

VoiceFinder

AP200 VoIP Gateway

(Residential Gateway)

User's Guide



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Getting into the VoiceFinder AP200 User's Manual

This chapter provides the overview of the VoiceFinder 200 Gateway user's manual and an explanation of the symbols and legends involved.

[Composition of the Manual]

The VoiceFinder AP200 user's manual serves to assist the operation of the 1100 Gateway. This manual is composed of 5 chapters and 4 Appendixes as the following :

Those experienced with Gateways may refer directly to the chapters as needed. But those less experienced are highly recommended to thoroughly understand the manual before operation of the Gateway.

- Chapter 1 『 **VoiceFinder AP200 Overview** 』 provides an introduction to the hardware and software features of VoiceFinder AP200 and technical support request method.
- Chapter 2 『 **Before Installation** 』 provides the installation environment and cable requirements, along with recommendations for safe operation of the equipment.
- Chapter 3 『 **Installation and Operation Environment** 』 explains the basics for connecting with LAN, WAN and Console Port
- Chapter 4 『 **Gateway Configuration and Commands** 』 explains in detail about configuring the User Interface and the corresponding commands along with configuration examples. This chapter provides important information and requires comprehensive understanding.
- Chapter 5 『 **Voice Configuration and Commands** 』 explains in detail about configuring the User Interface and the corresponding commands along with configuration examples for Voice Integration. This chapter provides important information about maintaining and optimizing quality of voice and also requires comprehensive understanding.
- Appendix A 『 **VoiceFinder AP200 Specifications** 』 provides detail specifications for the VoiceFinder AP200 Gateway.

- Appendix B 『 **Example of Gateway Port Configuration** 』 provides examples of basic VoiceFinder AP200 Gateway configurations.
- Appendix C 『 **Example of VoIP Configuration** 』 provides examples of basic VoiceFinder AP200 Gateway VoIP configuration.
- Appendix D 『 **Cable Specification** 』 describes the console cable, Ethernet cable specifications and Pin numbers for the VoiceFinder AP200 Gateway.
- Appendix E 『 **Miscellaneous Information** 』 defines the VoiceFinder AP200 Gateway certificate of quality and related policies.

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VoiceFinder AP200 Gateway User's Manual's Revision history are as follows.



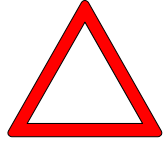
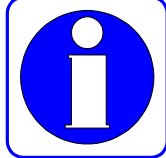
Revision No.	Date	Contents	Written By
Version 1.0	Oct. 8, 2001	Initial Released	AddPac R&D Part
Version 1.1	Jun. 1, 2002	Additional Command Released	AddPac R&D Part

[Symbols and Legends]

The symbols and legends used in this User's Manual are as follows :

- Commands and Keywords are typed in **Bold**.
- Variables that require user inputs are typed in *Italic*.
- Square brackets([]) are Optional values.
- Keywords that are required but need selection are grouped in braces({}) and are separated by Slashes(/).
- Angle brackets(<>) are required but appropriate parameters must be inputted.

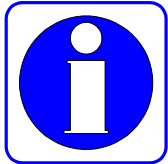
The following symbols are used for the user's reference in reading the user's manual.

<p>Danger</p> 	<p>Danger</p> <p>This symbol signals possible danger. Misuse could result in physical injuries. In procedures with this symbol, the user is strongly advised to follow safety regulations in order to avoid electrical shock.</p>
<p>Warning</p> 	<p>Warning</p> <p>This symbol warns the user that misuse in this procedure could result in hardware damage of the equipment or loss of data.</p>
<p>Caution</p> 	<p>Caution</p> <p>This symbol calls for the user's caution. Misuse in this procedure could result in software damage of the equipment, loss of data or system configuration.</p>
<p>Information</p> 	<p>Reference</p> <p>This symbol indicates reference, providing detail information for understanding the user manual.</p>

Chapter 1. VoiceFinder AP200 Overview

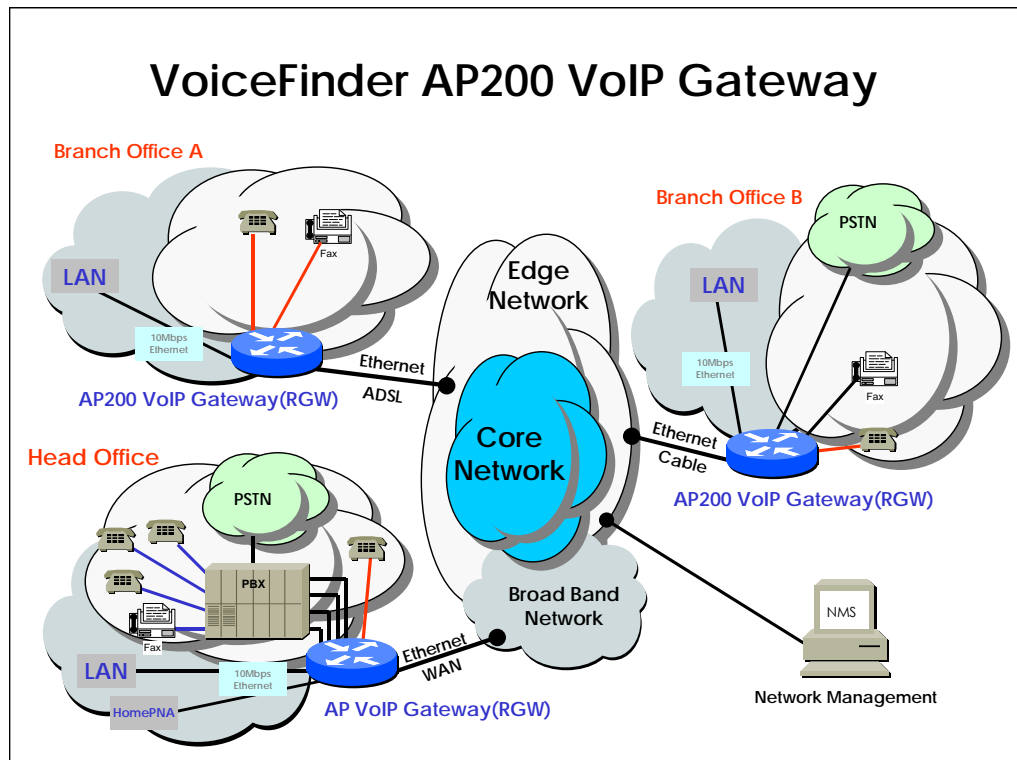
1.1. Introduction to the AP200 VoIP Gateway

Information



VoiceFinder AP200 VoIP Residential Gateway can be used in various network environments such as leased line, ADSL and Cable Modem networking with Static IP and dynamic IP address.

Figure 1-1 shows VoIP network inter-connection diagram as AP200 Residential Gateway is used for VoIP Service.



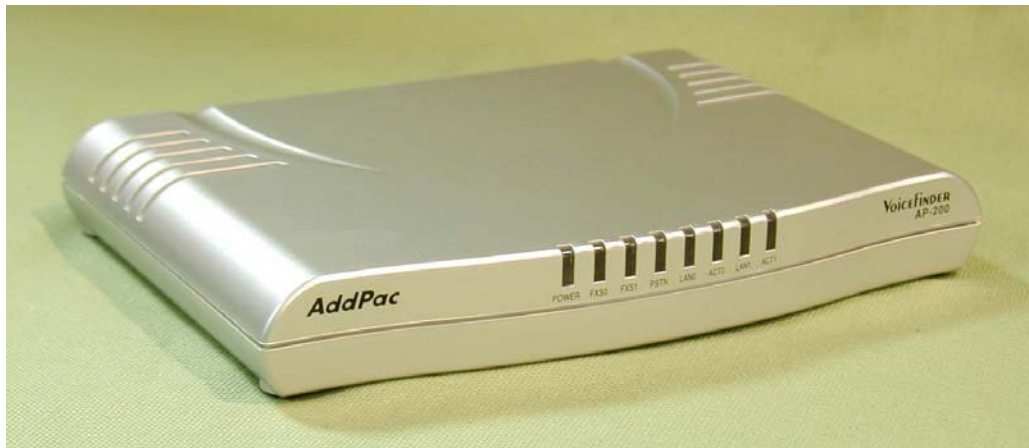
[Figure 1-1 Network Inter-connection Diagram using VoiceFinder AP200 RGW]

This equipment supports the static, default routing protocol and IEEE Spanning Tree bridging function. Therefore, AP200 RGW can be operated as a router equipment or a bridge system between WAN router system and local network.

And, this AP200 RGW supports inter-operability completely with carrier-class backbone gateways including small-size VoIP gateways such as CISCO, Clarent, 3COM, Lucent, etc. Also, this equipment provides the CISCO-style APOS Router Series operation and management features. This features help

to install the AP200 RGW easily.

Following Figure 1-2 shows the external form on the table of AP200 VoIP Residential Gateway.



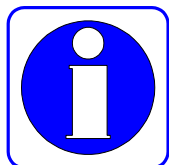
[Figure 1-2 VoiceFinder AP200 VoIP Residential Gateway's external form]

As supplementary service features of AP200 RGW, this equipment supports the Packet Filtering and Firewall function using Access List method. Firewall function restricts the local network access from outside network using source and destination IP address access list information at Network Layer(IP Layer) and Transport Layer(TCP/UDP Layer).

Also, using DHCP(Dynamic Host Configuration Protocol), this equipment can assign the IP address to network client automatically (DHCP server mode) and can receive the dynamic IP address of itself from DHCP Server(DHCP client mode). And AP200 VoIP RGW solves the IP address shortage problem due to explosive increment of internet user using NAT(Network Address Translation) protocol. The local IP address can be hidden from outside world when NAT protocol is used in AP200 VoIP RGW. This means that AP200 VoIP RGW can be used as a security function terminal simultaneously.

1.2. Main Features

Information



VoiceFinder AP200 VoIP Residential Gateway provides high-performance voice over IP(VoIP) solution including IP-routing services for small enterprise, government and other public offices as well as internet game room and cyber apartment.

Moreover, VoiceFinder AP200 VoIP Gateway maintains best voice quality under low bandwidth traffic using the latest voice compression algorithm and enhanced QoS management features of AddPac Technologies.

In addition, AP200 VoIP Gateway supports various network protocols such as IP routing, bridging, PPP, NAT/PAT, and network management features such as SNMP MIB v2, Web based Management, Cisco-Style CLI, etc.

This VoIP Gateway typically was designed using high-performance 32bit RISC Microprocessor based architecture with fixed network interface configuration : dual 10Mbps Ethernet, asynchronous serial port for console.

Hardware Description

VoiceFinder AP200 VoIP Residential Gateway supports the various network interface and voice interface module based on latest Embedded H/W technology and system memory. The main H/W features are followings.

- High Performance VoIP Residential Gateway
- High Performance WAN-to-LAN IP routing solution
- High Performance 32bit RISC Microprocessor Architecture
- Fixed 1-Port 10Mbps Ethernet for WAN Interface (RJ45)
- Fixed 1-Port 10Mbps Ethernet Interface for LAN Service (RJ45) /AP200B(Model B), AP200C(Model C)
- Fixed 4-Port 10Mbps Ethernet HUB Interface for LAN Service (RJ45) /AP200A(Model A)
- 1-Port FXS Voice Interface (1 x RJ11) / AP200A, AP200C
- 2-Port FXS Voice Interface (2 x RJ11) / AP200B
- Fixed 1-Port PSTN Backup Interface (1 x RJ11)

- Fixed 1-Port Asynchronous Serial Interface for Console Port (RJ45)
- Compact Design
- DC External Power Supply Adaptor
- Various System LED Display

Voice over IP Service

- Supports Voice and Data integration services
- Supports the maximum 2 FXS voice port
- ITU-T H.323 v2 VoIP Protocol with ITU-T H.235 Security Features
- Supports the G.723.1, G.729.A, G.711 Voice Compression Algorithms using High Performance DSP
- Supports Following Voice Processing Feature
 - ✓ VAD(Voice Activity Detection)
 - ✓ T.38 G3 FAX Relay(In-band 및 Out-band)
 - ✓ DTMF(Dual Tone Multi Frequency)
 - ✓ CNG (Comfort Noise Generation)
- G.168 Echo Cancellation
- Supports Enhanced QoS Management Features for Voice Traffics
- Supports the scalability, reliability, stability for H.323 based VoIP services
- Support Fast Connect Mode.
- Support H.323 call setup via normal connect mode if peer side does not support Fast Connect Mode.
- Support Voice Codec automatic negotiation function and Voice Codec Mode Setting by Operator.
- Support Adjustment of Frame Number per Packet
- Support the GK Discovery and Communication.(GRQ/GCF/GRJ)
- Support H.323 Endpoint Registration and Deregistration in GateKeeper (RRQ/RCF/RRJ, URQ/UCF/URJ).
- Support Lightweight RRQ Function
- Support VoIP Gateway H.323 ID Assignment, Modification and Transfer Function.
- Support Capital and Small letter Recognition of H.323 ID.
- Support H.323 E.164 Telephone Number Assignment, Modification and Transfer Function.
- Support Outbound Call routing function via GateKeeper.
- Support Inbound Call routing function using Phone number.
- Support Inbound/Outbound Call Number deletion and addition
- Support PBX side outbound call number Insertion in Calling Party Address.
- Support Voice Prompt for 1 Stage and 2 Stage Dialing
- Support Last digit recognition
- Call restriction about Specific Phone Number or Line
- Tone (Dial tone, Ring back tone, Busy tone, Congestion tone)

- LLO (Line Lock Out) Function
- Support Announcement Function for incorrect dialing, busy, network failure, Non-exist Phone number.
- Configuration Management and System Management
- Operation Data Automatic Backup and restore Function
- Port Diagnostic and Testing Function
- Support Secondary Gatekeeper in case of First Gatekeeper Failure
- H.225, RAS, H.245 Call Tracing Function
- DTMF transmission and Recognition Function (Out of band)
- FAX Tone automatic recognition
- Support Real-time FAX and Simultaneous FAX Transmission in All Channel (T.38)
- Redundancy Support in T.38 FAX
- Support PLAR(Private Line Auto Ring Down)
- Trunk Emulation for Broadcasting Equipment
- Ring Cadence Adjustment for Broadcasting Equipment
- BusyOut Function in case of LAN or Gatekeeper Failure (Rerouting at PBX)
- Call Pickup and Call Transfer between FXS port
- Polarity Inversion in FXS port
- Hook Flash Relay for supplementary service

IP Routing Protocols

VoiceFinder AP200 VoIP Residential Gateway supports various IP routing protocols. The main IP routing features are followings.

- High Performance IP routing capability with reliability
- Static, Default IP routing protocols
- Transparent Bridging (IEEE Spanning Tree Protocol) and VLAN routing protocols

Internet Service

VoiceFinder AP200 VoIP Residential Gateway supports following protocols and functions for internet access.

- Point-to-Point over Ethernet Protocol for ADSL Networking (PPPoE)
- Support DHCP Client function
- Support IP Sharer function

Network Managements

VoiceFinder AP200 VoIP Residential Gateway supports various network management protocols and functions.

- Standard SNMP Agent (MIB v2) Support
- Remote Management using Console, Rlogin, Telnet.
- Traffic Queuing
- Web based Managements using HTTP sever.

Security Functions

VoiceFinder AP200 VoIP Residential Gateway supports various security function.

- Standard & Extended IP Access List for network security function.
- Access Control and Data Protections
- Enable/Disable for Specific Protocols
- Multi-level User Account managements
- Auto-Disconnect for Telnet/Console Sessions

Operation and Managements

VoiceFinder AP200 VoIP Residential Gateway supports following operation and maintenance features.

- System Performance Analysis for Process, CPU, Connection I/F
- Configuration Backup & Restore for APOS Managements
- Debugging, System Auditing, and Diagnostics Support
- System Managements with Data Logging

- System Booting and Auto-rebooting with Watchdog Feature
- IP Traffic Statistics with Accounting

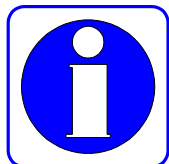
Other Scalability Features

VoiceFinder AP200 VoIP Residential Gateway supports various supplementary service functions.

- DHCP Server & Relay Functions
- Network Address Translation (NAT) Function
- Port Address Translation (PAT) Function
- Transparent Bridging (IEEE Standard) Function
 - Spanning Tree Bridging Protocol Support
 - Remote Bridging Support
 - Concurrent Routing and Bridging Support
- Cisco Style Command Line Interface(CLI)
- Load Balancing Support
- Network Time Protocol(NTP) Support
- Remote S/W Upgrade for APOS Management using FTP/TFTP
- Automatic remote S/W Upgrade for APOS Management using HTTP

1.3. APOS Internetworking Software

Information This part explains about Software Architecture of APOS(AddPac VoiceFinder Operating System).



APOS Internetworking Software for AP PassFinder Router, VoiceFinder Gateway

APOS Operating System is a best VoIP Gateway and Router software to provide scalability, reliability, stability, and QoS for IP(Internet Protocol) based Inter-networking solutions.

This APOS Operating System is designed using light, compact, and reliable Embedded Real-Time OS.

Also, APOS provides optimized performance functionality with easy-to-use, easy-to-installation, and maintenance.

Industry Standard Network Protocol Support

APOS Operating Software supports the industry standard network protocols. For example, WAN/LAN data networking protocols(HDLC,PPP,etc), ATM related protocols such as PPPoA (Point-to-Point Protocol over ATM), IPoA (IP over ATM), Network Management Protocols(SNMP v2,v3, Web based Management), VPN* based on IPsec, and various supplementary and usable protocols are supported by APOS.

Integrated Networking Solution

APOS Operating Software provides VoIP(Voice over Internet Protocol) solution including data networking solution. As VoIP protocols, H.323 v2, SIP*, and MGCP/MEGACO* protocols are supported by APOS. Using these Voice and Data Inter-networking protocols, APOS can be used for the various IAD(Integrated Access Device) application such as ATM Multi-service VoIP Router, Leased-Line VoIP Router, Frame-Relay VoIP Application, etc.

Optimized Performance and Functionality

APOS Operating Software supports the excellent packet processing performance, various QoS Schemes, and overload control capability. The Packet Processing Capability of APOS is implemented by Light Embedded Real-Time Kernel, Optimized Device Driver(LAN Device, DSP, etc), Cache Control Technique(I-CACHE, D-CACHE), Heuristic Polling and Interrupt based Device Access Control Schemes, Optimized TASK Priority Control, etc.

The VoiceFinder QoS schemes of APOS can be used to improve the voice quality. The technique used for QoS function can be divide to two side in view of VoIP gateway terminal. First, in transmit side, Priority Queuing, QoS Profile Scheme, Virtual Network Transmit Algorithm, RTP Packet Transmit Interval Control, RTP Redundancy Scheme, IP Header Field (TOS, DiffServ, etc) Control Techniques are applied in VoiceFinder VoIP Gateway for Transmit Side QoS Control. In receive side, APOS VoiceFinder Operating Software use Dynamic Jitter Buffering, Error Concealment and T.38 Fax data error recovery techniques to improve the receive side voice quality.

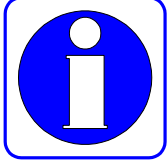
The overload control function supported by APOS sustains the reliable system operation when overload traffic is occurred in VoIP Gateway or Router System.

Easy to use, Easy to Installation, and Maintenance

For easy installation and simple usability, APOS Operating Software supports simple maintenance and inter-operability using Cisco-Style command line interface. Moreover, it is possible to simple maintenance using web based management schemes and remote management functions

1.4. Hardware Configuration and Network Interface

Information This Chapter explains the external form and network interface module parts of VoiceFinder AP200 VoIP Residential Gateway.



1.4.1. VoiceFinder AP200 Main Body

The external form of AP200 VoIP Residential Gateway is made by high degree of strength ABS material. The front panel of this equipment is designed to display the LED that indicates the system operation status. The back panel of AP200 VoIP gateway is designed to support the flexible voice and LAN interfaces including RS-232C serial port.

Front View of AP200

To check the system operation status easily, AP200 VoIP Residential Gateway provides the Front panel LED. Following Figure 1-3 and 1-4 shows the Front Panel external form and represents the explanation of each parts.



Figure 1-3 Front Panel Picture of VoiceFinder AP200 VoIP Residential Gateway

Table 1-1 shows the functional explanation about each LED of VoiceFinder AP200 VoIP Residential Gateway.

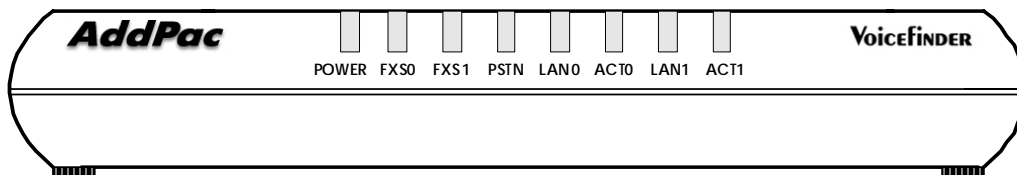


Figure 1-4 Front Panel Diagram of VoiceFinder AP200 VoIP RGW

No.	LEDs	Explanation
(1)	POWER	Power LED, display whether external power is supplied normally or not. (Green)
(2)	FXS 0	Display the Operation Status of "FXS 0" Voice Port. (Yellow)
(3)	FXS 1	Display the Operation Status of "FXS 1" Voice Port. (Yellow)
(4)	PSTN	Display the Operation Status of PSTN Backup Port. (Red)
(5)	LAN 0	LAN 0 LED, display the Ethernet port's link up status. (Green)
(6)	ACT 0	Display the LAN0 Ethemet port's activity status such as transmit and receive packet. (Yellow)
(7)	LAN 1	LAN 1 LED, display the Ethernet port's link up status. (Green)
(8)	ACT 1	Display the Ethemet port's activity status such as transmit and receive packet. (Yellow)

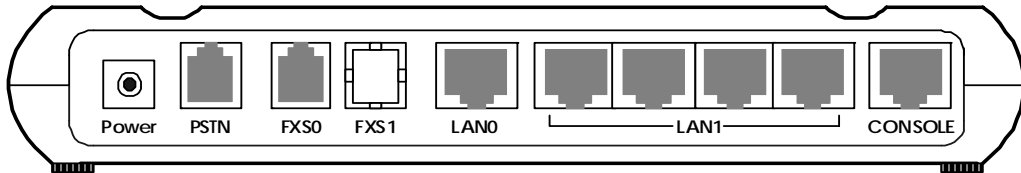
[Table 1-1 The explanation of front panel LEDs of AP200 VoIP Gateway]

Rear View of AP200

In Rear Side of AP200 VoIP Residential Gateway, this equipment provides the two 10Mbps Ethernet interfaces, RS-232C serial port for management, and FXS, FXO interface ports for voice signal processing. Two Ethernet ports are used for WAN side and LAN Side Ethernet interface each. Using two Ethernet ports, AP200 supports the various WAN-to-LAN Routing Service and QoS services.

AP200-A Type Model

Figure 1-5 shows the “Type A” Interface configuration of AP200 VoIP Residential Gateway. (AP200 Model A)



[Figure 1-5 Back Side Panel Diagram of VoiceFinder AP200 Model A VoIP RGW]

Following Table 1-2 explains the “Type A” interface of AP200 VoIP Residential Gateway.

No.	Interface	Explanation
(1)	DC Power	External DC Power Supply Input. (DC 5V)
(2)	PSTN	This port is used to connect PSTN Analog Line for PSTN backup. (RJ11)
(3)	FXS 0	1 FXS Port : As FXS Voice Interface Port, Phone or Fax Machine can be connected.(1x RJ11)
(4)	LAN0	10Mbps Ethernet Port for WAN interface (RJ45)
(5)	LAN1	10Mbps Ethernet Port for LAN interface. In case of AP200 Model A Model, 4-Port HUB Interface is provided (RJ45)
(6)	Console	RS-232C serial port for system management (RJ45)

[Table 1-2 The explanation of Back Side Panel Interfaces of AP200 Model A RGW]

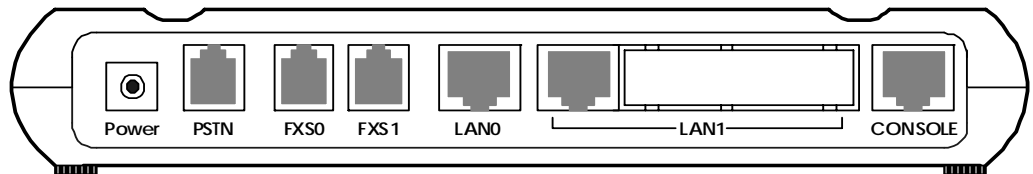
Following Figure 1-6 shows the Back panel interface picture of AP200 Model A VoIP Residential Gateway.



[Figure 1-6 Back Side Panel Picture of VoiceFinder AP200 Model A VoIP RGW]

AP200-B Type Model

Figure 1-7 shows the "Type B" Interface configuration of AP200 VoIP Residential Gateway. (AP200 Model B)



[Figure 1-7 Back Side Panel Diagram of VoiceFinder AP200 Model B VoIP RGW]

Following Table 1-3 explains the "Type B" interface of AP200 VoIP Residential Gateway.

No.	Interface	Explanation
(1)	DC Power	External DC Power Supply Input. (DC 5V)
(2)	PSTN	This port is used to connect PSTN Analog Line for PSTN backup. (RJ11)
(3)	FXS (0,1)	2 FXS Ports : As FXS Voice Interface Ports, Phone or Fax Machine can be connected. (2x RJ11)
(4)	LAN0	10Mbps Ethernet Port for WAN interface. (RJ45)
(5)	LAN1	10Mbps Ethernet Port for LAN interface. (RJ45)
(6)	Console	RS-232C serial port for system management. (RJ45)

[Table 1-3 The explanation of Back Side Panel Interfaces of AP200 Model B RGW]

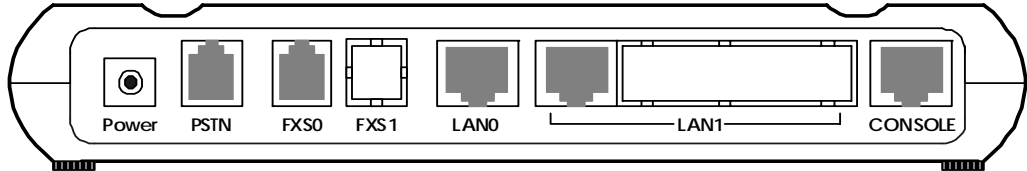
Following Figure 1-8 shows the Back panel interface picture of AP200 Model B VoIP Residential Gateway.



[Figure 1-8 Back Side Panel Picture of VoiceFinder AP200 Model B VoIP RGW]

AP200-C Type Model

Figure 1-9 shows the “Type C” Interface configuration of AP200 VoIP Residential Gateway. (AP200 Model C)



[Figure 1-9 Back Side Panel Diagram of VoiceFinder AP200 Model C VoIP RGW]

Following Table 1-4 explains the “Type C” interface of AP200 VoIP Residential Gateway.

No.	Interface	Explanation
(1)	DC Power	External DC Power Supply Input. (DC 5V)
(2)	PSTN	This port is used to connect PSTN Analog Line for PSTN backup. (RJ11)
(3)	FXS 0	1 FXS Port : As FXS Voice Interface Port, Phone or Fax Machine can be connected.(1x RJ11)
(4)	LAN0	10Mbps Ethernet Port for WAN interface. (RJ45)
(5)	LAN1	10Mbps Ethernet Port for LAN interface. (RJ45)
(6)	Console	RS-232C serial port for system management. (RJ45)

[Table 1-4 The explanation of Back Side Panel Interfaces of AP200 Model C RGW]

Following Figure 1-10 shows the Back panel interface picture of AP200 Model C VoIP Residential Gateway.



[Figure 1-10 Back Side Panel Picture of VoiceFinder AP200 Model C VoIP RGW]

1.4.2. Network Interface

The Standard Model of AP200 VoIP Residential Gateway supports network interfaces such as following lists.

- **2 Port LAN 0/1 10Mbps Ethernet Interface or 1port LAN0 and 4Port HUB (LAN1)**
- **1 Port RS232C Asynchronous Serial Interface**

Above listed network interface can be used in various WAN/LAN network environment for VoIP network integration. The Leased Line, xDSL, and Cable networks are one of networks supported by AP200 VoIP RGW.

The next explains the network interfaces of AP200 VoIP Residential Gateway.

2-Ports 10Mbps Ethernet Interface for LAN 0/1 (2 x RJ45)

AP200 VoIP Residential Gateway supports the two 10Mbps Ethernet Interfaces. Using these LAN Interfaces, various network application services are possible. In case of AP200 Model A RGW, LAN1 Interface supports the 4-Port HUB 10Mbps Ethernet Interface instead of 1-Port 10Mbps Ethernet Interface.

This interface is implemented using standard RJ-45 connector interface.

1-Port Async Serial Interface for Console Port (RJ45)

AP200 VoIP Residential Gateway provides the network management features using Asynchronous serial interface port.

This interface is implemented using standard RJ-45 connector interface.

Voice Processing Interfaces

The FXS type Voice Processing Modules on board of AP200 RGW are Analog Line Interface solutions which can interface to general Phone, FAX Machine, PBX.

- **FXS Interface Voice Processing Module**
- **PSTN back up port**

Using these Voice Interface Modules, AP200 RGW provides data networking service as well as VoIP service.

Note : To know the detailed Voice Interface configuration, please reference the order information of AP200 RGW.

1~2-Ports FXS Voice Interface(RJ11)

AP200 VoIP Residential Gateway supports One(1) or Two(2) Port FXS(Foreign Exchange Station) Voice Interface Module . Using these FXS Voice Interface Module, AP200 VoIP RGW provides analog Line Interface solutions which can interface to general Phone, FAX Machine, etc.

1-Port PSTN Backup Voice Interface Module

AP200 RGW supports One(1) Port RJ11 connector type PSTN backup Interface port which can be connected to analog subscriber module of PSTN switch directly. This backup Interface can be used for voice communication via PSTN when VoIP service is not available due to network fault, Gatekeeper problem, etc .

Chapter 2. Before Installation

2.1. Unpacking

Before unpacking, check for external damage of the packaging box .

If an external damage of the packaging has been found, please contact AddPac Technology Co. Ltd. sales department (sales@addpac.com, tel : +82-2-568-3848) for an immediate exchange of product.

If no external damage has found, confirm if the following items are enclosed.

No.	Item	Content	Q'ty
1	VoiceFinder AP200 Gateway Main Body		1
2	LAN Cable (for RJ45 to RJ45)		1
3	Console Port Cable (for RJ45 to DB9)		1
4	External Power and Power Cable (220V Power Cord)		1
5	AP200 User's Guide		1

[Table 2-1. VoiceFinder AP200 Package]

If any item is missing, immediately contact AddPac Technology Co. Ltd.
customer support

2.2. Installation Requirements

Warning

The following is the recommendation for safe operation of the equipment.



- Ensure the AP200 VoIP Gateway is in a dust-free environment before and after installation.
- Ensure the AP200 Gateway cover is opened on a flat and safe surface.
- To prevent accidents, avoid ties, scarf, sleeves, and any other loose clothing from entangling with the Chassis.
- Avoid any actions that may effect the equipment or the operator.

2.2.1. Electrical Requirements

Danger

There are two main sources of electrical problems with the AP200 Gateway : the power supply and static electricity.



This section describes safety recommendations for each case.

- **Electrical Safety**
 - ✓ In case of the occurrence of an electrical accident, operate at a position where immediate shut-off of power supply is possible.
 - ✓ Switch the power off when installing or taking the cover off the equipment.
 - ✓ Avoid operating the equipment alone at a potentially dangerous environment.
 - ✓ Do not assume the power is switched off, but always confirm the power status.
 - ✓ Be extremely cautious when operating in a humid environment or with an uncovered power extension cable.

- **Prevention of Static Electricity**
 - ✓ The main chip-set of the Gateway are very delicate and misuse may result in static electrical damage.
 - ✓ If a static prevention waist strap is available, strap it around the wrist and earth the cord before operating the equipment.
 - ✓ If no waist tap is available, earthing by holding a metal part of the Chassis will help prevent static electricity.

2.2.2. General Requirements

Warning



The VoiceFinder 1100 Gateway is ready for use where electronic products are used. However, a location with the following conditions are recommended for maximum performance.

- **A level and well ventilated location is recommended.**
- **Secure the equipment safely where intended to install.**
- **Avoid placing objects on top of the equipment.**
- Install the equipment in a cool location avoiding direct sunlight.
- Maintain distance from flammable, chemical, or magnetic objects

2.2.3. Prerequisites for Network Connection

Observe EIA standards and limitations when installing the Gateway

Warning



The following section describes the Ethernet Cable, and the Console Cable VoiceFinder AP200 supports.

2.2.3.1. Ethernet Port

2.2.3.2. RS-232C Serial Console Port

Chapter 3. Installation and Operating Environment

This chapter provides information about the basic installation procedure of VoiceFinder AP200 and related commands.

[Prerequisites] Unless a separate order is made, the tools and certain cables are not provided in the package. Prepare the following equipments and tools before installation.

- Standard screwdriver set

- Cable for LAN and WAN(Serial) port connection
 - ✓ RJ-45 to RJ-45 cable for LAN port
 - ✓ RS-232c console cable with RJ-45 connector (included in equipment box)

- Cable for connecting with phone port
 - ✓ RJ-11 to RJ-11 ordinary phone-line cable

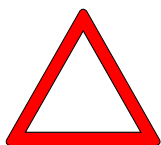
- PC with Console Terminal or Communication Emulator application (The Hyper Terminal Program in Windows will suffice. Configure as : 9,600 Baud, No Parity, 8Bit Data 1Stop Bit)

3.1. Installation

3.1.1. Installation Procedure

- Plug in the console cable and configure the console terminal. (Refer to 3.1.2 for details.)
- Connect the network to the desired port. Use RJ-45 cable with the LAN 1 port for connection with ADSL/Cable Modem or Router, and the RJ-45 cable with the LAN 0 port for connection with the HUB/Switch.
- Log in the Gateway after the booting message on the console with the root account. (Configuration is only possible when logged in with the root account.)
- Switch to Configuration Mode.
- Assign an internet address to the desired port of usage. (Refer to Interface Configuration.)
- Configure the settings for the routing and VoIP related. (Refer to Chapter 4 and 5.)
- Confirm the configured settings. (Refer to the Gateway administrative commands.)

Caution



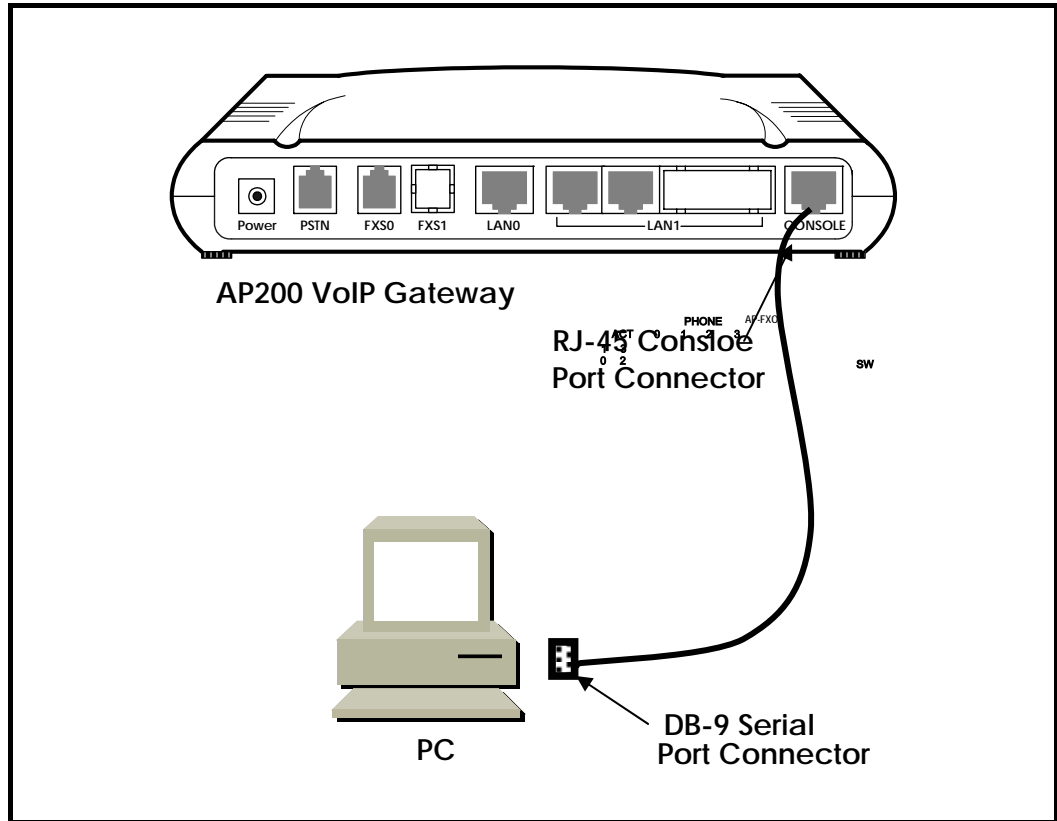
- Save the settings on Flash Memory. (**VoiceFinder AP200 immediately operates under the new settings, but under certain network environments rebooting is required.**) (Refer to the following instructions for this part.)
- Using commands such as Ping, Telnet, rlogin etc. check the status of other Gateways or PCs connected to the Gateway.

- Check the routing table to confirm if the Gateway is receiving the network information correctly.
- Use the Ping command to check other Gateways or PCs connection.
- This completes the basic configuration procedure. For optional functions, refer to the related chapter

3.1.2. Console Connection

- Connect the console port in the rear side of the Gateway with the serial port of the prepared console terminal. **(Refer to [Diagram 3.1 Console Cable Connection])**
 - ✓ Use console cable provided in the package.
 - ✓ If using a PC for the console terminal, connect with the PC serial port.
- In order to use the PC as a console terminal, a communication emulator application is required. Under normal circumstances, the Hyper Terminal Program in Windows will suffice.

The console terminal should be configured as : 9,600 Baud, No Parity, 8Bits Data, One(1) Stop Bit. The VoiceFinder AP200 Gateway is set to operate with the configurations above. Therefore, these settings are required for communication between the Gateway and the console terminal. **(Refer to [Diagram 3-2 Hyper Terminal Configuration])**
- When configuring the Hyper Terminal, select the Hyper Terminal menus in the following order: File → Configuration → Connection Target → Format and set each item.
- The console port serves the purpose of configuring the VoiceFinder 1100 Gateway settings and checking its operating status.



[Diagram 3.1 Console Cable Connection]

Port Configuration	Settings
Modem to Connect	Direct connection (Null Modem) to Com port
Bit per Second	9,600
Data Bit	8
Parity	None
Stop Bit	1
Flow Control	None

[Diagram 3.2 Hyper Terminal Configuration]

3.1.3. Plug-In Power

Warning



- The VoiceFinder AP200 Gateway is capable of recognizing and using both 110V and 220V.
- The package provides a 220V power cable. If the power supply is 110V, please use a 110V adapter.
- Switching the power switch on will turn the Power LED on the front side of the Gateway green.
- The supply of power will display the booting message on the console terminal and will turn the RUN LED green
- When the Gateway is being booted, the messages are displayed : (Refer to Diagram 3.3)
 - ✓ The booting title message is displayed. (This message contains information about the routing software version, Gateway status check results, memory size and status)
 - ✓ On the display of the log-in message, input the username "root" and the password "Gateway".
 - ✓ The completion of log-in will display the prompt "1router#" on the Gateway console terminal.
 - ✓ There are two types of prompts displayed for the VoiceFinder AP200 : "1router>" and "1router#". The " > " prompt indicates that the user is not an administrator. At this prompt, the user is unable to use certain commands : particularly the configuration commands. The " # " prompt indicates that the user is an administrator(or root), and is authorized to use all the functions and commands.

- ✓ Log-in as "Admin" allows Gateway setting configuration. Therefore, it is advised to change the Gateway password for security purposes. Refer to the Administrative contents for password change.

The display below is the message for initial booting of the VoiceFinder AP200 Gateway.

```
System Boot Loader, Version 1.3.6/1
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

VoiceFinder Gateway Series (AP200)
Serial Number: AP200-010102
32BIT RISC Processor With 33554432 Bytes System Memory
524288 Bytes System Flash Memory
4194304 Bytes 2nd System Flash Memory

1 RS232 Serial Console Interface
1 Ethernet/IEEE 802.3 Interface
1 Ethernet/IEEE 802.3 or HomePNA Interface

RGW System software Revision 5.44Tg
Released at Sun Sep 3 10:32:11 2000
Program is 3494964 bytes, checksum is 0x103c5579

Local Time : Thu Jan 1 02:01:56 1970

Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

The system is not configured yet or backup data is invalid.
Please login to system as a "root" and make configuration.

Voice Module (0): FXO
Voice Module (1): FXS

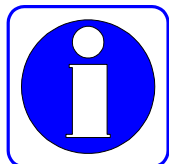
DSP S/W download
Voice Module (0): .... OK
Voice Module (1): .... OK

The System is ready. Please login to system.
login:
```

[Diagram 3.3 VoiceFinder AP200 Initial Screen]

3.2. Environment Configuration

Information



The Gateway requires various configurations of parameters according to the application. This section provides important information for using the Gateway and the user is highly advised to follow the following procedures before configuring the Gateway.

- Clarify the network address according to the IP with the network diagram.
- Determine which routing protocol to use. (e.g. Static, Default etc.) In doing so, discussion with the administrator of the connecting network is required.
- Determine the protocol to use with each LAN 1(WAN) port. (e.g. PPPoE, Ethernet etc.)
- When the conditions above are determined, thoroughly understand the related commands.

Environment configuration is required only once at initial installation. But when the network components have changed, settings must be reconfigured. At completion of configuration, always save the settings to prevent loss of data when switching the power on/off.

In order to log in at an unconfigured Gateway, the user must use a set username and password. The user access authority for VoiceFinder 1100 is divided into 4 levels : admin, high, normal, low. All users, other than Admin is prompted "1router>" at log-in.

The user must log-in at admin level for Gateway configuration. **Factory settings for admin level log-in uses "root" for username and "router" for password. The user is prompted with "1Gateway#" for Admin level log-in and is allowed to configure the equipment.**

VoiceFinder 1100 configuration is divided into two parts : Global Configuration,

which effects all the Gateways of the network, and Interface Configuration, which effects only the Gateway of configuration. By function, configuration is divided into: "User and Gateway Administration", "Interface Configuration", "Routing and Bridging Configuration", "System Status and Debugging Configuration", etc.

This manual describes configuration according to its functions.

3.2.1. User and Gateway Management Environment

The Gateway may be accessed through console connection or telnet. The VoiceFinder AP200 Gateway allows 1 connection through console session, and 512 connections through application sessions such as telnet, FTP, SNMP, etc. The sessions may effect the Gateway performance, therefore the user is advised not to connect more than 10 sessions

This configuration allows the setting of the user password. The default username for VoiceFinder AP200 Gateway is "root" and its password is "router". (This default setting is for "admin" level access.) When the Gateway configuration is completed, a change of password is advised. This is to prevent unauthorized users from reconfiguring the settings. The VoiceFinder AP200 Gateway saves the password and the configuration in safety area.

This configuration mode allows Gateway software upgrade, and commands related to system administration such as configuration saving and backup.

It also provides commands for checking the system status. These include commands for displaying CPU resource availability, Debugging commands to show packets received and dispatched by the Gateway, and Show commands to show the configuration status.

3.2.2. Interface Configuration Environment

In order to communicate in an Ethernet and WAN(LAN or HomePNA Port) environment, an IP address must be configured for each port. For commands related to port IP address connection, refer to interface related commands. For the WAN(SERIAL) port, configuration for lower level protocols is required as well as IP address.

The LAN 1(for WAN) ports for VoiceFinder AP200 are supports Legacy LAN, PPP(PPPoE) for ADSL connection. In order to connect to the network, the WAN protocol must match the one used at the other equipment, including configuration variables. Discussion with the administrator of the other equipment is recommended.

The Interface configuration mode, allows traffic management of particular packets per interface. For security related Access-List, DHCP information, refer to the "Configurations for Security and Internet" section.

For packet management information, refer to the "Configurations for Routing" section.

3.2.3. Routing Configuration Environment

The VoiceFinder AP200 Gateway supports Static, Default routing protocols. The routing protocol is responsible for assignment of packet route, and VoiceFinder AP200 is supports multi-protocols simultaneously. Therefore it is required to configure which protocol to use. Refer to "Routing Configuration Environment" and "Interface Configuration Environment".

3.2.4. Security and Internet Configuration Environment

The VoiceFinder AP200 Gateway supports additional functions for security and internet environment.

The functions provided include Packet Filtering, Access-List, NAT(Network Address Translation), PAT(Port Address Translation) and Multi-Level account for security and DHCP server, client and relay for internet connection. Refer to Chapter 4 for details

3.2.5. System Status and Debugging Environment

The VoiceFinder AP200 Gateway supports the "Show" command for checking the system operation status and the "Debug" command for locating system errors. The "Show" commands not only provides information about the status of interface, but also status for NAT configuration, Access-list, DHCP, registered user, buffers and all others assisting proper operation of the Gateway.

The "debug" command provides information regarding proper operation of the Gateway by displaying operating TCP/IP or Layer 2 on the terminal screen.

For more details, refer to Chapter 4.

3.2.6. Voice Integration Configuration Environment

The VoiceFinder AP200 Gateway allows integration of voice applications and data. The VoiceFinder AP200 Gateway provides a status check of voice and fax support / connection, voice gateway configuration, quality of voice control, PABX connection configuration, and other configuration / status report of voice related.

For detail information, refer to Chapter 5 "Voice Configuration and Related Commands".

Chapter 4. Gateway Configuration and Commands

This chapter describes how to configure the VoiceFinder AP200 Gateway and commands of the VoiceFinder AP200 Gateway.

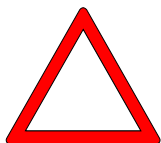
4.1. Gateway Booting

The operator can use all commands of the AP200 through the consol or Telnet connection.

When power is supplied, the Gateway is booted as follows :

- The Gateway performs a self-test and checks basic operations of the CPU, the memory and interfaces.
- The boot loader is executed, and the boot loader seeks for proper Gateway software image files. In the default configuration, the boot loader loads Gateway software in the flash memory.
- If the boot loader does not find any proper Gateway software image file from the flash memory, the boot loader stands by in the boot mode until it receives proper Gateway software from the system. (At this time, the boot loader can download Gateway software through TFTP or FTP protocol.)
- When Gateway software is loaded, Gateway starts to operate according to configuration information saved in the Gateway. However, if there is no configuration information, the Gateway operates according to the default values, and in this case, the operator shall set up related items for normal operation of the network.

Caution



When booting the system, set Gateway environment and save configuration information as using "copy running-config" command.

