FXSO GATEWAY
In SIP
User Manual
(1FXS/1FXO or 2FXS/2FXO ports)
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4.2.28 [auth]
Steps in configuration

**STEP 1  
Start Up**

To check out the peripheral equipments and understand the feature of this gateway. Please read this step very carefully before starting the configuring.

**STEP 2  
How to Setup and Connect basically**

Connecting the gateway and computer to start configuring by WEB GUI.

Setting the ip address for this gateway to make sure that it could connect with the internet.

Setting the configurations of dialing, including the Peer-To-Peer, GK mode and how to set these tables to make calls by this gateway easily. The other configurations of make call will be discussed in this step.

**STEP 3  
Advanced**

Advanced configurations and special functions of this gateway. Using the WEB GUI to show how to set this table and explain the meaning of these tables.

**STEP 4  
Command List**

To explain the meaning of the command in the command line interface and example the usage of the command.

To get more usages or configuration in this step and study about the command line configuration.
1. Start Up

1.1 Introduction

SIPFXSO 1S10/2S2O is a one-port/two-port FXS+FXO gateway. It supports an innovative intelligent call routing function that transparently routes calls to destination either through PSTN or Internet.

SIPFXSO 1S10/2S2O provides voice over IP and FAX over IP services for ITSP/ISP Internet Telephony Services Provider and Office/SOHO IP-PBX application.

**Application Architecture**

- **FXO ports can connect with PSTN Line or Extension Line of PBX**
- **FXS ports can connect with Phone Set or Trunk Line of PBX**
1.2 Features and specification

Features
- IETF RFC 3261
- Automatically Dial Path Selection (IP or PSTN)
- PSTN Line switch to telephone set when power is failure
- PPoE support
- Behind NAT router or IP sharing device
- DNS server inquiry
- Provide Peer-to-Peer Mode (Non SIP Proxy needed) selection
- E.164 Dial Plan
- TFTP/FTP software upgrade
- Remote configuration/ reset
- LED indication for system status
- Support Fix IP and DHCP

Audio feature
- Codec -- G.711 a/µ law, G.723.1 (6.3kbps), G.729, G.729A
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Completed voice band signaling support
- Provide In-band or RFC2833 DTMF generation/detection
- Provide call progress tone

Management Feature
- TELNET/Console port and Web Browser configuration

Certification
- UL, CE, FCC

FXS Features
- 2-wire loop start
- Programmable On-Hook voltage, Ring voltage/Cadence/Frequency, Loop current
- Line polarity reversal generation
FXO Features
- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)
- PSTN polarity reversal detection
- Disconnect tone detection
- Asking ping function with the incoming calls from PSTN side
- Record and analyze the Tone from PSTN side

Environmental
- Operation temp: 0°C to 40°C
- Humidity: 10% to 90% (Non-condensing)
1.3 Accessories and equipment

- The voice gateway in 2 FXS and 2 FXO ports or 1 FXS port and 1 FXO port models and two RJ-45 connector (WAN and LAN).

- The AC adapter.

- The CD of user manual.

- The connection cable in RS-232 interface.
1.4 Appearance

*Front panel:* The LED lights provide related system messages of the gateway.

**SIPFXSO 1S10**

**SIPFXSO 2S20**

*Power:* Light on means Gateway is power on, and vice versa.

*TEL:* Light on means the line is in use (off-hook), and vice versa.

*LINE:* Light on means the line is in use (off-hook), and vice versa.

**Status:**
1. LED light on means Gateway has successfully registered to Proxy when it is in the Proxy Mode.
2. LED flash means Gateway is not registered to the Proxy when it is in the Proxy Mode.
3. Or when Gateway is in downloading mode, LED should be flash as well.
4. LED light off means Gateway is in Peer-to-Peer Mode.

**Ready:**
1. Light on and slow flash means Gateway is in normal mode.
2. Light on and fast flash means Gateway is in downloading mode.

**WAN: Connected to Public Ethernet**
1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted / received.

**LAN: Switch to another device, such as PC**
1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted received.

*Back panel:*
1. Ethernet Port
   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T. The pin-out is as following:
   ![Ethernet Port Diagram]

   PIN 1, 2: Transmit
   PIN 3, 6: Receive

2. COM:
   RS232 console port (DB-9pin male connector)
   Note: use straightforward cable to connect to your computer.
   ![COM Connector]

   PINOUTS

<table>
<thead>
<tr>
<th>Pin</th>
<th>Name</th>
<th>Dir</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>RXD</td>
<td>←</td>
<td>Receive Data</td>
</tr>
</tbody>
</table>
3. **TEL:**
   RJ-11 connector, FXS interface is for connecting the analog phone sets or trunk line of PABX.

