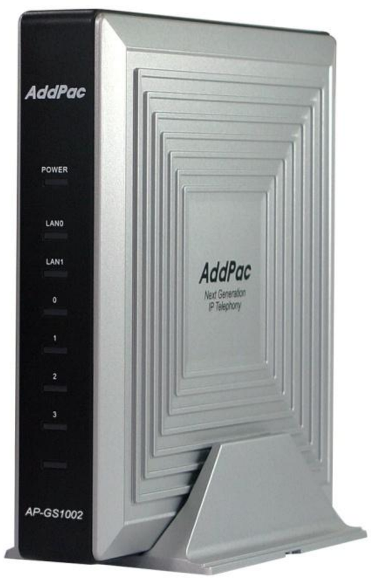
AP-GS1002™
2-Port FXS GSM Gateway

High Performance GSM Gateway Solution

WEB Setup 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

Console

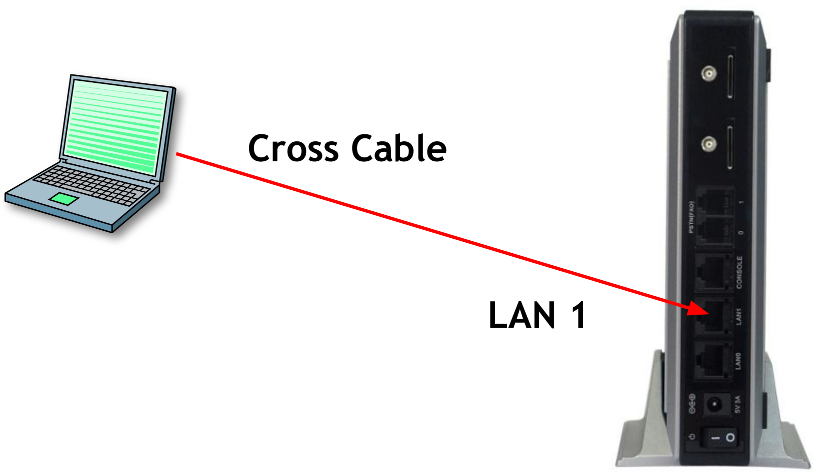
1. Connect to GS1002 via console port to enter IP address in order to set Interface 0/0
2. Input IP address of interface 0/0 using below command
3. Connect to the IP address via web after saving

login ID/Password : root/router

```
GS1002> enable                                   => Enter the enable mode
GS1002# configure terminal                       => Enter the configuration mode
GS1002(config)# interface FastEthernet 0/0
GS1002(config-if)# ip address <IP Address> <Subnet Mask>      => Ex) ip address 172.17.109.1 255.255.0.0
GS1002(config-if)# exit
GS1002(config)# ip route 0.0.0.0 0.0.0.0 <Defualt Gateway>    => Ex) ip route 0.0.0.0 0.0.0.0 172.17.1.1
GS1002(config)# write                            => saving
Proceed with write? [confirm]y
```

WEB Connection

2. Connection Web via LAN 1 Port



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

1. It is the way to connect to AP-GS1002 via LAN 1 port

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

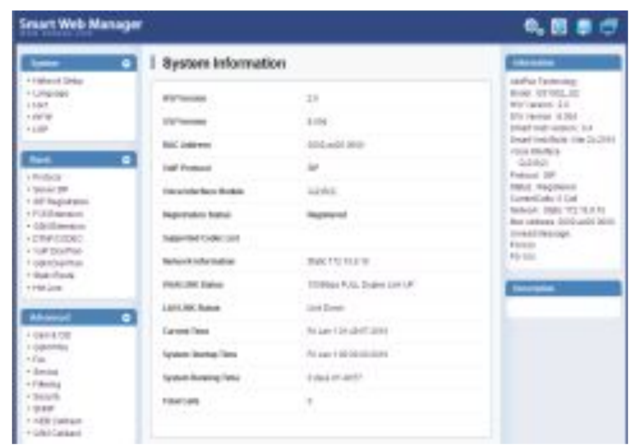
3. After set PC with same IP address subnet, connect to AP-GS1002
   - Connect PC to AP-GS1002 using Cross UTP-Cable. You may use Ethernet switch with normal UTP-cable
   - Enter IP address 192.168.10.1 on your web browser

WEB Connection

1. The screen of Web connection

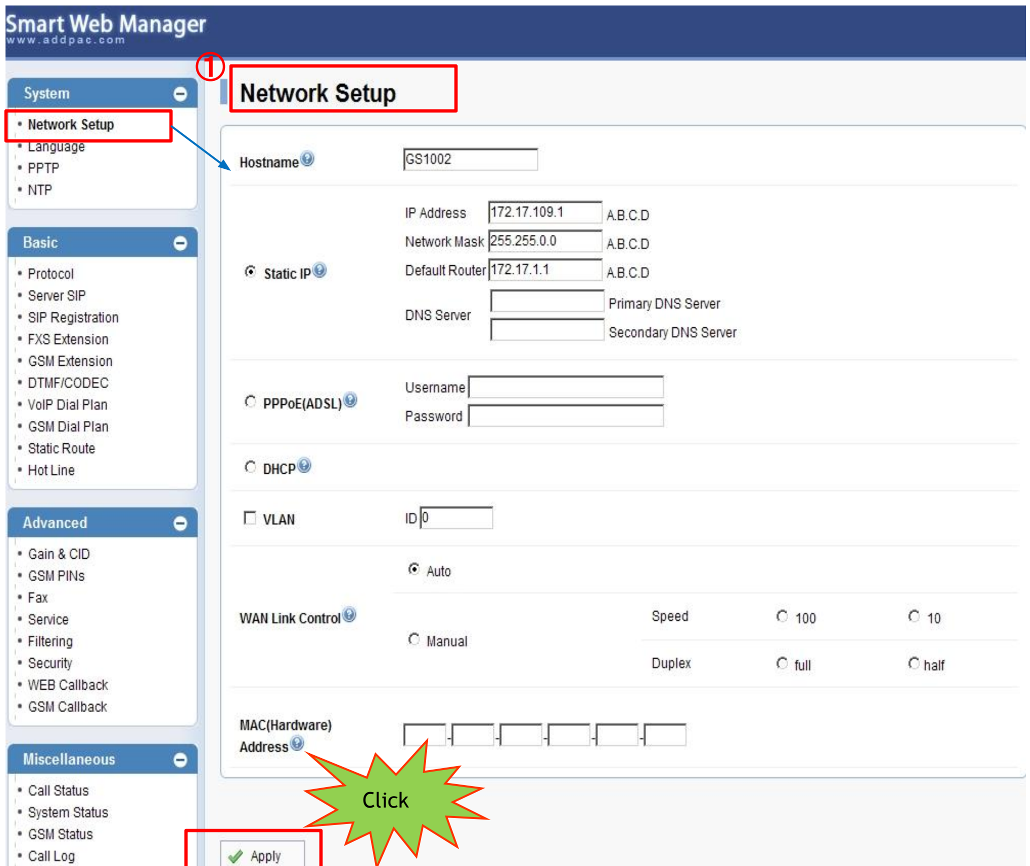
- The Shown log-in screen is connection to Web page.
  Please enter the below log-in information

  ID : root
  Password : router



Network Setup

1. Network Setup Screen



① Network Setup

-Hostname : Enter the device name of AP-GS1002

-Static IP
-PPPoE(ADSL)
-DHCP

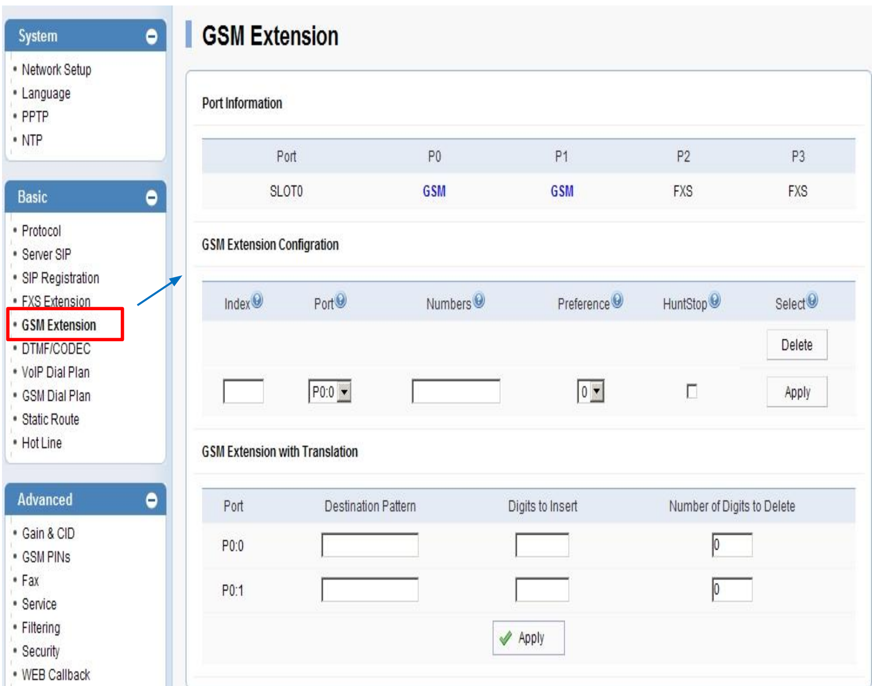
*** Please make sure to press the apply button for saving**

GSM Setup

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

GSM Setup > GSM Extension

1. 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 > GSM Extension

1. GSM Extension -2

GSM Extension

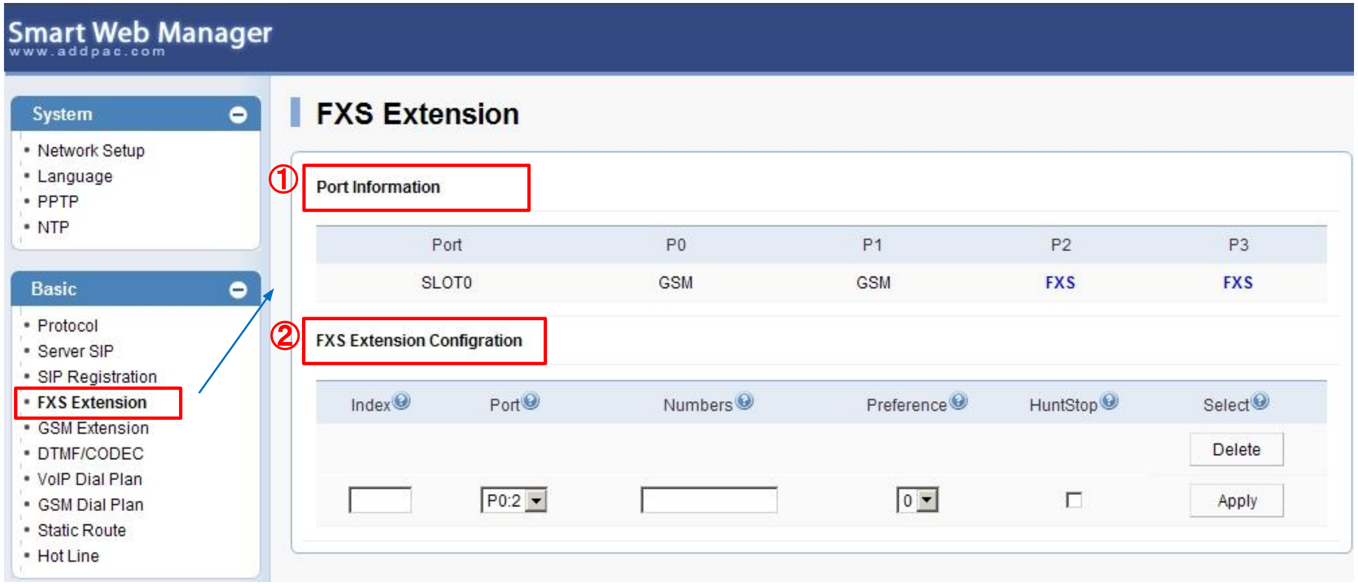
Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

Port Information
It displays the information of AP-GS1002 SLOT. You can refer to the above picture

