VoIP Gateway
Smart WEB

High Performance VoIP Gateway Solution

WEB Operation Guide

AddPac

www.addpac.com

Contents

WEB Connection

1. Web Connection via Console Port
2. Web Connection via LAN 1 Port

WEB Connection

1. Web Connection via Console Port

Serial port

Baud rate 9600
No parity
1 stop bit
No flow control

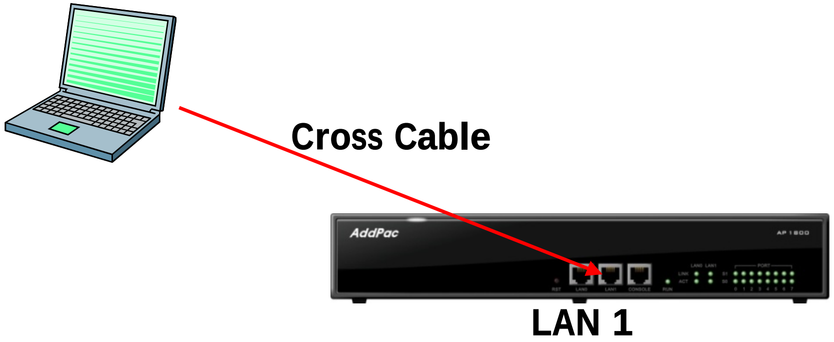
Login ID/Password : root/router

1. Connect to VoIP Gateway via console port to enter IP address in order to set Interface 0/0
2. Input IP address of interface 0/0 using below command
3. Connect to the IP address via web after saving

```
Router> enable                                           => Enter the enable mode
Router# configure terminal                               => Enter the configuration mode
Router(config)# interface FastEthernet 0/0
Router(config-if)# ip address <IP Address> <Subnet Mask>   => Ex) ip address 172.17.109.1 255.255.0.0
Router(config-if)# exit
Router(config)# ip route 0.0.0.0 0.0.0.0 <Defaullt Gateway>  => Ex) ip route 0.0.0.0 0.0.0.0 172.17.1.1
Router(config)# write                                    => saving
Proceed with write? [confirm]y
```

WEB Connection

2. Connection Web via LAN 1 Port



LAN 1 Default IP address
IP Address : 192.168.10.1
Subnet mask : 255.255.255.0

1. It is the way to connect to VoIP Gateway via LAN 1 port

2. The factory default of LAN 1 port
   - IP Address : 192.168.10.1
   - Subnet mask : 255.255.255.0

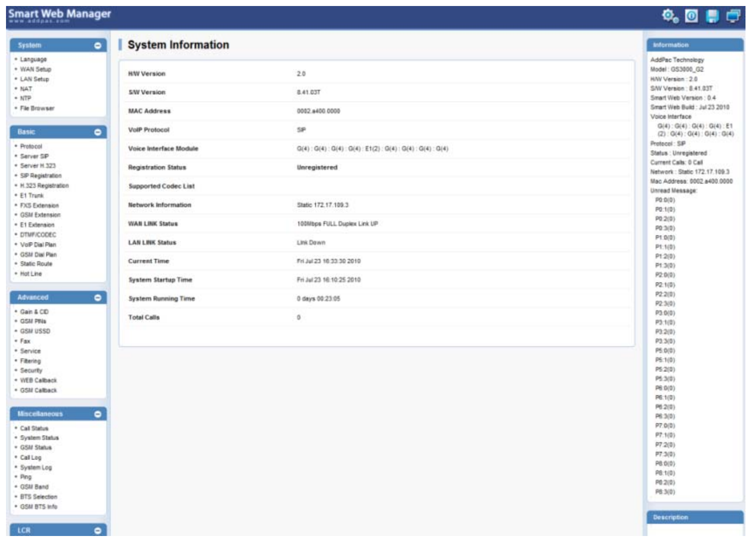
3. After set PC with same IP address subnet, connect to VoIP Gateway
   - Connect PC to VoIP Gateway using Cross UTP-Cable. You may use Ethernet switch with normal UTP-cable
   - Enter IP address 192.168.10.1 on your web browser

WEB Connection

The screen of Web connection

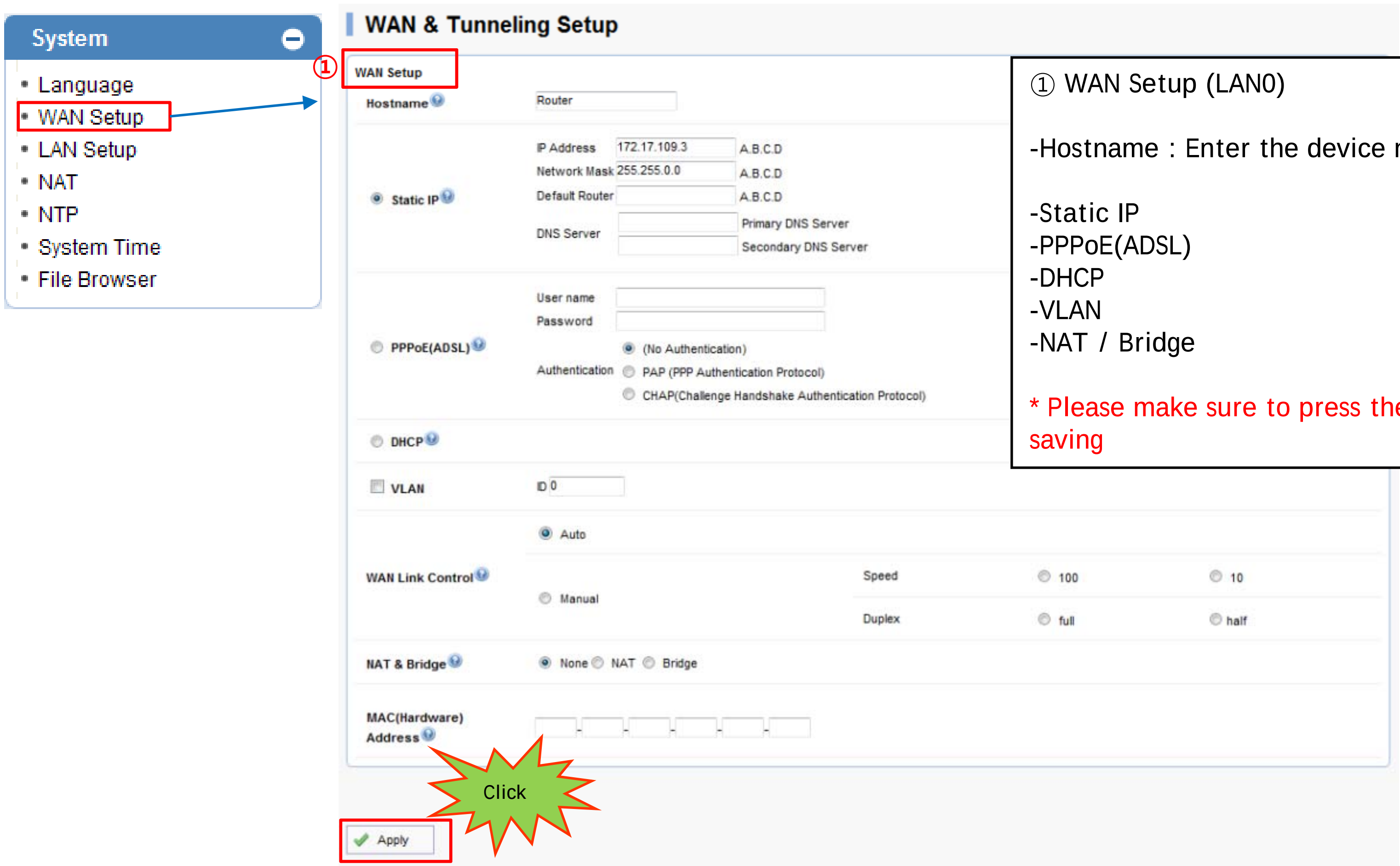
• The Shown log-in screen is connection to Web page.
Please enter the below log-in information

ID : root
Password : router



Network Setup

1. WAN Setup Screen -1



① WAN Setup (LAN0)

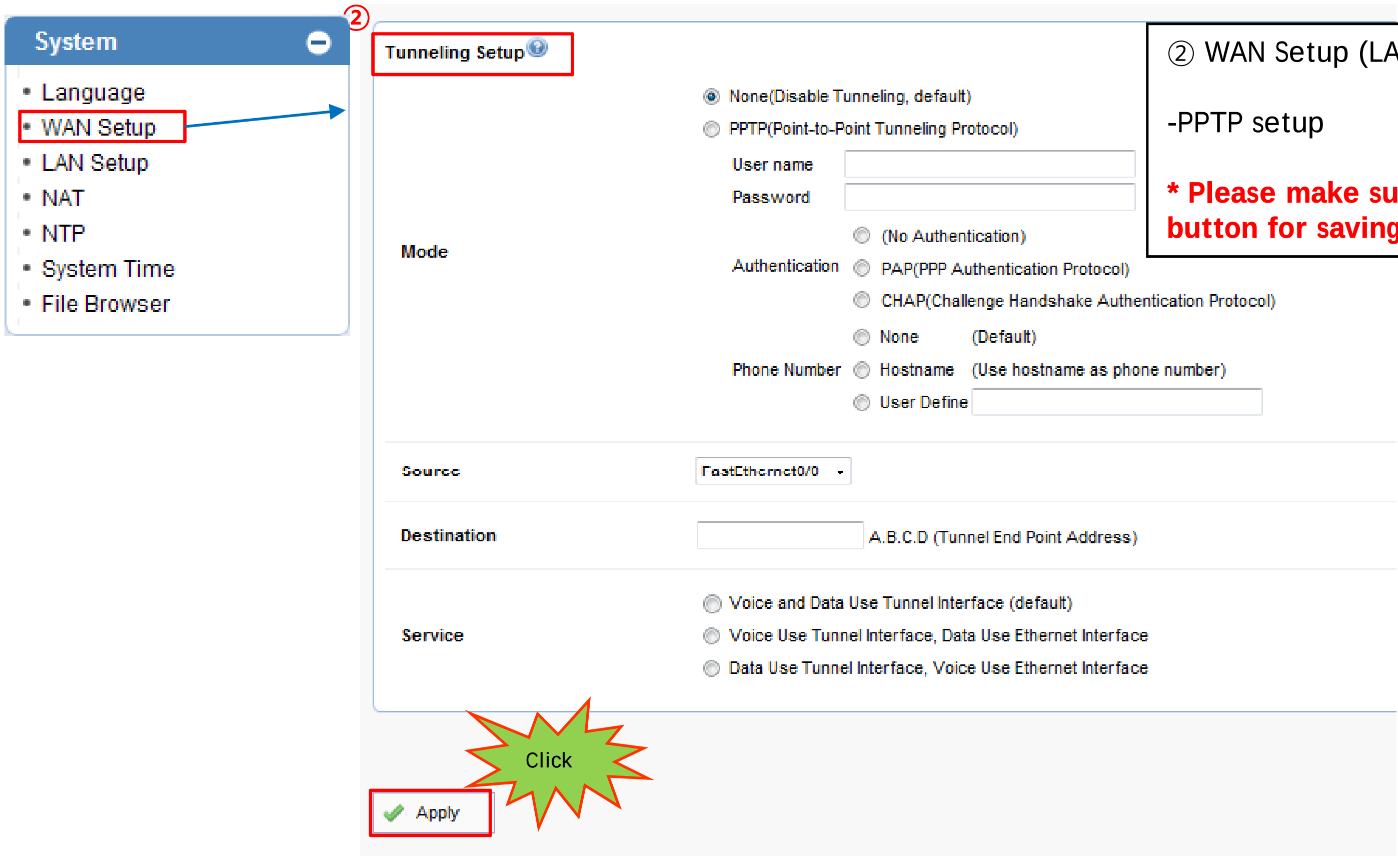
-Hostname : Enter the device name of Gateway

