# MOSA 3700
# Pure SIP Gateway
# User Manual

Version: 12.0

Firmware: 2.00

Update: 2008/05/26

DCC NO. 91710015012

# Table of Contents

**2. Safety Instructions** ........ 5
**3. Preface** ........ 5
3.1 What is SIP ........ 5
3.1.1 SIP Clients ........ 5
3.1.2 SIP Servers ........ 6
**4. Package Contents** ........ 7
**5. Panel Descriptions** ........ 7
5.1 Front Panel ........ 7
5.2 Rear Panel ........ 8
5.3 LED Indicators ........ 9
5.4 Connectors ........ 10
5.4.1 Connection of Network Cable ........ 10
5.4.2 Connection of Console Port ........ 11
5.5 Connection of 8/16 Ports Model ........ 12
5.5.1 Installation of Modules ........ 12
5.5.2 Numbering of Module ........ 12
5.5.3 IDC Connectors (Only for 3708/3716) ........ 13
5.5.4 Connection between IDC Connector and Phone Set ........ 13
**6. Initial Setting of a Single Machine** ........ 14
6.1 Connection of Basic Structure ........ 14
6.2 Phone Set Configuration (Phone Set Programming Mode) ........ 15
6.3 Configuration of Telecom Region ID ........ 15
6.4 Configuration of IP Address ........ 16
6.5 Restart ........ 16
6.6 Configuration of Router ........ 17
6.6.1 Configure PC to connect router ........ 17
6.6.2 Configure Router to connect Internet ........ 19
**7. SIP Configuration** ........ 21
7.1 Register to SIP Telephony Server Provider ........ 21
7.2 Channels and SIP entity ........ 23
7.2.1 Create Entity ........ 23
7.2.2 Assign Channel to Entity ........ 24
7.3 SIP Outbound Authentication ........ 25
7.4 Configure STUN for Client under NAT (Optional) ........ 26
7.5 Check SIP Entity Registration Status ........ 27

7.6 Phone Book ....28
7.6.1 General Phone Book .... 28
7.7 Make SIP Calls ....29
7.8 Make Inbound Transit Call ....30
7.9 Make SIP IP Call without SIP Proxy....33
**8. Advanced Parameters .... 33**
8.1 Hotline Function....33
8.2 Configuration of Dialing Plan ....36
8.2.1 Dialing Method.... 36
8.2.2 Digit Map .... 38
8.2.3 Dial in Rewriting Rule .... 39
8.3 Call Forward ....40
8.4 Inbound Authentication ....41
8.5 FAX....42
8.5.1 Connect FXO port to PSTN .... 42
8.5.2 Connect FXS port to FAX Machine.... 43
8.6 Non-SIP Call port seizure preference ....43
8.7 Call Waiting....44
8.8 Target the Media (RTP)....46
**9. File Management.... 47**
9.1 File Types ....47
9.2 Software Update ....48
9.2.1 Software update via Web .... 48
9.2.2 Software update via FTP .... 48
**10. WEB MANAGEMENT INTERFACE .... 51**
10.1 1.SIP Environment/1.1.Proxy/Trunk Mapping ....52
10.2 1.SIP Environment/1.2.Common ....54
10.3 2.Channel Config./2.1.Summary....55
10.3.1 Home.... 55
10.3.2 FXS Channel Setting.... 56
10.3.3 FXO Channel Setting.... 57
10.3.4 Transit in setting .... 59
10.4 2.Channel Config./2.2.Status ....60
10.5 3.SIP Advanced\3.1.Inbound Authen. ....61
10.6 3.SIP Advanced\3.2.Outbound Authen. ....62
10.7 3.SIP Advanced\3.3.SIP Phone Book ....64
10.8 3.SIP Advanced\ 3.4.WAN IP & STUN....65
10.9 4.System Config. ....68

10.10 5.PBX Advanced ........................................ 70
10.11 6.IP Settings ........................................ 72
10.12 7.Dial Plan ........................................ 74
10.13 8.File Transfer ........................................ 76
10.14 About Sample MEM File in Web ........................................ 78
**11. Appendix ........................................ 80**
11.1 Appendix A: Phone-Set Command ........................................ 80
11.2 Appendix B: Console Command ........................................ 82

| Change History: Software Version 2.00 | |
|---|---|
| New hardware with brand new management web | To work with new high performance hardware, management web is changed for new hardware. Old management web is discarded. |
| Change History: Software Version 1.10 or below | |
| Please read previous original manual | If you have previous MOSA 3700 hardware, please read MOSA 3700 User Manual (English) V11. |

# 2. Safety Instructions

WARNING

1. Do not attempt to service the product yourself. Any servicing of this product should be referred to qualified service personnel.
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.

**Notice**: The installation of MOSA 3700 is easy and quickly. Most of setting is pre-configured. Please read MOSA 3700 Quick Installation Guide for installation first. If you have further configuration, you can refer to this manual.

# 3. Preface

The MOSA 3700 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.

## 3.1 What is SIP

### 3.1.1 SIP Clients

SIP clients include the following:

(1) SIP Softphone: SIP client Software that runs at PC. It support SIP standard and can register to SIP Proxy for making calls.

(2) SIP Gateway: SIP client Software that runs at a box. It support SIP standard and can register to SIP Proxy. General phone-set that connect to this box can make SIP IP call.

(3) SIP IP Phone: SIP client Software that runs at a device that looks like general Phone-set. It support SIP standard and can register to SIP Proxy for making calls as using general phone-set.

(3) SIP Wi-Fi Phone: SIP client Software that runs at portable phone with wireless LAN connection.

It support SIP standard and can register to SIP Proxy. If wireless LAN connection keeps, the Phone can make calls in certain range without wiring.
MOSA 3700 is a SIP gateway with many FXS ports that can connect to general phone-set.

### 3.1.2 SIP Servers

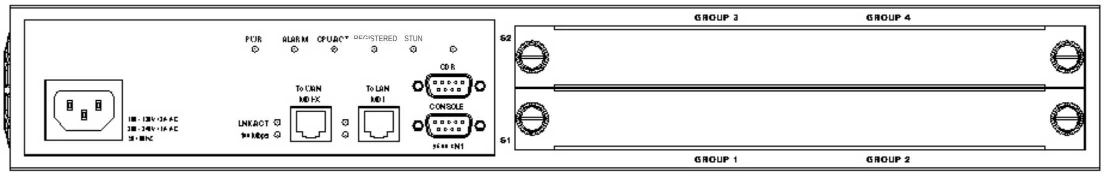
SIP servers include the following:
(1) Proxy server—The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
(2) 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.
(3) Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.
**Hint:** For most of ITSP (Internet Telephony Service Provider), the address (domain) of the servers above is consistent.
There are several series of products of VODTEL, such as MOSA 4600 Plus, MOSA 4600B, MOSA 4600D and MOSA 4600E, have SIP Proxy Server function for different purposes. Welcome to contact with distributor or VODTEL for detail.

# 4. Package Contents

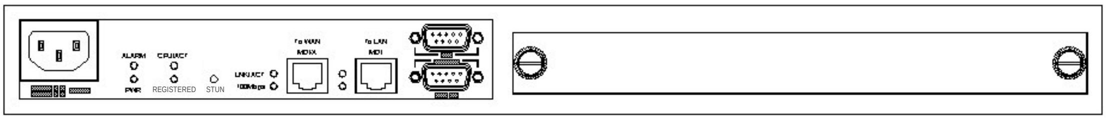
| | | | |
|---|---|---|---|
| The MOSA 3700 Gateway | X | 1 | |
| Power Core | X | 1 | |
| Accessories for rack support | X | 1 | (For 3708/3716) |
| System CD-ROM | X | 1 | |
| IDC Connector | | | (For 3708/3716) |
| Rubber footer | | | |
| RJ-45 Ethernet Cable | | | |
| RJ-11 Telephone Cable | | | |

# 5. Panel Descriptions

## 5.1 Front Panel



MOSA 3716 Front Panel

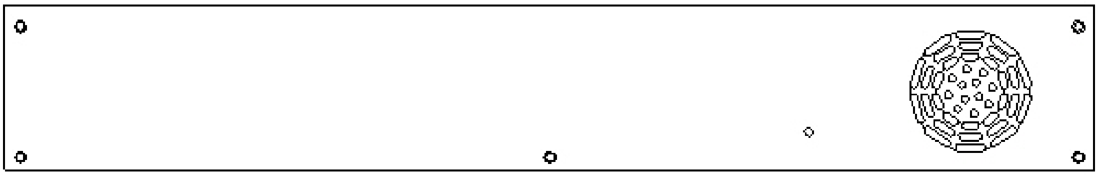

MOSA 3708 Front Panel

MOSA 3704 Front Panel

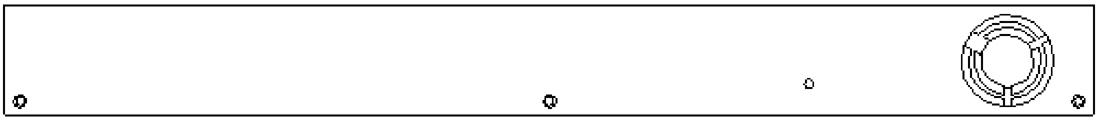
MOSA 3702 Front Panel

## 5.2 Rear Panel

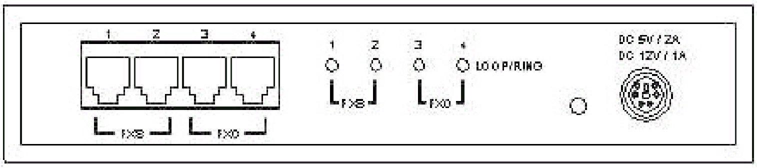
There is a button on the rear panel of gateway for special maintenance. Please don't touch this button under normal operation.



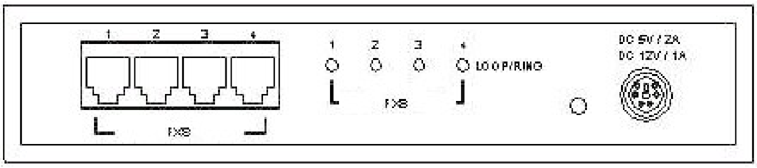
MOSA 3716 Rear Panel



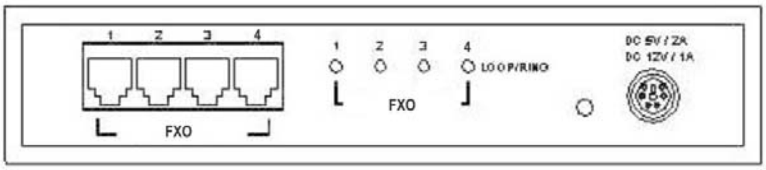
MOSA 3708 Rear Panel



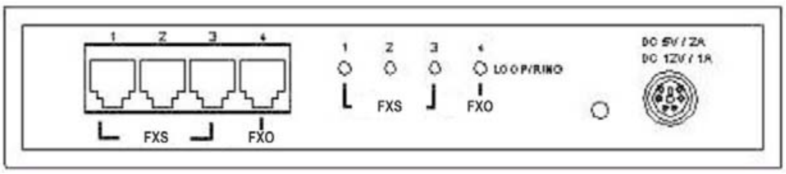

MOSA 3704A Rear Panel



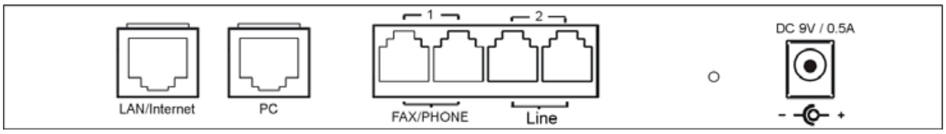

MOSA 3704B Rear Panel



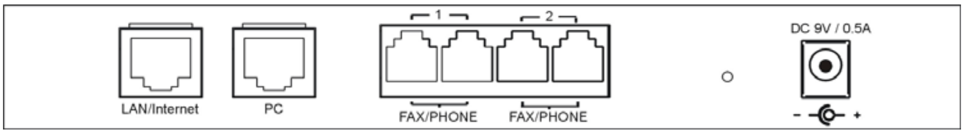

MOSA 3704C Rear Panel




MOSA 3704D Rear Panel




MOSA 3702A Rear Panel




MOSA 3702B Rear Panel

## 5.3 LED Indicators

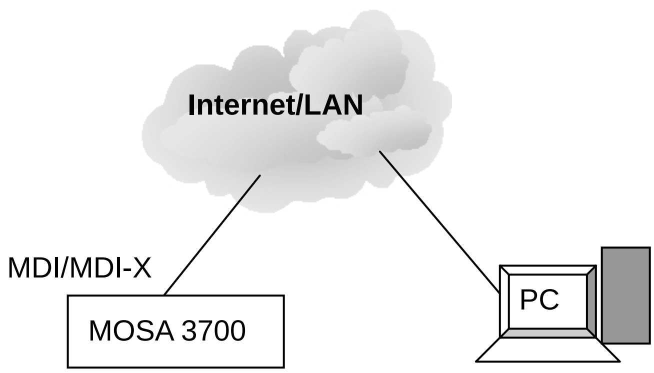
| LED | Label | Description | |
|---|---|---|---|
| 10/100 Ethernet | LNK/ACT | On | Link up |
| | | Off | Link down |
| | | Flash | Sending/Receiving data packets |
| | 100Mbps | On (LNK is on) | 100Mbps |
| | | Off (LNK is on) | 10Mbps |
| LOOP/RING | FXS | On | Off hook |
| | | Off | On hook |
| |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

## 5.4 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<br>MDI-X connects to a Modem |
| | PC | RJ-45 connector<br>MDI connects to a PC |
| Console Port<br>(Only 3704/3708/3716) | Console | RJ-45 connector/RS-232 Interface |

### 5.4.1 Connection of Network Cable

There are 2 kinds of LAN cable, straight through cable and crossover cable. Connectors are JR-45 type and they are all looked the same. It won't damage the machine if you mis-use the cable. The connection figure below is for your reference.




Please do confirm

- The Link/Act LED of PC network card is ON or blinking.
- LNK/ACT LED of MDI or MDI-X port of MOSA 3700 Plus is ON or blinking.

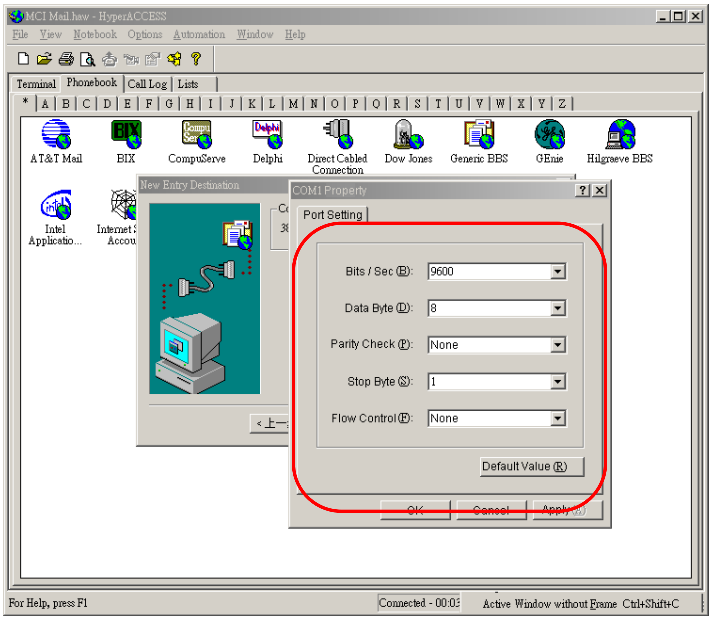
Otherwise, change port or LAN cable and retry it again

**Note: To connect PC is for the configuration of this product. When configuration is done, no PC is required to make or accept calls and all PCs can be shut down.**

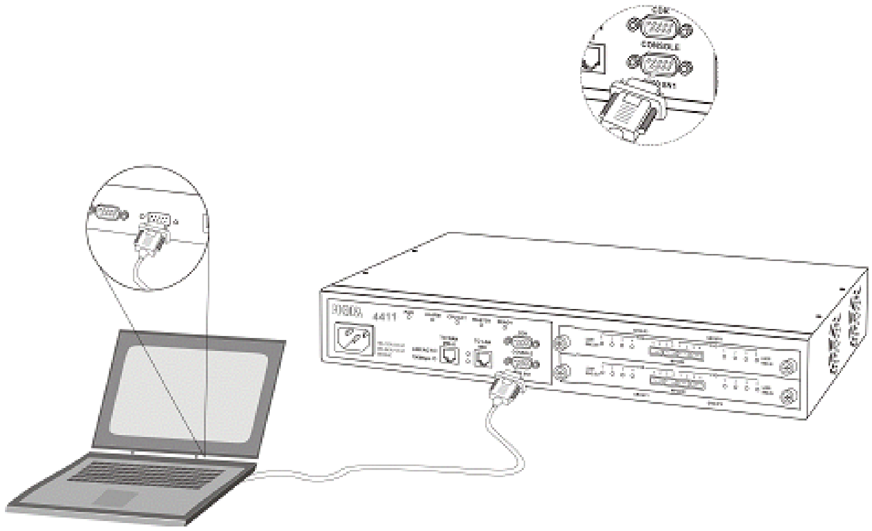
## 5.4.2 Connection of Console Port

To connect port, connect the PC with this machine via RS-232 Console cable, power on the PC and configure the PC parameters as following:

- Speed: 9600
- Data Bits: 8
- Parity Check: None
- Stop Bit: 1
- Flow Control: None




Console port is available for connecting to PC. It can configure some initial configuration. VODTEL had configured some initial value on this machine. You can configure this machine if this machine is connected to LAN network and Console Cable is not necessary



If Console cable is not available, run Telnet in PC for connection is OK.

In Windows system, Start --> Run --> Telnet 192.168.0.2

192.168.0.2 is the default IP of the machine; make sure that your PC is under the same subnet 192.168.0.X

## 5.5 Connection of 8/16 Ports Model

### 5.5.1 Installation of Modules

There are 3 available modules, MP3008+, MP3108+ and MP3208+。

Module can be installed on S1 or S2 bay freely (16 ports only) according to the needs of the structure. Loose the screw of bay cover and remove the cover, later, insert module into the bay and tighten the screw.

**Attention**: If the module is installed when you get it from VODTEL, don't change its position. When module type is changed (change position, add, remove), do Factory Reset is required and some setting needs to be configured again.

### 5.5.2 Numbering of Module

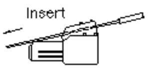
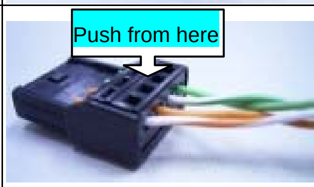
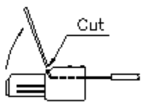
The port number is labeled on the front panel of the 19 inches rack model. For convenient management via Web management page and Console interface from remote side, the numbering is based by the port group; each group consists of four ports. The following table indicates the port number and the corresponding location:

| Model | Group | Location | Numbering for management | | | |
|---|---|---|---|---|---|---|
| 3716 | Group 1 | Lower module (S1), 4 ports of left side | 1 | 2 | 3 | 4 |
| | Group 2 | Lower module (S1), 4 ports of right side | 5 | 6 | 7 | 8 |
| | Group 3 | Upper module (S2), 4 ports of left side | 9 | 10 | 11 | 12 |
| | Group 4 | Upper module (S2), 4 ports of right side | 13 | 14 | 15 | 16 |
| 3708 | Group 1 | 4 ports of left side | 1 | 2 | 3 | 4 |
| | Group 2 | 4 ports of right side | 5 | 6 | 7 | 8 |

## 5.5.3 IDC Connectors (Only for 3708/3716)

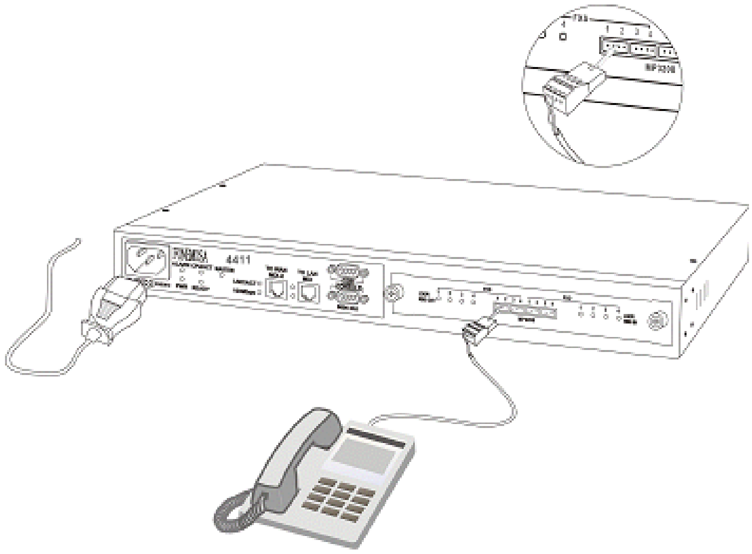
IDC connector is used for the voice interface (FXS and FXO) on the rack model. By IDC connector, PBX line and telephone wire can be easily connected to the VoIP gateway. No special tools are required; please follow the instruction to install:

(Remarks: For IDC connector, it's better to use No. 24 wire, e.g. CAT 5 and bind two wires for one port)

| | |
|---|---|
| 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 |   |

## 5.5.4 Connection between IDC Connector and Phone Set

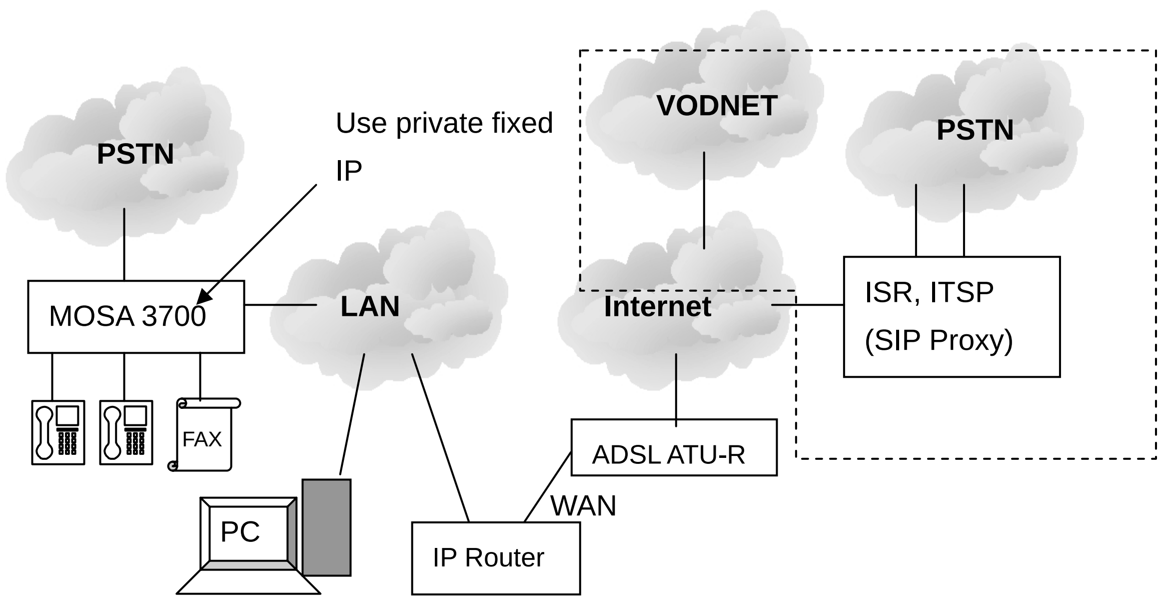
Voice interface (FXS and FXO) of this machine use IDC connector and it can be connected to extension phone set or PSTN.



# 6. Initial Setting of a Single Machine

## 6.1 Connection of Basic Structure

The figure below is the basic structure; please ignore the connection inside the dotted line. When your machine is configured correctly into Internet, these services exists (extra charged is required for some services)



## 6.2 Phone Set Configuration (Phone Set Programming Mode)

The system management has a special password (the default password is 0000). You can off-hook any extension phone set and dial "##0000" At this moment, the system management can dial the following item number for management.

## 6.3 Configuration of Telecom Region ID

The default Region ID of this machine is Taiwan. The purpose to configure Region ID is to adapt the PSTN specification of dial tone, busy tone, country code and area code for each country.

If this machine is not installed in Taiwan, change Region ID is required. The example below shows you how to change it to PRC.

