AP-GS501TM
1-Port FXS GSM Gateway

High Performance GSM Gateway Solution

WEB Setup Guide

AddPac Technology

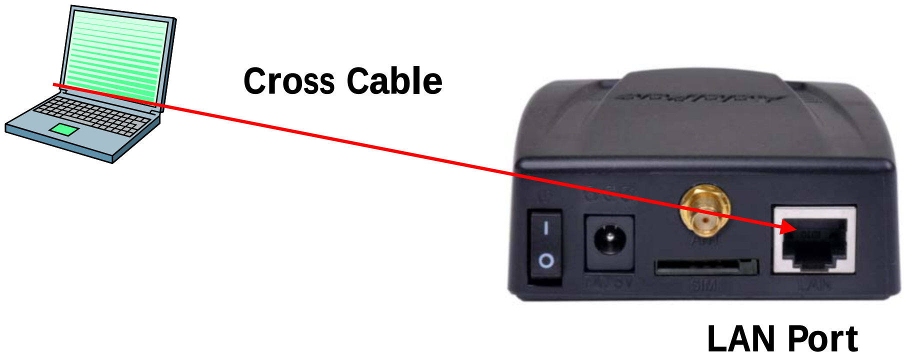
2012, Sales and Marketing

www.addpac.com

Contents

WEB Connection

1. Connection Web via LAN Port



LAN Default IP
IP Address : 192.168.10.1
Subnet mask : 255.255.255.0

1. It is the way to connect to GSM G/W via LAN port

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

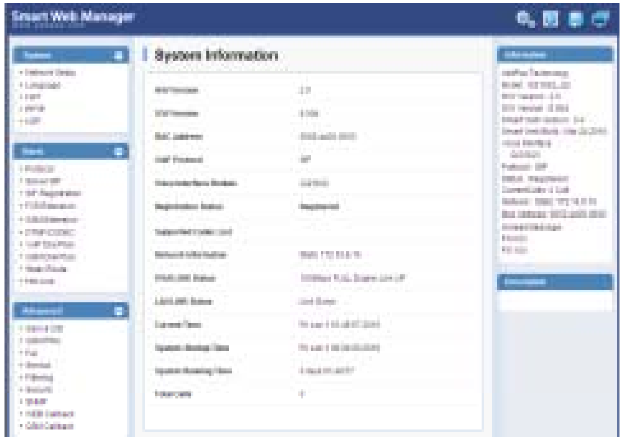
3. After set PC with same IP address subnet, connect to GSM G/W
   - Connect PC to GSM G/W using Cross UTP-Cable. You may use Ethernet switch with normal UTP-cable
   - Enter IP address 192.168.10.1 on your web browser

WEB Connection

1. The screen of Web connection

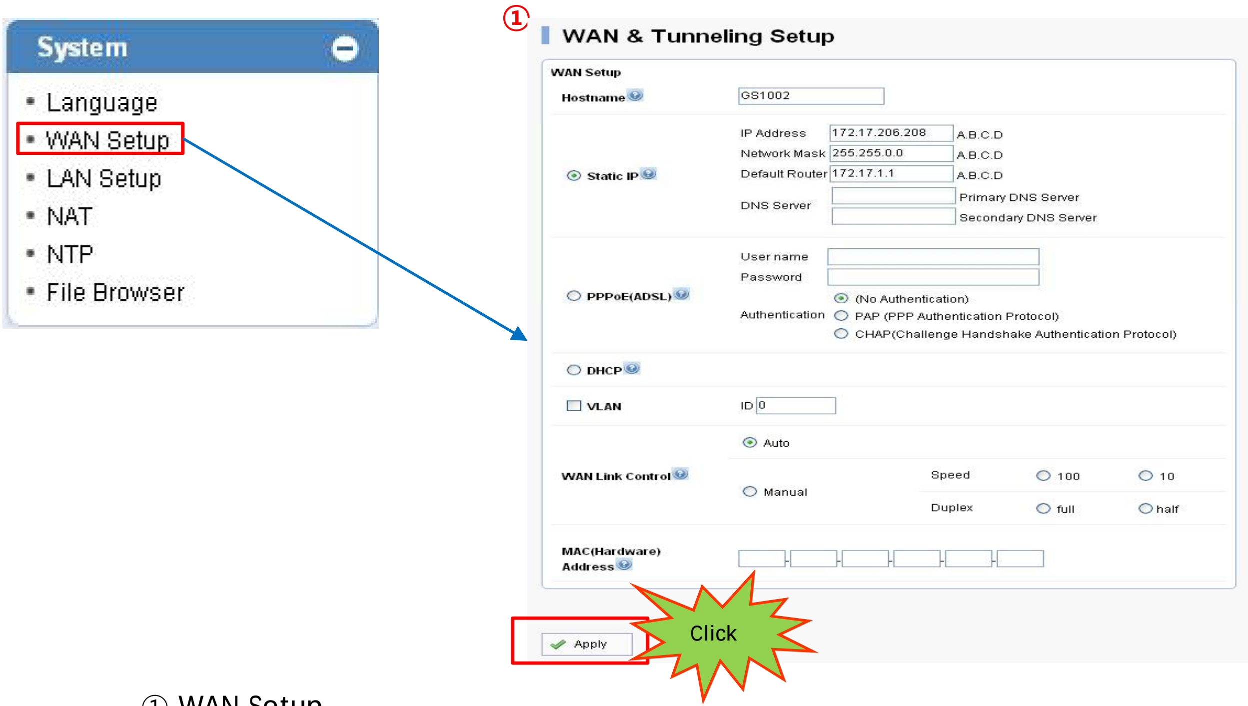
• The Shown log-in screen is connection to Web page.
Please enter the below log-in information

ID : root
Password : router



Network Setup

1. WAN Setup -1

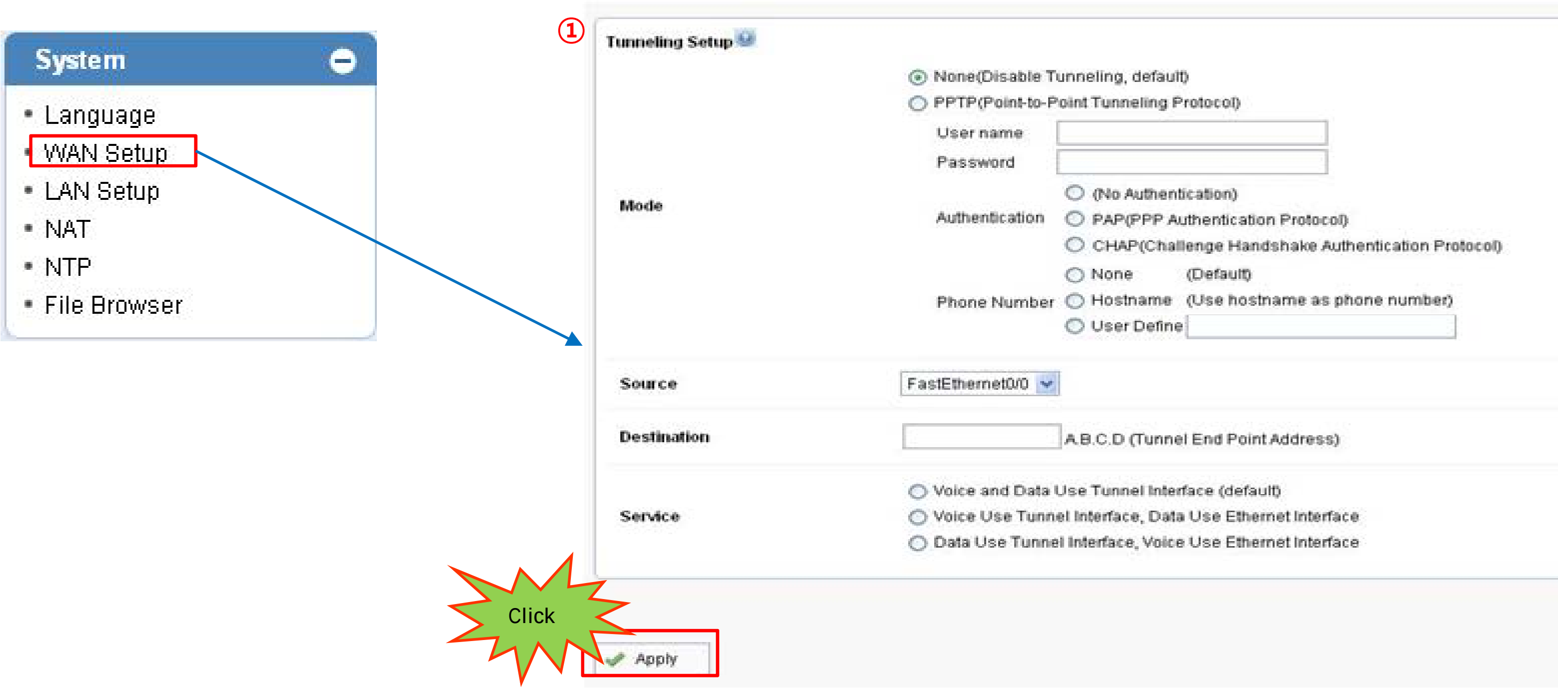


① WAN Setup
-Hostname : Enter the device name of GSM G/W
-Static IP
-PPPoE(ADSL)
-DHCP

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

Network Setup (Tunneling Setup)

1. WAN Setup -2



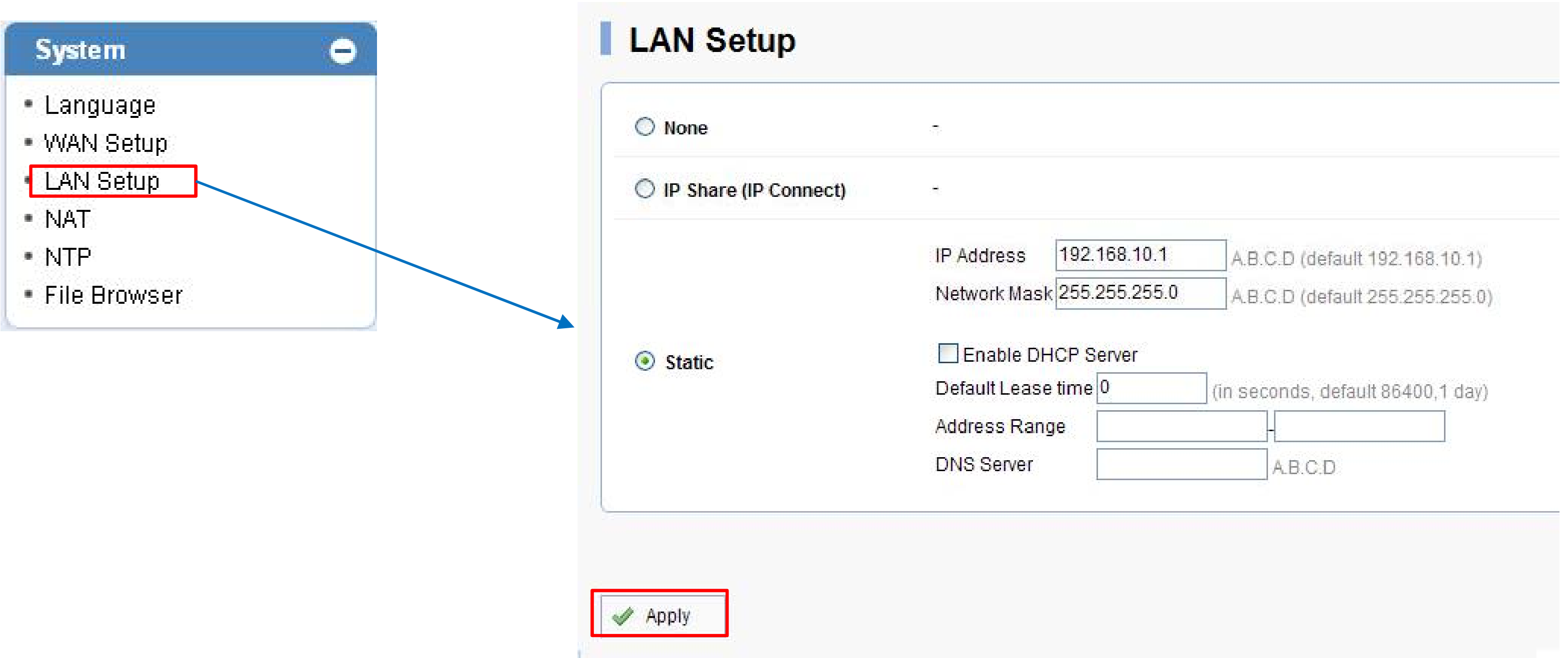
① PPTP (Point-Point Tunneling Setup)

- PAP (PPP Authentication Protocol) , CHAP (Challenge Handshake Authentication Protocol)
- Source : Setup the interface to connect with PPTP server.
- Destination : Enter the Tunnel End Point Address regard of Source.
- Service : Decide whether to apply Ethernet interface regard of Voice and Data communication.

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

Network Setup (LAN Setup)

2. LAN Setup



LAN Setup
- IP Share (IP Connect)
- Static : 1. Setup LAN 1 Port Static IP
  2. Setup DHCP Server

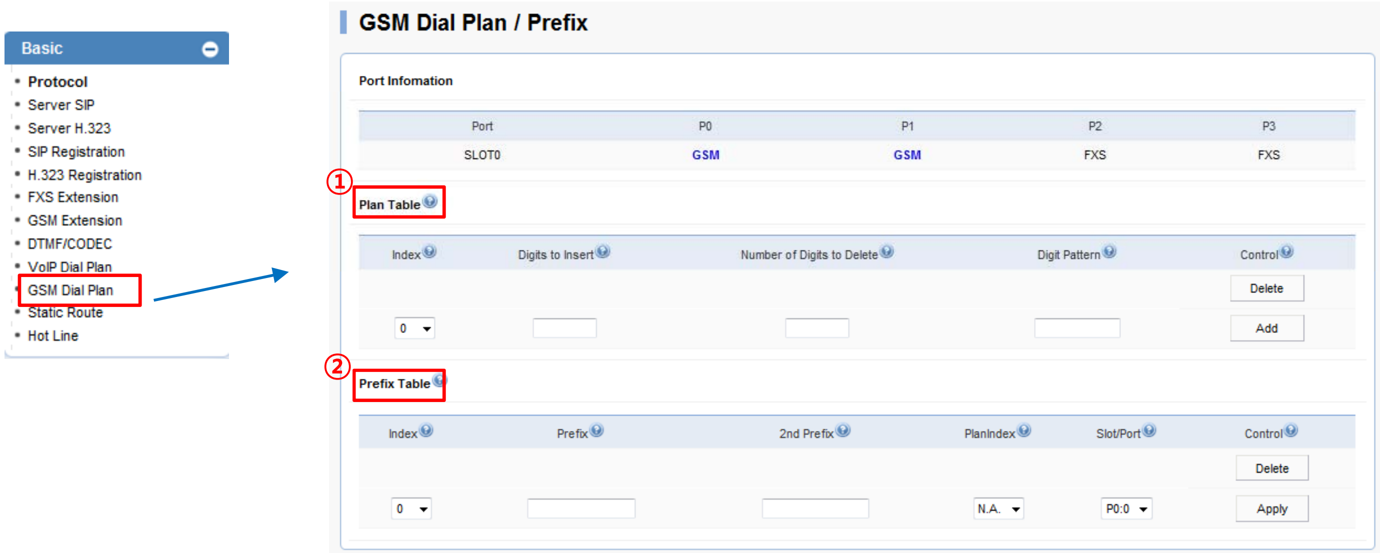
*** Please make sure to press the apply button for saving**

GSM Setup

1. GSM Dial Plan / Prefix
2. GSM Extension
3. FXS Extension
4. Hotline

GSM Setup > Dial Plan

1. GSM Dial Plan / Prefix -1



Dial Plan / Prefix : Setting for making outgoing call to GSM Networks using FXS or VoIP

