

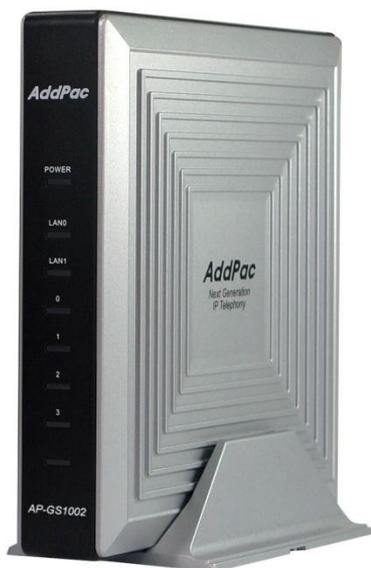


AP-GS1002™

2-Port FXS GSM Gateway

High Performance GSM Gateway Solution

WEB Setup Guide



AddPac

Contents

1. WEB Connection
2. Network Setup
3. GSM Setup
4. VoIP Setup
5. Callback Service
6. LCR
7. SMS
8. Advanced Service
9. Monitoring

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



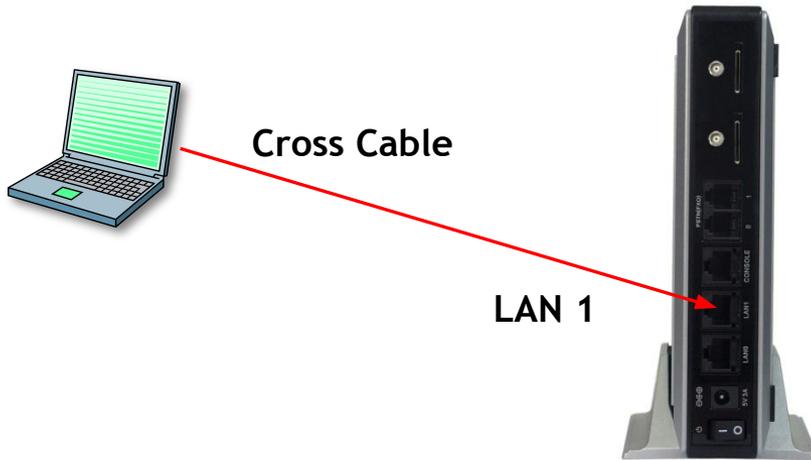
1. Connect to GS1002 via console port to enter IP address in order to set Interface 0/0
2. Input IP address of interface 0/0 using below command
3. Connect to the IP address via web after saving

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 <Default 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

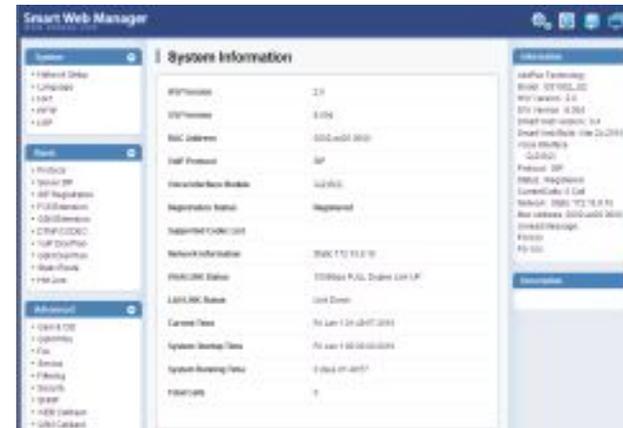
1. The screen of Web connection



- 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

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- GSM Status
- Call Log

Network Setup

Hostname

IP Address A.B.C.D

Network Mask A.B.C.D

Default Router A.B.C.D

Static IP

DNS Server Primary DNS Server

Secondary DNS Server

PPPoE(ADSL)

Username

Password

DHCP

VLAN ID

Auto

WAN Link Control

Manual

Speed 100 10

Duplex full half

MAC(Hardware) Address

Apply

Click

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

System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension**
- DTMF/CODEC
- VoIP Dial Plan
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- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- WEB Callback

GSM Extension

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

GSM Extension Configuration

Index	Port	Numbers	Preference	HuntStop	Select
	P0:0		0	<input type="checkbox"/>	Delete

GSM Extension with Translation

Port	Destination Pattern	Digits to Insert	Number of Digits to Delete
P0:0			0
P0:1			0