GSM Setup > FXS Extension

2. FXS Extension -1

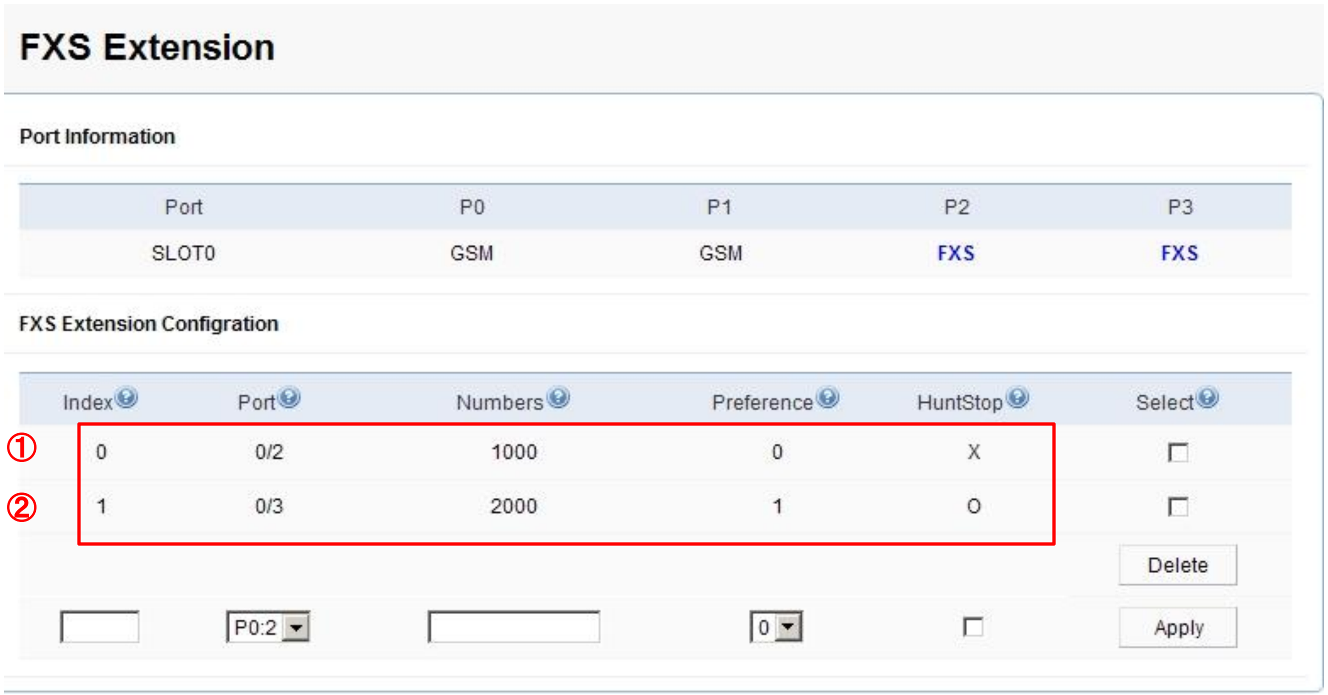


① The each port information of AP-GS1002

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

1. FXS Extension -2 (Example)



① Set the number to be used for FXS 0/2 port (ex. 2000)
② Set the number to be used for FXS 0/3 port (ex. 3000)

• 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

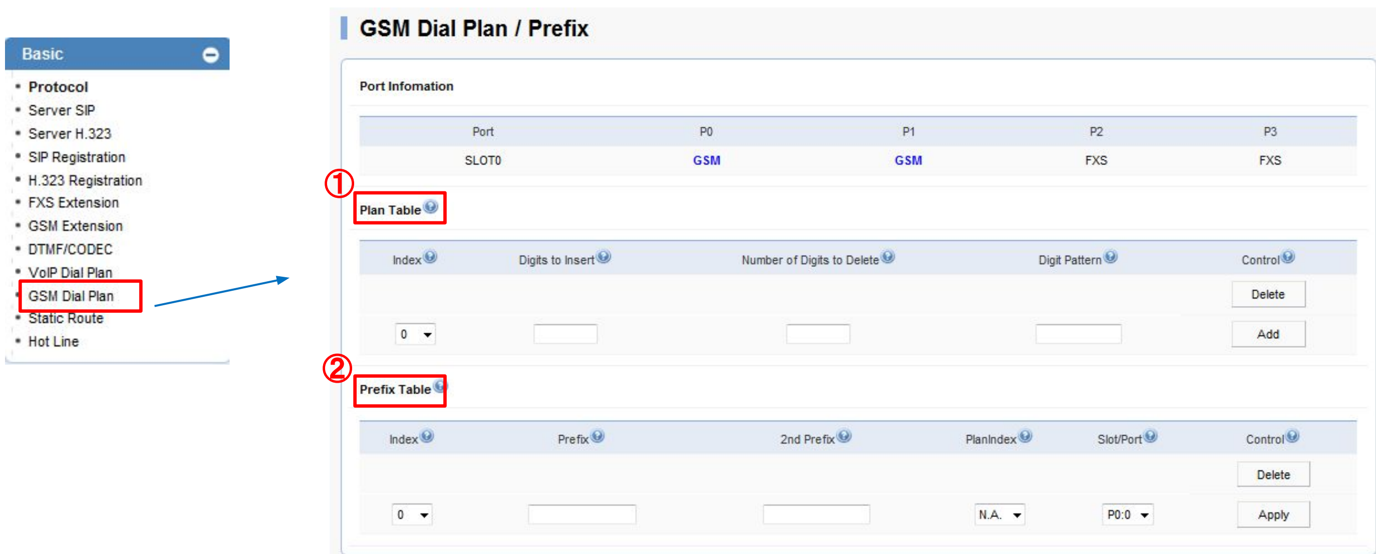
** T : Accept all number entered within Inter Digit Time (Default IDT : 3sec)
** Dot(.) : One dot(.) means one digit
** [ ] : The range of number

※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..

GSM Setup > Dial Plan

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 2^{nd} stage using, the Prefix can be used a number for hearing the Second dial-tone

GSM Setup > Dial Plan

GSM Dial Plan / Prefix - 2 (Example)

Plan Table

①

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0	82	1	025683848	☐
				Delete
0				Add

Prefix Table

②

Index	Prefix	2nd Prefix	PlanIndex	Slot/Port	Control
0	T		N.A.	0/0	☐
1	9	T	N.A.	0/1	☐
2	025683848		0	0/0	☐
					Delete
0			N.A.	P0:0	Apply

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

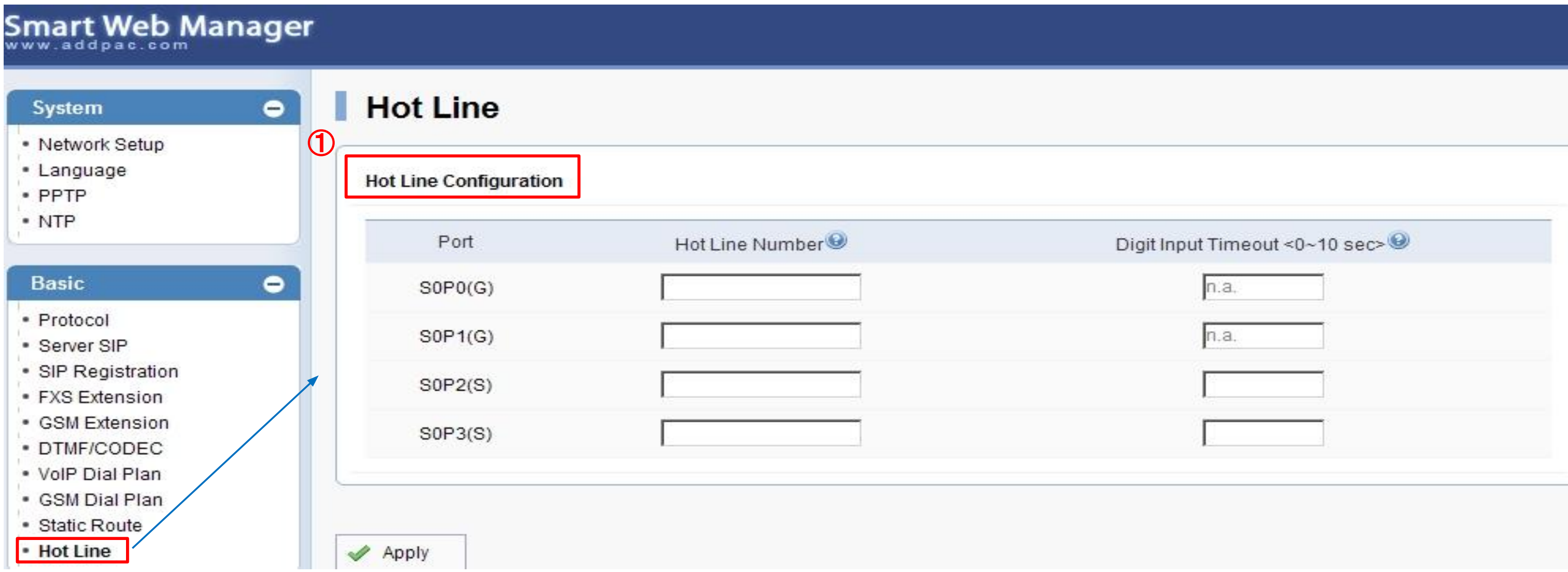
② Prefix , 2nd Prefix : Setting method of 1 stage and 2 stage

(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

Direct Incoming call

GSM Setup > Hot Line -1



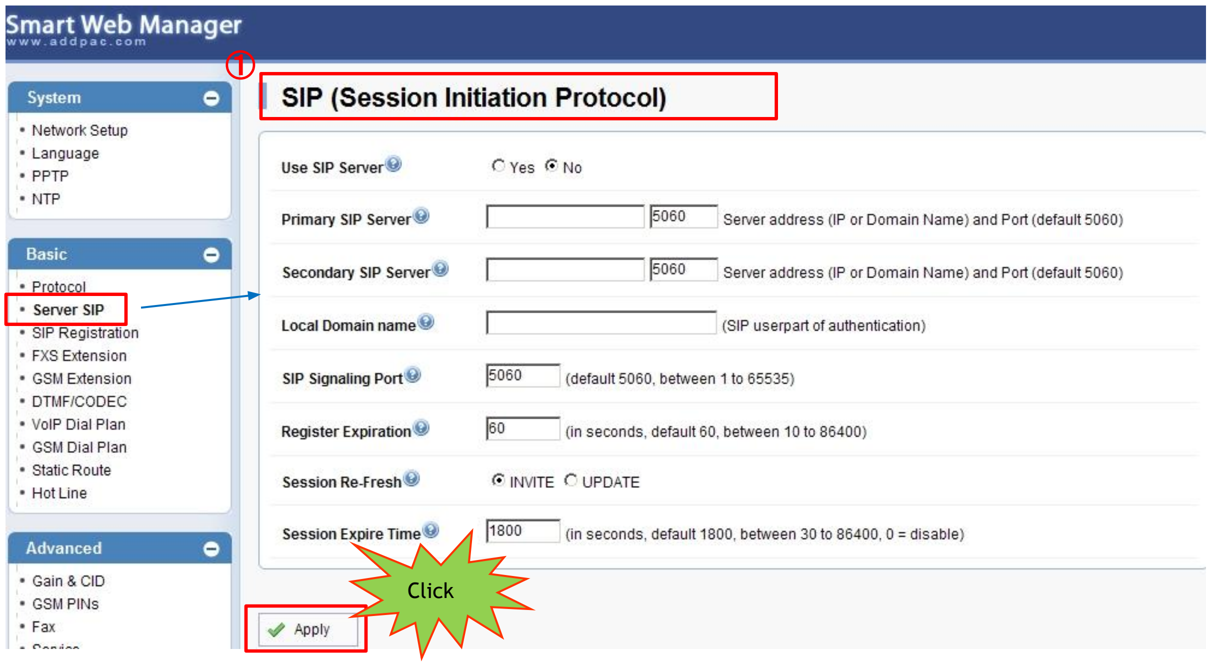
① Hot Line Configuration : Connect incoming and outgoing call directly
- Port : It means GSM and FXS port
- Hot Line Number : Forward call to entered number
  It connects to the number of GSM port in case of receiving call (Direct Incoming call)
- Digit Input Timeout : Time to make call to the Hot Line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

