VoiceFinder

VoIP Gateway

Configuration Guide

APOS 2.0 (G2)

AddPac Technology, Co. Ltd.

Note

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Preface – About This Guide

The APOS 2.0 (APOS G2) voice configuration guide collects in one place information that you need to implement APOS Release 3.1 voice features. It is organized in the following chapters:

- Chapter 1. 『**Overview**』
  This chapter describes its features and lists the hardware and software specifications of the AddPac VoiceFinder gateways.

- Chapter 2. 『**Configuring VoiceFinder Gateways for Operation**』
  This chapter describes how to use commands and all the related setting modes by presenting configuration examples to configure AddPac VoIP gateways as in a way to be suitable to the user's environment and interface. This chapter is very important and highly recommended to be studied thoroughly.

- Chapter 3. 『**Voice Port Configuration**』
  This chapter describes the type of connection being made and the type of signaling to take place over this connection. In addition to the commands for basic configuration, there are also fine-tuning for voice quality, enable special features. This chapter is very important and highly recommended to be studied thoroughly.

- Appendix A 『**Reason Code Value for H. 323 Call Termination**』
  This appendix explains the reason code value for H.323 termination of VoiceFinder Gateways and mapping Q.931 and H.225 call signaling and communication between endpoints (call signaling) and the gatekeeper.

- Appendix B 『**Reference Documents**』
  This appendix is organized with all the related RFC/OUT-T documents for SIP/H.323/MGCP of VoIP protocol and TCP/IP protocol

- Appendix C 『**Cable Specifications**』
  This appendix explains of console cable, V.35 cable and Ethernet cable specifications to be used for the gateways

- Appendix D 『**Acronyms and Glossary**』
  The acronyms and glossary of VoIP are organized in alphabetical order

The revision history of the VoiceFinder APOS Configuration Guide is listed as to follow:

Release No.		Revision	Prepared by
Release 1.0		Initial Released	AddPac R&D
Release 1.1		Added commands and revised	AddPac R&D
Release 1.2		Added commands and revised	AddPac R&D
Release 2.0		Added commands and combined the manuals	AddPac R&D
Release 3.0		Added commands (8.10/8.23/8.30)	AddPac R&D
Release 3.1		Release of APOS2.0 added commands and revised	AddPac R&D

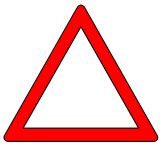
[Document Conventions]

This publication uses the following conventions to convey instructions and information:

Convention	Description
boldfast font	Commands and keywords
Italic font	Variable for which you supply values
[ ]	Keywords or arguments that appear within square brackets are optional
{x| y| z}	A choice of required keywords appears in braces separated by vertical bars. You must select one.
< >	This is the required variables to be replaced by numbers

[Safety Warnings]

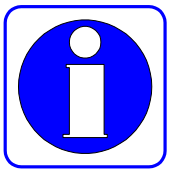
Safety warnings appear throughout this publication in procedures that, if performed incorrectly, might harm you. The following warning symbols precede each warning statement.

Danger 	This warning symbol means *danger*. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.
Warning 	This symbol means that if you do not follow the procedure presented with this symbol in this guide may result in a damage of the equipment or data loss.
Caution 	This symbol calls for the user's attention to be careful. If the user doe not follow the procedure presented with this symbol in this guide and misuses the product, the damage in the software or data loss or loss of system configuration can be resulted.

Chapter 1 . Overview

1.1. VoiceFinder Gateway Series

Information



The AddPac VoiceFinder Gateway Series are the voice over IP gateways which allow using a telephone at a low cost or even for free by supporting the voice communication using Internet and leased-line in enterprise (head office and branch), public office and Small and Medium Business (SMB) environment.

This gateway uses the latest voice compression and QoS algorithms of AddPac Technology's proprietary that allows maintaining the best voice quality regardless of whether the network is broadband or narrow band. This gateway has the various voice interface modules including FXS, FXO, E&M and digital E1/T1 that are suitable to the user's demands and provides a great flexibility to respond to the environmental changes of the user's network and protects the investment.

The AddPac VoiceFinder Gateways can be used in various network environments such as leased line, ADSL and cable modems networking with fixed and dynamic IP environments. And the gateway supports various network protocols such as static, RIP v1/2, OSPF v2 routing function and Internet application functions such as NAT/ PAT. Especially in the dynamic network environment, the VoIP and IP sharing platform provides the most economical and efficient solutions in the broadband networks.

Also the VoiceFinder gateways are interoperable with the major vendor's gatekeepers and large-scale gateways. The VoiceFinder gateways are easy and simple to use, operate and maintain. The gateway provides the voice integrated service solutions.

The gateways can support firewall in 2 ways, packet filtering and access list and limit the access from the outside network by using sauce and destination address information.

Also the gateways can allocate IP addresses automatically to the network clients below the router by using Dynamic Host Configuration Protocol (DHCP) and Network Address Translation (NAT) allows the gateways to solve the shortage problem of IP address due to an explosive increment of the users. At the same time, the internal IP address is hidden from the outside for the enhancement of security features.

The following figure is an example of the network using the VoiceFinder Gateways.

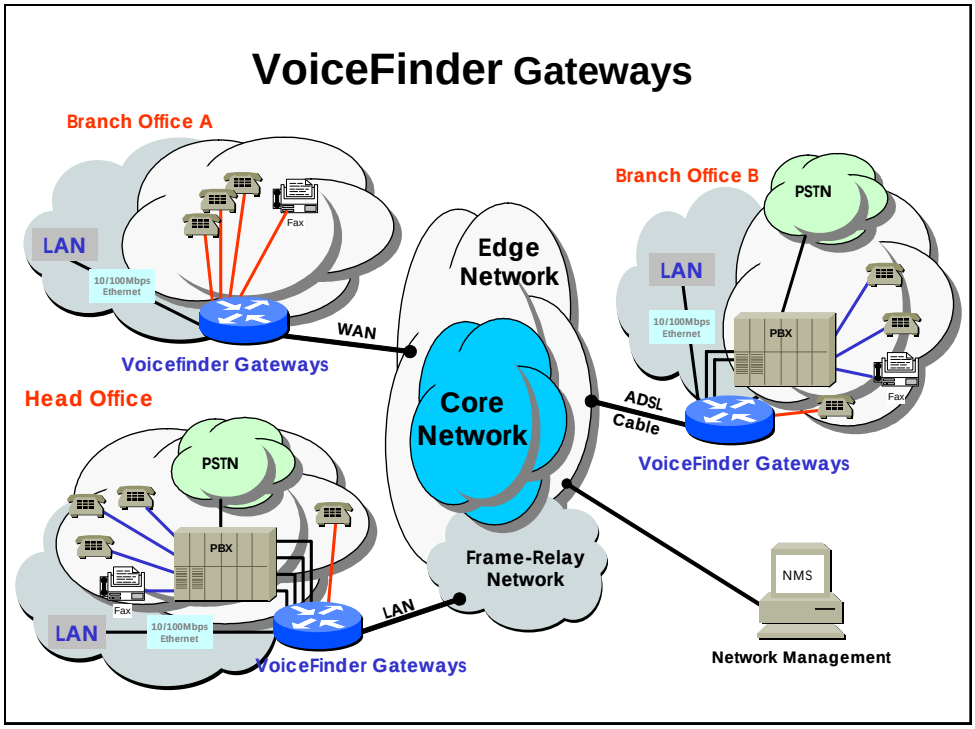
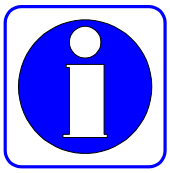


Figure 1.1 Network Configuration Example of VoiceFinder Gateways

1.2. Main Features

Information The main features and technical specification are described in the followings:



Network / Voice Interface

The AddPac Voice Finder Gateway Series have been designed with the system architecture which can provide enriched system memory and diversified voice interfaces.

- High performance VoIP service solution integrated with data/voice
- Hardware design of Extensible Modular Structure
  (Except stand alone types such as AP160,AP200,AP1100,AP1200)
- High Performance 32bit RISC Microprocessor
- 2 or 4-Voice Network Module independently (depending on product models)
  (Refer to the hardware specifications of the relevant product model)
- Fixed **1-Port 10 or 100Mbps Fast Ethernet Interface** for LAN Service (RJ45)
  (Refer to the hardware specifications of the relevant product model)
- Fixed **1-Port 10 or 100Mbps Ethernet Interface** for WAN Side Connection (RJ45)
  (Refer to the hardware specifications of the relevant product model)
- Fixed 1-Port Asynchronous Serial Interface for Console Port (RJ45)
  (Refer to the hardware specifications of the relevant product model)
- Optional 4-Ports FXS Voice Processing Network Module (4 x RJ11)
  (Refer to the hardware specifications of the relevant product model)
- Optional 4-Ports FXO Voice Processing Network Module (4 x RJ11)
  (Refer to the hardware specifications of the relevant product model)
- Optional 4-Ports E&M Voice Processing Network Module (4 x RJ11)
  (Refer to the hardware specifications of the relevant product model)
- Optional 2-Ports FXO and 2-Ports FXS Voice Processing Network Module (4 x RJ11)
  (Refer to the hardware specifications of the relevant product model)
- Optional 1-Ports Digital E1 ISDN-PRI/R2/DTMF Processing Network Module (1 x RJ48)
  (Refer to the hardware specifications of the relevant product model)
- Optional 1-Ports Digital T1 ISDN-PRI/R2/DTMF Processing Network Module (1 x RJ48)
  (Refer to the hardware specifications of the relevant product model)
- 1U x 19” Rack Mountable Standard Chassis
  (Refer to the hardware specifications of the relevant product model)

- AC Power Supply Unit (Free Voltage)
- Various System LED indicator

IP Routing Protocols

The following are the specifications for IP routing protocol supported by the Voice Finder VoIP Gateway:

- Static, RIP v1/v2, OSPF v2 Routing Protocol
- Transparent Bridging (IEEE Spanning Tree Protocol)
- IEEE802.1Q VLAN Routing

Voice over IP Service

The specification of Voice over IP features are listed as to follow:

- ITU-T H.323 v3 VoIP Protocol
- ITU-T H.235 Security Feature
- ITU-T H.323 Gateway, Gatekeeper
- Session Initiation Protocol (SIP)
- MGCP Protocol
- H.323, SIP, MGCP Concurrent Triple Stack
- G.723.1, G.729.A, G.711 Voice Compression
- Various Voice Processing Feature
  - ✓ VAD(Voice Activity Detection)
  - ✓ T.38 G3 FAX Relay(In-band and Out-band)
  - ✓ DTMF(Dual Tone Multi Frequency)
  - ✓ CNG (Comfort Noise Generation)
  - ✓ G.168 Echo Cancellation
- Enhanced QoS Management Features for Voice Traffic

WAN Service

The VoiceFinder VoIP Gateway Series supports the following WAN features:

- Point-to-Point Protocol over Ethernet (PPPoE)
- ADSL (Static IP and dynamic IP) and Cable Modem (DHCP)

Network Managements

The Gateway Series provide various network management features for systematic equipment management as to follow:

- Interoperation with AP-VPMS for systematic equipment management
- Standard SNMP Agent and standard MIB II, Bridge MIB
- Console feature through asynchronous port
- Telnet and login enabling remote control
- QoS through traffic queuing
- Web-based network management

Security Functions

The gateways also support diverse security features as to follows:

- Standard & Extended IP Access List for network security
- Enable/Disable a specific network protocol
- Account management features for multi-level users
- Auto-disconnect for Telnet/Console Sessions
- PPP User Authentication (PAP and CHAP)

Operation and Managements

The gateways support the following operation and Management:

- System Performance Analysis for Process, CPU, Connection I/F
- Configuration Backup & Restore for APOS Managements
- Debugging, System Auditing, and Diagnostics Support
- Diagnostic system enabling network packet analysis
- Debussing call process
- System Booting and Auto-rebooting with Watchdog Feature
- IP Traffic Statistics with Accounting
- IP Traffic Statistics with Accounting

Other Scalability Features

The gateway support the other scalability features as to follow:

- DHCP Server & Relay Functions
- Network Address Translation (NAT) Function
- Remote Upgrade for APOS Management using FTP/TFTP
- Cisco Style Command Line Interface(CLI)
- Network Time Protocol (NTP)

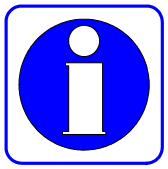
Interoperability Features

The AddPac VoiceFinder Gateway Series ensures reliable network interoperability of mutual operation with other major vendors' switches and voice gateway equipment

- CISCO AS5300 Series, CISCO 2600/3600 Series (H.323, SIP)
- Cisco Call Manager (MGCP)
- Xener System Softswitch (H.323, MGCP, SIP)
- 3Com Total Control Series(H.323, SIP)
- Clarent Gateway 3.0 Series (H.323)
- Soners Softswitch (H.323, SIP)
- NEC Softswitch (SIP)
- NTT Softswitch (SIP)
- major vendors' VoIP gateways, gatekeepers including Lucent

1.3. APOS Internetworking Software

Information



This section provides and an overview of AddPac Operating System (APOS) Internetworking Software as to follow:

APOS Internetworking Software for AP Router, Gateway

APOS Internetworking Software has been designed with the latest Embedded Real-time Operation System and implemented with the architecture based on continuous scalability of advanced networking software, outstanding reliability and stability and Quality of Service (QoS). In addition APOS internetworking software has been developed with the system architecture of Building Block concept which provides easy-to-upgrade for integrating various types of network interface or additional network protocols.

Industrial Standard Network Protocol Stack

APOS Internetworking Software supports the industrial standard network protocol stack. This protocol stack includes those protocols which can support the data networking for WAN/LAN and ATM and network management or security, VPN and other various protocols are included.

Integrated Networking Solution

APOS Interworking Software does not just support only data networking but also Voice over Internet Protocol: it provides the solution which can inter-work, with the different network infra-structure such as VoIP which is integrated with voice and data, ATM, Frame-Relay, ISDN and PSTN.

Optimized Performance and Functionality

APOS Internetworking Software provides the outstanding data processing capability and idea; bandwidth control ability for the complex traffic. This implementation of maximized functions can be used with the outstanding solution to design an ideal network together with many mandatory functions supported by APOS.

Easy to use, Easy to Install and Maintain

APOS Internetworking Software guarantees simple and compatible interoperability by using the industrial standard command for the user's convenience. In addition, maintenance and operations are easy with Web-based Management and remote management.

Chapter 2 . The Gateway Configuration and Its Commands

This chapter describes how to configure VoiceFinder Gateways and explains the commands.

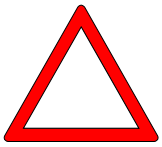
2.1. Booting the Gateway

This chapter describes how to configure VoiceFinder Gateways and explains the commands. All the commands for configuring the gateways can be used by accessing Telnet or connecting to console.

After the power is turned on, the gateways go through the following process:

- The gateways go through self-testing process then check their basic operation of CPU, memory and interface.
- After the Boot Loader is performed, the gateways look for the software image file. At the default configuration, the gateway is to load the software in the flash memory
- If the gateways can not find the software image file from the flash memory, the stand-by at the boot mode until they can download the proper software (at this time, FTP or TFTP protocol can be used to download the proper software for the gateways).
- The gateways operate basing on the setting information which is saved after the software is downloaded.

Caution



When the system is booted for the first time, the gateways perform the settings and these settings must be saves by the commands of ‘write’ or ‘copy running-config’.

After the booting is processed normally, you can see the following message:

```
System Bootstrap, Version 1.2
Decompressing the image:
########[OK]

System Boot Loader, Version 2.4.0/2. Board Rev 0
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

Ethernet port initialization complete
The "BOOT LOADER" is ready
BOOT_login:

System Bootstrap, Version 1.2
Decompressing the image:
############################################################
############################################################
################################################[OK]

VoiceFinder AP100 Series (AP100_G2)
Serial Number: AP100_G2-fffe7e
32BIT RISC Processor With 112MHz Clock
16 Mbytes System Memory.
512 Kbytes System Boot Flash Memory
2 Mbytes System Flash Memory

1 RS232 Serial Console Interface

AP100_G2 System software Revision 8.41.100
Released at Wed Nov  7 21:27:13 2007
Program is 1724824 bytes, checksum is 0xdd1b378

UTC Time is Thu Jan  1 00:00:00 1970
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

Allocating system mbuffer counter: 256
Loading file system(ver2.2), flash-base: 0xb01f0000 ram-base:
```

```
0x948e5098
Ethernet port initialization complete
Ethernet port initialization complete
System utilization reference (14/14/14/15)
Attach FastEthernet Interface at Slot 0, Port 0-1, <0-0>/<0-1>
Interface FastEthernet0/0, changed state to DOWN
Interface FastEthernet0/1, changed state to DOWN
Hardware Revision ID = 0
Slot (0) Module type : FXS

can't open configuration file [flash:/apos.cfg]
RTA Module Ready
Start HTTP Server (listen tcp/80)

Press RETURN to get started.

Start Target Debug Server
CPU internal DSP SRAM .... OK
Audio DSP S/W download ... OK

VoipGateway::Init1  - No IP address on the VoIP Interface

Welcome, APOS(tm)  Kernel Version 8.41.100.
Copyright (c) 1999-2006 AddPac Technology Co., Ltd.

Login:
```

2.2. Command Instructions

This chapter describes how to configure VoiceFinder Gateways and explains the commands.
All the commands for configuring the gateways can be used by accessing Telnet or connecting to console.

After the power is turned on, the gateways go through the following process:

- The gateways go through self-testing process then check their basic operation of CPU, memory and interface.

- After the Boot Loader is performed, the gateways look for the software image file. At the default configuration, the gateway is to load the software in the flash memory

- If the gateways can not find the software image file from the flash memory, the stand-by at the boot mode until they can download the proper software (at this time, FTP or TFTP protocol can be used to download the proper software for the gateways).

- The gateways operate basing on the setting information which is saved after the software is downloaded.

The gateway command marked with the asterisk mark '*' is not currently supported, but it will be supported in the near future.
The commands that are related to IPv6 can be applied for the product which supports IPV6. Some products can be applied with IPv6 in APOS 2.0 version.

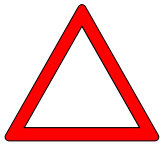
If there is no corresponding command, the product does not support the function.

Example) (name of the product model)# clear ?

- arp-cache Clear the entire ARP cache
- cdp CDP information
- counters Clear counters on one or all interfaces

- h323 [VoIP] Clear H323 call
- ip IP information
- ipv6 IPv6 information
- system APOS specific information
- utilization Clear system usage information
- voice-port [VoIP] Clear call on voice port

Caution



'No' command can be used to cancel the command which has been set already. In case of canceling the command with the default value by using 'no' command, the optional values of the command are to be set back to the default value.

Example) *(name of the product model)(config)# no ?*

- *access-list* *Add on access list entry*
- *access-list-all* *Add on access list entry*
- *arp* *Modify ARP table parameters and entry*
- *banner* *Set banner string*
- *call-diversion* *[VoIP] Remove call diversion profile*
- *cdp* *CDP information*
- *clock* *Configure time-of-day clock*
- *spe-id* *reset cpe-id*
- *debug* *debugging control*
- *dhcp* *Enable DHCP server or relay*
- *dial-peer* *[VoIP] Remove Dial Peer*
- *dialpattern-group* *[VoIP] Remove Dial Pattern Group*
- *dns* *host aliases*
- *ems-server* *[VoIP] Config EMS server*
- *enable* *Modify enable password parameters*
- *ftp* *file Transfer Protocol*
- *hostname* *Reset system's network names*
- *http* *Enable HTTP*
- *interface* *Select an interface to configure*
- *ip* *IP information*
- *ipv6* *IPv6 information*
- *key* *Authentication key management*

- *logging* *Modify message logging facilities*
- *mount* *Mount File system device*
- *num-exp* *[VoIP] Remove Number Expansion*
- *radius-server* *[VoIP] Config RADIUS server*
- *route-map* *Create route-map or enter route-map command mode*
- *script* *APOS script string*
- *serial* *Set system's configuration serial string*
- *service* *Set up miscellaneous service*
- *snmp* *Config a SNMP parameters*
- *system* *Set system parameter*
- *telnet* *Telnet port*
- *translation-rule* *[VoIP] Remove translation rule*
- *username* *Establish User Name Authentication*
- *utilization* *utilization*
- *voice* *[VoIP] Reset Voice class or service configuration*
- *voip-interface* *[VoIP] Set VoIP interface and address*

2.2.1. User Mode Commands

All the commands for configuring VoiceFinder Gateway can be used by accessing to Console or Telnet terminal (VT-100 terminal). And these commands can be used by Web-based (HTTP) management.

In the commands, there is the user mode for accessing data network, administrator's mode for looking up the configuration status or debugging and the configuration mode for changing the environment settings or create the new ones.

The followings are the attributes of commands for configuring VoiceFinder Gateways:

- You do not have to enter all the command letters and just entering a part of the command can be recognized automatically. For instance, if you want to enter 'show' command, just entering 'sh' or 'sho' can be automatically recognized as 'show'.

- On-line help provides the list of commands with usage sentences when a wrong command is entered.

- For the messages that can take more than one screen to be displayed, 'more' command is used to display the rest of messages in each additional screen.

- 'Help' and '?' command can used to see description of the command.

- There are 3 different types of modes for the gateway commands. The commands for each mode can be described in the following:

2.2.2. Administrator Mode Commands

The administrator mode command is used for the administrator only who logs into the gateway. To use this command, you must log in to the gateway by the root account. An entrance to the configuration of the gateway is possible only when you log in as administrator mode.

At the administrator, all the commands in the user mode can be used.

The prompt for the administrator mode can be displayed as '(product model)#'.

Command	Description
auto-upgrade	Sets to upgrade the image by using HTTP
clear	Initializes the initial counter and statistics
clock	Sets the present year, date and time
configure	Enters to the configuration mode
copy	Copies running config to startup config
debug	Debugs the overall system
disable	Enters to the user mode
disconnect	Closes VTY connection
dnsquery	Used for DNS Query test
dnsrv	Used for DNS SRV Record Test
end	Enters to the administrator mode
erase	Deletes config file
exit	Moves to a notch previous mode from the present
fsh	Enter File Shell
help	Displays APOS help screen
no	Deletes the present configuration
nsupdate	Transmits updated information to Name Server
ntpdate	Receives the clock information from ntp server
ping	Checks the network connection (IPv4)
ping6	Checks the network connection (IPv6)
quit	Equals to exit
reboot	Reboots the system
show	Checks the present status of the configuration
telnet	Connects remotely
terminal	Sets to display the terminal and debussing information
tftp	Transmits the file by tftp
traceroute	Checks the route (IPv4)
traceroute6	Checks the route (IPv6)
who	Displays a user's information who is currently connected
whoami	Displays a user's information for a terminal who is currently connected
write	Saves the present configuration

2.2.3. Configuration Mode Commands

Only the user with the root can access to the configuration mode. In this mode, the user can change the configuration. Largely the mode can be divided by the interface and general configuration mode.

The prompt of the general configuration mode can be displayed as 'product model name (config)#'. In this mode, the user can configure all the settings except the ones related to interface. In the interface mode, the user can configure the settings related to interface such as IP address, WAN protocol.

The prompt of the interface configuration mode can be displayed as 'product model name (config-if)'.

2.2.3.1. General Configuration (config) Commands

Command	Description
access-list	Creates access-list. The range of #1~99 is the standard access-list and #100~199 is the extended access list. Also the expanded range can be supported.
application	Configures the VoIP call termination cause value mappings.
arp	Deletes and adds a particular Ethernet address from ARP table. Also performs ARP table.
auto-upgrade	Sets to download the firmware and script file by using HTTP.
bridge	Configures the settings related to bridge.
call-diversion	Configures call-division.
cdp	Global CDP configuration subcommands
clock	Configures System Time of the gateway
console	Sets the serial console
controller	Configures the settings for E1/ T1 interfaces
dhcp	Configures settings for DHCP server and relay

dialpattern-group	Configures dial pattern group
dial-peer	Configures dial-peer as for VoIP command
dns	Configures the setting for DNS server
ems-server	Configures connection to AP-VPMS
exit	Returns to the previous mode
gatekeeper	Used for operating with H.323 Gatekeeper. It only works with the product that supports the gatekeeper.
gateway	Configure the settings for the voice gateway as for VoIP command
hostname	Changes a name of the gateway from the network
http	Configures the settings for HTTP server
Interface	Enters to the interface configuration mode or creates a logical interface
ip	Enables IP routing
ipv6	Configures the settings for IPv6 and others
ip-tos	Sets a value of IP Type of Server Field
key	Sets the authentication key for the routing protocol
logging	Changes or configures message logging
mgcp	Sets MGCP connection
no	Cancels the command which has been entered or return to the default value
num-exp	Configures VoIP settings of Phone Number Extension
radius-server	Configures the settings for connection with RADIUS server
recovery	Configures the settings for password recovery
remote-log	Configures the settings for syslog server
script	Configures the settings for auto-upgrade/ntp server
serial	When the settings of auto-config are used, it determines a serial number for the setting information.
sip-ua	Enters to the setting mode related to SIP User Agent
snmp	Configures the settings for SNMP
sscp	Configures the settings for SSCP Protocol related. The usage of this command is limited to the product that can support SSCP only.
translation-rule	Configures the setting for translation rule

username	Changes or registers a gateway user
utilization	This is an operation to set a time interval for checking the usage ratio of CPU, Ethernet, serial and others
voice	Configures the settings for VoIP service or codec
voice-port	Configures the setting for VoIP port
voip-interface	Configures the setting for VoIP interface
write	Save the present configuration

2.2.4. Network interface Configuration Commands

2.2.4.1. Ethernet interface Commands 1

Interface configuration mode assigns a particular interface for the settings of the configurations mode. The gateway has 1 Ethernet interface for the uplink and another Ethernet interface for the down link.

Commands	Description
bandwidth	Set the bandwidth informational parameter to kilobits
bridge-group	Specify the bridge parameter.
cdp	Enable CDP of the interface
description	Describe the text in interface configuration mode
encapsulation	Configure the encapsulation for a designated interface (supporting Ethernet, IEEE 802.1q VLAN, IEEE802.3 Encapsulation)
exit	Return to the previous mode
end	Return to the beginning mode
full-duplex	Set Ethernet to full- duplex
half-duplex	Set Ethernet to half-duplex
Interface	Configure another interface
Ip	To configure settings of IP service related and IP protocol.
ipv6	To set the settings of IPv6 service related and IP protocol
no	Cancel the command which has been entered or returns to the default value.
mac-address	Change a value of mac-address of a designated interface
multicast	Configure a designated interface to multicast flag
multicast-all	Configure a designated interface to receive all the multicast packets as a user's command
ppp	Specify a value for accessing Point-to-Point Protocol in interface configuration mode
peer	Allocate the addresses while PPP is in operation
pppoe	Specify a value for accessing PPP over Ethernet

qos-control	Specify a value of QoS traffic control of a designated interface.
shutdown	Perform the administrative down to a designated interface
speed	Specify a physical speed of a designated link

2.2.4.2. Ethernet interface Commands 2 (IP/IPv6 Configuration Mode)

IP related commands can be configured from the assigned interface. The prompt can be shown as product model name (config-if)#. The commands listed below can be shown by using (config-if)# ip ?.

Command	Description
access-group	Apply the access-list, which has been configured from global configuration mode, to an interface
accounting	Apply IP Account List to a designated interface
address	Change or configure IP/IPv6 address for a designated interface
dhcp	Specify DHCP configuration for interface
exit	Return to the previous configuration mode
nat	Enter NAT interface configuration mode
nd	Specify a default value for Pv6 ND (Neighbor Discovery)
mtu	Configure IP MTU of a designated interface
policy	Configure an ip route-map
policy-group	Set the policy which has been applied to a designated interface
route-cache	Specify an option whether to apply route-cache to a designated interface
tcp	Specify a MSS value of TCP header

2.2.4.3. Ethernet interface Commands 3 (PPP Configuration Mode)

This is the command to be configured for the designated interface with ppp encapsulation from

interface Fasrtethernet configuration mode.

Command	Description
accm	Configure the text map to be used for Async port
authentication	Configure the authentication method of ppp link
ccp	Enable PPP CCP
chap	Set a value of CHAP authentication
lcp	Enable PPP Link control negotiation
ipcp	Request for ipcp option parameters
Ipv6cp	Request for option parameters of IPv6CP
pap	Configure PAP authentication parameters+

2.2.5. VoIP Configuration Commands

2.2.5.1. voice service voip Commands

In this mode, VoIP related global configuration can be configured. The (config-vservice-voip)# prompt will be shown by typing "voice service voip" command at global configuration mode.

Commands	Description
accept-fse-at-connect	Accept only fast start element at CONNECT message. This prevents early listen of inband ring back tone
announcement	Enables a voice announcement of the voice-port
busyout	Enable busyout monitoring
cdr	Set syslog CDR (Call Deatail Record) format
counter	Set counter values
default	Set default values
delayed-connect	Enable delayed connect on FXO
display	Set display name option either H.323 id or E.164 address
dynamic-payload-type	Set dynamic payload type
end	Return to the beginning (exec)
exit	Return to the previous mode
fax	Set fax protocol and rate
force-h245address-at-setup	Set H.245 address at H.323 SETUP message when tunneling is disabled
force-starth245	Send startH245 message explicitly when H.323 tunneling is disabled
h323	Set H.323 specific mode settings (i.e., fast start/slow start, response message, H.245 tunneling mode, channel open mode)
ignore-dtmf-abcd-tone	Set ignoring ABCD DTMF tone
ignore-reverse-channel-info	Set ignoring reverse fast start element on ACK
inband-ringback-tone	Enable transmit inband ringback tone when receiving a voip call

local-ringback-tone	Enable pseudo ring back tone
max-call	Set maximum number of call limit
max-frame	Set maximum number of audio frames per Tx packet
minimize-voip-ports	Minimize UDP/TCP port range using for VoIP
no	Cancel the entered command and return to default value
modem	Set modem passthrough
qos-threshold	Set threshold value of delay, jitter, packet loss for sending SNMP trap
quit	Exit current mode and down to previous mode
remote-log	Configure remote syslog server for call logging
rtp-nat-pat	Set RTP NAT/PAT configuration
security	Set security parameters
hold-tone-play	Set tone play for hold call
static-jitter-buffer	Set static jitter buffer for RTP
timeout	Set timeout value
timing	Set timing value
translate-voip-incoming	Set number translation rule for VoIP incoming call
ttl-margin	Set TTL (Time To Live) margine
voip-response-on-pstn	Set VoIP response mode when call is on PSTN
write	Save the present configuration

2.2.5.2. voice-port Commands

To enter voice-port configuration mode, use the **voice-port** command in global configuration mode. Depending on the model, the gateway can have a number of FXS/FXO/E1/T1/E&M ports.

Commands	Description
announcement	Enables a voice announcement of the voice-port
busyout	Configure an action of voice-port when the gateway is in busyout status.

caller-id	Enable caller-id
clear-down-tone-detect	Detect clear down tone
comfort-noise	Enable comfort-noise to be generated
compand-type	Set compand type (A-law or U-law)
connection	Set connection plar
description	Enter a description of the voice-port
dial-tone-generate	Enable dial tone generation
did	Apply DID(Direct Inward Dialing) to a corresponding voice-port
echo-cancel	Enable echo cancelation
end	Return to the beginning (exec)
exit	Return to the previous mode
fax-early-detect	Apply fax-early-detect to a corresponding voice-port
force-clear-down	Enable forced clear down on low level signal (FXO or E&M)
help	Display APOS help
high-dtmf-gain	Set a high frequency dtmf gain value to DTMF tone
input	Adjust input gain (volume) value
low-dtmf-gain	Set a low frequency dtmf gain value to DTMF tone
no	Cancel the entered command and return to default value
non-confirmed-connect	Configure non-confirmed-connect from E&M module
output	Adjust output gain (volume) value
polarity-inverse	Configure polarity-inverse settings (FXS : generate, FXO : detect)
pstn-backup-port	Configure peer PSTN(FXO) backup port settings
quit	Return to the previous mode
ring	ring cadence and frequency
shutdown	Shut down the port
signal	Set signal type
timeout	Set timeout value
timing	Set timing value
translate-incoming	Set called/calling number translation-rule to the port
type	Set E&M type
write	Save the present configuration

2.2.5.2.1. E1/T1 Controller Commands

E1/T1 controller configuration mode assigns a specific controller port from global configuration mode, and then it does the settings. Depending on a model, the gateway can have a number of E1/T1 ports.

Commands	Description
chan-number-order	Set an order for opening a channels for the incoming call
channel-group	Organize a group of channels for use
clock-source	Set to ISDN clock-source
end	Return to the beginning mode (Exec)
exit	Return to the previous mode
help	Display APOS help
isdn	Set ISDN related options
no	Cancels the command has been enters and return to a default value
out-barred-group	Set the channel group for blocking an outgoing call
quit	Return to the previous mode
r2	Set R2 related options
signaling-type	Select a signaling type from ISDN/DTMF/R2
write	Save the present configuration

2.2.5.3. Pots peer Commands

The prompt changes to model name (config-dialpeer-pots-0)# after dial-peer voice [0~65535] pots is entered from global configuration mode.

Commands	Description
application	Set pots to MGCP
call-waiting	Enables call-waiting
destination-pattern	Set a destination pattern
diversion	Set a call diversion

forward-digits	Forward an incoming number arrived on the pots
huntstop	Set hunting stop on this peer when call routing
inbound-pots-peer	Set inbound pots peer when this peer call out
no	Cancels the pots peer command has been entered and return to the default value
numbering-type	Change a numbering type of pots peer
out-barred-group	Set outbound call barred group for the pots
outbound-notify	Set a beep sound for the outbound call
port	Set a voice port binding to this pots
preference	Set a precedence order for dial-peer hunt selection
prefix	Set the prefix in pots peer configuration mode
quit	Return to the previous mode
recording	Configure a recording related information
register	Register pots peer to server (H.323/SIP)
shutdown	Shut down pots peer
user-name	Enter an authentication ID when to register
user-password	Enter a password for the authentication ID when to register
url	Set url type to tel type
display-name	Display a name of the pots for SIP connection
to-display-name	to-display-name of pots for SIP connection
translate-outgoing	Configure a number translate rule for the outbound call
write	Save the present configuration

2.2.5.4. Voip peer Commands

Entering dial-peer voice [0~65535] voip from global configuration mode changes the prompt to model name (config-dialpeer-voip-1000)#.

Commands	Description
answer-address	Use a calling party number of the inbound call to find the voip peer call for an arriving VoIP inbound call on network
calling-pattern	Set calling pattern to outbound call
clid	Change CLID(Calling Line Identification) mode
codec	Specify a static codec type

codec-variant	Set a variant value of G.723.1/G.729 codec
description	Enter a description of voip peer configuration mode
destination-pattern	Specify a destination-patter in voip peer configuration mode
dtmf-relay	Specify the dtmf-relay usage
end	Return to the beginning mode (Exec)
exit	Return to the previous mode
huntstop	Set hunting stop on this peer when call routing
fax	Specify a rate and fax protocol in voip peer configuration mode
max-forward-hop	Set the maximum call forward hop count
no	Cancel the entered command and return to the default value
modem	Specify a modem passthrough
numbering-type	Specify a numbering type
out-barred-group	Specify an outbound call barred group
out-permit-group	Specify an outbound call permit group
preference	Specify a preference
quit	Return to the previous mode
recording	Specify a recoding related information
redundant-rtp	Enable redundant rtp for lossy network
session	Specify a session target address and protocol of this peer
shutdown	Shut down voip peer configuration mode
sid	Enable SID (Silence Insertion Descriptor) packet sending mode when VAD (Voice Activity Detection) is enabled
translate-outgoing	Specify the number translation rule for an outbound call
url	Configure url type to tel type
user-privacy	Configure the user-privacy function when SIP server is connected
vad	Enable VAD (Voice Activity Detection)
voice-class	Specify a codec class list in voip configuration mode
write	Save the present configuration

2.2.5.5. Gateway Commands

Gateway configuration mode is to configure the settings related to H.323 Gatekeeper. Entering gateway in global configuration mode changes the prompt to model name(config-gateway)#.

Commands	Description
acf-dest-info	Allow a destination information of the ACF (Admission Confirm) message from Gatekeeper
Arq	Configure ARQ (Admission Request) options
attach-src-e164	Enable to attach source e164 address to ARQ (Admission Request) and LRQ (Location Request)
Discovery	Configure to send GRQ (Gatekeeper Request) message for gatekeeper discovery
endpoint-type	Specify an H.323 endpoint type
fixed-ras-port	Set RAS source port to a fixed value
Gkip	Enter an IP address or FQDN domain of the gatekeeper
h323-id	Enter a H.323 ID
ignore-msg-from-other-gk	Ignore an deregistered message transmitted from the gatekeeper
keep-gk-on-rrj	Keep trying registration when RRJ (Registration Reject) message is received from the gatekeeper
Lightweight-irr	Enable lightweight IRR (Information Request Response)
Lrq	Specify LRQ (Location Request) option
nat-support	Apply NAT support option
No	Cancel the entered command and return to the default value
public-ip	Configure a public IP number mapped to a private IP under static NAT/PAT settings
Quit	Return to the previous mode
Register	Register a gatekeeper
Resource	Specify a threshold value of RAI (Resource Availability Indication)
Security	Specify gatekeeper security option

signaling-port	Change H.323 signaling port
tech-prefix	Add a technical prefix when the gatekeeper is registered
Write	Save the present configuration

2.2.5.6. sip-ua Commands

In sip-ua configuration mode, you can configure the settings related to a connection with SIP server. Entering sip-ua in global configuration mode changes the prompt to model name(config-sip-ua)#.

