



AP-GS501™

1-Port FXS GSM Gateway

High Performance GSM Gateway Solution

WEB Setup Guide



AddPac

AddPac Technology

2012, Sales and Marketing

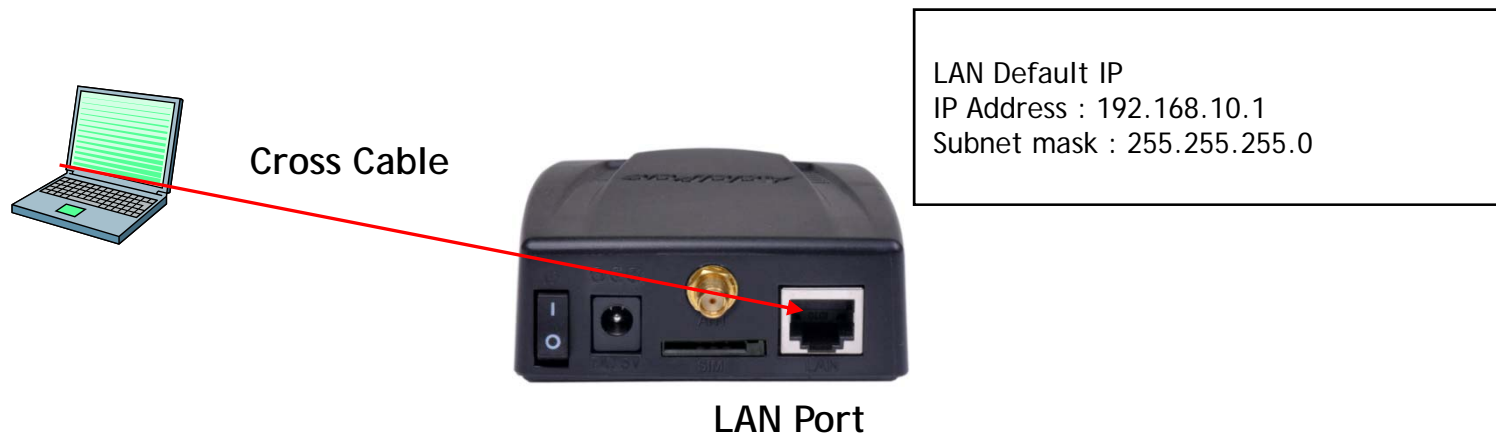
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1. WEB Connection
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5. VoIP Setup
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7. Advanced Service
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WEB Connection

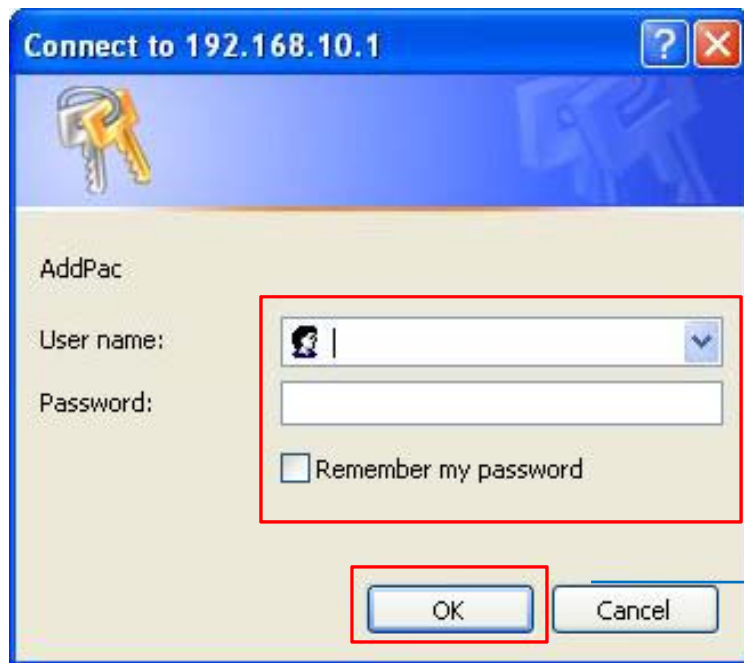
1. Connection Web via LAN Port



1. It is the way to connect to GSM G/W via LAN port
2. The factory default of LAN port
 - IP Address : 192.168.10.1
 - Subnet mask : 255.255.255.0
3. After set PC with same IP address subnet, connect to GSM G/W
 - Connect PC to GSM G/W using Cross UTP-Cable. You may use Ethernet switch with normal UTP-cable
 - Enter IP address 192.168.10.1 on your web browser

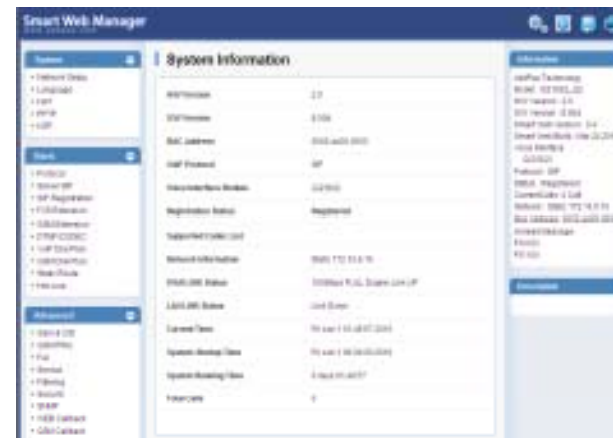
WEB Connection

1. The screen of Web connection



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

ID : root
Password : router



Network Setup

1. WAN Setup -1

The screenshot displays the 'WAN & Tunneling Setup' configuration page. On the left, a 'System' menu has 'WAN Setup' selected. The main configuration area includes:

