H-series VoIP Router

(H3000 / H1100 / H5xx / H6110)

Web Administor Interface

User's Manual

Version: 2.03

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1.   Web

1.1 Login Web UI

Welcome to buy and use ALLWIN Tech's VoIP Router, this manual will help you understand the operation using this device (hereinafter referred to as AwG) of the WEB management interface.
You can manage your VoIP Router by AwG built-in Web management interface.
Please prepare a computer connected to the LAN side of AwG. As AwG the default DHCP server service is ON, so keep your computer's TCP / IP settings to "Obtain an IP address automatically" in order to obtain the right from AwG DHCP IP.

  AwG default on the network which will become the Gateway, the default IP is 192.168.22.1, at the same time, it will be assigned to computers connected to the LAN side IP address of a 192.168.22.x.
To set your computer's TCP / IP, by following the path set (in Windows XP for example):

Start → Control Panel → Network Connections → Local Area Connection content
→ Internet Protorcol (TCP / IP) → content → click Obtain an IP address automatically

To access the management interface, on the computer, open IE browser in the address bar enter:
http://192.168.22.1/, as shown below:
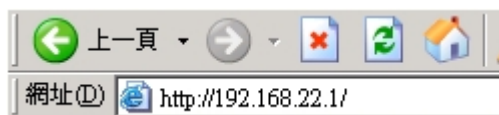
Then the screen will first ask you to enter an administrator account password, the default account is **voip**, the password is **1234**. Enter the correct password to enter the account management interface.

1.2 Web UI function list

AwG provide user-friendly management interface allows you to manage and configure your router and VoIP functionality. Web UI, there are two major key management project: VoIP functions, System Setup details of the items listed below:

● **VoIP Function**

■ Ports Status
■ Auto Provision
■ Line Configure
  ◆ Line Setting
  ◆ Line Interface `FXS/FXO`
  ◆ Tone Setting
  ◆ Tone Detect `FXS/FXO`
  ◆ Line Feature
  ◆ Line Voltage/Current
  ◆ Line Diagnostics
  ◆ Line Impedance
  ◆ PSTN Gain (FXS+PSTN only)
  ◆ Message Indicator
■ DAA Hybrid `FXO`
  ◆ DAA Hybrid Config
  ◆ DAA Hybrid Table
  ◆ Line Rejection
■ Routing Setup
  ◆ VoIP Call Out
  ◆ VoIP Call In
  ◆ VoIP Call In IVR
  ◆ VoIP Routing Profile
  ◆ Forwarding
  ◆ Authorization
■ Register Server
  ◆ Register Status
  ◆ Server #1
  ◆ Server #2
  ◆ Server #3
  ◆ Server #4
■ Advance Setup
  ◆ NAT Traversal
  ◆ Listen Port

- ◆ VoIP Package
- ◆ RTP Summary
- ◆ Flash & Call waiting
- ◆ Gain
- ◆ QoS
- ◆ CDR
- ◆ FoIP
- ◆ Prompt Voice & Beep
- ◆ Call Log
- ■ Application
  - ◆ Ping Test
  - ◆ Centrex
  - ◆ Telnet & SNMP
  - ◆ Call Timer

● **System Setup**

- ■ System
  - ◆ System Status
  - ◆ System Setting
  - ◆ Date & Time
  - ◆ Administrator Settings
- ■ WAN
  - ◆ WAN Settings
  - ◆ WAN Setting #2   (H3000 H5xx series only)
  - ◆ DNS
- ■ LAN
  - ◆ LAN Setting
  - ◆ DHCP Client List
- ■ Wireless (H6110n only)
  - ◆ Basic Settings
  - ◆ Advanced Settings
  - ◆ Security
  - ◆ Access Control
  - ◆ Site Survey
- ■ NAT
  - ◆ Virtual Server
  - ◆ Port Mapping
  - ◆ ALG

- Firewall
  - Denial-of-Service
- Routing
  - Routing Table
  - Static Routing
- Bandwidth & VLAN
  - Bandwidth Control
  - VLAN
- Backup/Restore
  - Configurations
  - VoIP module
- Reboot
- Language/語言
- Save Modification

2. VoIP Function

This section describes the VoIP Setup functions of device, the location of the menu items will be list to represent the range slash.

For example, */ Line Configure / Line Setting /* that is located in the *Line setting* below *Line Configure* menu item..

2.1 Portall Status

2.1.1 / Portall Status /

Port Status: Displays the current call status or a last call result.

Description:

a. **PC Time**: displays the date and time of connected computer.

b. **Gateway Time**: displays the current date and time in the device through the NTP Server on the network or manual set.

   You can set time on */ System / Date & Time /* menu item.

A. Port Message

c. **Port:** display line number.

d. **Type:** Line interface type (divided into 2 types):

   FXO: connected to PSTN or PBX analog e extension lines.

   FXS: connected to a telephone set or PBX Co. line.

e. **Display Name:** VoIP call user display name.

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f. **Status:** current line status display.

  Idle: non-use.

  Signal: Waiting for dial-up or VoIP call connection in progress.

  In: VoIP In-bound call in progress.

  Out: VoIP Out-bound call in progres.

g. **Connected IP:** display connedted remote side type for this call.

- PstnOut: Outbound call to analog line interface.
- PstnIn: Inbound call from analog line interface.
- rs: call via register server.
- IP: direct VoIP call by IP.

h. **Caller ID:** Caller ID.

i. **Start Time:** start time of VoIP call.

j. **End Time:** End time of VoIP call..

k. **Talking Sec:** Total VoIP talking time in seconds.

l. **Dialed number**:

- dial out numbers for out bound call.
- received dialed numbers from In bound call.

B. Error Message

Display the last error message of failure call.

2.2 Auto Provision

2.2.1 /Provision/

To use the auto provision function, the system have to install a dedicate Auto Provision Server for keep all parameters for installed gateways. When Enable the Auto Provision function, the System administer can modify all the Parameters of each gateway on the local Provision Server, and remote gateway will automatic download all the parameters from Provision Server.

The Gateways can link up to five provision servers simultaneously for Redundancy backup the system.

2.3 Line Configure

2.3.1 Line Setting

2.3.1 /Line Configure/Line Setting/

In this configuration page, you can set the name of each line, line number, voice volume, and physical port-related functions.

a. Port: display line numbers, such as the first line or second line of the state, and so on.

b. Interface: Line interface type (divided into 2 types):
  FXO : connected to PSTN lines or PBX analog extension lines.
  FXS : connected to analog telephone set or PBX co. lines.

c. Name: definable the line name, this name will display on the other side device during VoIP call.

d. Line Number: Define line extension number, can be given to each line as the extension number.

e. TxGain: Transmitter gain, adjustable playback on Local phone (handset) volume adjustment, increase the dB value of the local-end phones will increase the playback volume.

f. RxGain: receiver gain, adjustable Local phone microphone volume. Increase the receiver volume will amplifie microphone volume to transmit to the other end of the call.

g. Inbound: Enable or Disable Inbound (VoIP Call Out) function. default is Enable.

h. Outbound: Enable or Disable Outbound (VoIP Call In) function. default is Enable.

i.Hotline: Enable/Disable Hotline function. When Enable Hotline function, user do not need to dial number to make a VoIP call after seize the line (FXO: Ring in or FXS: off hook phone).

For example, If we want the first line to hotline, each time user seizes the first line (FXO: ring in, FXS: off hook phone), without dialing any number, will automatically make a VoIP call to gateway local on 168.56.109.22, and dial 600 automatically as extension. Then we should enable first line's hotline function, and on / Routing Setup / VoIP Call Out / added a dial rule. In the Area Code field to specify the first line input hl1 as Hot Line1, and remember to Strip field, enter 3 to mask out "hl1". In the Prefix field, enter the phone number you want to dial "600" See below:

Index	Remark	Area Code	Min Digits	Max Digits	Destination	Strip	Prefix	Profile	Delete
1	Hot_Line	hl1			10.1.1.1	3	600		Delete

2.3.2   Line Interface FXS/FXO

When using H3000-2 model, you can configure each port to use FXS or FXO.

Line Interface

1.	○ FXS ⊙ FXO
2.	○ FXS ⊙ FXO
3.	○ FXS ⊙ FXO
4.	○ FXS ⊙ FXO
5.	○ FXS ⊙ FXO
6.	○ FXS ⊙ FXO
7.	○ FXS ⊙ FXO
8.	○ FXS ⊙ FXO

[Modify]   [Reset]
(When Set , Save and reboot!)

2.3.2 /Line Configure/Line Interface/

This function must click / Save Modification /to re-start to take it effective.

2.3.3 Tone Setting

Tone	Type	Low Freq	High Freq	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
				Tone Setting			
				Call Progress Tone			
			Detect Tone Busy Cycle: 5				
1	Busy	480	620	500	500	0	0
2	Ring	440	480	1000	2000	0	0
3	Dial	440	440	5000	0	0	0
4	Busy	480	620	250	250	0	0

Modify   Reset

Insert new
Insert To: 5

Add   Delete   Reset

2.3.3/Line Configure/ Tone Setting

A. Call Progress Tone

Here to define the call progress tone for generate/detection. After modify, please click / Save Modification /to re-start to take it effective.

■ Detect Voice Busy Cycle: When detects a match of a busy tone, and the number of cycles up to the value of this setting, the device will be determined to confirm the receipt of a busy tone, that will pass this dropped calls.

B. define Call progress tone

Here you can set up 15 items of audio specification for tones generate and detection, such as dial tone, busy tone, ring tones, etc. Generally call progress tones are between 300 Hz to 2000 Hz.User can set up multiple groups of tone for detection, but only one group will be used as generation.

a. **Tone**: Tone item index, Maximum 15 items.
b. **Type**:
    ■ Dial: dial tone, tone generated to wait user dial.
    ■ Busy: busy tone, generate/detection for line busy.
    ■ Ring: ringback tone, Generate when waiting for answer.
c. **Low freq**: low frequency setting, set the lower frequency of tone.
d. **High freq**: high-frequency setting,set the higher frequency of tone.. Each tone can include two frequencis if only one frequency need, set the **High Freq.** to zero.
e. **T_ON_1, T_OFF_1, T_ON_2, T_OFF_2**:

■ tone cadence interval time : includes dual-band beat interval to four intervals (see below), the lowest is 30 milliseconds. (Unit in mS)
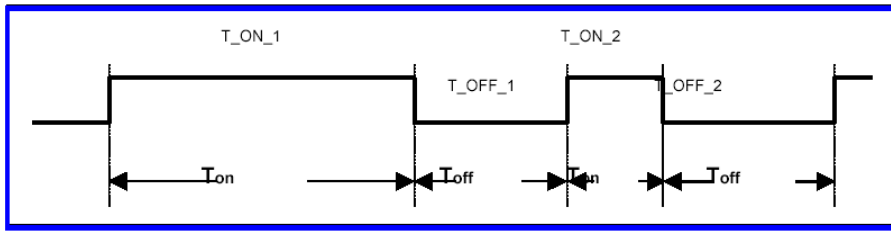
2.3.4 Tone Detect `FXS/FXO`

When the device has FXO interface, here define the buys tone and ring back tone detection parameters.

Tone	Enable	Level	Low Freq	High Freq	Accurate (hz)	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Accurate (%)
Busy1	☑	-40 db	480 hz	620 hz	50 hz	500 ms	500 ms			15 %
Busy2	☑	-40 db	480 hz	620 hz	50 hz	250 ms	250 ms			15 %
Busy3	☐	-35 db	0 hz	0 hz	50 hz	0 ms	0 ms			15 %
Busy4	☐	-35 db	0 hz	0 hz	50 hz	0 ms	0 ms			15 %
Ring1	☐	-35 db	0 hz	0 hz	50 hz	0 ms	0 ms	0 ms	0 ms	15 %
Ring2	☐	-35 db	0 hz	0 hz	50 hz	0 ms	0 ms	0 ms	0 ms	15 %

Modify   Reset

2.3.4 /Line Configure/Line Detect/

Device will disconnect the VoIP call when the busy tone was detected.

2.3.5 Line Feature

2.3.5.1 /Line Configure/ Line Feature

On this page, set the telephone line interface related parameters

A. **Dial Pause signal length [100 ~ 3000] ms**:

Defines the pause time (milliseconds, ms) defined for "," comma symbol used on Prefix field of */ Routing Setting / VoIP Call Out /* or */ VoIP Call In /* . By default, enter a comma "," will cause a pause time 1000 ms between DTMF digit, time can be set at least 100 ms, the maximum is 3000 ms. Users can use multiple consecutive "," to exend the dial pause time.

B. **Loop Current Drop & Polarity Reversal Generate**

If the remote party disconnect the VoIP call, Local FXS interface can enable/disable the following options:

- Disable: Disable FXS interface to generate both line polarity reversal signal and current interrupt signal function, just send busy tone.
- Polaryti Reversal-> Enable   FXS interface to generate line polarity reversal signal.
- Current Drop-> 1 S: Enable FXS interface to generate one second length current interrupt signal.
- Current Drop-> 2 S: Enable FXS interface to generate two seconds length current interrupt signal.
- Current Drop-> 3 S: Enable FXS interface to generate three seconds length current interrupt signal.