① Plan Table : Outgoing call to GSM network can be made with number conversion

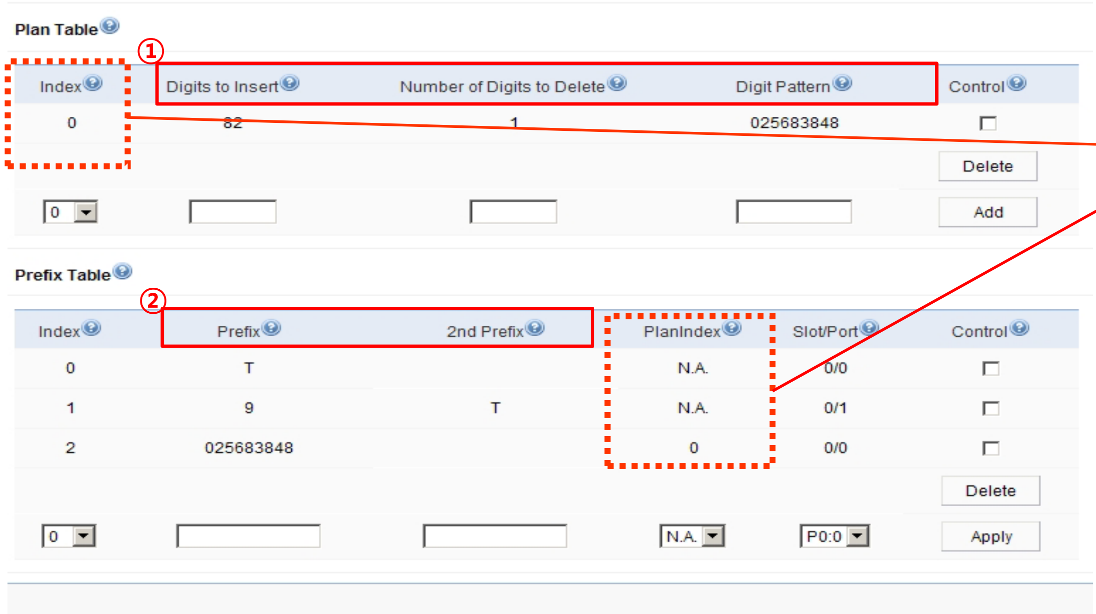
② Prefix Table : it is for outgoing call to GSM Networks. Both 1 Stage and 2 Stage are available

1 Stage : Making call after hearing the first dial-tone. Setting Prefix field is required

2 Stage : Making call after the second dial-tone. Setting 2nd Prefix field is required.
In case of 2nd stage using, the Prefix can be used a number for hearing the Second dial-tone

GSM Setup > Dial Plan

1. GSM Dial Plan / Prefix - 2 (Example)



It is required to set the same number on Plan Index of Prefix Table and Index number of Plan Table

① Digit to Insert : inserted Number
Number of Digit to Delete : Number of digit to delete
Digit Pattern : Number to apply for conversion

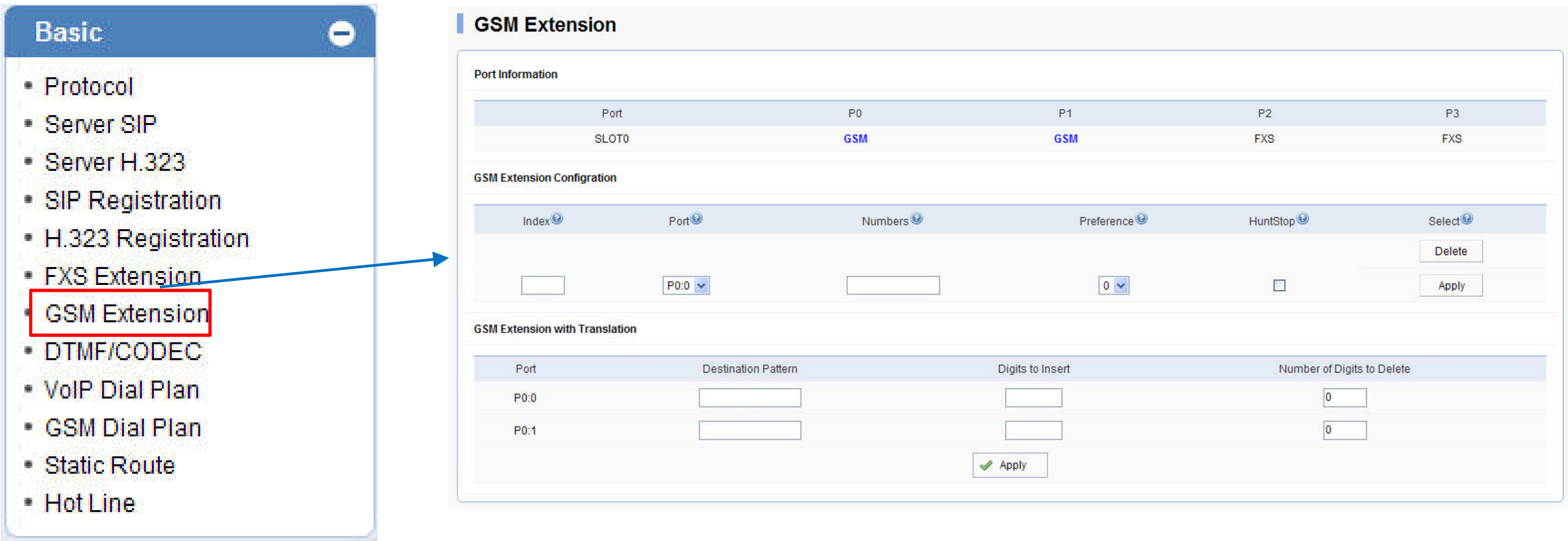
② Prefix , 2nd Prefix : Setting method of 1 stage and 2 stage

(ex : Prefix – T → 1 stage method – Forward call immediately
Prefix – 9, 2nd Prefix – T → Whe press 9, it is the method to press dial after hearing 2nd dial-tone

PlanIndex : Set index applied for Plan Table
SlotPort : Set GSM port

GSM Setup > GSM Extension

2. GSM Extension -1



GSM Extension Configuration
: Register GSM SIM Number
: Other party's number can be registered with Call back Service
- Index : Sequential number for each extension.
  Existed number makes configuration modified
- Port : Select port to set up
- Numbers : Register SIM number or mobile phone number
  to use callback service

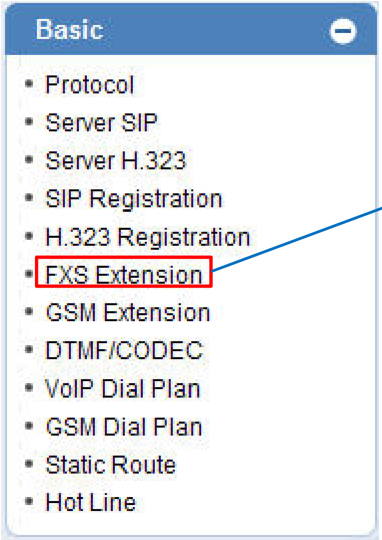
GSM Extension with Translation
: Use to convert mobile phone number for callback service
- Destination Pattern : Enter mobile phone number to convert
- Digits to Insert : Insert number to make calling number
- Number of Digits to Delete : Delete number to make calling number

ex) Destination Pattern : 025683848
Digits to Insert : 82
Number of Digits to Delete : 2

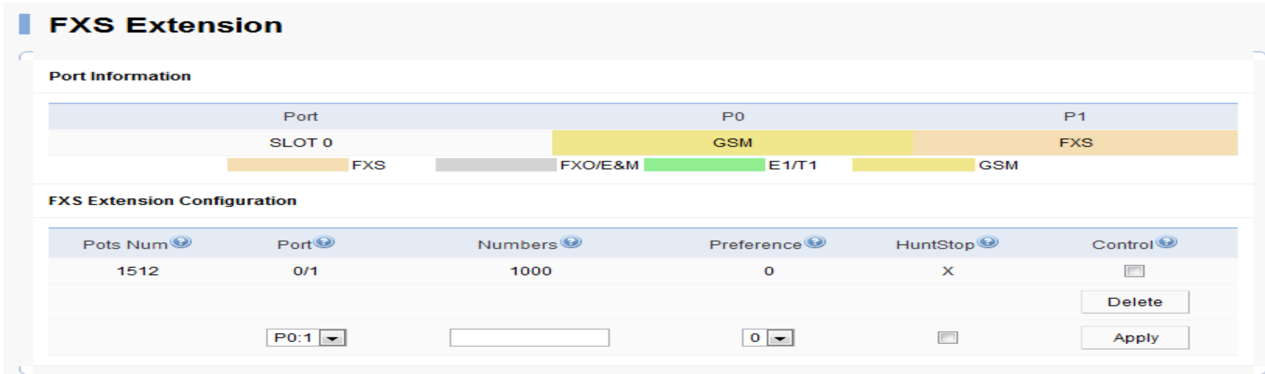
Result : 825683848

GSM Setup > FXS Extension

3. FXS Extension -1



①



②

① The each port information of GSM 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

GSM Setup > FXS Extension

3. FXS Extension -2 (Example)

FXS Extension

Port Information

Port	P0	P1
SLOT 0	GSM	FXS

FXS FXO/E&M E1/T1 GSM

FXS Extension Configuration

①

Pots Num	Port	Numbers	Preference	HuntStop	Control
1512	0/1	1000	0	X	☐
					Delete
	P0:1		0	☐	Apply

① Set the number to be used for FXS 0/1 port (ex. 1000)

• Setting number on each FXS port is required, so that Dial-tone can be heard on phone.

GSM Setup > FXS Extension

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

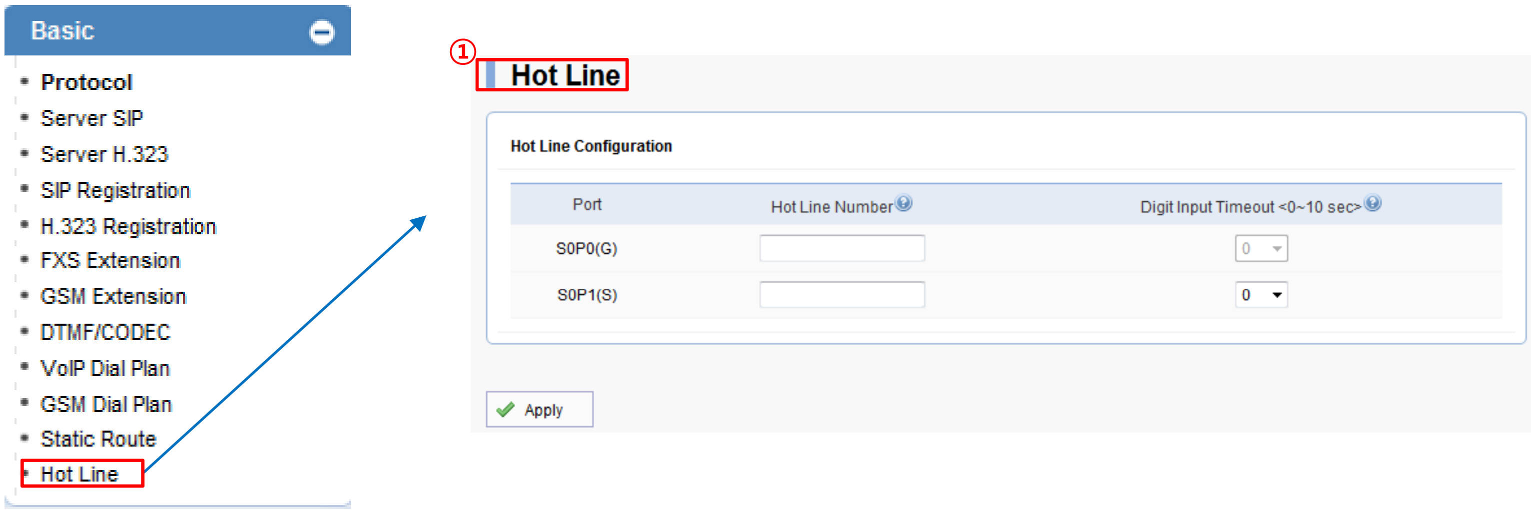
** T : Accept all number entered within Inter Digit Time (Default IDT : 3sec)
** Dot(.) : One dot(.) means one digit
** [ ] : The range of number

※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..

Direct Incoming call

4. GSM Setup > Hot Line -1



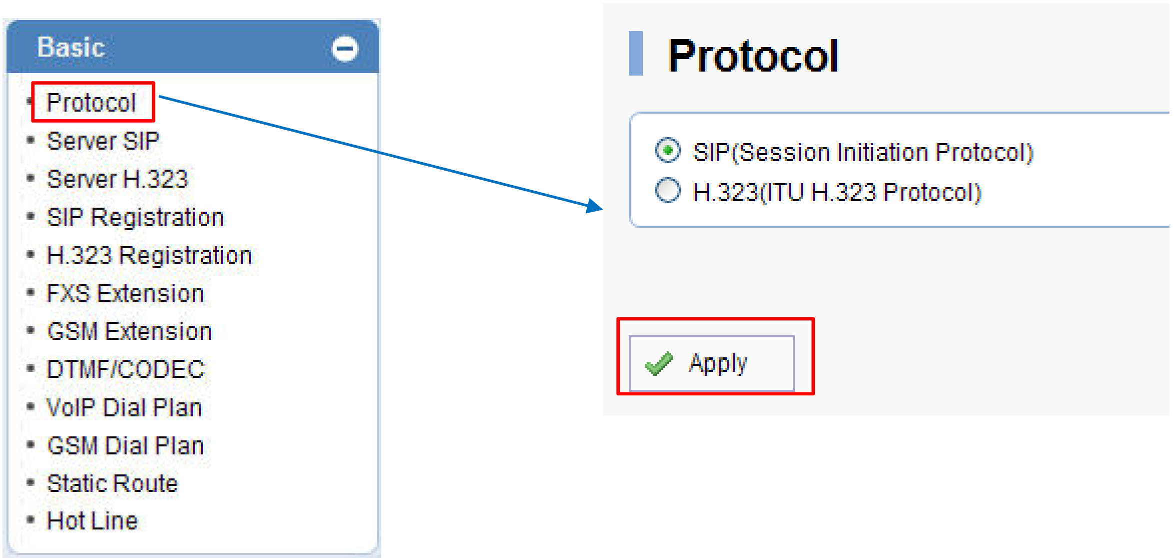
① Hot Line Configuration : Connect incoming and outgoing call directly
- Port : It means GSM and FXS port
- Hot Line Number : Forward call to entered number
  It connects to the number of GSM port in case of receiving call (Direct Incoming call)
- Digit Input Timeout : Time to make call to the Hot Line Number when user doesn't any action after off-hook

VoIP Setup

1. VoIP Setup
   - Server SIP
   - Server H.323
2. DTMF/CODEC
3. VoIP Dial Plan
4. Static Route

VoIP Setup (Protocol)

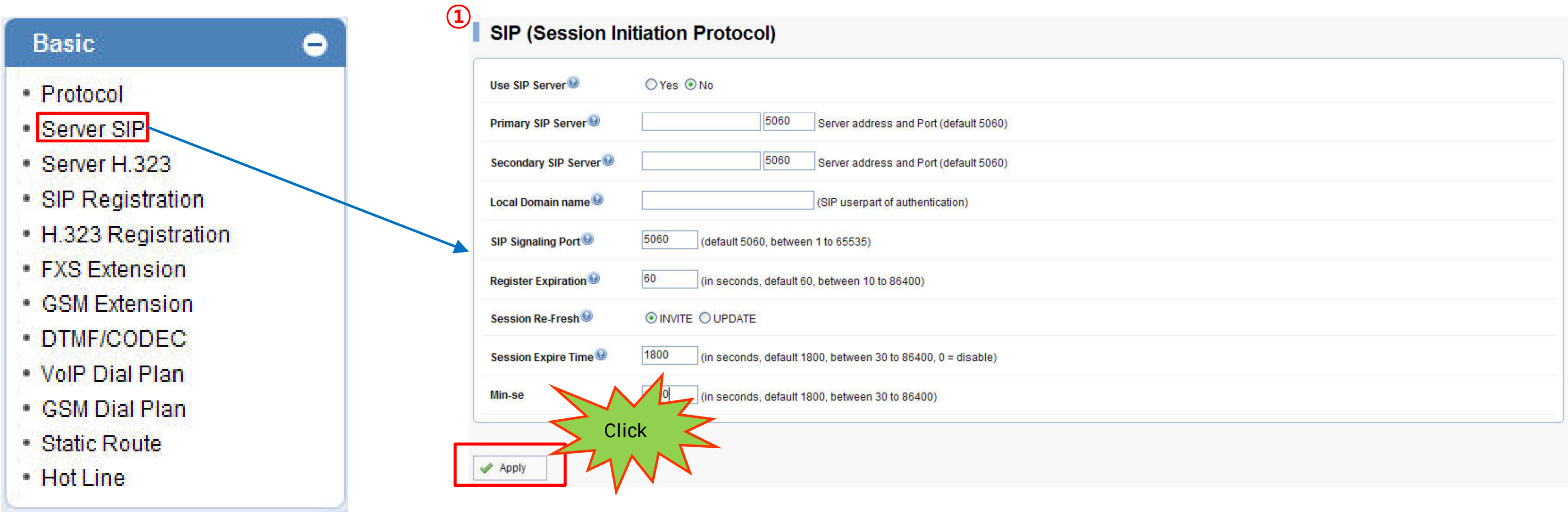
1. Protocol (SIP or H.323)



① Protocol
- Setup SIP or H.323 Protocol.