◆ **Step 1**

1. Dial “##0000" and hear Du Du Du tone.
2. Then dial 95 07 # and hear Du Du Du. (95: parameter, 07: PRC Region ID. For HK, dial 95 15. Refer the table below).
3. Dial 97 1# and hear Du Du Du tone. (97: parameter, 1 factory reset all)
4. Hook on, the machine restart automatically, and please wait until the CPU/ACT LED is ON.

Region ID Table

| Country | Region ID | Country | Region ID | Country | Region ID |
|---|---|---|---|---|---|
| Argentina | 01 | France | 12 | Singapore | 36 |
| Australia | 02 | Germany | 13 | Slovenia | 38 |
| Philippines | 03 | Hong Kong | 15 | South Africa | 39 |
| Portugal | 04 | India | 18 | Spain | 40 |
| Brazil | 05 | Italy | 22 | Switzerland | 42 |
| Canada | 06 | Japan | 23 | Taiwan | 43 |
| China | 07 | Korea | 24 | Thailand | 44 |
| Russia | 08 | Malaysia | 26 | British | 46 |
| Sweden | 09 | Mexico | 27 | USA | 47 |
| Vietnam | 10 | Netherlands | 28 | | |
| Belgium | 11 | New Zealand | 29 | | |

## 6.4 Configuration of IP Address

This Chapter tells you how to configure the **IP Address** of this machine
Before the configuration, apply an IP from system administrator. It can be a fixed Public IP or fixed private IP.

Assume the IP address you get is listed below. This IP address has to be the same as the Subnet of Router and it does not conflict with the IP address dispatched by DHCP.

IP Address: 192.168.1.11
Subnet Mask: 255.255.255.0
Default Gateway: 192.168.1.254

◆ **Step 2**

1. Dial ##0000 and hear Du Du Du tone.
2. 01 0# Du Du Du (01: parameter, 0: Fixed IP)
3. 02 192*168*1*11 # Du Du Du (02: parameter，later IP Address)
4. 03 255*255*255*0 # Du Du Du (03: parameter，later Subnet Mask)
5. 04 192*168*1*254 # Du Du Du (04: parameter，later Default Gateway)
6. Hold phone set.

Note: If you would like to use default IP address of this machine, they are IP Address: 192.168.0.2, Subnet Mask: 255.255.255.0. Please adjust the Subnet setting of PC to connect this PC.

## 6.5 Restart

To take effect those setting above, restart this machine is required.

◆ **Step 3**

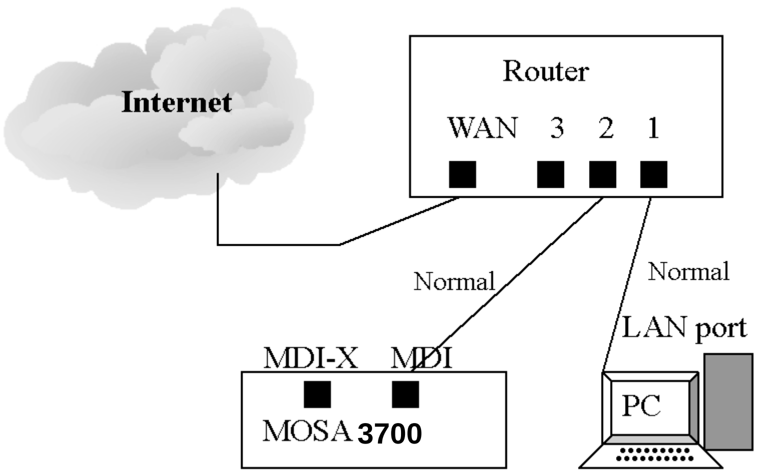
1. Continue from last section (still in phone set programming mode)
2. 98 1 # Du Du Du (98: parameter, 1: Warm-restart type)
3. Hook on phone set.

Wait a moment until the LED of CPU/ACT is ON, then the machine is ready.

## 6.6 Configuration of Router

Router is connection between LAN and Internet. It may also have some other function, such as Firewall, DHCP Server...
DHCP Server can dispatch IP Address to the PC and device in LAN environment. In this example, we assume that this router had activated function of DHCP Server.

Assume this machine is installed behind Router and connect it with others by straight through cable or crossover cable to Internet

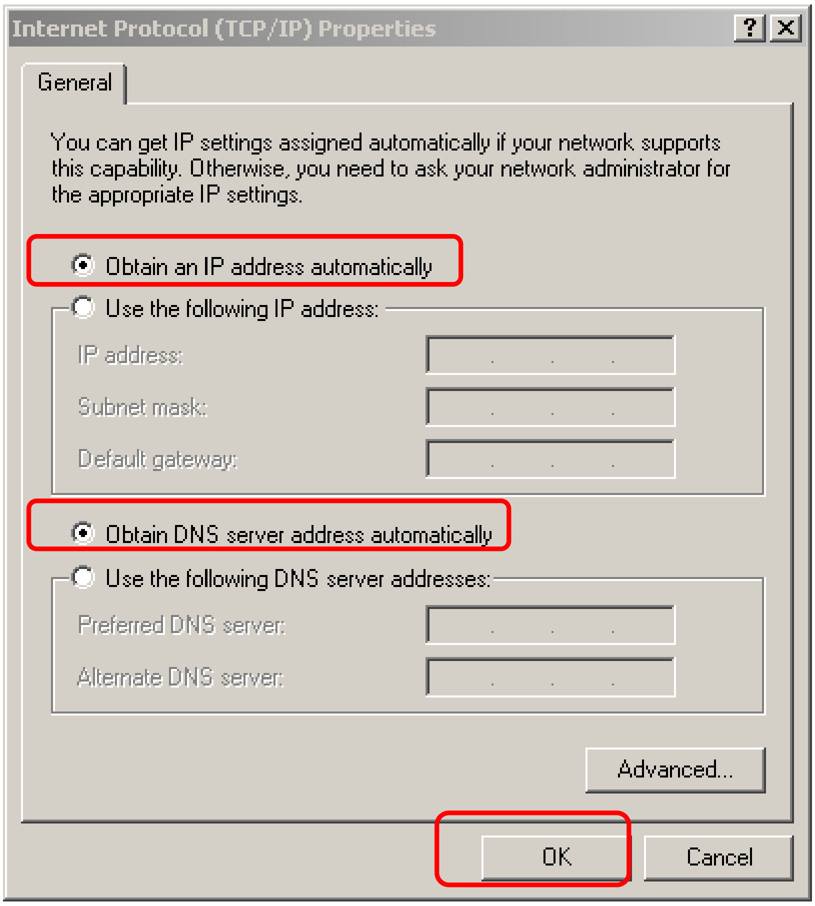



Then SIP device (SIP Phone, Softphone) user in Internet can communicate with desktop phone set from MOSA 3700. Desktop phone set user of MOSA 3700 also can make call to SIP device (SIP Phone, Softphone) user in Internet.

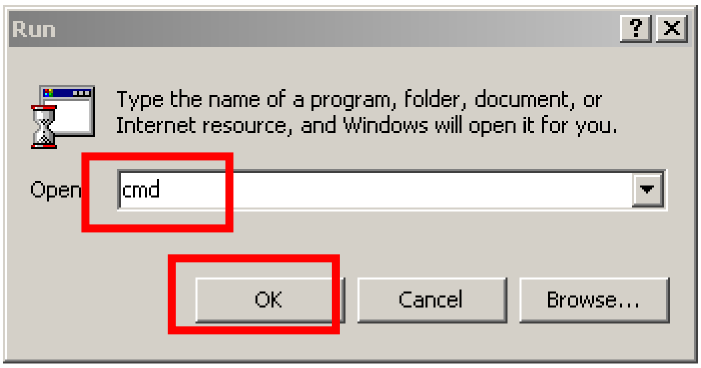
### 6.6.1 Configure PC to connect router

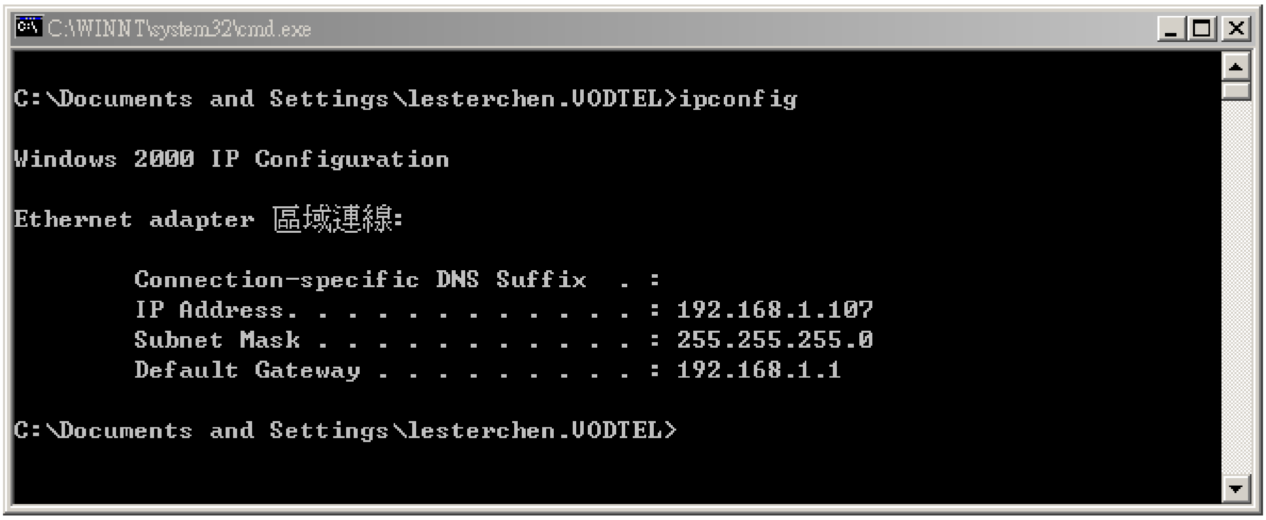
Before this step, make sure you had login into PC with administrator permission and the PC is connected to network

For the example of Window 2000, set the PC card to accept IP from DHCP.
For the example of Windows 2000, Set LAN card mode of PC to DHCP：Click of **Properties** of **Internet Protocol** (TCP/IP) (Start→Setting→Network and Dial-Up Connection→Right click "Local Area Connection"→Select "Properties" →Click Internet Protocol(TCP/IP) →Click "Properties"), Select "Obtain an IP address automatically" and " Obtain DNS Server address automatically ", Click "OK"



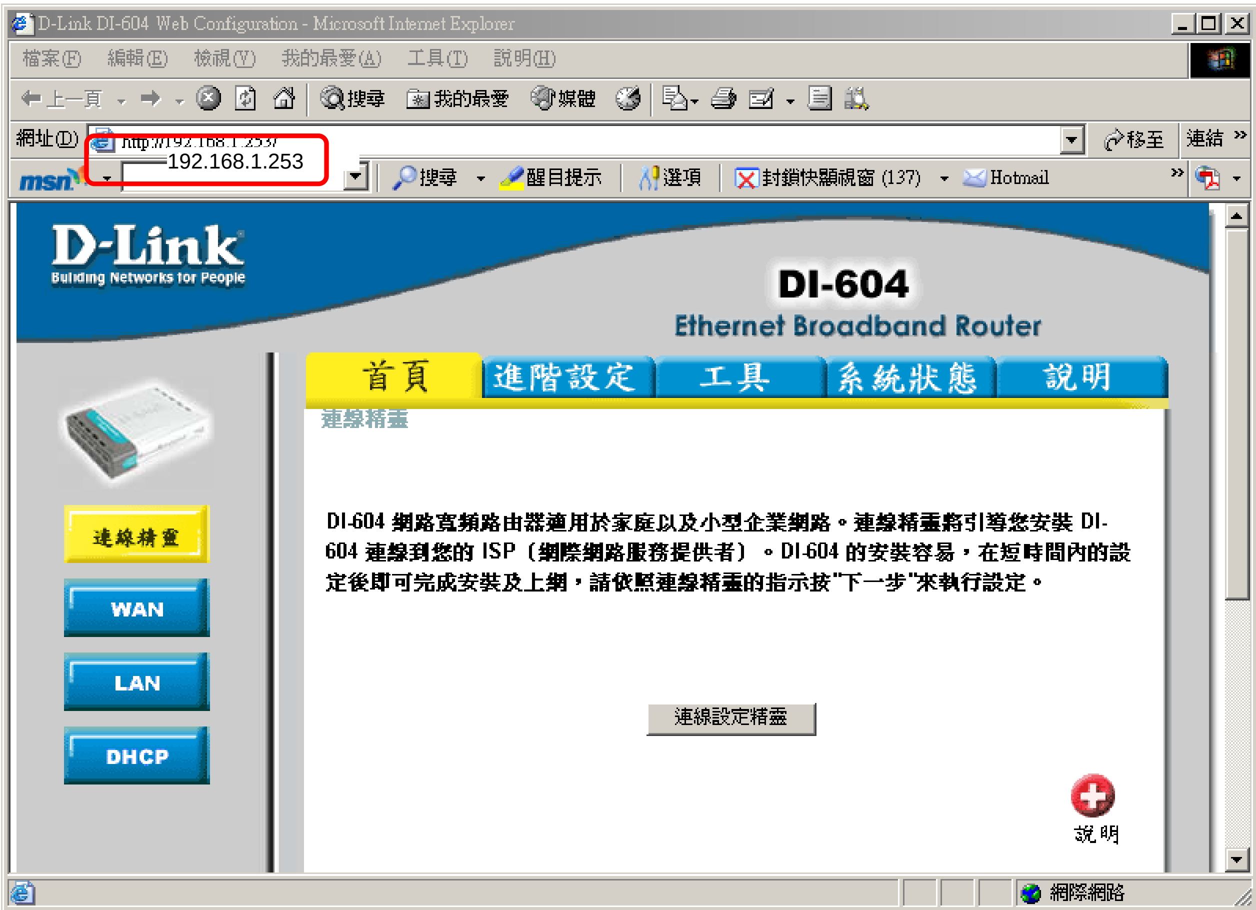

Confirm that PC had got the IP address from Router: Enter the Command Mode of PC (Start→Run→cmd) and input "ipconfig" and then press "Enter" key to know that you had got the IP or not (The IP you had got should not the same as this example)





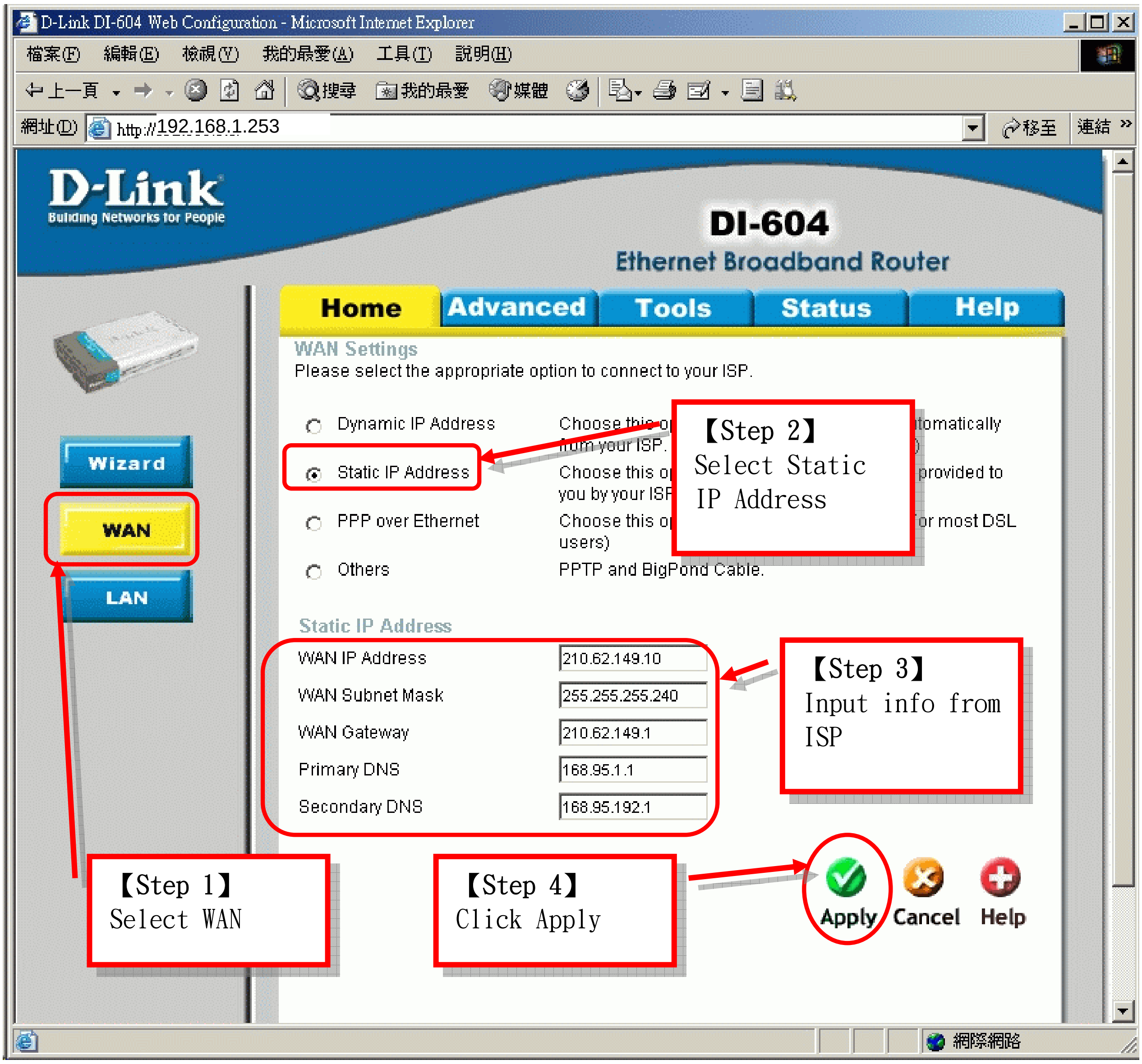

## 6.6.2 Configure Router to connect Internet

Enter management web of Router: Start browser and input address such as "http://192.168.1.253", and then press "Enter" key into the management web. (Assumes that the IP address of Router is 192.168.1.253, for the example of D-Link DI-604)



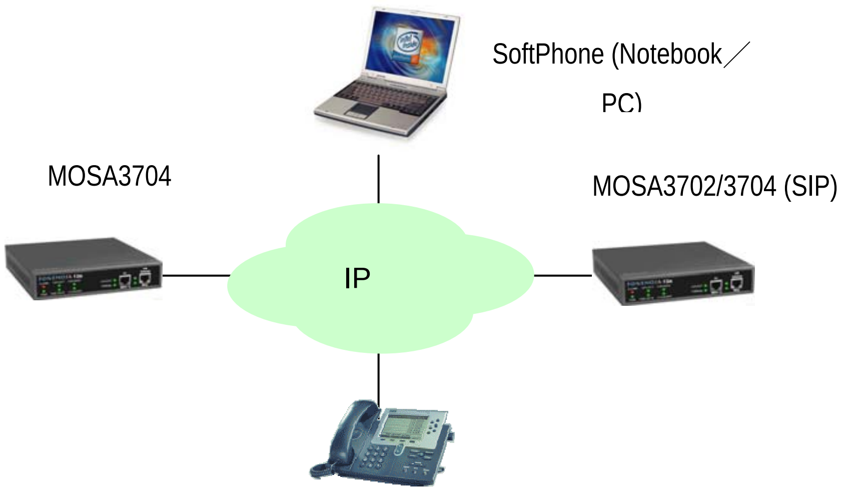

Then click WAN and configure related information according to the graph below

The information of **WAN IP Address, WAN Subnet Mask, WAN Gateway (Default Gateway)** can be got from ISP you apply.



# 7. SIP Configuration

MOSA 3700 not only can make regular PSTN calls, it also can communicate with IP Phones or Soft-Phones by using SIP protocol. This section shows you what parameters you need to configure for SIP calls and how to make the SIP calls.

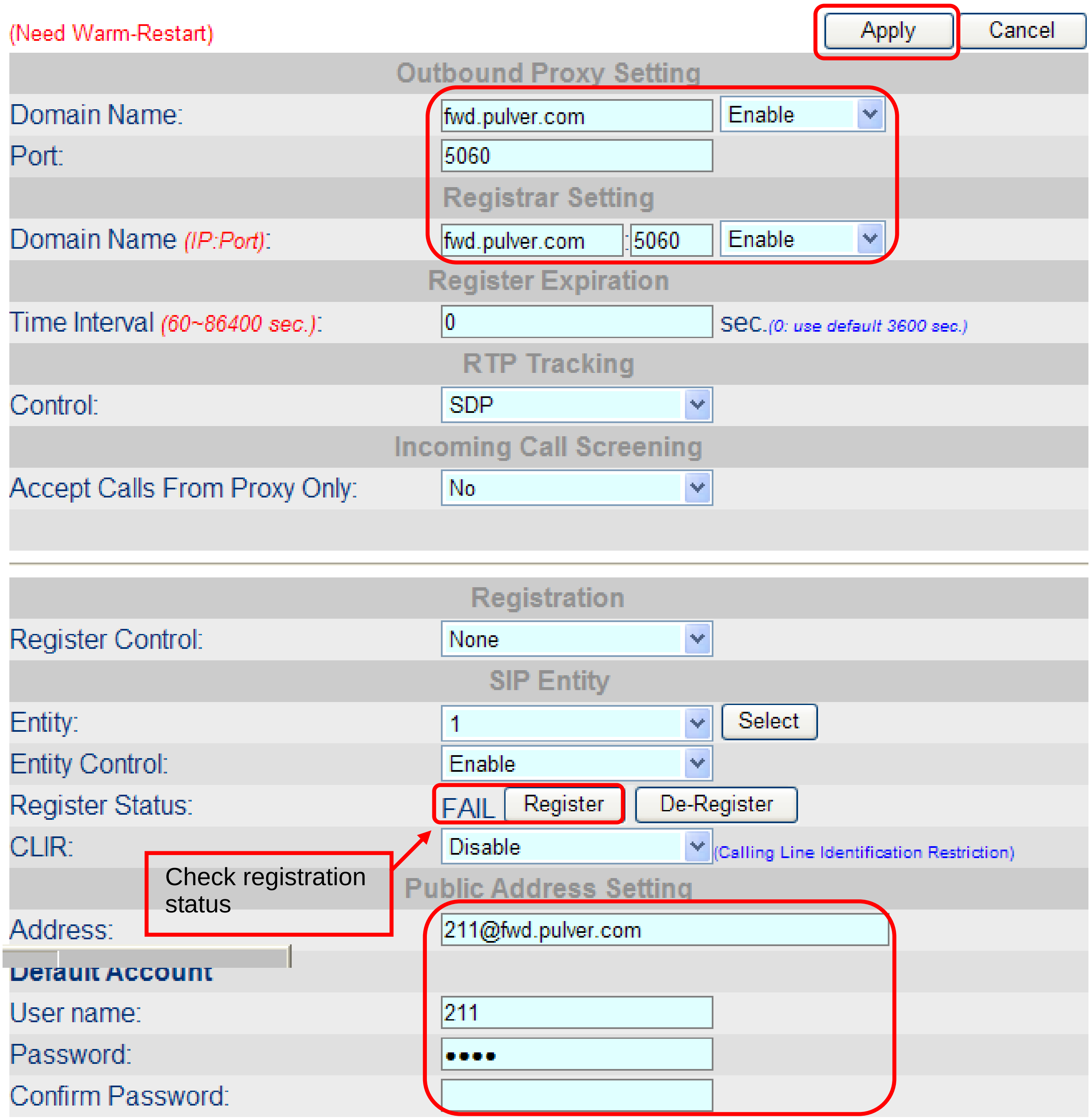



Cisco IP Phone

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.