Commands	Description
call-transfer-mode	Change call transfer mode from basic to attendant
conference-server	Configure an address of conference server
enable-ping	Deliver ping request to NAT/Firewall
end	Return to the beginning mode (Exec)
exit	Return to the previous mode
fault-tolerance	Configure redundancy for sip-server
help	Display APOS help
higher-priority-polling	Configure to send a ping message to a sip-server with higher priority
hook-flash-info-ignore	Do not send info message during hook-flash
keep-authentication-on-registration	keep authentication information on registration
media-channel	Enter media-channel mode
min-se	Specify a min-se value
mwi	Configure message waiting indication
no	Cancel the entered command or return to the default value
quit	Return to the previous mode
register	Register a SIP server
rel1xx	Send 100 Rel. message
remove-all-binding	Send a message to delete all the existing registration information prior to sending registration message

remote-party-id	Apply E.164, which is defined in destination-pattern, to user-name instead of field which INVITE is transmitted
response	Include SDP field in 183 progress message
retry-counter	Set a counter for retrying transmission when any reply is not received for all types of SIP request message (register,invite,200 OK and others)
rport	Configure rport parameter when NAT is supported
route-by-auxiliary	Route with a reference to user-id which is in to field details of the initial INVITE
session-expires	Specify a session-expires value
session-refresh	Specify a SIP message to be used for session refresh
set-local-domain	Create a URL of SIP message as a specific domain not as an IP address configured to the gateway
special-char	Change a special character of SIP URL (such as $!*#) to ASCII value.
set-local-host	Create a hostname configured to URL of SIP message
signaling-port	Change a SIP signaling port
force-forwarding	Process a call by selecting the following voip-peer commands when 403/404 is received for INVITE
sip-server	Enter a SIP server address
sip-username	Enter SIP username
sip-password	Enter a SIP password
srv	Enable SRV DNS query
timeout	Specify a timeout value related to REQUEST
user-register	Enable user-register
write	Save the present configuration
force-forwarding	Process a call by selecting the following voip-peer commands when 403/404 is received for INVITE
3way-conference	Set to 3-party conference
hook-flash-info-ignore	Configure not to send info message when hook-flash is on

2.2.5.7. mgcp Commands

In mgcp configuration mode, you can configure the settings related to the connection with MGCP server. Entering mgcp in global configuration mode changes the prompt to model name(config-mgcp)#.

Commands	Description
shutdown	Disable mgcp
busyout-timer	Specify a CallAgent busyout timer
call-agent	Specify a CallAgent address
codec	Specify a default codec to be used for mgcp
default-package	Configure a default package
digit-map	Configure the local digit map
dtmf-relay	Select dtmf-relay
end	Return to the beginning mode (Exec)
epid-type	Specify an epid-type
exit	Return to the previous mode
fast-connect	Set to fast connect
force-local-rt	Play Forced Local Ringback tone
restart-delay	Specify a restart-delay timer value
restart-each-endpoint	Configure the settings to send RSIP message to each port
timeout	Change a timeout value
sid	Enable sid
vad	Enable vad
areacode-no	Set up an area code
insert-areacode	Set to insert area code mode
discrimination-no	Set up a discrimination number
insert-discrimination	Set to Insert discrimination number mode

2.3. Gateway Configuration Startup

You must log as in configuration mode to set up the gateway and to log in configuration mode, you must use and administrator's password. If you do not know any proper command, you can use 'help'.

[Usage Procedure]

Steps	Workflow Description
1	Log in an administrator's account after booting up the gateway

2	Move to configuration mode
	`router # configure terminal`
	`router (config)#`

[Usage Example] Gateway Configuration Mode

```
Welcome, APOS(tm)  Kernel Version 8.41.100.
Copyright (c) 1999-2006 AddPac Technology Co., Ltd.
login: root ☞ Log in an administrator's account (it is set to root at the factory default)
password:****** ☞ Enter the administrator's password (it is set router at the factory default)
model name> enable
model name# configure terminal ☞ Enter the command to move to configuration mode
model name(config)# ☞ You can start configuring from this status
```

2.4. Configuring Ethernet

2.4.1. Basic Setups

Basically, VoiceFinder Gateway supports one or more RJ-45 ethernet ports. If the connecting device supports only AUI port, it is necessary to use 10Base-T MAU (Media Attach Unit). The ethernet port supports standard ARPA encapsulation, and it can be set to SNAP or IEEE802.3 Encapsulation.

More than one logical ports can be applied the physical ethernet port, and a logical port should be exist to use it.

[Usage Procedure]

Step	Workflow Description
1	Enter interface configuration mode
2	Assign an IP address to the interface (For using IPv6, an IPv6 address needs to be assigned)
3	(Optional) Specify an Encapsulation to be used
4	Up the interface
5	Set up other optional parameters

[Related Commands and Formats]

- **full-duplex/ half-duplex/ duplex {full| half| auto}**
  1. Set to operation mode of Ethernet Interface
  2. The default value is set to Auto

- **speed {10| 100| 1000| auto}**
  1. Set to speed of Ethernet Interface
  2. The default value is set to Auto
  3. Set to 1000 for supporting Gigabit

- **interface** { **Dialer /FastEthernet** / **loopback /Tunnel** } { **0 / 1** }.[*logical I/F #*]

  1. Choose an interface to be configures, then enter interface configuration mode
  2. {0/1} means the main interface and [logical I/F #] means the sub-interface
  3. Ethernet must be set to sub-interface

- **ip address** *{<ip_address> <net_mask>* | ***dhcp}*** **{ secondary}**

  1. Specify an IP address for the chosen interface
  2. This is the lower menu of the ip command
  3. If you want to use DHCP client, enter dhcp instead of <*ip_address*> <*net_mask*>
  4. The secondary function can be supported for APOS G2 version, but the secondary function can not be used for DHCP only

- **ipv6 address** { X:X::X:X/M | autoconfig }

  1. Specify an IPv6 address for the selected interface
  2. In this case, the IPv6 addresses can be automatically created by adding MAC addresses of the interface after receiving a network information from the inside of the pertinent network

[Usage Example] Ethernet (to start)

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

```
model name(config)# interface FastEthernet 0/0
model name(config-if)# ip address 192.20.1.1 255.255.255.0
model name(config-if)# ip address 210.10.2.1 255.255.255.0 secondary
model name(config-if)#
```

- **arp** <*ip-address*> H.H.H

  Configure Static ARP(MAC) function with the IP address. Enter 6 digit MAC address by a binary in hexadecimalcimal format (APOS G2)

- **arp** { keep <120-3600> | walk <1-600>}

Specify a timeout value and periodic forwarding time in APOS G2 version

- **shutdown / no shutdown**

  1. This is a command to up/down the presently located interface
  2. There is no way to shut down the main Ethernet interface. To link up and down a particular Ethernet interface, you can do it on the sub interface

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

  This is a commad to delete a logical interface. The physical interface which is created from the hardware equipment can not be eliminated (i.e. FastEthernet0/0, Loopback0 and others)

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

  Show interface status of if-name

[Usage Example] Ethernet Configuration Example

model name(config)#interface FastEthernet 0/0 ☞ Logical interface Ethernet 0.0 is created from Main interface Ethernet 0 and changes to the configuration status for that interface.
model name(config-if)# ☞ At this mode, the interface can be configured.
model name(config-if)# ip address 131.12.1.1 255.255.0.0 ☞ Set IP address to 131.12.1.1/16bit mask
model name(config-if)#no shutdown ☞ When Main interface up, also Sub-interface up automatically
model name(config-if)# end ☞Move to configuration mode

```
model name#show interface FastEthernet 0/0
Interface FastEthernet0/0
 flags=8003<UP,BROADCAST,MULTICAST>  index 2 metric 1 mtu
1500
 mac address  0002.a4ff.fe7e
 inet 131.12.1.1 255.255.0.0  broadcast 131.12.255.255
   FastEthernet0/0 is DOWN, Line protocol is DOWN
   QoS control is disabled
   interface type is 100Base-TX
   link status is 0 Mbps (HALF-DUPLEX)
   0 packets input, 0 bytes, 0 no buffers
```

```
    Received 0 runts, 0 giants
    0 input errors, 0 CRC, 0 frame
      0 length, 0 overrun, 0 ignored
    0 packets output, 0 bytes, 0 drops
    0 output errors, 0 collision, 0 interface resets
      0 collisions, 0 late collisions, 0 deferred
      0 lost carrier, 0 no carrier
model name#
```

2.4.2. Configuring PPPoE

Information

PPP(Point to Point Protocol) is one of the standards for transmitting data on WAN link, which is stated on RFC1661. It is the transmission protocol which can be used for not only Synchronous WAN(SERIAL) line but also Asynchronous WAN(Dial Up Line). PPP is the standard protocol which is different from HDLC and it guarantees the interoperability.

VoiceFinder Gateway can be connected to the rear part of ADSL modem. In this case, the gateway to support PPP and Encapsulation of Ethernet interface is to be configured to PPPoE.

PPP is contained in the following 2 different types of protocols:

- LCP (Link Control Protocol) : LCP is used to perform determination encapsulation format, limiting packet size, authentication on a link, determining normal operation time and failure time of the link, error detection and link termination
- NCP (Network Control Protocol) : NCP is used to communicate and connect to the protocols of the network layer of OSI 7 Model

When PPP Encapsulation Option is selected from Ethernet interface of VoiceFinder Gateway, PPPoE is enabled. The software of the gateway supports CHAP(Challenge Handshake Authentication Protocol) and PAP(Password Authentication Protocol) which are used for Authentication Option and Magic Number. The software always sends Magic Number Configuration Option only when it is set to Authentication Option.

[Usage Procedure]

Steps	Workflow Description
1	Enter interface configuration mode
2	Enable PPPoE of the interface
3	Add PPP Encapsulation protocol to the interface
4	Enable CHAP or PAP Authentication (optional)
5	Configure the setting for CHAP/PAP Parameter (optional)
6	Configure the IP setting for PPP Default Peer IP (optional)
7	Check whether the gateway operate normally by using debug command (when it is necessary)
8	Up the interface
9	Check whether the function operates normally on the interface by using show interface command
10	Use debug command to find any failure and correct it (when it fails)

[Related Commands and Format]

- **interface FastEthernet** { **0 / 1** }/ { **0 / 1** }

  Choose the interface to be configured, then enter interface configuration mode

- **pppoe enable**

  Enable PPPoE

- **pppoe-client local-interface**

  Assign the interface which is operated by PPPoE , to local

  In general, using local interface is recommended.

- **encapsulation ppp**

  Configure encapsulation mode for the interface to be used.

- **username** <username> **password** <password> {**administrator/operator/user**}

  1. This is a command to set a login name and password for authorizing an access to the

gateway when the gateway is connected to PPP PAP/CHAP server.

2. This commend is same as the one for an administrator to register a login user. Only the difference from the gateway user registration that the registered user's level is not used to connect to PPP even though the user registration is used for PPP connection.
3. In general, a function of adding user is not needed, because the network server performs the authentication function in ADSL network environment.

- **ppp ipcp { enable | default-router }**
  1. This is the command for configuring PPP IP address and default router settings from the interface configuration
  2. Set ipcp to enable/ default-router for IPv4 related settings in general
  3. When other ipcp options are not configured, it operated on a value which is determined by the system

- **ppp ipv6cp { enable }**
  1. This is the command for PPP IPv6 and default router from the interface configuration.
  2. Enable ipv6cp, which is IPv6 related configuration, only when it is used for IPv6
  3. When other ipcp options are not configured, it operated on a value which is determined by the system

- **ppp authentication** {**chap/pap**} [calling/{pap/chap}]
  1. This is the command to configure ppp authentication mechanism to chap or pap for the interface configuration.
  2. This calling option is to connect only incoming call to chap authentication
  3. {pap/chap} at the end is the option to respond to the calls requiring all chap/pap authentication

- **ppp chap hostname** *name*
  1. This command to configure PPP client device. When ppp chap is used to request for a connection to the server, it registers a user's name (an option for chap authentication)
  2. When this command is not used, the gateway name is sent, as a basic value, to the user name.

- **ppp chap password** *password*

  This command is to configure ppp client device. When ppp chap authentication is used for

requesting connection to ppp server, a password is registered for use (an option for using chap authentication).

- **ppp pap sent-username** *username* **password** *password*

  This is the command to configure ppp client device for using pap authentication. When the client device is configured, this account with a password is sent to sever for authentication. At this time the username and password must be same as the ones used for configuring the server (an option for using pap authentication)

- **shutdown / no shutdown**

  This is an option to link up/ down the present interface.

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

  Display a status of the Interface for if-name

- **debug ppp** { **negotiation/packet** }

  1. Display ppp low level packet as to decode
  2. 'negotiation' is to decode NCP protocol and LCP for configuring ppp link
  3. 'packet' is to decode ppp low level packet

[Usage Example] General Configuration of PPP and Usage Example

model name # configuration terminal ☞ moves to configuration mode

model name(config)# interface FastEthernet 0/0 ☞ moves to interface configuration status

model name(config-if)# ☞ config can be used from this status

model name(config-if)# pppoe enable ☞ Enable pppoe

model name(config-if)# pppoe-client local-interface ☞ Set the interface being operated by pppoe to local

model name(config-if)# encapsulation ppp ☞ Set to ppp mode

model name(config-if)# ppp ipcp default-router

model name(config-if)# ppp authentication chap calling ☞ Specify ppp authentication mode to chap for the interface

model name(config-if)# ppp chap hostname addpac ☞ If a user name which has been saved in the server is different from the one in client gateway, then the one saved in server is to be sent during ppp chap connection.

model name(config-if)# ppp chap password addpac ☞ Specify the username which has been configured in the server, for verifying the password sent from the server, during ppp chap connection

model name(config-if)# no shutdown ☞ Link up the interface

model name(config-if)# end ☞ Exit the configuration mode

model name# debug ppp packet ☞ Decode ppp packet

model name#

Ether0.0 LCP: TIMEOUT

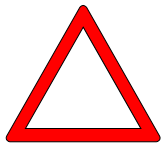
Ether0.0 LCP: O CONFREQ id=1

Ether0.0 BCP: TIMEOUT

Ether0.0 BCP: O CONFREQ id=1

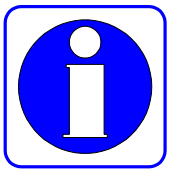
model name# debug ppp packet ☞ Turn off ppp packer debugging

Caution **An IP address must be assigned on the interface when DHCP notion is not used to the interface of the gateway.**



2.5. Routing Configuration

Information



VoiceFinder Gateway series support both of static routing and dynamic routing protocols. Dynamic routing protocol can be divided by 2 different types: one is IGP(Interior Gateway Protocol) which is used for routing between the networks within the same administrator's domain and the other is EGP(Exterior Gateway Protocol). RIP, OSPF, IS-IS are used for IGP and BGP is used for BGP. The VoiceFinder Gateway supports RIP and OSPF.

To use routing protocol from VoiceFinder Gateway, you must put the routing process on the gateway and specify the network to be used for.

Choosing the routing protocol to be used for the gateway is not that simple and consideration of the following details are recommended

- Network Size and Complexity : Static Routing is commonly used for the Edge Network. If you need to use Dynamic Routing, please use RIP for small network and use OSPF for large network.
- If many subnet classes may exist together within a network, please use the routing protocols which can support VLSM (Variable Length Subnet Mask) such as static route, RIPv2, OSPF.

For others please consider Convergence Time, Reliability Needs, and Internetwork Delay Characteristics

VoiceFinder can operate many routing protocols simultaneously. When a gateway uses many routing protocols simultaneously, each protocol can be assigned with a calculated path for its destination. The priority in routing table is Static route - OSPF Route - RIP Route - Default Route order.

2.5.1. Static Routing Configuration

Static route is a specified route to send a packet to go on a path which designated by an administrator. Static route can be used for the following 3 cases:

- A case where routing software fails to create a route towards a specific destination

properly.

- A case where network is small-size and its structure is not complicated, so configuring static route is relatively easy and a packet that gives a load on the network, such as Route Update Packet, is not desirable.
- A case where all the packets are to be sent to a specific next-hop address and destinations of the packets are not displayed in a routing table by using default route (or gateway of last resort).

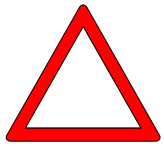
Once static route is configured, the gateway recognizes it as so until the route is deleted. If you want delete the route, you can use 'no' command.

Default route specifies the next path of the packet towards all the destinations that is not displayed in the routing table as a kind of static route. Default route gets the lowest in priority level of VoiceFinder Gateway.

[Usage Procedure]

Steps	Workflow Description
1	Enter configuration mode
2	Enable Static Gateway Process
3	Specify Static Path for Source Address and Destination network to be used
4	Check the routing table whether the desired route is configured by using Show command
5	Use Ping command for checking the packet is arrived on the designated network

Caution



1. **For configuring a static route, the next-hop address must be directly connected with the gateway, which is intended to be configured.**
2. **The default route is a kind of static route and the configuration is same as static route. Only the difference is that zero subnet (0.0.0.0.mask to 0.0.0.0 address) is used to the zero network which represents all the addresses for the destination address and the next-hop address is configured in the same way as the static route.**

[Related Commands and Format]

- **router** { **rip** | ospf | ripng | ospf6 }

  Enable or disable a specific routing process. If rip or ospf command does not present, the routing protocol is not supported. VoIP equipment does not support dynamic routing protocol operation in general.

- **ip route** <*destination-IP-network*> <*address-mask*> { <*next-hop-address*> / **FastEthernet / Dialer / Tunnel/ Loopback** } [(0/1)/<null_int_#>] [sub_int_#]

  1. Specify a route to send a packet towards a specific destination address

2. When a Candidate Default (default route), all the destination address and mask field are to be set to zero
3. The next-hop-address must be a location which can be recognized by the gateway (a port which can directly connected or a location to where the dynamic protocol can reach)
4. An interface port can be specified instead of the next-hop-address. It is recommended to apply to loopback and ppp interface. If it applies to FastEthernet, an unexpected problem may arise.
5. The static route using a Null Interface is to drop the packet towards a specific destination. The null interface is supported only in APOS G1 version

- **show ip route**

Check the route configured in a routing table

[Usage Example] Static Routing Configuration

Gateway# config terminal
Gateway(config)# ☞ In this status, config is possible

Gateway(config)# ip routing ☞ Enable IP Routing Process

Gateway(config)# ip route 130.2.0.0 255.255.0.0 131.20.1.1 ☞ Specify the packet with130.2.0.0/24bit to be sent to the equipment with the address of 131.20.1.1

Gateway(config)# ip route 0.0.0.0 0.0.0.0 132.20.1.1 ☞ Send all the packets that are not listed in the routing table to the address of 132.20.1.1

Gateway(config)# exit ☞ Exit configuration mode

Gateway(config)# show ip route ☞ Display the routing table

S> 0.0.0.0/0 [1/0] via 132.20.1.1 inactive

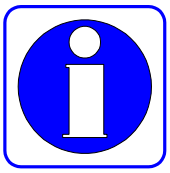
C>* 127.0.0.0/8 is directly connected, Loopback0

C>* 130.2.0.0/16 is directly connected, FastEthernet0/0

S 130.2.0.0/16 [1/0] via 131.20.1.1 inactive

2.6. Configuring Filter (Access-List)

Information



Packet Filtering is a function which enables an administrator to control the packet admission through a network. This function is used to block an unauthorized user's access from the outside to the inside of network.

VoiceFinder Gateway uses Access-list as a way to control the traffic from a specific network or equipment. The access-list can permit or deny the packet passing from one specific interface to another.

In access-list, there is the Standard Access-list which controls traffic by source and destination IP addresses and the Extended Access-List which uses application port number and protocol ID. The access-list is a group of permit/deny conditions to be applied an IP address. The VoiceFinder Gateway compares these conditions to the address field of the packet.

The first condition of the address field to be matched for the gateway is whether to accept or reject the packet. Process order of these conditions is very important for the access-list to operate normally, because the software may stop testing the conditions for the address after the first match. If all the conditions are not matched, the packet is rejected for that address.

VoiceFinder supports the standard access-list(List # 1~99, 1300 ~ 1999) and Extended Access-List(List # 100 ~ 199, 2000 ~ 2699).

[Usage Procedure]

Steps	Workflow Description
1	Enter configuration mode
2	Create an access-list with Access-List Number and Access Condition.
3	Move to interface configuration mode
4	Move to IP configuration mode
5	Apply the configured access-list to a target interface At this time, determine whether to apply the access condition to inbound packet or outbound packet
6	Use show access-list command to check whether access-list is configured with accuracy.

[Related Commands and Format]

Standard IP Access-List : Standard IP Access-List uses only Source IP Address to check the access condition.

- **access-list** <*access-list-number*> {**remark**/**deny**/**permit**} <*source-address* > <*source wildcard*>

  1. Create an access-list
  2. access-list-number : the number within the rage of 1~99, (expanded range : 1300 ~ 1999)
  3. source: Source Network Address, Source-wildcard : Inverse Mask of the source address
  4. any(any address), host(a specific host) can be used instead of each source, Source-Wildcard.
  5. Wildcard represents inverse mask. For instance ***132.1.20.1 255.255.255.0*** of a network can be written as ***132.1.20.1 0.0.0.255*** in wildcard format
  6. When all of the conditions can not be considered, use Permit Any Option to allow any packet which does not matched. Otherwise the unmatched packets will be discarded since the default option is deny.

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

  As an interface command, this command applies the access-list to the incoming and outgoing

packet of the interface.

Extended IP Access-List : Extended IP Access-List uses Source IP Address, Destination IP Address and Protocol ID, Application Port Number and Establish options to check access conditions.

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

1. Create an access-list
2. Description of Each Option
   1) access-list-number : The range of the numbers for Extended Access-List 100~199 (expanded range : 2000 ~ 2699)
   2) protocol : protocol ID Number or protocol name (example: TCP, ICMP, UDP IP)
   3) source : Source Network Address,
   4) Source-wildcard: Inverse Mask of Source Address
   5) Destination : Destination Network Address
   6) destination-wildcard : Inverse Mask of Destination Address
   7) operator : for Port #
      - ✓ ack ACK bit
      - ✓ established
      - ✓ fin FIN bit
      - ✓ psh PSH bit
      - ✓ rst RST bit
      - ✓ syn SYN bit
      - ✓ urg URG bit
      - ✓ eq: equal
      - ✓ gt: greater then
      - ✓ lt: less then
      - ✓ neq: not equal
      - ✓ range
   8) port-number: As an application port number, Well Known Port # is listed as to follow
      - ✓ bgp : Border Gateway Protocol (179)
      - ✓ chargen : Character generator (19)
      - ✓ cmd : Remote commands (rcmd, 514)
      - ✓ daytime : Daytime (13)

- ✓ discard : Discard (9)
- ✓ domain : Domain Name Service (53)
- ✓ echo : Echo (7)
- ✓ exec : Exec (rsh. 512)
- ✓ finger : Finger (79)
- ✓ ftp : File Transfer Protocol (21)
- ✓ ftp-data : FTP data connections (used infrequently, 20)
- ✓ gopher : Gopher (70)
- ✓ hostname : NIC hostname server (101)
- ✓ ident : Ident Protocol (113)
- ✓ irc : Internet Relay Chat (194)
- ✓ klogin : Kerberos login (543)
- ✓ login : Login (rlogin 513)
- ✓ lpd : Printer server (515)
- ✓ nntp: Network News Transport Protocol (119)
- ✓ pim-auto-rp PIM Auto-RP (496)
- ✓ 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)
- ✓ telnet : Telnet (23)
- ✓ time : Time (37)
- ✓ uucp : Unix-to-Unix Copy Program (540)
- ✓ whois : Nicname (43)
- ✓ www : World Wide Web (HTTP, 80)

9) established : established session

3. Any (any address), host (a specific host) can be used instead of source/destination, source-wildcard/destination-wildcard

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

Apply the configured access-list to the incoming and outgoing packet of the interface. This is an interface command

[Usage Example] Standard Access-List Configuration

model name(config)# ☞ Access-list Config is possible in this status

model name(config)# access-list 1 remark this is access-list ☞ The access-list 1 can be described.

model name(config)# access-list 1 deny 132.1.2.1 0.0.0.255 ☞ Deny any packet with the source address of 132.1.2.0/24bit**.**

model name(config)# access- list 1 deny 150.1.3.2 0.0.0.223 ☞ Deny any packet with the source address of 150.1.3.0/21bit

model name(config)# access- list 1 deny host 132.1.3.15 ☞ Deny any packet with the source address of 132.1.3.15 comes from a host

model name(config)# access-list 1 permit any ☞ Permit any packet which does not meet the condition of access-list 1 stated above. *Without this command, any default packet is to be denied.

model name(config)# interface FastEthernet 0/0

☞ Move to interface FastEthernet 0/0 configuration mode

model name(config-if)# ip access-group 1 in

☞ Apply configures Access-List 1 to any IP packet comes through the interface of FastEthernet 0/0

Router # show ip access-list ☞ Display the configured access-list

```
Standard IP access List 1
        deny      132.1.2.0 wildcard bits 0.0.0255
        deny      150.1.3.0 wildcard bits 0.0.0223
        deny      132.1.3.15
        permit    any
```

[Usage Example] Extended Access-List Configuration

model name(config)# ☞ You can start Access-list Config in this status

model name(config)# access-list 100 deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp ☞ Deny any TCP packet which accesses to the host and ftp port with the destination of 145.1.1.0/24Bit from the source address of 140.1.1.0/24bit.

model name(config)# access-list 100 deny tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp-data ☞ Deny any TCP packet which accesses to the host and ftp-data port with the destination of 145.1.1.0/24Bit from the source address of 140.1.1.0/24bit.

model name(config)# access-list 100 permit tcp 140.1.1.0 0.0.0.255 145.1.1.0 0.0.0.255 eq ftp ☞ Permit only the TCP packet configured with Session which accesses to the host and ftp port with the destination of 145.1.1.0/24Bit from the source address of 140.1.1.0/24bit

model name(config)# access-list 100 permit ip any any ☞ Permit all the other IP packets except the condition stated above

model name(config)# interface FastEthernet 0/0 ☞ Enter to interface FastEthernet 0/0

model name(config-if)# ip access-group 100 in ☞ Apply access-list100 to all the IP packets coming through Ethernet 0.0 interface

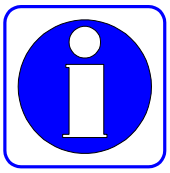
model name(config-if)# end

model name # show ip access-list 100 ☞ Display the configured access-list100

```
Extended IPaccess List 100
  deny tcp 140.1.1.0  0.0.0.255  145.1.1.0  0.0.0.255 eq ftp
  deny tcp 140.1.1.0  0.0.0.255  145.1.1.0  0.0.0.255 eq ftp-data
  deny tcp 140.1.1.0  0.0.0.255  145.1.1.0  0.0.0.255 eq ftp
  permit ip  any  any
```

2.7. Configuring NAT (Network Address Translation)

Information



One of the major problems with today's internet may be the shortage of supplying IP addresses to meet the demand. Network Address Translation (NAT) is one of the ways to resolve this problem. NAT translates the private IP address which is used within the network to a different type of IP address (public IP address) when the IP address goes out of the network. In the other way, NAT translates the public IP address coming from the outside into the inside of the network.

NAT can be used in the following several purposes:

- NAT enables a user to connect the private IP network to the global internet when authorized public IP address is not enough to assign to all terminals.

  NAT is to be configured with the gateway which is located between the border of Stub domain (commonly called as inside network) and the public network (commonly called as outside network). At this location, NAT converts the packet from the private IP address in the inside network to the unique public IP address before it sends out the packet to the outside network.

- Sometimes an administrator needs to change the inside network address for security and other management purposes. This change may require a lot of work, but using NAT, the address can be translated with easy.

- Sometimes an administrator needs to perform load-sharing of TCP traffic. In this case, TCP Load Distribution can be done by mapping many Local IP Addresses to one Global IP Address. To access from the outside, Global IP Address can be accessed as one, then it is divided by TCP sessions the load can be distributed.

[NAT Acronyms]

- ip nat inside : This is the IP Address to be configured for the Host of the Inside Network

- ip nat outside : This is the IP address authorized from Network Information Center (NIC) or Service Provider representing the inside Local IP Address to Outside Network.

- outside local address : This is the IP address of the Host in the Outside Network, which is represented in the Inside Network. The public address is not needed, but should be allocated the routing possible address.

- outside global address : This is the IP address which is allocated by the Host owner for the host in the Outside Network. The address is allocated to the globally routing possible address or network.

NAT has Static Address Translation and Dynamic Address Translation.

- Static Address Translation : When an access is requested from the Outside Network, this address translation converts the public IP address to the private IP address statically. On the other hand, when the inside host is accessed to the outside network, the reverse process is taken.

- Dynamic address Translation : When an access is requested from Inside Network to Outside Network, an unused public IP address is assigned from the IP address pool. If all the procured IP addresses are in use, the inside network can not allow any more access to the outside.

VoiceFinder Gateway supports both NAT, which translates many inside private IP addresses to the outside public IP address, and PAT (Port Address Translation) feature that translates many private IP address to the protocol numbers above the one single outside public IP address.

Interface PAT enables VoiceFinder Gateway to use DHCP of Cable Modem or PPPoE of ADSL environment dynamically. When PAT static entry is specified, not only a specific port, but also a constant port range can be connected to a specific host, which has been designed to be

suitable to VoIP equipment

1. The current version of VoiceFinder Gateway supports Dynamic Address Translation only.
2. The number of NAT addresses which are supported by VoiceFinder Gateway is 256.
3. The routing protocol related to NAT in the current version of VoiceFinder Gateway supports Static Routing and RIP only.

[Usage Procedure]

Steps	Workflow Description
1	Enter configuration mode
2	Create NAT/ PAT provisioned by the public IP address to be used. ✓ At first, decide whether the Global Address to be used outside or inside ✓ Determine and configure a entry to be matched statically between the Inside and outside Address ✓ Configure Session Time-Out to return the allocated address to be free, when NAT Session is in idle status
3	Move to interface configuration mode
4	Move to IP configuration mode
5	Apply the configured NAT/PAT to the interface
6	Use show running-config command to check whether NAT, access-list is configured correctly.

[Related Commands and Format]

- **ip nat** <inside/**pool**/translation><pool name><Start IP address><End IP address><netmask>

1. Configure nat pool to specify outside ip addresses when the inside packet communicates with the outside
2. Pool Name can be assigned by user.
3. When to communicate with the outside, specify ip address range. The user using the

inside private ip address takes the public ip address to communicate with the outside.

- **ip nat** <**inside**/pool/translation><destination/source><list/static><num/word interface/pool><dialer/fastethernet><slot/port><overload|cr>
  1. Configure NAT for the inside packet to go out to the outside.
  2. inside : Configure to specify the inside network
  3. list : Specify a list to be mapped with access-list. If the list does not match with the number of access-list, then nat configuration does not operate.
  4. Specify an interface for the inside packet applied with NAT configuration to be sent out to the outside

- **ip dhcp pool** <WORD>.
  1. Specify dhcp feature to active/de-active from global configuration location.

- **network FastEthernet** <interface-id>
  1. Specify a local interface
  2. interface id: ID of FastEthernet interface (0~1).

- **range FastEthernet** <interface-id>
  1. Specify a network side interface
  2. interface id: ID of FastEthernet interface (0~1).

- **show running-config**

  Display the settings including NAT configuration

[Usage Example] NAT – DHCP Configuration and Usage Example

```
model name# config terminal
model name(config)# ☞ NAT-list Config can be started from this status
model name(config)# ip nat inside source list 1 interface
FastEthernet0/0 overload ☞ Configure NAT so the inside packet can be
sent to the outside
model name(config)# access-list 1 permit 10.1.1.0 0.0.0.255 ☞
Apply the ip included in the inside ip address of 10.1.1.0/24 to standard access-list
model name(config)# ip dhcp pool addpac ☞ Configure dhcp mode for
```

the pool name with addpac

model name(config-dhcp-network)# network 10.1.1.0 255.255.255.0 ☞ Enter configuration mode of the inside network

model name(config-dhcp-network)# range 10.1.1.2 10.1.1.11 ☞ Specify the usage range of 10.1.1.2 ~ 10.1.1.11 from the inside network

model name(config-dhcp-network)# subnet-mask 255.255.255.0 ☞ Specify subnet-mask to be used in the inside network

model name(config-dhcp-network)# routers 10.1.1.1 ☞ Specify default route to send packet of the inside network

model name(config)# interface FastEthernet 0/0 ☞ Enter configuration mode for interface FastEthernet 0/0

model name(config-if)# ip address 210.98.100.1 255.255.255.0 ☞ Allocate the address to FastEthernet 0/0 interface

model name(config-if)# ip nat outside ☞Apply NAT configuration to FastEthernet 0/0 interface

model name(config-if)# exit ☞ Exit FastEthernet 0/0 interface configuration

model name(config-if)# interface FastEthernet 0/1 ☞ Enter configuration mode of interface FastEthernet 0/1

model name(config-if)# ip address 10.1.1.1 255.255.255.0 ☞ Allocate the address to FastEthernet 0/1 interface

model name(config-if)# ip nat inside ☞ Apply to NAT configuration to FastEthernet 0/1 interface

model name# show running-config ☞ Display the configuration for NAT – DHCP

```
interface FastEthernet0/0
 ip address 172.17.213.96 255.255.0.0
 ip nat outside

interface FastEthernet0/1
 ip address 10.1.1.1 255.255.255.0
 ip nat inside

access-list 1 permit 10.1.1.0 0.0.0.255
```

```
ip nat inside source list 1 interface FastEthernet0/0  overload


ip dhcp pool my
 network 10.1.1.0 255.255.255.0
  range 10.1.1.2 10.1.1.11
  subnet-mask 255.255.255.0
  routers 10.1.1.1


!
```

2.8. Configuring DHCP (Dynamic Host Configuration Protocol)

Information

DHCP (Dynamic Host Configuration Protocol) is a protocol which allocates IP address to DHCP Client automatically.

The DHCP feature of VoiceFinder has a role of allocating and manging IP addresses for DHCP Client by using address pool. If the VoiceFinder Gateway fails to respond to DHCP request, the gateway is able to forward this request to the other DHCP server.

The following picture basically shows the process of which DHCP Client requesting IP address from DHCP server

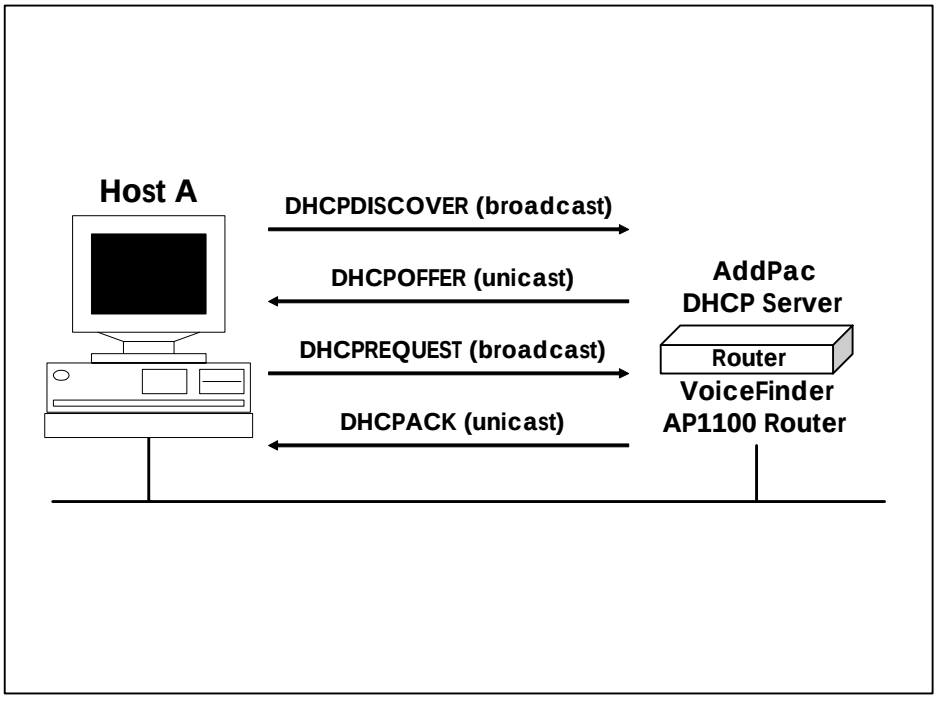


Figure 2.1 Communication between DHCP Server and Host A

Host A as a client sends a broadcast message which is named as DHCPDISCOVER to DHCP server. Then DHCP server returns DHSCPOFFER unicast message with the configuration details including IP address, MAC address, domain name, verification of the configuration details. DHCP server confirms the IP address which has been allocated to the client already by retiring DHCPACK unicast message.

DHCP feature of VoiceFinder Gateway complies with RFC2131, BOOTP of RFC951 and Bootstrap of RFC1542. By using these, you may gain the following advantages.

- DHCP configuration is relatively easy and you may save the time and cost to configure client.
- A network administrator manages only the central server, so the administrator can easily manage the addresses and its related details easily.

To deploy DHCP server feature, the following considerations needs to be completed:

- When to enable DHCP server features, you should divide the IP address which is to be allocated and the addresses which are not using DHCP feature (for example, the equipment such as server and printer which requires IP address to be fixed).
- Optionally, the DHCP options such as default gateway, DNS server need to be specified.

DHCP features of VoiceFinder Gateway can be used not only as DHCP server, but also as DHCP Client. In order to configure the features of DHCP Client, enable dhcp instead of set IP address directly.