VoIP Setup

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

VoIP Setup

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

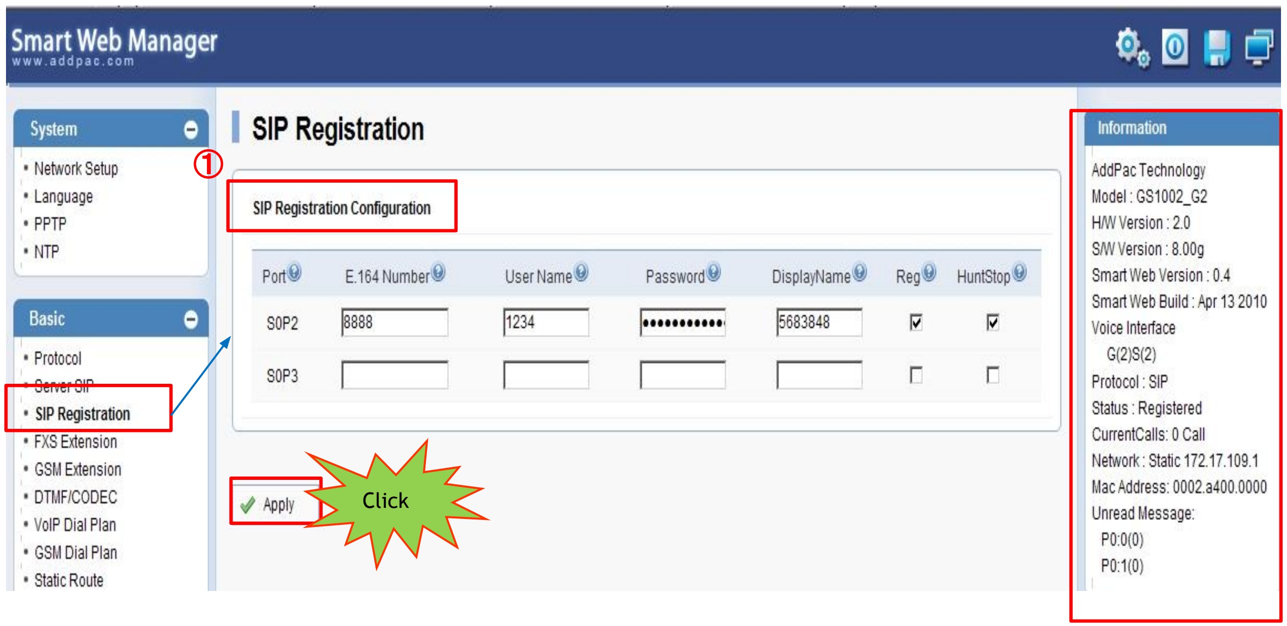
VoIP Setup

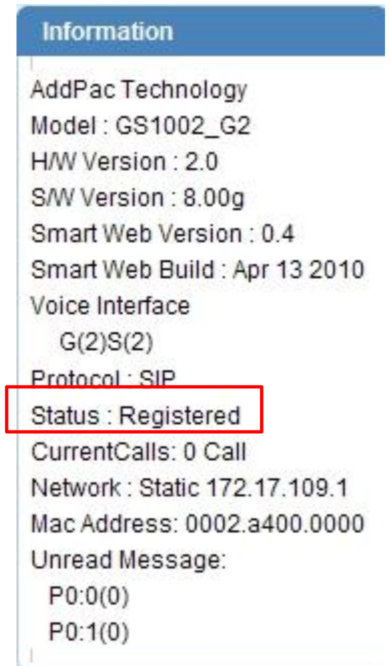
2. SIP Registration -1

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

2. SIP Registration -2 (Example)

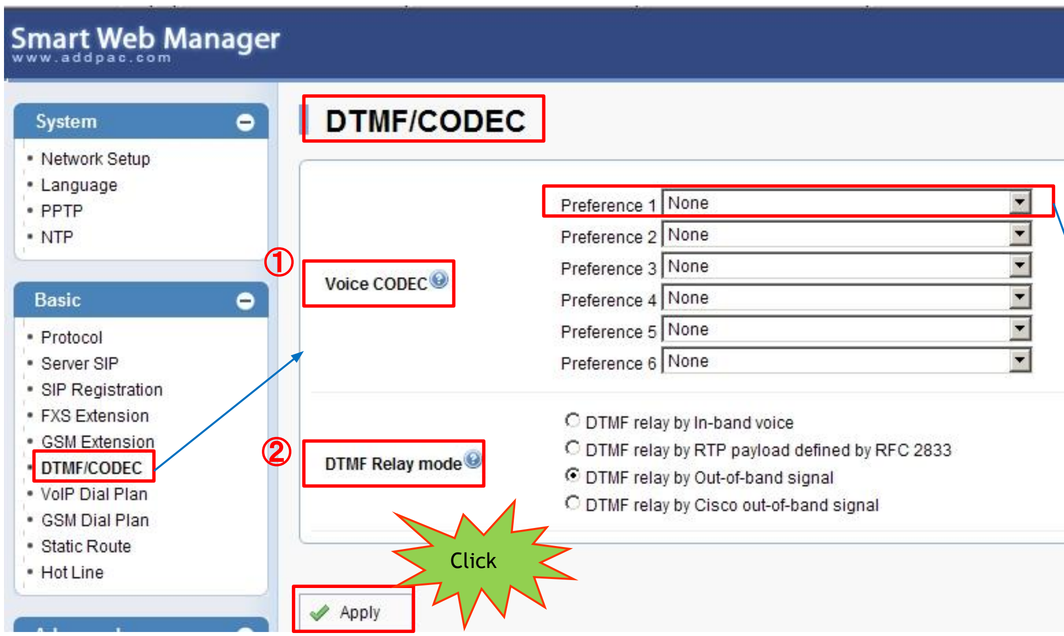




① Please click the apply button after enter information of SIP Registration
- You may check status of registration with reload web page using F5 key

VoIP Setup

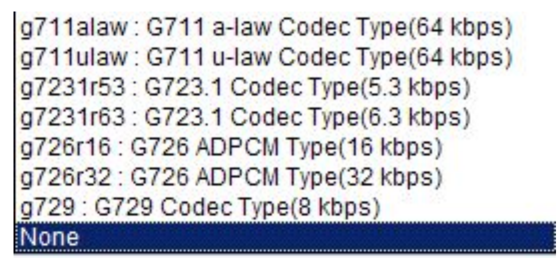
3. DTMF/CODEC



①Video Codec
-Select voice codec to be used

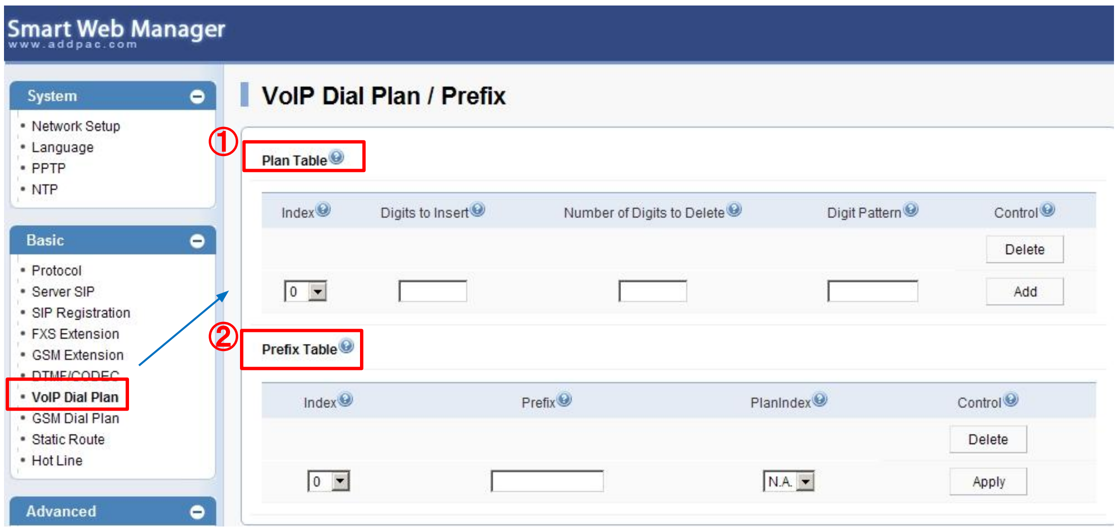
② DTMF Relay mode
-Select DTMF relay mode.
(Default : Out-of-band Signal)

* Please press the apply button to save



VoIP Setup

4. VoIP Dial Plan -1

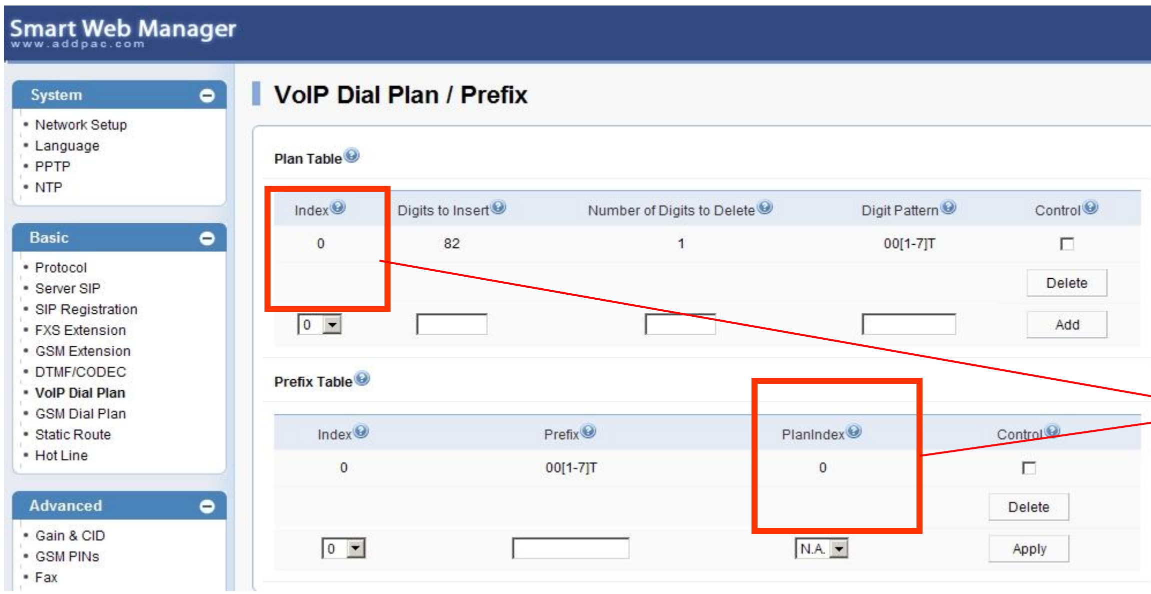


① Plan Table
- Digits to Insert : Number you want to enter
- Number of Digit to Delete : Number of digit to delete
- Digit Pattern : Number to apply for conversion