Apply

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

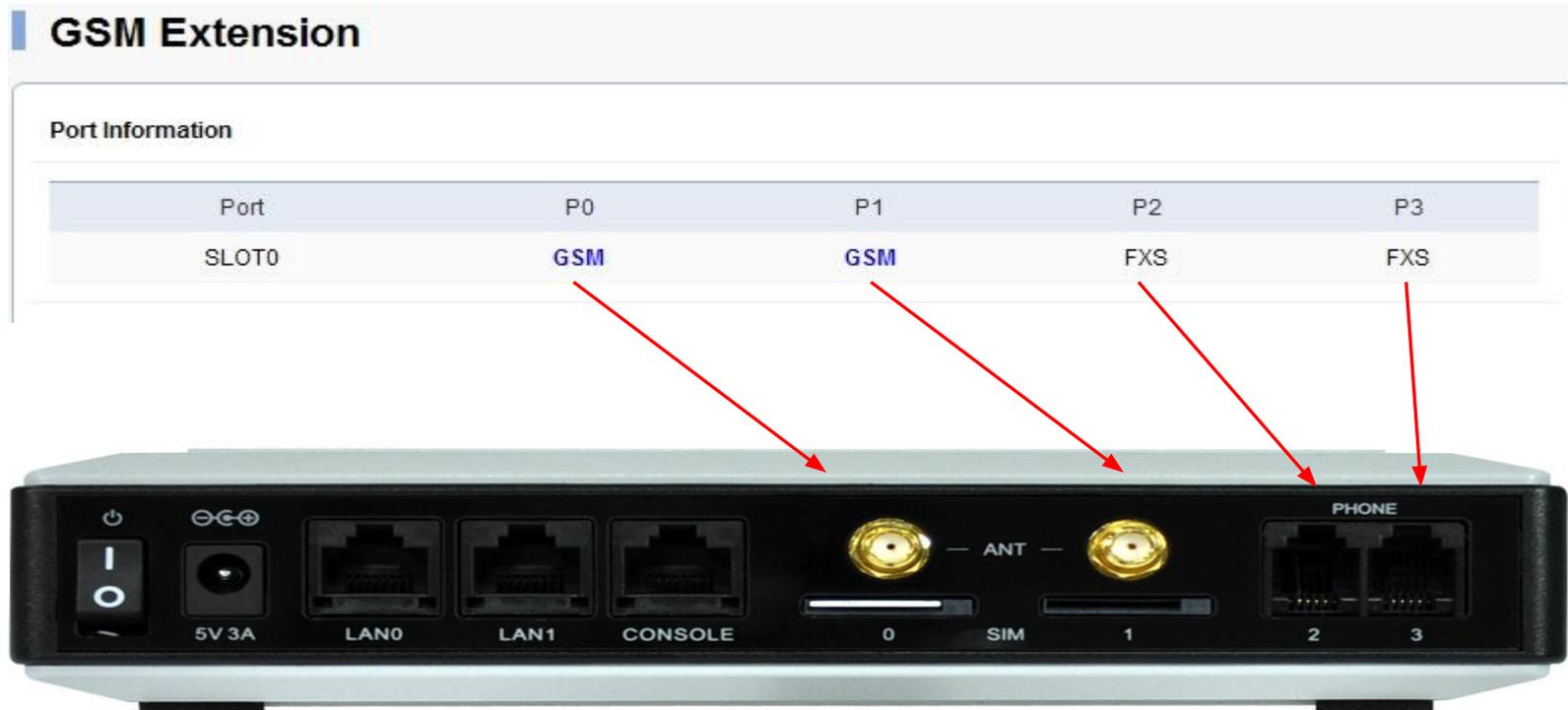
- : Use to convert mobile phone number for callback service
 - Destination Pattern : Enter mobile phone number to convert
 - Digits to Insert : Insert number to make calling number
 - Number of Digits to Delete : Delete number to make calling number

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

Result : 825683848

GSM Setup > GSM Extension

1. GSM Extension -2



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

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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension**
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

FXS Extension

① Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

② FXS Extension Configuration

Index	Port	Numbers	Preference	HuntStop	Select
<input type="text"/>	P0:2	<input type="text"/>	0	<input type="checkbox"/>	Delete

Apply

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)

FXS Extension

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

FXS Extension Configuration

Index	Port	Numbers	Preference	HuntStop	Select
① 0	0/2	1000	0	X	<input type="checkbox"/>
② 1	0/3	2000	1	0	<input type="checkbox"/>

- 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 transfer※

- 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

Basic

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- **GSM Dial Plan**
- Static Route
- Hot Line

GSM Dial Plan / Prefix

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

① Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0				Delete
				Add

② Prefix Table

Index	Prefix	2nd Prefix	PlanIndex	Slot/Port	Control
0			N.A.	P0:0	Delete
					Apply

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

GSM Dial Plan / Prefix - 2 (Example)

Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0	82	1	025683848	<input type="checkbox"/>

Prefix Table

Index	Prefix	2nd Prefix	PlanIndex	Slot/Port	Control
0	T		N.A.	0/0	<input type="checkbox"/>
1	9	T	N.A.	0/1	<input type="checkbox"/>
2	025683848		0	0/0	<input type="checkbox"/>

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

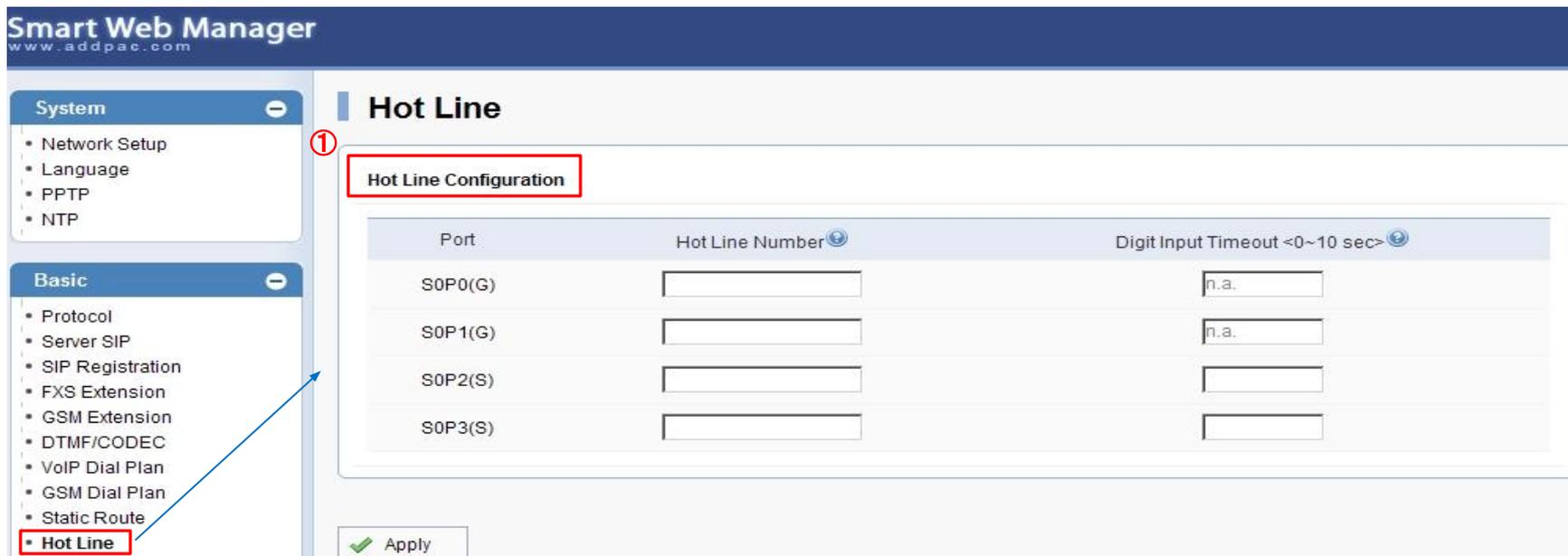
② Prefix , 2nd Prefix : Setting method of 1 stage and 2 stage

