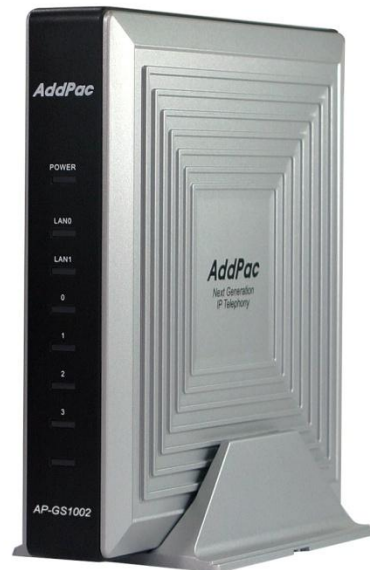


GSM Gateway

High Performance GSM Gateway Solution

WEB Setup Guide

AP-GS1001 / AP-GS1002 / AP-GS1004



AddPac

Contents

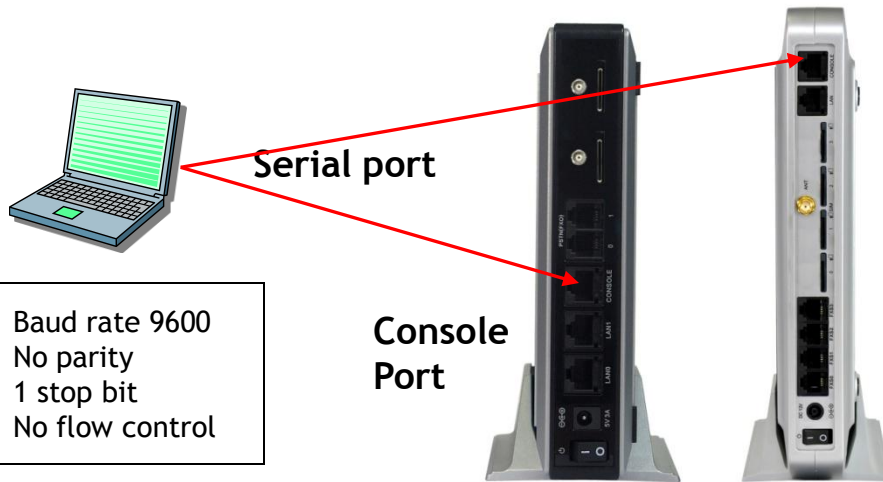
1. WEB Connection
2. Network Setup
3. Language
4. GSM Setup
5. VoIP Setup
6. Callback Service
7. LCR
8. SMS
9. Advanced Service
10. Monitoring

WEB Connection

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

WEB Connection

1. Web Connection via Console Port (AP-GS1002, AP-GS1004)



Baud rate 9600
No parity
1 stop bit
No flow control

1. Connect to GSM G/W 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

*** Following information is not applicable to AP-GS1001**

login ID/Password : root/router

```
GSM> enable
GSM# configure terminal
GSM(config)# interface FastEthernet 0/0
GSM(config-if)# ip address <IP Address> <Subnet Mask>
GSM(config-if)# exit
GSM(config)# ip route 0.0.0.0 0.0.0.0 <Default Gateway>
GSM(config)# write
Proceed with write? [confirm]y
```

=> Enter the enable mode
=> Enter the configuration mode

=> Ex) ip address 172.17.109.1 255.255.0.0

=> Ex) ip route 0.0.0.0 0.0.0.0 172.17.1.1
=> saving

WEB Connection

2. Connection Web via LAN 1 Port (AP-GS1001, AP-GS1002)



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

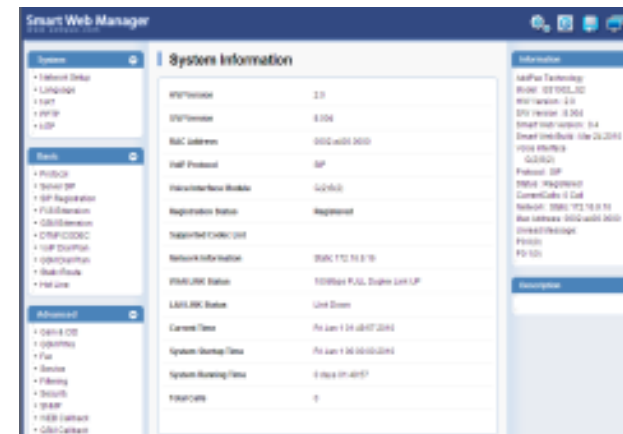
1. The screen of Web connection



- 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

System

- Language
- **WAN Setup**
- LAN Setup
- NAT
- NTP
- File Browser

① WAN & Tunneling Setup

WAN Setup

Hostname: GS1002

IP Address: 172.17.206.208 A.B.C.D
Network Mask: 255.255.0.0 A.B.C.D
Default Router: 172.17.1.1 A.B.C.D
DNS Server: Primary DNS Server
Secondary DNS Server

☒ Static IP

User name:
Password:
Authentication: ☒ (No Authentication)
☐ PAP (PPP Authentication Protocol)
☐ CHAP (Challenge Handshake Authentication Protocol)

☐ PPPoE(ADSL)

☐ DHCP

☐ VLAN ID: 0

☒ Auto

WAN Link Control: ☐ Manual Speed: ☐ 100 ☐ 10
Duplex: ☐ full ☐ half

MAC(Hardware) Address:

Apply

Click

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

The screenshot displays the 'System' menu on the left with 'WAN Setup' highlighted. A blue arrow points from 'WAN Setup' to the 'Tunneling Setup' page. The 'Tunneling Setup' page is titled 'Tunneling Setup' with a circled '1' next to it. It contains the following sections:

- Mode:** Radio buttons for 'None(Disable Tunneling, default)', 'PPTP(Point-to-Point Tunneling Protocol)', 'Authentication' (with sub-options: 'No Authentication', 'PAP(PPP Authentication Protocol)', 'CHAP(Challenge Handshake Authentication Protocol)', 'None (Default)'), and 'Phone Number' (with sub-options: 'Hostname (Use hostname as phone number)', 'User Define').
- Source:** A dropdown menu currently set to 'FastEthernet0/0'.
- Destination:** A text input field followed by the label 'A.B.C.D (Tunnel End Point Address)'.
- Service:** Radio buttons for 'Voice and Data Use Tunnel Interface (default)', 'Voice Use Tunnel Interface, Data Use Ethernet Interface', and 'Data Use Tunnel Interface, Voice Use Ethernet Interface'.
- Apply Button:** A green button with a checkmark and the text 'Apply', highlighted with a red box and a green starburst containing the word 'Click'.

- 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

System

- Language
- WAN Setup
- **LAN Setup**
- NAT
- NTP
- File Browser

LAN Setup

☐ None

☐ IP Share (IP Connect)

☒ Static

IP Address: 192.168.10.1 (A.B.C.D (default 192.168.10.1))

Network Mask: 255.255.255.0 (A.B.C.D (default 255.255.255.0))

☐ Enable DHCP Server

Default Lease time: 0 (in seconds, default 86400, 1 day)

Address Range: [] - []

DNS Server: [] (A.B.C.D)

☒ Apply

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

GSM Dial Plan / Prefix

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

Plan Table

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

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

1. 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"/>

0

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"/>

0

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

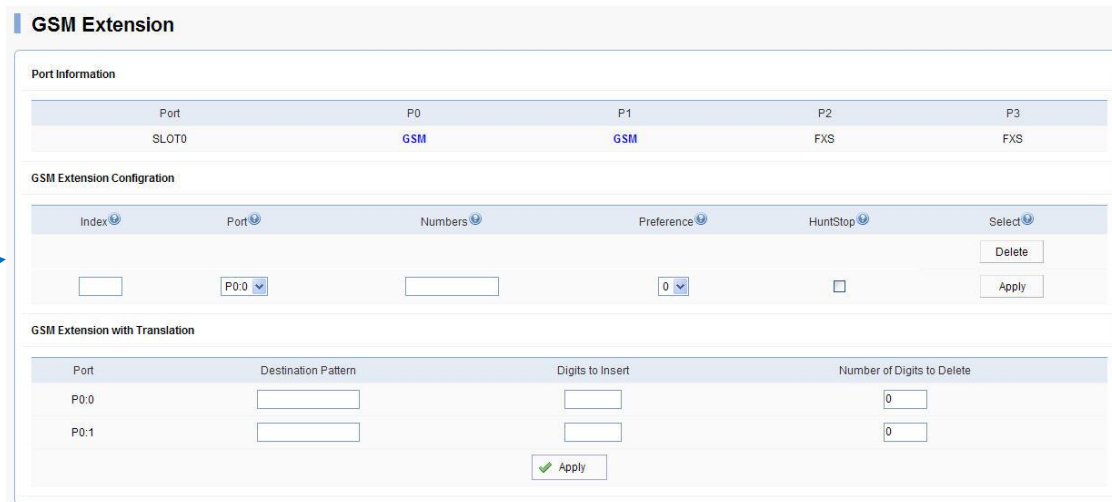
GSM Setup > GSM Extension

2. GSM Extension -1



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 Extension

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

GSM Extension Configuration

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

GSM Extension with Translation

Port	Destination Pattern	Digits to Insert	Number of Digits to Delete
P0:0	<input type="text"/>	<input type="text"/>	0
P0:1	<input type="text"/>	<input type="text"/>	0

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

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

FXS Extension

Port Information

Port	P0	P1	P2	P3
SLOT 0	GSM	GSM	FXS	FXS

FXS Extension Configuration

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

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	P2	P3
SLOT0	GSM	GSM	FXS	FXS

FXS Extension Configuration

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

P0:2

0

☐

Delete

Apply

- 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..

Direct Incoming call

4. GSM Setup > Hot Line -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**

Hot Line

① **Hot Line Configuration**

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

Apply

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)

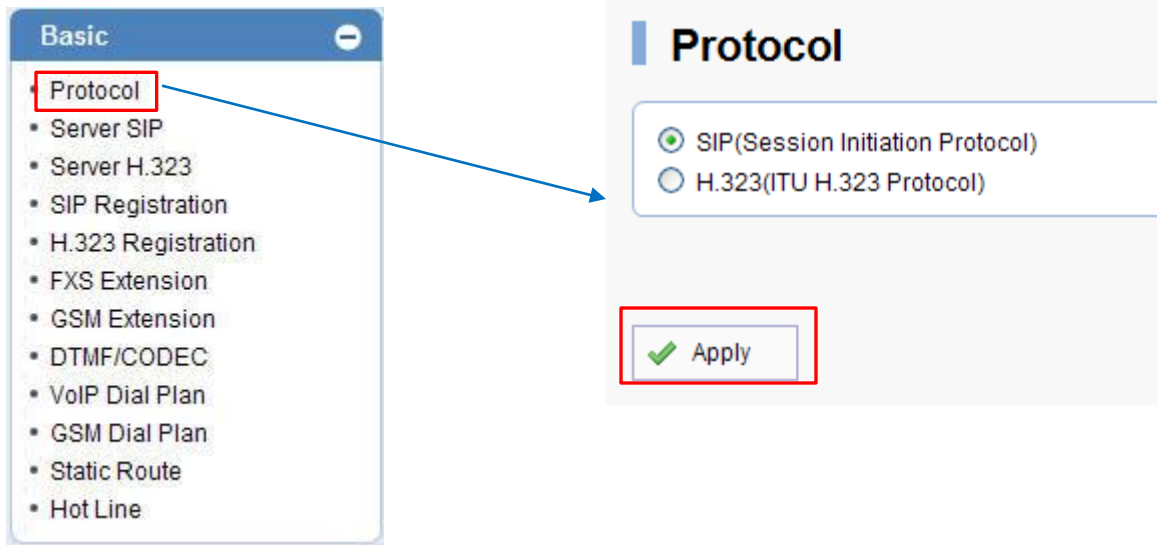
- Digit Input Timeout : Time to make call to the Hot Line Number when user doesn't any action after off-hook

