

# OvisLink

## ***VoIP-800/400***

***VoIP Gateway***

**User's Manual**

September , 2003  
Copyright © OvisLink Corp.

## **FCC Notice**

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed in accordance with the instruction manual, may cause harmful interference to radio communication. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at the user's own expense.

## **CE Declaration of conformity**

This equipment complies with the requirements relating to electromagnetic compatibility, EN 55022 Class A for ITE and EN 50082-1. This meets the essential protection requirements of the European Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

## **Rack Mount Installation Instruction**

### (a) Elevated Operating Ambient Temperature:

If installed in a closed or multi-unit rack assembly, the operating ambient temperature of the rack environment may be greater than room ambient. Therefore, consideration should be given to installing the equipment in an environment compatible with the manufacturer's maximum rated ambient temperature.

### (b) Reduced Air Flow

Installation of the equipment in a rack should be such that the amount of air flow required for safe operation of the equipment is not compromised.

### (c) Mechanical Loading

Mounting of the equipment in the rack should be such that a hazardous condition is not achieved due to uneven mechanical loading.

### (d) Circuit Overloading

Consideration should be given to the connection of the equipment to the supply circuit and the effect that overloading of the circuits might have on overcurrent protection and supply wiring. Appropriate consideration of equipment nameplate ratings should be used when addressing this concern.

### (e) Reliable Earthing

Reliable earthing of rack-mounted equipment should be maintained. Particular attention should be given to supply connections other than direct connections to the branch circuit (e.g., use of power strips).

## **Trademarks**

All brand, company and product names are trademarks or registered trademarks of their respective companies.

## **How to Use This Manual**

This manual was designed for the non-technical users of the Internet telephony gateway (VOIP-GATEWAY). It contains information about the functions operation and basic instructions for its installation.

### **Chapter 1: Overview**

This chapter provides conceptual overview and description of the VOIP-GATEWAY, explanation of how the VOIP-GATEWAY interacts with each for enabling VoIP services, and an overview of the required tasks

### **Chapter 2: Installing the VOIP-GATEWAY**

This chapter describes required steps to properly and safely install and configure the VOIP-GATEWAY on your network

### **Chapter 3: VOIP-GATEWAY Concepts**

This chapter gives information about the VOIP-GATEWAY technology and describes basic concepts using VOIP-GATEWAY for providing telephony services over IP network

### **Chapter 4: Configuring VOIP-GATEWAY from Web Browser**

This chapter explains procedures for configuring the VOIP-GATEWAY from a web browser

### **Chapter 5: Making a Call with VOIP-GATEWAY**

This chapter explains how to make Internet calls from telephony devices connected to the VOIP-GATEWAY directly or indirectly.

### **Chapter 6: Application Samples**

This chapter provides some application examples and information on how to configure the VOIP-GATEWAY under these samples.

### **Chapter 7: Troubleshooting Tips**

This chapter provides troubleshooting information in case the user has problems installing or maintaining the VOIP-GATEWAY.

## **Important Safety Instructions**

Before you plug the VOIP-GATEWAY into an electrical outlet, carefully read all the installation instructions in Chapter 2.

For your own safety and the safety of your equipment, always take the following precautions:

- Follow instructions and warnings in the documentation.
- Never push any object through the fan vent or other openings in the equipment. Such action may produce a short circuit, causing fire, electric shock, or equipment damage.
- Keep the VOIP-GATEWAY away from all chemicals and sources of liquids.
- “CAUTION – Always disconnect all telephone lines from the wall outlet before servicing or disassembling this equipment.”
- “CAUTION – Do not connected the TNV-2 connectors ( P1, P2 ) to wall TNV outlet.”
- “CAUTION – To reduce the risk of fire, use only No. 26 AWG or larger, telecommunication line cord.”

## **Warning**

- Connection of the RJ45 connector from an VOIP-GATEWAY to TNV circuits can cause permanent damage to the VOIP-GATEWAY.
- Incorrectly connecting telephony devices to the RJ11 port on the Telephony Interface Module can cause permanent damage to the module.

## Documentation Abbreviations

Throughout this guide, the user will come across a number of abbreviations that are common throughout the industry. The user should be familiar with the following abbreviations:

ATPM	Address Translation and Parsing Manager
CLI	Command Line Interface
DSP	Digital Signal Processor
DTMF	Dual Tone Multi-Frequency
E&M	Ear & Mouth
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
H.323	ITU specification for multimedia transmission over IP networks
ICMP	Internet Control Message Protocol
IMTC	International Multimedia Telecommunications Consortium
IP	Internet Protocol
VOIP-GATEWAY	Internet Telephony Gateway
KTS	Key Telephone System
LAN	Local Area Network
NVS	Non-Volatile Storage
LED	Light Emitting Diode
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RTP	Real-Time Transport
TCID	Telephony Channel Identifier
TFTP	Trivial File Transfer Protocol
TIM	Telephony Interface Modules
TNV	Telephone Network Voltage
UDP	User Datagram Protocol
UTP	Unshielded Twisted Pair
VAD	Voice Activity Detection
WAN	Wide Area Network

## Notation Conventions

Throughout this guide, different type styles and characters are used. These serve a variety of purposes as described below:

Convention	Description
<b>boldface</b>	Commands and keywords are in <b>boldface</b> .
<i>italic</i>	Arguments for which you supply values are in italics.
courier	Messages that the VOIP-GATEWAY CLI displays are in plain courier font.
[ ]	Elements in square brackets are optional.
{ x   y   z }	Alternative but required elements are grouped in braces ( { } ) and separated by vertical bars (   ).
[ x   y   z ]	Optional alternative keywords are grouped in brackets ( [ ] ) and separated by vertical bars (   ).
String	A non-quoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.
Key	A key on the VT-100 terminal of terminal emulator. For example <Enter> denotes the Enter key

# Table of Contents

<b>Chapter 1</b>	<b>Overview</b>	<b>1</b>
1.1	Default IP and Password Settings	1
1.2	Features	1
1.3	Networking Protocols	1
1.4	Package Contents	2
1.5	Front Panel	2
	LED Indicators	2
	Ports	3
	Reset Button	3
1.6	Rear Panel	3
<b>Chapter 2</b>	<b>Installing the VOIP-GATEWAY</b>	<b>4</b>
2.1	Network Requirements	4
2.2	Installing the VOIP-GATEWAY	4
2.3	Installing the telephony interface module	4
2.4	Connecting to the telephony devices	5
2.5	Connecting to the Network	5
2.6	Providing Power to the VOIP-GATEWAY	5
2.7	Assigning IP address to the VOIP-GATEWAY	5
<b>Chapter 3</b>	<b>VOIP-GATEWAY Concepts</b>	<b>6</b>
3.1	How the VOIP-GATEWAY Operates	6
3.2	ATPM	6
3.3	Destination	6
3.4	Hunt Group	7
3.5	Dial Plan	7
	Address Table	7
	Hunt Group Table(FXS4O4)	8
	Destination Table(FXS4O4)	8
3.6	DTMF Relay	8
3.7	Voice Codecs	8
<b>Chapter 4</b>	<b>Configuring VOIP-GATEWAY from a Web Browser</b>	<b>9</b>
4.1	Http setting mode	9

4.2	Web Browser setting sample	21
<b>Chapter 5</b>	<b>Making a Call with VOIP-GATEWAY</b>	<b>36</b>
5.1	Making a call with VOIP-GATEWAY FXS Module	36
5.2	Making a call with VOIP-GATEWAY FXO Module	36
5.3	Making a call with VOIP-GATEWAY Application Sample	37
<b>Chapter 6</b>	<b>Application Samples</b>	<b>41</b>
6.1	Slide-in Module Configuration	41
6.2	Application Samples	45
<b>Chapter 7</b>	<b>Troubleshooting Tips</b>	<b>52</b>
<b>Appendix A</b>		<b>54</b>
A.1	IP Parameters	54
A.2	ATPM Address Table	54
A.3	ATPM Hunt Group Table	55
A.4	ATPM Destination Table	56

## List of Figures

Figure 1-1	VOIP-GATEWAY Front Panel	2
Figure 1-2	VOIP-GATEWAY Rear Panel	3

# Chapter 1 Overview

This chapter gives an overview of the Internet Telephony Gateway (VOIP-GATEWAY) and a detailed description of its front panel and back panel.

## 1.1 Default IP and Password Settings

Default IP address: **192.168.0.1**

Default User's Name: **eitg**

Default Password: **123**

## 1.2 Features

The VOIP-GATEWAY is a cost-effective and highly reliable analog Voice over IP (VoIP) Gateway that offers toll quality voice and real-time fax data over IP networks. With its embedded architecture, the gateway is ideal for VoIP applications associated with Remote Office and Branch Office (ROBO) environments. With its built-in user-friendly interface, the gateway may be installed easily and conveniently to yield immediate cost savings. One Internet Telephony Gateway supports up to eight voice or Fax communications simultaneously.

Implemented with an efficient Real-Time Operating System (RTOS) and flash memory, the VOIP-GATEWAY provides upgradeable capabilities, so it may be programmed with updated firmware locally or via the network at any time. It comes equipped with remote management capabilities, configurable signaling to work with PBX, KTS, and/or telephone. The VOIP-GATEWAY utilizes advanced VoIP related technology. It includes various voice coders and fax algorithms, echo cancellation, Voice Activity Detection (VAD), Comfort Noise Generation (CNG), and lost packet recovery algorithms.

## 1.3 Networking Protocols

The VOIP-GATEWAY supports several industry-standard networking protocols required for voice communication. The following table describes these protocols.

Networking Protocol	Description
Internet Protocol (IP)	IP is a messaging protocol that addresses and sends packets across the network. To enable IP protocol, the VOIP-GATEWAY must have a Real IP address, subnet, and gateway assigned to it.
Voice over IP Protocol (VoIP)	VoIP enables the VOIP-GATEWAY to transfer voice communications over an IP network. The VOIP-GATEWAY employs ITU-T H.323 protocol for setting up call with one another.
Trivial File Transfer Protocol (TFTP)	TFTP allows you to transfer files over the network. The VOIP-GATEWAY implements a TFTP client allowing you to download

new revision firmware from a TFTP server. The TFTP client requires a TFTP server in your network.

**Real-Time Transport (RTP)** RTP is a standard for transporting real-time data over IP network. The VOIP-GATEWAY uses RTP protocol to send digitized and compressed voice packets.

## 1.4 Package Contents

The contents of your product should contain the following items:

- Internet Telephony Gateway 19" rack mountable version
- 90-240V AC power cord
- 9-pin straight through RS-232 cable
- Rack mount brackets (2) and screws (4)
- User's guide

## 1.5 Front Panel

The front panel of the VOIP-GATEWAY contains a RJ-45 Ethernet port, one DB9 RS-232 interface port, a push button and 16 LED indicators. The following figure illustrates the front panel of the VOIP-GATEWAY.

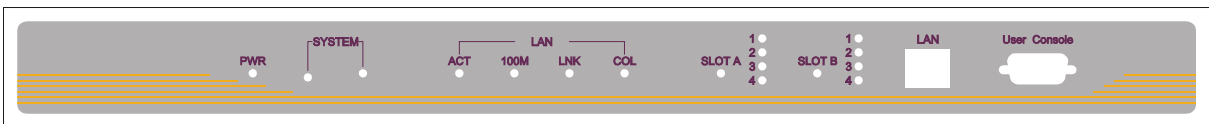


Figure 1-1 VOIP-GATEWAY Front Panel

### LED Indicators

The LED indicators on the front panel display the current status of the VOIP-GATEWAY as described in the following table:

Indicator	Color	Activity	Indication
PWR	Green	On	Power is supplied to the gateway.
SYSTEM	Green	Blinking	The system is running. (Heartbeat LED)
LAN			
• ACT	Green	On	Data is presented on LAN.
• 100M	Green	On	The gateway is connected to LAN at 100Mb/s.
		Off	The gateway is connected to LAN at 10Mb/s.
• LNK	Green	On	The gateway is connected to LAN.
• COL	Green	On	Data collision is occurring on the network connection.

Slot A	Green	On	A slide-in module is plugged into slot A.
Slot B	Green	On	A slide-in module is plugged into slot B.
Slot A/ Slot B		Blinking	The system is updating the firmware.
Slot A/B	Green	Off	The line is idle.
Channels		On	The line is being used.
1-4		Blinking	The line is ringing.

**Ports**

The VOIP-GATEWAY is equipped with an Ethernet interface with 10/100 Mbps auto-negotiation capability. The Ethernet interface port is located on the front panel. In addition to the Ethernet interface port, there is a 9-pin RS-232 interface port on the front panel. Their functions are described below:

Port	Label	Function
RJ45	LAN	Connecting the VOIP-GATEWAY to a 10/100 Mbps Ethernet network
9-pin RS-232	User Console	Connecting the VOIP-GATEWAY to a VT-100 terminal or terminal emulator for configuring the VOIP-GATEWAY

**Reset Button**

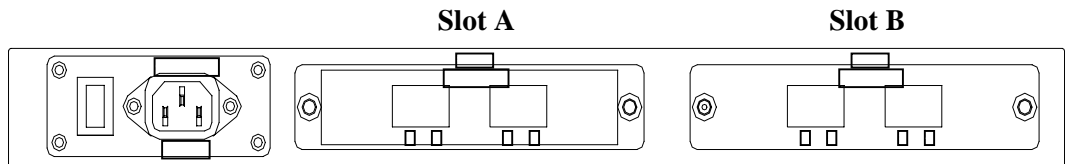
There is a push button located behind a small hole next to the SYSTEM LED. This button allows you to reset the VOIP-GATEWAY or force the VOIP-GATEWAY to enter firmware upgrade mode.