-Static IP
-PPPoE(ADSL)
-DHCP
-VLAN
-NAT / Bridge

* Please make sure to press the apply button for saving

Network Setup

1. WAN Setup Screen -2



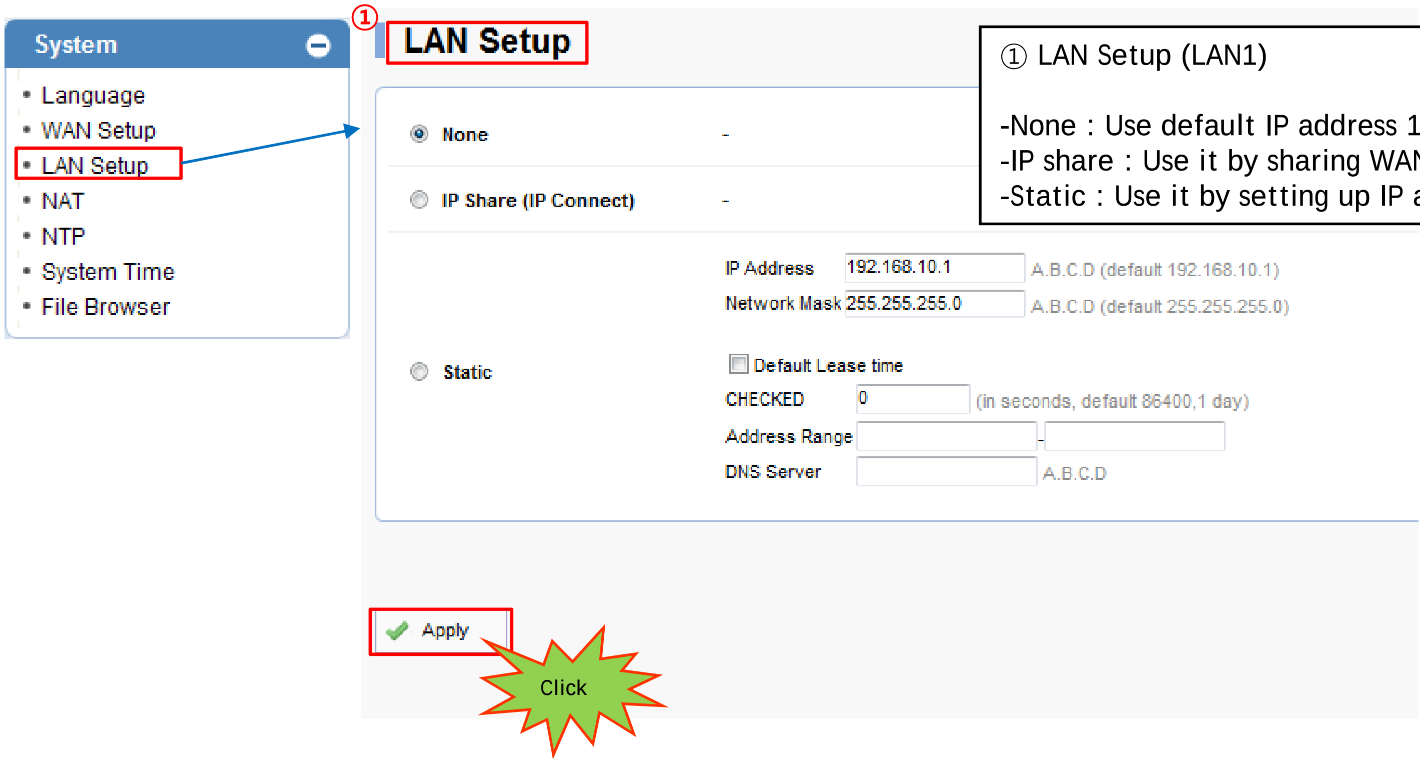
② WAN Setup (LAN0)

-PPTP setup

*** Please make sure to press the apply button for saving**

Network Setup

2. LAN Setup Screen



① LAN Setup (LAN1)

-None : Use default IP address 192.168.10.1.
-IP share : Use it by sharing WAN IP.
-Static : Use it by setting up IP address.

Port Setup

1. FXS Port Setup
2. FXO Port Setup
3. E1 / T1 Setup
4. Hot Line

Port Setup

AddPac Digit Structure

※Digit Structure※

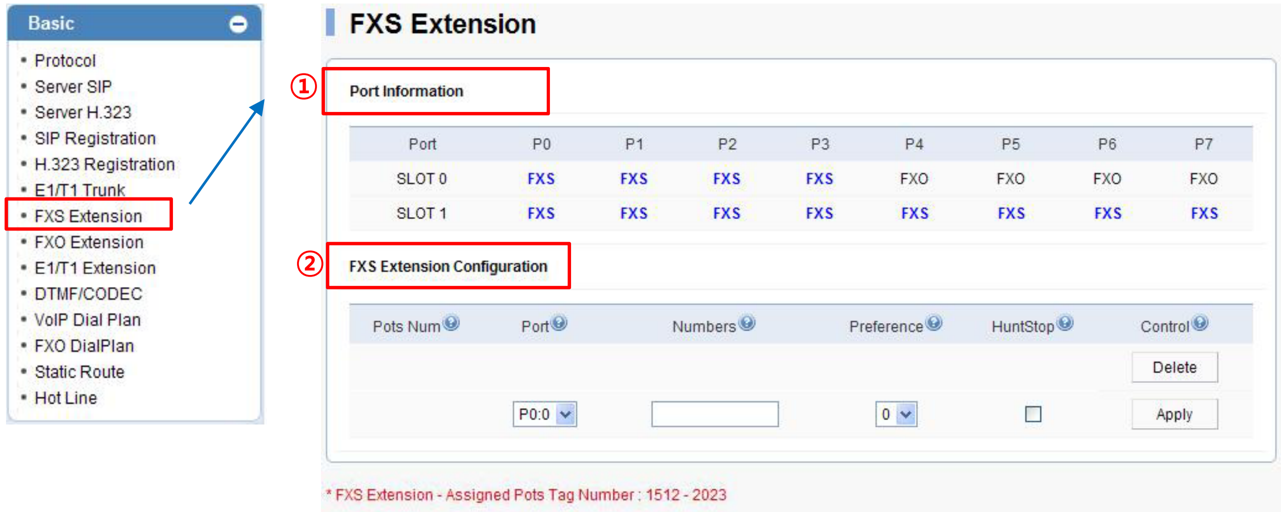
- 9T : All number started with 9 as the first digit
- 4.. : Three digit number started with 4 as the first digit
- [2-9]T : All number started with 2 to 9 as the first digit
- 00[127]T : All number started with 001, 002, 007 as the first digit

** T : Accept all number entered within Inter Digit Time (Default IDT : 3sec)
** Dot(.) : One dot(.) means one digit
** [ ] : The range of number

※Rule tranfer※
- Digit pattern : 025683848 / Digits to insert : 82 / Number of digits to delete : 1 → 8225683848
- Digit pattern : 00[127]T / Digits to insert : 123 / Number of digits to delete : 2 → 123[127]T
- Digit pattern : [2-9]4... / Digits to insert : 823848 / Number of digits to delete : 3 → 823848..

Port Setup > FXS Port

1. FXS Port - FXS Extension

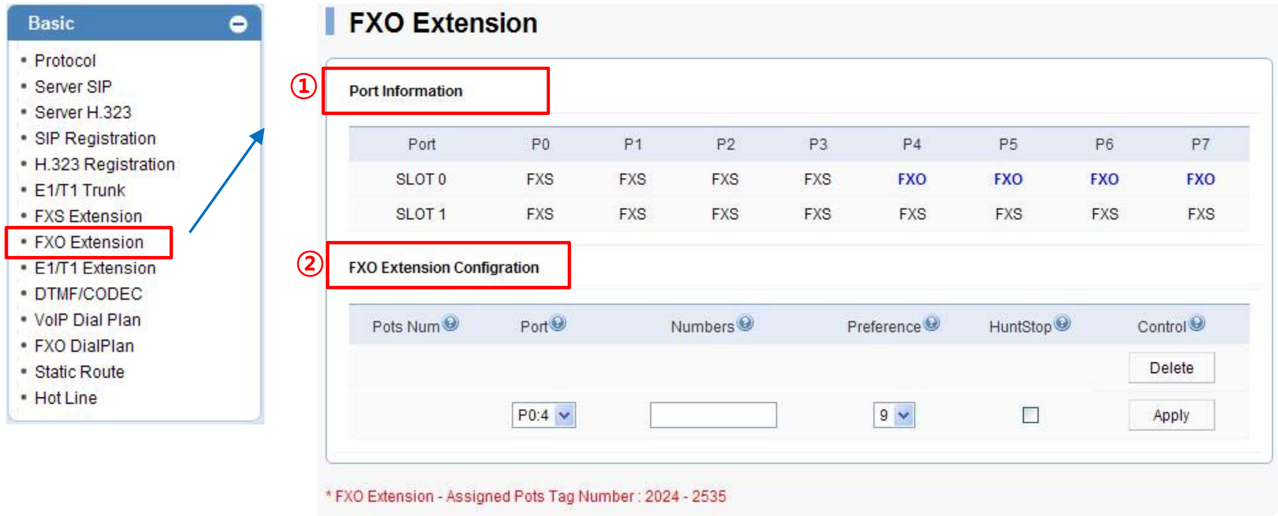


① The each port information of Gateway

② FXS Extension : Set the number of phone on FXS port
- Index : Enter number in order. Please make sure not to be duplicated
- Port : Select FXS port to be set
- Numbers : Enter FXS number
- Preference : Set priority for each number.
  If there is the same number at two ports, a port is selected by this priority
- Hunt Stop : It is a function of forward a call to other party in case of unavailable receiving call.
  Activation of this function is recommended

Port Setup > FXO Port

2. FXO Port - FXO Extension

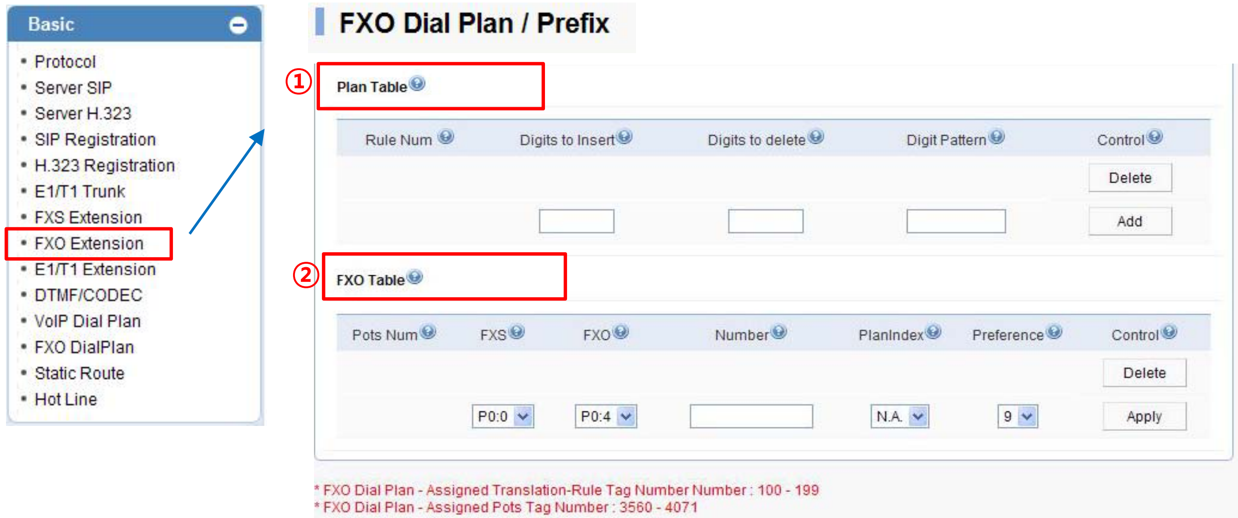


① The each port information of VoIP Gateway

② FXO Extension : Set the number of FXO port
- Index : Enter number in order. Please make sure not to be duplicated
- Port : Select FXO port to be set
- Numbers : Enter FXO number
- Preference : Set priority for each number.
  If there is the same number at two ports, a port is selected by this priority
- Hunt Stop : It is a function of forward a call to other party in case of unavailable receiving call.
  Activation of this function is recommended

Port Setup > FXO Port

2. FXO Port - FXO Dial Plan

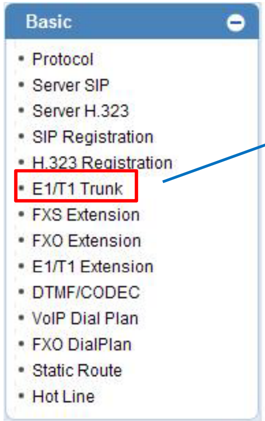


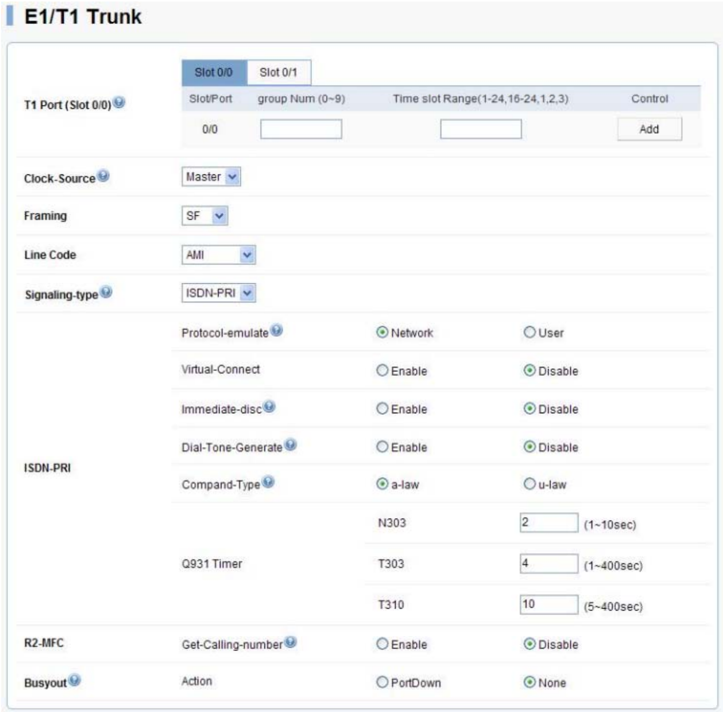
① Plan Table : use when number transformation is necessary.
- Digits to Insert : Number you want to enter
- Number of Digit to Delete : Number of digit to delete
- Digit Pattern : Number to apply for conversion

② FXO Table : use when setup backup dispatch with FXO port.
- FXS : specify FXS port for backup dispatch.
- FXO : specify FXO port for backup dispatch.
- Number : specify digit pattern for backup dispatch.
- Plan Index : specify relevant number when applying plan table.
- Preference : specify the order of priority for backup dispatch.