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

1. SIP Setup

VoIP Setup (SIP)

1. Server SIP

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

SIP (Session Initiation Protocol)

Use SIP Server ☐ Yes ☒ No

Primary SIP Server Server address and Port (default 5060)

Secondary SIP Server Server address and Port (default 5060)

Local Domain name (SIP userpart of authentication)

SIP Signaling Port (default 5060, between 1 to 65535)

Register Expiration (in seconds, default 60, between 10 to 86400)

Session Re-Fresh ☒ INVITE ☐ UPDATE

Session Expire Time (in seconds, default 1800, between 30 to 86400, 0 = disable)

Min-se (in seconds, default 1800, between 30 to 86400)

☒ Apply

Click

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

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

① SIP Registration

SIP Registration Configuration

Port	E.164 Number	User Name	Password	DisplayName	HuntStop	Reg
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"/>

☒ Apply

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)

SIP Registration

SIP Registration Configuration

Port	E.164 Number	User Name	Password	DisplayName	Reg	HuntStop
SIP2	8888	1234	*****	5683848	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
SIP3					<input type="checkbox"/>	<input type="checkbox"/>

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)

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

2. H.323 Setup

VoIP Setup (H.323)

1. Server H.323

The screenshot displays the 'Basic' configuration menu on the left, with 'Server H.323' highlighted and a blue arrow pointing to the 'H.323 (ITU H.323 Protocol)' configuration page on the right. The right page contains the following fields and options:

- Use H.323 Server:** Radio buttons for 'Yes' and 'No' (selected).
- Primary Gatekeeper:** Text box with '1719' and a label 'Server address and Port (default 1719)'.
- Secondary Gatekeeper:** Text box with '1719' and a label 'Server address and Port (default 1719)'.
- H.323 ID:** Text box with 'voip.172.17.206.208' and a label '(H.323 Identifier string)'.
- H.323 Signaling Port:** Text box with '1720' and a label '(default 1720, between 1 to 65535)'.
- H.323 Call start mode:** Radio buttons for 'Fast' (selected) and 'Slow'.
- H.323 Tunnel mode:** Radio buttons for 'Enable' (selected) and 'Disable'.
- Apply:** A green button with a checkmark icon, highlighted with a red box.

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

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

H.323 Registration

H.323 Registration

Port	Number	HuntStop	REG
S0P1	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>

☒ Apply

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

3. VoIP Setup

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)

Smart Web Manager
www.addpac.com

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"/>
				<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"/>
			<input type="button" value="Apply"/>

It must be the same with PlanIndex Number

VoIP Setup

2. Static Route -1

Smart Web Manager
www.addpac.com

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="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<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