## 7.1 Register to SIP Telephony Server Provider

Assume that the registration information of ITSP are
SIP Outbound Proxy: fwd.pulver.com  Port Number: 5060
Registrar: fwd.pulver.com  Port Number: 5060
The number you get is 211. So the Public Address of SIP Phone Number is "211@fwd.pulver.com"
Password is 1234
Input the information above to
Web Path：1.SIP Environment\1.1.Proxy/Trunk Mapping


(Need Warm-Restart)
Apply
Cancel
Outbound Proxy Setting
Domain Name:
fwd.pulver.com
Enable
Port:
5060
Registrar Setting
Domain Name (IP:Port):
fwd.pulver.com
5060
Enable
Register Expiration
Time Interval (60~86400 sec.):
0
sec.(0: use default 3600 sec.)
RTP Tracking
Control:
SDP
Incoming Call Screening
Accept Calls From Proxy Only:
No
Registration
Register Control:
None
SIP Entity
Entity:
1
Select
Entity Control:
Enable
Register Status:
FAIL
Register
De-Register
CLIR:
Disable
(Calling Line Identification Restriction)
Check registration status
Public Address Setting
Address:
211@fwd.pulver.com
Default Account
User name:
211
Password:
••••
Confirm Password:

## 7.2 Channels and SIP entity

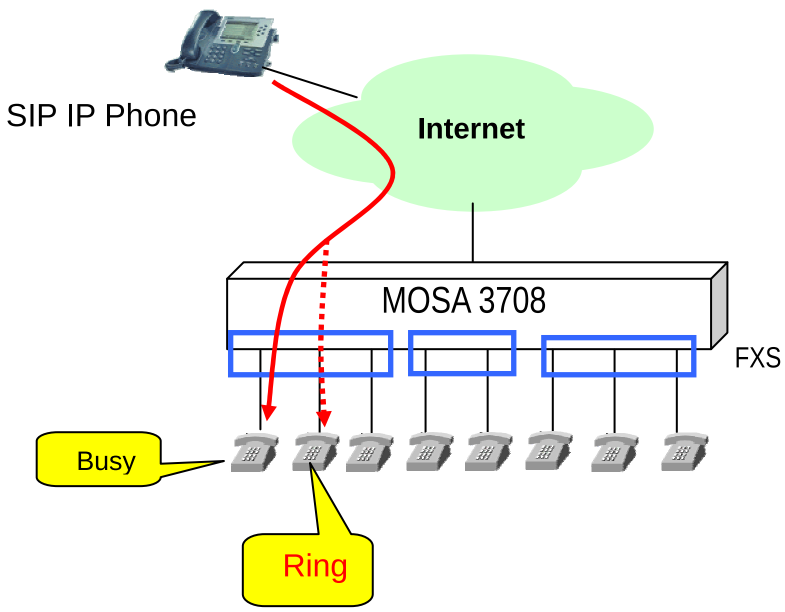
Many Channels can be assigned as one SIP Entity. Single Channel also can be assign as one SIP Entity.

SIP service provider will assign one or more SIP accounts (Entity) 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@vodtel.com.

Application example:

As the figure below, Channel 1-3 belongs to SIP Entity 1: 1001@vodtel.com. Channel 4 and Channel 5 belongs to SIP Entity 2: 1002@vodtel.com. , and Channel 6-8 belongs to SIP Entity 3: 1003@vodtel.com. When other device under SIP network dial into 1001@vodtel.com, the phone connect to Channel 1 is ringing. If Channel 1 is under conversation (busy), the line will be switched to Channel 2, and so on. So Channel 1~3 become a simple Hunting Group. (This feature needs the support of SIP Proxy Server).

Figure:



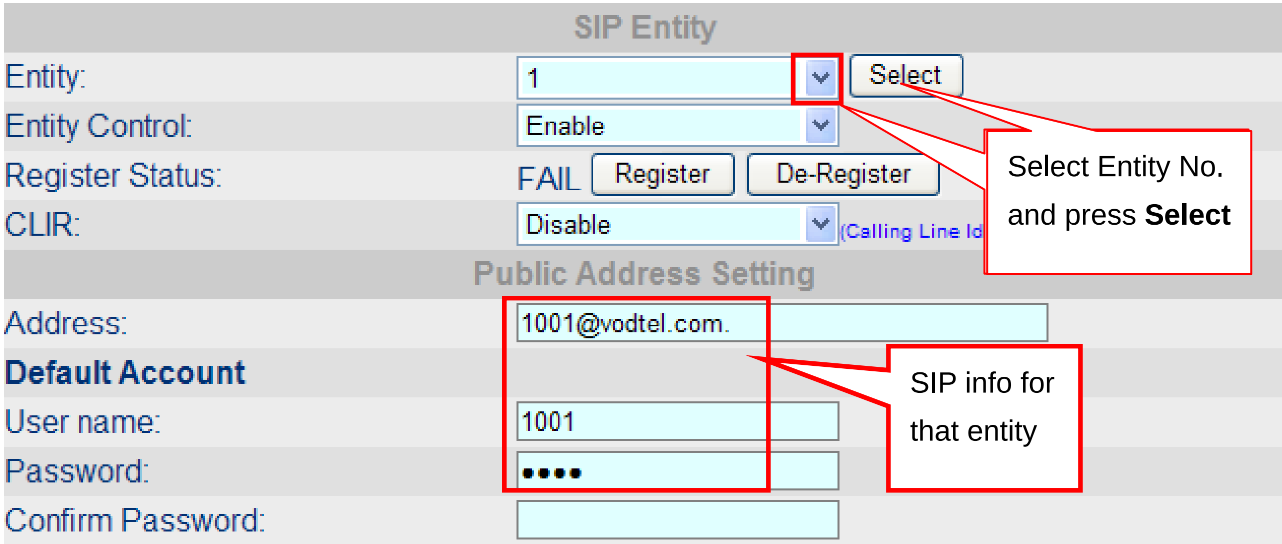

### 7.2.1 Create Entity

At previous section, you can select different entity and register with different entity name if ITSP provides

**Attention**: All entities have to belong to the same SIP Proxy. Such as the example above

1001@vodtel.com. and 1002@vodtel.com. Entity of different SIP Proxy (such as vodtel.com and fwd.pulver.com) is not allowed in one machine box.

WEB page: 1.SIP Environment\1.1.Proxy/Trunk Mapping



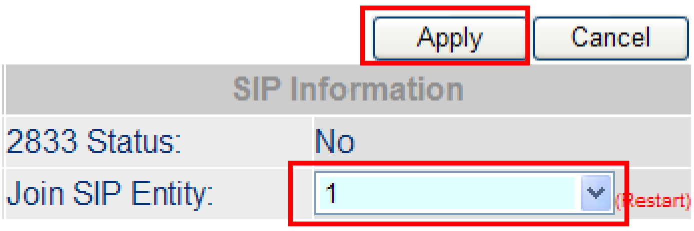

Click Apply finally.

### 7.2.2 Assign Channel to Entity

Each channel must belong to a SIP entity before it can make or receive call.
When Entity information is created, please assign channel(s) to join entity.
Select the Channel with FXS Type that will joins the entity and click its St (Status)
WEB page: 2.Channel Config.\2.1.Summary

| Analog Channel | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| Ch | St | Type | Entity | Reg. | 2833 Status | DND | T.38 | Statistics In/Out | VAD | Gain In/Out |
| 1 | | FXS | 16 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 2 | | FXS | 2 | - | - | - | - | 0/0 | ✓ | 0/0 |

Assign an Entity to that channel and then click **Apply**

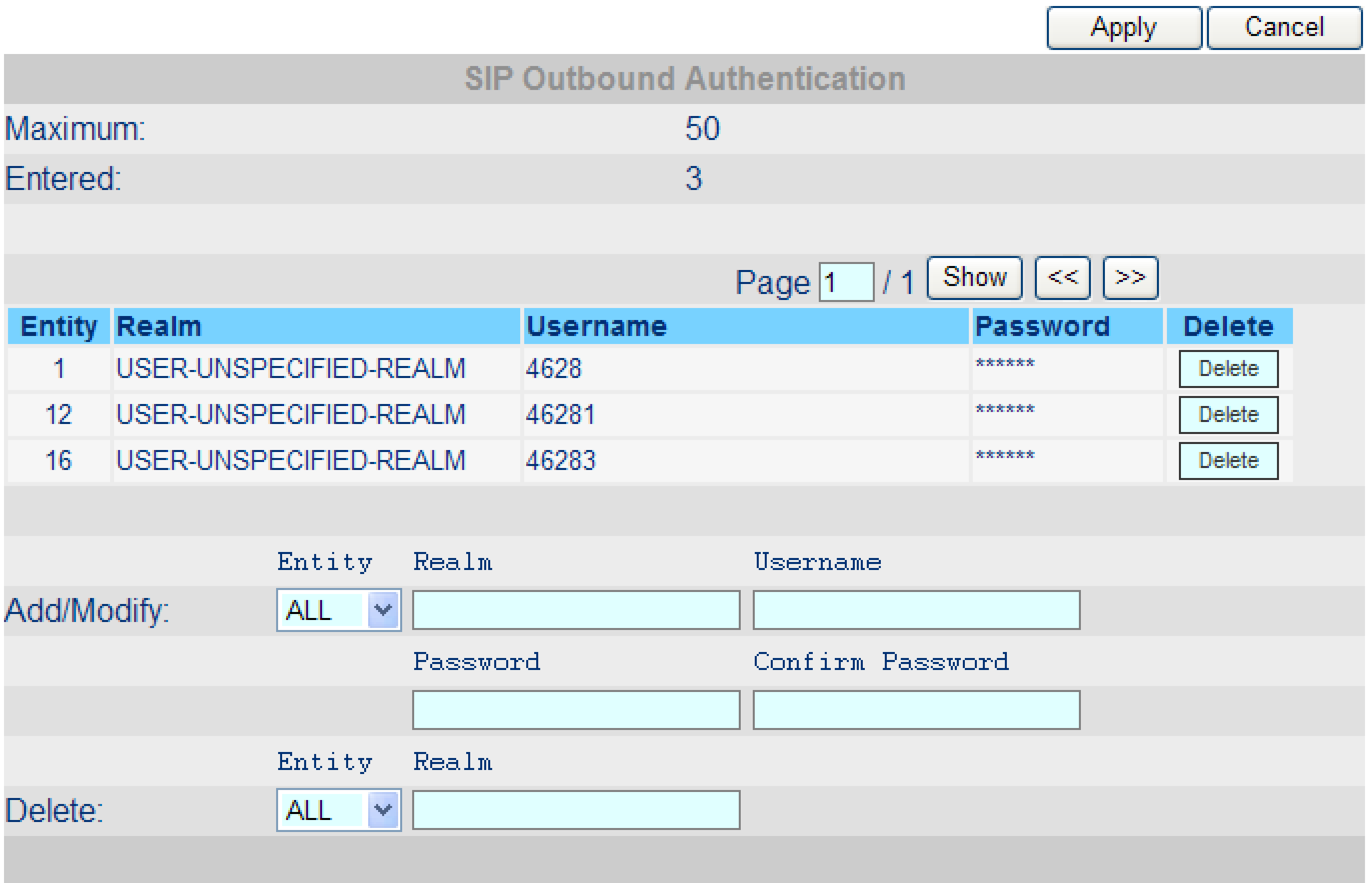


## 7.3 SIP Outbound Authentication

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.

"Realm" is a kind of verification for SIP Outbound Authentication. If SIP service provider does not provides this information. The gateway will create a default Realm (by string USER-UNSPECIFIED-REALM) automatically with your Username and Password mentioned on last section for SIP Outbound Authentication. If there are more than one SIP entity is created on this gateway. The gateway creates Realm for each entity. The default Realm helps you to register the SIP server successfully.

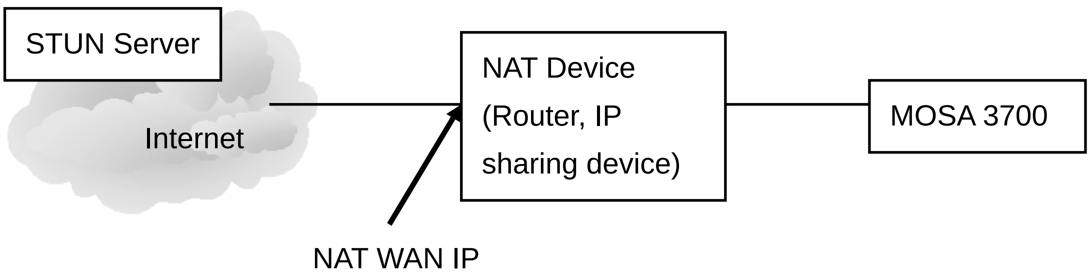
Configuration
WEB Page: 3.SIP Advanced\3.2.Outbound Authen.



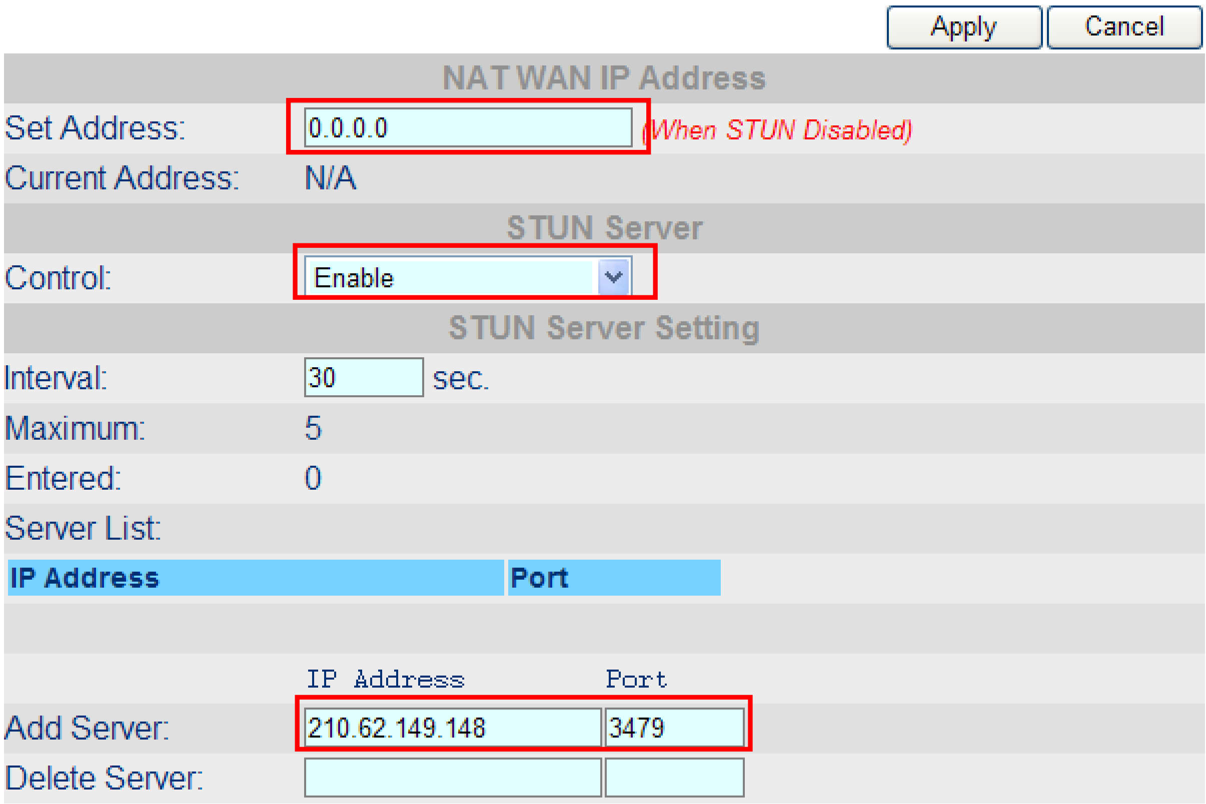
## 7.4 Configure STUN for Client under NAT (Optional)

STUN is an application-layer protocol that can determine the public IP Address of a NAT device that sits between the STUN client (MOSA 3700) and STUN server.
**Note**: MOSA 3700 use Media Relay technology to penetrate NAT. So configure STUN might not be required. Keep default value for management page of STUN if STUN is not required.

1. If your gateway is behind NAT (Use Private IP), please consult the SIP service provider to provide information of STUN server and also configure the parameter here, otherwise you need to input NAT WAN IP to penetrate NAT device. After configuring the parameters of STUN, please act Warm-Restart.
2. If no useable free STUN Server available, for most of ITSP (Internet Telephony Service Provider), their Outbound Proxy Server supports Media Relay, device under NAT can penetrate NAT without configuration.
3. STUN does not support Symmetric NAT.




**Configuration**
WEB Page: 3.SIP Advanced\3.4.WAN IP & STUN

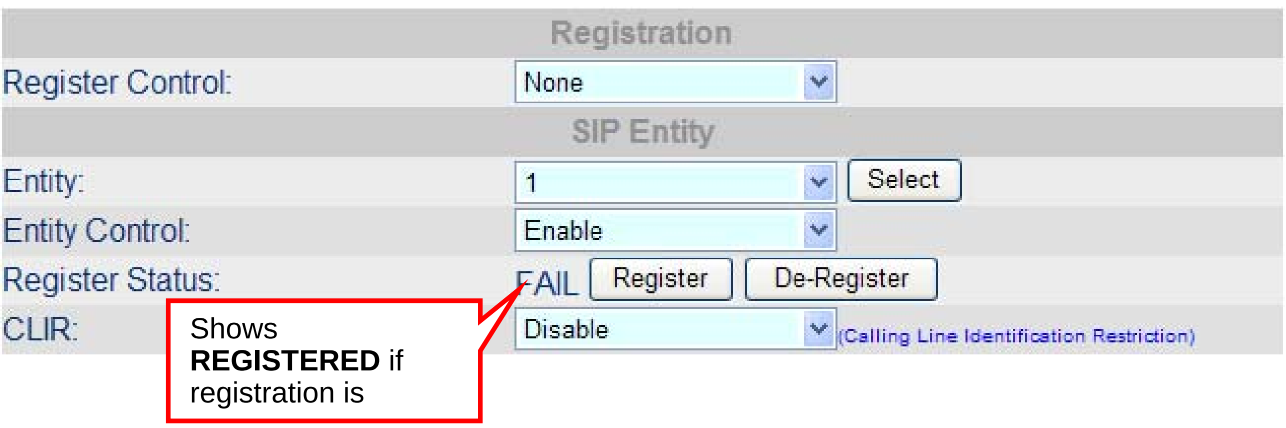



You can enable and disable the service on WEB page.

## 7.5 Check SIP Entity Registration Status

You can use the WEB page to check if the SIP entity is registered successful or not.

WEB Page: 1.SIP Environment\1.1.Proxy/Trunk Mapping



| Group | Field | Description | Default Value |
|---|---|---|---|
| Registration | Register Control | ◆ None: This machine does not register to SIP Proxy spontaneously. You can register each entity manually by the button below.<br>◆ Register All: All entities of this machine register to SIP Proxy spontaneously.<br>◆ De-Register All: All entities of this machine are forced to De-Register. | None |
| SIP Entity | Entity | Select the Entity you want to operate | |
| | Entity Control | Enable: The entity you select is enabled<br>Disable: The entity you select is disabled | |
| | Register Status | Shows the registration status<br>◆ Registered：Registration is successful<br>◆ Registering：Trying to register<br>◆ Fail：Registration is failed<br>◆ Idle：Means SIP trunk is disabled<br>Register (button)：Click to do manual registration<br>De-Register (button)：Click to quit registration manually. | |

# 7.6 Phone Book

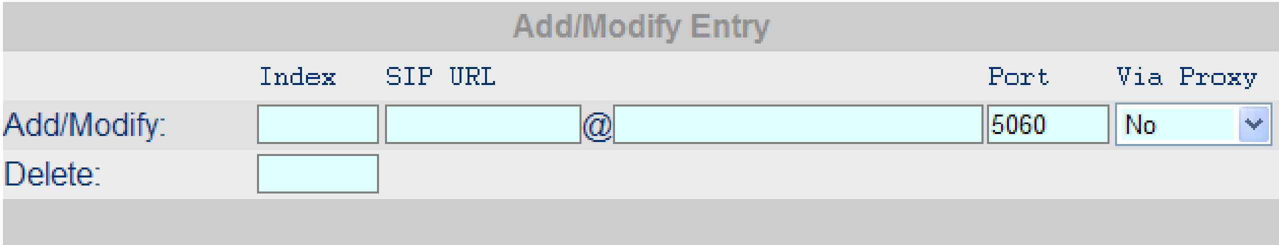
## 7.6.1 General Phone Book

Since the SIP phone number is not easy for regular phone to dial, MOSA 3700 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. User also can configure this index number to build the route by SIP Proxy or build the route without Proxy if destination gateway use fixed IP (Public IP or private IP in VPN)

For instance if the phone book is configure as below:

| Index | SIP URL | Port | Via Proxy | |
|---|---|---|---|---|
| 100 | 01@61.220.145.70 | 5060 | No | <-- GW1 |
| 200 | 73797@fwd.pulver.com | 5060 | Yes | <-- GW2 |
| 201 | 73797@61.222.217.5 | 5060 | No | <-- GW2 |

WEB Page: 3.SIP Advanced\3.3.SIP Phone Book




Notice: If your SIP account is digit type like 234@SIP.vodtel.com or 456@SIP.vodtel.com, and this MOSA 3700 register to SIP proxy: SIP.vodtel.com, you don’t need to configure the items.

## 7.7 Make SIP Calls

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

You can select the ways below to make SIP call:

**Standard Call**: Dial <numbers>+<#>.

1. Compare dialing plan (refer to 8.2 Configuration of Dialing Plan), check to see if the prefix number you dial is matched, such as example 050.
2. If the number in dialing plan is configured and matched, send the call to proxy. If the prefix number does not match dialing plan or the registration to the proxy is fail, then the call will be sent to PSTN.
3. If the prefix number is not in dialing plan, the call will be sent to PSTN.

**Example**: 050 is configured is Dialing Plan table
FXS channel user dial 0501234567, then call is sent to SIP proxy
FXS channel user dial 0968223371, then call is sent to PSTN
FXS channel user dial 0501234567, but registration to SIP is failed, then the call is sent to PSTN
※**Note**: There is a default "x" value in dialing plan table, means any digits that user dial are sent to SIP proxy.

**Phone Book Call:** Dial <#>+ <index>+<#>.

1. Compare SIP Phone books (refer to 7.6 Phone Book); check the number to see if it is in phone book.
2. If the number is configured in Phone Book 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 the index number you had configured to use **Via Proxy** but it communicates with proxy failed, you will hear 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

**Hotline Call**:

If the channel is configured to use Hotline function (refer to 8.1 Hotline Function), any dialing above is disabled. If the channel is hotlined to other SIP device, no dialing is needs after user picks up handset. Other SIP device rings immediately.

**Hotline Call to MOSA 4600 Plus/4600B/4600D** :

Dial <SIP extension number> or

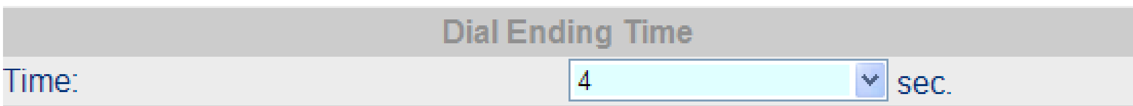
#<VODNET number> + # or

<Prefix number (configured in MOSA 4600 SIP Line)>

1. If you dial SIP extension number, other SIP device that register to MOSA 4600 SIP Line with that SIP extension number will ring.
2. If you dial # + VODNET number + #, the call is relay to the VODNET IP-PBX network.
3. If you dial Prefix number, the call is relay to the VODNET IP-PBX network according to the Prefix Map (also called extension table) specified in MOSA 4600 SIP Line.

**Notice:** If you do not want to dial "#" after numbers, please configure the 'Dial Ending Time' item. After the seconds configured here, the call will be sent automatically.

WEB Page: 5.PBX Advanced




## 7.8 Make Inbound Transit Call

To make an inbound transit call from PSTN to SIP, you have to enable Auto Answer function of this gateway

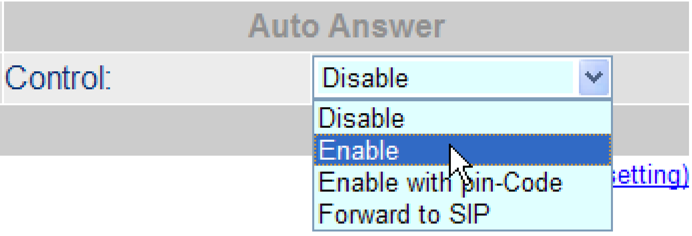
Please enable Auto Answer configuration at here

Select the Channel with FXO Type that will receive inbound transit call and click its St (Status)

WEB page: 2.Channel Config.\2.1.Summary

| Ch | St | Type | Entity | Reg. | 2833 Status | Auto Answer | T.38 | Statistics In/Out | VAD | Gain In/Out |
|---|---|---|---|---|---|---|---|---|---|---|
| 13 | | FXO | 13 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 14 | | FXO | 14 | - | - | - | - | 0/0 | ✓ | 0/0 |

Select Auto Answer Mode you want




If you don't enable the Auto Answer configuration, the inbound call from PSTN will be assigned to a free FXS port of this gateway directly. It makes Inbound Transit Call impossible.

