ArtDio

Voice Internet Phone Gateway

IPS 1000 Series

User Manual

Version:3.1 Update:2004/5/7

ARTDio Company Inc.

Contents

1. Safety Instructions

WARNING

1. Do not attempt to service the product yourself. Any servicing of this product should be referred to qualified service personal.
2. To avoid electric shock, do not put your finger, pin, wire, or any other metal objects into vents and gaps.
3. To avoid accidental fire or electric shock, do not twist power cord or place it under heavy objects.
4. The product should be connected to a power supply of the type described in the operating instructions or as marked on the product.
5. To avoid hazard to children, dispose of the product's plastic packaging carefully.
6. The phone line should always be connected to the LINE connector. It should not be connected to the PHONE connector as it may cause damage to the product.
7. Please read all the instructions before using this product.

2. Preface

The IPS 1000 unit is a personal SIP VoIP gateway developed using the latest in VoIP technology. It is also very simple to install and easy to operate.

2.1. What is SIP

Session Initiation Protocol (SIP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543& RFC 3621) that can be used to establish, maintain, and terminate calls between two or more end points. Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Session management* provides the ability to control the attributes of an end-to-end call.

SIP provides the following capabilities:
Determine the location of the target end point—Supports address resolution, name mapping, and call redirection.
Determine the media capabilities of the target end point—By using Session Description Protocol (SDP),

SIP determines the highest level of common services between the end points. Conferences are established using only the media capabilities that can be supported by all end points.

Determine the availability of the target end point—If a call cannot be completed because the target end point is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. It then returns a message indicating why the target end point is unavailable.

Establish a session between the originating and target end point—If the call can be completed, SIP establishes a session between the end points. SIP also supports mid-call changes, such as the addition of another end point to the conference or the changing of a media characteristic or Codec.

Handle the transfer and termination of calls—SIP supports the transfer of calls from one end point to another. During a call transfer, SIP simply establishes a session between the transferee and a new end point (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

2.1.1. Components of SIP

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function in one of the following roles:

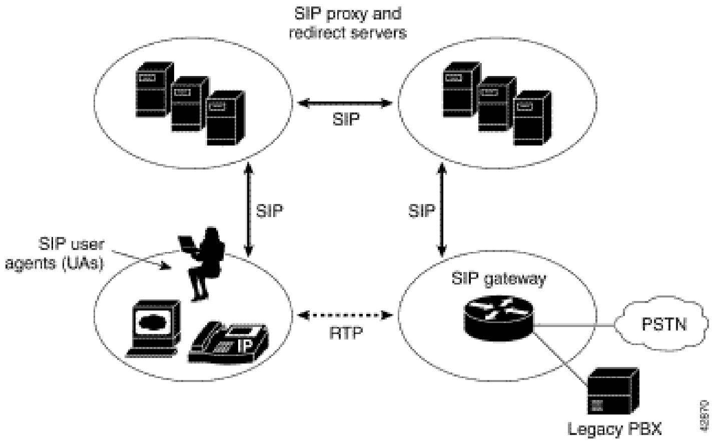
User agent client (UAC)—A client application that initiates the SIP request.

User agent server (UAS)—A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

Typically, a SIP end point is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request.

From an architecture standpoint, the physical components of a SIP network can be grouped into two categories: clients and servers.

Architecture



SIP Clients

SIP clients include the following:

Phones—Can act as either a UAS or UAC. Soft phones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.

Gateways—Provide call control. Gateways provide much functionality. The most common one is a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway also translates between audio and video Codec and performs call setup and clearing on both the LAN side and the switched-circuit network side.

SIP Servers

SIP servers include the following:

Proxy server—The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.

Redirect server—Provides the client with information about the next hop or hops that a message should take, then the client contacts the next hop server or UAS directly.

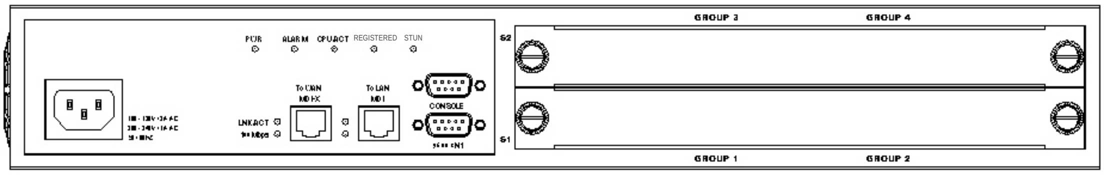
Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

3. Package Contents

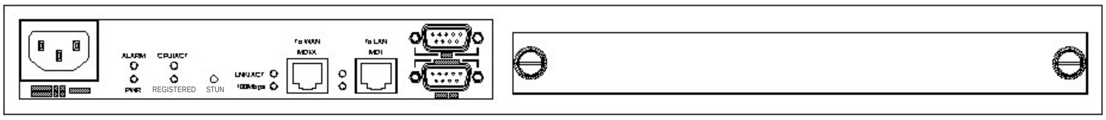
The IPS 1000 Gateway	x	1	
Power Core	X	1	
Accessories for fixing support	X	1	(For 1008/1016)
System CD-ROM	X	1	
5 IDC Connector	X	4	(For 1008/1016)
Rubber footer			
RJ-45 Ethernet Cable	X	1	
RJ-11 Telephone Cable	X	1	

4. Panel Descriptions

4.1. Front Panel



IPS 1016 Front Panel (16 ports)



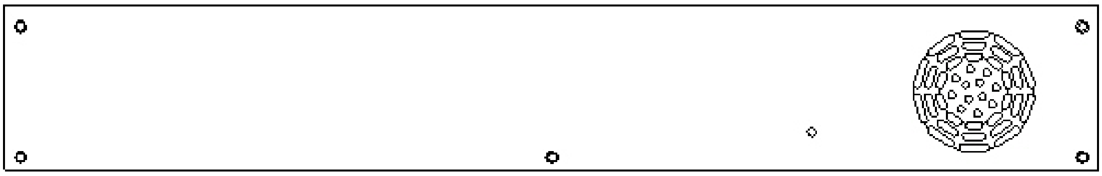
IPS 1008 Front Panel (8 ports)

IPS 1004 Front Panel (4 ports)

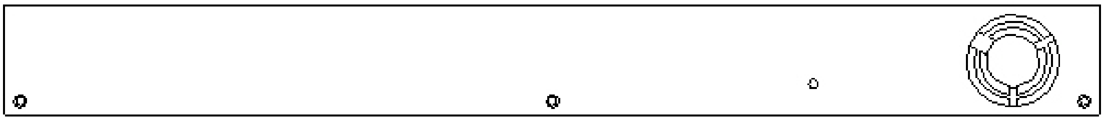
IPS 1002 Front Panel (2 ports)

4.2. Rear Panel

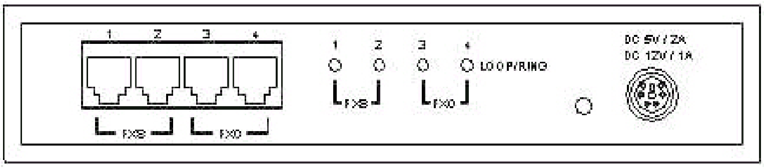
There is a button on the rear panel of gateway for special maintenance. Please don’t touch this button under normal operation.



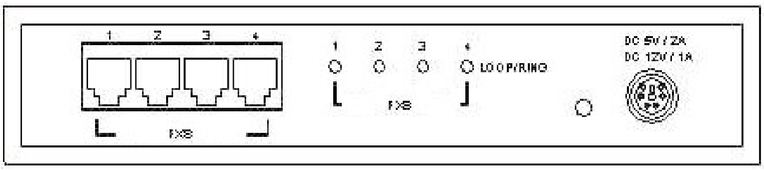
IPS 1016 Rear Panel (16 ports)



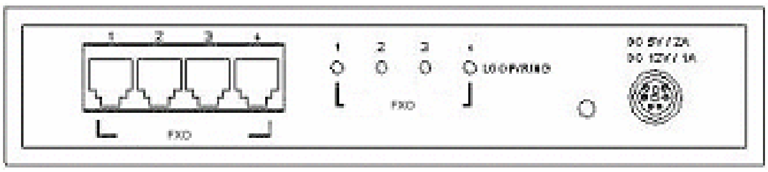
IPS 1008 Rear Panel (8 ports)



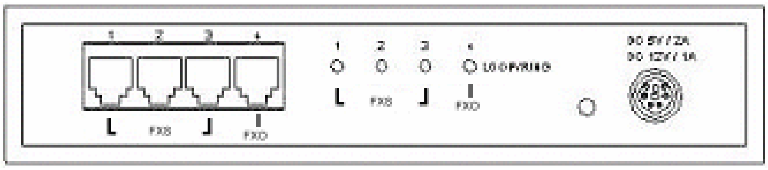
IPS 1202 Rear Panel (4 ports)



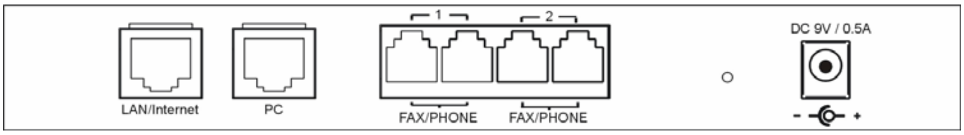
IPS 1004 Rear Panel (4 ports)



IPS 1400 Rear Panel (4 ports)



IPS 1103 Rear Panel (4 ports)



IPS 1101 Rear Panel (2 ports)

5. LED Indicators

<table>
<tr><td>LED</td><td>Label</td><td colspan="2">Description</td></tr>
<tr><td rowspan="5">10/100 Ethernet</td><td rowspan="3">LNK/ACT</td><td>On</td><td>Link up</td></tr>
<tr><td>Off</td><td>Link down</td></tr>
<tr><td>Flash</td><td>Sending/Receiving data packets</td></tr>
<tr><td rowspan="2">100Mbps</td><td>On (LNK is on)</td><td>100Mbps</td></tr>
<tr><td>Off (LNK is on)</td><td>10Mbps</td></tr>
<tr><td rowspan="6">LOOP/RING</td><td rowspan="3">FXS</td><td>On</td><td>Off hook</td></tr>
<tr><td>Off</td><td>On hook</td></tr>
<tr><td>Flash</td><td>Ringing out</td></tr>
<tr><td rowspan="3">FXO</td><td>On</td><td>Line is active</td></tr>
<tr><td>Off</td><td>Line is inactive</td></tr>
<tr><td>Flash</td><td>Ringing in</td></tr>
<tr><td rowspan="5">Device</td><td>Alarm</td><td colspan="2">The red light “On” indicates that system has some problem; please contact your vender.</td></tr>
<tr><td>Power</td><td colspan="2">“On” indicates that the power supply is working normally.</td></tr>
<tr><td>CPU/ACT</td><td colspan="2">“On” indicates that the CPU is working normally.</td></tr>
<tr><td>Registered</td><td colspan="2">“On” indicates that all SIP entities are registered successful.
“Off” indicates that all SIP entities are registered fail.
“Flash” indicates that one of these SIP entities is registered fail.</td></tr>
<tr><td>STUN</td><td colspan="2">“On” indicates communicate with STUN Server once.
“Off” indicates never communicate with STUN Server.</td></tr>
</table>

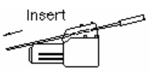
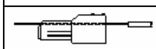
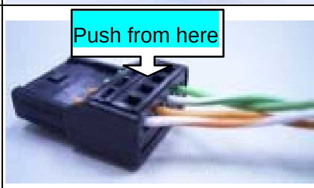
6. Connectors

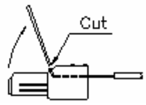
Ports	Label	Description
Voice Ports	FXS	Connects to a telephone set or fax machine
	FXO	Connects to the phone line
Ethernet Ports	LAN/Internet	RJ-45 connector MDI-X connects to a Modem
	PC	RJ-45 connector MDI connects to a PC
Console Port (Only 1004/1008/1016)	Console	RJ-45 connector/RS-232 Interface

7. IDC Connectors (Only for IPS-1000 series 8/16 ports)

IDC connector is used for the voice interface (FXS and FXO) on the frame model. IDC connector can easily connect PBX line and telephone wire together to the gateway. No special tools are required; please follow the instruction to install:

(Remarks: For IDC connector, it's better to use No. 24 wire, e.g. CAT 5)

Get the material ready	 
 Insert the insulated wires directly into the block for wire insertion	
 Push the block down until it is locked to flush the conductor with the probe	

Cut off the conductor outside the edge to avoid from causing the circuit shortage	

8. Information required before Installation

You need to prepare the following information before installing the gateway.

8.1. IP Address

The gateway requires an IP address for operation. Before installation you need to know how to obtain an IP address from your local ISP. Static IP, DHCP or PPPoE can be used. The following table helps you to decide what information you need. If your ISP offers static IP, you may need to obtain an IP from MIS personnel in order to prevent an IP conflict. Otherwise DHCP (most cable broadband providers offer this) and PPPoE (most ADSL broadband providers offer this) will work fine.