Port Setup > E1/T1

3. E1/T1 Port - E1/T1 Trunk -1

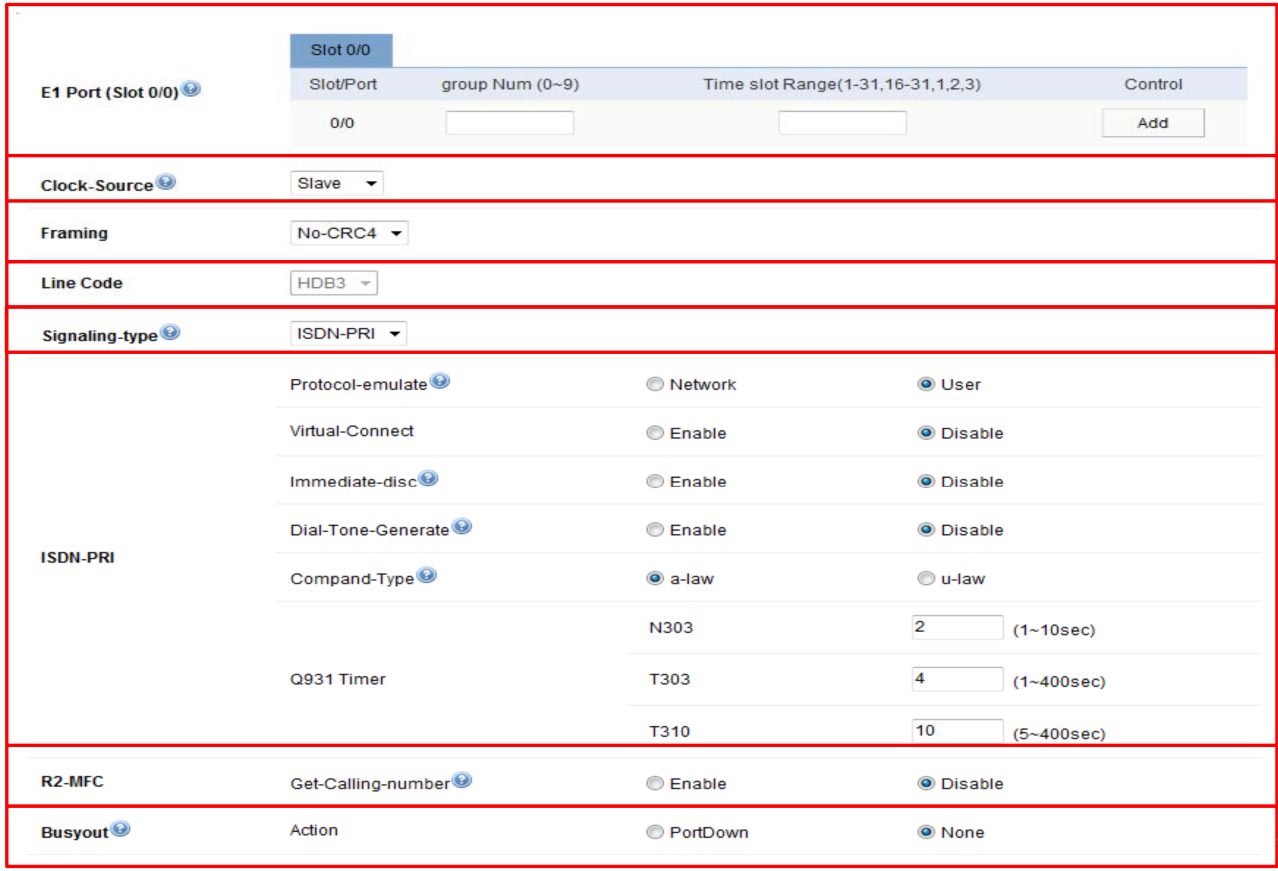




E1 / T1 Trunk : Be sure to setup first before using E1 or T1 (default is E1)
If E1/T1 Trunk is not setup, E1/T1 Extension and Registration cannot be setup.

Port Setup > E1/T1

3. E1/T1 Port - E1/T1 Trunk -2 (E1)



Group Setup for E1 Channel.
Be sure to setup before using E1.

Clock Setup : Master / Slave
Be sure to setup in pair with other device

E1 Framing : CRC4 / No CRC4

E1 Line Code : HDB3

E1 Signaling : ISDN-PRI, R2-MFC, R2-DTMF
(Default : ISDN-PRI)

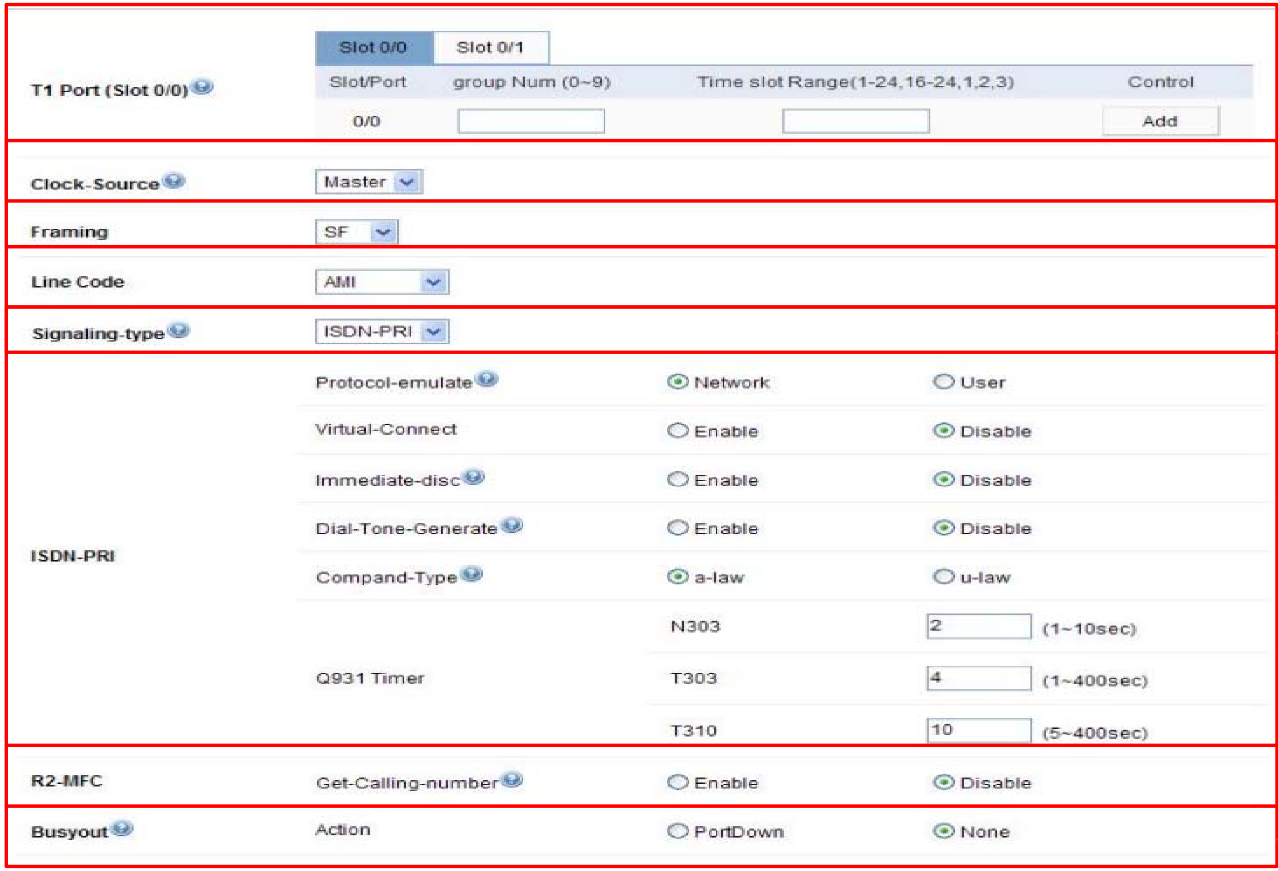
E1 PRI Setup
- Protocol-emulate : Be sure to setup in pair with other device (Network / User)
- Virtual-connect
- Immediate-disc
- Dial-Tone-Generate
- Compand-Type
- Q931 Timer

E1 R2 Setup
Get-Calling-Number : Calling number enable / disable

Setup for port down when the E1 port error occurs

Port Setup > E1/T1

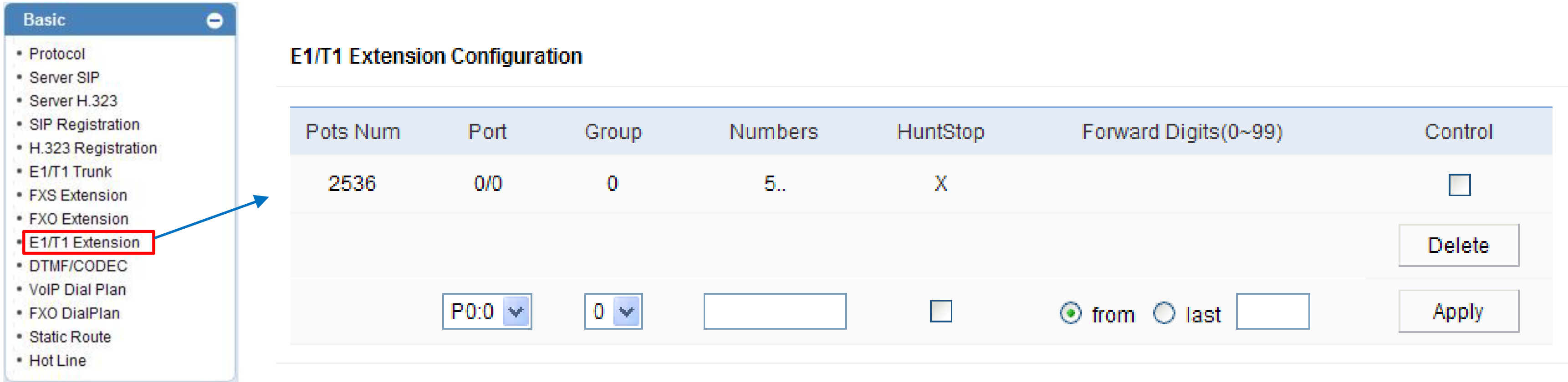
3. E1/T1 Port - E1/T1 Trunk -3 (T1)



- Group Setup for T1 Channel.
  Be sure to setup before using T1.
- Clock Setup : Master / Slave
  Be sure to setup in pair with other device
- T1 Framing : SF / ESF
- T1 Line Code : AMI / B8ZS
- T1 Signaling : ISDN-PRI, R2-MFC, R2-DTMF
  (Default : ISDN-PRI)
- T1 PRI Setup
  - Protocol-emulate : Be sure to setup in pair with other device (Network / User)
  - Virtual-connect
  - Immediate-disc
  - Dial-Tone-Generate
  - Compand-Type
  - Q931 Timer
- T1 R2 Setup
  Get-Calling-Number : Calling number enable / disable
- Setup for port down when the T1 port error occurs

Port Setup > E1/T1

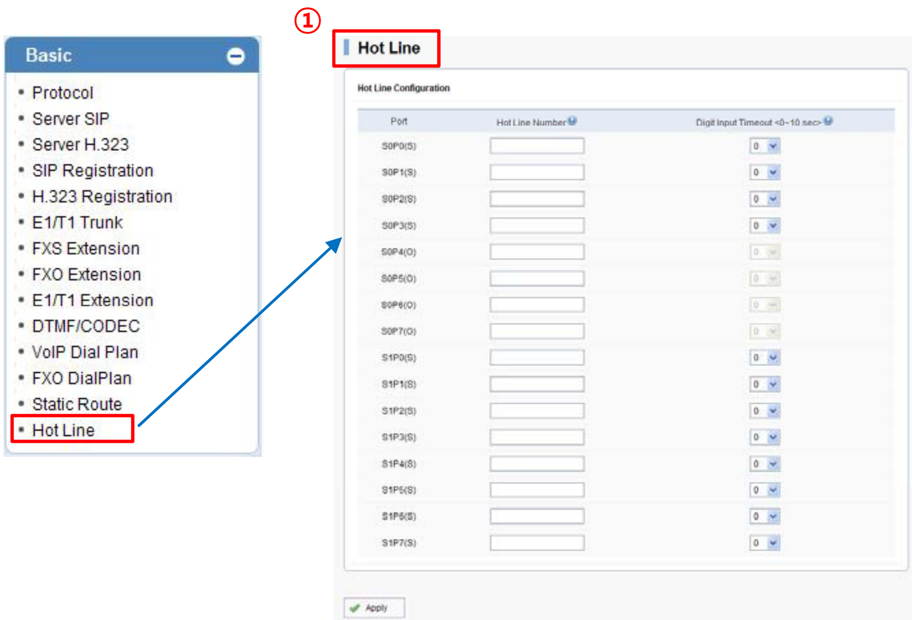
3. E1/T1 Port - E1/T1 Extension



① E1/T1 Extension : Configuration for incoming call with E1/T1
- Port : It means E1/T1 slot and port
- Group : It means group number which was setup in E1/T1 Trunk.
- Numbers : Setup the incoming number of E1.
- Hunt Stop : It is a function of forward a call to other party in case of unavailable receiving call. Activation of this function is recommended
- Forward Digits : Specify the digits for E1/T1 PABX.
  from : Send the number to E1/T1 PABX from the number you specified in front.
  last : Send the number to E1/T1 PABX from the number you specified at the back.