[Usage Procedure – DHCP Server]

Steps	Workflow Description
1	Enter configuration mode
2	Specify DHCP Type from the gateway
3	Create DHCP-List, which conform DHCP address pool or DHCP mode, to be used DHCP server configuration and the gateway. ✓ Specify DHCP Pool which conforms DHCP Start-Address and DHCP End-Address when DHCP is used as Server Type
4	Specify the necessary options which are related to other DHCP features
5	Move to configuration mode
6	Move to IP configuration mode
7	Apply the configured DHCP-List to the interface.

8	Check whether DHCP is configured correctly or not by using show running-config

[Related Commands and Format]

Mandatory Commands

- **dhcp** {**server/relay**}

  Enable DHCP of the gateway to operate as a server or DHCP Protocol Relay.

- **ip dhcp relay**

  1. Change DHCP protocol of the gateway which is broadcast, to relay-ip-address equipment, which is unicast

  2. relay-IP-address: the IP address of the equipment which relays DHCP broadcast to unicast message.

- **ip dhcp pool** *<pool name>*

  1. Configure DHCP pool to operate as DHCP server
  2. <subnet address><interface FastEthernet > : Specify the IP address or interface range of DHCP pool.

- **show running-config**

  Display the configured settings including DHCP

[Usage Procedure – DHCP Client]

Steps	Workflow Description
1	Enter configuration mode
2	Choose the interface to be specified to DHCP Client
3	Specify the IP address of the interface to dhcp

When you need to use only the static IP address and do not need to use DHCP Client feature, you may enter the IP address to the interface directly or no ip address dhcp, then DHCP Client feature stops automatically.

[Usage Example] Configuring DHCP Server Mode

```
model name# config terminal
model name(config)# ☞ DHCP Config can be started from this status
model name(config)# dhcp server ☞ Enable DHCP server
model name(config)# ip dhcp pool addpac ☞ Configure all the pool names with addpac to dhcp mode
model name(config-dhcp-network)# network 10.1.1.0 255.255.255.0 ☞ Move to the inside network configuration mode
model name(config-dhcp-network)# range 10.1.1.2 10.1.1.11 ☞ Specify the usage range to 10.1.1.2 ~ 10.1.1.11 from the inside network
model name(config-dhcp-network)# subnet-mask 255.255.255.0 ☞ Specify subnet-mask to be used in the inside network
model name(config-dhcp-network)# routers 10.1.1.1 ☞ Specify default route for sending the packet of the inside network
model name(config-if)# ip nat outside ☞ Apply NAT configuration to FastEthernet 0/0 interface
model name(config-if)# exit ☞ Exit FastEthernet 0/0 interface configuration
model name(config-if)# interface FastEthernet 0/1 ☞ Move to configuration mode of interface FastEthernet 0/1.
model name(config-if)# ip address 10.1.1.1 255.255.255.0 ☞ Allocate the address to FastEthernet 0/1 interface에 Address
model name(config-if)# ip nat inside ☞ Apply NAT configuration to FastEthernet 0/1 interface
```

[Usage Example] Configuring DHCP Relay Mode

model name# config terminal

model name(config)# ☞ DHCP Config can be started from this status

model name(config)# ip dhcp relay ☞ Enter dhcp relay mode

model name(config-dhcp)# relay ip 151.1.12.1 ☞ Configure DHCP Request Packet to be converted to Unicast Packet then sent to Host of IP address 151.1.12.1

model name(config-dhcp)# end ☞ Exit configuration mode

model name# show running-config ☞ Display the cofigured settings

!

[Usage Example] Configuring DHCP Client

model name# config terminal

model name(config)# interface FastEther0/0 ☞ Specify the interface to DHCP Client

model name(config-if)# ip address dhcp ☞ Specify interface IP address through DHCP without entering directly

model name(config-if)# end ☞ Move to the top of the tree to check the configured settings

model name# show running-config ☞ Check the configured settings

interface FastEther0/0

ip address dhcp

!

2.9. Configuring Transparent Bridging

VoiceFinder Gateway supports Transparent Bridging for Ethernet and Serial ports. Also it supports Bridge MIB based on RFC1286 standard for network management.

The followings are the bridge functions that VoiceFinder Gateway supports.

- The bridge functions comply with IEEE802.1 standard.
- The functions can segment Transparent Bridged Network to the logical VLAN
- The functions can support through not only FastEthernet, but also Serial, Frame relay networks.
- It supports the standard Spanning-Tree Algorithm by using BPDU(Bridged Protocol Data Unit) of IEEE standard (applied to some product models)

VoiceFinder Gateway Series support 1 Bridge-Group in general. Therefore, the concept of Bridge-Group is not used but the same id (generally 1). However, in case of supporting VLAN, many logical interfaces can be provided for the one physical interface, so more than 2 Bridge-Group can be supported.

[Usage Procedure]

Steps	Workflow Description
1	Move to configuration mode
2	Specify a value of option to be used in Bridge
3	Move to interface configuration mode
4	Apply configured Bridge-Group to the interface
5	Map Multi-Access interface such as Frame Relay
6	Apply Bridging Option parameters for other uses
7	Use show bridge or show span commands to check whether bridge is configured in a desired way and Spanning Tree Algorithm operates correctly

[Related Commands and Format]

- **bridge-group <1-255>**

  This is the interface command to configure the interface to operate as Bridge Group . Please configure the interface to be used for bridge. The same value is to be specified for connecting to the same network

- **frame-relay map bridge** <*dlci-number*>
  1. This is the interface command to configure Map, so Bridge Packet can be transmitted through the interface when the interface being used for Bridge is Frame-Relay.
  2. DLCI value is the range of 16~1007 numbers
  3. ***When bridge is used to Frame-Relay interface, Map Command must be used to enable Bridge***
  4. **When Frame-Relay is not used, the function is not provided.**

- **bridge priority** <*priority-number*>
  1. This is the interface command option to determine the priority order for blocking or forwarding when the interface is involved in Spanning Tree process.
  2. The range of the value is 0~255, as the number gets smaller, the priority order gets higher, the default value is 0.

- **bridge path-cost** <*path-cost-value*>
  1. This is the interface command option to determine the priority order for blocking or forwarding when the interface is involved in Spanning Tree process.
  2. The range of the value is 0~65535, as the number gets smaller, the priority order gets higher, the default value is Ethernet:100, Serial: 128

- **bridge hello-time** <*hello-interval*>
  1. This is a global command option to determine Hello Interval between BPDU.
  2. The range of value is 1~10 seconds by number, the default value is 2 seconds.

- **bridge forward-time**<*forward-interval*>
  1. This is a global command option to determine Forward Delay Interval.
  2. The range of value is 10~200 seconds by number, the default value is 30 seconds.

- **bridge max-age** <*max-age-time*>
  1. This is a global command option to determine a time interval for waiting from Root Bridge to BPDU.
  2. The range of the values is 100~200 seconds by number and the default value is 15 seconds.

- **no ip routing**
  1. This is a global command option to use the gateway for bridge only, without using routing functions.
  2. You must use **ip routing** command to use routing again
- **show bridge**

  This command displays Bridge Forwarding Database Entry.

- **show running-config**

  This commands displays the configured setting including bridging.

[Usage Example] Configuring Transparent Bridging

model name# config terminal

model name(config)# interface FastEthernet 0/0 ☞ Create FastEthernet interface 0/0, then Enter this configuration

model name(config-if)# bridge-group 1 ☞ Apply bridge to FastEthernet interface 0/0.

model name(config-if)# interface FastEthernet 0/1 ☞ Enter the configuration mode of FastEthernet interface 0/1

model name(config-if0)# bridge-group 1 ☞ Apply bridge to FastEthernet interface 0/1.

model name(config-if)# exit ☞ Return to global configuration.

model name(config)# no ip routing ☞ Use bridge only without using routing function

model name(config)# ip nat inside source list 1 interface FastEthernet0/0 ☞ Configure NAT, so the inside packet can go to the outside.

model name # show running-config ☞ Display the configuration

```
interface FastEthernet0/0
 ip address 172.17.213.96 255.255.0.0
 bridge-group 1

interface FastEthernet0/1
 no ip address
 bridge-group 1

no ip routing
ip route 0.0.0.0 0.0.0.0 172.17.1.1

ip nat inside source list 1 interface FastEthernet0/0
```

2.10. Configuring IP Share

Information

IP share function enables a user to share the public IP which has been allocated from VoIP gateway in IP network, which is a differentiated IP access method from NAT(network Address Translation)/PAT(Port Address Translation) using the private IP.

IP share can be divided by the dynamic IP access method, which is commonly used for IP access for ADSL or Cable Modem of the broadband network, and static IP access method which takes a fixed IP in advance to be used for ADSL modem or leased line service.

In case of dynamic IP access method, VoIP gateway uses PPPoE, DHCP to be allocated with the public IP, then the allocated public IP is transferred to the inside LAN user again. In case of static IP address, a fixed public IP, which is assigned from Network Service Provider or ISP, is to be allocated to both VoIP gateway and the inside LAN user in the same way.

For configuring the dynamic IP access, LAN0 Ethernet 0.0 port of the gateway is to be set properly to the access method (ADSL or Cable modem), then LAN1 Ethernet 1.0 port of the gateway is to be defined as DHCP server interface and deliver the allocated address without assigning IP address. For configuring the static IP access, the assigned IP address is to be configured on LAN0 Ethernet 0.0 port, no IP address is to be configured on LAN1 Ethernet 1.0.

IP Share function needs to have more than 2 Ethernet interfaces (LAN0, LAN1).

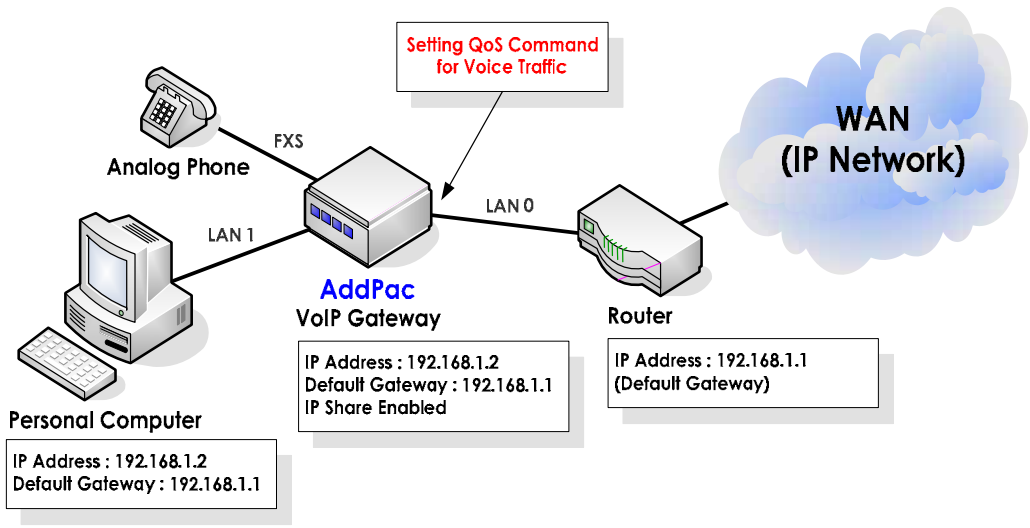


Figure 2.2 VoIP Network in IP Sharing Environment Diagram

Relaying packets transparently is similar to the bridge mode configuration. IP Share mode takes only VoIP packet of its own among the packets, which arrive on the determined destination on the network side interface. On the other hand, Bridge mode delivers the rest of the packets transparently, except those ones that are configured on the network interface side to arrive on the destination. Configuring QoS is possible by using APOS commands for voice traffic priority order control on LAN0 Ethernet 0/0 interface of the gateway accessing towards IP network. Such QoS configuration for all the incoming data and voice data to LAN1 Ethernet1/0 VoIP gateway supports maintaining the maximum voice quality through QoS provision for priority order and bandwidth control.

As no change in the existing user's environment is recommended basically, when the uplink port is configured to PPPoE Client, configuring the local interface to PPP server is recommended; when the uplink port is configured to DHCP client, configuring the local interface to DHCP server; when the uplink is configured to Static IP, configuring the local interface to Static IP as well is recommended.

[Usage Procedure]

Steps	Workflow Description
1	Move top configuration mode.
2	Configure the feature by using ip connect.
3	Generally inside is configured to LAN (FastEthernet1/0).
4	Generally outside is configured to WAN (FastEthernet0/0)

Uplink interface (LAN 0)	Local interface (LAN 1)	Configurability
DHCP	DHCP	O
	PPP	O
	Static	X
PPP	DHCP	O
	PPP	O
	Static	X
Static	DHCP	O
	PPP	O
	Static	O

Table 2.1 Configuration Chart for each Uplink Interface for VoIP Gateway

In APOS G2 version, when the uplink is static, the local interface is recommended to be static as

well and when the uplink is DHCP/PPP, the local interface is recommended to be DHCP

[Related Commands and Format]

- **ip dhcp pool** <WORD>

  1. Specify ip share active/de-active from global configuration location

- **network FastEthernet** <interface-id>

  1. Specify ip share local side interface
  2. interface id: ID of FastEthernet interface (0~1).

- **range FastEthernet** <interface-id>

  1. Specify ip share network side interface
  2. interface id: ID of FastEthernet interface (0~1).

[Usage Example] Getting the Public IP Address Assigned in DHCP Environment

You can get public IP address assigned in DHCP environment which can be used for Cable Modem application of broadband network.

APOS Command Script Screen

```
!
interface FastEthernet0/0
  ip address dhcp
  speed auto
!
interface FastEthernet0/1
 no ip address
 speed auto
!
!
ip dhcp pool default
 network FastEthernet0/1
  range FastEthernet0/0
  subnet-mast 255.255.255.0
!
dhcp server
!
ip connect default inside FastEthernet0/1 outside FastEthernet0/0
!
```

Step	Command	Description
1	# # **config terminal** Enter configuration commands, one per line. End with CNTL/Z (config)#	Change to APOS Command Configuration Mode
2	(config)# **interface FastEthernet0/0**	Start configuring Fast Ethernet interface 0/0
3	(config-if)# **ip address dhcp**	Get the public IP assigned by using DHCP
4	(config-if)# **exit**	Exit configuration mode for Fast Ethernet Interface 0/1
5	(config)# **ip dhcp pool default**	Start configuring DHCP for ip-share
6	(config-dhcp)# **network FastEthernet0/1**	Configure the interface to be connected to inside PC (or other equipment) to Fast Ethernet 0/1
7	(config-dhcp-network)# **range FastEthernet0/0**	Configure the interface assigned with the public IP to Fast Ethernet 0/0
8	(config-dhcp-network)#**subnet-mask 255.255.255.0**	Configure the interface to be connected to the inside PC to subnet-mask
9	(config-dhcp-network)# **end** #	Exit ip-share DHCP configuration mode

[Usage Example] Getting the Public IP Address Assigned in PPPoE Environment

You can get public IP address assigned in PPPoE environment which can be used for ADSL modem application of broadband network.

APOS Command Script Screen

```
!
Interface FastEthernet0/0
  no ip address
  pppoe enable
  encapsulation ppp
  pppoe-client local-interface
  ppp ipcp default-route
  ppp authentication pap calling
  ppp pap sent-username addpac password test
  ppp ipcp default-route
!
interface FastEthernet0/1
  no ip address
!
!
ip dhcp pool default
 network FastEthernet0/1
  range FastEthernet0/0
  subnet-mast 255.255.255.0
!
dhcp server
!
ip connect default inside FastEthernet0/1 outside FastEthernet0/0
!
```

Step	Command	Description
1	# # **configure terminal** Enter configuration commands, one per line. End with CNTL/Z	Change to APOS Command Configuration Mode
2	(config-# **interface FastEthernet0/0** (config-if)#	Start configuring Fast Ethernet interface 0/0
3	(config-if)# **pppoe enable**	Enable PPPoE
4	(config-if)# **pppoe-client local-interface**	Configure the present interface to pppoe of local-interface
5	(config-if)# **encapsulation ppp**	Configure PPP
6	(config-if)# **ppp ipcp default-route**	Configure to receive default router IP address from PPP server
7	(config-if)# **ppp authentication pap calling**	Configure PPP authentication to PAP
8	(config-if)# **ppp pap sent-username addpac password 1234**	Set PAP user ID to 'addpac' and password to '1234'
9	(config-if)# **exit**	Exit configuration mode of Fast Ethernet interface 0/1
10	(config)# **ip dhcp pool default**	Start configuring DHCP for ip-share
11	(config-dhcp)# **network FastEthernet0/1**	Configure the interface to be connected to the inside PC (or other equipment) to Fast Ethernet 0/1

12	(config-dhcp-network)# **range FastEthernet0/0**	Configure the interface assigned with the public IP to Fast Ethernet 0/0
13	(config-dhcp-network)# **subnet-mask 255.255.255.0**	Configure the interface to be connected to the inside PC to subnet-mask
14	(config-dhcp-network)# **end** #	Exit ip-share DHCP configuration mode

2.11. Configuring PPPoE + Bridge

PPPoE + Bridge function enables another network equipment to use a bridge function to access another PPPoE Session through xDSL modem when Multi-PPPoE is supported from one xDSL modem.

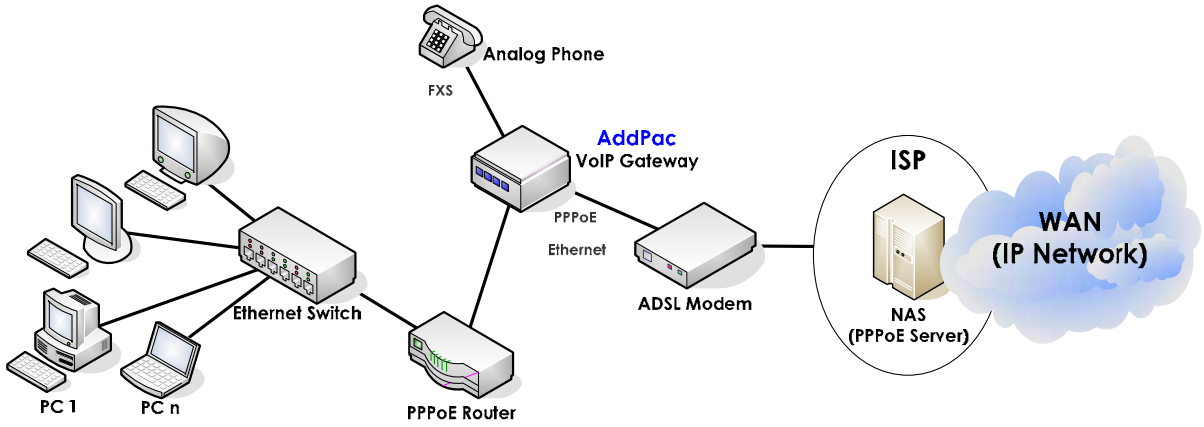


Figure 2.3 VoIP Network Diagram for PPPoE + Bridge Environment

Relaying packets transparently is similar to the bridge mode configuration.

PPPoE + Bridge mode takes only PPPoE Session, connected to VoIP gateway, of its own among the packets, which arrive on the determined destination on the network side interface and delivers the rest of the packets transparently.

Configuring QoS is possible by using APOS commands for voice traffic priority order control on LAN0 Ethernet 0.0 interface of the gateway accessing towards IP network. Such QoS configuration for all the incoming data and voice data to LAN1 Ethernet1.0 VoIP gateway supports maintaining the maximum voice quality through QoS provision for priority order and bandwidth control.

[Usage Procedure]

Steps	Workflow Description
1	Change to configuration mode
2	Configure PPPoE on WAN Interface
3	Configure bridge on WAN and LAN interfaces.
4	Disable ip routing

5	Disable Bridge Spanning Tree

[Usage Example]

APOS Command Script Screen

```
!
interface FastEthernet0/0
 no ip address
 pppoe enable
 encapsulation ppp
 pppoe-client local-interface
 ppp ipcp default-router
 ppp authentication pap calling
 ppp pap sent-username <username> password <password>
 bridge-group 1
 speed auto
!
interface FastEthernet0/1
 ip address 192.168.10.1 255.255.255.0
 bridge-group 1
 speed auto
!
no ip routing
!
```

[Related Commands]

Configuring PPPoE Bridge

The followings are the parameters for accessing Internet by using ISP in the configuration above:

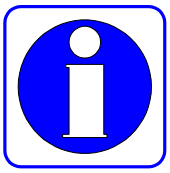
- Access ID: "**AddPac"**
- Access Password: "**1234**"
- get default-router IP (option)

Step	Command	Description
1	# # **config terminal** Enter configuration commands, one per line. End with CNTL/Z	Change to APOS command configuration mode
2	(config)# **no ip routing**	Enable IP routing
3	(config-# **interface FastEthernet0/0** (config-if)#	Start configuring Fast Ethernet interface 0/0
4	(config-if)# **pppoe enable**	Enable PPPoE
5	(config-if)# **pppoe-client local-interface**	Configure the present interface to local-interface of pppoe
6	(config-if)# **encapsulation ppp**	Configure PPP
7	(config-if)# **ppp ipcp default-route**	Configure to receive default router IP address from PPP server
8	(config-if)# **ppp authentication pap calling**	Configure PPP authentication to PAP
9	(config-if)# **ppp pap sent-username addpac password 1234**	Set PAP user ID to 'addpac' and password to '1234'

10	(config-if)# **bridge-group 1**	Enable bridge mode for Fast Ethernet 0/0 interface
11	(config-if)# **interface FastEthernet0/1** (config-if)#	Start configuring Fast Ethernet interface 0/1
12	(config-if)# **bridge-group 1**	Enable bridge mode for Fast Ethernet 0/1
13	(config-if)# **exit** (config)#	Exit the configuration mode for Fast Ethernet interface 0/1
14	(config)# **exit** #	Exit APOS command input mode

2.12. Configuring PPTP

Information



PPTP (Point-to-Point tunneling protocol) supported by APOS™ complies with RFC2637 standard.

If PPTP server can be accessed thorough LAN interface, Virtual Private Network can be configured by using PPTP.

PPTP supported by AddPac VoIP gateway is a client function which enables a user to access to PPTP server. Therefore PPTP server must exist in Internet.

Previously, VoIP and data were used to be transmitted though the tunnel by command of 'PPTP route tunnel'. However, Only VoIP can be transmitted though the tunnel by using a new command of 'PPTP route data'.

The following configuration must be taken first to transmit only VoIP packet through the tunnel and data packet to WAN.

Detailed configuration process can be verified in 'Related Commands' section.

(config-ether0.0)# encapsulation ppp

Some of AddPac Technology VoIP gateway series can support this feature.

2.12.1. Related Commands

Configuring PPTP route data

Step	Command	Description
1	# **config terminal**	Change to APOS Command Configuration Mode
2	(config)# **interface FastEthernet 0/0**	Change to interface FastEthernet 0/0 configuration mode
3	(config-if)# **ip address *IP-ADDRESS SUBNET-MASK***	Configure IP address (refer to Quick Operation Guide for configuring DHCP, PPPoE)
4	(config-if)# **exit**	Exit the mode for Fast Ethernet interface 0/0
5	(config)# **interface FastEthernet 0/1**	Change to interface Fast Ethernet 0/1 configuration mode
6	(config)# **ip address *IP-ADDRESS SUBNET-MASK***	Specify IP address.
7	(config-if)# **ip policy route-map *WORD***	Apply route-map tunnel to Fast Ethernet interface 0/1
8	(config-if)# **exit**	Exit the mode for Fast Ethernet interface 0/1.
9	(config)# **interface Dialer0**	Change to Interface Dialer 0 configuration mode
10	(config-if)# **no ip address**	No configuration for IP address
11	(config)# **encapsulation ppp**	Specify network protocol to PPP
12	(config-if)# **ppp authentication chap calling**	Set PPP authentication to CHAP (Please refer to Quick Operation Guide)
13	(config-if)# **ppp chap hostname *WORD***	Set Chap user ID to 'addpac'
14	(config-if)# **ppp chap password *LINE***	Set Chap password to '1234'
15	(config-if)# **exit**	Exit the mode for Dialer0 interface

16	(config)# **interface Tunnel0**	Change to Interface Tunnel 0
17	(config-if)# **no ip address**	No configuration for IP address
18	(config-if)# **tunnel source FastEthernet0/0**	Use FastEthernet0/0 for PPTP connection
19	(config-if)# **tunnel destination *IP-ADDRESS***	Configure PPTP Server IP address
20	(config-if)# **tunnel mode gre ppp**	Use PPP method for Tunnel
21	(config-if)# **tunnel pptp-client dial-pool-number 0**	Configure PPTP Client operation
22	(config-if)# **exit**	Exit Tunnel0 Interface mode
23	(config)# **access-list 110 deny ip any host *IP-ADDRESS*** (config)# **access-list 110 permit ip any**	Configure access-list for FastEthernet 0/1 Interface
24	(config)# **route-map *WORD* permit 1**	Change to route-map configuration mode
25	(config-route-map)# **match ip address 110** (config-route-map)# **set ip forwarding-interface Dialer0**	Apply access-list to dialer0

Cancelling PPTP

Step	Command	Description
1	(config)# **interface Dialer 0**	Change to interface dialer0 mode
3	(config-if)# **no encapsulation**	Cancelling PPP protocol in use
4	(config)# **exit**	Exit Dialer0 interface
5	(config)# **interface FastEthernet0/1**	Change to interface FastEthernet 0/1 configuration mode
6	(config-if)# **no ip policy route-map *WORD***	Cancelling route-map application

Caution : MS-Chap is not supported in PPP authentication

Default : Disable

2.13. Configuring SNMP

Information

SNMP is an application layer protocol which provides a message format for communication between SNMP Manager and SNMP Agent. There are 3 elements of SNMP system to manage network: SNMP Manager, SNMP Agent, MIB (Management Information Base)

SNMP Manager is composed of a portion of commonly used Network Management System (NMS) such as HP Openview. Agent and MIB are placed in Gateway. To configure SNMP to Gateway, you should understand relationship between SNMP Manager and Agent.

SNMP has MIB variables that SNMP Manager can request or change. SNMP Manager takes a value of management information, or gives a specific value to Agent for setting.

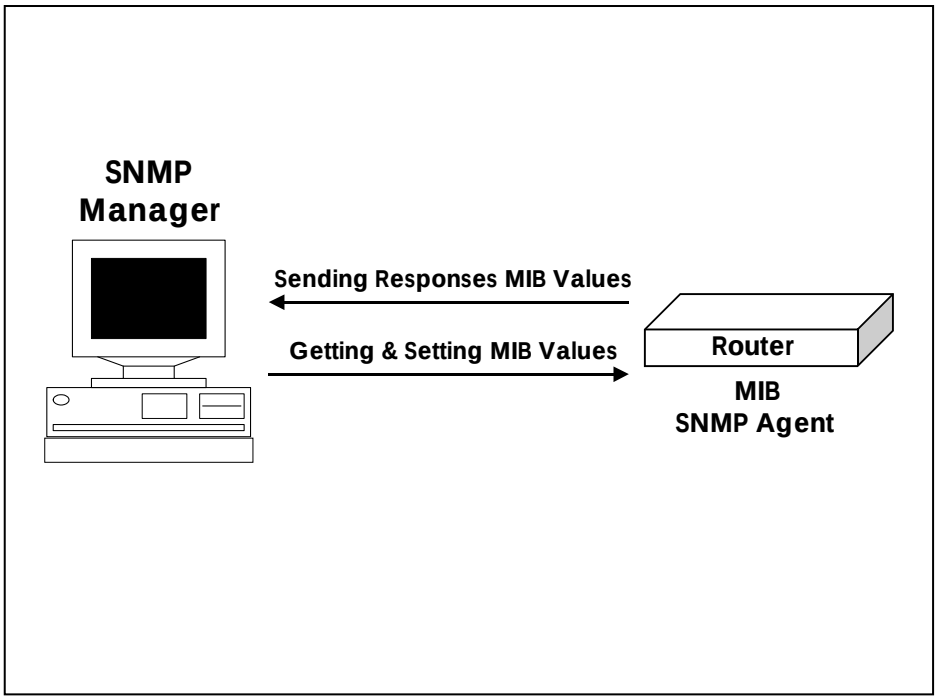


Figure 2.4 Communication between SNMP Manager and Agent

Agent is collecting data from MIB, which manages data or equipment parameters, and send it to manager or set the information by requesting from manager. An information sent from agent to manager without any request from Manager, is called Trap. Generally trap means a warning message representing the major events such as network failure, configuration changes.

The relationship between SNMP Agent and Manager is shown in Figure 2.4 . SNMP Manager sends requests to get or set MIB value to Agent and Agent responses. Agent also sends Trap for the important network events for an administrator to know.

SNMP standard is listed as to follow:

- SNMPv1 : Full Standard protocols complying to RFC1157
- SNMPv2C : organized in the following 2
  - ✓ SNMPv2: SNMP v2 Protocol complying with RFC 1902~1907 Internet Draft Standard
  - ✓ SNMPv2C: SNMPv2's Community Based Management Structure complying to RFC 1901

VoiceFinder Gateway supports all SNMPv1 and SNMPv2C.

[Usage Procedure]

Steps	Workflow Description
1	Change to configuration mode
2	Configure SNMP Community
3	Configure the settings for receiving SNMP Trap
4	Configure the rest of the parameters related to SNMP
5	Check the configured setting by using show snmp
6	Enable SNMP server

[Related Commands and Format]

- **snmp community** <*community-string*> {**ro/rw**} <*snmp-manager-ip/0.0.0.0*>
  1. Register the gateway to a specific SNMP Community
  2. Community-String : The string used for authentication when to communicate with SNMP
  3. {ro/rw} : Configure an option only to read gateway information or to change the

setting value of the gateway.

4. <snmp-manager-ip/0.0.0.0> : IP address of SNMP Manger, 0.0.0.0 sets an option to enable every NMS with the same Community-String value to work as a manager for this agent.

- **snmp host** <*trap-host-ip*> {**v1/v2c**} <*community-string*> <1-65535>
  1. SNMP version of a specific host and the time of occurrence for send trap to the gateway.
  2. <trap-host-ip> : IP Address of Trap Host (commonly known as SNMP Manager)
  3. {v1/v2c} : SNMP Version value
  4. Community-String : String The string use for authentication when to communicate with SNMP
  5. Specify a relevant port number when the standard port (162) is not in use.
- **snmp contact** <*string*>

  Specify a contact information of SNMP MIB-II System Group

- **snmp location** <*string*>

  Specify a location information of SNMP MIB-II System Group.

- **snmp name** <*string*>

  Specify a name of SNMP MIB-II System Group.

- **snmp engineID** <*string*>

  Set engine ID to be used for SNMPv3

- **snmp trap ip A.B.C.D**

  Configure SNMP Trap server in a simple way. Use snmp host commands to enable a user to configure various options.

- **snmp enable-authtrap**

  Send Authentication Violation information though one SNMP Manager when one SNMP Agent tries to access with a wrong community-string value.

- **snmp trap-community** <*string*>

  Set up the community to be used when SNMP Trap does not have community information

in SNMP Trap server.

- **snmp user WORD {ro|rw} { auth|noauth|priv}**

  Use SNMPv3 authentication.

- **snmp vacm { access|com2sec|group|view } OPTION**

  Use for the authentication by View-based Access Control Model (VACM). The usage is not recommended to due to a difficult configuration in general

- **snmp port <1-65535>**

  Configure a port of SNMP Agent. The standard value is 161.

- **snmp server**

  Enable SNMP agent

- **show snmp**

  Displays SNMP configured settings.

[Usage Example] Configuring SNMP

```
model name# config terminal
model name(config)# ☞ SNMP Config is possible from this status
model name(config)# snmp community ADDPAC-Domain1 rw ☞
Configure community-string to exchange information with all the SNMP managers
which is in ADDPAC-Domain1.
model name(config)# snmp host 131.23.1.1 v2c ADDPAC-Domain
☞ Send Trap from 131.23.1.1 of SNMP Manager to SNMP v2C protocol as to set
the string to Add-Domain
model name(config)# snmp contact HongKilDong ☞ Set up the
contact information of SNMP MIB-II System Group
model name(config)# snmp location 9FlofBuilding4 ☞ Set up
the contact information of SNMP MIB-II System Group.
model name(config)# snmp name Tac_Gateway1 ☞ Set up the contact
information of SNMP MIB-II System Group
model name(config)# snmp enable-authtrap ☞ Configure the
command to inform to all the other managers, if one of the mangers tries to
```

access an equipment with a wrong Community String

```
model name(config)# exit
Router # show snmp ☞ Display SNMP configuration status
Router# show snmp
snmp enable-authtrap
snmp community public ro
snmp community private rw
snmp community addpac ro
snmp host 10.1.1.1 version v2c
Router#
```

2.14. Gateway Management Commands

This section describes the commands, which are used in EXEC mode and Global Configuration Mode, to manage and operate the gateway in alphabetical order. Please look for each subject related to a specific setting of the gateway.

2.14.1. EXEC Mode Commands

[Command Format and Option Command]

- **clear { arp-cache/ cdp/ counters/ utilization }**

1. Reset a specific function or part of the gateway.

2. Command Options
   1) arp-cache : Initialize ARP table in use presently
   2) cdp : Initialize CDP information
   3) counters : Clear all the counters of the interfaces
   4) utilization : Clear System Utilization Information which the gateway has

- **clear { h323 / voice-port }**

1. Command Options
   5) h323 : Initialize the present H.323 call
   6) voice-port : Initialize all the ports or a specified port

- **clear { ip / ipv6 }**

1. Command Options
   7) ip : Initialize ip statistics (ip/tcp/udp/icmp) value and accounting, NAT, Flow Cache related tables
   8) Ipv6 : Initialize ipv6 statistics (ipv6/tcp/udp/icmp6) value and Ipv6 Neighbor table information

- **clock [current/running/start]**

1. Display System Clock of the gateway
2. Command Options
    1) current : shows the present time of the gateway
    2) running : shows the total operation time
    3) start: shows the starting operation time

- **clock time <1998-2099> <1-12> <1-31> <0-23> <0-59> <0-59>**
    1. Set the present time of the system. Enter the time unit by 24 hours

- **configuration terminal**

  Enter configuration mode.

- **copy running-config startup-config**

  Save the present information of operation

- **disconnect vty <1-65535>**

  Disconnect the presently connected VTY by force. VTY id is the id executed by who command.

- **Dnsquery host <option>**

  This is DNS Query test command.

- **Dnsrv SERVICE**

  This is DNS SERVICE test command.

- **Debug** <Option>
    1. Checks whether the gateway operates normally by decoding the packets staying in the gateway.
    2. Refer to the section of 4.13 equipment failure and debugging command for detailed usage and option
    3. Use ‘no debug' command’ to disable debug.

- **erase startup-config**
    1. Delete the configuration information stored in the present flash memory. If there is no change in information for the present operation, it turns into the initial mode when the

gateway is rebooted.

2. This command may cause a serious system error when it is executed in a wrong way. Therefore, use 'write' command to save the present operation information again.

- **exit**

1. Exit the present mode then move into the mode which is one case lower
2. Login again to use exit command in Exec mode

- **fsh**

1. This is the file system shell provided from the system. The usage is similar to the general user command.
2. The file system structure is same as the general file system structure with directory/ file
3. First the file system provide a similar function as DOS/UNIX as well as user command format
4. In case to access to the file system with fsh command in the beginning and executing chdir command, located in '/' directory, moves to the server directory. When the file system is access with fsh command afterwards, it is located at the end of the directory.

- **ftp**

Execute ftp function.

- **help**

This command describes Interactive Help System.

- **history**

1. This command shows all the command has been used
2. The gateway is able to keep 25 History in each mode
3. If you wan to use the command in History again, enter "! History#"

- **no {option}**

Negate the used or configured command.

- **nsupdate**

Change the information of the name server by the user command. This command is used frequently to change Dynamic DNS information.

The usage is same as nsupdate function. The script is executed, when 'update action'

command is entered. When help command is needed, enter '?'.

- **ntpdate**

  Synchronize the present time and NTP server by the user command.

- **ping** [**-flt**] [**-c** count] [**-s** packetsize] [**-S** *src_addr*] [**-t** timeout] ***Target-host-IP***

  1. Send Echo Message.
  2. Command Options.
     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]

- **Ping6**

  Execute ping function in IPv6 environment.

- **reboot**

  Reboot the gateway.

- **show {option}**

  1. Display all the information of the information of configured settings collected by the gateway. Use to grasp the operational status of the gateway.

  2. For more details of usage and option, please refer to 'Trouble Management and Debugging'.

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

  Open telnet connection from the remote host.

- **traceroute** Usage

  Usage [-6dFInrvx] [-g gateway] [-i iface] [-f first_ttl] [-m max_ttl] [-p port] [-q queries] [-s src_addr] [-t tos] [-w waittime] host [packetlen]

  Find any path to access to the remote host in IPv4 environment.

- **Traceroute6** Usage

  Usage [-dIlnrv] [-f firsthop] [-g gateway] [-m hoplimit] [-p port][-q

probes] [-s src] [-w waittime] target [datalen]

Find any path to access to the remote host in IPv6 environment.

- **who**

  Display the information of the user who is being connected to the system presently. The user id is in vty [id] format.

- whoami

  Display the information of the user who are being connected to and using the system presently.

2.14.2. Global Configuration Mode Commands

[Command Format and Options]

- **access-list { option }**
  1. Set up an access-list for the packets
  2. For more details of the content, please refer to the previous section of Access-List

- **application**
  1. Configure the user application
  2. The current version supports the mapping between H.323 Q850 Cause values

- **arp {option}**
  1. This command is to register ARP Entry statically or dynamically.
  2. Option
     1) **static** <*ip-address-number*> <*mac-address-number*> : Register Mac Address to ARP table statically for the IP host.
     2) **keep** <*keep-timer-value*> : The timer value that the gateway support is the value between 120-3600(sec).