- WAN Setup:** Hostname: GS1002
- Static IP:** IP Address: 172.17.206.208, Network Mask: 255.255.0.0, Default Router: 172.17.1.1, DNS Server fields.
- PPPoE(ADSL):** User name, Password, and Authentication options (No Authentication, PAP, CHAP).
- DHCP:** ID: 0
- VLAN:** ID: 0
- WAN Link Control:** Auto selected, Speed (100, 10), Duplex (full, half).
- MAC(Hardware) Address:** Five empty input fields.

A red box highlights the 'Apply' button at the bottom left, with a green starburst and the text 'Click' pointing to it.

- 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 shows the 'Tunneling Setup' configuration page. On the left, a 'System' menu has 'WAN Setup' selected. The main configuration area includes:

- 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)', 'Phone Number' (with sub-options: 'Hostname (Use hostname as phone number)', 'User Define')), and 'Service' (with sub-options: 'Voice and Data Use Tunnel Interface (default)', 'Voice Use Tunnel Interface, Data Use Ethernet Interface', 'Data Use Tunnel Interface, Voice Use Ethernet Interface').
- 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)'.
- Apply:** A button at the bottom right, highlighted with a red box and a green starburst labeled '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 A.B.C.D (default 192.168.10.1)

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

Enable DHCP Server

Default Lease time (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"/>

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

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
SLOTO	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"/>	Delete Apply

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

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 > FXS Extension

3. FXS Extension -1

The screenshot shows the 'Basic' menu on the left with 'FXS Extension' highlighted. The main configuration area is titled 'FXS Extension' and contains two sections:

Port Information

Port	P0	P1
SLOT 0	GSM	FXS
	FXS	FXO/E&M
	E1/T1	GSM

FXS Extension Configuration

Pots Num	Port	Numbers	Preference	HuntStop	Control
1512	0/1	1000	0	X	<input type="checkbox"/>

Below the table are input fields for 'P0:1', a text box, a dropdown menu with '0', and a checkbox. 'Delete' and 'Apply' buttons are at the bottom right.

① The each port information of GSM Gateway

② FXS Extension : Set the number of phone on FXS port

- Index : Enter number in order. Please make sure not to be duplicated
- Port : Select FXS port to be set
- Numbers : Enter FXS number
- Preference : Set priority for each number.

If there is the same number at two ports, a port is selected by this priority

- Hunt Stop : It is a function of forward a call to other party in case of unavailable receiving call.
Activation of this function is recommended

GSM Setup > FXS Extension

3. FXS Extension -2 (Example)

FXS Extension

Port Information

Port	P0	P1
SLOT 0	GSM	FXS

FXS Extension Configuration

POTS Num	Port	Numbers	Preference	HuntStop	Control
1512	0/1	1000	0	X	<input type="checkbox"/>

①

P0:1 0

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

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

GSM Setup > FXS Extension

AddPac Digit Structure

※Digit Structure※

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

** T : Accept all number entered within Inter Digit Time (Default IDT : 3sec)

** Dot(.) : One dot(.) means one digit

** [] : The range of number

※Rule tranfer※

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

Direct Incoming call

4. GSM Setup > Hot Line -1

The screenshot shows a web management interface for configuring a Hot Line. On the left, a sidebar menu is visible with the following items: 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, and Hot Line. The 'Hot Line' item is selected and highlighted with a red box. A blue arrow points from this item to the main configuration area. The main area is titled 'Hot Line' and contains a 'Hot Line Configuration' section. This section has a table with the following structure:

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

Below the table is an 'Apply' button with a green checkmark icon.

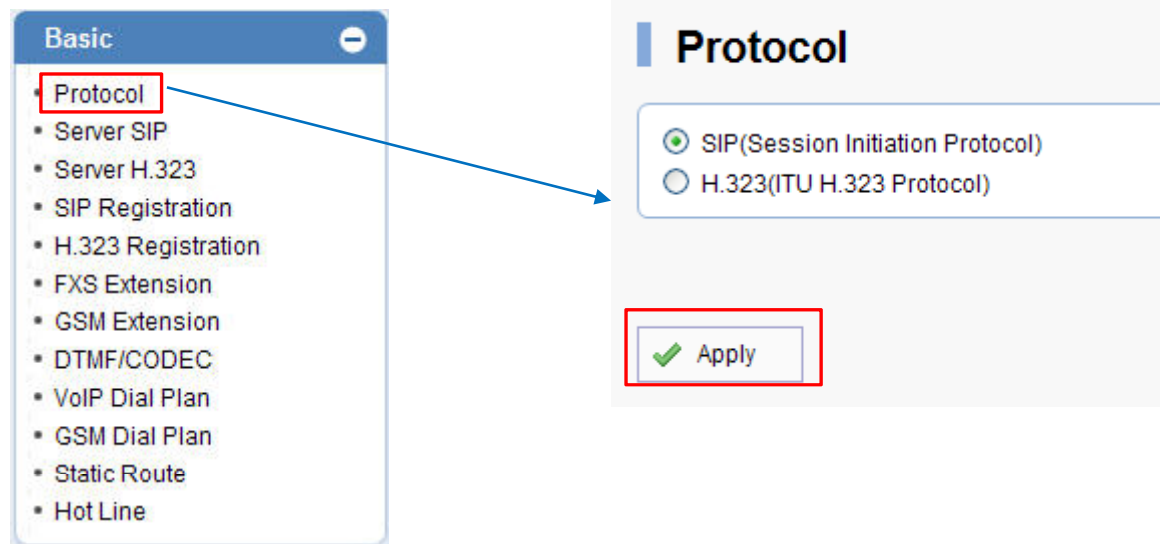
- ① Hot Line Configuration : Connect incoming and outgoing call directly
- Port : It means GSM and FXS port
 - Hot Line Number : Forward call to entered number
It connects to the number of GSM port in case of receiving call (Direct Incoming call)
 - Digit Input Timeout : Time to make call to the Hot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

