



# VoIP Gateway Smart WEB

High Performance VoIP Gateway Solution

## WEB Operation Guide



**AddPac**

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1. WEB Connection
2. Network Setup
3. Port Setup
4. VoIP Setup
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# WEB Connection

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

# WEB Connection

## 1. Web Connection via Console Port



Serial port

Baud rate 9600  
No parity  
1 stop bit  
No flow control



Login ID/Password : root/router

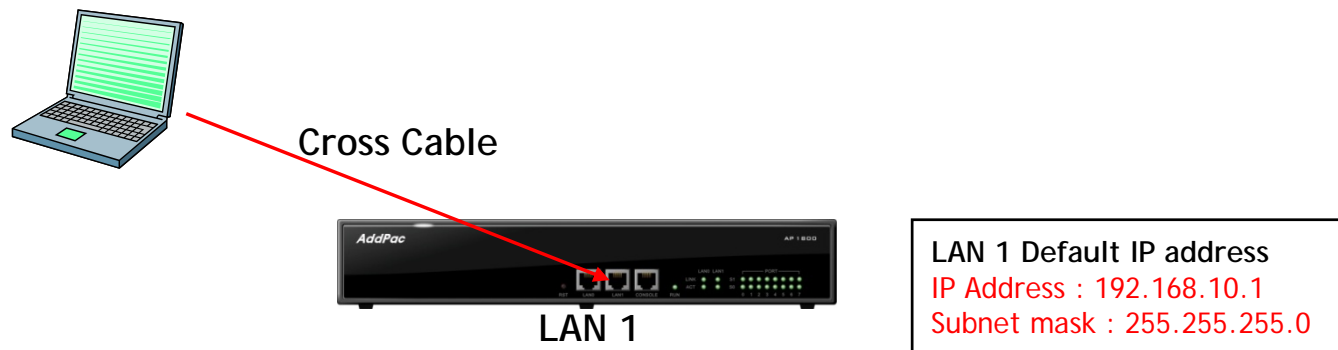
1. Connect to VoIP Gateway via console port to enter IP address in order to set Interface 0/0
2. Input IP address of interface 0/0 using below command
3. Connect to the IP address via web after saving

```
Router> enable
Router# configure terminal
Router(config)# interface FastEthernet 0/0
Router(config-if)# ip address <IP Address> <Subnet Mask>
Router(config-if)# exit
Router(config)# ip route 0.0.0.0 0.0.0.0 <Default Gateway>
Router(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



1. It is the way to connect to VoIP Gateway 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 VoIP Gateway
  - Connect PC to VoIP Gateway 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

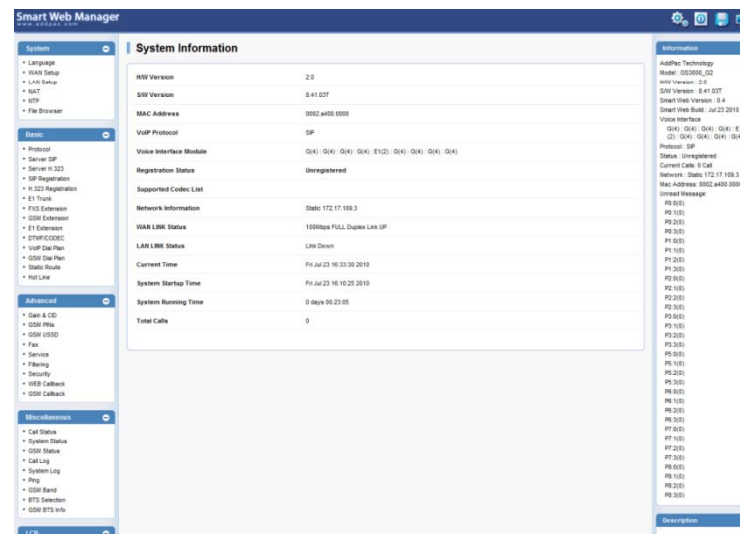
The screen of Web connection



- 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 Screen -1

**System**

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

**WAN & Tunneling Setup**

**WAN Setup**

Hostname Router

IP Address 172.17.109.3 A.B.C.D  
Network Mask 255.255.0.0 A.B.C.D  
Default Router A.B.C.D  
DNS Server Primary DNS Server  
Secondary DNS Server

Static IP

User name  
Password

PPPoE(ADSL)

(No Authentication)  
 PAP (PPP Authentication Protocol)  
 CHAP(Challenge Handshake Authentication Protocol)

DHCP

VLAN ID 0

Auto

WAN Link Control

Speed  100  10  
Duplex  full  half

NAT & Bridge  None  NAT  Bridge

MAC(Hardware) Address

Apply

**Click**

① WAN Setup (LAN0)

- Hostname : Enter the device name of Gateway
- Static IP
- PPPoE(ADSL)
- DHCP
- VLAN
- NAT / Bridge

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

# Network Setup

## 1. WAN Setup Screen -2

System

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

**Tunneling Setup**

Mode

- None(Disable Tunneling, default)
- PPTP(Point-to-Point Tunneling Protocol)

User name

Password

Authentication

- (No Authentication)
- PAP(PPP Authentication Protocol)
- CHAP(Challenge Handshake Authentication Protocol)
- None (Default)

Phone Number

- Hostname (Use hostname as phone number)
- User Define

Source

Destination  A.B.C.D (Tunnel End Point Address)

Service

- Voice and Data Use Tunnel Interface (default)
- Voice Use Tunnel Interface, Data Use Ethernet Interface
- Data Use Tunnel Interface, Voice Use Ethernet Interface

Apply

Click

② WAN Setup (LAN0)  
-PPTP setup  
\* Please make sure to press the apply button for saving



# Network Setup

## 2. LAN Setup Screen

**System**

- Language
- WAN Setup
- **LAN Setup**
- NAT
- NTP
- System Time
- 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)

Default Lease time

CHECKED  (in seconds, default 86400, 1 day)

Address Range  -

DNS Server  A.B.C.D

Apply

Click

① LAN Setup (LAN1)

- None : Use default IP address 192.168.10.1.