2. Static Route -2 (Example)

Smart Web Manager
www.addpac.com

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"/>

*

Click

* Please press the apply button to save

VoIP Setup

3. DTMF/CODEC

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

DTMF/CODEC

① Voice CODEC

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

② DTMF Relay mode

☐ 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

Apply

Click

* Please press the apply button to save

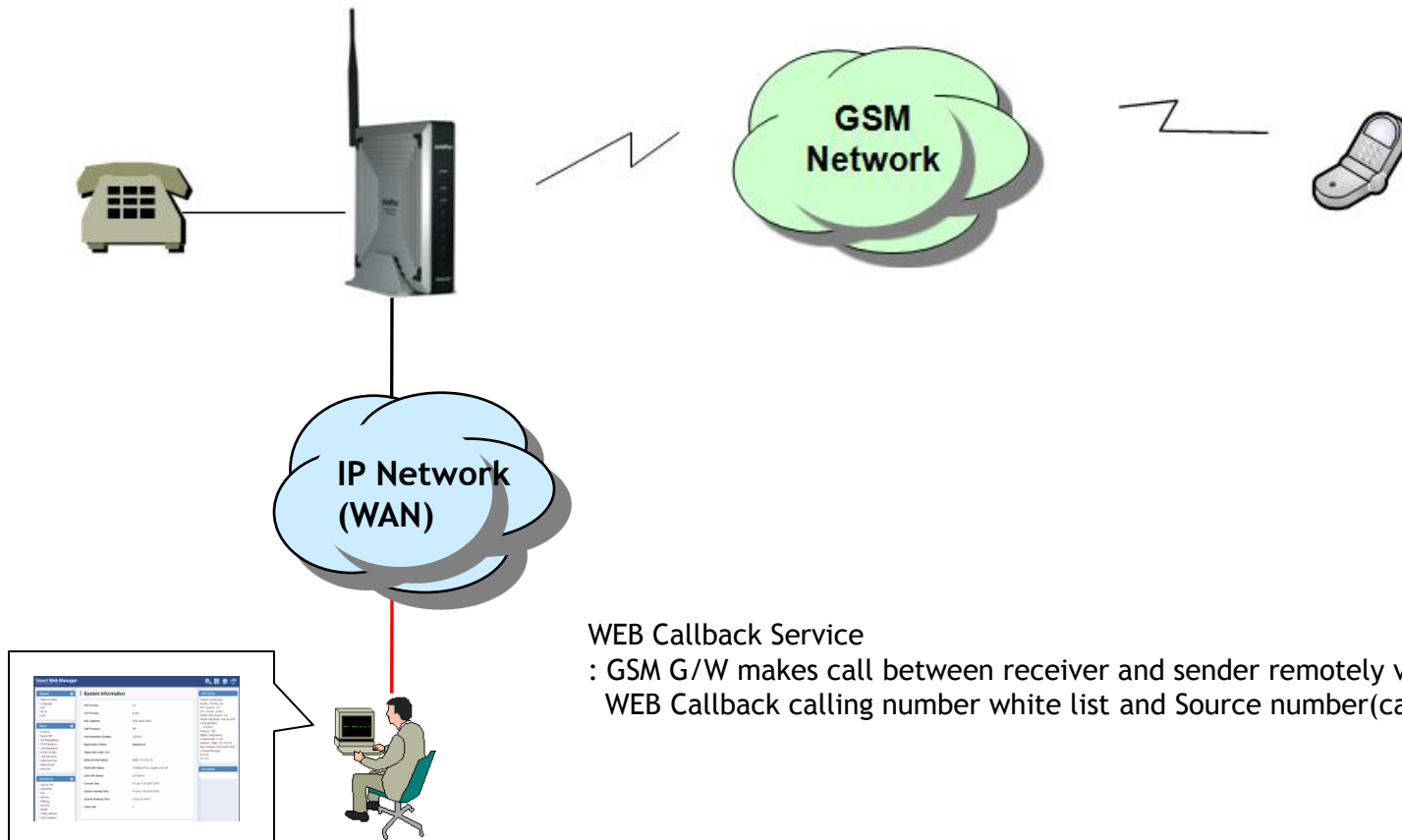
g711alaw : G711 a-law Codec Type(64 kbps)
g711ulaw : G711 u-law Codec Type(64 kbps)
g7231r53 : G723.1 Codec Type(5.3 kbps)
g7231r63 : G723.1 Codec Type(6.3 kbps)
g726r16 : G726 ADPCM Type(16 kbps)
g726r32 : G726 ADPCM Type(32 kbps)
g729 : G729 Codec Type(8 kbps)
None

Callback Service

1. WEB Callback Service
2. GSM Callback Service

Callback Service

WEB Callback Service



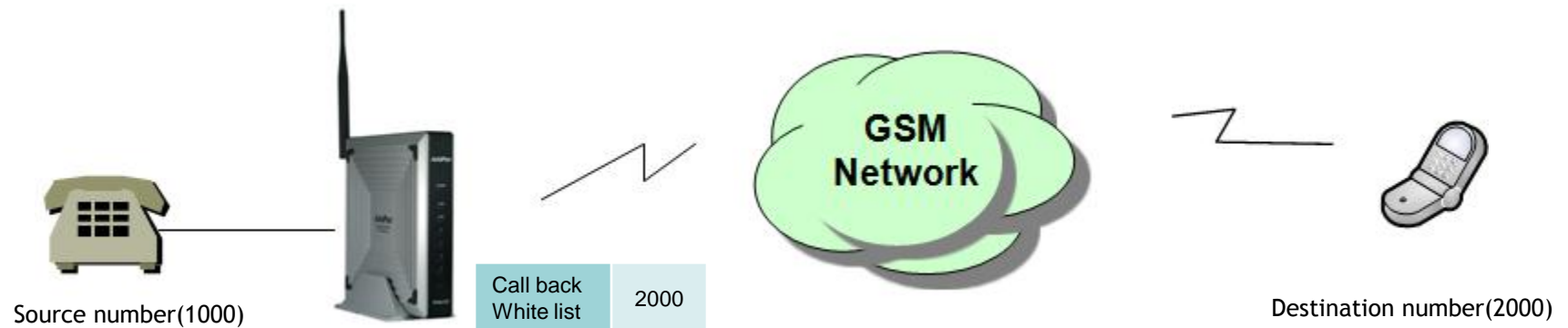
WEB Callback Service

: 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

Advanced

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

GSM WEB Callback

① **Calling Number Whitelist**

Index	DiaPattern	Control
0		Delete
		Add

② **WEB Callback**

Destination Numbers	Source Numbers	Control
		Apply


- 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)

The screenshot displays the 'Calling Number Whitelist' and 'WEB Callback' configuration sections. The 'Calling Number Whitelist' table has columns for 'index', 'DialPattern', and 'Control'. The 'DialPattern' column contains the value '500'. The 'WEB Callback' section includes input fields for 'Destination Numbers' (2000) and 'Source Numbers' (500), along with an 'Apply' button. A green starburst labeled 'Click' points to the 'Apply' button. A red dashed box highlights the 'Remote Trying' status at the bottom. Annotations with red lines point from text boxes to the 'DialPattern' field, the 'Apply' button, and the 'Remote Trying' status.

index	DialPattern	Control
0	500	
		Delete
		Add

WEB Callback

Destination Numbers: 2000

Source Numbers: 500

Control: Apply

Remote Trying

DialPattern must be the same as Source Number

Click

Check status of calling with executing WEB Callback

Callback Service > GSM Callback

2. GSM Callback Service -1

Advanced

- Gain & CID
- GSM PINs
- GSM USSD
- Fax
- Service
- Filtering
- Security
- 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

2. GSM Callback Service -1 (Example)

GSM Callback

Calling Number Whitelist

Group	Index	DialPattern	Control
3	0	01099116545	<input type="checkbox"/>
	1	8225683848	<input type="checkbox"/>

3 0

Delete Add

Callback

GSM Port	My Number	WhiteList Group
P0:0		3
P0:1		N.A.