- **arp A.B.C.D H.H.H**
  1. Register a static ARP entry
  2. Enter MAC address by 2 bytes out of 6 bytes in total

3. (example : arp 172.19.1.1 0002.a411.2233)

- **arp { keep <120-3600> | walk <1-600>}**
  1. This is a command to manage ARP entries.
  2. 'keep' is the time to maintain Dynamic ARP Entry. For a certain time passes, the entry is deleted from the table. The recommended time is 20 minutes (1200 seconds).
  3. 'walk' is a cycle time to check ARP table. If a cycle is too short and too many entries are made, the system can be over-loaded, so an appropriate value needs to be selected. The recommended value is 5 minutes (300 seconds).

- **auto-upgrade { configuration-serial | inf-serial}**
  1. Change a part of configuration as inf file format and update configuration file by using auto-upgrade.
  2. The value is used in the same format, which is used in the packing list file from auto-upgrade, as a sort of string.
  3. This is not related to the image update file.

- **Banner motd { WORD | default }**
  1. Specify a banner to be used for VTY access (in general telnet/console).
  2. Use a basic banner, which is provided from the system, in case of default
  3. If a user-specific character is needed to be used, enter the end of character of the banner to the WORD location, then it changes to input mode which can take the banner.
  4. If the ending character matches with a string, the input mode is changed to the general user command mode.
  5. The end of banner character used in general is c.

Router(config)# banner motd c ← **the end of banner character**
Enter TEXT message. End with the character 'c'
welcome system !!
thank you !!!

c ← **the end of banner character**

Router# show running-config
Building configuration...

Current configuration:

!
banner motd ^C
welcome system !!
thank you !!!

^C
!
welcome system !!
thank you !!!

Login:
Login:

- **bridge { option }**
  1. Configure bridge
  2. Please refer to bridge configuration in the previous section for more details.

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

  Set up the present system clock of the gateway.

- **clock time <1998-2099> <1-12> <0-23> <0-59> <0-59>**

  Set the clock to the present time. Enter by 24 hours unit.

- **clock timezone WORD <-23-23> <0-59>**

  Set the present time zone of the gateway. The name of the time zone is located in WORD and an offset value is entered from UTC.

Router(config)# clock timezone seoul 9

- **dhcp { server | relay }**
  1. Configure DHCP
  2. Configure the equipment to DHCP server and relay operation mode
  3. It must be a related configuration prior to DHCP sever and Relay operation mode.

4. Please refer to 'ip dhcp' section for the related configuration.
5. Please refer to 'Configuring DHCP' in previous section

- **dns domain-lookup**

  Enable DNS lookup. The default value is enabled for including DNS.

- **dns name-server { A.B.C.D | X:X::X:X }**

1. Specify DNS name-server. Possible in IPv4/ IPv6 format. First-in order determines the priority order.
2. 5 units can be entered at maximum. Even more that 5 units are entered, only 5 units can be applied. However, the order applies 5 units regardless of IPv4 / IPv6.

- **dns domain-name**

  Enter a domain name located in the equipment. It can be operated even without entering the option.

- **dns host WORD { A.B.C.D | X:X::X:X }**

1. Maintain the table without DNS Query.
2. When the name (WORD) to the table with the same host performs DNS Query, the table address replaces DNS Query.
3. When the 2 addresses have the same name, only one of them can be registered.

- **ems-server host { A.B.C.D|ALIAS }**

  Configure the server address for EMS server configuration. The server can support only one server in IP address and DNS server formats, all both of them are possible.

- **ems-server key WORD**

1. Enter an encryption key for the server authentication of EMS server configuration. If the encryption key is different, the server does not authorize.
2. The encryption key is a string and '?' is not allowed between the spaces.

- **ems-server retransmit <0-100>**

1. Set a counter for a number of retransmission for EMS server configuration.
2. Specify the counter for the maximum number of retransmission when the server transmits a message and no response is received during the timeout.
3. When this is not specified, it basically repeats the process 3 times.

- **ems-server timeout <1-60>**
  1. Set the server timeout for EMS server configuration.
  2. If this is not specified, it is basically set to 3 seconds.

- **ems-server status-interval <10-86400>**
  1. Set a frequency of reporting the status for EMS server configuration.
  2. Report the present status for the equipment to EMS server periodically.
  3. Without the setup, it is basically set to 5 seconds.

- **ems-server provisioning-required**
  1. This commend is to set up the server to request for provisioning for EMS server configuration.
  2. This setting is to be configured first in the beginning for the equipment configuration by using EMS.
  3. The configuration become disable automatically in EMS, when the configured settings of the equipment is executed normally.

- **ethernet [full-duplex|half-duplex]**
  1. This command is to set up Ethernet interface to full-duplex.
  2. The default is half-duplex.

- **exit**
  1. Exit the present mode and move to another one case lower
  2. Return to exec mode by using exit command from global configuration mode

- **help**

  This command describes Interactive Help System.

- **http server**

  Enable HTTP server.

- **http port** <1-65535>

  Change HTTP server port. When it is not specified, the number 80 is used.

- **http timeout <5-300>**

  This feature sets up a timeout when the client connected to HTTP server does not have any data in the channel. When it is not specified, it is set to 30 seconds.

- **http authentication**

  Configure HTTP server authentication. When it is not specified, all the clients can access without authentication. When it is not configures, it requires authentication. Use **no http authentication** command for not using authentication. Authentication performs its function, by using a user account and password, which are registered through username. RADIUS authentication also can be supported when it is set up to be used.

- **http access-class {ipv4|ipv6} WORD**

  This command defines the host which can access to HTTP server. When access-class is not set up, any host can access and only the host matches with the access-list can access.

- **http directory-index WORD**

  Set up default index file of HTTP server. When is not set, the directory is automatically set to index.html.

- **http document-root WORD**

  This command is to set up the root directory information in where HTTP documents are saved. The setting must start with '/'.

- **hostname** { host-name }

  Set up the name in the network of the gateway

- **interface** { **Dialer/FastEthernet/Loopback/Tunnel** } < *main-interface.sub-interface* >

  Enter configuration mode of a specific interface.

- **Line { console |vty }**

  Configure the information for a user remote access and console.

- **Exec-timeout <0-35791> <0-2147483>**

  Delete the terminal automatically when no entry is made for a certain time period. Without the setting, it is set to 10 minutes. If you do not wish to delete the terminal, even when there

is no entry, set the timeout to 0. The first entry is for minute and the second one is for second.

Router(config-line)# exec-timeout 0 0

- **access-class WORD , ipv6 access-class WORD**

  Define the host which can access to VTY server. If access-class is not configured and any host can access, only the host matching with access-list can access.

- **Session-limit <1-8>**

  Limit the maximum number of VTY which is possible to access. If the number to be configured exceeds the maximum limit, the basic setting value is the maximum. The number does not include the number of console connections.

- **logging { option }**

  1. Set up logging
  2. Refer to 'Fault Management and Debugging for more details.

- **ip nat { option }**

  1. Configure NAT(Network Address Translation)
  2. Refer to 'Configuring NAT' in the previous section for more details.

- **no {option}**

  Negate the used or configured commands.

- **ip route {option}**

  1. Configure static route.
  2. Refer to 'Routing Configuration' in the previous section for more details.

- **snmp { option }**

  1. Configure SNMP protocol for management
  2. Refer to 'Configuring SNMP' ion the previous section for more details.

- **username** { **Option** }

  1. Administer the gateway user
  2. Refer to 'User, Password, Software Image and Configured File Management' for more details.

- **utilization** { **cpu/FastEthernet** }
  1. Configure an option whether to check the availability rate, in a specific time interval, of a specific interface or CPU
  2. The default value is 1 minute.

- **write** <cr>
  1. Save the present settings.

- **erase startup-config** <cr>
  1. Initialize the settings of the equipment.
  2. Rebooting must be required.

- **script auto-upgrade WORD**

  Set up auto-upgrade. In order to use auto-upgrade, HTTP must have the packing list.

  1. **action-hour <0-23> <1-24>**
     - Record a relevant time when you want to perform auto upgrade in a specific time interval. The fist time is to start and the last time is to end. The starting and ending time can never be the same.
     - Perform auto-upgrade in the time interval when a value of action-hour is set.
     - When the initial booting and auto-upgrade are failed and retried, auto-upgrade is not performed in a specified time interval. When the time is set for action-hour, determine the time for auto-upgrade after auto-upgrade is successfully executed.
     - When action-hour is set, it may not match with the user specified retry time interval. There can 1 day difference.
     - Set the retry time interval to more than 2 days to use this function successfully.
  2. **authentication login WORD password WORD**
     - Enter the information for auto-upgrade and the server authentication.
     - Authentication is basic.
     - Not necessary to configure when authentication is not in use.
  3. **url URL**
     - Enter URL information of the server. When to enter URL, Include the packing list file.
     - The maximum string size is 120 bytes

- $MAC-ADDR$, $HOSTNAME$ changes MAC address or hostname of the equipment when a reserved string in advance is entered.

For MAC address, the address of WAN interface is used and the MAC address of FastEthernet0/0 or GgabitEthernet0/0 is used in general. The MAC address is changed to 6 byte-string. For $MAC-ADDR$, the MAC address can be changed to the same format as 0002a4112233.

- Change the hostname of the equipment for $HOSTNAME$ (examples: http://download.addpac.com/download/packing.lst)

4. **Interval retry <2-120>**
   - This is the time interval for retry when auto-upgrade fails. When it is not specified, the retry time interval is set to 10 minutes.
5. **Interval success <0-365> <0-23>**
   - This is the retry time interval when auto-upgrade is successful. When it is not specified, the retry interval is set to 30 days. The entry is to be made as day, hour in order. When 0 is entered to both day and hour, it is treated as a error
   - Generally, 2 days are recommended for the retry interval at minimum, in case of success.
6. **Auto-reboot**
   - When auto-upgrade is successfully executed, the system will do auto-reboot. Without configuration, rebooting is not carried out and just auto-upgrade can be performed.
   - When a call is in process and auto-reboot is set, the rebooting can be performed after the call is terminated.
7. **Server-port <1-65535>**
   - Specify HTTP server. If it is not specified, use the number 80. The number except 80 can be entered.

- **script ntpdate WORD**

  Set up the standard time by using NTP. To use this function, set up timezone information by using clock timezone command first.

Server { ip A.B.C.D | ipv6 X:X::X:X }

- Specify NTP server address. If more than 2 servers exist, enter the command more than twice.

1. **Version {3|4}**
   - Specify NTP server version information. When it is not specified, it is operated by 4. The version 4 is used in general.
2. **retry <2-120>**
   - This is the time interval for retry incase performing NTP fails. Without setup, the retry time interval is set to 10 minutes.

3. **resynchronize <0-72>**
   - This is the retry time interval when NTP is successfully performed. Without the setting, the retry is not performed.

- **radius-server host { A.B.C.D|ALIAS }**

  Specify the server address for RADIUS server configuration. Only one server is supported and all the IP and DNS address formats are possible.

- **radius -server key WORD**
  1. Enter an encryption key for the server authentication for RADIUS server configuration.
  2. The encryption is one string and no space or '?' between the characters are allowed.

- **radius -server retransmit <0-100>**
  1. Set a counter of retransmission for RADIUS server configuration.
  2. Specify the maximum number of retransmission when there is no response for the message has been sent for a certain timeout period.
  3. Without the setting, basically it repeats 3 times.

- **radius -server timeout <1-60>**
  1. Set the server timeout for RADIUS server configuration.
  2. Without the setting, basically it is set to 3 seconds.

- **radius -server accounting { telephony|voip} {start|stop}**
  1. Set the information related to RADIUS Accounting.
  2. '**telephony**' means the section for the interfaces of FXO/FXS/E&M/E1/T1 and '**voip**' means the section for VoIP.
  3. 'start' means the start of the call, 'stop' means the termination of the call.
  4. When this is not configured, RADIUS Accounting information is not transmitted.

5. RADIUS Account information is transmitted to the point of where the section is configured. In general, the accounting information is transmitted to the stop and start points.

2.15. Fault Management and Debugging

This section describes how to mange and process the fault when it takes a place in Voice Finder Gateway. The gateway provides Show, Debug and logging commands for its fault management.

2.15.1. Logging Commands

Logging command provides logging the operational status to manage the equipment, determines the level and end out the information to a specific host outside.

The followings are the related commands for Logging configuration.

- **logging on**

  Enable logging for all the possible destinations.

- **logging command**

  This the command used for logging the user commands which can be entered.

- **logging event {option}**

  1. This is the command to set the conditions for logging.
  2. Option
     1) **0-emergency** : Logging the level less than Emergency
     2) **1-alart** : Logging the level less than alert
     3) **2-critical** : Logging the level less than critical
     4) **3-error** : Logging the level less than error
     5) **4-warning** : Logging the level less than warning
     6) **5-notification**: Logging the level less than Notification
     7) **6-infonational** : Logging the level less than Informational
     8) **7-debug** : Logging the level less than debug

- **logging host server ip/ ipv6 <ip-address> <port>**

  1. This is the command to set the condition for the host to where the logging information is to be sent to.

2. Option
    1) **ip/ipv6** <*destination-ip-address*> : Specify IP/IPv6 Address of the remote host for sending logging information
    2) **port** [*port-number*] : Spicy the port number of Remote Host for sending the logging information

- **logging format addpac**

  Specify Syslog data format to be sent to the logging server. The server is not specified in general.

2.15.2. Show Commands

Show command enables the equipment administrator to see all the configured settings.

Show command can be sued for Exec mode. The followings are the syntax.

- **Show** {**option**} : Displays the option contents

The followings are the option commands related to show:

- **ip access-list** *[access-list-number]*
    1. This is the command to display the configured access-list.
    2. Please refer to 'Configuring Access-List' for more details.

- **arp** *[ip-address for ARP entry]*

  This is the command to display ARP table

- **bridge**
    1. Displays forwarding/blocking database of bridge.
    2. Please refer to 'Configuring Bridge' in the previous section for more details.

- **cdp** {entry|neighbor|<cr> }

  Display CDP related information.

- **clock** [**current/running/start**]

  This is the command to display the system clock of the present gateway.

- **ip dhcp** *[option]*

  1. Displays the configured setting of DHCP
  2. Please ref to 'Configuring DHCP' in the previous section for more details.

- **Ip icmp statistics**

  Display System Clock of the present gateway

- **Interface** [**Dialer/FastEthernet/Loopback/Tunnel**]

  [<*main-interface*>.<*sub-interface*>]

  Describe the status and settings of interface.

- **logging** [**option**]

  Display the content of Logging Buffer

- **ip nat** *[option]*

  1. Display NAT settings
  2. Please refer to 'Configuring NAT' in the previous section for more details.

- **ip route** {**connected/static/ospf/rip**}

  1. Display the route information table which has been determined.
  2. Displays each table formed by each algorithm by using the options of OSPF/RIP/Static.
  3. Refer to 'Configuring Routing' in the previous section for more details.

- **running-config**

  Displays currently running configurations.

- **startup-config**

  Displays the saved Configuration File.

- **snmp**

  Displays the configured setting for SNMP protocol status and options.

- **system task**

  Display the information and status for driving task from the present gateway.

- **ip tcp [statistics/port]**

  Display the information and status of the external system which is connected to TCP in the information for connection to the present gateway.

- **ip udp [statistics/port]**

  Display the information and status of the external system which is connected to UDP in the information of connection to the present gateway.

- **ip statistics**

  Display IP related statistic information of the equipment.

- **ip icmp statistics**

  Display ICMP related statistic information.

- **ip accounting**

  Display IP accounting related statistic information of the equipment.

- **ip local pool**

  Display the IP address Pool information of the equipment.

- **ip interface brief**

  Display IP interface information of the equipment briefly.

  Router# show ip interface brief

Interface	IP-Address	OK? Method Status	Protocol
Loopback0	127.0.0.1	YES manual up	up
FastEthernet0/0	172.16.9.26	YES manual up	down
FastEthernet0/1	unassigned	YES NVRAM up	down

  Router#

- **ipv6 statistics**

  Display IPv6 related statistic information of the equipment.

- **ipv6 tcp|udp port**

  Display IPv6 TCP/UDP port related information of the equipment.

- **ipv6 tcp|udp statistics**

  Display the statistic information related to IPv6 TCP/UDP of the equipment.

- **ipv6 neighbor**

  Display IPv6 Neighbor information of the equipment. IPv6 Neighbor information is similar to ARP information of IPv4.

- **ipv6 access-list**

  Display IPv6 access-list information of the equipment.

- **ipv6 route**

  Display IPv6 routing table information of the equipment.

- **username**

  Display the registered user information to the gateway.

- **utilization** { **FastEthernet/cpu** }

  Displays presently configured utilization status and its value.

- **version**

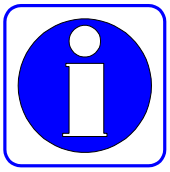
  Display the software driver version and hardware information.

- **voip-interface**

  Display all the status for VoIP interface of the present equipment.

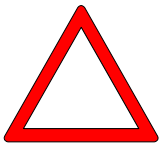
2.15.3. Debug Commands

Information



Debug command provides a function enables an administrator to see any fault in network or equipment settings by decoding a specific packet passing through the router equipment. Debug command can be used in Exec mode.

Caution



The caution for using Debug command is that the system takes a lot of resource. Therefore, it is recommended to use Debug command for the necessary portion at the minimum range. Turn off the debug functions that are not in use because they can degrade performance of the entire system.

For debugging IP or TCP packet especially from Telnet Virtual Terminal, debugging function can operate continuously and recursively by debugging TCP/IP packets continuously and displaying debugging message. In this case performance of the router can be dropped definitely.

To avoid this case for debugging IP or TCP packets from Telnet Virtual Terminal, it is strongly recommended to use access-list and remove such debugging function for TCP/IP packets of the Telnet terminal from the entire debugging function.

Debug command can be used in exec mode and the Syntax is listed as to follow:

- **debug** {**option**} : Enable debugging.
- **no debug** {**option**} : Disable debugging

The followings are the debugging related command options:

- **ppp** {**chap/error/negotiation/packet** }
  1. Display as to decode PPP settings or operational status
  2. The followings are the details of the option
     1) chap: show exchange of information during chap setup process by decoding
     2) error: show error information from PPP process by decoding
     3) negotiation: show the PPP Link Negotiation by decoding
     4) packet: show PPP packet by decoding

- **arp** packet

  Display ARP packet by decoding.

- **cdp {** packet|error }

  Display CDP packet by decoding.

  Display the error information if there is any during CDP packet process.

- **Domain-name {query|reply}**

  Display DNS query and Response packet information

- **ip** {**icmp/tcp/udp/nat/dhcp/packet** }

  1. Display TCP/IP packet passing through the gateway by decoding.
  2. The following are the details of the option:
     1) icmp: Display ICMP packet by decoding
     2) tcp: Display TCP/IP packet by decoding
     3) udp: Display UDP/IP Packet by decoding
     4) packet: Display IP Packet by decoding
     5) nat: Display NAT address translation process by decoding
     6) dhcp: Display DHCP Packet by decoding. Select an additional option for DHCP

- **ipv6** {**icmpv6/udp/rtadv/packet** }

  1. Display IPv6 packet passing through the gateway by decoding.
  2. The following are the details according to the option:
     1) Icmpv6: Display ICMv6 packet by decoding
     2) udp: Display UDP packet by decoding
     3) packet: Display IPv6 packet by decoding
     4) rtadv: Display IPv6 Routing Advertisement packet by decoding

- **ntp** packet

  Display NTP packet by decoding.

- **ppp { authentication|compress|error|fsm|negotiation|packet|state|user}**

  Display PPP packet by decoding.

  1. Authentication: Display PPP authentication related debugging information

2. Display PPP compression related debugging information
3. error: Display PPP error related debugging information
4. fsm: Display PPP status related debugging information
5. negotiation : Display PPP protocol negotiation related debugging information.
6. packet: Display PPP packet related debugging information
7. state: Display PPP inside status change related debussing information.
8. User: Display PPP user data (IP/IPv6/IPX) related debugging information

- **pppoe { error|information|packet }**

  Show PPPoE packet by decoding.

  1. Error: Display PPPoE error related debugging information
  2. Information: Display PPPoE protocol related information
  3. packet: Display PPPoE packet related debugging information.

- **radius**

  Show RADIUS packet by decoding.

- **snmp {all|error|info|packet}**

  Show RADIUS packet by decoding

  1. error: SNMP error related debugging information
  2. Info: Display SNMP protocol related information
  3. packet: SNMP packet related debugging information.
  4. Display all of the error, info, packet information

- **vlan packet <16-2048>**

  Show VLAN packet by decoding.

2.16. User, Password, Software Image and Configuration Files Management

This section describes user registration and change, recovering administrator's password, downloading the software image and backup, back, configuration file backup and restore.

2.16.1. User Registration and Change

This section describes the gateway user registration and change of password, change of user's root.

The followings are the related commands to user administration:

- **username** {**option**} : Register or change a user.

The followings are the related command options to username command:

- **username** <login-name> **nopassword user**
  1. Register the gateway user.
  2. The user's root level is user, login is possible without password

- **username** <login-name> **password <0/7>** <login-password> **<administrator/operator/user>**
  1. Register the gateway user
  2. Determine encryption of the password
     - 0: no encryption
     - 7: encrypt, can not be displayed in show running-config
  3. Set up the user's root level
     - administrator: all the privileges are given
     - operator: not allowed register a new user, limited to change GKIP/SIP-Server
     - user: only allowed to change one's own password and to read other information
- **username** <login-name> **password** <login-password> **<administrator/operator/user>**

1. Save the password by text
2. Set up the user's root level
   - administrator: all the privileges are given
   - operator: not allowed register a new user, limited to change GKIP/SIP-Server
   - user: only allowed to change one's own password and to read other information

2.16.2. Password Recovery

The password is mandatory to check the status of the gateway or to change the configuration of the gateway. The gateway administrator must memorize the password. This section describes the ways to recover the password when the password is lost due to an unavoidable circumstance.

The followings are the process and commands:

[Usage Procedure]

Steps	Workflow Description
1	Connect to the console and prepare for password recovery Password recovery can be performed only on the console.
2	Initialize the system (power off/on)
3	After the initial message is displayed, enter Ctrl+x and Ctrl+c once or twice repeatedly.
4	Wait a moment till Boot Mode is reached.
5	The root password can be displayed by 'show password command APOS G2 manages the password by file of **apos.cfg**.
6	Reboot the system.
7	Login by the verified password

Initialize the system. Initialize by the booter mode which is not the gateway program. To get into the booter mode, enter Ctrl+x and Ctrl+c onece or twice repeatedly after all the booter mode messages are displayed.

When you get into the booter mode, you may see the prompt screen with 'BOOT#'. Please refer to the screen below

The initial login name and password for the boot mode is **root/router**).

```
System Bootstrap, Version 1.2
Decompressing the image:
########[OK]

Welcome, APOS(tm) Boot Kernel Version 5.0.14.
Copyright (c) 1999-2005 AddPac Technology Co., Ltd.

Login:
Login: ETH0/0: Link is Down
Interface FastEthernet0/0, changed state to DOWN
ETH0/1: Link is Down
Interface FastEthernet0/1, changed state to DOWN

ETH0/0: Link is Up 10 Mbps Half Duplex
Interface FastEthernet0/0, changed state to UP

Login: root
Password: ****
Booter>
```

[Boot Mode Login Screen]

Enter '?' to see all the possible commands to be used in the booter mode.

BOOT# ?

clear	Reset functions
clock	Configure time-of-day clock
configure	Enter configuration mode
copy	copy configuration
disable	turn off privileged mode command
end	end current mode and change to enable mode
exit	Exit the EXEC
fsh	enter to local file system
ftp	internet standard file transfer protocol (ftp)
help	description of the interactive help system
no	negate a command or set its defaults

ping	Send echo messages
quit	exit current mode and down to previous mode
reboot	reboot system bys command
show	Show running system information
terminal	set terminal line parameters
tftp	transfer files to and from a remote machine using tftp
who	display who is on vty
write	write running configuration to memory, network, or terminal

Check the present commands of 'root', The following screen shows for the password of 'root' is router.

Use **fsh** to check the settings of **apos.cfg**. If you need to change to the factory default mode due to a problem, you can access again with a basic password by rebooting after deleting **apos.cfg**.

```
Booter#
Booter# fsh
fsh:/flash>ls
apos.cfg
booter.cfg
booter.cfg~
recentcall.cal
tmp
vp200_kr_g2_v8_41_026.bin

                        1 directories,  5 files
fsh:/flash>type apos.cfg
 #LN  LEN   Content
 ---  ---   ------------
   1    2   !
   2   40   ! APOS(tm) configuration saved from vty
   3   25   !  2007/10/24 20:55:22 !
   4   17   version 8.41.026
```

```
  5    2   !
  6   22   hostname VP200-112.31
  7    2   !
  8   44   username root password router administrator
  9    2   !
 10    2   !
 11   20   interface Loopback0
 12   32    ip address 127.0.0.1 255.0.0.0
 13    2   !
fsh:/flash>
```

[Password Verification Screen]

2.16.3. Software Image Upgrade and Backup

The software function of VoiceFinder gateway is too updated periodically due to an upgrade or bug fix. This chapter describes the process of how to upgrade or backup the software.

The following are the related commands to upgrade or backup the gateway software.

When FTP is used, you may need to use a user name and password to log into the gateway. If you need to upgrade the new gateway software by using FTP, use put. If you need to upgrade the gateway software

The following screen shows the case for downloading the gateway software which is in operation. Use 'put' command instead of 'get' after copying the software, to be upgraded, to the present directory.

[Usage Example] An Example of Software Backup by FTP

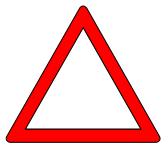
```
155 sun10:#> ftp 211.170.87.221
Connected to 211.170.87.221.
220 Gateway FTP server (Version 1.12) ready.
Name (211.170.87.221:noname): root
331 Password required for root.
Password:
230 User root logged in ok.
ftp> bin
200 Type set to I.
ftp> get Gateway.bin
200 PORT command successful.
150 BINARY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

The message on the user console is displayed as to follow:

```
" Gateway Software " is updated
```

Caution



The process of upgrading and backup the software image applies in the same way as from the gateway program and booter mode in operation. When there is a problem in the gateway program during its operation, the software image can be upgraded in the same way as above, in the booter mode.

2.16.4. Backup and Restoring Configuration File

The gateway has been designed to restore the configuration file to the flash memory in its inside. However, depending on the administrator's situation, sometimes it is necessary to restore the configuration which has been backed up already or back up the configuration file.

This chapter explains how to back up the gateway configuration file or restoring process and it related commands.

The process for backup of configuration file and restore is the same process as software image upgrade. The only difference is the configuration file name, which is **gateway.cfg**. As the process is same as software image upgrade and backup process, FTP/TFTP is used for the process. When the restoring process is completed, a message is displayed on the screen as 'Config Database is updated.

Use **put** command for restoring and **get** command for backing up the configuration file. The following is the example for using TFTP.

[Usage Example] Backup and Restoring Configuration by using FTP

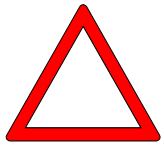
```
56 sun10#> ftp 211.170.87.221
Connected to 211.170.87.221.
220 Gateway FTP server (Version 1.12) ready.
Name (211.170.87.221:noname): root
331 Password required for root.
Password:
230 User root logged in ok.
ftp> bin
ftp> get apos.cfg
200 PORT command successful.
150 BINARY data connection for Gateway.bin (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:/#>
```

The message on the user console is displayed as to follow:

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

Caution



The process of upgrading and backup the software image applies in the same way as from the gateway program and booter mode in its operation. When there is a problem in the gateway program during operation, the software image can be upgraded in the same way as above, in the booter mode.

2.17. Auto-Upgrade

When a new feature is added to the gateway equipment, the software (firmware) needs to be upgraded.

For the first stage, the user can upgrade with a new software by using a transmission method such as ftp.

This feature configures the gateway to access to a specific server and compare its own OS and configuration, and then determines downloading and rebooting.

Step	Command	Description
1	model name# **configure terminal**	Get into configuration mode
2	model name(config)# **script auto-upgrade** *WORD*	Move to auto-upgrade configuration mode
2	model name(config-script)# ? *{action|authentication|auto-reboot| interval|proxy|url|verbose|action-hour}*	- action : a point of time for Auto-Upgrade (displayed after URL entry) - authentication : Web server (auto upgrade server) Login Name /Password - auto-Reboot: determine rebooting image after upgrade - interval : Set the retry time interval for fail or success - proxy : Enter proxy server IP Address - url : Enter Web server (auto upgrade server) url - verbose : screen display mode when to upgrade - action-hour: Set the time for auto-upgrade

The following details describe the features of auto-upgrade server to use auto-upgrade functions.

The followings are the configuration details for auto-upgrade server:

1. Install HTTP server program
2. Save APOS™ image
3. Create pacing list file (file name, size, version information are required)
4. Account and password setup (optional)

The configuration for the features of auto-upgrade server is limited to Packing List, because the rest of the contents are for Windows or HTTP. Therefore, the scope of APOS™ related configuration is limited to Packing List