C. **When using FXS to answer, decide to bring out the phone number by setting the**

following options:
- Drop out: Do not send, in order to avoid the phone to hear the unnecessary DTMF tones after answer the call..
- Resned: send the DTMF number for PBX to transfer the call.

D. **Method of FXS interface to generate CID (Caller ID),the following options**:
- Disable: turn off, do not send CID to phone
- DTMF: send DTMF CID signal to phone
- FSK Bell: send FSK Bell singal to phone
- FSK ETSI: send FSK ETSI to phone

E. **CID signal detection method for FXO Interfacel**:
- Disable: turn off, do not detect the CID
- DTMF: detection signal of the CID by DTMF
- FSK Bell: detection signal of the CID by FSK Bell
- FSK ETSI: detection signal of the CID by FSK ETSI

F. **Call waiting ID Generate type**：
- Disable: turn off, do not enable the Call Waiting Caller ID
- FSK Bell: generate Caller ID signals use FSK Bell
- FSK ETSI: generate Caller ID signals use FSK ETSI

G. **When VoIP call out，Send ANI by**
- Register Number: send the number of registered
- Line Number: send the setting Line number.
- PSTN CID: Only on the FXO interface, send the received number from the CID.

H. **Define how the FXS interface to ring the phone line when VoIP call in**:
- Free Random: Any unused available line.
- Line number Priority: The 1st line has high priority; it will always ring the 1st line if it is available. When 1st line is busy, it will try to ring 2nd line if it is free.
- Rotation: 1st line ring first, then 2nd line ring next time, when the latest line ring this time, it will come back to ring 1st line next time.
- All: Ring all phone lines if it is available.
- Sequence: Ring all the available phone line one by one, the ring period for ring each phone is definable.
- Period (sec.): define the ring period (seconds) when select "Sequence" ring.

I. **Polarity Detect Auto** FXS/PSTN ：
When enable, the device will detect the FXO interface polarity of the Tip and Ring endpoint,

and then automatically adjust the FXS interface on the Tip, Ring polarity to the same polarity as FXO endpoint. This feature is used for some PBX don't allow to change the polarity on its Co. line during call making..

J. **Define the default FXS phone port interface standby method, the following options** FXS/PSTN:

- **PSTN**:   Phone set on FXS port is standby on PSTN line connected on FXO interface. When the device set accept a VoIP call, the phone will switch to the FXS interface, and will be automatically on-hook the FXO line to avoid the call from PSTN. But if enable the Calling Waiting function on */ Routing Setup / VoIP Call In /* , the FXO line still can ring the phone on busy with call waiting tone.

If your device is not connected to the PSTN line, the phone set connected on FXS port will not hear a dial tone and can not be used in this mode.

- **FXS:** Standby the phone set on FXS interface. When make a PSTN call, the phone will switching to the PSTN Interface through relay.
- **FXS PCM**: Standby the phone set on FXS interface, when make a PSTN call, the device will connect FXS and FXO by internal PCM bus rather than relay,.

K. **Re-hook timeout for PSTN**: (H3000-1 / H1111p)：

When FXS port standby on PSTN port, The PSTN port connected on FXO port will become busy by off hook automatically if no enable call waiting function. Here define the interval time for FXO re-hook the PSTN line.

L. **Re-hook dela** (H3000-1 / H1111p)：

Define the on hook second for re-hook duration.

M. **Line OffHook Voltage** (H3000-1 / H1111p)：

When standby on PSTN, define the voltage value for hook detection. When voltage value is less then the define value, it will become off hook status.

2.3.5.2 /Line Configure/ Line Feature

N. **Loop Current Drop&Polarity Reversal Detect (FXO only)**:
    Enable to detect the current drop or line polarity reversal signal for terminal the call .

O. **FXO Wait dial Tone before dial? FXO (FXO only):**
    When enable, FXO line will wait for dial tone before dail the number.

P. **FXO Line Hunting Method，FXO** */VoIP Call In/* **(FXO only)**
    Define how the FXO interface to ring the phone line when VoIP call in:
    - Free Random: Any unused available line.
    - Line number Priority: The 1st line has high priority; it will always ring the 1st line if it is available. When 1st line is busy, it will try to ring 2nd line if it is free.
    - Rotation: 1st line ring first, then 2nd line ring next time, when the latest line ring this time, it will come back to ring 1st line next time.

Q. **Ring No Rehook**:
    When enable, FXO interface will not terminal the incoming call from PSTN by OFF-HOOK then ON-HOOK.

2.3.6 Line Voltage & Current

Detect FXS / FXO interface line voltage / current values.

Port	Voltage		Current (uA)	
	FXS	FXO	FXS	FXO
1	47	66	507	0
2	47	56	191	0
3	6	6	25059	20900
4	47	-63	557	0
5	48	65	22	0
6	47	-63	23	0
7	47	0	353	0
8	47	0	40	0

Reload

2.3.6.1/Line Configure/ Line Voltage Current

On-hook FXS voltage is about 47 ~ 50v normally, Off-hook voltage is about 6 ~ 12v

FXO Voltage is the value on the FXO interface, which is 0 if no PSTN line connected, and about 23~65 V during On-hook, about 5~13 V during On-hook.

FXS / FXO Off-hook current is greater than 20000uA typically.

H3000-2 / H1111 devices only measure the line voltage , no detect current.

Port	FXS	FXO
1		67
2		-66
3		-67
4		54
5		-50
6		-53
7		0
8		0

Reload

2.3.6.2/Line Configure/ Line Voltage Current

2.3.7   Line Diagnostics

H3000 / H1100 Series device supports line diagnostics as GR-909 specification.

Line Diagnostics

Port	Type	Enable	Hazardous Potential	Foreign Voltage	Resistive Faults	Offhook	REN	Time
1	FXS	☐						
2	FXS	☐						
3	FXS	☑	Pass	Pass	Pass	Pass	Low	2011-08-03 13:53:05
4	FXS	☐						
5	FXS	☐						
6	FXS	☐						
7	FXS	☐						
8	FXS	☐						

☐ Including Channel In Used
* Checking the checkbox will result in disconnection.

☐ (All)   Test   Detail   Clear

2.3.7.1/Line Configure/ Line Diagnostics

Provide a 5 test items: Hazardous Potential, Foreign Voltage, Resistive Faults, Offhook, REN
It will drop the call during the line testing.
Click [Detail] to view the detail test result.

Fxs Test Detail

Port	Tip AC	Ring AC	Tip Ring AC	Tip DC	Ring DC	Tip Ring DC	Tip Ring Resistance	REN
1								
2								
3	0.2	0.0	0.0	0.059	-1.474	-1.533	100000000	0.012
4								
5								
6								
7								
8								

Back

2.3.7.2 /Line Configure/ Line Diagnostics/Detail

2.3.8 Line Impedance

Configure the line impedance of FXS/FXO interface.

2.3.6 /Line Configure/ Line Impedance

2.3.9 PSTN Gain

Adjust the Line Volume when the phone standby on PCM function is enabled.

2.3.9/Line Configure/ PSTN Gain

■ TxGain: Transmitter gain, adjust the playback Local phone (handset) volume adjustment, the adjusted value is from +12 to-15dB, increase the dB value will increase the playback volume of the FXS telephone.

■ RxGain: Receiver gain, adjustable microphone volume. the adjusted value is from +12 to-15dB, increase the dB value will amplify the microphone volume of the FXS telephone.

2.3.10 Message Indicator

Define the method to notify the phone on FXS interface when receive a Voice mail message notify from VoIP server.

Port	Type	On		Off	
1	LAMP	50	ms	1000	ms
2	DTMF	B51C		B50C	
3	FSK	4120888		4120880	
4	Disable				
5	Disable				
6	Disable				
7	Disable				
8	Disable				

Message Indicator

Resend Data Time: 3 second.

Modify Reset

2.3.19 /Line Configure/ Message Indicator

a. Resend Data Time：Time interval between resend indicator message.

 .

b. voicemail notification message mode:
   - Disable :do not send the notification message.
   - LAMP: increase FXS interface voltage lighting message notification lamp on phone.
   - DTMF: send specific DTMF code to on/off the message notification on the phone, for the number specific to this function, please contact the phone manufacturer.
   - FSK: send specific FSK singnal to on/off the message notification on the phone, for the number specific to this function, please contact the phone manufacturer.

2.4 Routing Setup

2.4.1 VoIP Call Out

VoIP Call Out								
MaxDigits 20 FirstDigitTime(Sec): 30 OtherDigitTime(Sec): 3 Timeout for Re-entry route: Disable ▼ second.								

Index	Remark	Area Code	Min Digits	Max Digits	Destination	Strip	Prefix	Profile	Delete
1	IVR	1*123	5	5	ipivr	5			Delete
2	mobile	09	10	10	rs				Delete
3	pstn	0	9	10	pstn				Delete
4	DND	*74	3	3	rs				Delete
5	CLIR	#31			rs	1			Delete
6	transfer	#9	3	8	pstn		%23		Delete
7	backup	v			pstn		%s#		Delete

2.4.1 /Routing Setup/VoIP Call Out

This page let you define the routing rule for Call out to VoIP. (User press the phone number through phone set dial pad, then VoIP Router translate the phone number by the routing table setting here to destination IP & dial out number then Call out via network protocol).Here can define some special keyword like IPIVR, PSTN as destination for some special function also.

Each time when you off hook the phone connected to this VoIP Router, you will hear a dial tone or prompt voice to remind you to press the phone number, after you input the number you called, if digits of the number of you called is not exceed the Max Digits, please remember to press the # key for ending the input, if you do not press # key for enter, gateway will automatically call out the number after timeout of define on **OtherDigitTime**.

A. Time & Digits wait for dial out

The VoIP Router wait user input the number digits & time parameters as below:
Time & Digits wait for user Press.

a. **MaxDigits**: Define the maximum digits wait for user press for all VoIP Call Out, if user press digits match the number defined here. It will go to translate for call out rule without needed to press # key.

b. **FirstDigitTime**: Define the waiting time (seconds) for user press phone number first digit. User need to press first digits before the setting time (seconds) defined here, if VoIP Router wait for the defined seconds and there is no any digits press, the VoIP Router will stop to wait and feedback the user busy tone.

c. **OtherDigitTime**: Define the waiting time (seconds) for user press phone number secondary & the rest digits. User need to press the rest digits before the seconds defined here, if VoIP Router wait for the defined seconds and there is no any digits press, it will go to translate for call out rule without needed to press # key.

d. **Timeout for Re-entry route**: When one of the rules on the VoIP call out rules is matched and be execute, the device will wait the time(seconds) defined here for successful connection, but if time out defined there still failure connection, it will trying to reroute by another call rule setting by the "v"+ the number prefix.
For example as below, when the user try to call the destination number 12345678, it will try to call the gateway location at 168.11.22.33, but if wait 10 seconds and still can not successful connection, the gateway will abort the call and try call out by the PSTN line.

Timeout for Re-entry route: 10 second.

Index	Remark	Area Code	Min Digits	Max Digits	Destination	Strip	Prefix	Profile	Delete
1	Normal rule	8	8	8	168.11.22.33				Delete
2	Backup rule	v8			pstn				Delete

<The example that use "v" prefixes for reroute the call out>

When user enable the hot line function on */VoIP Setup/Line Configure/Line Setting/* menu, it will over ride the above parameters and direct call out by hot line call out rule.

___

B.VoIP call out Routing Table

a. **Remark**: Remark for this routing rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

b. **Area Code**: Define the Prefix number fit this rule, any phone number prefix digits matched with the rule will call out by this rule define. Please Notify there is a compare order rule on this routing table. That mean the VoIP Router will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user press will go to call out directly no regard to the rest rules below. For Example, if a rule item for area code 8862 is on Index 5, another rule item for area code 886 on Index 6 below that will be ignored.
By setting the hln (hl1 for hot line one, hl2 for hot line two) on the area code field and enable hot line function (*/VoIP Setup/Line Configure/Line Setting/*), the VoIP Router can service the hot line direct call.

i. **Min Digits**: define the minimum digits wait for user press for number fit this rule, if user press digits less the number defined here. It will keep waiting for input until exceed the **FirstDigitTime** defined time. If user press digits more then **Min Digits** here, the VoIP Router will wait time defined on **OtherDigitTime** then go to translate for call out rule without needed to press # key.

ii. **Max Digits**: define the maximum digits wait for user press for number fit this rule, if user press digits match the number defined here. It will go to translate for call out rule without needed to press # key.

iii. **Destination:** Define the destination IP for call out number fit this rule, user can input below format:

- IP address, for example: 168.56.9.22
  1. for sip → please add sip: before ip address, for example sip:168.56.9.22
  2. for h323 → please add h323: before ip address , for example h323:168.56.9.22

- URL, route via URL. For example: sip.fwd.com .This VoIP Router can setup to register to DDNS service (/*System Setup/Advanced/Dynamic DNS/*) to let user call out to another VoIP Router with dynamic IP by URL.

- **gkn** : route via gatekeeper, it will get the destination IP by gatekeeper setting (/*VoIP Setup/Gatekeeper/*) in advance. For example: gk1 for gatekeeper 1. gk2 for gatekeeper 2. gk for all the gatekeepers available (search sequence: gk1 > gk2 > gk3 > gk4). Gk3_2_1 will try gk3 first, then gk2, then gk1.
  All the setting above can be added by port number, for examples: 168.56.9.22:8495 will call to 8495 port.