Port Setup > Hot Line

4. Hot Line



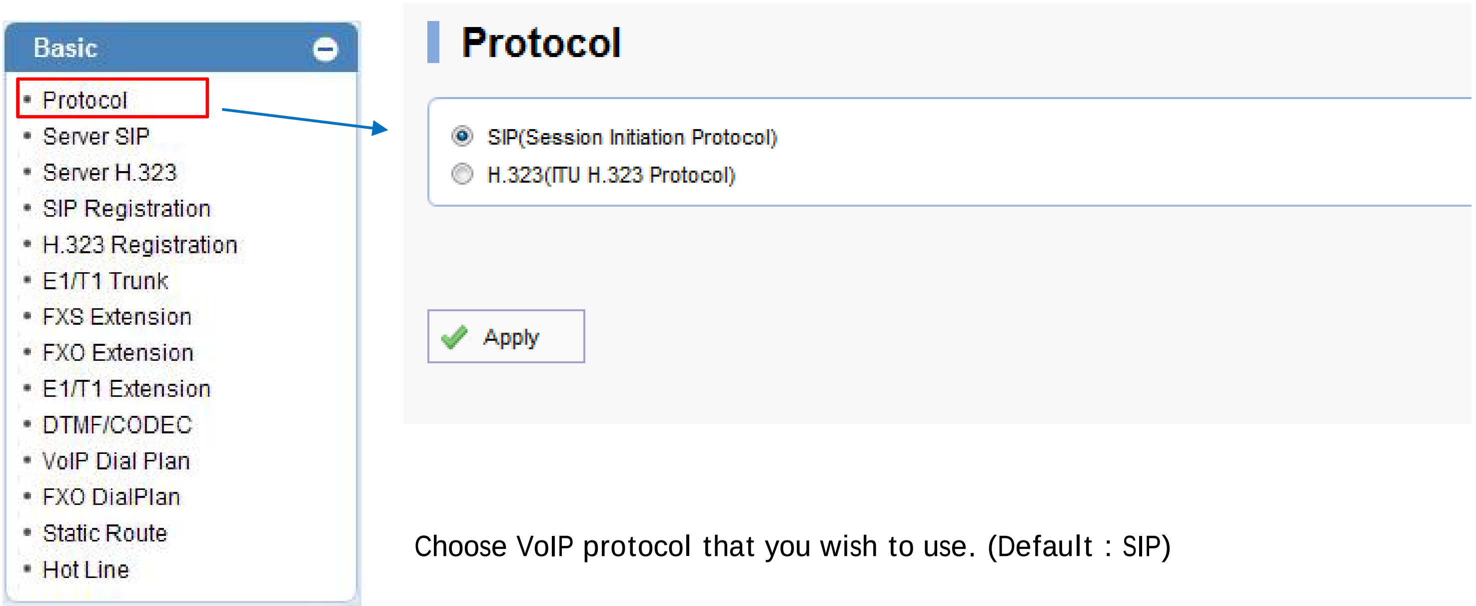
① Hot Line Configuration : Connect incoming and outgoing call directly

- Port : It means FXS and FXO port
- Hot Line Number : Forward call to entered number
  It connects to the number of FXO port in case of receiving call (Direct Incoming call)
- Digit Input Timeout : Time to make call to the Hot Line Number when user doesn't any action after off-hook
  FXS port is used only.

VoIP Setup

1. Server SIP
2. SIP Registration
3. Server H.323
4. H.323 Registration
5. DTMF/CODEC
6. VoIP Dial Plan
7. Static Route

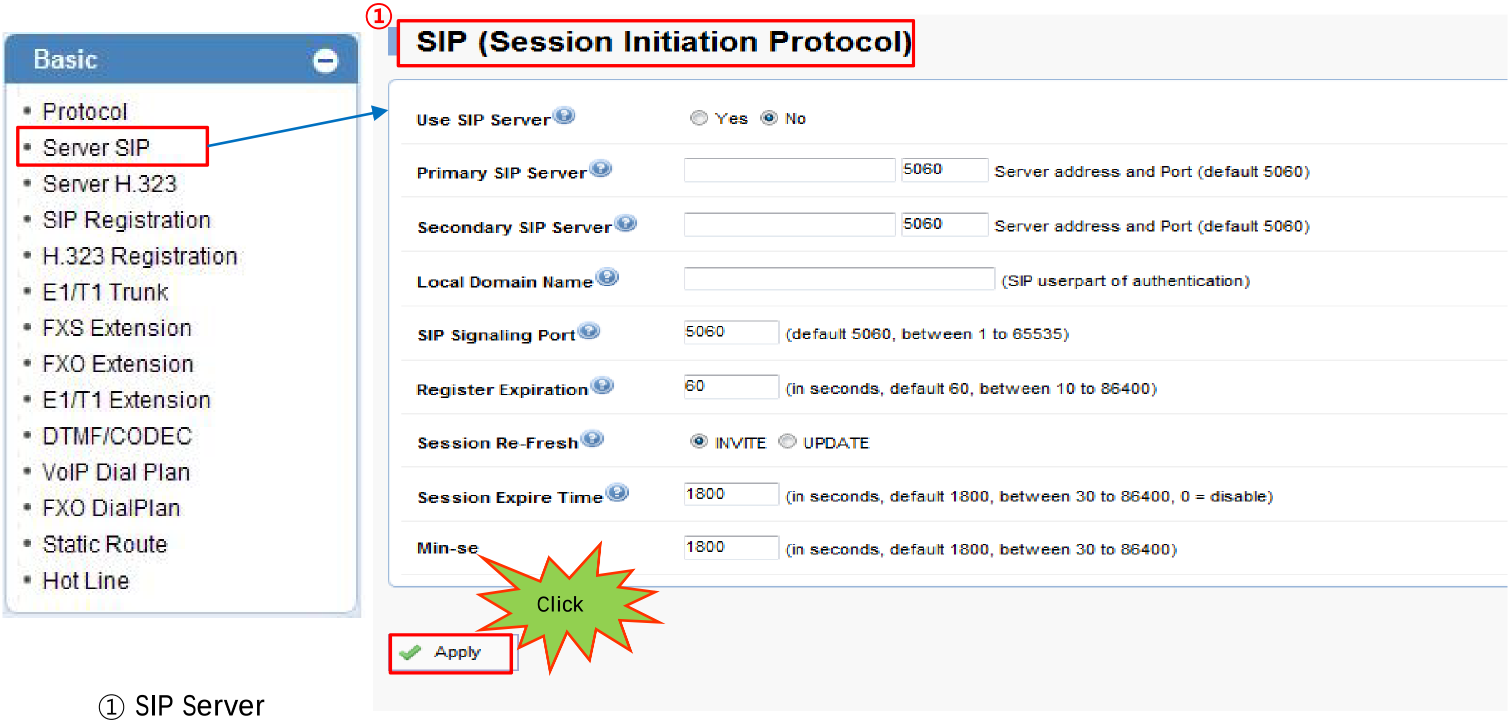
VoIP Setup > Protocol Selection



Choose VoIP protocol that you wish to use. (Default : SIP)

VoIP Setup > SIP

1. Server SIP -1



① SIP Server

-Use SIP Server : Select using SIP Server. Please click "Yes" to use SIP server

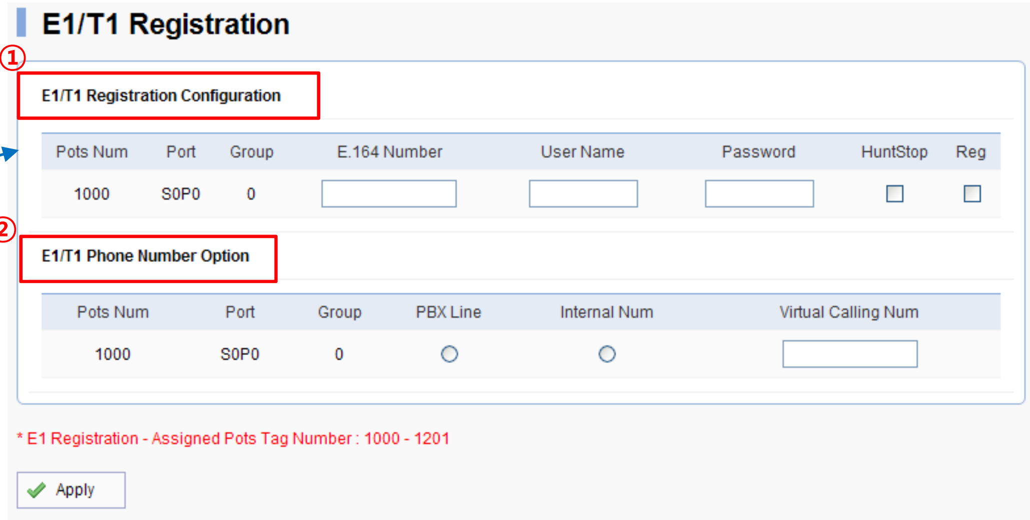
-Primary SIP server : Enter IP address of Primary SIP server

-Secondary SIP Server : Enter IP address of Secondary SIP server.
The secondary server is activated when Primary SIP Server is not available

-Local Domain name : Enter local domain when it is required on server authentication

-Default setting is recommended for other field

2. SIP Registration -1

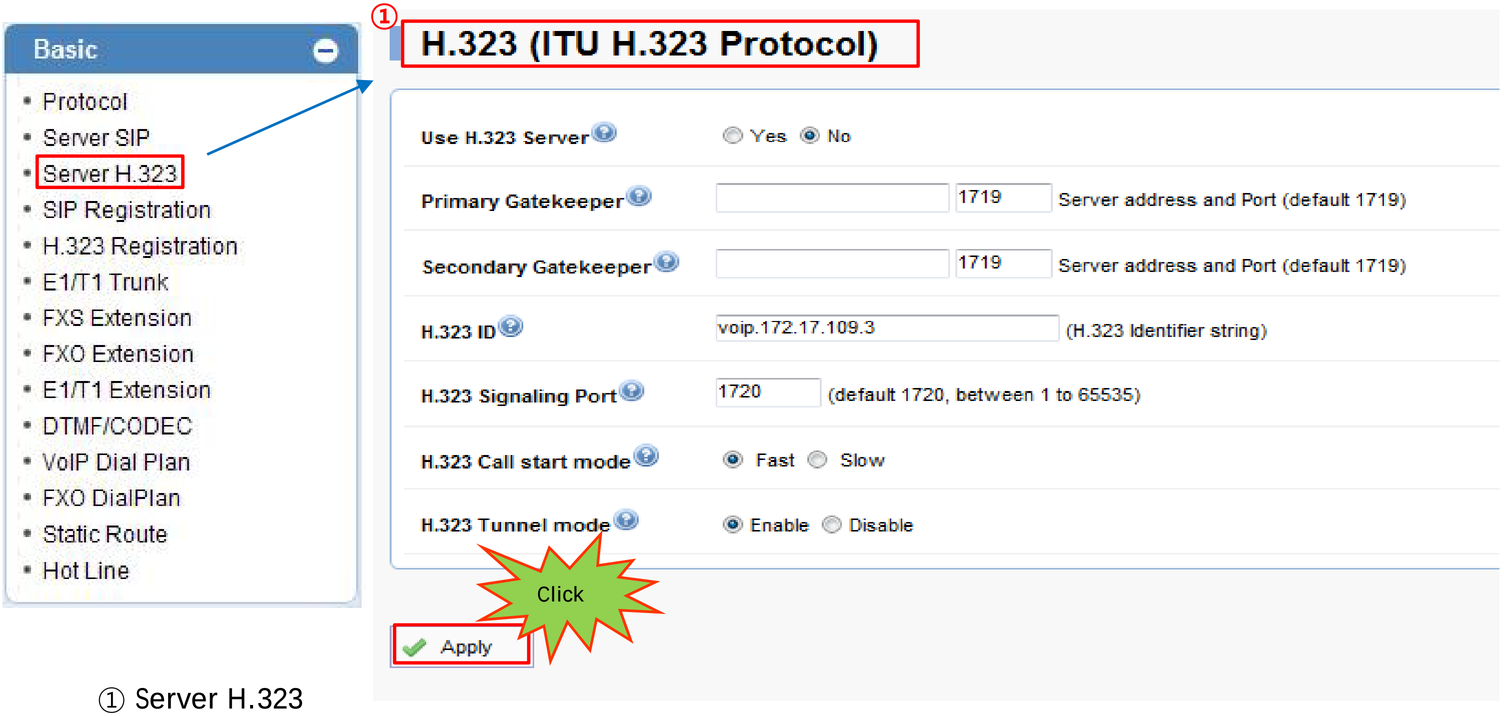


① SIP Registration Configuration
- Group : It means group number which was setup in E1 Trunk. You may choose it when you create several Group.
-E.164 Number : Enter SIP authentication number
-User Name : Enter authentication ID
-Password : Enter authentication Password .
-Reg : Checking this field is required to get authentication from SIP Server
-Hunt Stop : Forward call to other party when port is unavailable. It is recommended to use it.