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

VoIP Setup

4. VoIP Dial Plan -2 (Ex)



It must be the same with PlanIndex Number

VoIP Setup

5. 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

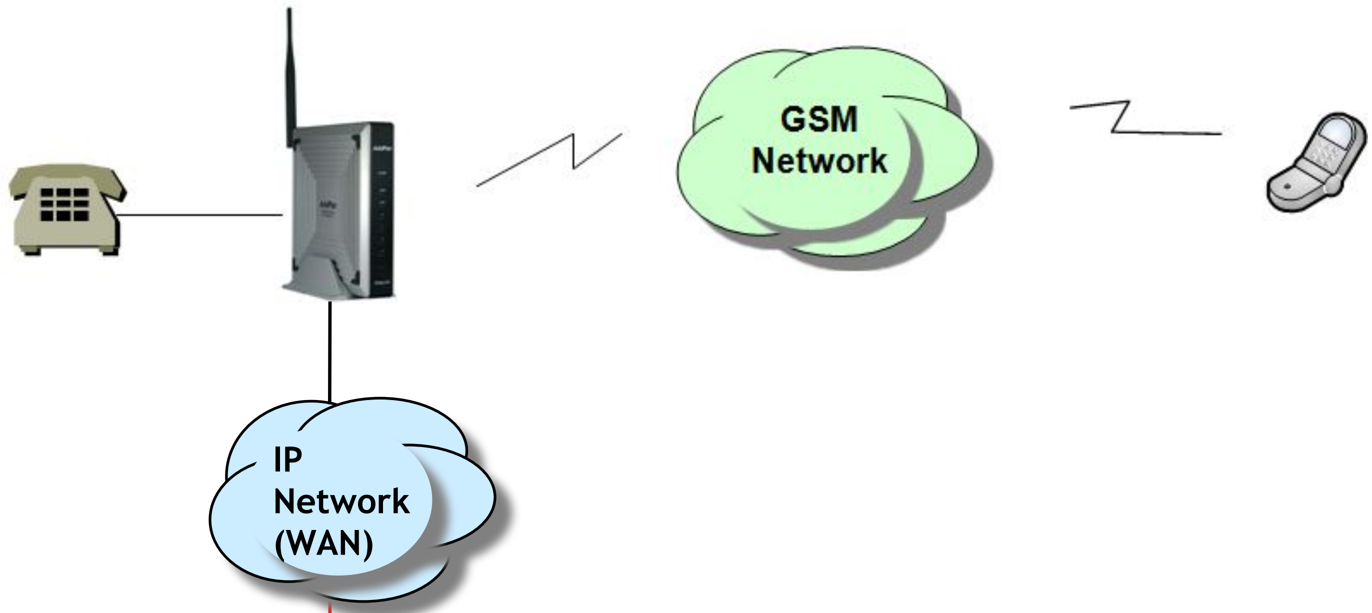
5. Static Route -2 (Ex)

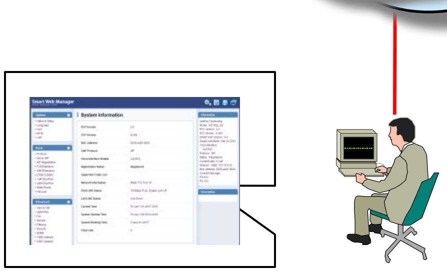
* Please press the apply button to save

Callback Service

1. WEB Callback Service
2. GSM Callback Service

1. WEB Callback Service

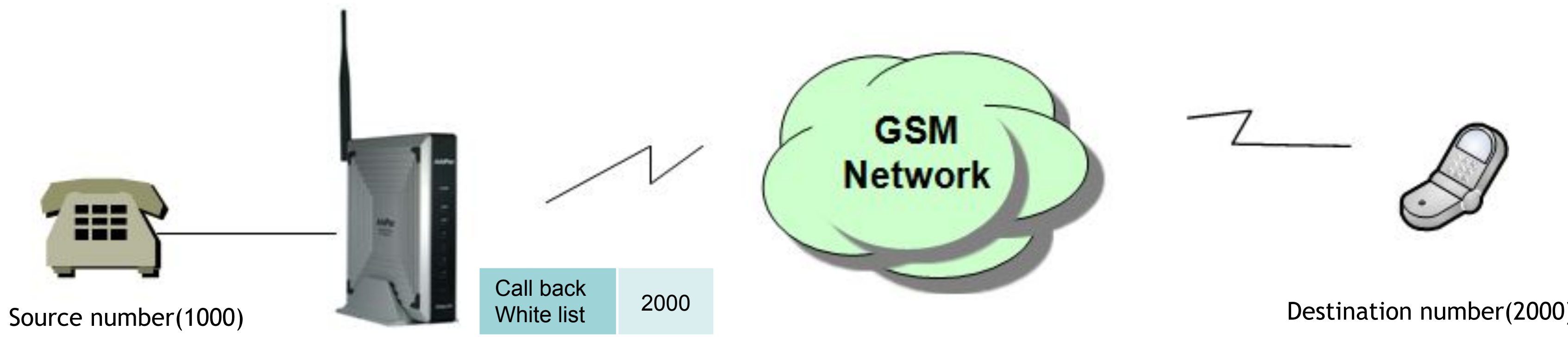




WEB Callback Service
: AP-GS1002 makes call between receiver and sender remotely via Web page.
WEB Callback calling number white list and Source number(call sender) must be the same

Callback Service

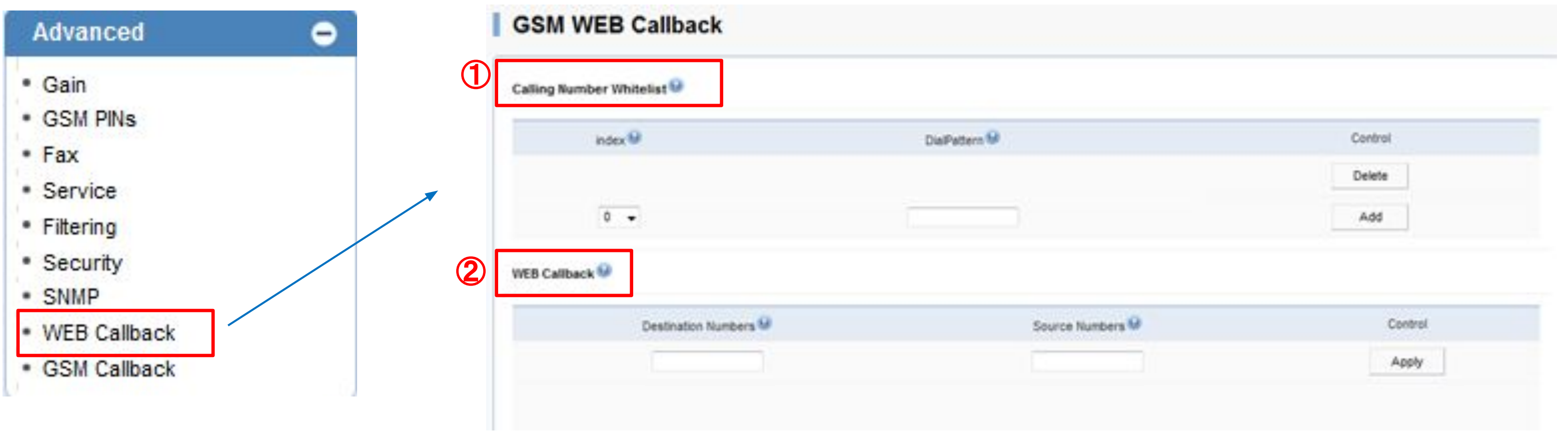
1. GSM Callback Service



GSM Callback Service
: The mobile user listed on the Callback white list can receive call back after disconnect the call by AP-GS1002

Callback Service > WEB Callback

1. WEB Callback Service -1



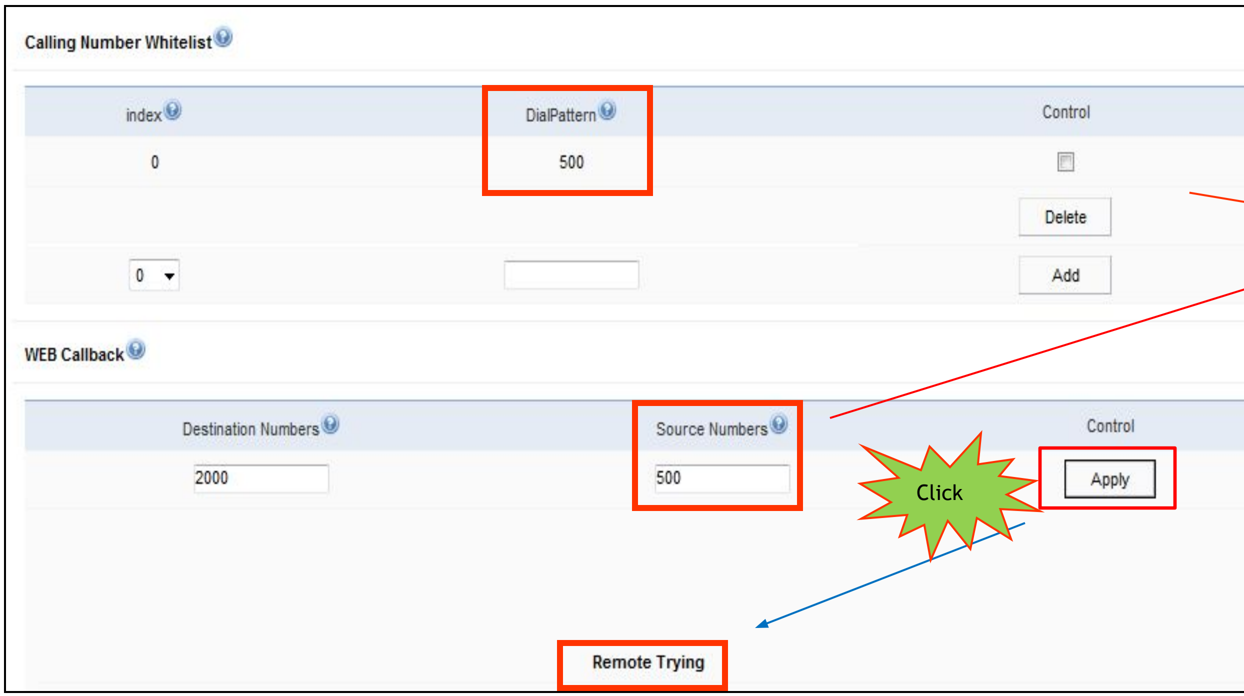
① Calling Number White List : Enter number of WEB Callback user
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Enter number to register to use WEB Callback Service

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 (Ex)