When Auto Answer function is enabled, the gateway will answer the call and calling side will hear the second dial tone. For the Auto Answer function, it is also divided into **Enable** and **Enable w/ Pincode** options. The configuration page is the same as above.

**Dial Inbound Transit Call when Auto Answer is configured as Enable**

Please dial the number below after the second dial tone:

1. SIP Number + '#',  Example: 73797#  or
2. '#' + Index Number + '#',  Example: #123#

If you still need to make a call to the FXS port of this gateway, please press "*" to seize a free FXS port.

**Dial Inbound Transit Call when Auto Answer is configured as Enable w/ PIN code**

This Auto Answer mode provides security control for the Inbound Transit call

Please dial the number below after the second dial tone:

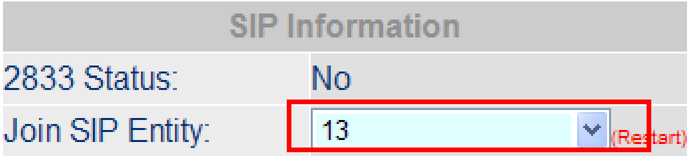
1. PIN code + '#'+ SIP Number + '#',  Example: 7742#73797#  or
2. PIN code + '#'+ '#' + Index Number + '#',  Example: 7742##123#

If you still need to make a call to the FXS port of this gateway, please press "*" to seize a free FXS port.

**Forward Inbound Call to other SIP Number**

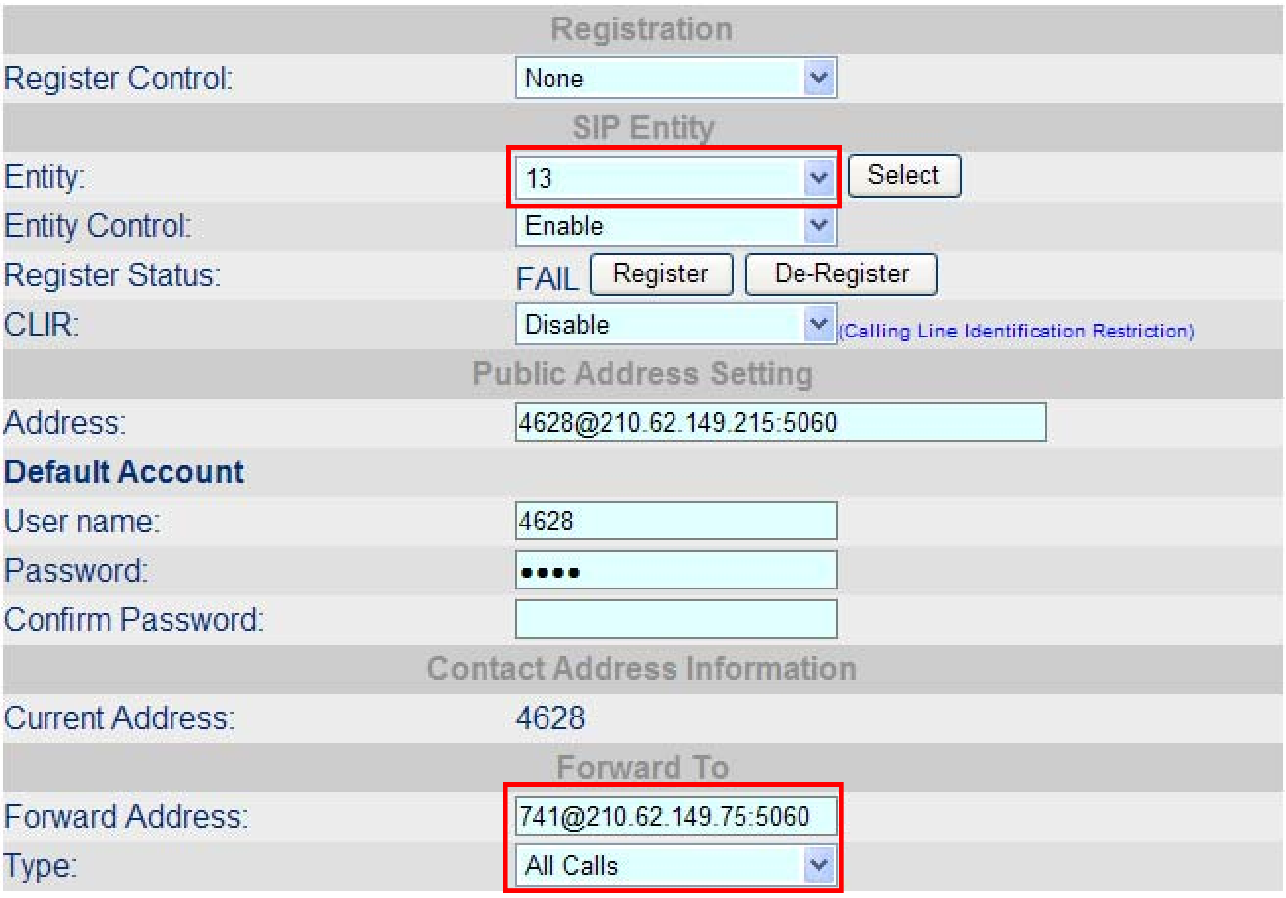
Select **Forward to SIP** at previous configuration

At the same page, watch which entity this FXO joins




And then configure its Forward To address number

Web Page: 1.SIP Environment\1.1.Proxy/Trunk Mapping



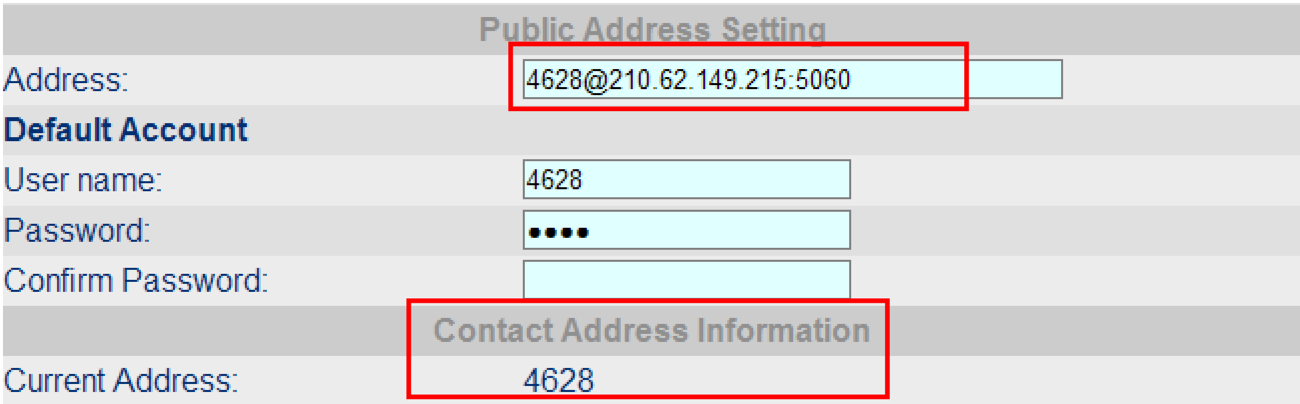

**Notice for the Inbound Transit Call**

1. If the SIP number that user dial does not match any prefix code configured in Dialing Plan page, the call is disconnected.
2. If the PIN Code does not match any passwords configured in Password For Inbound Transit page, the call is terminated.
3. If the Index Number does not match any pre-configured Phonebook Index in Phone Book page, the Index Number will be regarded as SIP number and create an IP call without applying any match rule configured in Dialing Plan.

For which free FXS port that this gateway will seize, please refer to 8.6 Non-SIP Call port seizure preference
The PIN code (Password for Inbound Transit) is configured at chapter 10.3.4 Transit in setting
The Dialing Plan is configured at chapter 8.2.1.1 Dialing Plan
The Index Number is configured at chapter 7.6 Phone Book

## 7.9 Make SIP IP Call without SIP Proxy

The main purpose of Contact Address is making SIP calls without proxy.
The Contact Address is the same as the "Username" of Public Address if that field is configured. For S/W version above 1.05, the value is read only. Generally speaking, "Username" of Default Account are digits and it is regarded as SIP number.
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 to connect other SIP gateway. The typical example is: 4628@210.62.149.215. Other SIP gateway that had already configured 4628@210.62.149.215 in Phone Book can connect this gateway by number 4628 without routing through SIP Proxy.
Notice: For this type of SIP calls, the destination device's IP address is already known and it is fixed.

# 8. Advanced Parameters

## 8.1 Hotline Function

A new Hotline function is added for MOSA 3700 Firmware Version 1.07 or above
When hotline function is enabled, the FXS channel is connected to specified SIP device or MOSA 4600Plus/4600B/4600D (if the MOSA 3700 is configured and register to MOSA series products as a

client) automatically when user of MOSA 3700 FXS channel picks up handset.

- If the FXS channel is Hotlined to other SIP device (SIP Phone, Softphone), other SIP device rings immediately when FXS channel user of MOSA 3700 picks up hand-set.
- If the FXS channel is Hotlined to MOSA 4600Plus/4600B/4600D, (skip this section if the MOSA 3700 don't register to MOSA series of products) FXS channel user of MOSA 3700 hear dialing tone from MOSA series of products when pick up hand-set, and then he/she can dial extension number to other SIP device or dial VODNET number or Outbound Call to PSTN via VODTEL IP-PBX environment.

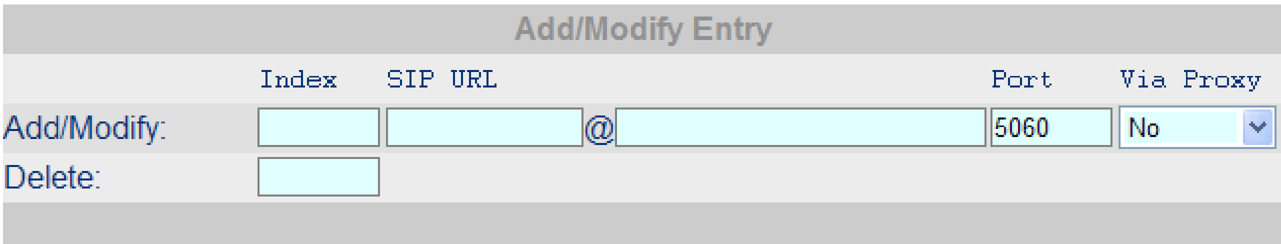
**Configuration of Hotline**

- Enable Hotline function

WEB page: 3.SIP Advanced\3.3.SIP Phone Book


- Setup index number

WEB page: 3.SIP Advanced\3.3.SIP Phone Book




When Hotline function is enabled, user also needs to specify which channels (FXS only) should join Hotline function and which SIP number (Public Address) the channel is hotlined to.

**Hotline mapping table**

| Channel (FXS) only | Index Number | Description |
|---|---|---|
| 1st FXS channel | 1 | Index number “1” maps the 1st FXS channel |
| 2nd FXS channel | 2 | Index number “2” maps the 2nd FXS channel |
| …. | …. | … |
| 16th FXS channel | 16 | Index number “16” maps the 16th FXS channel |

**Available Hotline index number**

| **Model** | **Available Hotline Index Number** | **Note** |
|---|---|---|
| MOSA 3702A | 1 | |
| MOSA 3702B | 1, 2 | |
| MOSA 3704A | 1, 2 | |
| MOSA 3704B | 1, 2, 3, 4 | |
| MOSA 3704C | None | No FXS channel is available |
| MOSA 3704D | 1, 2, 3 | |
| MOSA 3708 | Depends on module used. Please refer to table below. | Only FXS channel can be counted as index number |
| MOSA 3716 | Depends on module used. Please refer to table below. | Only FXS channel can be counted as index number |

MOSA 3708/MOSA 3716 channel mapping number

| Model | Group | Location | Channel Number (Please select FXS port only) | | | |
|---|---|---|---|---|---|---|
| 3716 | Group 1 | Lower module (S1), 4 ports of left side | 1 | 2 | 3 | 4 |
| | Group 2 | Lower module (S1), 4 ports of right side | 5 | 6 | 7 | 8 |
| | Group 3 | Upper module (S2), 4 ports of left side | 9 | 10 | 11 | 12 |
| | Group 4 | Upper module (S2), 4 ports of right side | 13 | 14 | 15 | 16 |
| 3708 | Group 1 | 4 ports from left | 1 | 2 | 3 | 4 |
| | Group 2 | 4 ports from right | 5 | 6 | 7 | 8 |

Any index number that is not listed in **Available Hotline Index Number** above is recognized as normal index number and they are not used as hotline function and not all of the channels have to join hotline function. Please see the example below

Example Model: MOSA 3704B

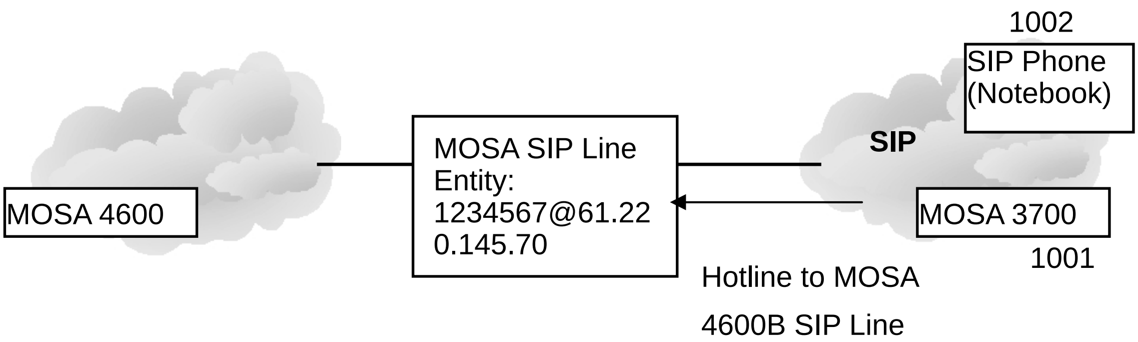
| Index | Public Address | Port | Via Proxy | Description |
|---|---|---|---|---|
| 1 | 01@61.220.145.70 | 5060 | No | Channel 1 Hotline to 01@61.220.145.70 without proxy |
| 2 | 73797@fwd.pulver.com | 5060 | Yes | Channel 2 Hotline to 73797@fwd.pulver.com by proxy, |

| | | | | |
|---|---|---|---|---|
| 100 | jack@fwd.pulver.com | 5060 | Yes | No hotline function for channel 3, 4 to dial |
| 200 | mike@fwd.pulver.com | 5060 | Yes | |
| 300 | Jason@fwd.pulver.com | 5060 | Yes | |

User of 1st FXS channel picks up hand set, and then 01@61.220.145.70 rings immediately
User of 2nd FXS channel picks up hand set, and then 73797@fwd.pulver.com rings immediately

**Hotline to MOSA 4600 Plus/4600B/4600D (skip this section if you don't register to MOSA 4600 SIP Line)**

Assume the Public Address of MOSA series product is 1234567@61.220.145.70 and it has extension number 1001 to 1002.




So we configure the Phone Book as below

| Index | Public Address | Port | Via Proxy | Description |
|---|---|---|---|---|
| 1 | 1234567@61.220.145.70 | 5060 | Yes | Channel Hotline to 1234567@61.220.145.70 MOSA SIP Line directly |
| 2 | 1234567@61.220.145.70 | 5060 | Yes | Channel Hotline to 1234567@61.220.145.70 MOSA SIP Line directly |

MOSA 3700 User hears dial tone from MOSA series products when pick up handset and then dial extension no. For example 1002, to other SIP device

## 8.2 Configuration of Dialing Plan

Dialing Plan controls the dialing number behavior of users

### 8.2.1 Dialing Method

According to different configuration, user needs to select different dialing method.

(1) Dialing Plan: When the first (few) digits that user dials matches Dialing Plan (described in

next section 8.2.1.1 Dialing Plan), number is send to SIP Proxy and build call route to SIP device, otherwise, make call route via local FXO port.

(2) Transparent: All numbers user dials are sent to SIP proxy server and all number that controls MOSA 3700 is disabled, including the end code # of each dialing.

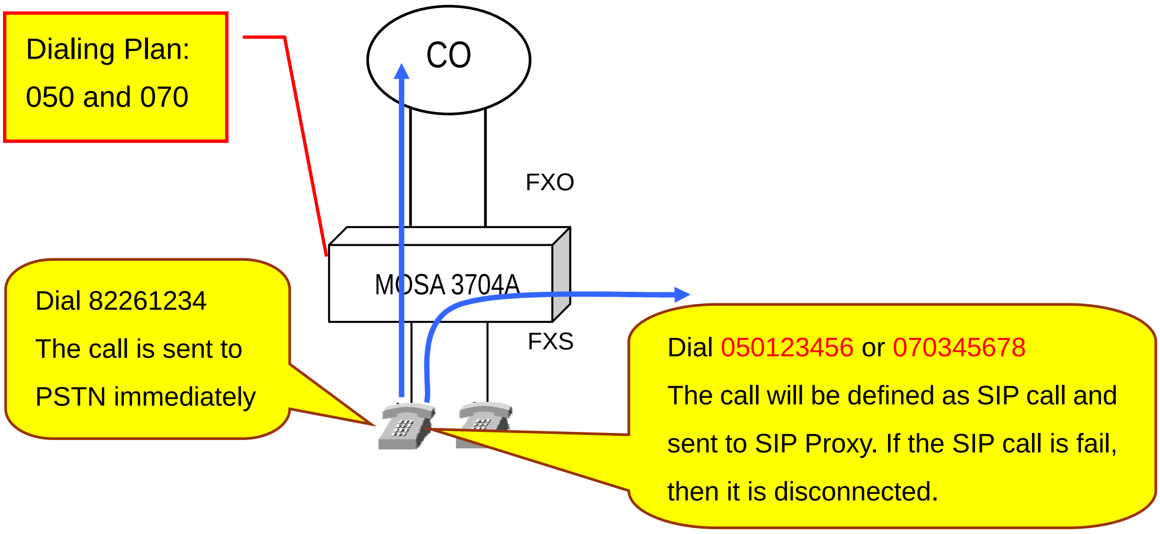
(3) Transparent with digitmap: All numbers user dials are sent to SIP proxy server and if any numbers match digitmap, number is send to SIP Proxy immediately without waiting dial end time. Please refer to 8.2.2Digit Map

Web Folder: 7.Dial Plan

| Dialing Method | |
|---|---|
| Control : | Dialing Plan |

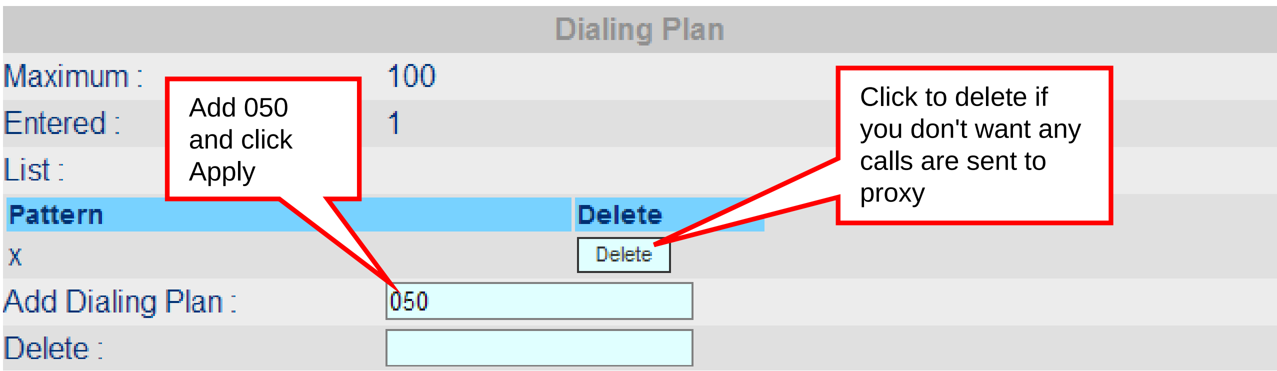
### 8.2.1.1 Dialing Plan

Entry "X" means all calls will be sent to SIP proxy, if the SIP call is fail, it is disconnected. Only if the registration to SIP Proxy is failed, then the gateway will try to connect the number by 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: 7.Dial Plan



### 8.2.2 Digit Map

**Advantage**

1. Able to create usable rule in digit map that is convenience for user to make calls. By this digit map, MOSA 3700 can simulate the FXS port of MOSA 4600 Plus product of VODTEL. It also makes dialing behavior more easily.
2. When digit map is enabled, the outgoing call that fits the rule goes immediately and wait dial ending time is not required.

**Select Dialing Method**

Before you start to use digit map, change dialing method is required. Change Dialing Method to Transparent (with Digitmap)

Web Folder: 7.Dial Plan

| Dialing Method | |
|---|---|
| Control : | Transparent(with Digitma |

Then you are able to use the transparent function with digitmap. Transparent means all numbers user dials are sent to SIP proxy server and all number that controls MOSA 3700 is disabled, including the end code # of each dialing. By this way, all numbers user dials compare with the digitmap. When the number dialed matches the digit map, the number send to Proxy Server immediately without waiting dial ending time.

| Digit Map | |
|---|---|
| Maximum : | 16 |
| Entered : | 1 |
| Length per pattern : | 24 |
| List : | |
| **Pattern** | **Delete** |
| [0-9*#ABCD]. | Delete |
| | Pattern |
| Add Digit Map Item : | |
| Delete Digit Map Item : | |

By the example figure above, we configure some example of digitmap

Here is the explanation of rule

(1) X means any digits

(2) [ ] means the digits in the [ ] are all acceptable, such as [479]

(3) [ ~ ] means the range between ~ are all acceptable. For example, [2~4] means the number 2, 3, 4 are all acceptable

(4) "." means the previous digit can appear again. For example, "X." means 22, 33, 44... are all acceptable.

| Example | Description |
|---|---|
| *[389*]X.[#8] | Number that match *+ 3 or 8 or 9 or * + any digit + repeat previous digit + # or 8 are send to proxy server without waiting dial ending time |
| *2XX | Number that match * +2 + any digit + any digit are send to proxy server without waiting dial ending time |

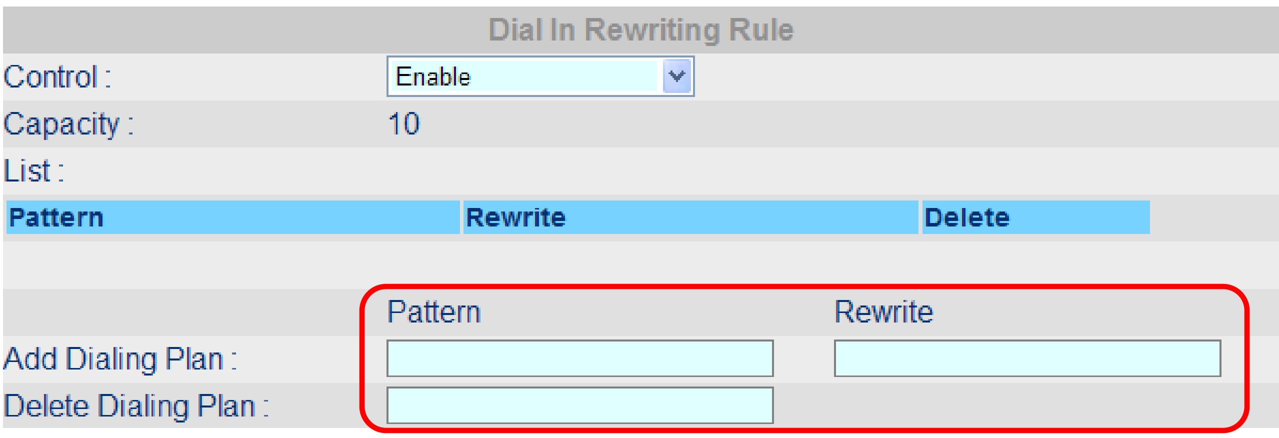
### 8.2.3 Dial in Rewriting Rule

Number dialed from MOSA 3700 can be converted to different number and sent to SIP Proxy. User can pre-define maximum 10 sets of prefix rewriting rule to convert the number that user dials before build the connection to SIP Proxy. It is useful to create a user-friendly dialing behavior and also can limit user to dial certain number. The rules below explain the judgment.

1. System will check the dialing plan on last page in advance to decide whether it is PSTN call or SIP call.
2. If the call will be send to SIP Proxy, then system will exams the number to see if it meets Rewriting Rule.
3. If the SIP call does not meets any Rewriting Rule, system will build the SIP call with the number that user dials.
4. If the numbers of the SIP call meets any Rewriting Rule, then the numbers is converted (or limited if it meets barring rule) and system build the SIP call by converted number.