- IP share : Use it by sharing WAN IP.
- Static : Use it by setting up IP address.

# Port Setup

1. FXS Port Setup
2. FXO Port Setup
3. E1 / T1 Setup
4. Hot Line

# Port Setup

## 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..

# Port Setup > FXS Port

## 1. FXS Port - FXS Extension

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- **FXS Extension**
- FXO Extension
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

**FXS Extension**

① **Port Information**

Port	P0	P1	P2	P3	P4	P5	P6	P7
SLOT 0	FXS	FXS	FXS	FXS	FXO	FXO	FXO	FXO
SLOT 1	FXS	FXS	FXS	FXS	FXS	FXS	FXS	FXS

② **FXS Extension Configuration**

Pots Num	Port	Numbers	Preference	HuntStop	Control
P0:0			0	<input type="checkbox"/>	Delete
					Apply

\* FXS Extension - Assigned Pots Tag Number : 1512 - 2023

① The each port information of 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

# Port Setup > FXO Port

## 2. FXO Port - FXO Extension

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- FXS Extension
- **FXO Extension**
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

**FXO Extension**

① **Port Information**

Port	P0	P1	P2	P3	P4	P5	P6	P7
SLOT 0	FXS	FXS	FXS	FXS	FXO	FXO	FXO	FXO
SLOT 1	FXS	FXS	FXS	FXS	FXS	FXS	FXS	FXS

② **FXO Extension Configuration**

Pots Num	Port	Numbers	Preference	HuntStop	Control
	P0:4		9	<input type="checkbox"/>	Delete

\* FXO Extension - Assigned Pots Tag Number : 2024 - 2535

① The each port information of VoIP Gateway

② FXO Extension : Set the number of FXO port

- Index : Enter number in order. Please make sure not to be duplicated
- Port : Select FXO port to be set
- Numbers : Enter FXO 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

# Port Setup > FXO Port

## 2. FXO Port - FXO Dial Plan

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- FXS Extension
- **FXO Extension**
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

**FXO Dial Plan / Prefix**

① **Plan Table**

Rule Num	Digits to Insert	Digits to delete	Digit Pattern	Control
	<input type="text"/>	<input type="text"/>	<input type="text"/>	Delete
				Add

② **FXO Table**

Pots Num	FXS	FXO	Number	PlanIndex	Preference	Control
P0:0	P0:4	<input type="text"/>	N.A.	9	Delete	
					Apply	

\* FXO Dial Plan - Assigned Translation-Rule Tag Number Number : 100 - 199  
\* FXO Dial Plan - Assigned Pots Tag Number : 3560 - 4071

- ① Plan Table : use when number transformation is necessary.
  - 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 FXO Table : use when setup backup dispatch with FXO port.
  - FXS : specify FXS port for backup dispatch.
  - FXO : specify FXO port for backup dispatch.
  - Number : specify digit pattern for backup dispatch.
  - Plan Index : specify relevant number when applying plan table.
  - Preference : specify the order of priority for backup dispatch.

# Port Setup > E1/T1

## 3. E1/T1 Port - E1/T1 Trunk -1

The screenshot displays the configuration interface for an E1/T1 Trunk. On the left, a 'Basic' menu is visible with 'E1/T1 Trunk' selected and highlighted in red. An arrow points from this menu item to the main configuration area. The main area is titled 'E1/T1 Trunk' and contains the following settings:

- T1 Port (Slot 0/0):** Slot 0/0, group Num (0-9) [ ], Time slot Range(1-24, 16-24, 1,2,3) [ ], Control [Add]
- Clock-Source:** Master
- Framing:** SF
- Line Code:** AMI
- Signaling-type:** ISDN-PRI
- Protocol-emulate:** Network (selected), User
- Virtual-Connect:** Enable, Disable (selected)
- Immediate-disc:** Enable, Disable (selected)
- Dial-Tone-Generate:** Enable, Disable (selected)
- ISDN-PRI:**
  - Compad-Type:** a-law (selected), u-law
  - N303:** 2 (1-10sec)
  - Q931 Timer:**
    - T303:** 4 (1-400sec)
    - T310:** 10 (5-400sec)
- R2-MFC:** Get-Calling-number, Enable, Disable (selected)
- Busyout:** Action, PortDown, None (selected)

E1 / T1 Trunk : Be sure to setup first before using E1 or T1 (default is E1)  
If E1/T1 Trunk is not setup, E1/T1 Extension and Registration cannot be setup.