If the system is normally booted, the following message will appear.

```
System Boot Loader, Version 1.3.6/1
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

VoiceFinder Router Series (AP200)
Serial Number: AP200-010102
32BIT RISC Processor With 33554432 Bytes System Memory
524288 Bytes System Flash Memory
4194304 Bytes 2nd System Flash Memory

1 RS232 Serial Console Interface
1 Ethernet/IEEE 802.3 Interface
1 Ethernet/IEEE 802.3 or HomePNA Interface

RGW System software Revision 5.44Tg
Released at Sun Sep 3 10:32:11 2000
Program is 3494964 bytes, checksum is 0x103c5579

Local Time   : Thu Jan 1 03:13:57 1970

Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

The system is not configured yet or backup data is invalid.
Please login to system as a "root" and make configuration.

Voice Module (0): FXO
Voice Module (1): FXS
```

```
DSP S/W download
Voice Module (0): .... OK
Voice Module (1): .... OK

The System is ready. Please login to system.
login:
Interface ethernet0.0, changed state to UP
login: root
password:*****
RGW - Login : root at Console on Thu Jan 1 03:14:59 1970

router#
```


4.2. Command Usage

The operator can use all commands of the VOICEFINDER AP200 Gateway through the consol or Telnet terminal (VTY100 terminal.)

There are three kinds of commands – commands of the user mode, commands of the manager mode and commands of the configuration mode. Commands of the user mode enable the operator to check limited information of the system and provide a connection function for data communication.

Commands of the manager mode enable the operator to check configuration status of the Gateway and perform debugging. And commands of the configuration mode enable the operator to change configuration environment and set new environment.

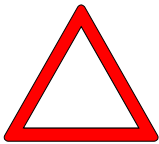
The VoiceFinder AP200 Gateway has following features regarding operator's entering commands.

- Although the operator enters only a part of the command, the VoiceFinder AP200 Gateway automatically recognizes the whole command. For example, if the operator enters only "sh" or "sho" instead of "show" the VoiceFinder AP200 Gateway automatically recognizes "sh" or "sho" as "show."
- The VoiceFinder AP200 Gateway provides an online help function so the user can check corresponding items for the command and command syntax.
- The VoiceFinder AP200 Gateway provides "More" function that divides a long message into several messages so that the operator can see the whole message by scrolling the screen.
- The VoiceFinder AP200 Gateway provides Help and "?" functions to display

available commands for the corresponding mode and descriptions of commands.

- The VoiceFinder AP200 Gateway provides "History" function. With the history function, the operator does not need to enter the command that the operator had used before. Instead, the operator only needs to use the numbers on the Gateway prompt.
- There are three modes for the Gateway commands, and in each mode, different commands are used. The following describes commands that can be used in each mode.

Caution

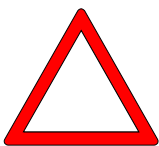


The commands indicated with "*" among the optional commands are not currently supported. They are to be supported in the higher version Gateways.

Ex) router# clear ?

- counters Clear counters on one or all interfaces
- *interface Clear the hardware logic on an interface
- logging Clear logging buffer
- utilization Clear system usage information

Caution



To cancel commands, use "no" command. If the operator uses "no" command for the commands that have default values, the optional values that had been set before will return to default values.

Ex) router(config)# no ?

- access-list : Add an access list entry
- arp : Remove a static ARP entry
- bridge : Set bridge Parameter to default value
- dhcp-list : Configure list entry
- ethernet : Configure Ethernet
- hostname : Set system's network name
- interface : Select an interface to configure

ip	: Set Ip routing mode
logging	: Modify message logging facilities
nat-list	: List NAT(Network Address Translation) lists
queue-list	: Build a custom queue list
route	: Establish static routes
router	: Enable a routing process
service	: Modify use of network based services
snmp	: Set SNMP community/configuration information
user	: Remove Gateway user
utilization	: System resource using information

4.2.1. Commands of the User Mode

Every person who logged in the Gateway can use commands of the user mode.

The prompts is indicated as "1router >" in the user mode.

Command	Description
?	Displays commands currently available and description of these commands.
clock	Indicates system time.
exit	Logs out the operator from the Gateway.
help	Explains command-using method in an interactive way.
history	Shows history of the used command lines.
ping	Sends an echo message to another network device and tests if the echo message reaches the destination.
rlogin	Establishes rlogin connection that is similar to the Telnet by an original login.
show	Shows configuration status and operating status of the Gateway. However, in the user mode, "show" command shows only limited information.
telnet	Establishes a protocol connection to log in a neighboring network device through a virtual terminal.
traceroute	Traces the path the packet passes through to reach the destination.
user	Adds a Gateway user or shows user information of the Gateway. With this command, the current user cannot check information of the user who has higher authority than him/herself.

4.2.2. Commands of the Manager Mode

The device manager who logged in the Gateway can use commands of the manager mode. To use commands of the manager mode, the user shall log in the Gateway with the root account or manager's ID. Only the manager can return to the configuration mode of the Gateway.

In the manager mode, commands usually show more information than in the user mode according to the options. For example, "show" command shows more information in the manager mode than in the user mode.

In the manager mode, the manager cannot use commands that are used in the user mode.

The prompt is indicated as "1router#" in the manager mode.

Commands	Description
?	Displays commands currently available and description of these commands.
clear	Clears statistical data saved in the Gateway.
clock	Indicates system time.
configure	Enters into the system configuration mode.
copy	Saves configuration data that is currently used in the non-volatile memory of the system.
debug	Displays packets and other information of the system for system debugging. Be careful in using the command since it can increase system load. (See "un-debug" command.)
Exit	Logs out the operator from the Gateway.
help	Explains command-using method in an interactive way.
history	Shows history of the used command lines.
load	VoIP related command. Loads the VoIP configuration script file to the VoIP Configuration of the Gateway.

no	Cancels commands entered in the command line or returns commands into default values.
ping	Sends an echo message to another network device and tests if the echo message reaches the destination.
reboot	Reboots the system.
rlogin	Establishes rlogin connection that is similar to the Telnet with by original login.
save	VoIP Command. Make VoIP Configuration Script File uses Gateway VoIP Running Configuration.
show	Shows configuration status and operating status of the Gateway. However, in the user mode, "show" command shows only limited information.
telnet	Logs in a neighboring network device through a virtual terminal.
test	Tests sub-systems of the Gateway – the memory, interfaces and so on.
tracert	Traces the path that the packet passes through to reach a certain destination.
who	Checks users who are currently online in the Gateway, login method and login time.
write	Saves the Gateway configuration file.
undebg	Stops execution of "debug" command.
user	Adds a Gateway user or shows user information of the Gateway. With this command, the current user cannot check information of the user who has higher authority than him/herself.

4.2.3. Commands of the Configuration Mode

Only the user who has the root account or equivalent authorities can access to the configuration mode. In the manager mode, the user cannot change the existing configuration of the Gateway nor make a new configuration of the Gateway.

The configuration mode can be divided into several kinds – the interface Configuration mode, the Global Configuration mode and the VoIP Configuration mode.

The prompt is indicated as "router(config)#" in the global configuration mode. In the global configuration mode, the user can make any configuration relating to the Gateway except the interface configuration. And in the interface configuration mode, the user can make any configuration relating to the interface – IP address configuration, WAN protocol configuration and so on.

The prompt is indicates as "router(config-serial0)" in the interface configuration mode.

4.2.3.1. Global Configuration(config) Commands

Command	Description
Access-list	Creates the access-list. From #0 to #29 are covered by the standard access-list, and from #30 to #59 are covered by the extended access-list.
Accounting-list	A configuration command to use the IP account.
Arp	Adds or deletes a certain Ethernet address in the ARP table.
Bridge	Sets bridge related items.
Clock	Sets system time of the Gateway.

debug-port	Display debug message into remote telnet terminal.
dhcp-list	Enables the Gateway to function as a DHCP server or send a DHCP packet broadcasting to other Gateways.
dial-peer	Sets dial-peer with "VoIP" command.
exit	Goes to the previous mode.
gateway	Makes voice gateway related configuration with "VoIP" command.
hostname	Changes the Gateway name of the network.
Interface	Enters into the interface configuration mode or creates a logical interface.
ip	Enables IP routing.
kill	Disconnects a certain Telnet session in the Telnet process.
logging	Changes or sets the message logging function.
nat-list	Creates Network Address Translation (NAT.)
no	Cancels commands entered in the command line or returns commands into default values.
Num-exp	Sets a phone number extension in the VoIP.
queue-list	Creates a queue-list to set the custom queue.
route	Adds or deletes static routes.
router	Enables a routing processor to use routing protocol.
service	Sets network-based service configuration – SNMP, Telnet, FTP and TFTP.
snmp	Sets "SNMP" command related items.
traceroute	Execute traceroute
translation-rule	Set translation rules in VoIP Service.
tll	Changes Time-To-Live (TTL) value.
user	Registers or changes Gateway users.
utilization	An optional command to set time intervals to check availabilities of the CPU, the Ethernet and the serial.
voice	Sets VoIP Service or Available Codecs.
VoIP-port	Sets the VoIP port.
VoIP-interface	Sets the VoIP Interface.

4.2.3.2. Commands of the Interface Configuration Mode 1

In the interface configuration mode, the user needs to define a certain interface before starting configuration.

* For voice related interface commands, see Chapter 5.

Command	Description
bridge	Sets the bridge parameters.
encapsulation	Sets and changes the encapsulation method of the interface. (AP200 supports Ethernet, IEEE802.1q VLAN, IEEE802.3 and PPPoE Encapsulation)
exit	Returns to the previous (configuration) menu.
end	Returns to the initial (Exec) mode.
Interface	Selects an interface to set additional interface.
ip	Sets IP protocol and IP service related items.
line-ctrl	Change Line Parameter settings for the interface.
no	Cancels commands entered in the command line or returns commands into default values.
mtu	Sets the size of the IP Maximum Transmission Unit (MTU.)
ppp	Sets PPP protocol related parameters.
rmon	Sets VLAN parameters for the interface.
Shutdown	Shuts down the selected interfaces.
vlan	Sets VLAN parameters for the interface.

4.2.3.3. Commands of the Interface Configuration Mode 2 (IP Configuration Mode)

The user can use IP related commands in the selected interface.

The prompt is indicated as router(config-ether0.0)#.

Command	Description
access-group	applies the access-list that has been set in the global configuration environment to the interface.
accounting	Apply IP Account List to the selected interface.
address	Sets or changes the IP address of the Interface.
dhcp-group	applies the DHCP-list that has been set in the global configuration environment to the interface.
exit	Returns to the previous (configuration) menu.
end	Returns to the initial (Exec) mode.
nat-group	applies the NAT-list that has been set in the global configuration environment to the interface.
no	Cancels the environment parameters that have been set in the configuration mode or returns them to the default values.
proxy-arp	Enables IP proxy ARP for the corresponding interface.

4.3. eZ-Setup(easy-setup)

AP200 VoIP Gateway supports a ez-setup command to provide the usability, easy configuration and fast equipment setting for general customers who do not understand the VoIP system concepts well. This feature supports the English and Korean language version commands. Also, ez-setup configuration can be performed via Web browser besides configuration setting via Console, Telnet protocol.

STEP	Command	Explanation
1	Router# configure	Enter to Configuration Mode
2	Router(config)# ez-setup <i>{English / korea}</i>	English: use the English. Korea: use the Korean language
3		Setting according to step by step configuration command guide

4.4. Starting Gateway Configuration

To set up the Gateway, log in the configuration mode. To log in the configuration mode, the user shall know the manager password. If the user does not know proper commands, the user can use "Help" or "?" function.

[Procedure]

Order	Operation and Related Commands
1	Boot the Gateway and log in with the manager's account.
2	Move to the configuration mode. 1 Gateway# <i>configure</i> 2 Gateway(config)#

[Example] Starting Gateway Configuration Mode

```

The System is ready. Please login to system.
login: root ☞ Enter the manager's account. (The manager's ID is
set as "root" in the factory.)
password:***** ☞ Enter the password. (The password is set as
"router" in the factory.)
AP200 Login: root at Console on Thu Jan 11 11:28:34 2001
1 router# configure ☞ Enter the command to move to the configuration
mode.
1 router(config)# ☞ Configuration is possible in this mode
    
```

4.5. Ethernet Configuration

4.5.1. Ethernet basic configuration

The Ethernet port of the VoiceFinderAP200 Gateway basically supports RJ-45. However, if the connection device of the other side supports only the AUI port, 10 Base-T Medial Attach Unit (MAU) shall be used in the other side. The Ethernet of the VoiceFinderAP200 Gateway uses the standard ARPA encapsulation method as the default. However, if necessary, the network manager can use SNAP or IEEE 802.3 encapsulation method.

The logical port can separate the Ethernet of the VoiceFinderAP200 Gateway. If the user wishes to use only one Ethernet port, the user must designate the logical port.

Do the following to use the Ethernet.

[Procedure]

Order	Operation
1	Enter into the interface configuration mode.
2	Enter into the IP configuration mode.
3	Give IP address to the interface.
4	Designate the encapsulation method to use (if necessary.)
5	Make the interface up.
6	Set other necessary optional parameters.

[Related Commands and Syntax]

- **Ethernet full-duplex**
 1. Sets the operation mode of the Ethernet interface.
 2. The default is half-duplex.

- **interface { Ethernet } { 0 / 1 }.[logical I/F #]**
 1. Selects the interface to set up and enters into the interface

configuration mode.

2. {0/1} represents the main interface while [logical I/F #] represents the sub-interface.
3. The Ethernet shall be set as a sub-interface, and if the manager needs to use the frame-relay encapsulation, the serial interface can be set as a sub-interface.

- **ip address <ip_address> <net_mask>**
 1. Sets the IP address for the selected interface.
 2. One of lower commands of "ip" command

[Example] Ethernet Configuration (Start)

When operating "Primary IP: 192.20.1.1/24bits, Secondary IP: 210.10.2.1/24Bits"

```
1Router(config)# interface ethernet 0 0
2Router(config-ether0.0)# ip address 192.20.1.1 255.255.255.0
3Router(config-ether0.0)# interface ethernet 0 1
4Router(config-ether0.1)# ip address 210.10.2.1 255.255.255.0
```

- **encapsulation {Ethernet/ieee/vlan/pppoe}**
 1. An optional command to change the encapsulation method for the current Ethernet interface
 2. The default is the Ethernet.
 3. VLAN supports 802.1Q VLAN.
 4. PPPoE supports ADSL Service
- **mtu <mtu-size>**
 1. Sets the MTU size for the current interface.
 2. The default is 1,500 Bytes.
- **arp request <ip-address>**

Forces the Gateway to send the ARP (MAC) request for the corresponding address. (Usually used for the test.)
- **arp static <ip-address> <hardware(MAC)-address>**

Forcefully registers information about the corresponding pair of the IP address and the hardware address in the ARP table.

- **arp table-size** *<table-size>*
 1. Defines the size of the ARP table for the corresponding interface.
 2. The default is 50. The size of the ARP table can be changed between 10 and 256. Adjust the size of the ARP table according to the number of PCs or terminals connected to the network.

- **shutdown / no shutdown**
 1. An optional command to make the current interface up/down.
 2. The Ethernet interface cannot shut down the main interface. To make a certain Ethernet interface up/down, go to the corresponding sub-interface.

- **no interface** *<if-name>*

An optional command to remove the logical interface. "If-name" represents the name of the logical interface.

- **show interface** *<if-name>*

Shows the interface status of "if-name."

[Example] Ethernet Configuration

```
Router(config)#interface ethernet 0.0
Router(config-ethernet0.0)#
Router(config-ethernet0.0)# ip address 131.12.1.1
255.255.0.0
Router(config-ethernet0.0)#no shutdown
Router(config-ethernet0.0)# mtu 2000
Router(config-ethernet0.0)# end
Router#show interface ethernet 0 0
    Interface Configuration Information for ethernet
    (131.12.1.1)
    Network = 130.100.0.0 NetMask = 255.255.0.0
    SubNetwork = 130.100.0.0 SubNetMask = 255.255.0.0
    Administrator Status = UP Operation Status = UP
    Ethernet CSMA-CD Speed - 10 Mbps
    MTU = 1500 Hardware Address = 00 00 00 00 00 42
    Secondary addresses : NONE
Router#
```


4.5.2. PPPoE Configuration

Information



Point to Point Protocol (PPP) is one of standard protocol to send data through the WAN link. RFC1661 describes PPP specifications. Not only in the synchronous WAN (serial) line but also in the asynchronous WAN (dial up line,) PPP can be used. Since PPP is a standard rules, it guarantees interoperability among different manufacturers' devices.

Nowadays PPP extended to not only Serial Line but also Ethernet and ATM Lines.

PPPoE(PPP over Ethernet) means PPP Protocol in the Ethernet Line.

PPP consists of two kinds of protocol as follows:

- Link Control Protocol (LCP): LCP decides the encapsulation format, limits the packet size, performs authentication in the link, decides normal operation time and breakdown time, detects loop-back link faults and other faults, and automatically terminates the link.
- Network Control Protocol (NCP): NCP communicates with various higher layer (network layer) protocol.

If a PPP encapsulation option is given to the VoiceFinder AP200 Gateway, PPP operation is possible. Current software installed in the Gateway supports Challenge Handshake Authentication Protocol (CHAP,) the authentication option that uses Password Authentication Protocol (PAP,) and the magic number configuration option. Software always sends the magic number configuration option. However, software sends the authentication option only when the authentication option is set.

[Procedure]

Order	Operation
1	Enter into the interface configuration mode.
2	Give PPP encapsulation protocol to the interface.
3	Move to the IP configuration mode.
4	Give an IP address to the interface.
5	Enable CHAP or PAP authentication. (Optional)
6	Set CHAP or PAP parameters. (Optional)
7	Set PPP default peer IP. (Optional)
8	(If necessary) use "debug" command to check if the Gateway is normally operating.
9	Make the interface up.
10	With "show interface" command, check if the interface is normally operating.
11	(For abnormal operation) find faults and recover faults as using "debug" command.

[Related Commands and Syntax]

- **interface { Ethernet / serial } { 0 / 1 }**
Selects the interface to set up and enters into the interface configuration mode.
- **encapsulation { hdlc / ppp / frame-relay }**
Sets the serial encapsulation mode for the interface.
- **ip address <ip_address> <net_mask>**
Selects the corresponding interface and enters into the IP related configuration mode.
- **user add <username> <password> {admin/high/normal/low}**
 1. Sets the login name and the password to authenticate a Gateway that is trying to access to another Gateway that function as a PPP PAP/CHAP server.

2. This command functions same as the command that the Gateway manager uses to register a login user. This is because the Gateway shares the PPP registered user database and the Gateway user database. The operator registers users as using the same command.
 3. The difference from the registration of Gateway users is that "user add" command does not use the registered user level for PPP connection in the user registration.
- **ppp authentication {CHAP/pap} [callin/{pap/CHAP}]**
 1. Sets the PPP authentication method as CAHP or PAP in the interface configuration mode.
 2. The "callin" option is to connect only incoming calls by CHAP authentication.
 3. {pap/CHAP} option is to respond to the calls which request both of CHAP and PAP authentication.
 - **ppp chap hostname *name***
 1. This command is for PPP client devices. This command registers a user name to request connection to the PPP server device when using PPP CHAP authentication. (An optional command for CHAP authentication)
 2. If this command is not used, the Gateway name (displayed in the Gateway prompt) will be considered as the user name.
 - **ppp chap password *password***

This command is for PPP client devices. This command registers a password to request connection to the PPP server device when using PPP CHAP authentication. (An optional command for CHAP authentication)
 - **ppp pap sent-username *username* password *password***

Sets PAP authentication in the PPP client device. When the client device sets a PPP call, the client device sends the user name and the password to the server for authentication. At this time, the user name and the password shall be the same with those set in the server. (An optional command for PAP authentication)

- **ppp peer default-ip-address** <ip-address>
 1. Sets the Gateway as a PPP server and the IP address to allocate to the serial interface of the other side. (An optional command)
 2. When the Gateway receives the IP address, the Gateway decides the subnet mask of the IP address that it received based on the IP subnet that has been set in its local interface.

- **ppp timeout** <second>
 1. Sets PPP negotiation timeout for PPP negotiation between two Gateways. (An optional command)
 2. The default is five seconds.

- **shutdown / no shutdown**

An optional command to make the current interface up/down.

- **show interface** <if-name>

Shows the interface status of "if-name."

- **debug ppp { chap/error/negotiation/packet }**
 1. Decodes PPP low level packets.
 2. "CHAP" option decodes challenge authentication related information.
 3. "Error" option decodes PPP protocol level errors and error statistics.
 4. "Negotiation" option decodes LCP and NCP protocol to set the PPP link.
 5. "Packet" option decodes PPP low level packets.

[Example] Normal PPP Configuration and Usage

```
Router# configuration ☞ Enters into the configuration mode
Router(config)#interface ethernet 0.0 ☞ Enters into the
interface configuration mode.
Router(config-ether0.0)# ☞ Configuration is possible in this
mode.
Router(config-ether0.0)# encapsulation pppoe ☞ Sets the PPPoE
mode
Router(config-ether0.0)# ip address 131.12.1.1 255.255.0.0
☞ Sets the IP address as "131.12.1.1/16 bit mask
Router(config-ether0.0)# no shutdown ☞ Makes the interface up.
Router(config-ether0.0)# end ☞ Exits from the configuration
menu.
Router # show interface ethernet 0 ☞ Checks the status of the
serial interface 0
router# sh int e 0 0
    Interface : ether0.0
        IP Address:211.238.72.221   Physical Inteface : Ethernet0
        Network : 211.238.72.0       Subnet Mask : 255.255.255.0
        Administrator Status : UP    Operation Status : UP
        Network Type : Ethernet      MTU : 1500
        Hardware Address : 00 02 a4 01 01 02
        Ethernet0 is UP, Line protocol is UP
        bandwidth : 10000 Kbit
        operating mode : HALF-DUPLEX
        operating speed : 10 Mbps
        last 1 minute data rate : tx 0 bps, rx 728 bps
        input : 95305 packets, 8979269 bytes, 0 no buffers
        error : 0 (0 length, 0 align, 0 short,
                0 crc, 0 overrun, 0 collision)
        output: 3 packets, 288 bytes, 0 drop
        error : 0 (0 underrun 0 deferred 0 collision)
```

[Example] Added Commands During PAP Configuration (Server)

If the Gateway functions as a server, it means the AP200 Gateway functions as the PPP authentication server.

```
Router (config)# user addpac password Router1 normal ☞ Registers the user name (addpac) and the password (Router1) with the normal priority in the server.
Router (config)#interface Ethernet 0.0 ☞ Enters into the interface configuration mode.
Router(config-ether0.0)# encapsulation pppoe ☞ Sets the PPP mode.
Router(config- ether0.0)# ppp authentication pap ☞ Sets the PPP authentication mode as PAP for the ethernet0.0 interface.
Router(config- ether0.0)# ip address 132.12.1.1 255.255.255.0 ☞ Sets the IP address as "130.1.1.1" and the subnet mask as "24Bit."
Router(config-ether0.0)# ppp peer default-ip address 132.12.1.2 ☞ When the other Router receives the serial interface IP from this Router, this command enables the Router to provide default address (130.1.1.2)to the other Router. (* If an IP address has been set already in the other Router, the operator does not need to use this command.)
Router(config- ether0.0)# ppp timeout 100 ☞ Sets PPP connection negotiation timeout value as 100 seconds.
Router(config-ether0.0)# end ☞ Exits from the configuration menu.
Router # debug ppp packet
    Ethernet0 LCP: TIMEOUT
    Ethernet0 LCP: O CONFREQ id=1
    Ethernet0 BCP: TIMEOUT
    Ethernet0 BCP: O CONFREQ id=1
    Ethernet0 LCP: TIMEOUT
    Ethernet0 LCP: O CONFREQ id=1
    Ethernet0 BCP: TIMEOUT
    Ethernet0 BCP: O CONFREQ id=1
Router # ubdebug ppp packet ☞ Stops PPP packet debugging.
```

[Example] Added Commands During PAP Configuration (Client)

This is when the AP200 Gateway is used as a PPP CallIn on the client side.

```

Router (config)#interface ethernet0.0 ☞ Enters into the
interface configuration mode.

Router(config-ethernet0.0)# encapsulation ppp ☞ Sets the PPP
mode.

Router(config-ethernet0.0)# ppp authentication pap ☞ Sets
the PPP authentication mode as PAP for the ethernet0.0 interface.

Router(config-ethernet0.0)# ppp pap sent-username addpac
password Router1 ☞ Sends the user name and the password that
were saved in the server during PPP connection.

Router(config-ethernet0.0)# ppp timeout 100 ☞ Sets PPP
connection negotiation timeout value as 100 seconds.

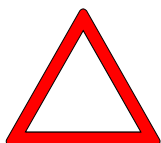
Router(config-ethernet0.0)# end ☞ Exits from the
configuration menu.

Router # debug ppp packet

Ethernet0 LCP: TIMEOUT
Ethernet0 LCP: O CONFREQ id=1
Ethernet0 BCP: TIMEOUT
Ethernet0 BCP: O CONFREQ id=1
Ethernet0 LCP: TIMEOUT
Ethernet0 LCP: O CONFREQ id=1
Ethernet0 BCP: TIMEOUT
Ethernet0 BCP: O CONFREQ id=1

Router # debug ppp packet ☞ Stops PPP packet debugging.

```

Caution

If the interface of the Gateway is not used as DHCP, the IP address must be set in the corresponding interface.

[Example] Added Commands During CHAP Configuration (Server)

This is when the AP200 Gateway functions as a PPP authentication server in the server side.

```
router (config)#user addpac password router1 normal ☞ Registers
the user name (addpac) and the password (router1) with the normal
priority in the server.
router (config)#interface ethernet0.0 ☞ Enters into the
interface configuration mode.
router (config-ether0.0)# encapsulation pppoe ☞ Sets the PPP
mode.
router (config- ether0.0)# ppp authentication CHAP ☞ Sets the
PPP authentication mode as CHAP for the ethernet0.0 interface.
router (config- ether0.0)# ip address 132.12.1.1 255.255.255.0
☞ Sets the IP address as "130.1.1.1" and the subnet mask as "24Bit."
router (config-s ether0.0)# ppp peer default-ip address
132.12.1.2 ☞ When the other Gateway receives the ethernet interface
ID from this Gateway, this command sets the IP address as the default
address "130.1.1.2."
router (config- ether0.0)# ppp timeout 100 ☞ Sets PPP connection
negotiation timeout value as 100 seconds.
router (config- ether0.0)# end ☞ Exits from the configuration
menu.
router # debug ppp packet
Ethernet0 LCP: TIMEOUT
Ethernet0 LCP: O CONFREQ id=1
Ethernet0 BCP: TIMEOUT
Ethernet0 BCP: O CONFREQ id=1
Ethernet0 LCP: TIMEOUT
Ethernet0 LCP: O CONFREQ id=1
Ethernet0 BCP: TIMEOUT
Ethernet0 BCP: O CONFREQ id=1
router # undebg ppp packet ☞ Stops PPP packet debugging.
```

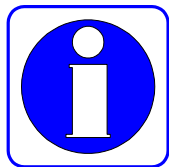

[Example] Added Commands During CHAP Configuration (Client)

This is when the AP200 Gateway functions as a PPP CallIn client in the client side.

```
router (config)#interface ether0.0 ☞ Enters into the interface
configuration mode.
router (config- ether0.0)# encapsulation ppp ☞ Sets the PPP mode.
router (config- ether0.0)# ppp authentication CHAP ☞ Sets the
PPP authentication mode as CHAP for the ethernet0.0 interface.
router (config- ether0.0)# ppp CHAP hostname addpac ☞ If the
user name that was saved in the server during PPP CHAP connection
is different from the user name of the client Gateway, this command
sends the user name of the server side.
router (config- ether0.0)# ppp CHAP password router1☞ Sets the
user name of the server side to check the password that the server
sends during PPP CHAP connection.
router (config- ether0.0)# ppp timeout 100 ☞ Sets PPP connection
negotiation timeout value as 100 seconds.
router (config- ether0.0)# end ☞ Exits from the configuration
menu.
router # debug ppp packet ☞ Decodes PPP packets.
router #
    Ethernet0.0 LCP: TIMEOUT
    Ethernet0.0 LCP: O CONFREQ id=1
    Ethernet0.0 BCP: TIMEOUT
    Ethernet0.0 BCP: O CONFREQ id=1
router # ubdebug ppp packet ☞ Stops PPP packet debugging.
```

4.6. Routing Configuration

Information



The VoiceFinder AP200 Gateway supports static routing protocol and dynamic routing protocol. There are two kinds of dynamic routing protocol – Interior Gateway Protocol (IGP) and Exterior Gateway Protocol (EGP.) IGP is used for routing among the networks in the same manager's domain while EGP is for routing among the network in different manager's domain. IGP includes RIP, OSPF and IS-IS, and EGP includes BGP. The VoiceFinderAP200 Gateway supports only IGP – RIP and OSPF.

To use routing protocol in the VoiceFinderAP200 Gateway, upload the routing process to the Gateway and designate the network that is going to use the routing process.

It is not easy to select routing protocol for each Gateway. Please note the following when selecting routing protocol.

- Network Size and Complexity – Normally, static routing is enough for edge network. However, to perform dynamic routing in a small scale network, use RIP. If the network is large or complex, use OSPF.
- Whether Variable Length Subnet Mask (VLSM) Is Supported or Not – If there are several subnet classes within the network, use routing protocol that support VLSM such as static route, RIP v2 and OSPF.

Besides, the user needs to consider convergence time, reliability needs and Internetwork delay characteristics.

The user can perform several kinds of routing protocol in the VoiceFinder AP200 Gateway at the same time. If several kinds of routing protocol is used in one Gateway, each kind of protocol may have its own path for the same destination. In this case, routing protocol has priority to be displayed in the routing table in order of static route, OSPF route, RIP route and default route.

4.6.1. Static Routing Configuration

The static route means a route that the manager designates to send the packet from a certain source to a certain destination. The static route is used for the following three cases:

- If routing software cannot create a proper route to send packets to a certain destination
- If the network is small or is not complex so it is easy to configure a static route, Or if the user does not want to have any packet such as route update packet and so on that may give load to the network
- If the user wants to send all packets of which destinations do not appear in the routing table to a certain next-hop address as using the default route (or gateway of last resort)

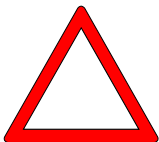
Once a static route is set in the Gateway, the Gateway keeps the static route until the manager forcefully removes that static route. To remove the static route, use "no" command and remove the static route from the route configuration.

The default route is one of static routes and designates the next-hop address of the packet of which destination is not displayed in the routing table. The default route has the least priority in the VoiceFinderAP200 Gateway. Therefore, only when the Gateway does not find any path, the Gateway uses the default route.

[Procedure]

Order	Operation
1	Go to the configuration mode.
2	Enable the static Gateway process.
3	Set the source address and the static route for the destination network.
4	Use "show" command and check if the route is correctly set in the routing table.
5	With "Ping" command, check if the route can reach the network.

Caution



The next-hop address to set during setting up the static route shall be directly connected with the Gateway to set.

The default route is one of static routes, and the setting method of the default route is same as that of the static route. However, the destination address shall be zero subnet (0.0.0.0 address and 0.0.0.0

mask) in the zero network, and the next-hop address shall be same with that of the static route.

[Related Commands and Syntax]

- **router static**
Enables or disables a certain routing process.
- **route <destination-IP-network> <address-mask> { <next-hop-address> / Ethernet / null } [(0/1)/<null_int_#>] [sub_int_#]**
 1. Designates the route to send the packet to the destination address.
 2. When using the candidate default (default route,) both of the destination address and the mask filed shall be zero.
 3. The Gateway shall be able to recognize the "next-hop-address" (directly connected port or where the Gateway can reach through dynamic protocol.)

4. The user can designate an interface port of the Gateway instead of the "next-hop-address."
5. To drop a packet that is headed for a certain destination, use the static route that uses a null interface.

- **show route**

Check the routes in the routing table.

- **show static**

Check the static route that has been set.

[Example] Static Routing Configuration and Usage

```

router# config
router(config)# Configuration is possible in this mode.

router(config)# ip routing Enables IP Routing.

router(config)# router static Enable static Routing Process

router(config)# route 130.2.0.0 255.255.0.0 131.20.1.1 Set
packet, whose destination address is 130.2.0.0/24bit, go to address
131.20.1.1.

router(config)# route 0.0.0.0 0.0.0.0 132.20.1.1 Set all
packets, whose address is not listed in routing table, go to address
132.20.1.1. (Candidate Default; Setting of Default route)

router(config)# exit Exit from setting mode.

router(config)# show route Shows routing table.

Destination Network-Mask Gateway Cost Interface TTL Protocol
-----
130.1.1.0 255.255.255.0 130.1.1.1 1 serial0 0 DIRECT
130.2.0.0 255.255.0.0 131.20.1.1 1 serial0 0 STATIC
0.0.0.0 0.0.0.0 131.20.1.1 1 serial0 0 STATIC

router(config)# showstatic Shows routing table setted as static.

Destination Network-Mask Gateway Cost Interface TTL Protocol
-----

```

130.2.0.0	255.255.0.0	131.20.1.1	1	serial0	0	STATIC
0.0.0.0	0.0.0.0	131.20.1.1	1	serial0	0	STATIC

4.7. Filter (Access-List) Configuration

Information



Packet filtering enables the manager to control packet movement on the network. With the packet filtering function, the manager can prevent unqualified user's access to the inside network from outside and disclosure of information.

The VoiceFinder AP200 Gateway uses the access-list to control traffic from a certain user (or an equipment or a network) to a certain network (or an equipment.) In this way, the Gateway can permit or deny packets passing through certain interfaces.

There are two kinds of access-list – the standard access-list and the extended access-list. The standard access-list uses IP addresses of the source and the destination in controlling traffic. And the extended access-list uses application port numbers and protocol IDs as well as IP addresses of the source and the destination in controlling traffic. The access-list is a continuous set of permit/deny conditions that are applied to the IP address. Software of the VoiceFinderAP200 Gateway checks these conditions with each address field of the packet.

With the first condition that matches with the address field, the Gateway decides to accept or reject the packet. After first matching, software stops testing the address. Therefore, orders of conditions are very important to normally operating the access-list. If there is no matching condition, software rejects the corresponding packet. (Default)

The VoiceFinderAP200 Gateway supports 30 standard access-lists (List # 0~29) and 30 extended access-list(List # 30 ~ 59.)

[[Procedure]

Order	Operation
1	Go to the configuration mode.
2	Create an access-list defining the access-list number and access conditions.
3	Go to the interface configuration mode.
4	Enter into the IP configuration mode.
5	apply the access-list that has been set to the corresponding interface. Decide where to apply the access conditions – inbound packets or outbound packets.
6	Use "Show access-list" command to check if the access-list has been correctly set.

[Related Commands and Syntax]

Standard IP Access-List: The standard IP address-lists uses only the source IP address in checking the access conditions.

- **access-list** *<access-list-number>* {deny/permit} *<source-address >* *<source wildcard>*
 1. Creates an access-list.
 2. access-list-number: Any number within 0 to 29, source: Source Network Address, Source-wildcard: Inverse mask of the source address
 3. Instead a pair of "source" and "source-wildcard," the user can use a pair of "any (any address)" and "host (a certain address.)"
 4. Wildcard is the inverse mask. For example, if the user writes 132.1.20.1 255.255.255.0 network in a wildcard form, the network will be 132.1.20.1 0.0.0.255.
 5. Since the default is the deny value, it is recommended to use "Permit Any Option" at the last line to permit all packets that do not satisfy conditions when every condition cannot be considered.

- **access-group** <access-list-number> [in/out]

As an interface command, applies the access-list to the incoming packet or the outgoing packet of the corresponding interface.

Extended IP Access-List: To check access conditions, the extend IP access-list uses source IP address, destination IP address, protocol ID, application port number and establishment status.

- **access-list** <access-list-number> {deny/permit}<protocol> <source> <source wildcard> <destination> <destination-wildcard> [operator] [port-number][established]

1. Creates Access-list.

2. Option explanation

1) access-list-number : Extended Access-List (Number in range of 30~59)

2) protocol : protocol ID Number 또는 protocol name (Ex: TCP, ICMP, UDP IP and so on)

3) source : Source Network Address,

4) Source-wildcard : Source Address의 Inverse Mask

5) Destination : Destination Network Address

6) destination-wildcard : Destination Address의 Inverse Mask

7) operator : operator for Port #

✓ eq : equal

✓ gt : greater then

✓ lt : less then

✓ neq : not equal

8) port-number: As application port number, well known port # is as follows:

✓ chargen : Character generator (19)

✓ daytime :Daytime (13)

✓ discard : Discard (9)

✓ domain : Domain Name Service (53)

✓ echo : Echo (7)

✓ finger : Finger (79)

✓ ftp : File Transfer Protocol (21)

- ✓ ftp-data: FTP data connections (used infrequently, 20)
- ✓ hostname: NIC hostname server (101)
- ✓ nntp: Network News Transport Protocol (119)
- ✓ pop2: Post Office Protocol v2 (109)
- ✓ pop3: Post Office Protocol v3 (110)
- ✓ smtp : Simple Mail Transport Protocol (25)
- ✓ sunrpc : Sun Remote Procedure Call (111)
- ✓ talk : Talk (517)
- ✓ time : Time (37)
- ✓ telnet : Telnet (23)
- ✓ uucp : Unix-to-Unix Copy Program (540)
- ✓ whois : Nicname (43)
- ✓ www : World Wide Web (HTTP, 80)

9) established : Established session

3. source/destination, Instead of source-wildcard/destination-wildcard pair, any(all Addresses), host(specified Host) can be used.

- **access-group** <access-list-number> {in/out}

applies the access-list that has been set by an interface command to the incoming packet or the outgoing packet of the corresponding interface.

[Example] Standard Access-List Configuration and Usage

router(config)# ☞ In this mode, Access-list Config is possible.

router (config)# access-1 1 deny 132.1.2.1 0.0.0.255 ☞ Denies all packets whose source addresses are "132.1.2.0/24bit."

router (config)# access-1 1 deny 150.1.3.2 0.0.0.223 ☞ Denies all packets whose source addresses are "150.1.3.0/21 bit."

router (config)# access-1 1 deny host 132.1.3.15 ☞ Denies all packets incoming from the host whose source address is "132.1.3.15."

router (config)# access-list 1 permit any ☞ Permits all packets that do not satisfy conditions of the Access-List 1 above. * If this command line does not exist, all default packets will be denied.

router (config)# interface Ethernet 0 0 ☞ Enters into the configuration mode of the interface Ethernet 0.0.

router (config-ether0.0)# ip access-group 1 in ☞ applies the Access-List 1 that has been set to all IP packets incoming through the Ethernet 0.0 interface.

router # show access-list ☞ Shows Access-List.

Standard Access List (Index = 1)

1 : deny 132.1.2.1 0.0.0.255

2 : deny 150.1.3.2 0.0.0.224

3 : deny host 132.1.3.15

4 : permit any

[Example] Extended Access-List Configuration and Usage

```
router (config)# ☞ In this mode, Access-list Config is possible.
```

```
router (config)# access-list 31 deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp ☞ Denies all TCP packets accessing to the host whose destination address is "145.1.1.0/24Bit" from "140.1.1.0/24bit" of the source address through the ftp port.
```

```
router (config)# access-list 31 deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp-data ☞ Denies all TCP packets accessing to the host whose destination is "145.1.1.0/24Bit" from "140.1.1.0/24bit" of the source address through the ftp-data port.
```

```
router (config)# access-list 31 permit tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp establish ☞ Permits only packets whose sessions are set already among the TCP packets accessing to the host whose destination is "145.1.1.0/24Bit" from "140.1.1.0/24bit" of the source address through the ftp port.
```

```
router (config)# access-list 31 permit ip any any ☞ Permits all IP packets except those matching conditions above.
```

```
router (config)# interface Ethernet 0 0 ☞ Enters into the configuration mode of the interface Ethernet 0.0
```

```
router (config-ether0.0)# ip access-group 31 in ☞ Applies the Access-List 31 that has been set for all IP packets incoming through the Ethernet 0.0 interface.
```

```
router (config-ether0.0)# end
```

```
router # show access-list 31 ☞ Shows the Access-List 31 that has been set.
```

```
Extended Access List (Index = 31)
```

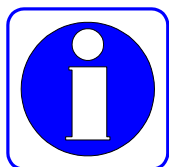
```
1 : deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255
```

```
2 : deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq  
ftp-data
```

```
3 : deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq  
    ftp established  
4 : permit ip any any
```

4.8. NAT(Network Address Translation) Configuration

Information



One of problems of the Internet is that available IP address space is decreasing. The Network Address Translation (NAT) changes uncertified address that is used in the inside network into another IP address (usually, registered address) when the address goes outside. Also, when a registered IP address comes to the inside network from outside, NAT changes it into an internal IP address.

NAT can be useful for the following cases:

- When the user wants to use Internet but the user cannot have a unique, certified address. In this case, NAT connects the private IP network that uses an unregistered IP address with the global Internet.
NAT shall be set in the Gateway that is located in the boarder between the public network (usually called an outside network) such as the Internet and the stub domain (usually called an inside network.) Before sending packets to outside networks, NAT converts internal private IP address into unique IP addresses.
- When the manager needs to change inside network address for the security reason or other reasons. In this case, without changing IP address that is a lot of work, the manager can translate addresses as using NAT.
- When the manager needs to distribute TCP traffic for load-sharing. In this case, the manager can map several local IP addresses into one global IP address as using the TCP load distribution function. When users access to the network from outside, they need to use the global IP address to access to the network and pass through the Gateway. Through the TCP session, load distribution is possible.

[NAT Glossaries]

- Inside local address: The address set in the host of the inside network
- Inside global address: An IP given by the Network Information Center (NIC) or the service provider. There are more than one IP address representing internal local IP addresses in the outside network.
- Outside local address: The IP address of the host in the outside network. The outside local address Appears in the inside network.
- Outside global address: The address that the host owner gave to the host in the outside network. The outside global address is allocated to the globally routable addresses or networks.

NAT supports static address translation and dynamic address translation.

- Static Address Translation: When there is any access request from the outside network, NAT regularly maps an unregistered IP address of the internal host to a certified IP address, and converts the registered IP addresses to an unregistered IP address. Also, if an internal host accesses to the outside network, NAT performs the opposite and converts addresses.
- Dynamic address Translation: NAT keeps registered IP addresses, and when an inside network requests to access to the outside network, NAT allocates one of IP addresses that it keeps. However, if all registered IP address that NAT keeps are in use, NAT cannot allocate any registered IP address to the inside network.

The VoiceFinder AP200 Gateway usually supports both of NAT function and Port Address Translation (PAT) function.

NAT function converts several internal, unregistered IP addresses into several external, registered IP addresses. And PAT function converts several internal,

unregistered IP addresses into a protocol port number on an external, registered IP address.

1. The VoiceFinder AP200 Gateway currently supports only dynamic address translation.
2. The VoiceFinder AP200 Gateway supports 256 NAT addresses.
3. The VoiceFinder AP200 Gateway only static routing for NAT.

[Procedure]

Order	Operation
1	Move to the configuration mode.
2	<p>Create the NAT/PAT-list defining the official address to use.</p> <ul style="list-style-type: none"> ✓ At this time, decide where to use the global address – inside or outside. ✓ Define and set the entry to statistically match address translation between the inside address and the outside address. ✓ The user needs to set timeout value of the session and recovery of the allocated address for the idle status when no data is transmitted through NAT.
3	Go to the interface configuration mode.
4	Enter into the IP configuration mode.
5	Apply the NAT/PAT-List that has been set to the corresponding interface.
6	Use "Show nat-list" command to check if correct access-list has been set.

[Related Commands and Syntax]

- **nat** *<nat-list-number>* **nat outside-global** *<start-address>* *<end-address>* *<mask>*
 1. Creates NAT pool for the outside global address in the global

configuration location.

2. NAT-list-number: Define any number between 0 and 7.
3. Start-address/End Address/Mask: Define the start address and the end address of NAT and the subnet masks for these addresses.

- **nat** *<nat-list-number>* **nat inside-global** *<start-address>* *<end-address>*
<local-ip address >

1. Creates a Nap pool for the outside global address in the global configuration location.
2. NAT-list-number: Define any number between 0 and 7.
3. Start-address/End Address: Designate the start address and the end address to use in the NAT.
4. Local -IP-Address: When there is a subnet that uses a registered IP in the local network, the user can register this subnet in the outside interface to route and advertise the subnet to the outside network.

- **nat** *<nat-list-number>* **natstatic-entry***<inside-local-address>*
<outside-global-address>

1. When it is necessary to access servers in the local network from outside, the user can define static entry for address translation with this command.
2. NAT-list-number: Define any number between 0 and 7.
3. Start-address/End Address: Define the start address and the end address that NAT uses.

- **nat** *<nat-list-number>* **nat time-out** *<timer-value>*

1. Defines time value for the NAT list to recover the address into free status when communication is idle.
2. The default is 300 seconds.

- **nat-group** *<nat-list-number>* {**nat/pat**}

As an interface command, Applies the NAT-list that has been set in the global mode to the corresponding interface.

- **nat** *<nat-list-number>* **pat** *<pat-address>*
 1. Sets the PAT list to use PAT in the global configuration location and PAT address.
 2. NAT-list-number: Define any number between 0 and 7.

- **nat** *<nat-list-number>* **pat static-entry { tcp/udp }** *<udp-port-number>* *<IP-address for PAT>* *<IP-address for PAT>*.....*<IP-address for PAT>*
 1. To Application such as Dial Pad, this command statically sets PAT translation between a certain port number and the IP address.
 2. NAT-list-number: Define any number between 0 and 7.
 3. Static-entry for TCP is to be implemented during the latter half of 2000.
 4. IP-Address for PAT: The address of a terminal whose port shall be statically set. IP-Address for PAT is one of local inner network addresses. As using this command, the user can set several IPs at the same time.

- **nat** *<nat-list-number>* **pat { fin-timeout / icmp-timeout / syn-timeout / tcp-timeout / udp-timeout }** *<timeout-value>*
 1. Selects options for timeout value when the session is in idle status during PAT conversion.
 2. Details of each option are as follows:
 - 1) Fin-timeout: Sets timeout after TCP Fin. The default is 10 seconds.
 - 2) icmp-timeout: Sets timeout after ICMP Session Idle. The default is 60 seconds.
 - 3) sys-timeout: Sets timeout after TCP sync Idle. The default is 60 seconds.
 - 4) tcp-timeout: Sets timeout after TCP Session Idle. The default is 3,600 seconds.
 - 5) udp-timeout: Sets timeout after UDP Session Idle. The default is 60 seconds.

- **show nat-list** [*nat-list-number*]
 1. Shows NAT-list that has been set.
 2. If no NAT-List-Number is designated, this command will show status of all NATs.