4. **LINE:**
   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.

5. **12V DC:**
   Input AC 100V~120V; output DC12V.
2. How to Setup and Connect basically

2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 RAM, Windows 98 or above.
   (b) Ethernet card or COM port
2. One standard straightforward RS-232 cable (female connector to Gateway side).
3. Analog telephone sets or the PBX trunk Lines.
4. PBX extension Lines or PSTN Lines.
5. Software tools – Hyper Terminal, TELNET, Web Browser.
2.2 IP Environment Setting

User must prepare a valid IP address, complied with IP Network, for Gateway’s proper operation.

For testing the validation of chosen IP address, using the same IP configuration in other PC or Notebook, and then try to connect to Public Internet (go to well-known website, receive Internet mail, or ping a specific public IP address). If it works, use the same IP address and network configuration for Gateway.
Please follow up the step for the configuration of your computer or notebook.

2.2.1 For Windows 2000/NT

Please make sure that the network interface of your computer is working fine and the cross over line (RJ-45) is connecting with the computer correctly or you could use a hub to connect with your computer and this gateway. Turn on your computer and configure the network parameter as follow:

1. Go to the start menu and enter the setting area. Click control panel.

2. Enter the network configuration.
Figure 2.1: Network Configuration

3 Select the **Property** of the LAN card.

4 Setup the ip address, subnet mask and default gateway as below:

![Network Configuration Settings](image)

Figure 2.2: Configure the network

5 Click OK after you finished the network setup.

The default ip address, netmask and default gateway address of the gateway is 10.1.1.3, 255.0.0.0, 10.1.1.254.
2.3 Network configurations in your gateway

1 Key in the ip address of the gateway (http://10.1.1.3) with the browser. (see figure 2.3)

Figure 2.3: WEB Browser

2 After key in the ip address, you have to enter the user name and password to enter the WEB configuration. **(Username: root ; No password)** (see figure 2.4)
Figure 2.4: Login the username and password

3 You will enter the main page of the configuration after key in the login name and password correctly: (see figure 2.5)

Figure 2.5: The main WEB configuration

4 Press the **Network Interface** to configure the networking of your gateway. (see figure 2.6)
Figure 2.6: The Network Interface

2.3.1 Static ip address

1. Please get the correct ip address, netmask and default gateway address from your ISP first. Press the OK button if you finished. (see figure 2.7)

Figure 2.7: Configure the static ip address
2 The 37 will auto commit data after pressing OK. This function only exists in 370x with the version sipfxso.106 and above.

3 **Press the reboot** if you want the configuration executed. (see figure 2.8)

![Figure 2.8: Reboot the system](image)

### 2.3.2 DHCP mode

1 Enable the DHCP if you are using the cable modem or DHCP server. (see figure 2.10)
Figure 2.10: Enable the DHCP function

2. The 37 will auto commit data after pressing OK. This function only exists in 370x with the version sipfxso.106 and above.

2.3.3 PPPoE mode

1. Switch to the PPPoE mode and press the “OK” button. Press the Network Interface button after the “OK” button. (see figure 2.11)
Figure 2.11: Switch to the PPPoE mode

2. Enter the Login account and password. Press the “OK” button if the configuration is finished. (see figure 2.12)

Figure 2.12: Enter the Account and password

3. The 37 will auto commit data after pressing OK. This function only exists in 370x with the version sipfxso.106 and above.
2.4 Making a VoIP Call

There are two modes that you could configure the gateway for making VoIP calls. One is the Peer-to-Peer mode, another is Proxy mode. The configurations and functions are different. Please make sure about the mode you want and follow up the step to configure your gateway.

2.4.1 Configure the gateway into the Peer-to-Peer mode

1. Enter the SIP Configuration table and change the mode to Peer-to-Peer.
Define the port numbers whatever you like. Press the “OK” button if the configuration is all finished. (see figure 2.13)

![Figure 2.13: Configure the Peer-to-Peer mode](image)

2. Enter the Phone Book configuration table and configure the name, ip address and phone number of the destination. (see figure 2.14)
Figure 2.14: Phone Book

【Example】

Figure 2.15: The example of Phone Book configuration

The name of the destination: **test**

The E164 number (phone number) of the destination: **123**

The ip address of the destination: **10.1.1.100**

Drop prefix: **Enable** – **The e164 number you define will be deleted**
Disable – The e164 number you define will be kept
Insert prefix: To add a number you define in this table

Press the “Add Data” button when you finished, and the new table will display on the first index if you press the Phone Book configuration button.