# Port Setup > E1/T1

## 3. E1/T1 Port - E1/T1 Trunk -2 (E1)

E1 Port (Slot 0/0)		Slot 0/0
Slot/Port	group Num (0~9)	Time slot Range(1-31,16-31,1,2,3)
0/0	<input type="text"/>	<input type="text"/>
		<input type="button" value="Add"/>
Clock-Source	Slave	
Framing	No-CRC4	
Line Code	HDB3	
Signaling-type	ISDN-PRI	
ISDN-PRI	Protocol-emulate	<input type="radio"/> Network <input checked="" type="radio"/> User
	Virtual-Connect	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
	Immediate-disc	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
	Dial-Tone-Generate	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
	Comband-Type	<input checked="" type="radio"/> a-law <input type="radio"/> u-law
	N303	<input type="text" value="2"/> (1~10sec)
	Q931 Timer	<input type="text" value="4"/> (1~400sec)
	T310	<input type="text" value="10"/> (5~400sec)
R2-MFC	Get-Calling-number	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Busyout	Action	<input type="radio"/> PortDown <input checked="" type="radio"/> None

Group Setup for E1 Channel.  
Be sure to setup before using E1.

Clock Setup : Master / Slave  
Be sure to setup in pair with other device

E1 Framing : CRC4 / No CRC4

E1 Line Code : HDB3

E1 Signaling : ISDN-PRI, R2-MFC, R2-DTMF  
(Default : ISDN-PRI)

E1 PRI Setup

- Protocol-emulate : Be sure to setup in pair with other device (Network / User)
- Virtual-connect
- Immediate-disc
- Dial-Tone-Generate
- Comband-Type
- Q931 Timer

E1 R2 Setup  
Get-Calling-Number : Calling number enable / disable

Setup for port down when the E1 port error occurs



# Port Setup > E1/T1

## 3. E1/T1 Port - E1/T1 Trunk -3 (T1)

T1 Port (Slot 0/0)		Slot 0/0	Slot 0/1
Slot/Port	group Num (0~9)	Time slot Range(1-24,16-24,1,2,3)	
0/0	<input type="text"/>	<input type="text"/>	
<input type="button" value="Add"/>			
Clock-Source	Master		
Framing	SF		
Line Code	AMI		
Signaling-type	ISDN-PRI		
ISDN-PRI	Protocol-emulate	<input checked="" type="radio"/> Network	<input type="radio"/> User
	Virtual-Connect	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
	Immediate-disc	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
	Dial-Tone-Generate	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
	Compand-Type	<input checked="" type="radio"/> a-law	<input type="radio"/> u-law
	Q931 Timer	N303	<input type="text" value="2"/> (1~10sec)
		T303	<input type="text" value="4"/> (1~400sec)
	T310	<input type="text" value="10"/> (5~400sec)	
R2-MFC	Get-Calling-number	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
Busyout	Action	<input type="radio"/> PortDown	<input checked="" type="radio"/> None

Group Setup for T1 Channel.  
Be sure to setup before using T1.