- **show nat-list** [**Ethernet/serial**] *<main-interface-number>* .
<sub-interface-number>

Shows NAT-list that has been set for a certain interface.

- **show running-config**

Shows configuration contents including the NAT-list that has been set.

[Example] NAT Configuration and Usage

```

router# config
router (config)# In this mode, NAT-list Config is possible.
router (config)# nat-list 0 nat outside-global 2.2.2.1
2.2.2.252 255.255.255.0 Defines NAT pool that the internal
packet can take "2.2.2.X/24bit" address as the source address when
going outside.
router (config)# nat-list 0 nat static-entry 1.1.1.253
2.2.2.254 Defines NAT pool that the internal packet can take
"2.2.2.254" address when going outside from the host whose source
is "1.1.1.253."
router (config)# nat-list 0 nat static-entry 1.1.1.254
2.2.2.253 Defines NAT pool that the packet can take "2.2.2.253"
address when going outside from the host whose source is "1.1.1.254."
router (config)# nat-list 0 nat timeout 250 When session is
in Idle status, sets the time value to turn into address free state
as 250 seconds.
router (config)# interface Ethernet 0 0 Enters into the
configuration mode of the interface Ethernet 0.0
router (config-ether0.0)# ip address 1.1.1.3 255.255.255.0
Allocates addresses to the Ethernet 0.0 interface. One of Indise
local addresses shall be selected.
router (config-ether0.0)# nat-group 0 nat Applies NAT pool
with NAT-list 0 to the Ethernet 0.0 interface. NAT shall be set
in the inside network always.
router (config-ether0.0)# end Exists from the configuration.
router # sh nat-list 0 Shows setting of NAT List No. 0.
NAT/PAT table Id: 0 Type : NAT TYPE
PAT Outside Public Address : 0.0.0.0
NAT Outside Public : 2.2.2.1 - 2.2.2.252 Netmask: 255.255.255.0
NAT Timer(secs) : 250
PAT Timer(secs) : ICMP(60) TCP(3600) UDP(60) TCPSYN(60) TCPFIN(10)
NAT static entry :

```

```

Local IP address : 1.1.1.254 Outside Global address : 2.2.2.253
Local IP address : 1.1.1.253 Outside Global address : 2.2.2.254
router # sh nat-list ethernet 0.0 ☞ Shows NAT format in ethernet
interface 0.0 and present NAT table.
NAT/PAT table Id: 0 Type : NAT TYPE
PAT Outside Public Address : 0.0.0.0
NAT Outside Public : 2.2.2.1 - 2.2.2.252 Netmask: 255.255.255.0
NAT Timer(secs) : 250
PAT Timer(secs) : ICMP(60) TCP(3600) UDP(60) TCPSYN(60) TCPFIN(10)
NAT static entry :
Local IP address : 1.1.1.254 Outside Global address : 2.2.2.253
Local IP address : 1.1.1.253 Outside Global address : 2.2.2.254
Local IP          Global IP          Timer
-----
1.1.1.2          2.2.2.3          120
1.1.1.1          2.2.2.2          15
router # sh running-config ☞ Shows present Configuration File.
interface ether0.0
ip address 1.1.1.3 255.255.255.0
Operation is UP
NAT/PAT table Id: 0 Type : NAT TYPE
NAT Outside Public : 2.2.2.1 - 2.2.2.252 Netmask: 255.255.255.0
NAT Timer(secs) : 250
NAT static entry :
Local IP address : 1.1.1.254 Outside Global address : 2.2.2.253
Local IP address : 1.1.1.253 Outside Global address : 2.2.2.254
interface serial0
ip address 132.1.1.1 255.255.255.0
Encapsulation HDLC
Operation is UP!
interface serial1
no encapsulation

```

[Example] PAT Configuration and Usage

```

router # config
router (config)# ☞ In this mode, NAT-list Config is possible.
router (config)# nat-list 0 pat 2.2.2.2 ☞ Sets PAT that the
internal packet can take IP address 2.2.2.2 as the source address
when going outside.
router (config)# nat 0 nat static-entry udp 1000 1.1.1.4 1.1.1.5
☞ Statically sets that packets should be sent to "1.1.1.4" and
"1.1.1.5" of internal host when Application (Dial Pad, Wow Call or
UDP No. 1000 port) tries to access to the network from outside. If
there are several internal hosts set, load is distributed in order.
router (config)# interface Ethernet 0 0 ☞ Enters into the
configuration mode of the interface Ethernet 0.0
router (config-ether0.0)# ip address 1.1.1.3 255.255.255.0
☞ Allocates addresses to Ethernet 0.0 interface. The address shall
be one of Indise local addresses.
router (config-ether0.0)# nat-group 5 pat ☞ Applies PAT pool
that is NAT-list 5 to Ethernet 0.0 interface. NAT/PAT shall be set
in the inside network always.
router (config-ether0.0)# end ☞ Exit from setting mode.
router # sh nat-list 5 ☞ Shows setting of NAT/PAT List #5.
NAT/PAT table Id: 5   Type : PAT TYPE

    PAT Outside Public Address : 2.2.2.2
    NAT Outside Public : 0.0.0.0 - 0.0.0.0 Netmask: 0.0.0.0
    NAT Timer(secs) : 300
    PAT Timer(secs) : ICMP(60) TCP(3600) UDP(60) TCPSYN(60) TCPFIN(10)
    PAT static entry :
        UDP port(1000) :
            1.1.1.4
            1.1.1.5
router # sh nat-list ethernet 0.0 ☞ Shows NAT/PAT setting in
ethernet interface 0.0 and Address Translation Table.

    NAT/PAT table Id: 5   Type : PAT TYPE
        PAT Outside Public Address : 2.2.2.2

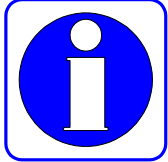
```

```

NAT Outside Public : 0.0.0.0 - 0.0.0.0 Netmask: 0.0.0.0
NAT Timer(secs) : 300
PAT Timer(secs) : ICMP(60) TCP(3600) UDP(60) TCPSYN(60) TCPFIN(10)
PAT static entry :
    UDP port(1000) :
        1.1.1.4
        1.1.1.5
STATE  PROTOCOL  TIMER      LOCAL-IP/Port      GLOBAL_IP/Port
-----
Dynamic TCP      120        1.1.1.2:1723      2.2.2.2:1723
Dynamic TCP      150        1.1.1.1:1024      2.2.2.2:1024
Dynamic TCP      120        1.1.1.2:1723      2.2.2.2:1723
Dynamic TCP      150        1.1.1.1:1024      2.2.2.2:1024
router # sh running-config ☞ Shows present Configuration File.
!
interface ether0.0
    ip address 1.1.1.3 255.255.255.0
    Operation is UP
    NAT/PAT table Id: 5  Type : PAT TYPE
    PAT Outside Public Address : 2.2.2.2
    PAT Timer(secs) : ICMP(60) TCP(3600) UDP(60) TCPSYN(60) TCPFIN(10)
    PAT static entry :
        UDP port(1000) :
            1.1.1.4
            1.1.1.5
interface serial0
    ip address 132.1.1.2 255.255.255.0
    Encapsulation HDLC
    Operation is UP
interface serial1
    no encapsulation
    
```

4.9. DHCP(Dynamic Host Configuration Protocol) Configuration

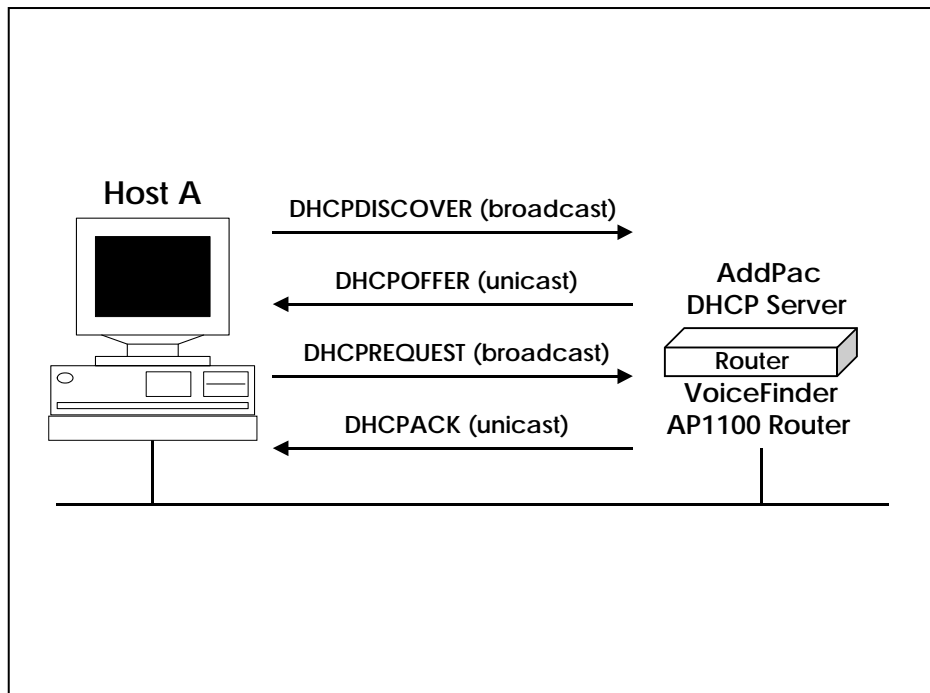
Information Dynamic Host Configuration Protocol (DHCP) automatically allocates IP addresses to DHCP clients.



The DHCP function of the VoiceFinder AP200 Gateway uses the address pool that has been set in the Gateway to allocate IP addresses to DHCP clients and manage IP addresses.

If software of the VoiceFinder AP200 Gateway does not respond to the request of DHCP through the database that is set in the Gateway, the Gateway will send the request to another DHCP server that the network manager has set.

The following figure shows the basic procedure that the DHCP client requests an IP address to the DHCP server.



Host A (a client) sends a broadcast message "DHCPDISCOVER" to the DHCP server of the Gateway. Then, the DHCP server sends the DHCPOFFER Unicast message that contains configuration information – IP address to be allocated, domain name and allocation status of the IP address – to the client. The

DHCP client sends official IP address request to the server through the DHCPREQUEST broadcast messages. The DHCP server sends back the DHCPACK Unicast message to the client, and confirms the IP address that has been allocated to the client.

The DHCP function of the VoiceFinder AP200 Gateway complies with DHCP of RFC 2131, BOOTP of RFC 951 and Bootstrap Specifications of RFC 1542. The DHCP function provides following benefits.

- It is easy to configure DHCP so the user can save time and cost in configuring clients.
- The network manager can easily manage addresses and other related items of the lower network by managing only the central server.

To implement the DHCP server function, the following conditions shall be satisfied.

- When the DHCP server function is enabled, IP addresses to be allocated to the server shall be separated from the addresses that will not use the DHCP function. (For example, servers and printers whose addresses shall be fixed.)
- If necessary, the user shall define DHCP options to use in the Gateway – the default Gateway and the DNS server.

VoiceFinder AP200 supports not only DHCP Server function but also DHCP Clients and Relay functions. If it configured DHCP in address field instead IP Address, AP200 operate in DHCP Client mode.

[Procedure-DHCP Server]

Order	Operation
1	Move to the configuration mode.
2	Define the DHCP-list type to use in the Gateway.
3	<p>Create a DHCP-list defining DHCP-list number, DHCP mode to use in the Gateway, or DHCP address pool.</p> <ul style="list-style-type: none"> ✓ If the server type is DHCP, set a DHCP pool that defines the DHCP start-address and the DHCP end-address. ✓ If the server type is DHCP, set a DHCP pool that defines the DHCP start-address and the DHCP end-address.
4	Set other DHCPs and other necessary options.
5	Go to the interface configuration mode.
6	Enter into the IP configuration mode.
7	Apply the DHCP-list that has been set to the corresponding interface.
8	Use "Show dhcp-list" to check if desired DHCP has been correctly set.

[Procedure – DHCP Client]

순서	작업 내용 Description
1	Move to the configuration mode.
2	Select the interface configure as a DHCP Client
3	Apply the DHCP-Client instead of IP Address to the corresponding interface

[Related Commands and Syntax]

Mandatory Commands

- **dhcp-list <dhcp-list-number> type {server/relay}**

Creates the DHCP list of the Gateway (dhcp-list-number: any number between 0 and 4) and sets DHCP in the list should function as a server or protocol relay.

- **dhcp-list** *<dhcp-list-number>* **address relay** *<relay-IP-address>*
 1. Sets that the Gateway should send broadcast DHCP protocol to an equipment of the relay-ip-address through Unicast.
 2. relay-IP-address: IP address of the equipment that is going to transfer DHCP broadcast through Unicast

- **dhcp-list** *<dhcp-list-number>* **address server** *<start-IP-address>*
<end-ip-address>
 1. Sets DHCP pool that the Gateway can function as a DHCP server.
 2. *<start-IP-address>*,*<end-IP-address>*: Defines IP address range of the DHCP pool.

- **dhcp-group** *<dhcp-list-number>*

As an interface command, binds DHCP-list with the interface to use.

- **ip address dhcp**

Set ip address with DHCP Clients to the interface to use.

- **show dhcp-list** [*dhcp-list-number*]

Shows a certain DHCP list or whole DHCP configuration.

- **show running-config**

Shows configuration contents in which DHCP is included.

Optional Commands

- **dhcp-list** *<dhcp-list-number>* **option** [option command]
 1. Sets options used in the DHCP-list set in the Gateway.
 2. Optional Commands
 - 1) **arp-cache-timeout** *<time(second)>*: Sets time value that the ARP cache table can keep the Mac address.
 - 2) **default-ip-ttl** *<time(second)>*: Sets IP TTL value of the packet.
 - 3) **dhcp-lease-time** *<time(second)>*: Sets time value to check how long each IP address allocated by the DHCP server is valid. The

default is one hour.

- 4) **dns** <*dns-address*>: Sets the addresses of the DNS server so that DHCP clients can use them.
- 5) **domain-name** <*domain-name*>: Designates the domain name to be used by DHCP clients. The domain name and the IP address are given together to DHCP clients.
- 6) **Ethernet-encapsulation** {*Ethernet/ieee*}: Sets that the DHCP clients should inform the Ethernet encapsulation method that the Gateway is going to use. The VoiceFinderAP200 Gateway supports Ethernet Version 2 and IEEE802.2 address. *The default is Ethernet Version 2.*
- 7) **interface-mtu** <*mtu-value*>: Sets MTU value for the interface.
- 8) **name-server** <*name-server-address*>: Sets the name server address.
- 9) **ntp-server** <*ntp-server-address*>: Sets the NTP server address.
- 10) **max-lease-time** <*time(second)*>: Sets time value to check how long each IP address allocated by the DHCP server is valid. After time passes, all addresses are recovered to be free regardless of the connection status.
- 11) **smtp-server** <*smtp-server-address*>: Sets the SMTP server address.
- 12) **pop3-server** <*pop3-server-address*>: Sets POP3 mail server address.
- 13) **Gateway-option** < *default-Gateway-address* >: After the DHCP client is booted, the DHCP client sends packets to its default Gateway. Therefore, the address and the default Gateway of the DHCP client shall be set. With this command, the user can set the address and the default Gateway.
- 14) **static-route** <*destination-address*> <*Gateway-address*>: Routes an initial DHCP packet to a certain address.
- 15) **time-server** < *time-server-address* >: Sets the time server address.
- 16) **www-server** < *www-server-address* >: Sets the web server address.

[Example] DHCP Server Mode Configuration and Usage

```
router(config)# ☞ In this mode, DHCP-list Config is possible.  
router (config)# dhcp-list 0 type server ☞ Sets that the  
Gateway should operate as a DHCP server.  
router (config)# dhcp-list 1 address server 211.1.1.1  
211.1.1.125 ☞ Sets the DHCP address pool. This command sets that the  
DHCP address pool can allocate addresses from "211.1.1.1" till  
"211.1.1.125."  
Router(config)# dhcp-list 1 option domain-name AddPac ☞ Sets  
that the Gateway should give AddPac as a domain name to the client  
when the Gateway functions as a DHCP server.  
Router(config)# dhcp-list 1 option gateway-option  
211.1.1.126 ☞ Sets that the Gateway should give "211.1.1.126" of default  
Gateway address to the client when the Gateway functions as a DHCP server.  
Router(config)# interface Ethernet 0 0 ☞ Makes the DHCP clients  
to enter into the configuration mode of the Ethernet 0.0 that is  
the interface that the DHCP clients are going to be connected.  
Router(config-ether0.0)# ip address 211.1.1.126  
255.255.255.127 ☞ Sets Ethernet 0.0 interface address as  
"211.1.1.126/25 Bit." At this time, the address shall have the  
same network address of DHCP address, and does not exist in the  
DHCP pool. IPs may overlap.  
Router(config-ether0.0)# dhcp-group 0 ☞ Sets that all DHCP  
packets incoming through Ethernet 0.0 interface should be allocated  
with addressed by rules of DHCP-0 that has been set already.  
Router(config-ether0.0)# end ☞ Exits from the configuration  
mode.  
Router# show dhcp-list 0 ☞ Shows configuration contents of DHCP  
List 0.  
DHCP Type = SERVER  
ADDRESS POOL Start = 211.1.1.1 End = 211.1.1.127  
DOMAIN NAME = addpac  
Lease Time = 3600, Max lease time = 268435455
```

```
ARP Timeout = 180, Enthnet Enc = 0  
Interface MTU = 1500 default-TTL = 255  
Gateways Option : 211.1.1.126
```

[Example] DHCP Relay Mode Configuration and Usage

```
Router# config  
Router(config)# ☞ In this mode, DHCP-list Config is possible.  
Router(config)# dhcp-list 1 type relay ☞ Sets that the Gateway  
should pass DHCP broadcast packets.  
Router(config)# dhcp-list 1 address relay 151.1.12.1 ☞  
Changes the DHCP request packet into a Unicast packet and sends  
it to a host whose IP address is "151.1.12.1."  
Router(config)# interface Ethernet 0 0 ☞ Enters into the  
configuration mode of the Ethernet 0.0 to which DHCP clients are  
going to be connected.  
Router(config-ether0.0)# ip address 211.1.1.126  
255.255.255.127 ☞ Sets the address of the Ethernet 0.0 interface  
as "211.1.1.126/25 Bit."  
Router(config-ether0.0)# dhcp-group 1 ☞ Relays all DHCP  
packets coming through Ethernet 0.0 interface according to rules  
of DHCP-List 1 that has been set already.  
Router(config-ether0.0)# end ☞ Exits from the configuration  
mode.  
Router# show dhcp-list 1 ☞ Shows configuration contents of the  
DHCP List 0.  
DHCP Type = RELAY  
Next Server = 151.1.12.1  
Router# show running-config ☞ Shows configuration contents that  
have been set.
```

[사용 예] DHCP Client 설정 및 사용의 예

```
Router# config
Router(config)# interface ether0.0 ☞ Enters into the
configuration mode of the Ethernet 0.0 that using as a DHCP clients.
Router(config-ether0.0)# ip address dhcp ☞ Set IP Address with
DHCP Client
Router(config-ether0.0)# end ☞ Exits from the configuration
mode.
Router# show run ☞ Shows configuration contents that have been
set.
interface ether0.0
    ip address dhcp

!
interface serial0
    ip address 132.1.1.2 255.255.255.0
    Encapsulation HDLC
    Operation is UP
!
interface serial1
    no encapsulation
!
```

4.10. Transparent Bridging Configuration

The VoiceFinder AP200 Gateway supports transparent bridging for Ethernet and serial ports. Also, to manage networks, the VoiceFinder AP200 Gateway supports Bridge MIB that is defined in RFC 1286.

The bridge functions that the VoiceFinder AP200 Gateway supports are as follows:

- Complying with IEEE 802.1D standard
- Segmenting transparent bridged network to the logical VLAN
- The bridge function is supported not only through the Ethernet but also through the serial network and the frame-relay network.
- Supporting the spanning-tree algorithm that adopts IEEE-based Bridged Protocol Data Unit (BPUD)

VoiceFinder AP200 series Gateways only support one bridge-group. Therefore, the concept of the bridge-group is not used.

[Procedure]

Order	Operation
1	Move to the configuration mode.
2	Set option values to use in the bridge.
3	Go to the interface configuration mode.
4	Apply the bridge-group that has been set to the corresponding interface.
5	For multi-access interfaces including frame-relay, make a map.
6	Apply other bridging option parameters to use.
7	Use "show bridge" or "show span" commands to check if the bridge has been correctly set and the spanning tree algorithm normally operates.

[Related Commands and Syntax]

- **bridge**

As an interface command, sets that the corresponding interface should function as a bridge group.

- **frame-relay map bridge <dlci-number>**
 1. An interface command. If the interface using the bridge is frame-relay, this command sets the map in a way that the bridge maps can be sent through this interface.
 2. DLCI values is any number between 16 and 1007.
 3. *If a bridge is used in the frame-relay interface, the user must use "MAP" command to enable the bridge.*

- **bridge priority <priority-number>**
 1. An option of the interface command. This command defines the priority of the interface to be blocked or forwarded while participating in the spanning tree procedure.
 2. The range is between 0 and 255. The lower the number is, the higher the priority is. The default is 0.

- **bridge path-cost <path-cost-value>**
 1. An option of the interface command. This command defines the priority of the interface to be blocked or forwarded while participating in the spanning tree procedure.
 2. The range is between 0 and 65535. The lower the number is, the higher the priority is. The default is 100 for Ethernet or 128 for Serial.

- **bridge hello-time <hello-interval>**
 1. An optional command of "Global" command. This command defines Hello Interval between BPDUs.
 2. The range is between 1 ~ 10 seconds. The default value is two seconds.

- **bridge forward-time***<forward-interval>*
 1. An option of the global command. This command decides the forward delay interval.
 2. The range is between 10 and 200 seconds. The default value is 30 seconds.

- **bridge max-age** *<max-age-time>*
 1. An option of the global command. This command decides standby time to wait until receiving BPDU from the root bridge.
 2. The range is between 100 to 200 seconds. The default value is 15 seconds.

- **no ip routing**
 1. An option of the global command. Use this command to operate the Gateway as a pure bridge without operating routing functions.
 2. For rerouting, the user must use "**ip routing**" command.

- **show bridge**

Shows bridge forwarding database entry.

- **show bridge**

Shows spanning-tree topology that the bridge is aware of.

- **show running-config**

Shows configuration contents that have been set including bridging.

[Example] Transparent Bridging Configuration and Usage

```
router# config
router(config)# interface Ethernet 0.0 ☞ Creates Ethernet
interface 0.0 and starts configuration.
router(config-ether0.0)# bridge ☞ Applies bridge to Ethernet
interface 0.0.
router(config-ether0.0)# bridge priority 2 ☞ Sets the
spanning tree priority of the interface as 2.
router(config-ether0.0)# interface serial 0 ☞ Enters into
the configuration mode of the interface serial 0.
router(config-serial0)# encapsulation frame-relay ☞
Encapsulates with the frame-relay.
router(config-serial0)# frame-relay map bridge 100 ☞
Enables the bridge in the frame-relay interface. This command
also encapsulates the bridge packet.
router(config-serial0)# exit ☞ Goes back to the global
configuration mode.
router(config)# bridge forward-time 150 ☞ Sets the bridge
forward delay interval as 150 seconds.
router(config)# bridge hello-time 5 ☞ Sets the bridge hello
BPDU interval as five seconds.
router(config)# bridge max-age 150 ☞ Sets the standby time
to wait until receiving BPDU from the root bridge as 150 seconds.
router(config)# exit ☞ Exits from the configuration mode.
router # show running-config ☞ Shows configuration contents.
interface ether0.0
    no ip address
    Operation is UP
    bridge
!
interface serial0
    no ip address
    Encapsulation FRAME-RELAY
```

```

    Operation is UP
    bridge
    !
    interface serial1
        no encapsulation
router # show bridge ⇨ Bridge Forwarding Database의 Entry를
    보여줍니다.

```

Address	type	status	Age	Port
1111.1111.1111	static	bppu0	0	--
FFFF.FFFF.FFFF	static	our mac	0	--
AA11.0000.1111	dynamic	single-port	3	e0
0000.0C06.1122	dynamic	single-port	10	e0
0000.0C06.1123	dynamic	single-port	144	s0
0000.0C12.125A	dynamic	single-port	11	e0

```

router # show spanning-tree ⇨ Bridge(Gateway)가 알고 있는
    Spanning Tree Topology를 보여줍니다.

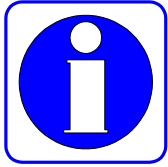
    Bridge is executing the IEEE compatible Spanning Tree protocol
    Bridge has priority 32768, address 0000.0000.0000.0000
    Configured hello time 2, max age 15, forward delay 30
    Current root has priority 128, address 0000.30c3.098a.f789
    [ We are the root of the spanning tree ]
    Topology change flag not set, detected flag not set
    Times: hold 1, topology change 30, notification 30
    Hello 2, max age 15, forward delay 30, ageing 300
    Timers: hello 1, topology change 0, notification 0

    Port 1(ETH0), forward status
    Port path cost 0, port priority 128
    Designated root has priority 128, address 0000. 304c.f686
    Designated bridge has priority 128, address 0000. 304c.f686
    Designated port is 1, path cost 0
    Timers : message age 0, forward delay 0, hold 0

```

4.11. SNMP Configuration

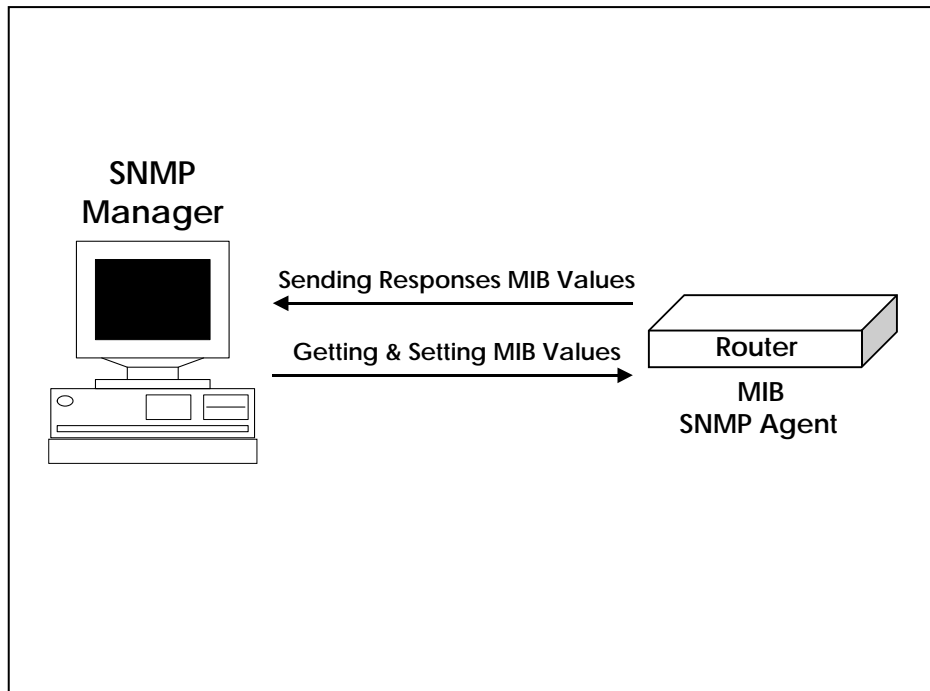
Information



SNMP is Application layer protocol providing a message format for the communication between the SNMP Manager and the SNMP Agent. Elements configuring an SNMP system to manage the network include the SNMP Manager, the SNMP Agent and the Management Information Base (MIB.)

The SNMP Manager composes a part of a commercialized Network Management System (NMS) such as the HP Openview. The SNMO Agent and the MIB are located in the Gateway. To configure SNMP in the Gateway, the user shall know the relations between the SNMP Manager and the SNMP Agent.

The SNMP Agent has MIB parameters that the SNMP Manager requests and changes. The SNMP Manager receives management information from the SNMP Agent or gives certain information to the SNMP agent for setting.



[Communication between SNMP Manager and SNMP Agent]

The SNMP Agent collects data from the MIB that manages data or equipment parameters for the routing function. According to the request of the SNMP Manager, the SNMP Agent gives or sets these data. However, if the SNMP Agent sends information without receiving any request from the SNMP Manager, it is called "Trap." Usually, trap is a warning message. A warning message is created upon fault occurrence in the network, configuration change or important event occurrence.

[Figure 3. Communication between SNMP Manager and SNMP Agent] shows the relations between the SNMP Agent and the SNMP Manager. The SNMP Manager sends requests to get or set MIB values to the SNMP Agent, and the SNMP Agent sends responses. Also, the SNMP Agent sends trap for important network events that the manager should know.

SNMP standards are as follows:

- SNMPv1: Full standard protocol defined in RFC1157
- SNMPv2C: Consisting of followings.
 - ✓ SNMPv2: SNMP v2 protocol defined in RFC 1902~1907. An Internet draft standard
 - ✓ SNMPv2C: Standard defined in RFC 1901. Community-based management structure of SNMPv2

The VoiceFinder AP200 Gateway supports both of SNMPv1 and SNMPv2C.

[Procedure]

Order	Operation
1	Move to the configuration mode.
2	Set a SNMP community.
3	Set the host to receive SNMP trap.
4	Set SNMP related parameters.
5	Moves to the IP configuration mode.
6	Apply the queue-group that has been set to the corresponding serial interface.
7	Use "show snmp" command to check if configuration has been correctly made.

[Related Commands and Syntax]

- **snmp community** *<snmp-manager-ip/0.0.0.0>* *<community-string>* {ro/rw}
 1. Register the Gateway in a certain SNMP community.
 2. *<snmp-manager-ip/0.0.0.0>*: IP address of the SNMP Manager. "0.0.0.0" is an option that enables all NMSs that have same community-string values to function as the manager of the SNMP.
 3. Community-String: String used for authentication during SNMP communication
 4. {ro/rw}: Sets whether to only read Gateway information or read and write setting values of the Gateway.
- **snmp host** *<trap-host-ip>* *<community-string>* {v1/v2c}
 1. Registers the host to which the Gateway sends trip and the SNMP version when trap occurs.
 2. *<trap-host-ip>*: IP address of the trap host (SNMP manager)
 3. Community-String: String to be used for authentication during SNMP communication
 4. {v1/v2c}: SNMP version

- **snmp contact** <string>
When the Gateway sends trap, the command indicates contact point (equipment manager) to solve trap problem.

- **snmp location** <string>
Indicates installation location of the faulty equipment when the Gateway sends trap.

- **snmp name** <string>
Indicates the faulty equipment when the Gateway sends trap.

- **snmp system-shutdown**
 1. Decides to shutdown (reboot) the Gateway or not by SNMP from a remote place
 2. This command can greatly affect equipment operation in the network in which security is not strong. Therefore, the user shall be careful in using this command. (This command is to be implemented during the latter half of 2000.)

- **snmp trap-authentication**
When another SNMP Manager accesses to the SNMP Agent with incorrect community-string value, this command sends authentication violation information.

- **show snmp**
Shows SNMP setting status.

[Example] SNMP Configuration and Usage

```

router# config
router(config)# ☞ In this mode, SNMP Config is possible.
router(config)# snmp community 0.0.0.0 AddPac-Domain1 rw
☞ Exchanges information with all SNMP Managers whose
community-string is AddPac-Domain1.
router(config)# snmp host 131.23.1.1 AddPac-Domain v2c ☞
Sends trap to the SNMP manager in "131.23.1.1" through SNMP v2c
protocol. At this time, the string is AddPac-Domain.
router(config)# snmp contact HongKilDong ☞ Sends a message
indicating that the contact point is "HongKilDong" when trap occurs
router(config)# snmp location 9FlofBuilding4 ☞ Sends a
message indicating that the installation location of the faulty
equipment is 9FlofBuilding4 when trap occurs
router(config)# snmp name Tac_Gateway1 ☞ Sends a message
indicating that the equipment name is Tac_Gateway1 when trap occurs
router(config)# snmp trap-authentication ☞ If an SNMP Manager
accesses to an equipment with incorrect string, this commands
informs this to all SNMP Managers that are set in the Gateway.
router(config)# exit
router # show snmp ☞ Shows SNMP configuration status.
TRAP version
  TYPE      Community-Name  IP-Address  Access Mode
-----
community public          0.0.0.0      read-only
community private       0.0.0.0      read-write
community proxy         0.0.0.0      read-write
community AddPac-Domain1 0.0.0.0      read-write
host        AddPac-Domain    131.23.1.1   SNMPv2c
contact    : HongKilDong
location   : 9FlofBuilding4
name       : Tac_Gateway1
system-shutdown : Not Implemented
trap-authentication : ENABLE

```


4.12. Gateway Management Command

This chapter describes commands used in the EXEC mode or the global configuration mode and necessary for management and operation of the Gateway in an alphabetical order. For commands regarding special configuration of the Gateway, refer to the previous chapters.

4.12.1. Command in the EXEC Mode

[Command Formats and Optional Commands]

- **clear { counters/interface/logging/utilization }**
 1. Resets certain functions or certain parts of the Gateway.
 2. Command options are as follows:
 - 1) counters: Clears counters of all interfaces.
 - 2) interface: Resets the hardware logic of a certain interface and functions as if rebooting certain interfaces.
 - 3) logging: Clears logging buffer. To be implemented later.
 - 4) utilization: Clears system utilization information of the Gateway.

- **clock [current/running/start]**
 1. Shows the system clock of the Gateway.
 2. Command options
 - 1) current: Shows current time of the Gateway.
 - 2) running: Shows total running time.
 - 3) start: Shows the time that the Gateway started operating.
 3. If there is no option, all values of the three options will be displayed.

- **configuration**

Enters into the configuration mode.

- **copy {startup-clear/running-config}**
 1. Saves or deletes configuration data.
 2. Command options are as follows:
 - 1) startup-clear: Deletes configuration data that is saved in the flash memory of the present Gateway.
 - 2) running-config: Saves currently running configuration information in the Gateway.

- **Debug <Option>**
 1. Decodes packets passing the Gateway and indicates if the Gateway normally operates.
 2. For more information and options, see "4.13 Fault Handling and Debugging."
 3. When disabling Debug, use "no debug" or "Un-debug" commands.

- **exit**
 1. Exits from the current mode, and enters into the next lower mode.
 2. If the user uses "exit" command in the Exec mode, the user needs to log in again.

- **help**

Describes the interactive help system.

- **history**
 1. Shows commands history.
 2. The AP200 Gateway keeps maximum 25 histories for each mode.
 3. To use a command again used in History, enter "! History#."

- **no {option}**

An important command to negate commands used or set.

- **ping** [-flt] [-s *source-ip-address*] *Target-host-IP* [*datasize(max:1500)*]
[*npackets*]

1. Sends echo messages.
2. Command options are as follows:
 - 1) [-f: fast send mode]
 - 2) [-l: loopback mode for HDLC]
 - 3) [-t: sends one datagram per seconds]
 - 4) [-s: specify the sending interface IP address]

- **reboot**

Reboots the Gateway system.

- **rlogin** [-l *username*] *target-host*

Opens the Rlogin connection.

- **show {option}**

1. Shows information that has been set in the Gateway or collected by the Gateway. With this function, the user can check operation status of the Gateway.
2. For more information and options, see "4.13 Fault Handling and Debugging."

- **telnet { target-host-ip }**

Opens a Telnet connection in the remote host.

- **test { memory/interface } [Ethernet/hdlc] [*main-interface.sub-interface*]**

1. Tests the Gateway itself.
2. Command options are as follows:
 - 1) memory: Tests the Gateway memory.
 - 2) interface: Performs the loopback test for the designated interface.

- **traceroute** [-w *waittime*] [-m *max_ttl*] [-s *src_addr*] *host* [*packetlen*]

Checks the path that can be accessed through a remote host.

- **undebug** <Option>
 1. Negates debugging configuration.
 2. For more information and options, see “ 4.13 Fault Handling and Debugging.”

4.12.2. Command in the Global Configuration Mode

[Command Formats and Optional Commands]

- **access-list { option }**
 1. Sets the access-list for the packet.
 2. For more information, see Access-List in the previous chapter.

- **arp {option}**
 1. Statically or dynamically registers ARP entries.
 2. Option
 - 1) **request** [*ip-address-number*]: Forcefully sends ARP requests for the host of a certain IP and registers it in the ARP table
 - 2) **static** <*ip-address-number*> <*mac-address-number*>: Statically registers Mac address in the ARP table for the IP host.
 - 3) **table-size** <*table-size-number*>: Sets the size of the ARP table. The AP200 Gateways supports 10 ~ 256 size.

- **bridge { option }**
 1. Sets the bridge.
 2. For more information, see Bridge Configuration in the previous chapter.

- **clock** [*yy mm dd hh mm ss*]

Sets the system clock of the present Gateway.

- **dhcp-list { option }**
 1. Sets the DHCP.
 2. For more information, see DHCP Configuration in the previous chapter.

- **Ethernet [full-duplex]**
 1. Sets the Ethernet interface as full-duplex.
 2. The default is half-duplex.

- **exit**
 1. Exits from the current mode, and enters into the next lower mode.
 2. If the user uses "exit" command in the global configuration mode, the user will be able to go back to the Exec mode.

- **help**

Describes the interactive help system.

- **history**
 1. Shows history of used commands.
 2. The AP200 Gateway keeps maximum 25 histories in each mode.
 3. To use the command again, enter "! History#."

- **hostname { host-name }**

Sets a name in the network of the Gateway.

- **interface { ethnet/null/loopback } < main-interface.sub-interface >**

Enters into a configuration mode of a certain interface.

- **logging { option }**
 1. Sets logging of the equipment.
 2. For more information, see "4.13 Fault Handling and Debugging."

- **nat-list { option }**
 1. Sets Network Address Translation (NAT.)
 2. For more information, see NAT Configuration in the previous chapter.

- **no {option}**

An important command to negates commands that the user used or have been set.

- **queue-list { option }**
 1. Sets the traffic queuing.
 2. For more information, see Traffic Queuing Configuration in the previous

chapter.

- **route {option}**

1. Sets the static route.
2. For more information, see Routing Configuration in the previous chapter.

- **router static**

1. Enables or disables static routing process.
2. For more information, see Routing Configuration in the previous chapter.

- **service {ftpd/snmpd/telnetd/tftpd}**

1. Enables Application demon for a certain service.
2. To disable the service, use "**no service**" command.

- **snmp { option }**

1. Sets SNMP protocol for management.
2. For more information, see SNMP Configuration.

- **user { Option }**

1. Commands to manage Gateway users.
2. For more information, see "4.14 User, Password, Software Image and Configuration File Management."

- **utilization { cpu/Ethernet/serial } [interface] [interface-number]**

[measuring-period]

1. Checks the availability of the CPU or a certain interface. With this command, the user can check the availability at a certain interval.
2. The default is five minutes.

4.13. Fault Management and Debugging

This chapter describes how to handle and process faults while operating the VoiceFinder AP200 Gateway. The AP200 Gateway provides "show" commands, "Debug" commands and "logging" commands for fault handling.

4.13.1. Logging Command

Logging commands log equipment operation status to manage equipments, and decide the level of log information. Logging commands also can send log information to a certain host outside. Logging configuration can be made in the global configuration mode.

Logging configuration related commands are as follows:

- **logging on**
Enables logging for all available destinations.

- **logging condition {option}**
 1. Sets commands to logging targets.
 2. Option
 - 1) **command**: Logs commands used.
 - 2) **event interface {Ethernet/serial} [interface-number]**: Logs changes of a certain interface.
 - 3) **event protocol {all/critical/icmp}**: Logs events of certain protocol.
 - 4) **alarm {all/critical/information/major/minor/warning}**: Sets logging targets for alarms of certain level.
 - 5) **debugging**: Logs debugging information.

- **logging destination {option}**
 1. Sets conditions of the destination host to send logging information.
 2. Option

- 1) **ip** *<destination-ip-address>*: Sets IP address of the remote host to send logging information.
- 2) **port** *[port-number]*: Defines the port number of the remote host to send logging information.
- 3) **on**: Enables logging in the remote host.

4.13.2. Show commands

With "show" command, the user can check configuration that the device manager has set and system status.

"Show" command can be used in the Exec mode and the syntaxes are as follows:

- **Show {option}**: Displays option contents.
"show" command related optional commands are as follows:
- **access-list** *[access-list-number]*
 1. Shows the access-list that has been set.
 2. For more information, see Access-List Configuration in the previous chapter.
- **arp** *[ip-address for ARP entry]*
Shows the contents of the ARP table.
- **bridge**
 1. Shows forwarding/blocking database of the bridge.
 2. For more information, see Bridge Configuration in the previous chapter.
- **clock** *[current/running/start]*
Shows the system clock of the current Gateway.
- **debug-port**
show current debug terminal information.

- **dhcp-list** [*dhcp-list-number*]
 1. Shows the DHCP that has been set.
 2. For more information, see DHCP Configuration in the previous chapter.

- **ethernet**

Shows the mode and operation rate of the Ethernet interface.

- **interface** [Ethernet/null/loopback] [<main-interface>.<sub-interface>]

Describes the status and the configuration of the interface.

- **logging** [history]
 1. Shows contents of the logging buffer.
 2. History option shows the contents of the system log history table.

- **nat-list** [*nat-list-number*]
 1. Shows the NAT that has been set.
 2. For more information, see NAT Configuration in the previous chapter.

- **proxy-arp**

Indicates if the proxy ARP is enabled.

- **route** [static]
 1. Shows decided route information table.
 2. Static option shows each table as using the algorithm of the corresponding option.
 3. For more information, see Routing Configuration in the previous chapter.

- **router**
 1. Displays enabled routing processes.
 2. For more information, see Routing Configuration in the previous chapter.

- **running-config**

Shows currently running configuration file.

- **session**
Displays information of the Telnet session that is currently connected to the Gateway.

- **service**
Displays enabled service processes in the current Gateway.

- **snmp**
Displays SNMP protocol state of the Gateway and options.

- **Spanning-Tree**
If the bridge is currently enabled in the Gateway, this command displays spanning-tree topology.

- **static**
Displays static routes that are set in the Gateway.

- **system task**
Shows information and the state of the task that is currently running in the Gateway.

- **tcp**
Displays information and the state of the external system that is connected to the TCP among information of the current Gateway.

- **udp**
Displays information and the state of the external system that is connected to the UDP among information of the current Gateway.

- **user**
Displays profiles of the users registered in the Gateway.

- **utilization** { *cpu/ethernet* } [**interface**] [*interface-number*]
[*measuring-period*]

Shows utilization state and values currently set.

- **version**

Shows hardware information of the Gateway and software version that is currently running in the Gateway.

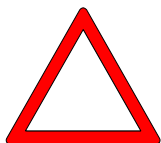
4.13.3. Debug Commands

Information



Debug commands decodes certain packets that pass through Gateway devices and indicate whether packets are normal or not to the manager. Therefore, the manager can check whether the network or device has been normally set up or not. Debug commands can be used in the Exec mode.

Caution



Note that Debug commands use a lot of system resources. Therefore, minimize the range of using Debug commands. Also, since Debug commands greatly lower general performances of the system, turn off Debug commands.

“debug” command can be used in the Exec mode, and its syntax is as follows:

- **debug {option}**: Enables debugging.
- **no debug {option}**: Enables debugging that has been set.
- **undebug {option}**: Disables debugging.

Debug related optional commands are as follows:

- **ppp {chap/error/negotiation/packet }**
 1. Decodes and shows configuration and operation status of PPP.
 2. Details of each option are as follows:
 - 1) chap: Decodes and shows information exchanged when CHAP is being set.
 - 2) error: Decodes and shows error information in the PPP process.
 - 3) negotiation: Decodes and shows PPP link negotiation information.
 - 4) packet: Decodes PPP packets.

- **tcpip {arp/icmp/tcp/udp }**
 1. Decodes and shows TCP/IP packets passing through the Gateway.
 2. Details of each option is as follows:
 - 1) arp: Decodes and shows ARP packets.
 - 2) icmp: Decodes and shows ICMP packets.
 - 3) tcp: Decodes and shows TCP/IP packets.
 - 4) udp: Decodes and shows UDP/IP packets.

4.14. User, Password, Software Image and Configuration File Management

This chapter describes how to register and change users, recover passwords, download and back up software image, and back up and restore configuration file, which are very useful in using the VoiceFinder AP200 Gateway.

4.14.1. User Registration and Change

This chapter describes how to register Gateway users, change passwords and change user's authorities.

Commands relating to managing Gateway users are as follows:

- **user {option}**: Registers of changes users.

User's command related optional commands are as follows:

- **add** <login-name> <password> [**admin/high/normal/low**]
 1. Registers Gateway users.
 2. Sets the user's authority level as admin, high, normal or low.
- **change** <login-name> <old-password> <new-password>
Changes the password of the Gateway user.
- **level** <login-name> <password> [**admin/high/normal/low**]
 1. Changes the authority level of the Gateway user.
 2. Changes the user authority level into admin, high, normal and low.

- **timeout** <login-name> <timeout-period>
 1. For the security reason, this command defines timeout value according to the Gateway user when the console of the Telnet session is idle.
 2. If timeout is 0, it means "forever."

4.14.2. Password Recovery

The Gateway manager shall know the password to change Gateway configuration and check the Gateway status. Therefore, the Gateway manager shall remember the password and keep it confidentially. This chapter describes how to recover the password when the Gateway manager forgets the password.

The following describes how to recover the password.

[Procedure]

Order	Operation
1	Connect the console and prepare to recover the password. Password recovery shall be made in the console only.
2	Initialize the system. (Turn on/off the system.)
3	After the initial messages are displayed, enter Ctrl+x and Ctrl+c once or twice.
4	Wait for a while until entering into the boot mode.
5	Use "Show password" command to check the root password.
6	Reboot the system.
7	Log in the system with verified password.

Initialize the system in the booter mode, not in the Gateway program state. *To enter into the booter mode, enter Control-X and Control-C keys once or twice when the booter mode message Appears.*

In the booter mode, "BOOT#" prompt Appears on the screen as in the following figure. See the following figure.

```
System Boot Loader, Version 1.10a
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

System Flash Memory is 4 Mbytes.
1 Ethernet/IEEE 802.3 Interface (10BaseTX).
1 RS232 serial console port, 2 Serial networks interface.

The "BOOT LOADER" is ready

1 BOOT# ?
configure : Enter configuration mode
copy : Copy configuration data
exit : Exit from the EXEC
history : Show command line history
ping : Send echo messages
reboot : reboot system
show : Show running system information
telnet : Open a telnet connection
2 BOOT#
```

[Boot Mode Login Screen]

Available commands in the booter mode: Enter "?" as in the normal Gateway mode.

```
1 BOOT# ?
configure : Enter configuration mode
copy : Copy configuration data
exit : Exit from the EXEC
history : Show command line history
ping : Send echo messages
reboot : reboot system
```

show : Show running system information
telnet : Open a telnet connection

Verify "root" command that is currently set. The following is when "root" password is "Gateway".

```
3 BOOT# sh password  
password = "Gateway"
```

To change current password in the booter, enter into the configuration mode and change the password as using "passwd" command.