4 The 37 will auto commit data after pressing OK. This function only exists in 370x with the version sipxso.106 and above.

![Phone Book Configuration Menu]

Figure 2.16: To show the Phone Book record

Phone Book is only for the Peer-to-Peer mode and could support forty records.

【The application in the drop and insert function】

<table>
<thead>
<tr>
<th>Input (E164)</th>
<th>Drop</th>
<th>Insert</th>
<th>Output</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>Disable</td>
<td>X</td>
<td>100</td>
</tr>
<tr>
<td>200</td>
<td>Disable</td>
<td>0</td>
<td>0200</td>
</tr>
</tbody>
</table>
2.4.2 **Configure the gateway into the Proxy mode**

1. Enter the SIP Config table and change the mode from Peer-to-Peer to Proxy.

   To change the Proxy information from your service provider (Ex: The Proxy IP, Domain and Line numbers). (see figure 2.17)

   ![SIP Configuration Settings](image)

   Figure 2.17: Configure the Proxy info

2. Press the OK button that is on the bottom of this page to save the configuration.

3. **Switch to the Security Config page and put the user account and password** in the correct table. Please get this info from your ITSP. Press the OK button if the configuration is finished. (see figure 2.18)
Figure 2.18: Configure the Security info

4 Press the Commit Data and **Reboot System** buttons when you finished the configuration.
3. Advanced

There are too many advanced commands for the advanced users. The following chapters are based on the application layer. Please get the info what you need. If you need the command, please watching the chapter of Command Line Interface.

3.1 Network Interface

Users have to configure the Network configurations in this page. This gateway will be work while it is connecting with the internet network. Please get more info from the following descriptions. (see figure 3.1)

![Network Interface](image)

**Figure 3.1:** Network Interface

- **IP Address** – Define the ip address for your networking if it is the fixed ip. Please get this info from your ISP.
- **Subnet Mask** – Define the mask address for your networking. Please get this info from your ISP.
- **Default Routing Gateway** – Define the default gateway for your networking. Please get this info from your ISP.
- **IP Mode** – To configure the fixed or dynamic ip address for this unit.
Please configure to PPPoE if the ADSL is using the PPPoE type.

◆ HTTP Port – To configure the HTTP port for access this unit from the remote side.
◆ DNS primary – To configure the first ip address for the DNS server.
◆ DNS secondary – To configure the second ip address for the DNS server.
◆ SNTP – Enable the SNTP server registering function if user wants to get the correct time from the Command Line Interface.
◆ SNTP Server Address – Enter the correct ip address of the SNTP server or get the incorrect time from the Command Line Interface.
◆ GMT – Configuring the time area for the time display in the Command Line Interface.
◆ IP Sharing – Enable this function if the gateway is behind the IP sharing device.
◆ IP Sharing Server Address – Enter the WAN IP address of the IP sharing device if it is the fixed ip.
◆ PPPoE User Name – To configure the user name for the PPPoE connection.
◆ PPPoE Password – To configure the password for the PPPoE connection.
◆ PPPoE IP Address – In the PPPoE mode, this table will show the ip address that this unit gets from the ISP.
◆ PPPoE Destination – In the PPPoE mode, this table will show the default gateway address that this unit gets from the ISP.
◆ PPPoE DNS primary – In the PPPoE mode, this table will show the DNS ip address that this unit gets from the ISP.
◆ After Remote Host Disconnection – This unit will reboot and re-connect to the ISP.
3.2 SIP Config

This WEB page will help user to configure the information about the dial mode, SIP information and the ports for the communication. Please get more info about this configuration from the below detail descriptions. (see figure 3.2)