IP Environment		Requiring information
Static IP	Public IP Address	IP Address Subnet Mask ***Default Gateway*** It is strongly suggested that you obtain an IP address from MIS personnel in order to prevent an IP conflict.
	Private IP Address	IP Address Subnet Mask Default Gateway It is strongly suggested that you obtain an IP address from MIS personnel in order to prevent IP conflicts. Your private IP requires an IP Sharing device and you must configure the IP Sharing device to treat the IPS unit and the IP that it is using as a virtual server.
Dynamic IP address (DHCP)		DHCP mode

PPPoE	Account Number Password Your ISP normally provides this information. If you don't have this information please contact your ISP.

8.2. SIP Information

Before configuring SIP, the IPS 1000 requires SIP information for operation. The following table helps you to decide what information you need.

Items	Description
1. SIP Proxy	If you want to make SIP calls through SIP proxy server, you will need to know the IP address or domain name of SIP proxy server. The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. If you don't know which SIP proxy for setting, contact your SIP service provider.
2. Public Address (SIP Account) Example: sip@artdioinc.com	The public address is like phone number, you can get the account from your SIP service provider.
3. Outbound Authentication	You will need the information when the SIP proxy server requires authentication. You can get this authentication information from SIP service provider when you apply for the service.

8.3. Prepare a password for Web Management

You will need to prepare a password for Web based Management. It can be a digit and/or letter combination ranging from 1 to 6 digits (E.g. 123). For security reason, password must be set to enter the Web Management page.

9. Installation and Configuration

After preparing the information you need as specified in section 5, follow the following steps to do the basic configuration. You can use either a telephone or a system console to perform basic configurations. It is simple to connect a telephone set to FXS port and configures the system. If you want to use system console to configure the system (Only 1004/1008/1016 support), you have to configure your VT100 terminal to match the settings of the IPS unit's console port. The console port's terminal connection is set to 9600 baud, 8 data bits, 1 stop bit and no parity. Turn on the IPS unit's power and wait for the terminal to display "Press Enter…" follow the directions to begin.

Here are several procedures to do:

1. Confirming the Region ID.
2. Configure IP address of gateway.
3. Enter into the WEB page.
4. Plan and configure the channels into SIP entity.
5. Configure SIP proxy and register information.
6. Configure SIP entity information.
7. Configure Outbound Authentication (If needs).
8. Configure STUN (If your gateway is behind NAT).
9. Check the SIP entity if is registered successful.
10. Configure Phone book (If needs)
11. Make a SIP call.

9.1. Confirming the Region ID

Skip this step if you are installing your IPS unit in the default region. The default Region ID is printed on the label located outside the box. If you are installing your IPS unit at any region other then the region ID specified on the label, you will then need to configure the IPS to the correct Region ID.

9.1.1. Phone Setting

1. Connect the power.
2. Connect the phone cable to the "Phone" socket on the rear panel as pictured above.
3. When the CPU/ACT LED is on, pick up the handset and listen for the dialing tone.
4. Dial "##0000" and listen for 3 short beep.

5. Dial "9507#"；Assuming you are modifying for China (The last 2 digits are the regional ID)
6. Dial "971#"；Sets the new regional ID.
7. Hang up the phone. The device will be updated with the new region setting after it restarts (restart time is about 10 seconds)

9.1.2. System console settings (Only for 4/8/16 ports)

SIP-RG>enable
SIP-RG #configure
Enter configuration commands, one per line. End with CNTL/Z
SIP-RG(config)#regional_id *07*
SIP-RG(config)#exit
SIP-RG#delete nvram
This command resets the system with factory defaults.
All system parameters will revert to their default factory settings. All static
and dynamic addresses will be removed.
Reset system with factory defaults, [Y]es or [N]o? Yes

Attention:
Before Changing the Region ID, the system has to be reset to the default value. Therefore this step should be done first.
The following instruction may keep the IP address unchanged after reset:
"delete nvram keep_ip"

9.2. IP Address Settings

We recommend using a traditional phone to configure the unit's parameters, as this is the easiest way. The following two sections contain the procedures used to configure the IPS unit according to how you obtain your IP address (Static IP; DHCP or PPPoE).

Attention:
Every time you set a parameter item and press the "#" key to complete it, a successful setting will be confirmed by three equal tones in succession. If your setting is unsuccessful you will be prompted with one long tone.

9.2.1. Static IP Mode

The following table shows an example.

IP Address	210.67.96.121
Subnet Mask	255.255.255.248
Default Gateway	210.67.96.120
Web Management Password	123

Using the information contained in the example above. The procedure is as follows:

1. Connect the IPS unit to a suitable Power source.
2. Connect a traditional phone set to the "FXS" connector located on the rear panel.
3. When the CPU/ACT light is on, pick up the phone to hear the dialing tone.
4. ##0000 ; you should hear three short tones.
5. 01<u>0</u># ; the digit "0" is used to enable "manual" IP mode.
6. 02<u>210*67*96*121</u># ; IP address.
7. 03<u>255*255*255*248</u># ; Subnet Mask.
8. 04<u>210*67*96*120</u># ; Default Gateway.
9. 15<u>123</u># ; "123" is the web management password.
10. 98<u>1</u># ; Warm-restarts.
11. Hang up the phone. The system should now restart.

You can also use console to configure IP address. But phone number can't be configured by console.(Only 1004/1008/1016)

SIP-RG>enable
SIP-RG#configure
Enter configuration commands, one per line. End with CNTL/Z
SIP-RG(config)#ip state user
SIP-RG(config)#ip address 210.67.96.121 255.255.255.248
System need to restart
SIP-RG(config)#ip default-gateway *210.67.96.120*
SIP-RG(config)#exit
SIP-RG#restart
This command resets the system. System will restart operation code agent.
Reset system, [Y]es or [N]o? Yes

9.2.2. DHCP Mode

1. Connect the IPS unit to a suitable Power source.
2. Connect a traditional phone set to the “FXS” connector located on the rear panel.
3. When the CPU/ACT light is on, pick up the phone to hear the dialing tone.
4. ##0000 ; you should hear three short tones.
5. 011# ; the digit “1” is used to enable “DHCP” IP mode.
6. 15123# ; “123” is the web management password.
7. 981# ; Warm-restarts.
8. Hang up the phone. The system should now restart.

You can also use console to configure IP address. But phone number can’t be configured by console.(Only for 4 ports gateway)

SIP-RG>enable
SIP-RG#configure
Enter configuration commands, one per line. End with CNTL/Z
SIP-RG(config)#ip state dhcp
SIP-RG(config)#exit
SIP-RG#restart
This command resets the system. System will restart operation code agent.
Reset system, [Y]es or [N]o? Yes

9.2.3. PPPoE Mode

If your network environment is using PPPoE, you need to prepare the information as specified in section 8. Information required before Installation.
The following table shows an example.

PPPoE Account	83721@hinet.net
PPPoE Password	123ab
Web management password	123

There are three ways to configure user name and password of PPPoE

1. Use phone set to configure:
You can configure the user name and password by using phone set. The command ‘09’ is used for username and ‘10’ is for password of PPPoE. Since the user name and password use characters and

digits are accepted by phoneset only, you need a mapping between characters and digits. You can find them at section 14.4 Mapping table of characters used in PPPoE.
Example user name：83721@hinet.net，Password：123ab，The procedure is below

1. Connect the phone to IPS
2. When CPU/ACT is light, pick up the phone and press
3. ##0000 ；You will hear 3 short tones.
4. 0938333732314068696*465742*46*46574# ；Set user name：83721@hinet.net
5. 103132336162# ；Set password is 123ab
6. 981# ；Save and restart.

2. Use Console to configure (Only for 4/8/16 ports Gateway)

SIP-RG>enable
SIP-RG#configure
Enter configuration commands, one per line. End with CNTL/Z
SIP-RG(config)#pppoe username 83721@hinet.net
SIP-RG(config)#pppoe password 123ab
SIP-RG(config)#exit
SIP-RG#restart
This command resets the system. System will restart operation code agent.
Reset system, [Y]es or [N]o? Yes

3. Use WEB Interface to configure:

You can configure the user name and password by using WEB interface. Follow the steps to finish configuration.

Step 1: Using a traditional phone set to configure the web management password and phone number
You will need to use a web browser to perform the PPPoE settings through the IPS unit's web based management interface. To enter the web based management interface you must have a previously configured password. Follow the next procedure to setup your password and phone number.

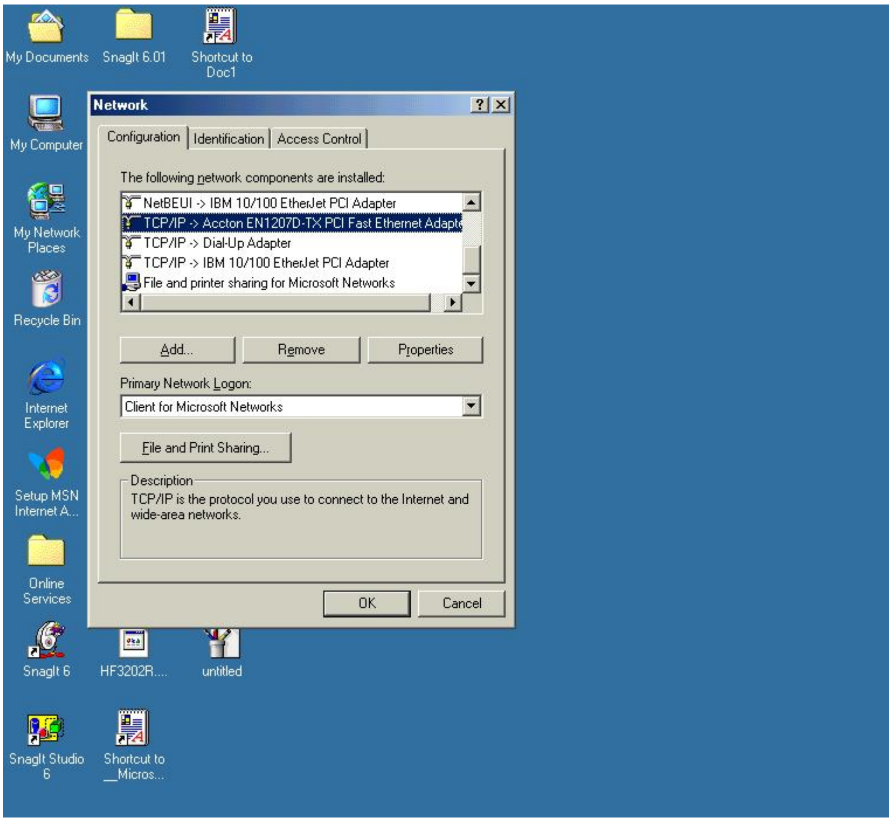
1. Connect the IPS unit to a suitable Power source.
2. Connect a traditional phone set to the "Phone" connector located on the rear panel.
3. When the CPU/ACT light is on, pick up the phone. You should hear the dialing tone.
4. ##0000 ; you should hear three short tones.
5. <u>15123</u> ; "123" is the web management password.
6. 010# ; "0" is to enable "manual" IP mode.

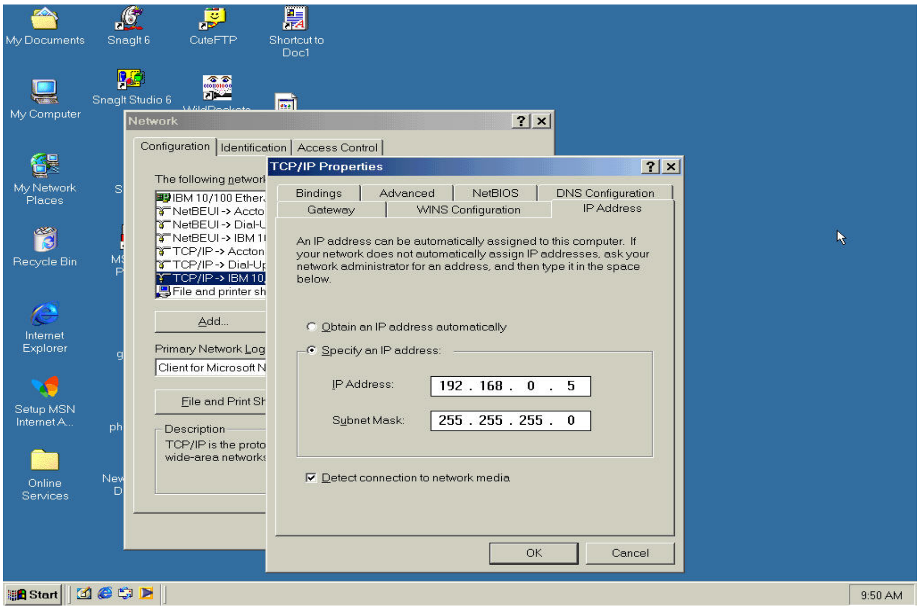
7. 02192*168*0*2# ; IP address.
8. 03255*255*255*0# ; Subnet Mask .
9. 981# ; Used to restart the IPS unit.
10. Hang up the phone to complete the configuration.