- **sr**n, **rs**n: same as **gkn**   , basically, it is used for SIP register server.

- **PSTN:** route this call via PSTN line interface. This is usually used for for the backup route for the rule setting on */Routing setup /VoIP Call out/* with "v" prefix.

- **ipivr**: Enter the Network parameter voice interactive setting mode. User can use this function to enter all the WAN network parameters without PC. (Please refer the application note " *IP IVR produce* " for more detail procedure).

■ **ldcfg**: Restore all parameters to the default values. User can assign a password to use this function to restore all the parameters to the default values.

■ **rect**: Enter to voice record procedure . User can assign a function code for enter the voice record procedure, when press this code to enter the voice record procedure, the device will record 30 seconds voice file and keep on sound wave file (G.711, uLaw), User can download the recorded wave file on */VoIP Setup/Advance setting/Prompt Voice/* and used this file to upload for customization voice file or used for busy tone analysis.

■ **agent**: agent code setting. When a VoIP call in made by this device, it will ring the assigned phone set. If the user want to use the different phone set (connected to same device, but did not ring) to answer the call, just off hook and enter this agent code to redirect the call to this phone you used for talk.

■ **lo**: assign the route to local loop back. The destination IP of this call will be the local host, i.e.:127.0.0.1

f. **Strip**: the number of digits will be ignored by user input. For example, if user press the number is 886212345678 and the STRIP field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 12345678.

g. **Prefix**: The numbers will be added on the prefix of the user press number. For examples, if user press the number is 12345678 and the PREFIX field is setting to 0028862, the actually call out number will be 002886212345678. Another example, if user press the number is 90, STRIP field is setting to 2, and the PREFIX field is setting to 0,12345678, the actually call out number will be 0,12345678 ("**,**" mean delay 1 second). This example is especially useful for speed dial function.

h. **Profile**: Define the optional special call out parameters on this destination. Please input the name you defined on the profile (*/VoIP Setup/Routing Setup/Routing Profile/*) list.

i. **Delete**: Delete this rule item on routing table.

To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE and IP address then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press */Syetem Maintenance/Save Modification/* .

C.Setting Examples

Here is some VoIP call out routing table setting examples below:

a.   Define wait time and digits for destination phone number

MaxDigits: `10` FirstDigitTime(Sec): `30` OtherDigitTime(Sec): `5`

In this case, when user picks up the phone, the VoIP router will generate 30 seconds (defined on FirstDigitTime) dial tone for user press DTMF for destination phone number, After user press first digit DTMF from phone set (for example, 0, the VoIP router will wait 5 seconds (defined on OtherDigitTime) to press the rest phone number digits, if user did not press any key within first 30 seconds, the VoIP Router will generate the busy tone to terminate the call. After user press first digit and did not key any key within 5 seconds, for example, like 601 it will call out 601 after 5 seconds, but if user press 601#, it will direct call out 601 immediately without waiting rest key.

In this case, the Max Digits is setting to 10, so if user dial 0212345678, 10 digits phone number, it will call out immediately without wait 5 seconds or # key, that mean it will not accept phone number more than 10 digits like 02123456781, if user press that phone number, it still call out the number to 0212345678 because maximum digits for phone number is 10.

b.   VoIP call out by IP:

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	NY_office	6			172.16.7.1				Delete

In this case, we assume that we have another VoIP router locate at New York office and the IP is 172.16.7.1 , when we press any phone number prefix is 6 will call to that VoIP router, for example, if we dial 601, the VoIP will Call out 601 to another VoIP router locate at IP 172.16.7.1, you can check the real call out IP and phone number at the VoIP Setup/ Port Status:

Port Message

Port	Type	Display name	Status	Connected IP	Caller ID	Start Time	End Time	Talking Sec	Dialed number	Release by
1	FXS		Idle	172.16.7.1		2004/02/19 13:55:10	2004/02/19 13:55:43	28	601	(146)onHangup
2	FXS		Idle							

c.   Call by Domain name:

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
2	Jack@SH	8621	4	4	Voiprouter.dyndns.org	4	013902440272		Delete
3	China	86	2	5	www.voipgateway.com				

In this case, by route rule 2, we set up a short cut number 8021 for dial out number 013902440272 to another VoIP router, user just press 8021 will cause cut 4 digits (8621) define on Strip, and add the number defined on Strip (013902440272), then call to that gateway(voiprouter.dyndns.org) and number(013902440272).

In this case, by route rule 3, we assume we have another VoIP Router locate at china.ezvon.com URL, and we use prefix 86 to call out for this gateway, the minimum digits for phone is 2 digits and the maximum phone number digits is 5, any phone number contain over 5 digits will be truncated to 5 digits like 862013 will be truncated to 86201 for call out.

- **Caution:**

    There is order rule on this routing table; the VoIP router will check the route table items by index order one by one. That mean, in above case, if user put the area code item 86(index 1) above 8621(index 2), then the route item 8621 will never been used.

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	Take_All	86			10.1.1.1				
2	Never_Used	8621			20.1.1.1				

never

d.   Strip and Prefix

User is easy to combine using Strip and Prefix define to modify the phone number from phone to real call out phone number, for example, if the VoIP router is installed on Taipei and use another Gatekeeper to service global service. When user just dial 10 digits Taipei phone number like 02-12345678(do not need to press # key because Max Digits setting is 10), and the VoIP router will stripe the 02 (2 digits defined on Strip) ,add the country code 8862 (defined on Prefix) then send 8862-12345678 out for VoIP call, see below example:

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	Taipei	02		10	gk	2	8862		Delete

By above setting, When you dial 0212345678, you can check the real call out IP and phone number will change to 886212345678 at the VoIP Setup/ Port Status:

Port Message

Port	Type	Display name	Status	Connected IP	Caller ID	Start Time	End Time	Talking Sec	Dialed number	Release by
1	FXS		Idle	172.16.7.1		2004/02/19 13:55:10	2004/02/19 13:55:43	28	886285123390	(146)onHangup
2	FXS		Idle							

e.   Call via Server /SIP Register Server

This VoIP router can register up to 4 servers, for example:

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	Via_RS2	1			rs2				Delete
2	RS1_3	2			rs1_3				Delete
3	RS_ALL	3			rs				Delete

By Index 1, if user input any phone number with prefix code is 1, The VoIP Router will call out via Gatekeeper 2.

By Index 2, if user input any phone number with prefix code is 2, The VoIP Router will try to call out by Gatekeeper 2 (if register to Gatekeeper 2 is successful), if Gatekeeper 2 is not available, it will check Gatekeeper 3, then check Gatekeeper 1.That mean if register to gk2 is

failure and register to gk3 & gk1 is successful, the VoIP router will call out via gk3.

You can check the Gatekeeper register status on */VoIP Setup/Register Server/Register Status.*

f.   Call to different IP port

The default IP port used by VoIP router is 1720 for H.323 and 5060 for SIP, if work with remote side of VoIP Router or gateway is change another port number for VoIP, please assign another port number after destination IP or URL. Please make sure both side use same port number for VoIP call, otherwise it can not make call. You can change the VoIP router default listen port on .

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	Port_1719	1			sip:10.1.1.1:1719				Delete
2	Port_8495	2			www.voipgateway.com:8495				Delete

g.   Profile:

Define the optional special VoIP parameters when calling to the destination. Please input the name you defined on the profile (*/VoIP Setup/Routing Setup/Routing Profile/*) list.

Example: if user set the VoIP Call Out & Routing Profile like below:

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1	UsePF1	1			rs1			pf1	Delete
2	UsePF2	2			10.1.1.2			pf2	Delete
3	UseDefaultPF	3			www.voipgateway.com				Delete

Index	Name	VAD	CODEC	H.245 Tunneling	DTMF Relay	T.38 FAX Relay	Package Frame	Q.931 Fast Start	
	ID1	AS	ID2	AS	ID3	AS	ID4	AS	Delete
1	pf1	ON	G.723.1	ON	Out band	OFF	3	ON	
	00001	H.323	1001	E.164					Delete
2	pf2	ON	G.723.1	ON	In band	OFF	3	OFF	
	00002	H.323	1002	E.164					Delete

When VoIP call out number with prefix 1 will use the Profile named PF1 (H.323 ID1 = 0001, E.164 ID=1001, DTMF Relay=Out band, Q.931 Fast Start=ON) to Call out VoIP.

When VoIP call out number with prefix 2 will use the Profile named PF2 (H.323 ID1 = 0002,

E.164 ID=1002, DTMF Relay=In band, Q.931 Fast Start=OFF) to Call out VoIP.

When VoIP call out number with prefix 3, because there is no Profile assigned, it will use the default value for VoIP out.

h. Delete：**Delete this rule item on routing table.**

To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press */Syetem Maintenance/Save Modification/*.

2.4.2 VoIP Call In

2.4.2 /Routing Setup/VoIP Call In/

This page let you define the routing rule for Call in from VoIP. (VoIP Router got a VoIP call required form Network, and then translates the phone number passed from remote side VoIP Router to the real dial out number & line base on this VoIP Call In routing table). Each time when the VoIP Router received a VoIP call from Network, it will check with Area Code to see which rule matched to service, if no rule matched, it will refuse to call out and will bound back the call.

When the VoIP Router received a VoIP called from network, it will check below rules fields then decide line and number to dial out.

a.   **Area Code**: Define the Prefix number this rule service, any VoIP called from network dialed number prefix digits matched with the rule will call out to phone by this rule define. Please Notify there is a compare order rule on this routing table. That mean the VoIP Router will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user press will go to call out directly no regard to the rest rules below.   For Example, if a rule item for area code 8862 is on Index 1, another rule below that like index 2 for area code 886 will be ignored.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	886	☐								☐					Delete
2	8862	☐								☐					Delete

b.   **Auth**：Authorization IP check enable. Enable IP range authorization function. When Enable, the gateway will check the remote caller IP range setting on */VoIP Setup/Routing Setup/Authorization/*, if it is within the authorization, the gateway will allow the call out, but if the remote caller's IP is not in the range, it will refuse to call out and terminate this call.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete

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Index	Area Code	Auth	Strip	Prefix	Max	Min	From	To	LineNo	Display name	CallWaiting	Alert	Profile	Forward	Delete
1	8862	☑	4							□					Delete

c. **Strip**: Number of digits will be ignored by user input. For example, if received VoIP call number is 886212345678 and the STRIP field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 12345678.

Index	Area Code	Auth	Strip	Prefix	Max	Min	From	To	LineNo	Display name	CallWaiting	Alert	Profile	Forward	Delete
1	8862	□	4							□					Delete

Ex: VoIP Call in number is 886212345678 and real dial out number is 12345678 by strip 4 digits.

d. **Prefix**: The numbers will be added on the prefix of received VoIP call number. For examples, if received VoIP call number is 12345678 and the **PREFIX** field is setting to 0028862, the actually call out number will be 002886212345678.

Index	Area Code	Auth	Strip	Prefix	Max	Min	From	To	LineNo	Display name	CallWaiting	Alert	Profile	Forward	Delete
1		□		0028862						□					Delete

Ex: VoIP Call in number is 12345678 and real dial out number is 0028862-12345678 by add 0028862 prefix.

Another example, if user VoIP router received a call number 90, **STRIP** field is setting to 2, and the **PREFIX** field is setting to 0,12345678, the actually call out number will be 0,12345678 (, mean wait 1 second for PBX get line for dial out to PSTN, the wait time for one , can be set on */VoIP Setup/Line Configure/Line Feature/*). This example is especially for speed dial function.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	90	□	2	0,85123390						□					Delete

Ex: VoIP Call in number is 90 and real dial out number is 0,12345678 by stripe 2 digits and add 0,12345678 prefix, so the real dial out number is 0,12345678.

e. **Maximum**: Define the maximum digits of call number allow to dial. If the length of dial number after pervious **STRIP & PREFIX** process is more than the setting, it will deny dialing out. For example, you can set the Maximum dial out digits is 8, for call to local area phone only, any VoIP call in attempt to dial 0712345678 out of 8 digits for call out long distance will been deny to call

out.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1		□			8					□					Delete

f. **Minimum**: Define the minimum digits of call number allow to dial. If the length of dial number after pervious STRIP & PREFIX process is less than the setting, it will deny dialing out. For example, if set **Minimum** to 4, any VoIP call in attempt to dial number less than 4 digits like 110, 911 will been deny to call out.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1		□			4					□					Delete

Ex: VoIP Call in number is 911 and Minimum setting to 4, the VoIP router will deny to call out.

g. **From**: Define the beginning line number for service this area code VoIP call. For example, if user assigned FROM 1 TO 1 for AREA CODE 601 in this routing table, then any VoIP call for call in number 601 will ring the line 1 only.

h. **To**: Define the ending line number for service this area code VoIP call.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	601	□					1	1		□					Delete
2	602	□					2	2		□					Delete
3		□								□					Delete

Ex. Any VoIP Call in number with prefix 601 will ring the line 1, and Any VoIP Call in number with prefix Call in number 602 will ring the line 2. any other numbers will ring any available (not busy) lines.

i. **Line No**: Click to enable if you want to force compare with the line number setting on <u>LINE CONFIGURE</u> menu (*/VoIP Setup/Line Configure/Line Setting/*). If the dial number after pervious STRIP & PREFIX process is matched with the line number setting, the VoIP call will ring the dedicate phone line that assigned with matched number.

j. **DisplayName**：Enable Display Name function ,the device can display the simple code.
For example :
From: 12736 <sip:+886987282721@ims.fchtet.net>

After enabled Display Name, you can display 12736。

k. **Call Waiting**：Enable or Disable the call waiting function。
   ■ **Enable:** During Talk, you can answer another phone call and hold the original conversation. When some one call in when you are busy on another phone call , you will hear a du-du call waiting signal, please use flash key on your phone to hold the original call and answer the incoming call, press again flask key will switch back the original call party.
   ■ **Disable:** Disalbe the Call waiting function. The gateway will reply the busy to remote side when the line is on used.