Here is the example

Web Folder: ADVANCED \ DIALING PLAN



Pattern: Add the pattern that user may dial
Rewrite: Add the converted number if user dials the same digits in pattern column.
Fill in digits and click the Apply button

By the operation above, we create a Rewriting Rule table below and it controls all SIP call.
The example table below illustrate that all call are converted to the phone number that includes Country Code + Area Code + Phone No., and then sent to proxy, and prefix phone number 0204 is forbidden.

| Pattern | Rewrite | X means any digits. ! means the call is terminated. |
|---|---|---|
| 00x | | If the prefix number dials from user are 001~009, then the 3 digits are removed. For example, if user dials 0028621123456, then the system dials 86211123456 to build SIP call. |
| 0 | 886 | If the prefix number dials from user are 0, then the digit is replaced with 886. For example, if user dials 0921123456, then the system dials 886921123456 to build SIP call. |
| x | 8862x | If the prefix number dials from user are 1~9, then add 8862 in front of the original number. For example, if user dials 82263368, then the system dials 886282263368 to built SIP call. |
| 0204 | ! | If the prefix number dials from user are 0204, then the call is terminated. |

**Matching Rule**

1. Best Match rule, the longest digits match first.
2. Wildcard ( x digits) match last

## 8.3 Call Forward

There are three forward types:

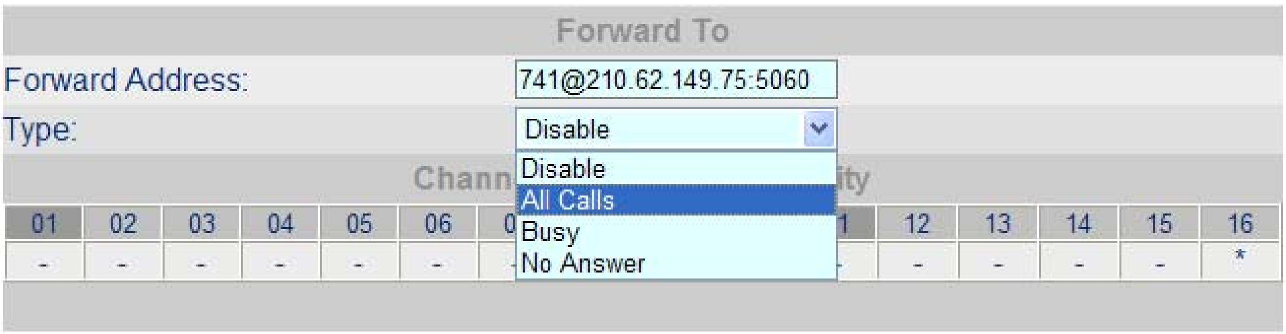
1. All: All incoming VoIP call to the SIP entity will be forward.
2. Busy: When the SIP entity is busy, the incoming VoIP call will be forward.
3. No Answer: When the SIP entity is no answer and after 30 seconds, the incoming VoIP call will be forwarded.

Notice:

- In order to let the caller identify that the port has been configured "forward", the caller will hear second dial tone, rather than normal dial tone.
- If Auto Answer function is disabled, incoming call from PSTN seizes a free FXS port. The call is not forwarded even the seized FXS port is part of Call Forward SIP Entity.
- If Auto Answer function is enabled, Incoming PSTN call dials "*" to seize a free FXS port after second dial tone. The call is not forwarded even the seized FXS port is part of Call Forward SIP Entity.
- If Auto Answer function is set to Forward to SIP, Incoming PSTN call is forward to new destination configured in the entity that this channel belongs to.

Configuration

WEB page: 1.SIP Environment\1.1.Proxy/Trunk Mapping



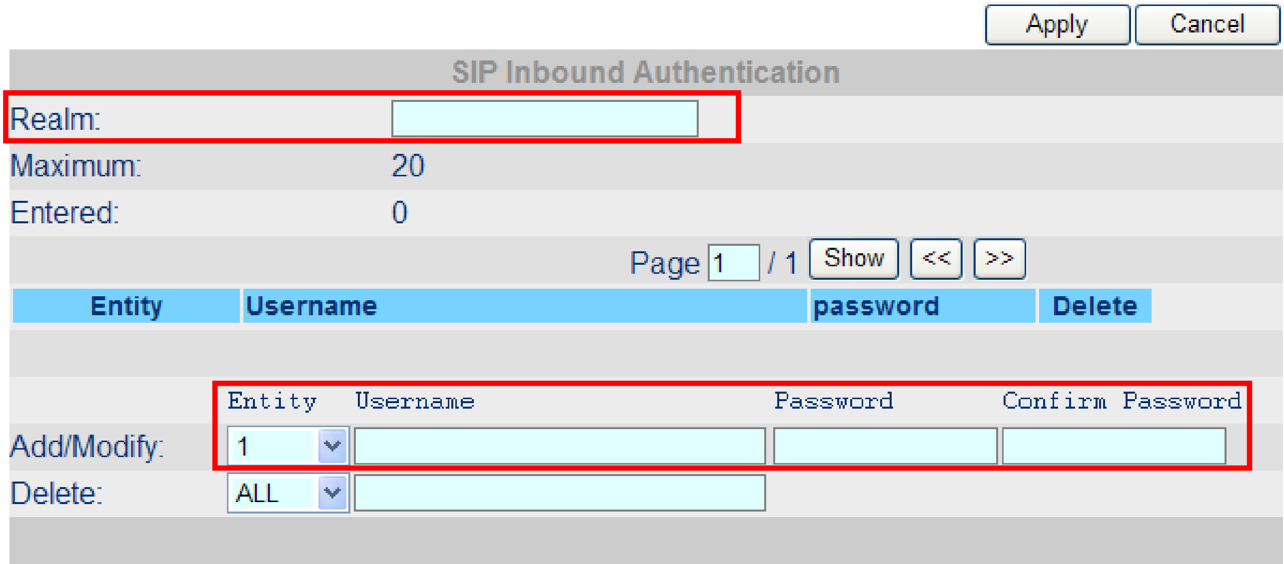

Phone Set: Please refer to section Appendix A: Phone-Set Command.

## 8.4 Inbound Authentication

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

Configuration

WEB Page: 3. SIP Advanced\3.1.Inbound Authen.




## 8.5 FAX

For MOSA 3700 software version 1.05 or above, SIP-based T.38 Fax protocol is applied. Any brand SIP gateway with SIP-based T.38 Fax protocol may transmit FAX with each other. T.38 is FAX protocol and it has better performance and better successful transmission rate. However, SIP device that does not support SIP-based T.38 still can transmit and receive FAX with MOSA 3700 by G.711 codec. G.711 codec uses more bandwidth, so it may not as good as SIP-based T.38 protocol if bandwidth control is the key factor of the network.

Setup method is listed below:

### 8.5.1 Connect FXO port to PSTN

Select the Channel with FXO Type that will receive FAX from PSTN and click its St (Status)

WEB page: 2.Channel Config.\2.1.Summary

| Ch | St | Type | Entity | Reg. | 2833 Status | Auto Answer | T.38 | Statistics In/Out | VAD | Gain In/Out |
|---|---|---|---|---|---|---|---|---|---|---|
| 13 | | FXO | 13 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 14 | | FXO | 14 | - | - | - | - | 0/0 | ✓ | 0/0 |

Configure Connected Device Type to FAX and then click **Apply**

### 8.5.2 Connect FXS port to FAX Machine

Select the Channel with FXS Type that will connect FAX and click its St (Status)

WEB page: 2.Channel Config.\2.1.Summary

| Analog Channel | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| Ch | St | Type | Entity | Reg. | 2833 Status | DND | T.38 | Statistics In/Out | VAD | Gain In/Out |
| 1 | | FXS | 16 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 2 | | FXS | 2 | - | - | - | - | 0/0 | ✓ | 0/0 |

Configure Connected Device Type to FAX and then click **Apply**

| Connected Device Type | |
|---|---|
| Type: | FAX |

Note: For FAX transmission, two gateways will change to SIP-Based T.38 Protocol automatically if both sides support SIP-based T.38.

**Note:**

If MOSA 3700 connects different SIP devices, some have T.38, but some use G.711 codec only, then user should enable G.711 codec support for FAX. Setup method is listed below:

1. The same step as above set Connect Device to Fax

2. Setup “Codecs Type“, Web Folder: 1.SIP Environment\1.2.Common

Select and mark “PCMU” and “PCMA” Codecs (G.711 Standard), than click “Apply” button

| Codecs Selection | | | | |
|---|---|---|---|---|
| Codec Type | G.729AB | G.723.1 | PCMU | PCMA |
| Selected | ☑ | ☑ | ☑ | ☑ |
| Codec Priority: | G729 - G723 - PCMU - PCMA | | | |

3. Warm-Restart the system

## 8.6 Non-SIP Call port seizure preference

For non-SIP Calls, the port seizure preference is listed below

**1. Inbound from PSTN**

If the inbound FXO port was configured as "Fax" device, it will also seize only FXS ports that "Connect Device" is configured as Fax. The Voice devices behave the similar way.

| From FXO port to FXS port | | Note |
|---|---|---|
| Connect Device at FXO port | Connect Device at FXS port | |
| Phone port | Select Phone port only | From the lowest port number upward |
| FAX port | Select FAX port only | From the lowest port number upward |

**2. Outbound to PSTN**

For the calls from FXS to FXO, the ports of the same "Connect Device" type will be the prior selection for the calls.

If there is no correct configured port is available, it will ignore the "Connect Device" setting and create a call as the rule below.

| From FXS port to FXO port | | Note |
|---|---|---|
| Connect Device at FXS port | Connect Device at FXO port | |
| Phone port | Select Phone port ($1^{st}$ priority) | From the highest port number downward |
| | Select FAX port ($2^{nd}$ priority) | |
| FAX port | Select FAX port ($1^{st}$ priority) | From the highest port number downward |
| | Select Phone port ($2^{nd}$ priority) | |

For the setting of "Connect Device", please refer to 8.5 FAX

## 8.7 Call Waiting

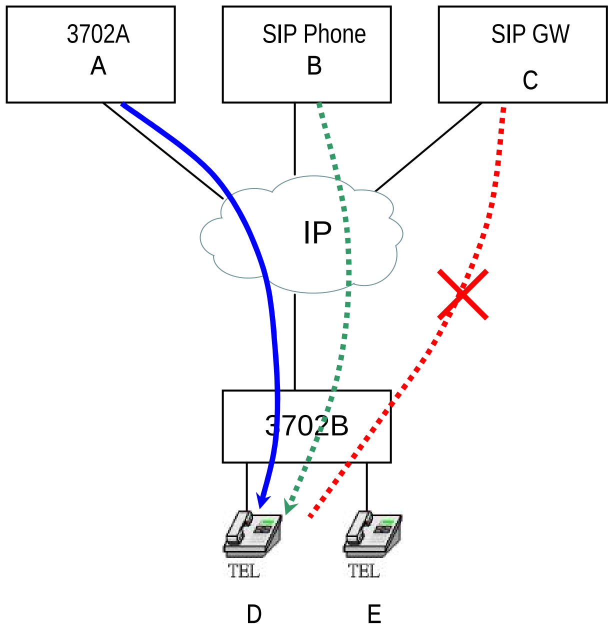
Call waiting function for a FXS port to answer two SIP calls.
When D answer a SIP call from other SIP phone or gateway, such as A. In normal condition, another incoming call dial to D will be busy, such as B to D. With Call Waiting function, the phone call dials from B to D will not be busy. Here is the possible situation.

- D keeps talking with A and hears Call Waiting Tone if B calls D.
- B hears normal ring back tone without sense any different.
- If D keep talking with A and ignore the Call Waiting Tone for more than 30 seconds, Call Waiting Tone stop and the phone call return to normal condition
- If D keep talking with A and ignore the Call Waiting Tone for more than 30 seconds, B keep hearing ring back tone for 30 seconds and listen busy tone finally.
- D can talk to B if D presses Flash button when hearing the Call Waiting Tone. Phone A is silent when D talk to B.

- D can talk to A or to B by keep pressing Flash button to switch the two sides.
- C will hear busy tone when C call to D if there is one line in call waiting status for A.

A: FXS port of MOSA 3700 Series

B, C: SIP Device (MOSA 3700 Series, other brand SIP gateway. SIP phone...), Normal PSTN phone call (special condition is described below)




**Configuration**

Enable the Call Waiting function of the FXS port (D) of MOSA 3700 gateway. This function can be configured for each FXS port individually.

Select the Channel with FXS Type that will has Call Waiting function and click its St (Status)

WEB page: 2.Channel Config.\2.1.Summary

| Analog Channel | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| Ch | St | Type | Entity | Reg. | 2833 Status | DND | T.38 | Statistics In/Out | VAD | Gain In/Out |
| 1 | | FXS | 16 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 2 | | FXS | 2 | - | - | - | - | 0/0 | ✓ | 0/0 |

Enable Call Waiting and click **Apply**

| Call Waiting Control | |
|---|---|
| Control: | Enable |

Call waiting function works only on SIP call. So PSTN call works when it is transited as SIP call. If inbound transit call is configured on MOSA 3700 (please refer to 7.8 Make Inbound Transit Call), then Call Waiting function is available when user dials the SIP number of this MOSA 3700 gateway itself. If no inbound transit call function is configured, it is impossible to do call waiting function.

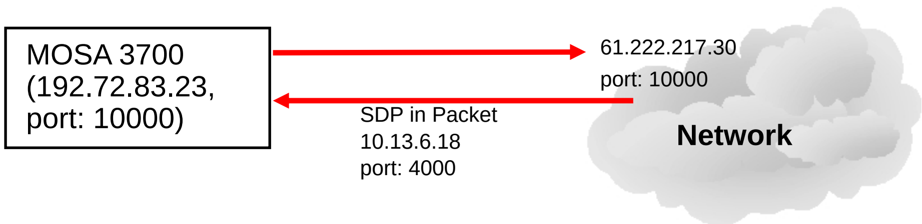
## 8.8 Target the Media (RTP)

For the SIP call passing through NAT, it is possible that the media would not deliver properly; owing to the RTP contact information (IP address, port number) is different from original RTP packet. This function selects different contact information for MOSA 3700 to send RTP Packets to other SIP device within far-end NAT. It designates whether to use the source contact information from the UDP/IP header (Symmetric RTP) or the contact information specified within the packet (SDP) when MOSA send RTP packet

Web Folder：ADVANCED\SIP COMMON, Default Value is SDP


**Example 1**: Via Symmetric RTP

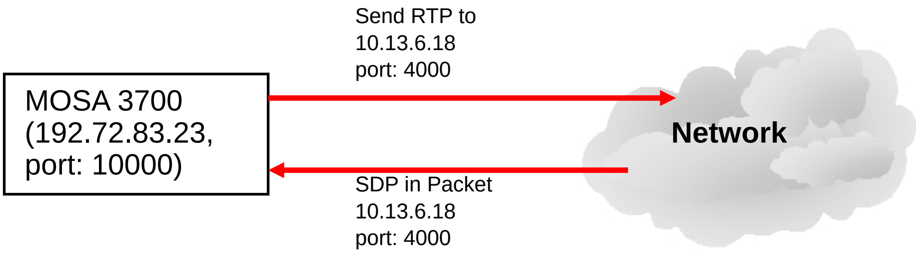
The source contact information (IP, port number) of RTP packet is IP: 61.222.217.30, port number: 10000, but the SDP in the packet is IP: 10.13.6.18, port: 4000. In this case, please Use **Symmetric RTP**




MOSA 3700 tries the contact information from SDP first (IP: 10.13.6.18, port number: 4000). If MOSA 3700 finds that the contact information from SDP is different from the source contact information, then it will try the source contact information, as the example above, use IP: 61.222.217.30, port number: 10000. It makes SIP call successful.

**Example 2**: Via SDP (Default)

This selection ignores the source contact information (IP, port number), which MOSA 3700 received. It always sends the RTP packet to the contact information (IP, port number) described in the packet (SDP) received.




# 9. File Management

## 9.1 File Types

The naming convention to the file type of FONEMOSA 4496 is listed in the following table:

| File Name | File Type | Description |
|---|---|---|
| SIP3302.CFG<br>SIP3304.CFG<br>SIP33XX.CFG | System configuration file | File of system configuration |
| SIP3302.RUN<br>SIP3304.RUN<br>SIP33XX.RUN | Executing file | System Software |
| SIP3302.WEB<br>SIP3304.WEB<br>SIP33xx.WEB | Web file | Page for web browser |
| SIP3302.MEM<br>SIP3304.MEM<br>SIP33xx.MEM | Text file | MEM setting file can be downloaded by Web or FTP to PC; open file and modify the contents using NOTEPAD or other word processing tool; then uploaded the file to system. |

# 9.2 Software Update

## 9.2.1 Software update via Web

For the most convenient way to update software, is to make it via Manage Web Page. Please refer to section 10.13 Web Page 8.File Transfer

## 9.2.2 Software update via FTP

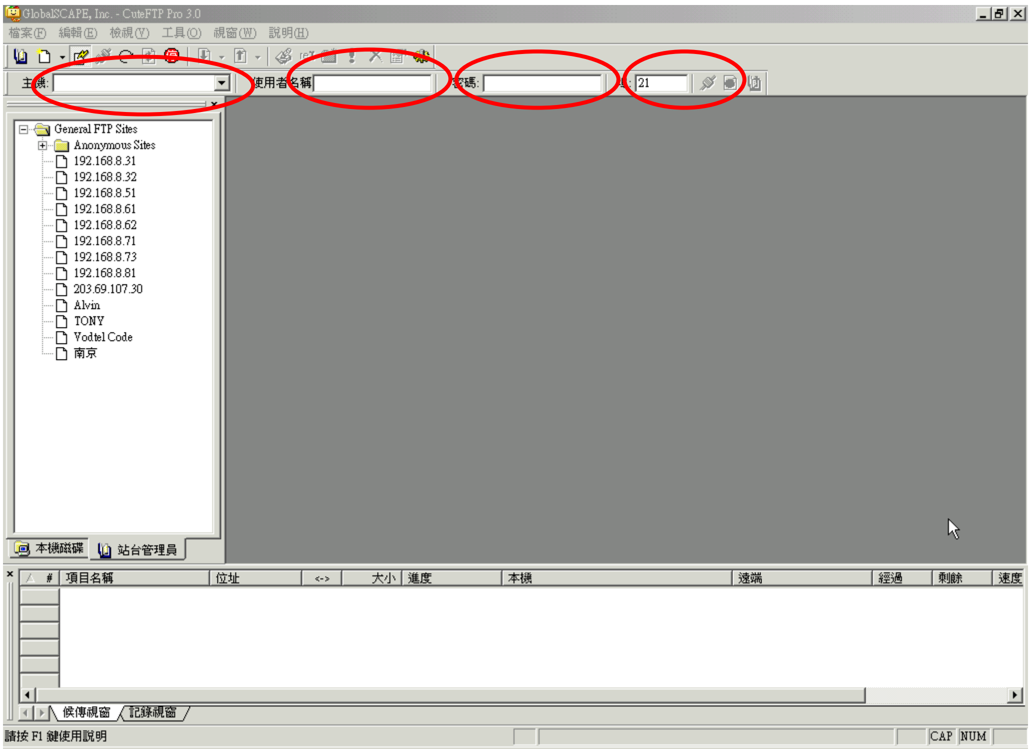
You can also do it by FTP client software

**Preparation before Updating FIRMWARE**

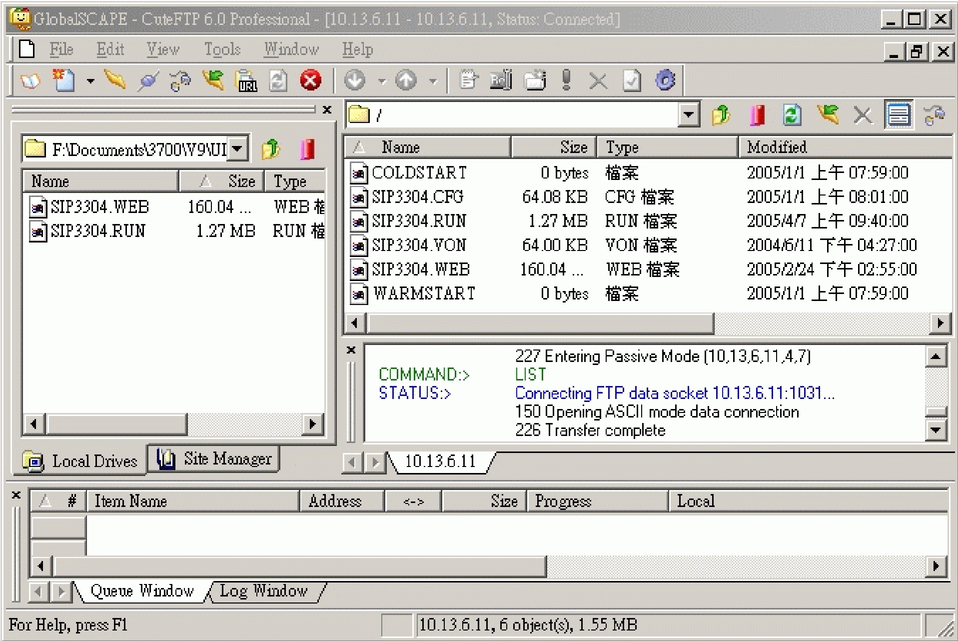
1. Power on this machine.
2. Get Windows based PC ready
3. LAN cable is well connected (for FTP)
4. Configure the IP, Subnet, and Default Gateway of this gateway and PC
5. Get the file for update ready.

**Software Update by FTP for File Type RUN, MEM and WEB**

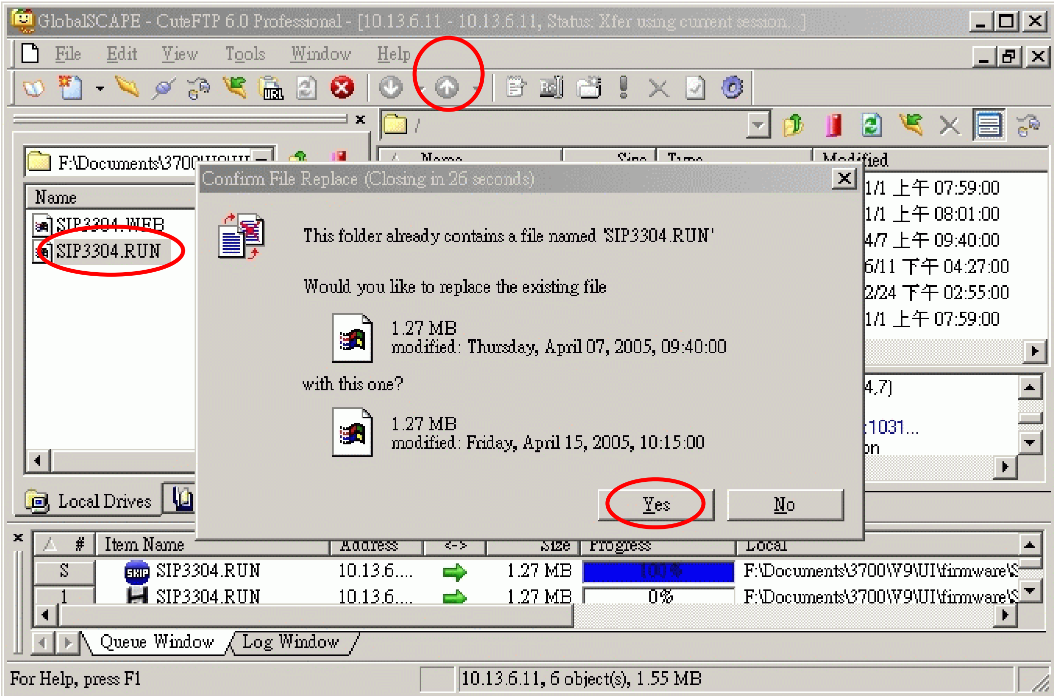
1. Execute FTP Client Software, e.g. CuteFTP
   Enter IP Address, User Name (default is FTP), Password (the password of FTP and Console is same, and the default is blank), and the Port Number to 21



2. Click button **Connect** to get connection between gateway and FTP Client. The files of the gateway will be displayed on the window if the connection is successful.




3. Select the file with extension of .RUN and click button Upload and then Yes to overwrite. (Please notice that the file name must be same as the file name in the Gateway, e.g. SIP3304.RUN).