Step 2：Configure IP address of PC

Use the provided Ethernet cable to connect your PC to the port labeled “PC”, located on the rear panel of the IPS unit.

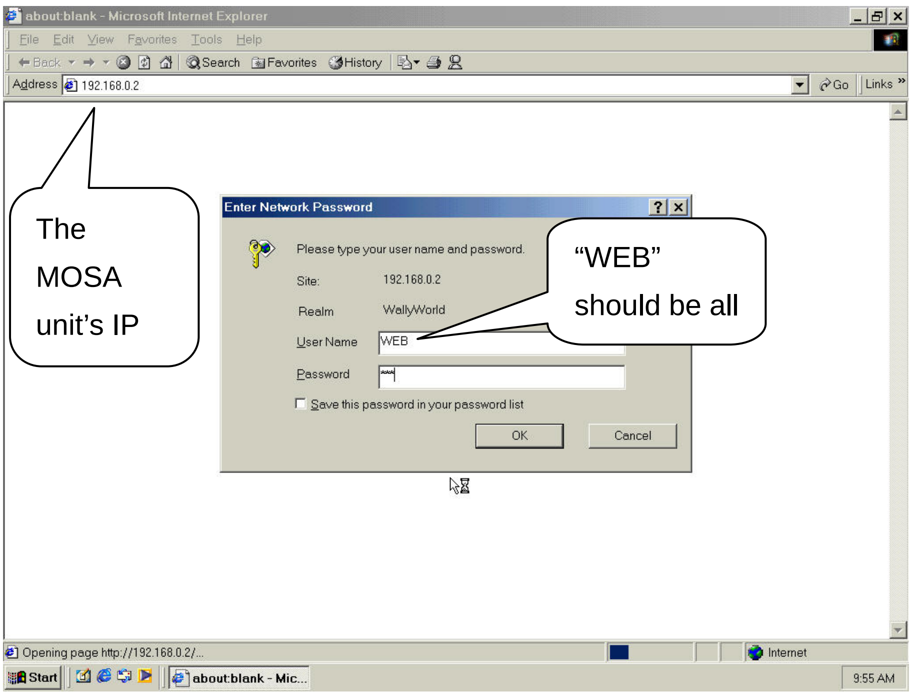
Because the IPS-1000 series unit’s default IP setting is 192.168.0.2, you must configure your PC to the same subnet. “192.168.0.x” for example. The following example uses 192.168.0.5 for the IP address and 255.255.255.0 for the subnet mask.





After you have completed the PC's IP address setting, you will be required to restart the PC in order for the new settings to take effect.

Step 3: Using the browser to configure the PPPoE Parameters of the gateway.
On the PC that is connected to the IPS unit, enter the IPS unit's IP address (Default 192.168.0.2) and press enter. The IPS will then prompt you with a dialogue box requesting that you enter a password. Use "WEB" (all capitals), for the User field and "123" for the password field that you have previously configured. Click the OK button; you should now have access to the IPS unit's web based management interface page.



Upon entering the web based configuration interface.

Click on “IP SETTING” at the top of the page and you will see the page as shown in the following image.

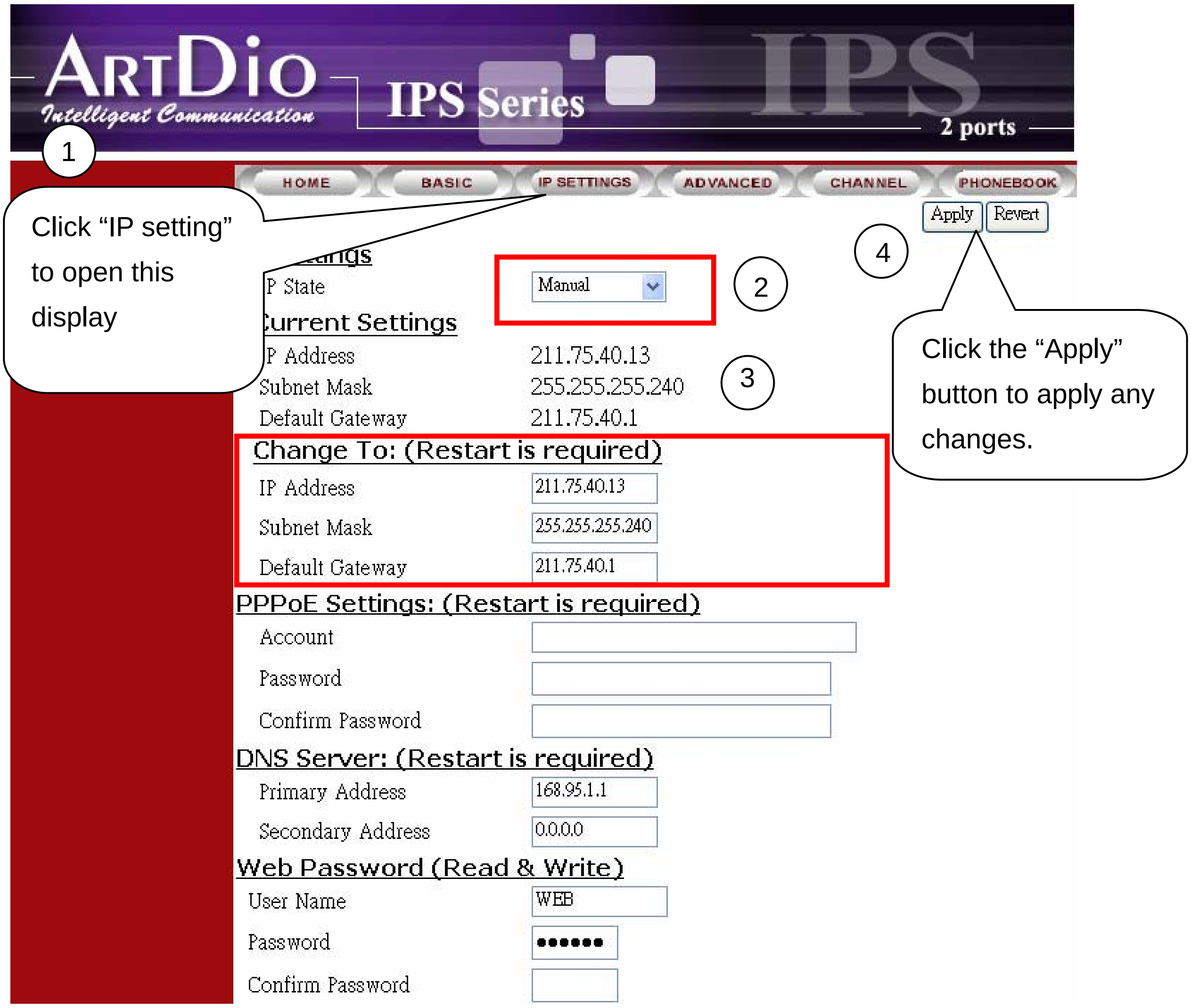
Select PPPoE from the “IP State” pull down menu.

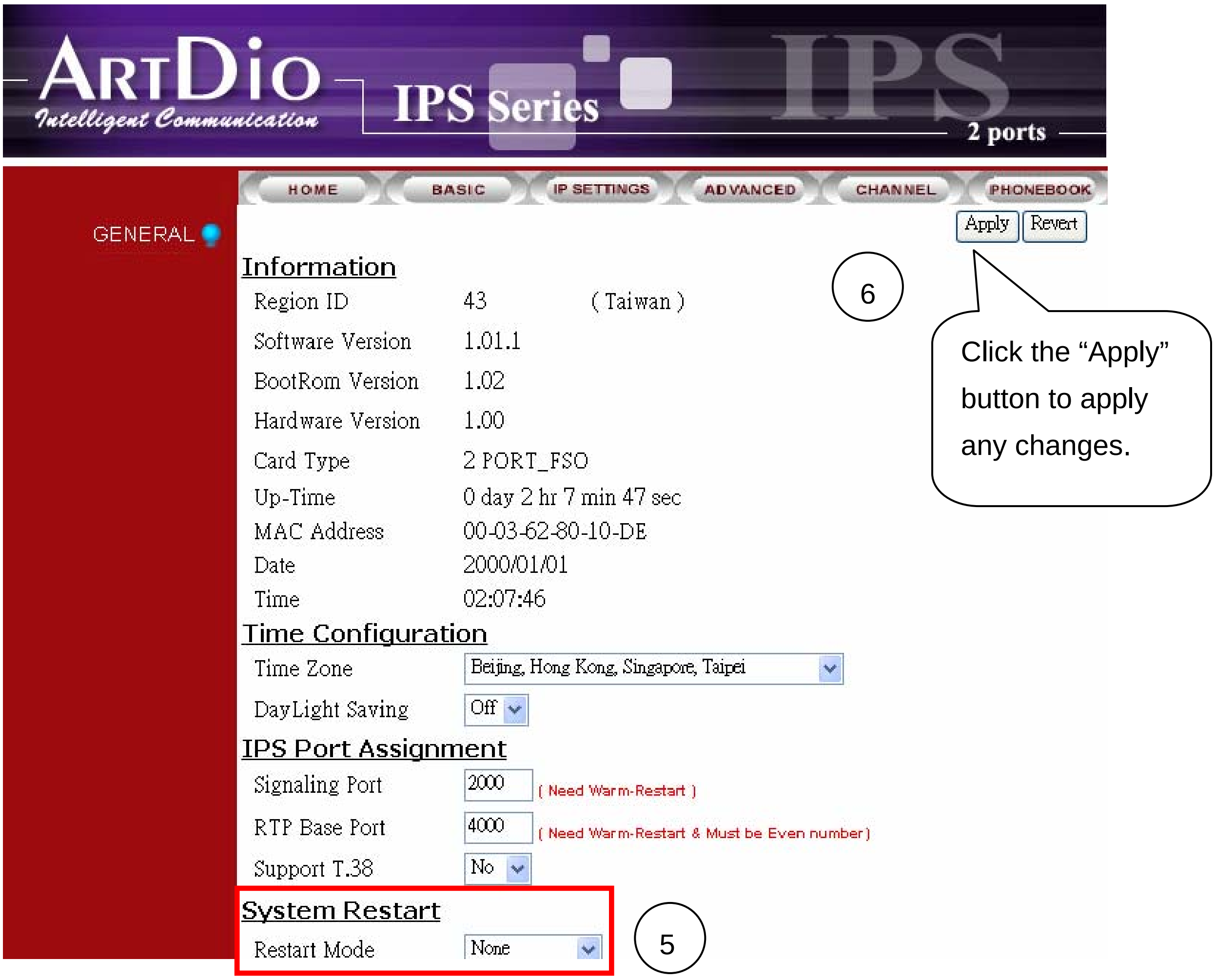
Fill in the “Account”, “Password”, and “Confirm Password” under the PPPoE Settings. You can obtain this information from your ISP.

Click on the Apply button.

Click the “BASIC” button at the top to go to the BASIC page and select “Warm Start” to restart the gateway. You can also perform a warm start using the phone by picking up the handset and dialing “##0000” then “981#”.

After restarting, the gateway will use PPPoE to obtain it’s IP address.

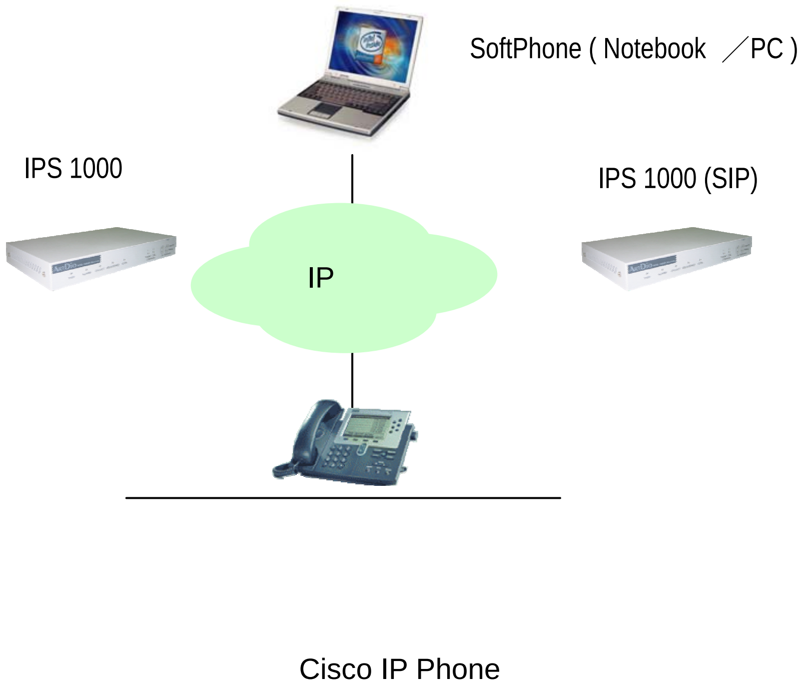




At this stage, your IPS should be able to use PPPoE to access the Internet. However, if you configured a wrong account number or password, your IPS cannot access the Internet. You are not able to use PC to access IPS by using the IP address of 192.168.0.2 because IPS has been set in PPPoE mode. You have to use phone set to configure IPS back to fix IP mode (##0000 010#) and use PC browser to configure correct parameters.