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	1	□								□	Enable				Delete
2	2	□								□					Delete
3	3	□								□					Delete

l. **Alert**: Control the Ring Back tone generate timing:
   ■ Mode 0: When this VoIP Router get ring back tone from phone line, it will send the ring Alert signal to remote VoIP Router for generate ring back tone.
      ■ Mode 1: Before this VoIP Router dial to phone line, it will send the ring Alert signal to remote VoIP Router for generate ring back tone.
      ■ Mode 2: After this VoIP Router finish dial out number to phone line, it will send Connect OK signal to remote VoIP Router.
      ■ Mode 3: Before this VoIP Router dial to phone line, it will send the ring Alert signal to remote VoIP Router for generate ring back tone, after this VoIP Router finish dial out number to phone line, it will send Connect OK signal to remote VoIP Router.

m. **Profile:** Define the optional special VoIP parameters when received on this destination. Please input the name you defined on the profile list (*/VoIP Setup/Routing Setup/Routing Profile/)*.

Example: if user set the VoIP Call in & Routing Profile like below:

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	1	□								□	Enable		pf1		Delete
2	2	□								□			pf2		Delete
3	3	□								□					Delete

Index	Name	VAD	CODEC	H.245 Tunneling	DTMF Relay	T.38 FAX Relay	Package Frame	Q.931 Fast Start	

	ID1	AS	ID2	AS	ID3	AS	ID4	AS	Delete
1	pf1	ON	G.723.1	ON	Out band	OFF	3	ON	
	00001	H.323	1001	E.164					Delete
2	pf2	ON	G.723.1	ON	In band	OFF	3	OFF	
	00002	H.323	1002	E.164					Delete

When VoIP call in number with prefix 1 will use the Profile named PF1 (H.323 ID1 = 0001, E.164 ID=1001, DTMF Relay=Out band, Q.931 Fast Start=ON) to answer the VoIP Call in.

When VoIP call in number with prefix 2 will use the Profile named PF2 (H.323 ID1 = 0002, E.164 ID=1002, DTMF Relay=In band, Q.931 Fast Start=OFF) to answer the VoIP Call in.

When VoIP call in number with prefix 3, because there is no Profile assigned, it will use the default value for VoIP Call Out.

n.   **Forward**: Define the profile name for forward the unanswerable VoIP call on this Call In rule. Please input the name you defined on the */Voip Setup/Routing Setup/ Forwarding/* .

Example: if user set the VoIP Call in & Forward Profile like below:

Index	Area Code	Auth	Strip	Prefix	Maximum	Minimum	From	To	LineNo	DisplayName	CallWaiting	Alert	Profile	Forward	Delete
1	601	□					1	1		□	Enable		PF1	CF1	Delete
2	602	□					2	2		□			PF2	CF2	Delete
3		□								□					Delete

Forwarding

Other: 10.1.1.1/104

No.	Name	Always	OnBusy	No Answer	No Answer Sec	Delete
1	CF1	hk.big.com/301				Delete
2	CF2		assist.big.com	assist.big. om/610	30	Delete

In this case, when the VoIP router received a VoIP call in number with prefix 501 (not 601 or 602 prefix defined on Call In Routing table), it will forward this call to the IP & number defined on **Other** filed(in this case, 10.1.1.1/104, it mean it will forward this call to IP 10.1.1.1 and calling number will change to 104).

When the line 1 user is going to have a tour to another location with same VoIP router equipment, user setup the called prefix number 601 forward to profile name CF1, and in CF1 profile, the **Always** field is set to hk.big.com/301, that mean any call number with prefix 601 will always be forward to another VoIP Router locate at hk.big.com and dial out number is 301.

When the line 2 is busy and another VoIP with prefix 602 Call in, it will forward it to the assist.big.com with same number (defined on **OnBusy**).

When VoIP Call In number with prefix 602, it will ring the line 2 for 30 seconds (**defined on No Answer Sec.**), if no one answer line 2 within 30 seconds, it will forward the call to another VoIP Router located at assist.big.com and dial out number is 610 (**defined on No Answer**)

o. **Delete**: Delete this rule item on routing table.
   To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press Save Modification (*/Syetem Maintenance/Save Modification/*).

2.4.3　VoIP Call In IVR

2.4.3 /Routing Setup/VoIP Call In IVR/

When Enable the [Prompt Voice for VoIP Call In function on */VoIP Setup/Advance Setup/Prompt Voice/* , all the remote party of VoIP caller will hear the customization upload voice file and need press the destination number. All the input number will be checked the number length and be strip/add prefix defined on this page. When a matched area code be processed, it will use this number to check on the */VoIP Setup/Routing Setup/VoIP Call In/* to decide the final route path. If no match area code rule defined on the table, the gateway will response busy tone and connect failure. User can use this function as the password authorization on the outbound gateway. For example, if we upload the voice file content on **Prompt voice for VoIP call in** of */VoIP Setup/Advance Setup/Prompt Voice/* is "Please input the password and destination number" and we set a compare rule as below:

Index	Remark	Area Code	Min Digits	Max Digits	Strip	Prefix	Delete
1	Password	8495	7	12	4	02	Delete

When a remote VoIP call in and want this gateway to outbound call, the remote side user will hear voice prompt like "Please input the password and destination number", because there is only a compare authorization rule, all the none 8495 prefix phone number will not accept to dial out and will be disconnect. (that mean we use 8495 as the outbound call authorization password), and the digits of user input phone number should between 7 to 12 (include 4 digits come from 8495),the number user input will strip the first 4 digits (8495) and add 02 prefix. Then this number will be checked by */VoIP Setup/Routing Setup/VoIP Call In/* . For Example, if user input the number is849512345678, it will strip 4 digits and add 02 prefix code, the use 0212345678 to find a call out rule.

A.　Time & Digits wait for user

The VoIP Router wait user input the number digits & time parameters as below:

Time & Digits wait for user Press.

a. **MaxDigits**: Define the maximum digits wait for user press for all VoIP Call Out, if user press digits match the number defined here. It will go to translate for call out rule without needed to press # key.

b. **FirstDigitTime**: Define the waiting time (seconds) for user press phone number first digit. User need to press first digits before the setting time (seconds) defined here, if VoIP Router wait for the defined seconds and there is no any digits press, the VoIP Router will stop to wait and feedback the user busy tone.

c. **OtherDigitTime**: Define the waiting time (seconds) for user press phone number secondary & the rest digits. User need to press the rest digits before the seconds defined here, if VoIP Router wait for the defined seconds and there is no any digits press, it will go to translate for call out rule without needed to press # key.

B.VoIP Call In IVR Routing Table

a. **Remark**: Remark for this routing rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

b. **Area Code**: Define the Prefix number fit this rule, any phone number prefix digits matched with the rule will call out by this rule define. Please Notify there is a compare order rule on this routing table. That mean the VoIP Router will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user press will go to call out directly no regard to the rest rules below. For Example, if a number 84951xxxxxx is fit the rule item 1&2 , it will be processed by rule 1 and never be processed by rule 2 , that mean that rule 2 is never been used.

Index	Remark	Area Code	Min Digits	Max Digits	Strip	Prefix	Delete
1	password	8495	7	12	4	02	Delete
2	No used rule	84951					

c. **Min Digits**: define the minimum digits wait for user press for number fit this rule, if user press digits less the number defined here. It will keep waiting for input until exceed the **FirstDigitTime** defined time. If user press digits more then **Min Digits** here, the VoIP Router will wait time defined on **OtherDigitTime** then go to translate for call out rule without needed to press # key.

d. **Max Digits**: define the maximum digits wait for user press for number fit this rule, if user press digits match the number defined here. It will go to translate for call out rule without needed to press # key.

e. **Strip**: the number of digits will be ignored by user input. For example, if user press the number is 886212345678 and the STRIP field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 12345678.

f. **Prefix**: The numbers will be added on the prefix of the user press number. For examples, if user press the number is 12345678 and the PREFIX field is setting to 0028862, the actually call out number will be 002886212345678.

g. **Delete**: Delete this rule item on routing table.

To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE and IP address then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press */Syetem Maintenance/Save Modification/* .

2.4.4 Routing Profile

				Routing Profile					
Index	Name	VAD	CODEC	H.245 Tunneling	DTMF Relay	T.38 FAX Relay	Package Frame	Q.931 Fast Start	
	ID1	AS	ID2	AS	ID3	AS	ID4	AS	Delete
1	711	OFF ▼	ALAW ▼	ON ▼	In band ▼	OFF ▼	3 ▼	ON ▼	
		E.164 ▼		E.164 ▼		E.164 ▼		E.164 ▼	Delete

Modify   Reset

Insert new
Insert To: 2   Name:

Add   Reset

2.4.4/Routing Setup/Routing Profile/

This page defines the optional special VoIP parameters when making/received a VoIP call. For define some special parameters for different VoIP equipment or authorize purpose, please add a profile at */VoIP Setup/Routing Setup/ Routing Profile/* and use the same name as the profile on the Call in Routing Table (*/VoIP Setup/Routing Setup/VoIP Call In/*) or Call out Routing table (*/VoIP Setup/Routing Setup/VoIP Call Out/*).

a. **Name**: Specify a profile name. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

b. **VAD**:
    - *ON*: turn on the VAD(Voice Active Detection) function.
    - *OFF*: turn off the VAD function, please select ON for save the bandwidth.

c. **CODEC**: Select different voice CODEC for VoIP communication. The bit rate of G.723.1 is 5.3k/6.3k, G.729 is 8k, uLaw and aLaw is 64k per second. The G.723.1 is default CODEC.

d. **H.245 tunneling**:
    - *ON* for enable H.245 tunneling.
    - *OFF* for disable H.245 tunneling.

e. **DTMF Relay**:
    - When select *In band* to transfer the DTMF during VoIP, the user pressed DTMF tone will be treat as general voice and been compressed then transmit to remote side to decompress play back,

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it maybe cause some problem on duplicate or missing DTMF receive.

■ When select *Out band* to transfer the DTMF during VoIP, the user pressed DTMF tone will be decode by local VoIP Router then transmit as signal, after received on received remote VoIP Router, it will be regenerate by remote VoIP Router. The default value is Out band.

f. **T.38 FAX Relay**:

■ ON: FAX will be transmitted by using T.38 FAX over IP protocol.

■ OFF: FAX over IP is disabled.

g. **Package Frame**: Select the number of voice payload frames on each UDP package VoIP transmit. More frames into one package mean save more bandwidth. The default frames on each package is 3.

h. **Q.931 Fast Start**:

■ ON: Enable Fast Start capability during Q.931 handshaking.

■ OFF: Disable Fast Start capability during Q.931 handshaking.

i. **ID1**: User defines ID #1 during this VoIP call.

j. **As**:

■ E.164: Parameter on ID1 field is the E.164 during this VoIP call.

■ H.323 ID: Parameter on ID1 field is the H.323 ID during this VoIP call.

■ Calling: Parameter on ID1 field is DID number during this VoIP call. If this optional is setting, it will override the LINE NUMBER on line setting menu.

■ Password: Parameter on ID1 field is the password for VoIP call. Parameter defined here will used as MD5 during H.235 and will not display on the Web UI

k. **ID2,ID3,ID4**: there are 4 fields for user define the ID parameters, please reference the ID1 setting above.

l. **Delete**: Delete this rule item on routing table.

To add new profile item on routing table, please assign the number you want to insert before, input profile NAME then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press Save Modification (*/Syetem Maintenance/Save Modification/*).

Here is VoIP Routing Profile setting examples below:

Index	Name	VAD	CODEC	H.245 Tunneling	DTMF Relay	T.38 FAX Relay	Package Frame	Q.931 Fast Start	
	ID1	AS	ID2	AS	ID3	AS	ID4	AS	Delete
1	pf1	ON	G.723.1	ON	Out band	OFF	3	ON	
	00001	H.323	1001	E.164					Delete
2	pf2	ON	G.723.1	ON	In band	OFF	2	OFF	
	00002	H.323	1002	E.164	****	Password			Delete

When using profile PF1 , the parameters will be used for H.323 ID1 = 0001, E.164 ID=1001, DTMF Relay=Out band, Q.931 Fast Start=ON) to call / answer the VoIP.