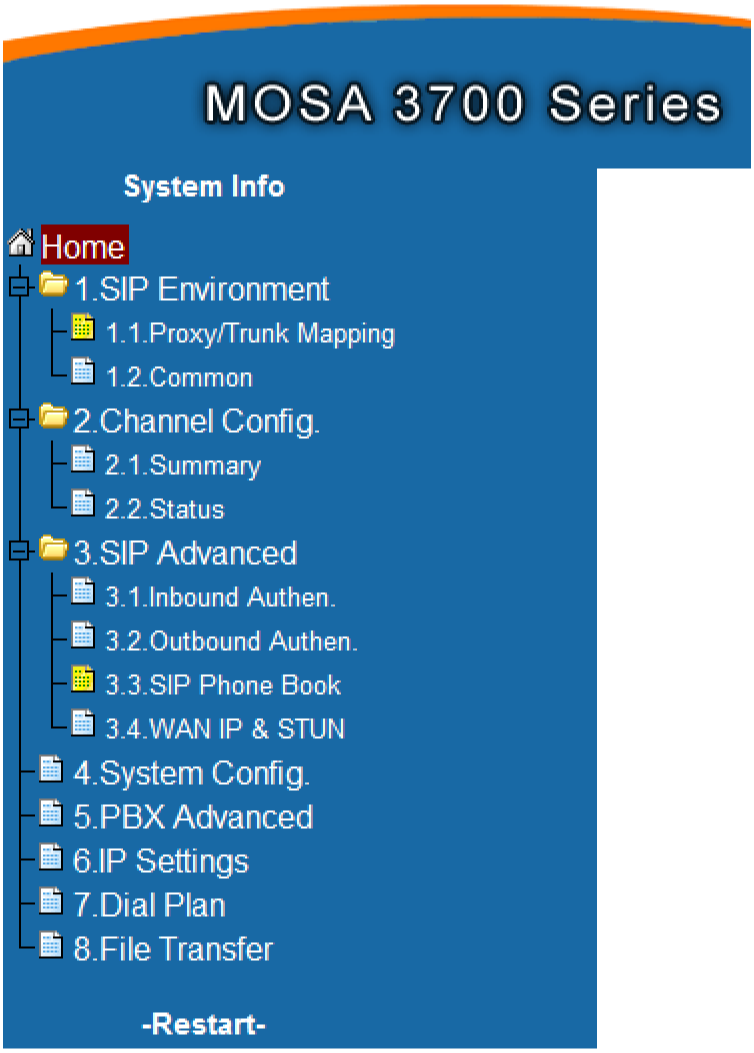

4. After the file is overwritten (you may check if the time of the file is updated), Gateway has to run Cold Start to store the configure file, then the updating is effective.
5. Select the file with extension of .WEB and click button **Upload** (Please notice that the file name must be same as the file name in the Gateway, e.g. SIP3302.WEB). And repeat the step 3 ~ 4.
6. Check if the uploading is successful, you enter the Web Management Page to examine the version of software. (Web Folder: 4.System Config.)

| Information | |
|---|---|
| Regional ID: | 0 (Taiwan) |
| Software Version: | 2.00.0 |
| BootRom Version: | 1.02 |
| Hardware Version: | 1.00 |
| Module Type: | 8 PORT_FXS / 8 PORT_FSO |
| Up-Time: | 5 day 2 hr 37 min 12 sec |
| MAC Address: | 00-03-62-00-00-48 |

Check if the version is correct

# 10. WEB MANAGEMENT INTERFACE

The Tree Architecture of Web Management is shown below



# 10.1 1.SIP Environment/1.1.Proxy/Trunk Mapping

(Need Warm-Restart) Apply Cancel

**Outbound Proxy Setting**

| | |
|---|---|
| Domain Name: | [ ] Enable |
| Port: | 5060 |

**Registrar Setting**

| | |
|---|---|
| Domain Name *(IP:Port)*: | [ ] : [ ] Enable |

**Register Expiration**

| | |
|---|---|
| Time Interval *(60~86400 sec.)*: | 0 sec. *(0: use default 3600 sec.)* |

**RTP Tracking**

| | |
|---|---|
| Control: | SDP |

**Incoming Call Screening**

| | |
|---|---|
| Accept Calls From Proxy Only: | No |

---

**Registration**

| | |
|---|---|
| Register Control: | None |

**SIP Entity**

| | |
|---|---|
| Entity: | 1 Select |
| Entity Control: | Enable |
| Register Status: | FAIL Register De-Register |
| CLIR: | Disable (Calling Line Identification Restriction) |

**Public Address Setting**

| | |
|---|---|
| Address: | |

**Default Account**

| | |
|---|---|
| User name: | |
| Password: | |
| Confirm Password: | |

**Contact Address Information**

| | |
|---|---|
| Current Address: | 4628 |

**Forward To**

| | |
|---|---|
| Forward Address: | |
| Type: | Disable |

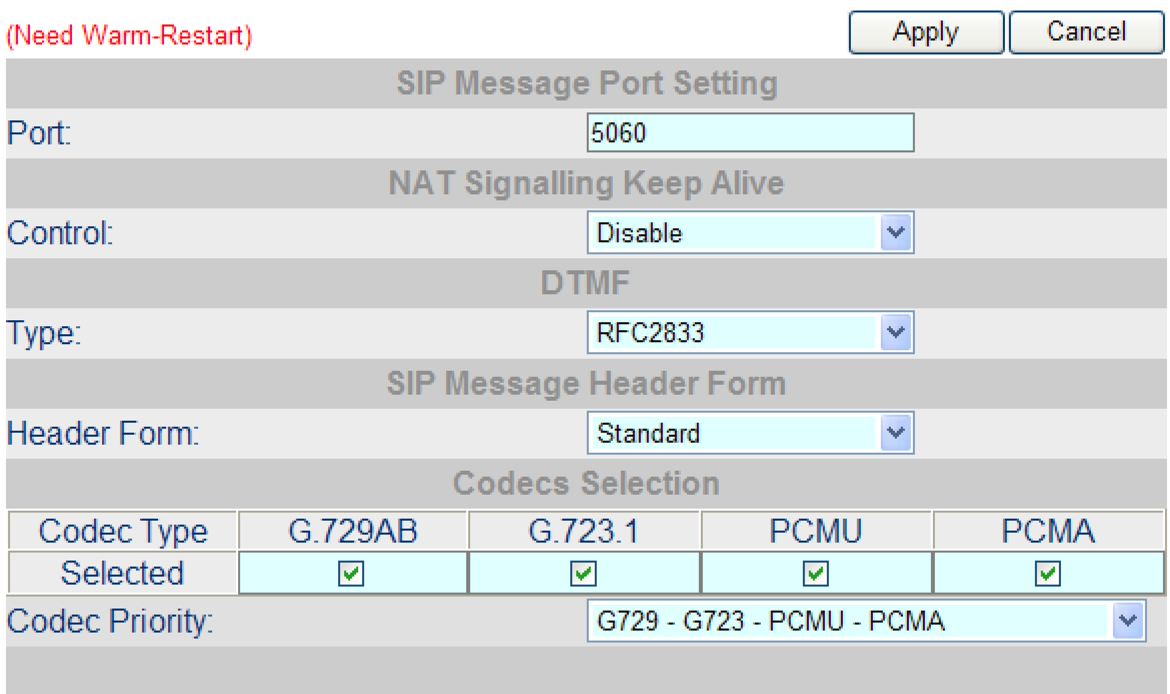
**Channel Member of This Entity**

| 01 | 02 | 03 | 04 | 05 | 06 | 07 | 08 | 09 | 10 | 11 | 12 | 13 | 14 | 15 | 16 |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| - | - | - | - | - | - | - | - | - | - | - | - | - | - | - | * |

| Type | Field | Description | Default Value |
|---|---|---|---|
| Outbound Proxy Setting | Domain Name | Input Domain name or IP address of SIP Proxy Server, and also Enable/Disable it. | Disable |
| | Port | Input control port number of SIP Proxy Server | 5060 |
| Registrar Setting | Domain Name(IP:Port) | Input Domain name or IP address of SIP Registrar Server, its Control Port and also Enable/Disable it. | Disable |
| Register Expiration | Time Interval (60~86400 sec) | When this machine is configured as Client Mode, it registers to SIP Proxy before timeout value configured here repeatedly. It keeps the registration status with SIP Proxy, because SIP Proxy may terminate connection when this machine is idle. | 0 (means default 3600) |
| RTP Tracking | Control | Send RTP packet to destination under the condition here<br>SDP：Retrieve destination info from incoming packet<br>Symmetric RTP：From IP/Port of original destination. | SDP |
| Incoming Call Screening | Accept Calls From Proxy Only | No: Accept all incoming SIP call<br>Yes: This machine only accepts incoming call through SIP Proxy. | No |
| Registration | Register Control | ◆ None: This machine does not register to SIP Proxy spontaneously. You can register each entity manually by the button below.<br>◆ Register All: All entities of this machine register to SIP Proxy spontaneously.<br>◆ De-Register All: All entities of this machine are forced to De-Register. | None |
| SIP Entity | Entity | Select the Entity you want to operate | 1 |
| | Entity Control | Enable: The entity you select is enabled<br>Disable: The entity you select is disabled | Disable |
| | Register Status | Shows the registration status<br>■ Registered：Registration is successful<br>■ Registering：Trying to register<br>■ Fail：Registration is failed<br>■ Idle：Means SIP trunk is disabled<br>Register (button)：Click to do manual registration<br>De-Register (button)：Click to quit registration manually. | |
| | CLIR | Calling Line Identification Restriction<br>Disable: Caller ID is sent<br>Enable: No caller ID is sent | Disable |
| Public Address Setting | Address | Input "SIP Entity number@Proxy Server". Please apply this number from ITSP | |

| Type | Field | Description | Default Value |
|---|---|---|---|
| | Default Account | The information to register SIP Proxy<br>Username: Account for registration<br>Password: Password to verify the account<br>Confirm Password: Double confirm password | |
| Contact Address Information | Current Address | Show the contract address (Read Only) currently uses | 01 |
| Forward To | Forward Address | Enter a complete SIP account (Public Address: SIP number@SIP Proxy, such as 234@10.13.6.21). When SIP user makes call to this SIP Entity, the call is forward to new SIP Entity configured here, such as other MOSA 3700 or SIP Phone. | |
| | Type | Disable: Disable Forward To function<br>All Calls: All incoming calls are forward.<br>Busy: forward calls when this Entity is busy.<br>No Answer: Call is forwarded when no one answer the phone for a period of time. | Disable |
| Channel Member of This Entity | | Show " * " means this channel had joined this SIP Entity. | 01 |

## 10.2 1.SIP Environment/1.2.Common



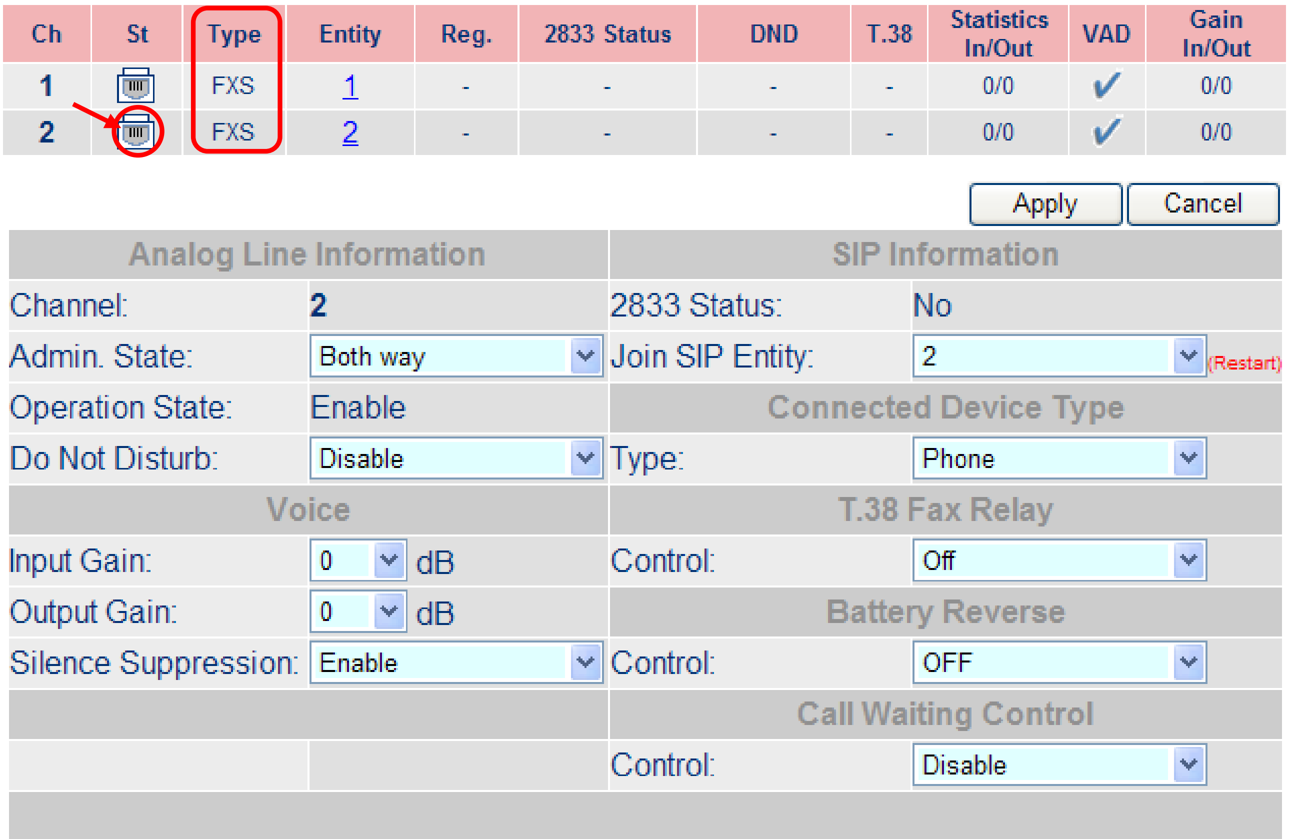
| Type | Field | Description | Default |
|---|---|---|---|
| SIP Message Port Setting | Port | Input SIP message port number | 5060 |
| NAT Signalling Keep Alive | Control | Check RTP packet to insure the connection status. System check RTP packet every 3 minutes. If no RTP packet is found, disconnect connection to release line Enable/Disable | Disable |
| DTMF | Type | Specification to send DTMF<br>RFC2833: Send DTMF by RFC2833, The called side SIP device or SIP Proxy server also have to support this standard<br>INFO: Embedded DTMF inside SIP packet. The called side SIP device or SIP Proxy server also have to support this standard | RFC2833 |
| SIP Message Header From | Header Form | Standard：Use standard SIP packet format<br>Compact: Use compact SIP packet that save bandwidth. The called side SIP device or SIP Proxy server also have to support this standard, otherwise, call can not be built. | Standard |
| Codecs Selection | Codec Type | G.729AB: Mark the selection to Enable this Codec | ✓ |
| | | G.723.1: Mark the selection to Enable this Codec | ✓ |
| | | PCMU: Mark the selection to Enable this Codec (G.711 u Law) | ✓ |
| | | PCMA: Mark the selection to Enable this Codec (G.711 A Law) | ✓ |
| Codec Priority | | You can select the codec priority for your requirement. Not selected items at the table above will not be used. | G729-G723-PCMU-PCMA |

# 10.3 2.Channel Config./2.1.Summary

## 10.3.1 Home

| Type | Field | Description | Default Value |
|---|---|---|---|
| Channel Status | Idle | Port is available | |
| | Conversation | Port is under conversation (when it is considered as successful IP call by this machine) | |
| | In Use | Port is in use | |
| | Ringing | Port is ringing | |
| | Disable | Port is disabled | |

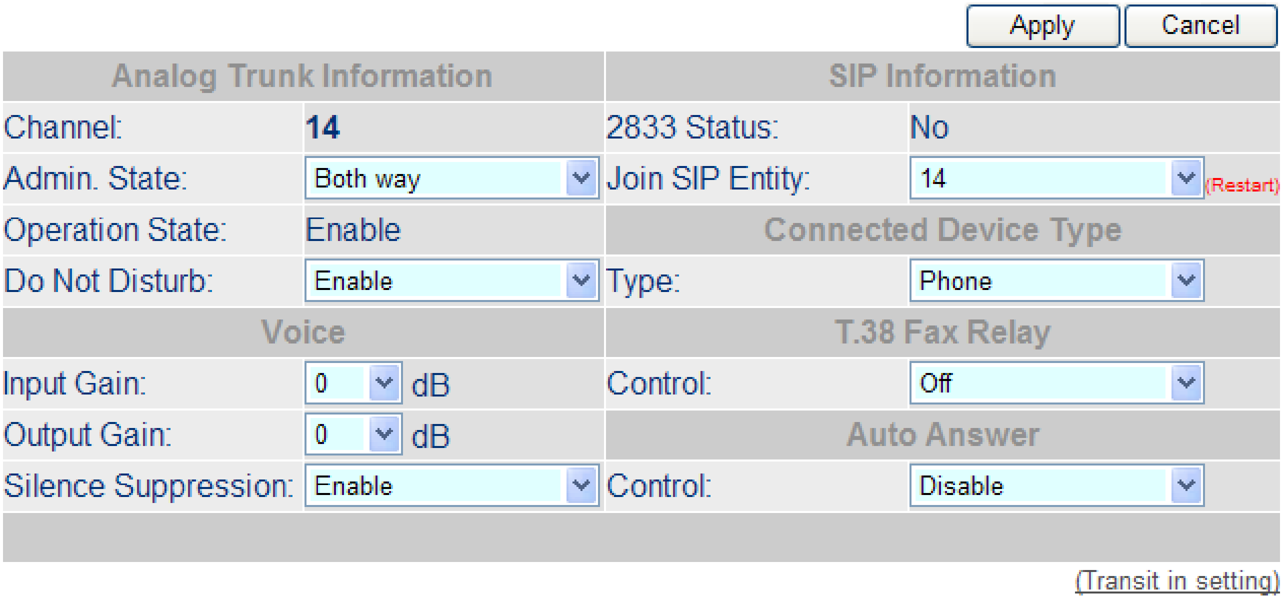
## 10.3.2 FXS Channel Setting



| Type | Field | Description | Default Value |
|---|---|---|---|
| Analog Line Information | Channel | Channel Number (Read Only) | |
| | Admin. State | Control the active status of this port<br>◆ Both way：Incoming and outgoing call is allowed<br>◆ Disable：Disable this port | Both way |
| | Operator State | The action status of this port (Read Only) | Enable |
| | Do Not Disturb | DND function<br>Disable: Disable DND function (accept incoming and outgoing call)<br>Enable: Enable DND (accept outgoing call only and incoming call is denied) | Disable |
| Voice | Input Gain | Voice volume control of input call | 0 |
| | Output Gain | Voice volume control of output call | 0 |
| | Silence Suppression | Silent control for voice packet<br>Enable: If silent happens in conversation, voice packet is not send (save bandwidth)<br>Disable: Keep sending packet even silent happens | Enable |
| SIP Information | 2833 Status | RFC2833 DTMF state (Read Only) | |
| | Join SIP Entity | Select SIP Entity that this channel joins. Do re-start is required if it is changed. | The same as channel no. |
| Connect Device Type | Type | Connection type of this FXS port<br>Phone：General phone line, analog phone-set<br>Fax：Fax machine (fax purpose) | Phone |
| T.38 Fax Relay | Control | Set it to ON is this channel connect FAX On/Off | Off |
| Battery Reverse | Control | Battery Reverse is a mechanism for traditional PBX to judge ON hook or OFF hook status.<br>**ON**: Battery reverse is enabled<br>**OFF**: Battery reverse is disabled | OFF |
| Call Waiting Control | Control | Call waiting function for more then one incoming call<br>Enable/Disable | Disable |

### 10.3.3 FXO Channel Setting

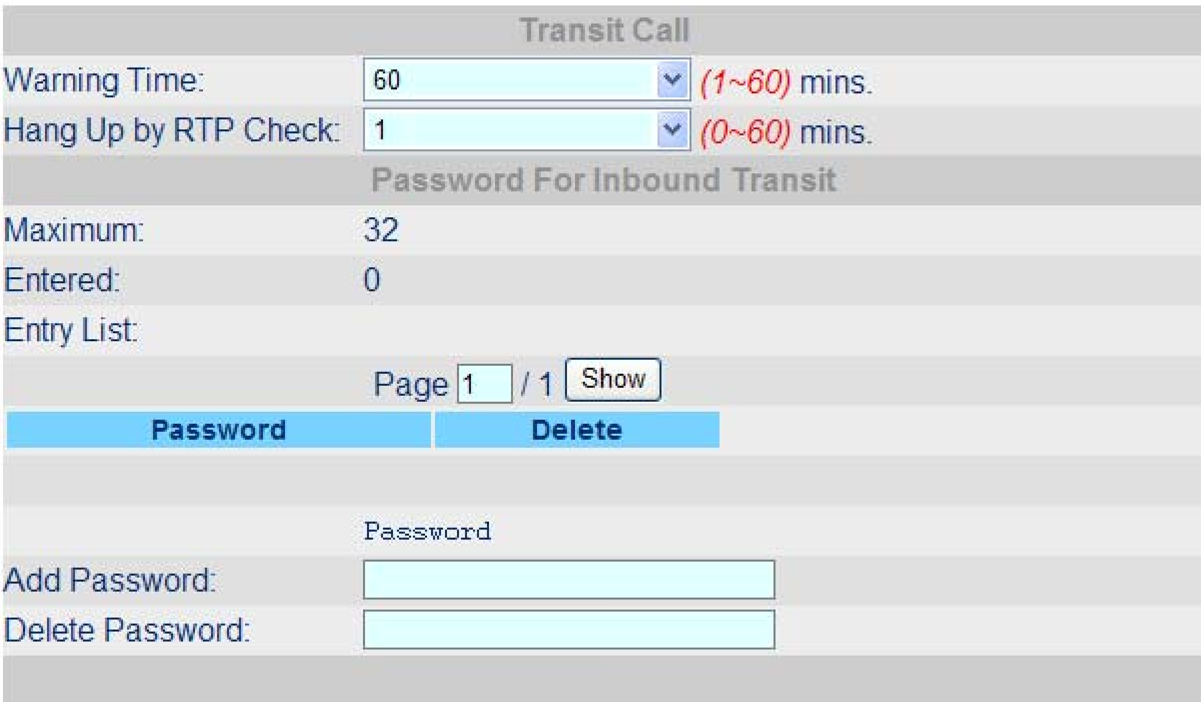
| Ch | St | Type | Entity | Reg. | 2833 Status | Auto Answer | T.38 | Statistics In/Out | VAD | Gain In/Out |
|---|---|---|---|---|---|---|---|---|---|---|
| 13 | | FXO | 13 | - | - | - | - | 0/0 | ✓ | 0/0 |
| 14 | | FXO | 14 | - | - | - | - | 0/0 | ✓ | 0/0 |




| Type | Field | Description | Default Value |
|---|---|---|---|
| Analog Trunk Information | Channel | Channel Number (Read Only) | |
| | Admin. State | Control the active status of this port<br>◆ Both way：Incoming and outgoing call is allowed<br>◆ Disable：Disable this port | Both way |
| | Operator State | The action status of this port (Read Only) | Enable |
| | Do Not Disturb | DND function<br>Disable: Disable DND function (accept incoming and outgoing call)<br>Enable: Enable DND (accept outgoing call only and incoming call is denied) | Enable |
| Voice | Input Gain | Voice volume control of input call | 0 |
| | Output Gain | Voice volume control of output call | 0 |
| | Silence Suppression | Silent control for voice packet<br>Enable: If silent happens in conversation, voice packet is not send (save bandwidth)<br>Disable: Keep sending packet even silent happens | Enable |