The following is when changing "root" command into "Gateway1".

[Example] Password Change Configuration and Usage

```
1 BOOT# conf  
1 BOOT(config)# ?  
    address : Set the IP address of an interface  
    clock : Manage the system clock  
    exit : Exit from the EXEC  
    history : Show command line history  
    passwd : Change password  
2 BOOT(config)# passwd ?  
    <new password> New Password  
3 BOOT(config)# passwd Gateway ?  
    <repeat new password> New Password for confirm  
4 BOOT(config)# passwd Gateway Gateway ?  
    < cr >  
5 BOOT(config)# passwd Gateway Gateway1  
    password changed  
6 BOOT(config)#
```

4.14.3. Software Image Upgrade and Backup

Software of the AP200 Gateway is regularly upgrade according to functional upgrade or bug fix. It is recommended for the existing users to upgrade software in this method. This chapter describes how to upgrade or back up Gateway software.

The following describes how to upgrade or back up Gateway software and related commands.

If the user uses FTP, the user must enter correct user ID and password when logging in the system. Firstly, if the user upgrades new Gateway software from the user consol of the PC or a workstation through FTP, the user shall use "put" command. Or, to download Gateway software that is currently in use to a PC or a workstation, the user shall use "get" command.

The following is when downloading Gateway software that is currently in use to a PC. Use "put" command to copy software to be upgraded to the current directory. Use "put" command instead of "get" command.

[Example] Software Backup through FTP

```
155 sun10:#> ftp 211.170.87.221
Connected to 211.170.87.221.
220 Gateway FTP server (Version 1.12) ready.
Name (211.170.87.221:noname): root
331 Password required for root.
Password:
230 User root logged in ok.
ftp> bi
200 Type set to I.
ftp> get Gateway.bin
200 PORT command successful.
150 BINARY data connection for Gateway.bin (211.170.87.99,44100).
```

```
226 BINARY Transfer complete.  
local: Gateway.bin remote: Gateway.bin  
201622 bytes received in 0.52 seconds (375.13 Kbytes/s)  
ftp> quit  
221 Goodbye.  
156 sun10:/#>
```

Software backup method through TFTP is same as FTP. However, login procedure is not necessary. The following is when using "put" command for software. When software upgrade is completed, "Gateway Software is updated" is displayed in the screen.

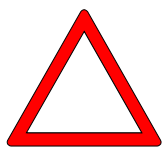
[Example] Software Upgrade through TFTP

```
156 sun10:##> tftp 211.170.87.221  
tftp> bi  
tftp> put addpac.bin  
Sent 201622 bytes in 0.4 seconds  
tftp> quit  
157 sun10:##>
```

The following message is displayed in the user's console.

```
"software image" is updated
```

Caution



To upgrade or backup software image, use a same procedure in the Gateway program that is currently in use or in the booter mode. If any fault occurs in the currently running Gateway program, upgrade software image by the procedure explained above.

4.14.4. Backup and Restore of Configuration File

The AP200 Gateway saves the configuration file in the flash memory of the Gateway. However, sometimes it is necessary to back up the configuration file or restore the backed up configuration file. This chapter describes how to back up or restore the configuration file and related commands. Backup and restoring procedures of the configuration file are same as upgrade and backup procedures of software image. However, the configuration file name is Gateway.cfg. The configuration file is backed up or restored through FTP/TFTP. When restoring is completed, "Config Database is updated" message is displayed in the screen. When backing up the configuration file, use "get" command, and when restoring the configuration file, use "put" command. The following is an example of backup and restore of configuration information through TFTP.

[Example] Software Backup through FTP

```
155 sun10: #> ftp 211.170.87.221
Connected to 211.170.87.221.
220 Gateway FTP server (Version 1.12) ready.
Name (211.170.87.221:noname): root
331 Password required for root.
Password:
230 User root logged in ok.
ftp> bi
200 Type set to I.
ftp> get Gateway.cfg
200 PORT command successful.
150 BINARY data connection for Gateway.cfg (211.170.87.99,44100).
226 BINARY Transfer complete.
local: Gateway.cfg remote: Gateway.cfg
2016 bytes received in 0.0 seconds (375.13 Kbytes/s)
ftp> quit
```

```
221 Goodbye.  
156 sun10:/#>
```

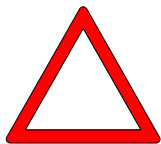
[Example] Backup and Restore of Configuration Information through TFTP

```
56 sun10#> tftp 211.170.87.221  
tftp> bi  
tftp> get Gateway.cfg  
Received 201622 bytes in 0.4 seconds  
tftp> quit  
157 sun10:/#>  
158 sun10:/#> tftp 211.170.87.221  
tftp> bi  
tftp> put Gateway.cfg  
Sent 201622 bytes in 0.5 seconds  
tftp> quit
```

The following message is displayed on the console.

```
"Config Database" is updated
```

Caution



To back up or restore the configuration file, use a same procedure in the Gateway program that is currently in use or in the booter mode. If any fault occurs in the currently running Gateway program, restore the configuration file in the booter mode by the procedure explained above.

4.15. Auto-Upgrade

When the software (firmware) of AP200 VoIP gateway is upgraded or changed, auto-upgrade feature reduces the time-consuming point-to-point S/W upgrading JOB via human being at remote side or local site. The point-to-point S/W upgrading at remote side is impossible practically if the deployed VoIP equipment number is large. This means the maintenance cost will be reduced if automatic S/W feature without human being intervene is available.

Generally, when the VoIP gateways are deployed at customer sites via VoIP service provider, VoIP service provider needs a lot of cost for VoIP equipment H/W and S/W maintenance. Among these maintenance cost, S/W upgrading cost is mostly charged.

STEP	Commands	Explanation
1	Router# configure	Enter to the configuration mode
2	Router(config)# auto-upgrade <i>{action / authentication / auto-reboot / interval / proxy / url / verbose}</i>	action : Auto-Upgrade timing (display after URL input) authentication : Web server (auto upgrade server) Login Name /Password auto-Reboot : rebooting decision after image upgrade interval : Retrial time at Failure, Success proxy : proxy server IP Address setting url : Web server (auto upgrade server) url setting verbose : display the status message at upgrade

Chapter 5. Voice Configuration and Command

This chapter explains voice configuration and commands to operate voice integration function of VoiceFinder AP200.

5.1. Voice Technologies and Concepts

5.1.1. Voice Over IP

Voice over IP enables a VoIP Gateway to carry voice traffic (for example, telephone calls and faxes) over an IP network. In Voice over IP, the DSP segments the voice signal into frames, which are then coupled in groups of two and stored in voice packets.

These voice packets are transported using IP in compliance with ITU-T specification H.323.

Because it is a delay-sensitive Application, you need to have a well-engineered network end-to-end to successfully use Voice over IP.

Fine-tuning your network to adequately support Voice over IP involves a series of protocols and features geared toward quality of service (QoS). Traffic shaping considerations must be taken into account to ensure the reliability of the voice connection.

Voice over IP is primarily a software feature; however, to use this feature on a VoIP Gateway, you must install a voice interface cards, each of which is specific to a particular signaling type associated with a voice port.

5.1.2. Codecs and MOS(Mean Opinion Score)

5.1.2.1. Codecs

PCM and ADPCM are examples of "waveform" CODEC techniques. Waveform CODECs are compression techniques that exploit the redundant characteristics of the waveform itself.

In addition to waveform CODECs, there are source CODECs that compress speech by sending only simplified parametric information about voice transmission; these CODECs require less bandwidth. Source CODECs include linear predicative coding (LPC), code-excited linear prediction (CELP), and multi-pulse, multi-level quantization (MP-MLQ).

Coding techniques are standardized by the ITU-T in its G-series recommendations. The most popular coding standards for telephony and voice packet are:

- G.711--Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs.
- G.723.1--Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This CODEC has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility.
- G.729--Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM.

5.1.2.2. Mean Opinion Score

Each CODEC provides a certain quality of speech. The quality of transmitted speech is a subjective response of the listener. A common benchmark used to determine the quality of sound produced by specific CODECs is the mean opinion score (MOS). With MOS, a wide range of listeners judge the quality of a voice sample (corresponding to a particular CODEC) on a scale of 1 (bad) to 5 (excellent). The scores are averaged to provide the mean opinion score for that sample. [Table 5-1](#) shows the relationship between CODECs and MOS scores.

Table 5-1: Compression Methods and MOS Scores

Compression Method	Bit Rate (kbps)	Processing (MIPS)	Framing Size	MOS Score
G.711 PCM	64	0.34	0.125	4.1
G.729 CS-ACELP	8	20	10	3.92
G.729a CS-ACELP	8	10.5	10	3.7
G.723.1 MP-MLQ	6.3	16	30	3.9
G.723.1 ACELP	5.3	16	30	3.65

Although it might seem logical from a financial standpoint to convert all calls to low-bit rate CODECs to save on infrastructure costs, you should exercise additional care when designing voice networks with low-bit rate compression. There are drawbacks to compressing voice. One of the main drawbacks is signal distortion due to multiple encoding (called tandem encoding). For example, when a G.729 voice signal is tandem encoded three times, the MOS score drops from 3.92 (very good) to 2.68 (unacceptable). Another drawback is CODEC-induced delay with low bit-rate CODECs.

One of the most important design considerations in implementing voice is minimizing one-way, end-to-end delay. Voice traffic is real-time traffic; if there is too long a delay in voice packet delivery, speech will be unrecognizable. Delay is inherent in voice-networking and is caused by a number of different factors. An acceptable delay is less than 200 milliseconds.

There are two kinds of delay inherent in today's telephony networks: propagation delay and handling delay. Propagation delay is caused by the characteristics of the speed of light traveling via a fiber-optic-based or copper-based media. Handling delay (sometimes called serialization delay) is

caused by the devices that handle voice information. Handling delays have a significant impact on voice quality in a packetized network.

CODEC-induced delays are considered a handling delay. [Table 5-2](#) shows the delay introduced by different CODECs.

Table 5-2: CODEC-Induced Delays

CODEC	Bit Rate (kbps)	Compression Delay (ms)
G.711 PCM	64	0.75
G.726 ADPCM	32	1
G.728 LD-CELP	16	3 to 5
G.729 CS-ACELP	8	10
G.729a CS-ACELP	8	10
G.723.1 MP-MLQ	6.3	30
G.723.1 ACELP	5.3	30

5.1.3. Dial Peer

The key to understanding our voice implementation is to understand the use of dial peers. Dial peers describe the entities to and/or from which a call is established. All of the voice technologies use dial peers to define the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection, as shown in [Figure 5-1](#) and [Figure 5-2](#). Four call legs comprise an end-to-end call, two from the perspective of the source Gateway as shown in [Figure 5-1](#), and two from the perspective of the destination Gateway, as shown in [Figure 5-2](#). You use dial peers to Apply specific attributes to call legs and to identify call origin and destination. Attributes Applied to a call leg include Quality of Service (QoS), compression/decompression (CODEC), Voice Activation Detection (VAD), and fax rate.

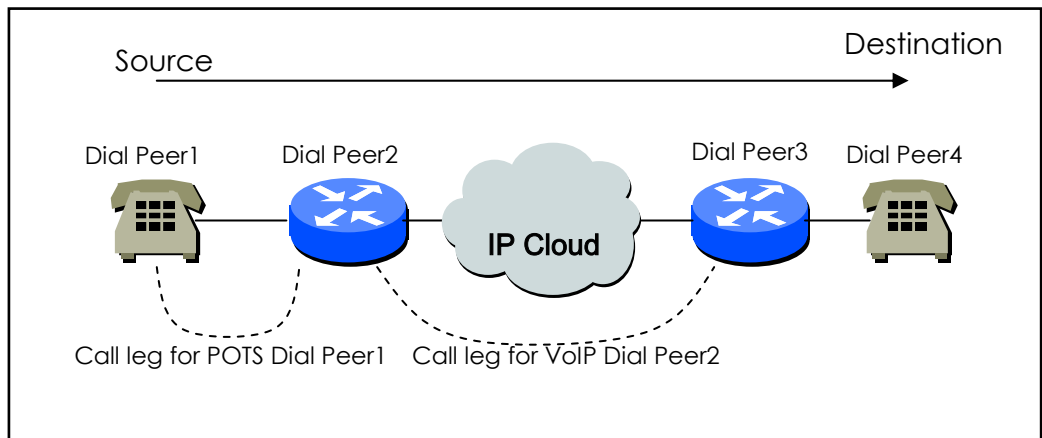


Figure 5-1: Dial Peer Call Legs from the Perspective of the Source Gateway

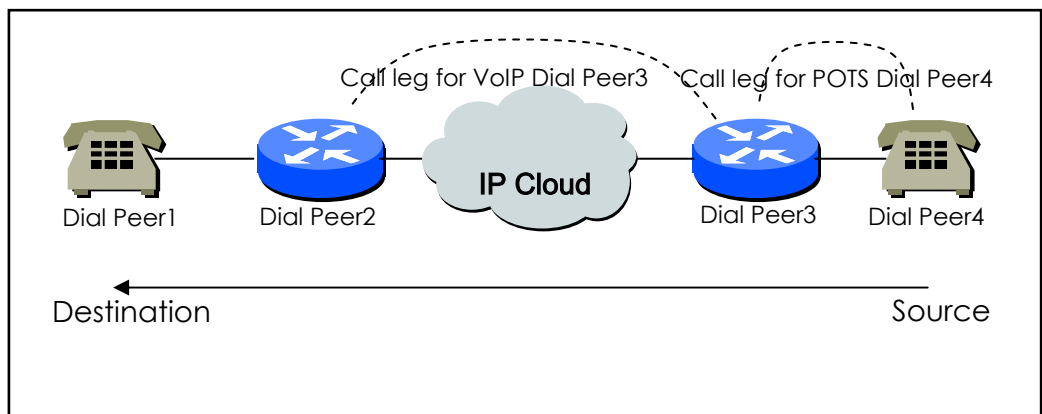


Figure 5-2: Dial Peer Call Legs from the Perspective of the Destination Gateway

There are basically two different kinds of dial peers with each voice implementation:

- POTS Dial peer : POTS Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device.

When configuring POTS dial peers, the key commands that must be configured are the **port** and **destination-pattern** commands. The **destination-pattern** command defines the telephone number associated with the POTS dial peer. The **port** command associates the POTS dial peer with a specific logical dial interface, normally the voice port connecting the AP200 VoIP Gateway to the local POTS network.

When configuring Voice over IP on the AP200 VoIP Gateway, direct inward dial can be configured on a POTS dial peer. In this case, the key commands that must be configured are the **destination-pattern** and **direct-inward-dial** commands.

- Voice Network Dial peer : Voice Network Dial peer describing the characteristics of a packet network connection; for example, in the case of Voice over IP, this is an IP network. Voice-network peers point to specific voice-network devices.

When configuring voice-network dial peers, the key commands that must be configured are the **destination-pattern** and **session-target** commands. The **destination-pattern** command defines the telephone number associated with the voice-network dial peer. The **session-target** command specifies a destination address for the voice-network peer.

If configuring a Voice over IP network peer, the session target is a destination IP address.

5.1.4. Voice Ports

Voice port commands for AP200 VoIP Gateway define the characteristics associated with a particular voice-port signaling type. Voice ports for both the AP200 VoIP Gateway provides support for three basic voice signaling formats:

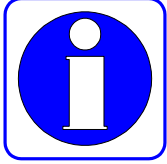
- FXO---Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the public switched telephone network's (PSTN's) central office (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premise extension Applications.
- FXS---The Foreign Exchange Station interface. The FXS interface is an RJ-11 connector that allows connection for basic telephone equipment, keysets, and PBXes; FXS connections supply ring, voltage, and dial tone.

The AP200 series currently provides only analog voice ports for its implementation of Voice over IP. The type of signaling associated with these analog voice ports depends on the interface module installed into the device.

The voice port syntax depends on the hardware platform is being configured. On the AP200 series, the voice-port syntax is **voice-port** *slot-number/port-number*.

5.2. VoIP Interface Configuration

Information



AP200 Gateway has various interfaces. They are Ethernet Interface and loop-back interface which are defined to provide VoIP service among those interfaces.

As Default, Ethernet 0.0 Interface is defined to provide VoIP service, and can be determined to use for another purpose by the following procedure.

In case VoIP Interface is changed during VoIP service, the signal connecting is terminated and reregistering process with gatekeeper is executed. Therefore, it is advised that VoIP Interface should not be changed after the initiate Setup of System is completed.

If the defined VoIP interface doesn't have IP address, you cannot configure and search those related to VoIP. Therefore, defining VoIP Interface and setting up IP Address is prior to make configuration those related VoIP.

In case Ip Address of VoIP Interface is changed during VoIP service, the signal connecting is terminated and reregistering process with gatekeeper is executed

Step	Commands	Description
1	Router# configure	Go to Configuration Mode.
2	Router(config)# voice-interface <i>interface-name</i>	Define Interface on Router. The names of Interfaces are, for instance, Ethernet 0.0, Ethernet 1.0, serial 0, and so on.

5.3. **Numbering Plan, Number Handling and Dial Peer Configuration**

5.3.1. **Numbering Plan**

The initiation of VoIP router (or gateway) configuration is to plan the number scable, efficient, and proper between routers.

Public telephone network has hierarchical structure, (Country Code) + (Area Code) + (Dialing Code) + (Directory Number) so that this hierarchical number planning is advantageous. As each router on Volp network is corresponding to switcher on telephone network, have a number plan in accordance the size of VoIP network.

It is important if router is confabulated with gatekeeper in number planning. In case of being configurated with the existing gatekeeper, you should follow the number planning defined on the gatekeeper.

The simplest number configuration makes router have the public telephone number already used in the setting place of router. This means a call trial to the public telephone number is advantageous by the number when co-operating with other VoIP routers or failing in VoIP call.

When configurating VoIP network with strong private feature, configurate network by having a private number planning.

5.3.2. Dial Peer Configuration

The key point to understanding how Voice over IP functions is to understand dial peers. Each dial peer defines the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. All the call legs for a particular connection have the same connection ID.

There are two different kinds of dial peers:

- POTS---Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device.
- VoIP---Dial peer describing the characteristics of a packet network connection; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

5.3.2.1. Inbound Dial Peer versus Outbound Dial Peer

Dial peers are used for both inbound and outbound call legs. It is important to remember that these terms are defined from the *router's* perspective. An inbound call leg originates *outside* the router. An outbound call leg originates *from* the router.

For inbound call legs, a dial peer might be associated to the calling number or the port designation. Outbound call legs always have a dial peer associated with them. The destination pattern is used to identify the outbound dial peer. The call is associated with the outbound dial peer at configuration time.

POTS peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received and outgoing calls can be placed. VoIP peers point to specific devices (by associating destination telephone numbers with a specific IP address) so that incoming calls can be received and outgoing calls can be placed. Both POTS and VoIP peers are needed to establish Voice over IP connections.

Establishing communications using Voice over IP is similar to configuring an IP static route: you are establishing a specific voice connection between two

defined endpoints. As shown in Figure 5.3, for outgoing calls (from the perspective of the POTS dial peer 1), the POTS dial peer establishes the source (via the originating telephone number or voice port) of the call.

The VoIP dial peer establishes the destination by associating the destination phone number with a specific IP address.

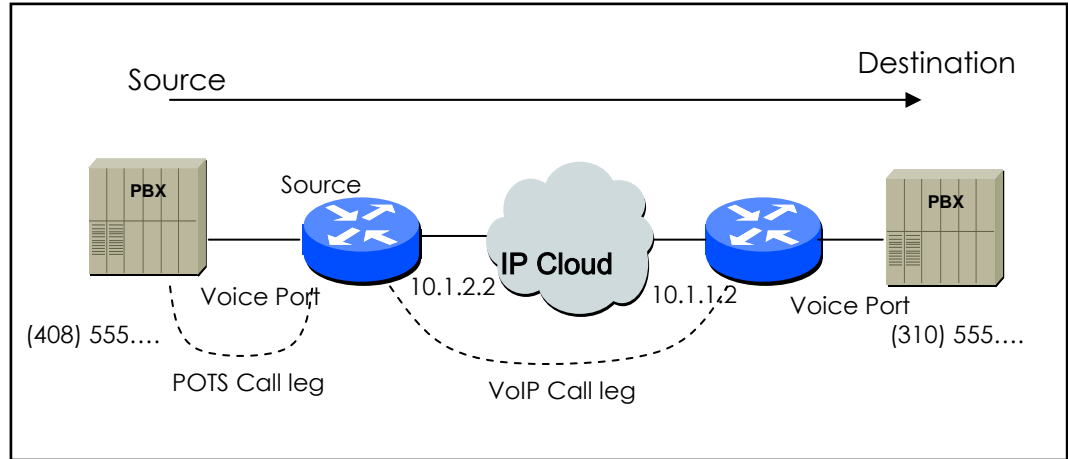


Figure 0.1 Outgoing Call in POTS Dial Peer 1's view

To configure call connectivity between the source and destination as illustrated in Figure 5.3, enter the following commands on router 10.1.2.2:

```
dial-peer voice 1 pots
  destination-pattern 1408555 . . . .
  port 0

dial-peer voice 2 VoIP
  destination-pattern 1310555 . . . .
  session target 10.1.1.2
```

In the previous configuration example, the last four digits in the VoIP dial peer's destination pattern were replaced with wildcards.

This means that from access server 10.1.2.2, calling any number string that begins with the digits "1310555" will result in a connection to access server 10.1.1.2.

This implies that access server 10.1.1.2 services all numbers beginning with those digits. From access server 10.1.1.2, calling any number string that begins with the digits "1408555" will result in a connection to access server 10.1.2.2.

This implies that access server 10.1.2.2 services all numbers beginning with those digits. For more information about stripping and adding digits, see the "Outbound Dialing on POTS Peers" section.

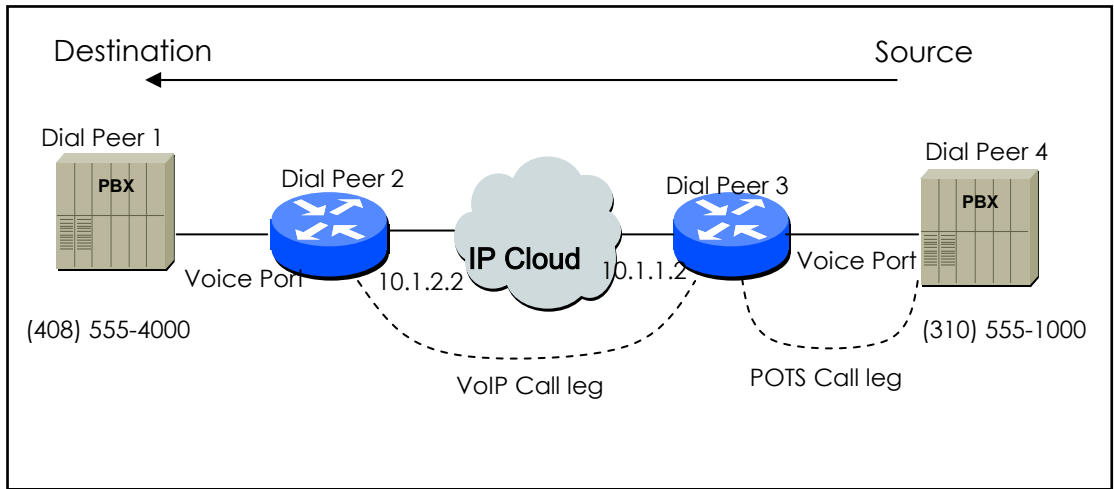


Figure 0.2 Outgoing Call in POTS Dial Peer 2's view

To complete the end-to-end call between dial peer 1 and dial peer 4 as illustrated in Figure 5.4 enter the following commands on router 10.1.1.2:

```
dial-peer voice 4 pots
  destination-pattern 1310555 . . . .
  port 0

dial-peer voice 3 VoIP
  destination-pattern 1408555 . . . .
  Session target 10.1.2.2
```

As explained above, call inside router is completed by selecting inbound dial and outbound.

While the selection of Outbound Dial peer is basically decided by matching

POTS Peer and VoIP Peer with destination pattern of Dial Peer, Inbound Dial Peer is decided by other ways.

First of all, the procedure of Inbound POTS Peer is as follows;

- Select POT peer assigned by voice port receiving Call.
- In case more than one POTS peer are assigned by voice port, the POT made at first gets selected.

The procedure of selection for VoIP is as follows;

- Select VoIP peer having the same IP address with receiving Router among VoIP peers.
- When the above selection fails, Select VoIP peer having answer-address which is matched with calling party number of Inbound call.
- When the above selection fails, Select VoIP peer having destination-pattern which is matched with calling party number of Inbound call.

The selection of Inbound Dial Peer is a proper measurement for receiving side. That is, parameters assigned by POTS or VoIP peer applies to the elected dial Peer. Ultimately, because the failure of the selection for VoIP means POTS peer related to choice port doesn't exist, Call doesn't advance. Meanwhile Inbound VoIP peer will proceed regardless of the selection of Inbound VoIP peer.

5.3.2.2. POTS Peer Configuration

Set the POTS peer as follows:

- Decide the dial peer tag value.
- Decide the destination pattern.
- Decide the port.

In most of the cases, other values than these shall be default values.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# dial-peer voice tag pots	Enters into the POTS configuration mode of the dial-peer configuration. The tag is the only identifier of the dial-peer of this system, and tag values range from 0 to 65535. The POTS means communication service configuration of the FXS and the FXO ports.
3	Router(config-dial-peer)# destination-pattern string [T]	Enters the telephone number of the corresponding dial peer. The string means the telephone number, and string values include 0 ~ 9, (#), (*) and the wildcard (.) The period (.) means the wildcard. Users can selectively enter (T) after the telephone number, and if a user enters (T) the system will collect dial digits till the end-of-dialing key (default #) or till the interdigit timer finishes.
4	Router(config-dial-peer)# port location	Maps the corresponding POTS with the port that the location indicates. The location is indicated by the slot-number or the port-number.
5	Router(config-dial-peer)# prefix string	(Selectively used.) When the corresponding POTS is selected as the termination side, the string is automatically dialed-out. String values include 0 ~ 9, (#), (*) and (.). When there is (.) the corresponding digit stops dialing-out for one second.
6	Router(config-dial-peer)# exit	Terminates the dial peer configuration mode.

5.3.2.3. VoIP Peer Configuration

Set the VoIP peer as follows:

- Decide the dial peer tag value.
- Decide the destination pattern.
- Decide the session target.

In most of the cases, other values than these shall be default values.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# dial-peer voice tag VoIP	Enters into the VoIP configuration mode of the dial-peer configuration. The tag is the only identifier of the dial-peer of this system, and tag values range from 0 to 65535. The VoIP means communication service configuration of the VoIP peer.
3	Router(config-dial-peer)# destination-pattern string [T]	Enters the telephone number of the corresponding dial peer. The string means the telephone number, and string values include 0 ~ 9, (#), (*) and wildcard (.). The period (.) means the wildcard. Users can selectively enter (T) after the telephone number, and if a user enters (T) the system will collect dial digits till the end-of-dialing key (default #) or till the inter-digit timer finishes.
4	Router(config-dial-peer)# session target destination-ip-address	Enters the IP address of the corresponding VoIP peer. The <i>destination-ip-address</i> shall be entered as a dotted decimal IP address. (Example: 123.321.1.2) If the <i>destination-ip-address</i> is "ras" the IP address of the corresponding VoIP peer will be found through the gatekeeper.
5	Router(config-dial-peer)# dtmf-relay [h245-alphanumeric]	(Selectively used.) Decides the DTMF tone transmission method for the corresponding VoIP-peer. The default value is h245-alphanumeric .

5.3.2.4. Setting CODEC and VAD in the Dial Peer

To set the COder-DECoder(CODEC) and the Voice Activity Detection (VAD) in the dial peer, how much bandwidth the voice session can have shall be defined. Normally, the CODEC converts analog signals into digital bit streams or vice versa. During this procedure, the CODEC defines the voice coder rate for the dial peer. The VAD prohibits silent packets (created while the caller/callee does not talk) from being sent.

5.3.2.4.1. Setting CODEC in the VoIP Dial Peer

To set the coder rate for the selected VoIP peer, use the following commands in the global setup mode (start.)

Step	Command	Description
1	<code>dial-peer voice tag VoIP</code>	Enters into the dial-peer setup mode to set the VoIP peer.
2	<code>codec [g711alaw / g711ulaw /g729 / g7231r63 /g7231r53]</code>	<ul style="list-style-type: none"> Selects the CODEC for the voice considering the coder rate.

Codec The default of "Codec" command is **g7231r63**. In normal cases, the default value is the most suitable. However, to connect to a network that has a high bandwidth or to get the highest voice quality, select **g711alaw** or **g711ulaw** from "Codec" command. These values allows better voice quality but requires more bandwidth for the voice session.

For example, to use a CODEC with G.711a-law Rate for the VoIP dial peer 108, set the CODEC as follows:

```
dial-peer voice 108 VoIP
destination-pattern 14085551234
codec g711alaw
session target 10.0.0.8
```

Besides doing the above, users can create CODEC classes and store them in

the VoIP peer. While the above method sets only one CODEC, creating a CODEC class makes several CODEC lists and enables flexible negotiation with the VoIP router.

Create CODEC classes as follows:

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# voice class codec <i>tag</i>	Enters into the CODEC class configuration mode. The tag is an identifier of the CODEC class.
3	Router(config-class)# codec preference <i>value</i> <i>codec-type</i>	Enters into the configuration mode.
4	Router(config-class)# codec preference <i>value</i> <i>codec-type</i>	Enters into the configuration mode.
5	Router(config-class)# exit	Terminates the CODEC class configuration mode. (After configuration is completed, configuration becomes valid.)

Then, save CODEC classes created above in a certain VoIP peer by the following method.

Step	Command	Description
1	dial-peer voice <i>tag</i> VoIP	Enters into the dial-peer setup mode to set the VoIP peer.
2	voice-class <i>codec-class-tag</i> codec	<ul style="list-style-type: none"> Selects a CODEC for the voice considering the voice coder rate.

The following example shows how to create CODEC class 99 and store it in the VoIP peer 108.

```
voice class codec 99
    codec preference 1 g7231r63
    codec preference 2 g729
dial-peer voice 108 VoIP
    voice-class codec 99
```

5.3.2.4.2. Setting VAD in the VoIP Dial Peer

To disable transmission of silent packets for the selected VoIP, use the following commands in the global setup mode (start.)

Step	Command	Description
1	<code>dial-peer voice <i>number</i> VoIP</code>	Enters into the dial-peer setup mode to set the VoIP peer.
2	<code>vad</code>	<ul style="list-style-type: none">Disables transmission of silent packets. In other words, enables the VAD.

In the default status, the **VAD** is enabled. Normally, the default is the most suitable.

However, to connect to a network with a high bandwidth or to get the best voice quality, disable the **VAD**.

By disabling the VAD, users can get better voice quality but more bandwidth is required for the voice session.

To enable the VAD for the VoIP dial peer 108, set the VAD as follows:

```
dial-peer voice 108 VoIP
  destination-pattern 14085551234
  vad
  session target 10.0.0.8
```

5.3.3. One-Stage Dialing versus Two-Stage Dialing

The VoIP network configuration inter-works with the normal telephone network or PABX of the office in most of cases so multi-staged dialing is made. To decrease dialing stages, users shall add the telephone number of the termination side and the telephone number of the next stage to the called party number when setting a call in the termination side.

Consider that a subscriber connected to the voice port of the Router A attempts to make a call to the subscriber who is using PABX line #100 that is connected to the other VoIP router B.

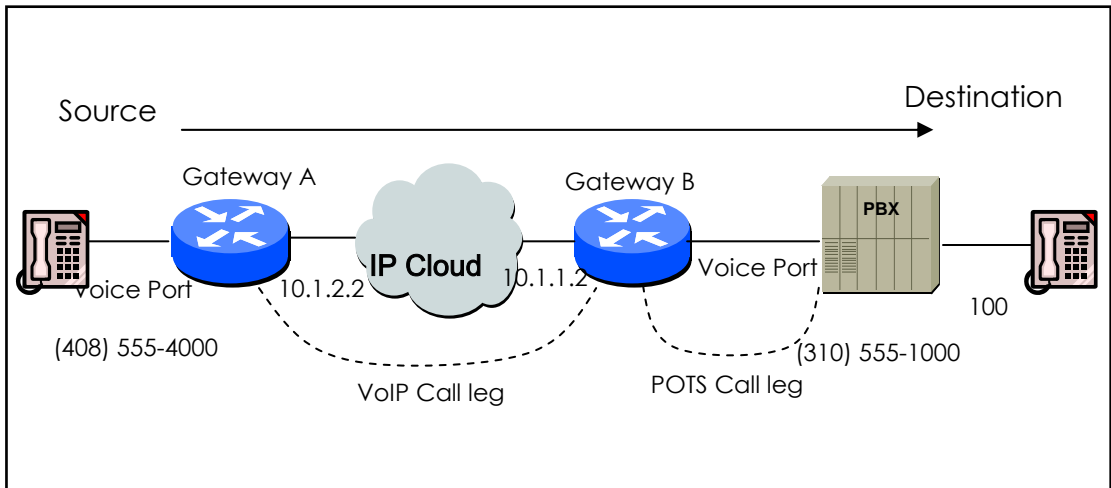


Figure 0.3 Two-Stage Dialing

Consider that the VoIP peer setup of the Router A is as follows:

```
dial-peer voice 555 VoIP
destination-pattern 310555....
```

In the above, as soon as the subscriber of Router A enters 3105551000, the outbound VoIP peer 555 will be decided and a call will be connected to Router B.

Consider that the POTS peer setup of Router B is as follows:

```
dial-peer voice 1000 pots
destination-pattern 3105551000
```

At this time, the originating subscriber hears the dial tone that the PABX sends and will enter 100.

To change two-stage dialing into one-stage dialing, users shall set the VoIP peer of Router A as follows:

```
dial-peer voice 555 VoIP
  destination-pattern 310555.....
```

In this kind of setup, the subscriber of Router A shall enter 3105551000100 to establish a call, and Router B sends called party number and other digits than fixed digit information except the wildcard of the destination pattern to the voice port when the outbound POTS peer is selected as 1000. In this case 100 is sent.

If the length of the inter-working number is not fixed, it is better to use "T" as the destination pattern.

Set the VoIP peer of Router A as follows:

```
dial-peer voice 555 VoIP
  destination-pattern 310555T
```

In this case, if the subscriber of Router A enters the termination digit (#) after entering 31055510001234567 or if the inter-digit is timed-out, a call will be connected to Router B and Router B will send 1234567 to the selected voice port.

5.3.4. Hunt Group-related Configuration

5.3.4.1. Basic Concept and Configuration

To select the outbound POTS that is going out of the router or to select the VoIP dial peer, the user shall compare the called party number of the inbound call and the destination pattern of the dial peer. At this time, more than one dial peers corresponding to the called party number belong to a hunt group, and dial peers of the hunt group attempts a call according to the given priorities.

In other words, the VoIP peer attempts a call to a dial peer of the hunt group when the call is failed due to network connection failure, gatekeeper failure, or gatekeeper rejection. And the POTS peer attempts a call to another dial peer of the hunt group when the call is failed due to corresponding voice port's being busy. Factors deciding priorities to attempt calls in the hunt group include the longest match, the explicit preference, the sequential, and the random.

The longest match decides priorities by the maximum digits matched between the origination number and the destination number of the dial peer. For example, consider that the origination number is 5683848, the destination number of dial peer 1 is 568T, the destination number of dial peer 2 is 568...., the destination number of dial peer 3 is 56838.., and the destination number of dial peer 4 is 5683848. Then, priorities of the dial peer by the longest match will be in order of dial peer 4 → dial peer 3 → dial peer 2 → dial peer 1.

In the explicit preference, the order set in the **preference** of the dial peer decides priorities. For example, consider that the preference of dial peer 1 is 3, the preference of dial peer 2 is 2, the preference of dial peer 3 is 1, and preference of dial peer 4 is 0. Then, priorities of the dial peer is in order of dial peer 4 → dial peer 3 → dial peer 2 → dial peer 1.

The random decides the dial peer within the hunt group randomly.

The sequential decides priorities according to the selection count. The less selected, the higher priority is given.

This priority algorithm operates using all this factors. For example, operation of dial-peer hunt 0 decides the first priorities according to the longest matching, checks the preference order within the same longest match priority, and then randomly selects the dial peer in the same preference order.

The first setup relating to the hunt group is to select the hunt algorithm.

Step	Command	Description
1	Router# configure	Enters into the global configuration mode.
2	Router(config)# dial-peer hunt <i>[0-7]</i>	Algorithm 0 ~ 7 are applied as follows: 0 – (default) longest match, explicit preference, random 1 - longest match, explicit preference, sequential 2 - explicit preference, longest match, random 3 - explicit preference, longest match, sequential 4 – sequential, longest match, explicit preference 5 - sequential, explicit preference, longest match 6 – random 7 - sequential

Users can also set priorities according to the **preference** or a huntstop in the corresponding peer according to the **huntstop**.

If a huntstop has been already set in a certain dial peer and if an outbound call going to the dial peer is failed, the call will be terminated without hunting another dial peer.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# dial-peer voice tag { pots VoIP }	Enters into the dial-peer configuration mode. The tag is the only identifier of the dial-peer of this system, and tag values range from 0 to 65535.
3	Router(config-dial-peer)# preference <i>number</i>	The value ranges from 0 to 9, and the lower the value is, the higher the priority is.

4	Router(config-dial-peer)# huntstop	Sets the huntstop in the corresponding dial peer.
---	---	---

5.3.4.2. Rerouting to the PSTN

With the hunt group explained before, PSTN rerouting through the FXO voice port can be made when connection with the VoIP network fails. The following figure shows PSTN rerouting.

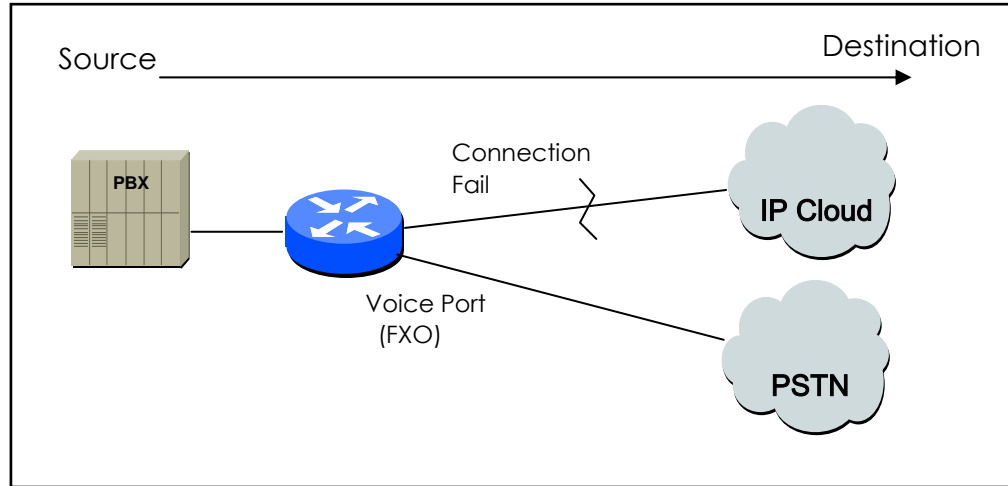


Figure 0.4 PSTN Rerouting

To make PSTN rerouting as shown in the above figure, set the dial peer as follows:

```
dial-peer voice 101 VoIP
 destination-pattern 472...
 session target 192.168.100.1
 preference 0
!
dial-peer voice 102 pots
 destination-pattern 472...
 prefix 472
 port 0
 preference 1
```

In the above example, VoIP peer 101 and POTS peer 102 exist in the same hunt group. Since the preference of the VoIP peer is low, the VoIP peer is selected first and used to attempt a call. However, if the VoIP peer fails, a call is

attempted through the POTS peer 102.

5.3.4.3. Call barring

Using the **huntstop** and the **shutdown** of the dial peer explained before, users can bar the outbound/inbound calls with certain patterns.

To bar calls of the outbound peer, define the pattern to bar in the destination pattern and set the shutdown and the huntstop. If necessary, set the preference and select all dial peers to bar first.

In the following example, VoIP peer 100 has been selected for all outbound calls. However, if the called party number starts with 526 or the called party number is 5441234, the call will not be processed any more.

```
dial-peer voice 100 VoIP
  destination-pattern T
  session-target ras
dial-peer voice 101 VoIP
  destination-pattern 526T
  session-target ras
  huntstop
  shutdown
dial-peer voice 102 VoIP
  destination-pattern 5441234
  session-target ras
  huntstop
  shutdown
```

To bar calls for the inbound VoIP peer, define the pattern to bar in the destination pattern and set the shutdown and the huntstop. If necessary, set the preference and select the dial peer to bar first.

In the above example, if the calling party number of the inbound call starts with 526 or is 5441234, the call will not be processed any more.

To bar the inbound VoIP call and allow the outbound call of the number starting with 538, use "**answer-address**" command as follows:

```
dial-peer voice 103 VoIP
  answer-address 538....
shutdown
```

5.3.5. Prefix and Forwarding Telephone Numbers

Forwarding numbers for the POTS peer has been explained already. When the number for the outbound POTS peer is forwarded, only digits except fixed digits of the destination-pattern of the outbound POTS peer are forwarded.

For example, if the destination-pattern is 444...., the fixed digit is 444. At this time, if the called party number of the inbound call is 444123456, only digits "123456" are forwarded to the voice port corresponding to the outbound POTS peer. (In case of an analog voice port, DTMF tones are outputted.)

If **prefix 99,,** is set in this outbound POTS peer, 99 is outputted first and 123456 is outputted in two seconds.

The above explains number forwarding operation for the default setup. For more precise operation of number forwarding, perform **forward-digit** setting in the POTS peer setup. The dial peer for which the **forward-digit** has been set does not check fixed digits of the destination-pattern and forwards the number according to the value set in the forward-digit.

The forward-digit setting can be made by **forward-digit from** and **forward-digit last**.

Forward-digit from "*" forwards numbers from the "*"th digit, and **forward-digit last "*"** forward only last "*" digits.

For example, if the called party number of the inbound call is 444123456 and "forward-digit from 4" is given, "123456" will be forwarded, and if the called party number is 444123456 and "forward-digit last 4" is given, "3456" will be forwarded.

5.3.6. Configuration Number Expansion

In most corporate environments, the telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Voice over IP can be configured to recognize extension numbers and expand them into their full E.164 dialed number by using two commands in tandem: **destination-pattern** and **num-exp**. Before you configure these two commands, it is helpful to map individual telephone extensions with their full E.164 dialed numbers. This task can be done easily by creating a number expansion table.

5.3.6.1. Number Expansion Table

In Figure 5.7, a small company wants to use Voice over IP to integrate its telephony network with its existing IP network. The destination pattern (or expanded telephone number) associated with Router 1 (located to the left of the IP cloud) are (408) 115-xxxx, (408) 116-xxxx, and (408) 117-xxxx, where xxxx identifies the individual dial peers by extension. The destination pattern (or expanded telephone number) associated with Router 2 (located to the right of the IP cloud) is (729) 555-xxxx.

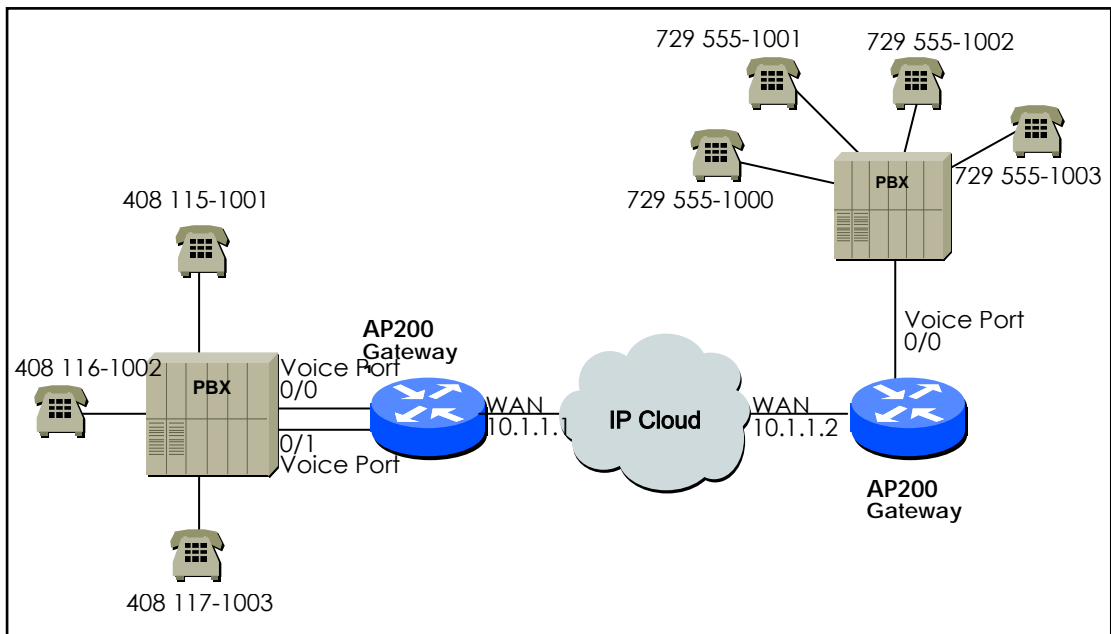


Figure 0.5 Sample VoIP Network

Sample Number Expansion Table for this scenario.

Extension	Destination Pattern	Num-exp Command Entry
5....	408115.....	num-exp 5 408115
6....	408116.....	num-exp 6 408116
7....	408117.....	num-exp 7 408117
1...	729555....	num-exp 2 729555

This information is used in configuration Gateway 1 및 Gateway 2.

5.3.6.2. Configuration Number Expansion

To define how to expand an extension number into a particular destination pattern, use the following command in global configuration mode:

Step	Command	Description
1	num-exp <i>extension-number</i> <i>extension-string</i>	Configure Number expansion.

You can verify the number expansion information by using the **show num-exp** command to verify that you have mapped the telephone numbers correctly.

5.3.7. Configuration Number Translation

5.3.7.1. Creating Translation Rules

Users can translate called/calling party number of the inbound/outbound calls in the AP200 Gateway. When translating called/calling party number of the inbound call, received called/calling party numbers are translated by the translation rule and used to select the outbound dial peer. When translating called/calling party numbers of the outbound call, called/calling party numbers that are used for originating a call are translated by the translation rule and calls are processed.

When changing private numbers into public numbers or vice versa or when extending numbers, number translation is used. Number translation provides more various conversions than number expansion. To translate numbers, a translation rule set shall be created first. Use "**translation-rule**" command on the global configuration mode to create a translation rule set.