② E1/T1 Phone Number Option
-PBX Line : use number that is used in E1 PABX.
-Internal Num : use when using virtual calling number.
-Virtual Calling Num : enter virtual calling number.

VoIP Setup > H.323

3. Server H.323 -1



① Server H.323

-Use H.323 Server : Select using H.323. Please click “Yes” to use H.323 server

-Primary Gatekeeper server : Enter IP address of Primary Gatekeeper server

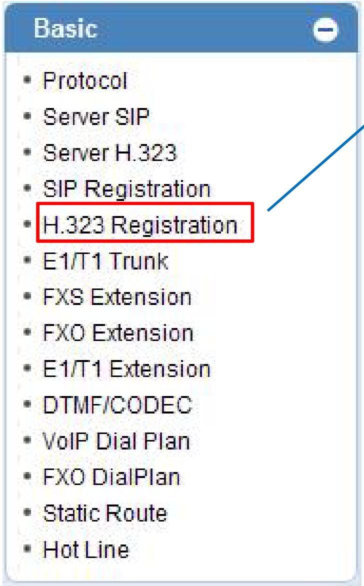
-Secondary Gatekeeper Server : Enter IP address of Secondary Gatekeeper server.
The secondary server is activated when Primary Gatekeeper Server is not available

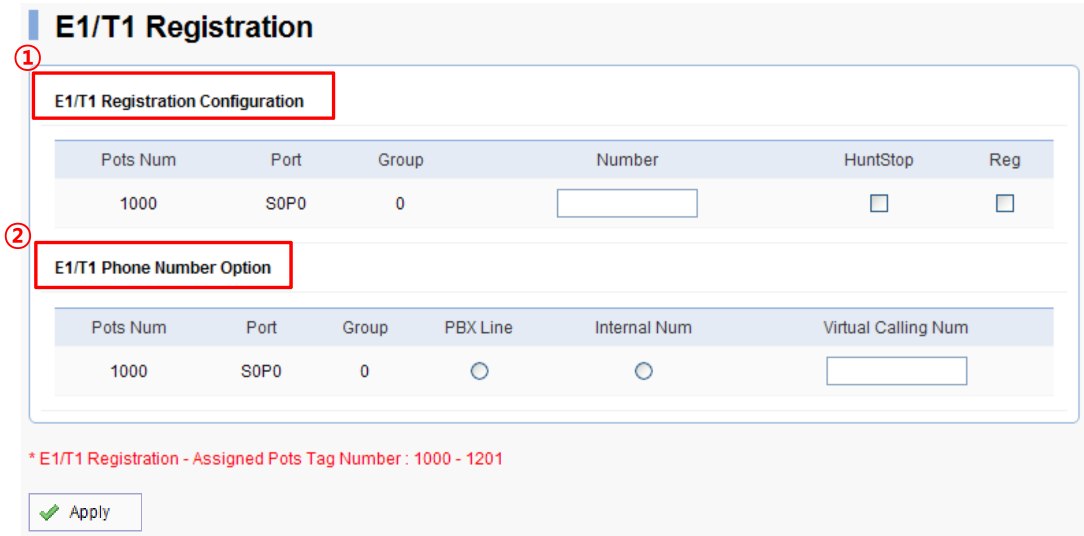
-H.323 ID : Please input Authentication ID to register Gatekeeper

-Default setting is recommended for other field

VoIP Setup > H.323

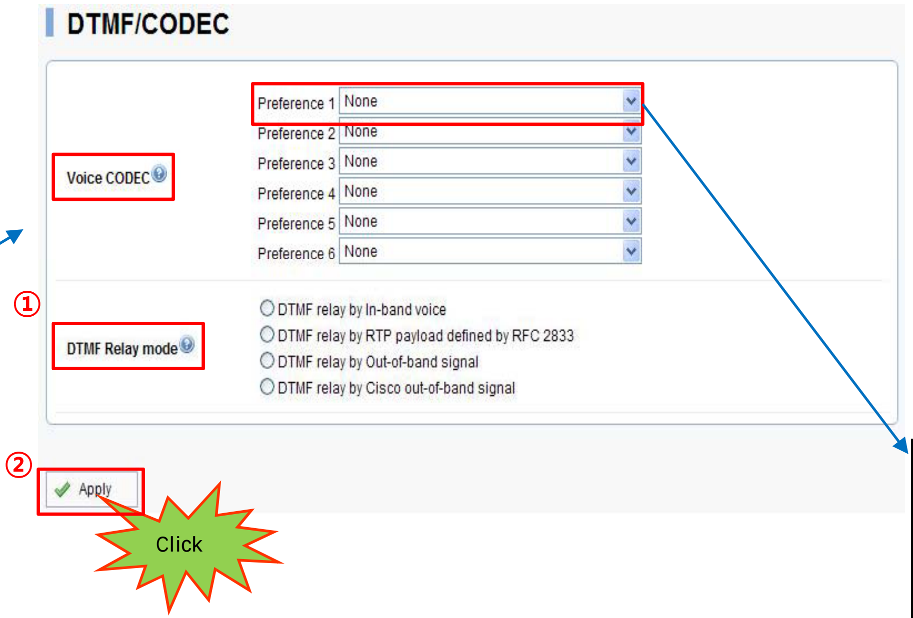
3. H.323 Registration -1

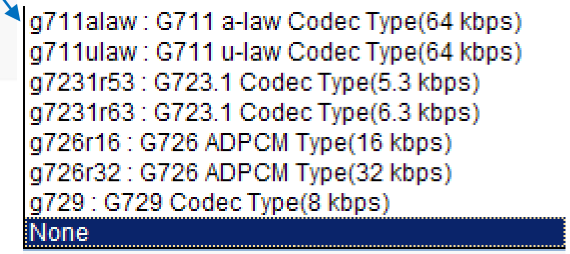




① H.323 Registration Configuration
-Number : Enter H.323 authentication number
-Hunt Stop : Forward call to other party when port is unavailable. It is recommended to use it.
-Reg : Checking this field is required to get authentication from Gatekeeper.
② E1/T1 Phone Number Option
-PBX Line : use the number for E1 PABX.
-Internal Num : use it for virtual calling number.
-Virtual Calling Num : enter virtual calling number.

4. DTMF/CODEC





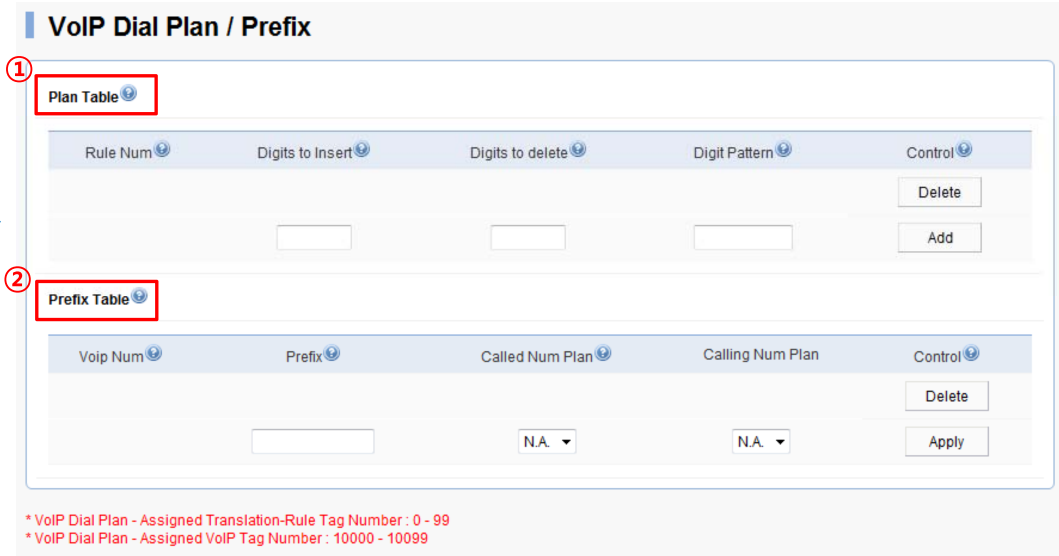
① Video Codec : Select voice codec to be used.

② DTMF Relay mode : Select DTMF relay mode.
(Default : Out-of-band Signal)

* Please press the apply button to save

VoIP Setup > VoIP Dial plan

6. VoIP Dial Plan -1

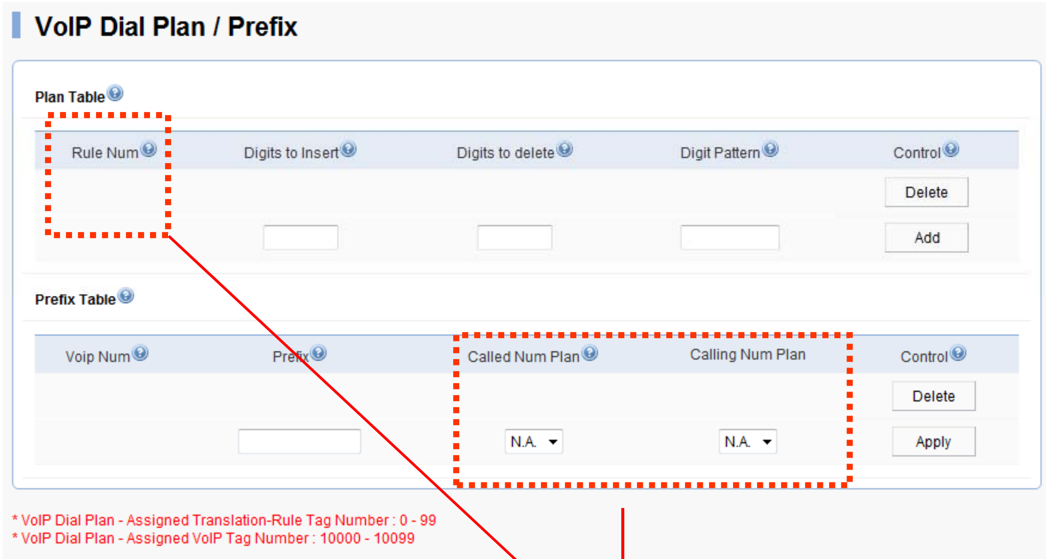


① Plan Table
- Digits to Insert : Number you want to enter
- Number of Digit to Delete : Number of digit to delete
- Digit Pattern : Number to apply for conversion

② Prefix Table
- Prefix : Number to make VoIP call
- Plan Index : Make the same number with Plan table

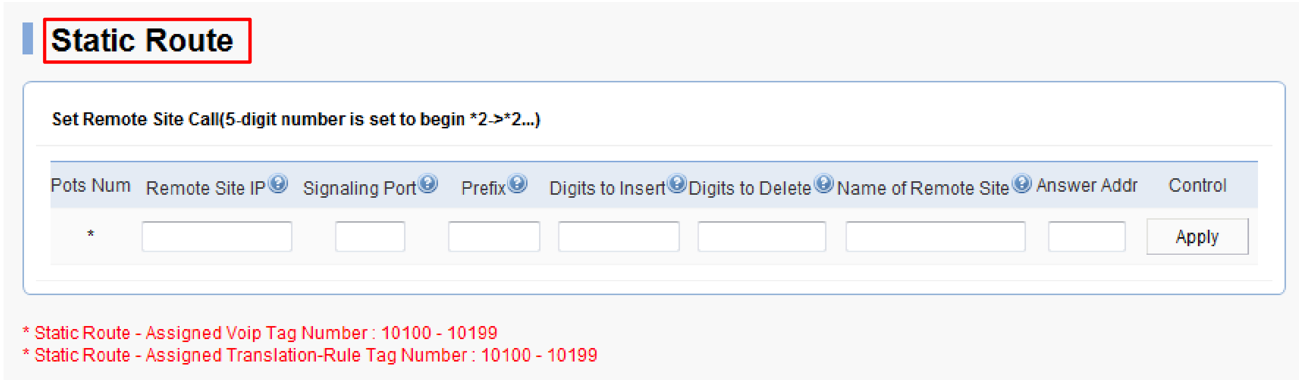
VoIP Setup > VoIP Dial Plan

6. VoIP Dial Plan -2 (Ex)



It must be the same with Plan Index Number

7. Static Route -1



① Static Route : : User can forward call to other party after enter IP address of them.
It can be done without SIP Server or other system