```
#-#auto-upgrade 101 Packing List for AddPac APOS 1.01
R/ap4820/packing.list
./ap4820 g2.bin 8.24 4008724 0x12345678 Tue, 05 May 1998 20:02:42
GMT
```

[Example of Packing List File]

The comment starts with '#', except **#-#auto-upgrade** which stands for the version of the packing list. The current version is 1.01 (101 next to **#-#auto-upgrade** represents the version).

The first letter 'R' starting with **R/ap4820/packing.list** mean 'Redirect'. The line with R tag should be placed always first in the packing list (except the comment line). R tag is used when location of a file in the lower case is different from the present location of the packing list. For an example of **ap4820_g2.bin**, no presence of 'R' tag means the file is in the directory located in the packing list On the other hand, presence of 'R' tag means **HTTP home / ap4820 / ap4820_g2.bin**.

'.' in the packing list represents the file information. In this case of the file, only APOS (.bin) is considered only, but the order is listed as to follow:

- File name: APOS Release formal version information (when the administrator changes the version name at discretion, it can be operated with error)
- Version Name: Displaying a string such as 8.24 format
- Size: decimal or hexadecimal. For hexadecimal, 0 x prefix must be used
- Checksum: hexadecimal, 0x prefix must be used (This is the field can not be checked at this present)
- Others: the information required by the administrator

Chapter 3 . Voice Configuration and the Related Commands

This chapter describes the configuration and commands for operating voice integrated features of VoiceFinder Gateway.

3.1. Overview

3.1.1. Voice over IP

VoIP is a technology to transmit voice traffic, such as voice telephone or fax, to IP network. By using Digital Signal Processor (DSP) integrated in VoIP equipment, a voice signal can be segmented by frame units and send to other devices connected to each other. DSP of other devices combines the voice packets each other and delivers the voice signal. The voice packets transmit to each other as to comply with H.323 of ITU-T standard.
Voice application is different from data application as it is responses more sensitively to delay. Therefore, when VoIP application is used, the network configuration needs to be well adjusted, so voice application can be used smoothly from VoIP equipment. These adjustments include Protocol Tuning to improve QoS, consideration of Traffic Shaping, adjustment of MTU Size.

All the commands being used for VoiceFinder Gateway can be accessed to console, telnet or web-client.

Voice over IP is a function basically processed from the software. Voice port of the gateway supports a specific signaling type to process voice.

3.1.2. Codec and Mean Opinion Score

Codec (Coder-Decoder) is a device which converts the voice analog signal to the digital beat stream and the digital beat stream to the analog voice signal.

In general, PSTN uses PCM Codec. PCM samples the analog signal 800 times per seconds (the sampling interval is 125 micro-seconds) and converts the analog sound to the digital signal by changing each sample to the numeric code. For this PSTN network, PCM uses 8 bit for the code, therefore, the standard bandwidth requirement is 64 Kbps.

Sometimes the other compression format, Adaptive Differential Pulse Code Modulation (ADPCM), is used. A typical example of ADPCM is to encode by using 4-bit in ITU-T G.726 standard, the bandwidth is 32kbps. This 4-bit compression format does not encode the voice amplitude directly, but it encodes the amplitude difference rate as a very elementary prediction method.

PCM and ADPCM utilize the characteristics of repetition with Waveform and they are the example of compression technique. The new compression techniques for utilizing the characteristics of voice generating source has been developed during the last 10 or 15 years. These techniques were used for the signaling process for compressing voice by sending the cyclic information which represents the original voice vibration and the lingual shape. So this information requires a bandwidth for transmission. These techniques can bind 'source' codec together which includes the variance formats such as LPC (Linear Predictive Coding), CELP (Code Excited Linear Prediction) and MP-MLQ (Multi-Pulse, Multi-Level Quantization).

CELP, MP-MLQ, PCM and ADPCM coding method has been standardized in the recommendation of the international standardization institution, G. series of ITU-T.

The followings are the formats of voice coding which are used most often in a call shop and packet voice.

- G.711 : This is the format that stipulated 64Kbps PCM voice coding, which has been explained earlier, and delivers voice through PSTN or PBX.
- G.726 : This format stipulates 40, 32, 24, 16Kbps of ADPCM voice coding and commonly used in PSTN, PBX. However, PSTN and PBX network must have ADPCM processing system.
- G.729 : This format stipulates CELP compression, which can code voice to 8Kbps stream. In this standard, there are 2 variance formats (G.729 and G.729a). Complexity involved in calculation has a big difference in these 2 variance formats, but the both provide 32 Kbps

of ADPCM voice quality.

- G.723.1 : This compression format is a part of the entire H.323 series standard which stipulates a technique to compress the voice of multimedia service or other audio signaling elements to a very low-bit speed. There are 2 different kinds of the bit speed in this coder, which are 5.3Kbps and 6.3 Kbps. 6.3 Kbps of the bit speed is based on MP-MLP technology and has an outstanding quality,. As 5.3Kbps is based on CELP technology, it has a good quality and provides more flexibility to the system designer.

As codec becomes more and more dependant on the subjectively tuned compression techniques, the quality measurement value-oriented standards, such as the total harmonic distortion or noise rate against signaling, become less and less relative to the recognized quality of codec. Therefore, the general bench mark, which is widely used in these days, to digitalize the performance of the voice codec, is Mean Opinion Score (MOS). Voice quality and sound quality is different from one audience to another and it is important to get a sample material and wide range of audience. MOS is carried out by an audience group who ranks from the score 1 (the worst) to 5 (the best) for each voice sample

The table blow represents MOS score for the compression format used most often and required Processing Power.

Compression Formats	Bit Speed (Kbps)	Processing (MIPS)	Frame Size	MOS Score
G.711PCM	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

Table 3.1 Compression Formats and MOS Scores

As far as the maintenance cost for configuring the infrastructure, which is required to maintain the general telephone quality level, is considered, it is effective to configure all the calls by using a coder with a low bit speed. However, there are some disadvantages over compressing voice repeatedly many times. As it is shown in Table 3.1, the one disadvantage is occurrence of Tandem Encoding when coding and decoding is performed. As an example of compressing voice many times by G.729 Coder, when encoding and decoding is performed 3 times, the signal of MOS deteriorates its quality from 3.92 (very good) to 2.68 (not to be used normally).

Another disadvantage to be considered is delay. There are 2 kinds of delay essentially in telephone or VoIP network. They are Propagation Delay and Handling Delay. Propagation delay is a kind of delay due a medium such as copper or fiber optic used in a network. The delay in delivery is so small, so human's ears can not feel, but when this delay is accumulated, it drops the voice quality as a result. A type of delay which can be managed by an administrator is Handling Delay. This delay can be divided by the delay, which take a place, in encoding and decoding process of the codec and the delay, which takes a place in the packet processing of the gateway. VoIP Gateway determines a path for the entered voice packet and it is configured to do the task of moving to the output part. You should consider the status of the network and QoS support of the equipment for Encoding/ Decoding process delay when you select a type of codec. The following Table 3.2 presents the delays in processing time for each codec.

Compression Format	Bit Speed(Kbps)	Compression Delays (ms)	MOS Scores
G.711PCM	64	0.75	4.1
G.729 CS-ACELP	8	10	3.92
G.729a CS-ACELP	8	10	3.7
G.723.1 MP-MLQ	6.3	30	3.9
G.723.1 ACELP	5.3	30	3.65

Table 3.2 Delays in Code for each Compression Formats

3.1.3. Dial Peer

One of the things that you must understand for VoIP installation and configuration is Dial Peer. A dial peer, also know as an addressable call endpoint, is a device that can originate or receive a call in a telephone network. The dial peer is used to set the characteristics contributed to a call leg in all the voice technologies, in which a telephone call can be executed. As it is shown in Figure 3.1 and Figure 3.2, the call leg a section in where the voice call connected between the 2 points. There are 4 call legs presented between the call originating telephone and ending telephone. The 2 call legs in Figure 3.1 are shown in a view point of source gateway and Figure 3.2 shows the call legs in a view point of destination gateway. An operator or administrator of voice network divides the call originating point and destination, then apply a specific optional function to the call leg. These optional functions to be applied to the call leg

include Quality of Service (QoS), Compression/decompression (Codec), Voice Activation Detection (VAD), FAX Rate and others.

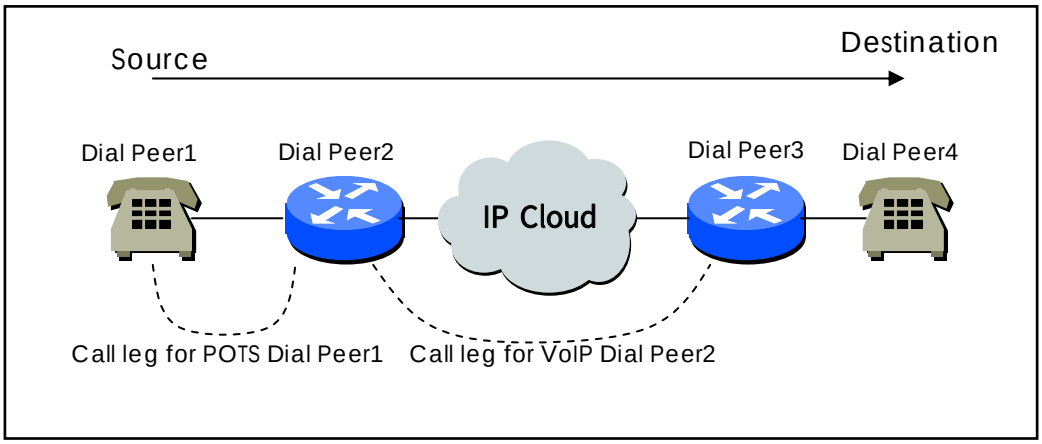


Figure 3.1 Dial Peer Call Leg from a View Point of Source Gateway

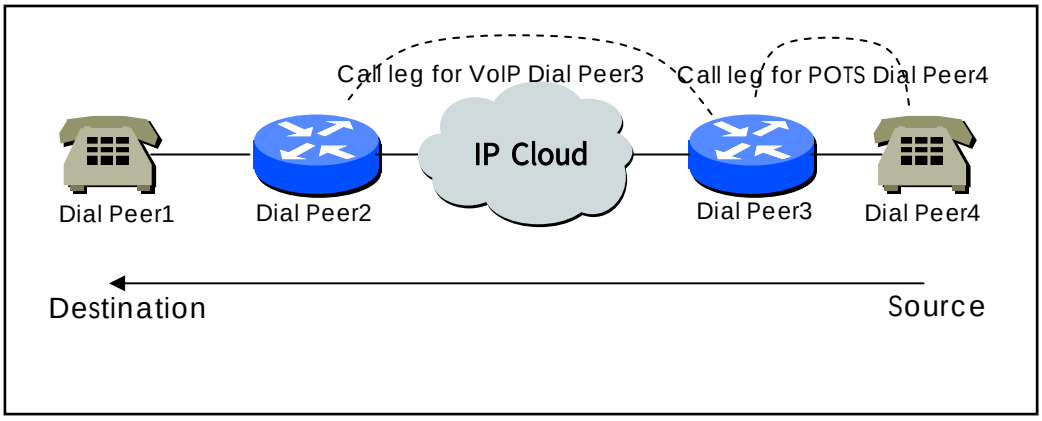


Figure 3.2 Dial Peer Call Leg from a View Point of Destination Gateway

In VoIP, the dial peer can be categorized as either VoIP (Voice over IP) dial peer or POTS (Plain Old Telephone Service) dial peers.

- POTS Dial Peer: This is a type of dial peer includes traditional telephone network devices such as phone sets, cell phones and fax machines. POTS dial peer means the characters contributed to the existing telephone network. POTS dial peer indicates a specific voice port in telephony network devices. When to configure POTS dial peer, **port** command and **destination-pattern** command must be set up first. **destination-pattern** command is, related to POTS dial peer configuration to set up a telephone number with connection to POTS dial peer. **Port** command is, related to POTS dial peer configuration, to configure a specific logical dial interface (a voice port of the gateway connected to the local POTS network)

- VoIP Dial Peer: This is a type of dial peer which means the characteristics contributed to packet network connection - for instance, IP network within Voice over IP network. Voice-Network dial peer refers to a specific voice device such as VoIP capable computers, routers and gateways within a network. When to configure VoIP dial peer, **session-target** command, **session-protocol** command and **destination-pattern** command must be set up first. **destination-pattern** command is telephone number or pattern designated to be routed to this peer. **session-target** command is to set up the destination static IP address or server for voice-network dial peer. **session-protocol** command should be set to use SIP protocol for this VoIP peer.

3.1.4. Voice Port

Voice port commands configure the characteristics contributed to voice-port signaling type. The analog voice port of the gateway supports the following two basic voice signaling formats:

- FXS (Foreign Exchange Station) Interface : This is an interface of RJ-11 connector type which connects the basic telephone devices, such as telephone, key phone and PBX, to be connected. FXS connection provides ring, voltage and dial tone.
- FXO (Foreign Exchange Office) Interface : This is an interface of RJ-11 connector type which connects PSTN line, standard private PBX to be connected.

The signaling type used for these analog ports is different for each module structure. **voice-port** *port-number* is used as a command line to configure the voice ports of the gateway. In addition to above analog interfaces, E1, T1 and E&M interfaces are supported by module type.

3.2. Configuring VoIP interface

Information

VoIP gateway with APOS G2 version has many different network interfaces. Most of all, it is necessary to specify an interface for VoIP service. At a default, FastEthernet 0/0 interface is specified for providing VoIP service and other interfaces can be specified by the following procedure. When VoIP interface is changed while in VoIP service, the call in present connection is terminated, and then registration to gatekeeper (SIP server) is performed. Therefore, it is recommended to specify VoIP interface at the time of the initial settings for the system and not to make any change afterwards. If the specified VoIP interface does not have IP address assigned, the contents of VoIP related settings can not be verified. Therefore, specifying VoIP interface and setting up IP address must be done prior to configuring VoIP related settings. During the provision of VoIP service, any change in IP address of VoIP interface also terminated the call in present connection and registration to the gatekeeper (SIP server) is performed.

Step	Commands	Description
1	model name# **configure terminal**	Get into configuration mode
2	model name(config)# **voice-interface** *interface-name*	Specify an interface of the installed gateway. For instance, FastEthernet 0/0, FastEthernet 1.0 are the names of interface.

3.3. Numbering Plan, Dialing Operation and Configuring Dial Peer

3.3.1. Numbering Plan

Configuration of VoIP gateway (or router) starts with setting up an efficient and expandable numbering plan.

A general telephone network has a hierarchical number structure of (country code) + (area code) + (prefix number) + (subscriber's number) and this hierarchical numbering plan is advantageous in expansion.

Each gateway in VoIP network belongs to PBX of the general telephone network, so set up the number plan which can be suitable to the size of VoIP network.

When to set the numbering plan, it is important to consider an option whether to configure the gateway with a gatekeeper.

If the existing gateway is to interoperate with the gatekeeper, the gatekeeper should follow the pre-defined numbering plan.

The simplest way of number setting is to match with the pre-existing public telephone number of the location, where the gateway will be installed. This way is advantageous for retrying the call to the public telephone network, when to interoperate with the other VoIP gateway or when a VoIP call fails.

Besides this plan, you need to configure a network by setting up a private numbering plan where private telephone network is deployed.

3.3.2. Configuring Dial Peer

3.3.2.1. Inbound Dial Peer and Outbound Dial Peer

The dial peer can be used for both of inbound and outbound call legs. Please be reminded that the vocabulary of 'inbound' and 'outbound' are defined as in a position of the gateway. In other word, inbound call leg means a call leg directing from the outside of the gateway towards the gateway, on the other hand, outbound call leg means a call leg directing from the gateway toward the outside. In case of inbound call leg, the dial peer always should have the setting for calling number and port destination.

Outbound call leg always should have the dial peer connected to the outbound call leg.

Generally, destination pattern is used to define the outbound dial peer.

POTS peer relates to a telephone number of a specific voice port, which enables a call to be incoming and outgoing. VoIP peer is to designate a specific VoIP device and connects the incoming call to be received and outgoing call to be sent out. To set up VoIP connection, both of POTS and VoIP peers are needed.

Configuring a VoIP communication is very similar to configuring IP static router. In other words, a specific voice connection needs to be configured for both of the pre-defined end points on each side.

As it is shown in the following Figure 3.3, (from a view point of POTS Dial Peer1), POTS dial peer fixes the outgoing call (telephone number or the call originated from the voice port). VoIP dial peer fixes destination by connecting destination phone number to a specific IP address.

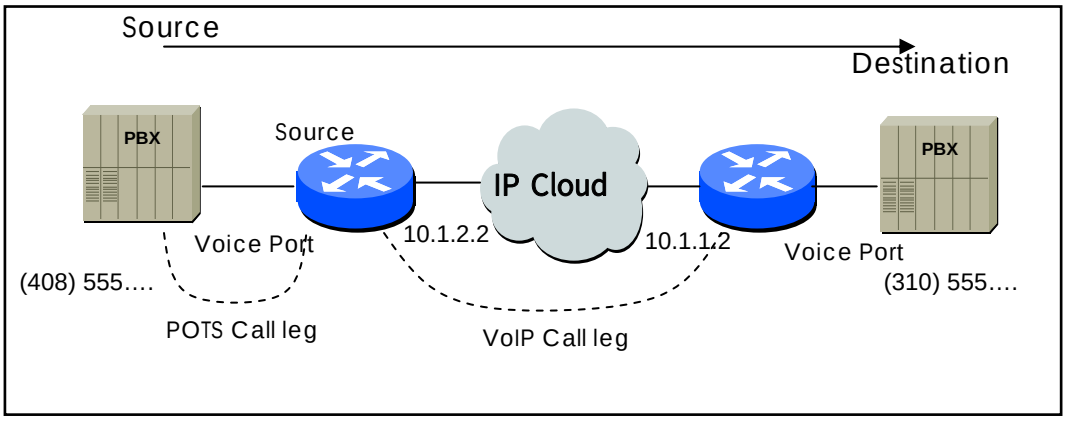


Figure 3.3 Outgoing Call from a View Point of POTS Dial Peer 1

To configure call connection between source and destination in the above Figure 3.3, you may use the following commands to Gateway 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
```

From the example above, the last 4 digits in destination patter of VoIP dial peer can be replaced with wildcard. This means all the calls with dial number of '1310555' originating from VoIP gateway with 10.1.2.2 are to be connected to 10.1.1.2 VoIP gateway. In other words, 10.1.1.2 VoIP gateway provides service for all the dial numbers starting with '1310555'

The following Figure 3.4 shows how end-to-end call, between Dial Peer 1 and Dial Peer 4, is completed.

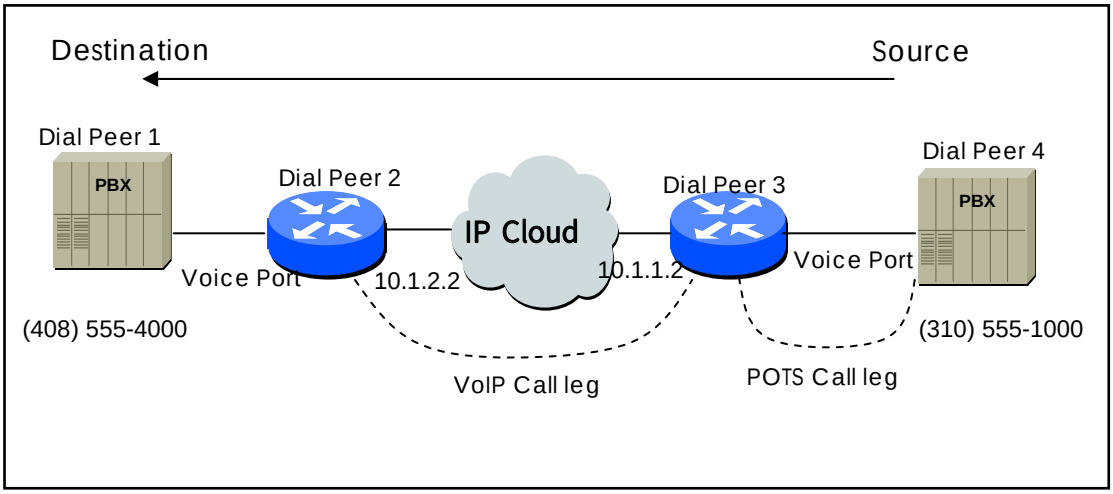


Figure 3.4 Outgoing Call from a View Point of POTS Dial Peer 2

Please use the following commands to complete end-to-end call, between Dial Peer 1 and Dial Peer 4 in the configuration of Figure 3.4:

```
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 it is described above, call completion in the gateway can be accomplished basically by selection of inbound and outbound dial peers.

Selection of outbound dial peer is determined by pattern matching between dialed digits and destination patterns of POTS peer and VoIP peer.

Selection of inbound dial peer is determined by following rules different from POTS peer and VoIP peer.

The followings are the selection procedure of inbound POTS peer:

- Choose POTS peer specified with voice port and the call has been received.

- When more than one POTS peers are specified to the same voice port, the POTS peer created first is to be chosen.

The followings are the procedure of selecting inbound VoIP peer:

- Select the VoIP peer with answer-address matching with calling party number of the inbound call.
- When the above selection fails, select the VoIP peer with destination-pattern matching with calling party number of the inbound call.
- When the above selection fails, select the VoIP peer configured with the same IP address as the outgoing gateway

Selection of inbound dial peer should be done for proper call control at inbounding phase.

In other words, each parameter configured to POTS or VoIP peer is applied to the chosen dial peers. When the inbound POTS peer could not be found, the call will not be processed since the voice port has no POTS peer. In case of inbound VoIP peer, the call can be still processed even the final section of inbound VoIP fails

3.3.2.2. Configuring POTS Peer

Configuration of POTS peer can be accomplished in the following procedure:

- Determine dial peer tag value
- Determine destination pattern
- Determine the port

In most of cases, the values other than these are set to a default value.

Steps	Commands	Description
1	model name# **configure terminal**	Get into configuration mode.
2	model name(config)# **dial-peer voice** *tag* **pots**	Move to POTS configuration mode in dial-peer. *'tag'* in this system is the only identifier and get the value of 0~65535. 'pots' represents configuration for the communication service of FXS, FXO port
3	model name(config-dial-peer)# **destination -pattern** *string* **[T]**	Enter a relevant telephone number of dial peer. 'string' means the telephone number 0~9, (#), (*) value and wildcard (.) is possible, Period '.' represents wildcard. 'T' can be entered, as an option, after a telephone number. When it is entered, the system collects the dial digits until end-of-dialing key (default#) or interdigit timer is finished.

4	model name(config-dial-peer)# **port** *location*	Map the pots to the port representing location Indicate location as port_number
5	model name(config-dial-peer)# **prefix** *string*	(optional) When the pots is selected as an incoming side, string automatically dial-out The range of string value is 0~9, (#),(*) and (,) is possible. With presence of (,), dial-out stops for 1 second at the digit
6	model name(config-dial-peer)# **exit**	dial peer configuration mode is finished

3.3.2.3. Configuring VOIP Peer

Configuration of VoIP peer can be achieved in the following procedure:

- Determining dial peer tag value
- Determining destination pattern
- Determining session target

In most of cases, the values other than these are set to a default value.

Step	Commands	Description
1	model name# **configure**	Get in to configuration mode.
2	model name(config)# **dial-peer voice** *tag* **voip**	Move to POTS configuration mode in dial-peer. *'tag'* in this system is the only identifier and get the value of 0~65535. 'pots' represents configuration for the communication service of FXS, FXO port
3	model name(config-dial-peer)# **destination -pattern** *string* **[T]**	Enter a relevant telephone number of dial peer. 'string' means the telephone number 0~9, (#), (*) value and wildcard (.) is possible, Period '.' represents wildcard. 'T' can be entered, as an option, after a telephone number. When it is entered, the system collects the dial digits until end-of-dialing key (default#) or interdigit timer is finished.
4	model name(config-dial-peer)# **session target** *destination-ip-address*	Enter ip address of the voip peer. Enter *destination-ip-address* to dotted decimal ip address (example, 123.321.1.2) If *destination-ip-address* 'ras', ip address of the voip peer can be known through the gatekeeper. If *destination-ip-address* 'sip-server, ip address of the voip peer can be known through the sip-server.
5	model name(config-dial-peer)# **dtmf-relay**	(optional)

	[h245-alphanumeric]	Determine DTMF transmission method to the voip-peer. The default value is **h245-alphanumeric**

3.3.2.4. Configuring Codec and VAD from Dial Peer

The Codec change the analog signal to digital bit stream and reversly, change the digital bit stream to analog signal. VAD (Voice Activity Detection) is processed when making digital bit stream from analog signal. Using VAD, the bit stream is not generated during silent duration and it save the bandwidth.

3.3.2.4.1. Configuring CODEC from VoIP Dial Peer

If you need to set a coder rate for the selected VoIP peer, please use the following commands in the beginning of global configuration mode.

Step	Command	Usage
1	**dial-peer voice** *tag* **voip**	Move to dial-peer configuration mode to configure VoIP peer
2	**codec [**g711alaw / g711ulaw /g729 / g7231r63 /g7231r53**]**	Select a codec with consideration of coder rate to be used for voice

The default value of codec command is g7231r63 and the default value is the most appropriate value.

However, when you face a circumstance that you are to connect to the network with high bandwidth and consider voice quality as top priority, please select g711alaw or g711ulaw in codec commands.

For instance, if you intend to use G.711a-law Rate of codec for VoIP Dial Peer 108, please configure the settings as to follow:

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

In addition, there is another way to create codec class and specify it to VoIP peer. The ways

describes above can configure only one codec, but on the other hand, codec class has an advantage of being flexible as it has many codec in a list with priority order.

The following is the procedure to create codec class first:

Step	Commands	Description
1	model name# **configure**	Get into configuration mode
2	model name(config)# **voice class codec** *tag*	Move to codec class configuration mode. 'tag' is the only codec class identifier.
3	model name(config-class)# **codec preference** *value codec-type*	Get into configuration mode
4	model name(config-class)# **codec preference** *value codec-type*	Get into configuration mode
5	model name(config-class)# **exit**	Finish codec class configuration mode (When the configuration is finished, the configuration has the effect.

The following are the procedure for specifying the codec class, which has been created through the procedure above, to a specific VoIP peer.

Step	Commands	Usage
1	**dial-peer voice** *tag* **voip**	Move to dial-peer configuration mode to configure VoIP peer
2	**voice-class codec** *codec-class-tag*	Select a codec with consideration of coder rate to be used for voice

The followings show an example of creating codec class 99 and specifying it to VoIP peer 108:

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

3.3.2.4.2. Configuring VAD from VoIP Dial Peer

Use the following commands in the beginning of global configuration mode to disable transmission of silence packet to VoIP peer.

Step	Commands	Usage
1	**dial-peer voice** *number* **voip**	Move to dial-peer configuration mode to configure VoIP peer
2	**vad**	Disable transmission of silence packer. In other

		words, enable VAD

The default is set to VAD. The default value is the most appropriate value. However, when you face a circumstance that you are to connect to the network with high bandwidth and consider voice quality as you top priority, disable vad. This setting provides a better voice quality; it requires more bandwidth than voice session. For an example, if you want enable VAD for VoIP Dial Peer 108, please configure the followings:

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

3.3.3. One-Stage Dialing and Two-Stage Dialing

Dialing can take many stages basically in VoIP network with many configurations, because of interoperation with a general telephone network or PABX in the office.

One way to reduce many stages of dialing is to give out the incoming telephone number together with the next stage of number information to the information of called party number when to configure an incoming call.

The below Figure 3.5 assumes that a call user connected to the voice port of Gateway A wants to make a call to the other call users connected to VoIP Gateway B and PABX with the extension number of 100.

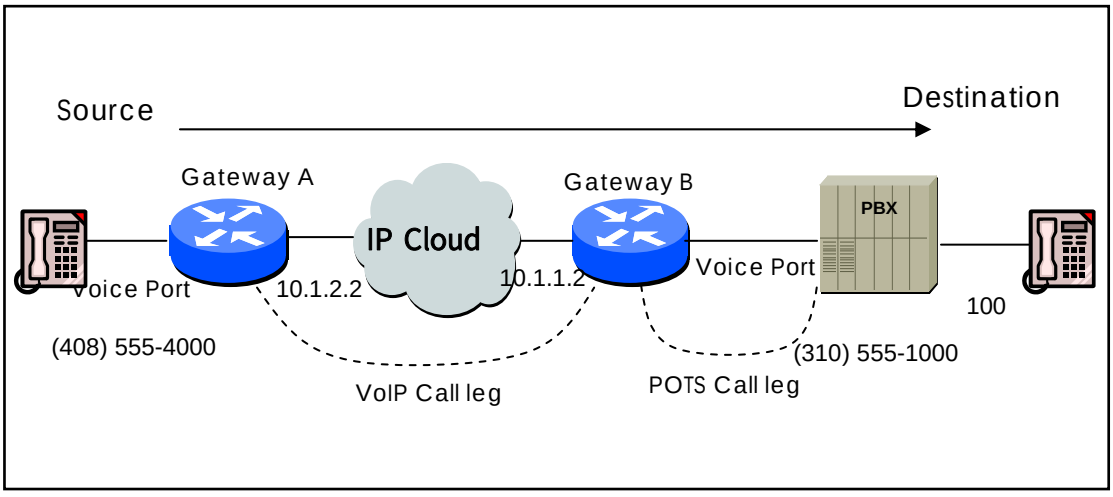


Figure 3.5 Two-Stage Dialing

VoIP peer setting of Gateway A is assumed as to follow:

```
dial-peer voice    555    voip
   destination-pattern    310555….
```

In the configuration above, as soon as, the call user Gateway A enters 3105551000, outbound VoIP peer 555 is determined and connected to Gateway B.

At this time, it is assumed that the settings of POTS peer of Gateway B are assumed as to follow:

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

In this case, the user at the outgoing side listens to the dial tone sent from PABX, and then enters the extension number of 100.

To make the two-stage dialing to one-stage, VoIP peer of Gateway A needs to be configured as to follow:

```
dial-peer voice    555    voip
   destination-pattern    310555……..
```

In these settings, the call can be connected only when the call user at Gateway A enters all of 3105551000100 and Gateway B delivers the digits (except fixed digit information) 100, except called party number information and wild card of destination pattern, to its voice port, when outbound POTS peer chooses 1000.

If a length of number is not fixed, you can use ‘T’ from the destination pattern

VoIP peer settings of Gateway A can described as to follow:

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

In this case, after the user of Gateway A enters 31055510001234567, terminating-digit (#) or inter-digit timeout, the call is connected to Gateway B and Gateway B delivers 1234567 to the selected voice port.

3.3.4. Hunt Group

3.3.4.1. Basic Concept and Configuration

Selection of outbound POTS or VoIP dial peer, which is sent out of the gateway, can be accomplished by comparing called party number of the inbound call to destination pattern of

dial peer.

At this time, more than one dial peer, which is coincided with called party number, is called hunt group and attempts to make calls according to the priority order based on the consistent policy.

In other words for the case of VoIP peer, When the call attempt fails due to network connection fail, gatekeeper reject, another call attempt can be made by other dial peers in the hunt group.

In case of POTS peer, when the attempt is failed due to the port being busy of the voice port, another call attempt can be made to another dial peer in the hunt group.

The elements for determining the priority order, of the call attempts in the hunt group, are longest match, explicit preference, sequential, random.

First, longest match is the priority order according to the maximum digit matching with the destination number of dial peer and outgoing number.

For example, when the outgoing number is 5683848, destination number of dial peer 1 is 568T, the destination number of dial peer 2 is 568…, and the destination number of dial peer 3 is 56838.. and the destination number of dial peer 4 is 5683848, the priority order basing on longest match is dial peer 4 --> dial peer 3 --> dial peer 2 --> dial peer 1.

Explicit preference determines the priority order basing on a preference of the dial peer.

For example, when the preference of dial peer 1 is 3, dial peer 2 is 2, dial peer 3 is 1 and dial peer 4 is 0, the priority order basing on the explicit preference is dial peer 4 --> dial peer 3 --> dial peer 2 --> dial peer 1.

The random priority order determines the dial peer in the hunt group randomly. When the random priority order is sequential, the order determines the lowest frequency first, which is selected previously. Such algorithm of priority order is processed in combination, for instance, processing the default setting of dial-peer 0 can be determined by longest matching in the first stage, explicit preference in the second and random in the third.

The first setting related to the hunt group is selection of hunt algorithm.

Step	Command	Usage
1	model name# **configure terminal**	Get into global configuration mode
2	model name(config)# **dial-peer hunt** *[0-7]*	Apply the priority order algorithm from 0 to 7 as to follow: 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

As next step, you may configure the explicit priority order by preference or huntstop which can be configured to the dial peer. If huntstop is configured to a specific dial peer and the outbound call of that dial peer fails, then terminate the call without hunting to other dial peer.

Step	Commands	Description
1	model name# **configure terminal**	Get into configuration
2	model name(config)# **dial-peer voice** *tag* **{ pots | voip }**	Move to dial-peer configuration tag is the only identifier of dial-peer and it take the rage of value 0~65535
3	model name(config-dial-peer)# **preference** *number*	The range of the value is 0-9 and less the value, higher the priority is.
4	model name(config-dial-peer)# **huntstop**	Set up huntstop to the dial peer

3.3.4.2. Rerouting to PSTN

As it is explained previously, hunt group enables PSTN rerouting through FXO voice port when the connection to VoIP network fails. The following Figure 3.6 exhibits PSTN rerouting.

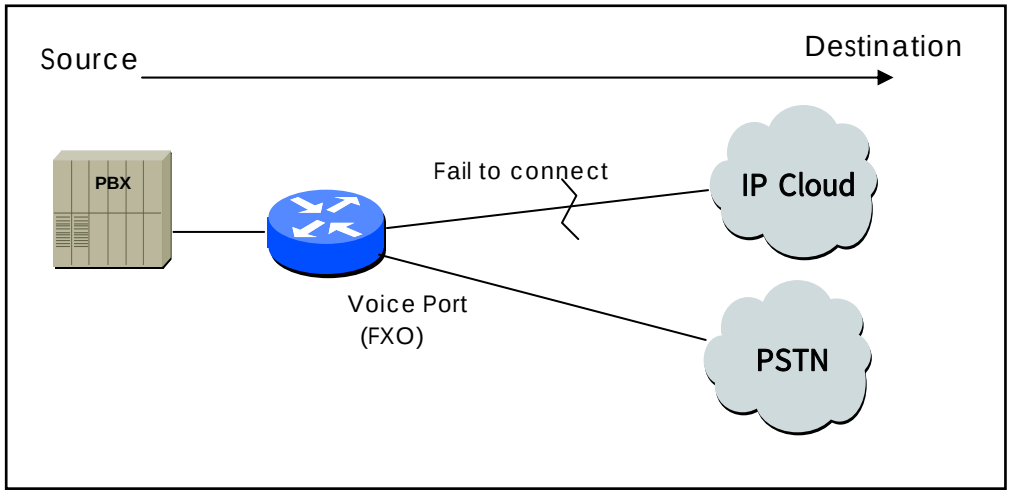


Figure 3.6 PSTN Rerouting

To make this PSTN routing to happen, configure the dial peer as to follow:

```
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 this example above, VoIP peer 101 and POTS peer 102 exist in the same hunt group. Since the preference value of VoIP peer is lower and this is selected first for the call attempt and when it fails, another call attempt is to be made to POTS peer 102.

3.3.4.3. Call bar

As it has been explained previously, you can block the outbound and inbound call process with a specific pattern by using **huntstop** and **shutdown**.

If you want to place restrict a call for the outbound peer, set the pattern to destination pattern and use the commands to configure shutdown and hunt stop.

All the outbound calls, in the below example, are set to choose VoIP peer 100, but the call can not be processed for the called party number starting with 526 or 5441234.

```
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
```

If you want to place restrict a call for the inbound VoIP peer, set the pattern, which you want to restrict, to the destination pattern and use the commands to configure shutdown and hunt group. If it is necessary, the dial peer, that you intend to restrict, is selected as a top priority.

From the example above, if the calling party number of the inbound call, starting with 526, or is 5441234, the call is not processed.

If you want block the inbound VoIP call starting with the number 538 and allow the outbound call, use **answer-address** as to follow:

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

3.3.5. Number Forwarding and Prefix

In the previous section of 3.3.3, forwarding of the number for POTS peer was mentioned.

The forwarding of the number for the outbound POTS peer forwards the digits except the fixed digit of destination-patter of the outbound POTS peer.

For example, if destination-pattern is 444…, the fixed digit is 444. If the call party number of the inbound call is 444123456, the digits of 123456 is forwarded to the voice port to the outbound POTS peer. (In case of analog voice port, DTMF tones of forwarding digits are generated and in case of digital E1 voice port, forwarding digits are set to the called party number field to other side)

If this outbound POTS peer is set to **prefix** 99,, the digit forwarding of 99 is forwarded first, and then delayed 2 seconds by ‘,,’ , after then forward 123456.

The number forwarding process, which has been explained above, is about the default settings. For more detailed operation of the number forwarding, configure **forward-digit** to POTS peer configuration.

The dial peer configured with **forward-digit** does not check the fixed digit of destination pattern and forwards the number depending on the value, which is set by **forward-digit**. The settings of forward digit can be operated by **forward-digit from** and **forward-digit last**. **forward-digit from** forwards all the digits to the number, which has been set, of the called part number of the inbound call.

forward-digit last forwards the last portion of the numbers as many as it is set from the called party number of the inbound call.

For example, if the called party number of the inbound call is 444123456 and **forward-digit from 4**, 123456 is delivered and if it is **forward-digit last 4**, 3456 is delivered.

3.3.6. Configuring Number Expansion

In most of business environment, the telephone number has been structured to allow dialing only a portion (Number Extension) of the entire E.164. VoIP (voice over IP) can be configured to recognize an extension number and expand the extension number to full E.164 dialed number by using both **destination-pattern** and **num-exp**. Prior to configuring these 2 numbers, it would be very helpful to draw a configuration diagram of each telephone extension number. This can be easily done by preparing Number Expansion Table.

3.3.6.1. Preparing Number Expansion Table

The following Figure 3.7 shows an example of integrating PSTN network to existing IP network for a small-sized company using VoIP. The figure below shows the destination pattern (or Expanded Telephone Number), being used in Gateway1 assumes (408) 115-xxxx, (408) 116-xxxx, (408) 117-xxxx, xxxx is the command to configure to each dial peer using extension. Also the figure below shows the destination pattern (or Expanded Telephone Number) configured to (729) 555-xxxx.

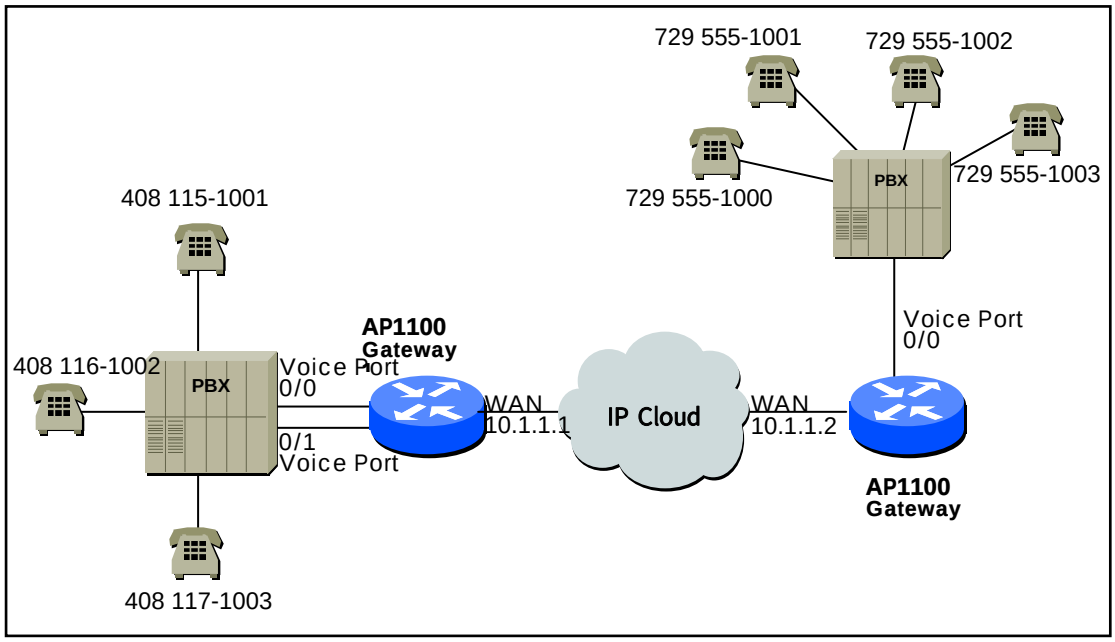


Figure 3.7 An Example of VoIP Network

The number extension table for this scenario is listed in the table as to follow:

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 to configure Gateway1 and 2.

3.3.6.2. Configuring Number Expansion

Use the following commands in global configuration mode to expand an extension number for a specific destination patter.

Step	Command	Usage
1	**num-exp** *extension-number extension-string*	Configure number expansion

To check whether a telephone number is mapped correctly, you can check the information for the number expansion by using **show num-exp** and **show dialplan number**.

3.3.7. Configuring Number Translation

3.3.7.1. Creating Translation Rule

You can apply Number Translation to called party number and calling party number of the inbound call and outbound call. When it applies to the inbound call, the incoming called (or calling) party number is translated depending on a translation rule. Then it is used for selection of outbound dial peer.

When it is applies to the outbound call, the outgoing called party (or calling) is translated depending on a translation rule and process the call.

Number translation is needed, when to change a private number to a public number (or a public number to a private number) or it can be used for number translation and number interworking and more various changes are possible than number expansion.

In order to translate a number, first create Translation Rule Set by using the command **translation-rule** in global configuration mode.

This translation rule set can define more than one rule and this can configured by using **rule** command in translation-rule configuration mode.

Step	Commands	Description
1	model name# **configure terminal**	Get into configuration mode.
2	model name(config)# **translation-rule** *tag*	Move to translation–rule configuration mode *tag* is the only identifier to specify translation rule set
3	Router(translation-rule)# **rule** *rule-tag input-matched-pattern substituted-pattern*	*rule-tag* is an identified to specify rule in the rule set. The rage of value is 0 – 65535. *input-matched-pattern* is the digits to be entered for pattern matching The texts for entry are 0-9#*[].T *substituted-pattern* is the pattern to be translated when pattern matching succeeds..Valid text entry is 0-9#*%.T.

If more than one rule is matched with the called (or calling) party number in a rule set, the rule, which has the most matches with *input-matched-pattern*, is selected. Range expression (for instance [1-9]) can be used by *input-matched-pattern*

Also wildcard '.' can be used for application of the digit number of called (calling) party number.

If *input-matched-pattern* is configured only with '.' or 'T', number translation applies to all of called (or calling)-party-number.

substituted-pattern translates the fixed digit (the digit excluding wildcard) of *input-matched-pattern* to a string of substituted-pattern. The substituted-pattern can be divided by 2 formats.

In first case, *substituted-pattern* is configured only with IA5 texts (0-9#), the fixed digit part of input-matched-pattern is translated into the string part of *substituted-pattern*, the rest of digits, except the fixed digits of the called (or calling) party number, are to added next to the end.

In next case, *substituted-pattern* uses '%' to enable configuration of the number by substituting each digit of the called (or calling) party number to %xx variable.

substituted-pattern is only configured with '.'or 'T', the called (or calling) -party-number is to be configured with the digits except the fixed patter of *input-matched-pattern*.

For an example, if the rules are set as to follow and 00181463701234 is entered to a translation rule set, then it is translated to 81463701234. If 0313961234 is entered, then it is translated to 82313961234. If 5261234 is entered, it is translated 8225261234.

```
translation-rule 100
  rule 1  001……  .
  rule 2  0……..  82
  rule 3  [1-9]…… 822%01%02%03%04%05%06%07%08
```

The created translation rule can be verified by **show translation-rule**.

For instance, when show translation-rule 100 is entered, the rules in translation rule set 100 are displayed.

If you want to see the result of the applied translation rule set, just enter the number that you intend to test. If you want to see the result of the number 100 applied to translation set 100, just enter show translation-rule 100 1234. The result is 1234.

3.3.7.2. Applying Translation Rule to Inbound POTS Call

To apply the translation rule set to all the incoming calls to a specific voice port, configure as to follow:

Step	Commands	Purpose of Usage
1	model name# **configure terminal**	Get into global configuration mode
2	model name(config)# **voice-port** *location*	Enter the specified voice port configuration mode Indicate location by port -number
3	Router(voice-p ort)# **translate-incoming {called-number |**	**called-number :** Apply translation rule to the inbound called party number

	calling-number} *tag*	**calling-number** : Apply translation rule to the inbound calling party number *tag* is to reference the rule set. The range of value is 0-65535.

If the translations is applied to the called party number and the number information is entered in the order of the voice port, check whether number translation takes a place, wherever the information is entered.

3.3.7.3. Applying Translation Rule to Inbound VOIP Call

You may need to configure the following setting in order to apply the translation rule to all the incoming calls from a network.

Step	Commands	Purpose of Usage
1	model name# **configure terminal**	Get into global configuration mode
2	model name(config)# **voice service voip**	Enter voice service voip configuration mode
3	Router(service-voip)# **translate-voip-incoming {called-number | calling-number}** *tag*	**called-number :** Apply translation rule to the inbound called party number **calling-number** : Apply translation rule to the inbound calling party number *tag* is to reference the rule set. The range of value is 0-65535.

3.3.7.4. Applying Translation Rule to Outbound Call

You may need to configure the following settings in order to apply the translation rule to the outbound call of a specific dial peer (POTS peer or VoIP peer).

Steps	Commands	Purpose of Usage
1	model name# **configure terminal**	Get into global configuration mode.
2	model name(config)# **dial-peer voice** *tag* **{ pots | voip }**	*tag* is the only identifier of dial-peer in this system and its range of value is 0-6535.
3	Router(dial-peer-config)# **translate-outgoing {called-number | calling-number}** *tag*	**called-number :** Apply translation rule to the outbound called party number **calling-number** : Apply translation rule to the outbound calling party number *tag* is to reference the rule set. The range of value is 0-65535.