VoIP Setup

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

VoIP Setup (Protocol)

1. Protocol (SIP or H.323)



- ① Protocol
 - Setup SIP or H.323 Protocol.

VoIP Setup (SIP)

1. Server SIP

① SIP (Session Initiation Protocol)

Use SIP Server Yes No

Primary SIP Server 5060 Server address and Port (default 5060)

Secondary SIP Server 5060 Server address and Port (default 5060)

Local Domain name (SIP userpart of authentication)

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

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

Session Re-Fresh INVITE UPDATE

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

Min-se 0 (in seconds, default 1800, between 30 to 86400)

Apply

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

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"/>

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)

The screenshot displays the SIP Registration configuration interface. It includes a configuration table, an information panel, and a status panel.

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"/>

Information Panel:

- 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)

Status Panel:

- Information
- AddPac Technology
- Model : GS501_G2
- H/W Version : 2.0
- S/W 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 (H.323)

1. Server H.323

The screenshot displays the 'Basic' configuration menu on the left, with 'Server H.323' highlighted and a red box around it. A blue arrow points from this menu item to the 'H.323 (ITU H.323 Protocol)' configuration page on the right. The right page is titled 'H.323 (ITU H.323 Protocol)' and contains the following settings:

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

A red box highlights the 'Apply' button at the bottom left of the configuration page.

① Server H.323

-Use H.323 Server : Select using H.323 Server. Please click "Yes" to use SIP server

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

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

VoIP Setup (H.323)

2. H.323 Registration

① H.323 Registration

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

1. VoIP Dial Plan -1

The screenshot shows the 'VoIP Dial Plan / Prefix' configuration page. On the left, a 'Basic' menu is visible with 'VoIP Dial Plan' selected. The main configuration area contains two tables:

- Plan Table:** This table has columns for Index, Digits to Insert, Number of Digits to Delete, Digit Pattern, and Control. It includes 'Delete' and 'Add' buttons.
- Prefix Table:** This table has columns for Index, Prefix, PlanIndex, and Control. It includes 'Delete' and 'Apply' buttons.

- 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="checkbox"/>

Prefix Table

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

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"/>	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
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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