VoIP Setup (SIP)

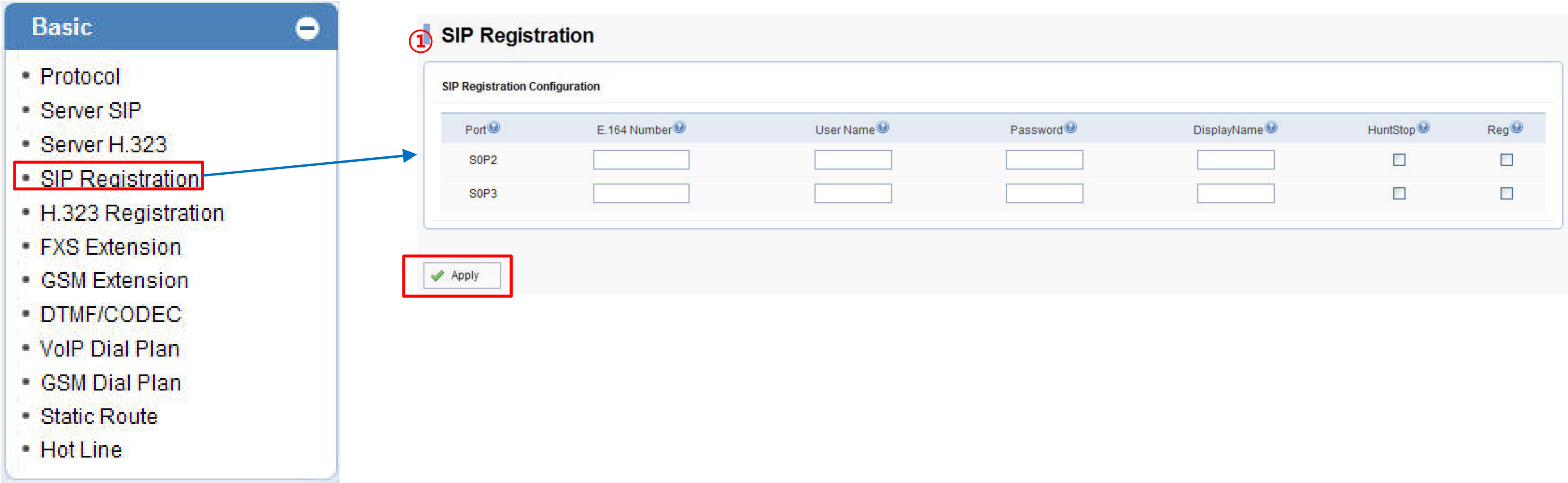
1. Server SIP



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

VoIP Setup (SIP)

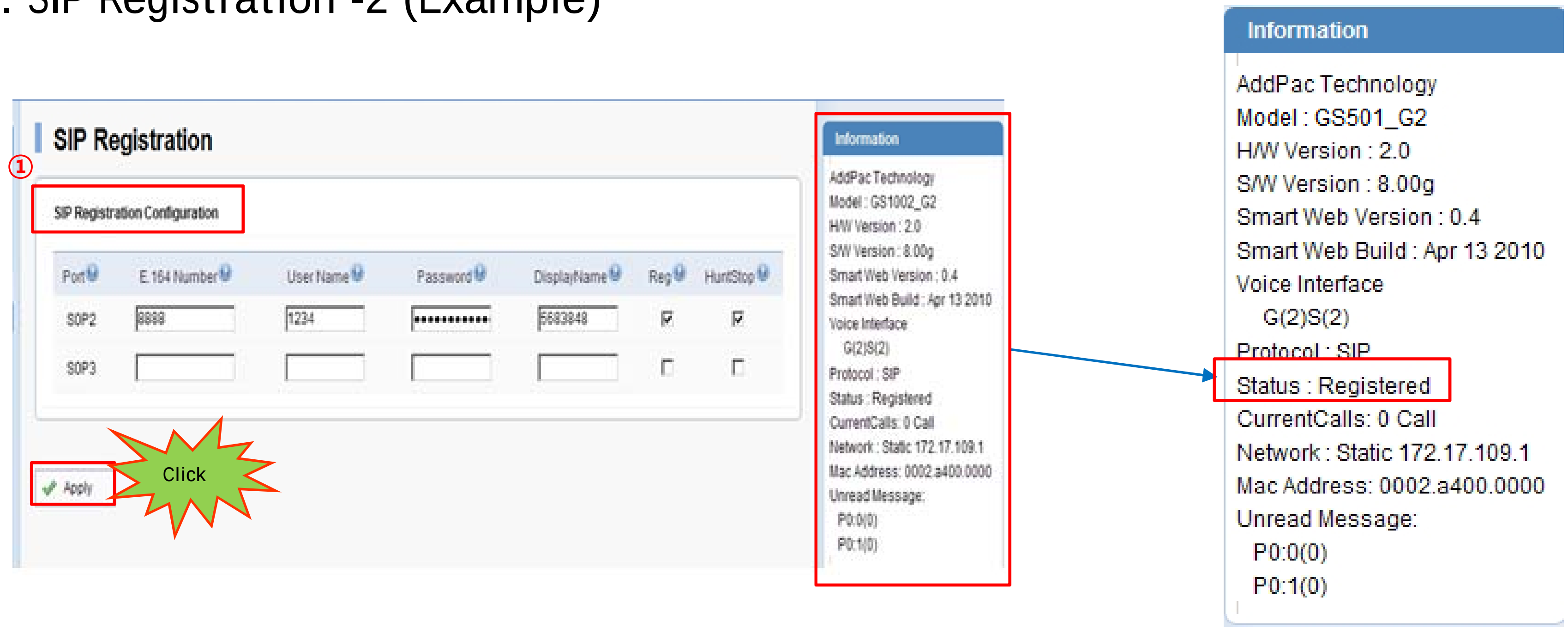
2. SIP Registration -1



① SIP Registration Configuration
-E.164 Number : Enter SIP authentication number
-User Name : Enter authentication ID
-Password : Enter authentication Password .
-Display Name : Use it when virtual number
-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.

VoIP Setup (SIP)

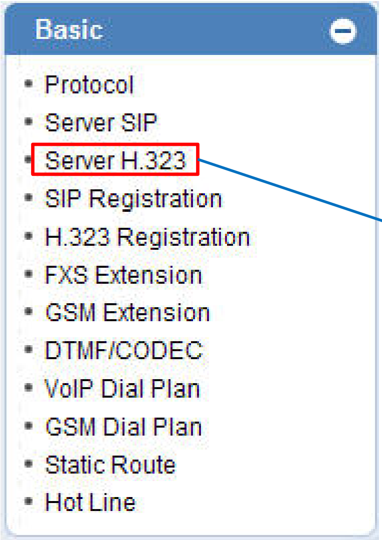
2. SIP Registration -2 (Example)



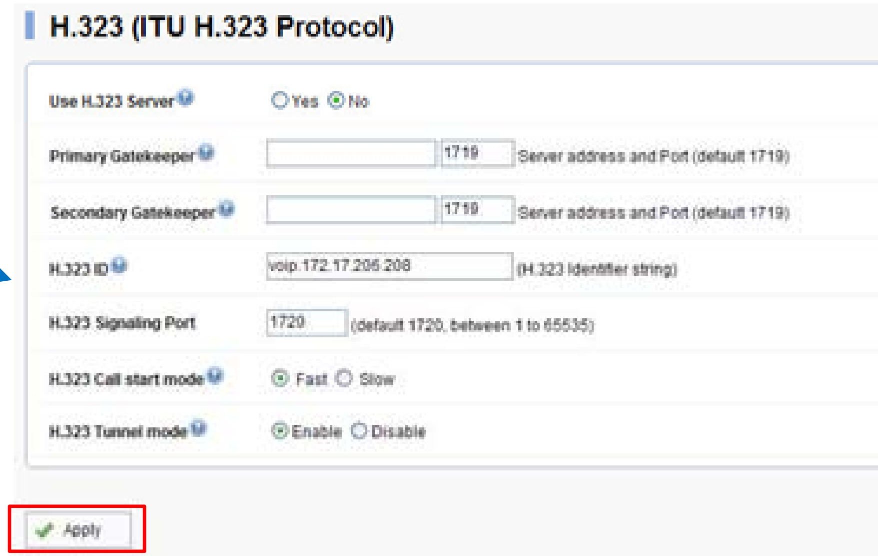
① Please click the apply button after enter information of SIP Registration
- You may check status of registration with reload web page using F5 key

VoIP Setup (H.323)

1. Server H.323



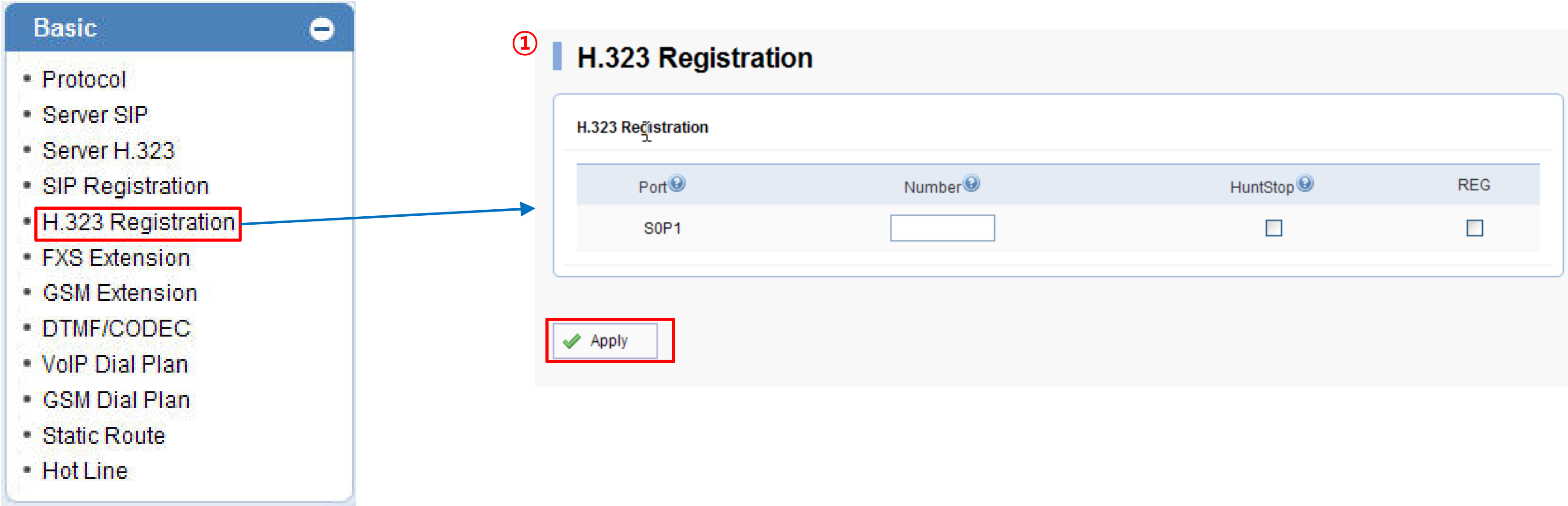
①



① Server H.323
-Use H.323 Server : Select using H.323 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

VoIP Setup (H.323)

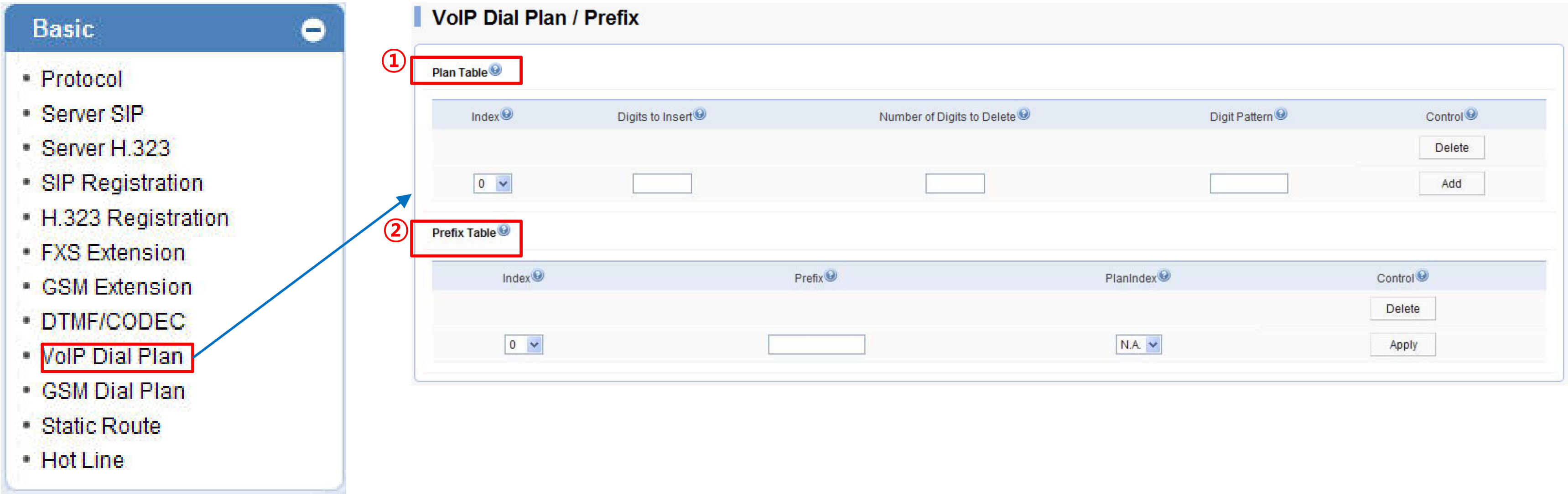
2. H.323 Registration



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 SIP Server

VoIP Setup

1. VoIP Dial Plan -1

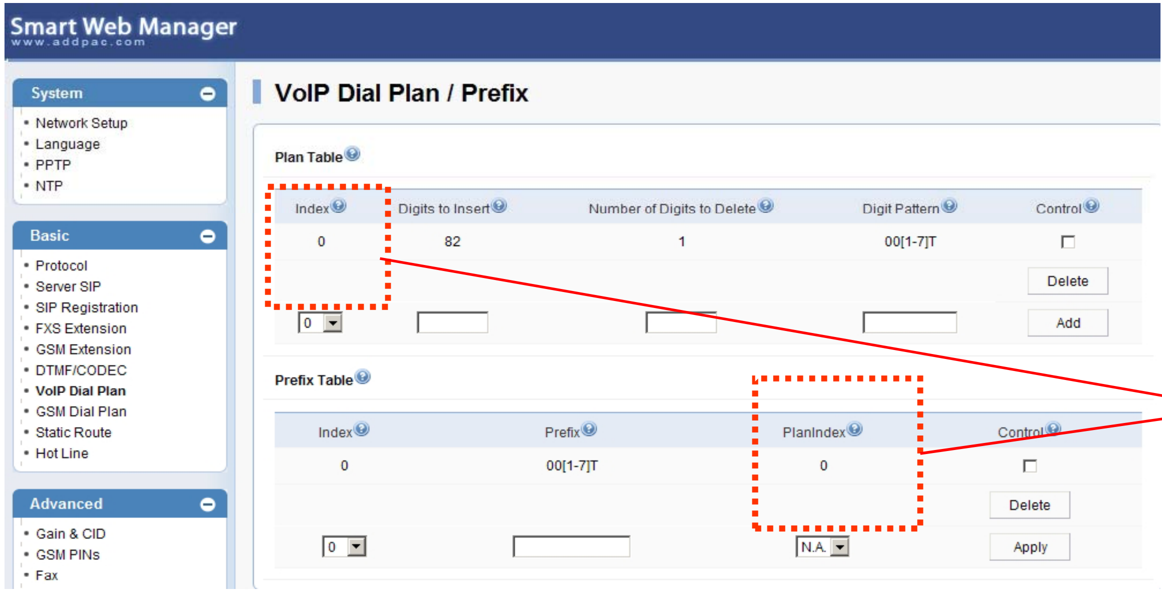


① Plan Table
- Digits to Insert : Number you want to enter
- Number of Digit to Delete : Number of digit to delete
- Digit Pattern : Number to apply for conversion

