Tone:](#)
- [Dial Tone:](#)

FXO Configuration Web Server - Microsoft Internet Explorer

File Modifica Visualizza Preferiti Strumenti ?

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4FXO Gateway Configuration Menu

- [Network Interface](#)
- [H323 Information](#)
- [System Config](#)
- [Voice Setting](#)
- [Tone Pattern](#)
- [Support Functions](#)
- [Phone Book](#)
- [Type Of Service](#)
- [Hot Lines](#)
- [PPPoE Setting](#)
- [Password](#)
- [ROM Upgrade](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Tone Configuration

Busy Tone I:	High(freq) 620	Low(freq) 480	High(lev) 8	Low(lev) 8	On1 50	Off1 50	On2 50	Off2 50
Busy Tone II:	High(freq) 0	Low(freq) 400	High(lev) 8	Low(lev) 8	On1 50	Off1 50	On2 0	Off2 0
Reorder Tone I:	High(freq) 620	Low(freq) 480	High(lev) 8	Low(lev) 8	On1 25	Off1 25	On2 25	Off2 25
Reorder Tone II:	High(freq) 440	Low(freq) 350	High(lev) 8	Low(lev) 8	On1 25	Off1 25	On2 25	Off2 25
Ring Tone:	High(freq) 480	Low(freq) 440	High(lev) 13	Low(lev) 13	On1 200	Off1 400	On2 0	Off2 0
Dial Tone:	High(freq) 0	Low(freq) 400	High(lev) 0	Low(lev) 10	On1 50	Off1 0	On2 50	Off2 0

OK

1.6 Support Functions (Both side must support)

- **T.38:** Enable for T.38 FAX
- **Fast Start:** Enable to do Fast Start
- **H.245 Tunneling:** Enable to open H.245 Tunneling

Support Configuration	
T.38 FAX:	<input type="radio"/> enable <input checked="" type="radio"/> disable
Fast Start:	<input type="radio"/> enable <input checked="" type="radio"/> disable
H.245 Tunneling:	<input type="radio"/> enable <input checked="" type="radio"/> disable
<input type="button" value="OK"/>	

1.7 Phone Book (For Peer-to-Peer mode only)

Input the Name, IP Address and E.164 No. for the destination device.

Please Note: The E.164 No. will be carried together to the destination side. It is said if the destination side is requested to match its E.164 No. (Line No.), user can not input any digit he wished.

Phone Book			
Index	Name	IP_Address	e164

New Record			
Index	Name	IP Address	E164 No.
<input type="text"/>	<input type="text"/>	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>	<input type="text"/>
<input type="button" value="Add Data"/> <input type="button" value="Delete Data"/>			

1.8 Type Of Service

Adjust the parameter in IP Header for router identity purpose.

If the version has PPPoE function, ToS is not available to do any configuration.

TOS Configuration				
Signaling TOS:	Precedence Flash Override(4) ▼	Delay Normal(0) ▼	ThruPut Normal(0) ▼	Reliability Normal(0) ▼
RTP TOS:	Precedence CRITIC/ECP(5) ▼	Delay Normal(0) ▼	ThruPut Normal(0) ▼	Reliability Normal(0) ▼
RTCP TOS:	Precedence Routine(0) ▼	Delay Normal(0) ▼	ThruPut Normal(0) ▼	Reliability Normal(0) ▼
OK				

1.9 Hot Lines

Select HOST Port and set Destination Address. The Remote Number is subject to the Destination's configuration.

Bureau		
PORT	DESTINATION_ADDRESS	Remote_TEL(FXS) / Port_No(FXO)
1	192.168.4.69	628
2	192.168.4.69	628
3	192.168.4.69	628
4	192.168.4.69	628

New Record		
Port	Destination Address	Remote Number
<input type="text"/>	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>	<input type="text"/>
SET		

1.9 Password

First select login name as root or administrator, then enter current password , new password and confirm new password again.

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

1.10 ROM Upgrade

- **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

ROM Configuration	
TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

1.11 flash Clean

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

Please be noted: Once execute this function, user must re-configure all other commands except IP Address.



1.12 Commit Data

After configuration, user has to commit data then reboot machine.

It is an important step after every configuration.



1.13 Reboot System

After commit configuration, user has to REBOOT device.

It is an important step after every configuration.