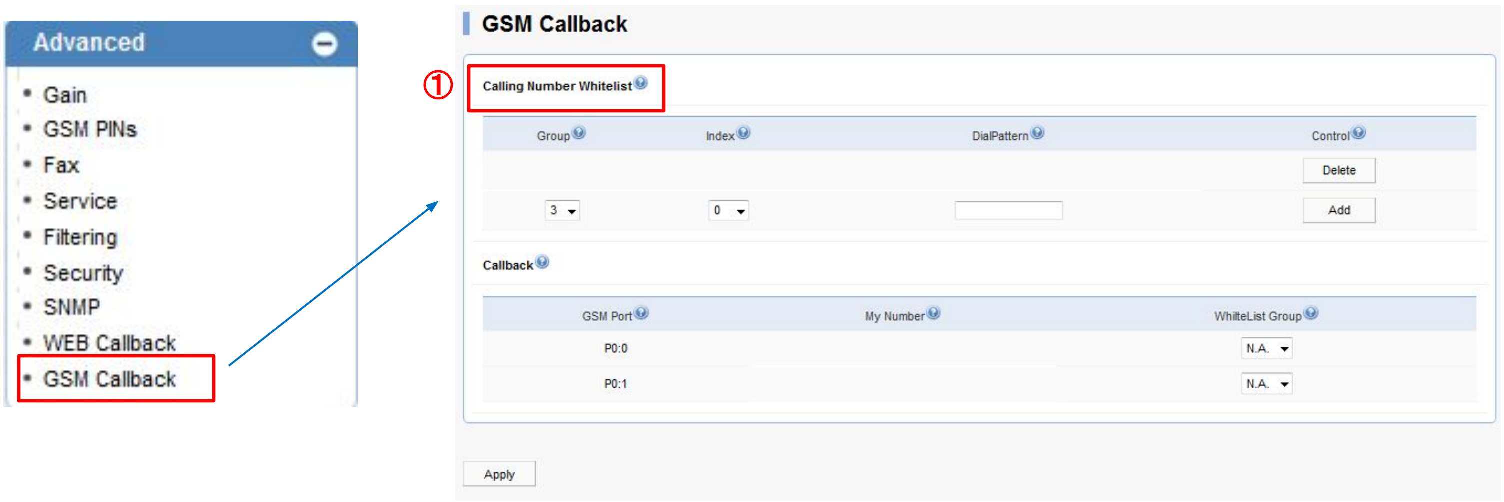


DialPattern must be the same as Source Number

Check status of calling with executing WEB Callback

Callback Service > GSM Callback

1. GSM Callback Service -1

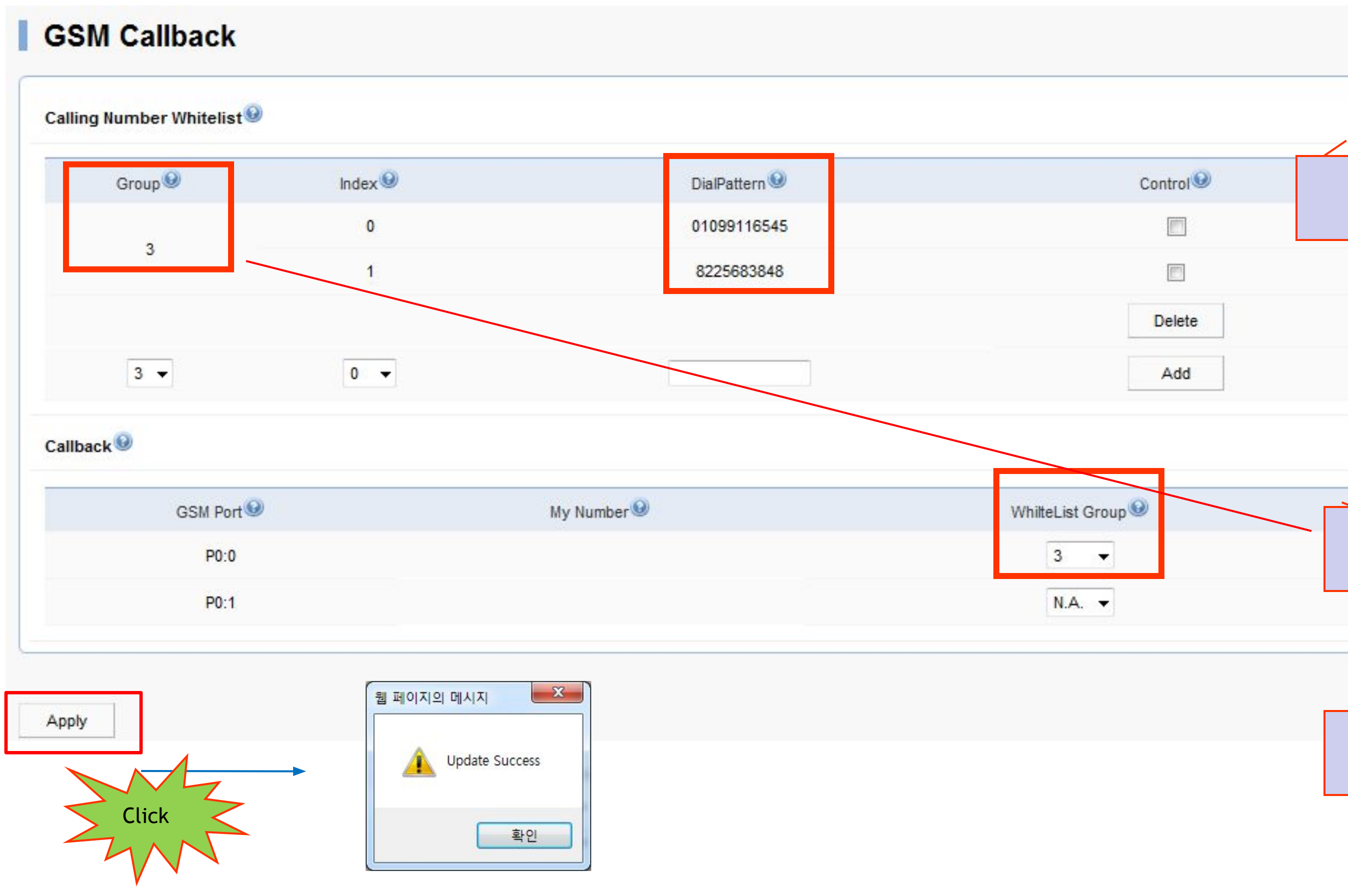


① Calling Number White List : Enter number to use GSM Callback
- Group : Enter Group Number (default : 3)
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Enter mobile phone number to use callback service

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

Callback Service > GSM Callback

1. GSM Callback Service -1 (Ex)



Register number to use Callback Service

Enter group number of port to be used

Please press the apply button and check pop-up screen

LCR(Least Cost Routing)

1. LCR(Least Cost Routing)

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

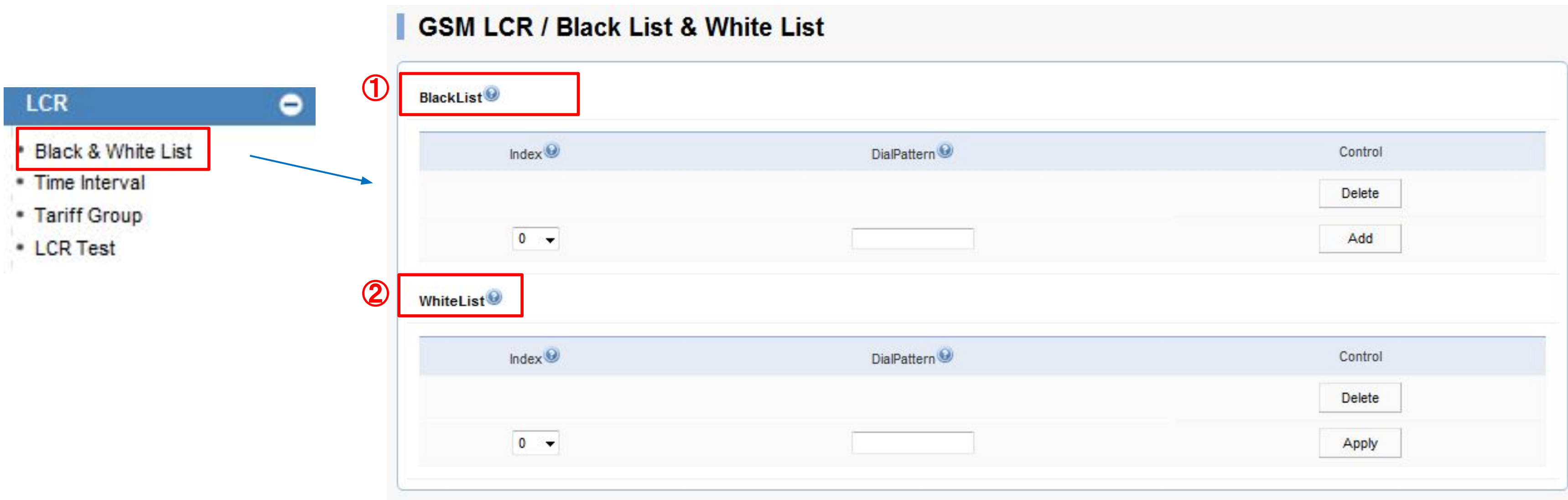
GSM LCR Time Interval
: The function to allow user to make call on GSM networks at specific time

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)

1. Black List / White List-1

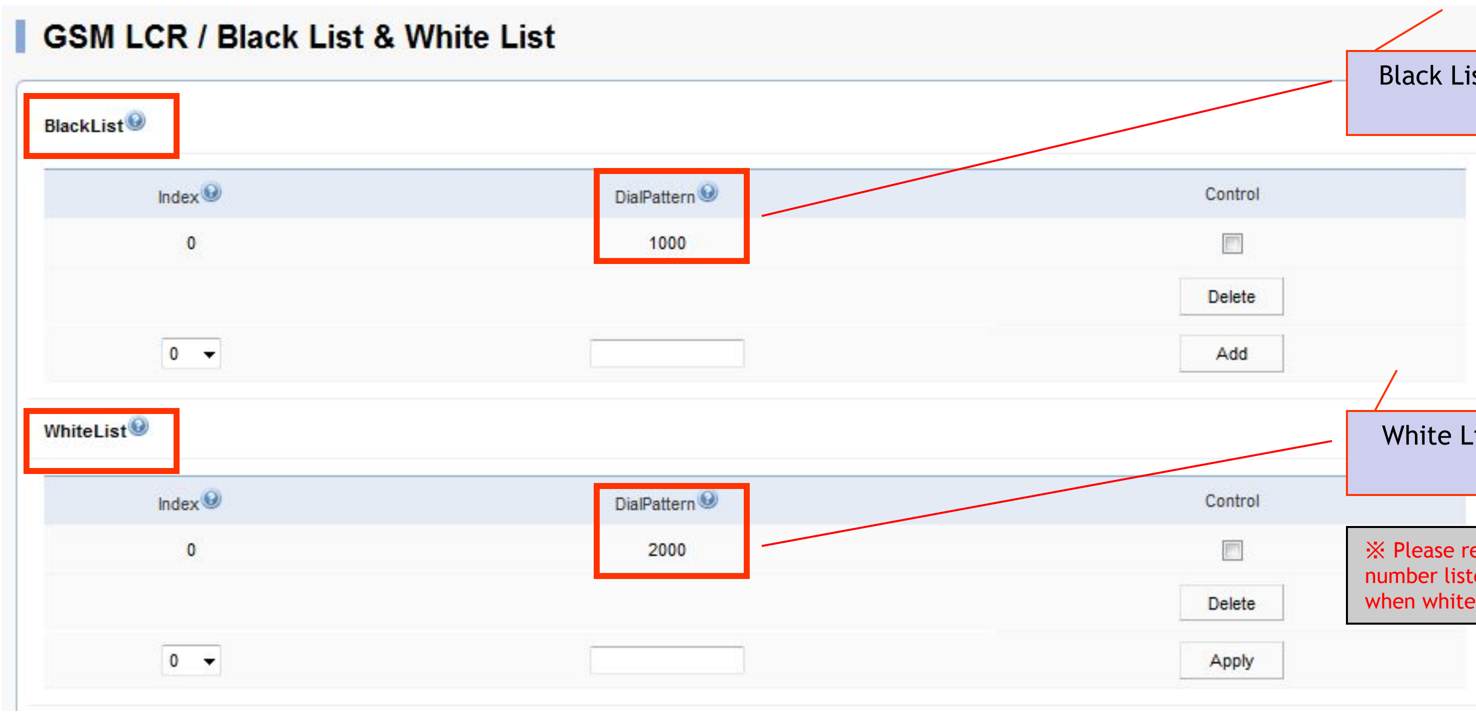


① Black List : Reject call from specific number
- Index : Sequential Index number. Entering the existed number updates the same field.
- DialPattern : Set number to reject

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

LCR(Least Cost Routing)

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



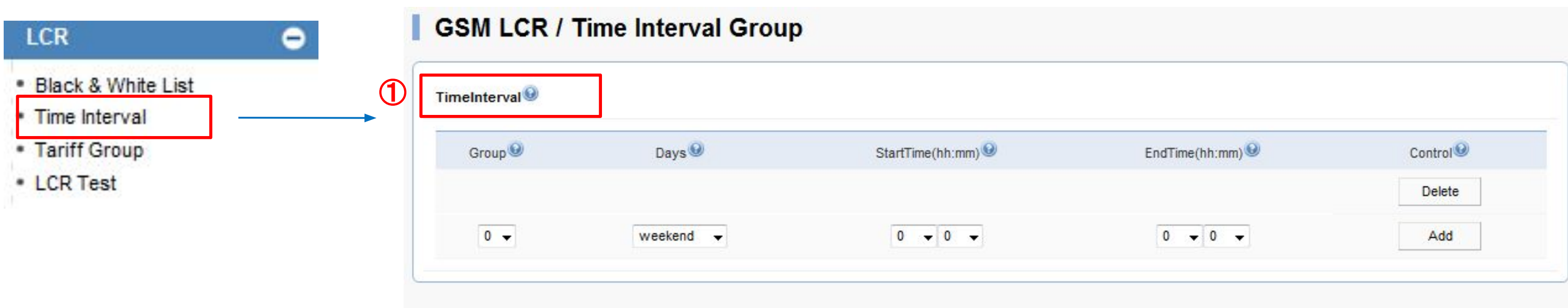
Black List : Reject call from specific number

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)

1. 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)