Users can define more than one rules for the translation rule set using "**rule**" command on the translation-rule configuration mode. The following table shows how to define rules for the translation rule set.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# translation-rule tag	Enters into the translation rule configuration mode. The <i>tag</i> is an identifier of the translation rule set.
3	Router(translation-rule)# rule rule-tag <i>input-matched-pattern</i> <i>substituted-pattern</i>	The <i>rule-tag</i> is an identifier of the rule within the rule set. The rule-tag values range from 0 to 65535. The <i>input-matched-pattern</i> means the digit to be inputted for pattern-matching. 0 ~ 9, "#", "*", "[]", "." and T can be entered. The <i>substituted-pattern</i> is a pattern to be converted upon successful pattern matching. 0 ~ 9, "#", "*", "%", "." and T can be entered.

When one or more rules of a rule set match with the called/calling party number, the rule that most matches with the input-matched-pattern is selected.

The input-matched-pattern can perform range expression. (eg. [1-9]) Also, the wildcard (.) can be used to apply number of digits of the called/calling party number. If the input-matched-pattern is made of only (.) or (T) all called/calling-party-numbers will be translated.

The *substituted-pattern* converts the fixed digits (excluding the wildcard) of the *input-matched-pattern* into the string of the *substituted-pattern*. There are two forms of the substituted-pattern.

For example, if the substituted-pattern is composed only of IA5 characters (0 ~9, #, and *) the substituted-pattern will convert the fixed digits of the input-matched-pattern into the string of the substituted-pattern and add other digits than fixed digits of the called/calling party number at the end.

Or, if the substituted-pattern uses “%” form, each digit of the called/calling party number is replaced by “%xx” to make a number. At the time, % values range from %01 to %99 (from the 1st digit to the 99th digit of the called/calling party number.)

If the *substituted-pattern* is composed only of (.) or (T) the called/calling-party-number will be made of other digits than fixed digits of the input-matched-pattern.

In the following example, if 00181463701234 is entered into the translation rule set 100, the number will be translated into 81463701234. In this way 0313961234 is translated into 82313961234, and 5261234 is translated into 8225261234.

```
translation-rule 100
  rule 1 001..... .
  rule 2 0..... 82
```

rule 3 [1-9]..... 822%01%02%03%04%05%06%07%08

Created translation rules can be verified by “**show translation-rule**” command. For example, “show translation-rule 100” command will show rules of the translation rule set 100. To view the translation result, enter the number to test. To check the translation result of “1234” in the translation rule set 100, enter “show translation-rule 100 1234.” At this time, the result will be 1234.

5.3.7.2. Applying Translation Rules to the Inbound POTS Calls

To apply the translation rule set to all calls received in the voice port, make following setting.

Step	Command	Description
1	Router# configure	Enters into the global configuration mode.
2	Router(config)# voice-port <i>location</i>	Enters into the designated voice port configuration mode. The location is indicated by the slot-number/port-number.
3	Router(voice-port)# translate-incoming {called-number calling-number} <i>tag</i>	called-number: Applies the translation rule to the inbound called party number. calling-number: Applies the translation rule to the inbound calling party number. The <i>tag</i> refers to the rule set and tag values range from 0 to 65535.

If the translation rule is applied to the called party and if numbers are entered into the voice port in order, check if translation is made for each number entered. At this time, translation shall be made only once.

5.3.7.3. Applying Translation Rules to the Inbound VoIP Calls

To apply the translation rule set to all calls received from the network, make following setting.

Step	Command	Description
1	Router# configure	Enters into the global configuration mode.
2	Router(config)# voice service VoIP	Enters into the voice service VoIP setup mode.
3	Router(vservice-VoIP)# translate-VoIP-incoming {called-number calling-number} tag	called-number: Applies the translation rule to the inbound called party. calling-number: Applies the translation rule to the inbound calling party number. The <i>tag</i> refers to the rule set and tag values range from 0 to 65535.

5.3.7.4. Applying Translation Rules to the Outbound Calls

To apply the translation rule set to the outbound calls going to a certain dial peer (POTS peer or VoIP peer) make following setting.

Step	Command	Description
1	Router# configure	Enters into the global configuration mode.
2	Router(config)# dial-peer voice tag { pots VoIP }	Enters into the dial-peer configuration mode. The tag is the only identifier of the dial-peer of this system, and tag values include 0 ~ 65535.
3	Router(dial-peer-config)# translate-outgoing {called-number calling-number} tag	called-number: Applies the translation rule to the outbound called party number. calling-number: Applies the translation rule to the outbound calling party number. The <i>tag</i> refers to the rule set and tag values range from 0 to 65535.

5.3.8. call-diversion Command Setting

5.3.8.1. call-diversion

To enter the call diversion mode, use the **call-diversion** command in Global command mode. To delete the call diversion configuration, use the **no call-diversion** command.

call-diversion *tag*

no call-diversion *tag*

5.3.8.1.1. Syntax

Keyword / Argument	Explanation
tag	call-diversion identifier

5.3.8.1.2. Default Value

No default value.

5.3.8.1.3. Command Mode

Global Setting Mode

5.3.8.1.4. User Guideline

This command can be used to enter into call-diversion information setting mode. The generated call-diversion identifier can be referenced in dial peer part.

5.3.8.1.5. Usage Example

Following example shows the call-diversion 100 setting procedures

```
call-diversion 100
    cfb ipaddr 211.111.111.1
```

5.3.8.2. max-forward-hop

When call diversion is set and inbound call busy is occurred, VoIP gateway performs call forwarding to the other gateway. However, if call-forwarding loop is existed or cascaded VoIP gateway performs the call forwarding continuously, VoIP inbound call fall in deadlock. This command can be used to restrict maximum hop count to protect deadlock state.

max-forward-hop number

no max-forward-hop

5.3.8.2.1. Syntax

Keyword / Argument	Decription
number	Call-forwarding maximum number

5.3.8.2.2. Default Value

Default value is Four(4)

5.3.8.2.3. Command Mode

Dial-Peer Setting Mode

5.3.8.2.4. User Guideline

This command can used to assign the call forwarding maximum number in max-forward-hop information setting mode by VoIP peer side request..

5.3.8.2.5. Example

Following example shows **max-forward-hop** 100 setting procedures
max-forward-hop 10

5.3.9. Call Transfer Related Command Setting

Generally, PBX supports **call transfer** function when VoIP gateway provides the VoIP service via PBX. However, if analog phone is connected to VoIP gateway's FXS ports directly, AddPac VoIP gateway supports the **call transfer** function for PBX's **call transfer** function role.

AddPac VoIP gateway supports the **call transfer** service as a default function or activates the **call transfer** service using following commands

dial-peer call-transfer h

In above example, **call transfer** function is activated by hook-flash. If user wants to deactivate the **call transfer** function, **no dial-peer call-transfer** or **dial-peer call-transfer n** commands can be used.

To explain the **call transfer** function, we should know **called party**s that participate the call transfer scenario. Basically, **call transfer** scenario is extension of **call hold** scenario.

In Figure 5-3-9, in first step, Call AB is occurred between user A and user B, in second step, Call BC is occurred between user B and user C by user B hook-flash, in final step, Call AC is occurred between user A and user C by user B hook-on

STEP	Commands	Explanation
1	Router# configure	Enter into configuration setting mode
2	Router(config)# dial-peer call-transfer <h> <n>	Call-transfer Operating Mode h : hookflash n : no action

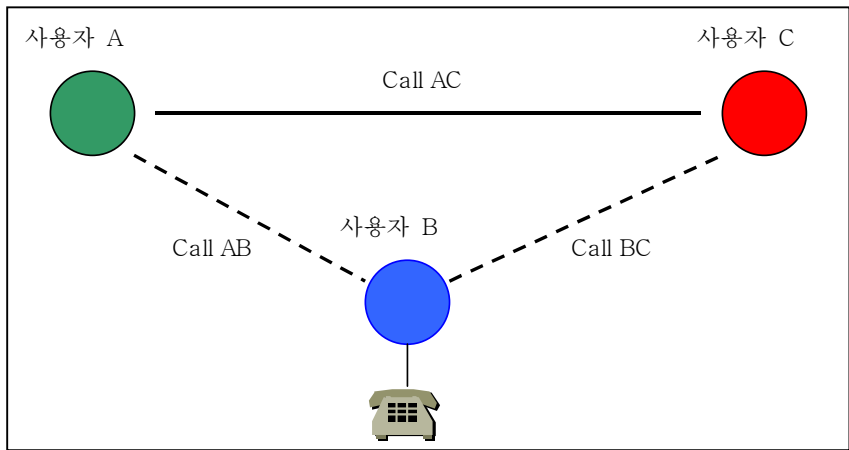


Figure 5.3.9 Call transfer scenario

There are two types of **call transfer**. The first type, confirmed method represents that user B hook-on occurs call transfer after conversion between user B and user C, non-confirmed call transfer represents that user B hook-on occurs call transfer at **ringing** state of user C after dialing.

AddPac VoIP gateway supports **confirmed and non-confirmed call transfers** all.

5.3.10. Call Pickup Related Command Setting

Generally, PBX supports **call pickup** function when VoIP gateway provides the VoIP service via PBX. However, if analog phone is connected to VoIP gateway's FXS ports directly, AddPac VoIP gateway supports the **call pickup** function for PBX's **call pickup** function role.

AddPac VoIP gateway supports the **call pickup** service as a default function or activates the **call pickup** service using following commands

dial-peer call-pickup #11

In above example, **call pickup** code #11, when a specific FXS port is ringing state in multi-port VoIP gateway, if call pickup code is pressed sequentially after hook off in the other FXS port in VoIP gateway, the other FXS port can receive ringing inbound VoIP call.

If user wants to deactivate the **call pickup** function, **no dial-peer call-pickup** command can be used.

STEP	Commands	Explanation
1	Router# configure	Enter into configuration setting mode.
2	Router(config)# dial-peer call-pickup <0-9 # *> + IA5 digits	Call-pickup operation mode

5.3.11. Inbound-pots-peer Related Command Setting

When network interfaces of VoIP gateway are normal and communication between VoIP gateway and gatekeeper is normal, all call conversion is occurred via VoIP network.

However, when network interface of VoIP gateway and communication with gatekeeper is abnormal, user does not use the VoIP service via Internet. In case, user can use PSTN service via PSTN back-up features. AddPac VoIP gateway supports the **inbound-pots-peer** for PSTN backup features.

AddPac VoIP gateway deactivates **inbound-pots-peer** service as a default function or activates the **inbound-pots-peer** service using following commands

inbound-pots-peer 0
no inbound-pots-peer

This command can be used to minimize the configuration complexity when translation rule is applied in 1:1 port mapping at PSTN backup configuration

STEP	Commands	Explanation
1	Router# configure	Enter into configuration setting mode.
2	Router(config)# dial-peer voice tag pots	dila-peer configuration mode tag is 0~65535 value as a unique identifier of dial peer in this system.
3	Router(config-dialpeer-pots-<tag>)# inbound-pots-peer <tag>	Inbound-pots-peer is applied. tag is 0~65535 value as a unique identifier of dial peer in this system.

5.3.12. PSTN Backup Related Command Setting

When network interfaces of VoIP gateway are normal and communication between VoIP gateway and gatekeeper is normal, all call conversion is occurred via VoIP network.

However, when network interface of VoIP gateway and communication with gatekeeper is abnormal, user does not use the VoIP service via Internet. In case, user can use PSTN service via PSTN back-up features. AddPac VoIP gateway supports the **busyout monitor** , **busy action** services for PSTN backup features.

5.3.12.1. Busyout Monitor

The VoIP gateway performs call transfer to PSTN when network interface of VoIP gateway and communication with gatekeeper is abnormal.

This Busyout monitor command can assign the PSTN backup condition and pre-determines the operation mode of VoIP gateway at PSTN backup.

STEP	Commands	Explanation
1	Router# configure	Enter into configuration setting mode.
2	Router(config)# voice service voip	Enter into Voice service voip mode
3	Router(config-vservice-voip)# busyout monitor {gatekeeper voip-interface}	Gatekeeper : busyout setting applied in gatekeeper. Voip-interface: busyout setting applied in voip-interface

5.3.12.2. Busyout Action

User can set a specific action such as busyout tone or port-down in a specific port when VoIP gateway is busyout. If busyout action is set as busyout tone and situation that meet this condition is occurred, PBX does not call try in this port because PBX recognizes as busy the state of Internal or external PBX line connected to gateway port.

In this time, if PBX HUNT function is operated and a call is transferred to external PSTN line via PBX when gateway port is busy, the perfect PSTN backup service is possible.

STEP	Commands	Explanation
1	Router# configure	Enter into configuration setting mode
2	Router(config)# voice-port slot-id/port-id	Enter into Voice-port configuration mode
3	Router(config-voice-ports-0/0)# busyout action {none tone port-down}	None : no action Tone : busy ton generation Port-down : port down

5.4. Configuration Voice Ports

5.4.1. Configuration Voice Ports on the AP200 Gateway

In general, voice port commands define the characteristics associated with a particular voice port signaling type. Under most circumstances, the default voice port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network.

5.4.2. Voice Ports Configuration Task List and Steps

5.4.2.1. Configuring FXS or FXO Voice Ports

Under most circumstances the default voice port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. The first two tasks are required; the third task is optional.

- 1) Identify the voice port and enter the voice-port configuration mode.
- 2) Configure the following mandatory voice-port parameters.
- 3) Configure one or more of the following optional voice-port parameters.
 - PLAR(Private Line Auto Ringdown) Connection Mode
 - Description
 - Ring Number
 - Input Gain
 - Output Gain

Step	Command	Description
1	configure	Enter global configuration mode.
2	voice-port <i>location</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	ring number <i>number</i>	(For FXO ports only) Specify the maximum number of rings to be detected before answering a call.
4	connection plar <i>string</i>	(Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The <i>string</i> value specifies the destination telephone number.
5	description <i>string</i>	(Optional) Attach descriptive text about this voice port connection.
6	<ul style="list-style-type: none"> ● input gain <i>value</i> 	Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -31 to 31.
7	output gain <i>value</i>	Specify (in decibels) the amount of gain at the transmit side of the interface.

5.4.2.2. Activating/Deactivating the Voice Ports

To activate a voice port, use the following commands in voice-port configuration mode:

Step	Command	Description
1	no shutdown	Activate the voice port

To cycle a voice port, use the following command in voice-port configuration mode

Step	Command	Description
1	voice-port <i>location</i>	Identify the voice port you want to activate and enter the voice-port configuration mode.
2	no shutdown	Activate the voice port.

5.5. FAX Application

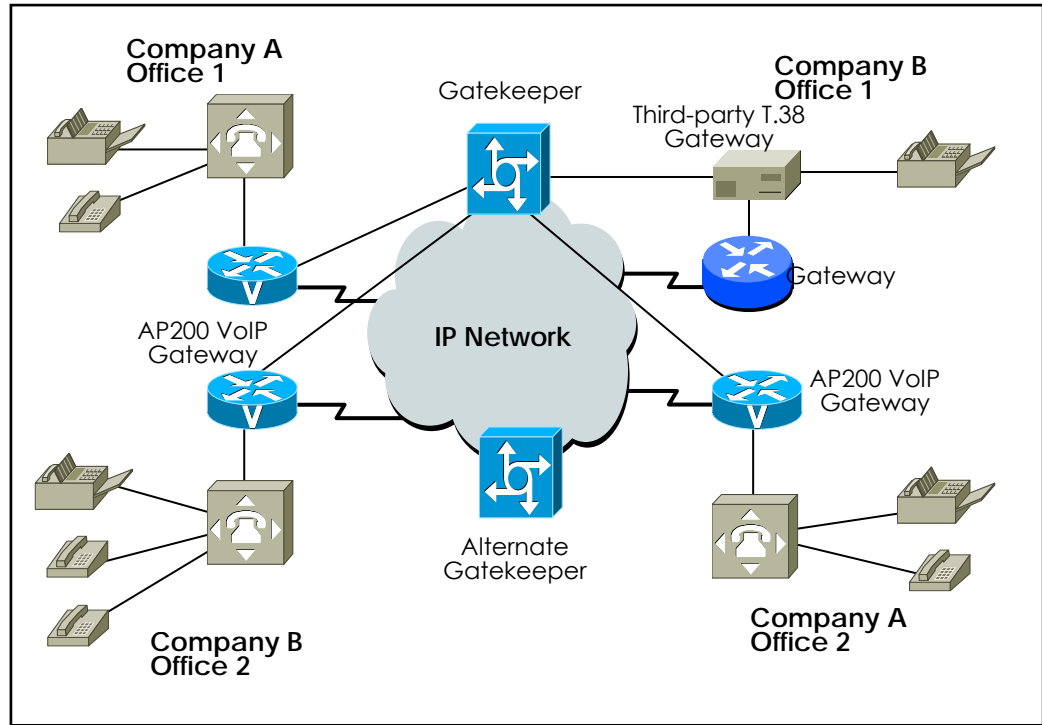
5.5.1. T.38 FAX Relay using VoIP H.323

The T.38 Fax Relay for VoIP H.323 feature provides standards-based fax relay protocol support on the most of VoIP router or Gateway including AP200.

Because the T.38 fax relay protocol is standards based, TN2520 can interoperate with third-party T.38-enabled gateways and gatekeepers in a mixed-vendor network when real-time fax relay capabilities are required.

Figure 5.9 shows an IP H.323 network with TN2520 routers and third-party gateways and gatekeepers using T.38 fax relay functionality. By using T.38 fax

relay, all gateways and gatekeepers in this network are able to send faxes to other remote offices or to the offices of another company on the IP network.



[Figure 5-9] IP Network for T.38 Fax Relay

For example, when a fax is sent from the originating gateway, a voice call is established. The terminating gateway detects the fax tone generated by the answering fax machine. The VoIP H.323 call stack then starts a T.38 mode request using H.245 procedures. If the opposite end of the call acknowledges the T.38 mode request, the initial audio channel is closed and a T.38 fax relay channel is opened. When the fax transmission is completed, the call reverts to voice mode.

5.5.2. Configuring T.38 FAX Relay for VoIP H.323

To configure T.38 Fax Relay for VoIP H.323 for all the connections of a gateway, which is required, use the following commands in the global configuration mode:

Step	Command	Description
1	Router(config)# voice service VoIP	Enters voice-service configuration mode.
2	Router(config-vservice-VoIP)# fax protocol {t38 [redundancy value] }	Specifies the global default fax protocol. . t38 : Enable T.38 Fax relay protocol . redundancy : (Optional) Configuration Redundant T.38 Fax packet . value : Redundancy Value. Range : 0 ~ 5, Default Value : 0 .
3	Router(config-vservice-VoIP)# fax rate {2400/ 4800 / 7200 / 9600 /12000 / 14400 / disable }	Selects the maximum fax transmission speed.
4	Router(config-vservice-VoIP)# exit	Exits voice-service configuration mode and returns to the global configuration mode.

If redundancy is set as 1 or higher, not 0, in the above setting, contents of T.38 packet will be duplicated as many times as the redundancy number and transmitted.

In this case, higher bandwidth will be required. Therefore, set redundancy as 1 or higher only when there is UDP packet loss in the network. In other cases, set redundancy as 0.

Set the maximum speed for the fax rate. However, the fax rate is decided by the transmission cable quality and the transmission speeds of the VoIP gateway and two fax machines on both sides. If the fax rate is set disabled, T.38 session will not be opened.

5.5.3. FAX Relay setting by Bypass

To set fax relay with G.711 PCM clean channel in addition of T.38 fax relay, use following commands in Global Configuration Mode. To do fax relay in this mode voice channel shall be opened with g711alaw or g711ulaw. So, codec and codec-class setting of dial-peer needs to support g711alaw or g711ulaw interface and counterpart needs to set in G.711 mode.

Step	Command	Description
1	Router(config)# voice service VoIP	Converts into Voice Service Setting Mode
2	Router(config-vservice-VoIP)# fax protocol bypass	<ul style="list-style-type: none">● Sets Global default FAX protocol
3	Router(config-vservice-VoIP)# exit	<ul style="list-style-type: none">● Exits from Voice-Service Setting Mode and return to Global setting Mode

5.6. Other VoIP Configuration

5.6.1. Setting H.323 Gateway

H.323 gateway inter-works with the gatekeeper and receives the Registration Admission and Security (RAS) service. The AP200 Gateway can set static IP address in the VoIP peer and operate without any gatekeeper. Also, the AP200 Gateway can dynamically bring the IP address (the other party's number) through the gatekeeper without setting any IP address.

For this, h323 ID of the gateway is necessary, and h323 ID is the only identifier in the gatekeeper. If VoIP IP address of the router is 211.123.1.2, the AP200 Gateway sets the default h323 ID as VoIP.211.123.1.1. Users can set H323 ID in the gateway setup mode using "**h323-id**" command.

The AP200 Gateway uses "**gkip**" command to designate gatekeepers. With "**gkip**" command, users can designate more than one gatekeeper and register them by priorities. There is only one gatekeeper that can be registered at the same time.

For the security between the gateway and the gatekeeper, users can set the secure token using "**security password**" command. However, if the password is enabled, the gateway will add Crypto Token to the message and send the message to the gatekeeper. Only when CryptoH323Token is set for the gatekeeper and cryptoEPPwdHash is supported, this security setup can be made. At this time, the password is given by the administrator off line.

When the user exits the gateway setup mode by "**exit**" or "**end**" gateways start to be registered in the gatekeeper.

To cancel registration of gateway in the gatekeeper, use "**no gateway**" command in the global setup mode.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# gateway	Enters into the gateway configuration mode, and registers the gateway in the gatekeeper.
3	Router(config-gateway)# gkip <i>gatekeeper-ip-address [port] [priority]</i>	Designates the gatekeeper ip address.
4	Router(config-gateway)# h323-id <i>h323-id</i>	Designates h323 id of the gateway.
4	Router(config-gateway)# security password <i>password</i>	Sets the H.235 security password.
5	Router(config-gateway)# exit	Exits configuration.

5.6.2. Configuring H323 Call Start Mode

In H.323 version2, logical channel negotiation procedures made by the fast start mode upon the start of the H.323 call is explained.

In the voice service VoIP setup mode, users can select the fast start procedures by using "**h323 call start**" command in the AP200 Gateway.

The fast start mode is the default, and H245 tunneling and the fast start is disabled when the slow start is set.

To find out the capability (T.38 fax, DTMF relay capability, etc.) of the other side in the fast start mode, users can perform H.245 procedures.

Step	Command	Description
1	Router# configure	Enters into the global configuration mode.
2	Router(config)# voice service VoIP	Enters into the voice service VoIP setup mode.
3	Router(vservice-VoIP)# h323 call start {fast slow}	Sets the fast mode or the slow start mode.

5.6.3. Configuring User Class

User-class setting is used to reject receiving calls from the unauthorized users when an origination call arrives in the FXO from the network. When a user attempts a call to the FXO port through the network while no user-class is set, the user will have to enter the extension code digit after hearing dial tones generated by the PBX if the FXO is connected to the internal line of the PBX. However, if the FXO is connected to the PSTN, the user will hear dial tones created by the PSTN switching machine and will have to enter the other side number of the PSTN.

If any user-class is set, the user who makes the first call will hear "beep" sound instead of the dial tone. If the user enters the **password** at this time and if the password is authorized, the user can enter as many numbers as max-digits defined in the user-class. ("Beep" sound may not occur due to the gateway on the origination side.) In this way, users can control internal calls, local calls, long-distance calls, and international calls by controlling "**max-digit**." Users can set more than one user-classes and set call limits for other user-classes.

To keep the security of calls incoming to the FXO port through the network, users can use this command and **security permit-FXO** in the AP2520. Since it is possible to directly attempt calls to the PSTN through this FXO port or indirectly attempt calls to the PSTN through the PBX internal line, unauthorized remote users can attempt calls as well.

To prevent unauthorized users' attempting calls, the security shall be kept. Two security systems that the AP200 Gateway provides have following advantages and disadvantages.

With "security permit-FXO" command, remote users does not need to enter the password so they can easily access the network. However, IP address of the VoIP peer on the other side shall be registered and the gatekeeper cannot be used at the same time. Also, it is not possible to bar calls of the registered peers by class.

With the "voice class user" users need to enter the password digit but stronger and multi-classed call barring is possible.

Step	Command	Description
1	Router# configure	Enters into the configuration mode.
2	Router(config)# voice class user <i>tag</i>	Enters into the user class configuration mode. The tag is a user class identifier.
3	Router(config-class)# password <i>digits</i>	Sets the password. At this time, digits include IA5 characters (0~9, #, and *) and the digit length is 4.
4	Router(config-class)# max-digits <i>value</i>	Sets the maximum number of digits when generating origination signals with the FXO. By adjusting the maximum number of digits, users can set PBX internal calls, local calls, long-distance calls and international calls.
5	Router(config-class)# exit	Terminates the user class configuration mode.

5.7. VoIP Configuration Command

5.7.1. VoIP-Related whole Command

clear	h323	call	all					
			tag <0-4294967295>					
	voice-port	<0-1>/<0-3>						
		all						
configure								
	dial-peer	hunt	<0-7>					
		ipaddr-prefix	#,*, n					
		terminator	#,*, n					
		voice	tag <0-65535>	pots				
					destination-pattern	string		
					forward-digits	from	<0-99>	
						last	<0-99>	
					huntstop			
					no	destination-pattern		
						forward-digits		
						huntstop		
						port		
						preference		
						prefix		
						register	e164	
						shutdown		
						translate-outgoing	called-number	
							calling-number	
					port	slot / port	<0-3>/<0-3>	
					preference		<0-9>	
					prefix		string	
					register		e164	
					shutdown			
					translate-outgoing	called-number	tag	<0-65535>
						calling-number	tag	<0-65535>
				VoIP				
					answer-address		string	
					codec		g711alaw g711ulaw g729 g7231r63 g7231r53	
					description		string	

					destination-pattern	string		
					dtmf-relay	h245-alphanumeric		
					huntstop			
					no	answer-address		
						codec		
						description		
						destination-pattern		
						dtmf-relay		
						huntstop		
						preference		
						session	target	
						shutdown		
						sid		
						translate-outgoing	called-number	
							calling-number	
						vad		
						voice-class	codec	
					preference	<0-9>		
					session	target	ip-addr	
							ras	
					shutdown			
					sid			
					translate-outgoing	called-number	tag <0-65535>	
						calling-number	tag <0-65535>	
					vad			
					voice-class	codec	tag <0-65535>	
	gateway							
		discovery						
		gkip	ip-addr	port<0-65536>	priority<0-254>			
					<cr>			
				<cr>				
		lightweight-irr						
		h323-id	string					
		no	discovery					
			gkip	ip-addr				
			lightweight-irr					
			public-ip					
			register					
			security	password				
		public-ip						
		register						
		security	password					
	no	dial-peer	hunt					

			ipaddr-prefix					
			terminator					
			voice	tag <0-65535>	pots			
					VoIP			
		gateway						
		num-exp	string					
		translation-rule	tag <0-65535>					
		voice	class	clear-down-tone	tag <0-1>			
				codec	tag <0-65535>			
				user	tag <0-10>			
		VoIP-interface						
	num-exp	string	string					
	translation-rule		tag <0-65535>					
			rule	tag <0-65535>	string	string		
			no	rule	tag <0-65535>			
	voice	class	clear-down-tone	tag <0-1>	low-num <300-1980>	high-num <0-300-1980>	on-num <0-10000>	off-num <0-10000>
			codec	tag <0-65535>				
					codec	preference	num <1-5>	
					no	codec	preference	num <1-5>
			user	tag <0-10>				
					password	digits <4 digits>		
					max-digits	num <0-100>		
					no	password		
						max-digits		
		service	VoIP					
				announcement				
				counter	cras	<1-5> default :3		
				default				
				fax	protocol	bypass		
						t38	redundancy	num <0-5>
							<cr>	
						inband-t38	redundancy	num <0-5>
							<cr>	
					rate	2400 4800 7200 9600 12000 14400 disable		
				h323	call			
						channel	early	
							late	
						response	alert	
							progress	
							none	
						start	fast	
							slow	

							preferred-slow	
				no	announcement			
					counter	cras		
					fax	protocol		
						rate		
					h323	call	channel	
							response	
							start	
					security	permit-FXO		
					timeout	t301		
						t303		
						tras		
						tttl		
						tidt		
						treg		
					translate-VoIP-incoming	called-number		
						calling-number		
				security	permit-FXO			
				timeout	t301	<5-600> default :180		
					t303	<5-60> default :8		
					tras	<2-30> default :6		
					tttl	<10-600> default :60		
					tidt	<1-600> default :10		
					treg	<10-600> default :30		
				translate-VoIP-incoming	called-number	tag <0-65535>		
					calling-number	tag <0-65535>		
	voice-port	slot/port						
				comfort-noise				
				connection	plar	string		
				description	string			
				echo-cencel				
				input	gain	num <-13 - 31>		
				no	comfort-noise			
					connection	plar		
					description	string		
					echo-cencel			
					input	gain		
					operation			
					output			
					ring	number		
					shutdown			
					signal			
					timing	dialout-delay		

					delay-duration			
					delay-start			
					wait-wink			
					wink-duration			
					wink-wait			
				translate-incoming	called-number			
					calling-number			
				type				
			operation	2-wire 4-wire				
			output	gain	num <-31 - 31>			
			ring	number	num <1-255>			
			shutdown					
			signal	delay-dial immediate wink-start				
			timing	dialout-delay	num <50-5000>			
				delay-duration	num <100-5000>			
				delay-start	num <20-2000>			
				wait-wink	num <100-5000>			
				wink-duration	num <30-5000>			
				wink-wait	num <100-5000>			
			translate-incoming	called-number	tag <0-65535>			
				calling-number	tag <0-65535>			
			type	1 2 3 5				
	VoIP-interface	interface	(default ether 0.0)					
show	call	active	all					
			summary					
		history	all	last	num <1-100>			
				<cr>				
		clear-down-tone						
	codec-classes	tag <0-65535>						
		<cr>						
	dialplan	number	string					
		port	slot / port					
	dial-peer	pots	tag <0-65535>					
			summary					
			<cr>					
		voice	tag <0-65535>					
			summary					
			<cr>					
		VoIP	tag <0-65535>					
			summary					
			<cr>					
	gateway							
	num-exp							
	translation-rule	tag <0-65535>	string					

			<cr>					
		<cr>						
	user-class							
	voice	port	slot/port					
			summary					
			<cr>					
	VoIP-interf ace							

5.7.2. Global Configuration Command

5.7.2.1. dial-peer hunt

To set the priorities for selecting the dial peer hunt, use “**dial-peer hunt**” command on the global setup mode.

To return to the default setting, use “**no**” command before this command.

dial-peer hunt *hunt-order-number*

no dial-peer hunt

5.7.2.1.1. Syntax

Keyword / Argument	Description
<i>hunt-order-number</i>	Priority algorithm 0 ~ 7 are applied as follows: 0 – (default) longest match, explicit preference, random 1 - longest match, explicit preference, sequential 2 - explicit preference, longest match, random 3 - explicit preference, longest match, sequential 4 – sequential, longest match, explicit preference 5 - sequential, explicit preference, longest match 6 – random 7 - sequential

5.7.2.1.2. Default Value

0 – longest match, explicit preference, random

5.7.2.1.3. Command Mode

Global Configuration Mode

5.7.2.1.4. Usage Guideline

To select the outbound POTS going out of the router or the VoIP dial peer, the called party number of the inbound call and the destination pattern of the dial

peer are compared. At this time, more than one dial peers corresponding to the called party number belong to a hunt group, and dial peers in the hunt group attempt calls according to given priorities.

In other words, the VoIP peer attempts a call to a dial peer of the hunt group when the call is failed due to network connection failure, gatekeeper failure, or gatekeeper rejection. And the POTS peer attempts a call to another dial peer of the hunt group when the call is failed due to corresponding voice port's being busy.

Factors deciding priorities to attempt calls in the hunt group include the longest match, the explicit preference, the sequential, and the random.

The longest match decides priorities by the maximum digits matched between the origination number and the destination number of the dial peer. For example, consider the origination number is 5683848, the destination number of dial peer 1 is 568T, the destination number of dial peer 2 is 568...., the destination number of dial peer 3 is 56838.., and the destination number of dial peer 4 is 5683848. Then, priorities of the dial peer by the longest match will be in order of dial peer 4 → dial peer 3 → dial peer 2 → dial peer 1.

In the explicit preference, the order set in the **preference** of the dial peer decides priorities. For example, consider the preference of dial peer 1 is 3, the preference of dial peer 2 is 2, the preference of dial peer 3 is 1, and preference of dial peer 4 is 0. Then, priorities of the dial peer is in order of dial peer 4 → dial peer 3 → dial peer 2 → dial peer 1.

The random decides the dial peer within the hunt group randomly.

The sequential decides priorities according to the selection count. The less selected, the higher priority is given.

This priority algorithm operates using all factors mentioned above. For example, operation of hunt-order-number 0 decides the first priorities according to the longest matching, checks the preference order within the same longest match priority, and then randomly selects the dial peer in the

same preference order.

5.7.2.1.5. Example

In the following example, "longest match" "explicit preference" and "sequential" algorithms are applied to the hunting group.

```
dial-peer hunt 1
```

5.7.2.2. dial-peer ipaddr-prefix

To designate a special character to be used as a ipaddr-prefix for making call by ip address, use the **dial-peer ipaddr-prefix** global configuration command. Use the **no** form of this command to default setting.

```
dial-peer ipaddr-prefix character
```

```
no dial-peer ipaddr-prefix character
```

5.7.2.2.1. Syntax

Keyword / Argument	Description
character	Designates ip address prefix. Valid numbers and characters are #, *

5.7.2.2.2. Default Value

Character (*)

5.7.2.2.3. Command Mode

Global Configuration Mode

5.7.2.2.4. Usage Guideline

In normal case, making a remote call is handled by number digits those are configured at dial-peer destination pattern and session target by VoIP router

operator. Although this pre-configured method is easy and secure, making a call by end-user with IP address of destination terminal is useful to call to ubiquitous VoIP terminal and gateway.

This prefix is used to discriminate normal call with number digits and direct call with IP address. To solve conflicting with terminator character, terminator character and ip address prefix character are changed automatically when configuring it.

5.7.2.2.5. Examples

The following example configures * as the special ip address prefix character:

```
configure
dial-peer ipaddr-prex *
```

The following example shows digit sequence for making a call by IP address. If IP address is 127.0.1.1 and called party number is 1234 then sequence of digits to making that call is:

```
* 10 * 0 * 0 * 1 * 1234 #
```

In above sequence, the first digit is **ipaddr-prefix** and the **ipaddr-prefix** character is used to discriminate IP address dot. And the last digit is **terminator** character.

When the destination terminal is simple VoIP phone like Microsoft **Netmeeting**, sequence of digits would be:

```
* 10 * 0 * 0 * 1 #
```

5.7.2.3. dial-peer terminator

To designate a special character to be used as a terminator for variable length dialed numbers, use the **dial-peer terminator** global configuration command. Use the **no** form of this command to default setting.

dial-peer terminator *character*
no dial-peer terminator *character*

5.7.2.3.1. Syntax

Keyword / Argument	Description
character	Designates the terminating character for a variable-length dialed number. Valid numbers and characters are #, *.

5.7.2.3.2. Default Value

Character (#)

5.7.2.3.3. Command Mode

Global Configuration Mode

5.7.2.3.4. Usage Guideline

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed-number, the system waits until the configured value for the **timeouts interdigits** command has expired before placing the call.

To avoid waiting until the interdigit timeout value has expired, you can designate a special character as a terminator---meaning that when you dial that character, the system no longer waits for any additional digits and places the call.

Use the **dial-peer terminator** global configuration command to designate a particular character as a terminator.

5.7.2.3.5. Example

The following example configures # as the special terminating character for variable-length dialed-numbers:

```
configure
dial-peer terminator #
```

5.7.2.4. dial-peer voice

To enter dial-peer configuration mode (and specify the method of voice-related encapsulation), use the **dial-peer voice** global configuration command.

```
dial-peer voice number {VoIP/pots}
```

5.7.2.4.1. Syntax

Keyword / Argument	Description
number	Digit(s) defining a particular dial peer. Valid entries are from 1 to 2147483647.
VoIP	Indicates that this is a VoIP peer using voice encapsulation on the POTS network.
pots	Indicates that this is a POTS peer using Voice over IP encapsulation on the IP backbone.

5.7.2.4.2. Default Value

No Default Value

5.7.2.4.3. Command Mode

Global Configuration Mode

5.7.2.4.4. Usage Guideline

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global

configuration mode.

5.7.2.4.5. Examples

The following example accesses dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
configure
dial-peer voice 10 pots
```

5.7.2.5. gateway

To enable the H.323 Voice over IP gateway, use the **gateway** command in global configuration mode. Use the **no** form of this command to unregister this gateway with the gatekeeper.

```
gateway
no gateway
```

5.7.2.5.1. Syntax

This command has no keywords or arguments.

5.7.2.5.2. Default Value

The gateway is unregistered.

5.7.2.5.3. Command Mode

Global Configuration Mode

5.7.2.5.4. Usage Guideline

Use the **gateway** command to enable H.323 VoIP gateway functionality. After you enable the gateway, it will attempt to discover a gatekeeper by using the H.323 RAS GRQ message. If you enter **no gateway**, the VoIP gateway will unregister with the gatekeeper via the H.323 RAS URQ message.

To register the dial peer or change the number by using “**load VoIP**” command and the script file in the gateway currently operating and registered in the gatekeeper, use “**no gateway**” command. Unregister the gateway from the gatekeeper, and then load configuration or state “**no gateway**” at the beginning of the script file. Otherwise, messages between the gateway and the gatekeeper may get congested to update changed information of the gateway.

5.7.2.5.5. Examples

The following example enables the gateway:

```
gateway
```

5.7.2.6. num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** global configuration command. Use the **no** form of this command to cancel the configured number expansion.

num-exp *extension-number expanded-number*

no num-exp *extension-number expanded-number*

5.7.2.6.1. Syntax

Keyword / Argument	Description
Extension-number	Digit(s) defining an extension number for a particular dial peer. (max length 55) Available character' s are 0-9#*%T.
Expanded-number	Digit(s) defining the expanded telephone number or destination pattern for the extension number listed. (max length 55) Available character' s are 0-9#*%T.

5.7.2.6.2. Default Value

No Default Value

5.7.2.6.3. Command Mode

Global Configuration Mode

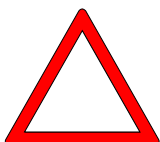
5.7.2.6.4. Usage Guideline

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, an extension number) into a particular destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing less than seven digits.

Number expansion is applied to the called party number of the inbound call. The called party number of the inbound call from the voice port or the network is translated by number expansion, and the dial peer is selected according to the translation result.

If more than one number expansions match with the called party number of the inbound call, the number expansion that matches most with the fixed pattern of the expansion-number will be selected.

Caution



It is recommended to take cautions when using number expansion with **translation-incoming** of the voice port or **translate-VoIP-incoming** of the network at the same time. If translation of the number is clear enough, it is not recommended to use two methods at the same time. If a user uses two methods, **translation-incoming** or **translate-VoIP-incoming** will be applied first and then number expansion will be applied.

Expansion-number can perform range expression. (eg. [1-9]) Also, the

wildcard (.) can be used to apply number of digits of the called party number. If extension-number is configured only with (.) or (T) number translation is applied to all called -party-number.

Expanded-number is to convert fixed digits (excluding the wildcard) of the extension-number into the string of the expanded-number. There are two forms of the expanded-number. See the following:

If the expansion-number is composed only of IA5 characters (0 ~ 9, # and *) fixed digits of the extension-number will be converted into the string of the expanded-number, and other digits than the fixed digits of the called-party-number will be added at the end.

Or, if the expansion-number uses "%" form, each digit of the extension-number will be replaced by "%xx" to make a number. At this time, % values range from %01 to %99 (from the 1st digit to the 99th digit of the called-party-number.)

If the expanded-number is composed of (.) or (T) only, the called -party-number is made of other digits than the fixed pattern of the extension-number.

5.7.2.6.5. Examples

The following example expands the extension number 55541 to be expanded to 1408555541: In the following example, the inbound called party number 5554123 is extended into 140855554123.

```
num-exp 55541 1408555541
```

In the following example, if the inbound called party number is 5551, the translation rule will not be applied. However, if the inbound called party number is 5551234, it will be translated into 14085551234.

```
num-exp 555.. 1408555
```

In the following example, if the inbound called party number is 1251234, it will be translated into 1408551234. And 3551234 will be translated into

14085551234.

```
num-exp [1-3][25]5.. 1408555
```

In the following example, if the inbound called party number is 5551234, it will be translated into 4441234.

```
num-exp 555.. 444%04%05%06%07%08%09%10%11%12
```

In the following example, if the inbound called party number is 55512, 5551234 or 555123456, it will be translated into 444.

```
num-exp 555.. 444%99
```

In the following example, if the inbound called party number is 5551234, it will be translated into 3334.

```
num-exp 555.. 111
```

```
num-exp 55512 222
```

```
num-exp 555[0-9][0-9][0-9] 333
```

In the following example, if the inbound called party number is 5551234, it will be translated into 1234.

```
num-exp 555 .
```

```
num-exp 555 T
```

In the following example, if the inbound called party number is 5551234, it will be translated into 95551234.

```
num-exp . 9
```

```
num-exp T 9
```

5.7.2.7. translation-rule

To enter into the translation rule setup mode, use "**translation-rule**" command in the global setup mode. To delete the translation rule that has been set, use "**no**" command before "**translation-rule**" command.

```
translation-rule tag
```

```
no translation-rule tag
```

5.7.2.7.1. Syntax

Keyword / Argument	Description
tag	An identifier to designate the translation rule set

5.7.2.7.2. Default Value

No Default Value

5.7.2.7.3. Command Mode

Global Configuration Mode

5.7.2.7.4. Usage Guideline

This command is to enter into a mode to set the translation rule of the called/calling party number of the inbound/outbound call.

5.7.2.7.5. Examples

The following example is to set the translation rule set 100.

```
translation-rule 100
    rule 0 2 822
```

5.7.2.8. voice-port

To enter the voice-port configuration mode, use the **voice-port** global configuration command.

voice-port *port_number*

5.7.2.8.1. Syntax

Keyword / Argument	Description
slots	Specifies the slot number in the router where the voice network module is installed. Valid entries are

	from 0 to 1, depending on the voice interface card you have installed.
port	Specifies the voice port. valid entries are 0 to 3.

5.7.2.8.2. Default Value

No Default Value

5.7.2.8.3. Command Mode

Global Configuration Mode

5.7.2.8.4. Usage Guideline

Use the **voice-port** configuration command to switch to the voice-port configuration mode from the global configuration mode. Use the exit command to exit the voice-port configuration mode and return to the global configuration mode.

For more information about the physical characteristics of your voice network module, or how to install it, refer to the installation documentation that came with your voice network module.

5.7.2.8.5. Examples

The following example accesses the voice-port configuration mode for slot 1 and port 3:

```
configure
voice-port 1/3
```

5.7.2.9. voice class clear-down-tone

To configure clear down tone on the FXO port, use the **voice class clear-down-tone** global configuration command.

`voice class clear-down-tone tag lowFreq highFreq onTime offTime`

`no voice class clear-down-tone tag`

5.7.2.9.1. Syntax

Keyword / Argument	Description
tag	Specifies the clear down tone. Valid entries are from 0 to 1.
lowFreq	Specifies the low frequency, of Hz, of the clear-down tone provided by the local switch or PBX. The frequency range is from 300Hz to 1980 Hz
highFreq	Specifies the high frequency, of Hz, of the clear-down tone provided by the local switch or PBX. The frequency range is from 300Hz to 1980 Hz. In case of single tone, value is 0.
onTime	Specifies the on-time duration of clear down tone.
offTime	Specifies the off-time duration of clear down tone. In case of long duration tone, value is 0.

5.7.2.9.2. Default Value

No Default Value

5.7.2.9.3. Command Mode

Global Configuration Mode

5.7.2.9.4. Usage Guideline

Call termination of FXO voice-port is established through detection of clear-down tone from PSTN or PBX, which is connected with FXO port. This clear-down tone(Ex: busy tone, fast busy tone) is different from each PSTN and PB. So, handle it as registering by this command.

This command is to enable to user to set detection tones in addition to clear-down-tone provided by the system. If the default detection tone provided by the system and shown in the clear-down-tone is enough,

additional setting is not necessary.

To operate newly added detection tones, reboot the system.

5.7.2.9.5. Examples

Following example shows setting of clear down tone whose dual-tone 350 Hz and 420 Hz is on time 250 msec, off time 250 msec.

```
configure
```

```
voice class clear-down-tone 0 350 420 250 250
```

5.7.2.10. voice class codec

To enter voice-class configuration mode and assign an identification tag number for a codec voice class, use the **voice class codec** command in global configuration mode. To delete a codec voice class, use the **no** form of this command.

```
voice class codec tag
```

```
no voice class codec tag
```

5.7.2.10.1. Syntax

Keyword / Argument	Description
tag	The unique number you assign to the voice class. The valid range is 1 to 10,000. Each tag number must be unique on the Gateway.

5.7.2.10.2. Default Value

No Default Value

5.7.2.10.3. Command Mode

Global Configuration Mode

5.7.2.10.4. Usage Guideline

This command only creates the voice class for codec selection preference and assigns an identification tag. Use the **codec preference** command to specify the parameters of the voice class, and use the **voice-class codec dial-peer** command to apply the voice class to a VoIP dial peer.

5.7.2.10.5. Examples

The following example shows how to enter voice-class configuration mode and assign a voice class tag number starting from global configuration mode:

```
voice class codec 10
```

After you enter voice-class configuration mode for codecs, use the **codec preference** command to specify the parameters of the voice class.

The following example creates preference list 99, which can be applied to any dial peer:

```
configure
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729
codec preference 4 g7231r63
codec preference 5 g7231r53
exit
```

5.7.2.11. voice class user

To enter user-class configuration mode and assign an identification tag number for a user class, use the **voice class user** command in global configuration mode. To delete a user voice class, use the **no** form of this command.