When using profile PF2 , the parameters will be used for H.323 ID1 = 0002, E.164 ID=1002, DTMF Relay=In band, Q.931 Fast Start=OFF, Password=1234 but be hidden) to call / answer the VoIP.

2.4.5 Forwarding

2.4.5 /Routing Setup/Forwarding/

This page defines the forwarding behavior include:

- get an unmatched prefix number for VoIP call in,
- Line busy
- No answer

Please add a profile at */VoIP Setup/Routing Setup/Routing Profile/* and put the name of profile on the Call out Routing table (*/VoIP Setup/Routing Setup/VoIP Call Out/*).

a. **Other**: Define the forward IP and forward phone number when there is no match rule setting on *VoIP Call Out Routing* table. The format is *IP/phone number* or *URL/phone numbe*r. i.e. all the phone number can not match a prefix rule will be forward to the IP& phone number define on here.

b. **Name**: Specify a profile name. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

c. **Always:** Always redirect forward to this IP(or URL)/phone number, All incoming call will be forward to IP assigned here.

d. **On Busy**: Redirect forward to this IP(or URL)/phone number when busy, an incoming VoIP call will forward to IP assigned here when this line is busy.

e. **No Answer**: Redirect forward to this IP(or URL)/phone number when no answer over the time **No Answer Sec** , an incoming VoIP call will forward to IP assigned here when ring time over the defined on **No Answer Sec.**

f. **No Answer Sec**. Defined the wait seconds for redirect forward to another IP(or URL).

g. **Delete**: Delete this rule item on routing table.

h. To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press Save Modification (*/System Maintenance/Save Modification/*).

Example: if user set the VoIP Call in & Routing Profile like below:

Index	Area Code	Strip	Prefix	Maximum	Minimum	From	To	LineNo	Gatekeeper	Alert	Profile	Forward	Delete
1	601					1	1					cf1	Delete
2	602					2	2					cf2	Delete

Other: 10.1.1.1/104

No.	Name	Always	OnBusy	No Answer	No Answer Sec	Delete
1	cf1	hk.big com/301				Delete
2	cf2		assist.big.com	assist.big.com/610	30	Delete

In this case, when the VoIP router received a VoIP call in number with prefix 501 (not 601 or 602 prefix defined on Call In Routing table), it will forward this call to the IP& number defined on **Other** filed(in this case, 10.1.1.1/104, it mean it will forward this call to IP 10.1.1.1 and calling number will change to 104).

When the line 1 user is going to have a tour to another location with same VoIP router equipment, user setup the called prefix number 601 forward to profile name CF1, and in CF1 profile, the **Always** field is set to hk.big.com/301, that mean any call number with prefix 601 will always be forward to another VoIP Router locate at hk.big.com and dial out number is 301.

When the line 2 is busy and another VoIP with prefix 602 Call in, it will forward it to the assist.big.com with same number (defined on **OnBusy**).

When VoIP Call In number with prefix 602, it will ring the line 2 for 30 seconds (**defined on No Answer Sec**.), if no one answer line 2 within 30 seconds, it will forward the call to another VoIP Router located at assist.big.com and dial out number is 610 (**defined on No Answer**) .

2.4.6　Authorization

	Authorisation		
Index	From	To	Delete
1	202.153.167.1	202.153.167.255	Delete

2.4.6 /Routing Setup/Authorization/

When this gateway has been used for outbound call, it can enable to check the remote caller gateway's IP to decide accept or refuse the call. If define the IP range here and enable the [Auth] option on the*/VoIP Setup/Routing Setup/VoIP Call In/* , only the IP in range will allow to call out by this gateway.

2.5 Register Server

2.5.1 Register Status

2.5.1 /Register Server/Register Status

You Can check the register status of this gateway on this page.

a. MAC: this gateway' MAC Address。

b. RS1-4: Indicate the status of 4 server register.

- ■ **SIP&H323**: The protocol used for registering the server, this gateway supports both H.323 and SIP protocol.
- ■ Green Indicator: Successful to register server and the register phone number.
- ■ Red Indicator: Failure to register server and the failure reason.
- ■ Yellow Indicator: Disable the register function.

Example: for Status display as above, it indicates:
1. The register to Server #1 function is disabled (SIP).
2. Use SIP protocol to register to register as RS2, the register method is 4 lines independent. Each lines use different number to register: 25618801, 25618802, 5618803, 25618804. Line 2 and Line 4 are disabled to register, Line 1 and Line 3 are successful to register.
3. Use H.323 protocol to register as RS3, all 4 lines share same register 102003
4. Use SIP protocol to register as RS4, each lines use different number as 102002, 102003, 77201111. You can see that line 2 and line 3 register failures. The line 2 failure reason is "unauthorized" and Line 4 failure reason is "not number".

Please setup each register parameters at */VoIP Setup/Register Status/Server#1~4/*。

2.5.2　Register Server—SIP Protocol

Enable SIP Proxy: ☑								
SIP Proxy URL	**Port[1 - 65535]**	**Thought Outbound Proxy**		**Port[1 - 65535]**		**Ping[10 - 600]**		
153.20.44.125	5060			5060		0	Sec	
TTL (Registration interval) [10 - 7200 s]		**Domain**		**Proxy Require**				
120								

Line	Type	Remark	Number	Account	Password	Extn.	Conference ID	Enable
1	FXO		440	440				☑
2	FXO		441	441				☑
3	FXO		442	442				☑
4	FXO		443	443				☑
5	FXO		444	444				☑
6	FXO		445	445				☐
7	FXO		446	446				☑
8	FXO		447	447				☑

Modify

圖 *2.5.2　/Register Server/Server*

If you need use this gateway to register to the H.323 gatekeeper or SIP register/proxy server, you can setup the account for register here. This gateway can register up to four Servers simultaneously.

a. **Protocol**: Select use SIP or H.323 protocol to register to server, by different protocol, the gateway will adjust the page for different parameters for input.

b. **Register** Method:
■ Global: All the lines of the gateway share same account to register.
■ Independent: Each lines can set different/same account independently for register.

c. **Enable SIP Proxy** :
■ ☑ Enable Register SIP Proxy server function.
■ ☐ Disable Register SIP Proxy server function.

d. **SIP Proxy URL**: Please input the IP/URL of the SIP proxy server.

e. **Port [1~65535]**:　Port number used for register to server. The SIP protocol default is 5060, please make sure you have same port number setting on the gateway and server.

f. **Through Outbound Proxy:**

When your gateway is installed behind the firewall or NAT, you maybe need use Proxy server to relay your call. If so, please input the Outbound proxy server's IP here.

g. **Prot[1~65535]**: Port number used for register Outbound Proxy Server.

h. **TTL(Registration interval)[10-7200s]**: Some SIP Server need you set the time interval (seconds) for send the expire signal to register server keep alive.

i. **Domain:** Some SIP Server need you input the Domain for register, please input here.

j. **Proxy Require:** Some SIP Server (Nortel) need you input the more information for proxy function, please input here.

k. **Line:** Number index of lines.

l. **Type:** Interface type of the line::
- FXO: Analog phone interface for connect to PSTN or PBX extension line
- FXS: Analog phone interface for connect to phone set or PBX Co. line.

m. **Remark**: Remark for this routing rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

n. **Number**: Register phone number, Some SIP Server needs this to parameters for register.

o. **Account**： Account for register to SIP server.

p. **Password**: Password for register to SIP server.

q. **Conference ID:** Some SIP Server requires an ID to enable the conference function, please input the ID here to enable that.

r. **Enable:** Enable or disable independently each line for register.

2.5.3 Register Server—H.323 Protocol

1.14 /VoIP Setup/Register Server/Server

When Select use H.323 to register gatekeeper, please input the flow information for register:

a. **Protocol**:
   - SIP
   - H323

b. **Register**:
   - Global: All the lines of the gateway share same account to register.
   - Independent: Each line can set different/same account independently for register.

c. **Enable H323 Gatekeeper** :
   - ☑ Enable Register H.323 Gatekeeper function.
   - ☐ Disable Register H.323 Gatekeeper function.

d. **Gatekeeper URL**: Please input the IP/URL of the Gatekeeper server.

e. **Prot[1~65535]**: Port number used for register to server. The H.323 protocol default is 1719, please make sure you have same port number setting on the gateway and server.

f. **GK ID**: Some Gatekeeper Server need you input an ID for register, please input here

g. **Proxy for NAT**: When your gateway is installed behind the firewall or NAT, you maybe need use Proxy server to relay your call. If your gatekeeper supports this proxy function, you can enable gateway function here to use that.

h. **Line:** Number index of lines.

i. **Type:** Interface type of the line:
- FXO: Analog phone interface for connect to PSTN or PBX extension line.
- FXS: Analog phone interface for connect to phone set or PBX Co. line.

j. **Remark**: 註解： Remark for this routing rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation..

k. **E.164:** phone number used for register to server.

l. **H.323 ID:** Account name used for register to gatekeeper.

m. **Account:** Account name used for register to gatekeeper

n. **Password:** Password used for register to gatekeeper.

o. **Enable:** Enable or disable independently each line for register.

Please remember to press the **Modify** button to take it effect. For store back to flash memory, please press Save Modification (*/Syetem Maintenance/Save Modification/*).

2.6 Advance Setup

2.6.1 NAT Traversal

2.6.1 /Advance Setup/NAT Traversal

If your VoIP gateway is installed behind NAT, you may need a special configuration and server to establish the VoIP communication, this gateway support several method for NAT Traversal as below:

● By Outbound Proxy:
  User can appoint an Outbound Proxy Server to handle the NAT traversal on *VoIP Setup/Register Sever/Server #/*

● Declare NAT IP address:
  Select to enable the input the NAT router IP of the network.

● Use STUN server
  Enable STUN and input the STUN server's IP for handle the NAT traversal, you can input 2 sets of STUN servers IP.

The Gateway will display the system found NAT IP address.

For store back to flash memory, please press Save Modification (*/System Maintenance/Save Modification/*).

2.6.2 Listen Port

2.6.2 /Advance Setup/Listen Port

In this page, user can define the usage port for setup the VoIP communication. Both side of gateways need use the same port for begin VoIP communication.

a. **SIP Listen Port** : Define the listen port for SIP protocol, the default port is 5060, input range from 1024 to 65535.

b. **H.323 Call Signal Port**: Define the Call signal port for H.323 protocol, the default port is 1720, input range from 1024 to 65535.

c. .**H.323 Gatekeeper Listen Port**: Define the Gatekeeper listen port for H.323 protocol, the default port is 1719, input range from 1024 to 65535.

d.   **RTP Initial Port**: Define the RTP package initial port, the input range from 1024 to 65535. the gateway will display the used UDP ports due to multiple lines connection.

After modify and press Modify, system will save and reboot automatically to take it effective.

2.6.3 VoIP Package

2.6.3/Advance Setup/VoIP Package

User can define the parameters relative about VoIP package on this page.

- **Jitter Buffer(ms):**
  Define the Jitter buffer size, input range is from 20 to 200ms.

- **VoIP DTMF Relay Mode:**
  Define the relay mode for DTMF signal:
  - *In band*: When local gateway detects a DTMF signal, it will not decode it . The DTMF signal will been compress/decompress as VoIP voice package.
  - *Out band:* When local gateway detects a DTMF signal, it will decode it, and relay it as a data package separately. The remote gateway will regenerate the DTMF signal after receive the DTMF data package. System default is relay DTMF by out band mode.

- **VoIP DTMF Relay Mode (Out band),:**
  Define 2 methods to relay DTMF when select Out band relay mode:
  - by SIP:RFC2833 (SIP protocol) or H.323:H.245 (H.323 protocol)
  - by SIP INFO   (SIP protocol) or Q.931 (H.323 protocol)
- **RFC2833: Payload number for DTMF[96~127]:**

Define the DTMF token on RFC2833, input range form 96 to 127.

- **RTP Packet NAT Detect by:**
  User can select the packet transmit method when under the NAT network environment, by IP (NAT 255 fake IP), or transmit by TCP port, usually suggest select the option "By IP or By Port".

- **Extension Number SIP Header To:**
  Use the number in the SIP header column as the dial out number.

- **Supported SIP 100rel:**
  The name of the option tag for the SIP extension that is supported by this SIP protocol is 100rel, it is used for Sending Reliable Provisional Responses. Enable this item if needed.

- **RTP Packet Monitor Time [0 Disable](sec)**
  If set 60 sec, when network failed over 60 seconds, device will cut this call automatically.

- **Silence Detection / Suppression:**
  Enable or disable the Silence Detection/VAD function. When Enable, if local gateway detect a silence situation (no talk), it will send a VAD package rather than a full voice package for remote side to active CNG (Comfort Noise Generation) to save the bandwidth. The default is Enable to save the bandwidth.

- **Prefer CODEC** :
  In this table, you can define the prefer CODEC. The priority 1 selection is highest priority. By different CODEC, user can select different payload size per package as below:
  - G.711 uLaw: 20,30,40,50,60,70,80ms
  - G.711 aLaw: 20, 30,40,50,60,70,80ms
  - G.723.1:   30,60,90ms
  - G.729a: 20,30,40,50,60,70,80ms
  - G.726:   20,30,40,50,60,70,80ms
  - None: none

  The gateway will calculate and show approximately bandwidth for one VoIP call.