To reset the gateway, push a small, stiff object into the hole until the SYSTEM LED stops blinking, then release the button.

Powering on the gateway while pressing down the button for 5 seconds forces the VOIP-GATEWAY to enter download mode

**1.6 Rear Panel**

The rear panel of the VOIP-GATEWAY has two slots allowing the installation of up to two Telephony Interface Modules (TIM). Each module is equipped with 4 telephony interface ports. Depending on the type of the telephony interface module, the telephony interface ports may be connected to telephony devices, such as PBX, KTS, and telephone sets of central office. In addition to the TIM slots, there is an AC power receptacle and a power switch on the rear panel.



*Figure 1-2 VOIP-GATEWAY Rear Panel*

## Chapter 2 Installing the VOIP-GATEWAY

This chapter gives information on how to install the VOIP-GATEWAY.

### 2.1 Network Requirements

For the VOIP-GATEWAY to successfully operate in your network, your network must meet the following requirements:

1. A working 10/100 Base-T Ethernet. The VOIP-GATEWAY connects to Internet via an Ethernet LAN.
2. IP network that supports gateway, and subnet mask. You'll need static IP address to assign the VOIP-GATEWAY.

### 2.2 Installing the VOIP-GATEWAY

The VOIP-GATEWAY can be used in a desktop configuration or mounted in a standard 19-inch rack enclosure. Ensure that the VOIP-GATEWAY is placed in a clean, well-ventilated, and vibration-free environment.

When the VOIP-GATEWAY is placed on a desktop, be certain that the unit is placed on a sturdy, flat surface, near a grounded power outlet. At least three inches of clearance must be provided on both sides of the VOIP-GATEWAY for well ventilation.

### 2.3 Installing the telephony interface module

The VOIP-GATEWAY allows the installation of up to two TIMs. The following steps illustrate how to install a TIM:

1. Unfasten the two screws holding the cover of the TIM slot A on the rear panel.
2. Remove the cover.
4. Before proceeding further, wear a grounded wrist strap.
5. Remove the TIM from its static-free bag. Holding the module such that the metal bracket is facing toward you.
6. Carefully align the edge of the module facing away from you with the slide-in guide inside the VOIP-GATEWAY, and insert the module into the slide-in slot until it is fully sited.
7. Fasten the screws on the module.
8. Repeat step 1 through 6 for slot B if you have two TIMs. If you have only a single TIM, be certain to install it in slot A, otherwise, the VOIP-GATEWAY won't function.

## 2.4 Connecting to the telephony devices

The VOIP-GATEWAY supports three types of TIM. One is FXO, FXS and the other is E&M. Each module has four ports for connecting to telephony devices.

The FXO module is designed for connecting to PBXs or central office switches.

The FXS module is designed for connecting to analog telephone sets or G3 fax machines. Connecting the telephony devices to the appropriate RJ11 ports on the TIM.

The E&M module is designed for connecting to equipment (such as PBX or channel bank) providing the tie-line connection. Connecting the tie-line port to the appropriate RJ45 port on the TIM.

**Warning:** *connection of incorrect telephony devices to the ports on the TIM can cause permanent damage to the TIM and/or the VOIP-GATEWAY.*

## 2.5 Connecting to the Network

The RJ45 network port on the front panel supports 10/100 Mbps half-duplex connection to Ethernet. You can use either Category 3 or 5 straight-through UTP cable for 10 Mbps connection, but use Category 5 for 100 Mbps connection. To connect to Ethernet, insert one end of the Ethernet cable to the RJ45 port on the front panel of the VOIP-GATEWAY and other end of the cable to an Ethernet switching hub or repeater hub.

## 2.6 Providing Power to the VOIP-GATEWAY

To provide AC power to the VOIP-GATEWAY complete the following steps:

1. Turn the power switch on the rear panel of the VOIP-GATEWAY to the off position.
2. Connect one end of the power cord that came with the VOIP-GATEWAY to the power receptacle on the rear panel.
3. Connect the other end of the power cord to an AC power outlet.
4. Turn the power switch on. The VOIP-GATEWAY will perform power on self-test. After completing the self-test, the SYSTEM LED blinks continuously and the TIM indicators on the front panel will turn on.

## 2.7 Assigning IP address to the VOIP-GATEWAY

The IP address is the unique logical address identifying each IP node, such as the VOIP-GATEWAY, on an IP network. An IP address is a 32-bit number expressed as four decimal numbers from 0 to 255 separated by periods. The VOIP-GATEWAY needs static IP address and be aware of the subnet mask and default gateway (typically a router) of your network to be able to send to and receive data from the IP network. Consult your network manager to obtain a unique and static IP address for the VOIP-GATEWAY, the IP subnet mask and default gateway of your network, and fill out the work sheet in Appendix A before configuring the gateway. Procedures for assigning IP address, default gateway and subnet mask is available in Chapter 4.

## Chapter 3 VOIP-GATEWAY Concepts

The VOIP-GATEWAY enables the transmission of voice and fax traffic over any IP network by digitizing voice and fax signals, encapsulating the information within IP packets, and then sending the packets across the IP network

### 3.1 How the VOIP-GATEWAY Operates

1. The TIM inside the VOIP-GATEWAY digitizes analog voice signals at 8 Kbps.
2. VOIP-GATEWAY system software handles the:
  - Capture of telephone number presented as DTMF tones.
  - Mapping the telephone number to the IP address of remote VOIP-GATEWAY.
  - Setting up calls with remote VOIP-GATEWAYs utilizing H.323 call control protocol.
  - Digitizing, compressing and encapsulating the voice into IP packets and transmission of the IP packets onto the Ethernet LAN.
3. A router attached to the LAN forwards the IP packets across the WAN, where they will be received by another VOIP-GATEWAY at the remote.
4. The process is reversed at the remote VOIP-GATEWAY.

### 3.2 ATPM

To allow you to easily dial a telephone or fax on the network, the VOIP-GATEWAY maps a series of dialed digits to the IP address of the remote VOIP-GATEWAY whose phone or fax you are calling. This mapping information is contained in a database inside each VOIP-GATEWAY called the dial plan.

Based on the dial plan the Address Translation and Parsing Manager (ATPM) inside the VOIP-GATEWAY translate telephony numbers to IP addresses of remote VOIP-GATEWAYs. The ATPM collects telephone number dialed by users, decides whether the dial string is part of the dial plan and, if it is, maps it a remote VOIP-GATEWAY. When the call is set up to the destination, a sub-string of the original dial string will be sent along to the remote VOIP-GATEWAY.

### 3.3 Destination

The destination is where a call is terminated. Typically, for inbound calls from IP network, the VOIP-GATEWAY terminals the call at one of the telephony ports. The destination for the call is the telephony port where the call terminated. For calls initiated from telephony ports, the VOIP-GATEWAY forward the call to a remote VOIP-GATEWAY via IP network, and the remote VOIP-GATEWAY terminal the call. The destination of the call is the remote VOIP-GATEWAY.

### 3.4 Hunt Group

Instead of directly mapping a phone number to a destination, the ATPM first maps the phone number to a group of destinations known as a Hunt Group. A hunt group is a group of destinations that are equivalent. For example, the customer support group of a company might have 20 people who can handle support calls. Access to customer support is through a single phone number but the next available support person is actually connected upon each incoming call. These 20 phones would be configured as a hunt group. A hunt group consists of a phone number and a list of destinations (members of the group). When an incoming phone number matches the phone number of the hunt group, the VOIP-GATEWAY attempts to terminate the call at each of the destinations in the hunt group, one at a time until a call is successfully completed.

Every destination that can be reached by dialing a phone number is a member of at least one hunt group. When an address is presented to ATPM for lookup, the output is a hunt group ID number. As a second step, the hunt group ID is presented to ATPM to get the list of members. To effectively bypass the hunt group feature, simply make a unique hunt group for each destination and one member in each hunt group.

### 3.5 Dial Plan

The dial plan is a database inside the VOIP-GATEWAY for the ATPM to map telephony numbers users dialed to the IP address of remote VOIP-GATEWAYS. The dial plan consists of the destination table, hunt group table and the address table. Users need to setup these tables, so that the VOIP-GATEWAY knows how to setup calls with remote VOIP-GATEWAYS.

#### Address Table

The address table maps a phone number to a hunt group. The table contains entries that specify the following information:

- Telephone number
- The hunt group the phone number maps to.
- The minimum number of digits to collect before the ATPM starting address lookup.
- The maximum number of digits the ATPM collects before it considers the dial string is complete.
- Number of digits forward to the destination.

Address table sample:(FXS4O4)

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
200	1	3	3	0	None
201	3	3	3	0	None
800	5	0	3	3	None
810	5	0	3	3	None
20	11	2	5	2	None

### Hunt Group Table(FXS404)

The hunt group table maps a hunt group to a list of destinations.

Hunt group sample

Group id	Type	#Members	Member ids
1	2	1	1
3	2	1	3
5	2	1	5
11	2	1	11

### Destination Table(FXS404)

The destination table maps a destination to a telephony port or the IP address of a remote VOIP-GATEWAY.

Destination table sample

Dest id	Mode	Destination
1	Local	PORT = 0
3	Local	PORT = 2
5	Local	PORT = 4
11	H.323	Dest = 192.168.0.55/1720 TCP
12	DNS	Dest = OvisLink0021.dyndns.org /1720 TCP

## 3.6 DTMF Relay

Voice from PSTN is compressed by the VOIP-GATEWAY before sending across the IP network and then decompressed by the destination VOIP-GATEWAY. The voice coders supported by the VOIP-GATEWAY are designed for ideally compressing and decompressing human voice. If the compression / decompression process is performed on DTMF tone which needs to be conveyed across IP network, distortion might be too significant to be not cognizable in the receiving end. To overcome the shortcoming that the voice coders can not perfectly encode DTMF tone, the VOIP-GATEWAY encodes DTMF tone into special packets. The packets are then sent to the destination VOIP-GATEWAY via a separate IP connection. The destination VOIP-GATEWAY decodes the packets, generates the DTMF tone, and then sends the tone to the PSTN. The way the VOIP-GATEWAY handles DTMF tone is so called DTMF relay.

The VOIP-GATEWAY handles DTMF relay per H.323 specifications. Certain third party VoIP devices may handle DTMF relay per IMTC standard. For the VOIP-GATEWAY to interoperate with those VoIP devices, users need to specify which remote VoIP devices uses IMTC conforming DTMF relay technique. Refer to CLI command on Chapter 8 for detailed information on how to select DTMF relay mode.

## 3.7 Voice Codecs

Voice codecs supported by the VOIP-GATEWAY include G.711, G.723.1 5.3kbps, G.723.1 6.3kbps and G.729 AB. When setting up a call, two VOIP-GATEWAY automatically negotiate with each other until an agreed upon codec is determined.

## Chapter 4 Configuring VOIP-GATEWAY from a Web Browser

This chapter explains procedures for configuring the VOIP-GATEWAY from the web browser.

### 4.1 Http setting mode

This section describes the processes for setting up Internet Telephony Gateway once it has been installed. Microsoft Explorer version 4 or higher, or Navigator version 4.5 or higher can be used in this section to view and change parameters.

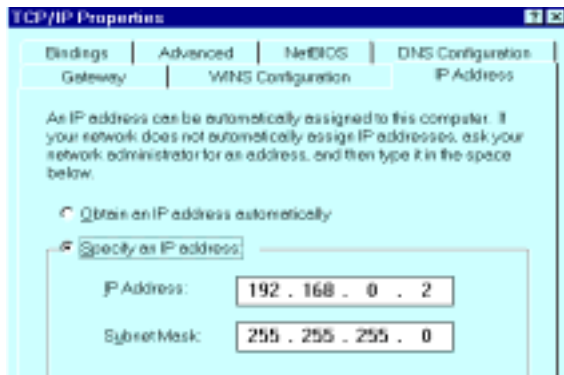
#### PC Setup

In order to configure the VOIP-GATEWAY, PC needs to have TCP/IP protocol and a compatible IP Address.

1. Connect the VOIP-GATEWAY to network with a RJ-45 UTP cable. Power it on.
2. Find a PC, for example, Windows 95/98. Under Windows 95/98, select the *Network Neighborhood* icon on the desktop, then select *Properties*. We will see a screen like below:



3. If a line like the one highlighted ("TCP/IP -> Network Card") is not listed, select *Add-Protocol-Microsoft-TCP/IP-OK* to add it.
4. Select *Properties* for the "TCP / IP -> Network card" entry. You will see a screen like the following:



5. On the *IP Address* table, enter values as follows:

- *Specify an IP address* set ON.
- *IP Address*: 192.168.0.2
- *Subnet Mask*: 255.255.255.0

Restart your PC and Start your WEB browser.