1. Time Interval-2(Ex)

GSM LCR / Time Interval Group

TimeInterval

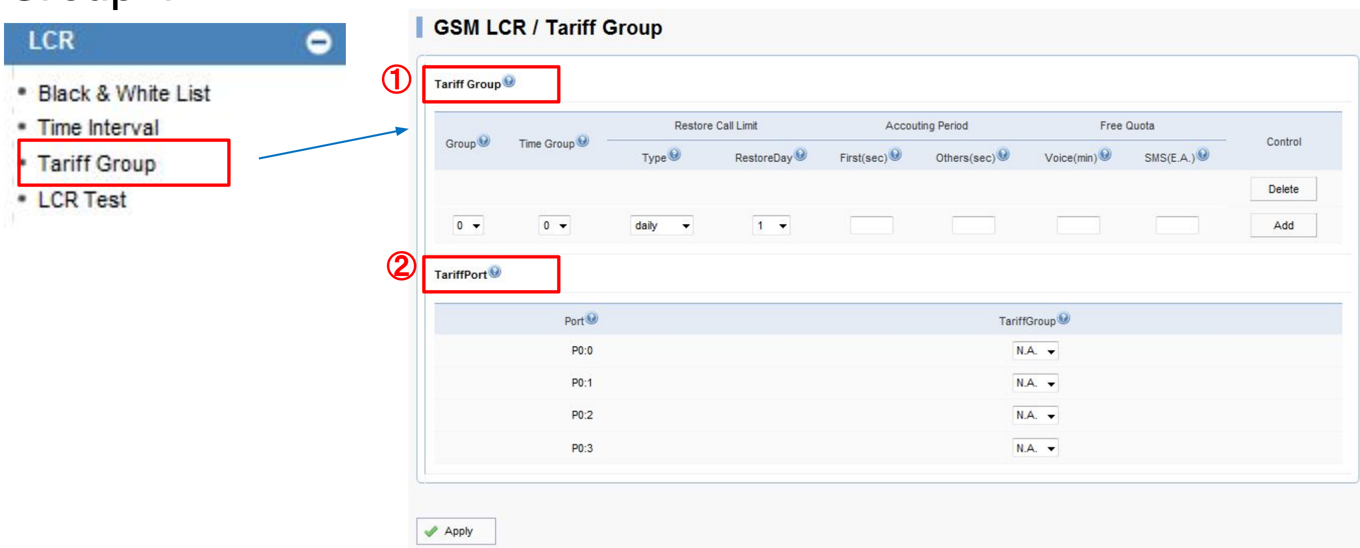
Group	Days	StartTime(hh:mm)	EndTime(hh:mm)	Control
0	Weekdays	09:00	18:00	☐
1	SUN	10:00	13:00	☐
1	SAT	10:00	13:00	☐
				Delete
0	weekend	0 0	0 0	Add

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)

1. Tariff Group-1

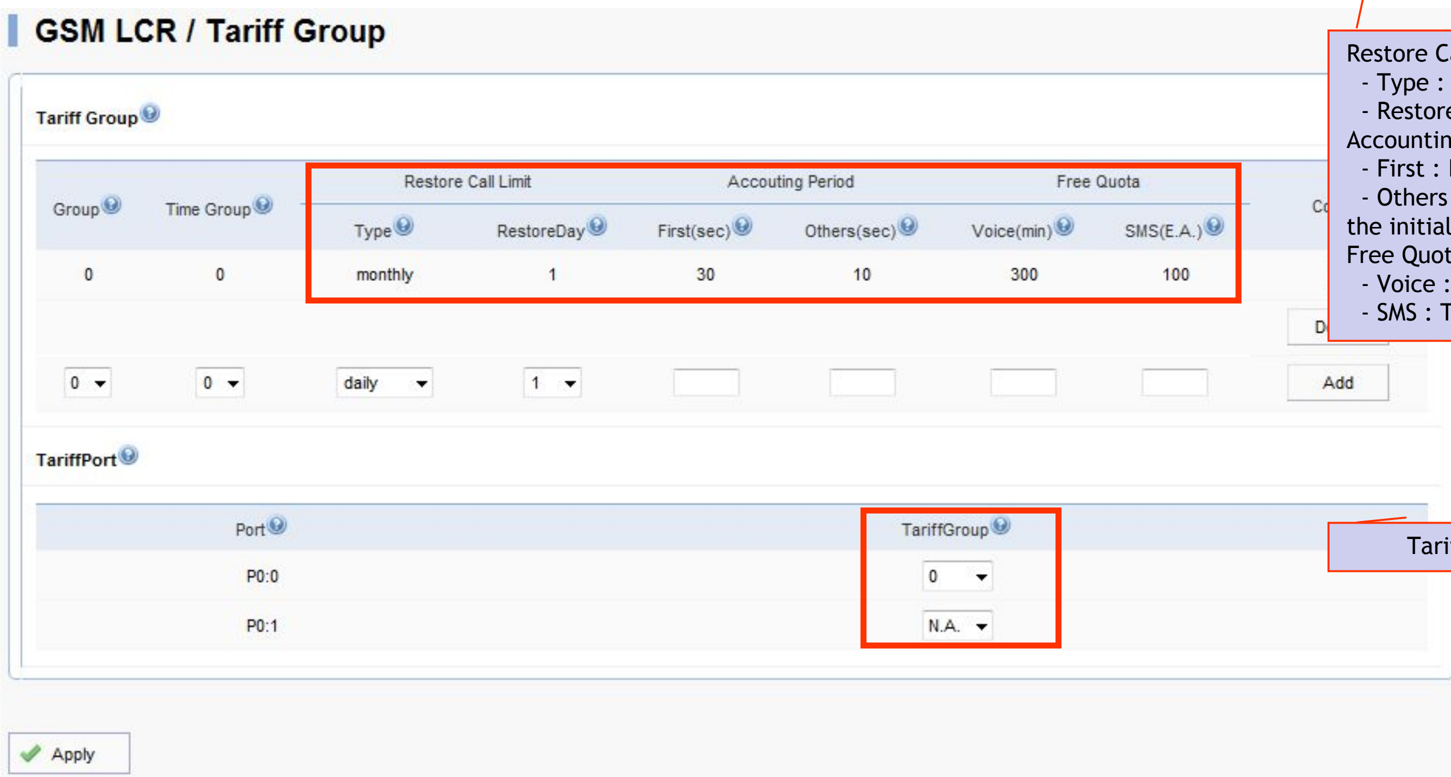


① Tariff Group : Set Time Group and toll-free limitaion
- Group : Generate group
- Time Group : Select group generated at Time Interval
- Restore Call Limit : Set point to restore set limitation
- Accounting Period : Tim period to charge (sec)
- Free Quota : Set toll-free time (min)

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

LCR(Least Cost Routing)

1. Tariff Group-2 (Ex)



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

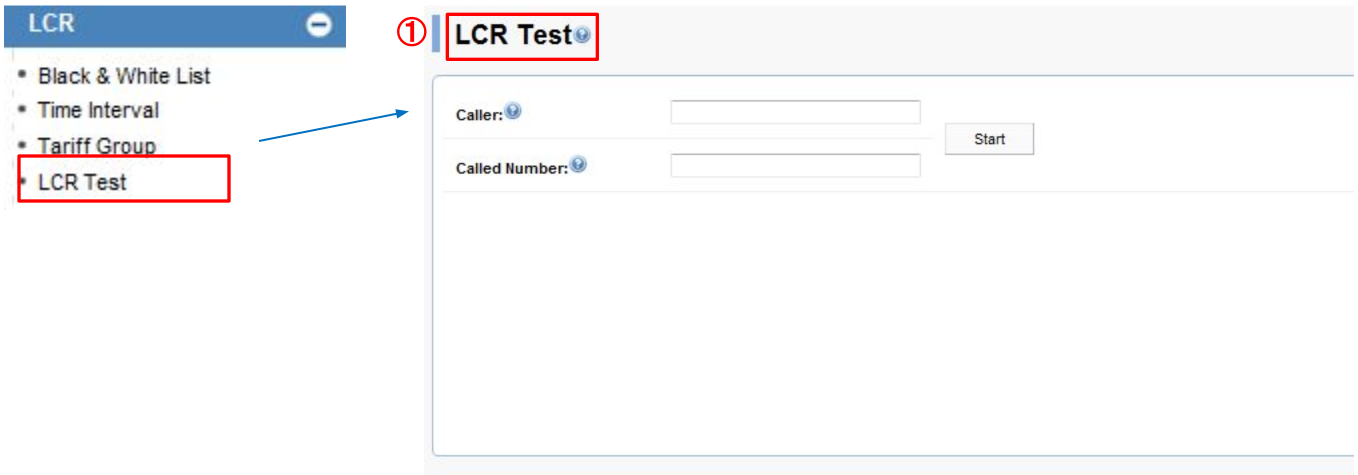
Accounting Period
- First : Initial period to charge
- Others : Second period to charge after the initial period

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

Tariff Group : Set group to port

LCR(Least Cost Routing)

1. 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)

1. LCR Test -2

(Ex)

LCR Test

Caller: 1000

Called Number: 2000

Start

```
<  1> LCR : ========================================================
<  2> LCR : == GSM LCR(Least Cost Route) Simulator Start        ==
<  3> LCR : ========================================================

<  4> LCR : -- src digits : 1000(GSM) -> dst digits : 2000(GSM)
<  5> LCR : -- MatchAllProcess After Sorted
<  6> LCR :                <0>  id(4584) dest(T) prefer(0) selected(1)
<  7> LCR : -- Trying : <0> id(4584) dest(T)
<  8> LCR : -- Error : Denied by Time Interval Restriction

<  9> LCR : ----------------------------------------------------
< 10> LCR : -- Result : Fail
< 11> LCR : ========================================================
< 12> LCR : == GSM LCR(Least Cost Route) Simulator End          ==
< 13> LCR : ========================================================
```

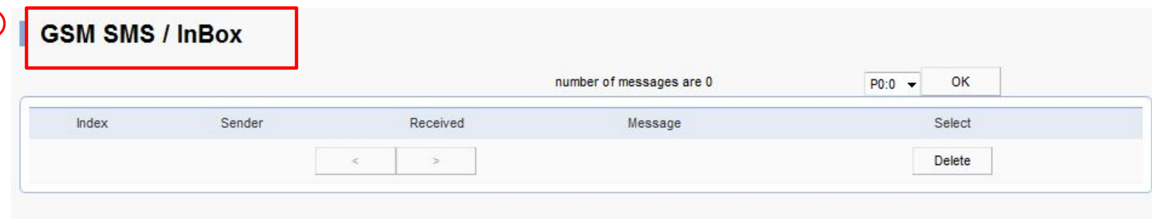
Problem can be monitored to make virtual call

The above message is shown that failure occurred by time interval

SMS

1. SMS -1

①



받은 메시지들이 저장됩니다.

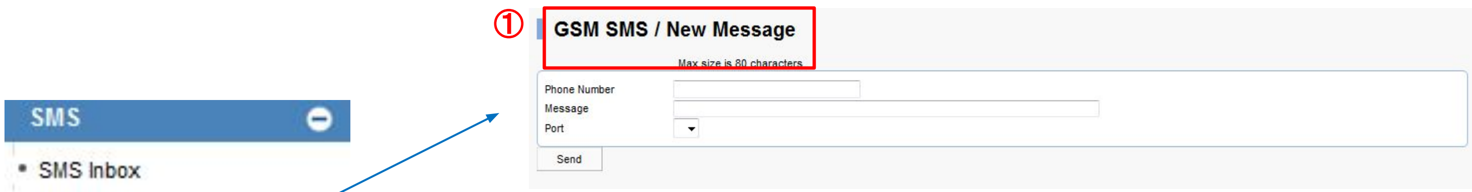
②

보낸 메시지들이 저장됩니다.