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

VoIP Setup

1. VoIP Dial Plan -2 (Example)



It must be the same with PlanIndex Number

VoIP Setup

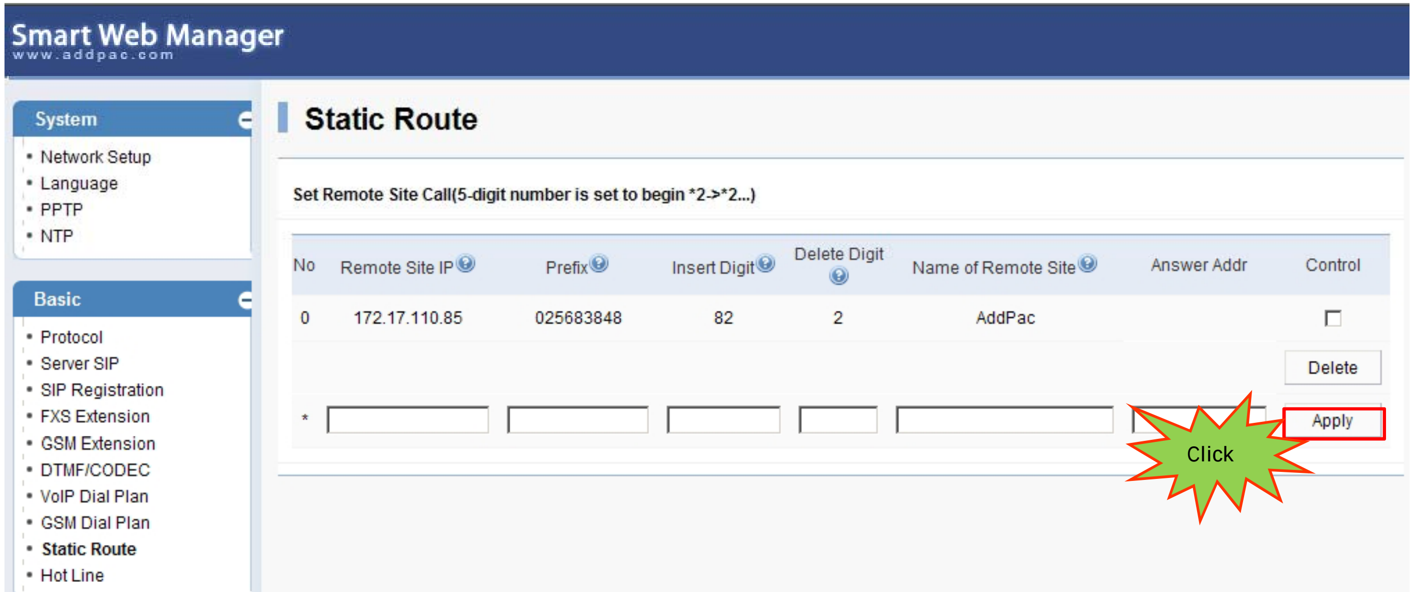
2. Static Route -1

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'

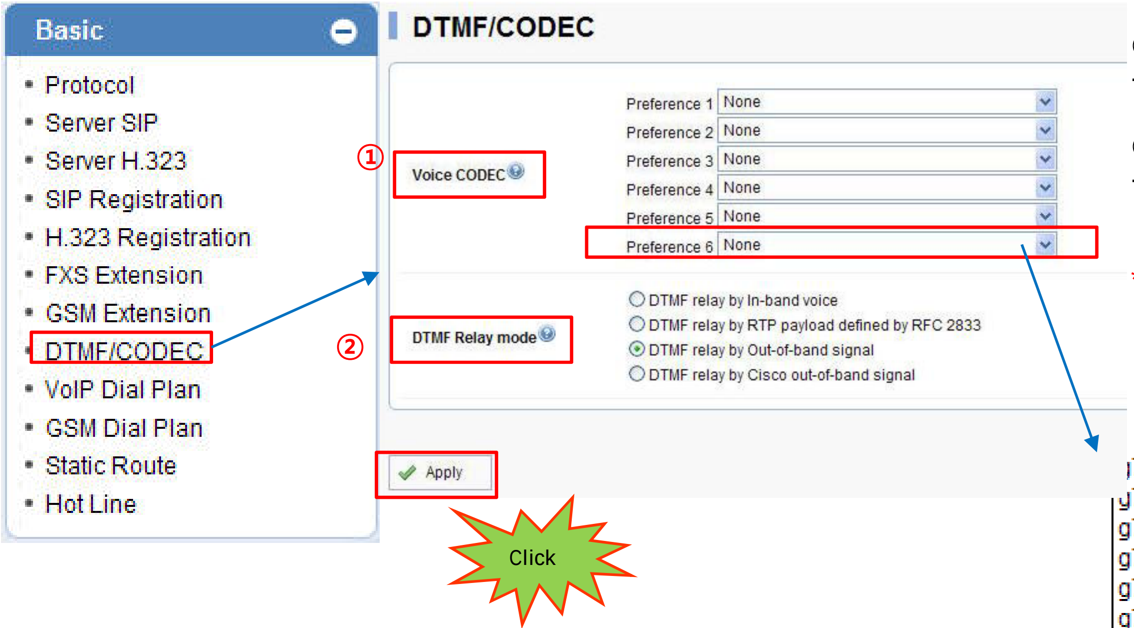
VoIP Setup

2. Static Route -2 (Example)



* Please press the apply button to save

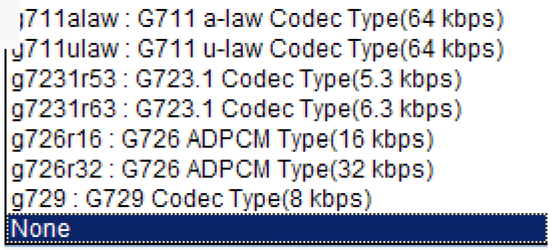
3. DTMF/CODEC



①Video 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

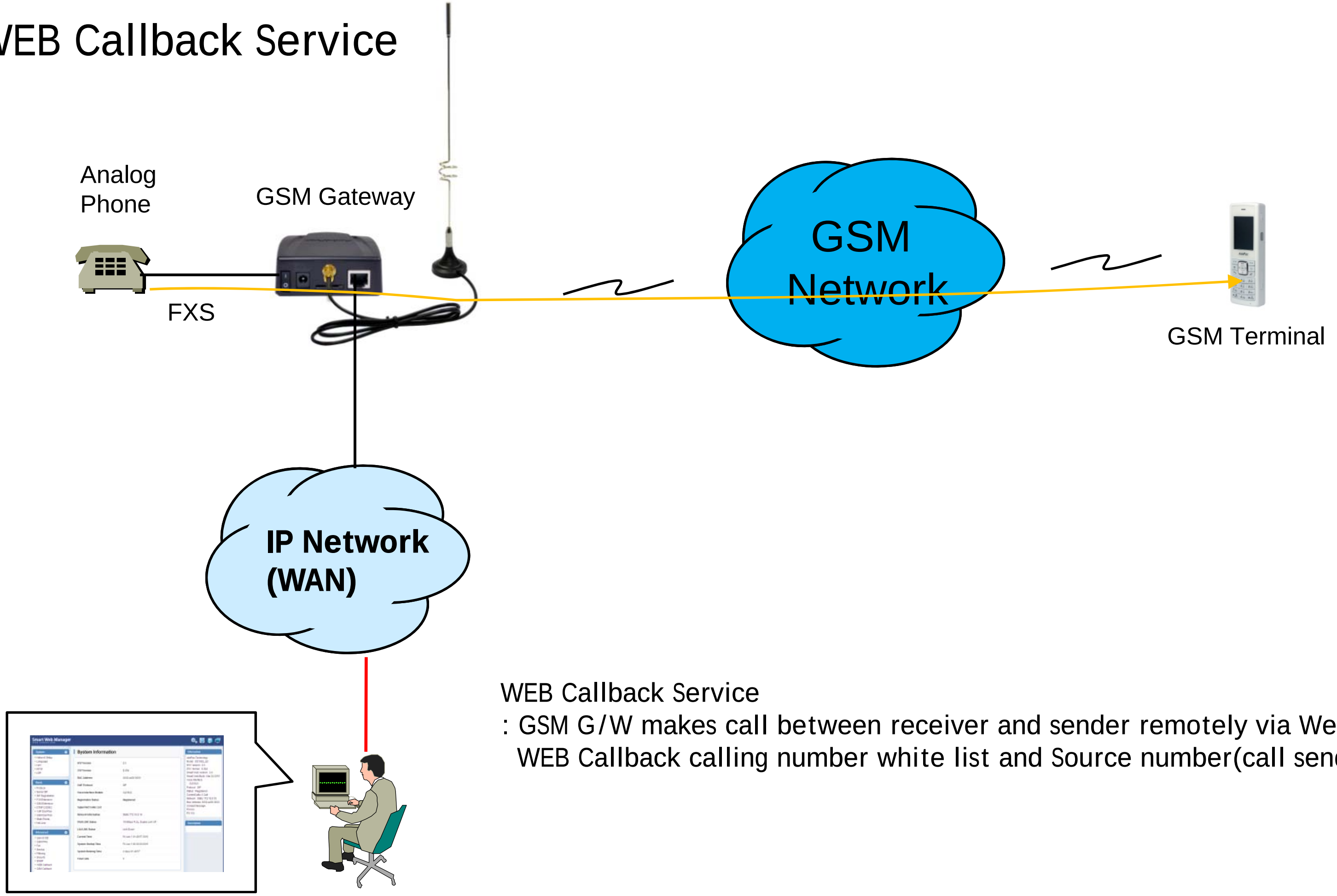


Callback Service

1. WEB Callback Service
2. GSM Callback Service

Callback Service

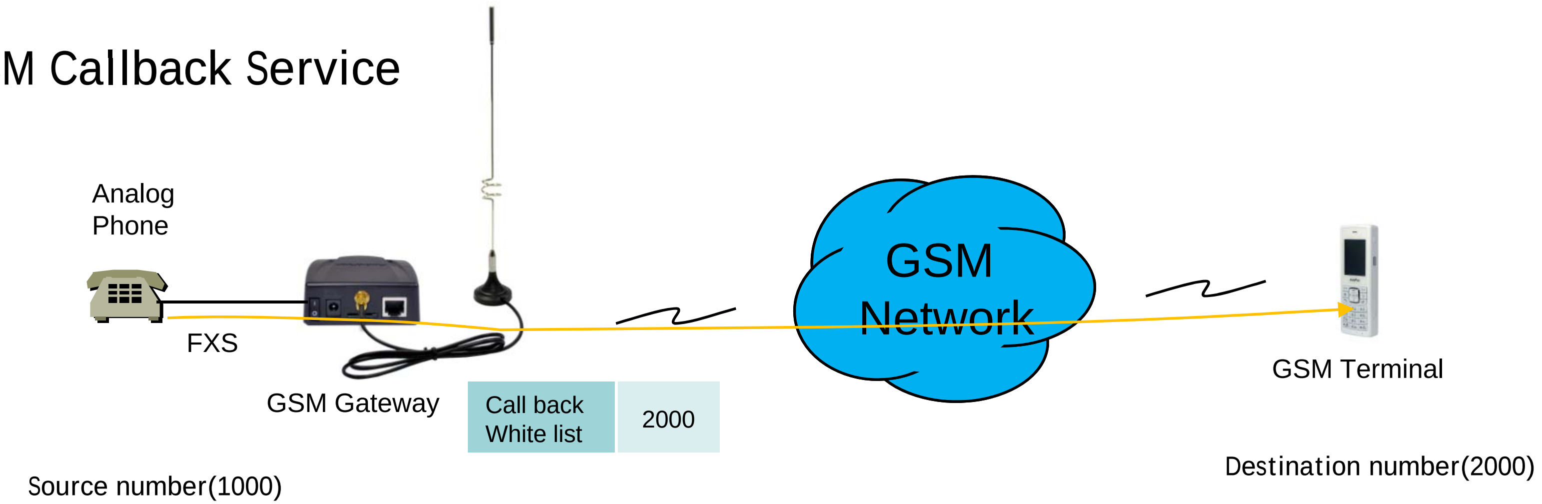
WEB Callback Service



WEB Callback Service
: GSM G/W makes call between receiver and sender remotely via Web page.
WEB Callback calling number white list and Source number(call sender) must be the same

Callback Service

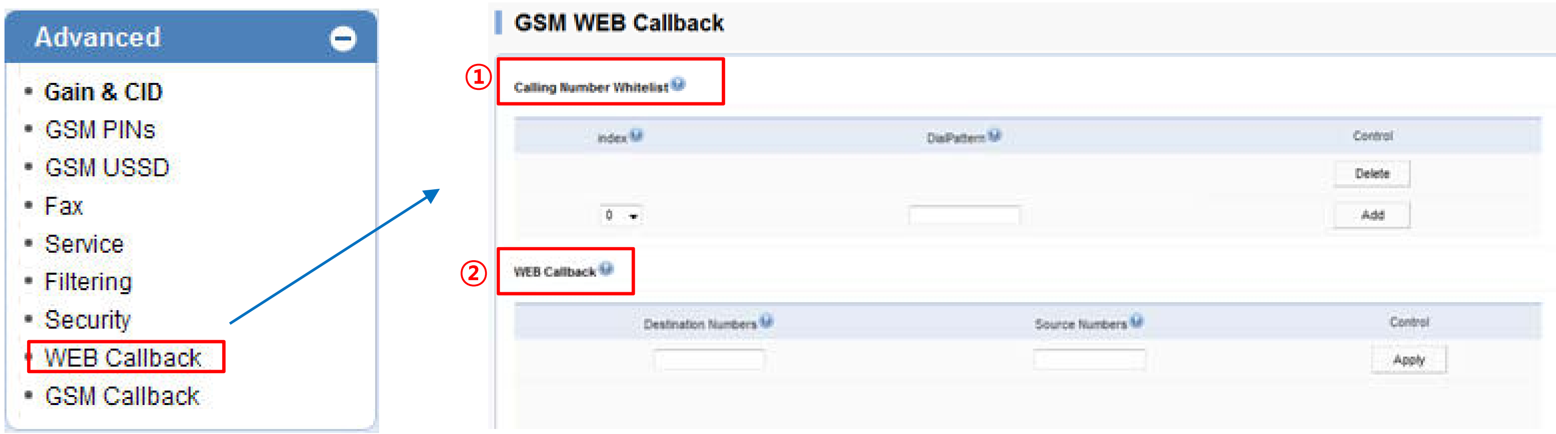
GSM Callback Service



GSM Callback Service
: The mobile user listed on the Callback white list can receive call back after disconnect the call by GSM G/W

Callback Service > WEB Callback

1. WEB Callback Service -1



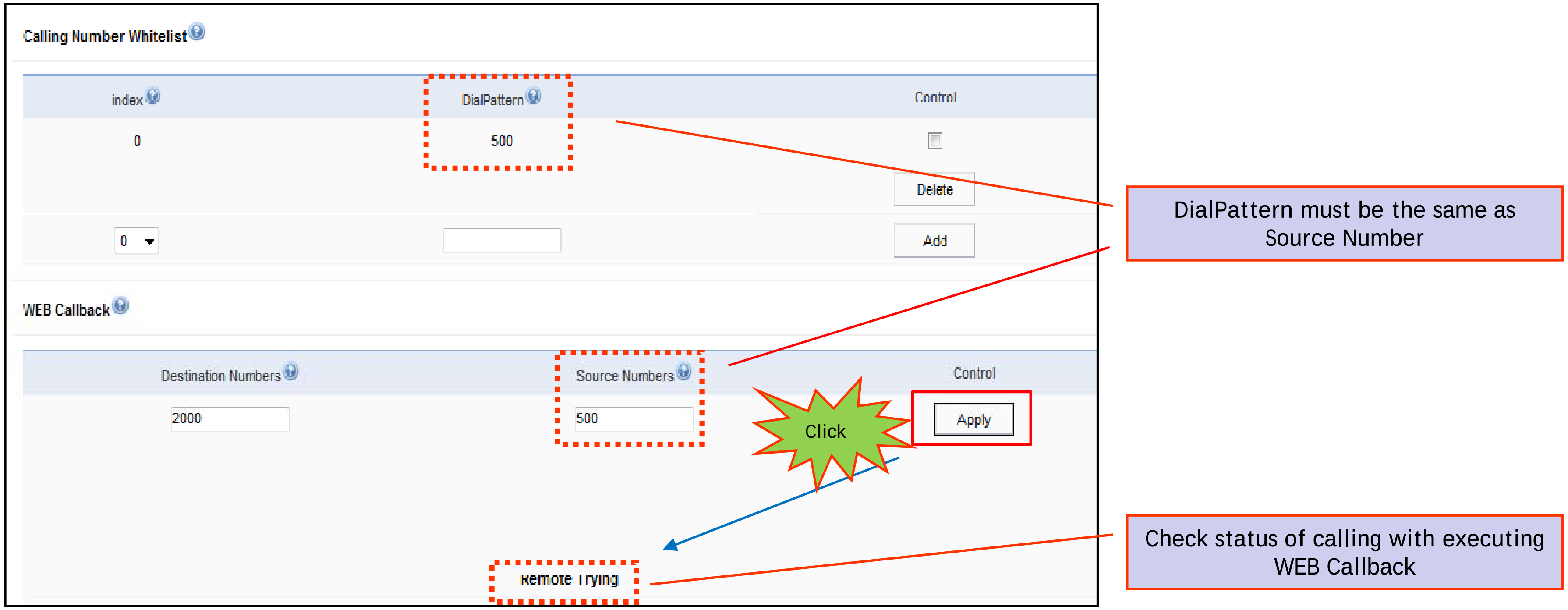
① Calling Number White List : Enter number of WEB Callback user
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Enter number to register to use WEB Callback Service

② WEB Callback :
- Destination Number : Enter number to receive call
- Source Number : Enter number of call maker. It must be the same as Dialpattern number
- Apply : Connect call between sender and receiver. Waiting tone is heard until call connected

※ You must need CID Enable to use Callback service.