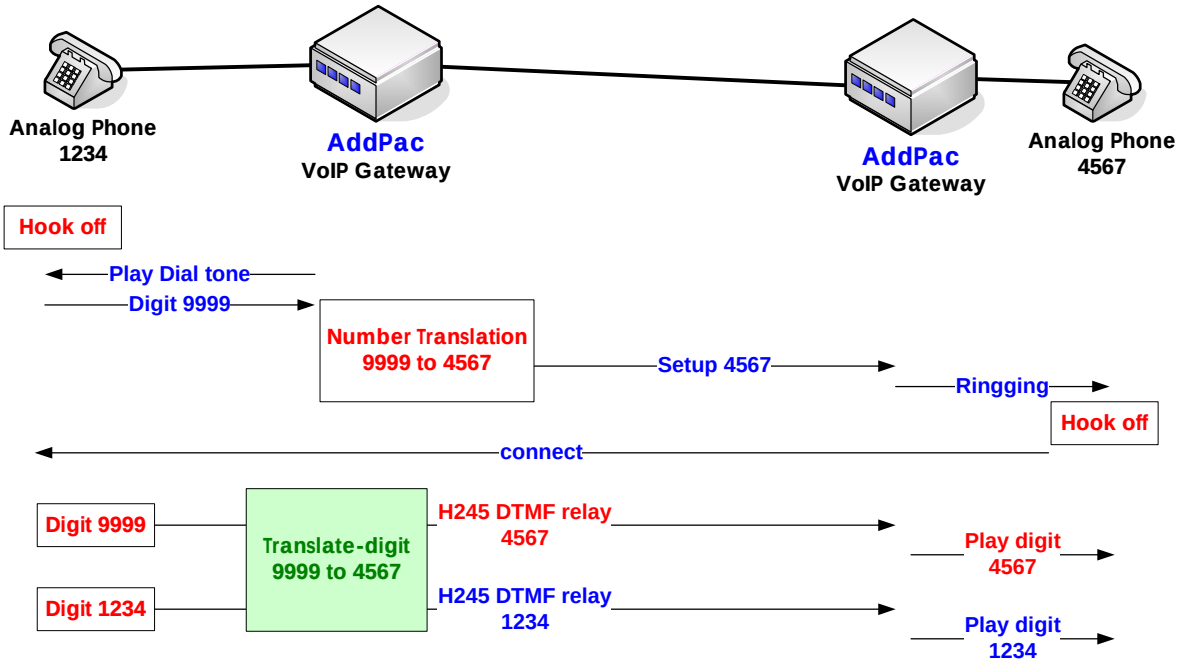
3.3.7.5. Applying Translation Rule to Connect Call

This feature is to support the translation rule for the relayed digit (DTMF relay) in the state of which the call has been configured.

When to configure the call, this feature is distinguished from number translation which translates calling party number or called party number.

The following figure describes the difference between **number translation** (translation-rule) and **translation-digit**. When the called-party number (destination number) applies to the outgoing side of the gateway 999 to be translated to 4567, by the rule for number translation (translation-rule), this rule is used to translate the called-party number of e.164 number, which is also used for the setup.

In other word, this is the translation for a telephone number. If the same rule applies to translation-digit, it is used for translating the digit to be delivered by DTMF relay. Therefore, translation-digit has nothing to do with e.164 number which is used for the setup.



Step	Steps	Commands
1	model name# **configure terminal**	Get into global configuration mode.
2	model name(config)# **dial-peer voice** *tag* **voip }**	Enter dial-peer configuration mode *tag* is the only identifier of dial-peer in this system and its range of value is 0 ~ 65535.
3	Router(dial-peer-config)#	*tag* is to reference the rule set. The range

	translate-outgoing digit-in- call *tag*	of value is 0-65535.

3.3.8. Configuring and Applying call-diversion

To configure the related settings to the call relay, when all the incoming call share busy, as an adjacent device

3.3.8.1. call-diversion

Transfer from global configuration mode to its setting mode . To delete call diversion, add **no** command.

call-diversion *tag*

no call-diversion *tag*

3.3.8.1.1. Syntax

Keyword / Argument	Description
tag	This is the identifier to specify call diversion.

3.3.8.1.2. Command Default

No default value.

3.3.8.1.3. Command Modes

Global configuration mode

3.3.8.1.4. User Guideline

No default value.

3.3.8.1.5. Example

The following is the example for call diversion 100.

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

3.3.8.2. max-forward-hop

To set the maximum number of attempts for call forwarding requested when the correspond voip-peer is determined for outbound

max-forward-hop number

no max-forward-hop

3.3.8.2.1. Syntax

Keyword / Argument	Description
number	the maximum number of attempts for call forwarding

3.3.8.2.2. Default Value

The default value is 4.

3.3.8.2.3. Command Modes

Dial-Peer configuration mode.

3.3.8.2.4. Usage Guideline

To specify the maximum number of call-forwarding requested at call diversion configuration mode.

3.3.8.2.5. Examples

The following is the example to set up max-forward-hop 10.

```
max-forward-hop 10
```

3.3.9. Configuring and Applying Call Transfer

Call transfer, which is used by an end user as VoIP interoperates with IP-PBX, is the feature provided from IP-PBX. However, the call transfer in here is the configuration for a telephone connected to FXS port of the gateway directly.

The default setting of the AddPac gateway is to disable the call transfer, but using the following commands, you can enable the call transfer:

dial-peer call-transfer h

The above example for the call transfer can be enabled by hook-flash and if you want to disable the enabled call transfer, enter **no dial-peer call-transfer or dial-peer call-transfer n.**

To describe call transfer, you need the description for each party to be configured to call transfer. Basically, the scenario of call transfer is the scenario of call hold. In Figure 3.8 Call AB exists between 5.3.9, first the call, Call AB exists between User A and User B, User B creates a new call, Call BC, by hook-flash, at last, User B exits by hook on, Call AC exist between User B and C.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **dial-peer call-transfer** **<h>** **<n>**	The is the operation mode for call-transfer h : operated by hookflash n : not in operation (call-transfer does not work)

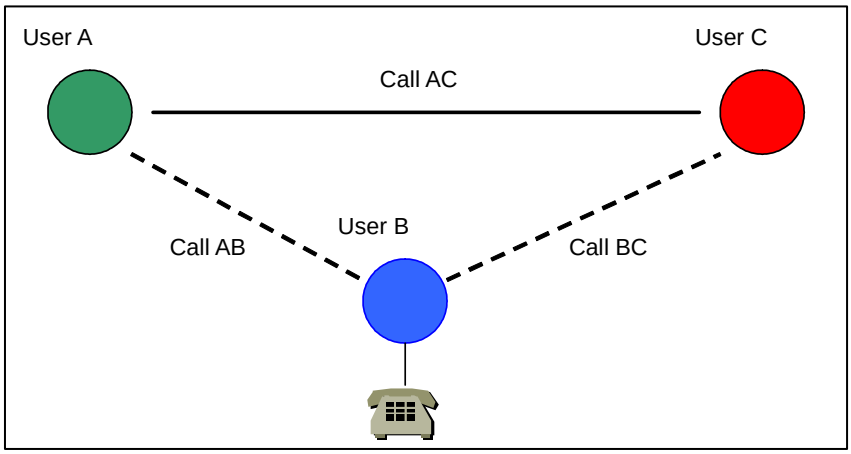


Figure 3.8 Call transfer Scenario

User B should be connected to FXS port and all the configuration is possible except the configuration that User B and C both are remote users.

Call transfer has 2 different types; one is confirmed and non-confirmed is the other.

Call transfer takes a place by hook on, after User B makes a call to User C in the confirmed

type.

In the non-confirm type, after User B makes a call to User C, while the phone is ringing, the call transfer takes a place. The AddPac gateway supports both types.

3.3.10. Configuring and Applying Call Pickup

Call pickup, which is used by an end user as VoIP interoperates with PBX, is the feature provided from PBX. However, the call pickup in here is the configuration for a telephone connected to FXS port of the gateway directly.

The default setting of the AddPac gateway is set to disable the call pickup, but using the following commands, you can enable the call pickup:

dial-peer call-pickup #11

From the example above, call pickup code is #11 and when a call is processed to be ringing, another telephone connected to another FXS hooks off, then you may press the call pickup code to pick up the call of the port while it is ringing.

If you want to disable this feature, you can set up **no dial-peer call-pickup**.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **dial-peer call-pickup** <0-9|#|*> + IA5 digits	This is the operation mode call pick-up.

3.3.11. Configuring and Applying Inbound-pots-peer

When VoIP interface seems well and its communication with gatekeeper is normal, all the calls are delivered through this network.

However, when communication of VoIP interface with gatekeeper can be placed in abnormal status due to a network failure or a certain exception and the call can not be delivered properly, the feature of **inbound-pots-peer** related to PSTN-Backup, which transfers to PSTN automatically, is provided by the gateway.

The default setting of the AddPac gateway is to disable inbound-pots-peer, but using the following commands, you can enable the call :

inbound-pots-peer 0

no inbound-pots-peer

This features is used to minimize the complexity involved with PSTN backup configuration, by using translation rule, when port to port needs to be mapped 1:1 in some special cases.

Step	Steps	Commands
1	model name# **configure**	Enter configuration mode
2	model name(config)# **dial-peer voice** *tag* **pots**	Move to dial-peer configuration mode. *tag* is the only identifier of dial peer in this system and it takes the range of value, 0~65535.
3	model name(config-dialpeer-pots-<tag>)# **inbound-pots-peer** <tag>	Apply inbound-pots-peer *tag* is the only identifier in this system and it takes the range of value, 0~65535.

3.3.12. Configuring and Applying PSTN Backup

When VoIP interface seems well and its communication with gatekeeper is normal, all the calls are delivered through this network.

However, when communication of VoIP interface with gatekeeper can be placed in abnormal status due to a network failure or a certain exception and the call can not be delivered properly, the features of busyout monitor, busyout action related to PSTN-Backup, are provided by the gateway.

The default setting of the AddPac gateway is to disable inbound-pots-peer, but using the following commands, you can enable the call :

3.3.12.1. busyout monitor

Busyout state means that VoIP gateway lost its function as the gateway depending on a network status. Busyout determines by monitoring a status of binding or voip-interface .

When binding is disconnected with the gatekeeper and you want to busyout, set the gatekeeper to busyout monitor, use no to exclude busyout monitor.

If you want to monitor both voip-interface and gatekeeper at the same time, set both busyout monitor gatekeeper and busyout monitor voip-interface.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **voice service voip**	Move to voice service voip mode
3	model name(config-vservice-voip)# **busyout monitor {callagent | gatekeeper | sip-server | voip-interface}**	Callagent : apply busyout monitoring to mgc Gatekeeper : apply busyout monitoring to gatekeeper Sip-server : apply busyout monitoring to sip-server Voip-interface : apply busyout monitoring to voip- interface

3.3.12.2. busyout action

When the gateway is in busyout state, a specific action (busy-tone, port-down) can be configured.

If busyout action is set to busy-tone and operates under this condition, the call is not attempted to the port of the gateway at PBX, because the extension or central office line, which is connected to the port, is recognized as busy. At this time, if hunt feature is configured, when the port of the gateway is busy, PSTN back-up can be configured in a way to attempt the call, which is

connected to the central office line and PSTN, where the port of the gateway is busy.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **voice-port slot-id/port-id**	Move to voice-port configuration mode
3	model name(config-voice-ports-0/0)# **busyout action {none | tone | port-down}**	None : do nothing Tone : generate busy tone Port-down : port down (mute)

3.4. Configuring Voice Port

3.4.1. Configuring the Gateway Voice Port

Generally, voice port command is to configure the characteristics of a specific voice port signaling type. In most of telephone network environments, a proper value is fixed with the default voice port command for FXO and FXS ports to transmit voice data to the existing IP network.

3.4.2. Voice Port Configuration Items and Order

3.4.2.1. Configuring FXS and FXO Port

In most of telephone network environments, the default value of voice ports are properly configured to FXO and FXS. If you need to change the default values of these voice ports, process the following procedure. The first 2 items are mandatory and the third one is optional.

1) Specify a voice port and enter the voice port configuration mode.
2) Set up the necessary parameters of the voice port
3) The followings are the commands to set up the optional voice port parameters:
   - PLAR(Private Line Auto Ringdown) connection mode
   - Description
   - Ring Number
   - Input Gain
   - Output Gain

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode.
2	**voice-port** *location*	Move to the specified voice port configuration mode Indicate location as slot-number/port-number
3	**ring number** *number*	(Use only to FXO port) Set the maximum number of ringing prior to responding to a call
4	**connection plar** *string*	(Optional Command) Set up PLAR (Private Line Auto Ringdown) connection for the port The string value is Destination Phone Number
5	**description** *string*	(Optional Command)

		Add tests for description of the voice port connection.
6	**input gain** *value*	(Optional Command) Set up an amount of gain by decibel units for the incoming signal to the voice port. The rage of the possible value is –31~31.
7	**output gain** *value*	(Optional Command) Set up an amount of gain by decibel units for the outgoing signal to the voice port. The range of the possible is -31~31.

3.4.2.2. Configuring E&M Port

Distinguished from FXO or FXS voice port, the parameters of default E&M voice port are not enough to transmit voice data though a user's IP network.

The setting value of E&M voice must be adjusted to the characteristic of a specific PBX, which is to be connected to this port. For type 5 of PBX installation, the main frame of the gateway should be grounded.

When to use E&M port, a proper setting value of E&M port must be found the existing PBX equipment.

Take the following procedure for configuring E&M port:

1) Specify a voice port and Enter the voice port configuration mode.
2) Find the proper values of the parameters and set the mandatory parameter values as to follow:
   - Signal Type
   - Operation
   - Type (For AP-E&M module, jumper setting is used for the cabling scheme.

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode
2	**voice-port** *location*	Enter the specified voice port configuration mode Indicate location as slot-number/port-number
3	**signal {wink-start | immediate | delay-dial }**	Select a proper signal type for the interface.
4	Select a proper cabling scheme for the VoIP port. Select whether to use 2-wire or 4-wire. For AP-E&M module, jumper setting is used for the cabling scheme. At the factory default option, this part is set 2-wire. For more details, refer to E&M Module Jumper Settings.	

5	Select a proper type of E&M for the port. Select which of Type 1, 2, 3, 5 to be used. Jumper setting is used for this cabling option of AP-E&M. At the factory default mode, it is set to Type-5. For more details, refer to E&M Module Jumper Settings. The followings are each signal configuration for each E&M Type. Type 1 - E : output, relay to ground - M : input, referenced to ground Type 2 - E : output, relay to SG(Signal Ground) - M : input, referenced to ground - SB(Signal Battery) : feed for M, connected to –48V - SG(Signal Ground) : return for E, basically isolated from ground Type 3 - E : output, relay to ground - M : input, referenced to ground - SB(Signal Battery) : connected to –48V - SG(Signal Ground) : connected to ground Type 5 - E : output, relay to ground - M : input, referenced to –48V	
6	**operation {2-wire| 4-wire}** **no operation**	*This command is for Informational description and does not have an actual effect. The actual operation can be effected depending on the jumper setting in the number 4. This command provides the information of a wiring type to be used without verifying jumper.*
7	**type {1|2|3|5}** **no type**	*This command is for Informational description and does not have an actual effect. The actual operation can be effected depending on the jumper setting in the number 5. This command provides the information of a wiring type to be used without verifying jumper.*

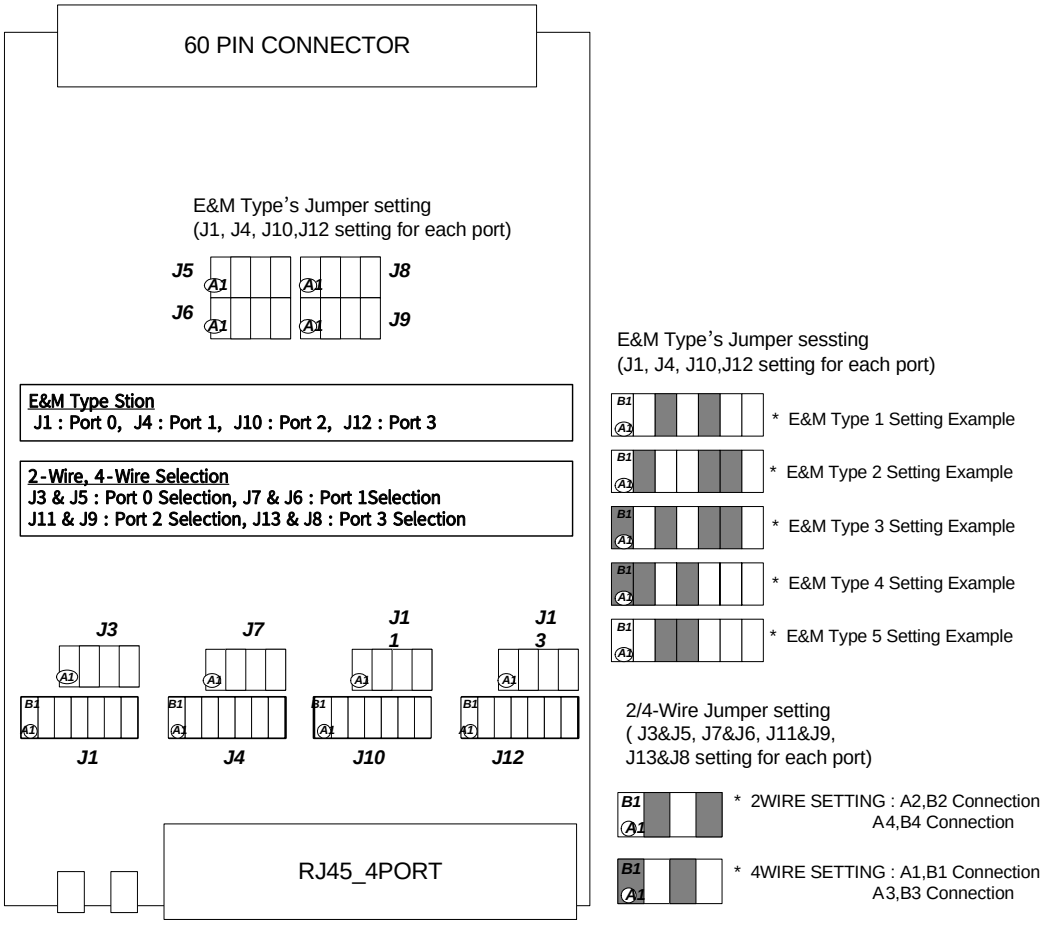


Figure 3.9 Setup for Jumper of E&M Card

3.4.2.3. Tuning E&M Voice Port

In most of cases, the default value of E&M voice port is enough, but depending on an environment of the existing telephone network, the voice parameters, including timing, input gain and output gain, need to be adjusted for E&M voice port. In this case, use the following commands to perform tuning:

1) Specify a voice port and enter voice port configuration mode.
2) Select a proper value of parameter and configure the optional voice port parameters as to follow:
   - Input Gain
   - Output Gain

- Timing other then Timeout

Use the following commands for tuning E&M.

Deactivate/ Activate the port by using **shutdown / no shutdown** command after changing the settings of the voice port by using voice-port command.

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode
2	**voice-port** *location*	Enter a specified voice port configuration mode Indicate location as slot-number/port-number
3	**input gain** *value*	Specify a gain value to be added to the input interface, by decibel units The range to be used is -18 ~ 9
4	**output gain** *value*	Specify a gain value to be added to the output interface by decibel units The range for use is -18 ~ 9
5	**timing delay-duration** *milliseconds*	Specify delay signal duration for delay dialing signaling The range for use is 100~5000 msec.
6	**timing delay-start** *milliseconds*	Specify a minimum delay time sending the outgoing signal after the incoming seizure is detected. The range for use is 20~2000 msec.
7	**timing wink-duration** *milliseconds*	Specify maximum wink signal duration. The value range for possible use is 50~3000 msec.
8	**timing wink-wait** *milliseconds*	Specify maximum win-wait duration for wink start signal. The value range for possible use is 100~5000 msec.
9	**timing dialout-delay** *milliseconds*	Specify dial-out delay for cut-through or sending the number to E&M The value range for possible use is 100~5000 msec.
10	**timing wait-wink** *milliseconds*	Specify a maximum wait value for wink signal. The value range for possible use is 100~5000 msec.

3.4.2.4. Configuring E1 Voice

All the settings, which can be applied to analog FXS port, can be also applied to all the channels in E1.

The followings are the commands to be applied to only E1 voice port

Step	Commands	Purpose
1.	**configure terminal**	Enter global configuration mode
2.	**voice-port** *location*	Enter the specified voice configuration mode Indicate location as slot-number/port-number

3.	**compand-type {a-law | u-law}**	Set up PCM compand type for PCM channel such as E1. - a-law : European type - u-law : North America type PCM compand type should be same as the other party's E1 The default is in a-law status.

3.4.2.5. Activating/Deactivating Voice Port

Use the following commands to activate the deactivated voice port in voice port configuration mode.

Step	Commands	Purpose of Usage
1	**no shutdown**	Activate the voice port

Use the following commands to deactivate the activated voice port.

Step	Commands	Purpose of Usage
1	**voice-port** *location*	Enter the specified voice configuration mode to activate/deactivate the voice port Indicate location by slot-number/port-number
2	**no shutdown**	Activate the voice port

3.5. Configuring E1 controller

This section describes the AddPac E1 voice module related settings. The product models of VoiceFinder Gateway Series which can integrate E1 modules are AP2520R, AP2520G. AP2110, AP2620, AP2640, AP2650.

E1 configuration related work process can be carried out in the following order mostly:

1) Verifying E1 module of the other party: ISDN / R2 / DTMF

2) Cabling E1 interface : RJ 45 connector

3) Configuring E1 controller : signaling type, channel group and others

4) Configuring Dial-peer : general pots peer configuration

5) Configuring voice port: command type or other selective setting details

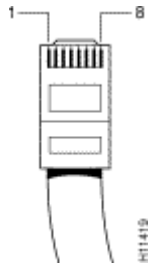
3.5.1. Connecting to PBX / PSTN

E1 voice module operates in the network mode as to be connected with E1 interface of PBX. Also it can operate in the subscriber mode as to be connected with E1 interface of PSTN

Prior to connecting with PBX, the following details need to be verified:

- Verify whether a type of E1 line board of PBX is ISDN PRI or R2. In most cases, the model of R2 board is different from ISDN PRI board.
- Specify the 30 channels supported by E1 board as the central office line group and assign a specific call number (for instance 9 or 8 and others) to it.
- Specify ISDN number entry method for ISDN PRI board to enbloc or overlap. The AddPac E1 module supports both enbloc and overlap based number entry.
- Check PCM command type. AddPac E1 module is set to a-law as default.
- Check E1 clock source. The most of PSTN is master and PBX is slave.

Cabling E1 interface of RJ-45 connector applied to AddPac E1 module is described in the followings:



Pinouts for T1/E1 Trunk and Digital Voice Port (RJ-45) Pin	signal
1	RX (tip)
2	RX (ring)
3	-
4	TX (tip)
5	TX (ring)
6	-
7	-
8	-

3.5.2. Common Configuration

Below is the configuration which is commonly applied, does not matter with the signaling type of ISDN PRI/ R2.

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode
2	**controller e1** *location*	Enter the specified controller configuration mode Indicate location as slot-number/port-number
3	**channel-group timeslots** *expr*	(**mandatory**) Configure the channels of E1 to be used. At default, the channel is not configured; therefore, it must be configured. If E1 module is placed in slot 0, use the following commands to configure all the channels (the following settings are used for most of cases). The signaling channel, the number 16 is ignored from the channel group configuration and does not need to be considered. model name(config)# controller e1 0/0 model name(config-controller-e1-0/0)# **channel-group timeslots 1-31** If channel 1, 2, 3, 8, 20, 21 are used as an exceptional case, use the following commands: model name(config-controller-e1-0/0)# **channel-group timeslots 1-3,8,20,21**

4	**signaling-type { dtmf | isdn | r2 }**	The first thing to configure is the signaling type. The configuration possible methods are ISDN PRI and R2, DTMF If the signaling type is set to ISDN PRI at default and change to the other type , then save the settings and reboot
5	**chan-number-order { ascending | descending }**	When the call is initialized through E1 of the module, the following commands are used to configure the order. Use the channel 1 for ascending and the channels 31 for descending. Configuring in opposite direction from PBX is recommended. The AddPac E1 module is set to descending at default.
6	**clock-source { master | slave }**	Specify the clock source of E1 to master or slave. Fax communication may not work if E1 clock source does not operate as master on one side and slave on the other. The command default is master.
7	**out-barred-group timeslots** *expr*	Configure the channels of E1 for not allowing the outbound At default, all the channels are configured in both directions. Verify direction of the channels by show voice port *slot/port* model name(config)# controller e1 0/0 model name(config-controller-e1-0/0)# **out-barred-group timeslots 20-31**

3.5.3. Configuring ISDN PRI

ISDN PRI related settings are listed as to follow:

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode
2	**controller e1** *location*	Enter the specified controller configuration mode Indicate location as slot-number/port-number
3	**isdn protocol-emulate {network | user}**	ISDN PRI protocol is operated on the interface as a center: one side is for network and user is the other. In case of PBX, the PRI directing PBX is for user and PSTN directing PRI is network. The default is set to network to react with PBX. However, it needs to be set to user for connecting to PSTN.

4	**isdn {n303 | t303 | t310}** *value*	This command is used for timer and counter from ISDN PRI protocol ● N303 : SETUP retry counter ● T303 : The time out value until reply message is received after transmitting SETUP message ● T310 : The time out value until the next message is received after receiving CALL PROCEEDING message
5	**isdn virtual-connect**	Under ISDN PRI enbloc case, this option enables routing by user's DTMF entry before end-to-end connection.

3.5.4. Configuring R2

R2 related setting is listed as to follow:

Step	Commands	Purpose of Usage
1	**configure terminal**	Enter global configuration mode
2	**controller e1** *location*	Enter the specified controller configuration mode. Indicate location as slot-number/port-number
3	**r2 get-calling-number**	Configure this command to receive calling party number for R2

3.6. Configuring FAX Applications

3.6.1. H.323 or SIP-Based T.38 FAX Relay

T.38 Fax Relay feature used from H.323 protocol is the standard based Fax Relay Protocol which is supported by VoiceFinder Gateway and most of other gateways or routers.

For VoIP gateway to use fax relay, the configuration for voice routers and gateways need to be changed to T.38 protocol.

The following shows IP H.323 network, which is composed of VoiceFinder Gateway and the companies with gateways and gatekeepers in different locations, using T.38 fax relay feature. By using T.38, all the gateways and gatekeepers in this network can send fax to other offices in remote locations

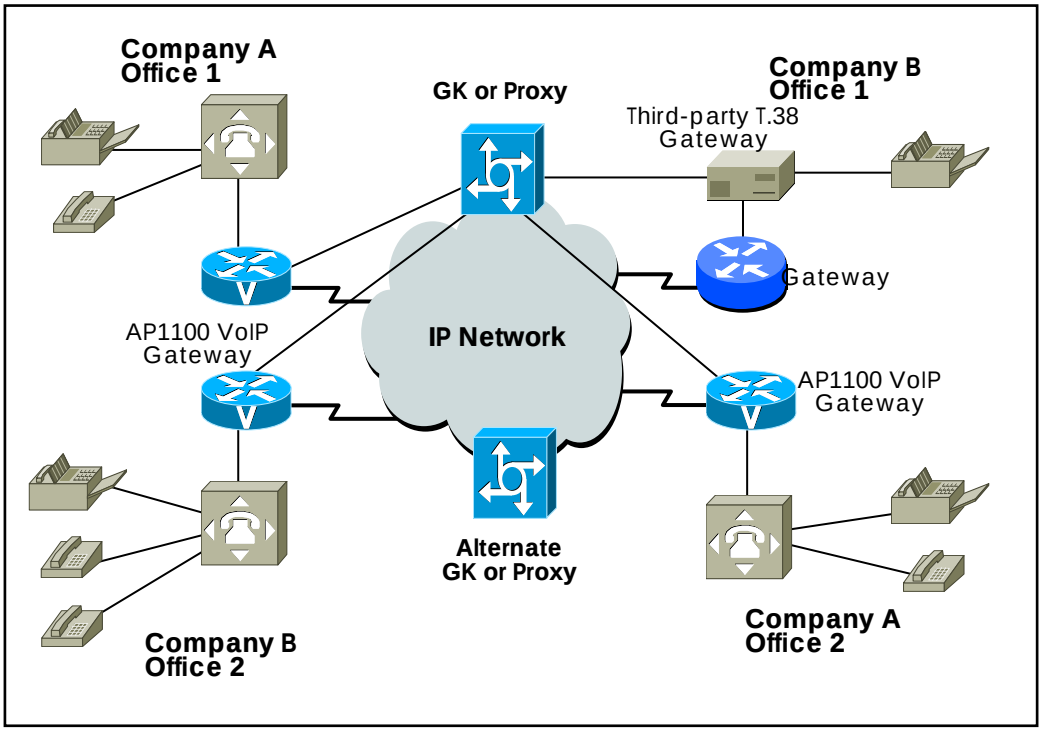


Figure 3.10 IP Network for T.38 FAX Relay

A voice call is established when the gateway transmits fax. The transmitting gateway detects fax tone which is generated from the fax response, and then it starts T.38 Mode Request by processing H.245. At this time, the receiving side recognizes T.38 Mode Request and terminates the voice channel. After terminates the voice channel it opens T.38 Fax Relay Channel.

3.6.2. Configuring T.38 Fax Relay

Below is a configuration procedure of T.38 fax relay;

Step	Commands	Purpose of Usage
1	(config)#**voice service voip**	Change to voice service configuration mode
2	(config-vservice-voip)#fax protocol {t38 [redundancy value] }	Set globalty applied default FAX protocol . t38 : Enable T.38 Fax relay protocol. . redundancy : (Optional) Redundant T.38 Fax packet . value : Redundancy value. 0 ~ 5, Default is 0.
3	(config-vservice-voip)#fax rate {2400/ 4800 / 7200 / 9600 /12000 / 14400 / disable }	Set maximum FAX rate
3	(config-vservice-voip)#**exit**	Exit voice-service configuration mode and return to global configuration mode

From above configuration, using redundancy need more bandwidth for sending copied fax packets. Therefore, the redundancy option is necessary for packet lossy network but should consider the network bandwidth consumption.

The fax rate option is a maximum value and the real fax transmission rate is negotiated automatically by two fax machines. The 'fax rate disable' means disable T.38 fax relay.

Above option is globally applied to all voip peers and overrided by option at voip peer.

3.6.3. Configuring FAX Relay with Bypass

When to configure the fax relay by G.711 PCM clean channel besides T.38 fax relay, use the following commands in global configuration mode. To operate fax relay in this mode, the voice channels needs to be opened by g711alaw or g711ulaw, so you need to check whether it is possible to connect from codec and codec-class of dial-peer configuration to this mode.

Step	Commands	Purpose of Usage
1	model name(config)#**voice service voip**	Change to voice service configuration mode
2	model name(config-vservice-voip)#**fax protocol bypass**	Specify global default fax protocol
3	model name(config-vservice-voip)#**exit**	Exit voice-service configuration mode and return to global configuration mode

3.7. Service Related Settings

3.7.1. ftp

VoiceFinder Gateway Series of AddPac Technology are capable of working as FTP server. A remote device of AddPac can access to VoiceFinder receiving and transmitting files.

This feature is disabled by default. To enable this feature, use **ftp server** commands and use **no service ftpd** command to disable it.

If a user needs to change FTP port information of the gateway, the user can change it by port control.

When to change, use **ftp port control <FTP Control Port> <FTP Data Port>** command and use **ftp port control 21 20** or **no ftp port** command.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **ftp server**	Enable FTP Server
3	model name(config)# **ftp port control <control port number> <data port>**	Change FTP server port to specified value
4	model name(config)# **exit**	End configuration

3.7.2. http

VoiceFinder Gateway Series of AddPac Technology are capable of working as web server. By using HTTP, a remote device can access to VoiceFinder Gateway Series of AddPac Technology and use web management feature.

This feature is disabled by default, but to enable it, use **http server** command and to disable it, use **no http server** command.

If a user needs to change HTTP port information of the gateway, use service-port.

When to change, use **http port <TCP/UDP Server Port>** and use **http port 80** or **no http port** command for default HTTP port (80).

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **http server**	Enable web server
3	model name(config)# **http port <port number>**	Change web server port to designated port by user.
4	model name(config)# **exit**	End configuration

3.7.3. ntp

VoiceFinder Gateway Series of AddPac Technology products support Network Time Protocol. This command is to set the gateway time to network time server same by taking the time information from a time server of the network.

This feature is basically disabled, but to enable ntp related settings, use **script ntpdate *WORD*** command. To disable it, use **no script ntpdate *WORD*** command to delete ntp related settings.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **script ntpdate *WORD***	Enter script mode to set NTP options
3	model name(config-script)# **server ip <ip address>**	Add server address

3.7.4. snmp

The AddPac VoiceFinder Gateway Series can perform the work of SNMP agent.
A remote equipment can access to the gateway by using SNMP and use the feature.
This feature is disabled at default. To enable, use snmp server command and to return to the default, use no snmp server command.

If a user wants to change snmp port information of the gateway, then service-port feature can be used.
When to change, **snmp port <TCP/UDP Server Port>** command is used, to use SNMP port(161) at default, use **snmp port 161** or **no snmp port** command.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **snmp server**	Enable SNMP agent
3	model name(config)# **snmp port <port number>**	Change to the designated SNMP port by user
4	model name(config# **exit**	Ends configuration

3.7.5. telnet

VoiceFinder Gateway Series of AddPac Technology are capable of working as telnet server.
A remote device can access to VoiceFinder Gateway Series of AddPac by using telnet for changing information and monitoring.
This is feature is enabled at default and to enable it, use **telnet server** command. To disable it, use **no telnet server** command.

If a user needs to change telnet port information of the gateway, the user may use service-port feature to change.
When to change, use **telnet port <TCP/UDP Server Port>** command and to use default telnet(23), use telnet port 23 or no telnet port.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **telnet server**	Enable telnet server
3	model name(config)# **telnet port <port number>**	Change telnet Server port to a user's specified port
4	model name(config)# **exit**	End configuration

3.8. Other VoIP Related Settings

3.8.1. Configuring H.323 Gateway

The gateway can take RAS (Registration, Admission, and Security) feature as to interoperate with a gatekeeper. This VoIP gateway can set a static IP address to VoIP peer to operate without the gatekeeper. Also, interoperating with gatekeeper, it can call to other side without knowledge of IP address of the other side.

To carry out this interoperation, h323 ID of the gateway is required and it must be a unique identifier. In case IP address of this VoIP gateway is 211.123.1.2, h323 ID is set to voip.211.123.1.2 at default. When you need this ID, you can set up **h323-id** command to use.

This VoIP uses **gkip** command to address a gatekeeper. By using **gkip** command, more than one gatekeeper can be addressed and registration is attempted to set in a priority order. Only one gatekeeper can be registered at the same time.

When security is need between the gateway and gatekeeper, security password command can be used to configure secure token. If this password is enabled, the gateway adds Crypto Token to the message and sends it to the gatekeeper. This security related settings can be configured when CryptoH323 Token is configured and cryptoEPPwdHash is supported.

register command can be used in gateway configuration mode for starting registration of the gatekeeper. To cancel registration of the gatekeeper in gateway configuration mode, use no register command.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **gateway**	Enter gateway configuration mode and register the gateway to the gatekeeper.
3	model name(config-gateway)# **gkip** *gatekeeper-ip-address [port] [priority]*	Specify IP address of the gatekeeper.
4	model name(config-gateway)# **h323-id** *h323-id*	Specify H323id of the gateway
5	model name(config-gateway)# **security password** *password*	Set H.235 security password
6	model name(config-gateway)# **register**	Register the gateway to the gatekeeper.
7	model name(config-gateway)# **exit**	End configuration (When to end, it interacts with the gatekeeper)

3.8.2. Configuring H323 Call Start Mode

The H.323 Version 2 describe the negotiation procedure by Fast Start Mode when to start H.323 call. This VoIP gateway can choose the procedure of Fast Start by using h323 call start in

voice service voip configuration mode. This Fast Start mode is set as default and slow start mode disables H245 Tunneling and Fast Start when to set up H245.

At all start modes, the H.245 procedure is proceeded to find out the other party's capability (T.38, DTMF relay capability).

Step	Commands	Purpose of Usage
1	model name# **configure terminal**	Enter global configuration mode
2	model name(config)# **voice service voip**	Enter voice service voip configuration mode
3	Router(vservice-voip)# **h323 call start {fast | slow| preferred-slow }**	Set up fast or slow start mode

3.8.3. Configuring SIP User Agent

The gateway can be supported with registration and call control (registration, INVITE, Security) features through interaction with SIP proxy server. This VoIP gateway can set a static IP address to VoIP peer to operate without the proxy server. Also, interoperating with SIP proxy server, it can call to other side without knowledge of IP address of the other side.

To carry register this gateway, user name, password and e164 of the gateway are required and the ID must be a unique identifier. When you need this ID, you can set up by using **sip-username** and **sip-password** in gateway SIPUA configuration mode and **destination number** of POTS peer commands. If you want each authentication for e.164 number for each dial-peer, use the same commands as in dial-peer. You must take precaution not to set SIP-UA related commands. You may ignore the settings of dial-peer, if **sip-username** and **sip-password** are set in SIP-UA and dial-peer

Use **sip-server** command to specify SIP Proxy Server for the gateway. By using **sip-server**, more than one SIP server can be addressed and registration is attempted to set in a priority order. Only one SIP server can be registered at the same time.