Clock Setup : Master / Slave  
Be sure to setup in pair with other device

T1 Framing : SF / ESF

T1 Line Code : AMI / B8ZS

T1 Signaling : ISDN-PRI, R2-MFC, R2-DTMF  
(Default : ISDN-PRI)

T1 PRI Setup  
- Protocol-emulate : Be sure to setup in pair with other device (Network / User)  
- Virtual-connect  
- Immediate-disc  
- Dial-Tone-Generate  
- Compand-Type  
- Q931 Timer

T1 R2 Setup  
Get-Calling-Number : Calling number enable / disable

Setup for port down when the T1 port error occurs

# Port Setup > E1/T1

## 3. E1/T1 Port - E1/T1 Extension

**E1/T1 Extension Configuration**

Pots Num	Port	Group	Numbers	HuntStop	Forward Digits(0~99)	Control
2536	0/0	0	5..	X		<input type="checkbox"/>

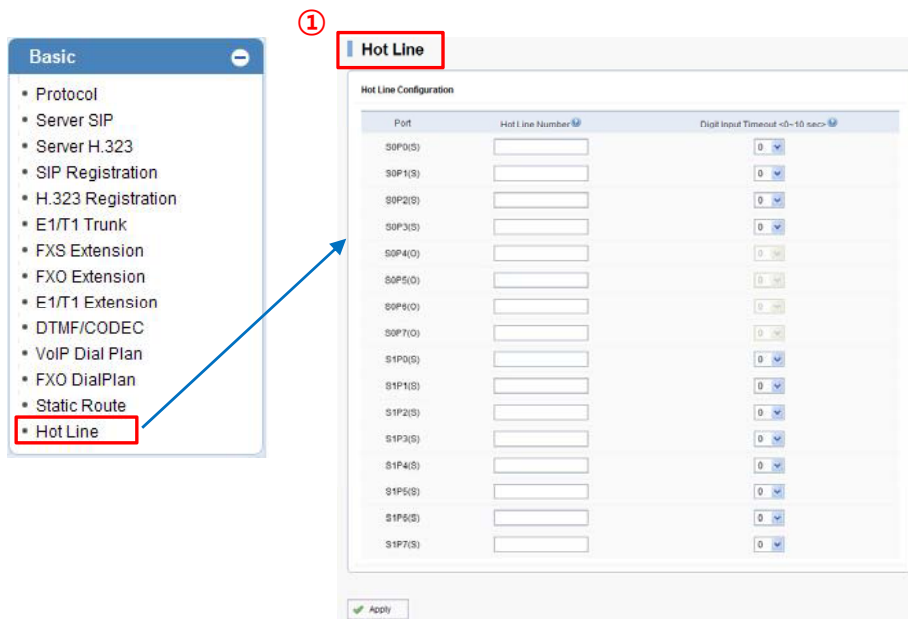
Port: P0:0 | Group: 0 | Numbers: | HuntStop:  | Forward Digits: from  last  |

Buttons: Delete, Apply

- ① E1/T1 Extension : Configuration for incoming call with E1/T1
- Port : It means E1/T1 slot and port
  - Group : It means group number which was setup in E1/T1 Trunk.
  - Numbers : Setup the incoming number of E1.
  - 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
  - Forward Digits : Specify the digits for E1/T1 PABX.
    - from : Send the number to E1/T1 PABX from the number you specified in front.
    - last : Send the number to E1/T1 PABX from the number you specified at the back.

# Port Setup > Hot Line

## 4. Hot Line

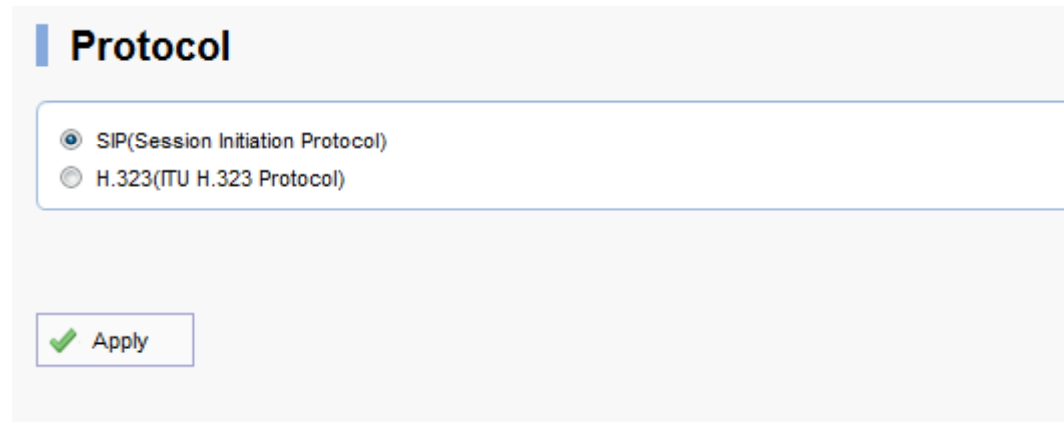


- ① Hot Line Configuration : Connect incoming and outgoing call directly
- Port : It means FXS and FXO port
  - Hot Line Number : Forward call to entered number  
It connects to the number of FXO port in case of receiving call (Direct Incoming call)
  - Digit Input Timeout : Time to make call to the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

# VoIP Setup

1. Server SIP
2. SIP Registration
3. Server H.323
4. H.323 Registration
5. DTMF/CODEC
6. VoIP Dial Plan
7. Static Route

# VoIP Setup > Protocol Selection



Choose VoIP protocol that you wish to use. (Default : SIP)

# 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  1800 (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
- E1/T1 Trunk
- FXS Extension
- FXO Extension
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

### E1/T1 Registration

**E1/T1 Registration Configuration**

Pots Num	Port	Group	E.164 Number	User Name	Password	HuntStop	Reg
1000	S0P0	0	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>

**E1/T1 Phone Number Option**

Pots Num	Port	Group	PBX Line	Internal Num	Virtual Calling Num
1000	S0P0	0	<input type="radio"/>	<input type="radio"/>	<input type="text"/>

\* E1 Registration - Assigned Pots Tag Number : 1000 - 1201

### ① SIP Registration Configuration

- Group : It means group number which was setup in E1 Trunk. You may choose it when you create several Group.

-E.164 Number : Enter SIP authentication number

-User Name : Enter authentication ID

-Password : Enter authentication Password .

-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.

### ② E1/T1 Phone Number Option

-PBX Line : use number that is used in E1 PABX.

-Internal Num : use when using virtual calling number.