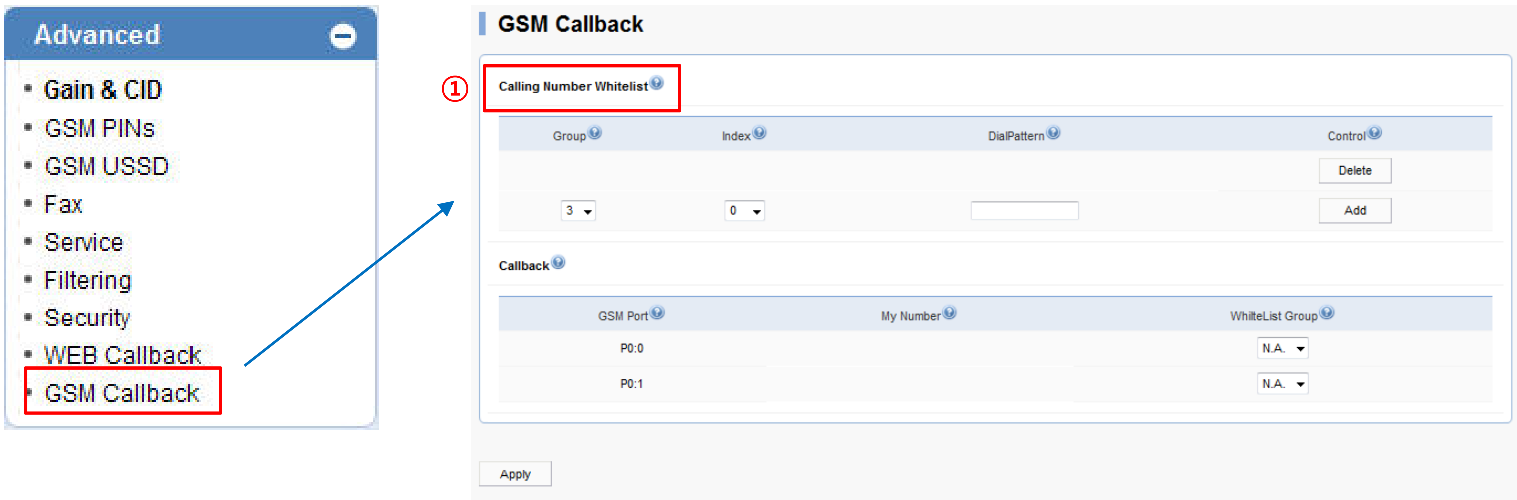
Callback Service > WEB Callback

1. WEB Callback Service -2 (Example)



Callback Service > GSM Callback

2. GSM Callback Service -1

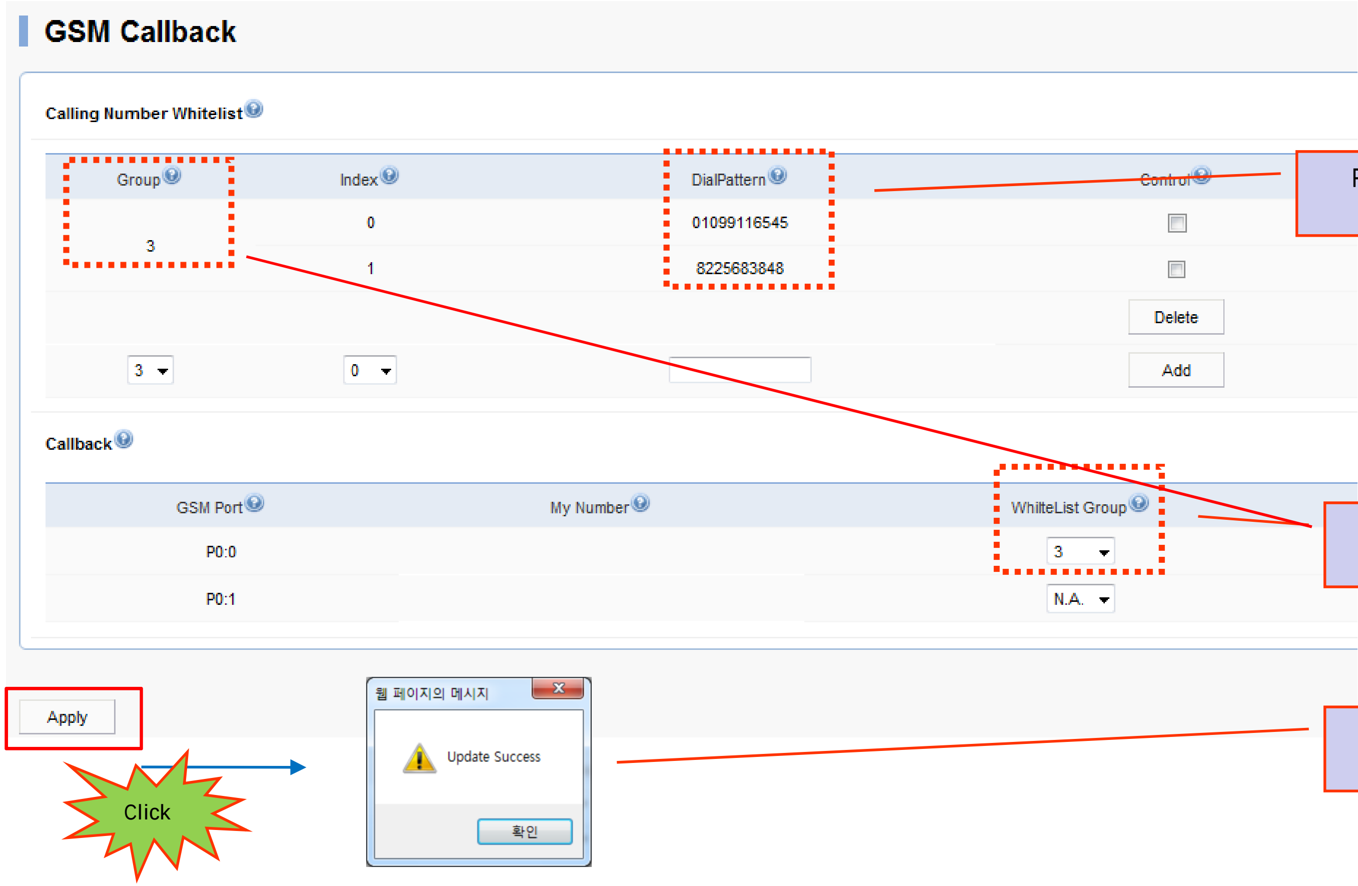


① Calling Number White List : Enter number to use GSM Callback
- Group : Enter Group Number (default : 3)
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Enter mobile phone number to use callback service

② Callback :
- White List Group : Enter group number of port to be used

Callback Service > GSM Callback

2. GSM Callback Service -1 (Example)



Register number to use Callback Service

Enter group number of port to be used

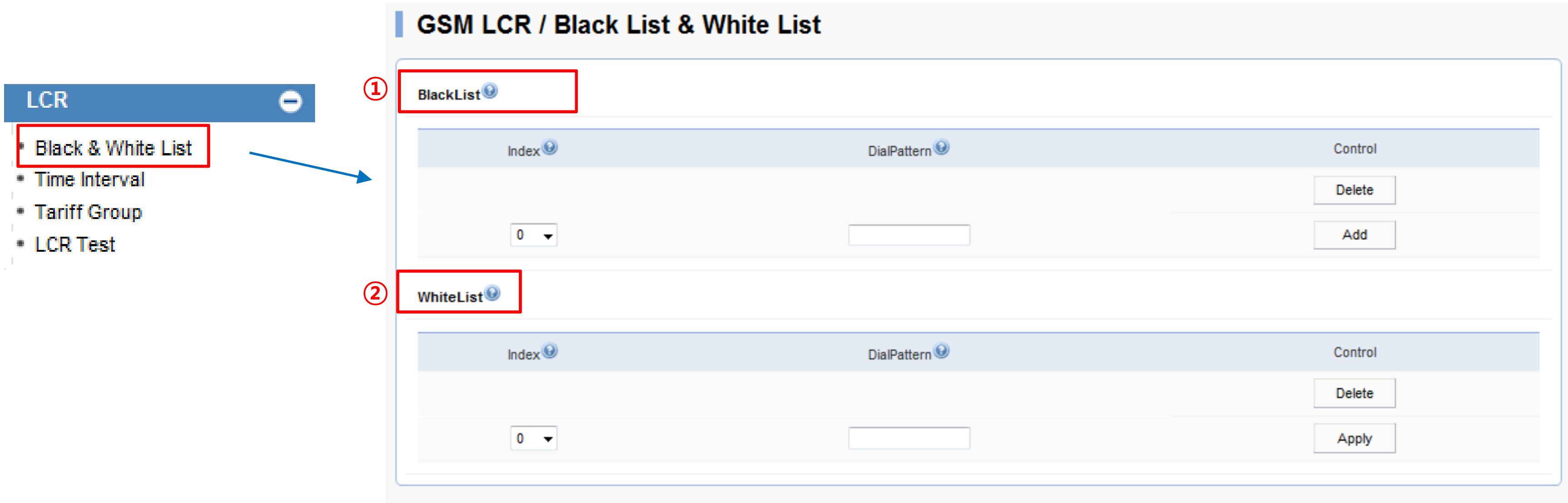
Please press the apply button and check pop-up screen

Advanced Service

1. Black / White List
2. LCR
3. SMS

Black / White List

1. Black List / White List-1

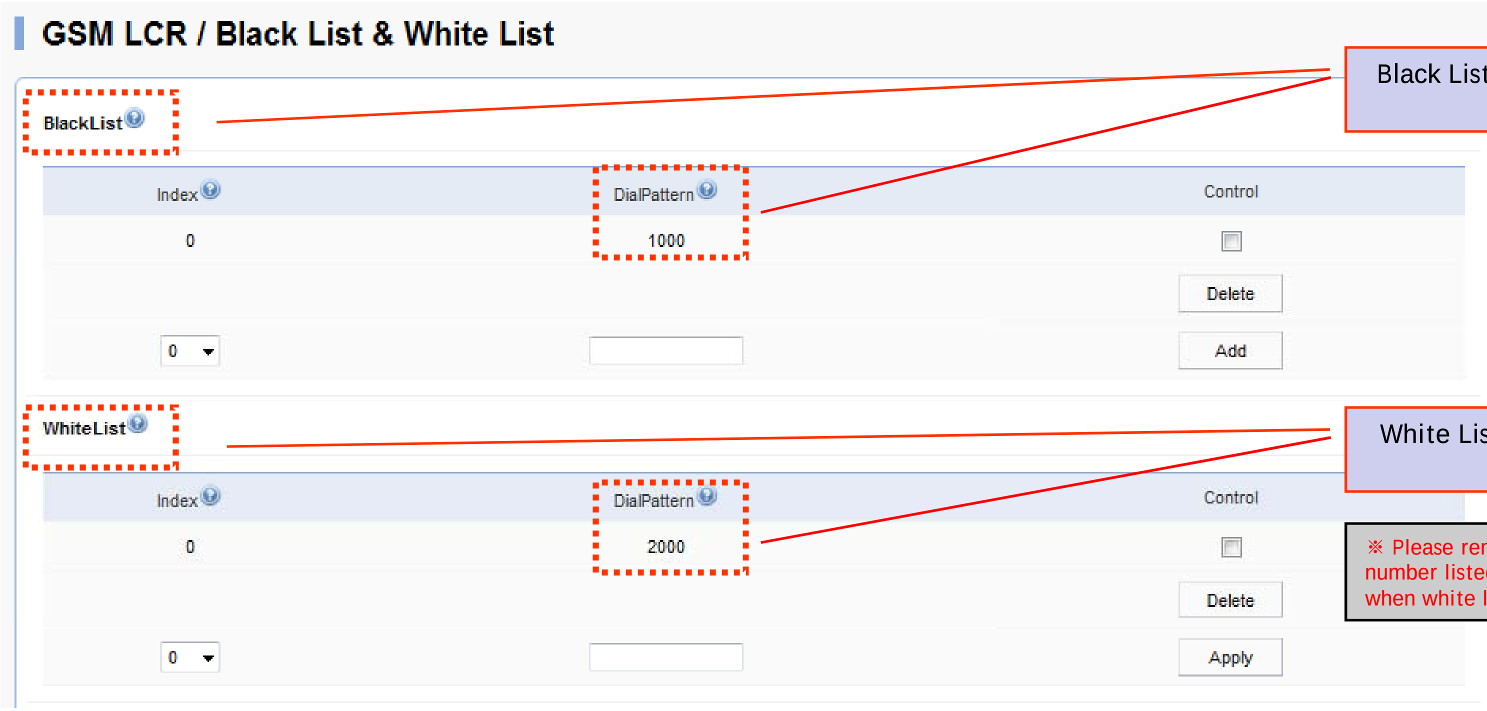


① Black List : Reject call from specific number
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Set number to reject

② White List : Allow call from specific number
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Set number to allow

Black / White List

1. Black List / White List-1 (Example)



Black List : Reject call from specific number

White List : Allow call from specific number

※ Please remind that all of call except the number listed on White List is not available when white list function is activated

LCR(Least Cost Routing)

2. LCR(Least Cost Routing)

Black List & White List
: The function to reject and to accept calling for specific number

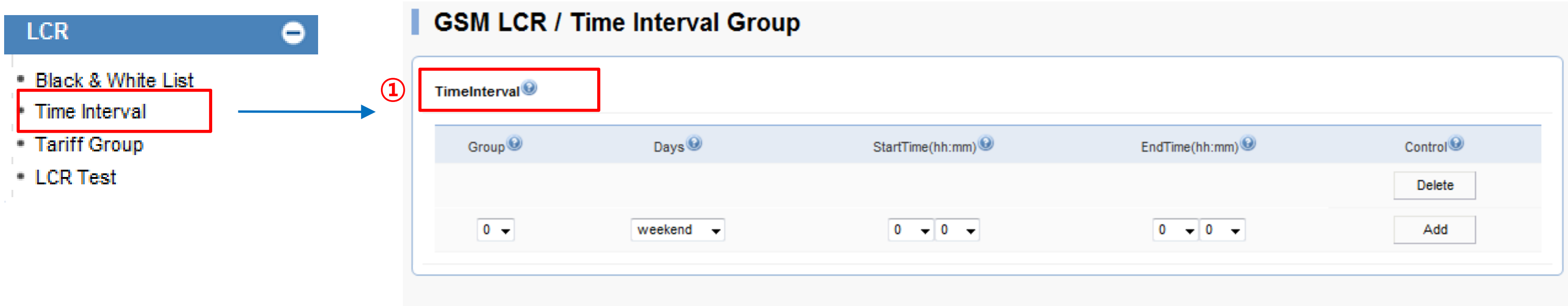
GSM LCR Time Interval
: The function to allow user to make call on GSM networks at specific time

GSM LCR Time Tariff
: The function to check the available time and used time. Set Restore Call limit

GSM LCR Simulator
: The function to test call function to virtual number via WEB GUI

LCR(Least Cost Routing)

2. Time Interval-1



① Time Interval : Set date and time to use LCR
- Group : Set Time Group (Default : 0)
- Days : Set day to apply LCR
  (weekdays / weekend / Monday / Tuesday / Wednesday / Thursday / Friday / Saturday / Sunday)
- Start Time : Set time to start (hh:mm)
- End Time : Set time to end (hh:mm)

LCR(Least Cost Routing)