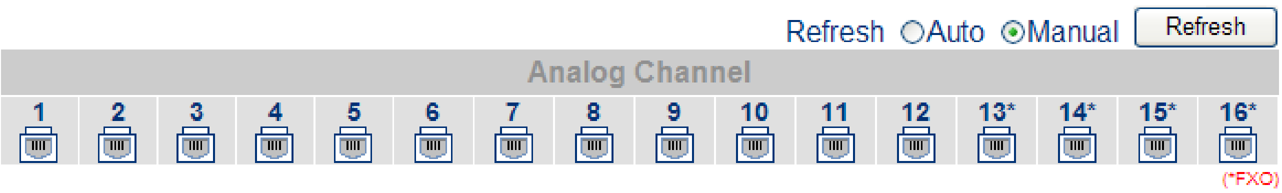
| Type | Field | Description | Default Value |
|---|---|---|---|
| SIP Information | 2833 Status | RFC2833 DTMF state (Read Only) | |
| | Join SIP Entity | Select SIP Entity that this channel joins. Do re-start is required if it is changed. This entity is for inbound transit call purpose only. Other call can not do outbound call to PSTN via this entity | The same as channel no. |
| Connect Device Type | Type | Connection type of this FXS port<br>Phone：General phone line, analog phone-set<br>Fax：Fax machine (fax purpose) | Phone |
| T.38 Fax Relay | Control | Set it to ON is this channel connect FAX On/Off | Off |
| Auto Answer | Control | When Auto Answer is enabled, user can make inbound transit call when machine answer the FXO incoming call.<br>Enable/Disable | Disable |
| Transit in setting | | Please see next section | |

## 10.3.4 Transit in setting



| Type | Field | Description | Default Value |
|---|---|---|---|
| Transit Call | Warning Time | PSTN inbound Call is disconnected to release call after the time configured here,<br>When busy tone detection is failed, this is the only way to release the call. | 60 |
| | Hang Up by RTP Check | If the machine fails to detect the RTP packet for more then the time configure here, the machine disconnect this PSTN inbound Call. | 0 |
| Password For Inbound Transit | Maximum | Maximum number of entries allowed | 32 |
| | Entered | The number of entries had been entered | 0 |
| | Entries List | Display the detail data in the list<br>Password: Shows entries<br>Delete (button): Click it to delete that entry | |
| | Add Passwords | Password：Input the password you want to add, can be digit 1~9 or *, #, max 8 digits | |
| | Delete Passwords | Enter the password to be deleted, refer the detail data under Entries List | |

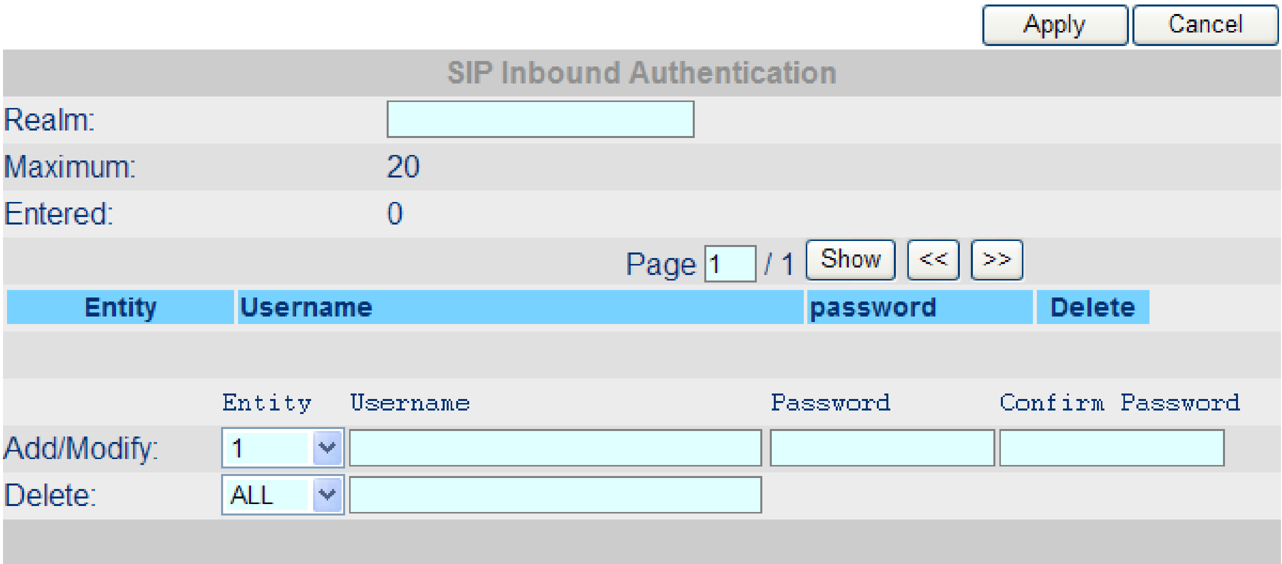
## 10.4 2.Channel Config./2.2.Status


| Type | Field | Description | Default Value |
|---|---|---|---|
| Channel Status | Idle | Port is available | |
| | Conversation | Port is under conversation (when it is considered as successful IP call by this machine) | |
| | In Use | Port is in use | |
| | Ringing | Port is ringing | |
| | Disable | Port is disabled | |

| | | | |
|---|---|---|---|
| Refresh (button) | Refresh | Select Refresh Mode<br>Auto: Web Page update every 10 seconds<br>Manual: Web Page update when you click Refresh button | Manual |
| Analog Channel | 1~16 | Status of each port. Number with "*" mark are FXO channel | |

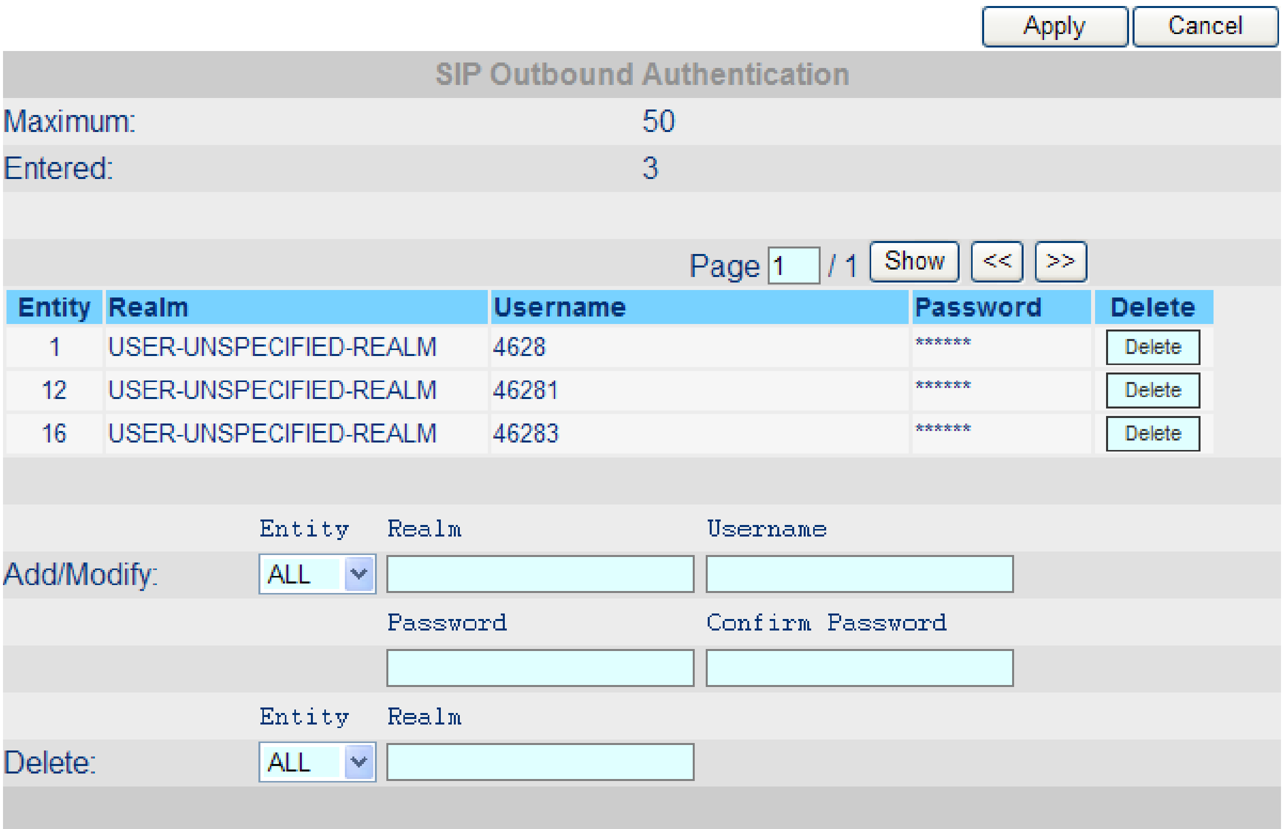
## 10.5 3.SIP Advanced\3.1.Inbound Authen.




| Type | Field | Description | Default |
|---|---|---|---|
| SIP Inbound Authentication | Realm | Enter domain name or IP address of this machine<br>(Such as SIPLine.vodtel.com.tw) or IP Address | |
| | Maximum | Maximum number of entries allowed (Read only) | 20 |
| | Entered | Number of entries of authentication entered. (Read only) | 0 |
| | | It shows the detail of Inbound authentication below (Read only)<br>◆ Entity：SIP group number<br>◆ Username：Account name<br>◆ Password：Password, shows ****<br>◆ Delete：Click Delete button to remove this entry | |

| Type | Field | Description | Default |
|---|---|---|---|
| | Add/Modify | Enter entries of authentication<br>◆ Entity: Which SIP entity that you select.<br>◆ Username: Username of authentication.<br>◆ Password: Password of authentication.<br>◆ Confirm Password: Enter password again for confirmation. | |
| | Delete | Delete data of Inbound authentication<br>■ Entity：SIP group number<br>■ Username：Account name | |

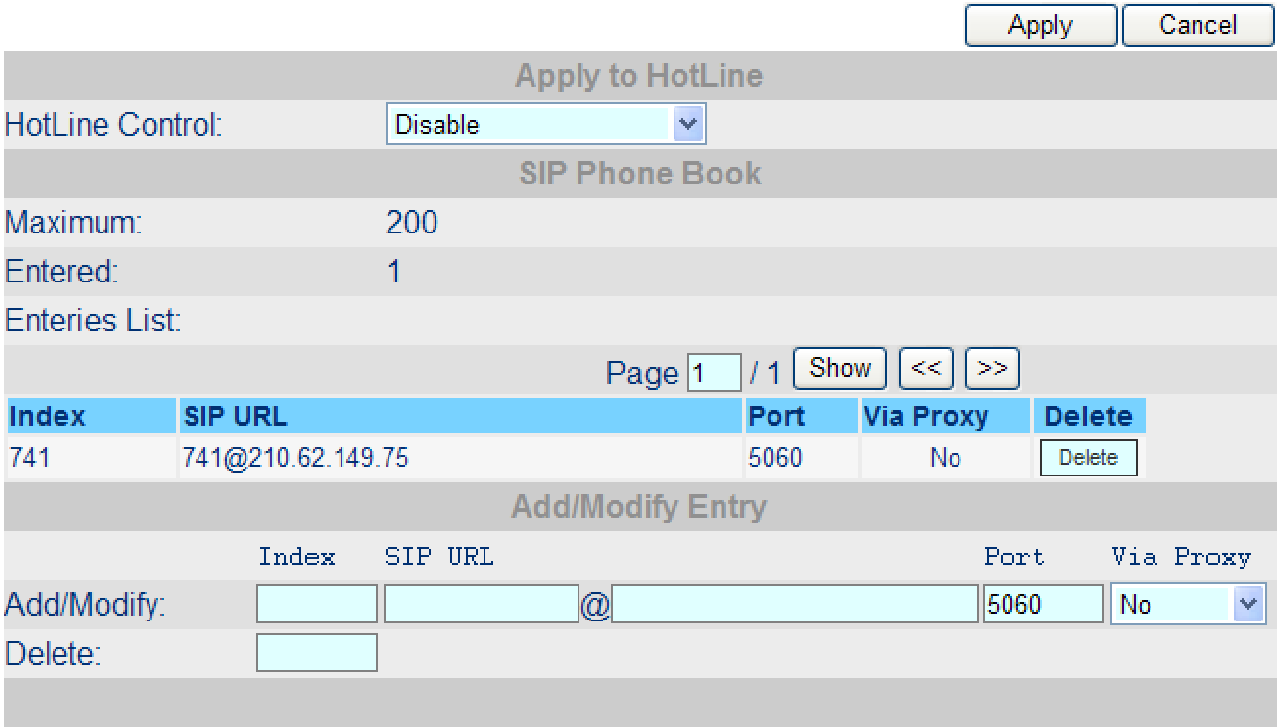
## 10.6 3.SIP Advanced\3.2.Outbound Authen.




| Type | Field | Description | Default |
|---|---|---|---|
| SIP Outbound Authentication | Maximum | Maximum number of entries allowed (Read only) | 50 |
| | Entered | Number of entries of authentication entered. (Read only) | 0 |

| Type | Field | Description | Default |
|---|---|---|---|
| | | It shows the detail of Outbound authentication below (Read only)<br>◆ Entity：SIP Group Number<br>◆ Realm：In most of case, it is the domain name or IP address of SIP Proxy. If none is entered, system create a default value<br>◆ Username: Username of authentication.<br>◆ Password: Password of authentication.<br>◆ Delete: Click Delete button to remove this entry | |
| | Add/Modify | Enter the information of outbound authentication<br>◆ Entity: Select an entity.<br>◆ Realm: Domain name or IP address, however, for some proxy, it use special characters.<br>◆ Username: Enter Username of authentication.<br>◆ Password: Enter password of authentication.<br>◆ Confirm Password: Enter password again for confirmation. | |
| | Delete | Delete Outbound Authentication data<br>◆ Entity：SIP group number<br>◆ Username：Account name | |

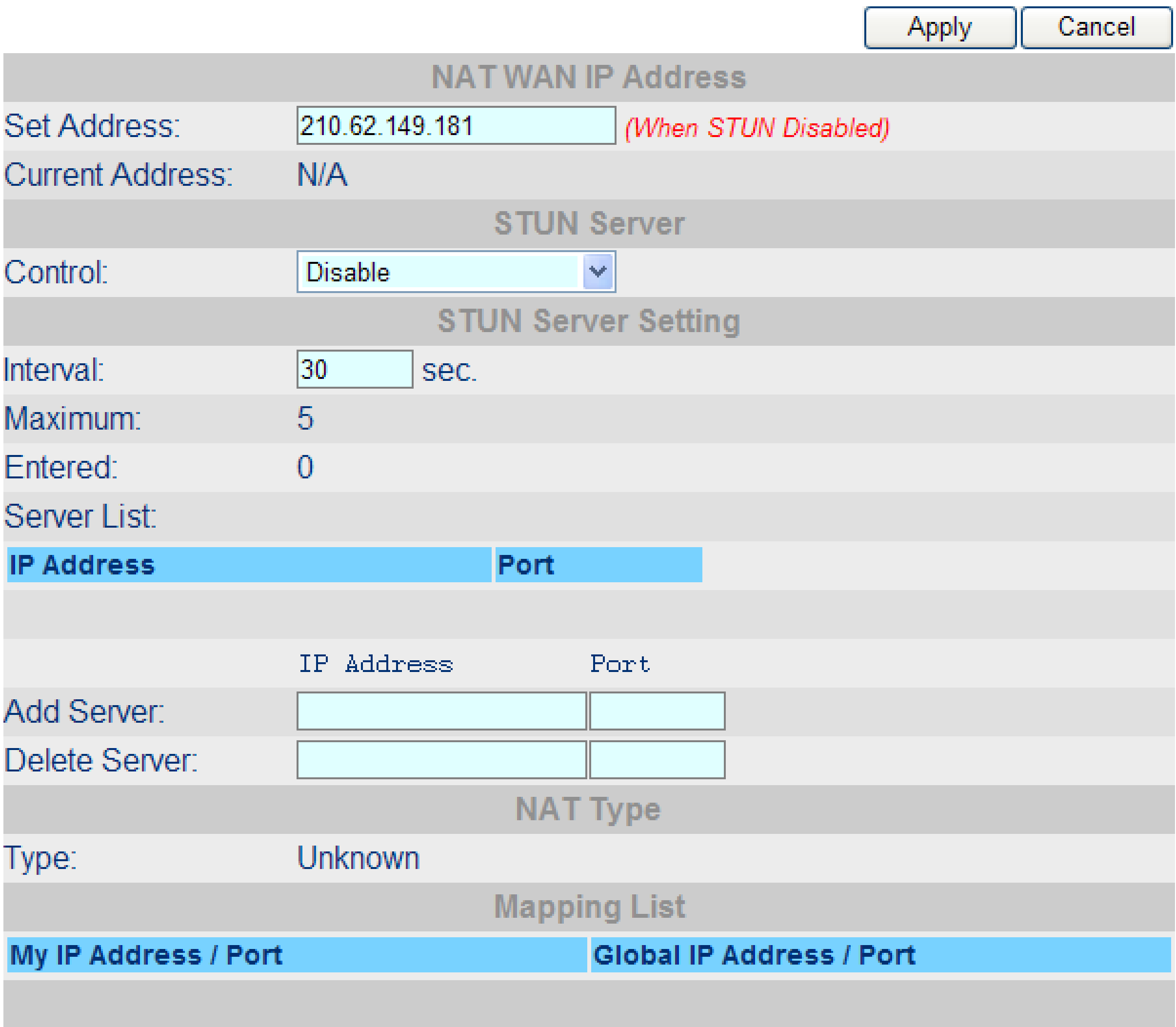
# 10.7 3.SIP Advanced\3.3.SIP Phone Book




| Section | Item Field | Description | Default |
|---|---|---|---|
| Apply to Hotline | Hotline Control | Enable or Disable the hotline function to MOSA 4600 SIP Line or other SIP device to make hotline call. | Disable |
| 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)<br>Index: Dialing number<br>SIP URL: SIP account.<br>Port: Port number.<br>Via Proxy: Via proxy or not. | Empty |

| Section | Item Field | Description | Default |
|---|---|---|---|
| | Add/Modify Entry | Add/Modify Entry<br>Index: Enter dialing number<br>SIP URL: Enter SIP account.<br>Port: Enter port number<br>Via Proxy: Select via Proxy or not | <br><br><br>5060<br>No |
| | Delete Entry | Delete entries<br>Index: Enter the index for delete. | Empty |

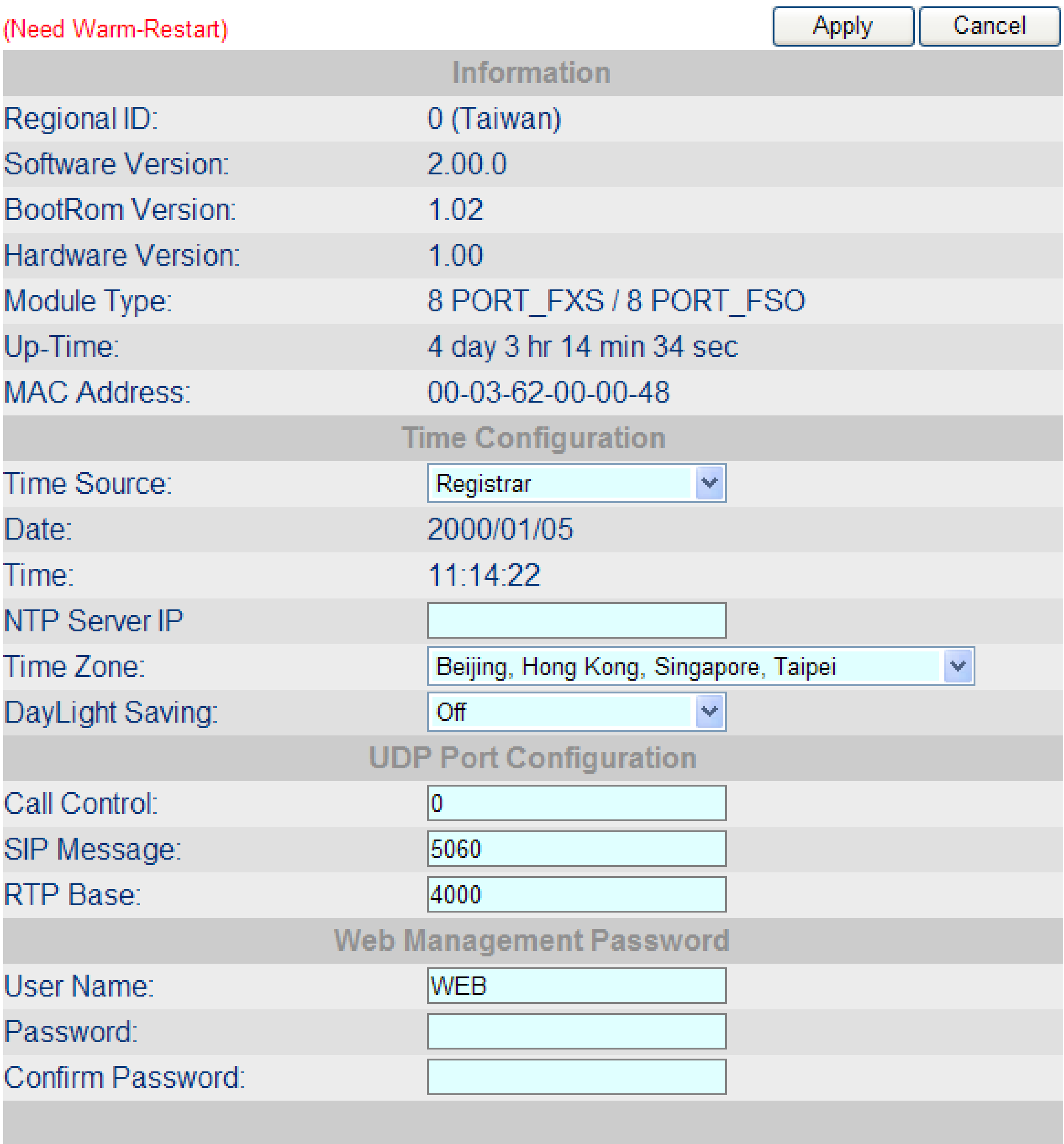
## 10.8 3.SIP Advanced\ 3.4.WAN IP & STUN



| Type | Field | Description | Default |
|---|---|---|---|
| NAT WAN IP Address | Set Address *(When STUN Disable)* | Input NAT WAN IP helps this machine to penetrate NAT without using STUN Server. For different network condition and registration requirement, there are 4 kinds of conditions for configuration<br>1. The machine uses fix private IP and it is for LAN user only. Configure it to 0.0.0.0<br>2. The machine uses fix public IP directly. Configure it to 0.0.0.0<br>3. The machine is installed under NAT and it use fix private IP under NAT. Connection out side NAT use dynamic public IP and this machine is for users at both inside and outside NAT. Then configure it to 255.255.255.255 and work with DDNS is suggested, to prevent registration fail when IP Address is changed.<br>4. The machine is installed under NAT and it use fix private IP under NAT. Connection out side NAT use fix public IP and this machine is for users at both inside and outside NAT. Then configure it to the Public IP of NAT device.<br><br>Attention: Improper configuration cause connection problem. Please configure it carefully according to real network situation and registration requirement. | 0.0.0.0 |
| | Current Address (When NAT IP is 255.255.255.255) | When NAT WAN IP is configured to 255.255.255.255, and LED of Time Srvr keeps ON, it shows the external public IP (Read only)<br>N/A: When NAT WAN IP is not configured to 255.255.255.255 | |
| STUN Server | Control | Use the service provided by STUN Server. When this function is activated, NAT WAN IP is disabled.<br>Enable/Disable | Disable |
| STUN Server Setting | Interval | How frequent does this box query IP info to STUN Server | 30 |
| | Maximum | Maximum number of entries allowed (Read only) | 5 |
| | Entered | Number of entries entered. (Read only) | 0 |

| Type | Field | Description | | Default |
|---|---|---|---|---|
| | Server List | List all entered data | (Read only) | |
| | IP Address | Shows all entered IP Address of server | (Read only) | |
| | Port | Shows all entered control port number of server | (Read only) | |
| | Add Server | Add an entry of a new STUN Server<br>◆ IP Address: IP address of server<br>◆ Port: Control port of server | | |
| | Delete Server | Delete an entry of a STUN Server<br>◆ IP Address: IP address that is going to delete<br>◆ Port: Control port that is going to delete | | |
| NAT Type | Type | Show the current status of NAT type | (Read only) | Unknown |
| Mapping List | My IP Address / Port | Shows the NAT mapping table The private IP Address/ Port Number that is used under NAT | (Read only) | |
| | Global IP Address / Port | The Public IP Address/ Port Number that is used for this machine | (Read only) | |