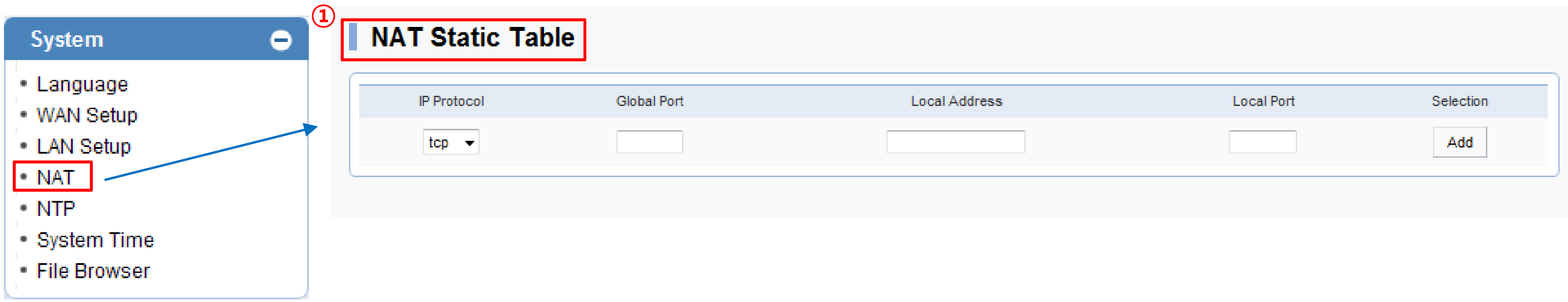
-Remote Site IP : Enter IP address of other party device
-Prefix : Enter number of other party
-Insert Digit : Enter number of digit to add
-Delete Digit : Enter number of digit to delete
-Name of Remote Site : Enter name of other party'

Additional Service

1. NAT Static Table
2. NTP
3. System Time
4. File Browser
5. Gain & CID
6. FAX
7. Service
8. Filtering
9. Security

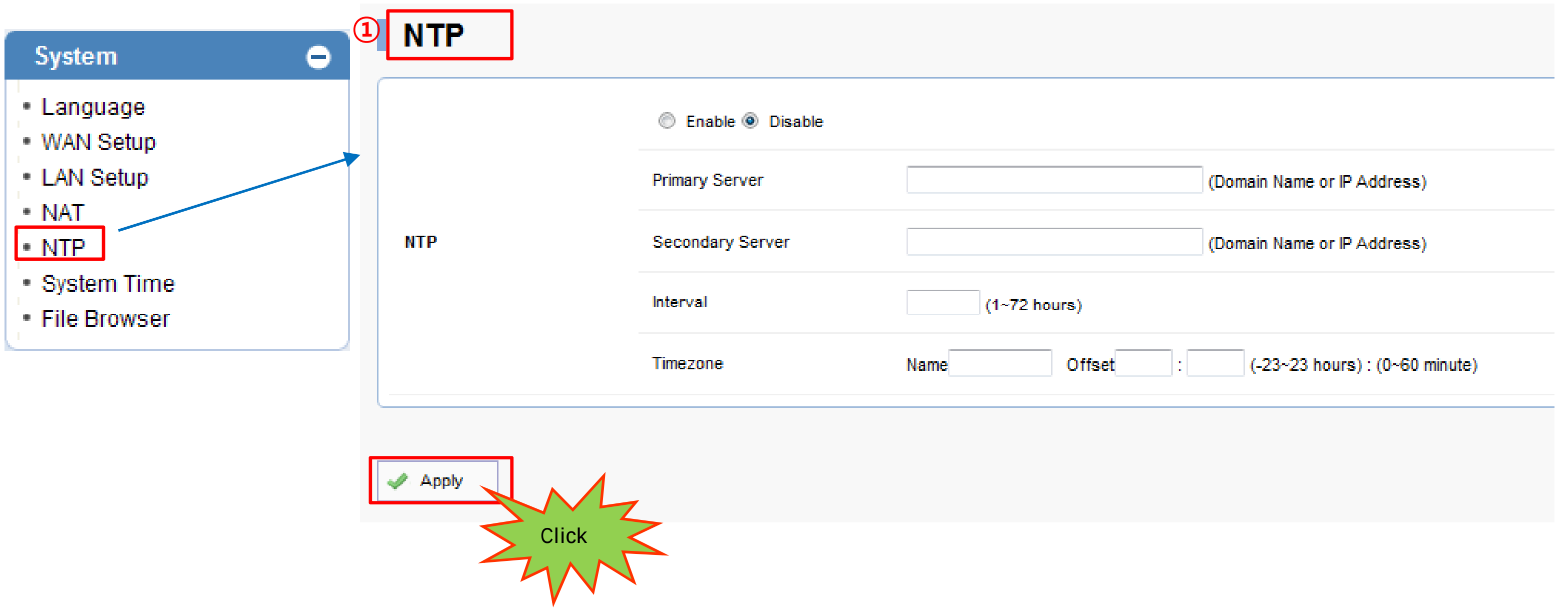
Additional Service

1. NAT Static Table



① NAT Static Table
Use when port forwarding is necessary for special IP address.

2. NTP



① NTP : Input information of NTP Server
- Click the apply button for NTP activation
- Primary Server : Input IP or domain name of NTP Server
- Interval : Interval to request and receive data from NTP server

* Please click the apply button after set up

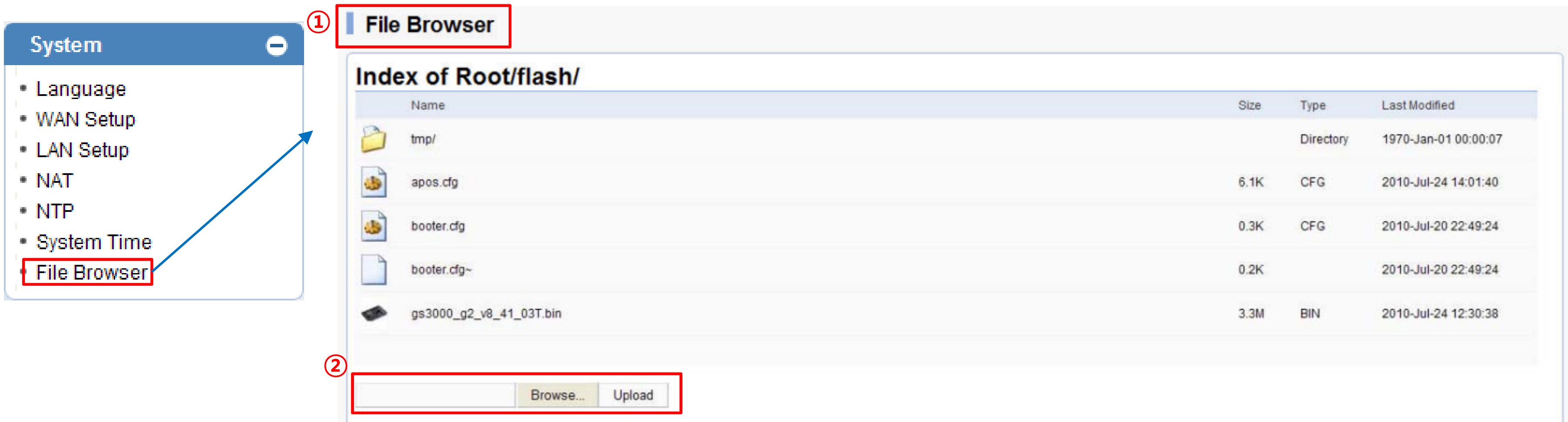
Additional Service

3. System Time

① You may setup the device local time.

Additional Service

4. File Browser

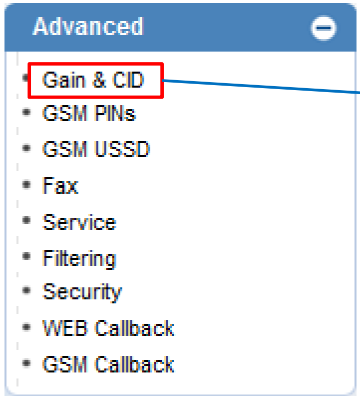


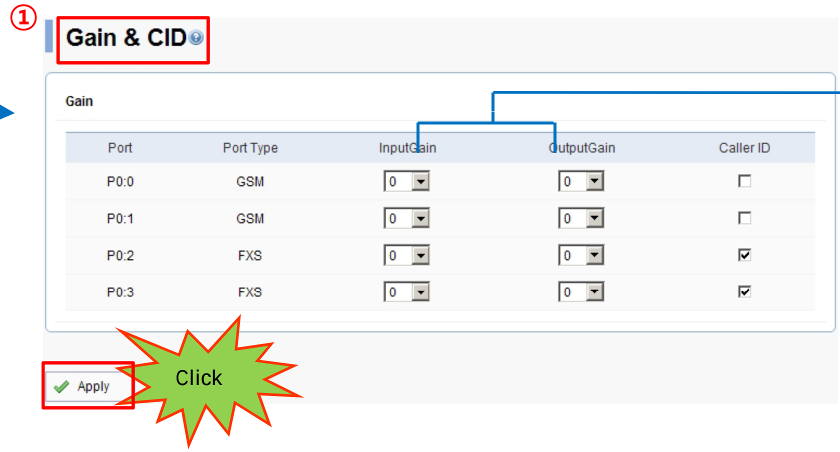
① File Browser
  - You may view firmware and configuration file in Gateway.

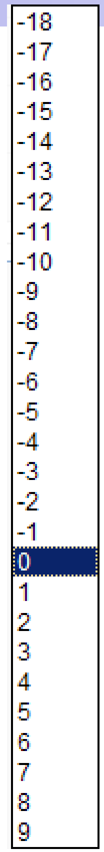
② Firmware Upload
  - Use when uploading Gateway firmware.
  - Process upload by finding PC related firmware.
  - Uploaded firmware will be applied after rebooting.

Additional Service

5. Gain & CID







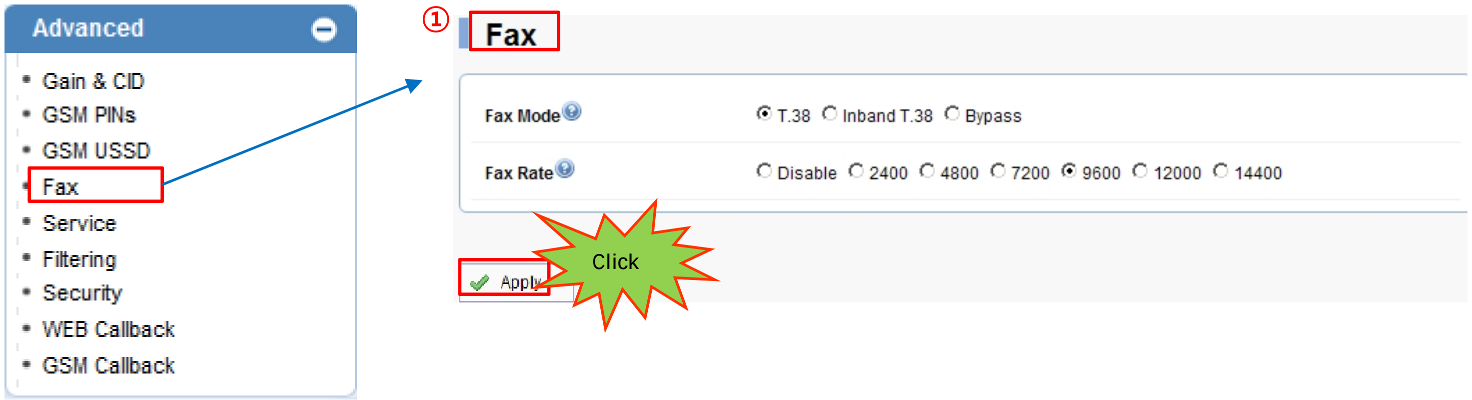
① Gain & CID : Adjustment output voice level of each port(FXS / FXO)
(You may reduce the level when echo and noise occurred)
In addition, call number can be detected by Caller-ID

- Input Gain : Please adjust input gain when sending call is too loud or too low
- Output Gain : Please adjust output gain when receiving call is too loud or too low
- Caller-ID : It is a function to display number of callers