Figure 3.2: SIP Configuration

- **Mode** – Pick up the calling mode for this gateway.
  - Peer-2-Peer : It only supports the peer-to-peer mode and users have to define the phone book for this mode.
  - Proxy : Users have to register on the Proxy if users picked up this option.
- **Primary Proxy IP Address** – Enter the proxy ip if users pick up the proxy mode.
- **Primary Proxy port** – Set Proxy port for 380x to send message, default value is 5060, if there is no special request of Proxy server, please don’t change this value.
- **Secondary Proxy IP Address** – Set secondary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to Network Interface). When 380x fail to register to primary Proxy, it will try to register to secondary Proxy, when it fails again, it will retry to register to Primary Proxy.
◆ Secondary Proxy port – Set the secondary proxy port for every SIP message, default value is 5060.
◆ Outbound Proxy – This version could support the outbound proxy, users could define the ip address or domain name in this table.
◆ Outbound Proxy port – Set outbound Proxy port for 380x to send message, default value is 5060, if there is no special request of Proxy server, please don’t change this value.
◆ Prefix String – Users could define this if the registration name was a phonetic alphabet not the numbers.
◆ Line 1 Number – The phone number of the Tel 1.
◆ Line 2 Number – The phone number of the Line 1.
◆ Line 3 Number – The phone number of the Tel 2.
◆ Line 4 Number – The phone number of the Line 2.
◆ SIP port – Users could change the sip port of this unit for the registration.
◆ RTP port – Users could change the beginning RTP ports in this table.
◆ Expire – Users could change the expire time for the register message sending.
3.3 Security Config

Some proxy will include the security policy. The endpoint may need the user account and password for the registration. If these are necessary, users could put the correct account and password in the correct table. (see figure 3.3)

![Figure 3.3: Security Configuration](image)

- Line 1 Account – The user name for the line 1 account.
- Line 1 Password – The password for the line 1 account.
- Line 2 Account – The user name for the line 2 account.
- Line 2 Password – The password for the line 2 account.
- Line 3 Account – The user name for the line 3 account.
- Line 3 Password – The password for the line 3 account.
- Line 4 Account – The user name for the line 4 account.
- Line 4 Password – The password for the line 4 account.
3.4 Line Configuration

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table (see figure 3.4)

![Figure 3.4: Line Configuration](image)

- **Type** – Show the type of this port. There are only two types of this gateway. One is FXS type another is FXO type. It couldn’t be changed.
- **Hunting Group** – Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group.
- **Hotline** – Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- **FWD Type** – This version could support the call forward with the Proxy. Users could define the forward type in this table. The call forward could only support the FXS port as so far.
  - **Disable** – This selection will make the forward function disabled.
  - **Unconditional** – This selection will enable the un-conditional forward function. Every call will be routed to another destination whatever the port is busy or not.
  - **Busy** – The selection will enable the busy forward function. Every call will be routed to another destination if this port is busy.
The busy forward couldn't be work if users define the hunt group function. This will be work without the hunt group. Please make sure about that the group number is different before you enable the busy forward function.

- No Answer Forward – This selection will enable the no answer forward function. Every call will be routed to another destination if the call didn’t answer by someone.

The no answer forward function should be used with the forwardtime command (in the System configuration table). The range of the forward time is 5 to 65535. If the function is enabling, the No Answer forward number is 123, and the forward time is 5 seconds. The call will be forwarded to the destination with 123 numbers when the origin port didn’t answer the call.

- Fwd Number – Please put the number for the call forwarding in this table.

The no answer forward function could support the IP or PSTN side. If users want the calls routed to the PSTN side, please define the routing table for some special number.

- Registration – To show the gateway registered on the GK or not.
- Status – To show the port is busy or ready.
3.5 System Configuration

There are some parameters in the system configurations, please get more detail as following. (see figure 3.5)

Figure 3.5: System Configuration

- Keypad type – There are two types for the Keypad. On is the In-Band type, another is the RFC2833 type. User could define the keypad type for the dialing.
- Inter Digit Time – It’s the time for the time out during the dialing numbers.
- Forward Time – It’s the time for the no-answer-forward. Users have to configure it with the no-answer-forward function.
- Ring Time – FXO will detect the ring tone according this time.
- End of Dial – It will transfer the digit “#” if this function is disabling.
- Hardware Type – It’s for the hardware issue.
3.6 Voice Setting

User could define some parameters about the voice in this voice-setting page. (see figure 3.6)

![Voice Setting Configuration](image)

Figure 3.6: Voice Setting

- **Codec Priority**: It’s for the codec setting. User could use the codec, which they want by the setting.

---

There are two firmware versions for the 2S2O. One is for G.723 only, another is for G.729 only. It with no meanings if users use the G.723 version and configure the codec in G.729. User could check out the firmware version first and configure the codec second. 1S1O could support all the codec at the same times.