2. Time Interval-2(Example)

GSM LCR / Time Interval Group

TimeInterval

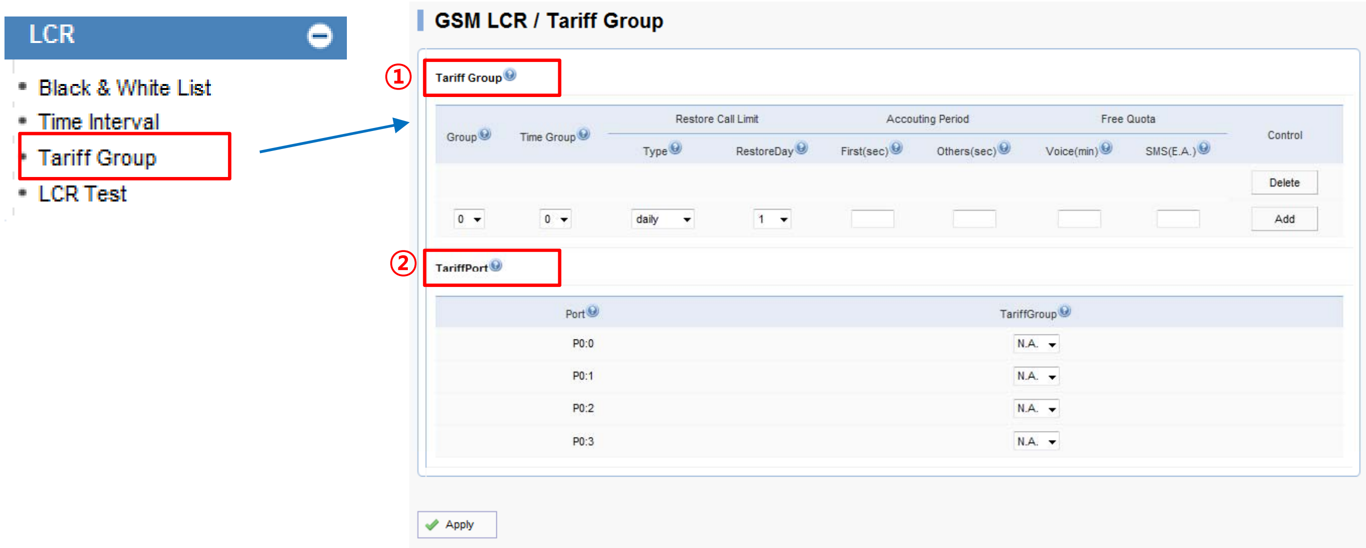
Group	Days	StartTime(hh:mm)	EndTime(hh:mm)	Control
0	Weekdays	09:00	18:00	☐
1	SUN	10:00	13:00	☐
1	SAT	10:00	13:00	☐
				Delete
0	weekend	0 0	0 0	Add

Description
Group 0 : Available time is Monday to Friday, 9AM to 6PM

Description
Group 1 : Available time is Saturday and Sunday, 10AM to 1PM

LCR(Least Cost Routing)

3. Tariff Group-1

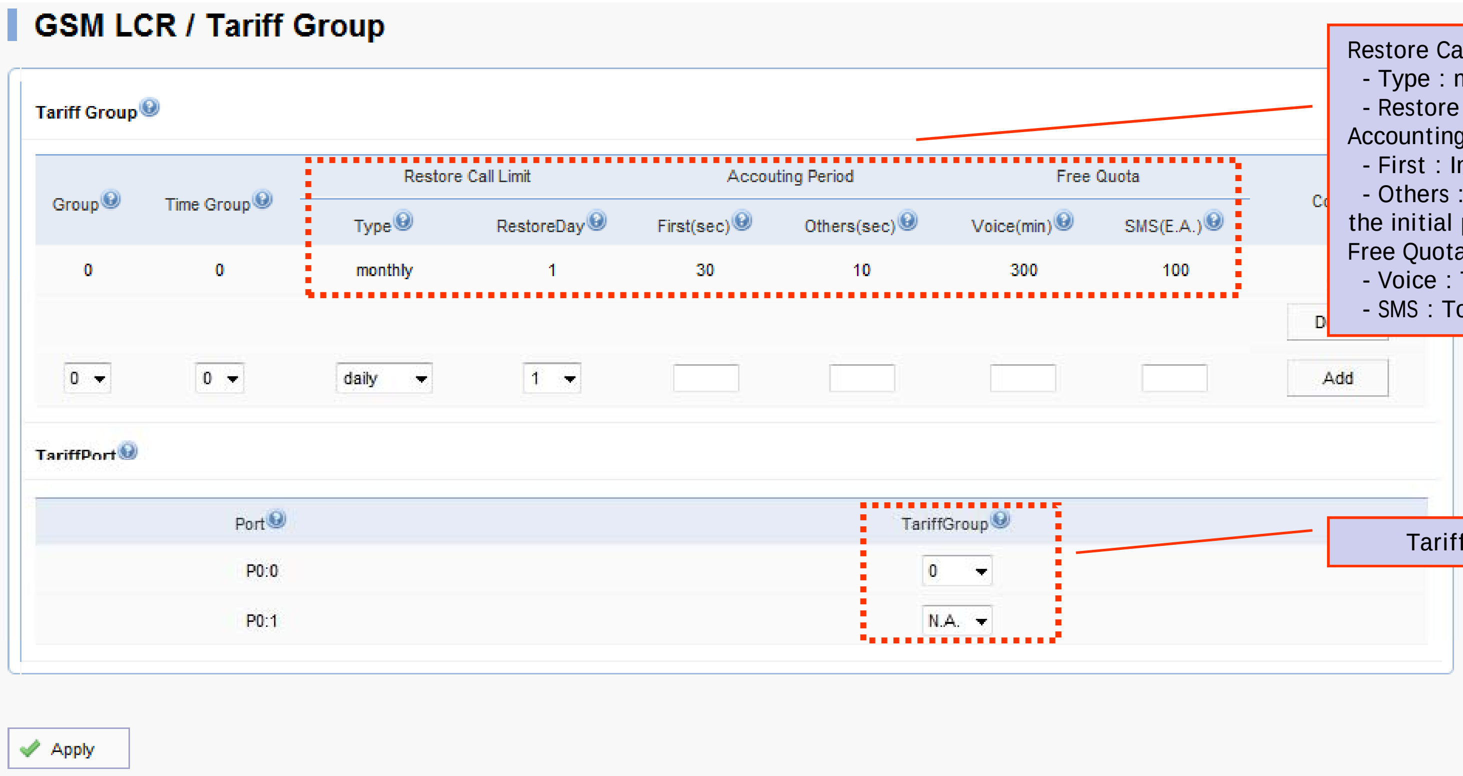


① Tariff Group : Set Time Group and toll-free limitaion
- Group : Generate group
- Time Group : Select group generated at Time Interval
- Restore Call Limit : Set point to restore set limitation
- Accounting Period : Tim period to charge (sec)
- Free Quota : Set toll-free time (min)

② Tariff Port : Apply information on Tariff Group to specific port

LCR(Least Cost Routing)

3. Tariff Group-2 (Example)



Restore Call Limit
- Type : monthly or daily
- Restore Day : Set date

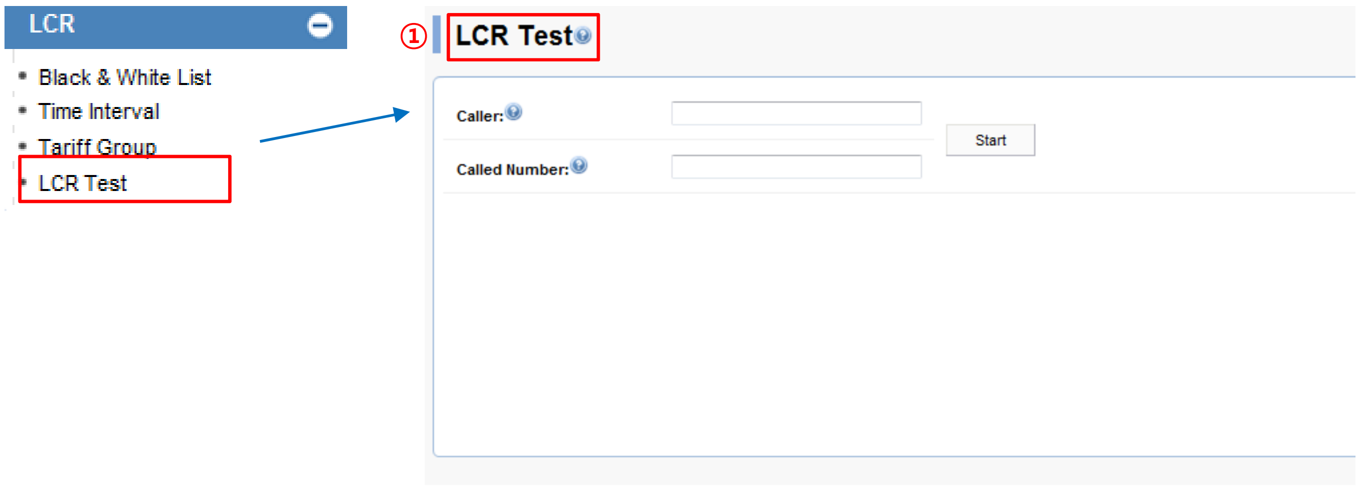
Accounting Period
- First : Initial period to charge
- Others : Second period to charge after the initial period

Free Quota
- Voice : Toll-free call time
- SMS : Toll-free SMS

Tariff Group : Set group to port

LCR(Least Cost Routing)

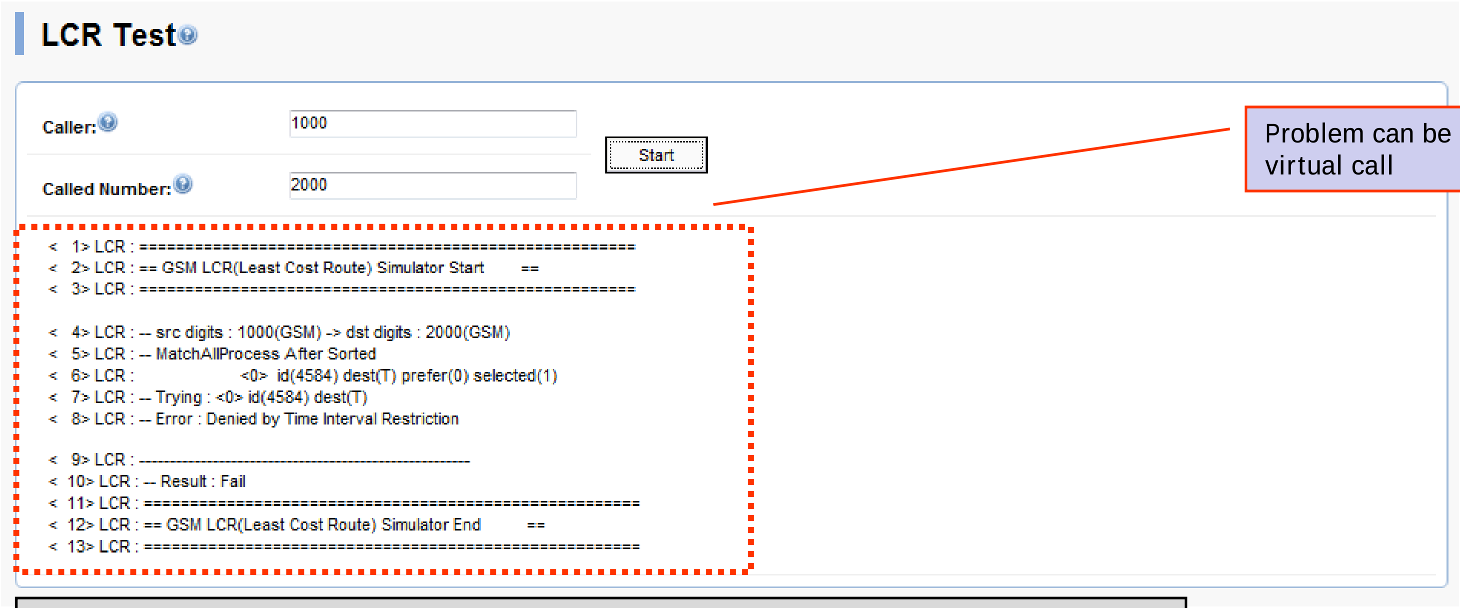
4. LCR Test -1



① LCR Test : The function to make call to virtual number for testing
- Caller : Enter number of sender
- Called Number : Enter number of receiver
- GSM Networks status can be monitored to make virtual call

LCR(Least Cost Routing)

4. LCR Test -2 (Example)

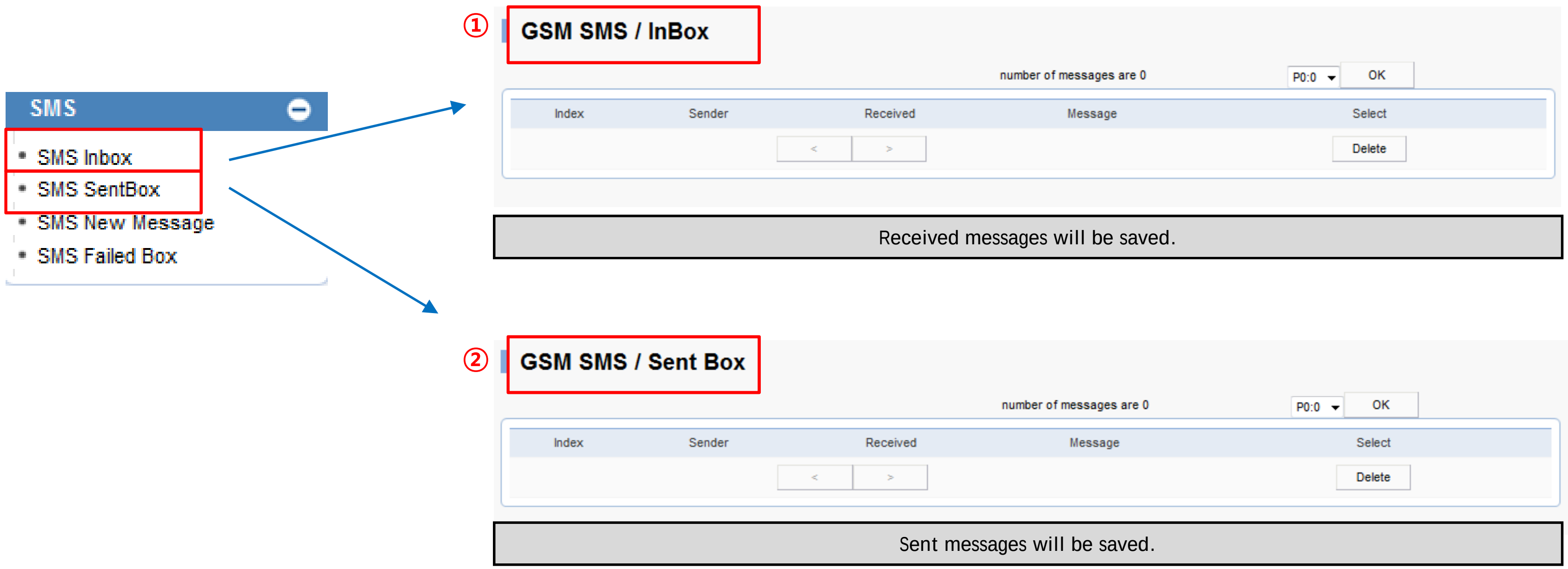


Problem can be monitored to make virtual call

The above message is shown that failure occurred by time interval

SMS

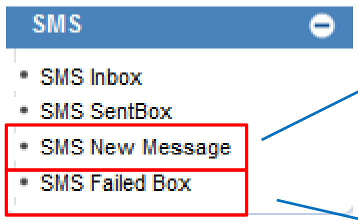
5. SMS -1

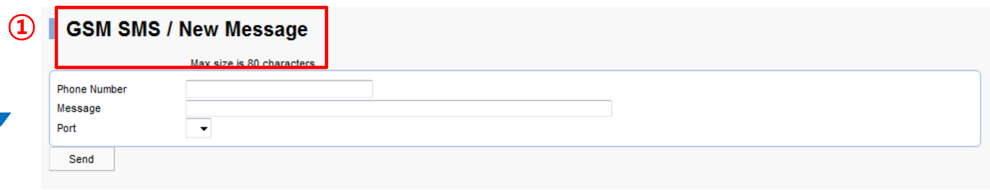


※ The number of message can be stored in InBox, Sent Box and Fail Box is 17

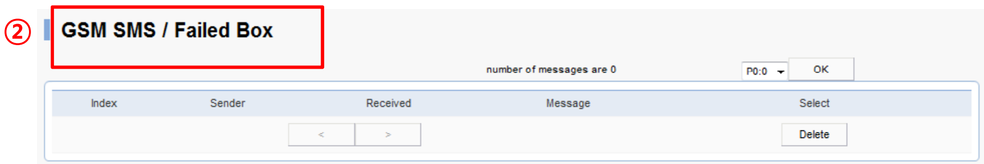
SMS

5. SMS -2





You may send the messages.



Failed messages will be saved.