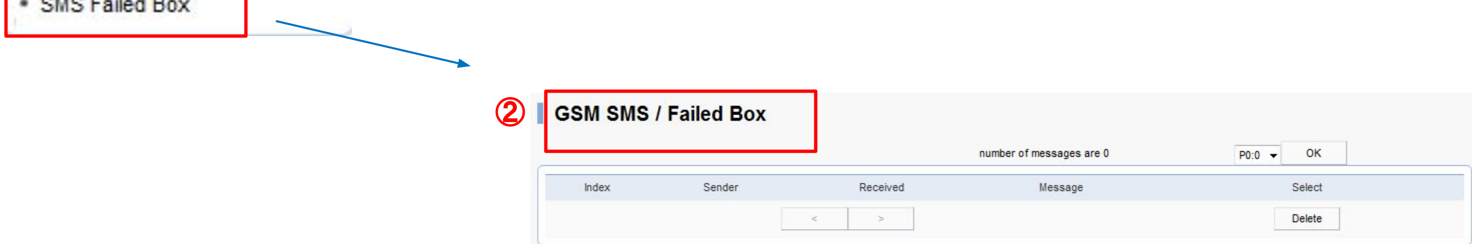
※ The number of message can be stored in InBox, Sent Box and Fail Box is 17

SMS

1. SMS -2



메시지를 보낼 수 있습니다.



실패한 메시지들이 저장됩니다.

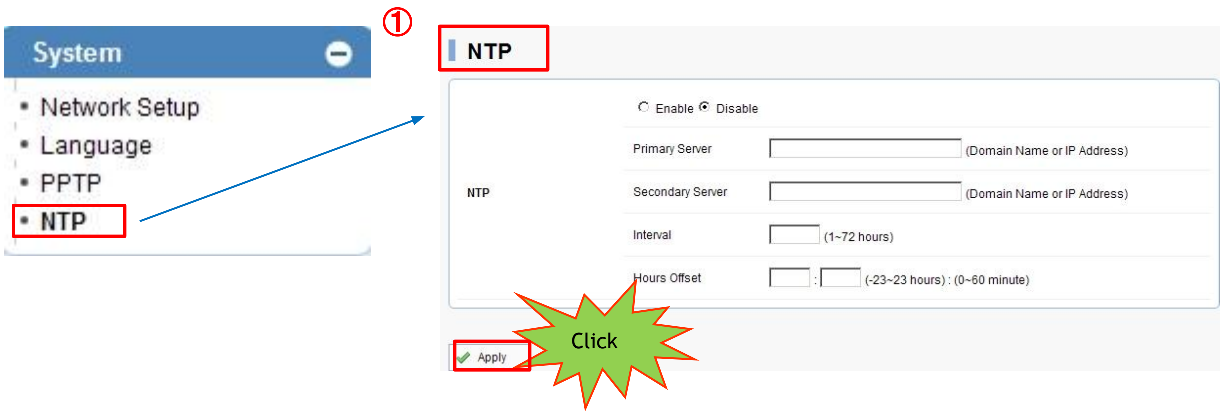
※ SMS Support Language : Korean, English, Russian, Spanish and Portuguese

Advanced Service

1. NTP
2. Gain & CID
3. GSM Pins
4. GSM Band
5. BTS
6. FAX
7. Service
8. Filtering
9. Security

Advanced Service

1. NTP

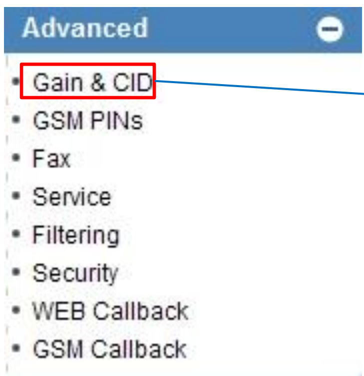


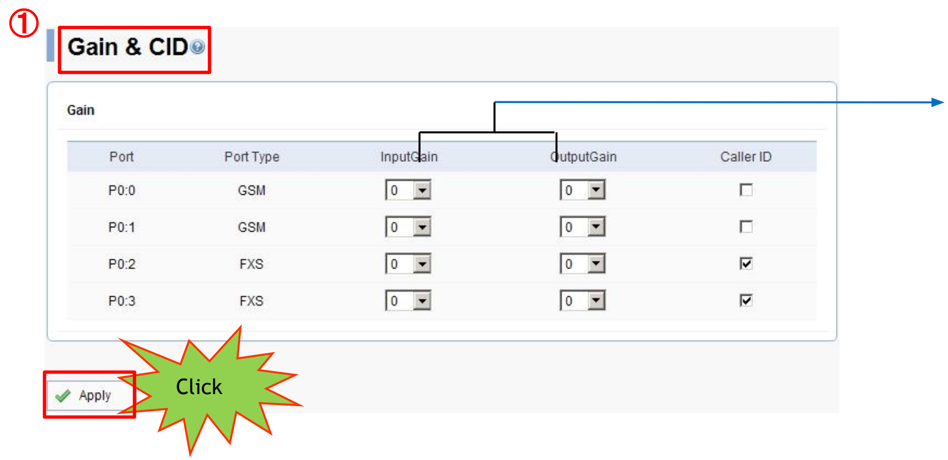
① NTP : Input information of NTP Server
- Click the apply button for NTP activation
- Primary Server : Input IP or domain name of NTP Server
- Interval : Interval to request and receive data from NTP server

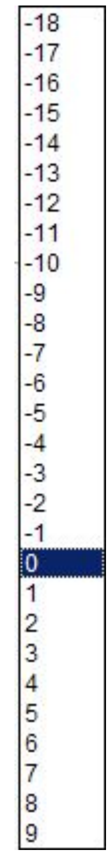
* Please click the apply button after set up

Advanced Service

2. 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

3. GSM PINs

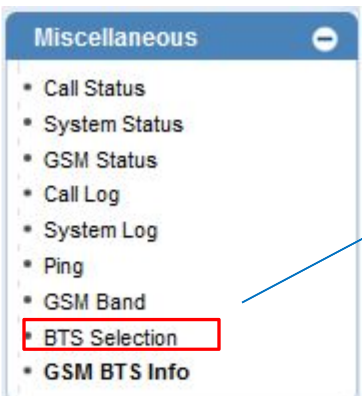
Advanced Service

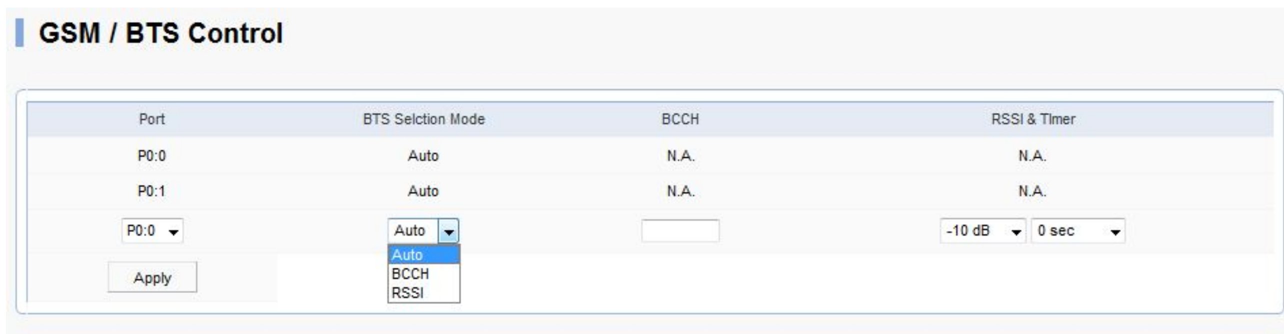
4. GSM Band

GSM Band Selection : Setting for bandwidth of GSM Networks
- Auto (Default)
- 900Mhz + DCS 1800Mhz
- 900Mhz + PCS 1900Mhz
- 850Mhz + DCS 1800Mhz
- 850Mhz + PCS 1900Mhz

Advanced Service

5. BTS(Base Terminal Station) -1

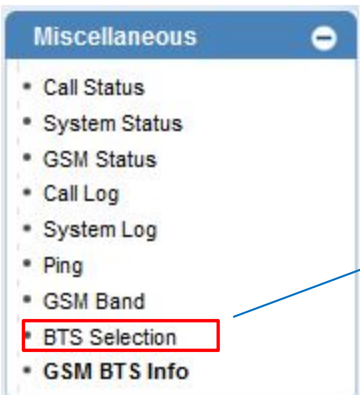


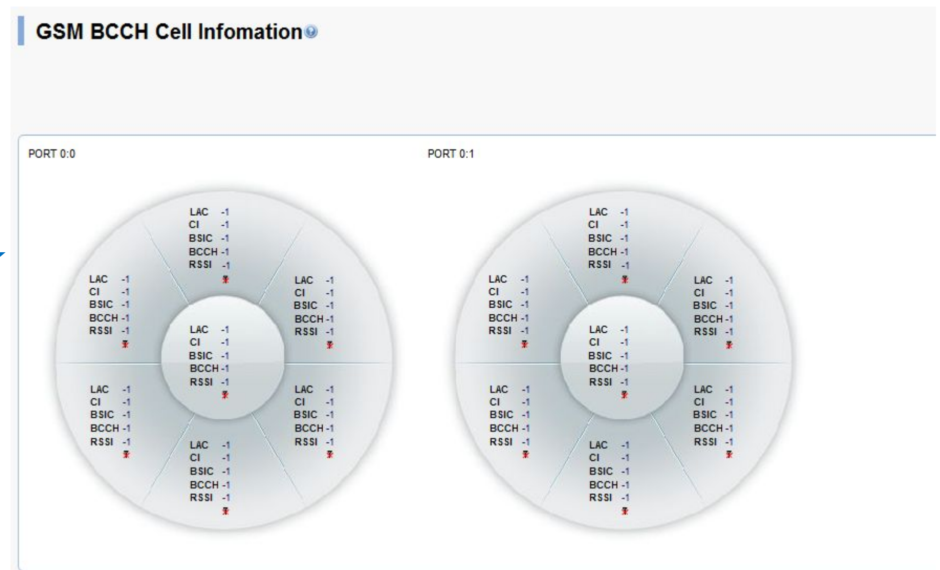


BTS Control : Setting for selection method of BTS cell
- Auto (Default) : The most highest power cell will be selected
- BCCH : Choose specific cell by entering BCCH value manually
- RSSI : Choose cell which has specific RSSI level
  Cell will be selected in accordance with set interval

Advanced Service

5. BTS(Base Terminal Station) -2





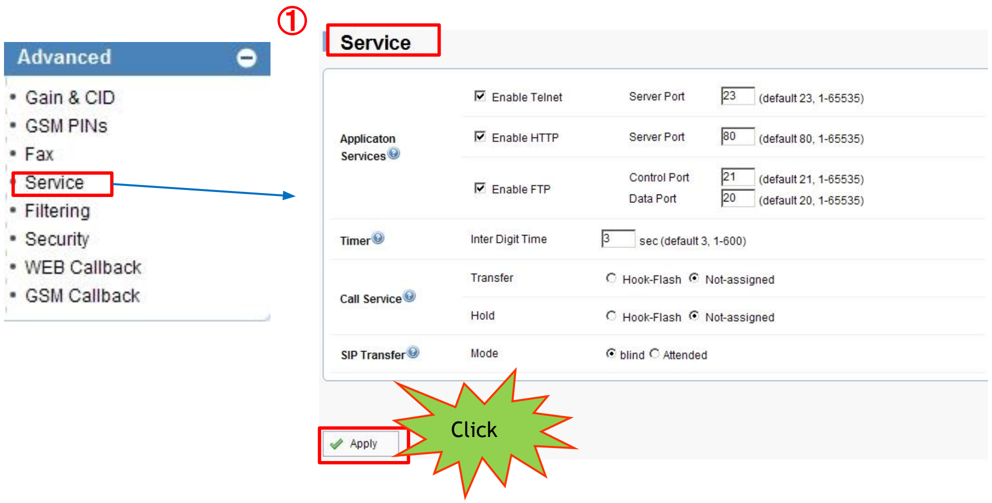
BTS Cell Information
: The information of serve cell and neighbor cell can be shown