- **Frame Size**: It’s the packet size for all codec. It will take more bandwidth if users configure the packet size in the minimum value.
- **G723 Silence Suppression**: For the VAD and CNG function support.
- **Volume**: To adjust the gain of the output, input and dtmf.
- **Echo Canceller**: To enable the echo cancellation function.
- **Jitter Buffer**: To adjust the Jitter Buffer size to avoid the packets losing.
A large jitter buffer causes increase in the delay and decreases the packet loss. A small jitter buffer decreases the delay but increases the packet loss. The size of the jitter buffer depends on the condition of the network, which varies with time. Typically the packet loss should be less than 10% for a good quality of speech.
3.7 Phone Pattern

The FXSO could generate some tones, such like the busy tone, dial tone, ring back tone and second dial tone…etc. Users could adjust these tones or get the detail info from this page. (see figure 3.7)

Figure 3.7: Phone Pattern

- **Ring Cadence** – Adjust the pattern for the cadence of the Ring tone. Including the Frequency, On time, Off time and the gain level.
- **Ring Back Tone** – Adjust the pattern for the Ring Back Tone. Including the High, Low frequency, High, Low Level, and the On, Off time.
- **Busy Tone** – Adjust the Busy Tone.
- **Dial Tone** – Adjust the parameters for the dial tone playing.
- **2nd Dial Tone** – Adjust the second dial tone.
- **Flash Frequency** – Adjust the High, Low frequency for the Flash. (unit : ms)

The Flash will affect the call transfer function. Users could define a range for the low and high flash time. Before you change this range, please make sure about that the analog phone set could generate the flash signal in this range. The FXS will know that is the flash signal if the flash time will in this range. For the FXO, it will generate the flash between this range. For example, if the low flash time is 400ms and the high is 800ms, the FXO will generate the 600ms for the flash time.
3.8 Phone Book

The Phone Book configuration is only support the gateway in Peer-to-Peer mode. Please refer the chapter 2 about the Peer-to-Peer mode. (see figure 3.9)

![Phone Book Configuration Screen]

Figure 3.9: Phone Book

- Index – The list number of the Phone Book.
- Name – The name for this contact number.
- E164 – The dialing number for the calling side.
- IP Address – The destination IP address for this phone number.
- Port – The call signal port of the destination.
- Drop – Support the drop function. Enable is for enable this drop function; Disable is for disable this drop function. The Drop Prefix will drop the E164 number, which you had configured in the E164 table.
- Insert – Support the insert digits function.

1. It will be the drop function if user enable the Drop Prefix function and put nothing into the Insert Prefix table.
2. It will be the insert function if user disable the Drop Prefix function and put the digits into the Insert Prefix table.
3. It will be the replace function if user enable the Drop Prefix function and put the digits into the Insert Prefix table.
◆ Add Data – Press this button if users fill the entire information table above.
◆ Delete Date – If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted.
3.9 Prefix Configuration

The Prefix functions are using the drop and insert function (see figure 3.6).

Figure 3.10: Prefix Configuration

- Index – The list number of the Phone Book.
- Prefix – The prefix number of the whole numbers that could be into this gateway
- Drop – The drop function. Enable this function by the Enable button; Disable this function by the Disable button.
- Insert – The insert function. Users could enter the digits that you want to insert in this number.
- Add Data – Press this button if users fill the entire information table above.
- Delete Data – If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted.

This function is just like the Phone Book configuration. But it will make the drop and insert function in the GK routed mode. All the numbers into this gateway will check out the prefix table first and find out the destination in the Routing Table.

There is an example about the configuration, please follow up these steps.
1 Press the Prefix Configuration button to enter the configuration table (see figure 3.10)

2 Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert (see figure 3.11)

![Prefix Drop/Insert Configuration](image)

Figure 3.11: Configure the Prefix Table

The usage is as same as the drop, insert function of the Phone Book.