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

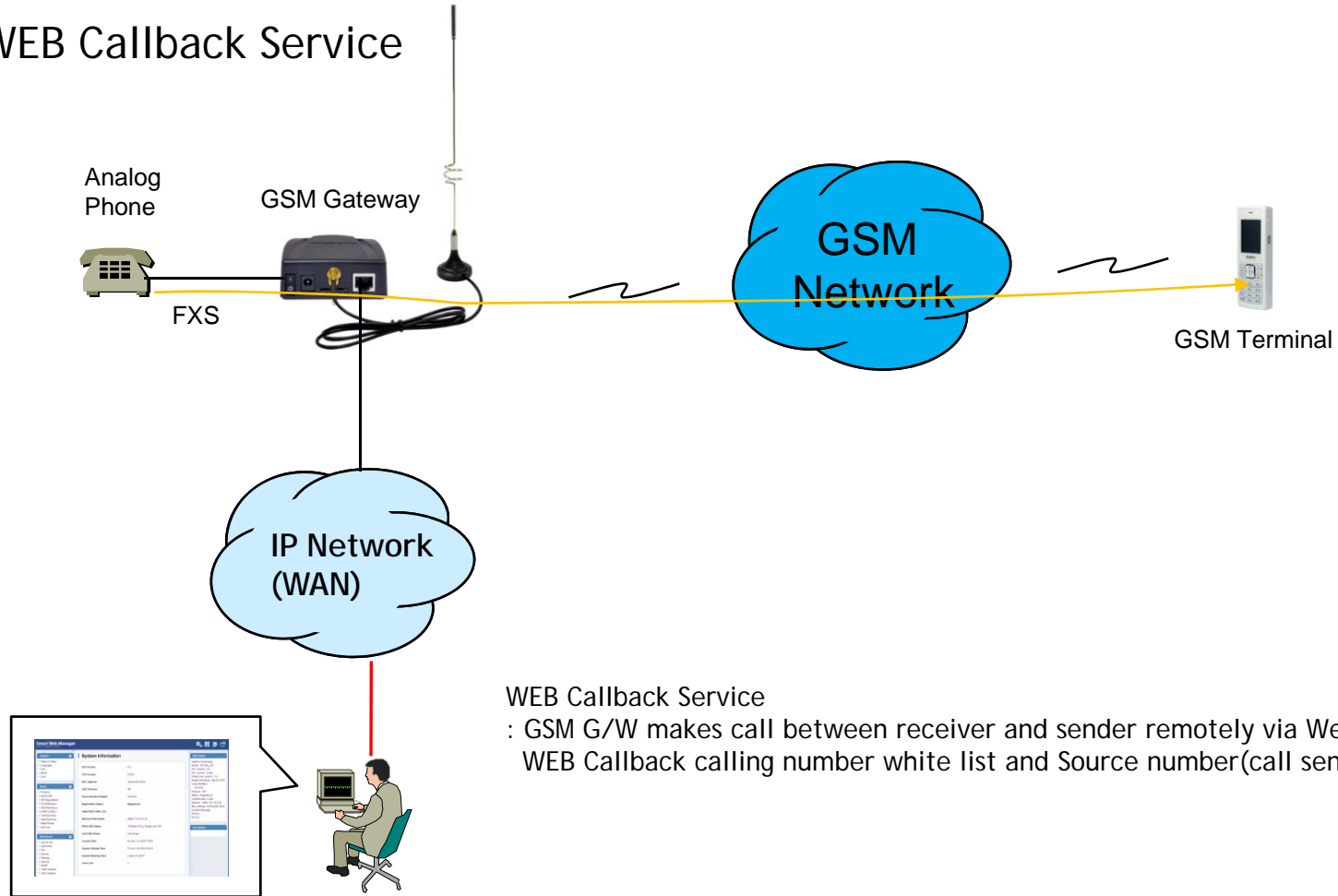


Callback Service

1. WEB Callback Service
2. GSM Callback Service

Callback Service

WEB Callback Service



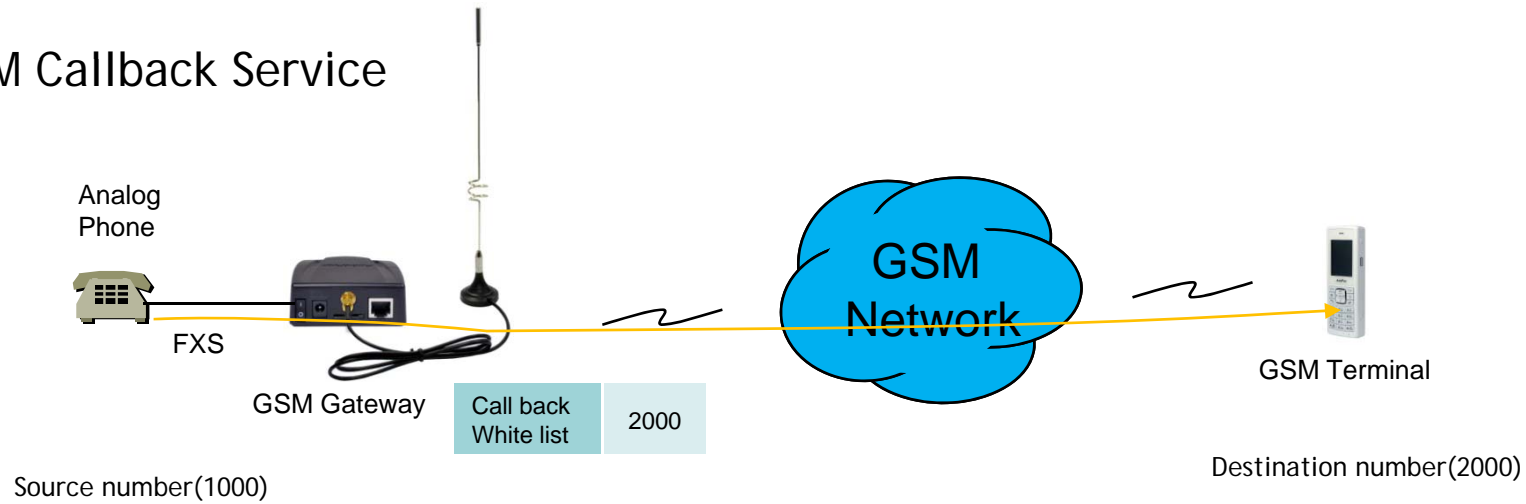
WEB Callback Service

: 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

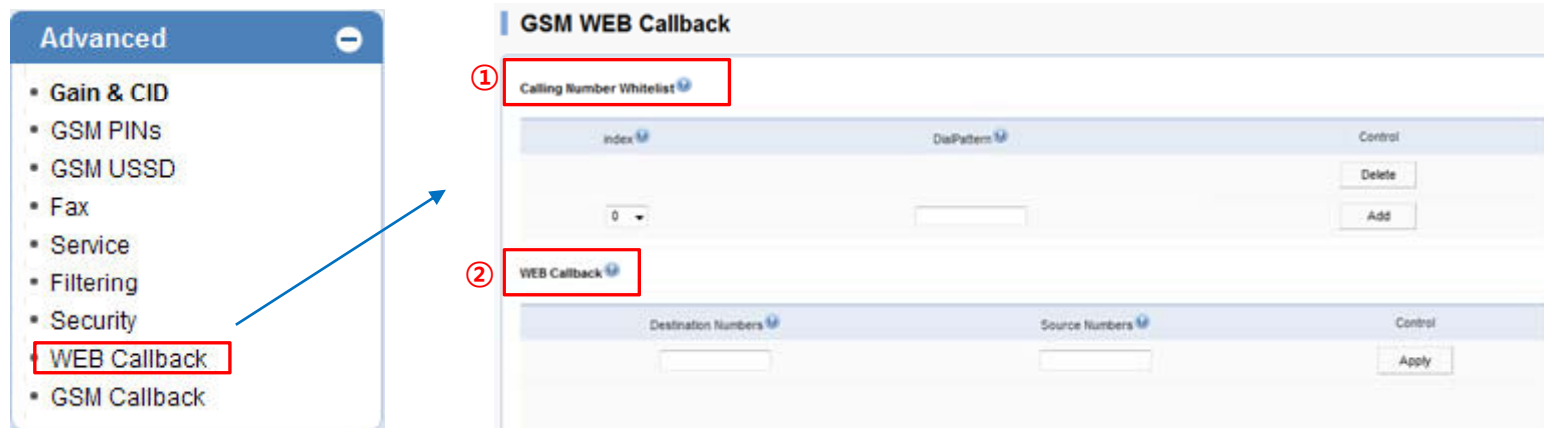


GSM Callback Service

: 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



- 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 two main sections: 'Calling Number Whitelist' and 'WEB Callback'. In the 'Calling Number Whitelist' section, a table has columns for 'index', 'DialPattern', and 'Control'. The first row shows 'index' as 0 and 'DialPattern' as 500. Below the table are input fields for 'index' (set to 0) and 'DialPattern', and buttons for 'Delete' and 'Add'. In the 'WEB Callback' section, there are input fields for 'Destination Numbers' (2000) and 'Source Numbers' (500), and an 'Apply' button. A green starburst labeled 'Click' points to the 'Apply' button. Below this, the status 'Remote Trying' is displayed. Red dashed boxes highlight the 'DialPattern' field in the whitelist and the 'Source Numbers' field in the callback section. Red arrows point from these fields to callout boxes. One callout box states 'DialPattern must be the same as Source Number'. Another callout box states 'Check status of calling with executing WEB Callback'. A blue arrow points from the 'Remote Trying' status back to the 'Apply' button.

index	DialPattern	Control
0	500	

0 ▾

Delete Add

Destination Numbers: 2000 Source Numbers: 500

Apply

Remote Trying

DialPattern must be the same as Source Number

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)

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'.

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

Below the table are dropdown menus for 'Group' (set to 3) and 'Index' (set to 0), and 'Add' and 'Delete' buttons.

Callback: A table with columns 'GSM Port', 'My Number', and 'WhiteList Group'.

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

Annotations:

- A red box highlights the 'Group' column in the 'Calling Number Whitelist' table, with a callout: "Register number to use Callback Service".