Apply

Click

웹 페이지의 메시지

Update Success

확인

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

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

Black / White List

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

Black / White List

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

GSM LCR / Black List & White List

BlackList

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

Black List : Reject call from specific number

WhiteList

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

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

The screenshot displays the 'LCR' configuration menu on the left, with 'Time Interval' highlighted. An arrow points from this menu item to the 'GSM LCR / Time Interval Group' configuration page. On this page, the 'TimeInterval' tab is selected and highlighted with a red box and a circled '1'. Below the tab is a table with columns: Group, Days, StartTime(hh:mm), EndTime(hh:mm), and Control. The table contains one row with the following values: Group: 0, Days: weekend, StartTime: 0, EndTime: 0. There are 'Delete' and 'Add' buttons in the Control column.

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

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	<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)

3. Tariff Group-1

LCR

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

GSM LCR / Tariff Group

① **Tariff Group**

Group	Time Group	Type	Restore Day	First(sec)	Others(sec)	Voice(min)	SMS(E.A.)	Control
0	0	daily	1					

Buttons: Add, Delete

② **TariffPort**

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

Buttons: Apply

- 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)

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)

4. 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)

4. LCR Test -2 (Example)

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

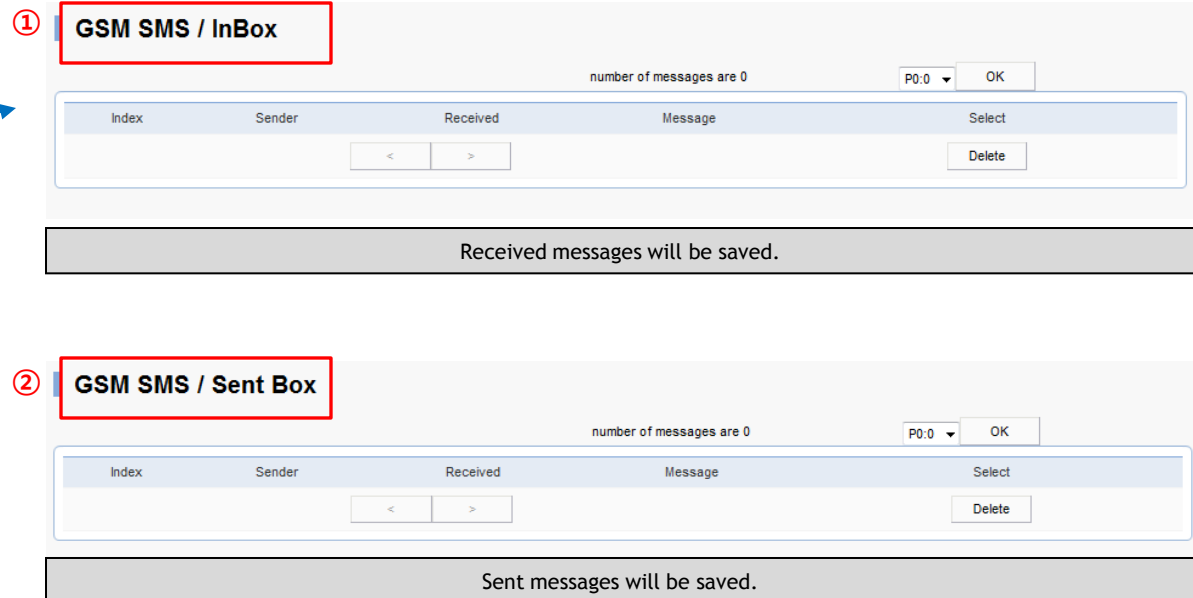
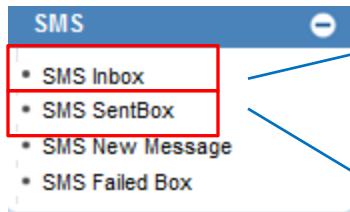
< 9> LCR : -----
< 10> LCR : -- Result : Fail
< 11> LCR : =====
< 12> LCR : == GSM LCR(Least Cost Route) Simulator End ==
< 13> LCR : =====
```

The above message is shown that failure occurred by time interval

Problem can be monitored to make virtual call

SMS

5. SMS -1



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

SMS

5. SMS -2

The screenshot displays the SMS management interface. On the left, a blue sidebar contains the 'SMS' menu with a minus icon. Below it, a list of options is shown: '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

You may send the messages.

② GSM SMS / Failed Box

number of messages are 0

P0.0 OK

Index	Sender	Received	Message	Select
<div>< ></div>				

Delete

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**