<table>
<thead>
<tr>
<th>Input (Prefix)</th>
<th>Drop</th>
<th>Insert</th>
<th>Output</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>Disable</td>
<td>X</td>
<td>100</td>
</tr>
<tr>
<td>200</td>
<td>Disable</td>
<td>0</td>
<td>0200</td>
</tr>
<tr>
<td>300</td>
<td>Enable</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>400</td>
<td>Enable</td>
<td>500</td>
<td>500</td>
</tr>
</tbody>
</table>
3 Press the Prefix Configuration button to reload the configuration table (see figure 3.12)

![Figure 3.12: Show the added table](image)

4 Please Commit it and Reboot the system if the configuration is finished.
3.10 Routing Table

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table (see figure 3.13). In fact, there are three directions of the incoming calls (from IP, FXS and FXO side). The explanation of the default routing is as below:

<table>
<thead>
<tr>
<th>The location with the incoming calls</th>
<th>The location with the destination</th>
<th>The explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP (Default)</td>
<td>FXS</td>
<td>The destination will be the FXS port when the calls from the IP side without any define rules.</td>
</tr>
<tr>
<td>FXS (Default)</td>
<td>IP</td>
<td>The destination will be the IP side when the calls from the FXS port without any define rules.</td>
</tr>
<tr>
<td>FXO (Default)</td>
<td>IP</td>
<td>The destination will be the IP side when the calls from the FXO port without any define rules.</td>
</tr>
</tbody>
</table>

*Figure 3.13: Routing Table Configuration*

◆ **Index** – The list number of the Route Table.
Default – For change the default setting. Users have to pick the direction for the default setting changed.

Prefix – The prefix number for the dialed digits. The call will be followed this route table if the prefix number was matched.

Destination – To decide the destination for this route table.

E.164 – The E.164 number of the destination.

Mini Digits – The mini digits requirement for this route table.

Max Digits – The max digits requirement for this route table.

Hunt Method – Enable the Hunt Group function and pick up the hunt type.
  ■ NONE : Disable the Hunt Group function.
  ■ GROUP : The Hunt Group function will working for the same group.
  User could configure the group in the Line Configuration table.
  ■ ALL : The Hunt Group will working for the same type.

None – Disable this function

Group – The call will search other ports to be the destination with the same
group if the origin destination is busy.

All – The call will search other ports to be the destination with the same type if
the origin destination is busy.

About the Group setting, Please get more info from the Line Configuration.

Add Data – Add a new record for the route table.

Delete Data – Delete a record for the route table.

Change Default – Change default route table.

3.10.1 Change the default routing

Please follow up the steps if you want to change the default routing:

Pick up the side for the incoming calls and define the destination of this side.

Press the Change Default to save the data. (see figure 3.14)
Figure 3.14: Change the Default Setting

2 The default setting is changed after you press the Change Default button. Please press the Routing Table button again to show the new setting. (see figure 3.15)

Figure 3.15: The Default Setting Changed

3 Please Commit it and Reboot the system if the configuration is finished.
3.10.2 Add a new Routing Table

1. The default setting is changed after you press the Change Default button.

Please press the Routing Table button again to show the new setting. (see figure 3.16)

2. Press Add Data button to save the configuration and press the Routing Table button again to reload the configuration. (see figure 3.13)
Figure 3.17: New Special Routing

The explanation of figure 3.17 is as below:

When the user dial 8 with the first digit of the numbers from FXS or IP side. And the numbers you dial is between 1 and 8 digits. If this number matches the rule, it will be transferred to the FXO port whose E164 number is 1003.

3 Please Commit it and Reboot the system if the configuration is finished.
3.11 IP Packet ToS

The Type of Service should be worked with the network router. The router will check all the packets if it supports the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure these values to make the packets with the correct values for the TOS service from the gateway. (see figure 3.19)

Figure 3.19: ToS

According to the RFC 1349 document, the ToS value as following:

1000 – minimize delay
0100 – maximize throughput
0010 – maximize reliability
0001 – minimize monetary cost
0000 – normal service

These values are the Binary format. Please change to the Decimal and put these values in to the correct table.
3.12 **Password**

There are two-login accounts in this unit. One is the account root another is administrator. **The default setting for these two accounts are empty.** Users could define the passwords for these two accounts. Please get more info from the following description. (see figure 3.20)

![Password Configuration](image)

**Figure 3.20: Password**

- **root** – The password for the root account.
- **administrator** – The password for the administrator account. This account couldn’t upgrade the 2M and boot rom file.
- Current Password – Enter the original password for the account.
- New Password – Enter the new password for the account.
- Confirm New Password – Enter the new password again.
- Change – This button will make the configurations saved and next time login will need the new password.
- Abort – Abort the configuration of the password changing.