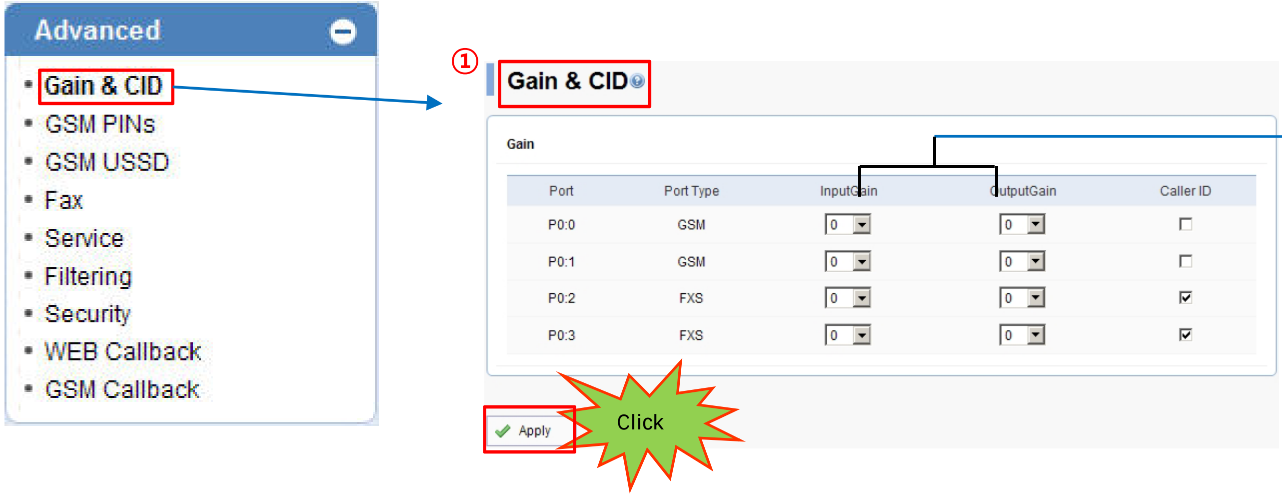
※ **SMS Support Language : Korean, English, Russian, Spanish and Portuguese**

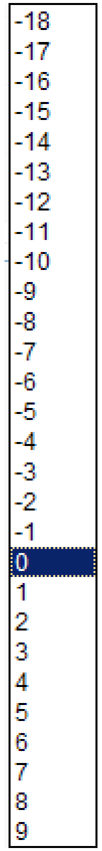
Advanced Service

1. Gain & CID
2. GSM Pins
3. GSM Band
4. BTS

Advanced Service

1. Gain & CID





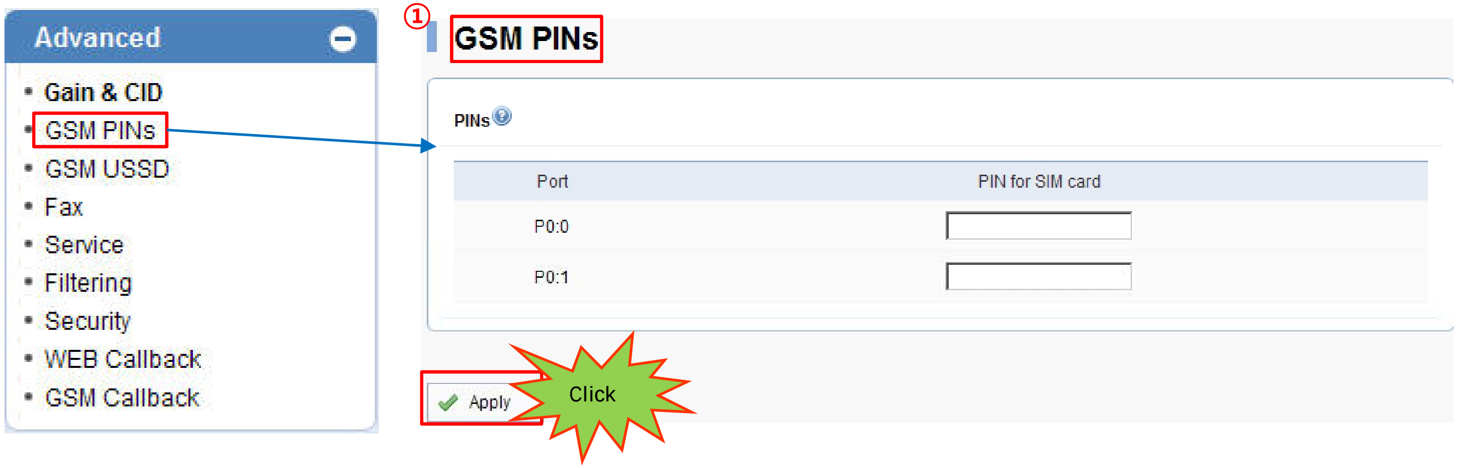
① Gain & CID : Adjustment output voice level of each port(GSM, FXS)
(You may reduce the level when echo and noise occured)
In addition, call number can be detected by Caller-ID

- 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

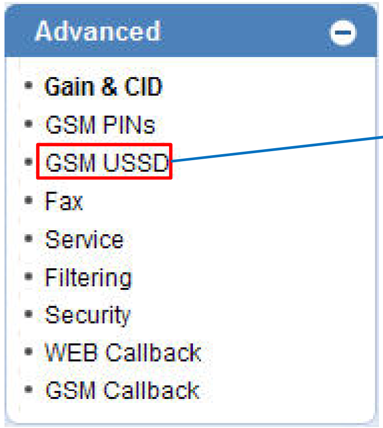
Advanced Service

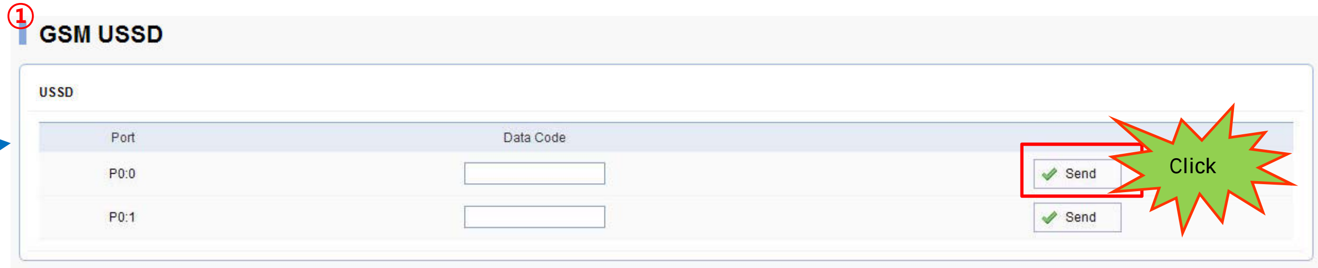
2. GSM PINs



Advanced Service

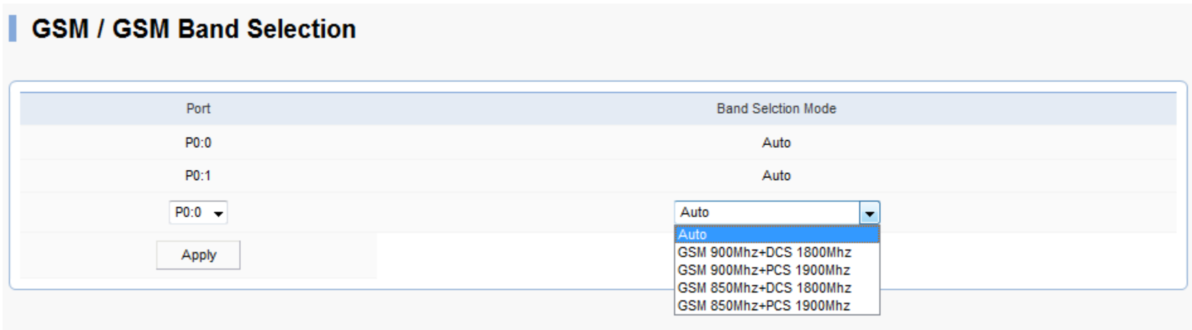
3. GSM USSD





Advanced Service

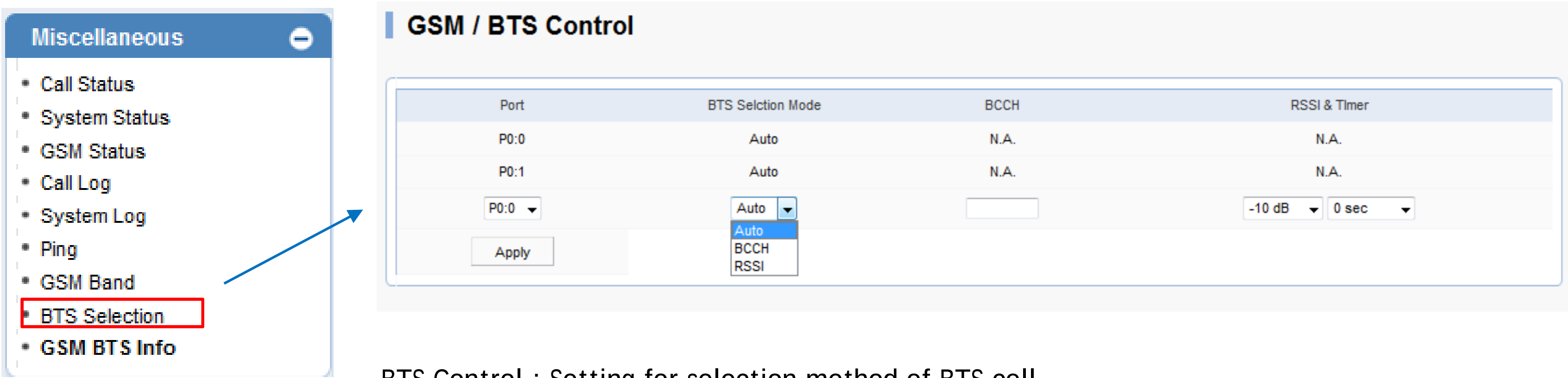
4. GSM Band



GSM Band Selection : Setting for bandwidth of GSM Networks

- Auto (Default)
- 900Mhz + DCS 1800Mhz
- 900Mhz + PCS 1900Mhz
- 850Mhz + DCS 1800Mhz
- 850Mhz + PCS 1900Mhz

4. BTS(Base Terminal Station) -1



BTS Control : Setting for selection method of BTS cell
- Auto (Default) : The most highest power cell will be selected
- BCCH : Choose specific cell by entering BCCH value manually
- RSSI : Choose cell which has specific RSSI level
  Cell will be selected in accordance with set interval

Advanced Service(etc)

1. NAT Static Table
2. NTP
3. File Browser
4. FAX
5. Service
6. Filtering
7. Security
8. Language

Advanced Service (etc)

1. NAT Static Table

① NAT Static Table

- You may setup IP Protocol (TCP, UDP). You may send the desirable data information (comes through Global Port) by changing (Local Address, Local Port)

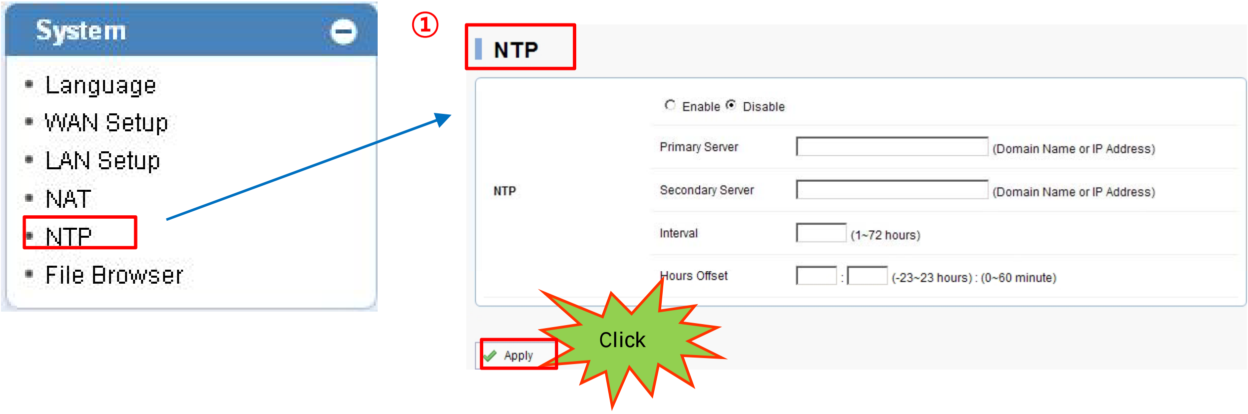
*** Example**

- **Global port : 5060**
- **Local Address : 10.1.1.1**
- **Local Port : 5070**

--> Forward by changing the information which comes through 5060 port to 10.1.1.1, 5070 port.

Advanced Service (etc)

2. NTP

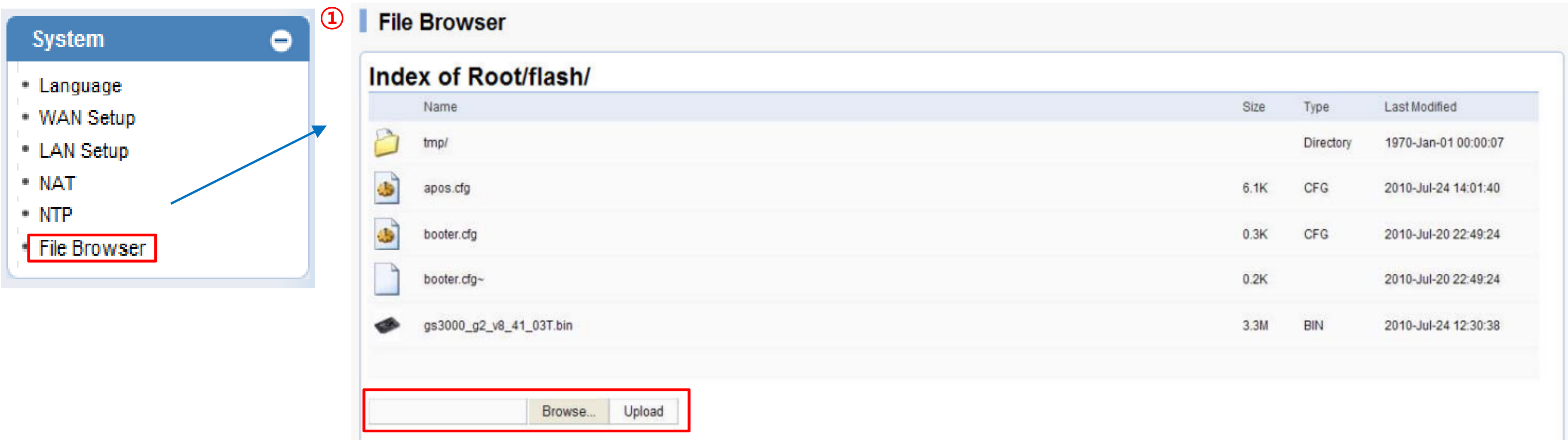


① NTP : Input information of NTP Server
- Click the apply button for NTP activation
- Primary Server : Input IP or domain name of NTP Server
- Interval : Interval to request and receive data from NTP server

* Please click the apply button after set up

Advanced Service (etc)

3. File Browser



① File Browser

You may download or upload GSM Gateway image file.

- Download : Click on image file then download will be processed.
- Upload : Click browse to search image file. Click on upload button then image file upload will be processed.

Advanced Service (etc)

4. FAX

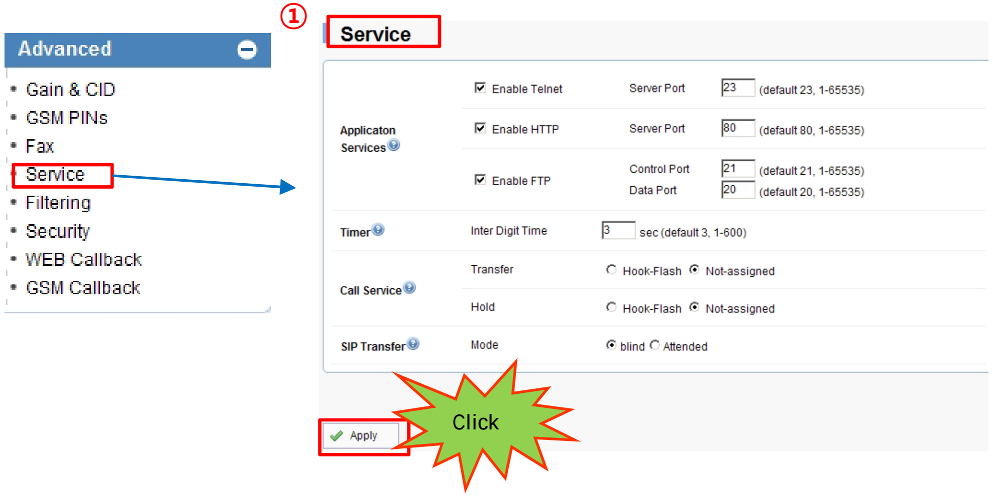
① Fax : Setting the property of FAX mode

- 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

Advanced Service (etc)

5. Service



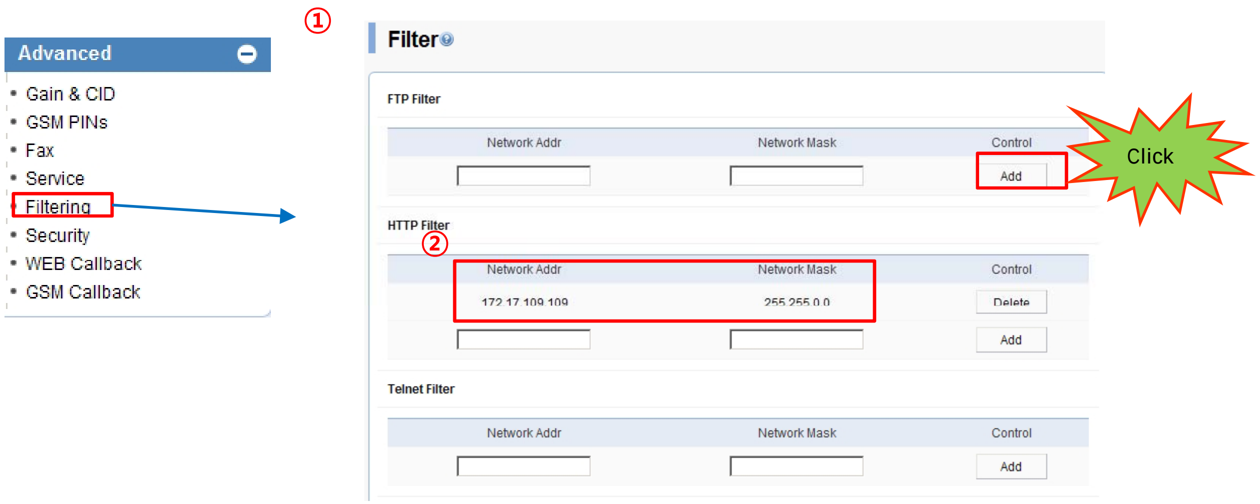
① Service : Set extra features

- Application Services : The port setting for each Telnet , HTTP, FTP
- Timer : Adjust digit time for phone connected to GSM G/W
  Set time limitation between Digit and Digit
- Call Service : Call-Transfer Set Activation and Call-Hold function
  (Hook-Flash – Activate , Not-assigned – Inactivate)
- Transfer Mode : GSM G/W supports blind mode and attended mode. To use this function, call transfer mode must be activated.