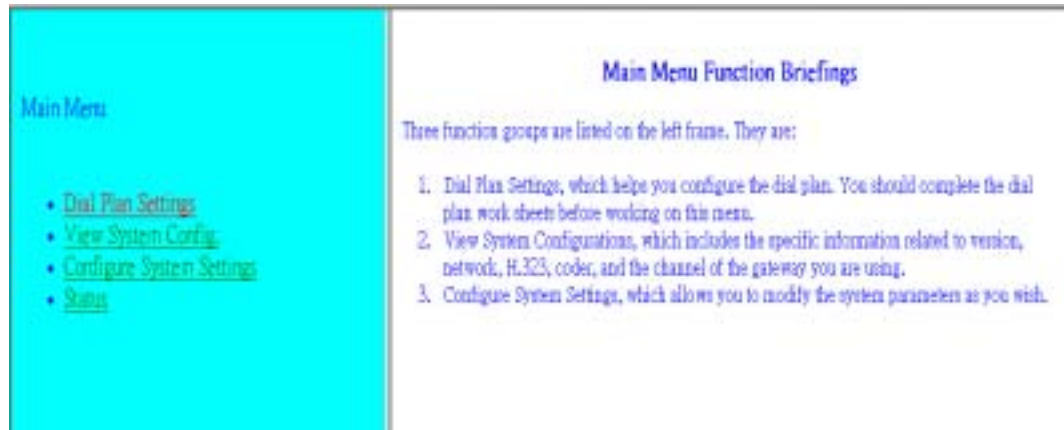
6. In the *Address* box, enter the following:

[HTTP://192.168.0.1](http://192.168.0.1) (This is a default IP in the gateway from factory)

7. Press enter to confirm and you should find the screen below.



8. The default User Name is **eitg** (all lower case). Password is **123**. For security reasons, please change and memorize the new password after this first setup.
9. Click "OK". The main screen will appear as below.



### Main Menu Function Briefings

1. **Dial Plan Settings**, which helps you configure the dial plan. You should complete the dial plan work sheets before working on this menu.
2. **View System Configurations**, which includes the specific information related to version, network, H.323, coder, and the channel of the gateway you are using.
3. **Configure System Settings**, which allows you to modify the system parameters as you wish.

**Select Main Menu\Dial Plan Settings**, we can find the following Dial Plan Menu.



### Dial Plan Settings Function Briefings

1. **Telephone Settings**, which allows one to Add, Delete, Find, or List telephone numbers.
2. **Hunt Group Settings**, which allows one to Add, Delete, Find, or List hunt group settings
3. **Destination Settings**, which allows one to Add, Delete, Find, or List destination settings

4. Common Parameter Settings, which defines the desired total dial time, first digit wait time, inter-digit wait time, and the termination digit.
5. Clear All Entries, which clears all telephone numbers, hunt groups, and destinations settings, for both old and new ones
6. Restore All Entries, which retrieves all dial plan settings from the flash, temporary settings will be cleared
7. Store Settings To Flash, which stores all settings that you have specified permanently.

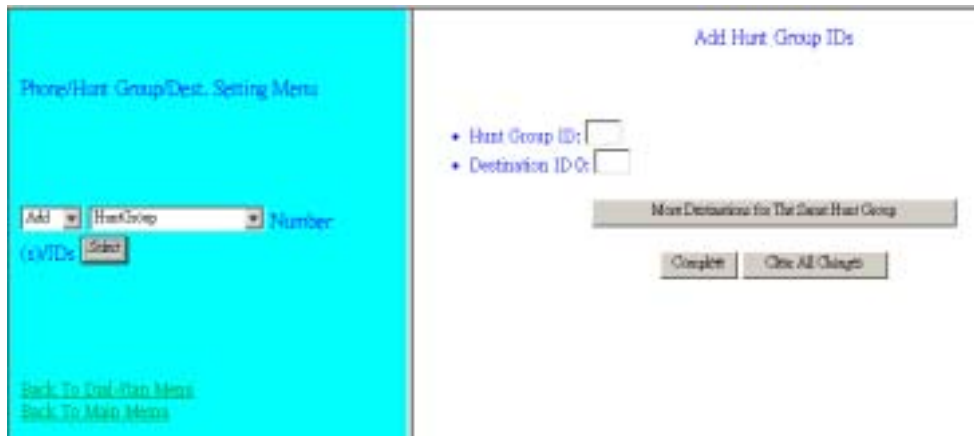
To ensure your new settings are actually accepted, please check the configuration page carefully after your submit the new settings. If the new settings are not present, it probably conflicts with the old settings. Or, there is not enough space for storing your settings.

**Please select Main Menu\Dial Plan\Phone/Hunt Group/Destination Settings, we can find Phone/Hunt Group/Dest. Setting Menu.** We are doing VOIP-GATEWAY telephone address table management.

We can add (delete, find or list) desired telephone number mapping to hunt group at this menu.

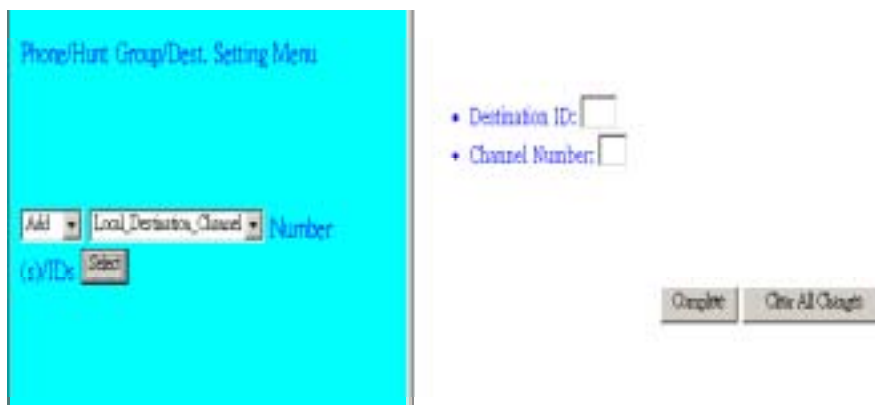
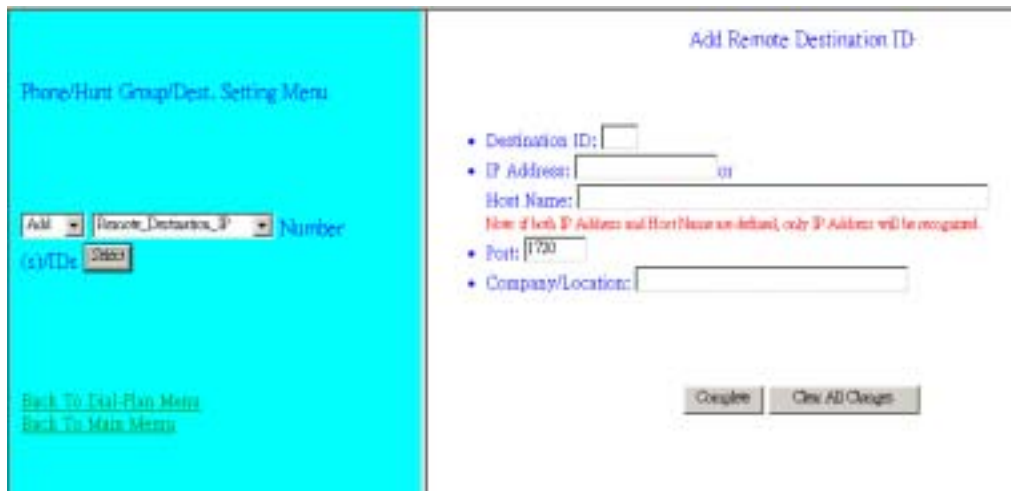
Telephone Number	Telephone number to match. This is only part of the total dialed string.
Hunt Group ID	For each hunt group ID, you need to assign it a unique identifier between 0 and 99.
Min. Digits	Minimum number of digits to be collected before the ATPM starting matching the dial string with entries in the address table.
Max. Digits	Maximum number of digits to be collected before the ATPM starting matching the dialed string with entries in the address table.
Strip Length	The number of digits to be stripped at the beginning of the collected dial string before forwarding the string to the destination.
Append Prefix	(Optional) Digit to be added before the beginning of the collected dial string before forwarding it to the destination.

**Please select Main Menu\Dial Plan\Phone/Hunt Group/Destination Settings, we can find Phone/Hunt Group/Dest. Setting Menu.** We are doing the VOIP-GATEWAY hunt group table management.



We can add (delete, find or list) desired hunt group ID mapping to destination ID at this menu.

**Please select Main Menu\Dial Plan\Phone/Hunt Group/Destination Settings, we can find Phone/Hunt Group/Dest. Setting Menu.** We are doing the VOIP-GATEWAY destination table management. One is Remote Destination IP and the other is Local Destination Channel.



We can add (delete, find or list) desired destination ID mapping to Remote Destination IP or Local Destination Channel at this menu.

Please back to Dial Plan Menu.

**Please select Main Menu\Dial Plan\Common Parameter Settings**

This Common Parameter Settings is the same as previous ATPM system parameters description. .

**Please select Main Menu\Dial Plan\Clear All Dial Plan Settings**

Clear All Dial Plan Settings will clear all the settings you made before. However, the 'Clear All' is only temporary if you do not store to the Flash Memory afterwards.

That is, to restore all the settings from the Flash Memory, you need only to click on the 'Restore All Entries' on the left frame, or to reboot the system (the settings can be restored ONLY IF YOU DID NOT STORE the entries).

**Please select Main Menu\Dial Plan\Restore All Dial Plan Settings**

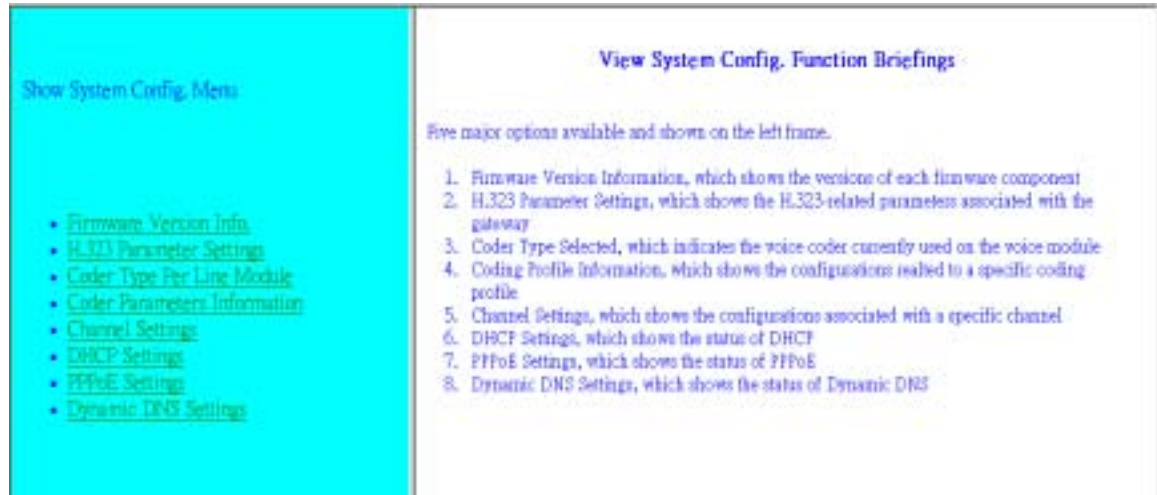
Retrieve all dial plan settings from the flash memory. All temporary settings will be cleared.

**Please select Main Menu\Dial Plan\Store Settings To Flash.**

Ready to store Dial Plan settings to Flash Memory

Please Back To Main Menu

**Please select Main Menu\View System Config.**



**View System Config. Function Briefings.** Five options available

1. Firmware Version Information, which shows the versions of each firmware component
2. Network Settings, which shows the IP-related settings, and the status of associated management tools
3. H.323 Parameter Settings, which shows the H.323-related parameters associated with the gateway
4. Coding Profile Information, which shows the configurations related to a specific coding profile
5. Channel Settings, which shows the configurations associated with a specific channel
6. DHCP Settings, which shows the status of DHCP
7. PPPoE Settings, which shows the status of PPPoE
8. Dynamic DNS Settings, which shows the status of Dynamic DNS

**Please select Main Menu\View System Config.\Firmware Version Info.**

This will show current VOIP-GATEWAY firmware version information.

Firmware Information:

Internet Telephony Gateway Version: 3.23  
Boot Loader Version: 4.13  
RTOS Version: 2.5.0./BE  
H323 Stack Version: 3.0.9.0  
DSP image Version: 8.1.2.1.  
TSG Version : R8.0 gateway(Build 4)

**Please select Main Menu\View System Config.\H.323 Parameter Settings**

The H.323 Parameters Settings:

```

h323 display_name = 'Customer'
h323 h245_term_type = 60
h323 rtp_port_base = 30000
h323 out_fast_start = off
h323 in_fast_start = off
h323 h245_tunneling=off
h323 cisco_t38=on
h323 callSignalPort=1720
h323 nat_call = on
h323 call_name=
h323 local_alert=off
h323 default_dtmf = H323 V2 Signal
No Alternate IP Defined!
h323 dns_ip = 0
h323 gk_mode = off
h323 h245_timeout = 30000
h323 term_id =
h323 wan_refresh=120
h323 nat_name=
h323 wan_mode=off
h323 wan_port_gk=0
h323 wan_port_gw=0
h323 wan_max_retries=10
h323 wan_lost_timer=1
h323 wan_serv_ip=0.0.0.0

```

**Please select Main Menu\View System Config.\Coder Type Per Line Module**

The Voice Coder For Each Line Module:

Voice coder is G.723 for Line Module 1.

Voice coder is G.723 for Line Module 2.