(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



The screenshot shows the Smart Web Manager interface for configuring Hot Lines. The left sidebar contains a 'System' menu with 'Hot Line' selected, and a 'Basic' menu with 'Hot Line' also selected. The main content area is titled 'Hot Line' and contains a 'Hot Line Configuration' section. This section features a table with three columns: 'Port', 'Hot Line Number', and 'Digit Input Timeout <0~10 sec>'. The table has four rows corresponding to ports S0P0(G), S0P1(G), S0P2(S), and S0P3(S). The 'Hot Line Number' and 'Digit Input Timeout' fields are currently empty or contain 'n.a.'. An 'Apply' button is located at the bottom left of the configuration area.

Port	Hot Line Number	Digit Input Timeout <0~10 sec>
S0P0(G)	<input type="text"/>	<input type="text" value="n.a."/>
S0P1(G)	<input type="text"/>	<input type="text" value="n.a."/>
S0P2(S)	<input type="text"/>	<input type="text"/>
S0P3(S)	<input type="text"/>	<input type="text"/>

- 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

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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration**
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan

SIP Registration

① SIP Registration Configuration

Port	E.164 Number	User Name	Password	DisplayName	Reg	HuntStop
S0P2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>
S0P3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>

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)

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route

SIP Registration

SIP Registration Configuration

Port	E.164 Number	User Name	Password	DisplayName	Reg	HuntStop
S0P2	8888	1234	••••••••	5683848	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
S0P3					<input type="checkbox"/>	<input type="checkbox"/>

Apply

Click

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00g
Smart Web Version : 0.4
Smart Web Build : Apr 13 2010

Voice Interface
G(2)S(2)

Protocol : SIP

Status : Registered

CurrentCalls : 0 Call

Network : Static 172.17.109.1

Mac Address : 0002.a400.0000

Unread Message:
P0:0(0)
P0:1(0)

- 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

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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC**
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

DTMF/CODEC

Voice CODEC

DTMF Relay mode

Apply

Click

Preference 1	None
Preference 2	None
Preference 3	None
Preference 4	None
Preference 5	None
Preference 6	None

DTMF relay by In-band voice
 DTMF relay by RTP payload defined by RFC 2833
 DTMF relay by Out-of-band signal
 DTMF relay by Cisco out-of-band signal

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

g711alaw : G711 a-law Codec Type(64 kbps)
g711ulaw : G711 u-law Codec Type(64 kbps)
g7231r53 : G723.1 Codec Type(5.3 kbps)
g7231r63 : G723.1 Codec Type(6.3 kbps)
g726r16 : G726 ADPCM Type(16 kbps)
g726r32 : G726 ADPCM Type(32 kbps)
g729 : G729 Codec Type(8 kbps)
None

VoIP Setup

4. VoIP Dial Plan -1

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- **VoIP Dial Plan**
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

VoIP Dial Plan / Prefix

Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0				Delete
				Add

Prefix Table

Index	Prefix	PlanIndex	Control
0		N.A.	Delete
			Apply

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)

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax

VoIP Dial Plan / Prefix

Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0	82	1	00[1-7]T	<input type="checkbox"/>
0				<input type="button" value="Delete"/>
0				<input type="button" value="Add"/>

Prefix Table

Index	Prefix	PlanIndex	Control
0	00[1-7]T	0	<input type="checkbox"/>
0		N.A.	<input type="button" value="Delete"/>
0			<input type="button" value="Apply"/>

It must be the same with PlanIndex Number

VoIP Setup

5. Static Route -1

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route**
- Hot Line

Static Route

Set Remote Site Call(5-digit number is set to begin *2->*2...)

No	Remote Site IP	Prefix	Insert Digit	Delete Digit	Name of Remote Site	Answer Addr	Control
*	<input type="text"/>	<input type="button" value="Apply"/>					

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)

Smart Web Manager
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System

- Network Setup
- Language
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route**
- Hot Line

Static Route

Set Remote Site Call(5-digit number is set to begin *2->*2...)