-Virtual Calling Num : enter virtual calling number.

# VoIP Setup > H.323

## 3. Server H.323 -1

**H.323 (ITU H.323 Protocol)**

Use H.323 Server  Yes  No

Primary Gatekeeper  Server address and Port (default 1719)

Secondary Gatekeeper  Server address and Port (default 1719)

H.323 ID  (H.323 Identifier string)

H.323 Signaling Port  (default 1720, between 1 to 65535)

H.323 Call start mode  Fast  Slow

H.323 Tunnel mode  Enable  Disable

Apply

### ① Server H.323

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

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

-Secondary Gatekeeper Server : Enter IP address of Secondary Gatekeeper server.

The secondary server is activated when Primary Gatekeeper Server is not available

-H.323 ID : Please input Authentication ID to register Gatekeeper

-Default setting is recommended for other field



# VoIP Setup > H.323

## 3. H.323 Registration -1

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration**
- E1/T1 Trunk
- FXS Extension
- FXO Extension
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

### E1/T1 Registration

① **E1/T1 Registration Configuration**

Pots Num	Port	Group	Number	HuntStop	Reg
1000	S0P0	0	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>

② **E1/T1 Phone Number Option**

Pots Num	Port	Group	PBX Line	Internal Num	Virtual Calling Num
1000	S0P0	0	<input type="radio"/>	<input type="radio"/>	<input type="text"/>

\* E1/T1 Registration - Assigned Pots Tag Number : 1000 - 1201

### 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 Gatekeeper.

### ② E1/T1 Phone Number Option

-PBX Line : use the number for E1 PABX.

-Internal Num : use it for virtual calling number.

-Virtual Calling Num : enter virtual calling number.

# VoIP Setup > DTMF/Codec

## 4. DTMF/CODEC

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- FXS Extension
- FXO Extension
- E1/T1 Extension
- **DTMF/CODEC**
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

**DTMF/CODEC**

Voice CODEC

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

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

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

- ① Video Codec : Select voice codec to be used.
- ② DTMF Relay mode : Select DTMF relay mode.  
(Default : Out-of-band Signal)

\* Please press the apply button to save

# VoIP Setup > VoIP Dial plan

## 6. VoIP Dial Plan -1

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- FXS Extension
- FXO Extension
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan**
- FXO DialPlan
- Static Route
- Hot Line

**VoIP Dial Plan / Prefix**

① **Plan Table**

Rule Num	Digits to Insert	Digits to delete	Digit Pattern	Control
	<input type="text"/>	<input type="text"/>	<input type="text"/>	Delete
				Add

② **Prefix Table**

Voip Num	Prefix	Called Num Plan	Calling Num Plan	Control
	<input type="text"/>	N.A. ▾	N.A. ▾	Delete
				Apply

\* VoIP Dial Plan - Assigned Translation-Rule Tag Number : 0 - 99  
\* VoIP Dial Plan - Assigned VoIP Tag Number : 10000 - 10099

- 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 > VoIP Dial Plan

## 6. VoIP Dial Plan -2 (Ex)

**Basic**

- Protocol
- Server SIP
- Server H.323
- SIP Registration
- H.323 Registration
- E1/T1 Trunk
- FXS Extension
- FXO Extension
- E1/T1 Extension
- DTMF/CODEC
- VoIP Dial Plan
- FXO DialPlan
- Static Route
- Hot Line

**VoIP Dial Plan / Prefix**

**Plan Table**

Rule Num	Digits to Insert	Digits to delete	Digit Pattern	Control
	<input type="text"/>	<input type="text"/>	<input type="text"/>	Delete
				Add

**Prefix Table**

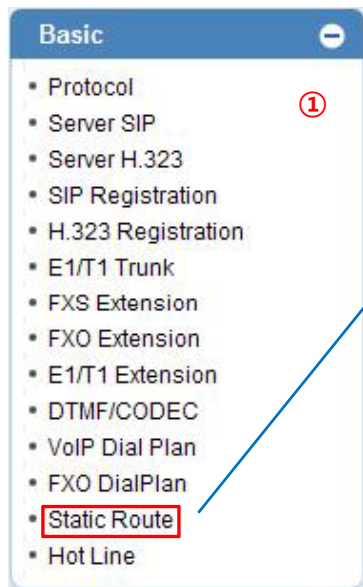
Voip Num	Prefix	Called Num Plan	Calling Num Plan	Control
	<input type="text"/>	N.A.	N.A.	Delete
				Apply

\* VoIP Dial Plan - Assigned Translation-Rule Tag Number : 0 - 99  
\* VoIP Dial Plan - Assigned VoIP Tag Number : 10000 - 10099

It must be the same with Plan Index Number

# VoIP Setup > Static Route

## 7. Static Route -1



A screenshot of the 'Static Route' configuration page. The page title is 'Static Route'. Below the title is a section titled 'Set Remote Site Call(5-digit number is set to begin \*2->\*2...)' with a table of input fields. The table has the following columns: Pots Num, Remote Site IP, Signaling Port, Prefix, Digits to Insert, Digits to Delete, Name of Remote Site, Answer Addr, and Control. The 'Pots Num' field contains an asterisk (\*). Below the table are two red asterisked notes: '\* Static Route - Assigned Voip Tag Number : 10100 - 10199' and '\* Static Route - Assigned Translation-Rule Tag Number : 10100 - 10199'. An 'Apply' button is located at the bottom right of the table.