Please remember the password you configure for the account. It will be more difficult to access it if you forgot the password.
3.13 RTP Pay Load Type Configuration

There are more types for the RTP Payload. This web page could support users define the Payload Type for some special payload. (see figure 3.21)

![Figure 3.21: Pay Load Type Configuration](image)

- RFC2833 Payload Type – To define the payload type for RFC2833 type.
- DTMF Payload Type – To define the payload for the DTMF type.
- FAX Payload Type – To define the payload for the FAX type.
- FAXByPass Payload type – To define the payload for the FAX by Pass type.
- MODEMByPass Payload Type –To define the payload for the Modem by Pass type. (This is no use for the hardware as so far.)
- Redundancy Payload Type – To define the payload for the Redundancy type.
- MODEMRelay Payload Type – To define the payload for the FAX by Pass type. (This is no use for the hardware as so far.)

The modem function couldn’t be work in this kind of VoIP unit.
3.14 **Version and Information**

Users could get more detail about the software version for all the parts in this web page. (see figure 3.22)

![Figure 3.22 Version and Information](image)

- **Boot Rom** – The version of the Boot Rom layer.
- **Application Rom** – The version of the Application Rom layer.
- **DSP Application** – The version of the DSP Application Rom layer.
- **DSP Kernel** – The version of the DSP Kernel layer.
- **DSP Test Code** – The version of the DSP Test Code layer.
- **Greeting** – The version of the Greeting file.
- **ASK Pin** – The version of the ASK Pin file.
3.15 ROM Upgrade

User could update the firmware just by the web configuration interface. There are two type for the upgrading procedure. One is using the TFTP server, another is using the FTP server. Please follow the step to update the gateway firmware version. (see figure 3.23)

Figure 3.23 ROM Upgrade

- TFTP/FTP server IP Address – Put the ip address of the TFTP or FTP server in this table.
- Target File name – Put the target file name in this table.
- Method – There are two upgrade methods for the upgrade procedure. One is TFTP and another is FTP. Please change to correct method for the upgrading.
- FTP Login – Please enter the login name and password for the FTP upgrade method. This is necessary if user change the method to the FTP.
- Target File Type – Please pick up the correct file type for upgrading. If the file name and the file type is unconformable, the upgrade procedure will be failed.
- OK – Press the OK button if all the info above are correct. The unit will start to download the firmware file from the TFTP or FTP and write to the flash after the downloading.
【Updating the firmware by the FTP server】

1. Pick up the “Rom Upgrade” button to enter the upgrading web page and switch to the FTP method. (see figure 3.24)

![ROM Configuration](image)

Figure 3.24: ROM Upgrade for FTP

2. Key in the IP address, the login name, password of your FTP server and the correct file name, file type. (see figure 3.25)

![ROM Configuration](image)
Figure 3.25: FTP information

Please pay more attentions about the red blank. The Target File Type has to be matched with the Target File name. Please put the correct info about the Target file in this table.

3 Press the OK button to execute the upgrade procedure.

4 Please press the “Reboot System” button to make it reboot.

【Updating the firmware by the TFTP server】

1 Downloading the TFTP program from our web site and install it first.
   Executing the TFTP program before you want to use the TFTP upgrade method.

2 Pick up the “Rom Upgrade” button to enter the upgrading web page and switch to the TFTP method. (see figure 3.26)

Figure 3.26: ROM Upgrade for TFTP
3. Key in the IP address of your TFTP server, pick up the file type for your upgrade file and the correct file name for upgrading. (see figure 3.27)

![Figure 3.27: TFTP information](image)

4. Press the OK button to execute the upgrade procedure.

Please pay more attentions on the file name you used. The file name with the prefix “2m” is the complete firmware, it will take more times for the downloading and upgrading. About the file name without the prefix “2m”, it’s only for the application layer firmware. In fact, the latest version firmware is only changed in the application layer.

Please use the flash clean web to make all the configuration back to the default setting if the upgrade procedure was finished.

5. Please press the “Flash Clean” button when the procedure is finished.

6. After pressing the “Flash Clean” button, please press the “Reboot System” button to make it reboot.
3.16 Flash Clean

Users could make all the configurations back to the default setting by this button. The password of the account and the networking configuration couldn’t be back to the default setting by this command. (see figure 3.28)

Figure 3.28: Flash Command
3.17 **Reboot System**

This web page will restart the whole system. This is the necessary step for the changing the configurations and makes it executed. (see figure 3.30)

![Reboot System Diagram]

Figure 3.30: Reboot System