**Please select Main Menu\View System Config.\Coder Parameters Information**



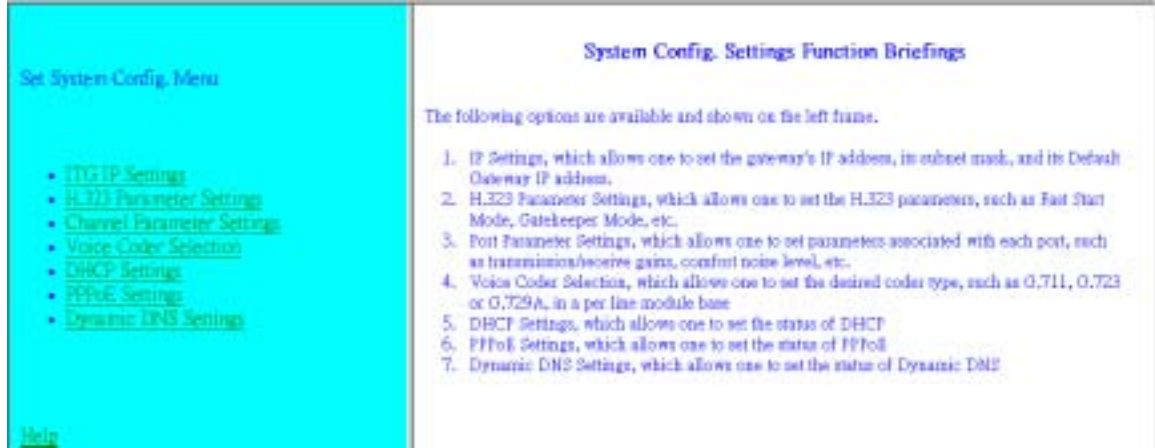
**Please select Main Menu\View System Config.\Channel Settings**

This command can show the desired TCID setting of VOIP-GATEWAY, value is from 0 to 7.

Please refer the appendix C to find the factory default setting.

Please Back To Main Menu

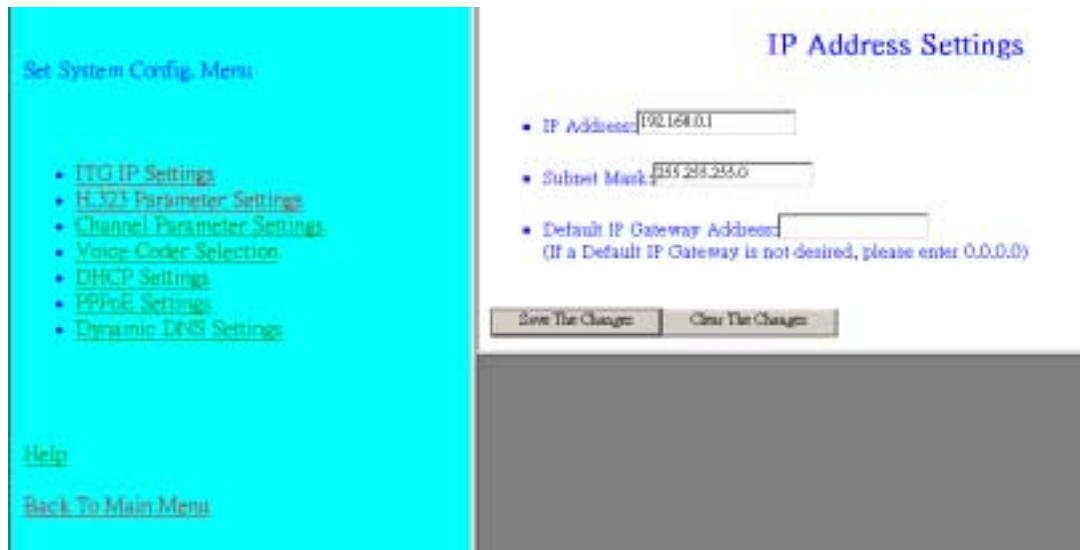
**Please select Main Menu\Configure System Settings**



**System Config. Settings Function Briefings**

1. IP Settings, which allows one to set the gateway's IP address, its subnet mask, and its Default Gateway IP address.
2. H.323 Parameter Settings, which allows one to set the H.323 parameters, such as Fast Start Mode, Gatekeeper Mode, etc.
3. Port Parameter Settings, which allows one to set parameters associated with each port, such as transmission/receive gains, comfort noise level, etc.
4. Voice Coder Selection, which allows one to set the desired coder type, such as G.711, G.723 or G.729A, in a per line module base.
5. DHCP Settings, which allows one to set the status of DHCP
6. PPPoE Settings, which allows one to set the status of PPPoE
7. Dynamic DNS Settings, which allows one to set the status of Dynamic DNS

**Please select Main Menu\Configure System Settings\VOIP-GATEWAY IP Settings**

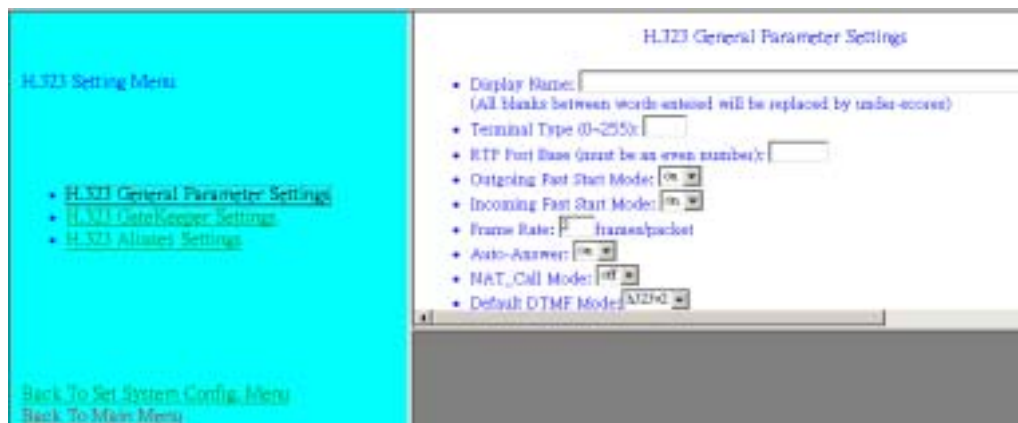


This screen can setup your VOIP-GATEWAY IP Address, Subnet Mask and Default IP Gateway Address. These values will be valid after VOIP-GATEWAY reboot.

**Please select Main Menu\Configure System Settings\H.323 Parameter Settings**

There are H.323 General Parameter Settings, H.323 Gatekeeper Settings, H.323 Aliases Settings.

**Please select Main Menu\Configure System Settings\H.323 Parameter Settings\H.323 General Parameter Settings**



1. Display Name: The default string is “Customer”.

This field is to set the display name information that is carried in the H.323 setup messages. Up to 48 characters can be entered.

2. Terminal Type: The default value is 60.

This field is to set the H.245 terminal type, which is used as part of the master/slave determination process of H.245. Typically, setting a value of less than 50 will force slave operation, and a value of greater than 200 will force the master operation. For more details, please refer to H.323-related standard documents.

3. RTP Port Base: The default value is 30000.

This field is to select the starting port number for assignment of RTP and RTCP ports. According to the H.323 specification, RTP port number should be even in value, and the RTCP port number should be one greater than the RTP port. Typically, numbers from 0 to 1023 are reserved on most systems.

4. Outgoing Fast Start: The default setting is off.  
This field is to enable or disable the Fast start mode on the outgoing side of the link.
5. Incoming Fast Start: The default setting is off.  
This field is to enable or disable the Fast start mode on the incoming side of the link.  
You'll have to reboot the system to make your changes active!
6. Frame Rate = 2 frames/packet
7. Auto-Answer: Enables quick H.225 to H.245 transition without waiting for receiver picking up the phone, default is on
8. NAT Call Mode: Enables calls from remote sites which use NAT routers with private IP networks behind, default is on
9. Default DTMF Mode: H323 V2 Signal or IMTC. The default is H323 V2 Signal
10. DNS IP Address
11. Alternate DTMF IP Address:
12. DTMF Duration: default is 300

**Please select Main Menu\Configure System Settings\H.323 Parameter Settings\H.323 GateKeeper Settings**



1. Gate Keeper Mode: The default setting is off. This field is to select the co-operation mode with some gatekeeper(s). Three options available:
  - off: disables gatekeeper co-operation,
  - auto: enables auto-discovery of the gatekeeper, and
  - manual: enables gatekeeper co-operation in manual operation (the gatekeeper address must be properly assigned).

The following fields must be filled up when the Gate Keeper Mode is set to either auto or manual.

2. Gate Keeper Address: Specify the gatekeeper address when configured to manual mode.  
When auto mode is desired, this field should be set to auto.
3. Allow Calls Without Gate Keeper: This field is to inform the H.323 stack to allow calls when the endpoint is not registered with a gatekeeper.

4. Registration Type: This field is to set the endpoint registration type. This specifies how the endpoint will register itself with the gatekeeper.
5. Max Registration Retries: This field is to control how many registration attempts will be made before the endpoint considers itself to have failed registration.
6. Time To Live:
7. Endpoint Prefix: This command is used to set the H.323 prefix that the VOIP-GATEWAY uses when registering to an H.323 gatekeeper.
8. Terminal ID: To specify GW's ID.
9. GateKeeper ID: To specify Gate Keeper's ID.

You'll have to reboot the system to make your changes active!

**Please select Main Menu\Configure System Settings\H.323 Parameter Settings\H.323 Aliases Settings**



Alias: This field is to create or delete aliases that are registered with the gatekeeper.

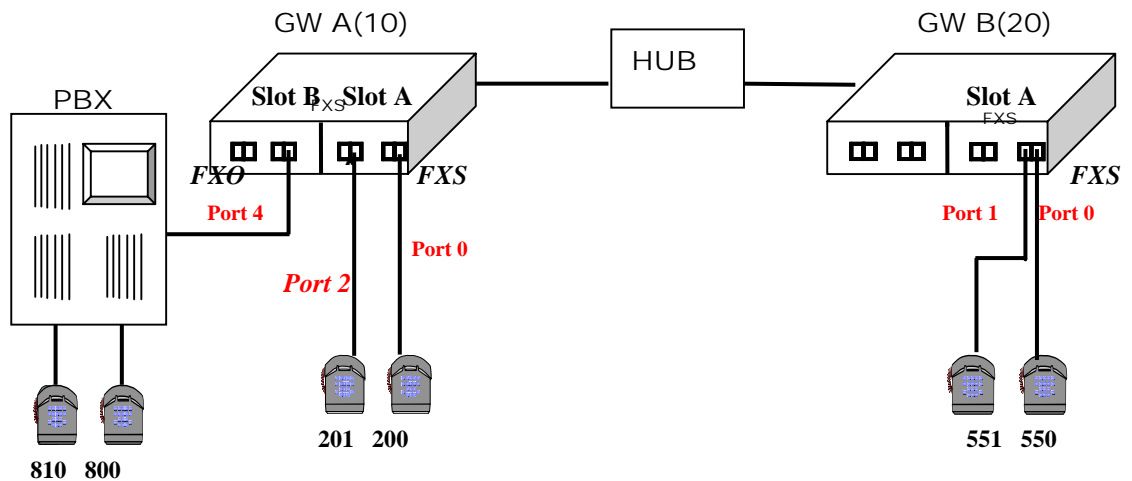
## 4.2 Web Browser setting sample

This section describes how to use web browser to build a sample scenario dial plan in VOIP-GATEWAY. We suggest some steps to setup dial plan at web browser setting,

1. Please draw down scenario to have a VoIP call application
2. Local gateway: IP setup
3. Local gateway: local telephone number setup
4. Local gateway: remote gateway IP & remote telephone number setup

### Please draw down scenario to have a VoIP call application

In order to describe the dial plan of Internet Telephony Gateway, we would like to have the following scenario to step by step the gateway setting by web browser. In case, you don't have PBX on hand, please ignore GW A Port 5 setting.



Scenario description: There are two gateways connected by a Hub.

Gateway A is configured as following:

1. Gateway A, ID=10, Gateway A IP: 192.168.0.20, mask IP: 255.255.255.0, gateway IP 0.0.0.0 (virtual IP).
2. Slot A plugs FXS module and slot B plugs FXO module
3. Port 0 of FXS in slot A has a telephone set connected, its phone number is 200
4. Port 2 of FXS in slot A has a telephone set connected, its phone number is 201
5. Port 4 of FXO in slot B is connected to PBX. There are two telephone sets connected by PBX, their phone numbers are 800 and 810.

Gateway B, ID=20, Gateway B is configured as following:

1. Gateway B IP: 192.168.0.55, mask IP: 255.255.255.0, gateway IP 0.0.0.0 (virtual IP), ID: 20
2. Slot A plugs FXS module and slot B is empty
3. Port 0 of FXS in slot A has a telephone set connected, its phone number is 550
4. Port 1 of FXS in slot A has a telephone set connected, its phone number is 551

Local gateway: IP setup, we are setting gateway A.