No	Remote Site IP	Prefix	Insert Digit	Delete Digit	Name of Remote Site	Answer Addr	Control
0	172.17.110.85	025683848	82	2	AddPac		<input type="checkbox"/>

*

Delete Apply

Click

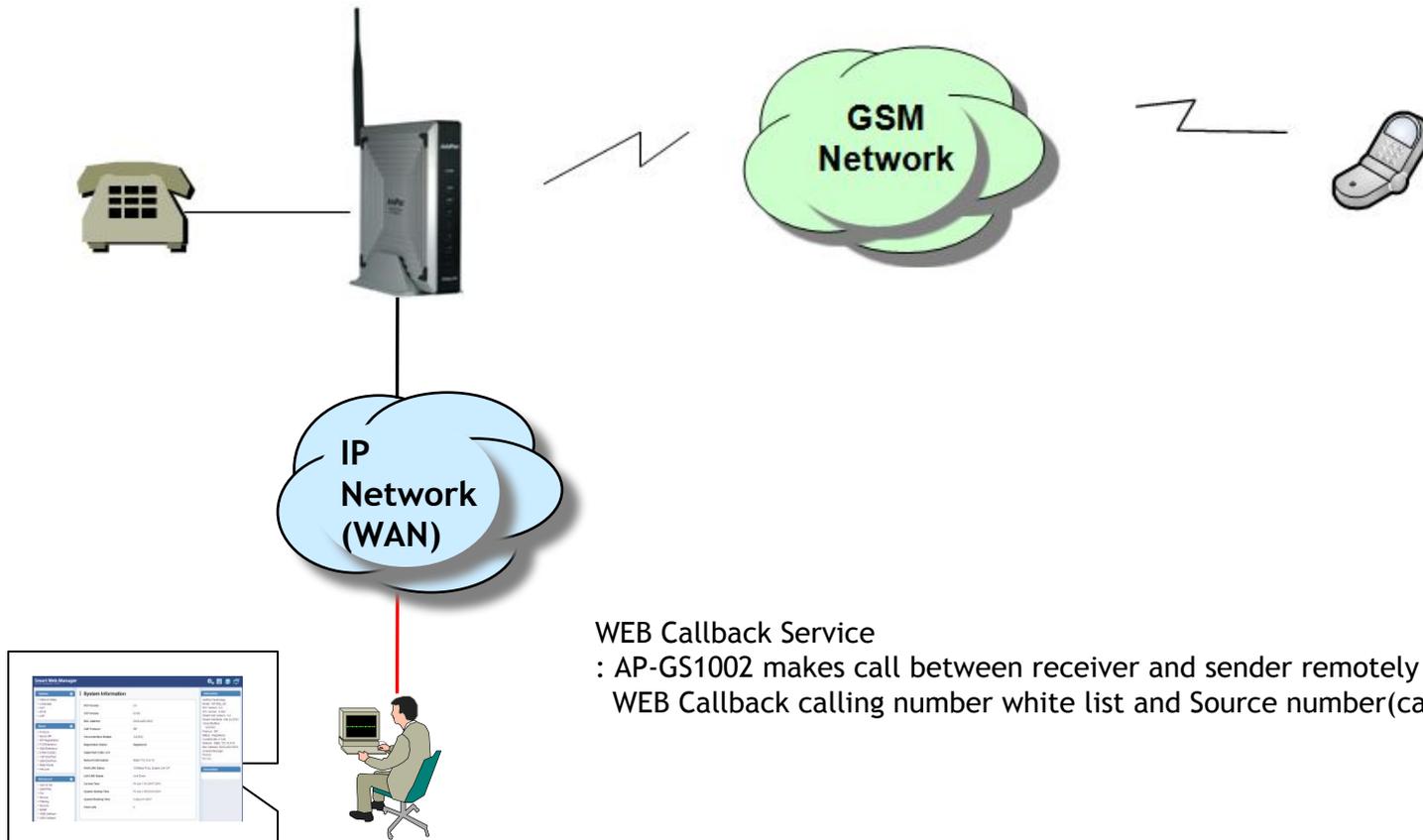
* Please press the apply button to save

Callback Service

1. WEB Callback Service
2. GSM Callback Service

Callback Service

1. WEB Callback Service



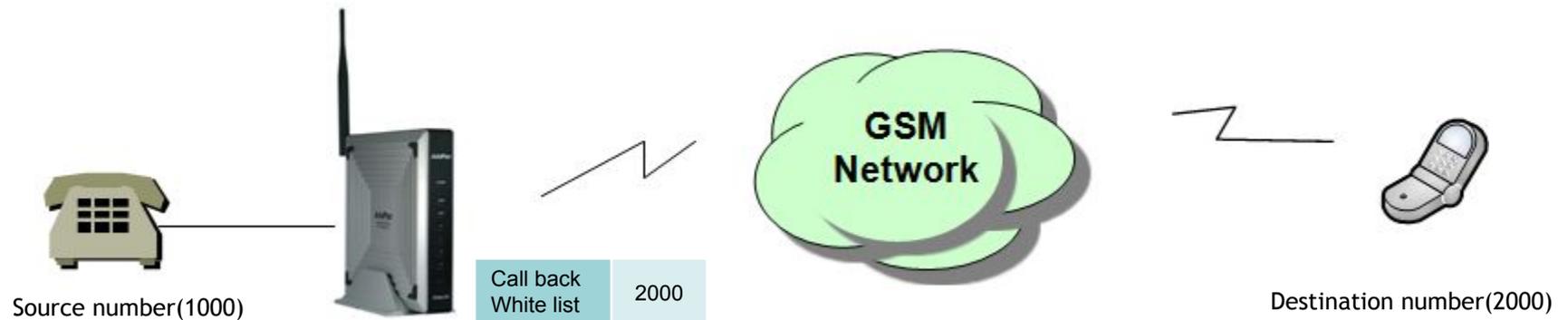
WEB Callback Service

: 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

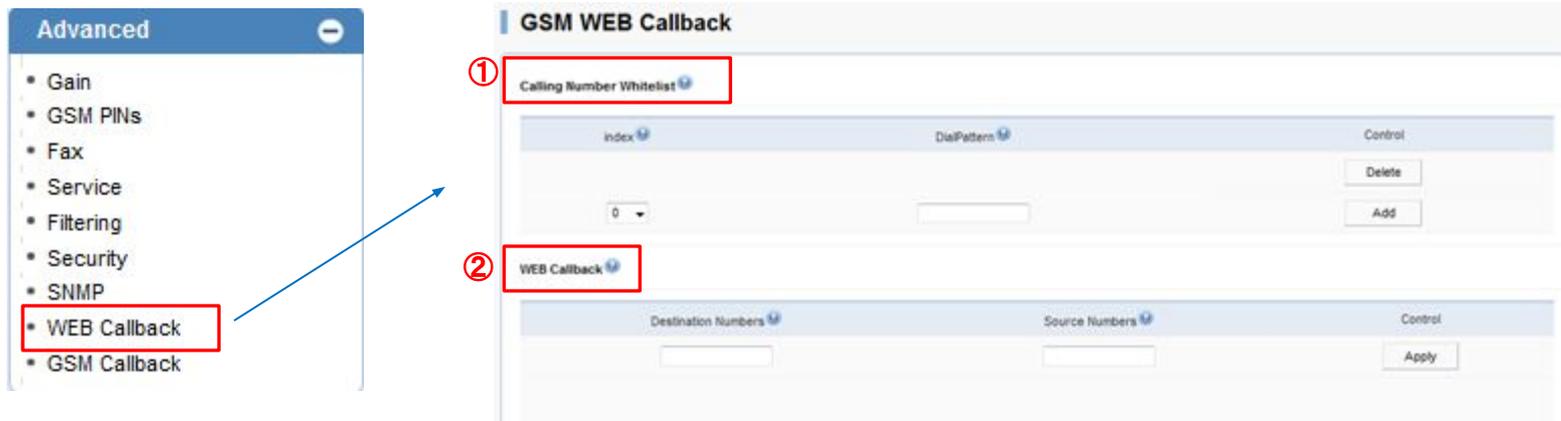


GSM Callback Service

: 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)

The screenshot displays two main sections: 'Calling Number Whitelist' and 'WEB Callback'.

Calling Number Whitelist:

index	DialPattern	Control
0	500	

Below the table are a dropdown menu with '0' selected and an 'Add' button.

WEB Callback:

Destination Numbers:

Source Numbers:

Control:

A green starburst labeled 'Click' points to the 'Apply' button.

At the bottom, a box labeled 'Remote Trying' is highlighted.

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

Advanced

- Gain
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback**

GSM Callback

① Calling Number Whitelist

Group	Index	DialPattern	Control
3	0		Delete
			Add

Callback

GSM Port	My Number	WhiteList Group
P0:0		N.A.
P0:1		N.A.

Apply

- 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)

The screenshot shows the 'GSM Callback' configuration page. It is divided into two main sections: 'Calling Number Whitelist' and 'Callback'.

Calling Number Whitelist: A table with columns: Group, Index, DialPattern, and Control. The 'Group' column has a value of 3. The 'DialPattern' column has two entries: 01099116545 and 8225683848. Below the table are dropdown menus for Group (3) and Index (0), and buttons for 'Delete' and 'Add'.

Callback: A table with columns: GSM Port, My Number, and WhiteList Group. The 'GSM Port' column has entries P0:0 and P0:1. The 'WhiteList Group' column has a dropdown menu with the value 3 selected.