* Please click the apply button after set up

Advanced Service (etc)

6. Filtering



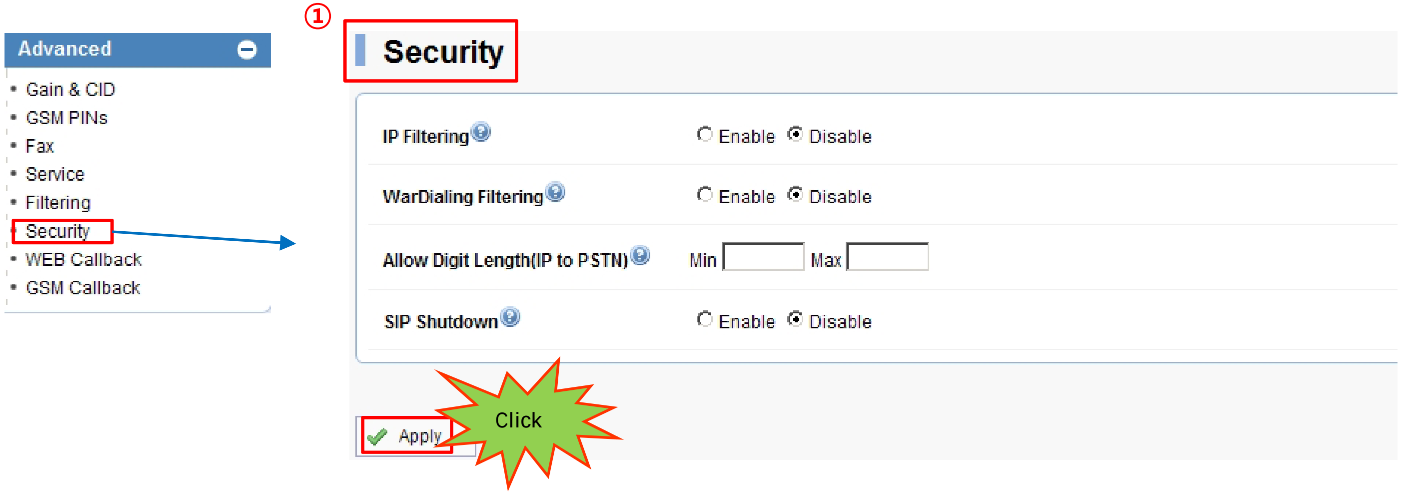
① Filter : Setting IP address authorized by administrator for connection

- 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

Advanced Service (etc)

7. Security

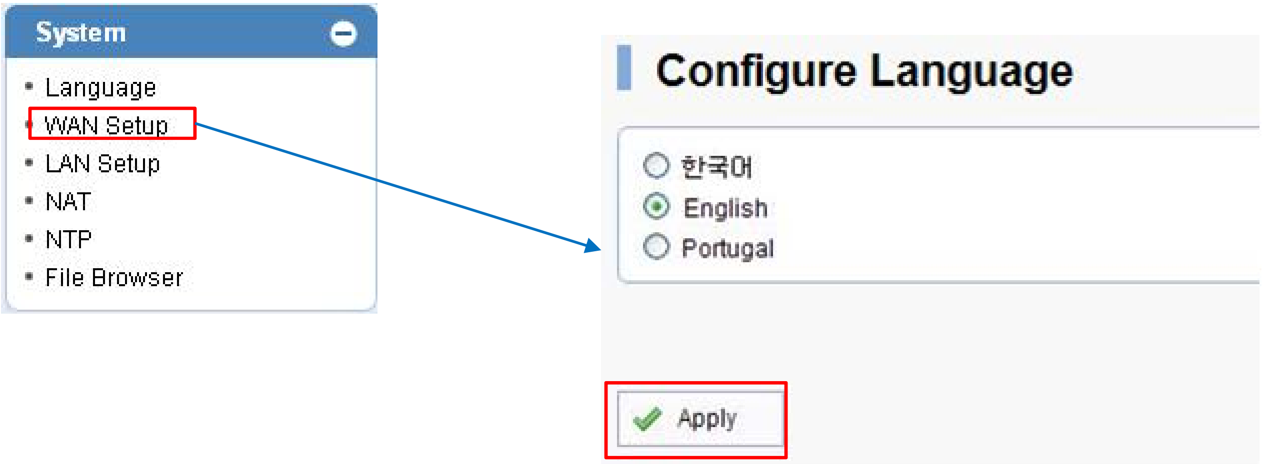


① Security : Set security to block unauthorized call
- IP Filtering : The only call made from the device with IP address listed on GSM G/W is available to make call
- War Dialing Filtering : The only receiving call listed on dial plan is available to make call
- Allow digit Length(IP to PSTN) : The only 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

Advanced Service (etc)

8. Web Connection via Console Port



① You may choose three different languages.
- Korea
- English
- Portugal

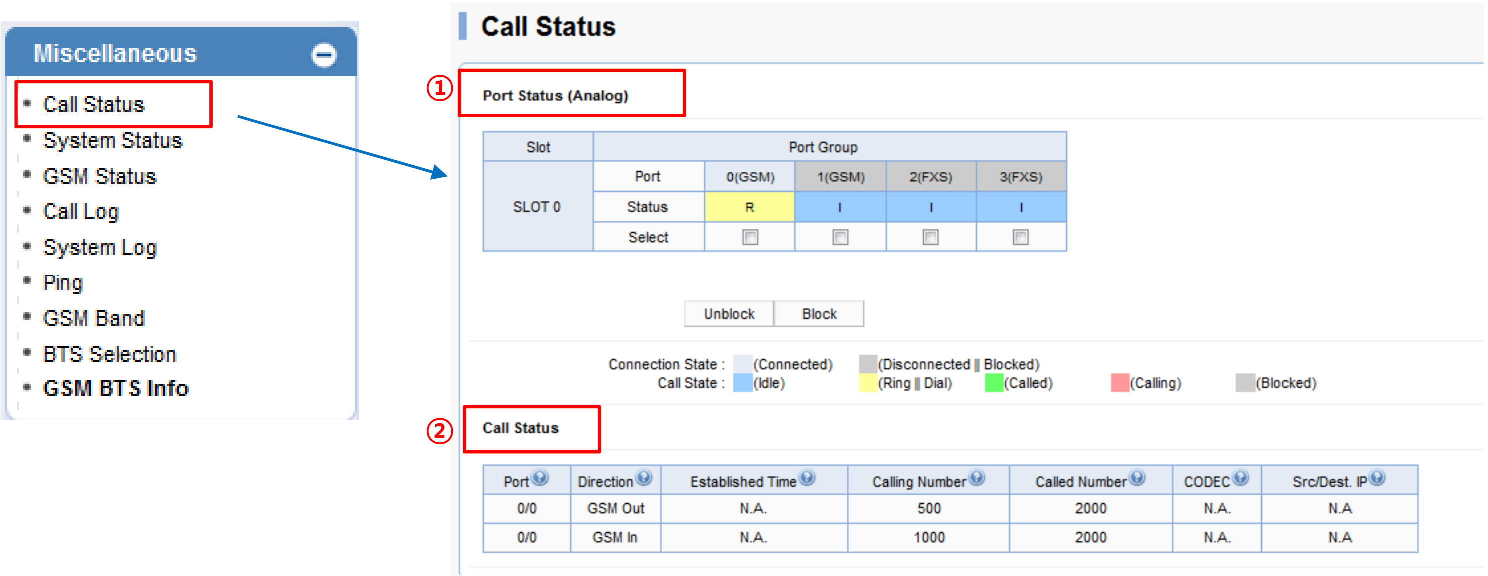
* Language setup will be applied immediately once you click on apply.

Monitoring

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

Monitoring

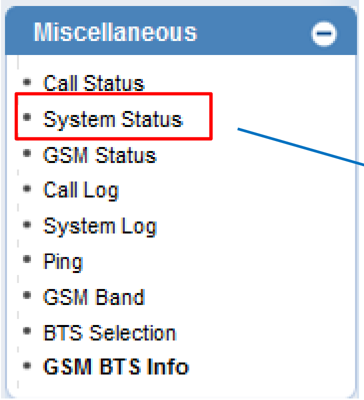
1. Call Status

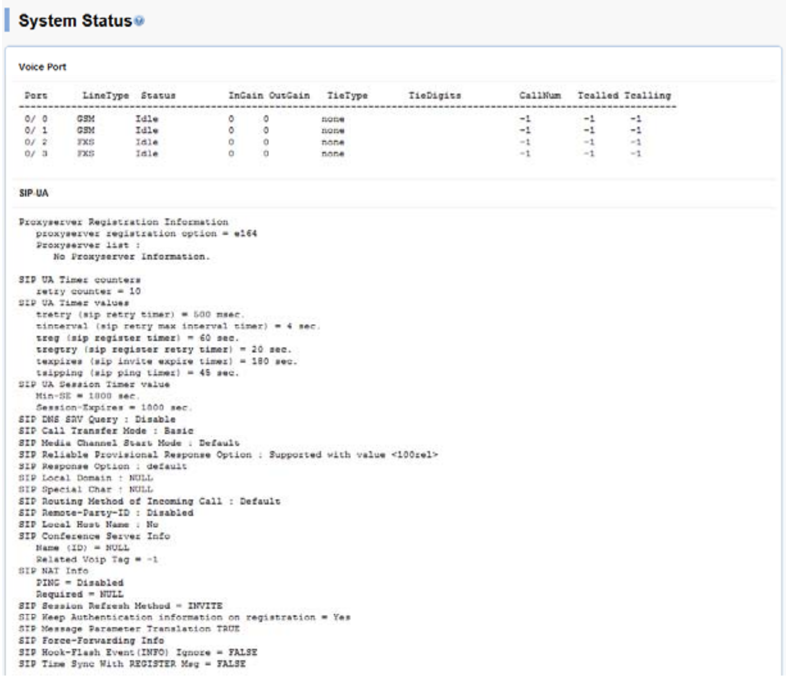


Call Status : The status of GSM G/W port and call can be monitored in real-time
① Port Status : Monitoring GSM G/W port
② Call Status : Monitoring call status

Monitoring

2. System Status

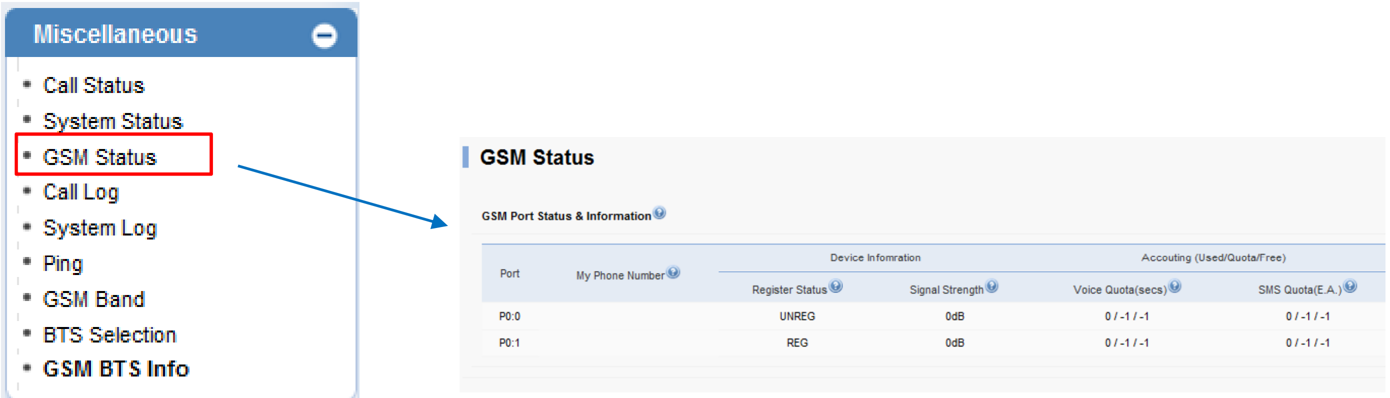




System Status : The system status of GSM G/W can be monitored

Monitoring

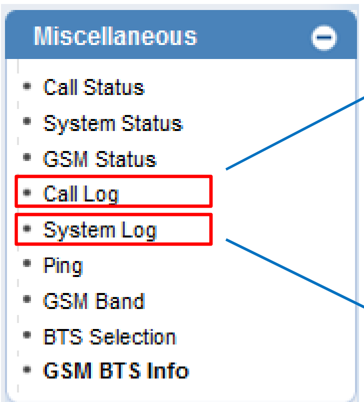
3. GSM Status

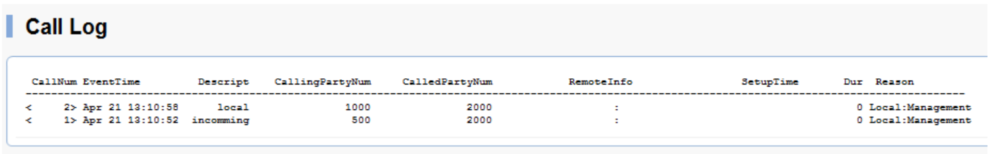


GSM Status : GSM Networks status, Usage can be monitored

Monitoring

4. Call Log / System Log

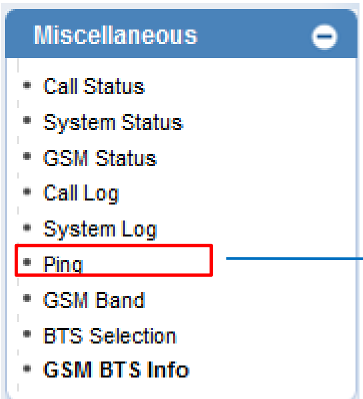


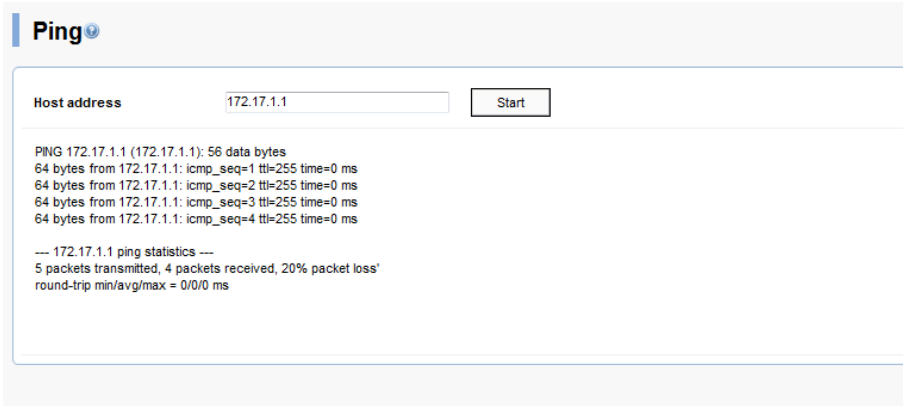


Call Log : Monitoring all of call history
※ Call history will be clear with rebooting

System Log : Monitoring GSM G/W 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)

5. Ping





Ping : Network status can be checked by pinging

Thank you