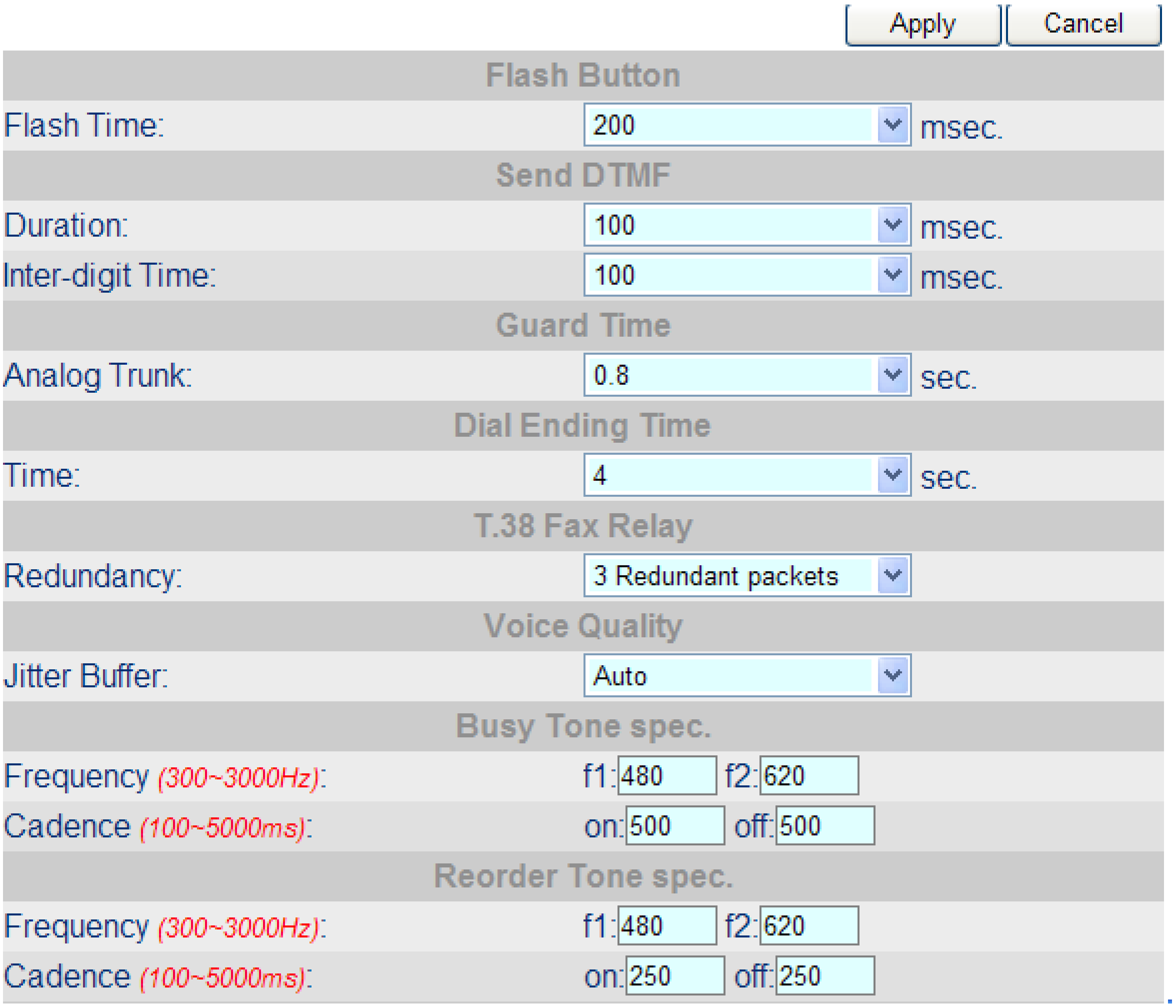
## 10.9 4.System Config.




| Type | Field | Description | Default Value |
|---|---|---|---|
| Information | Region ID | Displays the Region ID (Country ID) of this machine. The ID on the screen is what the machine now using. | 0 |
| | Software Version | Displays the Software Version of this machine (Read Only) | |

| Type | Field | Description | Default Value |
|---|---|---|---|
| | BootRom Version | Displays hardware BootRom Version of this machine (Read Only) | |
| | Hardware Version | Displays hardware Version of this machine (Read Only) | |
| | Module Type | Display the type of module card (Read Only) | |
| | Up-Time | Display the elapse time since last start (Read Only) | |
| | MAC Address | Display the MAC address of HW equipment (Read Only) | |
| Time Configuration | Time Source | Select the time source to synchronize the system date and time<br>**Registrar:** Get time source from the Registrar the box register to<br>**NTP Server:** Get time source from Public NTP time server | Registrar |
| | Date | Current date of this box | |
| | Time | Current time of this box | |
| | NTP Server IP | Input domain name or IP address of NTP server for time sync. | |
| | Time Zone | Select the time zone which the system is located | |
| | DayLight Saving | Select if daylight saving applied<br>**ON**：daylight saving applied<br>**OFF**：daylight saving not applied | OFF |
| UDP Port Configuration | Call Control | Call Control UDP port number for MOSA protocol | 0 |
| | SIP Message | Define SIP call port number for message control | 5060 |
| | RTP Base | Define UDP port number for voice packet transmission. The port number must be even and between the range of 0 – 65534.<br>(It is activated after system re-started) | 4000 |
| Web Management | User Name | User Name to login Web | WEB |
| | Password | Password to login Web | |
| | Confirm Password | Double confirm the password to login Web (has to be consistent with the Password above) | |

## 10.10 5.PBX Advanced



| Type | Field | Description | Default |
|---|---|---|---|
| Flash Button | Flash Time | The time interval for “Flash” that system may accept | 200 ms |
| Send DTMF | Duration | Duration time for DTMF transmit | 100 ms |
| | Inter-digit Time | Inter-digit time between two DTMF | 100 ms |
| Guard Time | Analog Trunk | The minimum time interval between two trunk calls | 0.8 sec |
| Dial Ending Time | Dial Ending Time | Generally “#” is the last character of the number, and that means “end of dialing”. If no “ # ” is dialed, system will wait until dial ending time out. | 4 sec |
| T.38 Fax Relay | Redundancy | Select the volume of re-send redundant packet<br>No Redundant Packet<br>1 Redundant Packet<br>2 Redundant Packet<br>3 Redundant Packet<br>4 Redundant Packet | 3 Redundant Packet |
| Voice Quality | Jitter Buffer | Select the method to suppress voice vibration<br>◆ Auto, the system detects it automatically.<br>◆ Other selection from 20ms~460 ms | Auto |
| Busy Tone Spec | Frequency | Specification of the frequency of busy tone | (300 ~ 3000 Hz) |
| | Cadence | Specification of the cadence of busy tone, system will base this cadence to detect the FXO port | (100 ~ 5000 ms) |
| Reorder Tone Spec | Frequency | Specification of the frequency of reorder tone | (300 ~ 3000 Hz) |
| | Cadence | Specification of the cadence of reorder tone. System will base this cadence to detect the FXO port | (100 ~ 5000 ms) |

## 10.11 6.IP Settings

(Need Warm-Restart) Apply Cancel

| IP Settings | |
|---|---|
| IP State: | Auto (DHCP) |
| **Public IP Address** | |
| IP/Port: | 210.62.149.181/ 5060 |
| **Current Settings** | |
| IP Address: | 10.13.6.61 |
| Subnet Mask: | 255.255.255.0 |
| Default Gateway: | 10.13.6.130 |
| **Change To** | |
| IP Address: | 10.13.6.61 |
| Subnet Mask: | 255.255.255.0 |
| Default Gateway: | 10.13.6.130 |
| **DNS Server** | |
| Primary Address: | 10.13.6.129 |
| Secondary Address: | 0.0.0.0 |

| Type | Field | Description | Default |
|---|---|---|---|
| IP Settings | IP State | The type of IP Address get:<br>Manual：User enters the assigned static IP address<br>Auto(DHCP)：Dynamic IP address from DHCP server | Manual |
| | Public IP Address IP/Port | IP Address / Port current used for this machine | |
| | Current Setting | Display the current setting (current using) IP information, including IP Address, Subnet Mask and Default Gateway. (Display only) | 192.168.0.2<br>255.255.255.0<br>192.168.0.1 |
| | Change To | Enter the information to be updated to, including:<br>1. IP Address<br>2. Subnet Mask<br>3. Default Gateway<br>(IP State must be at state “Manual”)<br>After you had filled out these parameters, click button “Apply” to activate the updated value and the system must be restarted. (Warm Start) | |
| DNS Server | Primary Address | IP Address of Primary DNS server. | 168.95.1.1 |
| | Secondary Address | IP Address of Secondary DNS server. | 0.0.0.0 |

## 10.12 7.Dial Plan

Apply Cancel

| Dialing Method | |
|---|---|
| Control : | Dialing Plan |

| Dialing Plan | |
|---|---|
| Maximum : | 100 |
| Entered : | 1 |
| List : | |

| Pattern | Delete |
|---|---|
| x | Delete |

| | |
|---|---|
| Add Dialing Plan : | |
| Delete : | |

| Dial In Rewriting Rule | |
|---|---|
| Control : | Disable |
| Capacity : | 10 |
| List : | |

| Pattern | Rewrite | Delete |
|---|---|---|

| | Pattern | Rewrite |
|---|---|---|
| Add Dialing Plan : | | |
| Delete Dialing Plan : | | |

| Digit Map | |
|---|---|
| Maximum : | 16 |
| Entered : | 1 |
| Length per pattern : | 24 |
| List : | |

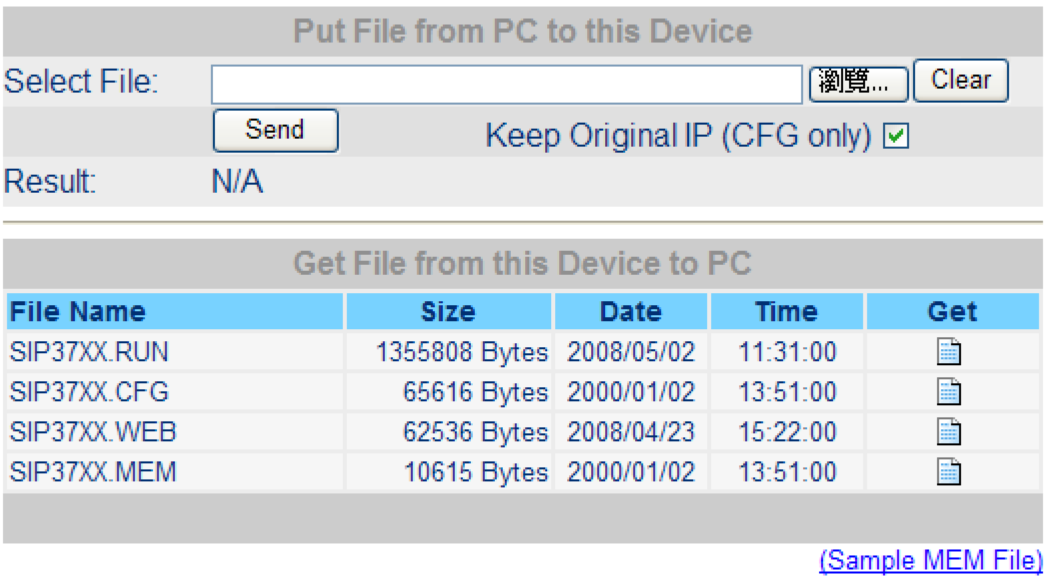
| Pattern | Delete |
|---|---|
| [0-9*#ABCD]. | Delete |

| | Pattern |
|---|---|
| Add Digit Map Item : | |
| Delete Digit Map Item : | |

| Section | Item Field | Description | Default |
|---|---|---|---|
| Dialing Method | Control | ◆ Dialing Plan: Use dialing plan rule<br>◆ Transparent: All number dialed is passed to Proxy Server<br>◆ Transparent (with Digitmap): All number dialed is passed to Proxy Server with digit map control<br>For the detail of this function, please refer to 8.2 Configuration of Dialing Plan | Dialing Plan |
| Dialing Plan | Maximum | Maximum number of entries allowed (Read Only) | 100 |
| | Entered | Number of entries of authentication entered. (Read Only) | 1 |
| | List | Pattern: Display the entries. The default value "x" means that all numbers that you dial will first go through SIP proxy. (Read Only)<br>Delete: Click Delete button to remove entry | x |
| | Add Dialing Plan | Enter numbers. Example: 050. | |
| | Delete | Enter numbers for delete. | |
| Dial In Rewriting Rule | Control | Digits dialed from MOSA 3700 can be rewrite to different digits and sent to SIP Proxy.<br>Enable/Disable | Disable |
| | Capacity | The max set of rewrite number | 10 |
| | List | List the entries of original digits and the rewrite digits<br>◆ Pattern: the pattern that user may dial<br>◆ Rewrite: the converted number if user dials the same digit in pattern column. | |

| Section | Item Field | Description | Default |
|---|---|---|---|
| | Add Dialing Plan (button) | ◆ Pattern: Add the pattern that user may dial<br>◆ Rewrite: Add the converted number if user dials the same digit in pattern column.<br>Fill in digits and click the Add Dialing button | |
| | Del Dialing Plan (button) | Fill in the Pattern digit that will be deleted and click Del Dialing button | |
| Digit Map | Maximum | Maximum number of entries allowed (Read Only) | 16 |
| | Entered | Number of entries of authentication entered. (Read Only) | 1 |
| | Length per pattern | Max digits length of each entries (Read Only) | 24 |
| | List | Display the entries (Read Only) | [0-9*#ABCD]. |
| | | Delete: Click Delete button to remove entry | |
| | Add Digit Map Item | Enter digit map pattern | |
| | Delete Digit Map Item | Delete digit map pattern | |

## 10.13 8.File Transfer



| Type | Field | Description | Default |
| --- | --- | --- | --- |
| Put File from PC to this Device | Select file | Browse (button): Select the file that will upload to this machine<br>Send (button): Execute upload action<br>Clear (button): Clear the file and path that had been input<br>Keep Original IP (CFG only): When you upload other CFG Configuration file. The IP keep intact<br>Attention: Run Cold Restart is required when .RUN and .Web file is uploaded | |
| | Result | Shows the upload status<br>Success: file is uploaded successful and take effect immediately<br>Need Warm Restart: Warm restart is required, such as file: WEB<br>Need Cold Restart: Cold restart is required, such as file: RUN<br>File ID Error: File uploaded is not for this machine.<br>N/A: No action | N/A |
| Get File From this Device to PC | File Name | Shows the file information of in this machine currently.<br>File Name | |
| | Size | File Size | |
| | Date | File date | |
| | Time | File time | |
| | Get | Select file that can be download to PC | |
| | Sample MEM file | MEM configuration file that can upload to this box without manually configure each item one by one. Please see next section | |

# 10.14 About Sample MEM File in Web

| [SIP-COMM] | | |
|---|---|---|
| ***Key Word*** | ***Value*** | ***Comments*** |
| Header-Form | = 0 | #(0/1, Standard/Compact) |
| Out-Proxy-Domain | = "outboundproxy.com" | |
| Out-Proxy-Status | = 1 | #(0/1, Disable/Enable) |
| Out-Proxy-Port | = 5060 | |
| Registrar-Domain | = "registrar.com" | |
| Registrar-Status | = 1 | #(0/1, Disable/Enable) |
| Out-Of-Band-DTMF | = 0 | #(0/1, Disable/Enable) |
| Incoming-Call-Screen | = 0 | #(0/1, Disable/Enable) |
| NAT-Keep-Alive | = 0 | #(0/1, Disable/Enable) |
| Target-The-Media | = 0 | #(0/1, SDP/Symmetric RTP) |
| Codecs-Selection | = "1111" | #(G729:1000,G723:100,G711U:10,G711A:1) |
| Codec-Priority | = 0 | #(0~23) (refer to webpage) |
| Hotline-Control | = 0 | #(0/1, Disable/Enable) |
| RTP-Base-Port | = 10000 | |
| Time-Source | = 1 | #(0/1, Registrar/NTP Server) |
| NTP-Server | = "ntpserver.com" | |
| Time-Zone | = 24 | #(0~29) (refer to webpage) |
| DayLight-Saving | = 0 | #(0/1, Off/On) |
| Register-Expire | = 60 | #(60~86400) |
| **[SIP-ENTITY]** | | |
| ***Key Word*** | ***Value*** | ***Comments*** |
| Entity-No | = 1 | |
| Entity-Control | = 1 | #(0/1, Disable/Enable) |
| CLIR | = 0 | #(0/1, Disable/Enable) |
| Public-Address | = "user@registrar.com" | |
| Default-Account-User | = "username" | |
| Default-Account-PASS | = "password" | |
| RFC-2833-DTMF | = 1 | #(0/1, Never/Negotiate) |
| Forward-Address | = "user2@registrar.com" | |
| Forward-Type | = 0 | #(0/1/2/3, None/All/Busy/No Answer) |
| **[INBOUND-PASSWORD]** | | |
| ***Key Word*** | ***Value*** | ***Comments*** |
| Channel-No | = 1 | |
| Join-SIP-Entity | = 1 | #(0 for None) |
| Control | = 1 | #(0/1/2, IN_Only/BothWay/Disable) |
| DND | = 0 | #(0/1, Disable/Enable) |
| Slience-Suppress | = 0 | #(0/1, Disable/Enable) |
| Connect-Device | = 0 | #(0/1, Phone/Fax) |
| Battery-Reverse | = 0 | #(0/1, Off/On) |
| Auto-Answer | = 1 | #(0/1/2, Disable/Enable/Enable w/Pincode) |
| Call-Waiting | = 1 | #(0/1, Disable/Enable) |
| T38-Fax | = 1 | #(0/1, No/Yes) |
| Voice-Input-Gain | = 1 | #(0~12, -6~6) |
| Voice-Output-Gain | = 1 | #(0~12, -6~6) |
| **[STUN]** | | |
| ***Key Word*** | ***Value*** | ***Comments*** |
| STUN-Server-Control | = 0 | #(0/1, Disable/Enable) |
| NAT-WAN-IP | = "223.223.223.223" | |
| STUN-Refresh-Time | = 60 | #unit: seconds |
| **[TELEPHONY]** | | |
| ***Key Word*** | ***Value*** | ***Comments*** |
| DIAL-END-TIME | = 1 | #(1~10)(refer to webpage) |
| T38-RELAY | = 0 | #(0~4)(refer to webpage) |
| VOICE-JITTER-Buffer | = 0 | #(0~23)(refer to webpage) |
| **[SIP-INBOUND-INFO]** | | |
| ***Key Word*** | ***Value*** | ***Comments*** |
| Realm | = "realm.com" | |

| [SIP-OUTBOUND-AUTH] | | | | |
|---|---|---|---|---|
| ***# format:*** | ***entity(0 for all)*** | ***realm*** | ***username*** | ***password*** |
| | 1 | "realmA" | "realmA_user" | "0000" |
| | 2 | "realmB" | "realmB_user" | "1111" |

| [SIP-INBOUND-AUTH] | | | |
|---|---|---|---|
| ***# format:*** | ***entity(0 for all)*** | ***username*** | ***password*** |
| | 1 | "1010" | "1010" |
| | 2 | "1011" | "1011" |

| [STUN-SERVER] | | |
|---|---|---|
| ***# format:*** | ***ip_address*** | ***Port*** |
| | 223.223.223.224 | 3478 |

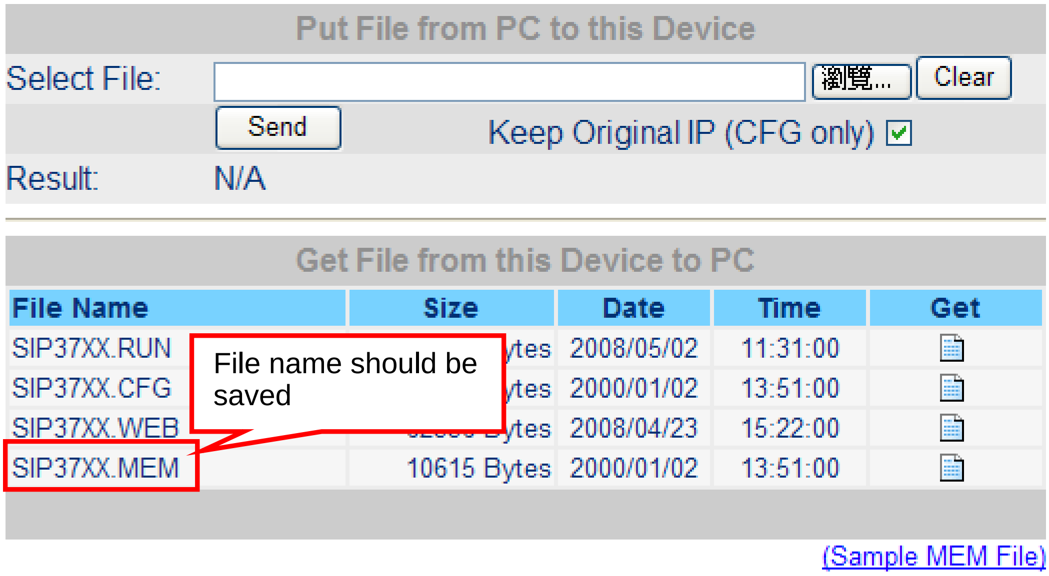
| [SIP-PHONE-BOOK] | | | | | |
|---|---|---|---|---|---|
| ***# format:*** | ***index*** | ***user_part*** | ***host_part*** | ***port*** | ***via_proxy(0:No/1:Yes)*** |
| | 8888 | "user" | "registrar.com" | 5060 | 0 |

Mem file template is the text data of your customized configuration. You can keep it for backup purpose. Configure other MOSA 3700 with this text file can save time to re-configure it. You can see comments after "#" of each command.

For this purpose, open and copy Sample MEM file at previous web page, paste to Windows Notepad (add # as remark for useless command) and save it as SIP33XX.MEM (for example, SIP3304.MEM. Depend on the original file this box has, you can see the file list on the Web Page: 8.File Transfer)

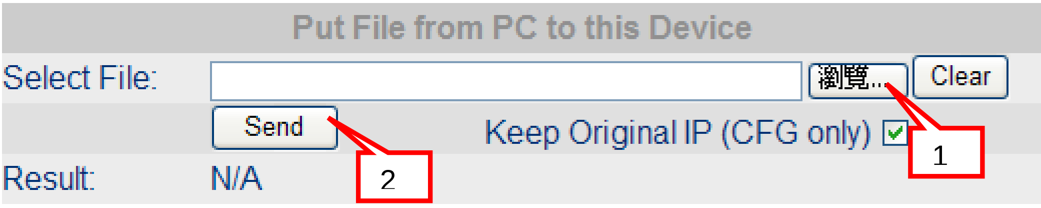
Add "#" as remark


Web Page: 8.File Transfer




This content of this filez can be modified, then upload the file to Gateway via Management Web or FTP. If there is lots of data need to create or modify, use this way can save lots of time.

From Web Page




**Note:** After you had uploaded MEM file back to gateway, for those setting that need not to restart the machine, it will take effect immediately; for those setting that need to restart the machine, you have to restart the machine to take effect.

# 11. Appendix

## 11.1 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.

**Note**: If Dialing Mode is configured to Transparent, (refer to section 8.2 Configuration of Dialing Plan) then all digits you dial is passed to SIP Proxy and all Phone-Set Command is disabled.

| 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) |
| 11 | DND | Do not Disturb, this line accept dial out call only. All incoming call is terminated. 0 : Disable ; 1: Enable |
| 12 | SIP Forward State | 0 : Disable ; 1: Enable; 2: Busy; 3: No Answer |
| 13 | SIP Forward To Number | The SIP number that this line will forward to. The Forward To address is "key in phone-set |

| | | |
|---|---|---|
| | | number@SIP proxy registered". For example, 73796@fwd.pulver.com, 73796 is the number you key-in by phone-set. fwd.pulver.com is the registered proxy of this gateway. |
| 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 |
| 17 | Register or De-register (quit) the SIP Entity registration | 0: De-Register; 1: Register |
| 40 | Listen for the IP Address | (Ending ”#” is not required) |
| 41 | Listen for the Subnet Mask | (Ending ”#” is not required) |
| 42 | Listen for the Default Gateway | (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) |
| 95 | Region ID | 2 digits |
| 97 | Reset unit to Factory Default values | 1: reset all; 2: keep IP; 3: region specific |
| 98 | System Warm Restart | 1: do it |

## 11.2 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 | FastEthernet 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 busytone cadence |
| Probe-remove | stop probe busytone cadence |
| Reload | Halt and perform cold start |
| Restart | Halt and perform warm start |
| Show | Show running system information |

Global Mode

| | |
|---|---|
| Dbflush | DataBase 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 |
| regional_id | Set regional id |
| service_port | Set service port number |