Annotations:

- A red box highlights the 'Group' value '3' in the 'Calling Number Whitelist' table. A callout box says: "Register number to use Callback Service".
- A red box highlights the 'WhiteList Group' dropdown menu in the 'Callback' section. A callout box says: "Enter group number of port to be used".
- A red box highlights the 'Apply' button. A callout box says: "Please press the apply button and check pop-up screen".

Pop-up Screen: A small window titled '웹 페이지의 메시지' (Web Page Message) with a yellow warning icon and the text 'Update Success'. It has a '확인' (Confirm) button.

Click: A green starburst graphic with the word 'Click' and an arrow pointing to the 'Apply' button.

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

LCR

- Black & White List
- Time Interval
- Tariff Group
- LCR Test

GSM LCR / Black List & White List

① BlackList

Index	DialPattern	Control
0		Delete
		Add

② WhiteList

Index	DialPattern	Control
0		Delete
		Apply

- 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)

GSM LCR / Black List & White List

BlackList

Index	DialPattern	Control
0	1000	<input type="checkbox"/>
		Delete
0		Add

WhiteList

Index	DialPattern	Control
0	2000	<input type="checkbox"/>
		Delete
0		Apply

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

The screenshot shows the LCR configuration interface. On the left, a sidebar menu lists 'Black & White List', 'Time Interval', 'Tariff Group', and 'LCR Test'. The 'Time Interval' option is highlighted with a red box and a blue arrow pointing to the main configuration area. The main area is titled 'GSM LCR / Time Interval Group' and contains a table with columns: Group, Days, StartTime(hh:mm), EndTime(hh:mm), and Control. The 'TimeInterval' label is highlighted with a red box and a circled '1'. The table has a 'Delete' button and an 'Add' button.