10. SIP Configuration

IPS 1000 not only can make regular PSTN calls, it also can communicate with IP Phones or Soft-Phones by using SIP protocol. Previous paragraphs have described the way to make regular IP calls. This section shows you what parameters you need to configure for SIP calls and how to make the SIP calls.



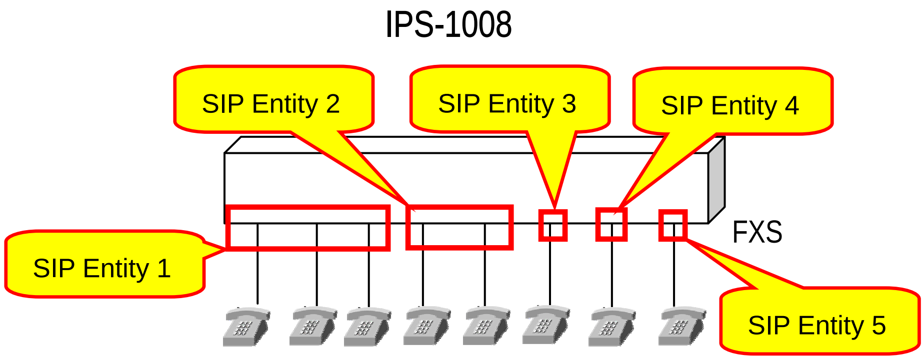
Notice: These configurations on WEB page, after select or input value in the field, please press “Apply” button to save and confirm the setting. Some parameters need “Warm-restart”, please process the restart action, thanks.

10.1. Channels and SIP entity

Select the channel and join a SIP entity.

Figure:

-



Configuration:

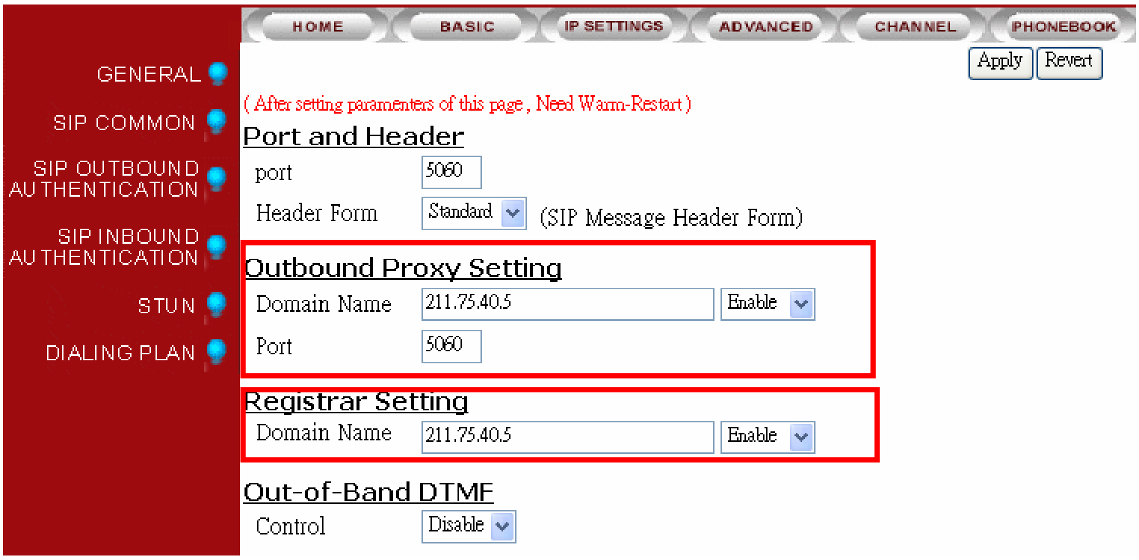
WEB page: CHANNEL\

Notice: Each channel must belong to a SIP entity.

10.2. SIP Proxy and Register Parameters

You need to configure IP address or Domain name of Registrar and Outbound Proxy server, please check the information is right.
SIP service provider will give you an IP address or Domain name of Registrar and Outbound proxy when you apply for the service.

Configuration
WEB Page: ADVANCED\SIP COMMOM



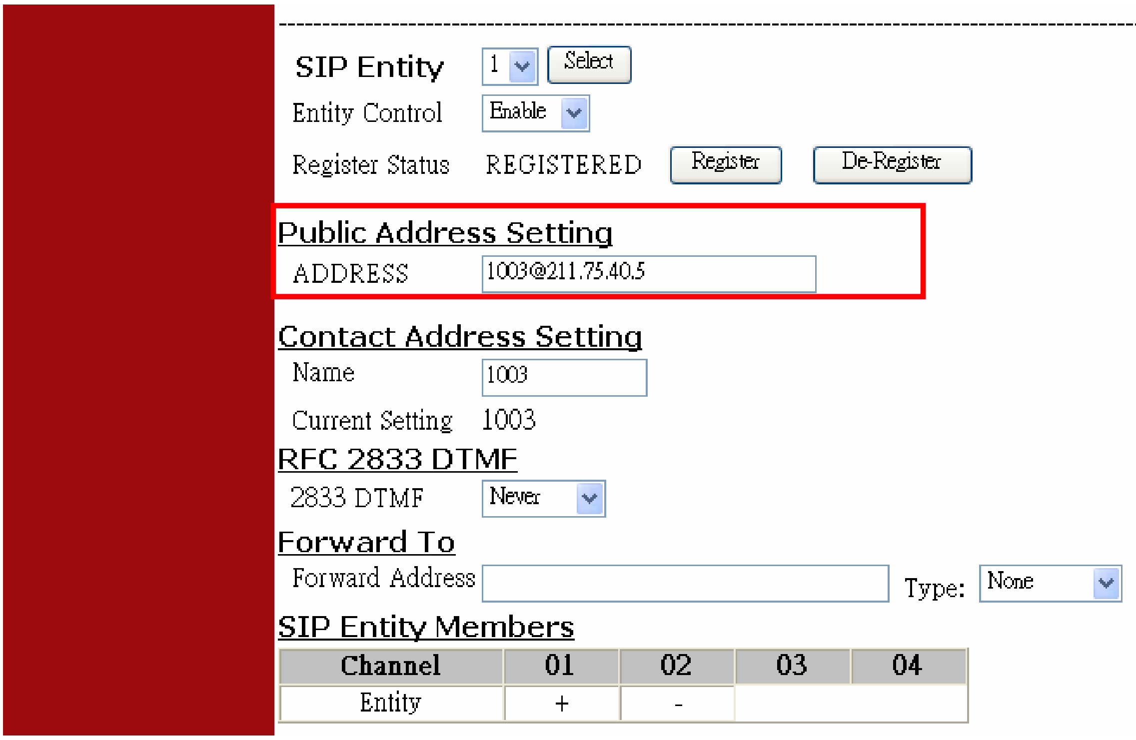
Notice: The Registrar Server is only for SIP entity registering. If the SIP entity register is fail, please check the item. SIP calls are all through Outbound Proxy Server, if the parameter is not configured, the SIP call will fail. So the two parameters must be configured.

10.3. SIP Entity

SIP service provider will assign one or more SIP accounts for you when you apply for the service. In standard, the SIP account is called ‘Public Address’, so you need to configure the account information in ‘Public Address’ item. The format is like an E-mail address such as mary@artdioinc.com.

Configuration

WEB Page: ADVANCED \ SIP COMMON



You can control the SIP entity on WEB page, just select 'Enable' or 'Disable'.

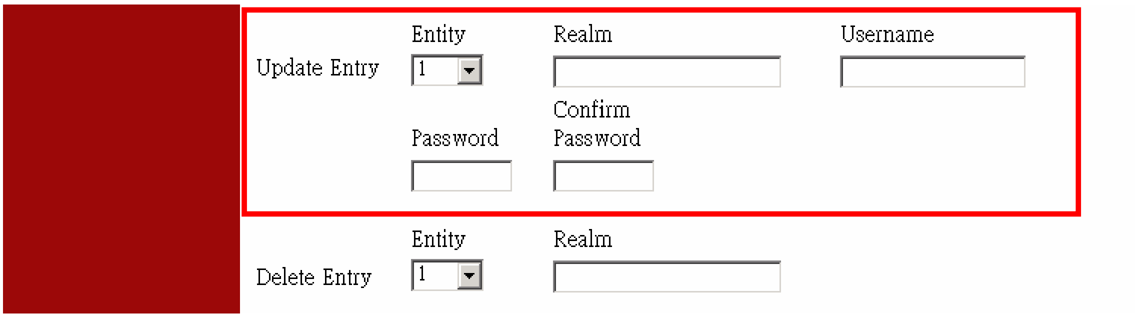
10.4. SIP Outbound Authentication

You need to configure outbound authentication for each SIP entity if SIP proxy server or other SIP phone request for authentication. Please check with SIP service provider if you need the setting.

Please select the entity then input information includes realm, username, and password.

Configuration

WEB Page: ADVANCED \ SIP OUTBOUND AUTHENTICATION

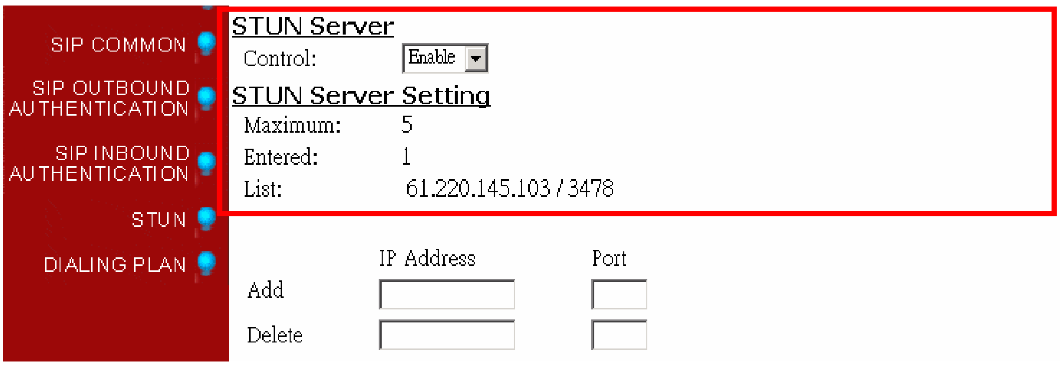


10.5. Configure STUN

The STUN (Simple Traversal UDP through NAT) server is an implementation of the STUN protocol that enables STUN functionality in SIP-based systems. The STUN server also includes a client API to enable STUN functionality in SIP endpoints.
STUN is an application-layer protocol that can determine the public IP and nature of a NAT device that sits between the STUN client and STUN server.

Notice: If your gateway is behind NAT (Use Private IP), must configure the parameter.
After configuring the parameters of STUN, please act Warm-Restart.

Configuration
WEB Page: ADVANCED\STUN



You can enable and disable the service on WEB page.
The STUN refresh time defines how long the device will send a binding request packet with discard flag on to STUN server. A binding packet with discard flag off will be sent each time when the number of binding request packet with discard flag on reach the Rebinding counts. The binding request packet is used to let the STUN server keep the most fresh client information.

10.6. Check SIP entity Status

You can use the WEB page to check the SIP entity is registered successful or unsuccessful.

WEB Page: ADVANCED\SIP COMMOM

If the status shows “REGISTERED” means successful, otherwise means fail; please notice that.

When you find the registration is fail, first check the “Registrar Setting” configuration is normal, or not “Enable”.
Then check the “Public Address” and “Outbound Authentication” configuration is in normal status.
If the configurations are all right, please check the situation with your SIP service provider.

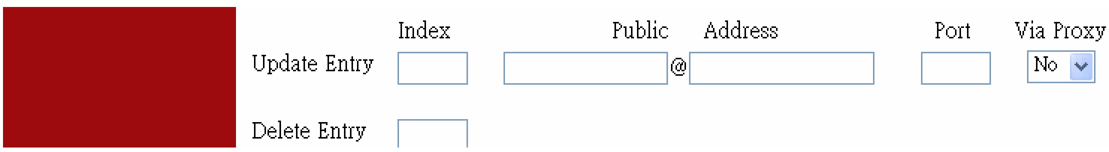
10.7. Phone Book

Since the SIP phone number is not easy for regular phone to dial, IPS 1000 provide a SIP phone book to let standard phone to make a SIP call easier. The phone book uses index number to map SIP account. For instance if the phone book is configure as below:

Index	Public Address	Port	Proxy
2231	mary@artdioinc.com	5060	Yes
331	John@artdioinc.com	5060	No

Notice: If your SIP account is number type like 234@artdioinc.com or 456@artdioinc.com, you don’t need to configure the items.