* Please click the apply button after set up

Additional Service

6. FAX



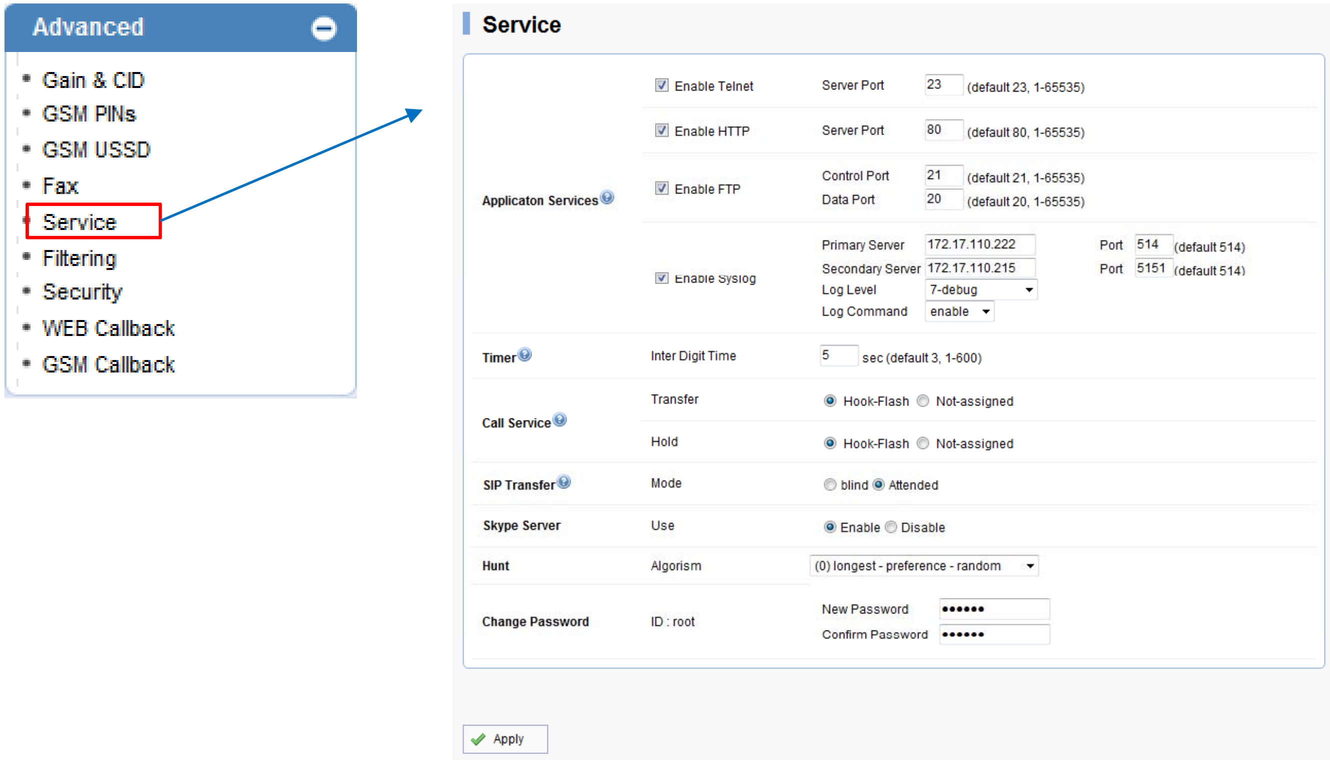
① Fax : Setting the property of FAX mode

- Fax Mode :
  - T.38 : FAX signal is being sent by T.38 packet with new session opening. In case of using T.38, FAX Rate is needed to be set
  - Bypass : FAX signal is being sent by RTP. FAX Rate setting is not required
- Fax Rate : Setting FAX transmit rate. Default is 9600bps and the range is from 2400bps to 14400bps

* Please click the apply button after set up

Additional Service

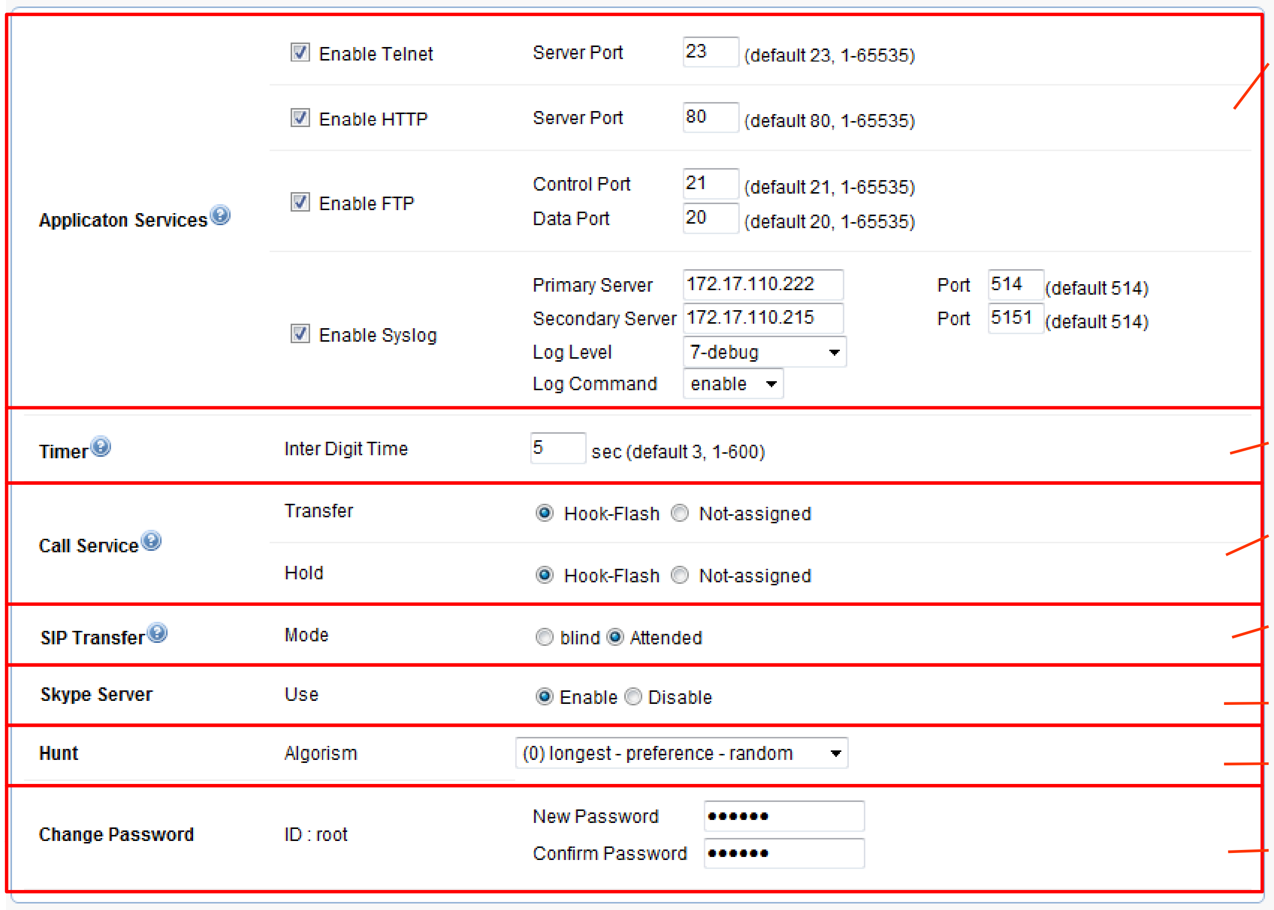
7. Service -1



Please refer to next page about description of function.

Additional Service

7. Service -2



Application Service Use
: Telnet / HTTP / FTP / Syslog

PDD Time (default : 3 sec)

Call Hold / Transfer Use
: Enable / Disable

SIP Call transfer mode : Blind / Attended

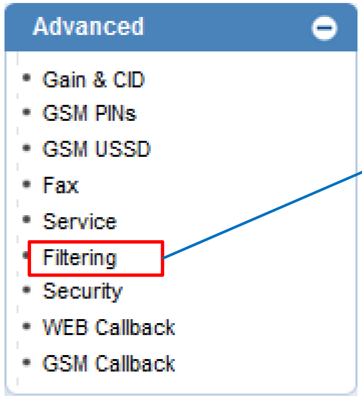
Skype SIP Sever Use : enable / disable

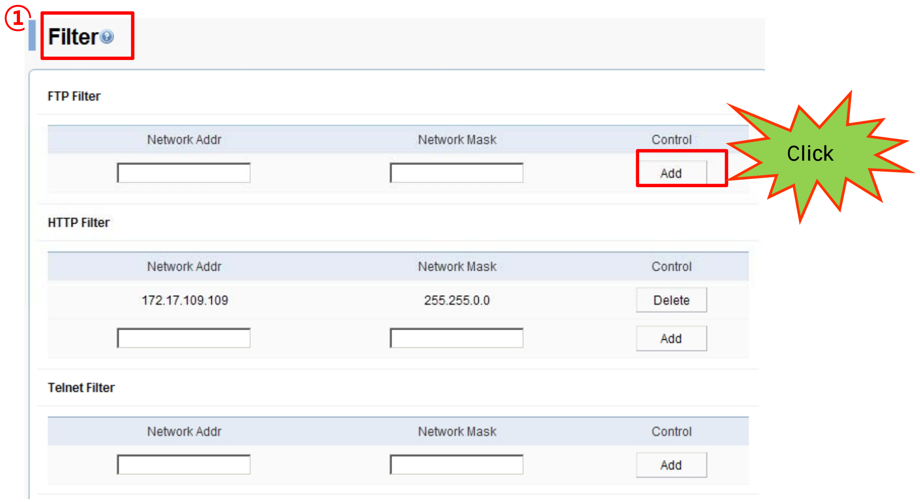
Call Routing Algorism

Password change regard of WEB / Telnet access

Additional Service

8. Filtering



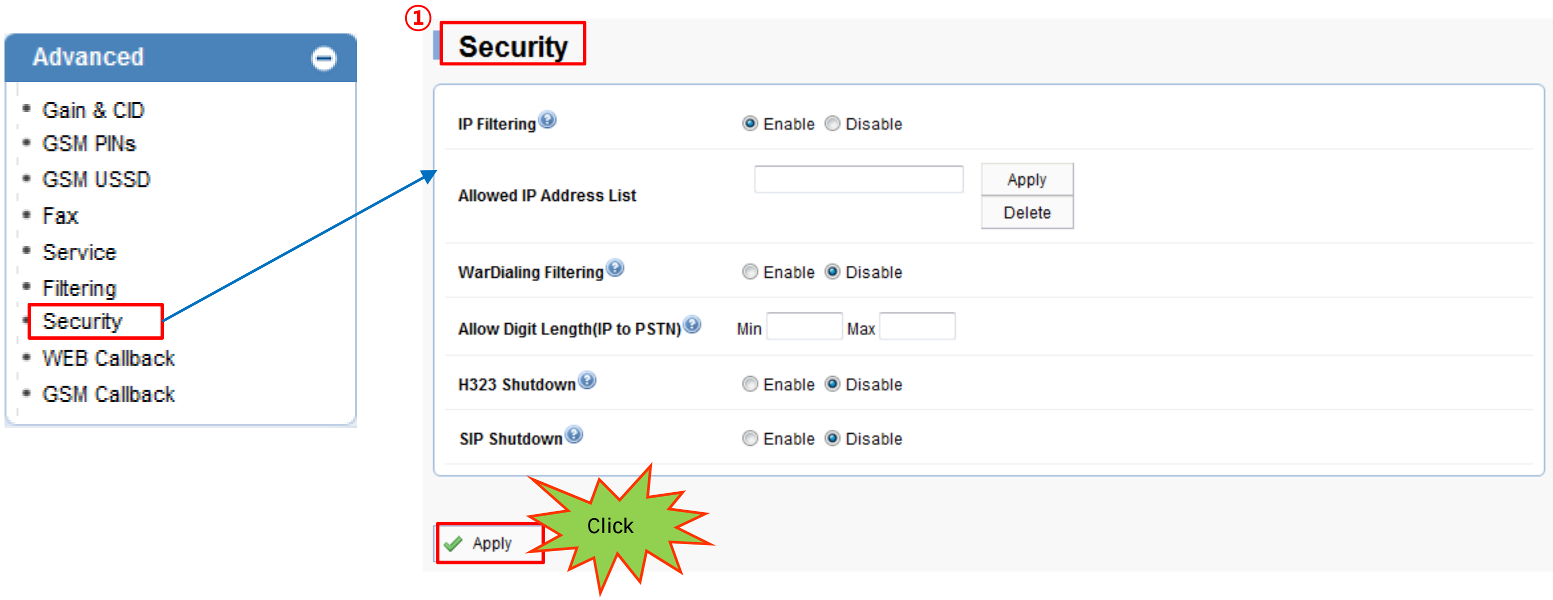


① Filter : Setting IP address authorized by administrator for connection
- FTP Filter : The only device with the IP address authorized by administrator can access FTP connection
- HTTP Filter : The only device with the IP address authorized by administrator can access WEB connection
- Telnet Filter : The only device with the IP address authorized by administrator can access Telnet connection

* Please click the apply button after set up

Additional Service

9. Security



① Security : Set security to block unauthorized call
- IP Filtering : The only call made from the device with IP address listed on Gateway is available to make call
-Allowed IP address list : Allow VoIP call for added IP address.
- War Dialing Filtering : The only receiving call listed on dial plan is available to make call
- Allow digit Length(IP to PSTN) : The only receiving call within range of set number is available to make call
- SIP Shutdown : Set using SIP Signaling. It must be enabled with SIP communication