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

7. Service

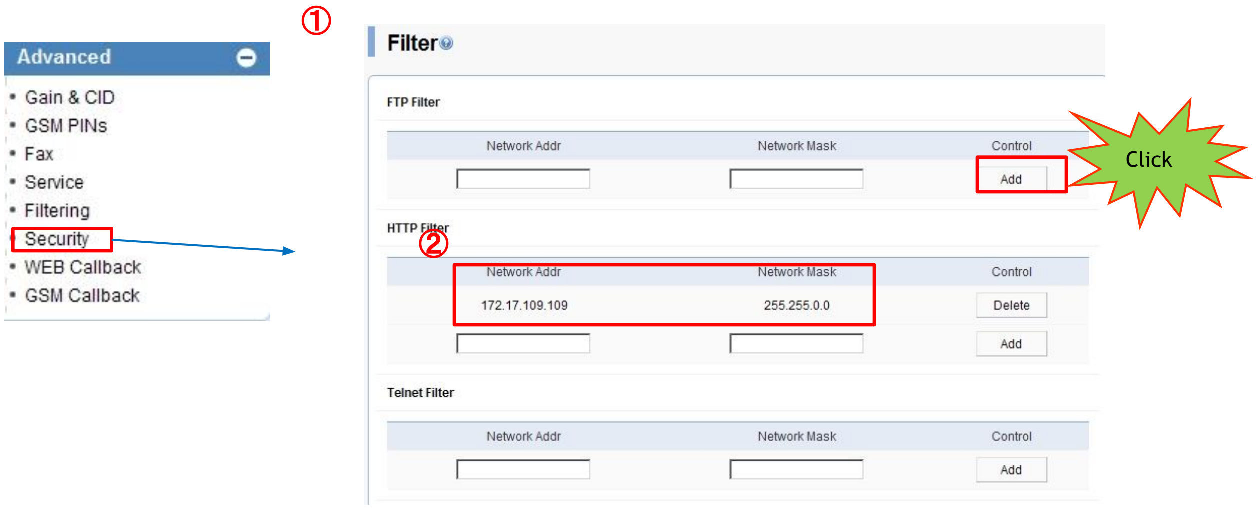


① Service : Set extra features
- Application Services : The port setting for each Telnet , HTTP, FTP
- Timer : Adjust digit time for phone connected to AP-GS1002
  Set time limitation between Digit and Digit
- Call Service : Call-Transfer Set Activation and Call-Hold function
  (Hook-Flash – Activate , Not-assigned – Inactivate)
- Transfer Mode : AP-GS1002 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

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

Advanced 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 AP-GS1002 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

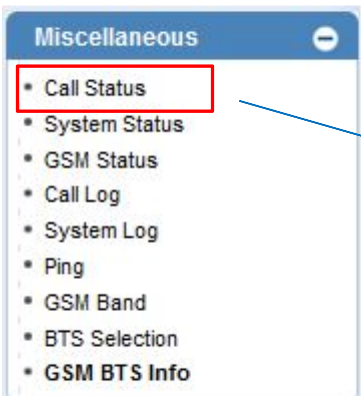
* Please click the apply button after set up

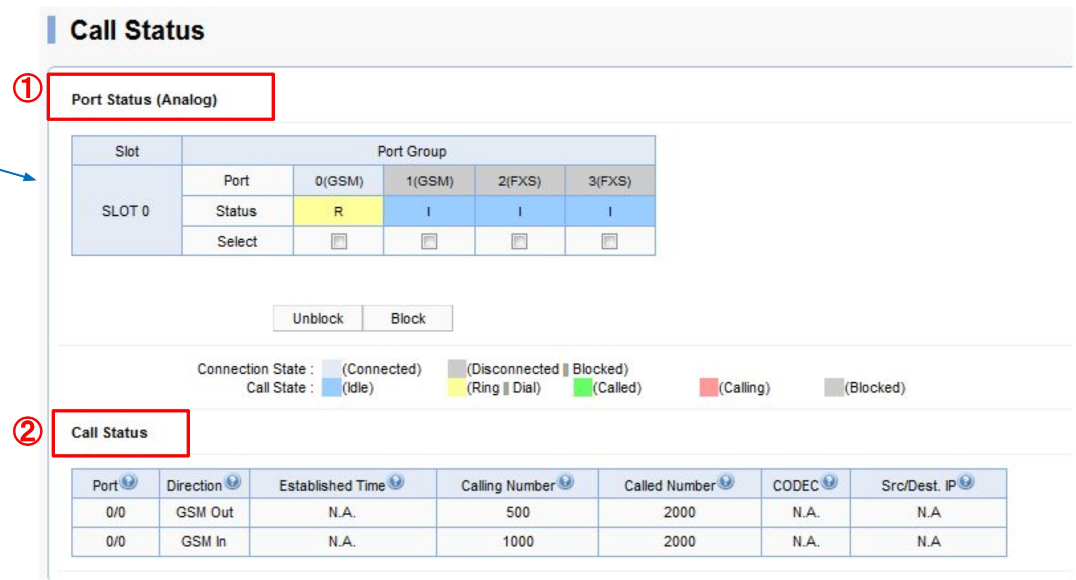
Monitoring

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

Monitoring

1. Call Status

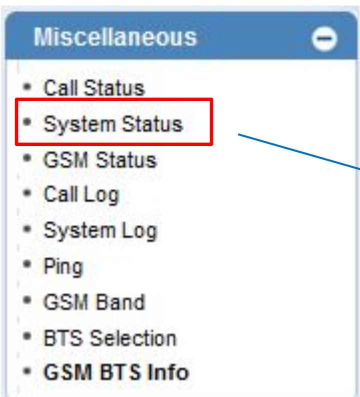


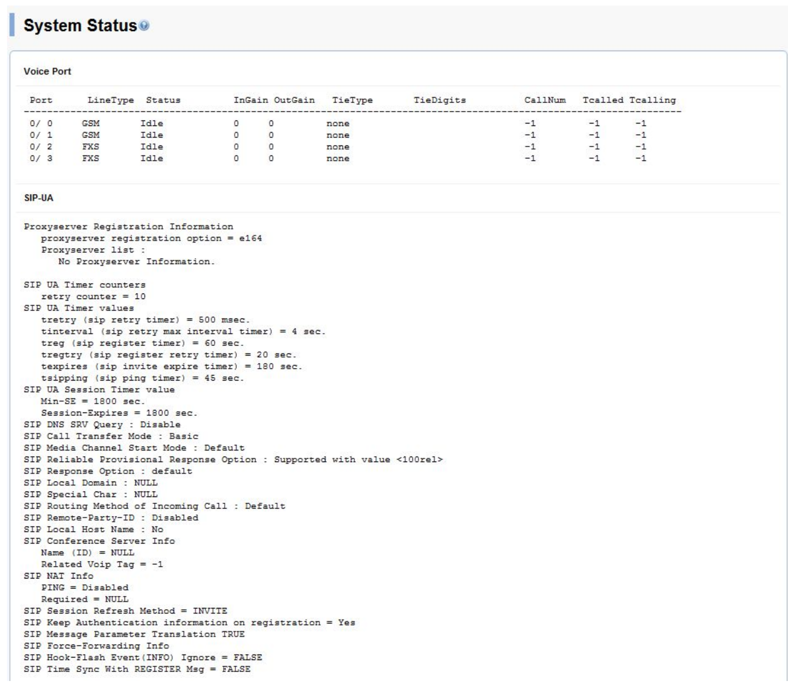


Call Status : The status of AP-GS1002 port and call can be monitored in real-time
① Port Status : Monitoring AP-GS1002 port
② Call Status : Monitoring call status

Monitoring

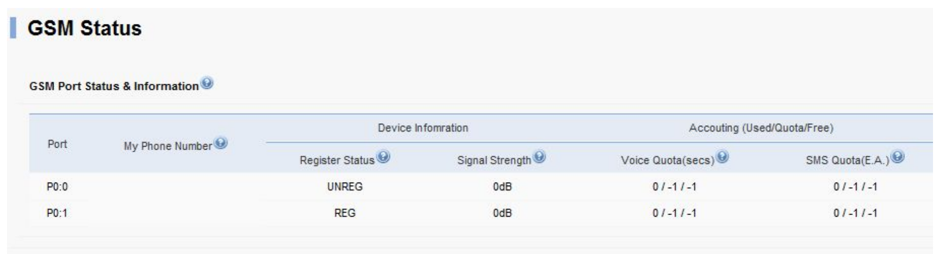
2. System Status





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

3. GSM Status



GSM Status

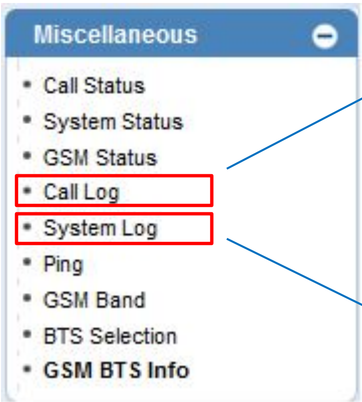
GSM Port Status & Information

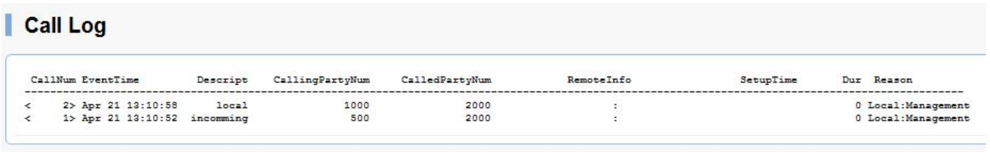
Port	My Phone Number	Device Infomration		Accouting (Used/Quota/Free)	
		Register Status	Signal Strength	Voice Quota(secs)	SMS Quota(E.A.)
P0:0		UNREG	0dB	0 / -1 / -1	0 / -1 / -1
P0:1		REG	0dB	0 / -1 / -1	0 / -1 / -1

GSM Status : GSM Networks status, Usage can be monitored

Monitoring

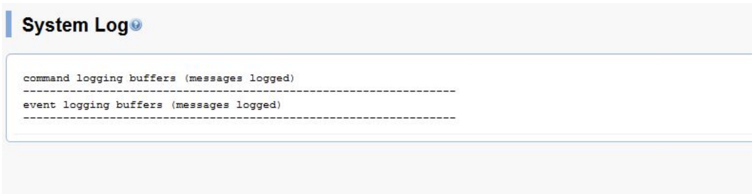
4. Call Log / System Log





Call Log : Monitoring all of call history

※ Call history will be clear with rebooting



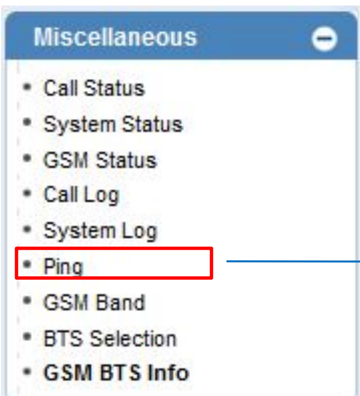
System Log : Monitoring AP-GS1002 System log

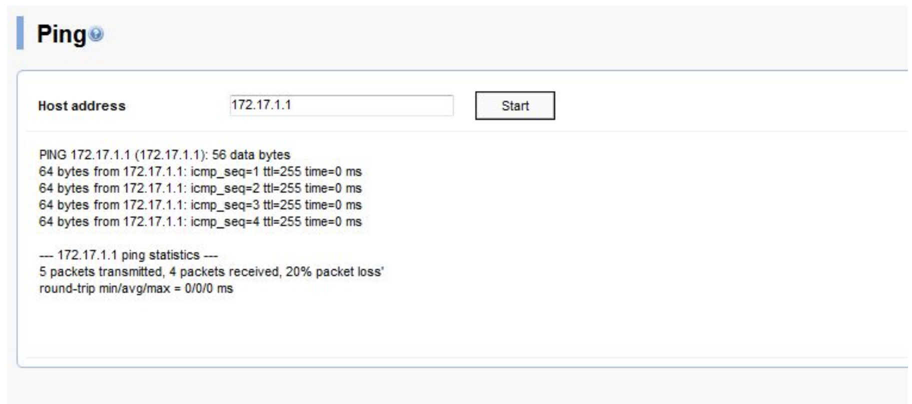
Default setting is off. System log can be monitored by telnet connection and entering CLI command is required

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

Monitoring

5. Ping





Ping : Network status can be checked by pinging

Thank you