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

2. GSM PINs

Advanced

- Gain & CID
- **GSM PINs**
- GSM USSD
- 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

3. GSM USSD

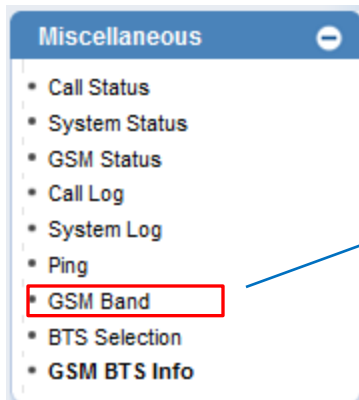
The screenshot displays the 'Advanced' service configuration interface. On the left, a sidebar lists various services: Gain & CID, GSM PINs, **GSM USSD** (highlighted with a red box and a blue arrow), Fax, Service, Filtering, Security, WEB Callback, and GSM Callback. The main content area is titled 'GSM USSD' and contains a table for configuring USSD services.

Port	Data Code	
P0:0	<input type="text"/>	<input checked="" type="checkbox"/> Send
P0:1	<input type="text"/>	<input checked="" type="checkbox"/> Send

A red box highlights the 'Send' button for the 'P0:0' port, with a green starburst and the word 'Click' pointing to it.

Advanced Service

4. GSM Band



A screenshot of the 'GSM / GSM Band Selection' configuration page. It features a table with two columns: 'Port' and 'Band Selection Mode'. The table has two rows for ports P0:0 and P0:1, both set to 'Auto'. Below the table, there is a dropdown menu for 'P0:0' and an 'Apply' button. A dropdown menu is also shown, listing the available band selection modes.

Port	Band Selection Mode
P0:0	Auto
P0:1	Auto

P0:0 ▼
Apply

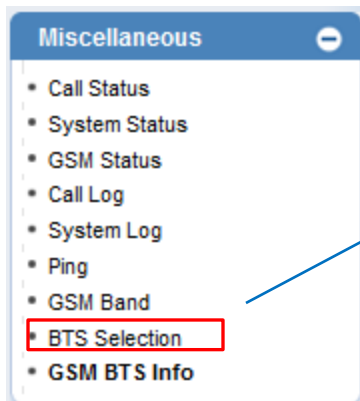
Auto ▼
Auto
GSM 900Mhz+DCS 1800Mhz
GSM 900Mhz+PCS 1900Mhz
GSM 850Mhz+DCS 1800Mhz
GSM 850Mhz+PCS 1900Mhz

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

4. BTS(Base Terminal Station) -1



A screenshot of the 'GSM / BTS Control' window. It contains a table with columns: Port, BTS Selection Mode, BCCH, and RSSI & Timer. The table has two rows for P0:0 and P0:1, both with 'Auto' in the BTS Selection Mode column and 'N.A.' in the BCCH column. Below the table, there is a dropdown menu for 'P0:0' showing 'Auto' selected, and an 'Apply' button. The 'RSSI & Timer' column has a dropdown menu showing '-10 dB' and a dropdown menu showing '0 sec'.

Port	BTS Selection Mode	BCCH	RSSI & Timer
P0:0	Auto	N.A.	N.A.
P0:1	Auto	N.A.	N.A.

P0:0 ▼ Auto ▼
Auto
BCCH
RSSI

Apply

-10 dB ▼ 0 sec ▼

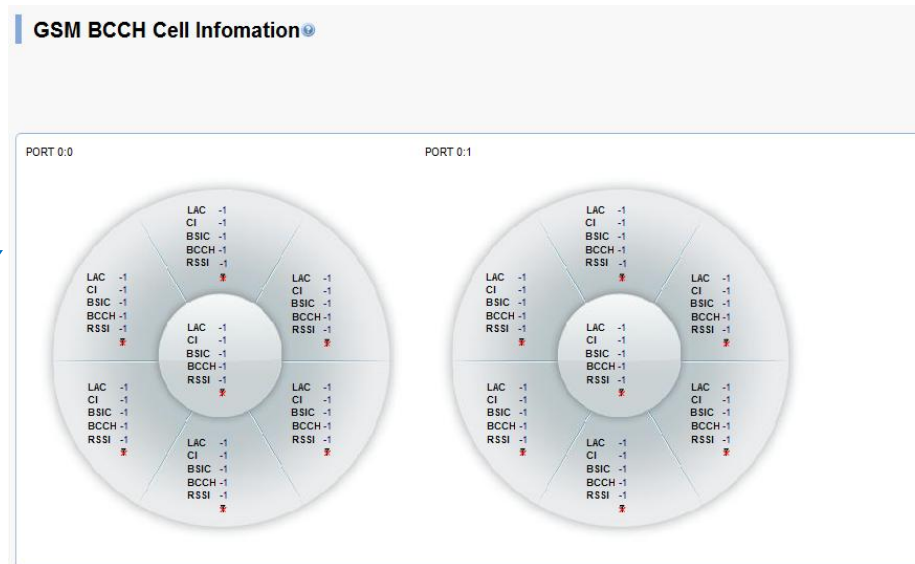
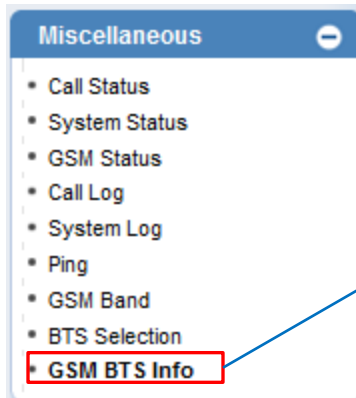
BTS Control : Setting for selection method of BTS cell

- 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. GSM BTS Info



BTS Cell Information

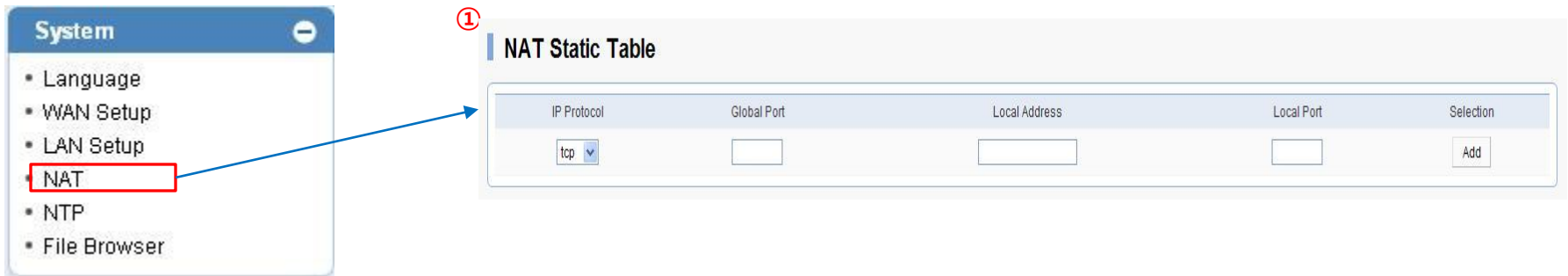
: The information of serve cell and neighbor cell can be shown

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