2.6.4 RTP Packet Summary

Line1	G.729A 8kbps	Packet Send:	4213	Packet Received:	4081	Packet Lost:	0
	Source IP:	163.20.15.125		Source Port:	10170	Packet Interval[ms]:	20
Line2	G.729A 8kbps	Packet Send:	142	Packet Received:	138	Packet Lost:	0
	Source IP:	163.20.15.125		Source Port:	19844	Packet Interval[ms]:	20
Line3	G.729A 8kbps	Packet Send:	9451	Packet Received:	9456	Packet Lost:	0
	Source IP:	163.20.15.125		Source Port:	16888	Packet Interval[ms]:	20
Line4	G.729A 8kbps	Packet Send:	2471	Packet Received:	2323	Packet Lost:	0
	Source IP:	163.20.15.125		Source Port:	15444	Packet Interval[ms]:	20
Line5	G.729A 8kbps	Packet Send:	86679	Packet Received:	86561	Packet Lost:	0
	Source IP:	163.20.15.125		Source Port:	12664	Packet Interval[ms]:	20
Line6		Packet Send:	0	Packet Received:	0	Packet Lost:	0
	Source IP:			Source Port:	0	Packet Interval[ms]:	0
Line7		Packet Send:	0	Packet Received:	0	Packet Lost:	0
	Source IP:			Source Port:	0	Packet Interval[ms]:	0
Line8		Packet Send:	0	Packet Received:	0	Packet Lost:	0
	Source IP:			Source Port:	0	Packet Interval[ms]:	0

Reload

2.6.4 /Advance Setup/RTP Packet Summary

On this page, user will know the RTP package summary about last VoIP call.

- Line#: number of line
- Using CODEC: ex.: G.729a
- Source IP: Remote side IP
- Source Port: Remote side port
- Packet Interval: interval time between 2 packets.(ms)
- Packet Send: number of packets sent.
- Packet Received: number of packets received.
- Packet Lost: number of lost packets.

2.6.5　Flash & Call waiting

2.6.5 /Advance Setup/Flash & Call waiting

On this page, user can define the parameters relative to the FLASH key and Call Waiting function. These functions usually work with PBX

- **Token for flash key on VoIP(!):**

    Define the token for flash key during VoIP protocol (use "! " by default).

- **Flash Signal generate length :**

    Define the pause time (ms) for one "," symbol at */Routing Setting/VoIP Call Out/*. This pause till is useful for PBX seize the trunk line from extension line. The default time is 1000ms, Input range from 100 to 3000ms.

- **Flash Signal Detect Threshold:**

    Define the threshold for valid FLASH signal. Only the flash time length between setting between min. to max. is accept by the gateway.

- **VoIP User Agent:**

    You can fill in the value to this packet, usually it used for the communicating recognization.

- **VoIP Transfer:**

    Select the proper transfer method for VoIP in this item, Unattended (For INVITE), Attended (For INVITE) , Attended (For INFO Mode1) ,Attended (For INFO Mode2)　or Flash+N(For CHT).

2.6.6 Gain

2.6.6/VoIP Setup/Advance Setup/Gain

This page defines different function gain on the gateway.

● **Gain when Dial tone phase**：

When phone off hook, user will hear the dial tone generated from the gateway, sser can adjust the play/record gain during this phase for stable DTMF detection. After connection, the gains setting here is no use, the gateway will adjust the gain setting on */Line Configure/Line setting/*

■ **Play:** Transmit gain from network to line. Adjust the speaker volume on the handset. Higher value will louder the speaker on local side.

■ **Record**: Receive gain from line to network. Adjust the microphone volume on the handset. Larger value which will amplifier the MIC volume on local site.

Incorrect value will cause the gateway can not receive the DTMF user pressed on phone set, please use the default 0dB if no other issue.

● **DTMF Generate DSP play Gain** [-29~3]

Setting the internal gain used by DSP for generate the DTMF signal, incorrect value will cause the DTMF can not accept by other telephone equipment. Please use the default value if no other issue.

● **Call progress Tone DSP play Gain**[-31~0]

Setting the internal gain used by DSP for generate the CPT (Call Progress Tone), Incorrect value will cause the DTMF can not accept by other telephone equipment. Please use the default value if no other issue.

● **Caller ID Detection record Gain** [13~-3]

Setting the Caller ID Receiver gain. Incorrect value will cause the Caller ID signal can not be receive, please use the default value if no other issue.

- **Gain when VoIP DSP Talking phase**
  This item allows user to adjust the talking volume of the device, which means it will affects all ports of this device, User can select the db value of Play and Record, it will increase or decrease the talking and listen volume.

  User can adjust individual port talking or listen volume in page *Line Configure* → *Line Setting*.

2.6.7 QoS

2.6.7 /Advance Setup/QoS

User can define the ToS field on the VoIP packet for Quality of Service control. The ToS field is included these 2 parameters:

- Precedence: 0~7
- DSCP(Diffserv Code Point):0~46

User can select inptut either IP Precedence or DSCP value. Or input the ToS binary code directlly.

The QoS function works depending on the router in front of the VoIP device, which means the router need to support the Tos, diffserv recognition also.
User can select Tos or diffserv to define the QoS method, voip device need to match the Tos tag or value of diffserv with the router, so that the voice packets can be sent with priority.

2.6.8   CDR

2.6.8 /Advance Setup/CDR

The Gateway can export all the CDR (Call detail Record) to external CDR server by HTTP protocol. The gateway supports up to 2 CDR servers for keep the record.

- **Export to CDR Server:** Please install and enable this function if you want to keep CDR of this gateway.
- **CDR Server IP1:**   Please input the IP of first CDR server, if installed.
- **CDR Server IP2:**   Please input the IP of second CDR server, if installed.

2.6.9 FoIP

2.6.9/Advance Setup/FoIP

User can define the parameters relative FAX Over IP function.

- FXS Mode:Disable , T.38 , T.30 , T38+T30
- **Maximum FoIP Rate**

  Appoint the maximum FAX transceiver rate during FoIP:
  - ◆ Disable: Only the VoIP function is supported on the gateway. FoIP is disabled.
  - ◆ Auto: Gateway will negotiation the maximum speed for FoIP with FAX machine..
  - ◆ Appoint the Maximum speed:2400,4800,9600,12000, 14000

- **CED Tone(2100 Hz) Detection Gain**: [-45~0]

2.6.10 Prompt Voice & Beep

2.6.10 /Advance Setup/Prompt Voice & Beep

This gateway can use voice or beep to prompt the user different situation. User can download/upload their own prompt voice wave files also.

For prompt beep enable function, it can be enable by:
- **VoIP Call out Beep**:
  When enable, the gateway will generate a beep for call out for VoIP

- **VoIP Call out Failure twice Beep**
  When enable, the gateway will generate twice beep if failure to call out for VoIP

- **No PSTN line warming twice Beep** :
  When enable, the gateway will generate twice beep if failure to call out for PSTN, usually mean there is no trunk line connect to the PSTN line port.

- **Can not register to server warming twice Beep:**
  When enable, the gateway will generate twice beep when end user off hook the phone if the gateway failure to register to Register Server.

For prompt voice enable function, it can be enable by:
- **Prompt voice for replace dial tone**:
  Use a customize voice file to replace the dial tone

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- **Warming Prompt after VoIP out failure**:
  Annunciate a customize voice file when VoIP call failure.
- **No PSTN Line Connected** :
  Annunciate a customize voice file when there is no trunk connected to the PSTN port and failure to call via PSTN line.
- **Can not register to server warming prompt** :
  Annunciate a customize voice file when the gateway failure to register to Register Server.
- **Prompt voice for VoIP call in** :
  When Remote gateway call in the gateway, if enable this function, the gateway will annunciate a customize voice to remote gateway user to ask the destination number , this function must work with the setting rule on */VoIP Setup/Routing Setup/VoIP Call In IVR*.
- **Warning Prompt after Check ANI number failure** :
  Annunciate a customize voice file when check ANI number failure.

For enable the prompt beep or voice annunciation function, please select ☑ to enable the function and click Modify

Caution: If enable both prompt beep and Voice annunciation on the same function, only the voice annunciation will work and will not hear the beep sound.

Procedure to upload customize voice wave file:
  1.Select the function index you want to modify.
  2.press brows , select the content voice file.
  3 Press Restore to upload and save.
  4.To keep the voice file permanently, press Save Flash to save it

  *. The gateway only accepts the G.723.1 or G.711 format voice file, and all the 5 files size totally can not exceed 384KB.

2.6.11　Call log

2.6.11/Advance Setup/Call log

- The gateway can export status and performance to remote system.
- Syslog supports eight leavels:
  Disable、Emergency、Alert、Critical、Error、Warning、Notice、Info、Debug

2.7   Application

2.7.1   Ping Test

2.7.1 /Application/Ping Test

User can use the Ping Test function to test the network status or remote device

■   ***Ping Destination:***
Target device IP for ping test.

■   ***Number of Ping[1-100]:***
Number of ping test, maximum is 100.

■   ***Ping Packet Size[56-5600 bytes]:***
Size of ping test packet, input range is between 56 to 5600 bytes。

2.7.2   Centrex

2.7.2/VoIP Setup/Application/Centrex

If you are connected to the PSTN device is applied Centrex service , please enable Centrex access code, when you press the access code , you will hear a dial tone.

2.7.3 Telnet & SNMP

Telnet and Ftp Services Settings

Enable telnet server :	Disable
Enable SNMP server :	Disable

Modify Reset

User Name:	
Login Password:	
Confirm Password:	

Modify Reset

2.7.3 /Application/Telnet & SNMP

User can enable and set the user account for the SNMP and Telnet function of this gateway on this page.

a. **Enable telnet server**
   - Enable: Enable telnet service function, user can telnet this gateway.
   - Disable: Disable telnet service function.

b. Enable SNMP Server
   - Enable: Enable SNMP service function, user can use SNMP on the gateway.
   - Disable: Disable SNMP service function.

c. User Name：Set a user name for Telnet & SNM login.
d. Login Password：Set the password for Telnet & SNMP login.
e. Confirm Password：Check the password again.

● How to use Telnet :
   1. Using telnet login::
voip login:
password:

welcome
2.   telnet command:

Help

Help:
sini
cat
save
ping
tracert
help
exit
;
3.   Use the ping command to test ping time:
ping www.hinet.net
pinging www.hinet.net (203.66.88.89) With 32 bytes:
32 byes from 203.66.88.89: seq=1 ttl=249 time=40ms
32 byes from 203.66.88.89: seq=2 ttl=249 time=40ms
32 byes from 203.66.88.89: seq=3 ttl=249 time=40ms
ping statistics www.hinet.net:
packets: send=4 received=4 lost=0

4.   Use the tracert command to test the routing rule :
tracert www.hinet.net
[tracer   hinet.net]-10...
tracert www.hinet.net
tracert to www.hinet.net (61.219.38.89), 30 hops max, 40 byte packets
  1   172.16.7.1 (172.16.7.1) 0ms 0ms 10ms
  2   220-139-40-254.dynamic.hinet.net. (220.139.40.254) 40ms 40ms 50ms
  3   tp-sc-c6r1.router.hinet.net. (168.95.170.34) 40ms 40ms 50ms
  4   tp-sc-c12r1.router.hinet.net. (203.75.134.2) 40ms 50ms 40ms
  5   tp-s2-c12r1.router.hinet.net. (210.65.2.162) 50ms 40ms 40ms
  6   tp-s2-c6r9.router.hinet.net. (211.22.35.1) 50ms 40ms 50ms
  7   * * *
tracert end

2.7.4   Call Timer

2.7.4 /Application/Call Timer

If youneed to limit the user's talk time per call , you can set the call timer.

- ■ **Pre Alam**.
- ■ **Line**
- ■ **Remark**
- ■ **Area Code**
- ■ **Miniute**
- ■ **Enabel**
- ■ **Delete**

3. System Setup

3.1 System

3.1.1 System Status

System Status	
Reload (WAN1)	
INTERNET	
Cable/DSL	Connected
WAN IP	163.20.172.32
Subnet Mask	255.255.255.0
Gateway	163.20.172.254
DNS	168.95.1.1
Secondary DNS	168.95.192.1
Domain Name	
Connection Type	Static IP
INTERNET #2	
Cable/DSL	Disconnected
WAN IP	0.0.0.0
Subnet Mask	0.0.0.0
Connection Type	Disable
WAN Gateway Select	WAN1
GATEWAY	
IP Address	192.168.22.1
Subnet Mask	255.255.255.0
DHCP Server	Enabled
NAT	Enabled
Firewall	Disable
INFORMATION	
System Up Time	78 days 14:48:17
System Date	08/17/2011 11:28:35
Connected Clients	0
Runtime Code Version	2.6.30.9 #301 Wed Apr 20 14:22:28 CST 2011
Boot Code Version	0.1.0.f
LAN MAC Address	00:02:46:61:0D:AD
WAN MAC Address	00:02:46:61:0D:AC
Hardware Version	0.1.2.3
Serial Number	06360492

3.1.1 /System/System Status

This page reveals the status of the gateway including WAN, LAN and some hardware information.