1. Please select Main Menu\ Configure System Settings\ VOIP-GATEWAY IP Settings from Main Menu
2. Key-in IP Address: 192.168.0.20, Subnet Mask: 255.255.255.0 and Default IP Gateway Address: 0.0.0.0 in the related fields.
3. And push “Save The Change” button

#### Local gateway: local telephone number setup

Gateway A's telephone number 200 setting. 200 locates at Port 0 of FXS in slot A

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add telephone” from Main Menu

2. Push “Complete” button. And screen will show “Telephone Number Added 200 OK” table. Telephone number 200 is added in gateway dial plan.
3. Purpose: add telephone address to hunt group mapping table of phone number 200 into dial plan

Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu

2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of phone number 200 into dial plan

Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Local\_Destination\_Channel” from Main Menu

2. Push “Complete” button. And screen will show “Add Destination ID Result:” table  
The Destination ID 1 is bound to local channel 0.
3. Purpose: add destination ID to destination IP mapping table of phone number 200 into dial plan. Channel Number can have values from 0 to (number of ports –1)

Back To Dial-Plan Menu

Gateway A's telephone number 201 setting. 201 locates at Port 2 of FXS in slot A

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add telephone” from Main Menu

2. Push “Complete” button. And screen will show “Telephone Number Added 201 OK” table. Telephone number 201 is added in gateway dial plan.
3. Purpose: add telephone address to hunt group mapping table of phone number 201 into dial plan

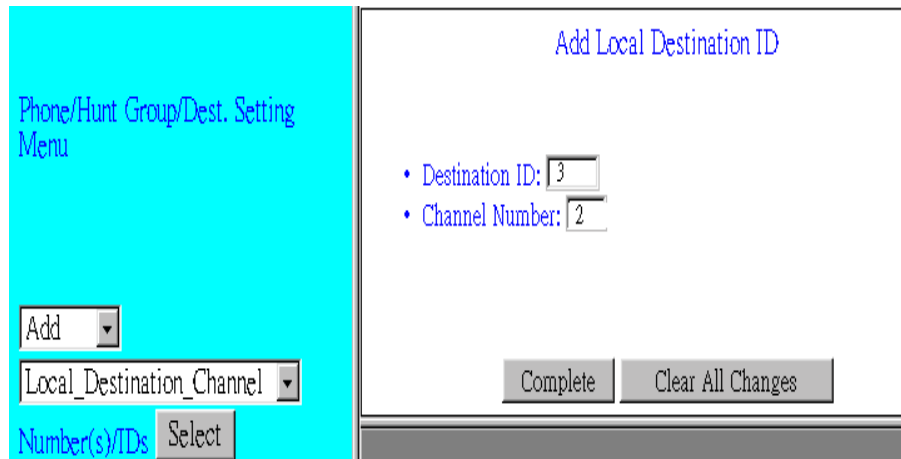
#### Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu

2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of phone number 201 into dial plan

#### Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest . Setting Menu\ Add Local\_Destination\_Channel” from Main Menu

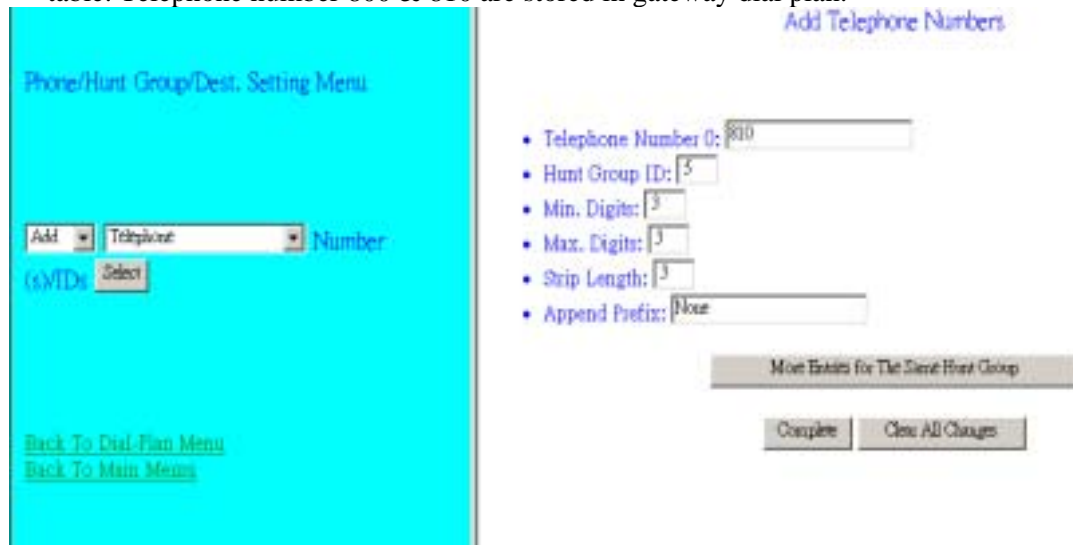


2. Push “Complete” button. And screen will show “Add Destination ID Result:” table  
The Destination ID 3 is bound to local channel 2.
3. Purpose: add destination ID to destination IP mapping table of phone number 201 into dial plan

**Back To Dial-Plan Menu**

Gateway A’s telephone number 800 & 810 setting. 800 & 810 locate in Port 4 of FXO in slot B. Due to slot B is the second slide-in module of this gateway, we usually define its hunt group number by beginning at number 5 at this dial plan setting.

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Telephone” from Main Menu
2. Key-in 800 firstly, and push “More Entries for The Same Hunt Group” button and key-in 810.
3. Push “Complete” button. And screen will show “Telephone Number Added 800 & 810 OK” table. Telephone number 800 & 810 are stored in gateway dial plan.



4. Purpose: add telephone address to hunt group mapping table of phone number 800 & 810 into dial plan

## Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu

2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of phone number 800 & 810 into dial plan

## Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Local\_Destination\_Channel” from Main Menu

2. Push “Complete” button. And screen will show “Add Destination ID Result:” table  
The Destination ID 5 is bound to local channel 4.
3. Purpose: add destination ID to destination IP mapping table of phone number 800 & 810 into dial plan

**Local gateway: remote gateway IP & remote telephone number setup**

Gateway A's remote gateway IP & remote telephone number 550 & 551 setting

Back To Dial-Plan Menu

Gateway A's remote telephone number 550 & 551 setting. Due to 550 & 551 locate in remote gateway, we usually define its hunt group number is beginning at number 11 at this dial plan setting.

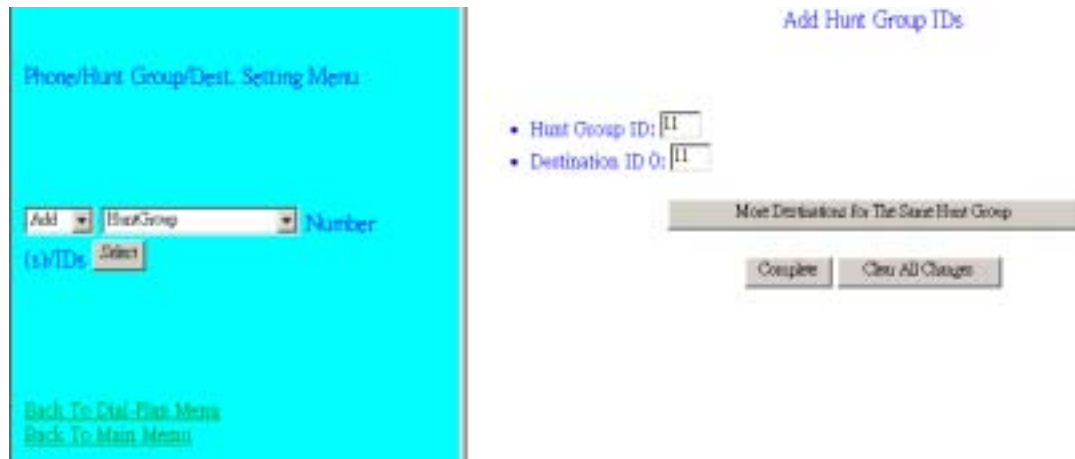
1. Please select "Main Menu\Dial Plan\Phone/Hunt Group/Destination Settings\Phone/Hunt Group/Dest. Setting Menu" from Main Menu



2. Key-in 20 , and related information in each filed.
3. Push "Complete" button. And screen will show "Telephone Number Added 20 OK" table. Remote ID number 20 is stored in gateway dial plan.
4. Purpose: add telephone address to hunt group mapping table of the number 20 into dial plan

Back To Dial-Plan Menu

1. Please select "Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest . Setting Menu\ Add Hunt Group" from Main Menu



2. Push "Complete" button. And screen will show "List All Hunt Group IDs Result:" table.
3. Purpose: add hunt group to destination ID mapping table of phone number 20 into dial plan

Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Remote\_Destination\_IP” from Main Menu

Phone/Hunt Group/Dest. Setting Menu

Add > Remote\_Destination\_IP > Number

(x)IDs Select

Back To Dial-Plan Menu  
Back To Main Menu

- Destination ID: 11
- IP Address: 192.168.0.55 or
- Host Name: \_\_\_\_\_
- Note: If both IP Address and Host Name are obtained, only IP Address will be recognized.
- Port: 1720
- Company/Location: \_\_\_\_\_

Complete Clear All Changes

2. Push “Complete” button. And screen will show “Add Destination ID Result: The Destination ID 11 is bound to IP address 192.168.0.55 for H.323 VoIP application.”
3. Purpose: add destination ID to destination IP mapping table of phone number 20 into dial plan

Back To Dial-Plan Menu

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ List Telephone” from Main Menu, we can obtain:

Phone/Hunt Group/Dest. Setting Menu

List > Telephone > Number

(x)IDs Select

List All Phone Numbers Result:

Phone #	Hunt Group ID	Min. Digits	Max. Digits	Strip	Prefix
20	11	2	5	2	Note
200	1	0	3	3	Note
201	3	0	3	3	Note
800	5	0	3	3	Note
810	5	0	3	3	Note

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ List Hunt Group” from Main Menu, we can obtain:

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu” from Main Menu and select List Remote\_Destination\_IP or Local\_Destination\_Channel, we can obtain:

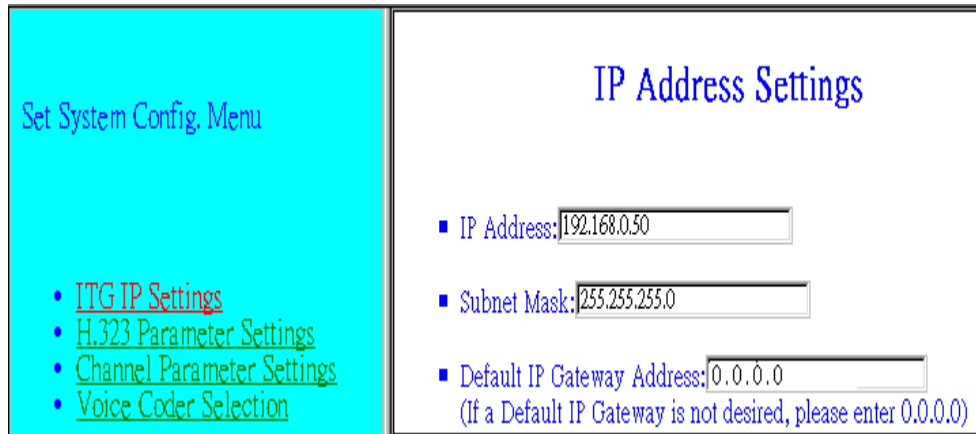


Please select Main Menu\Dial Plan\Store Settings To Flash. The above setting will be stored to flash memory.

At this moment we have finished gateway A VoIP call application setting. We are going to have the same process to setup gateway B.

**Local gateway: IP setup, we are setting gateway B.**

1. Please select Main Menu\ Configure System Settings\ VOIP-GATEWAY IP Settings from Main Menu
2. Key-in IP Address: 192.168.0.50, Subnet Mask: 255.255.255.0 and Default IP Gateway Address: 0.0.0.0 in the related fields.
3. And push “Save The Change” button



**Local gateway: local telephone number setup**

Gateway B's telephone number 550 setting. 550 locates at Port 0 of FXS in slot A

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Telephone” from Main Menu

2. Push “Complete” button. And screen will show “Telephone Number Added 550 OK” table. Telephone number 550 is stored in gateway dial plan.
3. Purpose: add telephone address to hunt group mapping table of phone number 550 into dial plan

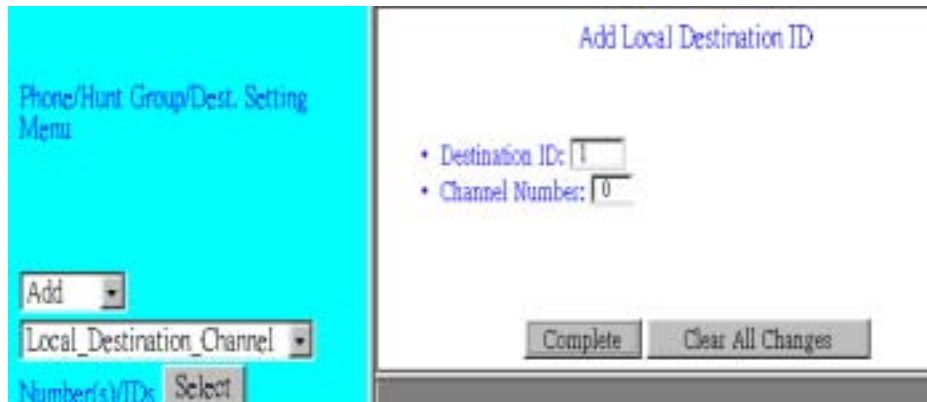
#### Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu

2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of phone number 550 into dial plan

#### Back To Dial-Plan Menu

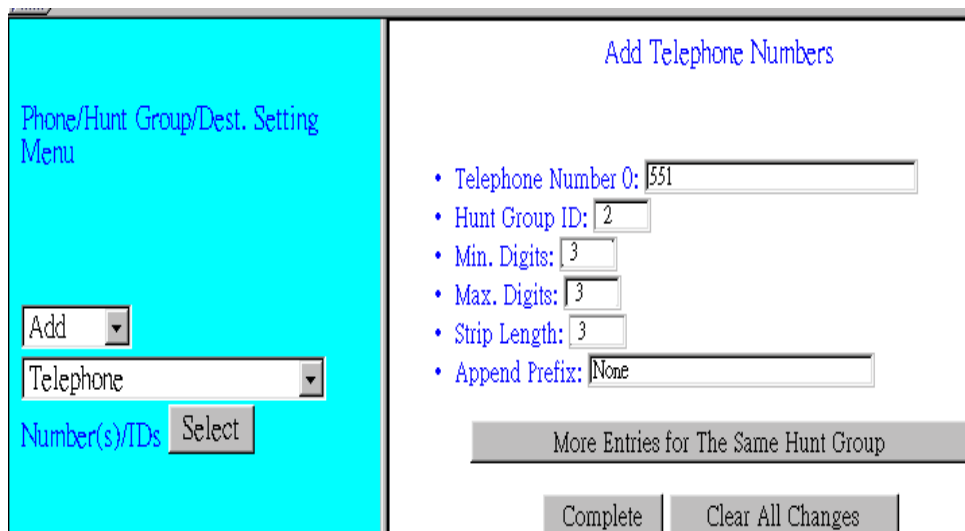
1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Local\_Destination\_Channel” from Main Menu