Configuration
WEB page: PHONEBOOK \



10.8. Make SIP Calls

After you have configured the SIP phone on the SIP phone book, you can easily make SIP calls.

You can select one way to make SIP call following these ways:

Standard Call: Only dial <numbers>+<#>.

1. Compare dialing plan, check the number if it is in setting. Example 050.
2. If the number is in setting, send the call to proxy. If communicate with proxy is fail, then the call will be sent to PSTN.
3. If the number is not in dialing plan, the call will be sent to PSTN.

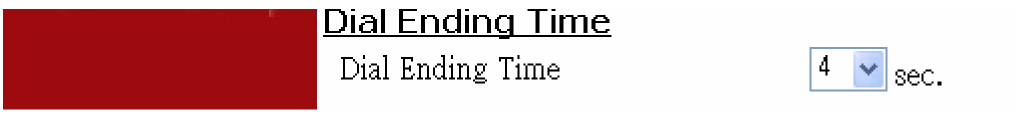
Force SIP Call: Dial <#>+ <numbers>+<#>.

1. Compare SIP Phone books; check the number if it is in phone book.
2. If the number is in setting and Proxy selection is set to "No", you will hear a busy tone. If Proxy selection is set to "Yes", then send the call to proxy.
3. If communicate with proxy is fail, you will hear a busy tone.
4. If the number is not in phone book, you will hear busy tone.

Force PSTN Call: Dial <*>+<numbers>+<#>.
Always go through PSTN

Notice: If you do not want to dial “#” after numbers, please configure the ‘Dial Ending Time’ item. After the seconds, the call will be sent automatically.

WEB Page: ADVANCED\GENERAL



10.9. Contact Address

The main purpose of Contact Address is making SIP calls without proxy.

The Contact Address can be any numbers or characters such as 'Mary' or '1003'.

WEB Page: ADVANCED\SIP COMMOM

Making SIP calls without proxy server:

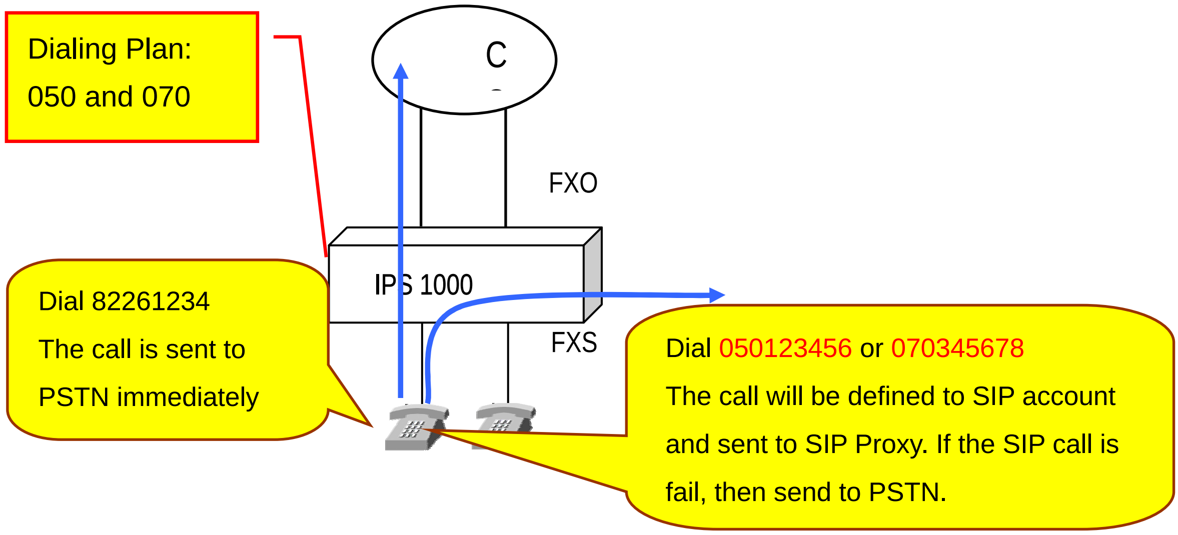
The SIP protocol allows you to make SIP calls directly to the destination number without through the proxy server. You can simply dial the SIP number and domain name or IP address. The typical example is: mary@artdioinc.com or 1003@artdioinc.com.

Notice: For this type of SIP calls, the destination device's IP address is already known and fixed.

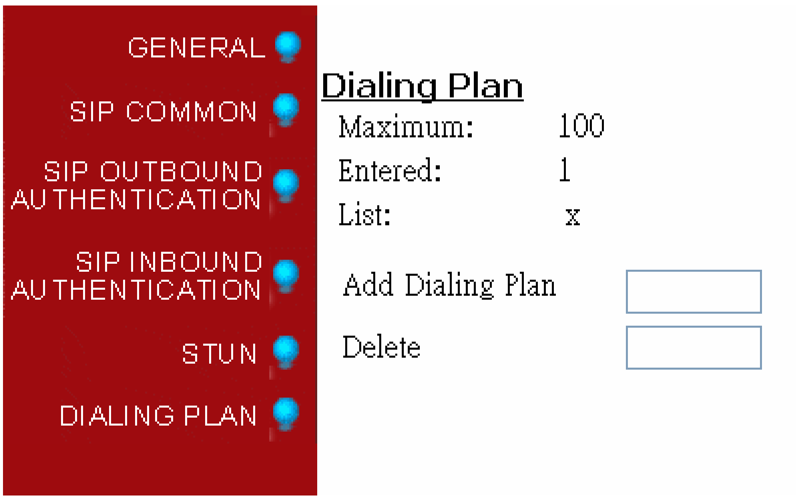
11. Other SIP Parameters

11.1. Dialing Plan

X means all calls will be send to SIP proxy first, if the SIP call is fail, and then sent to PSTN.
If the configuration is only '050' means the numbers like 050xxxxx will send to SIP proxy, if you dial any other numbers like 100, the number will send to PSTN immediately.



Configuration
WEB Page: ADVANCED\Dialing Plan



11.2. Call Forward

There are three forward types:

1. All: All incoming call to the SIP entity will be forward.
2. Busy: When the SIP entity is busy, the incoming call will be forward.
3. No Answer: When the SIP entity is no answer and after 30 seconds, the incoming call will be forward.

Notice: In order to let the caller identify the port has been configured "forward"; the caller will hear second dial tone, rather than normal dial tone.

Configuration

WEB page: ADVANCED\SIP COMMOM

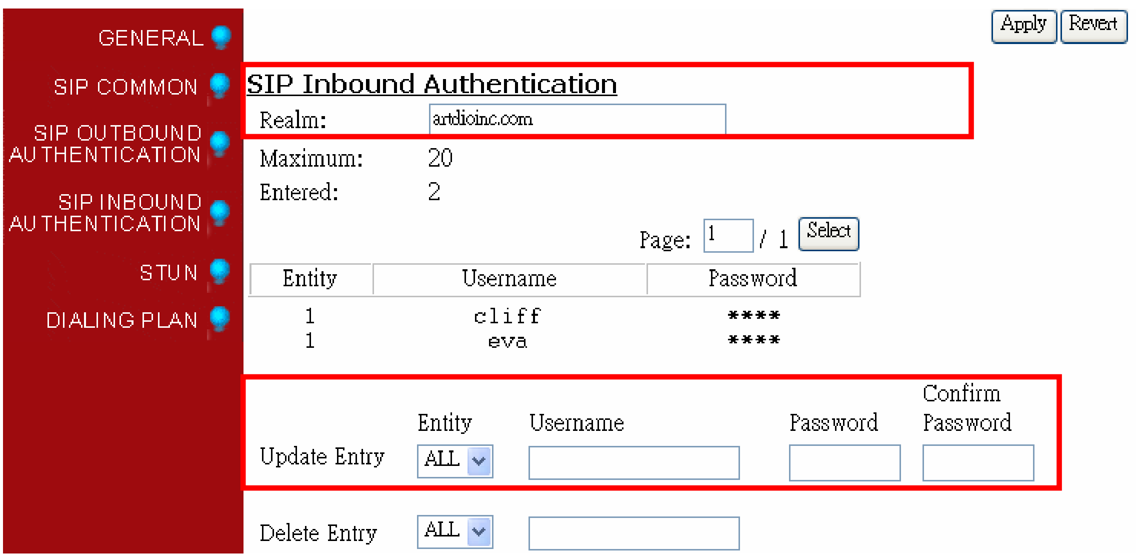
Phone Set: Please refer to section Appendix A: Phone-Set Command.

11.3. Inbound Authentication

You need to configure inbound authentication if you request authentication for other SIP phone to call you.

Configuration

WEB Page: ADVANCED \ SIP INBOUND AUTHENTICATION



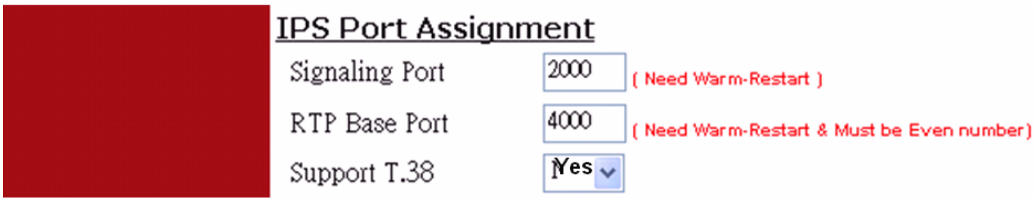
11.4. FAX

11.4.1. The devices at two sides are all IPS 1000 series gateway

Use the FAX protocol that is the proprietary protocol of IPS (supporting T.38). Setup method is listed below:

1. Web Folder: “Connect Device” in “Channel” folder. Select “FAX” and then click “Apply” button

2. Web folder: “IPS Protocol” in “Basic” folder
   Signaling Port: input “2000”
   Support T.38: select “Yes”
   Click “Apply” button



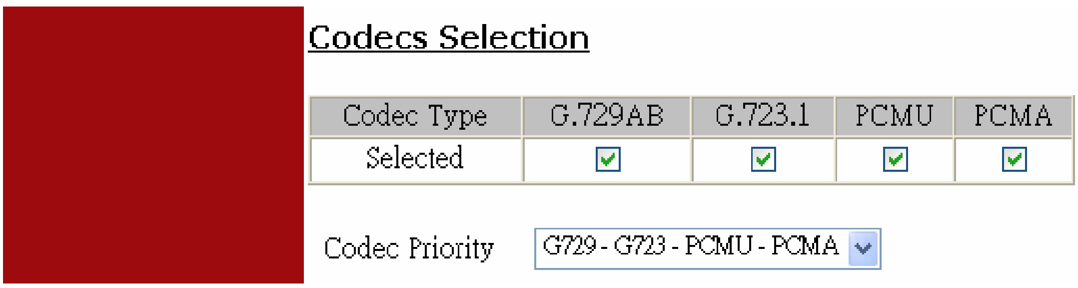
3. Warm-Restart the system

11.4.2. The devices at two sides are IPS 1000 and the other brands

Use the FAX protocol as G.711 (non-supporting T.38). Setup method is listed below:

1. Web folder: “Connect Device” in “Channel” folder. Select “FAX” and then click “Apply” button

2. Setup “Check Protocol”, web folder: ADVANCED\SIP COMMON
Select and mark “PCMU” and “PCMA” Codecs, than click “Apply” button

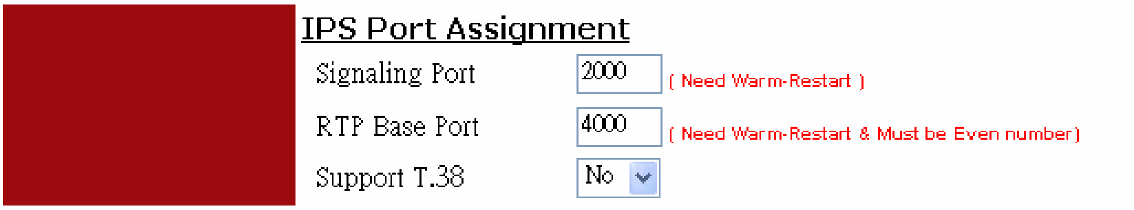


3. Web folder: “IPS Protocol” in “Basic” folder

Signaling Port: input “0”

Support T.38: select “No”