Internet

This sub-block shows the Internet information of your home gateway.   It depends on the WAN mode connecting to your ISP.   The different items correspond to each WAN mode will be revealed after the common part of the Internet status sub-block.

<u>***Common Part:***</u>

Refresh

Clicking this button, the browser refreshes the Internet status page to get the most update information.

Cable/DSL

This field indicates the Internet connection status.   Its value is Connected , Disconnected or Connecting.

WAN IP

Connected to the Internet through Cable or ADSL modem, the ISP will offer the home gateway a WAN IP address to communicate with other hosts in the Internet.

Subnet Mask

This field indicates a <u>mask</u> used to determine what <u>subnet</u> the WAN <u>IP address</u> belongs to. An <u>IP address</u> has two components, the network address and the <u>host</u> address.   For example, consider the IP address 192.168.168.182 with subnet mask is 255.255.255.0, the first three numbers (192.168.168) represent the Class C network address, and the forth number (182) identifies a particular host on this network.

Gateway

"Gateway" is a <u>node</u> on a <u>network</u> that serves as an entrance to another <u>network</u>.   For the home gateway, The "Gateway" is the next device, which routes the traffic to the Internet.

DNS

Domain Name System (or Service or Server) is an Internet service that translates domain names into IP addresses.   Because the domain names are alphabetic, they are easier to remember.   However, the Internet is based on <u>IP addresses</u>.   Every time you use a domain name, a <u>DNS</u> service must translate the name into the corresponding IP address. For example, the domain name www.example.com might translate to 198.105.232.4.   The DNS system has its own <u>network</u>.   If one <u>DNS server</u> doesn't know how to translate a particular domain name, it will ask its upper stream server, and so on, until the correct IP address is returned or timed-out.

Secondary DNS

This is the secondary DNS to use when the primary DNS does not work.

Domain Name

Domain name is a name, which identifies one or more IP addresses.   This field represents the domain name obtained from your ISP.

Connection Type

There are five ways to get the WAN IP address. They are DHCP, STATIC, PPPoE, PPTP and L2TP. This field indicates the way to get the WAN IP address.   Through Figure 3-2 to 3-6 detail all specific items of each mode.

Gateway

IP Address

This field is the LAN IP address of the home gateway.

Subnet Mask

This field is the subnet mask of the network in the LAN side.

DHCP Server

The home gateway supports DHCP service.   This field indicates the enabled status of the DHCP Server.

NAT

This field shows whether the NAT is enabled or not.

Firewall

The gateway supports firewall service. This field indicates firewall service is enabled or not.

Information

System Up Time

Shows the time in *hh:mm:ss* format from when the home gateway was powered up to the web browser requests this page.

System Date

Shows the data and time in *mm/dd/year hh:mm:ss* when the web browser requests this page.

Connected Clients

This field shows how many clients in the LAN clients connect to the home gateway.

Runtime Code Version

Shows the version of runtime code.

Boot Code Version

Shows version of boot code.

LAN MAC Address

Short for Media Access Control address, a hardware address that uniquely identifies each node of a network.   This field indicates gateway's LAN MAC address.

WAN MAC Address

This field indicates gateway's WAN MAC address.

Hardware Version

Tells the version of hardware of the gateway.

Serial Number

This field indicates serial number.

3.1.2 ystem Settings

3.1.2 /System/System Settings

This page is used to configure the names given by the ISP, if any, to represent the gateway.

◆ *Host Name*

Some ISPs request the host name to represent the home gateway.   Fill the host name given by the ISP, or you may not be able to access the Internet successfully.   The maximum length of the host name is 32 bytes.

◆ *NTP Server*

Network Time Protocol is used to obtain the time from the Internet NTP server.   The home gateway will resolve the NTP server from the internal URL lists.   If you know a better NTP server, you can enter it.   Domain name and IP address format are both acceptable.

3.1.3 Date&Time

Date&Time	
Time establishing	
It sets for the systematic time of VOIP	
Systematic time at present	**August 17, 2011 12:24:35**
Correct the time way:	
⊙ SNTP corrects time	
○ Correct time with your computer	
○ Insert date and time manually	
Time zone	(GMT+08:00) Hong Kong, Perth, Singapore, Taipei ▾
When automatic with NTP server is right	
NTP server preserved	time.stdtime.gov.tw (Do not fill out)
Time	Year 2011 ▾ Month August ▾ Day 17 ▾
	Hour 12 ▾ Minute 24 ▾ Second 27 ▾
	Modify Reset

3.1.3/System Setup/System/Date&Time

This page is used to setting the system time of VoIP gateway; it can define the correct time by which ways.

3.1.4 Administrator Settings

3.1.4 /System/Administrator Settings

This page allows you to change the user and administration password used to manage this router for security reasons.

3.2 WAN

3.2.1 WAN Settings

Wan Settings	
○ Dynamic IP Address	Obtain an IP address automatically from your service provider.
● Static IP Address	Uses a static IP address. Your service provider gives a static IP address to access Internet services.
○ PPPoE	PPP over Ethernet is a common connection method used for xDSL.
○ Bridge	Participate WAN & LAN interface in a bridge group.

Modify   Reset

3.2.1.1/WAN/WAN Settings

There are four ways to connect to the Internet.
They are Dynamic IP , Static IP, PPPoE, Bridge.

IP   Dynamic IP

Dynamic IP	
Domain Name	
DHCP Vendor Class ID	voip
Client ID	
Request IP address	. . .
MTU(576-1500)	

Modify   Reset

3.2.1.2/WAN/WAN Settings/Dynamic IP

◆ **Domain Name**
   User configured domain name of the network maintained by gateway.
◆ **DHCP Vendor Class ID**
   This is a packet label which can be used to recognize for different Vendors, fill in this area if needed. The value of this class ID can be checked by software like ethereal.
◆ **Client ID (Option 61) ,** *Option 61 allows the DHCP Server to be setup to provide a* preprogrammed IP address to a field device based on its Media Access Control (MAC) Address or an ID entered into the field equipment.
◆ **Request IP address**

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You can specify the IP address you desired.　　But the ISP has the right to neglect it and provides you a different one.

◆ *MTU (576-1500)*

You can specify the MTU (maximum transmission unit) of your home gateway.　　The default value is 1500 bytes and in normal case, you don't have to change.

Static IP

Static IP

IP address assigned by your ISP	59	119	110	50
Subnet Mask	255	255	255	248
ISP Gateway Address	59	119	110	49

[Modify] [Reset]

3.2.1.3/WAN/WAN Settings/Static IP

◆ *IP address assigned by your ISP*
   Set the IP address that assigned by the ISP.

◆ *Subnet Mask*
   Set the subnet mask of the network.

◆ *ISP Gateway Address*
   Set the ISP's gateway IP address. This address routes packets to Internet.

PPPoE

PPPoE

User Name	1232
Password	••••••••
Please retype your password	••••••••
Service Name	
MTU (546-1492)	1492

[Modify] [Reset]

3.2.1.4/WAN/WAN Settings/PPPoE

This page is the PPPoE configuration page.   Most of the ISPs request the user to connect to central office (CO) side via PPPoE, acronym of Point-to-Point Protocol over Ethernet, which provides authentication, authorization and accounting.

◆ *User Name*
   Enter the user name provided by your ISP to identify the computer using PPPoE.
◆ *Password*

Enter the password provided by your ISP to identify the computer using PPPoE.

◆ *Please retype your password*

Retype the password to make sure type correct password.

◆ *Service Name*

Some ISP provides the service name of this PPPoE connection.   If so, enter this item, or make it blank.

◆ *MTU (546-1492)*

Maximum Transmission Unit (MTU) is the largest physical packet size measured in bytes, which a network can transmit.   Any messages larger than the MTU are divided into smaller packets before sent.   In ordinary, that the user does not have to worry about the MTU size, the gateway routing engine will handle the MTU differences between PPP and the LAN Ethernet side.   But for some old PPPoE server, you have to make the MTU size of the PPPoE side smaller than the default value, or some Web side is not able to access.

PPPoE MTU should be set between 546 and 1492.

Bridge

Bridge IP				
Bridging group IP address	172	16	5	2
Bridging group Subnet Mask	255	255	0	0
Bridging group Gateway Address	172	16	7	1
	Modify	Reset		

3.2.1.5/WAN/WAN Settings/Bridge

User can set this device work as a bridge, set the necessary information.

A network bridge connects multiple network segments at the data link layer (layer 2) of the OSI model, and the term layer 2 switch is often used interchangeably with bridge.

Bridging group IP address

Enter the bridging IP address in this column.

Bridging group Subnet Mask

Enter the bridging Subnet Mask in this column.

Bridging group Gateway Address

Enter the bridging gateway IP address in this column.

3.2.2 WAN Settings #2

3.2.2/WAN/WAN Settings #2

◆ **Enable**

 WAN2 settings enabled

◆ **Backup**

 When WAN1 network error , it will automatically switch to WAN2。

◆ **WAN1 Ping check address**

◆ **WAN2 Ping check address**

There are six ways to connect to the Internet.They are Dynamic IP, Static IP, PPPoE and Bridge.

3.2.3 DNS

This page sets the primary and secondary DNS servers, which were given by your ISP or known to you.   When a domain name request received, the gateway tries to resolve to it from the Primary Domain Name Server.   Resolving failed, the gateway tries the Secondary sever again.

◆ *DNS Proxy*
    Enable or disable DNS Proxy.

◆ *Domain Name Server*
    Set your primary DNS in this field.

◆ *Secondary DNS Address(optional)*
    Set your secondary DNS to use when the primary DNS does not work

3.3 LAN

3.3.1 LAN Settings

3.3.1 /LAN/LAN Settings

The home gateway is an IP sharing device, which provides the home users share the same public IP address.   While in the LAN side, each network device must have one private IP address to do network communication.   This page is to set the configuration of the LAN interface of the gateway.

◆ **IP 位址 IP Address**

Set this to be gateway's LAN interface IP address.   The LAN interface address is also aced as the default gateway address to the computers in you private network.

◆ **Subnet Mask**

This field indicates a mask used to determine what subnet the LAN IP address belongs to.

◆ **The Gateway acts as DHCP Server**

Check this item, if the home gateway supports DHCP server service.   This is the normal case that can make you free from installing a DHCP server in your home network.   And you can connect to the Internet just as what you have done in your company environment.

◆ **IP Pool Starting Address**

The starting address provided by DHCP service

◆ **IP Pool Ending Address**

The ending address provided by DHCP sevice.

◆ **Lease Time**

Set the lease time of the IP address to renew.

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◆ *Local Domain Name*

Set the gateway's local Domain Name.

◆ *DNS Proxy*

The home gateway acts as a DNS Proxy.　In this case, the DHCP service will set LAN interface IP address as the DNS server address, and inform the clients in the DHCP renew process.

3.3.2 DHCP Client List

3.3.2 /LAN/DHCP Clients List

This page lists all the DHCP clients in the LAN side. The DHCP server is capable of administering 253 clients.

◆ *Host Name*

This column shows the host name of the DHCP clients.

◆ *IP Address*

This column shows the IP Address of the DHCP clients.

◆ *MAC Address*

This column shows the MAC Address of the DHCP clients.

◆ *Remaining Time*

This is the amount of time the lease is valid if not renewed before it expires.

3.4 Wireless

3.4.1 Basice Settings

Wireless Basic Settings

☑ **Enable Wireless LAN Interface**

Band:	2.4GHz (B+G+N) ▼
Mode:	AP ▼
SSID:	ALLWIN
Channel Width:	40MHz ▼
Control Sideband:	Upper ▼
Channel Number:	Auto ▼
Broadcast SSID:	Enable ▼
Data Rate:	Auto ▼
Associated Clients:	Show Active Clients

Modify   Reset

This page is used to configure the parameters for wireless LAN clients which may connect to your Access Point. Here you may change wireless encryption settings as well as wireless network parameters.

3.4.1 /Wireless/Basic Setting

This page is used to configure the parameters for wireless LAN clients which may connect to your Access Point. Here you may change wireless encryption settings as well as wireless network parameters.

◆ *Enable Wireless LAN Interface*

Check the box to enable wireless funtion.

◆ *Band*

Allows you to set the AP fixed at 802.11b , 802.11g or 802.11n mode. You can also select B+G+C mode to allow 80211b , 802.11g and 802.11n clients at the same time.

◆ *Mode*

◆ *SSID*

Set a name you would like to call your wireless.SSID can be any text numbers.

◆ *Channel Width*

The amount of data that can be transmitted also depends on the width of the channel used in data transmissionis the amount of time the lease is valid if not renewed before it expires. In 802.11n, two adjacent channels, each of 20 MHz are bonded to get a total bandwidth of 40

MHz

◆ *Control Sideband*

Specify if the extension channel should be in the Upper or Lower sideband.

◆ *Channel Number*

It means the channel of frequency of the wireless LAN. Sets the channel on which the gateway operates. If you are experiencing interference or wireless network problems, changing the channel may solve the issue. It is recommended that you keep the control channel set at "Automatic" to avoid interference with adjacent networks.

◆ *Broadcast SSID*

Chooes Enable to make the SSID being seen by wireless clients.