To cancel registration of SIP server in SIP-UA configuration mode, use **no register** command.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **sip-ua**	Enter SIP User Agent configuration mode and register the gateway to the SIP proxy server
3	model name(config-sip-ua)# **sip-server** *server-ip-address or domain name or sip-server-ipv6-address [port] [priority]*	Specify SIP Server ip address
4	model name(config-sip-ua)# **sip-username** *user-id*	Specify sip user id of the gateway

5	model name(config-sip-ua)# **sip-password** *password*	Specify sip password of the gateway
6	model name(config-sip-ua)# **register** <*e164* | *gateway* >	Register the gateway to SIP server
7	model name(config-sip-ua)# **exit**	End configuration

3.8.4. Configuring User Class

Configuration of user-class is used, when the outgoing call to FXO is received from the network, to reject receiving the call from an unauthorized user. When user-class is not configured and a user tries a call to a FXO port through the network and the FXO is connected to the extension of PBX, the user listens to the dial tone generated by the PBX and enters the desired digits of an extension number. When that FXO is connected to PSTN, the user listens to the dial tone generated from PSTN switch and enters the other party's number.

If any of user-class is configured, the user listens to a beep sound instead of the dial tone for the first call. When **password** is entered at this time and the call passes, the user can enter digits up to the number of max-digits which is explicit in the user-class (you may not be able to listen to the beep sound depending on the gateway in the outgoing side). Consequently, restriction of extension call, local area call, toll call, international call is possible by adjusting this **max-digit**. More than one user-class is available; therefore, setting call limit is possible for other user classes which are different to each other.

The reason, that the security is needed for the incoming call on FXO, is that there can be a misuse of unauthorized remote user by the direct call attempt, which is possible through this FXO port, and the indirect call attempt to PSTN through an extension of PBX is also possible. The gateway provided 2 type of security system and of the types can described in the following advantages and disadvantages

security permit-FXO is simple because the remote user doe not need to enter a password. On the other hand, all the IP addresses of VoIP peer must be registered and can not be registered together with a gatekeeper and can not perform the call limit to classify the registered peer.

Voice class user many be inconvenient in a way as for the user to enter password digits, but the security can be stronger and classification of call limit is possible.

Step	Commands	Description
1	model name# **configure terminal**	Enter configuration mode
2	model name(config)# **voice class user** *tag*	Enter user class configuration mode tag is unique identifier for user class
3	model name(config-class)# **password** *digits*	Set up a password. Digits are the texts of IA5 (0~9,#,*) with length of 4.
4	model name(config-class)# **max-digits** *value*	Set a maximum number of digits for an outgoing call to FXO. It is possible to configure extension call, local area call, toll call, international call

5	model name(config-class)# **exit**	End user class configuration mode

3.9. Interoperable Features with IP-PBX

The features described in this section are well interoperable with AddPac's IP-PBX IPNext Series.

3.9.1. Synchronizing Call-Forwarding Service of IP-PBX with PBX

This feature is generally provided from PBX to set up (*88*) and cancel (#88) call-forwarding feature by the same number.

A call can be carried out, by a scenario through IP-PBX and the feature can be applied to PBX with the same number, by controlling FXO port of the gateway basing on the scenario.

Step	Commands	Description
1	model name(config)#**service port-group <group-num>**	Specify service port group
2	model name(config-service-pgroup)#**port slot / port**	Specify FXO port to take the interoperable features. Start operating features by assigning the FXO in idle state.
3	model name(config-dialpeer-voip-9)# **forwarding-service port-group <group-num>**	Designate a port group number of the features generated from VoIP peer

3.9.2. IP-PBX Polling among IP-PBX Cluster

When many IP-PBX (i.e., SIP server) are in service, you need to set a priority order. When IP-PBX with a low priority is in service due to a failure of IP-PBX with a high priority, using this option the gateway can polling high priority IP-PBX status through 'PING' message.

When a PING response is received, try to register IP-PBX with a high priority to take the service.

Step	Commands	Description
1	model name(config-sip-ua)# **sip-server 172.16.1.88 5060 0 sip-server 172.16.1.89 5060 1**	Register SIP servers to SIP User Agent with a priority which different to each other
2	model name(config-sip-ua)# **higher-priority-polling enable**	Enable IP-PBX polling with a high priority
3	model name(config-sip-ua)# **timeout higher-priority-polling <5-3600>**	Set a frequency (by seconds)

3.9.3. Fault-Tolerant Call Attemption

When many IP-PBX (i.e., SIP server) are in service, keep providing services continuously by registering IP-PBX and maintaining the call

Step	Commands	Description
1	model name(config-sip-ua)# **sip-server 172.16.1.88 5060 0** **sip-server 172.16.1.89 5060 1**	Register SIP server to SIP User Agent
2	model name(config-sip-ua)# **fault-tolerance <1-10> <100-4000 (msec)>**	Set timeout value of call attempt and the counter of the call attempts

3.10. VoIP Related commands

3.10.1. VoIP Related Overall Commands

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					
	sip-ua	min-se	<60-86400>					

		register	e.164					
			gateway					
		retry-counter	<3-10>					
		signaling-port	<port number>					
		sip-server	<ip-address>	<port number>	<priority>			
		sip-username	<string>					
		sip-password	<string>					
		timeout	tretry	<10-86400>				
			treg	<10-86400>				
			tregtry	<10-86400>				
		user-register						
		end						
		exit						
	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-class	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-interface							

3.10.2. Global Configuration Commands

3.10.2.1. dial-peer call-hold

To use **call-hold** command in global configuration mode to place a hold for VoIP call. To restore the default selection order, use the no form of command.

dial-peer call-hold *character*

no dial-peer call-hold

3.10.2.1.1. Syntax

Keyword / Argument	Description
character	Specify call-hold. The texts can be used are h, n. - h(hook-flash) : Hold the present call by pressing hook-flash - n(not assigned) : not using hold as the same command as no dial-peer call-hold

3.10.2.1.2. Command Default

Character (n)

3.10.2.1.3. Command Modes

Global configuration

3.10.2.1.4. Usage Guidelines

Place a hold on the present call and use for other operation (taking a new call or start a new call). This command is used for passing the present call to the other number by interworking with call-transfer.

3.10.2.1.5. Examples

The following example configures to use hook-flash to call-hold.

```
dial-peer call-hold h
```

dial-peer call-hold h

3.10.2.2. dial-peer call-pickup

To pick up an incoming call to the other port in the same device, use the dial-peer call-pickup command in global configuration mode.

dial-peer call-pickup *character*

no dial-peer call-pickup

3.10.2.2.1. Syntax

Keyword / Argument	Description
character	Specify pickup. The texts can be used are <0-9>, #, *

3.10.2.2.2. Command Default

No default behavior or value.

3.10.2.2.3. Command Modes

Global configuration

3.10.2.2.4. Usage Guidelines

To pick up an incoming call to the other voice port of the gateway, use this command. To transfer the call, which has been picked up, to the other number, as to interwork with call-hold/call-transfer features?

3.10.2.2.5. Examples

The following example sets call pick up to '**'.

```
dial-peer call-pickup **
```

3.10.2.3. dial-peer call-transfer

To transfer a busy call to the other port, use dial-peer call-transfer command in global configuration mode. To restore the default selection order, use the no form of command.

dial-peer call-transfer *character*

no dial-peer call-transfer

3.10.2.3.1. Syntax

Keyword / Argument	Description
character	Specify call transfer, The value could be h, or n. - **h** (hook-flash) : Transfer the present cal by pressing hook-flash - **n** (not assigned) : Not to use transfer as the same command as no dial-peer call-transfer

3.10.2.3.2. Command Default

Character (n)

3.10.2.3.3. Command Modes

global configuration

3.10.2.3.4. Usage Guidelines

This command is used for transferring a busy call to the other number. Enable call-hold first because the feature interworks with call-hold.

3.10.2.3.5. Examples

The following example sets hook-flash to call transfer

```
dial-peer call-transfer h
```

3.10.2.4. dial-peer hunt

To specify a hunt selection order for dial peers, use the **dial-peer hunt command in** global configuration mode. To restore the default selection order, use the **no** form of this command.

dial-peer hunt *hunt-order-number*

no dial-peer hunt

3.10.2.4.1. Syntax

Keyword / Argument	Description
hunt-order-number	Apply a priority order algorithm from o to 7 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

3.10.2.4.2. Command Default

0 – longest match, explicit preference, random

3.10.2.4.3. Command Modes

global configuration

3.10.2.4.4. Usage Guidelines

Selection of outbound POTS or VoIP dial peer, which is sent out of the gateway, can be accomplished by comparing called party number of the inbound call to destination pattern of dial peer.

At this time, more than one dial peer, which is coincided with called party number, is called hunt group and attempts to make calls according to the priority order based on the consistent policy.

In other words for the case of VoIP peer, When the call attempt fails due to network connection fail, gatekeeper reject, another call attempt can be made by other dial peers in the hunt group.

In case of POTS peer, when the attempt is failed due to the port being busy of the voice port, another call attempt can be made to another dial peer in the hunt group.

The elements for determining the priority order, of the call attempts in the hunt group, are longest match, explicit preference, sequential, random.

First, longest match is the priority order according to the maximum digit matching with the destination number of dial peer and outgoing number.

For example, when the outgoing number is 5683848, destination number of dial peer 1 is 568T, the destination number of dial peer 2 is 568…, and the destination number of dial peer 3 is 56838.. and the destination number of dial peer 4 is 5683848, the priority order basing on longest match is dial peer 4 --> dial peer 3 --> dial peer 2 --> dial peer 1.

Explicit preference determines the priority order basing on a preference of the dial peer.

For example, when the preference of dial peer 1 is 3, dial peer 2 is 2, dial peer 3 is 1 and dial peer 4 is 0, the priority order basing on the explicit preference is dial peer 4 --> dial peer 3 --> dial peer 2 --> dial peer 1.

The random priority order determines the dial peer in the hunt group randomly. When the random priority order is sequential, the order determines the lowest frequency first, which is selected previously. Such algorithm of priority order is processed in combination, for instance, processing the default setting of dial-peer 0 can be determined by longest matching in the first stage, explicit preference in the second and random in the third.

3.10.2.4.5. Examples

The following example configures the dial peers to hunt in the following order: (1) longest match in phone number, (2) explicit preference and (3) random selection.

```
dial-peer hunt 1
```

3.10.2.5. dial-peer ipaddr-prefix

To make a call by an IP address and to specify a specific text, use dial-peer ipaddr-prefix command in global configuration mode.

dial-peer ipaddr-prefix *character*

no dial-peer ipaddr-prefix *character*

3.10.2.5.1. Syntax

Keyword / Argument	Description
character	Specify a prefix of IP address. The texts can be used are #, * n.

3.10.2.5.2. Command Default

Character (*)

3.10.2.5.3. Command Modes

global configuration

3.10.2.5.4. Usage Guidelines

A remote call can be carried out by pressing the predetermined numbers in dial-peer destination pattern and session target in general.

This way of predetermined setting is easy and secure, but also useful to make a call by using destination IP address for calling to those VoIP terminals and gateways installed in allocation near to the end-users.

This prefix is divided by a general call using number digit and direct call using IP address. To avoid collision with terminator text, IP address prefix is configured for changing terminator text and IP address prefix automatically.

To disable this IP address prefix, use dial-peer ipaddr-prefix n command.

3.10.2.5.5. Examples

The following example uses “*” as a specific prefix of IP address

```
configure terminal

dial-peer ipaddr-prex *
```

The followings describe how to make a call by using IP address from the gateway configured with the settings above. If the IP address is 10.0.0.1 and the called party number is 1234, the following numbers can be pressed:

```
* 10 * 0 * 0 * 1 * 1234 #
```

The first number represents **ipaddr-prefix** and its text is used for distinguishing from the dot of IP address.

When Destination terminal a simple VoIP phone such as Microsoft Netmeeting, the numeric order can be written as to follow:

```
* 10 * 0 * 0 * 1 #
```

3.10.2.6. dial-peer terminator

To change the character used as a terminator for variable-length dialed numbers, use the **dial-peer terminator**

command in global configuration mode. To restore the default terminating character, use the **no** form of this command.

dial-peer terminator *character*

no dial-peer terminator *character*

3.10.2.6.1. Syntax

Keyword / Argument	Description
character	Designates the terminating character for a variable-length dialed number. The valid characters are #, *.

3.10.2.6.2. Command Default

Character (#)

3.10.2.6.3. Command Modes

global configuration

3.10.2.6.4. Usage Guidelines

There are certain areas in the world (for example, in certain European countries) where telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed-number, the system does not place a call until the configured value for the **timeouts interdigits** command has expired or until the caller dials the terminating character. Use the **dial-peer terminator** global configuration command to change the terminating character. To disable the terminator, use dial-peer terminator n command.

3.10.2.6.5. Examples

The following example shows that "#" has been specified as the terminating character for variable-length dialed numbers:

```
configure terminal
dial-peer terminator #
```

3.10.2.7. dial-peer voice

To define a particular dial peer, to specify the method of voice encapsulation, and to enter dial-peer configuration mode, use the **dial-peer voice** command in global configuration mode. To delete a defined dial peer, use the **no** form of this command.

dial-peer voice *number* **{voip/pots}**

3.10.2.7.1. Syntax

Keyword / Argument	Description
number	The number to define a specific dial peer. Valid entry ranges from 0~65535.
voip	This is the command to indicate the dial peer as VoIP peer using voice encapsulation.
pots	This is the command to indicate the dial peer as POTS peer using voice encapsulation in IP backbone

3.10.2.7.2. Command Default

No default behavior or value.

3.10.2.7.3. Command Modes

global configuration

3.10.2.7.4. Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to dial-peer configuration mode from global configuration mode and to define a particular dial peer. Use the **exit** command to exit dial peer configuration mode and return to global configuration mode.

3.10.2.7.5. Examples

The following example specifies POTS peer to dial-peer 10

```
Configure terminal
dial-peer voice 10 pots
```

3.10.2.8. gateway

To enter gateway configuration mode, use the **gateway** command in global configuration mode. To deregister the gateway from the gatekeeper, use the **no** form of this command.

gateway

no gateway

3.10.2.8.1. Syntax

This command has no arguments or keywords.

3.10.2.8.2. Command Default

The gateway is deregistered

3.10.2.8.3. Command Modes

global configuration

3.10.2.8.4. Usage Guidelines

To enter gateway configuration mode, use the **gateway** command. If you enter **no gateway voip**, the VoIP gateway deregisters with the gatekeeper via the H.323 RAS URQ message.

3.10.2.8.5. Examples

The following example enables the gateway:

```
gateway
```

3.10.2.9. num-exp

To define how to expand a telephone extension number into a particular destination pattern, use the **num-exp** command in global configuration mode. To cancel the configured number expansion, use the **no** form of this command.

num-exp *extension-number expanded-number*

no num-exp *extension-number expanded-number*

3.10.2.9.1. Syntax

Keyword / Argument	Description
Extension-number	One or more digits that define an extension number for a particular dial peer. The valid characters are 0-9#*[].T.
Expanded-number	One or more digits that define the expanded telephone number or destination pattern for the extension number listed. The valid characters are 0-9#*%.T.

3.10.2.9.2. Command Default

No default behavior or value.

3.10.2.9.3. Command Modes

global configuration

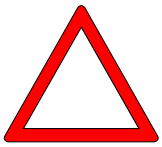
3.10.2.9.4. Usage Guidelines

Use this command to define how to expand a particular set of numbers (for example, a telephone extension number) into a particular destination pattern. With this command, you can bind 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 sent from a network or voice port is translated by number expansion and chooses the dial peer as a result of translation.

If more than one number expansion matches with the called party number, then the number expansion with most matches with a fixed patter of the expansion-number is chosen.

Caution



When to apply, a translation-incoming of the voice port or a translate-voip-incoming of the network, with a number expansion, requires an attention. When the number translation is clear, mixing the two is not recommended. When they are mixed together, one of either translation-incoming or translate-voip-incoming are applied first, then the number expansion afterwards. Range expression is possible for extension-number. You can apply a wildcard (.) to a number of digits in the called party number. When an extension number is configured with (.) or (T), number expansion is applied to all the called party numbers.

Expanded-number translates a fixed digit of extension-number to a string of expanded-number. The expanded-number has 2 different types.

The first type is configuring the expanded number with only IA5 texts (0-9#*). In this type, the fixed digit part of the extension-number is translated into the string part and the rest is added to the last digit.

The second type is using '%'. In this type, the number can be configured by replacing each digit to %xx variable. The value ranges from 1%~99%, which means, from the first digit of called-party-number to the 99th digit.

When expended number is configured with (.) or (T) only, the called-party-number can be configured with the digits except the fixed pattern of the extension-number.

3.10.2.9.5. Examples

The following example expands the extension number 55541 to the number 14085555541:

If inbound called party number is 5554123, it is expanded to 140855554123

```
num-exp 55541 14085555541
```

The following example does not apply when the inbound party number is 5551, but when it is 14085551234, it translates

```
num-exp 555.. 1408555
```

The following example is translates inbound called party number of 1251234 to 14085551234 and 3551234 to 14085551234.

```
num-exp [1-3][25]5.. 1408555
```

The following example translates the inbound called party number of 5551234 to 4441234.

```
num-exp 555.. 444%04%05%06%07%08%09%10%11%12
```

The following example translates all the inbound called party numbers of 55512, 5551234, and 555123456 to 444.

```
num-exp 555.. 444%99
```

The following example translates the inbound called party number of 5551234 새 3334

```
num-exp 555.. 111

num-exp 55512 222

num-exp 555[0-9][0-9][0-9] 333
```

The following example translates the inbound called party number of 5551234 to 1234.

```
num-exp 555 .

num-exp 555 T
```

The following example translates the inbound called party number of 5551234 to 9551234.

```
num-exp . 9

num-exp T 9
```

3.10.2.10. translation-rule

To create a translation name and enter translation-rule configuration mode to apply rules to the

translation name, use the **translation-rule** command in global configuration mode. To disable the translation rule, use the **no** form of this command.

translation-rule *tag*

no translation-rule *tag*

3.10.2.10.1. Syntax

Keyword / Argument	Description
tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number.

3.10.2.10.2. Command Default

No default behavior or value.

3.10.2.10.3. Command Modes

global configuration

3.10.2.10.4. Usage Guidelines

This command transfers to a mode to configure the setting s for the translation rule of the inbound and outbound called party number or calling party number

3.10.2.10.5. Examples

The following example creates translation rule set 100 and applies a rule to it:

```
translation-rule 100
   rule  0  2  822
```

3.10.2.11. voice-port

To enter voice-port configuration mode, use the **voice-port** command in global configuration

mode.

voice-port *port_number*

3.10.2.11.1. Syntax

Keyword / Argument	Description
slot-number/port-number	Voice port number. Valid entries are from 0 to 7

3.10.2.11.2. Command Default

No default behavior or value.

3.10.2.11.3. Command Modes

global configuration

3.10.2.11.4. Usage Guidelines

Use the **voice-port** global configuration command to switch to voice-port configuration mode from global configuration mode. Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

3.10.2.11.5. Examples

The following example accesses voice-port configuration mode for port 3, installed in slot 0:

```
configure terminal
voice-port 0/3
```

3.10.2.12. voice class clear-down-tone

3.10.2.12.1. Syntax

Keyword / Argument	Description

tag	Specify to clear down tone. The value ranges from 0 to 1.
lowFreq	Specifies low frequency value by Hz units which are provided from local switch or PBX. The valid entry ranges from 300Hz~1980Hz.
highFreq	Specifies low frequency value by Hz units which are provided from local switch or PBX. The valid entry ranges from 300Hz~1980Hz.Single tone value is 0.
onTime	On-time duration of clear down tone
offTime	On-time duration of clear down tone. Long duration value is 0.

3.10.2.12.2. Command Default

No default behavior or value.

3.10.2.12.3. Command Modes

global configuration

3.10.2.12.4. Usage Guidelines

Clear-down-tone detects call termination of FXO port connected to and generated from PSTN or PBX. The value of clear-down-tone (busy tone, fast busy tone) is different for each PSTN and PBX. So use voice class clear-down-tone for registration process in global configuration mode.

This command configures the tone detection by a user, besides the clear-down-tone provided by the system. If show clear-down-tone displays enough tone detection provided by system, at default, no additional settings are needed.

For the actual operation of the tone detection, which is added by this command, reboot the system after write the settings.

3.10.2.12.5. Examples

The following example configures the clear-down-tone for the dual tone of 350and 420Hz which are on time 250msec, off time 250msec.

```
configure terminal
voice class clear-down-tone 0 350 420 250 250
```

3.10.2.13. 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*

Keyword / Argument	Description
tag	Unique number that you assign to the voice class. Range is from 1 to 65,535.

3.10.2.13.1. Command Default

No default behavior or value.

3.10.2.13.2. Command Modes

global configuration

3.10.2.13.3. Usage Guidelines

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.

3.10.2.13.4. 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 codec, 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 terminal
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
```

3.10.2.14. voice class user

To create a tag number which classifies user voice class in voice-class configuration mode, use **voice class user** command in global configuration mode. To delete a codec voice class user, use the **no** form of this command.

voice class user *tag*

no voice class user *tag*

3.10.2.14.1. Syntax

Keyword / Argument	Description
tag	Assigns a unique number to voice class user. The valid entry ranges from 0 to 10. Only one tag exists in each gateway system

3.10.2.14.2. Command Default

No default behavior or value.

3.10.2.14.3. Command Modes

Global configuration mode

3.10.2.14.4. Usage Guidelines

User-class rejects receiving a call from unauthorized user when the outgoing signal of FXO is received in the network. If user-class is not configured and a user makes a call attempt on FXO port through the network, then the user is connected to an extension of PBX and listens to a dial tone.

The use enters the digits of the extension number afterwards. If FXO is connected to PSTN, the user listens to a dial tone generated by PSTN switch, and then enters the other party's number of PSTN.

If use class is configured, the user may hear a beep sound instead of a dial tone. If the user passes a password entry, the number can be entered as many as the max-digits which is explicated in the user-class (depending on the transmit side of the gateway, the beep sound may not be heard). Therefore, adjust a number of max-digit for extension calls, local calls, toll calls and international calls.

More than one user-class is possible, so it can place limits on other user-classes which are different to each other.

This command and security permit-FXO command provides security for the incoming call directing FXO port through the network. Such security is necessary for the incoming call to FXO, because a direct call attempt to PSTN is possible through FXO port and an indirect call attempt through PBX extension to PSTN are possible for a misuse of an unauthorized remote user.

These 2 security system provided by the gateway has an advantage and disadvantage.

Security permit-FXO is very simple and the remote user does not need to enter a password, but all IP addresses of VoIP peer for other parties have to be registered and it can not be used with a gatekeeper. Also call limitation is not possible to rank the registered peer.

On the other hand, voice class user provides an enhanced security and call limitation by many ranks.

3.10.2.14.5. Examples

The following example creates user class 1 and changes to user class configuration mode:

```
voice class user 1
  password 1234
  max-digits 10
  exit
```

3.10.2.15. voice class clear-down-cadence

Clear-down-tone detects call termination of FXO port connected to and generated from PSTN or PBX. The value of clear-down-tone (busy tone, fast busy tone) is different for each PSTN and PBX. So use voice class clear-down-tone for registration process in global configuration mode.

If the clear-down-tone generated from PBX or PSTN does not match with the gateway and a user is not able to know the information of the clearance-down-tone, then use this command. This command configures the detection of a tone by user besides the on provided from the system. To delete voice class clear-down-cadence, please use the no form of this command.

voice class clear-down-cadence <Detect Count> <Tone Level> < ActiveTimeDuration> < Idle Time Duration> < Active Power Variance> < Idle Power Variance>

no voice class clear-down-cadence

3.10.2.15.1. Syntax

Keyword / Argument	Description
Detect Count	mute detect cycle 주기
Tone Level	mute detect tone level
ActiveTimeDuration	clear-down tone play duration
Idle Time Duration	clear-down tone idle duration
Active Power Variance	clear-down tone play error range
Idle Power Variance	clear-down tone idle error range

3.10.2.15.2. Command Default

No default behavior or value.

3.10.2.15.3. Command Modes

Global configuration

3.10.2.15.4. Usage Guidelines

If a call is terminated on analog side and a voice interface of the gateway is FXO, the call is terminated by re-order-tone or clear-down-tone which comes from PABX. If clear dial tone does not recognize a tone, the remote gateway can not provide release signal, as a result, the port can be busy, as long as, a local user does not receive call termination signal from the gateway in the remote side. To resolve such problem, the gateway needs to be configured with an accurate clear-down-tone cycle and cadence value. When the user is informed with tone frequency and cadence, the user can use clear-down-tone or re-order-tone commands to change the tone.

This command does not perform clear-down-tone in frequency but analyzes an energy level and

clears the FXO port by force.

Clear-down-tone is analyzed by its cadence (on/off time) and energy level deviation. On/off borderline of tone and depend on a type codec to be used, energy level deviation can be really different, so you would better get familiar with its usage then apply.

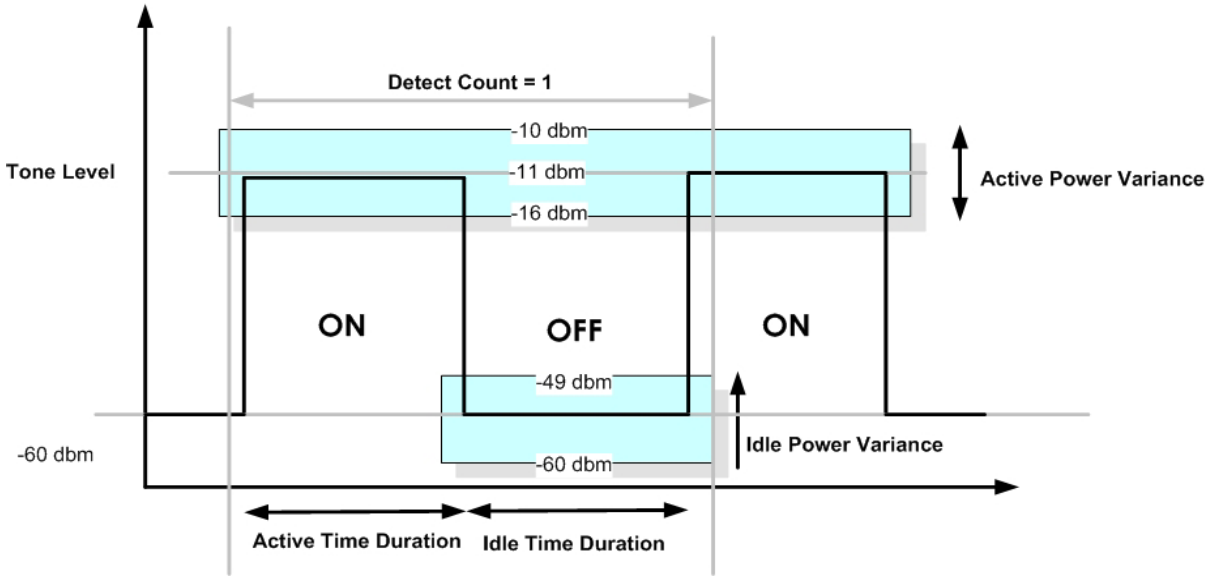
(Example (In case of codec = g7231))

Codec=G711,G729 (10msec * count), G7231 (30msec * count)

debug rta voice
Make PABX play clear-down-tone by hook-on the phone which connected extension line.
8 18 18 17 17 17 17 17 17 18 18 18 18 18 18 14 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 16 49 58 56 57 54 58 56 53 56 57 56 56 55 57 56 57 55 56 55 57 55 57 55 57 55 15 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 16 50 56 55 56 56 57 56 57 56 57 56 58 56 57 56 57 54 57 55 57 56 57 54 58 15 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11 12 10 12 11 11

Calculation

count of Tone Level (-10 ~ -16dbm) = 25
Active Time Duration (25 * 30ms) = 750ms, if codec is g711 or g729, it will be codec 250ms (25 * 10ms)
count of Idle level (-49 ~ -60dbm) = 25
Idle Time Duration (25 * 30ms) = 750ms, if codec is g711 or g729, it will be codec 250ms (25 * 10ms)
Tone Level = -11
Active Power Variance = 5 (-6 ~ 16dbm)
Idle Power Variance = 11 (-49 ~ -60)
Idle Tone Level is set as -60 dbm internally, so it was calculated as (11 = (-60(Min)) - (49(Max))).



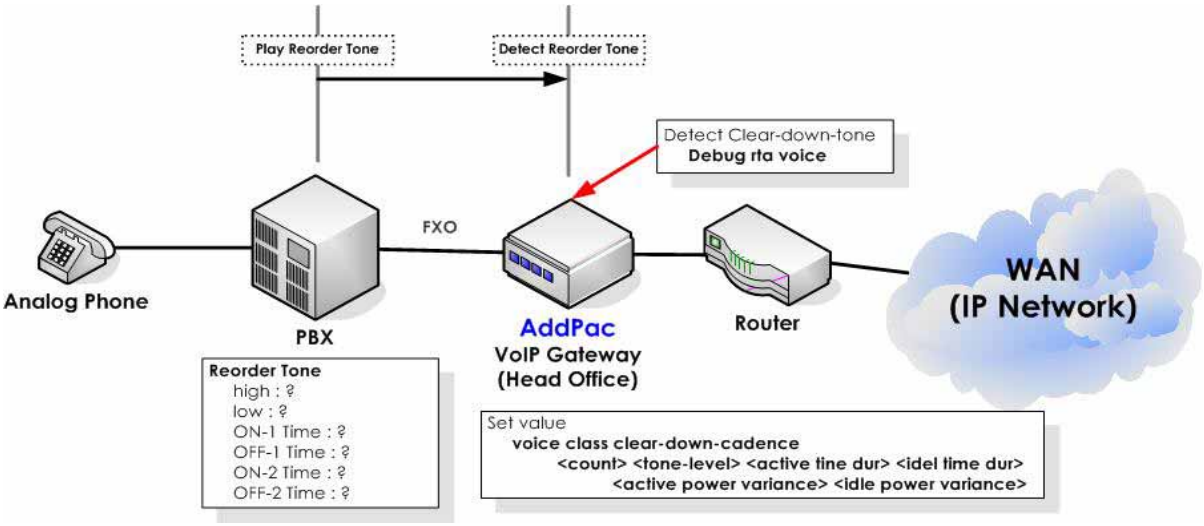


Figure 3.11 Detecting clear-down-tone parameter by using VoIP Gateway

3.10.2.15.5. Examples

The following example analyzes clear-down-tone parameter by using VoIP gateway for the figure above

```
voice class clear-down-cadence 1 -11 750 750 5 11
exit
```

3.10.2.16. voice service

To enter voice-service configuration mode and to specify a voice-encapsulation type, use the **voice service** command in global configuration mode.
Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

voice service voip

3.10.2.16.1. Syntax

Keyword / Argument	Description
voip	The keyword to enter VoIP configuration mode to specify VoIP parameters

3.10.2.16.2. Command Default

No default behavior or value.

3.10.2.16.3. Command Modes

global configuration

3.10.2.16.4. Usage Guidelines

This command is used for transferring from global configuration mode to voice-service mode and specifying voice encapsulation type. Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

Voice-service configuration mode is used for packet telephony service commands that affect the gateway globally.

3.10.2.16.5. Examples

The following example shows how to enter voice-service configuration mode and assign a VoIP parameter starting from global configuration mode:

```
voice service voip
```

3.10.2.17. voip-interface

To specify VoIP interface, use voip-interface command in global configuration mode. To restore the default selection order, use the no form of command.

voip-interface *<ip/ipv6><interface-name><slot number><port number>*

no voip-interface *<ip/ipv6>*

3.10.2.17.1. Syntax

Keyword / Argument	Description
interface-name	Specify an interface of the gateway. The interface names re Ethernet 0.0, Ethernet 1.0, Serial 0 and others

3.10.2.17.2. Command Default

FastEthernet 0 /0 is the default interface

3.10.2.17.3. Command Modes

global configuration

3.10.2.17.4. Usage Guidelines

This command specifies a particular VoIP interface. The interface uses the specified interface for VoIP service. If IP address is not specified, checking VoIP related setting and details is not possible.

3.10.2.17.5. Examples

The following example specifies VoIP service to ethernet 1.0 interface.

```
configure
  voip-interface ip FastEthernet 1/0
```

The following example specifies VoIP service to FastEthernet 0/0 IPv6 interface:

```
Configure terminal
voip-interface ipv6 FastEthernet 0/0
```

3.10.3. Voice Port Configuration Commands

3.10.3.1. announcement

To enable announcement to a particular port, use this command. To disable this feature, use the no form of this command.

announcement

no announcement

3.10.3.1.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.3.1.2. Command Default

Enabled

3.10.3.1.3. Command Modes

Voice-port configuration

3.10.3.1.4. Usage Guidelines

Enable announcement first in voice service voip configuration mode, so the configured feted on the port can operate normally.

3.10.3.1.5. Examples

The following example enables announcement on voice-port 0/0:

```
voice-port 0/0
  announcement
```

3.10.3.2. busyout action

To place a voice port in busyout state, use **busyout** action command.

busyout action {none| port-down| tone}

3.10.3.2.1. Syntax

Keyword / Argument	Description
action	Specify the action when it is in busyout state - none: do nothing - port-down: set the port to muting operation - tone: notifying busyout by tone

3.10.3.2.2. Command Default

Default busyout action tone

3.10.3.2.3. Command Modes

voice-port configuration mode

3.10.3.2.4. Usage Guidelines

To find out busyout state, enable busyout monitoring first which acts in voice service VoIP configuration mode, so the specified port act in normal behavior.

3.10.3.2.5. Examples

The following example shows the analog voice-port busyout state set to 0/0:

```
voice-port 0/0
 busyout action port-down
```

3.10.3.3. busyout backup

To enable one voice-port configured with back-up port, as soon as the other voice-port is set to off-hook, acts same as off-hook at the same time.

busyout backup {none| off-hook}

3.10.3.3.1. Syntax

Keyword / Argument	Description
backup	Specifies an action to a configured port with pstn-backup-port - none: do nothing - off-hook: off-hook at the same time

3.10.3.3.2. Command Default

Busyout backup none

3.10.3.3.3. Command Modes

voice-port configuration mode

3.10.3.3.4. Usage Guidelines

Enable pstn-backup-port first, so the configured port with this feature can act normally

3.10.3.3.5. Examples

The following sets an action of off-hook at the same time to the voice-port configured with pstn-backup-port when voice-port 0/0 is off-hook:

```
voice-port 0/0
  busyout backup off-hook
```

3.10.3.4. caller-id

To enable caller ID, use the **caller-id** command in dial peer configuration mode. To disable caller ID, use the **no** form of the command.

caller-id { enable | name | type }

no caller-id { enable | type }

3.10.3.4.1. Syntax

Keyword / Argument	Description
enable	Enable caller-id
name *enable* | *disable*	Indicate to do not indicate name field of caller-id
type *bellcore|etsi|etsi-dtmf|etsi-dtmf-prior-ring|ntt*	Specifies a caller-id type. Type is different for each country. Adjust configuration to be suitable to user's environment.

3.10.3.4.2. Command Default

Caller-id disable | caller-id name disable |caller-id type bellcore

3.10.3.4.3. Command Modes

Voice-port configuration mode

3.10.3.4.4. Usage Guidelines

The command is effective only if you subscribe to caller ID service. If you enable caller ID on a router without subscribing to the caller ID service, caller ID information does not appear on the telephone display. The configuration of caller ID must match the device connected to the POTS port. That is, if a telephone supports the caller ID feature, use the command **caller-id** to enable the feature. If the telephone does not support the caller ID feature, use the command default or disable the caller ID feature. Odd ringing behavior might occur if the caller ID feature is disabled when it is a supported telephone feature or enabled when it is not a supported telephone feature.

3.10.3.4.5. Examples

The following example specifies caller-is-type to bellcore and enables the caller-id and applies to voice port 0/0. So the caller's number is delivered to the voice port.