Click “Apply” button



4. Warm-Restart the system

12. WEB MANAGEMENT INTERFACE

The Tree Architecture of Web Management

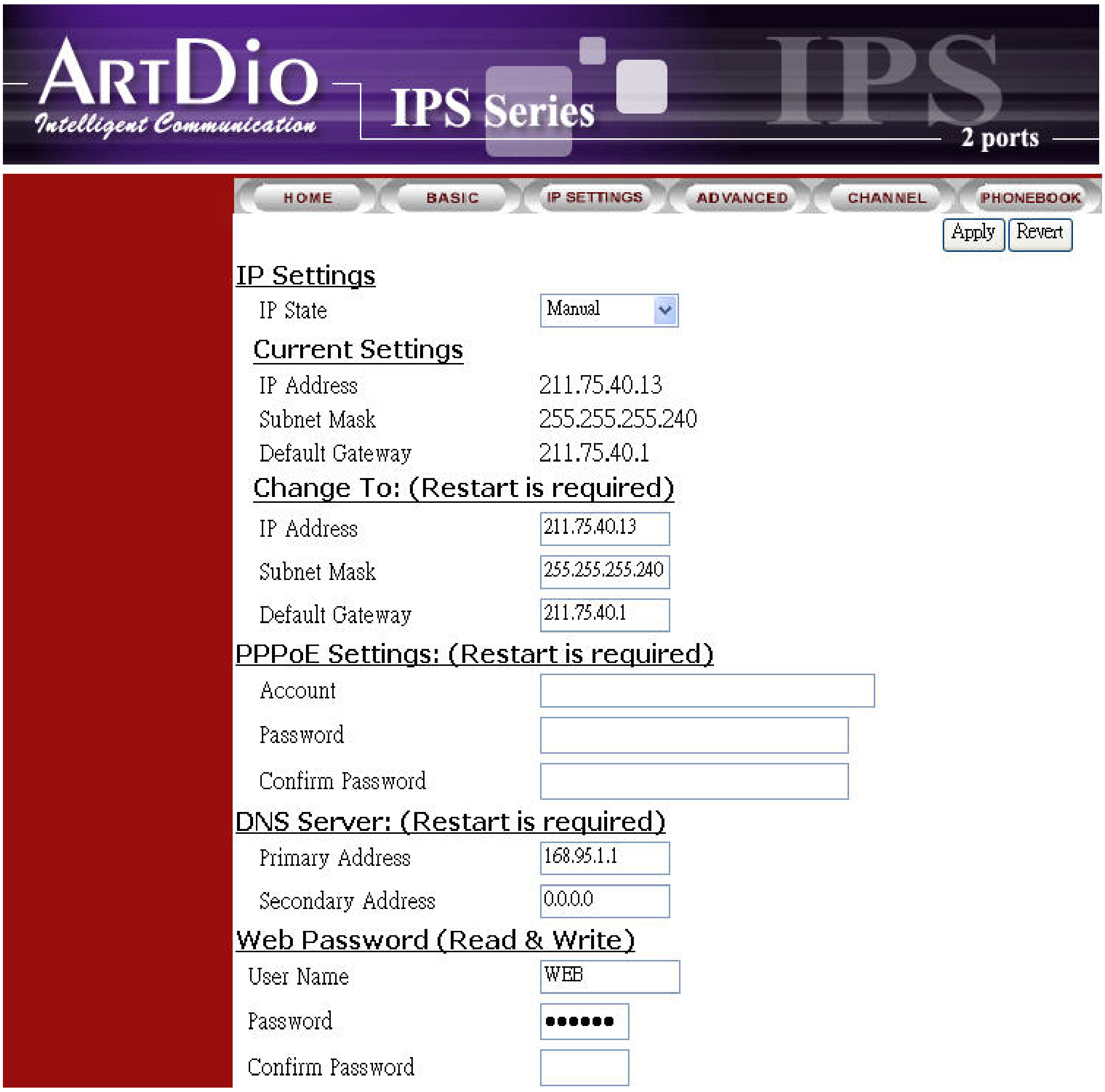
HOME	BASIC	GENERAL
	IP SETTING	
	ADVANCED	General
		SIP COMMON
		SIP OUTBOUND AUTHENTICATION
		SIP INBOUND ATHENTICATION
		STUN
		Dialing Plan
	CHANNEL	
	PHONE BOOK	
	ACCESS CODE	

12.1. BASIC / GENERAL

Category	Section	Description	Default Setting
Information	Region ID	Display region ID.(Read only)	0
	Software Version	Display software version.(Read only)	
	BootRom Version	Display BootRom Version.(Read only)	
	Hardware Version	Display hardware Version.(Read only)	
	Card Type	Display card type. (Read only)	

	Up-Time	Display the use time since from system reboot.(Read only)	
	MAC Address	Display MAC address.(Read only)	
	Date	Show the date	
	Time	Show the time	
Time Configuration	Date	Manually Input date, only effected in Manual Mode. yyyy / mm / dd	Empty
	Time	Manual input time, only effected in Manual Mode of Time Source. hh : mm : ss	Empty
	Time Zone	Select local system time zone. Select correct Time Zone.	
	Daylight saving	ON: Enable daylight saving. OFF: Disable daylight saving.	OFF
IPS protocol	Signaling Port	UDP port to transfer signal packets. It can be setting in the range of 0 to 65535. (Must reboot system to apply changes)(Only support IPS device)	0
	RTP Base Port	Base of UDP port to receive RTP packets. It can be setting in the range of 0 to 65534.(Must be Even, after setting this item, please reboot system to apply changes)	4000
	Support T.38	Enable/Disable the FAX relay (T.38) of IPS Protocol	No
System Restart	Restart Mode	None: Not to restart system. Cold restart: Cold restart. Warm restart: Warm restart.	None

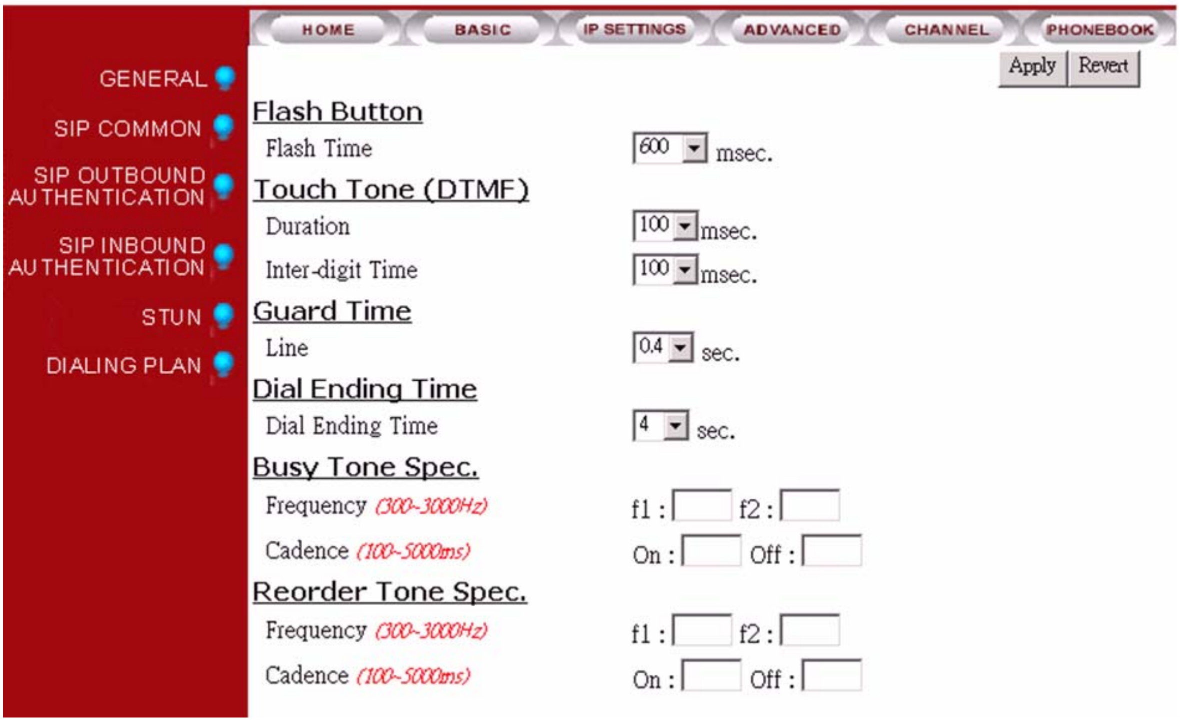
12.2. IP SETTING



Category	Section	Description	Default Setting
IP Settings	IP State	The way to obtain IP address: Manual: Entered by user (Static IP) Auto(DHCP): Assigned by DHCP server PPPoE: Assigned by PPPoE of ISP	Manual

	Current Setting	Display the configured IP address, subnet mask address and default gateway. (Read only)	192.168.0.2 255.255.255.0 192.168.0.1
	Change To	Enter the IP address that will be used after next restart, Including: IP Address Subnet Mask Address Default Gateway (This item is used only on Manual mode of IP Setting.)	
PPPoE Settings	Account	The user's account of PPPoE protocol, provided by ISP.	
	Password	The user's password of PPPoE protocol.	
	Confirm Password	Confirm the user's password of PPPoE protocol.	
	Service Name	The service name of PPPoE account, provided by ISP. (Most ISP doesn't need this)	
DNS Server	Primary Address	The primary address of DNS server. The default setting would be different according to the local area. In Taiwan, the default setting is 168.95.1.1.	168.95.1.1
	Secondary Address	The secondary address of DNS server.	
Web Password	User Name	The user's name of Web Management Interface.(12 character)	WEB
	Password	The password of Web Management Interface.(6 character)	
	Password Confirm	Enter the password again to confirm it.	

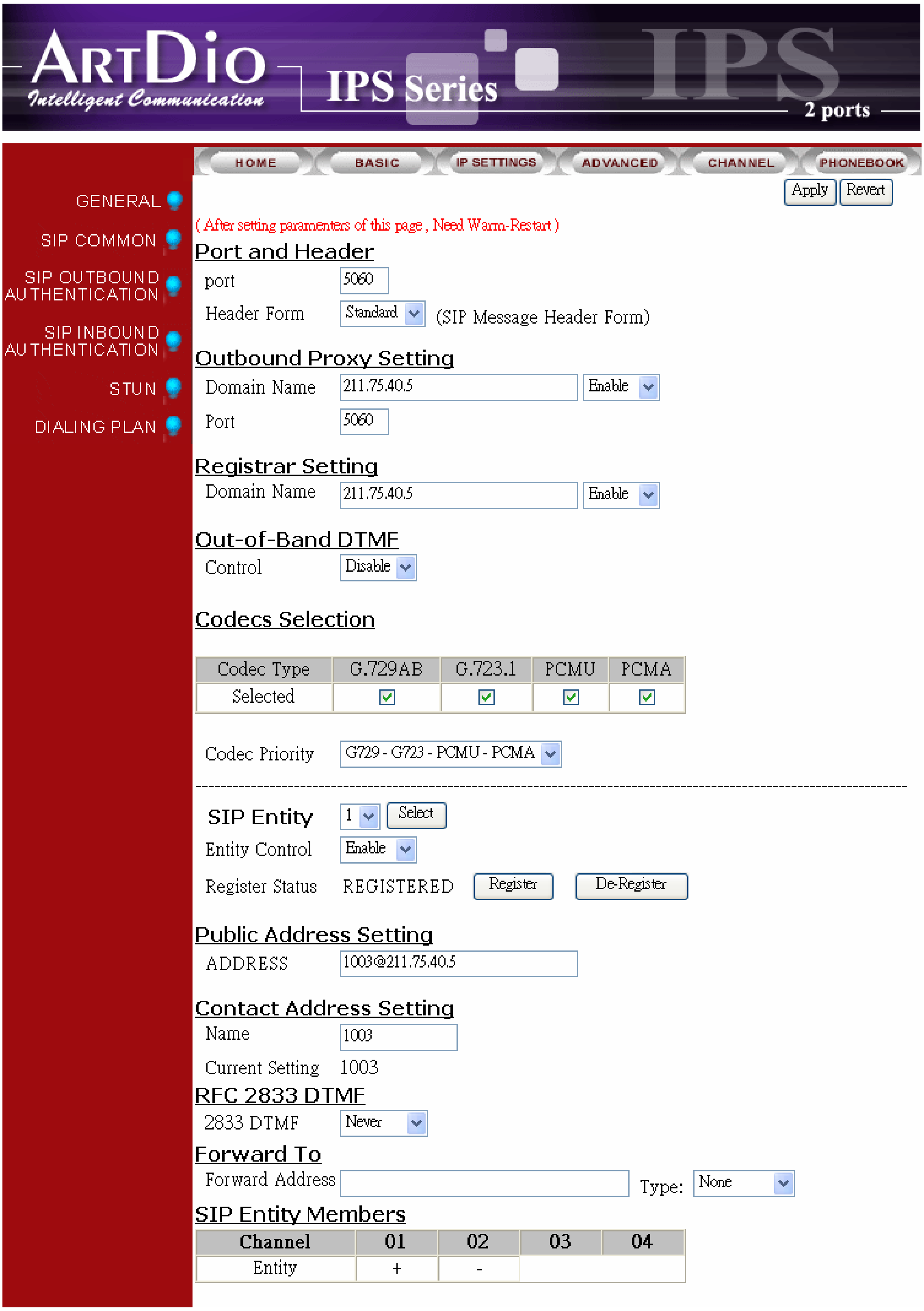
12.3. ADVANCED / GENERAL



Category	Section	Description	Default Setting
Flash Button	Flash Time	System confirmed “Flash” time.	200 msec
Touch Tone (DTMF)	Duration	The duration to send a DTMF.	100 msec
	Inter-digit	The inter-digit time of sending string of DTMF digits.	100 msec
Guard Time	Line	The time defines how long the system will not take incoming call after call has been disconnected.	0.8 sec
Dial Ending Time	Dial Ending Time	The time specifies how long to end the dialing number if a ‘#’ digit is missing.	4 1-10 (seconds)
Busy Tone Spec	Frequency	f1, f2	(300 ~ 3000Hz)