```
voice class user tag
```

```
no voice class user tag
```

5.7.2.11.1. Syntax

Keyword / Argument	Description
tag	The unique number you assign to the user class. Each tag number must be unique on the Gateway. Valid entries are from 0 to 10.

5.7.2.11.2. Default Value

No Default Value

5.7.2.11.3. Command Mode

Global Configuration Mode

5.7.2.11.4. Usage Guideline

Setting of user-class uses to deny call receivance from unpermitted users when originating call is received from network. Without user-class setting, anybody, who tries to call trough network to FXO port, can hear a dial-tone from PBX, which is connected with that port, and put desired extension number. (Or anybody, who tries to call trough network to FXO port, can hear a dial-tone from PSTN exchanger, which is connected with that port, and put desired PSTN number.)

If at least one of user-class is configured, initial caller shall listen beep sound instead of dial tone and after passing password, caller can input numbers as much as max-digits, which is defined in user-class. So, using max-digit number, it is possible adjusting of extension call, intown call, local area call, long distance call, and international call.

One or more user-class can be configured. So, different call limitations are capable for different user classes.

To keep the security of calls incoming to the FXO port through the network, users can use this command and "security permit-FXO" command in the AP200 Gateway. Since it is possible to directly attempt calls to the PSTN through this

FXO port or indirectly attempt calls to the PSTN through the PBX internal line, unauthorized remote users can attempt calls as well. To prevent unauthorized users' attempting calls, the security shall be kept. Two security systems that AP200 provides have following advantages and disadvantages.

With "security permit-FXO" command, remote users does not need to enter the password so they can easily access the network.

However, IP address of the VoIP peer on the other side shall be registered and the gatekeeper cannot be used at the same time. Also, it is impossible to bar calls of the registered peers by class.

With the "voice class user" users need to enter the password digit but stronger and multi-classed call barring is possible.

5.7.2.11.5. Examples

Following example shows generation of user class 1 and setting of user class mode.

```
voice class user 1
  password 1234
  max-digits 10
  exit
```

5.7.2.12. voice service

To specify the voice encapsulation type, use the **voice service command** in global configuration mode. To exit voice-service configuration mode, use the **exit** command.

voice service VoIP

5.7.2.12.1. Syntax

Keyword / Argument	Description
VoIP	Specifies Voice over IP (VoIP) parameters.

5.7.2.12.2. Default Value

No Default Value

5.7.2.12.3. Command Mode

Global Configuration Mode

5.7.2.12.4. Usage Guideline

Use the **voice service** command to switch to voice-service configuration mode from global configuration mode and to specify a voice encapsulation type. Use the **exit** command to exit the voice-service configuration mode and return to the global configuration mode.

5.7.2.12.5. Examples

The following example shows how to access voice-service configuration mode and specify VoIP parameters, beginning in global configuration mode:

```
voice service VoIP
```

5.7.2.13. VoIP-interface

To set an interface in which the VoIP is going to operate, use "**VoIP-interface**" command in the global setup mode.

To set the interface as the default, use "**no**" command before this command.

```
VoIP-interface interface-name
```

```
no VoIP-interface
```

5.7.2.13.1. Syntax

Keyword / Argument	Description
--------------------	-------------

<i>interface-name</i>	Designates the interface installed in the router. Interfaces include Ethernet 0.0, Ethernet 1.0, Serial 0 and so on.
-----------------------	--

5.7.2.13.2. Default Value

The default interface is Ethernet 0 .0.

5.7.2.13.3. Command Mode

Global Configuration Mode

5.7.2.13.4. Usage Guideline

With this command, users can designate the VoIP service in a certain interface.

VoIP service is provided using the IP address stored in the VoIP interface.

If no IP address is designated in the corresponding VoIP interface, VoIP-related setup or search is impossible.

5.7.2.13.5. Examples

The following example shows how to designate the VoIP service in the Ethernet 1.0 interface.

```
configure
  VoIP-interface ethernet 1 0
```

The following example shows how to designate the VoIP service in the serial 0 interface.

```
configure
  VoIP-interface serial 0
```


5.7.3. Voice Port Configuration Command

5.7.3.1. **comfort-noise**

To generate background noise to fill silent gaps during calls if voice activity detection (VAD) is activated, use the **comfort-noise** command in voice-port configuration mode. To provide silence when the remote party is not speaking and VAD is enabled at the remote end of the connection, use the **no** form of this command.

comfort-noise

no comfort-noise

5.7.3.1.1. **Syntax**

This command has no arguments or keywords.

5.7.3.1.2. **Default Value**

Enabled

5.7.3.1.3. **Command Mode**

Voice-port configuration Mode

5.7.3.1.4. **Usage Guideline**

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If the **comfort-noise** command is not enabled, and VAD is enabled at the remote end of the connection, the user will hear dead silence when the remote party is not speaking.

The configuration of the **comfort-noise** command only affects the silence generated at the local interface; it does not affect the use of VAD on either end of the connection or the silence generated at the remote end of the connection.

5.7.3.1.5. Example

The following example enables background noise on the AP200 Gateway:

```
voice-port 1
    no comfort-noise
```

5.7.3.2. connection

To specify a connection mode for a voice port, use the **connection** command in voice-port configuration mode. To disable the selected connection mode, use the **no** form of this command.

```
connection { plar } string
no connection { plar } string
```

5.7.3.2.1. Syntax

Keyword / Argument	Description
plar	Specifies a private line automatic ringdown (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off-hook, a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX.
string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

5.7.3.2.2. Default Value

No connection mode is specified.

5.7.3.2.3. Command Mode

Voice-port configuration Mode

5.7.3.2.4. Usage Guideline

Use the **connection** command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all calls coming in over this connection. The destination peer is determined by called number.

5.7.3.2.5. Examples

The following example selects PLAR as the connection mode on the AP200, with a destination telephone number of 555-9262: In this example, if the receiver connected to the voice port 1/0 is hooked off, a call will be automatically sent to 5559262.

```
voice-port 1
connection plar 5559262
```

5.7.3.3. description (voice port)

To include a description of what this voice port is connected to, use the **description** voice-port configuration command. Use the **no** form of this command to disable this feature.

```
description string
no description
```

5.7.3.3.1. Syntax

Keyword / Argument	Description
string	Description Character String on the Port. (max length 255)

5.7.3.3.2. **Default Value**

Enabled with a null string

5.7.3.3.3. **Command Mode**

Voice-port configuration Mode

5.7.3.3.4. **Usage Guideline**

Use the **description** command to include descriptive text about this voice-port connection. This information is displayed when you issue a **show** command and does not affect the operation of the interface in any way.

5.7.3.3.5. **Examples**

The following example identifies voice port 1/0/0 on the Cisco 3600 series as being connected to the Purchasing department:

```
voice-port 0
description marketing_dept
```

5.7.3.4. **echo-cancel**

To enable the cancellation of voice that is sent out the interface and is received back on the same interface, use the **echo-cancel** command in voice-port configuration mode. To disable echo cancellation, use the **no** form of this command.

```
echo-cancel
no echo-cancel
```

5.7.3.4.1. **Syntax**

This command has no arguments or keywords

5.7.3.4.2. **Default Value**

Enabled for all interface types.

5.7.3.4.3. **Command Mode**

Voice-port configuration Mode

5.7.3.4.4. **Usage Guideline**

The **echo-cancel** command enables cancellation of voice that is sent out the interface and is received back on the same interface; sound that is received back in this manner is perceived by the listener as an echo. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

The **echo-cancel** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a 4-wire ear and mouth (E&M) interface. The echo canceller should be disabled for this interface type.

5.7.3.4.5. **Examples**

The following example enables the echo cancellation feature on voice-port 3.

```
voice-port 3
  no echo-cancel
```

5.7.3.5. **high-dtmf-gain**

For high frequency dtmf tone gain of setting, use the **high-dtmf-gain** command. To disable the **high-dtmf-gain** configuration, use the **no high-dtmf-gain** command.

high-dtmf-gain *value*
no high-dtmf-gain *value*

5.7.3.5.1. Syntax

Keyword / Argument	Description
value	Dtmf gain value, decibel unit. Value range is integer between -31 and 3.

5.7.3.5.2. Default Vaule

-5

5.7.3.5.3. Command Mode

Voice-Port Setting Mode

5.7.3.5.4. Usage Guideline

Using **high-dtmf-gain** and **low-dtmf-gain** commands, customized **dtmf tone** generation is possible

5.7.3.5.5. Example

Following example shows the 3 decibel gain addition to current level in the receive port 4 of gateway.

```
port 4
high-dtmf-gain 3
```

5.7.3.6. input gain

To configure a specific input gain value, use the **input gain** voice-port configuration command. Use the **no** form of this command to disable the selected amount of inserted gain.

Input gain *value*
no Input gain *value*

5.7.3.6.1. Syntax

Keyword / Argument	Description
value	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable value is any integer from -31 to 31.

5.7.3.6.2. Default Value

0

5.7.3.6.3. Command Mode

Voice-port configuration Mode

5.7.3.6.4. Usage Guideline

A system-wide loss plan must be implemented using both **input gain** and **output gain** commands. Other equipment (including PBXs) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6dB attenuation between phones. Connections are implemented to provide -6dB of attenuation when the **input gain** and **output gain** commands are configured with the default value of 0.

You can't increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation

You can increase the gain of a signal coming in to the router. If the voice level is too low, you can increase the input gain.

5.7.3.6.5. Examples

The following example configures a 3-decibel gain to be inserted at the receiver side of the interface in the router.:

```
port 1/4
input gain 3
```

5.7.3.7. low-dtmf-gain

For high frequency dtmf tone gain of setting, use the **low-dtmf-gain** command. To disable the **low-dtmf-gain** configuration, use the **no low-dtmf-gain** command.

```
low-dtmf-gain value
no low-dtmf-gain value
```

5.7.3.7.1. Syntax

Keyword / Argument	Description
value	Dtmf gain value, decibel unit. Value range is integer between -31 and 3.

5.7.3.7.2. Default Vaule

-5

5.7.3.7.3. Command Mode

Voice Port Setting Mode

5.7.3.7.4. Usage Guideline

Using **high-dtmf-gain** and **low-dtmf-gain** commands, customized **dtmf tone** generation is possible

5.7.3.7.5. Example

Following example shows the -10 decibel gain addition to current level in the receive port 4 of gateway.

```
port 4
low-dtmf-gain -10
```

5.7.3.8. output gain

To configure a specific output gain value, use the **output gain** voice-port configuration command. Use the **no** form of this command to disable the selected output gain value.

```
output gain value
no output gain value
```

5.7.3.8.1. Syntax

Keyword / Argument	Description
value	The amount of gain in decibels at the transmit side of the interface. Acceptable value is any integer from -31 to 31. The default value for FXO, FXS, and E&M ports is 0..

5.7.3.8.2. Default Value

0

5.7.3.8.3. Command Mode

Voice-port configuration Mode

5.7.3.8.4. Usage Guideline

A system-wide loss plan must be implemented using both **input gain** and **output gain** commands. Other equipment (including PBXs) in the system must be taken into account when creating a loss plan.

This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6 dB attenuation between phones.

Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output gain** commands are configured with the default value of 0.

You can't increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

5.7.3.8.5. Examples

The following example on the router configures a 6-decibel gain to be inserted at the transmit side of the interface:

```
port 1/4
output gain 3
```

5.7.3.9. polarity-inverse

To enable polarity-inverse function for a FXS voice port', use the **polarity-inverse** voice-port configuration command. Use the **no** form of this command to disable polarity-inverse.

```
polarity-inverse
no polarity-inverse
```

5.7.3.9.1. Syntax

This command has no arguments or keywords

5.7.3.9.2. Default Value

```
disable .
```

5.7.3.9.3. Command Mode

Voice-port configuration Mode

5.7.3.9.4. Usage Guideline

General case, there os no needs to enable this functions.

5.7.3.9.5. Examples

To enable the polarity-inverse function to voice-port 1/0 :

```
voice-port 1/0
    polarity-inverse
```

5.7.3.10. ring number

To specify the number of rings for a specified FXO voice port, use the **ring number** voice-port configuration command. Use the **no** form of this command to restore the default value.

ring number *number*

no ring number *number*

5.7.3.10.1. Syntax

Keyword / Argument	Description
number	Number of rings detected before answering the call. Valid entries are numbers from 1 to 255. The default is 1.

5.7.3.10.2. Default Value

One Ring

5.7.3.10.3. Command Mode

Voice-port configuration Mode

5.7.3.10.4. Usage Guideline

Use the **ring number** command to set the maximum number of rings to be detected before answering a call over an FXO voice port. Use the **no** form of this command to reset the default value, which is one ring. Normally, this command should be set to the default so that incoming calls are answered quickly. If you have other equipment available on the line to answer incoming calls, you might want to set the value higher to give the equipment sufficient time to respond. In that case, the FXO interface would answer if the equipment on line did not answer the incoming call in the configured number of rings. This command is not applicable to FXS or E&M interfaces because they do not receive ringing to receive a call.

5.7.3.10.5. Examples

The following example on the router sets five rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1/1
ring number 5
```

5.7.3.11. shutdown (voice-port)

To take the voice ports for a specific voice interface card offline, use the **shutdown** voice-port configuration command. Use the **no** form of this command to put the ports back in service.

```
shutdown
no shutdown
```

5.7.3.11.1. Syntax

This command has no arguments or keywords

5.7.3.11.2. Default Value

No shutdown

5.7.3.11.3. Command Mode

Voice-port configuration Mode

5.7.3.11.4. Usage Guideline

When you enter the **shutdown** command, all ports on the voice interface card are disabled. When you enter the **no shutdown** command, all ports on the voice interface card are enabled. A telephone connected to an interface will hear dead silence when a port is shut down.

5.7.3.11.5. Examples

The following example takes voice port 1/3 on the router offline:

```
configure
voice-port 1/3
shutdown
```

5.7.3.12. translate-incoming

To apply the translation rule to the inbound POTS call coming to the corresponding voice port, use this command. To stop applying the translation rule, use “**no**” command before “**translate-incoming**” command.

```
translate-incoming { called-number | calling-number } tag
no translate-incoming { called-number | calling-number }
```

5.7.3.12.1. Syntax

Keyword / Argument	Description
called-number	Applies the translation rule to the inbound called party number.
calling-number	Applies the translation rule to the inbound calling

	party number.
<i>tag</i>	Refers to the rule set. Tag values range from 0 to 65535.

5.7.3.12.2. Default Value

No translation rule is applied.

5.7.3.12.3. Command Mode

Voice-port configuration Mode

5.7.3.12.4. Usage Guideline

This command uses the number translation rules that have been set by "translation-rule" command for the inbound call of the corresponding voice-port.

If the translation rule is applied to the called party number and if numeric data is entered into the voice port in order, check if translation is made for every number entered. At this time, translation shall be made only once.

5.7.3.12.5. Examples

In the following example, translation rule set 10 is created and applied to the calling party number of the voice port 1/1.

Therefore, if the calling party number of the inbound call is 93450, it is translated into 9563450.

```
translation-rule 10
    rule 0 9 956
    rule 1 8 878
voice-port 1
    translate-incoming calling-number 10
```

5.7.4. Dial Peer Commands

5.7.4.1. answer-address

To find the VoIP dial peer for the VoIP inbound call incoming to the network as using the calling party number of the inbound call, use “**answer-address**” command in the dial-peer setup mode. To disable the number that has been set, use “**no**” command before “**answer-address**” command.

answer-address *string*

no answer-address

5.7.4.1.1. Syntax

Keyword / Argument	Description
String	<p>Number string defined in the E.164 or private telephone number plan. Numeric data (0 to 9) “#” and “*” can be used.</p> <ul style="list-style-type: none"> ● (*) & (#): These characters can be found on the standard button-type telephone. They cannot be located at the first place of the string. (For example, *650 is invalid.) ● (.): The period means a value that can match with any numeric entered. In other words, the period cannot be placed at the first place of the string. (For example, .650 is invalid.) ● ([]): Brackets are to indicate the range. The Range is the character sequence within the bracket and only numeric data (0 to 9) can be used for the range. This is similar to the regular expression rule.

5.7.4.1.2. Default Value

The default is null string.

5.7.4.1.3. Command Mode

Dial-peer setup mode (VoIP dial peer)

5.7.4.1.4. Usage Guideline

This command is applied to the VoIP dial peer of the AP200 VoIP Gateway.

The “**answer-address**” command is used to find the VoIP dial peer for the VoIP inbound call incoming to the network.

The VoIP dial peer for the VoIP inbound call from the network is selected as follows:

Firstly, the VoIP dial peer that has the **session target** matching with the IP address of the inbound call is searched.

Secondly, if the corresponding peer is not found, the VoIP dial peer that has **answer-address** matching with the calling party number of the inbound call will be searched.

Lastly, if no peer is found, the VoIP dial peer matching with the **destination-pattern** of the inbound call will be searched.

5.7.4.1.5. Examples

In the following example, if the calling party number of the inbound VoIP call is “5263848” VoIP peer 10 will be selected.

```
dial-peer voice 10 VoIP
  answer-address 526....
```

5.7.4.2. codec

To specify the voice coder rate of speech for a dial peer, use the **codec** dial-peer configuration command. Use the **no** form of this command to reset the default value.

```
codec {g711alaw / g711ulaw / g729r8 / g7231r63 / g7231r53 }
no codec
```

5.7.4.2.1. Syntax

Keyword / Argument	Description
G711alaw	G.711 A-Law 64Kbps Codec
G711ulaw	G.711 u-Law 64Kbps Codec
G729	G.729 8Kbps Codec
G7231r63	G.723.1 6300 bps. This is the default CODEC for AP200 Gateway.

G7231r53	G.723.1 5.3Kbps Codec
----------	-----------------------

5.7.4.2.2. Default Value

G.723.1 6.3Kbps

5.7.4.2.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.2.4. Usage Guideline

Use the **codec** command to define a specific voice coder rate of speech for a dial peer.

For toll quality, use **G.711.alaw** or **G.711.ulaw**. These values provide high-quality voice transmission but use a significant amount of bandwidth. For almost toll quality (and a significant savings in bandwidth), use the **G.729** value.

If **codec** values for the VoIP peers of a connection do not match, the call will fail.

5.7.4.2.5. Examples

The following example configures a voice coder rate that provides toll quality and uses a relatively high amount of bandwidth:

```
dial-peer voice 10 VoIP
  codec g711alaw
```

5.7.4.3. description (dial-peer)

To include a description of what this VoIP dial peer is connected to, use the **description** dial-peer configuration command. Use the **no** form of this command to disable this feature.

description *string*
no description

5.7.4.3.1. Syntax

Keyword / Argument	Description
string	Character String for Dial-Peer. (max length 255)

5.7.4.3.2. Default Value

Enabled with a null string

5.7.4.3.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.3.4. Usage Guideline

Use the **description** command to include descriptive text about this dial-peer connection.

This information is displayed when you issue a **show** command and does not affect the operation of the interface in any way.

5.7.4.3.5. Examples

The following example identifies dial peer 10 of the Seoul office:

```
dial-peer voice 10 VoIP
  description Seoul_office
```

5.7.4.4. destination-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer, use the **destination-pattern** dial-peer configuration command. Use the **no** form of this command to disable the

configured prefix or telephone number.

destination-pattern *string* [T]

no destination-pattern

5.7.4.4.1. Syntax

Keyword / Argument	Description
String	<p>Series of digits that specify the E.164 or private dialing plan telephone number. (max length 55) Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:</p> <ul style="list-style-type: none"> ● The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. These characters cannot be used as leading characters in a string (for example, *650). ● Period (.), which matches any entered digit (this character is used as a wildcard). The period cannot be used as a leading character in a string (for example, .650). ● Brackets ([]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. This is similar to a regular expression rule.
T	(Optional) Control character indicating that the destination-pattern value is a variable length dial string

5.7.4.4.2. Default Value

enabled with a null string.

5.7.4.4.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.4.4. Usage Guideline

This command is applicable to both VoIP and POTS dial peers on all platforms.

Use the **destination-pattern** command to define the E.164 telephone number

for this dial peer.

This pattern is used to match dialed digits to a dial peer. The dial peer is then used to complete the call. When a router receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the voice-telephony peer. The router then strips out the left-justified numbers corresponding to the destination pattern. If you have configured a prefix, the prefix is appended to the front of the remaining numbers, creating a dial string, which the router then dials. If all numbers in the destination pattern are stripped-out, the user receives a dial tone.

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. Use the optional control character **T** to indicate that a particular **destination-pattern** value is a variable-length dial-string. In this case, the system does not match the dialed numbers until the interdigit timeout value has expired.

5.7.4.4.5. Examples

The following example configures the E.164 telephone number, "555-7922," for a dial peer:

```
dial-peer voice 10 pots
  destination-pattern 5557922
```

The following example shows configuration of a destination pattern in which the digit numbers range between 5553409 and 5559499:

```
dial-peer voice 3 VoIP
  destination-pattern 555[3-9]4[0-9]9
```

The following example shows configuration of a destination pattern in which the digit numbers range between 5551439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice 4 VoIP
  destination-pattern 555[13579]439
```

5.7.4.5. dtmf-relay

To specify how an H.323 gateway relays dual tone multi-frequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** command in dial-peer configuration mode. Use the **no** form of this command to remove all signaling options and to send the DTMF tones as part of the audio stream.

dtmf relay { h245-alphanumeric }

no dtmf relay

5.7.4.5.1. Syntax

Keyword / Argument	Description
h245-alphanumeric	(Optional) Forwards DTMF tones by using the H.245 "alphanumeric" User Input Indication method. Supports tones 0-9, *, #, and A-D.

5.7.4.5.2. Default Value

No default behavior or values.

5.7.4.5.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.5.4. Usage Guideline

DTMF is the tone generated when you press a digit on a touch-tone phone. This tone is compressed at one end of a call; when the tone is decompressed at the other end, it can become distorted, depending on the codec used. The DTMF relay feature transports DTMF tones generated after call establishment out of band using a standard H.323 out-of-band method.

The **dtmf-relay** command determines the outgoing format of relayed DTMF tones. The gateway automatically accepts all formats.

The principal advantage of the **dtmf-relay** command is that it sends DTMF tones with greater fidelity than is possible in-band for most low-bandwidth codecs, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated DTMF-based systems, such as voice-mail, menu-based ACD systems, and automated banking systems.

5.7.4.5.5. Examples

The following example configures DTMF relay with the h245-alphanumeric option when sending DTMF tones to dial-peer 103:

```
dial-peer voice 103 VoIP
    dtmf-relay h245-alphanumeric
```

The next example configures the gateway to send DTMF in-band (the default) when sending DTMF tones to dial-peer 103:

```
dial-peer voice 103 VoIP
    no dtmf-relay
```

5.7.4.6. forward-digits

To set the number of the forwarding digit of the outbound POTS as a random number not by using the default method, use "**forward-digits**" command in the dial-peer setup mode. To forward digits that do not match with the destination pattern by the default method, add "**no**" command.

```
forward-digits { from | last } number
no forward-digits
```

5.7.4.6.1. Syntax

Keyword / Argument	Description
from	Forwards the called party number from the designated digit.
last	Forwards last digits of the called party number as designated.
<i>number</i>	Number of digits to forward. The value ranges from 0 to 100. If the designated value is higher

	than the maximum digits that can be forwarded, only the digits that can be forwarded at maximum are forwarded.
--	--

5.7.4.6.2. Default Value

In the default, if the called party number matches with the destination pattern of the outbound POTS peer, only digits that are not matched will be forwarded.

5.7.4.6.3. Command Mode

Dial-Peer Configuration Mode (POTS peer)

5.7.4.6.4. Usage Guideline

This command is a dial-peer setup mode command and applied only to the POTS peer.

This command is used to define number of digits when relaying last digits of the called party number of an inbound call as the called party number of the outbound call. The default is no forward-digit, and in the default status, number forwarding is made.

5.7.4.6.5. Examples

If POTS peer 10 is decided for the outbound, call and if the called party number of the inbound call is 100123456789, number 123456789 will be forwarded as set in the default. This is because "**forward-digit**" has not been set.

```
dial-peer voice 10 pots
  destination-pattern 100...
```

If "**forward-digit from**" is added to the above setting, only "456789" (from the 7th digit to the last digit) will be forwarded.

```
forward-digit from 7
```

In the following example, all digits (100123456789) will be forwarded.

forward-digit from 1

In the following example, no digit will be forwarded.

forward-digit from 99

If “**forward-digit last**” is added as below, only last four digits “6789” will be forwarded.

forward-digit last 4

In the following example, no digit is forwarded.

forward-digit last 0

In the following digit, all digits “100123456789” are forwarded.

forward-digit last 99

5.7.4.7. huntstop

To stop dial peer hunting in the hunting group, use “**huntstop**” command in the dial-peer setup mode. To use default setting, use “**no**” command before “**huntstop**” command.

huntstop

no huntstop

5.7.4.7.1. Syntax

This command has no arguments or keywords.

5.7.4.7.2. Default Value

Hunting can be made in the default state.

5.7.4.7.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.7.4. Usage Guideline

If an outbound dial peer is selected for an inbound call and if more than one dial peers are selected, a hunting group will be made.

If a **huntstop** has been set already in a certain dial peer and if the outbound call to that dial peer fails, a call will be terminated without hunting other dial peers.

5.7.4.7.5. Examples

The following example is to perform huntstop in VoIP peer 110.

```
dial-peer voice 110 VoIP
    huntstop
```

5.7.4.8. port

To associate a dial peer with a specific voice port, use the **port** dial-peer configuration command. Use the **no** form of this command to cancel this association.

```
port slot/port
no port
```

5.7.4.8.1. Syntax

Keyword / Argument	Description
Slot	Slot number where the voice interface card is installed. Valid entries are 1 or 0.
port	Port number for voice port number of voice interface module. Valid entries are 1 to 7.

5.7.4.8.2. Default Value

No port is configured.

5.7.4.8.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.8.4. Usage Guideline

This command is applicable only to POTS peers .

Use the **port** configuration command to associate the designated voice port with the selected dial peer.

This command is used for calls incoming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

5.7.4.8.5. Examples

The following example associates a dial peer with voice port:

```
dial-peer voice 10 pots
port 1/1
```

5.7.4.9. preference

To clearly designate priorities within the hunt group for a certain dial-peer, use “**preference**” command in the dial-peer setup mode. To use default priorities, use “**no**” command before “**preference**” command.

```
preference value
no preference
```

5.7.4.9.1. Syntax

Keyword / Argument	Description
<i>value</i>	Values range from 0 to 9, and the lower the value is, the higher the priority is.

5.7.4.9.2. Default Value

The default is 0 and has the highest priority.

5.7.4.9.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.9.4. Usage Guideline

Setting priorities within the hunt group by the preference enables users to adjust the priority of a certain dial peer.

5.7.4.9.5. Examples

Let's assume that there is a dial peer as follows:

```
dial-peer voice 10 pots
  destination-pattern 5551234
  preference 3
```

```
dial-peer voice 11 pots
  destination-pattern 555....
  preference 0
```

If the called party number of the inbound call is 5551234, all dial peers will be selected. However, if the preference is prior in selecting the hunt algorithm by "dial-peer hunt" command, dial peer 11 will be selected first.

5.7.4.10. prefix

To specify the prefix of the dialed digits for this dial peer, use the **prefix** dial-peer configuration command. Use the **no** form of this command to disable this feature.

```
prefix string
```

```
no prefix
```


5.7.4.10.1. Syntax

Keyword / Argument	Description
string	Integers representing the prefix of the telephone number associated with the specified dial peer. Valid numbers are 0 through 9, and a comma (,). Use a comma to include a pause in the prefix. (max length 55)

5.7.4.10.2. Default Value

Null string.

5.7.4.10.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.10.4. Usage Guideline

Use the **prefix** command to specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

5.7.4.10.5. Examples

The following example specifies a prefix of "9" and then a pause:

```
dial-peer voice 10 pots
prefix 9,
```

5.7.4.11. register

To configure a gateway to register or deregister a fully qualified POTS dial-peer

E.164 address with a gatekeeper, use the **register e164** command in dial-peer configuration mode. To deregister an E.164 address, use the **no** form of this command.

register e164

no register e164

5.7.4.11.1. Syntax

This command has no keywords or arguments

5.7.4.11.2. Default Value

No default value

5.7.4.11.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.11.4. Usage Guideline

Use this command to register the E.164 address of an analog telephone line attached to a Foreign Exchange Station (FXS) port on a router. The gateway automatically registers fully qualified E164 addresses. Use the **no register e164** command to deregister an address. Use the **register e164** command to register a deregistered address.

Before you automatically or manually register an E.164 address with a gatekeeper, you must create a dial peer (using the **dial-peer** command), assign an FXS port to the peer (using the **port** command), and assign an E.164 address (using the **destination-pattern** command).

The E.164 address must be a fully qualified address. For example, 5551212, and 4085551212 are fully qualified addresses; 408555.... is not a fully qualified address. E.164 addresses are registered only for active interfaces—those that are not shut down

If an FXS port or its interface is shut down, the corresponding E.164 address is deregistered.

5.7.4.11.5. Examples

The following command sequence places the gateway in dial-peer configuration mode, assigns an E.164 address to the interface, and registers that address with the gatekeeper:

```
dial-peer voice 110 pots
port 1
destination-pattern 5551212
register e164
```

The following commands deregister an address with the gatekeeper:

```
dial-peer voice 110 pots
no register e164
```

5.7.4.12. session target

To specify a network-specific address for a specified dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

```
session target destination-address
no session target
```

5.7.4.12.1. Syntax

Keyword / Argument	Description
destination-address	IP address of the dial peer.

5.7.4.12.2. Default Value

No Default Value

5.7.4.12.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.12.4. Usage Guideline

Use the **session-target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

5.7.4.12.5. Examples

The following example configures a session-target IP address "211.238.1.1".

```
dial-peer voice 10 VoIP
session-target 211.238.1.1
```

5.7.4.13. shutdown (Dial-Peer)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** dial-peer configuration command. Use the **no** form of this command to change the administrative state of this dial peer from down to up.

```
shutdown
no shutdown
```

5.7.4.13.1. Syntax

This command has no arguments or keywords.

5.7.4.13.2. Default Value

No shutdown

5.7.4.13.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.13.4. Usage Guideline

When a dial peer is shut down, you cannot initiate calls to that peer. This command is applicable to both VoIP and POTS peers.

5.7.4.13.5. Examples

The following example changes the administrative state of voice telephony dial peer 10 to down:

```
configure
dial-peer voice 10 pots
shutdown
```

5.7.4.14. sid

When SID Packet transmission function is enable during silence processing under VAD function activation for call processing of specific dial-peer, use **sid** command in dial-peer configuration mode. If you want to set disable mode, add **no** command in front of **sid** command.

```
sid
no sid
```

5.7.4.14.1. Syntax

Keyword / Argument	Description
There is no any specific keyword and argument in this command.	

5.7.4.14.2. Default Value

The default value is enable mode status.

5.7.4.14.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.14.4. Usage Guideline

If the VAD function is enable, the silence traffic is not transmitted via VoIP network environments, tin this time, only it is possible to aware to hear voice packet. But during silence processing, intermittently SID packet is transmitted. If you don't want this SID packet transmission function for reason of matching problem with interoperability or unnecessary comfort noise generation, do disable command for this function non activation mode.

5.7.4.14.5. Examples

The following command example is disable mode of SID packet transmission function.

```
dial-peer voice 10 VoIP
no sid
```

5.7.4.15. translate-outgoing

To apply the translation rule to the outbound call of the corresponding dial peer, use this command. To stop applying the translation rule, add "no" command.

```
translate-outgoing { called-number | calling-number } tag
no translate-outgoing { called-number | calling-number }
```

5.7.4.15.1. Syntax

Keyword / Argument	Description
called-number	Applies the translation rule to the outbound called party number.
calling-number	Applies the translation rule to the outbound calling party number.
<i>tag</i>	Refers the rule set. Tag values range from 0 to 65535.

5.7.4.15.2. Default Value

No translation rule is applied.

5.7.4.15.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.15.4. Usage Guideline

This command is applied to the POTS peer and the VoIP peer. Use “**Translation-rule**” command for the outbound call of the corresponding dial peer, and apply the number translation rule that has been set before.

5.7.4.15.5. Examples

In the following example, translation rule set 10 is created and it is applied to the calling party number of dial-peer 200. Therefore, if the calling party number of the outbound call is 93450, it will be translated into 9563450.

```
translation-rule 10
    rule 0 9 956
    rule 1 8 878
dial-peer voice 200 VoIP
    translate-outgoing calling-number 10
```

5.7.4.16. vad

To enable voice activity detection (VAD) for the calls using this dial peer, use the **vad** dial-peer configuration command. Use the **no** form of this command to disable VAD.

```
vad
no vad
```

5.7.4.16.1. Syntax

This command has no arguments or keywords.

5.7.4.16.2. Default Value

Enable

5.7.4.16.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.16.4. Usage Guideline

Use the **vad** command to enable voice activity detection. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

If you use the **no** form of this command, VAD is disabled and voice data is continuously transmitted to the IP backbone.

5.7.4.16.5. Examples

The following example enables VAD:

```
dial-peer voice 10 VoIP
vad
```

5.7.4.17. voice-class codec

To assign a previously configured codec selection preference list (codec voice class) to a VoIP dial peer, enter the **voice-class codec command** in dial-peer configuration mode. To remove the codec preference assignment from the dial peer, use the **no** form of this command.

```
voice-class codec tag
no voice-class codec tag
```

5.7.4.17.1. Syntax

Keyword / Argument	Description
tag	The unique number assigned to the voice class. The valid range for this tag is 1 to 10,000. The <i>tag</i> number maps to the tag number created using the voice class codec global configuration command.

5.7.4.17.2. Default Value

Dial peers have no codec voice class assigned.

5.7.4.17.3. Command Mode

Dial-Peer Configuration Mode

5.7.4.17.4. Usage Guideline

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

5.7.4.17.5. Examples

The following example shows how to assign a previously configured codec voice class to a dial peer:

```
dial-peer voice 100 VoIP
voice-class codec 10
```

5.7.5. Gateway, Voice Service, Voice Class and Rule Configuration Command

5.7.5.1. announcement

To use following command for voice announcement function, use this command. To make disable this functionality, use the **no** form of this command.

announcement

no announcement

5.7.5.1.1. Syntax

Keyword / Argument	Description
There is no any keyword and argument in this command.	

5.7.5.1.2. Default Value

Default value is disable status mode.

5.7.5.1.3. Command Mode

Voice Service VoIP Configuration Mode

5.7.5.1.4. Usage Guideline

If voice announcement function is applied to voice configuration with abnormal finishing of call processing, inputting password for FXO voice ports, and PSTN rerouting, AP200 VoIP gateway could process voice announcement during PSTN rerouting. The kind of voice announcements is different from each APOS versions. Other AP VoIP gateway may be not supports this function.

5.7.5.1.5. Examples

The following command example is voice announcement function for enable mode.

```
voice service VoIP
  announcement
```

5.7.5.2. codec preference

To specify a list of preferred codecs to use on a dial peer, use the **codec preference** command in voice-class configuration mode. To make disable this functionality, use the **no** form of this command.

```
codec preference value codec_type
no codec preference value codec_type
```

5.7.5.2.1. Syntax

Keyword / Argument	Description
value	Specifies the order of preference, with 1 being the most preferred and 5 being the least preferred.
Codec_type	Specifies the codec preferred. . G711alaw : G.711 A-Law 64Kbps Codec . G711ulaw : G.711 u-Law 64Kbps Codec . G729 : G.729 8Kbps Codec . G7231r63 : G.723.1 6.3Kbps Codec.. . G7231r53 : G.723.1 5.3Kbps Codec

5.7.5.2.2. Default Value

No default behavior or values.

5.7.5.2.3. Command Mode

Voice-class Configuration Mode

5.7.5.2.4. Usage Guideline

The gateway at opposite ends of the WAN may have to negotiate the codec selection for the network dial peers. The codec preference command specifies the order of preference for selecting a negotiated codec for the connection.

5.7.5.2.5. Examples

The following example creates codec preference list 99 and applies it to dial peer 1919:

```
voice class codec 99
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g7231r63
codec preference 4 g729
end
dial-peer voice 1919 VoIP
voice-class codec 99
```

5.7.5.3. counter

To set VoIP-related counter parameter value, use “**counter**” command in the voice service setup mode. To convert this setup into the default state, use “**no**” command before this command.

```
counter { cras } value
no counter { cras }
```

5.7.5.3.1. Syntax

Keyword / Argument	Description
cras <i>value</i>	RAS message retransmission counter with the gatekeeper. The value ranges from 1 to 5, and the default is 3.

5.7.5.3.2. Default Value

See the above..

5.7.5.3.3. Command Mode

Voice-Service Configuration Mode

5.7.5.3.4. Usage Guideline

This command is to partially set the global voice-service for the VoIP service. The **cras** counter is to retransmit message if no reply is received during **timeout tras** after sending RAS message – GRQ, RRQ, ARQ and DRQ – to the gatekeeper.

5.7.5.3.5. Examples

In the following message, the RAS message is tried two times.

```
voice service VoIP
  counter cras 2
```

5.7.5.4. discovery

This command is applicable to message transmission function for GRQ(Gatekeeper Request). Use the **no** form of this command to set disable mode.

```
discovery
no discovery
```

5.7.5.4.1. Syntax

Keyword / Argument	Description
There is no any keyword and argument in this command set.	

5.7.5.4.2. Default Value

The default status is enable mode setting.

5.7.5.4.3. Command Mode

Gateway Configuration Mode

5.7.5.4.4. Usage Guideline

When the VoIP gateway is registered gatekeeper, if this function is activated, after sending GRQ, and then receiving GCF, and finally sending RRQ. However, if this is not activated, directly send RRG.

5.7.5.4.5. Examples

The following command example is to perform disable discovery command.

```
gateway
no discovery
```

5.7.5.5. fax protocol

To specify the global default fax protocol for all the Voice over IP (VoIP) dial peers, use the **fax protocol** command in voice-service configuration mode. To return to the default fax protocol, use the **no** form of this command.

```
fax protocol { t38 [redundancy value] | bypass | inband-t38 [redundancy
value]}
no fax protocol
```

5.7.5.5.1. Syntax

Keyword / Argument	Description
t38	ITU-T T.38 standard fax protocol.
Inband-t38	A conversion of ITU-T T.38 standard fax protocol sending T.38 information on RTP payload.

bypass	Fax protocol on clean voice channel (i.e., G.711)
redundancy	(Optional) redundancy for the T.38 fax protocol. The <i>value</i> can be from 0 to 5. The default is 0..
value	The <i>value</i> can be from 0 to 5. The default is 0.

5.7.5.5.2. Default Value

T.38 fax protocol

5.7.5.5.3. Command Mode

Voice-Service Configuration Mode

5.7.5.5.4. Usage Guideline

Use the **fax protocol t38** command to configure T.38 Fax Relay for VoIP. The **t38** keyword enables the T.38 Fax Relay protocol.

Optional parameters **redundancy** is used to send redundant T.38 fax packets.

5.7.5.5.5. Examples

The following example shows T.38 fax protocol for VoIP, beginning in global configuration mode:

```
VoIP service VoIP
fax protocol t38
```

5.7.5.6. fax rate

To establish the rate at which a fax is sent to the specified dial peer, use the **fax rate** command in dial-peer configuration mode.

To reset the dial peer for voice calls, use the **no** form of the command.

```
fax rate { 2400 | 4800 | 7200 | 9600 | 12000 | 14400 | disable }
no fax rate
```

5.7.5.6.1. Syntax

Keyword / Argument	Description
2400	Specifies a fax transmission speed of 2400 bps.
4800	Specifies a fax transmission speed of 4800 bps.
7200	Specifies a fax transmission speed of 7200 bps.
9600	Specifies a fax transmission speed of 9600 bps.
12000	Specifies a fax transmission speed of 12,000 bits per second (bps).
14400	Specifies a fax transmission speed of 14,400 bps.
disable	Disables Fax Relay transmission capability.

5.7.5.6.2. Default Value

14400 bps

5.7.5.6.3. Command Mode

Voice-Service Configuration Mode

5.7.5.6.4. Usage Guideline

Use the **fax rate** command to specify the fax transmission rate to all dial peers.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher transmission speed values (14,400 bps) provide a faster transmission speed but monopolize a significantly large portion of the available bandwidth. The lower transmission speed values (2400 bps) provide a slower transmission speed and use a relatively smaller portion of the available bandwidth.

This command is meaningful only under T.38 Fax Relay. If the fax rate is disabled, T.38 fax relay does not operate. The fax relay made by the bypass mode performs no operation in the gateway (as it does nothing for the fax communication in the PSTN network) setting the rate does not mean anything.

Although T.38 is set as 14400 bps by this command, if two fax machines on both sides operate at 9600, actual fax rate will be 9600 bps.

5.7.5.6.5. Examples

The following example shows a fax rate transmission speed of 9600 bps for faxes sent using a dial peer:

```
voice service VoIP
fax rate 9600
```

5.7.5.7. force-h245address-at-setup

When H.245 no tunneling call is started, this command can be used to include H.245 address information element in SETUP message. To enable this feature, use **force-h245address-at-setup** command. To disable this feature, use **no force-h245address-at-setup** command. Default setting value is **enable**

```
force-h245address-at-setup
no force-h245address-at-setup
```

5.7.5.7.1. Syntax

Keyword / Argument	Description
No argument and keyword in this command	

5.7.5.7.2. Default value

enable

5.7.5.7.3. Command Mode

Voice Service Setting Mode

5.7.5.7.4. Usage Guideline

For inter-operability with peer side gateway, VoIP user can use this command.

5.7.5.7.5. Example

Following example shows command setting so that SETUP message does include the h245Address information element.

```
voice service voip
    no force-h245address-at-setup
```

5.7.5.8. force-starth245

When a call is established by no tunneling configuration, startH245 procedure for TCP connection used in H.245 is enabled by this command.

To enable this feature, use **force-h245address-at-setup** command. To disable this feature, use **no force-h245address-at-setup** command. Default setting value is **enable** mode.

```
force-starth245
no force-starth245
```

5.7.5.8.1. Syntax

Keyword / Argument	Description
	No argument and keyword in this command.

5.7.5.8.2. Default Value

disable

5.7.5.8.3. Command Mode

Voice Service Setting Mode

5.7.5.8.4. Usage Guideline

This force-starth245 command can be used for inter-operability with Peer VoIP gateway. When a call is established by No tunneling configuration, perform

starth245 call procedure if TCP connection for H.245 procedure is not OK before CONNECT message.

5.7.5.8.5. Example

Following example shows that force-starth245 command activates the starth245 VoIP signaling procedure.

```
voice service voip
    force-starth245
```

5.7.5.9. h323 call start

To force the H.323 Version 2 gateway to use Fast Connect or Slow Connect procedures for all H.323 calls, use the **h323 call start** command in voice-service configuration mode. To restore the default condition, use the **no** form of this command.

```
h323 call start { fast | slow | preferred-slow }
no h323 call start
```

5.7.5.9.1. Syntax

Keyword / Argument	Description
fast	Gateway uses H.323 Version 2 (Fast Connect) procedures
slow	Gateway uses H.323 Version 1 (Slow Connect) procedures.
preferred-slow	If you configure this option, at the time to make a call, Gateway use slow start (normal start) procedure. At the time to receive a call, Gateway use slow start (normal start) or fast start procedure depends on calling party's configuration.

5.7.5.9.2. Default Value

The default is **fast**.

5.7.5.9.3. Command Mode

Voice-Service Configuration Mode

5.7.5.9.4. Usage Guideline

This **h323 call start** command is configured as part of the global voice-service configuration for VoIP services. It does not take effect unless the **call start system** voice-class configuration command is configured in the VoIP dial peer.

5.7.5.9.5. Examples

The following example selects Slow Connect procedures for the gateway:

```
voice service VoIP
h323 call start slow
```

5.7.5.10. inband-ringback-tone

The calling party user hear the ring-back tone indicating a called party's ringing. The called party send inband ring-back tone to calling party via RTP channel. To disable inband ring-back tone, use **no inband-ringback-tone** command.

```
inband-ringback-tone
no inband-ringback-tone
```

5.7.5.10.1. Syntax

Keyword / Argument	Description
No Keyword / Argument in this command	

5.7.5.10.2. Default Value

Default is disable

5.7.5.10.3. Command Mode

Voice-Service setting mode

5.7.5.10.4. Usage Guideline

The calling party user hear the ring-back tone indicating a called party's ringing. The ring- back tone is generated from calling party side or called party side. The called party send ring-back tone to calling party using two methods. The first method is in-band audio packet that is transmitted to receive side via RTP channel by fast start procedure. The second method is ALERT message tone received from called party.

5.7.5.10.5. Examples

Following example show inband ring-back tone setting procedure

```
voice service voip
    inband-ringback-tone
```

5.7.5.11. local-ringback-tone

The calling party user hear the ring-back tone indicating a called party's ringing. This command is used for hearing ring-back tone generated in local calling party not received from called party. To disable local ring-back tone, use **no local-ringback-tone** command.

```
local-ringback-tone [ early ]
no local-ringback-tone
```


5.7.5.11.1. Syntax

Keyword / Argument	Description
early	The calling party generate the ring-back tone after SETUP message transmission
Alert	The calling party generate the ring-back tone after ALERT message receiving
<cr>	The calling party generate the ring-back tone after whether inband tone or ALERT message is received.

5.7.5.11.2. Default Value

Default is enable

5.7.5.11.3. Command Mode

Voice-Service Setting Mode

5.7.5.11.4. Usage Guideline

The calling party user hear the ring-back tone indicating a called party's ringing. The ring- back tone is generated from calling party side or called party side. The called party send ring-back tone to calling party using two methods. The first method is in-band audio packet that is transmited to receive side via RTP channel by fast start procedure. The second method is ALERT message tone received from called party.

At first, this VoIP gateway's calling party hear the ring-back tone using ALERT message(default setting), and then, if inband ring-back tone is received from called party, calling party can hear inband ring-back tone. However, if called party send a CALL PROCEEDING/ ALERT/ PROCESS message that include a progress indicate information element indicating "inband announcement available", calling party does not generate local ring-back tone.

Therefore, we recommend the **local-ringback-tone** configuration setting. If user want to generate virtual ring-back tone immediately after SETUP message

transmission, use the **local-ringback-tone early** command. And, if user want to hear the inband announcement or tone, use **no local-ringback-tone** command.

5.7.5.11.5. Examples

Following example show virtual **local ring-back tone** configuration.

```
voice service voip
    local-ringback-tone early
```

5.7.5.12. minimize-voip-ports

This command can be used for minimizing the TCP/UDP port number used in VoIP gateway system. If user want to disable this command, use the **no minimize-voip-ports** command.

```
minimize-voip-ports
no minimize-voip-ports
```

5.7.5.12.1. Syntax

Keyword / Argument	Description
No Argument or Keyword in this command	

5.7.5.12.2. Default value

Default is disable.

5.7.5.12.3. Command Mode

Voice Service setting mode

5.7.5.12.4. Usage Guideline

The VoIP system use TCP/UDP ports for signaling, RAS and RTP. In default mode, TCP/UDP port selects one(1) port number from wide range TCP/UDP port number pool. The **show gateway** command show the TCP/UDP port range. However, under a specific application such as firewall, complex NAT/PAT environment and fixed **source port** depend on gatekeeper, this command restricts the port number range of TCP/UDP port for operation. The **show gateway** command show the fixed TCP/UDP port or reduced TCP/UDP port number pool.

5.7.5.12.5. Examples

Following show **minimize-voip-ports** command example

```
voice service voip
    minimize-voip-ports
```

5.7.5.13. max-frame

To set the maximum length of audio frame in RTP packet, this command will be used. If user want to disable this feature, use the **no max-frame { }** command.