- A red box highlights the 'WhiteList Group' dropdown in the 'Callback' table, with a callout: "Enter group number of port to be used".
- A red box highlights the 'Apply' button, with a callout: "Please press the apply button and check pop-up screen".
- A green starburst labeled 'Click' points to the 'Apply' button.
- A screenshot of a pop-up message box titled '웹 페이지의 메시지' (Web Page Message) with a yellow warning icon and the text 'Update Success' and a '확인' (Confirm) button is shown.



Advanced Service

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

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)

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 'GSM LCR / Time Interval Group' configuration page. On the left, a sidebar menu under 'LCR' includes 'Black & White List', 'Time Interval' (highlighted with a red box and an arrow pointing to the main content), 'Tariff Group', and 'LCR Test'. The main content area shows a table with columns: Group, Days, StartTime(hh:mm), EndTime(hh:mm), and Control. The 'TimeInterval' section is highlighted with a red box and a circled '1' icon. Below the table, there are input fields for Group (0), Days (weekend), StartTime (0:00), EndTime (0:00), and buttons for 'Delete' and 'Add'.

Group	Days	StartTime(hh:mm)	EndTime(hh:mm)	Control
0	weekend	0:00	0:00	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	Restore Call Limit	Accounting Period	Free Quota	Control	
Type	RestoreDay	First(sec)	Others(sec)	Voice(min)	SMS(E.A.)	
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 limitaiion
 - 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

The screenshot displays a web interface for LCR Test. On the left, a sidebar menu is visible with the following items: Black & White List, Time Interval, Tariff Group, and LCR Test. The 'LCR Test' item is highlighted with a red box. A blue arrow points from this item to the main content area. The main content area has a header 'LCR Test' with a red box around it and a circled '1' icon. Below the header, there are two input fields: 'Caller:' and 'Called Number:'. A 'Start' button is located to the right of the 'Called Number' field. The main content area is otherwise empty.