Group	Days	StartTime(hh:mm)	EndTime(hh:mm)	Control
0	weekend	0 0	0 0	Delete Add

- 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	<input type="checkbox"/>
1	SUN	10:00	13:00	<input type="checkbox"/>
1	SAT	10:00	13:00	<input type="checkbox"/>

0 | weekend | 0 | 0 | 0 | 0

Delete | 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

LCR

- Black & White List
- Time Interval
- Tariff Group**
- LCR Test

GSM LCR / Tariff Group

① Tariff Group

Group	Time Group	Type	RestoreDay	First(sec)	Others(sec)	Voice(min)	SMS(E.A.)	Control
0	0	daily	1					Delete Add

② TariffPort

Port	TariffGroup
P0:0	N.A.
P0:1	N.A.
P0:2	N.A.
P0:3	N.A.

Apply

- ① Tariff Group : Set Time Group and toll-free limitiaon
 - 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)

GSM LCR / Tariff Group

Tariff Group

Group	Time Group	Restore Call Limit		Accounting Period		Free Quota	
Type	RestoreDay	First(sec)	Others(sec)	Voice(min)	SMS(E.A.)		
0	0	monthly	1	30	10	300	100

0 0 daily 1 [] [] [] [] Add

TariffPort

Port	TariffGroup
P0:0	0
P0:1	N.A.

Apply

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

LCR

- Black & White List
- Time Interval
- Tariff Group
- LCR Test**

① **LCR Test**

Caller:

Called Number:

Start

- 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:

Called Number:

```
< 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 : =====
```

The above message is shown that failure occurred by time interval

Problem can be monitored to make virtual call

SMS

1. SMS -1

The image shows a software interface for managing SMS. On the left is a menu with the following items:

- SMS
- SMS Inbox
- SMS SentBox
- SMS New Message
- SMS Failed Box

Two red boxes highlight 'SMS Inbox' and 'SMS SentBox' in the menu. Blue arrows point from these boxes to two larger screenshots on the right:

① **GSM SMS / InBox**

This screenshot shows a table with the following columns: Index, Sender, Received, Message, and Select. The table is currently empty. Below the table, there is a grey bar with the text: 받은 메시지가 저장됩니다. (Received messages are stored.)

② **GSM SMS / Sent Box**

This screenshot shows a table with the same columns as the InBox view. The table is currently empty. Below the table, there is a grey bar with the text: 보낸 메시지가 저장됩니다. (Sent messages are stored.)

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

SMS

1. SMS -2

The screenshot displays the SMS management interface. On the left, a blue 'SMS' menu is expanded to show four options: 'SMS Inbox', 'SMS SentBox', 'SMS New Message', and 'SMS Failed Box'. The 'SMS New Message' and 'SMS Failed Box' options are highlighted with red boxes. Two blue arrows point from these options to the right-hand panels.

① GSM SMS / New Message
Max size is 80 characters.
Phone Number:
Message:
Port:
Send

메시지를 보낼 수 있습니다.

② GSM SMS / Failed Box
number of messages are 0 P0:0 OK
Index Sender Received Message Select
< > Delete

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

The screenshot shows the NTP configuration page. On the left, the 'System' menu is expanded, and 'NTP' is selected. The main configuration area has a title 'NTP' and a 'Enable/Disable' toggle. Below are four input fields: 'Primary Server' (Domain Name or IP Address), 'Secondary Server' (Domain Name or IP Address), 'Interval' (1-72 hours), and 'Hours Offset' (-23-23 hours : 0-60 minute). At the bottom left, the 'Apply' button is highlighted with a red box and a green starburst labeled 'Click'.

- 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

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- WEB Callback
- GSM Callback

Gain & CID

Gain

Port	Port Type	InputGain	OutputGain	Caller ID
P0:0	GSM	0	0	<input type="checkbox"/>
P0:1	GSM	0	0	<input type="checkbox"/>
P0:2	FXS	0	0	<input checked="" type="checkbox"/>
P0:3	FXS	0	0	<input checked="" type="checkbox"/>

Apply

Click

-18
-17
-16
-15
-14
-13
-12
-11
-10
-9
-8
-7
-6
-5
-4
-3
-2
-1
0
1
2
3
4
5
6
7
8
9

① Gain & CID : Adjustment output voice level of each port(GSM, FXS)

(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

Advanced Service

3. GSM PINs

Advanced ⊖

- Gain & CID
- GSM PINs**
- Fax
- Service
- Filtering
- Security
- WEB Callback
- GSM Callback

GSM PINs

PINs ⓘ

Port	PIN for SIM card
P0:0	<input type="text"/>
P0:1	<input type="text"/>

Apply **Click**

Advanced Service

4. GSM Band

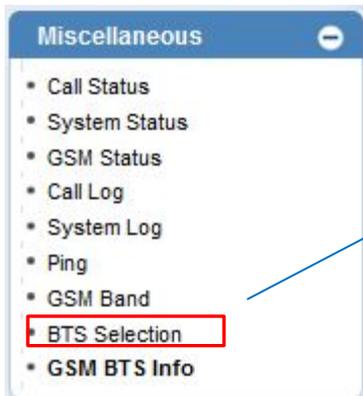


GSM Band Selection : Setting for bandwidth of GSM Networks

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

Advanced Service

5. BTS(Base Terminal Station) -1



GSM / BTS Control

Port	BTS Selection Mode	BCCH	RSSI & Timer	
P0:0	Auto	N.A.	N.A.	
P0:1	Auto	N.A.	N.A.	
P0:0	Auto	<input type="text"/>	-10 dB	0 sec