```
max-frame { g7231 | g729 } value
no max-frame { g7231 | g729 }
```

5.7.5.13.1. Syntax

Keyword / Argument	Description
g7231	Frame length setting in G.7231 codec
g729	Frame length setting in G.729 codec
<i>value</i>	Frame length, Value is 1~8

5.7.5.13.2. Default value

In case of G.7231, default value is 1. This means that audio frame is generated and transmitting RTP packet at every 30ms interval.

Default value is 2 in case of G.729. This means that audio frame is generated at 10ms interval and transmitting RTP packet at 20ms interval.

5.7.5.13.3. Command Mode

Voice-Service setting mode

5.7.5.13.4. Usage Guideline

Generally, if bandwidth is low or congestion is occurred in internet, increase the max-frame length. At first step, increase the max-frame length as 2 in G.7231, 4 in G.729. If this does not work well, increase the max-frame length as 3 in G.7231, 6~8 in G.729. But, if max-frame is too large, this feature have transmission delay problem. In cable network that have poor upstream network environment, this command with qos-control command will enhance the voice quality.

5.7.5.13.5. Examples

Qos-control and **max-frame** setting examples for cable modem network.

```
interface ethernet 0 0
    qos-control 128 100
voice service voip
    max-frame g729 4
    max-frame g7231 2
```

5.7.5.14. gkip

To specify the gatekeeper associated with a proxy and control how the

gatekeeper is discovered, use the **gkip** command in gateway configuration mode.

gkip *ip-addr* [*port*] [*priority*]

no gkip *ip-addr*

5.7.5.14.1. Syntax

Keyword / Argument	Description
<i>ip-addr</i>	The gatekeeper discovery message will be unicast to this address. and, optionally, the UDP port specified.(Default Port Value is 1719)
<i>port</i>	Optionally, the UDP port of gatekeeper specified.(Default Port Value is 1719)
<i>priority</i>	Optionally designates priorities of several alternate gatekeepers. The priority value ranges from 0 to 254, and the lower the value is, the higher the priority is. The default priority is 128.

5.7.5.14.2. Default Value

No gatekeeper is configured for the proxy.

5.7.5.14.3. Command Mode

Gateway Configuration Mode

5.7.5.14.4. Usage Guideline

The VoIP gateway is registered in the gatekeeper that is the Registration Admission and Status (RAS) server and receives the charging service. The AP200 VoIP Gateway can designate maximum ten gatekeepers in a gateway using this command. To view the list of the gatekeepers, use "**show gateway**" command. If more than one gateway is designated, gateways try to register themselves in the gatekeeper using the GRQ message according to their priorities. There is only one gatekeeper that can be registered at the same time. When a gateway receives the re-registration failure message or does not receive any message, the next gateway tries to register itself in the gatekeeper.

Users can designate the gatekeeper using this command or using the Alternate GK list included in the message that the currently registered gatekeeper sends. For reader's reference, official gatekeeper multicast IP address based on H.323 standard is 224.0.1.41 and the port is 1718.

5.7.5.14.5. Examples

The following example sets up a unicast discovery to a gatekeeper:

```
gkip 192.7.5.1
```

In the following example, the gatekeeper has 224.0.1.41 IP address (a multicast IP address) and port 1718, and its priority is 0.

```
gkip 224.0.1.41 1718 0
```

5.7.5.15. h323-id

To configure the H.323 name of the gateway identifying this gateway to its associated gatekeeper, use the **h323-id** command in gateway configuration mode.

```
h323-id h323_id
```

5.7.5.15.1. Syntax

Keyword / Argument	Description
h323-id	H.323 name (ID) used by this gateway when this gateway communicates with its associated gatekeeper. Usually, this ID is the name of the gateway with the gatekeeper domain name appended to the end: name@domain-name . (max length 95)

5.7.5.15.2. Default Value

```
VoIP.ip_address
```

5.7.5.15.3. Command Mode

Gateway Configuration Mode

5.7.5.15.4. Examples

The following example configures the gateway ID is GW13@addpac.com

```
gateway
gkip 211.238.1.1
h323-id GW13@addpac.com
```

5.7.5.16. lightweight-irr

To send IRR(Information Request Response) message transmission as simple information, use this command. To make disable this command, use the **no** form of this command to set disable mode.

```
lightweight-irr
no lightweight-irr
```

5.7.5.16.1. Syntax

Keyword / Argument	Description
	There is no any keyword and argument in this command set.

5.7.5.16.2. Default Value

The default status is disable mode.

Default로 disable 되어있습니다.

5.7.5.16.3. Command Mode

Gateway Configuration Mode

5.7.5.16.4. Usage Guideline

AP200 VoIP gateway is able to send IRR message for response IRQ message from gatekeeper. Originally, this IRR message gives service for status confirmation of VoIP gateway with call information, and so on. However, this message sequence is able to send essential information in case of IRR message period to be shot, not to need large information for call processing.

5.7.5.16.5. Examples

The following command example shows to send essential information as a IRR message.

```
gateway
  lightweight-irr
```

5.7.5.17. h323 call channel

If you want to open voice channels before voice CONNECT In case of normal voice processing, use **h323 call channel early** command in voice service configuration mode. To return default configuration mode, use the **no** form of this command.

```
h323 call channel { early | late }
no h323 call channel
```

5.7.5.17.1. Syntax

Keyword / Argument	Description
early	If you want to open voice channel before voice CONNECT in case of normal(slow) voice processing, use this command.
late	If you want to open voice channel after voice

	CONNECT in case of normal(slow) voice processing, use this command.
--	---

5.7.5.17.2. Default Value

The default status value is late mode.

5.7.5.17.3. Command Mode

Voice-Service Configuration Mode

5.7.5.17.4. Usage Guideline

This command supports one of global voice service parts for VoIP services. This command is able to use at a point of time under transmission CONNECT of H.245 call processing based logical call using the AP200 VoIP gateway or remote other gateway doing starting h323 call.

It may be possible to cut a front of real voice sound in case of opening voice channel after CONNECT message with hooking off other side under default configuration late mode. To avoid this phenomenon, use **h323 call channel early** mode to open voice channel before other side hooking off.

5.7.5.17.5. Examples

The following command example shows fast voice channels

```
voice service VoIP
  h323 call channel early
```

5.7.5.18. h323 call response

If you want to define message besides ALERT message after call proceeding message about response Q.931 setup message, use h323 call response command in voice service configuration mode. To return default configuration mode, use the **no** form of this command.

h323 call response { alert | progress | none}

no h323 call response

5.7.5.18.1. Syntax

Keyword / Argument	Description
alert	Sending ALERT message as response message.
progress	Sending PROGRESS message as response message.
none	Sending CONNECT message as response message after CALL PROCEEDING .

5.7.5.18.2. Default Value

The default status value is alert mode.

5.7.5.18.3. Command Mode

Voice-Service Configuration Mode

5.7.5.18.4. Usage Guideline

This command supports to setup configuration for one of global voice service parts for VoIP service. The AP200 VoIP gateway is able to send message whether gateway send ALERT or PROGRESS message according to command configuration before completing CONNECT message after sending call proceeding message when gateway operate as receiving gateway side with getting setup message. This command configuration setting recommend

default mode besides special case.

5.7.5.18.5. Examples

The following command example shows how to make configuration response message to change progress message.

```
voice service VoIP
  h323 call response progress
```

5.7.5.19. max-digits

This command is used to limit the number of digits for a user class, which in turn adds to the security of specific out-going signal to the FXO port. The "no" default value for this command is "0", meaning there is no limitation.

max-digits *number*

no max-digits

5.7.5.19.1. Syntax

Keyword / Argument	Description
number	Maximum number of digits for out-going signals

5.7.5.19.2. Default Value

The default value is number 0 digit.

5.7.5.19.3. Command Mode

User-class configuration Mode

5.7.5.19.4. Examples

The following describes configuring the maximum digits for user class 1 as "10".

```
voice class user 1
max-digits 10
```

5.7.5.20. password

This command is used to configure a 4 digit security password for the security of out-going signals to FXO port. The "no" default value for this command is "null", which runs no security test for all out-going signals to the FXO port. If a security digit is configured for at least one user class, the password is checked.

```
password string
no password
```

5.7.5.20.1. Syntax

Keyword / Argument	Description
string	security code which could be sequence of IA5 characters. (4 digits)

5.7.5.20.2. Default Value

Null

5.7.5.20.3. Command Mode

User-class configuration Mode

5.7.5.20.4. Examples

The following is the configuration for password "1234" for user class 1.

```
voice class user 1
password 1234
```

5.7.5.21. public-ip

To define public IP address with mapping private IP address of VoIP gateway under static NAT/PAT network environment, use this command. To make disable mode, use the **no** form of this command.

```
public-ip addr  
no public-ip
```

5.7.5.21.1. Syntax

Keyword / Argument	Description
addr	Define IP address setting for example 211.238.72.3.

5.7.5.21.2. Default Value

The default status value is disable mode.

5.7.5.21.3. Command Mode

Gateway Configuration Mode

5.7.5.21.4. Usage Guideline

In case of using private network environment with NAT/PAT, IP address of VoIP gateway should be set in VoIP interface of AP200 VoIP gateway. On the other hand, using public network environment under gatekeeper and other gateway, AP200 VoIP gateway should be defined static NAT or static PAT, and public IP address.

5.7.5.21.5. Examples

The following command example shows public IP configuration mode.

```
gateway
  public-ip xxx.xxx.xxx.xxx
```

5.7.5.22. register

To make registration into gatekeeper, uses **register** command in gateway configuration mode. To cancel registration into gatekeeper, use **"no"** command before this command.

```
register
no register
```

5.7.5.22.1. Syntax

Keyword / Argument	Description
There is no any specific keyword and argument in this command set.	

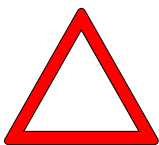
5.7.5.22.2. Default Value

The default status is disable mode.

5.7.5.22.3. Command Mode

Gateway Configuration Mode

5.7.5.22.4. Usage Guideline



To make enable function for H.323 VoIP gateway, uses register command. If VoIP gateway is enable, this VoIP gateway try to look for gatekeeper using H.323 RAS GRQ or RRQ message. If you use no register command, AP200 VoIP gateway is able to cancel registration from gatekeeper using H.323 RAS URG message.

If you want to change or register new dial peer using script file with operating gatekeeper, use no register (or no gateway) command in initial stage of script file or loading configuration after doing un-registration from gatekeeper.

Otherwise, messages may be crashed between gateway and gatekeeper for updating changing information

5.7.5.22.5. Examples

The following command example shows setting registration mode.

```
gateway
  register
```

5.7.5.23. rule

To apply the translation rule to the calling/called party number of the inbound/outbound call, use “**rule**” command in the translation-rule setup mode. To remove the rule that has been set, add “**no**” command to the above command.

```
rule tag input-matched-pattern substituted-pattern
no rule tag
```

5.7.5.23.1. Syntax

Keyword / Argument	Description
<i>tag</i>	An identifier to designate the rule within the rule set. Tag values range from 0 to 65535.
<i>input-matched-pattern</i>	Input digits for pattern matching. Characters that can be entered include numeric data (0 to 9) “#” “*” “[]” “.” and “T.”
<i>substituted-pattern</i>	A pattern to be converted upon successful pattern matching. Characters that can be entered include numeric data (0 to 9) “#” “%” “.” And “T.”

5.7.5.23.2. Default Value

There is no default.

5.7.5.23.3. Command Mode

Translation-rule Configuration Mode

5.7.5.23.4. Usage Guideline

This command is used to apply the translation rule to the calling party/called party number of the inbound/outbound call.

If one or more rules match with the called (or calling) party number, the rule which matches most with the *input-matched-pattern* will be selected.

"Input-matched-pattern" can perform range expression. (eg. [1-9]) Also, the wildcard (.) can be used to apply number of digits of the called/calling party number. If "input-matched-pattern" is configured only with (.) or (T) number translation will be applied to all called/calling-party-number.

"Substituted-pattern" is to convert fixed digits (excluding the wildcard) of the input-matched-pattern into the string. There are two forms of the *substituted-pattern*. See the following:

If the substituted-pattern is composed only of IA5 characters (0 ~ 9, # and *) fixed digits of the input-matched-pattern will be converted into the string part of the substituted-pattern and other digits than the fixed digits of the called/calling party will be attached at the end.

Or, if the substituted-pattern uses "%" form, each digit of the called/calling party number is replaced by "%xx" to make a number. At this time, % values range from %01 to %99 (from the 1st digit to the 99th digit of the called/calling party number.)

If the *substituted-pattern* is composed of (.) or (T) only, the called/calling-party-number is made of other digits than the fixed pattern of the input-matched-pattern.

5.7.5.23.5. Examples

In the following example, 5554123 is extended into 14085554123.

```
rule 0 55541 1408555541
```

In the following example, the translation is not applied when the number is 5551. However, 5551234 is translated into 1408551234.

```
rule 0 555.. 1408555
```

In the following example, 1241234 is converted into 14085551234 and 3551234 is converted into 14085551234..

```
rule 0 [1-3][25]5.. 1408555
```

In the following example, 5551234 is converted into 4441234.

```
rule 0 555.. 444%04%05%06%07%08%09%10%11%12
```

In the following example, 55512 5551234 and 555123456 are all converted into 444.

```
rule 0 555.. 444%99
```

In the following example, 5551234 is converted into 3334.

```
rule 0 555.. 111
```

```
rule 1 55512 222
```

```
rule 2 555[0-9][0-9][0-9] 333
```

In the following example, 5551234 is converted into 1234.

```
rule 0 555 .
```

```
rule 0 555 T
```

In the following example, 555123 is converted into 95551234.

```
rule 0 . 9
```

```
rule 0 T 9
```

5.7.5.24. security password

To configure the secure token with gatekeeper, this **security password** command is used. If this password is enabled, this gateway add CryptoToken element to the message to gatekeeper. This CryptoToken is MD5 hashed token which also should be enabled in gatekeeper when registration, and permit call. If **no** form of this command is used, security between gateway and gatekeeper is disabled.

security password *string*
no security password

5.7.5.24.1. Syntax

Keyword / Argument	Description
string.	security code which could be sequence of ASCII characters. (max length 55)

5.7.5.24.2. Default Value

Disabled

5.7.5.24.3. Command Mode

Gateway Configuration Mode

5.7.5.24.4. Examples

The following example set password "okok1234":

```
gateway
security password okok1234
```

5.7.5.25. security permit-FXO

The outgoing call from remote side to FXO of this system to PSTN or PABX route is security concerned. To protect a call from un-secure user in remote side, when **security permit-FXO** is disabled, the call from remote side that is not registered in session target list of VoIP dial-peer will be dropped.

When session target is set to "ras" with gatekeeper, this command should be set to permit all call to the FXO (This command is only useful without gatekeeper). If **no** form of this command is used, security is enabled and do not permit a call from unregistered VoIP peer.

security permit-FXO

no security permit-FXO

5.7.5.25.1. **Syntax**

This command has no arguments or keywords.

5.7.5.25.2. **Default Value**

Permit All Calls

5.7.5.25.3. **Command Mode**

Voice Service Configuration Mode

5.7.5.25.4. **Usage Guideline**

To keep the security of the calls coming to the FXO port through the network, the AP200 Gateway provides two methods – “Security permit-FXO” command and the voice class user. Since it is possible to directly attempt calls to the PSTN through this FXO port or indirectly attempt calls to the PSTN through the PBX internal line, unauthorized remote users can attempt calls as well. To prevent unauthorized users’ attempting calls, the security shall be kept. Two security systems that the AP200 Gateway provides have following advantages and disadvantages.

With “security permit-FXO” command, remote users does not need to enter the password so they can easily access the network. However, IP address of the VoIP peer on the other side shall be registered and the gatekeeper cannot be used at the same time. Also, it is impossible to bar calls of the registered peers by class.

With the “voice class user” users need to enter the password digit but stronger and multi-classed call barring is possible.

5.7.5.25.5. **Examples**

The following example permit all call to FXO:

voice service VoIP

security permit-FXO

5.7.5.26. signaling-port

5.7.5.27. timeout

To set VoIP-related timer parameters, use “**timeout**” command in the voice service setup mode. To return this setup to the default state, use “**no**” command before this command.

timeout { t301 | t303 | tras | ttl | tidt | treg } *value*

no timeout { t301 | t303 | tras | ttl | tidt | treg }

5.7.5.27.1. Syntax

Keyword / Argument	Description
t301 <i>value</i>	Timeout value from Q.931 Alert message reception till Connect message reception. Values range from 5 to 600, and the default is 180. T301 value is expressed in seconds.
t303 <i>value</i>	Timeout value from Q.931 Setup message transmission till initial message reception. Values range from 5 to 60 and the default value is 8. T303 value is expressed in seconds.
tras <i>value</i>	Timeout value from RAS message transmission till reply message reception. Values range from 2 to 30, and the default value is 6. Tras value is expressed in seconds.
ttl <i>value</i>	RAS time-to-live timeout value. Values range from 10 to 600, and the default value is 60. Values are expressed in seconds. This value is updated by the gatekeeper.
tidt <i>value</i>	Inter-digit timeout value to enter digits into the analog voice port. Values range from 1 to 600, and the default value is 10. Tidt value is expressed in seconds.
treg <i>value</i>	Timeout value for re-registration attempt upon registration failure in the gatekeeper. Values range from 10 to 600 and the default value is 30. Treg value is expressed in seconds.

5.7.5.27.2. Default Value

See the above.

5.7.5.27.3. Command Mode

Voice-Service Configuration Mode

5.7.5.27.4. Usage Guideline

This command is to partially set the global voice-service for the VoIP service. A proper value has been set as the default value of the timeout. It is recommended to use the default value in most of cases.

5.7.5.27.5. Examples

In the following example, the timeout value of the RAS message has been set as three seconds..

```
voice service VoIP
    timeout tras 3
```

5.7.5.28. translate-VoIP-incoming

Use this command to apply the translation rule to every inbound VoIP call. To remove application of the translation rule, add "no" command to the above command.

translate-VoIP-incoming { called-number | calling-number } *tag*

no translate-VoIP-incoming { called-number | calling-number }

5.7.5.28.1. Syntax

Keyword / Argument	Description
called-number	Applies the translation rule to the inbound called party number.
calling-number	Applies the translation rule to the inbound calling party number.
<i>tag</i>	Refers the rule set. Ranges from 0 to 65535.

5.7.5.28.2. Default Value

No translation rule is applied.

5.7.5.28.3. Command Mode

Voice-Service Configuration Mode

5.7.5.28.4. Usage Guideline

This command is to apply the translation rule that has been set by using "translation-rule" command for the inbound VoIP call incoming from the network.

5.7.5.28.5. Examples

In the following example, translation rule set 10 is created and it is applied to the calling party number of the VoIP inbound call. Therefore, if the calling party number of the inbound call incoming from the network is 93450, it will be translated into 9563450.

```
translation-rule 10
    rule 0 9 956
    rule 1 8 878
voice service VoIP
    translate-VoIP-incoming calling-number 10
```

5.7.5.29. ttl-margin

The light-weight RRQ interval is TTL vaule. This command can be used for guide margin of TTL value.

If user want to disable the **ttl-margin**, use **no ttl-margin** command.

`ttl-margin value`
`no ttl-margin`

5.7.5.29.1. Syntax

Keyword / Argument	Description
<i>value</i>	Value is 0 ~50, unit is % value. Default value is 20%.

5.7.5.29.2. Default value

Default value is 20 %.

5.7.5.29.3. Command Mode

Voice-Service setting mode

5.7.5.29.4. Usage Guideline

The RRQ message is keep alive message between gateway and gatekeeper. The RRQ message interval is TTL value. If gateway does not send RRQ message to gatekeeper within TTL time due to transmission delay, VoIP gateway registration expiration will be occurred in gatekeeper. To protect registration expiration due to small transmission delay, guard-band margin of TTL value is necessary.

For example, if TTL is 100 sec and **ttl-margin** is 20, RRQ interval is 80sec.

5.7.5.29.5. Example

```
voice service voip
  ttl-margin 0
```


5.7.6. Miscellaneous Commands

5.7.6.1. clear h323 call

To force a disconnect on a specific call or for all calls active with remote user, use the **clear h323 call** command in Administrator command mode.

```
clear h323 call { all | local_call_ID }
```

5.7.6.1.1. Syntax

Keyword / Argument	Description
all	Forces all active calls currently associated with this gatekeeper to be disconnected.
local_call_ID	Specifies the local call identification number (CallID) that identifies the call to be disconnected.

5.7.6.1.2. Default Value

No Default Value

5.7.6.1.3. Command Mode

Administrator command

5.7.6.1.4. Usage Guideline

If you want to force a particular call to be disconnected (as opposed to all active calls on the gatekeeper), use the CallID number to identify that specific call. You can find the local CallID number for a specific call by using the **show call active all** command; the ID number is displayed in the CallID column.

5.7.6.1.5. Examples

The following example forces all active calls:

```
clear h323 call all
```

5.7.6.2. clear voice port

To force a disconnect on a call on a specific voice port, use the **clear voice port** command in Administrator command mode. If port is not specified, disconnect all calls on the system.

```
clear voice port [port]
```

5.7.6.2.1. Syntax

Keyword / Argument	Description
port	Specifies a port to clear calls on the port..

5.7.6.2.2. Default Value

None

5.7.6.2.3. Command Mode

Administrator command

5.7.6.2.4. Usage Guideline

None

5.7.6.2.5. Examples

The following example forces all active calls:

```
clear voice port
```

5.7.6.3. show call active

To display active call information for voice calls or fax transmissions in progress, use the **show call active** command in Administrator command

```
show call active { all/summary }
```

5.7.6.3.1. Syntax

Keyword / Argument	Description
all	Display all Information about all active calls
summary	Display summarized Information about all active calls.

5.7.6.3.2. Default Value

No default behavior or values..

5.7.6.3.3. Command Mode

Administrator command

5.7.6.3.4. Usage Guideline

Use the **show call active** command to display the contents of the active call table. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information.

5.7.6.3.5. Examples

The following is sample output from the **show call active voice** command:

```
show call active summary
```

5.7.6.4. show call history

To display the call history table for voice calls or fax transmissions, use the **show call history** command in Administrator command.

```
show call history { all } { last number }
```

5.7.6.4.1. Syntax

Keyword / Argument	Description
all	Displays all history information of the call history table
last	(Optional) Displays the last calls connected.
number	the number of calls that appear is defined by the <i>number</i> argument. Valid values are from 1 to 100.

5.7.6.4.2. Default Value

No default behavior or values

5.7.6.4.3. Command Mode

Administrator command

5.7.6.4.4. Usage Guideline

The **show call history** command displays a call history table containing a list of voice or fax calls connected through the Gateway in descending time order. Each call record is aged out of the table after a configurable number of minutes has elapsed,

5.7.6.4.5. Examples

The following is sample output from the **show call history voice** command:

```
show call history all last 10
```

5.7.6.5. show clear-down-tone

To show clear-down-tone classes , use the **show clear-down-tone** command in Administrator command mode. Without number, all clear-down-tone classes will be displayed.

```
show clear-down-tone
```

5.7.6.5.1. Syntax

This command has no arguments or keywords.

5.7.6.5.2. Default Value

No default behavior or values

5.7.6.5.3. Command Mode

Administrator command

5.7.6.5.4. Usage Guideline

This command will display not only user added clear-down-tone by **voice class clear-down-tone** command, but also system providing clear-down-tones.

5.7.6.5.5. Examples

The following is to display all Clear-down-tone Class.

```
show clear-down-tone
```

5.7.6.6. show codec class

To show codec classes , use the **show codec-class** command in Administrator command mode. Without number, all codec classes will be displayed.

```
show codec class [ number ]
```

5.7.6.6.1. Syntax

Keyword / Argument	Description
number	(Optional) codec class tag number.

5.7.6.6.2. Default Value

No default behavior or values

5.7.6.6.3. Command Mode

Administrator command

5.7.6.6.4. Examples

The following is to display all Codec Class.

```
show codec class
```

5.7.6.7. show dial-peer

To display configuration information for dial peers, use the **show dial-peer voice** command. If there is no options, display all information of all Dial-Peer..

```
show dial-peer {voice | pots | VoIP} [ number | summary ]
```

5.7.6.7.1. Syntax

Keyword / Argument	Description
voice	Display VoIP and POTS Dial-peer information
pots	Display POTS Dial-peer information
VoIP	Display VoIP Dial-peer information
number	Optional) A specific dial peer. This option displays configuration information for a single dial peer identified by the <i>number</i> argument. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.
summary	Optional) Displays a summary of all voice dial peers

5.7.6.7.2. Default Value

No default behavior or values

5.7.6.7.3. Command Mode

Administrator command

5.7.6.7.4. Usage Guideline

Use the **show dial-peer voice** Administrator command to display the configuration for all Voice over IP (VoIP) and plain old telephone service (POTS) dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

5.7.6.7.5. Examples

The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
show dial-peer voice
```

5.7.6.8. show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dialplan number** command in Administrator command mode.

```
show dialplan number dial_string
```

5.7.6.8.1. Syntax

Keyword / Argument	Description
dial_string	Specifies a particular destination pattern (telephone number).

5.7.6.8.2. Default Value

No default behavior or values.

5.7.6.8.3. Command Mode

Administrator command

5.7.6.8.4. Usage Guideline

The **show dialplan number** command is used to test whether the dial plan configuration is valid and working as expected.

5.7.6.8.5. Examples

The following is sample to show all dial-peers matching telephone number 4441234:

```
show dialplan number 4441234
```

5.7.6.9. show dialplan port

To show which POTS dial peer is matched for a specific calling number or voice port, use the **show dialplan port** command in Administrator command mode.

```
show dialplan port voice-port
```

5.7.6.9.1. Syntax

Keyword / Argument	Description
voice_port	Specifies the voice port location. (slot number / port number)

5.7.6.9.2. Default Value

No default behavior or values.

5.7.6.9.3. Command Mode

Administrator command

5.7.6.9.4. Usage Guideline

Use the **show dialplan port** command as a troubleshooting tool to determine which POTS dial peer is matched to an voice-port.

5.7.6.9.5. Examples

To show all dial-peers matching port 1/1:

```
show dialplan port 1/1
```

5.7.6.10. show gateway

To show gateway related information , use the **show gateway** command in Administrator command.

```
show gateway
```

5.7.6.10.1. Syntax

This command has no arguments or keywords.

5.7.6.10.2. Default Value

No default behavior or values.

5.7.6.10.3. Command Mode

Administrator command

5.7.6.10.4. Usage Guideline

This command will display not only gatekeeper interaction information which are gatekeeper IP address, registration status, registered aliases, but also system resource information about VoIP gateway (i.e., number of dial-peers, number of voice ports, number of codec classes, ...)

5.7.6.10.5. Examples

To show gateway related information of this system:

```
show gateway
```

5.7.6.11. show num-exp

To show all number expansions information, **show num-exp** Administrator command mode.

show num-exp

5.7.6.11.1. Syntax

This command has no arguments or keywords.

5.7.6.11.2. Default Value

No default behavior or values.

5.7.6.11.3. Command Mode

Administrator command

5.7.6.11.4. Usage Guideline

Even though you create number expansion with wildcard(*), show num-exp will not display wildcard.

5.7.6.11.5. Examples

To show number expansion information of this system:

show num-exp

5.7.6.12. show translation-rule

To view whole or total application result of the translation rule, use "**show translation-rule**" command that is one of administrator's commands.

show translation-rule *[tag]* *[dial_string]*

5.7.6.12.1. Syntax

Keyword / Argument	Description
<i>tag</i>	Designates a certain rule set. If not, all translation rules will be displayed..
<i>dia_string</i>	If a certain destination pattern (telephone number) is entered, application result of the rule will be shown.

5.7.6.12.2. Default Value

No default behavior or values.

5.7.6.12.3. Command Mode

Administrator command

5.7.6.12.4. Usage Guideline

This command is to check if the translation rule has been properly set and to test operations of the translation rule.

5.7.6.12.5. Examples

In the following example, the result of applying the translation rule to the telephone number 4441234 will be displayed.

```
show translation-rule 10 4441234
```

5.7.6.13. show user-class

To show user classes , use the **show user-class** command in Administrator command mode. All user classes will be displayed.

```
show user-class
```

5.7.6.13.1. Syntax

This command has no arguments or keywords.

5.7.6.13.2. Default Value

No default behavior or values.

5.7.6.13.3. Command Mode

Administrator command

5.7.6.13.4. Usage Guideline

This command shows user class tag, password, and max digits can input

5.7.6.13.5. Examples

To display User Class information :

```
show user-class
```

5.7.6.14. show voice port

To show voice port information , use the **show voice port** command in Administrator command mode. Without slot/port, all voice port available in this system will be displayed.

```
show voice port [summary | slot/port ]
```

5.7.6.14.1. Syntax

Keyword / Argument	Description
summary	(Optional) Brief information.
slot/port	(Optional) slot number and port number.

5.7.6.14.2. Default Value

No default behavior or values.

5.7.6.14.3. Command Mode

Administrator command

5.7.6.14.4. Usage Guideline

This command can be used not only Administrator command, but also Voice-port configuration mode.

5.7.6.14.5. Examples

To show brief voice port information of this system:

```
show voice port summary
```

5.7.6.15. show VoIP-interface

To view the VoIP interface that is currently designated, use "show VoIP-interface" command that is one of administrator's commands.

```
show VoIP-interface
```

5.7.6.15.1. Syntax

This command has no arguments or keywords

5.7.6.15.2. Default Value

No default behavior or values.

5.7.6.15.3. Command Mode

Administrator command

5.7.6.15.4. Usage Guideline

Shows the VoIP interface currently in service.

5.7.6.15.5. Examples

In the following example, VoIP interface information of the corresponding system will be displayed.

```
show VoIP-interface
```

5.7.6.16. debug VoIP call

To trace VoIP related events, use the **debug VoIP call** command in Administrator command mode.

```
debug VoIP call
```

```
no debug VoIP call
```

5.7.6.16.1. Syntax

This command has no arguments or keywords

5.7.6.16.2. Default Value

No default behavior or values.

5.7.6.16.3. Command Mode

Administrator command

5.7.6.16.4. Usage Guideline

The trace will be displayed by console port. Q.931 events, H.245 events, User interface events will be displayed. This trace makes system performance degraded. This should be disabled on normal operational state.

5.7.6.16.5. Examples

To trace VoIP events:

```
debug VoIP call
```

To stop trace:

```
undebug VoIP call
```

5.7.6.17. debug VoIP

To trace events related to VoIP ASN.1, use the administrator command mode "debug VoIP"

```
debug VoIP { h225-asn1 | h245-asn1 | ras-asn1 }
```

```
no debug VoIP { h225-asn1 | h245-asn1 | ras-asn1 }
```

5.7.6.17.1. Syntax

Keyword / Argument	Description
h225-asn1.	Trace H.225 ASN.1 event
h245-asn1	Trace H.245 ASN.1 event.
ras-asn1	Trace RAS ASN.1 event

5.7.6.17.2. Default Value

No default behavior or values.

5.7.6.17.3. Command Mode

Administrator command

5.7.6.17.4. Usage Guideline

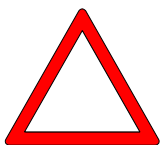
This command traces H.225 ASN.1, H.245 ASN.1, and RAS ASN.1 events, to display it on the console port. This command may effect system performance, therefore it is advised to disable this function under normal circumstances.

Information Users can see VoIP-related messages and call tracing through the console port.



This is default setting. However, if the user wishes to view call tracing through the telnet terminal from a remote place, the user shall use “**debug-port**” command (one of the global commands) from the remote terminal. Message tracing operates either in the console or in the remote terminal wherever “debug-port” command has been used. If the telnet terminal is terminated, message tracing will automatically operate in the console. “No debug-port” displays call tracing on the default display console.

Caution



Message tracing with using the debug command gives a high load to the router so it is recommended not to use the debug command under normal conditions. After tracing in the telnet terminal, do not exit the terminal without using “no debug” or “undebug” command. Otherwise, message tracing will be displayed in the console.

5.7.6.17.5. Examples

The following is an example of tracing an H.225 ASN.1 event for the system.

```
debug VoIP h225-asn1
```

The following is an example of switching an enabled H.245 ASN.1 Debugging function off for the system.

```
undebug VoIP h245-asn1
```


Appendix A. AP200 VoIP Gateway Specifications

This chapter explains the technical specification of VoiceFinder AP200 VoIP Residential Gateway.

IP Routing Service

IP Routing Protocols	Static
	Default routing
	Dynamic Default Routing Configuration for DHCP Client
	IEEE 802.1Q VLAN Routing

LAN Service

Ethernet Interface	10Mbps Ethernet Interface 10Mbps Ethernet HUB Interface
<ul style="list-style-type: none"> ● Ethernet Interface Configuration 	Port configuration
	Secondary/Subnet Support
	<ul style="list-style-type: none"> ● MTU size Change ● ARP Entry Revalidate Function
	<ul style="list-style-type: none"> ● Transmit/Receive Connection Recovery Function

Voice over IP Service

VoIP Protocols	ITU-T H.323 v2 Protocol with ITU-T H.235 Security Feature
	Session Initiation Protocol (SIP)
Voice Compression	G.723.1 MP-MLQ, 6.3Kbps, 5.3Kbps
	G.729.A CS-ACELP, 8Kbps
	G.711 PCM, 64Kbps
Voice Processing	Voice Activity Detection (VAD)
	T.38 Real Time Protocol(In band, Out Band Mode)
	Dual Tone Multi Frequency (DTMF)
	Comfort Noise Generation (CNG)
	Echo Cancellation

Network Managements

SNMP	Standard SNMP Agent MIB v2
RMON	Remote Monitoring, RFC1271 Support
Web	Web Based Management using HTTP Server Interface
Others	Traffic Queuing
	RTP/RTCP Flow Control

Security Functions

IP Access List	Standard and Extended IP Access List, IP Packet Filtering
PPP User Authentication	Password Authentication Protocol (PAP)
	Challenge Handshake Authentication Protocol (CHAP)
Others	Access Control and Data Protections
	Enable/Disable for Specific Protocols
	Multi-level User Account Management
	Auto-disconnect for Telnet/Console Sessions
*VPN	High-performance VPN Features

Operation and Managements

Console Port	RS-232C Based Async Serial Interface Support
Remote Management	Console, Rlogin, Telnet
System Performance Analysis	Process, CPU, and Connection Interface
APOS Management	APOS Configuration Back-up and Restore
	Remote Upgrade Function using FTP/TFTP
Others	Debugging and System Auditing
	Data Logging and Diagnostics
	System Booting, Auto-rebooting with Watch-dog Timer
	IP Traffic Statistics with Accounting

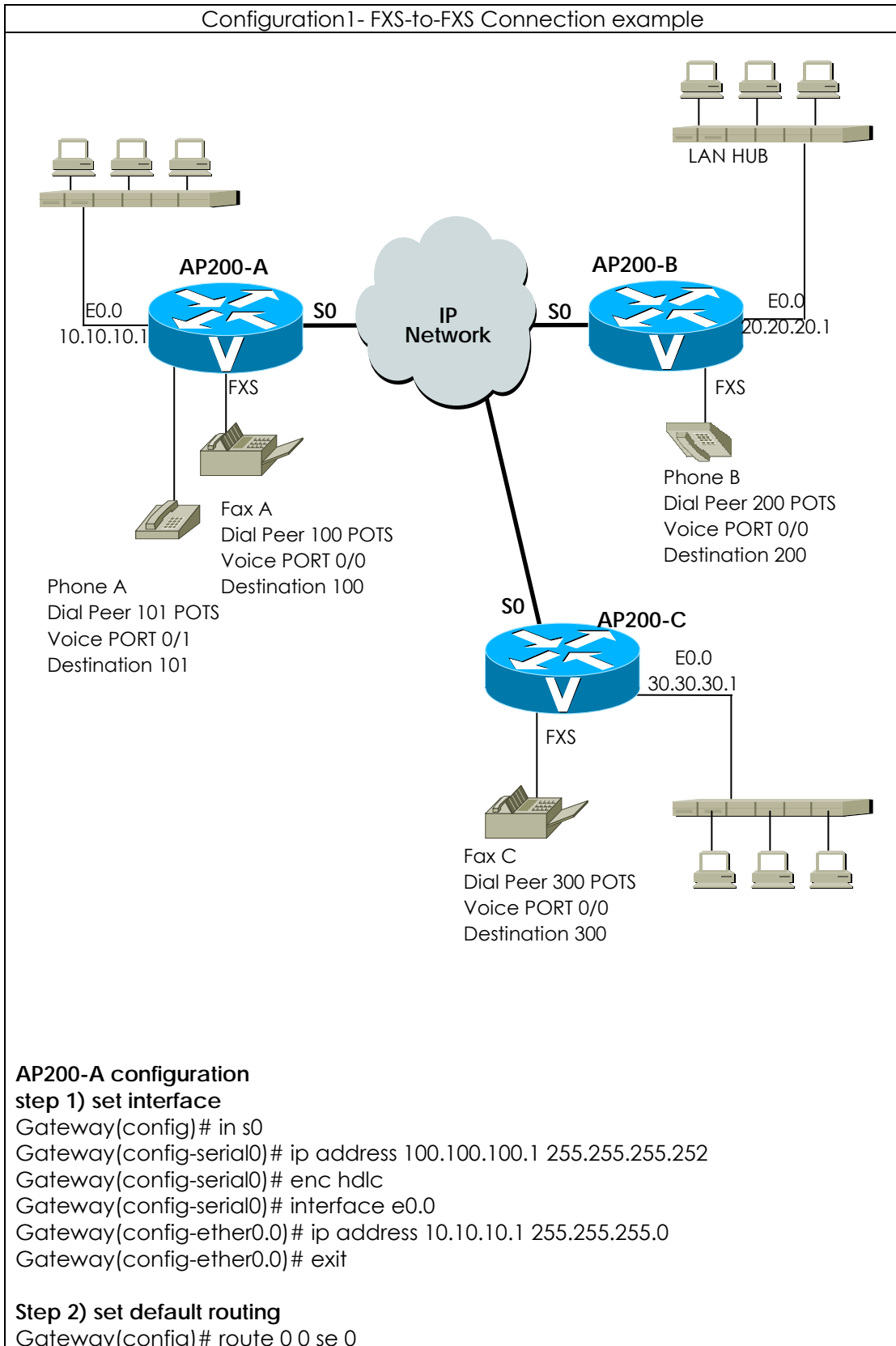
Other Scalability Features

DHCP	Dynamic Host Configuration Protocol (DHCP) Server and Relay Functions
ADSL	PPPoE (Static IP, Dynamic IP)
NAT/PAT	Network Address Translation (NAT) Protocol Port Address Translation (PAT) Protocol
Bridging	IEEE Standard Spanning Tree Bridging Protocol Remote Bridging Support Concurrent Bridging Support
User Interface	Industry Standard Command Line Interface (CLI)
Others	Network Time Protocol (NTP) Support

Hardware Specification

Microprocessor	32bit RISC Microprocessor
Network Interface	1-Port 10Mbps Ethernet Interface for WAN (RJ45) 1-Port 10Mbps Ethernet Interface for LAN (RJ45) or 4-Port 10Mbps Ethernet HUB for LAN (RJ45) 1-Port Async Serial Interface for RS-232C Console Port (RJ45)
Voice Interface	1-Port or 2-Port FXS Voice Interface (1 x RJ11) 1-Port PSTN Backup Voice Interface (1 x RJ11)
Memory	2MB Flash Memory 16MB SDRAM / Main Memory 512KB Boot Flash Memory
System LED	WAN, LAN, Power LED Support (Front Panel)
Power	DC External Power Supply (5V x 1.2A)
Power Requirement	7 Watt
Operating Temperature	0°C ~ 55°C
Storage Temperature	-40°C ~ 85°C
Relative Humidity	5% ~ 95%
Cooling Method	Natural Air Cooling
Depth x Width x Height	134 X 200 X 26 (mm)
Weight	350g(Main Body)/485g (Including Power Adapter)

Appendix B. VoIP(Voice over IP) Config. Example



Step 3) set POTS Peer

```
Gateway(config)# dial-peer voice 100 pots
Gateway(config-dialpeer-pots-100)# port 0/0
Gateway(config-dialpeer-pots-100)# destination-pattern 100
Gateway(config-dialpeer-pots-100)# dial-peer voice 101 pots
Gateway(config-dialpeer-pots-101)# port 0/1
Gateway(config-dialpeer-pots-101)# destination-pattern 101
Gateway(config-dialpeer-pots-101)# exit
```

Step 4) set VoIP Peer

```
Gateway(config)# dial-peer voice 200 VoIP
Gateway(config-dialpeer-VoIP-200)# destination-pattern 2..
Gateway(config-dialpeer-VoIP-200)# session target 20.20.20.1
Gateway(config-dialpeer-VoIP-200)# dial-peer voice 300 VoIP
Gateway(config-dialpeer-VoIP-300)# destination-pattern 3..
Gateway(config-dialpeer-VoIP-300)# session target 30.30.30.1
Gateway(config-dialpeer-VoIP-300)# exit
```

Step 5) configuration Confirmation

```
Gateway(config)# show run
interface loopback0
 ip address 127.0.0.1 255.0.0.0
!
interface ether0.0
 ip address 10.10.10.1 255.255.255.0
!
interface serial0
 ip address 100.100.100.1 255.255.255.252
 Encapsulation HDLC
 Operation is DOWN
!
!
```

```
Gateway(config)# sh route
```

Destination	Network-Mask	Gateway	Interface	Protocol
0.0.0.0	0.0.0.0		serial0	STATIC
10.10.10.0	255.255.255.0	10.10.10.1	ether0.0	DIRECT
100.100.100.0	255.255.255.252	100.100.100.1	serial0	DIRECT
127.0.0.0	255.0.0.0	127.0.0.1	loopback0	DIRECT

```
Gateway(config)# show dial-peer voice
POTS Peers :
```

```
Pots peer 100
 dest-pattern = 100
 port = 0/0 (0)
 prefix =
 register E.164 = yes
 administrative status = up
```

```
Pots peer 101
 dest-pattern = 101
 port = 0/1 (1)
```

```
prefix =  
register E.164 = yes  
administrative status = up
```

VoIP Peers :

```
VoIP peer 300  
dest-pattern = 3..  
session-target = 30.30.30.1  
codec = default  
codecClass = default  
dtmfRelay = h245-alphanumeric  
vad = yes  
translation-outgoing called-number = -1  
translation-outgoing calling-number = -1  
description =  
administrative status = up
```

```
VoIP peer 200  
dest-pattern = 2..  
session-target = 20.20.20.1  
codec = default  
codecClass = default  
dtmfRelay = h245-alphanumeric  
vad = yes  
translation-outgoing called-number = -1  
translation-outgoing calling-number = -1  
description =  
administrative status = up
```

AP200-C configuration

step 1) set interface

```
Gateway(config)# in s0  
Gateway(config-serial0)# ip address 100.100.102.1 255.255.255.252  
Gateway(config-serial0)# enc hdlc  
Gateway(config-serial0)# interface e0.0  
Gateway(config-ether0.0)# ip address 30.30.30.1 255.255.255.0  
Gateway(config-ether0.0)# exit
```

Step 2) set default routing

```
Gateway(config)# route 0 0 se 0
```

Step 3) set POTS Peer

```
Gateway(config)# dial-peer voice 300 pots  
Gateway(config-dialpeer-pots-300)# port 0/0  
Gateway(config-dialpeer-pots-300)# destination-pattern 300  
Gateway(config-dialpeer-pots-300)# exit
```

Step 4) set VoIP Peer

```
Gateway(config)# dial-peer voice 100 VoIP  
Gateway(config-dialpeer-VoIP-100)# destination-pattern 1..  
Gateway(config-dialpeer-VoIP-100)# session target 10.10.10.1  
Gateway(config-dialpeer-VoIP-100)# dial-peer voice 300 VoIP  
Gateway(config-dialpeer-VoIP-200)# destination-pattern 2..
```

```

Gateway(config-dialpeer-VoIP-200)# session target 20.20.20.1
Gateway(config-dialpeer-VoIP-200)# exit

Step 5) configuration Confirmation
Gateway(config)# show run
interface loopback0
  ip address 127.0.0.1 255.0.0.0
!
interface ether0.0
  ip address 30.30.30.1 255.255.255.0
!
interface serial0
  ip address 100.100.102.1 255.255.255.252
  Encapsulation HDLC
  Operation is DOWN
!
!

Gateway(config)# sh route
Destination      Network-Mask    Gateway          Interface        Protocol
-----
0.0.0.0          0.0.0.0        serial0          STATIC
30.30.30.0      255.255.255.0  30.30.30.1     ether0.0         DIRECT
100.100.102.0   255.255.255.252 100.100.102.1  serial0          DIRECT
127.0.0.0       255.0.0.0      127.0.0.1      loopback0        DIRECT

Gateway(config)# show dial-peer voice
POTS Peers :

Pots peer 300
  dest-pattern = 300
  port = 0/0 (0)
  prefix =
  register E.164 = yes
  administrative status = up

VoIP Peers :

VoIP peer 200
  dest-pattern = 2..
  session-target = 20.20.20.1
  codec = default
  codecClass = default
  dtmfRelay = h245-alphanumeric
  vad = yes
  translation-outgoing called-number = -1
  translation-outgoing calling-number = -1
  description =
  administrative status = up

VoIP peer 100
  dest-pattern = 1..
  session-target = 10.10.10.1
  codec = default
  codecClass = default
    
```

```
dtmfRelay = h245-alphanumeric  
vad = yes  
translation-outgoing called-number = -1  
translation-outgoing calling-number = -1  
description =  
administrative status = up
```

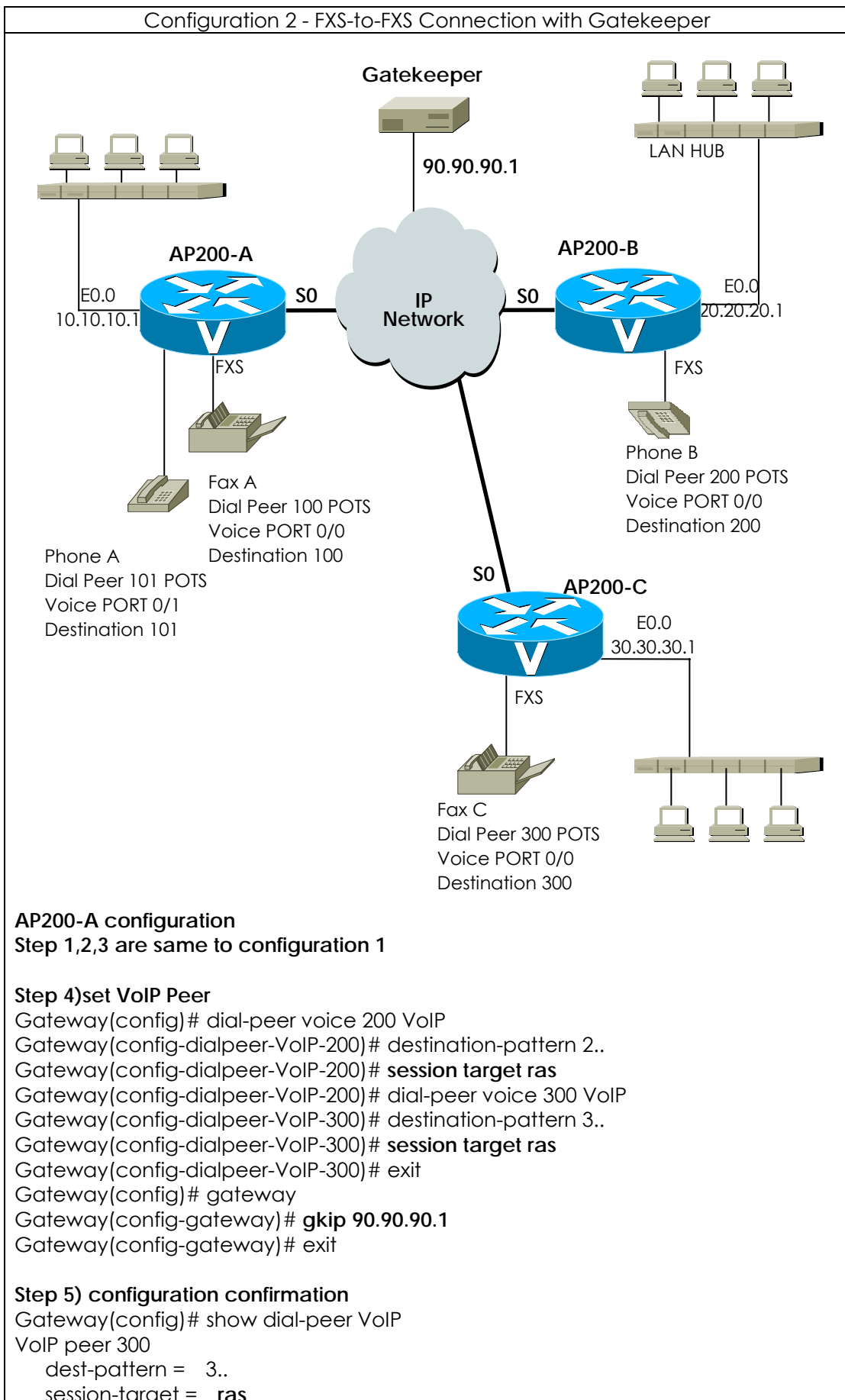
Call Scenario

1) Phone A and Phone B connection

- User A hangs up phone A.
- user A press No. 200 dial up.
- Confirm RING sound.
- User B hangs up phone B.
- Speak over the telephone.
- Finish speaking out.