- 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:

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 : =====
```

Problem can be monitored to make virtual call

The above message is shown that failure occurred by time interval

SMS

5. SMS -1

The screenshot displays the SMS management interface. On the left, a menu titled "SMS" contains four items: "SMS Inbox", "SMS SentBox", "SMS New Message", and "SMS Failed Box". The "SMS Inbox" and "SMS SentBox" items are highlighted with red boxes. Two blue arrows point from these items to the main interface. The main interface is divided into two sections:

① **GSM SMS / InBox**: This section shows a table with columns for "Index", "Sender", "Received", "Message", and "Select". The "number of messages are 0" is displayed at the top right. Below the table, there are navigation buttons "<" and ">". A "Delete" button is located at the bottom right. A grey bar at the bottom of this section contains the text "Received messages will be saved."

② **GSM SMS / Sent Box**: This section shows a table with columns for "Index", "Sender", "Received", "Message", and "Select". The "number of messages are 0" is displayed at the top right. Below the table, there are navigation buttons "<" and ">". A "Delete" button is located at the bottom right. A grey bar at the bottom of this section contains the text "Sent messages will be saved."

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

SMS

5. SMS -2

The image shows a navigation menu on the left with the following items:

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

Two red boxes highlight 'SMS New Message' and 'SMS Failed Box'. Blue arrows point from these boxes to two screenshots of the application interface:

① GSM SMS / New Message
This screenshot shows a form for sending a new message. It includes fields for 'Phone Number', 'Message', and 'Port', along with a 'Send' button. A note indicates 'Max size is 80 characters'. Below the form is a confirmation message: 'You may send the messages.'

② GSM SMS / Failed Box
This screenshot shows a table of failed messages. The table has columns for 'Index', 'Sender', 'Received', 'Message', and 'Select'. The 'Select' column contains 'Delete' buttons. Above the table, it says 'number of messages are 0' and 'P0:0'. Below the table is a confirmation message: '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

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"/>