2. Push “Complete” button. And screen will show “Add Destination ID Result:” table  
The Destination ID 1 is bound to local channel 0.
3. Purpose: add destination ID to destination IP mapping table of phone number 550 into dial plan

**Gateway B’s telephone number 551 setting. 551 locates at Port 1 of FXS in slot A**

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings \ Phone/Hunt Group/Dest. Setting Menu\ Add Telephone” from Main Menu
2. Push “Complete” button. And screen will show “Telephone Number Added 551 OK” table.  
Telephone number 551 is stored in gateway dial plan.
3. Purpose: add telephone address to hunt group mapping table of phone number 551 into dial plan



**Back To Dial-Plan Menu**

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu

2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of phone number 551 into dial plan

#### Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Local\_Destination\_Channel” from Main Menu
2. Push “Complete” button. And screen will show “Add Destination ID Result:” table  
The Destination ID 2 is bound to local channel 1.
3. Purpose: add destination ID to destination IP mapping table of phone number 551 into dial plan

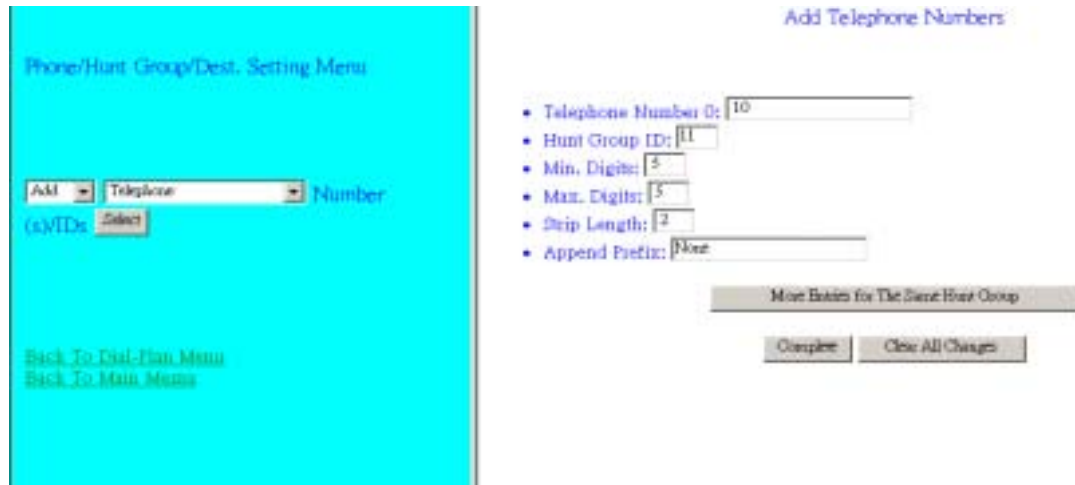
#### Local gateway: remote gateway IP & remote telephone number setup

Gateway B's remote gateway IP & remote telephone number 200 & 201 & 800 & 810 setting

#### Back To Dial-Plan Menu

Remote Gateway ID number setting. Due to 10 is remote gateway A ID. Due to 200 & 201 & 800 & 810 locate in remote gateway, we usually define its hunt group number is beginning at number 11 at this dial plan setting.

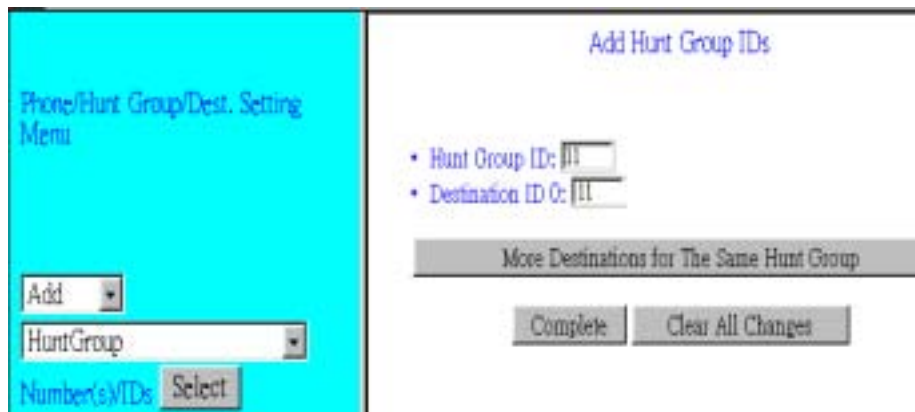
1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Telephone” from Main Menu



2. Key-in 10 firstly, and related information in each filed.
3. Push “Complete” button. And screen will show “Telephone Number Added 10 OK” table. GW ID number is stored in gateway dial plan.
4. Purpose: add telephone address to hunt group mapping table of ID number 10 into dial plan

Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Hunt Group” from Main Menu



2. Push “Complete” button. And screen will show “List All Hunt Group IDs Result:” table.
3. Purpose: add hunt group to destination ID mapping table of number 10 into dial plan

Back To Dial-Plan Menu

1. Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ Add Remote\_Destination\_IP” from Main Menu

2. Push “Complete” button. And screen will show “Add Destination ID Result: The Destination ID 11 is bound to IP address 192.168.0.20 for H.323 VoIP application.”
3. Purpose: add destination ID to destination IP mapping table of number 10 into dial plan

#### Back To Dial-Plan Menu

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ List Telephone” from Main Menu, we can obtain telephone address table:

Phone #	Hunt Group ID	Min. Digits	Max. Digits	Strip	Prefix
10	11	2	5	2	None
550	1	0	3	3	None
551	2	0	3	3	None

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu\ List Hunt Group” from Main Menu, we can obtain hunt group table:



List All Hunt Group IDs Result:

Hunt Group ID	Hunt Type	# of Dest ID(s)	Dest. ID(s)
1	2	1	1
2	2	1	2
11	2	1	11

Please select “Main Menu\ Dial Plan\ Phone/Hunt Group/Destination Settings\ Phone/Hunt Group/Dest. Setting Menu” from Main Menu and select List Remote\_Destination\_IP or Local\_Destination\_Channel, we can obtain:



List All Destination IDs Result:

Destination ID	Mode	Destination
1	Local	PORT = 0
2	Local	PORT = 1
11	H.323	Dest = 192.168.0.20/1720 TCP

Please select Main Menu\Dial Plan\Store Settings To Flash. The above setting will be stored to flash memory.

We have finished gateway B VoIP call application setting. We can enjoy VoIP call between gateway A and gateway B application now.

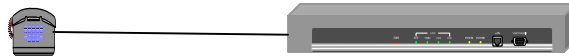
## Chapter 5 Making a Call with VOIP-GATEWAY

This chapter how to make phone calls from telephony devices connected to the VOIP-GATEWAY directly or indirectly.

### 5.1 Making a call with VOIP-GATEWAY FXS Module

VOIP-GATEWAY has two slide-in modules. FXS module can connect to analogue phone directly, and FXO module can connect to PBX.

Connection: analogue telephone set connects to FXS module port



Operation:

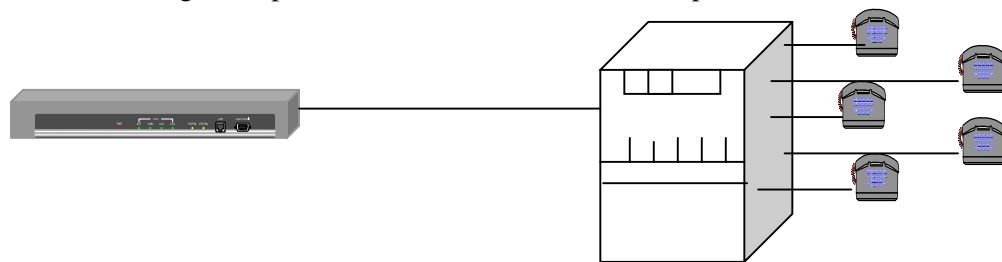
1. Pick up this analogue telephone set, you can hear the VOIP-GATEWAY dial tone at the telephone set
2. The related LED will be lighted at the green color when the telephone set is pick up.

For Example: If the telephone set is connected to the second port of FXS in slot B. When you pick up the telephone set, the second LED will be at the green color, it means on, in slot B of front panel.

3. We can dial the desired destination phone number at the telephone set. If the desired destination phone number is legal, the VOIP-GATEWAY will play two quick address ack tones, Du Du, to destination. If the desired destination phone number is illegal, the VOIP-GATEWAY will play three quick out of service tone, please check the dial plan and your desired destination phone number.

### 5.2 Making a call with VOIP-GATEWAY FXO Module

Connection: analogue telephone set connects to FXO module port



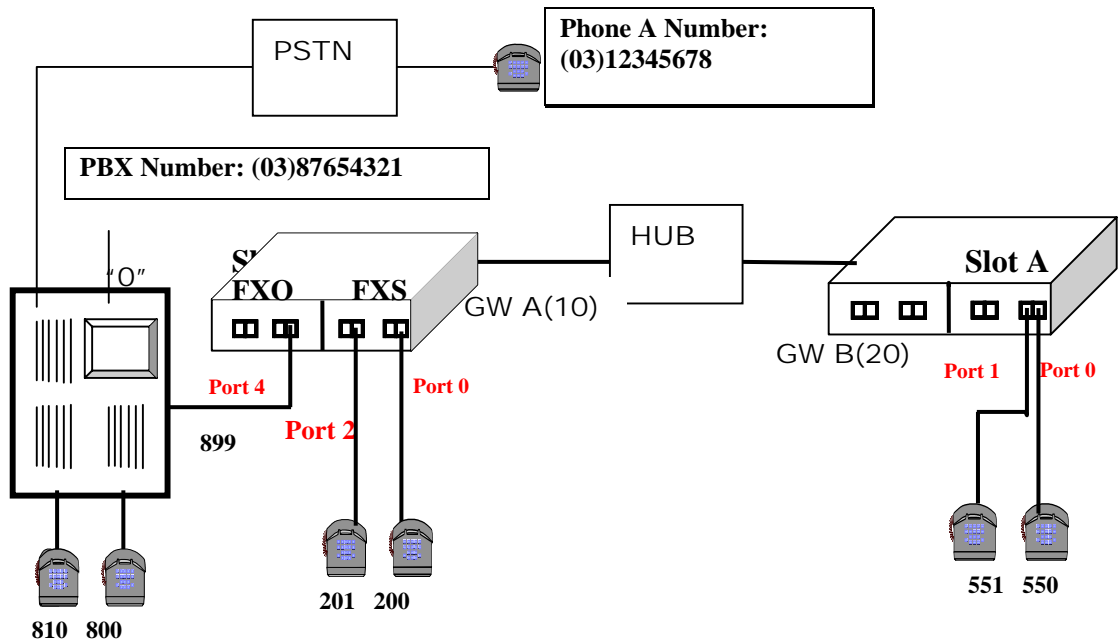
Operation:

1. Pick up this analogue telephone set, you can hear the PBX dial tone at the telephone set. In some cases, you have to dial a specific number to get the connection between telephone set and PBX.
2. Dial the phone number which is connected between PBX and VOIP-GATEWAY FXO module port.
3. When the VOIP-GATEWAY is connected, the related LED will be lighted green.

For Example: If the telephone set is connected to the third port of FXO in slot A. When the PBX is connected to VOIP-GATEWAY FXO port, the third LED will be green, it means on, in slot A of front panel.

4. We can dial the desired destination phone number on the telephone set. If the desired destination phone number is legal, the VOIP-GATEWAY will plan two quick sounds, “Du Du”, to destination. If the desired destination phone number is illegal, the VOIP-GATEWAY will play three out of service tones, please check the dial plan and your desired destination phone number.

### 5.3 Making a call with VOIP-GATEWAY Application Sample



**Scenario description:**

There are two gateways connected by a Hub. Gateway A is configured as following:

- 1 Gateway A IP: 192.168.0.20, mask IP: 255.255.255.0, gateway IP 0.0.0.0 (virtual IP)
- 2 Slot A plugs FXS module and slot B plugs FXO module
- 3 Port 0 of FXS in slot A has a telephone set connected, its phone number is 200

- 4 Port 2 of FXS in slot A has a telephone set connected, its phone number is 201
- 5 Port 4 of FXO in slot B is connected to PBX.
  - 5.1 There are two telephone sets connected by PBX, their phone numbers are 800 and 810.
  - 5.2 PBX has a number, 899, is connected to Port 4 of FXO in slot B.
  - 5.3 PBX has an outside call to PSTN. Digit 0 is connected to outside call.
  - 5.4 Phone number: (03) 87654321 is an PBX representative number. 03 is an area code.
- 6 There is telephone set, define phone A, its number is (03) 12345678.