	Cadence	on, off. The on and off duration in playing the tone	(100 ~ 5000ms)
Reorder Tone Spec	Frequency	f1, f2	(300 ~ 3000Hz)
	Cadence	on, off. The on and off duration in playing the tone	(100 ~ 5000ms)

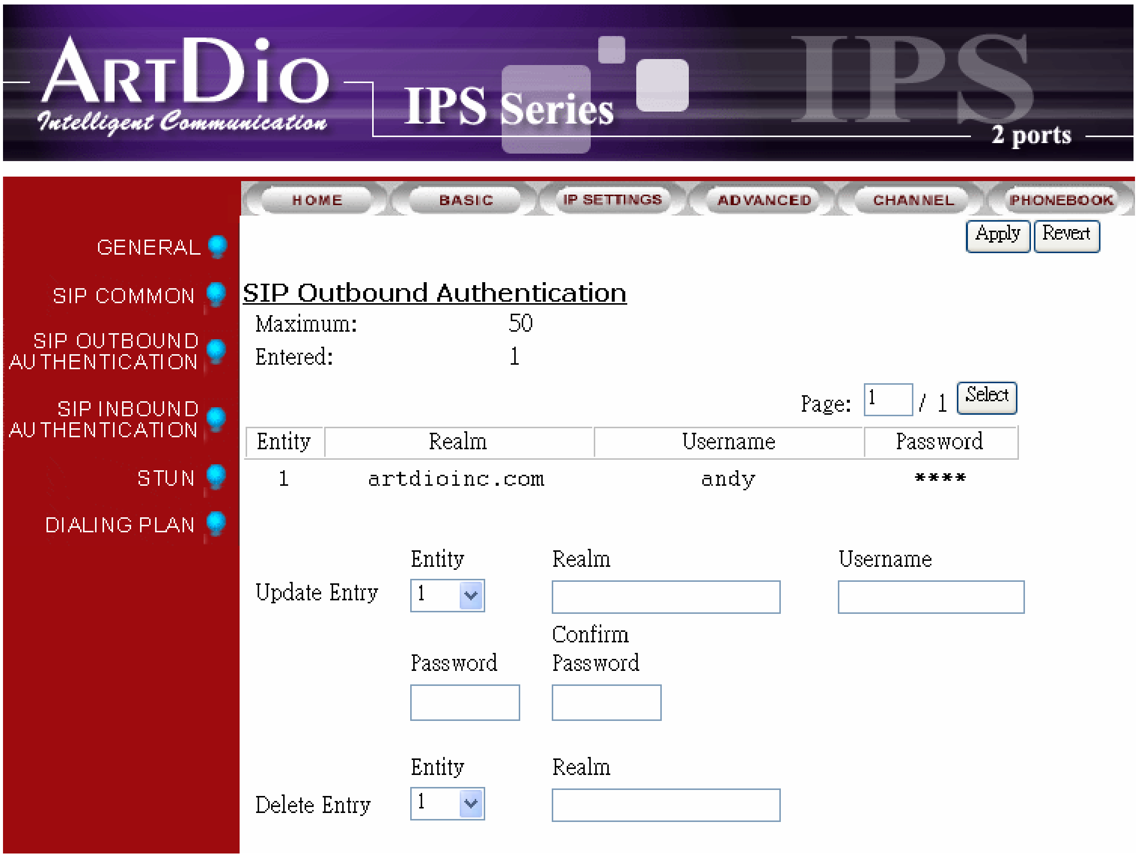
12.4. SIP COMMON



Section	Item Field	Description	Default
Port and Header	Port	The control port number of SIP protocol.	5060
	Header Form	Select ‘Standard’ or ‘Compact’ to be the header format of SIP packet. When Compact is selected, the header will be shorter and it saves bandwidth.	Standard
Outbound Proxy Setting	Domain Name	Domain name or IP address of proxy.	Empty Disable
	Port	Control port number of SIP protocol.	5060
Registrar Setting	Domain Name	Domain name or IP address of proxy that you want to register.	Empty Disable
Out-band DTMF	Control	Enable/Disable	Disable
Codecs Selection	Codec Type	G.729AB: Mark the selection to Enable G.729AB Codec	Enable
		G.723.1: Mark the selection to Enable G.723.1 Codec	Enable
		PCMU: Mark the selection to Enable PCMU Codec	Enable
		PCMA: Mark the selection to Enable PCMA Codec	Enable
	Codec Priority	You can select the codec priority for your requirement.	G729-G723-PCMU-PCMA
SIP Entity		Select an entity Select: Select Button Register: Register Button De-Register: Cancel Register Button	1
Entity Control		Select Enable/Disable	Enable
Register Status	Register Status	Show the register status, if it shows Registered means successful. (Read only) Register: Register Button De-Register: Cancel Register Button	Empty
Public Address Setting	Address	Enter SIP phone number of the port. The phone number general assigned by SIP service provider.	Empty

Section	Item Field	Description	Default
Contact Address Setting	Name	Enter Contact Address. You can assign a name for the port. However, it should be unique within per channels.	Empty
	Current Setting	Display current setting of (Read Only) Contact Address	01
RFC 2833 DTMF	2833 DTMF	Enable/Disable RFC 2833 DTMF.	Never
	2833 In Use	Display current status of (Read Only) DTMF configuration.	
Forward To	Forward Address	Enter a SIP account (Public Address) forward. When users dial into the SIP Entity, the call will be forwarded to the number.	Empty
	Type	N/A: All incoming calls are forward. Busy: When the SIP entity is busy, the calls will be forward. No Answer: When the SIP entity is no answer about 30 seconds, the calls will be forwarded.	N/A
SIP Entity Members	Channel	Show the all channels	Depend on gateways
	Entity	Show ‘+ ‘means the SIP entity is for the channel.	Empty

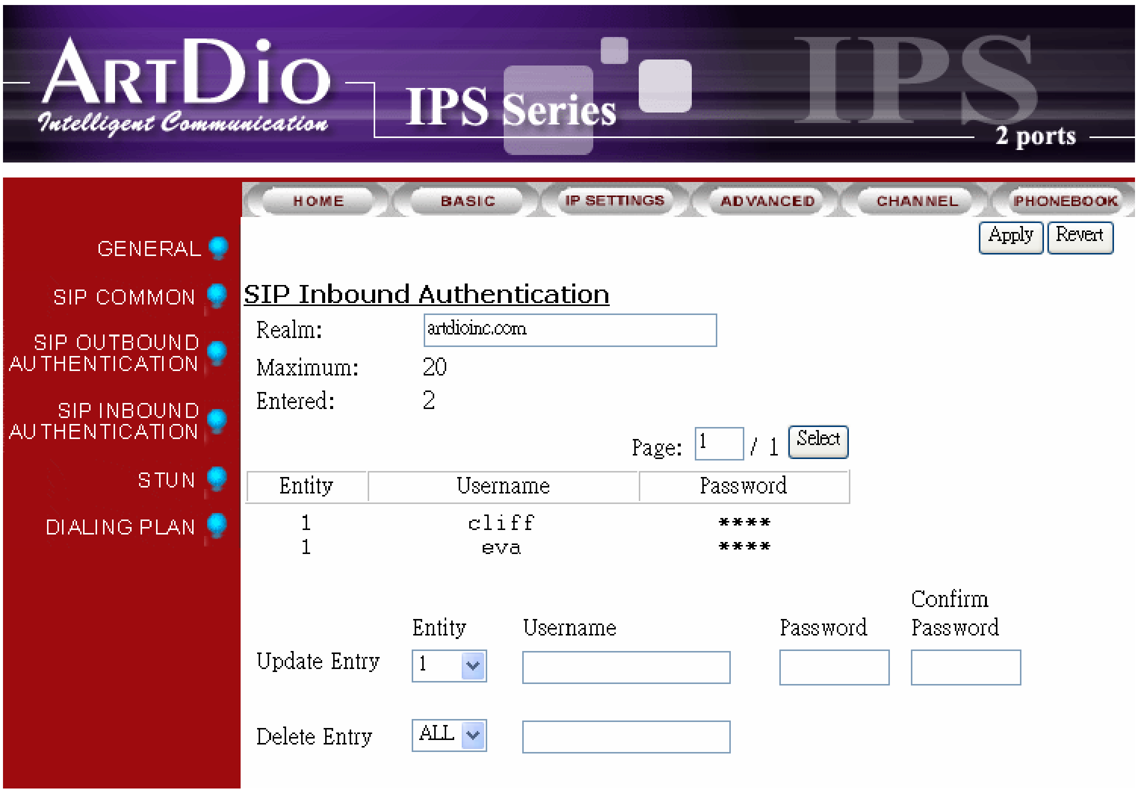
12.5. SIP OUTBOUND AUTHENTICATION



Section	Item Field	Description	Default
SIP Outbound Authentication	Maximum	Maximum number of entries (Read Only) allowed	50
	Entered	Number of entries of (Read Only) authentication entered.	0
	Entries List	List of entries (Read Only) Entity: Which entity that you select. Realm: Domain name or IP address. Username: Username of authentication. Password: Password of authentication.	Empty

Section	Item Field	Description	Default
	Update Entry	Enter the information of outbound authentication Entity: Select an entity. Realm: Domain name or IP address. Username: Enter Username of authentication. Password: Enter password of authentication. Confirm Password: Enter password again for confirmation.	Empty
	Delete Entry	Delete the information of outbound authentication Entity: Select an entity. Realm: Domain name or IP address.	Empty

12.6. SIP INBOUND ANTHENTICATION

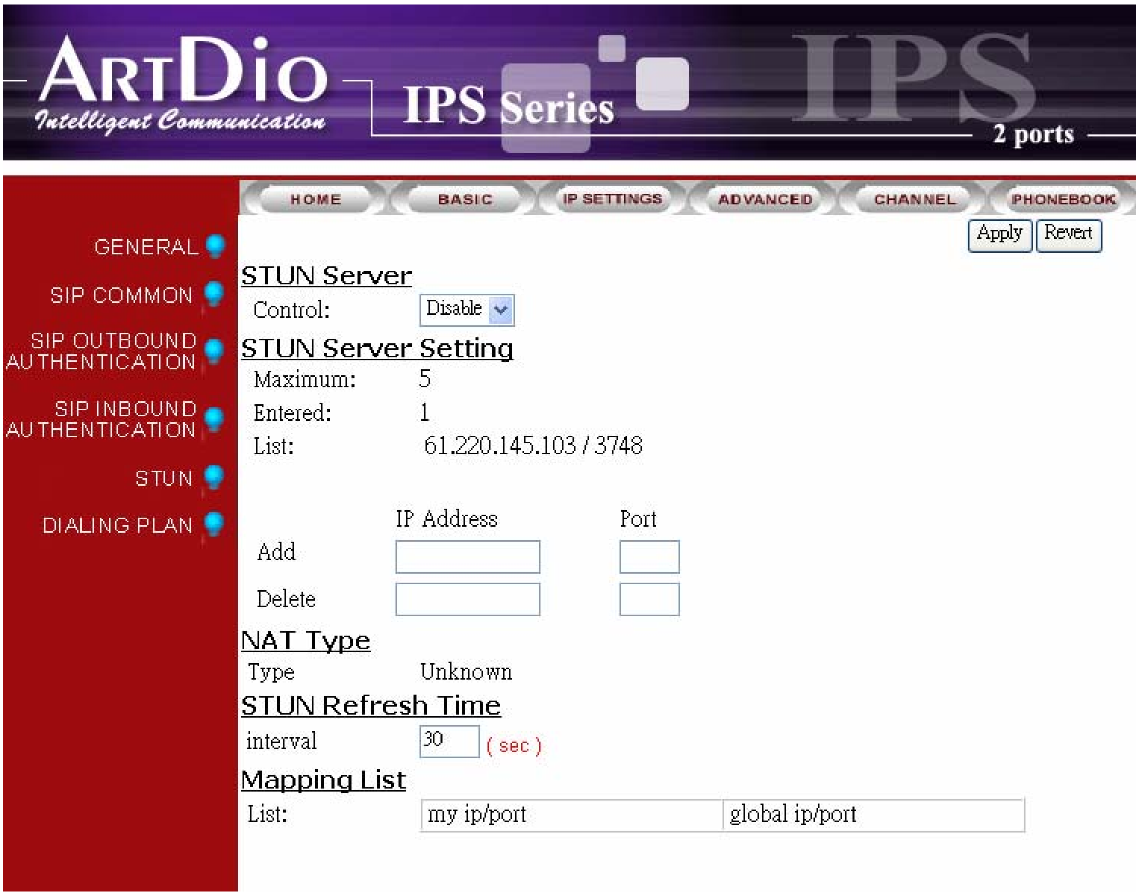


Section	Item Field	Description	Default
SIP Inbound Authentication	Realm	Enter domain name or IP address	Empty
	Maximum	Maximum number of entries allowed (Read Only)	20
	Entered	Number of entries of authentication entered. (Read Only)	0
	Entries List	Display the entries (Read Only) Entity: Which entity that you select. Username: Username of authentication. Password: Password of authentication.	Empty
	Update Entry	Enter entries of authentication Entity: Which entity that you select. Username: Username of authentication. Password: Password of authentication. Confirm Password: Enter password again for confirmation.	Empty
	Delete Entry	Delete entries of authentication Entity: Which entity that you want to delete. Username: Username of authentication.	Empty