Gain Level Scale:

- 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

Advanced

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

1 GSM USSD

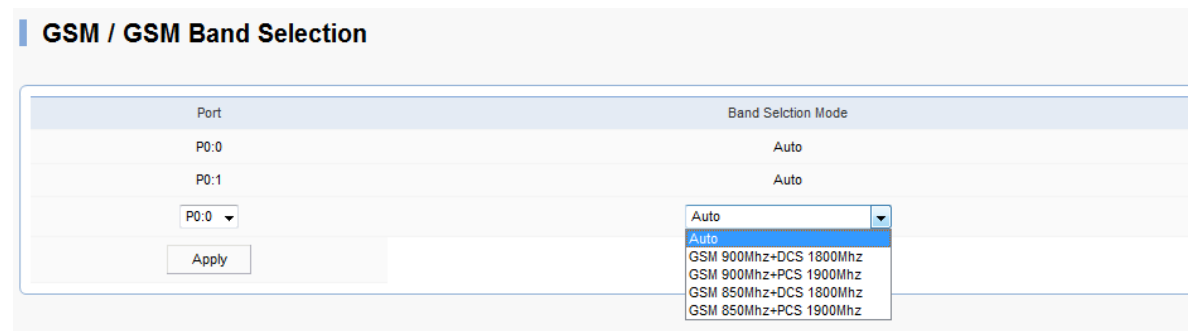
USSD

Port	Data Code	
P0:0	<input type="text"/>	<input type="button" value="✓ Send"/>
P0:1	<input type="text"/>	<input type="button" value="✓ Send"/>

Click

Advanced Service

4. GSM Band

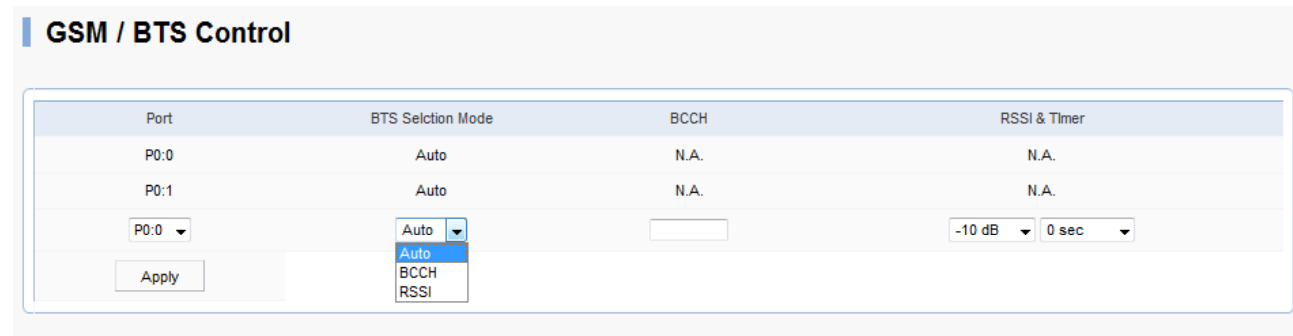
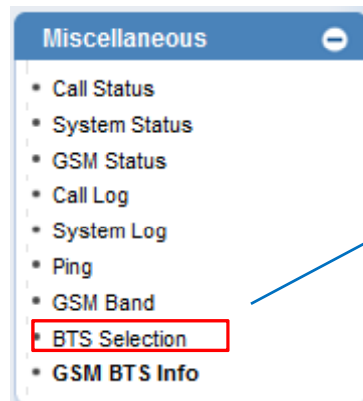


GSM Band Selection : Setting for bandwidth of GSM Networks

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

Advanced Service

4. BTS(Base Terminal Station) -1



BTS Control : Setting for selection method of BTS cell

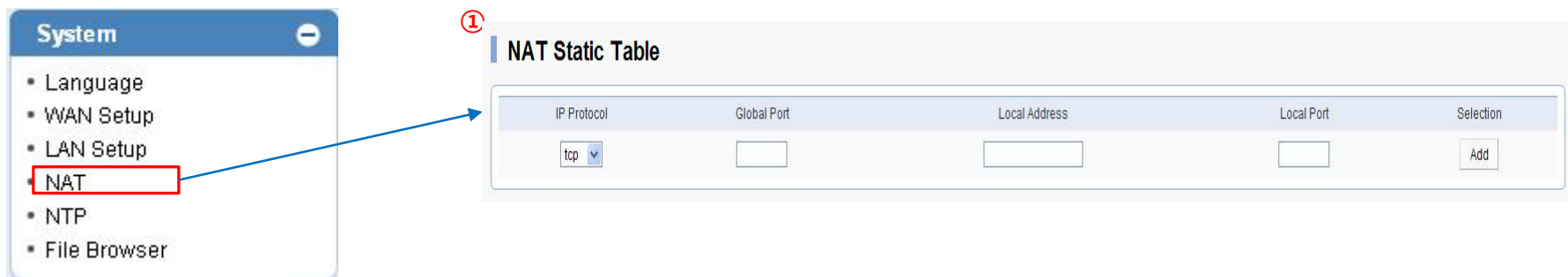
- Auto (Default) : The most highest power cell will be selected
- BCCH : Choose specific cell by entering BCCH value manually
- RSSI : Choose cell which has specific RSSI level
Cell will be selected in accordance with set interval

Advanced Service(etc)

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

Advanced Service (etc)

1. NAT Static Table



① NAT Static Table

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

* Example

- Global port : 5060

- Local Address : 10.1.1.1

- Local Port : 5070

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

Advanced Service (etc)

2. NTP

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

* Please click the apply button after set up

Advanced Service (etc)

3. File Browser

①

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



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**

<input checked="" type="checkbox"/> Enable Telnet	Server Port	<input type="text" value="23"/> (default 23, 1-65535)
<input checked="" type="checkbox"/> Enable HTTP	Server Port	<input type="text" value="80"/> (default 80, 1-65535)
<input checked="" type="checkbox"/> Enable FTP	Control Port	<input type="text" value="21"/> (default 21, 1-65535)
	Data Port	<input type="text" value="20"/> (default 20, 1-65535)
Timer	Inter Digit Time	<input type="text" value="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
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

HTTP Filter

②

Network Addr	Network Mask	Control
172.17.100.100	255.255.0.0	<input type="button" value="Delete"/>
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Telnet Filter

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