Gateway B is configured as following:

- 1 Gateway B IP: 192.168.0.55, mask IP: 255.255.255.0, gateway IP 0.0.0.0 (virtual IP)
- 2 Slot A plugs FXS module and slot B is empty
- 3 Port 0 of FXS in slot A has a telephone set connected, its phone number is 550
- 4 Port 1 of FXS in slot A has a telephone set connected, its phone number is 551

**Gateway A and Gateway B dial plan setting**

Gateway A address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
200	1	3	3	3	None
201	3	3	3	3	None
8	5	3	3	3	None
0	5	1	1	0	None
20	11	5	5	2	None

Gateway A hunt group table

Group id	Type	#Members	Member ids
1	2	1	1
3	2	1	3
5	2	1	5
11	2	1	11

Gateway A destination table

Dest id	Mode	Destination
1	Local	PORT = 0
3	Local	PORT = 2
5	Local	PORT = 4
11	H.323	Dest = 192.168.0.55/1720 TCP

VOIP-GATEWAYB address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
550	1	3	3	3	None

551	2	3	3	3	None
10	11	2	5	2	None

VOIP-GATEWAYB hunt group table

Group id Type #Members Member ids

```
-----
```

1	2	1	1
2	2	1	2
11	2	1	11

VOIP-GATEWAYB destination table

Dest id Mode Destination

```
-----
```

1	Local	PORT = 0
2	Local	PORT = 1
11	H.323	Dest = 192.168.0.20/1720 TCP

**Making a call between gateway A and Gateway B**

**Case 1: Gateway Phone to Gateway Phone,**

Gateway B phone 550 calls to Gateway A phone 200

Human operation at GW B Caller side	Equipment operation	Human operation at GW A Receiver Side
Pick up phone 550	1.VOIP-GATEWAY dial tone is heard. 2.the first LED will be at the green color in slot A of front panel.	
Dial 10200	1.Du Du is heard 2.VoIP communication is going	
Ring back tone is heard	1.The first LED is flashing in slot A of front panel.	Phone 200 is ringing
		Pick up phone 200
Enjoy VoIP		Enjoy VoIP

The above process is the same as Gateway B phone 550 calls to Gateway A phone 201, 800 and 801.

**Case 2: Gateway Phone to PSTN Phone,**

Gateway B phone 550 calls to phone A number (03) 12345678

Human operation at GW B Caller side	Equipment operation	Human operation at Phone A Receiver Side
Pick up phone 550	1.VOIP-GATEWAY dial tone is heard. 2.the first LED will be at the green	

	color in slot A of front panel.	
Dial 100	1.Du Du is heard 2.VoIP Communication is going	
Ring back tone is heard	1.The first LED is flashing in slot B of GW A front panel.	
PSTN dial tone is heard	1.Gateway B is connected to gateway A by Hub 2.Gateway A is connected to PSTN	
Dial 12345678	1.PSTN communication is going	Phone 12345678 is ringing
Ring back tone is heard		Phone 12345678 is ringing
		Phone 12345678 picks up
Enjoy VoIP		Enjoy VoIP

The above process is the same as Gateway A phone 201 (301) calls to phone A number (03) 12345678

### Case 3: PSTN Phone to Gateway Phone,

Phone A number (03) 12345678 calls to Gateway B phone 551

Human operation at Phone A, Caller side	Equipment operation	Human operation at Gateway B phone 551, Receiver Side
Pick up phone A	1.PSTN dial tone is heard.	
Dial 87654321	1.Communication is going 2.PBX plays voice greeting	
Dial 899		
	1.The first LED lights in green color in slot B of front panel. 2.VOIP-GATEWAY dial tone is heard	
Dial 20551	1.Du Du is heard 2.VoIP Communication is going	
Ring back tone is heard		
	1.The second LED is flashing in slot A of GW B front panel.	Phone 551 is ringing
		Pick up phone 551
Enjoy VoIP		Enjoy VoIP

The above process is the same as Phone A calls to phone numbers 201, 301, 800, 801 or 550

## Chapter 6 Application Samples

This section discusses the various environments and situations where you can use the Internet Telephony Gateway.

### 6.1 Slide-in Module Configuration

Internet Telephony Gateway supports FXS and FXO modules.

FXS (Foreign Exchange Station)

FXS provides power and ringing signals to its interfacing equipment. It is *not* intended for connection to the Public Switched Telephone Network. It is a station loop start operation that provides a connection to:

1. a standard, single-line analog telephone system
2. the line circuit of a key telephone system
3. a loop start trunk circuit of a Private Branch Exchange (PBX) that normally connects to incoming Central Office circuits

FXO (Foreign Exchange Office)

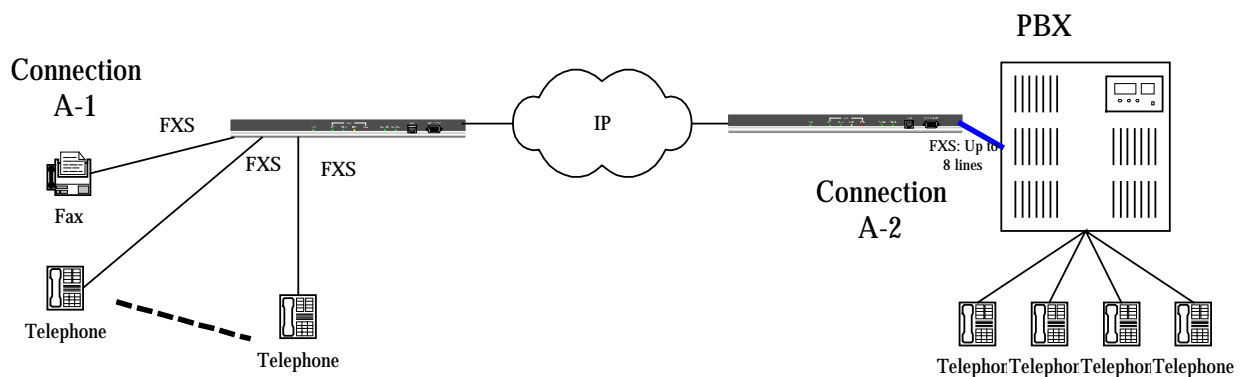
FXO recognizes signals and draws current to indicate an active state. It is a trunk loop start operation that emulates a single-line telephone to:

1. Central Office lines
2. PBX station

There are three basic environments that the VOIP-GATEWAY can be applied to. They are described in the following sections.

## Application A: FXS

Application-A has two possible connections. The A-1 connection shows the connection between the gateway to telephone set (POTS) or FAX machine. The FXS module is used in the A-1 connection. The other connection A-2, on the right hand side, is used to connect the gateway and PBX that has the CO trunk interface. FXS is also used in this connection. The telephone sets connected to PBX can be either POTS or digital telephone depending upon the PBX capability.



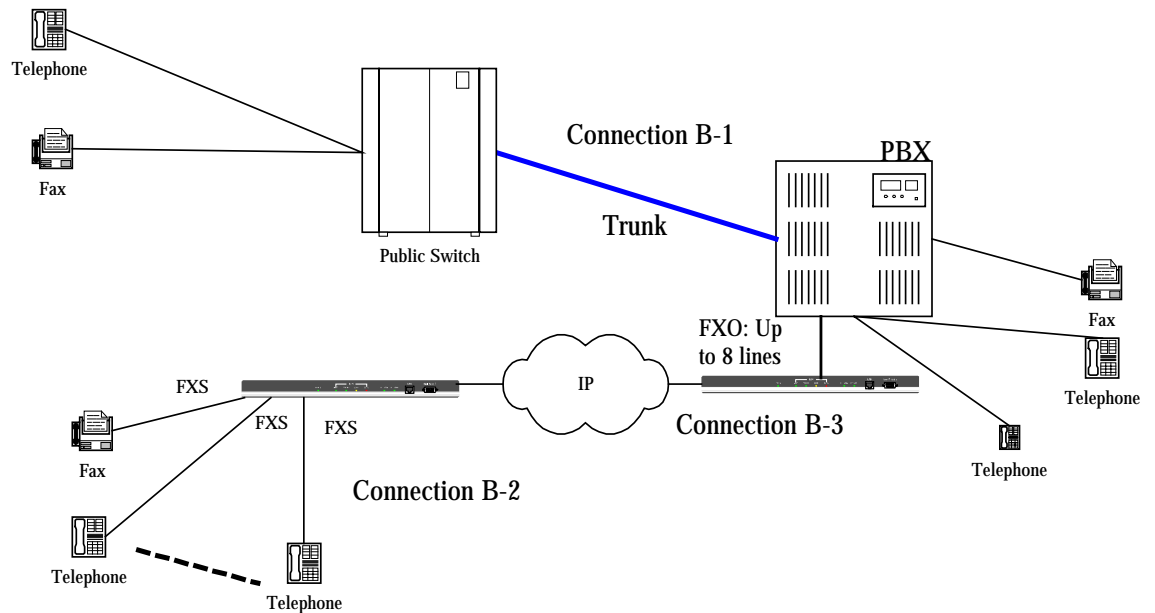
**Application B: FXS / FXO combination**

Application-B has three possible connections. The B-1 connection is a normal trunk connection between the CO switch and PBX. The B-2 connection is identical to the A-1 connection mentioned in Application A. The B-3 connection is used to connect the PBX and gateway through the internal analogue lines provided by the PBX. In this case, the FXO module has to be used, and the FXO ports will serve the same roles as the analogue telephone sets.

*Note: If the PBX used in the B-3 connection is a digital type, a certain analog terminal adapter is needed to provide an analog interface to the FXO module.*

In this application, The caller in the B-2 connection can use the Gateway through its FXS module to communicate with the party on the other end in the B-3 connection. The phone call is forwarded through the FXS module in the B-2 Gateway and is received by the FXO module in the B-3 gateway, which is connected to the PBX, which will then switch to the dedicated terminal.

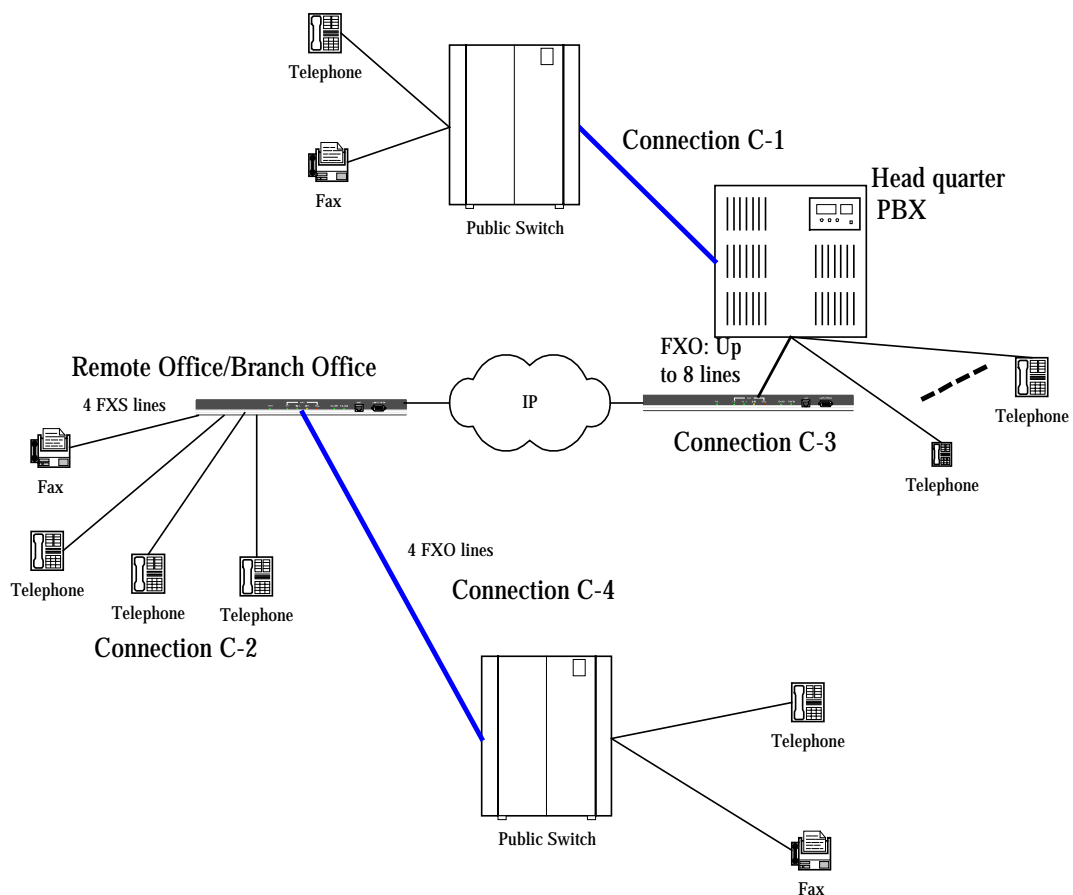
The same caller can also use the feature that the Gateway provides to redial the local number through the PBX, to the public switch, in order to reach the person inside the region that the public switch serves.



## Application C: FXS/FXO Extended Combination

Application-C has four possible connections. The C-1 connection is a normal trunk connection between the CO switch and PBX (identical to connection B-1 in the previous application.) The C-2 connection is identical to the A-1 connection mentioned in Application A. The C-3 connection connects the PBX and gateway through the internal analogue lines provided by the PBX (identical to the B-3 connection). The C-4 connection connects the CO switch to the gateway, in this connection the FXO module is also used (similar to the C-3 connection).

Of course, the C-2 connection can replace the C-3 connection, but it must then use an FXO module instead of an FXS module. In this application, the user can benefit from the VoIP application even when outside the office. With a proper dial plan setting, the user can make a phone call through a public switch back to the office, the PBX will relay the phone call to the other end which may be located in another city, state, or even country.