◆ *Data Rate*

Specify the rate at which multicast packets are transmitted and received on your wireless network.

◆ *Associated Clients*

This table shows the MAC address, transmission, reception packet counters and encrypted status for each associated wireless client.

Active Wireless Client Table

MAC Address	Mode	Tx Packet	Rx Packet	Tx Rate (Mbps)	Power Saving	Expired Time (s)
00:15:00:1d:f8:3e	11g	29947	29735	54	no	300
78:a3:e4:db:66:3f	11n	309	540	65	yes	291
18:87:96:b9:31:c1	11n	119	456	65	yes	291
64:a7:69:70:ab:19	11g	1385	1100	48	yes	270

3.4.2 Advanced Settings

Wireless Advanced Settings

Fragment Threshold:	2346 (256-2346)
RTS Threshold:	2347 (0-2347)
Beacon Interval:	100 (20-1024 ms)
Preamble Type:	⊙ Long Preamble ○ Short Preamble
IAPP:	○ Disabled ⊙ Enabled
Protection:	⊙ Disabled ○ Enabled
Aggregation:	○ Disabled ⊙ Enabled
Short GI:	○ Disabled ⊙ Enabled
WLAN Partition:	⊙ Disabled ○ Enabled
STBC:	⊙ Disabled ○ Enabled
20/40MHz Coexist:	⊙ Disabled ○ Enabled
RF Output Power:	⊙ 100% ○ 70% ○ 50% ○ 35% ○ 15%

Modify   Reset

3.4.2 /Wireless/Advanced Settings

◆ **Control Sideband**

The maximum level the gateway will reach when sending information in packets before the packets are broken up in fragments. Typically, if you are experiencing problems sending information, it is because there is other traffic on the network and the data being transmitted is colliding. This might be corrected by the information being broken into fragments. The lower the level that the Fragmentation threshold is set to, the smaller a packet has to be before it is broken into fragments.

◆ *RTS Threshold*

The RTS Threshold is the minimum size in bytes for which the Request to Send/Clear to Send (RTS/CTS) channel contention mechanism is used. In a network with significant radio interference or large number of wireless devices on the same channel, reducing the RTS Threshold might help in reducing frame loss. The RTS threshold is 2347 bytes by default, which is the maximum value.

◆ *Beacon Interval*

The amount of time between beacon transmissions. A beacon is basically a heartbeat for a wireless client or gateway, sending out a signal informing wireless clients that the wireless network is still active. The default beacon interval is 100 ms.

◆ *Preamble type*

If you are not using any 802.11b devices in your network, you can configure the Preamble

type to Short for optimum performance. The Long Preamble type should be used when both 802.11g and 802.11b devices exist on your network.

◆ *IAPP*

◆ *Protection*

The 802.11g standards provide a protection method so 802.11g and 802.11b devices can co-exist in the same network without "speaking" at the same time. Do not disable 802.11n protection if there is a possibility that 802.11b or 802.11g devices will use your wireless network.

◆ *Aggregation*

◆ *Short GI*

◆ *WLAN Partition*

◆ *STBC*

◆ *20/40MHz Coexist*

◆ *RF Output Power*

The Power level sets the strength of the wireless signal that the gateway transmits.

3.4.3　Security

3.4.3 /Wireless/Wireless Security Setup

Encryption

　　Select an appropriate encryption mode to improve the security and privacy of your wireless data packets. Each encryption mode will bring out different web page and ask you to offer additional configuration.

Authentication Mode

　　Enterprise(RADIUS) or Personal (Pre-Shared Key)

　　Enterprise(RADIUS):

　　Every user must use a valid account to login to this Access Point before accessing the wireless LAN. The authentication is processed by a RADIUS server.

WPA Cipher Suite

WPA2 Cipher Suite

Pre-Shared Key Format

Pre-Shared Key

3.4.4 Wireless Access Control

3.4.4 /Wireless/Wireless Access Control

Access Control allows you to configure AP to give exclusive access to wireless stations or exclude them from accessing AP. Select Allow Listed to allow the wireless stations to access the wireless network. Otherwise, select Deny Listed to deny the wireless stations to access the wireless network.

3.4.5 Site Survey

Wireless Site Survey

Site Survey

SSID	BSSID	Channel	Type	Encrypt	Signal(Max.100)
conplus	00:0a:79:98:20:33	1	2.4GHz (B+G)	WEP	28
Superior	00:1e:58:25:d5:3b	1	2.4GHz (B+G+N)	WEP	24

This page provides tool to scan the wireless network. If any Access Point or IBSS is found, you could choose to connect it manually when client mode is enabled.

3.4.5 /Wireless/Site Survey

This page provides tool to scan the wireless network. If any Access Point or IBSS is found, you could choose to connect it manually when client mode is enabled.

3.5   NAT

3.5.1   Virtual Server

Index	Private IP	Private Port	Type	Public Port	Comment	Enabled
1	192.168.22.		BOTH ▾			☐
2	192.168.22.		BOTH ▾			☐
3	192.168.22.		BOTH ▾			☐
4	192.168.22.		BOTH ▾			☐
5	192.168.22.		BOTH ▾			☐
6	192.168.22.		BOTH ▾			☐
7	192.168.22.		BOTH ▾			☐
8	192.168.22.		BOTH ▾			☐
9	192.168.22.		BOTH ▾			☐
10	192.168.22.		BOTH ▾			☐

Modify   Reset

3.5.1/NAT/Virtual Server

This is the virtual server page, which set the rules translate the private IP/private port pairs to the public ports.

◆ *Private IP*

Set the IP address of the virtual server.

◆ *Private Port*

Set the port of the virtual server to connect to Internet.

◆ *Type*

Set the protocol of the virtual server.   Valid options are TCP, UDP and Both.

◆ *Public Port*

Set gateway's WAN port to connect to virtual server.

◆ *Comment*

Let you put some notes to describe this entry.

◆ *Enabled*

Enable this entry.

3.5.2　Port Mapping

3.5.2/NAT/Port Mapping

The Port Mapping page does a port range mapping.

◆　*Server IP*

Set one of LAN client's IP address to do port mapping.

◆　*Mapping Ports*

Set the ports to mapping from WAN to LAN.　You can enter multiple ports, delimited by comma, or a range of port by dash.　White space will be neglect.

◆　*Type*

Defines the protocol type, TCP/UDP/Both, of the public ports.

◆　*Comment*

Let you put some notes to describe this entry.

◆　*Enabled*

Enable this entry.

3.5.3 ALG

ALG	
FTP:	☐
Non-Standard FTP Port:	[ ]
	Modify Reset

3.5.3 /NAT/ALG

Some applications have to do Application Level Gateway (ALG) to monitor the transaction or monitor the payload of the packet.   This page let you setup them.

We use mnemonic nouns to describe the specific ALGs.   The checked entries will be enabled by the ALG processing.

For the Non-Standard FTP Port, you have to set the port number to let the ALG take it as an FTP session.

3.6 Firewall

3.6.1 Denial-of-Service

3.6 /Firewall/Denial-of-Service

◆ **Enable DoS Prevention**

DoS Setup Denial of Service A "denial-of-service" (DoS) attack is characterized by an explicit attempt by hackers to prevent legitimate users of a service from using that service.

3.7 Routing

3.7.1 Routing Table

*圖 3.7.1/ **Routing/Routing Table***

◆ **Destination LAN IP**

This field indicates the destination IP of this route entry.

◆ **Subnet Mask**

This field indicates the Subnet Mask of this route.

◆ **Gateway**

If a packet's destination IP address do "bit and" operation with the "Subnet Mask" equals to the "Destination LAN IP", the packet will send to the Gateway.

◆ **Interface**

This field indicates which network interface deal the connection.

◆ **Reload**

Reload the routing information from the home gateway again.

3.7.2　Static Routing

3.7.2/ Routing/Static Routing

◆　***Destination LAN IP***

This field indicates the destination IP of this route entry.

◆　***Subnet Mask***

This field indicates the Subnet Mask of this route.

◆　***Gateway***

If a packet's destination IP address do "bit and" operation with the "Subnet Mask" equals to the "Destination LAN IP", the packet will send to the Gateway.

◆　***Add***

Click this button to add this entry to routing table statically.

3.8 Bandwidth&VLAN

3.8.1 Bandwidth Control

*3.8.1/ **Bandwidth&VLAN/Bandwidth Control***

You can enable WAN or LAN bandwidth control in this configuring page, so far only the Upstream Bandwidth is workable, you can fill in the limit value in Upstream Bandwidth to make limitation.

3.8.2   VLAN

VLAN				
Enable VLAN	☐			
	Port Enable	VLAN ID[1-4094]	Priority[0-7]	CFI
LAN 1	☐	22	1 ▾	☐
LAN 2	☐	33	2 ▾	☐
VOICE	☐	66	5 ▾	☐
		Modify		

*3.8.2/ **Bandwidth&VLAN/VLAN***

This function is used for special purpose of application. For more details about VLAN, please refer to application note for VLAN, which is in the end of this manual.

3.9　Backup / Restore

3.9.1　Configurations

3.9.1 / Backup/Restore /Configurations

This page let you backup / Restore all of your configuration parameters on the VoIP Router. It is very good idea to back up all of your VoIP Router configuration parameters after install.

a. To Backup, press Download setting backup file, and input the file name you want and file location to save.

b. To Restore, press the Browse button the select the backup configuration parameters file to upload then press Restore . After you upload the file, Press Saved modification to save your current configuration to Flash ROM (Usually used to save currently WAN configuration).After save, please remember to Reboot the VoIP Router to let the restored parameters take effective.

*** Caution: Never power off the VoIP Router when during Restore configure or upgrade VoIP module or System, it will cause permanent damage when power off during writing Flash inside VoIP Router.

3.9.2 VoIP module


*3.9.2 / **Backup/Restore /VoIP Module***

This page displays the current firmware module version and let you backup / Restore all of your VoIP firmware module on the VoIP Router. Please use this page to update the VoIP module firmware if necessary.

a. To Restore from local file, press the Browse button the select the VoIP module file to upload then press Restore . After you upload the file, Press Saved modification to save your current configuration to Flash ROM (Usually used to save currently WAN configuration).After save, please remember to Reboot the VoIP Router to let the restored parameters take effective.

b. To Restore from Upgrade server, please input the URL address of upgrade server and press the APServer to link to upgrade server to get latest version firmware and upgrade automatically.

*** Caution: Never power off the VoIP Router when during Restore configure or upgrade VoIP module or System, it will cause permanent damage when power off during writing Flash inside VoIP Router.

3.10   Reboot

Reboot System

Are you sure you have already **saved modification**?
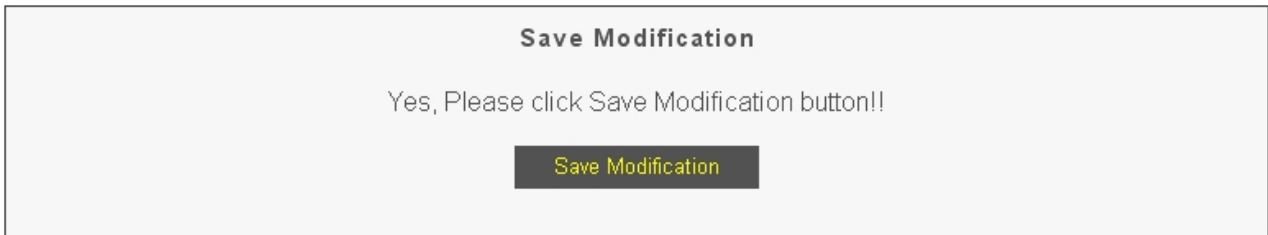Yes, Please click Reboot button!
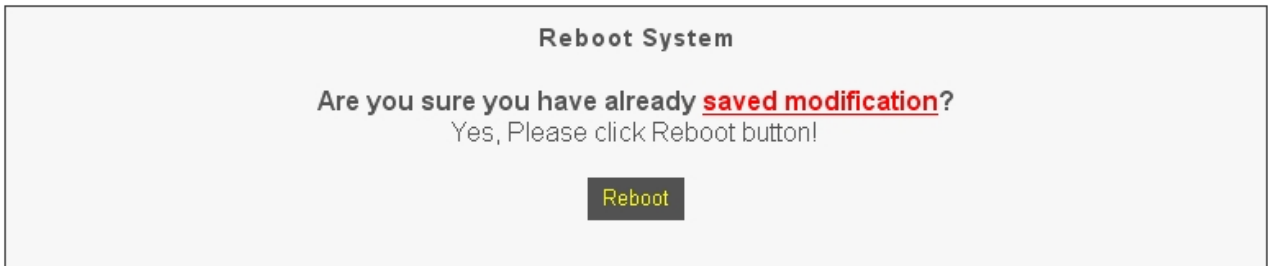
Reboot

3.10 /Reoobt

Use the Reboot button on this page to reboot your VoIP Router, before you reboot, please make sure you have to press the Saved modification to save your current configuration to Flash ROM, otherwise all the change will be disappear after reboot.

3.11 Save Modification

3.11 / Save Modification/

Most of the VoIP Router parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the VoIP Router, to save the parameters into Flash ROM and let it take effective forever, please remember to press the **Save Modification** button after you modify the parameters.