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'

# Additional Service

1. NAT Static Table
2. NTP
3. System Time
4. File Browser
5. Gain & CID
6. FAX
7. Service
8. Filtering
9. Security

# Additional Service

## 1. NAT Static Table

System

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

**NAT Static Table**

IP Protocol	Global Port	Local Address	Local Port	Selection
tcp	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

① NAT Static Table  
Use when port forwarding is necessary for special IP address.

# Additional Service

## 2. NTP

System

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

① NTP

Enable  Disable

Primary Server  (Domain Name or IP Address)

Secondary Server  (Domain Name or IP Address)

Interval  (1-72 hours)

Timezone Name  Offset  :  (-23~23 hours) : (0~60 minute)

Apply

Click

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

\* Please click the apply button after set up



# Additional Service

## 3. System Time

①

**System Time**

Current Time Sat Sep 11 01:47:59 2010

Set System Time  Year  Month  Day  Hour  Min  Sec

Apply

Click

① You may setup the device local time.

# Additional Service

## 4. File Browser

① File Browser

②

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 view firmware and configuration file in Gateway.
- ② Firmware Upload
  - Use when uploading Gateway firmware.
  - Process upload by finding PC related firmware.
  - Uploaded firmware will be applied after rebooting.

# Additional Service

## 5. Gain & CID

Advanced

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

① Gain & CID

Gain

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

Apply

Click

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

- ① Gain & CID : Adjustment output voice level of each port(FXS / FXO)  
(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

# Additional Service

## 6. FAX

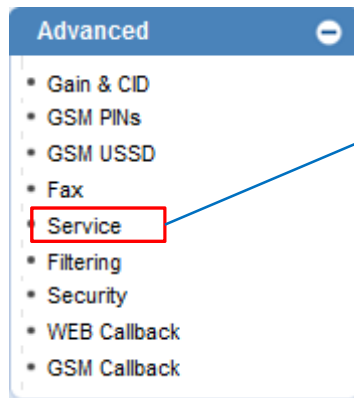


- ① Fax : Setting the property of FAX mode
- Fax Mode :
    - T.38 : FAX signal is being sent by T.38 packet with new session opening. In case of using T.38, FAX Rate is needed to be set
    - Bypass : FAX signal is being sent by RTP. FAX Rate setting is not required
  - Fax Rate : Setting FAX transmit rate. Default is 9600bps and the range is from 2400bps to 14400bps

\* Please click the apply button after set up

# Additional Service

## 7. Service -1



A screenshot of the 'Service' configuration page. The page is titled 'Service' and contains several sections of settings:

- Application Services:**
  - Enable Telnet: Server Port 23 (default 23, 1-65535)
  - Enable HTTP: Server Port 80 (default 80, 1-65535)
  - Enable FTP: Control Port 21 (default 21, 1-65535), Data Port 20 (default 20, 1-65535)
  - Enable Syslog: Primary Server 172.17.110.222 (Port 514, default 514), Secondary Server 172.17.110.215 (Port 5151, default 514), Log Level 7-debug, Log Command enable
- Timer:** Inter Digit Time 5 sec (default 3, 1-600)
- Call Service:** Transfer:  Hook-Flash  Not-assigned; Hold:  Hook-Flash  Not-assigned
- SIP Transfer:** Mode:  blind  Attended
- Skype Server:** Use:  Enable  Disable
- Hunt:** Algorithm: (0) longest - preference - random
- Change Password:** ID: root, New Password: [masked], Confirm Password: [masked]

At the bottom of the page, there is an 'Apply' button with a green checkmark icon.

Please refer to next page about description of function.