* Please click the apply button after set up

Monitoring

1. Call Status
2. System Status
3. Call Log / System Log
4. Test Call
5. Ping

Monitoring

1. Call Status

Miscellaneous

- Port & Call Status
- System Status
- Call Log
- System Log
- Test Call
- Ping

E1

Port Status (E1)

Port#		Channel Group																															Control
SLOT 4/0	Channel	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	Unblock
	Status	I	I	I	I	I	I	I	I	I	I	I	I	I	I	I	B	I	I	I	I	I	I	I	C	I	I	I	I	I	I	I	Block
SLOT 4/1	Channel	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	Unblock
	Status	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	Block

Call Status : The status of Gateway port and call can be monitored in real-time

Analog

Port Status

SLOT		Port Group							
SLOT 0	Port	0(FXS)	1(FXS)	2(FXS)	3(FXS)	4(FXO)	5(FXO)	6(FXO)	7(FXO)
	Status	C	I	I	I	I	I	I	I
	Select	☐	☐	☐	☐	☐	☐	☐	☐

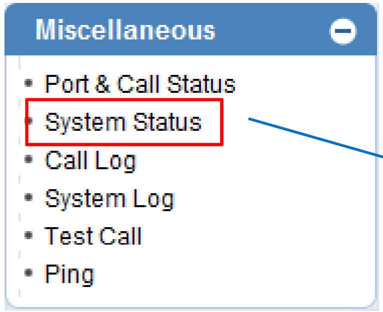
Connection State : (Connected) (Disconnected \| Blocked)
Call State : (Idle) (Ring \| Dial) (Called) (Calling) (Blocked)

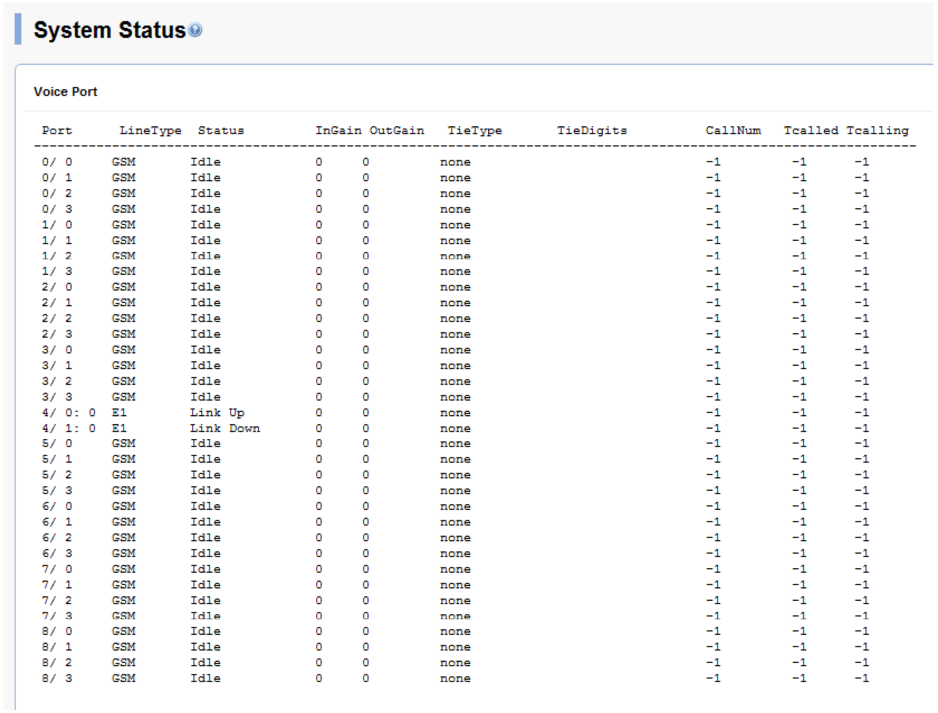
Call State

Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
0/0	GSM In	07/23 17:28:27	821021140183	500	g711u	N.A

Monitoring

2. System Status





System Status

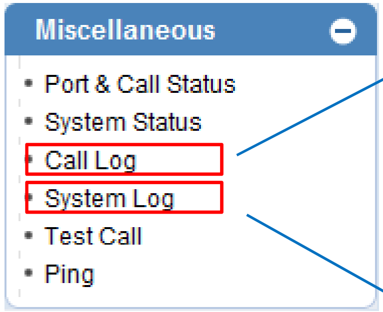
Voice Port

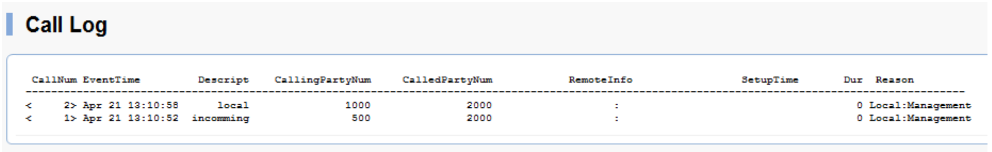
Port	LineType	Status	InGain	OutGain	TieType	TieDigits	CallNum	Tcalled	Tcalling
0/ 0	GSM	Idle	0	0	none		-1	-1	-1
0/ 1	GSM	Idle	0	0	none		-1	-1	-1
0/ 2	GSM	Idle	0	0	none		-1	-1	-1
0/ 3	GSM	Idle	0	0	none		-1	-1	-1
1/ 0	GSM	Idle	0	0	none		-1	-1	-1
1/ 1	GSM	Idle	0	0	none		-1	-1	-1
1/ 2	GSM	Idle	0	0	none		-1	-1	-1
1/ 3	GSM	Idle	0	0	none		-1	-1	-1
2/ 0	GSM	Idle	0	0	none		-1	-1	-1
2/ 1	GSM	Idle	0	0	none		-1	-1	-1
2/ 2	GSM	Idle	0	0	none		-1	-1	-1
2/ 3	GSM	Idle	0	0	none		-1	-1	-1
3/ 0	GSM	Idle	0	0	none		-1	-1	-1
3/ 1	GSM	Idle	0	0	none		-1	-1	-1
3/ 2	GSM	Idle	0	0	none		-1	-1	-1
3/ 3	GSM	Idle	0	0	none		-1	-1	-1
4/ 0: 0	E1	Link Up	0	0	none		-1	-1	-1
4/ 1: 0	E1	Link Down	0	0	none		-1	-1	-1
5/ 0	GSM	Idle	0	0	none		-1	-1	-1
5/ 1	GSM	Idle	0	0	none		-1	-1	-1
5/ 2	GSM	Idle	0	0	none		-1	-1	-1
5/ 3	GSM	Idle	0	0	none		-1	-1	-1
6/ 0	GSM	Idle	0	0	none		-1	-1	-1
6/ 1	GSM	Idle	0	0	none		-1	-1	-1
6/ 2	GSM	Idle	0	0	none		-1	-1	-1
6/ 3	GSM	Idle	0	0	none		-1	-1	-1
7/ 0	GSM	Idle	0	0	none		-1	-1	-1
7/ 1	GSM	Idle	0	0	none		-1	-1	-1
7/ 2	GSM	Idle	0	0	none		-1	-1	-1
7/ 3	GSM	Idle	0	0	none		-1	-1	-1
8/ 0	GSM	Idle	0	0	none		-1	-1	-1
8/ 1	GSM	Idle	0	0	none		-1	-1	-1
8/ 2	GSM	Idle	0	0	none		-1	-1	-1
8/ 3	GSM	Idle	0	0	none		-1	-1	-1

System Status : The system status of AP-GS1002 can be monitored

Monitoring

3. Call Log / System Log





Call Log : Monitoring all of call history

※ Call history will be clear with rebooting

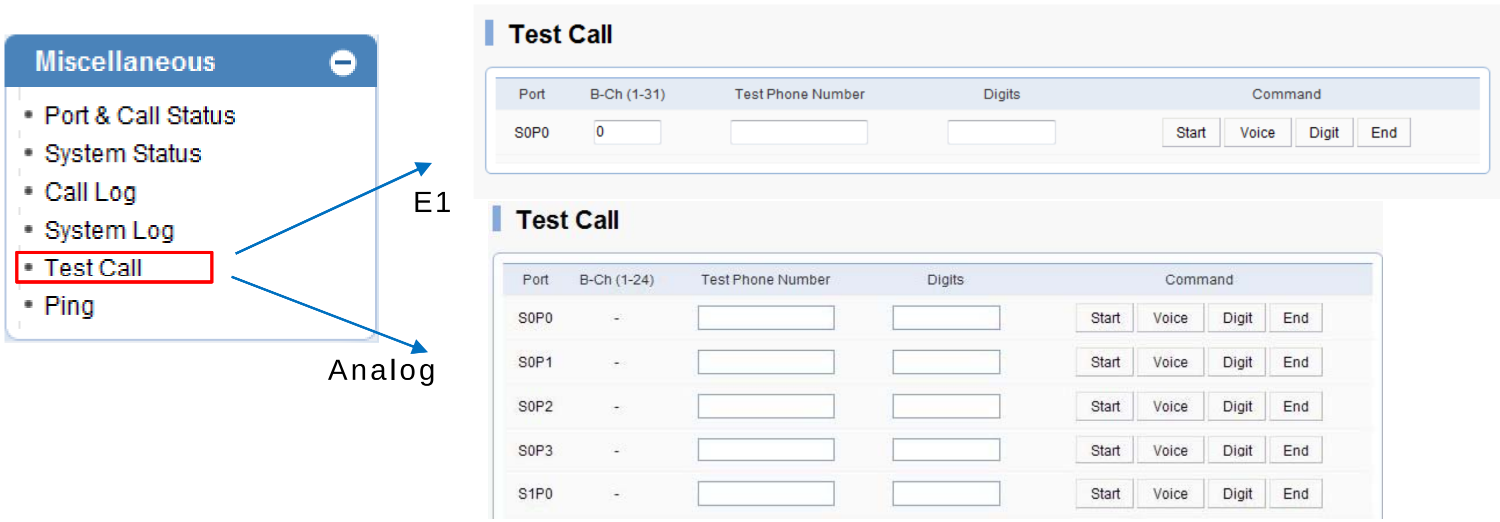
System Log : Monitoring Gateway System log

Default setting is off. System log can be monitored by telnet connection and entering CLI command is required

(Please contact to AddPac technical support team for more detail)

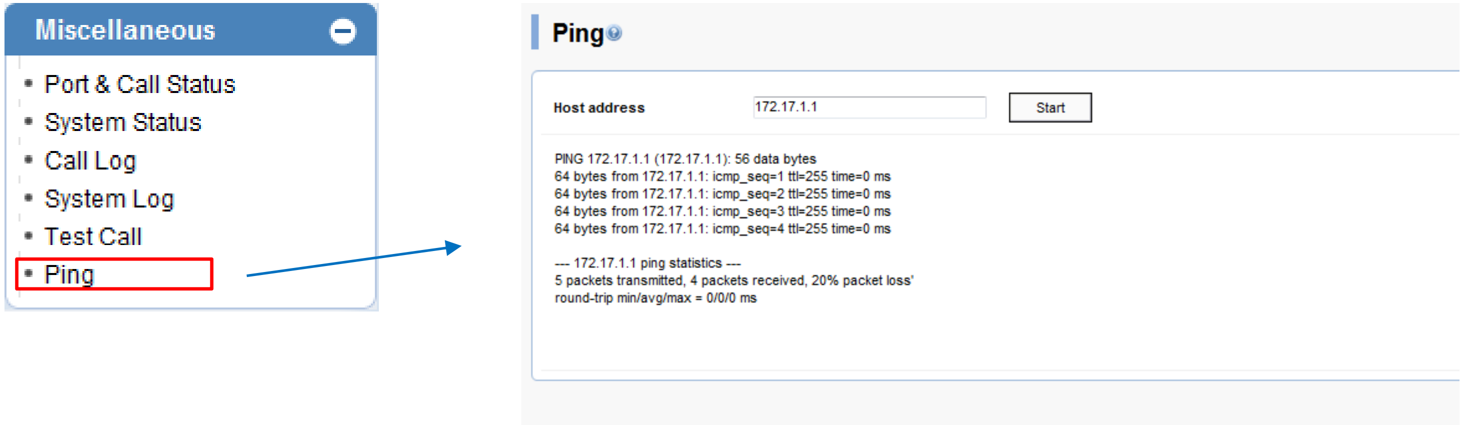
Monitoring

4. Test Call



Test Call : remote diagnose the proper operation of VoIP call.
: Enter the test phone number in relevant port then click start button.
- Test Phone Number : enter receiving number for test processing.
- Digits : enter the desire digit once the call is connected.
- Command
  - Start : start test call.
  - Voice : sample voice play under connection status.
  - Digit : play the entered digit underconnection status.
  - End : end test call.

5. Ping



Ping : Network status can be checked by pinging

Thank you