12.7. Dialing Plan

Section	Item Field	Description	Default
DIALING PLAN	Maximum	Maximum number of (Read Only) entries allowed	100
	Entered	Number of entries of (Read Only) authentication entered.	0
	List	Display the entries (Read Only) The default value “x“means that all numbers that you dial will first go through SIP proxy. If the call communicates with SIP proxy is fail, it will be transferred to PSTN.	x
	Add Dialing Plan	Enter numbers. Example: 050.	Empty
	Delete Entry	Enter numbers for delete.	Empty

12.8. STUN



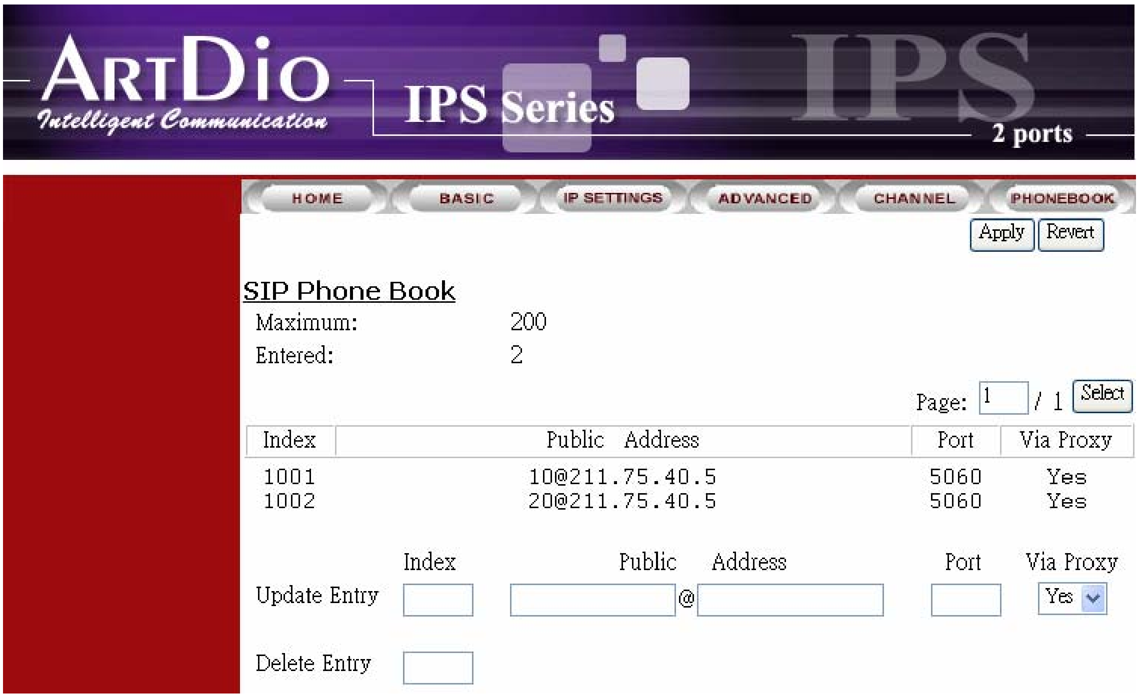
Section	Item Field	Description	Default
STUN Server	Control	Enable or Disable STUN Server service.	Disable
STUN Server Setting	Maximum	Maximum number of entries allowed (Read Only)	5
	Entered	Number of entries of STUN server that have been entered. (Read Only)	0
	List	Display all of servers that have been entered. (Read Only)	
	Add	Add a stun server IP Address: Enter IP address or Domain Name Port: Enter port number of service.	Empty

Section	Item Field	Description	Default
	Delete	Delete a stun server IP Address: Enter IP address or Domain Name. Port: Enter port number of service.	Empty
NAT Type	Type	Display NAT type (Read Only)	Unknown
Stun Refresh Time	Interval	It defines how long the device will send a binding request packet with discard flag on to STUN server.	30
Mapping List	List	My ip/port: shows the (Read Only) private IP and port number. Global ip/port: Display public IP and port number.	Empty

12.9. CHANNEL

Category	Section	Description	Default Setting
	Channel	Channel number:	1
Information	Channel Type	Display port type. (Read only) Phone: FXS Interface, connect to telephone set or Fax machine. Line: FXO Interface, connect to phone line. NA: Not available.	
	Channel Control	Enable/Disable all functions of this port. Enable/Disable	Enable
	Current State	Display the current state of this port. (Read only) Enable/ Disable.	
	Do not Disturb	Enable/Disable does not disturb function	Disable
	Silence Suppression	Enable/Disable the function.	Disable
	2833 In use	Yes: (Read only) No:	
	Join SIP Entity	Select an Entity for SIP.	1
	Connect Device	Phone: Connect to the FXS port is regular phone FAX: Connect to the FXS port is FAX machine	Phone
Voice	Input Gain	Adjust Voice input Gain	0
	Output Gain	Adjust Voice output Gain	0

12.10. PHONE BOOK



Section	Item Field	Description	Default
SIP Phone Book	Maximum	Maximum number of entries (Read Only) allowed	200
	Entered	Number of entries of phone (Read Only) books entered.	0
	Entries List	Display phone books (Read Only) Index: Dialing number Public Address: SIP account. Port: Port number. Via Proxy: Via proxy or not.	Empty
	Update Entry	Enter entries Index: Enter dialing number Public Address: Enter SIP account. Port: Enter port number Via Proxy: Select via Proxy or not	Empty
	Delete Entry	Delete entries Index: Enter the index for delete.	Empty

13. Use Private IP (Behind NAT)

Using a Private IP in a NAT Environment
The IPS unit is able to communicate with other IPS units under a NAT environment using Private IP addresses on the LAN side of your IP Sharing device. However you must configure the IP Sharing device to treat the IPS unit as a Virtual Server using UDP port 5060, 2000.

You will have to ask MIS personnel to enable the ports listed in the following table.

Packet Modes	Using Ports
SIP Signal Packets	UDP 5060
IPS Signaling Port	UDP 2000
IPS RTP Base Port	UDP 4000
FTP software upgrade	TCP 21
Web management	TCP 80

If you want to use private IP behind NAT and Proxy Server is in Internet, you must need to enable STUN service. If the system is installed in VPN, it is not necessary to Enable Stun.

14. Appendix

14.1. Appendix A: Phone-Set Command

Pick up the handset and listen for the dialing tone. Dial “##0000 and listen for three consecutive tones before setting the following parameters. After input the parameters, please dial ‘# to end the configuration.

Command	Description	Parameters
01	IP State	0 : static; 1: DHCP; 2: PPPoE
02	IP Address	xxx*xxx*xxx*xxx
03	Subnet Mask	xxx*xxx*xxx*xxx
04	Default Gateway	xxx*xxx*xxx*xxx
05	Primary DNS Server IP	xxx*xxx*xxx*xxx
06	Second DNS Server IP	xxx*xxx*xxx*xxx
07	Select Signaling Port	0~65535
08	Select RTP Base Port	0~65534 (limit to even port number only)
09	PPPoE username	User name (use the mapping table to map character into digits)
10	PPPoE password	Password (use the mapping table to map character into digits)
11	DND	0 : Disable ; 1: Enable
12	SIP Forward State	0 : Disable ; 1: Enable; 2: Busy; 3: No Answer
13	SIP Forward Target	6 Digits
14	Change Service Port	1:FTP; 2:HTTP 3:Telnet (Port: 0-65535)
15	Change WEB Password	6 digits
16	Change FTP Password	6 digits
40	Listen for the IP Address	(ending ”#” is not required)
41	Listen for the Subnet	(ending ”#” is not required)

	Mask	
42	Listen for the Default Gateway	(ending "#" is not required)
43	Listen for Current Signaling Port	(ending "#" is not required)
44	Listen for Global IP Address	(ending "#" is not required)
45	Listen for Global Signaling Port	(ending "#" is not required)
46	Listen for WEB, FTP, Telnet Port	1:FTP; 2:HTTP 3:Telnet
47	Listen for Current Public Address	(ending "#" is not required)
96	Region ID	2 digits
97	Reset unit to Factory Default values	1: reset all; 2: keep IP
98	System Warm Restart	1: do it

14.2. Appendix B: Console Command

User Exec commands

Enable	Turn on privileged commands
Exit	Exit from the EXEC
Help	Description of the interactive help system
Show	Show running system information

show

Dns	Show the IP address of domain name server
ethernet	Fast Ethernet port status and configuration
history	Display the session command history
Ip	Display IP configuration
running-config	Show current operating configuration
version	System hardware and software status

Privileged Mode

Configure	Enter configuration mode
Delete	Reset configuration
Disable	Turn off privileged commands
Exit	Exit from the EXEC
Help	Description of the interactive help system
Ping	Send echo request to destination
Probe-hook	probe busy tone cadence
Probe-remove	stop probe busy tone cadence
Reload	Halt and perform cold start
Restart	Halt and perform warm start
Show	Show running system information

Global Mode

Dbflush	Data Base flush
Dns	Set the IP address of domain name server
End	Exit from configure mode to privileged mode
Exit	Exit from configure mode
Help	Description of the interactive help system
Ip	Global IP configuration subcommands
Log	Control log output
No	Negate a command or set its defaults
pppoe	PPPoE configuration subcommands
regional_id	Set regional id
service_port	Set service port number

14.3. Specifications

Voice Interface	
FXS interface	Loop start, 2 wire Feeding Voltage: 20V Feeding Current: 30 mA
FXO interface	Loop start, 2 wire
Connectors	RJ-11 Connectors (3702/3704) IDC Connectors (3708/3716)
Voice compression	G.711/G.723/G.729AB
Silence suppression	VAD, CNG
Echo cancellation	G.165/G.168 16ms
Jitter buffer	Adaptive jitter buffer management
Gain control	In/Out +/-6db
Transport protocols	RTP, RTCP
Call control protocol	Pure SIP
Network Interface	
Number of ports	Two Ethernet ports
Interface	10BASE-T/100BASE-TX Auto-negotiation
Connectors	RJ-45 Connectors
General Spec	
Dimension	IPS 3702: 190mm x 110mm x 25 mm IPS 3704: 172mm x 177mm x 35 mm IPS 3708: 440mm x 44mm x 254 mm IPS 3716: 440mm x 66mm x 254 mm
Power	Voltage: 100-240 VAC, Frequency: 50/60 Hz
Power consumption	IPS 3702: 8 W IPS 3704: 12W IPS 3708/3716: 70W
Working environment	Operating temperature: 0 to 50℃ Storage temperature: -10 to 70℃
EMI	FCC part 15 Class B . CE Mark
PTT	FCC part 68 , NALTE , iDA , JATE
Safety	cUL , CCIB , CB

14.4. Mapping table of characters used in PPPoE

Character	Digits to key-in	Character	Digits to key-in
0	30	X	58
1	31	Y	59
2	32	Z	5*0
3	33	a	61
4	34	b	62
5	35	c	63
6	36	d	64
7	37	e	65
8	38	f	66
9	39	g	67
@	40	h	68
A	41	i	69
B	42	j	6*0
C	43	k	6*1
D	44	l	6*2
E	45	m	6*3
F	46	n	6*4
G	47	o	6*5
H	48	p	70
I	49	q	71
J	4*0	r	72
K	4*1	s	73
L	4*2	t	74
M	4*3	u	75
N	4*4	u	76
O	4*5	w	77
P	50	x	78
Q	51	y	79
R	52	z	7*0
S	53	=	3*3
T	54	.	2*4

U	55	@hinet.net	**01
V	56		
W	57		

14.5. Region ID

Country	Region ID	Country	Region ID
Australia	02	Korea	24
Philippines	03	Malaysia	26
Canada	06	Singapore	36
China	07	Slovenia	38
Vietnam	10	Spain	40
France	12	Taiwan	43
Germany	13	Thailand	44
Hong Kong	15	British	46
Italy	22	USA	47
Japan	23		

15. Contact Information

If you have any question please feel free to ask our technical support, or visit our website http://www.artdioinc.com.

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