# Additional Service

## 7. Service -2

Applicaton Services ⓘ	<input checked="" type="checkbox"/> Enable Telnet	Server Port	<input type="text" value="23"/> (default 23, 1-65535)	Application Service Use : Telnet / HTTP / FTP / Syslog		
	<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)			
<input checked="" type="checkbox"/> Enable Syslog	Primary Server	<input type="text" value="172.17.110.222"/>	Port	<input type="text" value="514"/> (default 514)	PDD Time (default : 3 sec)	
	Secondary Server	<input type="text" value="172.17.110.215"/>	Port	<input type="text" value="5151"/> (default 514)		
	Log Level	<input type="text" value="7-debug"/>				
	Log Command	<input type="text" value="enable"/>				
Timer ⓘ	Inter Digit Time	<input type="text" value="5"/> sec	(default 3, 1-600)			Call Hold / Transfer Use : Enable / Disable
Call Service ⓘ	Transfer	<input checked="" type="radio"/> Hook-Flash <input type="radio"/> Not-assigned				SIP Call transfer mode : Blind / Attended
	Hold	<input checked="" type="radio"/> Hook-Flash <input type="radio"/> Not-assigned				
SIP Transfer ⓘ	Mode	<input type="radio"/> blind <input checked="" type="radio"/> Attended				Skype SIP Sever Use : enable / disable
Skype Server	Use	<input checked="" type="radio"/> Enable <input type="radio"/> Disable				Call Routing Algorism
Hunt	Algorism	<input type="text" value="(0) longest - preference - random"/>				Password change regard of WEB / Telnet access
Change Password	ID : root	New Password	<input type="password" value="....."/>			
		Confirm Password	<input type="password" value="....."/>			

# Additional Service

## 8. Filtering

The screenshot displays the 'Filtering' configuration page in the AddPac interface. On the left, the 'Advanced' menu is expanded, with 'Filtering' selected. The main content area is titled 'Filter' and contains three filter sections: FTP Filter, HTTP Filter, and Telnet Filter. Each section has a table with columns for 'Network Addr', 'Network Mask', and 'Control'. The 'Add' button in the FTP Filter section is highlighted with a red box and a green starburst labeled 'Click'. A red circle with the number '1' is positioned above the 'Filter' title.

- 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

# Additional Service

## 9. Security

The screenshot displays the 'Security' configuration page. On the left, the 'Advanced' menu is open, with 'Security' selected. The main configuration area includes the following options:

- IP Filtering:**  Enable  Disable
- Allowed IP Address List:** A text input field with  and  buttons.
- WarDialing Filtering:**  Enable  Disable
- Allow Digit Length(IP to PSTN):** Min  Max
- H323 Shutdown:**  Enable  Disable
- SIP Shutdown:**  Enable  Disable

A green starburst with the text 'Click' points to the 'Apply' button at the bottom left of the configuration panel.

① Security : Set security to block unauthorized call

- IP Filtering : The only call made from the device with IP address listed on Gateway is available to make call
- Allowed IP address list : Allow VoIP call for added IP address.
- War Dialing Filtering : The only receiving call listed on dial plan is available to make call
- Allow digit Length(IP to PSTN) : The only receiving call within range of set number is available to make call
- SIP Shutdown : Set using SIP Signaling. It must be enabled with SIP communication

\* Please click the apply button after set up



# Monitoring

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

# Monitoring

## 1. Call Status

**Miscellaneous** -

- Port & Call Status
- System Status
- Call Log
- System Log
- Test Call
- Ping

E1 Analog

Port Status (E1)

Port#	Channel Group																															Control		