2) Fax A and FAX C connection

- Insert paper for transmission from FAX B to FAX C.
- Press No. 300 dial up from FAX B to FAX C.
- Confirm transmission and FAX quality



```
codec = default
codecClass = default
dtmfRelay = h245-alphanumeric
vad = yes
translation-outgoing called-number = -1
translation-outgoing calling-number = -1
description =
administrative status = up
```

VoIP peer 200

```
dest-pattern = 2..
session-target = ras
codec = default
codecClass = default
dtmfRelay = h245-alphanumeric
vad = yes
translation-outgoing called-number = -1
translation-outgoing calling-number = -1
description =
administrative status = up
```

Gateway(config)# show gateway

Gatekeeper Registration Information

```
this gateway's H.323 id = VoIP.10.10.10.1
gatekeeper registration option = yes
gatekeeper registration status :
    not registered.
gatekeeper address = 90.90.90.1
gatekeeper security = disabled
local aliases
    [1] VoIP.10.10.10.1
    [2] 100
    [3] 101
```

Gateway Information

```
number of ports = 8
number of pots peers = 2
number of VoIP peers = 2
number of number expansions = 0
number of codec classes = 0
number of user classes = 0
number of current calls = 0
end of digit = #
ip address prefix = *
permit unregistered h323 incoming call to FXO = yes
h323 call start mode = fast
system fax mode = t38
system fax rate = 14400 bps
system T.38 fax redundancy = 0
```

AP200-B configuration

**Step 1,2,3 are same to configuration 1
Step 4) set VoIP Peer**

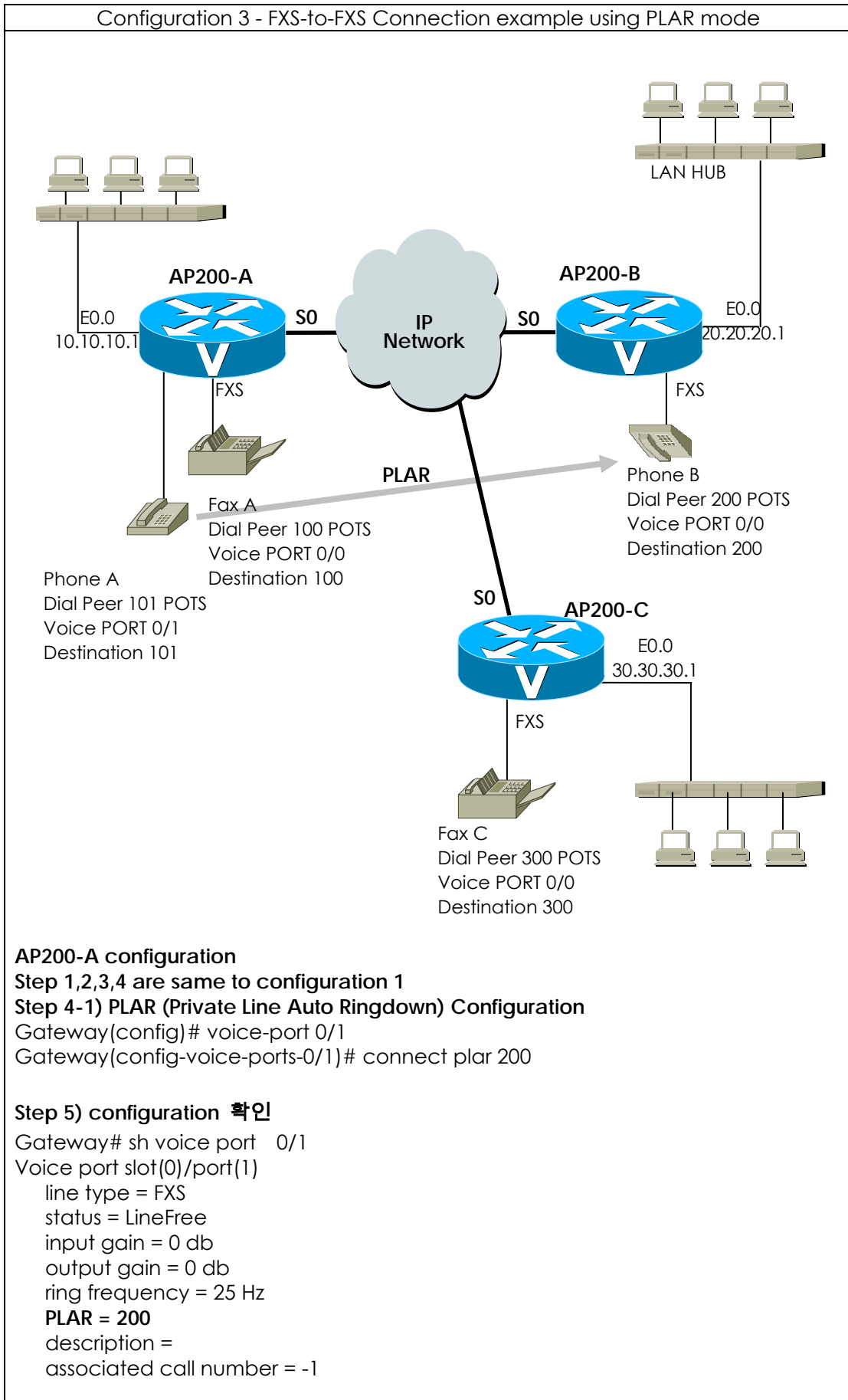
```
Gateway(config)# dial-peer voice 100 VoIP
Gateway(config-dialpeer-VoIP-100)# destination-pattern 2..
Gateway(config-dialpeer-VoIP-100)# session target ras
Gateway(config-dialpeer-VoIP-200)# dial-peer voice 300 VoIP
Gateway(config-dialpeer-VoIP-300)# destination-pattern 3..
Gateway(config-dialpeer-VoIP-300)# session target ras
Gateway(config-dialpeer-VoIP-300)# exit
Gateway(config)# gateway
Gateway(config-gateway)# gkip 90.90.90.1
```

AP200-C configuration

Reference Gateway A,B configuration

Call Scenario

Configuration 3 - FXS-to-FXS Connection example using PLAR mode

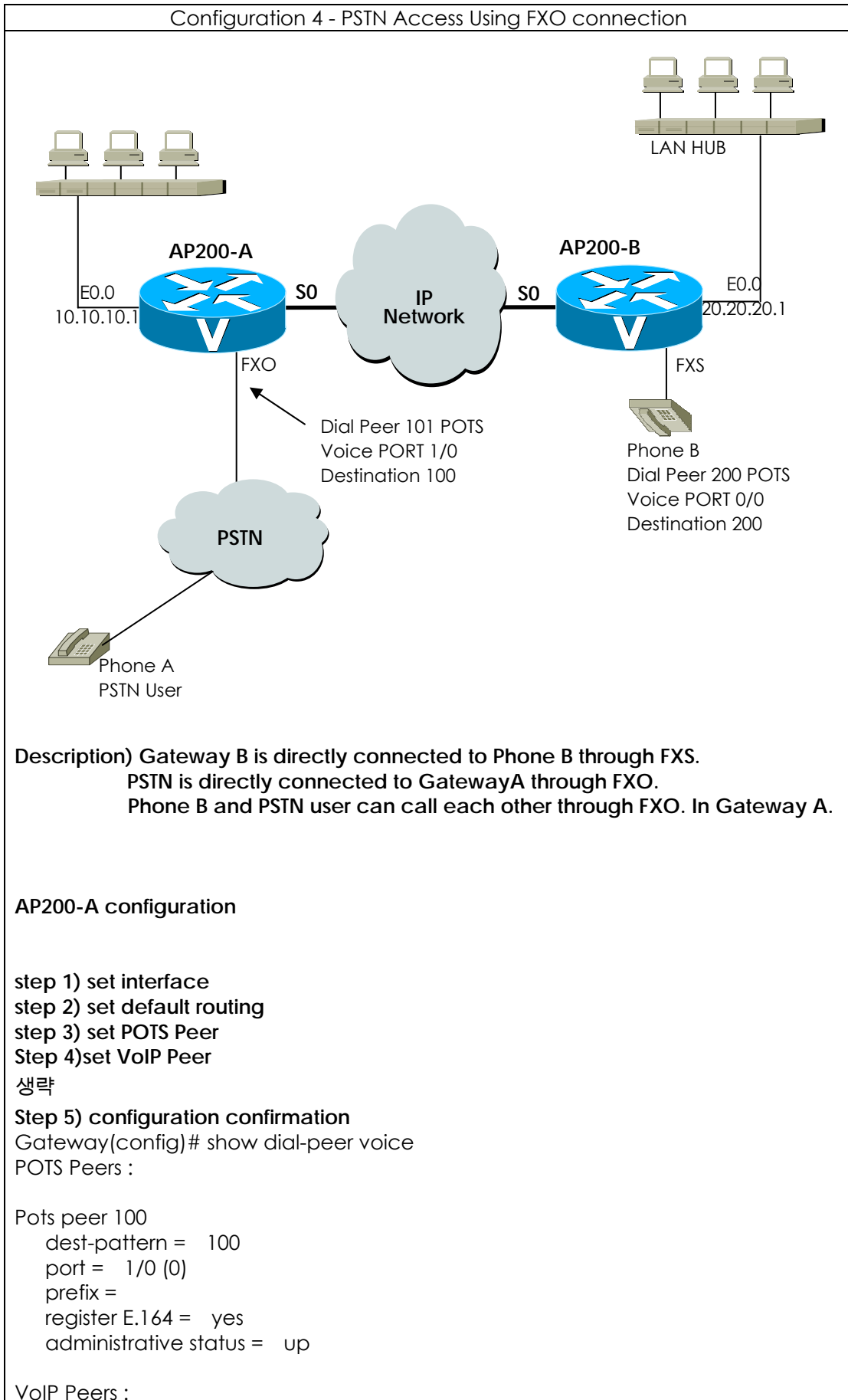


AP200-B configuration
same to configuration 1

AP200-C configuration
same to configuration 1

Call Scenario

1) Phone A and Phone B connection



```
VoIP peer 200
  dest-pattern = 2..
  session-target = 20.20.20.1
  codec = default
  codecClass = default
  dtmfRelay = h245-alphanumeric
  vad = yes
  translation-outgoing called-number = -1
  translation-outgoing calling-number = -1
  description =
  administrative status = up
```

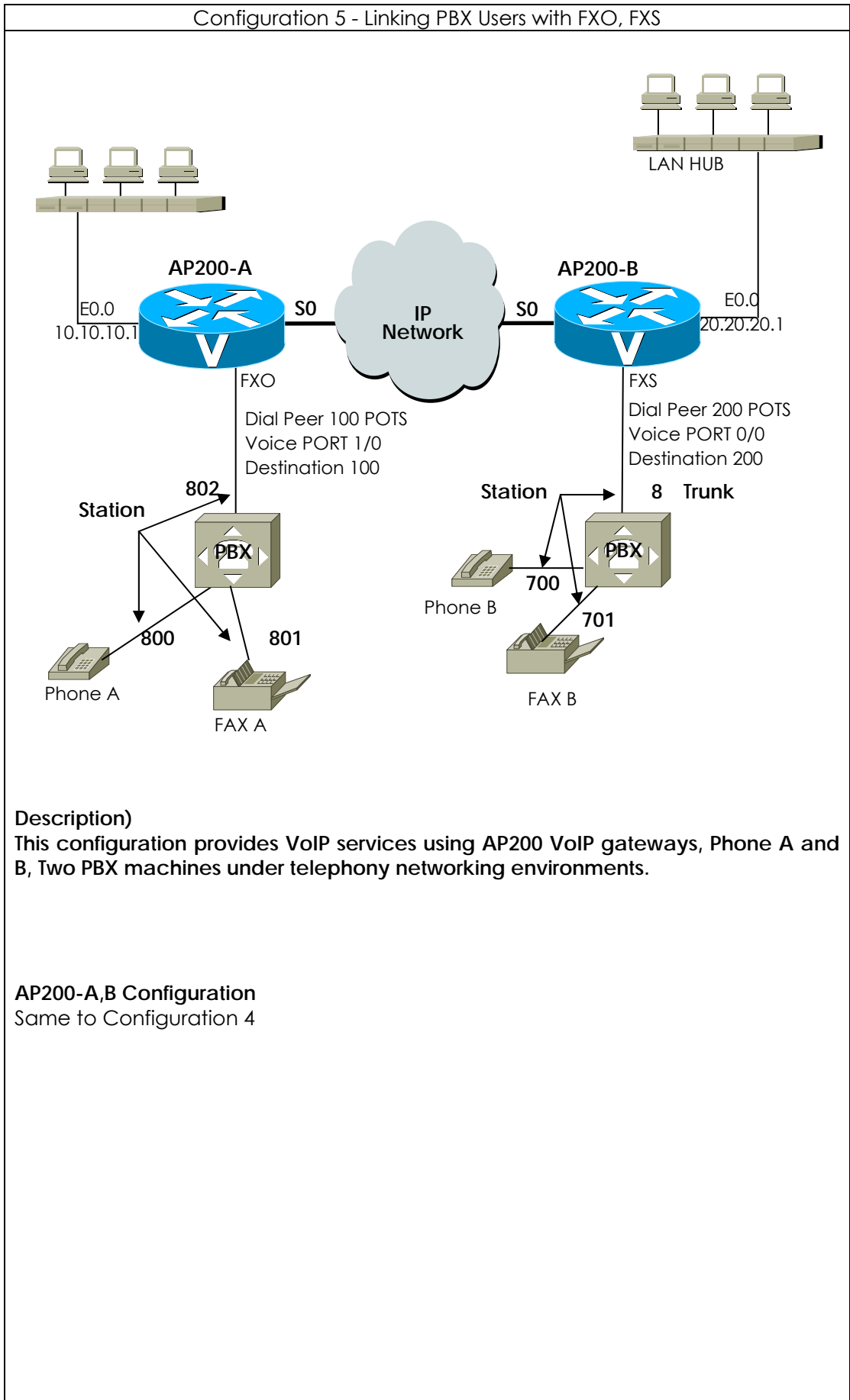
AP200-B configuration

Same to configuration 1

Call Scenario

- 1) Call Connection from Phone A to Phone B under PSTN
 - Press dial No. 568-3848 in connecting PSTN phone.
 - Listen dial tone, press dial No. 200 of phone B.
 - Confirm RING sound.
 - Hand up phone B.
 - Speak over the telephone.
 - Finish speaking out.

- 2) Call Connection from Phone B to PSTN user Phone
 - Hang up Phone B.
 - Press dial No. 100 in connecting VoIP gateway FXO voice port 1/0 under PSTN.
 - Confirm dial tone sound, and press PSTN dial number.
 - Speak over the telephone.
 - Finish speaking out.



Call Scenario

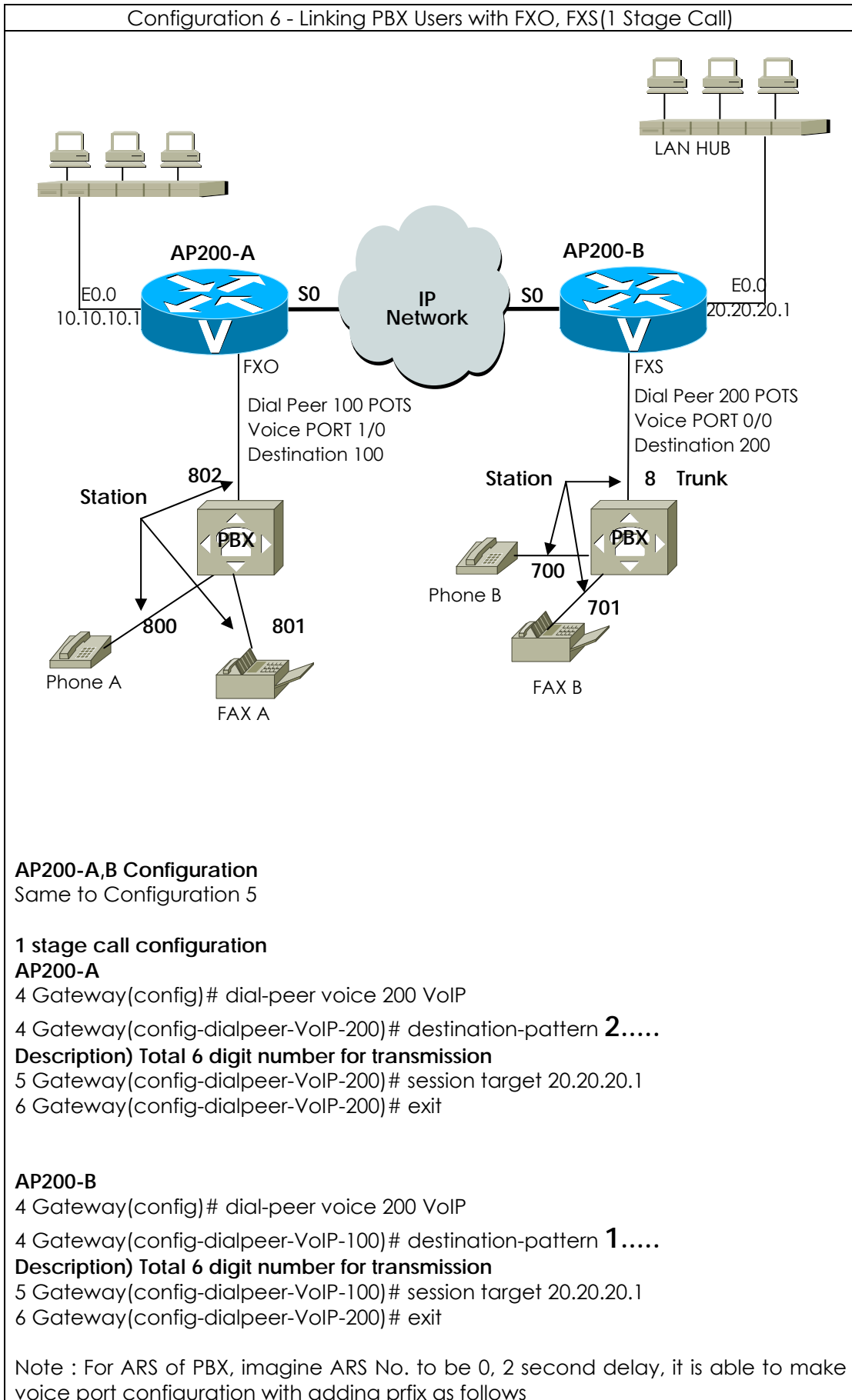
1) Phone A To Phone B connection

- Hang up Phone A.
- Listen dial tone sound of PBX A machine, and then press dial No. 802 in connecting VoIP gateway A.
- Listen dial tone sound of gateway A, and then press dial No. 200 in connecting VoIP gateway FXS voice port.
- Listen dial tone sound of PBX B machine, and then press dial No. 700 (internal Number)
- Hang up Phone B
- Speak over the telephone.
- Finish speaking out.

2) Phone B To Phone A connection

- Hang up Phone B.
- Listen dial tone sound of PBX B machine, and then press dial No. 8 in connecting VoIP gateway FXO port.
- Listen dial tone sound of VoIP gateway, and then press dial No. 100 in connecting VoIP gateway A FXO port.
- Listen dial tone sound of PBX A machine, and then press dial No. 800 (internal number)
- Speak over the telephone.
- Finish speaking out.

Note : Each call scenario is made up setting of PBX configuration.



Gateway(config-dialpeer-pots-100)# prefix ,,0,,

Description) Press 100200 at Phone B

Gateway A use No. "100"-> PBX connecting

2second delay

Gateway A dial up "0" to PBX

2 second delay

Gateway A dial up "200" to PBX

....

Call Scenario

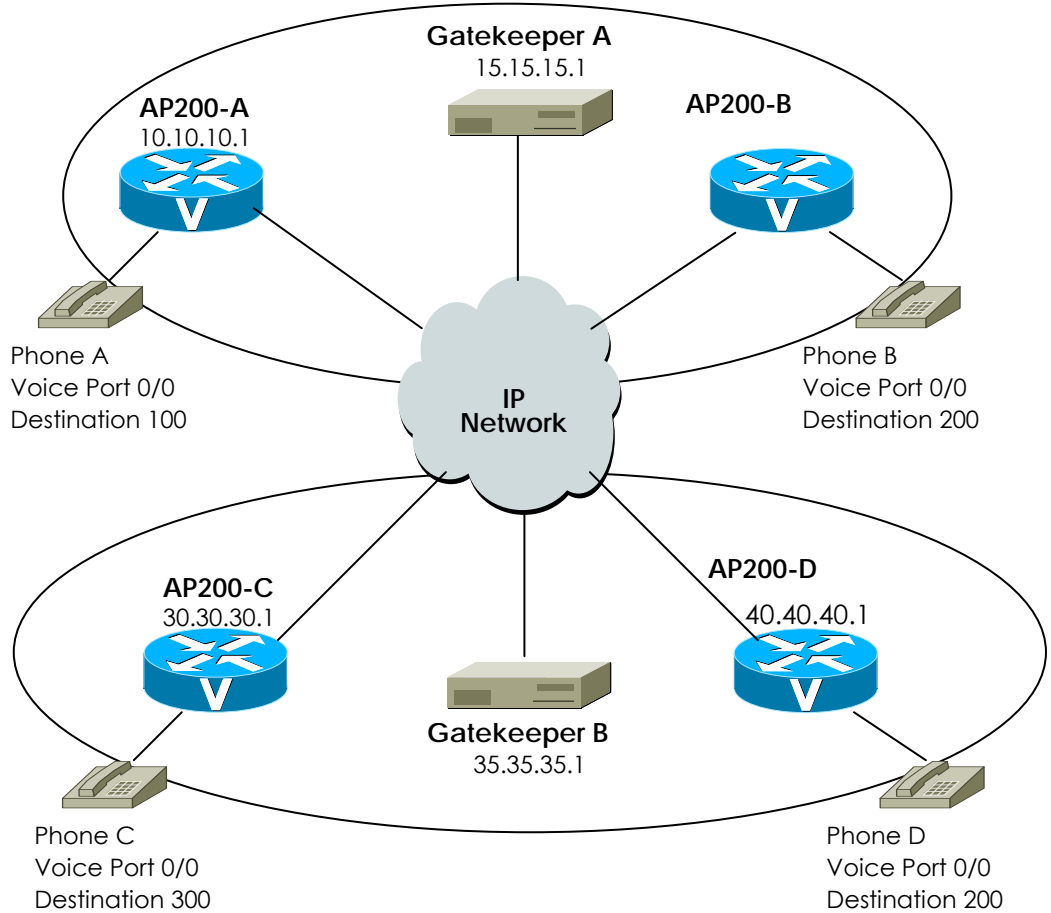
1) Phone A To Phone B connection

- Hang up Phone A.
- Listen dial tone sound of PBX machine, and then press dial No. 802 in connecting VoIP gateway A FXO port.
- Listen dial tone sound of VoIP gateway A, and then press dial No. **200700** in connecting part of VoIP gateway FXS voice port phone No. 200 and other connecting part of PBX internal phone number 700. (2 stage)
- Confirm RING sound.
- Speak over the telephone.
- Finish speaking out.

2) Phone B To Phone A connection

- Hang up Phone B.
- Listen dial tone sound of PBX B, and then press No. 8 in connecting VoIP gateway FXO voice port.
- Listen dial tone sound of VoIP gateway, and then press No. 100800 in connecting part of VoIP gateway FXO voice port phone No. 100, and other connecting part of PBX internal phone No. 800.
- Confirm RING sound.
- Speak over the telephone.
- Finish speaking out.

Configuration 7 – Calling under 2 network domain & Translate-outgoing call



Configuration & Description)

VoIP gateway A and B uses gatekeeper A, VoIP gateway C and D uses gatekeeper B. There are 3 way following call path as follows.

- 1) Phone A to Phone B
Normal path.

Configuration
Refer to Configuration 2

- 2) Phone A to Phone C
VoIP gateway A and C is connected different gatekeeper each other. To connect from phone A to phone C, gateway configuration should be setup static VoIP peer to peer configuration for each other gateway and gatekeeper.

AP200-A Configuration

```
Gateway(config)# dial-p voice 200 VoIP
Gateway(config-dialpeer-VoIP-200)# destination-pattern ...
Gateway(config-dialpeer-VoIP-200)# session target ras
Gateway(config-dialpeer-VoIP-200)# dial-p voice 300 VoIP
Gateway(config-dialpeer-VoIP-300)# destination-pattern 3..
Gateway(config-dialpeer-VoIP-300)# session target 30.30.30.1
Gateway(config-dialpeer-VoIP-300)# exit
```

```
Gateway(config)# gateway
Gateway(config-gateway)# gkip 15.15.15.1
Gateway(config-gateway)# exit
```

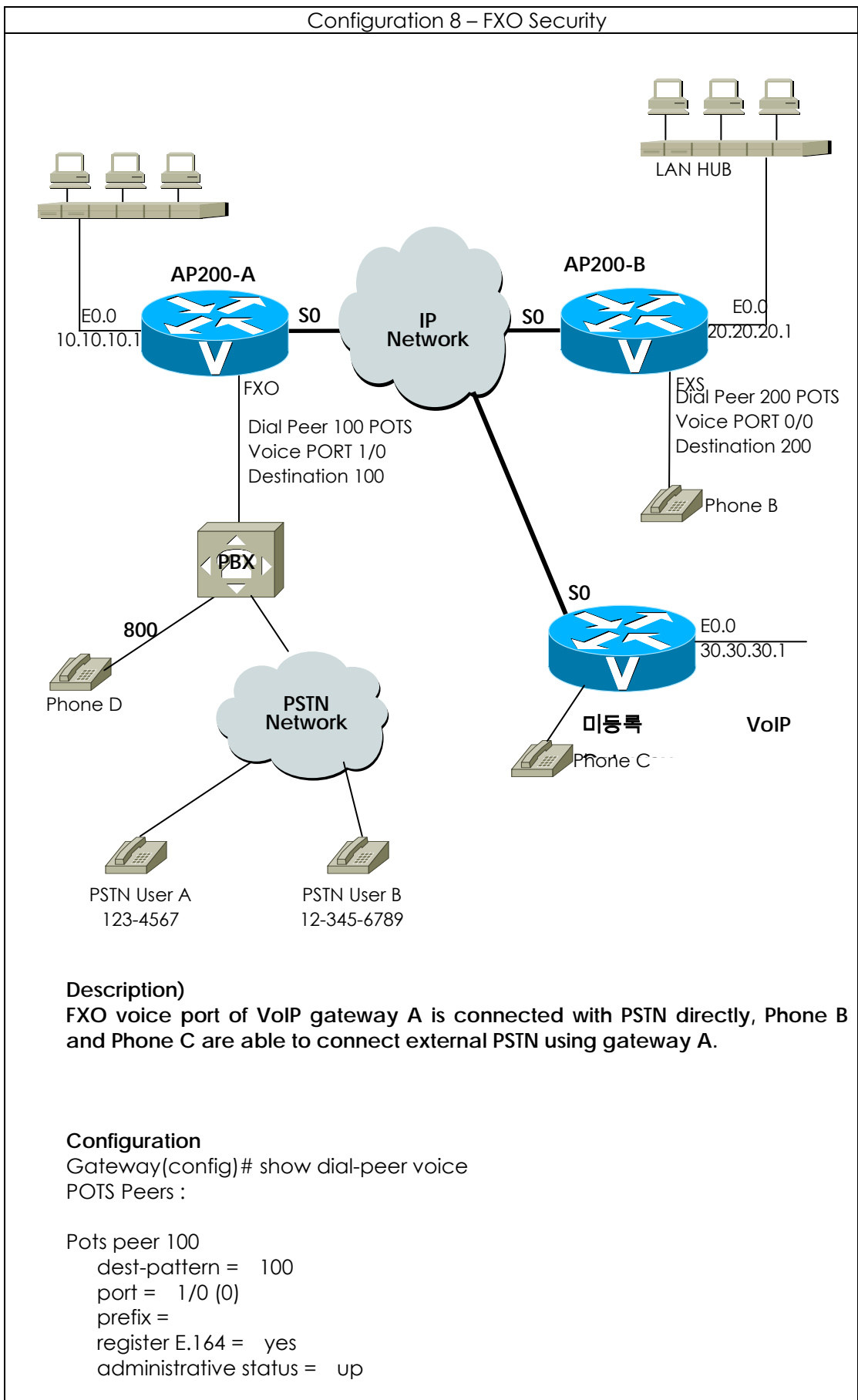
3) Phone A to Phone D

AP200-A Configuration (추가)

```
Gateway (config)# translation-rule 1
Gateway (config-trans-rule-0)# rule 0 8.. 2
Gateway (config-trans-rule-0)# exit
Gateway (config)# dial-p voice 800 VoIP
Gateway (config-dialpeer-VoIP-800)# destination-pattern 8..
Gateway (config-dialpeer-VoIP-800)# session target 40.40.40.1
Gateway (config-dialpeer-VoIP-800)# translate-outgoing called-number 1
```

Call Scenario

- 1) Call from Phone A to Phone B
 - Hang up Phone A.
 - Press dial No. 200 of Phone B
 - Confirm RING sound of Phone B
 - Hang up Phone B
 - Speak over the telephone.
 - Finish speaking out.
- 2) Call from Phone B to Phone C
 - Hang up Phone B.
 - Press dial No. 300 of Phone C.
 - Confirm RING sound of Phone C.
 - Hang up Phone C
 - Speak over the telephone.
 - Finish speaking out.
- 3) Call from Phone A to Phone D
 - Hang up Phone A.
 - Press dial No. 800 of Phone D.
 - Confirm RING sound of Phone D.
 - Hang up Phone D.
 - Speak over the telephone.
 - Finish speaking out.



VoIP Peers :

VoIP peer 200

```
dest-pattern = 2..
session-target = 20.20.20.1
codec = default
codecClass = default
dtmfRelay = h245-alphanumeric
vad = yes
translation-outgoing called-number = -1
translation-outgoing calling-number = -1
description =
administrative status = up
```

Gateway(config)# **voice service VoIP**

Gateway(config-vservice-VoIP)# **no security permit-FXO**

% default is permit-FXO

call scenario)

- 1) call Phone B to Phone D
 - Hang up Phone A
 - Press dial No. 100 of Phone B FXO voice port phone number.
 - Press dial No. 800 of PBX machine internal phone number.
 - Speak over the telephone.
 - Finish speaking out.

- 2) call Phone C to Phone D
 - Hang up Phone C
 - Press dial No. 100 of Phone C FXO voice port phone number.
 - Confirm silence sound
 - Finishing

설정 2) User Class Configuration.

Password check for every call under VoIP gateway A FXO voice port

Configuration)

Configuration

Gateway(config)# show dial-peer voice

POTS Peers :

Pots peer 100

```
dest-pattern = 100
port = 1/0 (0)
prefix =
register E.164 = yes
administrative status = up
```

VoIP Peers :

VoIP peer 200

```
dest-pattern = 2..
session-target = 20.20.20.1
codec = default
codecClass = default
```

```
dtmfRelay = h245-alphanumeric
vad = yes
translation-outgoing called-number = -1
translation-outgoing calling-number = -1
description =
administrative status = up
```

```
VoIP peer 300
dest-pattern = 3..
session-target = 30.30.30.1
codec = default
codecClass = default
dtmfRelay = h245-alphanumeric
vad = yes
translation-outgoing called-number = -1
translation-outgoing calling-number = -1
description =
administrative status = up
```

```
Gateway(config)# voice class user 1
Gateway(config-vclass-user#1)# password 1234
Gateway(config-vclass-user#1)# max-digits 3
Gateway(config-vclass-user#1)# exit
Gateway(config)# voice class user 2
Gateway(config-vclass-user#2)# password 4567
Gateway(config-vclass-user#2)# max-digits 8
Gateway(config-vclass-user#2)# exit
Gateway(config)# voice class user 3
Gateway(config-vclass-user#2)# password 7890
Gateway(config-vclass-user#2)# max-digits 0
Gateway(config-vclass-user#2)# exit
```

call scenario)

- 1) Phone B to PSTN user A using user class 1
 - Hang up Phone B.
 - Press dial No. 100 of Phone B
 - Listen shot dial tone, and then press password No. 1234 as user class 1
 - Listen normal dial tone of PBX machine, and then press dial No. 800
 - Speak over the telephone
 - Finish speaking out.

- 2) Phone B to PSTN user A using user class 1
 - Hang up Phone B
 - Press dial No. 100 of Phone B FXO
 - Listen shot dial tone, and then press password No. 1234 as user class 1
 - Listen normal dial tone of PBX machine, and then press dial No. 9 for external PSTN connection
 - Listen dial tone, and then press dial No. 1234567 for PSTN call.
 - Speak over the telephone
 - Finish speaking out.

- 3) Phone B to PSTN user A using user class 2
 - Hang up Phone B
 - Press dial No. 100 of Phone B FXO voice port
 - Listen shot dial tone, and then press password No. 4567 as user class 2.

- Listen normal dial tone of PBX machine, and then press dial No. 9 for external PSTN connection.
 - Listen dial tone, and then press dial No. 12345678 for PSTN call.
 - Speak over the telephone
 - Finish speaking out.
 -
- 4) Phone B to PSTN user B using user class 3
- Hang up Phone B
 - Press dial No. 100 of Phone B FXO voice port
 - Listen shot dial tone, and then press password No. 7890 as user class 3.
 - Listen normal dial tone of PBX machine, and then press dial No. 9 for external PSTN connection.
 - Listen dial tone, and then press dial No. 123456789 for PSTN call.
 - Speak over the telephone
 - Finish speaking out.

Appendix C AP200 Call Finishing Cause Code

The following table shows description for call finishing cause code and mapping information for Q.931 cause or H.225 cause.

AP200 Call Finishing Cause Code	Call Finishing Master	Reason of Call Finishing	Defined Code
RemoteNoBandwidth	remote side	RELCOM* receiving cause noBandwidth(H225) NoCircuitChannelAvailable (Q931:34)	RELCOM transmission cause H225 destinationRejection
RemoteGatekeeperResourceUnavailable	remote side	RELCOM receiving cause gatekeeperResources(H225) ResourceUnavailable (Q931:47)	RELCOM transmission cause H225 destinationRejection
RemoteUnreachableDestination	remote side	RELCOM receiving cause unreachableDestination (H225) NoRouteToDestination (Q931: 3)	RELCOM transmission cause H225 destinationRejection
RemoteCallClear	remote side	RELCOM receiving cause destinationRejection (H225) NormalCallClearing (Q931: 16)	RELCOM transmission cause H225 destinationRejection
RemoteIncompatibleDestination	remote side	RELCOM receiving cause invalidRevision (H225) IncompatibleDestination (Q931: 88)	RELCOM transmission cause H225 destinationRejection
RemoteNoPermission	remote side	RELCOM receiving cause noPermission (H225) InterworkingUnspecified (Q931: 127)	RELCOM transmission cause H225 destinationRejection
RemoteUnreachableGatekeeper	remote side	RELCOM receiving cause unreachableGatekeeper (H225) NetworkOutOfOrder (Q931: 38)	RELCOM transmission cause H225 destinationRejection
RemoteResourceUnavailable	remote side	RELCOM receiving cause gatewayResources (H225) SwitchingEquipmentCongestion (Q931: 42)	RELCOM transmission cause H225 destinationRejection
RemoteInvalidNumber	remote side	RELCOM receiving cause badFormatAddress (H225) InvalidNumberFormat (Q931: 28)	RELCOM transmission cause H225 destinationRejection
RemoteAdaptiveBusy	remote side	RELCOM receiving cause adaptiveBusy (H225) TemporaryFailure (Q931: 41)	RELCOM transmission cause H225 destinationRejection
RemoteUserBusy	remote side	RELCOM receiving cause inConf (H225) UserBusy (Q931: 17)	RELCOM transmission cause H225 destinationRejection
RemoteUnknown	remote side	RELCOM receiving cause	RELCOM transmission cause

		undefinedReason (H225) NormalUnspecified (Q931: 31) 또는 unspecified reason from remote side	H225 destinationRejection
RemoteCallDeflection	remote side	RELCOM receiving cause facilityCallDeflection (H225)	RELCOM transmission cause H225 destinationRejection
RemoteSecurityDenial	remote side	RELCOM receiving cause securityDenied (H225)	RELCOM transmission cause H225 destinationRejection
RemoteCalledPartyNotRegistered	remote side	RELCOM receiving cause calledPartyNotRegistered (H225) SubscriberAbsent (Q931: 20)	RELCOM transmission cause H225 destinationRejection
RemoteCallerNotRegistered	remote side	RELCOM receiving cause callerNotRegistered (H225)	RELCOM transmission cause H225 destinationRejection
GkCalledPartyNotRegistered	gatekeeper	Gatekeeper ARJ ** cause calledPartyNotRegistered	RELCOM transmission cause H225 alledPartyNotRegistered
GkInvalidPermission	gatekeeper	Gatekeeper ARJ cause invalidPermission	RELCOM transmission cause H225 noPermission
GkRequestDenied	gatekeeper	Gatekeeper ARJ cause requestDenied	RELCOM transmission cause H225 noPermission
GkUndefinedReason	gatekeeper	Gatekeeper ARJ cause undefinedReason	RELCOM transmission cause H225 undefinedReason
GkCallerNotRegistered	gatekeeper	Gatekeeper ARJ cause callerNotRegistered	RELCOM transmission cause H225 callerNotRegistered
GkRouteCallToGatekeeper	gatekeeper	Gatekeeper ARJ cause routeCallToGatekeeper	RELCOM transmission cause H225 unreachableGatekeeper
GkInvalidEndpointIdentifier	gatekeeper	Gatekeeper ARJ cause invalidEndpointIdentifier	RELCOM transmission cause H225 undefinedReason
GkResourceUnavailable	gatekeeper	Gatekeeper ARJ cause resourceUnavailable	RELCOM transmission cause H225 gatekeeperResources
GkSecurityDenial	gatekeeper	Gatekeeper ARJ cause securityDenial	RELCOM transmission cause H225 securityDenied
GkQosControlNotSupported	gatekeeper	Gatekeeper ARJ cause qosControlNotSupported	RELCOM transmission cause H225 gatekeeperResources
GkIncompleteAddress	gatekeeper	Gatekeeper ARJ cause incompleteAddress	RELCOM transmission cause H225 badFormatAddress
GkAliasesInconsistent	gatekeeper	Gatekeeper ARJ cause aliasesInconsistent	RELCOM transmission cause H225 undefinedReason
GkDisengageRequested	gatekeeper	Gatekeeper DRQ	RELCOM transmission cause H225 undefinedReason
LocalCallClear	local side	Hang on in normal local voice port	RELCOM transmission cause H225 destinationRejection
LocalResourceUnavailable	local side	Required local resources (exceed Max. opening call processing)	RELCOM transmission cause H225 gatewayResources
LocalPortBusy	local side	busy condition on local voice port	RELCOM transmission cause H225 inConf

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LocalPortNoConnect	local side	No response voice port (ringing timer expired)	RELCOM transmission cause H225 destinationRejection
LocalPortShutdowned	local side	shutdown condition on local voice port	RELCOM transmission cause H225 unreachableDestination
LocalPeerShutdowned	local side	shutdown condition on local dial peer	RELCOM transmission cause H225 unreachableDestination
LocalInterdigitTimerExpired	local side	Local inter-digit timer expired	N/A
LocalSecurityDenial	local side	Call finishing by local security	RELCOM transmission cause H225 securityDenial
LocalInvalidGatekeeperRoute	local side	Local gateway decide abnormal condition on transport address from receiving gatekeeper.	RELCOM transmission cause H225 unreachableGatekeeper
LocalUnreachableGatekeeper	local side	Local gateway could not operate call processing due to registration failure in gatekeeper.	RELCOM transmission cause H225 unreachableGatekeeper
LocalUnreachableDestination	local side	Local gateway connecting failure on other side gateway	N/A
LocalNoAnswerFromDestination	local side	Local gateway receiving message failure from other side gateway(T303 Expired)	N/A
LocalNoConnectFromDestination	local side	Local gateway CONNECT message receiving message failure from other side gateway (T301 Expired)	RELCOM transmission cause H225 destinationRejection
LocalUnknown	local side	Local unknown reason	RELCOM transmission cause H225 undefinedReason
LocalProtocolError	local side	Message & protocol error on local side	RELCOM transmission cause H225 undefinedReason
LocalInvalidNumber	local side	Invalid number on local side	RELCOM transmission cause H225 badFormatAddress
LocalT38FaxError	local side	T.38 fax error on local side	RELCOM transmission cause H225 undefinedReason
LocalManagement	local side	Call finishing by management on local side	RELCOM transmission cause H225 undefinedReason
LocalUnavailableDestination	local side	Call finishing by destination invalid on local side (Ex. Call for FXO – FXO, Call for H323 – H323)	RELCOM transmission cause H225 undefinedReason
LocalAbortedDestination	local side	Local gateway aborting call connection to other side gateway	N/A
LocalCapabilityNegotiationFail	local side	Local gateway capability negotiation failure to other side gateway	RELCOM transmission cause H225 undefinedReason

*RELCOM : Q.931 Release Complete message

**ARJ : H.225 Admission Reject message

For your reference, the following table shows recommendation of ITU-T for H.225 cause and Q.931 cause mapping for H.323.

H225 Cause	Q931 Cause
noBandwidth	NoCircuitChannelAvailable (34)
gatekeeperResources	ResourceUnavailable (47)
unreachableDestination	NoRouteToDestination (3)
destinationRejection	NormalCallClearing (16)
invalidRevision	IncompatibleDestination (88)
noPermission	InterworkingUnspecified (127)
unreachableGatekeeper	NetworkOutOfOrder (38)
gatewayResources	SwitchingEquipmentCongestion (42)
badFormatAddress	InvalidNumberFormat (28)
adaptiveBusy	TemporaryFailure (41)
inConf	UserBusy (17)
undefinedReason	NormalUnspecified (31)
facilityCallDeflection	NormalCallClearing (16)
securityDenied	NormalUnspecified (31)
calledPartyNotRegistered	SubscriberAbsent (20)
callerNotRegistered	NormalUnspecified (31)

Appendix D Cable Specifications

This Appendix provides information about the Pinout specifications of the following cables used with the VoiceFinder 1100 Gateway.

- Console Port Signal and Pinout(RJ-45 to DB9)
- Ethernet Cable Assemble(RJ-45 to RJ-45) Pinout

[Console Port Signal and Pinout]

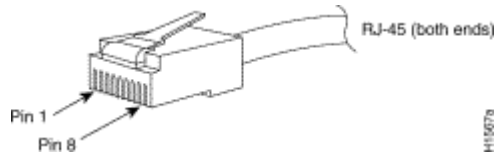
In order to connect the Gateway console port with the Terminal Emulating PC, the RJ-45 to DB9(Female DTE Connector) cable is used. The transferred signal and Pinout specifications are enlisted in the following Table C-1 "Console Port Signal and Pinout".

Gateway Console (DTE)	RJ-45	DB-9	Console Device (PC)
Signal	RJ-45 Pin	DB-9 Pin	Signal
RTS	1	8	CTS
DTR	2	6	DSR
TxD	3	2	RxD
GND	4	5	GND
GND	5	5	GND
RxD	6	3	TxD
DSR	7	4	DTR
CTS	8	7	RTS

Table D-1 "Console Port Signal and Pinout"]

[Ethernet Cable Assemble(RJ-45 to RJ-45) Pinout]

In order to connect the Gateway with other equipments (i.e. HUB), the RJ-45 to RJ-45 Ethernet Cable is used. The RJ-45 Connector Pin sequence is provided in Diagram C-1 and the transferred signal and Pinout specifications are enlisted in Table C-2 "Serial Ethernet Cable Signal and Pinout".



[Diagram D-1 10Base-T RJ-45 Connector]

RJ-45	Signal	Direction	RJ-45 Pin
1	Tx +	→	1
2	Tx -	→	2
3	Rx +	←	3
4	-	-	4
5	-	-	5
6	Rx -	←	6
7	-	-	7
8	-	-	8

1. These specifications are for serial cables connecting the Gateway and the HUB.
2. For Gateway to Gateway or Gateway to PC connection, the Cross Cable must be used.

[Table D-2 Serial Ethernet Cable Signal and Pinout]