## 6.2 Application Samples

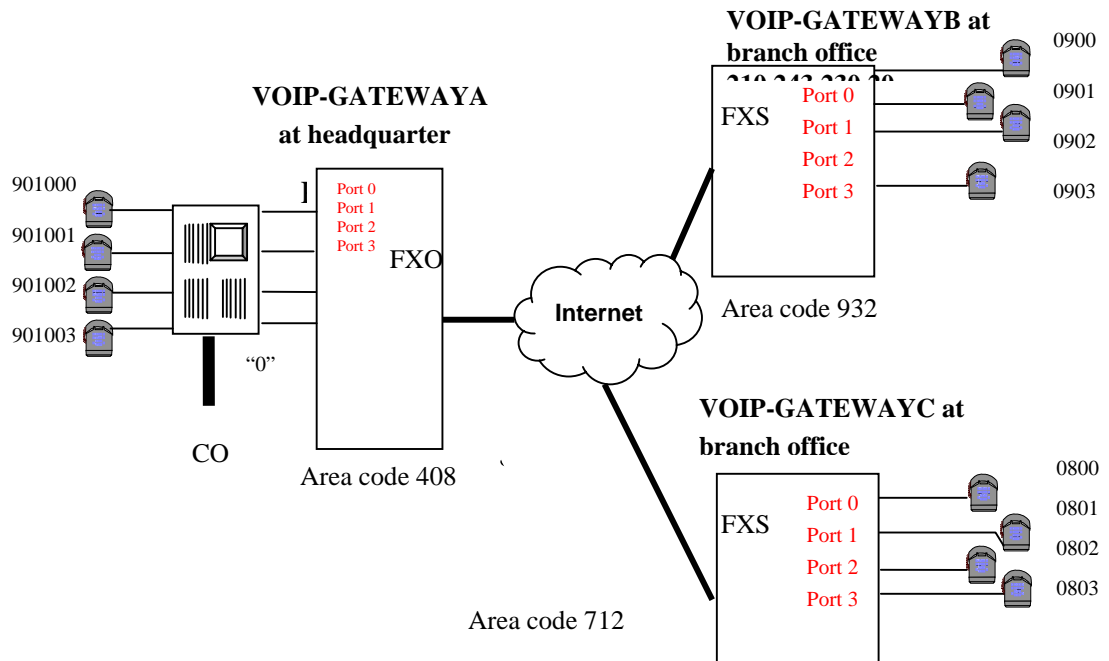
### Scenario 1 description

- 3 sets of VOIP-GATEWAYs were connected to Internet. One of VOIP-GATEWAY was built in a FXO module and the other two equipped with a FXS module.
- Telephone address description

VOIP-GATEWAYA: 408 and 415 are address specific PSTN connections on PBX CO trunk, and 90 are address specific connections on PBX internal lines on VOIP-GATEWAYA FXO ports

VOIP-GATEWAYB: 0900, 0901, 0902, 0903 are specific telephone address on VOIP-GATEWAYB FXS ports

VOIP-GATEWAYC: 0800, 0801, 0802, 0803 are specific telephone address on VOIP-GATEWAYC FXS ports



VOIP-GATEWAYA address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
90	1	2	6	2	None
408	1	3	10	3	"0"
415	1	3	10	3	"0"

09	11	2	4	0	None
08	12	2	4	0	None

VOIP-GATEWAYA hunt group table

Group id Type #Members Member ids

Group id	Type	#Members	Member ids
1	2	4	1 2 3 4
11	2	1	11
12	2	1	12

VOIP-GATEWAYA destination table

Dest id Mode Destination

Dest id	Mode	Destination
1	Local	PORT = 0
2	Local	PORT = 1
3	Local	PORT = 2
4	Local	PORT = 3
11	H.323	Dest = 210.243.230.20/1720 TCP
12	H.323	Dest = 210.243.230.30/1720 TCP

VOIP-GATEWAYB address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
0900	1	4	4	4	None
0901	2	4	4	4	None
0902	3	4	4	4	None
0903	4	4	4	4	None
90	11	2	6	0	None
408	11	3	10	0	None
415	11	3	10	0	None
08	12	2	4	0	None

VOIP-GATEWAYB hunt group table

Group id Type #Members Member ids

Group id	Type	#Members	Member ids
1	2	1	1
2	2	1	2
3	2	1	3
4	2	1	4
11	2	1	11
12	2	1	12

VOIP-GATEWAYB destination table

Dest id Mode Destination

Dest id	Mode	Destination
---------	------	-------------

1	Local	PORT = 0
2	Local	PORT = 1
3	Local	PORT = 2
4	Local	PORT = 3
11	H.323	Dest = 210.243.230.10/1720 TCP
12	H.323	Dest = 210.243.230.30/1720 TCP

VOIP-GATEWAYC address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
0800	1	4	4	4	None
0801	2	4	4	4	None
0802	3	4	4	4	None
0803	4	4	4	4	None
90	11	2	6	0	None
408	11	3	10	0	None
415	11	3	10	0	None
09	12	2	4	0	None

VOIP-GATEWAYC hunt group table

Group id Type #Members Member ids

```

-----
1      2      1      1
2      2      1      2
3      2      1      3
4      2      1      4
11     2      1      11
12     2      1      12
    
```

VOIP-GATEWAYC destination table

Dest id Mode Destination

```

-----
1      Local      PORT = 0
2      Local      PORT = 1
3      Local      PORT = 2
4      Local      PORT = 3
11     H.323      Dest = 210.243.230.10/1720 TCP
12     H.323      Dest = 210.243.230.20/1720 TCP
    
```

Add a FXO module to both VOIP-GATEWAYB and VOIP-GATEWAYC.

**Scenario 2 description**

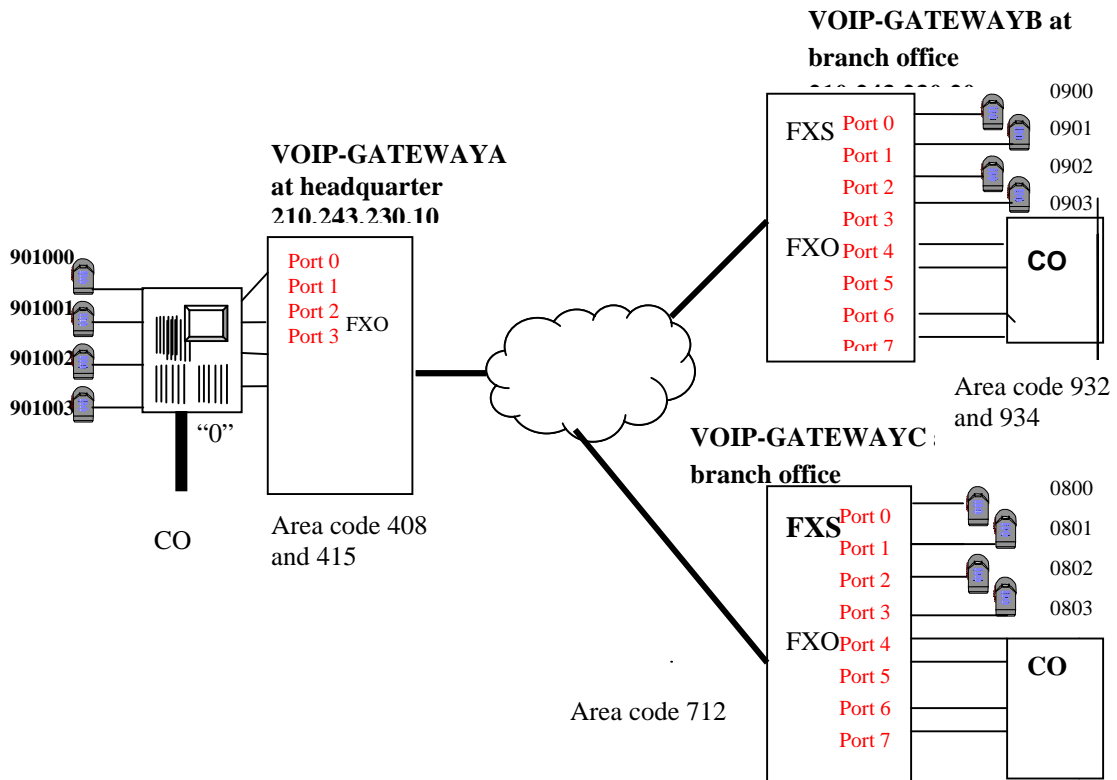
1. 3 sets of VOIP-GATEWAYs were connected to Internet and all of VOIP-GATEWAYs are built-in FXO and FXS modules.

2. Telephone address description

VOIP-GATEWAYA: 408 and 415 are address specific PSTN connections on PBX CO trunk, and 90 are address specific connections on PBX internal lines on VOIP-GATEWAYA FXO ports

VOIP-GATEWAYB: 0900, 0901, 0902, 0903 are address specific telephone on VOIP-GATEWAYB FXS ports, 932 and 934 are address specific PSTN connections on VOIP-GATEWAYB FXO ports

VOIP-GATEWAYC: 0800, 0801, 0802, 0803 are address specific telephone on VOIP-GATEWAYC FXS ports and 712 are address specific PSTN connections on VOIP-GATEWAYC FXO ports



VOIP-GATEWAYA address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
90	1	2	6	2	None
408	1	3	10	3	"0"
415	1	3	10	3	"0"
09	11	2	4	0	None
08	12	2	4	0	None
932	11	3	10	0	None
934	11	3	10	0	None

712	12	3	10	0	None
-----	----	---	----	---	------

VOIP-GATEWAYA hunt group table

Group id	Type	#Members	Member ids
1	2	4	1 2 3 4
11	2	1	11
12	2	1	12

VOIP-GATEWAYA destination table

Dest id	Mode	Destination
1	Local	PORT = 0
2	Local	PORT = 1
3	Local	PORT = 2
4	Local	PORT = 3
11	H.323	Dest = 210.243.230.20/1720 TCP
12	H.323	Dest = 210.243.230.30/1720 TCP

VOIP-GATEWAYB address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
0900	1	4	4	4	None
0901	2	4	4	4	None
0902	3	4	4	4	None
0903	4	4	4	4	None
90	11	2	6	0	None
408	11	3	10	0	None
415	11	3	10	0	None
08	12	2	4	0	None
932	5	3	10	3	None
934	5	3	10	3	None
712	12	3	10	0	None

VOIP-GATEWAYB hunt group table

Group id	Type	#Members	Member ids
1	2	1	1
2	2	1	2
3	2	1	3
4	2	1	4
5	2	4	5 6 7 8
11	2	1	11

12 2 1 12

VOIP-GATEWAYB destination table

Dest id	Mode	Destination
1	Local	PORT = 0
2	Local	PORT = 1
3	Local	PORT = 2
4	Local	PORT = 3
5	Local	PORT = 4
6	Local	PORT = 5
7	Local	PORT = 6
8	Local	PORT = 7
11	H.323	Dest = 210.243.230.10/1720 TCP
12	H.323	Dest = 210.243.230.30/1720 TCP

VOIP-GATEWAYC address translation table

Address Entry	Hunt Grp_Id	Min. Digits	Max. Digits	Prefix strip	Prefix Address
0800	1	4	4	4	None
0801	2	4	4	4	None
0802	3	4	4	4	None
0803	4	4	4	4	None
90	11	2	6	0	None
408	11	3	10	0	None
415	11	3	10	0	None
09	12	2	4	0	None
712	5	3	10	3	None
932	12	3	10	0	None
934	12	3	10	0	None

VOIP-GATEWAYC hunt group table

Group id	Type	#Members	Member ids
1	2	1	1
2	2	1	2
3	2	1	3
4	2	1	4
5	2	4	5 6 7 8
11	2	1	11
12	2	1	12

VOIP-GATEWAYC destination table

Dest id Mode Destination

-----  
1 Local PORT = 0  
2 Local PORT = 1  
3 Local PORT = 2  
4 Local PORT = 3  
5 Local PORT = 4  
6 Local PORT = 5  
7 Local PORT = 6  
8 Local PORT = 7  
11 H.323 Dest = 210.243.230.10/1720 TCP  
12 H.323 Dest = 210.243.230.20/1720 TCP

## Chapter 7 Troubleshooting Tips

This section provides solutions for problems you may possibly encounter while installing and connecting your Internet Telephony Gateway.

Power (PWR) LED is not illuminate

1. Check the power adapter connection.
2. If the power adapter or power cord is connected to the gateway, check that the cord is securely plugged into the power socket on the rear of the console.
3. Check that the other end of the power adapter or power cord is securely plugged into the power outlet.
4. If both ends of the power cord are properly connected and there is still no power, the gateway might have a faulty power outlet, power adapter, or power cord.
5. Change to another power outlet or obtain another power adapter or power cord.

LNK LED is not illuminate

1. Check the VOIP-GATEWAY was connected correctly:
  - a. Powered on
  - b. Correctly connected
2. Check if the cable connects to an end device is a standard straight through network Ethernet cable.
3. Make sure connectors at both ends are securely seated.
4. Check the VOIP-GATEWAY IP setting is correct

Slot A/B LED is not illuminate

1. Check if the slide-in module card plugged into the port is securely seated.
2. Make sure that the slide-in module card is not damaged.
3. If problems still occur, then the slide-in module card may not be defined.

Line module port LED is not illuminate when pick up the phone

1. Check Slot A/B LED illuminate
2. Check the RJ11 telephone line is connected correctly between phone set and line module port

There is no dial tone when pick up the phone

1. Check line module port LED illuminate
2. Check the RJ11 telephone line is connected correctly between phone set and line module port

There is an out-of-service tone when dialing destination phone number

1. Check the dial plan setting (address table, hunt group table and destination table) at the destination phone number

There is no connected tone when dialing destination phone number

1. Check the IP network (Ethernet cable, Ethernet port and GW IP setting) is connected correctly
2. Check the destination VOIP-GATEWAY and phone is available

#### Final Steps

If the procedures in this section have not solved the problem, reset the gateway several times by turning the power on and off. If the problem still exists, contact customer support.