SLOT 4/0	Channel	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	Unblock	Block
	Status	I	I	I	I	I	I	I	I	I	I	I	I	I	I	I	B	I	I	I	I	I	I	I	I	C	I	I	I	I	I	I		
SLOT 4/1	Channel	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	Unblock	Block
	Status	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B		

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

Port Status

SLOT	Port	0(FXS)	1(FXS)	2(FXS)	3(FXS)	4(FXO)	5(FXO)	6(FXO)	7(FXO)
SLOT 0	Status	C	I	I	I	I	I	I	I
	Select	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

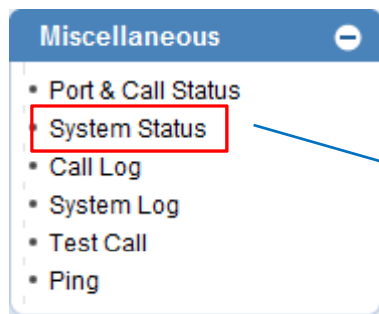
Connection State : (Connected) (Disconnected | Blocked)  
 Call State : (Idle) (Ring | Dial) (Called) (Calling) (Blocked)

Call State

Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
0/0	GSM In	07/23 17:28:27	821021140183	500	g711u	N.A

# Monitoring

## 2. System Status



**System Status**

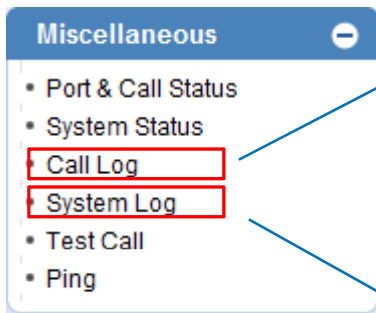
Voice Port

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	GSM	Idle	0	0	none		-1	-1	-1
0/ 3	GSM	Idle	0	0	none		-1	-1	-1
1/ 0	GSM	Idle	0	0	none		-1	-1	-1
1/ 1	GSM	Idle	0	0	none		-1	-1	-1
1/ 2	GSM	Idle	0	0	none		-1	-1	-1
1/ 3	GSM	Idle	0	0	none		-1	-1	-1
2/ 0	GSM	Idle	0	0	none		-1	-1	-1
2/ 1	GSM	Idle	0	0	none		-1	-1	-1
2/ 2	GSM	Idle	0	0	none		-1	-1	-1
2/ 3	GSM	Idle	0	0	none		-1	-1	-1
3/ 0	GSM	Idle	0	0	none		-1	-1	-1
3/ 1	GSM	Idle	0	0	none		-1	-1	-1
3/ 2	GSM	Idle	0	0	none		-1	-1	-1
3/ 3	GSM	Idle	0	0	none		-1	-1	-1
4/ 0: 0	E1	Link Up	0	0	none		-1	-1	-1
4/ 1: 0	E1	Link Down	0	0	none		-1	-1	-1
5/ 0	GSM	Idle	0	0	none		-1	-1	-1
5/ 1	GSM	Idle	0	0	none		-1	-1	-1
5/ 2	GSM	Idle	0	0	none		-1	-1	-1
5/ 3	GSM	Idle	0	0	none		-1	-1	-1
6/ 0	GSM	Idle	0	0	none		-1	-1	-1
6/ 1	GSM	Idle	0	0	none		-1	-1	-1
6/ 2	GSM	Idle	0	0	none		-1	-1	-1
6/ 3	GSM	Idle	0	0	none		-1	-1	-1
7/ 0	GSM	Idle	0	0	none		-1	-1	-1
7/ 1	GSM	Idle	0	0	none		-1	-1	-1
7/ 2	GSM	Idle	0	0	none		-1	-1	-1
7/ 3	GSM	Idle	0	0	none		-1	-1	-1
8/ 0	GSM	Idle	0	0	none		-1	-1	-1
8/ 1	GSM	Idle	0	0	none		-1	-1	-1
8/ 2	GSM	Idle	0	0	none		-1	-1	-1
8/ 3	GSM	Idle	0	0	none		-1	-1	-1

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

# Monitoring

## 3. Call Log / System Log



### Call Log

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

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

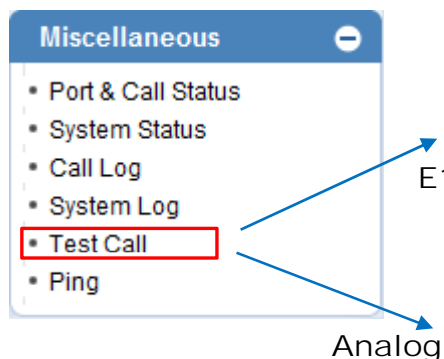
### System Log

command logging buffers (messages logged)
event logging buffers (messages logged)

System Log : Monitoring Gateway 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

## 4. Test Call



### Test Call

Port	B-Ch (1-31)	Test Phone Number	Digits	Command			
S0P0	0	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End

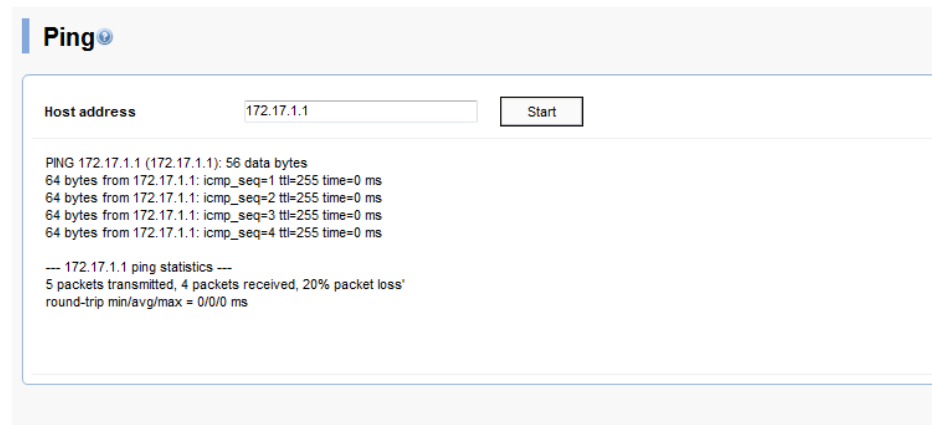
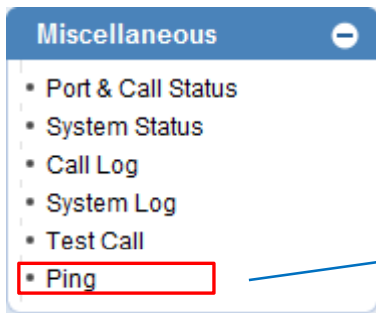
### Test Call

Port	B-Ch (1-24)	Test Phone Number	Digits	Command			
S0P0	-	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End
S0P1	-	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End
S0P2	-	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End
S0P3	-	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End
S1P0	-	<input type="text"/>	<input type="text"/>	Start	Voice	Digit	End

- Test Call : remote diagnose the proper operation of VoIP call.
- : Enter the test phone number in relevant port then click start button.
  - Test Phone Number : enter receiving number for test processing.
  - Digits : enter the desire digit once the call is connected.
  - Command
    - Start : start test call.
    - Voice : sample voice play under connection status.
    - Digit : play the entered digit underconnection status.
    - End : end test call.

# Monitoring

## 5. Ping



Ping : Network status can be checked by pinging



# Thank you