Apply

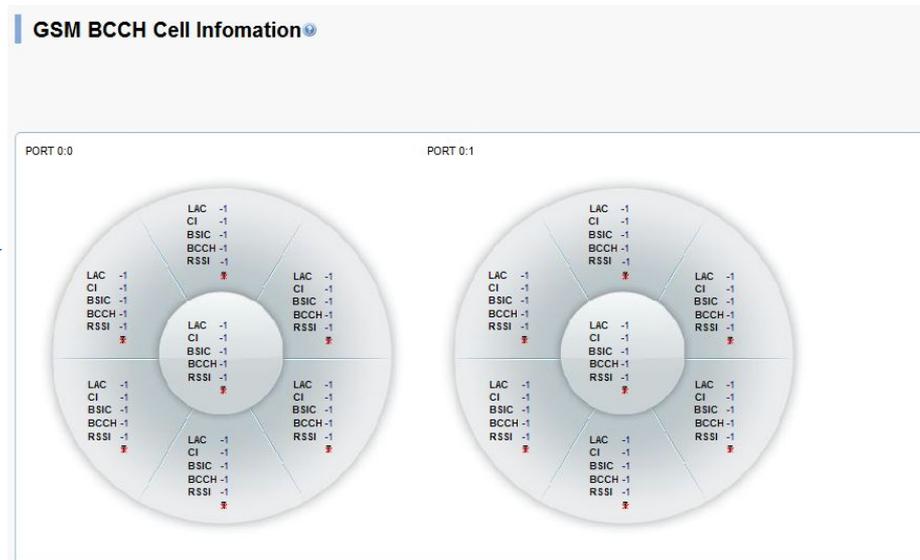
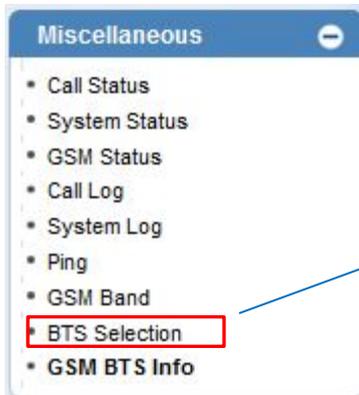
Auto
BCCH
RSSI

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

Advanced Service

6.

FAX

The screenshot shows the 'Advanced' configuration menu on the left, with 'Fax' highlighted. A blue arrow points from 'Fax' to the 'Fax' configuration page. The 'Fax' page has a red box around the title and a circled '1' above it. The 'Fax Mode' section has radio buttons for 'T.38' (selected), 'Inband T.38', and 'Bypass'. The 'Fax Rate' section has radio buttons for 'Disable', '2400', '4800', '7200', '9600' (selected), '12000', and '14400'. At the bottom, there is a green checkmark icon and the text 'Apply', with a green starburst and the word 'Click' pointing to it.

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

7. Service

①

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service**
- Filtering
- Security
- WEB Callback
- GSM Callback

Service

Application Services	<input checked="" type="checkbox"/> Enable Telnet	Server Port	23 (default 23, 1-65535)
	<input checked="" type="checkbox"/> Enable HTTP	Server Port	80 (default 80, 1-65535)
	<input checked="" type="checkbox"/> Enable FTP	Control Port	21 (default 21, 1-65535)
		Data Port	20 (default 20, 1-65535)
Timer	Inter Digit Time	3 sec (default 3, 1-600)	
Call Service	Transfer	<input type="radio"/> Hook-Flash <input checked="" type="radio"/> Not-assigned	
	Hold	<input type="radio"/> Hook-Flash <input checked="" type="radio"/> Not-assigned	
SIP Transfer	Mode	<input checked="" type="radio"/> blind <input type="radio"/> Attended	

Apply

Click

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

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- WEB Callback
- GSM Callback

Filter

FTP Filter

Network Addr	Network Mask	Control
<input type="text"/>	<input type="text"/>	Add

HTTP Filter

②

Network Addr	Network Mask	Control
172.17.109.109	255.255.0.0	Delete
<input type="text"/>	<input type="text"/>	Add

Telnet Filter

Network Addr	Network Mask	Control
<input type="text"/>	<input type="text"/>	Add

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

The screenshot shows the 'Advanced' service configuration page. On the left, a sidebar menu lists various settings: Gain & CID, GSM PINs, Fax, Service, Filtering, Security, WEB Callback, and GSM Callback. The 'Security' option is highlighted with a red box and a blue arrow pointing to the main content area. The main content area is titled 'Security' and contains four configuration sections: 'IP Filtering' with radio buttons for 'Enable' and 'Disable' (selected); 'WarDialing Filtering' with radio buttons for 'Enable' and 'Disable' (selected); 'Allow Digit Length(IP to PSTN)' with 'Min' and 'Max' input fields; and 'SIP Shutdown' with radio buttons for 'Enable' and 'Disable' (selected). At the bottom left, there is an 'Apply' button with a green checkmark icon, which is highlighted with a red box and a green starburst graphic with the word 'Click' written inside.

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

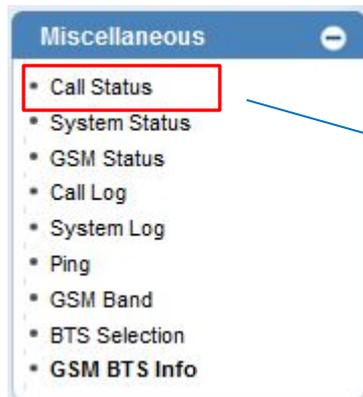
* Please click the apply button after set up

Monitoring

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

Monitoring

1. Call Status



Call Status

① Port Status (Analog)

Slot	Port	Port Group			
		0(GSM)	1(GSM)	2(FXS)	3(FXS)
SLOT 0	Status	R	I	I	I
	Select	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Unlock Block

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

② Call Status

Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
0/0	GSM Out	N.A.	500	2000	N.A.	N.A.
0/0	GSM In	N.A.	1000	2000	N.A.	N.A.