The image shows a screenshot of the AddPac web interface. On the left, a 'System' menu is visible with options: Language, WAN Setup, LAN Setup, NAT, NTP (highlighted with a red box), and File Browser. A blue arrow points from the 'NTP' menu item to the NTP configuration page on the right. The NTP configuration page has a title bar 'NTP' (also highlighted with a red box) and a sub-header '①'. Below the title bar, there are radio buttons for 'Enable' and 'Disable'. The 'Enable' option is selected. Below this, there are two input fields for 'Primary Server' and 'Secondary Server', both with placeholder text '(Domain Name or IP Address)'. Below these, there is an 'Interval' input field with a placeholder '(1~72 hours)'. Below that, there is a 'Hours Offset' input field with a placeholder '(-23~23 hours) : (0~60 minute)'. At the bottom left of the configuration area, there is a green 'Apply' button with a checkmark icon, highlighted with a red box. A green starburst graphic with the word 'Click' is placed over the 'Apply' button.

- 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

System

- Language
- WAN Setup
- LAN Setup
- NAT
- NTP
- File Browser**

File Browser

Index of Root/flash/

Name	Size	Type	Last Modified
tmp/		Directory	1970-Jan-01 00:00:07
apos.cfg	6.1K	CFG	2010-Jul-24 14:01:40
booter.cfg	0.3K	CFG	2010-Jul-20 22:49:24
booter.cfg~	0.2K		2010-Jul-20 22:49:24
gs3000_g2_v8_41_03T.bin	3.3M	BIN	2010-Jul-24 12:30:38

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

The screenshot shows the 'Advanced' configuration menu on the left, with 'Fax' selected. The main configuration area for 'Fax' is on the right. It includes a 'Fax Mode' section with radio buttons for 'T.38' (selected), 'Inband T.38', and 'Bypass'. Below that is a 'Fax Rate' section with radio buttons for 'Disable', '2400', '4800', '7200', '9600' (selected), '12000', and '14400'. At the bottom of the configuration area is an 'Apply' button with a green checkmark icon. A green starburst with the word 'Click' is pointing to the 'Apply' button.

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

Advanced

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

① 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 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

Advanced

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

Filter

FTP Filter

Network Addr	Network Mask	Control
		Add

HTTP Filter

②

Network Addr	Network Mask	Control
172.17.109.109	255.255.0.0	Delete
		Add

Telnet Filter

Network Addr	Network Mask	Control
		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 (Etc)

7. Security

Advanced

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

Security

IP Filtering ☐ Enable ☒ Disable

WarDialing Filtering ☐ Enable ☒ Disable

Allow Digit Length(IP to PSTN) Min Max

SIP Shutdown ☐ Enable ☒ Disable

☒ Apply

Click

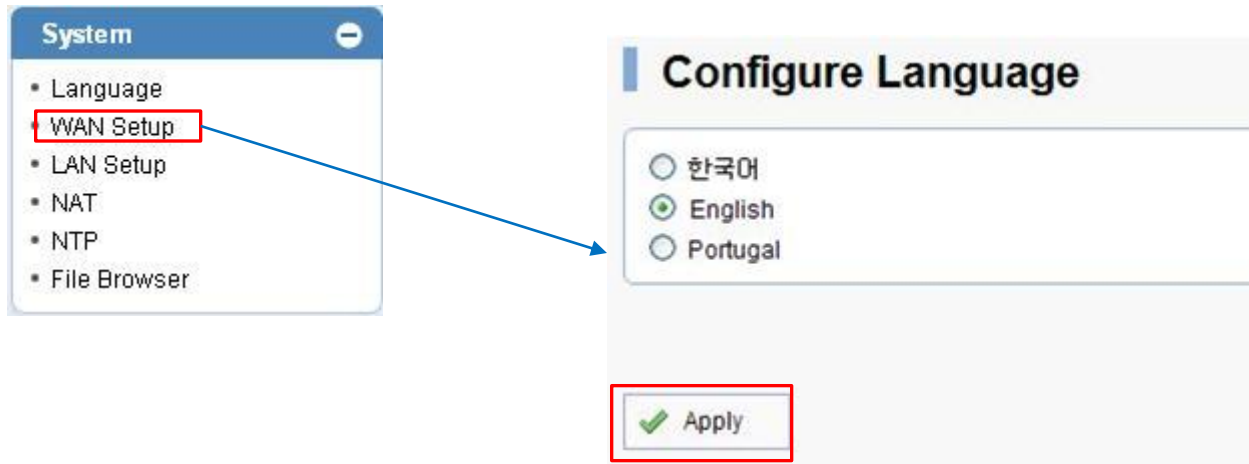
① Security : Set security to block unauthorized call