Click

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

The screenshot shows a web interface for configuring security settings. On the left, a sidebar menu under 'Advanced' includes 'Gain & CID', 'GSM PINs', 'Fax', 'Service', 'Filtering', 'Security' (highlighted with a red box and a blue arrow), 'WEB Callback', and 'GSM Callback'. A red circle with the number '1' is next to the 'Security' menu item. The main content area is titled 'Security' and contains four 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, an 'Apply' button with a green checkmark is highlighted with a red box, and a green starburst with the word 'Click' is overlaid on it.

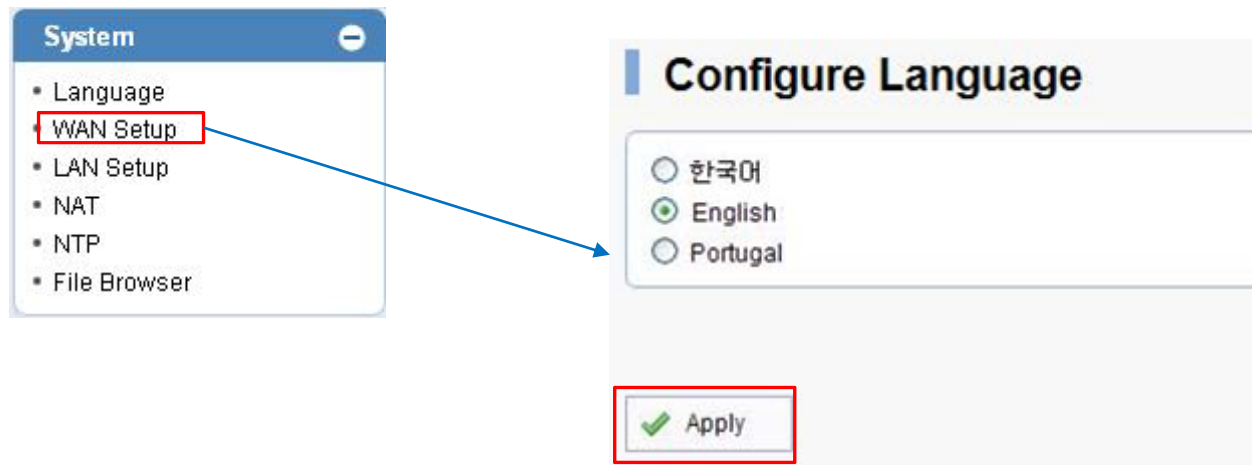
① Security : Set security to block unauthorized call

- 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.

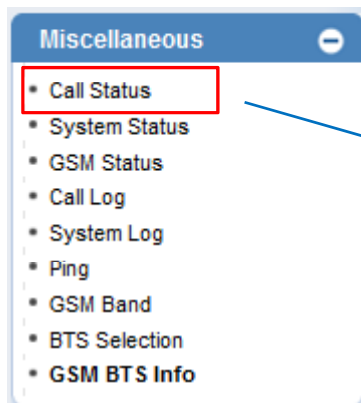


Monitoring

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"/>

Unlock Block

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

② Call Status

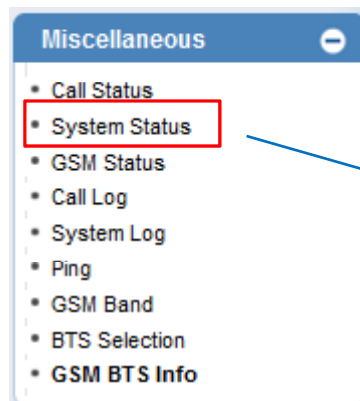
Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
0/0	GSM Out	N.A.	500	2000	N.A.	N.A.
0/0	GSM In	N.A.	1000	2000	N.A.	N.A.

Call Status : The status of 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	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 = *164
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.
tshipping (sip ping timer) = 45 sec.

SIP UA Session Timer value
MinSSE = 1800 sec.
Session-Expires = 1800 sec.

SIP DNS SVF 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

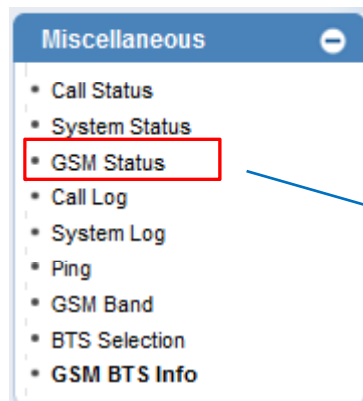
SIP NAT Info
PNIC = Disabled
Required = NULL

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



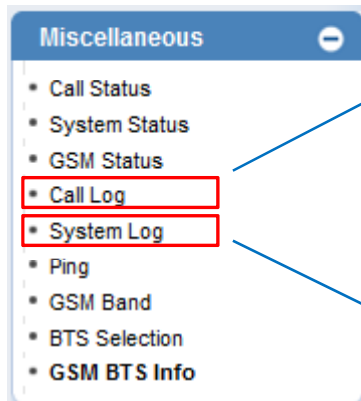
The 'GSM Status' page displays 'GSM Port Status & Information'. It features a table with columns for 'Port', 'My Phone Number', 'Register Status', 'Signal Strength', 'Voice Quota(secs)', and 'SMS Quota(E.A.)'. The table has two rows: P0:0 and P0:1.

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

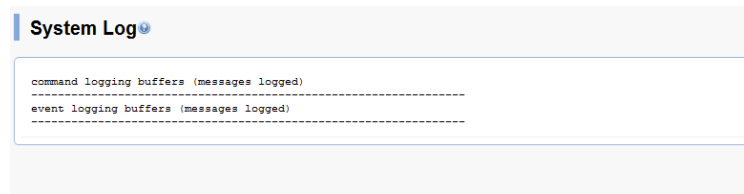
4. Call Log / System Log



Call Log

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

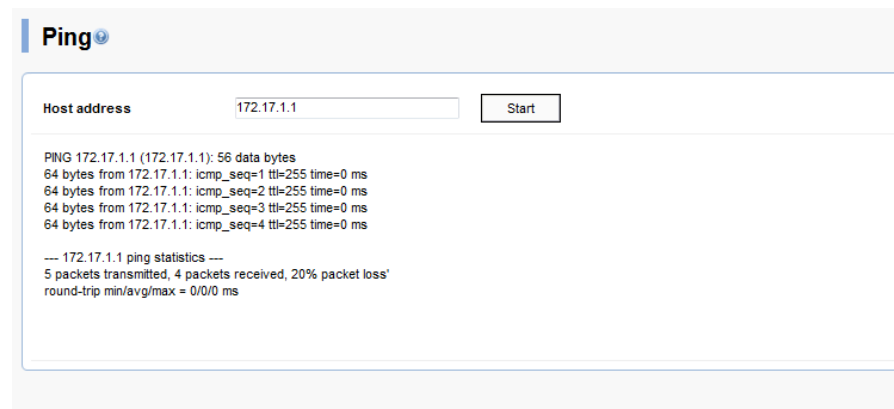
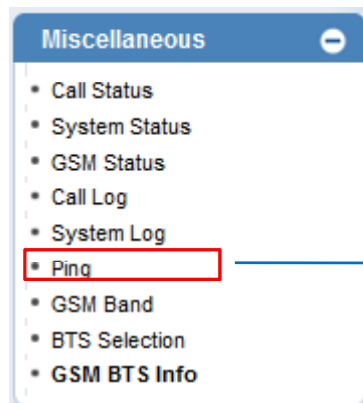
Call Log : Monitoring all of call history
※ Call history will be clear with rebooting



System Log : Monitoring 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