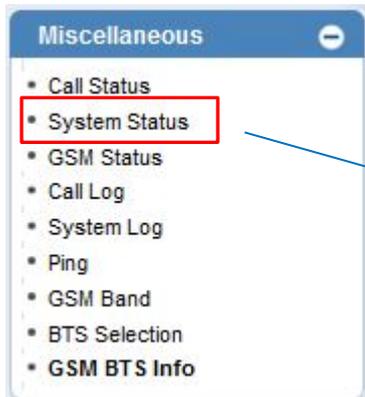
Call Status : The status of AP-GS1002 port and call can be monitored in real-time

① Port Status : Monitoring AP-GS1002 port

② Call Status : Monitoring call status

Monitoring

2. System Status



System Status

Voice Port

Port	LineType	Status	InGain	OutGain	TieType	TieDigits	CallNum	Toalled	Toalling
0/0	GSM	Idle	0	0	none		-1	-1	-1
0/1	GSM	Idle	0	0	none		-1	-1	-1
0/2	FXS	Idle	0	0	none		-1	-1	-1
0/3	FXS	Idle	0	0	none		-1	-1	-1

SIP-UA

Proxyserver Registration Information
proxyserver registration option = e164
Proxyserver list :
No Proxyserver Information.

SIP UA Timer counters
retry counter = 10

SIP UA Timer values
tretry (sip retry timer) = 500 msec.
tinterval (sip retry max interval timer) = 4 sec.
treg (sip register timer) = 60 sec.
tcregtry (sip register retry timer) = 20 sec.
texpiries (sip invite expire timer) = 180 sec.
tstipping (sip ping timer) = 45 sec.

SIP UA Session Timer value
Min-SE = 1800 sec.
Session-Expires = 1800 sec.

SIP DNS SRV Query : Disable
SIP Call Transfer Mode : Basic
SIP Media Channel Start Mode : Default
SIP Reliable Provisional Response Option : Supported with value <100rel>
SIP Response Option : default
SIP Local Domain : NULL
SIP Special Char : NULL
SIP Routing Method of Incoming Call : Default
SIP Remote-Party-ID : Disabled
SIP Local Host Name : No
SIP Conference Server Info
Name (ID) = NULL
Related Voip Tag = -1

SIP NAT Info
PING = Disabled
Required = NULL

SIP Session Refresh Method = INVITE
SIP Keep Authentication information on registration = Yes
SIP Message Parameter Translation TRUE
SIP Force-Forwarding Info
SIP Hook-Flash Event (INFO) Ignore = FALSE
SIP Time Sync With REGISTER Msg = FALSE

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

Monitoring

3. GSM Status



A screenshot of the 'GSM Status' page. It features a table titled 'GSM Port Status & Information' with columns for Port, My Phone Number, Register Status, Signal Strength, Voice Quota(secs), and SMS Quota(E.A.).

Port	My Phone Number	Device Information		Accounting (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

CallNum	EventTime	Describe	CallingPartyNum	CalledPartyNum	RemoteInfo	SetupTime	Dur	Reason
< 2>	Apr 21 13:10:58	local	1000	2000	:			0 Local:Management
< 1>	Apr 21 13:10:52	incoming	500	2000	:			0 Local:Management

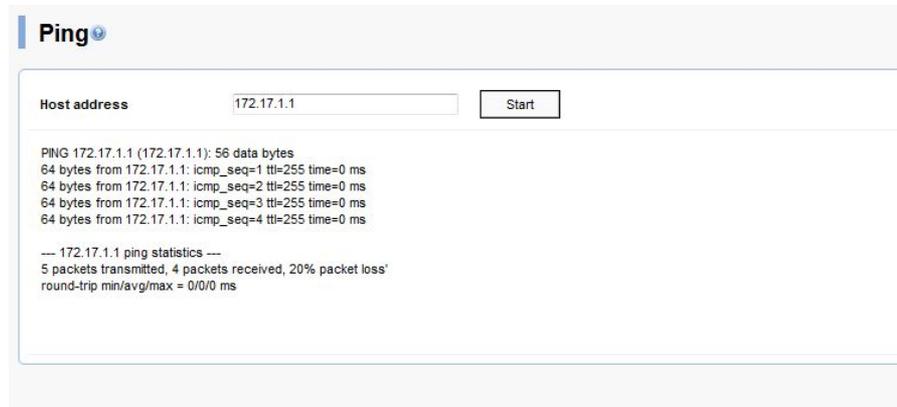
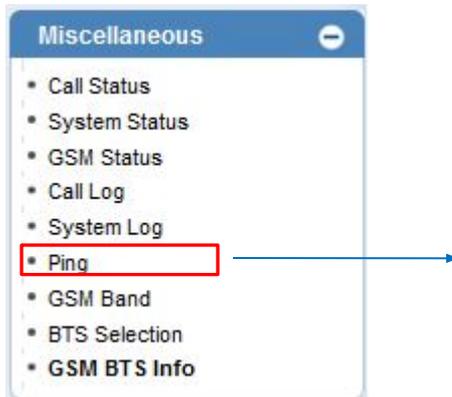
Call Log : Monitoring all of call history
✘ Call history will be clear with rebooting



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