- 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

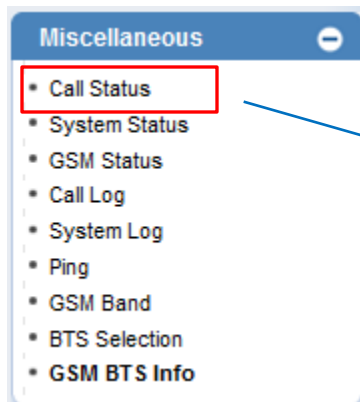
* Language setup will be applied immediately once you click on apply.

Monitoring

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

Monitoring

1. Call Status



Call Status

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"/>

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

② Call Status

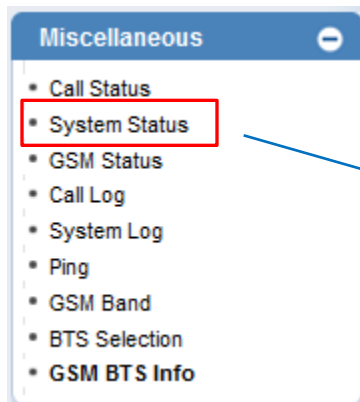
Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
0/0	GSM Out	N.A.	500	2000	N.A.	N.A
0/0	GSM In	N.A.	1000	2000	N.A.	N.A

Call Status : The status of 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

Voice Port

Port	LineType	Status	InGain	OutGain	TieType	TieDigits	CallNum	Tcalled	Tcalling
0/0	GSM	Idle	0	0	none		-1	-1	-1
0/1	GSM	Idle	0	0	none		-1	-1	-1
0/2	FMS	Idle	0	0	none		-1	-1	-1
0/3	FMS	Idle	0	0	none		-1	-1	-1

SIP-UA

Proxyserver Registration Information
proxyserver registration option = el64
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.
tretry (sip register retry timer) = 20 sec.
texpiries (sip invite expire timer) = 180 sec.
tshipping (sip ping timer) = 45 sec.

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

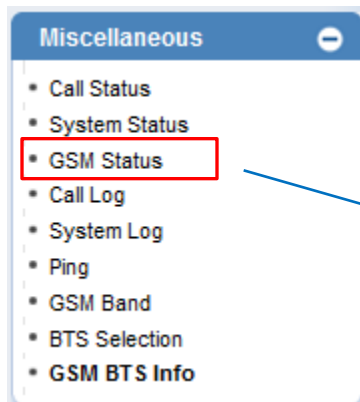
SIP DNS SRV Query : Disable
SIP Call Transfer Mode : Basic
SIP Media Channel Start Mode : Default
SIP Reliable Provisional Response Option : Supported with value <100rel>
SIP Response Option : default
SIP Local Domain : NULL
SIP Special Char : NULL
SIP Routing Method of Incoming Call : Default
SIP Remote-Party-ID : 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 GSM G/W can be monitored

Monitoring

3. GSM Status



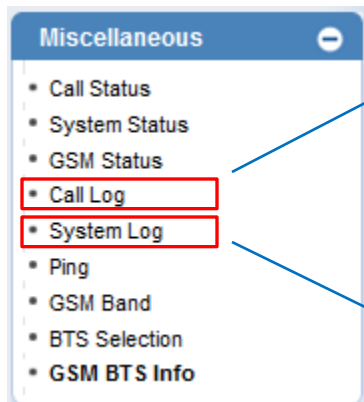
A screenshot of the 'GSM Status' page. It features a table titled 'GSM Port Status & Information' with columns for Port, My Phone Number, Register Status, Signal Strength, Voice Quota(secs), and SMS Quota(E.A.). The table shows two rows: P0:0 (UNREG, 0dB, 0 / -1 / -1) and P0:1 (REG, 0dB, 0 / -1 / -1).

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

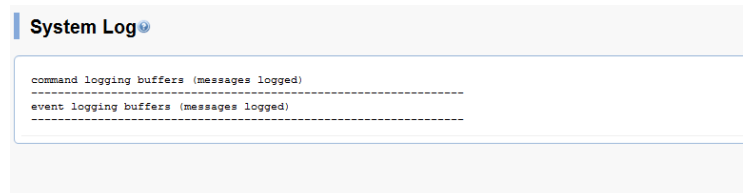
4. Call Log / System Log



Call Log								
CallNum	EventTime	Descript	CallingPartyNum	CalledPartyNum	RemoteInfo	SetupTime	Dur	Reason
< 2>	Apr 21 18:10:58	local	1000	2000	:			0 Local:Management
< 1>	Apr 21 18:10:52	incoming	500	2000	:			0 Local:Management

Call Log : Monitoring all of call history

※ Call history will be clear with rebooting



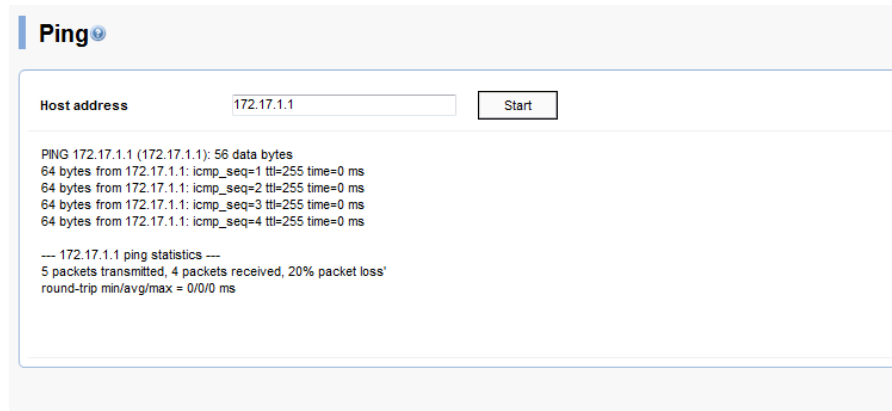
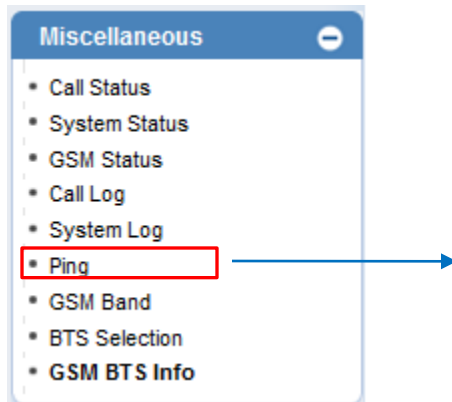
System Log : Monitoring GSM G/W System log

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)

Monitoring

5. Ping



Ping : Network status can be checked by pinging

Thank you