```
voice-port 0/0
        caller-id enable
        caller-id type bellcore
```

3.10.3.5. 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

3.10.3.5.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.3.5.2. Command Default

Comfort noise is generated and enabled by default.

3.10.3.5.3. Command Modes

Voice-port configuration mode

3.10.3.5.4. Usage Guidelines

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 hears dead silence when the remote party is not speaking. The configuration of the **comfort-noise** command affects only 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.

3.10.3.5.5. Examples

The following example disables comfort noise on voice-port 1

```
voice-port 1
  no comfort-noise
```

3.10.3.6. connection plar

To specify a plar for a voice port, use the **connection plar** 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 }

3.10.3.6.1. 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 afar-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 numbers that specify the E.164 telephone number.

3.10.3.6.2. Command Default

No connection mode is specified.

3.10.3.6.3. Command Modes

Voice-port configuration

3.10.3.6.4. Usage Guidelines

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 incoming calls over this connection. The destination peer is determined by the called number.

3.10.3.6.5. Examples

The following example shows PLAR as the connection mode with a destination telephone number

of 555-9262. In this example, a telephone connected with voice-port 1 id off-hook then the telephone make a call to 5559262 automatically

```
voice-port 1/0
   connection plar 5559262
```

3.10.3.7. connection trunk

To specify a plar for a voice port, use the **connection plar** 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 }

3.10.3.7.1. 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 afar-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 numbers that specify the E.164 telephone number.

3.10.3.7.2. Command Default

No connection mode is specified.

3.10.3.7.3. Command Modes

Voice-port configuration

3.10.3.7.4. Usage Guidelines

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 incoming calls over this connection. The destination peer is determined by the called number.

3.10.3.7.5. Examples

The following example shows PLAR as the connection mode with a destination telephone number

of 555-9262. In this example, a telephone connected with voice-port 1 id off-hook then the telephone make a call to 5559262 automatically

```
voice-port 1/0
   connection plar 5559262
```

3.10.3.8. description (voice port)

To add a description of connection for the port, use description command in voice-port configuration mode. To disable this feature, use the no form of this command.

description *string*

no description

3.10.3.8.1. Syntax

Keyword / Argument	Description
string	Character string from 1 to 255

3.10.3.8.2. Command Default

Enabled with a null string

3.10.3.8.3. Command Modes

Voice-port configuration mode

3.10.3.8.4. Usage Guidelines

Use the **description** command to include descriptive text about this interface connection. This information is displayed when you issue a **show** command and does not affect the operation of the interface in any way.

3.10.3.8.5. Examples

The following example identifies voice port 0 on the VoIP gateway as being connected to the marketing department:

```
voice-port 0
description marketing_dept
```

3.10.3.9. did

To enable the direct inward dialing (DID) call treatment for an incoming called number, use the **direct-inward-dial command in** dial peer configuration mode. To disable DID on the dial peer, use the **no** form of this command.

did { normal | none | ntt-modem | ntt-pb }

3.10.3.9.1. Syntax

Keyword / Argument	Description
normal	Sends the number after hook-off. DID was introduced from the initial version
none	No DID.
ntt-modem	Japanese NTT specific treatment. Dials in a specific order by using FSK
ntt-pb	Japanese NTT specific treatment. Dials in a specific order by using PB (Push Button)

3.10.3.9.2. Command Default

Normal

3.10.3.9.3. Command Modes

Voice-port configuration mode

3.10.3.9.4. Usage Guidelines

Use the **direct-inward-dial** command to enable the DID call treatment for an incoming called number. When this feature is enabled, the incoming call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller. Use the **no** form of this command to disable DID on the dial peer. When disabled, the called number is used to select the outgoing dial peer. The caller is prompted for a called number via dial tone. This command is applicable only to plain old telephone service (POTS) dial peers.

3.10.3.9.5. Examples

The following example enables did normal for the incoming called number on voice port 0/0. When treatment DID is applied to the voice port, it runs in the normal form (sending digit after hook-off):

```
voice-port 0/0
        did normal
```

- This command is applicable only to plain old telephone service (POTS) dial peers.
- The following example delivers a specific number for the call arriving to port 0/0: `prefix <Input Prefix String>`
- Use the following command to deliver a number of destination pattern <Destination Pattern String>:

```
forward-digit {from | last} {number}
```

  - For more details of the commands, refer to 5.7.4.6 forward-digit and 5.7.4.10 prefix

3.10.3.10. echo-cancel

To enable echo cancel, use this command. To disable this feature, use the no form of this command.

echo-cancel

no echo-cancel

3.10.3.10.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.3.10.2. Command Default

Enabled

3.10.3.10.3. Command Modes

Voice-port configuration

3.10.3.10.4. Usage Guidelines

The **echo-cancel enable** command enables cancellation of voice that is sent out the interface and received back on the same interface; sound that is received back in this manner is perceived by the listener as an echo. In most cases, this feature is enabled ad it is set to be enabled at default.

3.10.3.10.5. Examples

The following example disables the echo cancel in voice-port 3:

```
voice-port 3
  no echo-cancel
```

3.10.3.11. fax-early-detect

To apply **fax-early-detect** to the designated port, use this command. To delete fax-early-detect, use the no form of this command.

fax-early-detect

no fax-early-detect

3.10.3.11.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.3.11.2. Command Default

disable

3.10.3.11.3. Command Modes

Voice-port configuration mode

3.10.3.11.4. Usage Guidelines

The AddPac Voice Finder Gateway Series are configured at default, as in a way to prevent an error of transferring to fax mode by side tone (CED tone) and the gateways do not transfer to fax mode as long as HDLC is not detected even after fax tone received However, some conventional fax machines does not have retransmission function of HDLC data. The user may recognize this situation as an error, because HDLC data transmit can be swept away even before VoIP setup. fax-early-detect. Fax-early-detect command allows transferring to fax mode even before HDLC is detected, at the stage of receiving CED Tone.

This feature is not recommended for a general environment, in where other fax machines would be connected, during telephone conversation, unexpectedly side tone (fax tone) is detected and transfer to fax mode.

3.10.3.11.5. Examples

The following example configures fax-early-detect to voice port 0/0, then applies

```
voice-port 0/0
```

3.10.3.12. high-dtmf-gain

To specify high frequency dtmf gain value of DTMF tone to a designated port in voice-port configuration mode, use **high-dtmf-gain** command. To disable this feature, use the no form of this command.

high-dtmf-gain *value*

no high-dtmf-gain *value*

3.10.3.12.1. Syntax

Keyword / Argument	Description
value	Specify an amount of gain by decibel unit for an interface. Range is integers from -31 to 3

3.10.3.12.2. Command Default

-5 decibels

3.10.3.12.3. Command Modes

Voice-Port configuration

3.10.3.12.4. Usage Guidelines

Implement dtmf tone for the entire system by using the command set of **high-dtmf-gain** and **low-dtmf-gain**.

3.10.3.12.5. Examples

The following example adds the gain value of 3 decibels from receiver's side of the gateway interface.

```
port 4
high-dtmf-gain 3
```

3.10.3.13. input gain

To configure a specific input gain value or enable automatic gain control, use the **input gain** command in voice-port configuration mode.

To disable this feature, use the no form of this command.

Input gain *value*

no Input gain *value*

3.10.3.13.1. Syntax

Keyword / Argument	Description
value	Gain, in decibels (dB), to be inserted at the receiver side of the interface. Range is integers from –31 to 31. The default is 0.

3.10.3.13.2. Command Default

0 decibels

3.10.3.13.3. Command Modes

Voice-Port Configuration

3.10.3.13.4. Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. You must consider other equipment (including PBXs) in the system when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there is typically a minimum attenuation of –6 dB between phones, especially if echo cancellers are present. Connections are implemented to provide 0 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

You cannot increase the gain of a signal to the public switched telephone network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

3.10.3.13.5. Examples

The following example inserts a 3-dB gain at the receiver side of the interface in the gateway:

```
port 4
input gain 3
```

3.10.3.14. low-dtmf-gain

To specify low frequency dtmf gain value of DTMF tone to a designated port in voice-port configuration mode, use **low-dtmf-gain** command. To disable this feature, use the no form of this command.

low-dtmf-gain *value*

no low-dtmf-gain *value*

3.10.3.14.1. Syntax

Keyword / Argument	Description
value	Specify an amount of gain by decibel unit for an interface. Range is integers from -31 to 3

3.10.3.14.2. Command Default

-5 decibel

3.10.3.14.3. Command Modes

Voice-port configuration

3.10.3.14.4. Usage Guidelines

Implement dtmf tone for the entire system by using the command set of **high-dtmf-gain** and **low-dtmf-gain**.

3.10.3.14.5. Examples

The following example adds the gain value of -10 decibels from receiver's side of the gateway interface.

```
port 4
low-dtmf-gain -10
```

3.10.3.15. output gain

To configure a specific output gain value or enable automatic gain control, use the **output gain** command in voice-port configuration mode. To disable the selected output gain value, use the **no** form of this command.

output gain *value*

no output gain *value*

3.10.3.15.1. Syntax

Keyword / Argument	Description
value	Attenuation, in decibels (dB), at the transmit side of the interface. Range is integers from –31 to 31. For Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), the default is 0. ports: *decibels*: 0 decibels

3.10.3.15.2. Command Default

For Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), the default is 0.

3.10.3.15.3. Command Modes

Voice-Port Configuration

3.10.3.15.4. Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output gain** commands. You must consider other equipment (including PBXs) in the system when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB 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 dB. You cannot increase the gain of a signal to the public switched telephone network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output gain.

3.10.3.15.5. Examples

On the gateway, the following example configures a 3-dB loss to be inserted at the transmit side of the interface:

```
port 4
output gain 3
```

3.10.3.16. polarity-inverse

To enable polarity inverse of FXS voice port. To disable this feature, use the no form of this command.

polarity-inverse

no polarity-inverse

3.10.3.16.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.3.16.2. Command Default

Disabled

3.10.3.16.3. Command Modes

Voice-port configuration

3.10.3.16.4. Usage Guidelines

This command enables PBX, which is connected to FXS, to perform accounting by inversing the both of the beginning and end of billing feature

3.10.3.16.5. Examples

The following example enables polarity inverse on voice-port 1/0.

```
voice-port 1/0
  polarity-inverse
```

3.10.3.17. pstn-backup-port

To configure FXO port to pstn-backup-port, use this command. To disable this feature, use the no form of this command.

pstn-backup-port *slot/port*

no pstn-backup-port

3.10.3.17.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.3.17.2. Command Default

Disabled

3.10.3.17.3. Command Modes

Voice-port configuration

3.10.3.17.4. Usage Guidelines

3 causes of busyout state for gateway are to follow:

First power supply is suspended.

Second, the link is down on LAN interface of the gateway

Third, the link is down on gatekeeper, MGC or proxy server, so the gateway can not be connected.

SIP proxy server does not have a feature to detect busyout. Hunt feature can be used when voip-peer connection fails. Hunting can be carried out by the pots-peer which is configured to PSTN-backup-port.

VoIP call can not be made when the gateway is in busyout state. The telephone network can be opened continually through PSTN.

When PSTN backup feature is used, enable busyout monitor in voice-service-voip configuration mode.

If PSTN port presents in the gateway, no need to configure the feature. If the gateway has only FXS port and FXO, enable “PSTN-Backup-port” on FXS port. By using this feature, you can make FXO port to work as PSTN.

3.10.3.17.5. Examples

The following example sets FXO 1/0 port from FXS 0/0 to pstn-backup-pot.

```
voice-port 0/0
  pstn-backup-port 1/0
```

3.10.3.18. ring number

To specify the number of rings for a specified Foreign Exchange Office (FXO) voice port, use the **ring number** command in voice port configuration mode. To reset to the default, use the **no** form of this
command.

ring number *number*

no ring number *number*

3.10.3.18.1. Syntax

Keyword / Argument	Description
number	Number of rings detected before answering the call. Range is from 1 to 255. This feature is useful for VoIP devices and ARS. The default is 1.

3.10.3.18.2. Command Default

1 ring

3.10.3.18.3. Command Modes

Voice-port configuration

3.10.3.18.4. Usage Guidelines

Use this 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 online did not answer the incoming call in the configured number of rings.

This command is not applicable to Foreign Exchange Station (FXS) or E&M interfaces because they do not receive ringing on incoming calls.

3.10.3.18.5. Examples

The following example sets 5 as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1
ring number 5
```

3.10.3.19. shutdown (voice-port)

To take the voice ports for a specific voice interface card offline, use the **shutdown** command in voice-port configuration mode. To put the ports back in service, use the **no** form of this command.

shutdown

no shutdown

3.10.3.19.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.3.19.2. Command Default

No shutdown

3.10.3.19.3. Command Modes

Voice-port Configuration

3.10.3.19.4. Usage Guidelines

When you use this command, all ports on the voice interface card are disabled. When you use the **no** form of the command, all ports on the voice interface card are enabled. A telephone connected to an interface hears dead silence when a port is shut down.

3.10.3.19.5. Examples

The following example takes voice port 3 offline:

```
configure
voice-port 3
shutdown
```

3.10.3.20. timeout

To set a timer on a voice-port, use **timeout** command. To delete timer, use the no form of this command.

timeout {tterm| tvcc}

no timeout {tterm| tvcc}

3.10.3.20.1. Syntax

Keyword / Argument	Description
tterm	Starts timer from the point of time when a designated voice-port is off-hook. The call is terminated after a certain configured time.
tvcc	Voice-confirmed-connect timer is applied for the voice-port.

3.10.3.20.2. Command Default

No default behavior or value.

3.10.3.20.3. Command Modes

Voice-port configuration

3.10.3.20.4. Usage Guidelines

Once timeout tterm is set up in voice-service-voip configuration mode to limit call duration, it is applied to all ports of the gateway for transmit/ receiving. To limit call duration on outgoing call, you need to configure voice-port configuration mode.

In a voip configuration for passing through PBX, when a use in PBX extension (call receiver) side acts off-hook and an actual voice is not connected, use timeout tvcc command to deliver a disconnect message to the transmit side after a certain period of time. When PBX does not deliver a connect message (for instance, when a user does not take a call on PBX extension side), use this command to transmit a disconnect message after a certain time of period set by timer (one-stage-dialing should be carried out from transmit telephone.

Set up voice-confirmed-connect command first in voice-service-voip configuration mode to use timeout tvcc.

3.10.3.20.5. Examples

The following command limits the outgoing call duration to 3 minutes for voice-port 0/0 of the gateway:

```
voice-port 0/0
timeout tterm 180
```

3.10.3.21. translate-incoming

To apply a translation rule to manipulate dialed digits on an inbound POTS call leg, use the **translate** command in voice-port configuration mode. To remove the translation rule, use the **no** form of this command.

translate-incoming { called-number | calling-number } *tag*

no translate-incoming { called-number | calling-number }

3.10.3.21.1. Syntax

Keyword / Argument	Description
called-number	Translation rule applies to the inbound calling party number.
calling-number	Translation rule applies to the inbound called party number.
tag	Tag by which the rule set is referenced. Range is from 0 to 65535. There is no default value.

3.10.3.21.2. Command Default

No default behavior or values

3.10.3.21.3. Command Modes

Voice-port configuration

3.10.3.21.4. Usage Guidelines

To apply a translation rule to the inbound call of voice-port, use **translation-rule** command. When the translation rule applies to the called party number, it checks whether the translation takes a place every time, the number information is entered to the voice port in the order. At this time, the number translation takes a place only once.

3.10.3.21.5. Examples

The following example creates translation rule set 10 and applies it to the calling party number of voice port 1. When the calling party number of the voice port is 93450, the translation rule set translates if to 9563450.

```
translation-rule 10
```

```
        rule 0  9  956
        rule 1  8  878
voice-port 1
        translate-incoming calling-number 10
```

3.10.4. Dial Peer pots / voice Configuration Commands

3.10.4.1. answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** command in dial peer configuration mode. To disable the configured telephone number, use the **no** form of this command.

answer-address *string*

no answer-address

3.10.4.1.1. Syntax

Keyword / Argument	Description
String	Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the following special characters(#, *) : ● The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. ● Period (.) which matches any entered digit (this character is used as a wildcard). These characters can not come first in a string. (example: .650 is not the valid entry) ● 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, which is similar to Regular Expression Rule.

3.10.4.1.2. Command Default

The default value is enabled with a null string

3.10.4.1.3. Command Modes

Dial peer voice configuration (VOIP dial peer)

3.10.4.1.4. Usage Guidelines

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: by identifying either the interface through which the call is received or the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface is associated with the incoming call.

For calls that come in from a plain old telephone service (POTS) interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

There are certain areas in the world (for example, certain European countries) where valid telephone numbers can vary in length. Use the optional control character **T** to indicate that a particular **answer-address** 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.

3.10.4.1.5. Examples

The following example shows the calling party number of the inbound VoIP 555-0104 as the VoIP peer 10 of an incoming call being configured:

```
dial-peer voice 10 voip
 answer-address 526....
```

3.10.4.2. codec

To specify the voice coder rate of speech for a dial peer, use the **codec** command in dial peer configuration mode. To reset the default value, use the **no** form of this command.

codec {g711alaw / g711ulaw / g729r8 / g7231r63 / g7231r53 }

no codec

3.10.4.2.1. 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 6.3Kbps codec which is the default codec of the gateway
G7231r53	G.723.1 5.3Kbps codec

3.10.4.2.2. Command Default

G.723.1 6.3Kbps codec

3.10.4.2.3. Command Modes

Dial peer configuration

3.10.4.2.4. Usage Guidelines

Use this command to define a specific voice coder rate of speech and payload size for a dial peer.

A specific codec type can be configured on the dial peer as long as it is supported by the setting used with the **codec complexity** voice-card configuration command. The **codec complexity** command is voice-card specific and platform specific. The **codec complexity** voice-card configuration command is set to either high or medium. If the **codec complexity** command is set to high, the following keywords are available: **g711alaw, g711ulaw.**

The **codec** dial peer configuration command is particularly useful when you must change to a small-bandwidth codec. Large-bandwidth codec, such as G.711, do not fit in a small-bandwidth link. However, the g711alaw and g711ulaw codec provide higher quality voice transmission

than other codec. The g729r8 codec provides near-toll quality with considerable bandwidth savings.

If codec values for the dial peers of a connection do not match, the call fails.

3.10.4.2.5. Examples

The following example configures a voice coder rate for VoIP dial peer 10 that provides toll quality but uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
  codec g711alaw
```

3.10.4.3. description (dial-peer)

To add a description to a dial peer, use the **description** command in dial peer configuration mode. To remove the description, use the **no** form of this command.

description *string*

no description

3.10.4.3.1. Syntax

Keyword / Argument	Description
string	The character string for description of the dial-peer, range is from 1 to 255characters

3.10.4.3.2. Command Default

The default value is enabled with a null string

3.10.4.3.3. Command Modes

Dial-Peer Configuration

3.10.4.3.4. Usage Guidelines

Use this command to include descriptive text about the dial peer. The description displays in **show** command output and does not affect the operation of the dial peer.

3.10.4.3.5. Examples

The following example shows a description as of dial peer 10 of the gateway in Seoul office:

```
dial-peer voice 10 voip
  description Seoul_office
```

3.10.4.4. destination-pattern

To specify either the prefix or the full E.164 telephone number to be used for a dial peer, use the **destination-pattern** command in dial peer configuration mode. To disable the configured prefix or telephone number, use the **no** form of this command.

destination-pattern *string [* **T** *]*

no destination-pattern

3.10.4.4.1. Syntax

Keyword / Argument	Description
String	Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the following special characters(#, *) : ● The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. ● Period (.) which matches any entered digit (this character is used as a wildcard). These characters can not come first in a string. (example: .650 is not the valid entry) ● 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, which is similar to Regular Expression Rule.
T	Control character that indicates that the **destination-pattern** value is a variable-length dial string.

3.10.4.4.2. Command Default

Enabled with a null string

3.10.4.4.3. Command Modes

Dial peer configuration

3.10.4.4.4. Usage Guidelines

Use the **destination-pattern** command to define the E.164 telephone number for a dial peer.

The pattern you configure 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 that

correspond 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, 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.

3.10.4.4.5. Examples

The following example shows configuration of 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 possible numeric values are 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 possible numeric values are between 5551439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice 4 voip
  destination-pattern 555[13579]439
```

3.10.4.5. diversion

To apply the configured settings call-diversion to a particular pots-peer, use **diversion** command. To delete diversion, use the no form of this command.

diversion *tag*

no diversion

3.10.4.5.1. Syntax

Keyword / Argument	Description
tag	Specify a tag same as the tag of call-diversion that you intend to apply.

3.10.4.5.2. Command Default

No default behavior or value.

3.10.4.5.3. Command Modes

Dial peer configuration

3.10.4.5.4. Usage Guidelines

This command is connected to call diversion settings. To use this command, create call-diversion first.

3.10.4.5.5. Examples

The following example configures the created call-diversion 100 in global configuration mode to posts-peer 100.

```
call-diversion  100
   cfb ipaddr  211.111.111.1
dial-peer voice 0 pots
   diversion 100
```

3.10.4.6. display-name

To include a registered user ID in INVITE message, use **display-name** command. To reset to the default, use the **no** form of this

display-name *string* {non-quoted|<cr>}

no display-name

3.10.4.6.1. Syntax

Keyword / Argument	Description
string	Add a registered user to from field of INVITE - non-quoted: output the same setup value as input - <cr>: Output the setup value “string” form

3.10.4.6.2. Command Default

No default behavior or value.

3.10.4.6.3. Command Modes

Dial peer configuration

3.10.4.6.4. Usage Guidelines

To verify the use id, which has been used for authentication, with actual number in INVITE message in a particular proxy, use this command.

The user id used for authentication represented in From field of INVITE message. Also the id can be represented in To field by using to-display-name command.

The AddPac VoiceFinder Gateway Series basically use E.164 for registration try and authentication. To use the value, which is entered directly, instead of E.164 used for authentication, use user-name command from pots-peer command first for the configuration?

3.10.4.6.5. Examples

.The following example registers the user id as ‘addpac_user’ instead of E.164 in pots-peer 0 and delivers INVITE message including the user id.

```
dial-peer voice 0 pots
    user-name addpac_user
    display-name addpac_user
```

3.10.4.7. dtmf-relay

To specify how an H.323 or Session Initiation Protocol (SIP) gateway relays dual tone multi-frequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** command in dial peer voice configuration mode. To remove all signaling options and send the DTMF tones as part of the audio stream, use the **no** form of this command.

dtmf relay { h245-alphanumeric }

no dtmf relay

3.10.4.7.1. Syntax

Keyword / Argument	Description
h245-alphanumeric	(Optional) Forwards DTMF tones by using the H.245“alphanumeric” User Input Indication method. Supports tones from 0 to 9, *, #, and from A to D.

3.10.4.7.2. Command Default

No default behavior or values

3.10.4.7.3. Command Modes

Dial peer configuration

3.10.4.7.4. Usage Guidelines

DTMF is the tone generated when you press a button 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 either a standard H.323 out-of-band method.

This command determines the outgoing format of relayed DTMF tone, so the gateway takes the format automatically.

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 codec, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth codec may have trouble accessing automated DTMF-based systems, such as voice mail, menu-based Automatic Call

Distributor (ACD) systems, and automated banking systems.

3.10.4.7.5. Examples

The following example configures DTMF relay with the **cisco-rtp** keyword when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
  dtmf-relay h245-alphanumeric
```

The following example configures DTMF relay with the **cisco-rtp** and **h245-signal** keywords when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
  no dtmf-relay
```

3.10.4.8. forward-digits

To specify which digits to forward for voice calls, use the **forward-digits** command in dial peer configuration mode. To specify that any digits not matching the destination-pattern are not to be forwarded, use the **no** form of this command. To reset to the default, use the **default** form of this command.

forward-digits { from | last } *number*

no forward-digits

3.10.4.8.1. Syntax

Keyword / Argument	Description
from	Forwards all the digits of the called party number, which are specified by *number* .
last	Forward the digits in a number which are counted from the last digit.
number	The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. Range is 0 to 32. Setting the value to 0 is equivalent to entering the **no forward-digits** command.

3.10.4.8.2. Command Default

Dialed digits not matching the destination pattern are forwarded

3.10.4.8.3. Command Modes

Dial peer configuration (POTS peer)

3.10.4.8.4. Usage Guidelines

This command applies only to POTS dial peers. This command specifies a number of digits for relaying the last part of the called party number of the inbound call to the called party number of the outbound. The destination pattern includes both explicit digits and wildcards if present. Use the **default** form of this command if a no default digit-forwarding scheme was entered previously and you wish to restore the default.

3.10.4.8.5. Examples

The following example shows that the outbound call is POTS peer 10 and the called party number of the inbound is 100123456789, the number 123456789 is forwarded because forward-digit is not set.

```
dial-peer voice 10 pots
  destination-pattern 100...
```

When **forward-digit from** is set in the following example, the first seven of the digits, 456789 in the destination pattern of a POTS dial peer are forwarded:

```
  forward-digit from 7
```

When **forward-digit from** is set in the following example, all the digits 100123456789 are forwarded:

```
  forward-digit from 1
```

If **forward-digit from** is set in the following example, none of the digit is forwarded:

```
  forward-digit from 99
```

If **forward-digit from** is set in the following example, the last 4 of the digits, in other words, 6789 are forwarded.

```
  forward-digit last  4
```

If **forward-digit from** is set in the following example, none of the digit is forwarded:

```
  forward-digit last  0
```

If **forward-digit from** is set in the following example, all of the digits 100123456789 are forwarded

```
  forward-digit last  99
```

3.10.4.9. huntstop

To disable all dial-peer hunting if a call fails when using hunt groups, use the **huntstop** command in dial peer configuration mode. To enable dial-peer hunting, use the **no** form of this command.

huntstop

no huntstop

3.10.4.9.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.4.9.2. Command Default

Enabled hunting at default

3.10.4.9.3. Command Modes

Dial peer configuration

3.10.4.9.4. Usage Guidelines

When you choose the outbound dial peer for the inbound, more than one hunting group is formed. Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

3.10.4.9.5. Examples

The following example shows how to stop hunting on voip peer 110.

```
dial-peer voice 110 voip
  huntstop
```

3.10.4.10. port

To associate a dial peer with a specific voice port, use the **port** command in dial peer configuration mode. To cancel this association, use the **no** form of this command.

port *port*

no port

3.10.4.10.1. Syntax

Keyword / Argument	Description
port	Voice port number. Valid entries are 0 and 7.

3.10.4.10.2. Command Default

No default behavior or value.

3.10.4.10.3. Command Modes

Dial peer configuration

3.10.4.10.4. Usage Guidelines

This command is used for calls that come from a telephony interface to select an incoming dial peer and for calls that come from the VoIP network to match a port with the selected outgoing dial peer. This command applies only to POTS peers.

This command chooses the incoming dial-peer for the incoming call to the telephone interface and matches with the chosen port number of the outbound dial peer with the incoming call from VoIP network.

3.10.4.10.5. Examples

The following example associates the dial peer with the voice port:

```
dial-peer voice 10 pots
port 1
```

3.10.4.11. preference

To indicate the preferred order of a dial peer within a hunt group, use the **preference** command in dial peer configuration mode. To remove the preference, use the **no** form of this command.

preference *value*

no preference

3.10.4.11.1. Syntax

Keyword / Argument	Description
value	Integer from 0 to 10, where the lower the number, the higher the preference. Default is 0 (highest preference).

3.10.4.11.2. Command Default

0 (highest preference)

3.10.4.11.3. Command Modes

Dial peer configuration

3.10.4.11.4. Usage Guidelines

Setting the preference within a hunt group enables adjustment of the preference for a specific dial peer.

3.10.4.11.5. Examples

The following example shows the dial peer:

```
dial-peer voice 10 pots
  destination-pattern 5551234
  preference 3


dial-peer voice 11 pots
  destination-pattern 555....
  preference 0
```

The above settings describes that when the called party number 5551234 of the inbound, all the

configured dial peers in hunt algorithm are chosen, dial-peer hunt command chooses dial peer 11 as the first preference.

3.10.4.12. prefix

To specify the prefix of the dialed digits for a dial peer, use the **prefix** command in dial peer configuration mode. To disable this feature, use the **no** form of this command.

prefix *string*

no prefix

3.10.4.12.1. Syntax

Keyword / Argument	Description
string	Integers that represent the prefix of the telephone number associated with the specified dial peer. Valid values are 0 through 9 and a comma (,). Use a comma to include a pause in the prefix.

3.10.4.12.2. Command Default

Null String

3.10.4.12.3. Command Modes

Dial peer configuration

3.10.4.12.4. Usage Guidelines

Use this 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.

3.10.4.12.5. Examples

The following example specifies a prefix of 9 and then a pause for 1 second:

```
dial-peer voice 10 pots
prefix 9,
```

3.10.4.13. register

To configure a gateway to register or deregister a fully-qualified dial-peer E.164 address with a gatekeeper, use the **register e164** command in dial peer configuration mode. To deregister the E.164 address, use the **no** form of this command.

register e164

no register e164

3.10.4.13.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.4.13.2. Command Default

No default behavior or values

3.10.4.13.3. Command Modes

Dial peer configuration

3.10.4.13.4. Usage Guidelines

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 E.164 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, 55501212, and 4085550112 are fully qualified addresses; 408555.... is not. E.164 addresses are registered only for active interfaces, which are those that are not shut down. If an FXS port or its interface is shut down, the corresponding E.164 address is deregistered.

3.10.4.13.5. 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
```

3.10.4.14. session target

To designate a network-specific address to receive calls from a VoIP dial peer, use the **session target** command in dial peer configuration mode. To reset to the default, use the **no** form of this command.

session target *destination-address*

no session target

3.10.4.14.1. Syntax

Keyword / Argument	Description
destination-address	IP address of the dial peer to receive calls.

3.10.4.14.2. Command Default

No default behavior or value.

3.10.4.14.3. Command Modes

Dial peer configuration

3.10.4.14.4. Usage Guidelines

Use this command to specify a network-specific destination for a dial peer to receive calls from the current dial peer. You can select an option to define a network-specific address or domain name as a target, or you can select one of several methods to automatically determine the destination for calls from the current dial peer.

3.10.4.14.5. Examples

The following example creates a session target with IP address 211.238.1.1

```
dial-peer voice 10 voip
session-target 211.238.1.1
```

3.10.4.15. shutdown (Dial-Peer)

o change the administrative state of the selected dial peer from up to down, use the **shutdown** command in dial peer configuration mode. To change the administrative state of this dial peer from down to up, use the **no** form of this command.

shutdown

no shutdown

3.10.4.15.1. Syntax

Keyword / Argument	Description
This command has no arguments or keyword.	

3.10.4.15.2. Command Default

No shutdown

3.10.4.15.3. Command Modes

Dial peer configuration

3.10.4.15.4. Usage Guidelines

This command applies to both VoIP and POTS peer. When dial peer is shut down, you can not initiate a call to the peer.

3.10.4.15.5. Examples

The following example changes the administrative state of voice telephony (plain old telephone service [POTS]) dial peer 10 to down:

```
configure
dial-peer voice 10 pots
shutdown
```

3.10.4.16. sid

To enable Silence Insertion Description (SID) packet transmission in silence, when VAD is activated for a call in a particular dial-peer (VoIP peer), use sid command in dial-peer configuration mode. To disable this feature, use the no form of this command.

sid

no sid

3.10.4.16.1. Syntax

Keyword / Argument	Description
This command has no arguments or keyword.	

3.10.4.16.2. Command Default

Enabled

3.10.4.16.3. Command Modes

Dial peer configuration

3.10.4.16.4. Usage Guidelines

When VAD is enabled, silence does not transmit through network, but only voice. Actually, sid packets are transmitted during silence. If sid packet does not interworks together and creates a call problem or comfort noise generation is not needed, disable this feature.

3.10.4.16.5. Examples

The following example disables sid packet transmission:

```
dial-peer voice 10 voip
no sid
```

3.10.4.17. translate-outgoing

To apply a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg, use the **translate-outgoing command in** dial peer configuration mode. To disable the translation rule, use the **no** form of this command.

translate-outgoing { called-number | calling-number } *tag*

no translate-outgoing { called-number | calling-number }

3.10.4.17.1. Syntax

Keyword / Argument	Description
called-number	Apply translation rule to the outbound called party number.
calling-number	Apply translation rule to the outbound calling party number.
tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is 1 to 2147483647. There is no default value.

3.10.4.17.2. Command Default

No default behavior or values

3.10.4.17.3. Command Modes

Dial peer configuration

3.10.4.17.4. Usage Guidelines

This command applies to both POTS peers and VoIP peer. Use **translation-rule** command to configure the number translation rule set.

3.10.4.17.5. Examples

The following example creates a translation rule set 10, and then applies to the calling part number of dial-peer 200. so if the calling party number of the outbound call is 93450, then translates to 9563450.

```
translation-rule 10

        rule 0  9  956
```

```
        rule 1  8  878
dial-peer voice 200 voip
        translate-outgoing calling-number 10
```

3.10.4.18. vad

To enable voice activity detection (VAD) for the calls using a particular dial peer, use the **vad** command in dial peer configuration mode. To disable VAD, use the **no** form of this command.

vad

no vad

3.10.4.18.1. Syntax

Keyword / Argument	Description
This command has no arguments or keyword.	

3.10.4.18.2. Command Default

VAD is enabled

3.10.4.18.3. Command Modes

Dial peer configuration

3.10.4.18.4. Usage Guidelines

Use this command to enable voice activity detection. With VAD, voice data packets fall into three categories: speech, silence, and unknown. Speech and unknown packets are sent over the network; silence packets are discarded. The sound quality is slightly degraded with VAD, but the connection monopolizes much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously sent to the IP backbone.

3.10.4.18.5. Examples

The following example enables VAD

```
dial-peer voice 10 voip
vad
```

3.10.4.19. voice-class codec

To assign a previously configured codec selection preference list (codec voice class) to a Voice over IP (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*

3.10.4.19.1. Syntax

Keyword / Argument	Description
tag	Unique number assigned to the voice class. Range is from 1 to 65533. The *tag* number maps to the tag number created using the **voice class codec** global configuration command.

3.10.4.19.2. Command Default

Dial peers have no codec voice class assigned.

3.10.4.19.3. Command Modes

Dial peer configuration

3.10.4.19.4. Usage Guidelines

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.

3.10.4.19.5. 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
```

3.10.4.20. user-name

When to use SIP proxy, you need authentication, the SIP proxy requests for "WWW-Authenticate Digest"

At this time, user-id and user-name are configured same without setup.

To send user-id and user-name differently for the SIP proxy, use this command.

Use use-name command in dial-peer configuration mode.

If you want use user-name and user-id same, use the no form of this command.

user-name *string*

no user-name

3.10.4.20.1. Syntax

Keyword / Argument	Description
sting	An user ID that is registered to SIP Proxy.

3.10.4.20.2. Command Default

The user-name information configured in dial peer at default is set up same as destination-pattern <string>.

3.10.4.20.3. Command Modes

Dial peer

3.10.4.20.4. Usage Guidelines

Use user-name for registering and authenticating each dial peer. Think of each dial peer in UA(User Agent) concept.

If authentication of the AddPac VoiceFinder Gateway Series requires only one user-name, use sip-username in SIP-UA configuration mode rather than this command

The user-name is necessary information for REGISTER which requires authentication. If user-id and user-name are same, you do not have to configure.

3.10.4.20.5. Examples

The following example shows that the user-id is 025683848 and the user-name is addpac.

```
dial-peer voice 10 pots

destination-pattern 025683848

port 0/0

user-name addpac
```

3.10.4.21. user-password

When to use SIP proxy, you need authentication, the SIP proxy requests for "WWW-Authenticate Digest"

At this time, user-id and user-name are authenticated together.

Use **use-password** command in dial-peer configuration mode.

user-password *string*

no user-password

3.10.4.21.1. Syntax

Keyword / Argument	Description
tag	Enter the registered user-password to SIP proxy

3.10.4.21.2. Command Default

No default behavior or value

3.10.4.21.3. Command Modes

Dial peer configuration

3.10.4.21.4. Usage Guidelines

Use user-password for registering and authenticating each dial peer. Think of each dial peer in UA(User Agent) concept.

If the AddPac VoiceFinder Gateway Series need only one user-password for authentication, use sip-userpassword in SIP-UA configuration mode rather than this command

The user-password is necessary information for REGISTER which requires authentication. If authentication is not required, you do not have to use this command.

3.10.4.21.5. Examples

The following example shows that the user-id 025683848 and user-name requiring authentication is 025683848 and user password is addpac.

```
dial-peer voice 10 pots
```

```
destination-pattern 025683848

port 0/0

user-password addpac
```

3.10.4.22. CLID(Calling Line Identification)

To control the calling party number presentation field to Q.931 message in H.323. and use the **clid** command in dial peer configuration mode. To remove CLID controls, use the **no** form of this command.

clid *{network-number | restrict | strip}*

no clid

3.10.4.22.1. Syntax

Keyword / Argument	Description
network-number	Network number. Establishes the calling-party network number in the CLID for the gateway.
restrict	Restricts presentation of the caller ID in the CLID.
strip	Strips the calling-party number from the CLID.

- Reference : Details of Field for CLID

	Presentation Indicator	Screening Indicator	Calling party number
CLID disable	X	X	O
CLID network-number	Presentation allowed	Network provided	O
CLID restrict	Presentation restricted	User-provided, verified and failed	O
CLID strip	X	X	X

3.10.4.22.2. Command Default

disable

3.10.4.22.3. Command Modes

Dial peer configuration

3.10.4.22.4. Usage Guidelines

Presentation indicator and screening indicator are supported for the calling party information in Q.931 message of H.323

When a number type is “unknown” in Q.931 setup message in general, only calling party number is transmitted without presentation indicator, screening indicator field. The configured prefix or escape digit is transmitted to the receiver side of the trunk network. So a sender’s information is not known to the receiver’s side in this way and this feature is used to resolve this problem. Refer to ITU-T Recommendation Q.931 Calling party number for more details.

3.10.4.22.5. Examples

The following example shows that the user-id 025683848 and user-name requiring authentication is 025683848 and user password is addpac.

```
dial-peer voice 10 pots
destination-pattern 025683848
port 0/0
user-password addpac
```

3.10.4.23. call-wafting

To enable call waiting, use the **call-waiting** command in interface configuration mode. To disable call waiting, use the **no** form of this command.

call-waiting

no call-waiting

3.10.4.23.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.4.23.2. Command Default

Disabled

3.10.4.23.3. Command Modes

Dial peer configuration

3.10.4.23.4. Usage Guidelines

Alerts the phone user to incoming call while the phone user is on another call. Call waiting provides an audio alert and displays incoming call information (visual alert) on the phone screen. When the user press hook-flash-button of the telephone, the previous is place in hold state and connects to a new call.

If you press hook-flash button one more time, the call returns to the previous user. The feature of switch-to-pstn-on-call and switch-to-voip-on-call are to be disabled when to use this feature in the VoiceFinder Gateway Series with PSTN back-up port-AP200, AP1000 and AP160.
This feature is supported by SIP only.

3.10.4.23.5. Examples

The following example shows how to configure call waiting:

```
dial-peer voice 10 pots
call-wafting
```

```
call-wafting
```

3.10.4.24. out-barred-group

This command limits the outbound called party number to be transmitted through pots peer or voip only when one matching number is found in the list of pattern group

If you do not use this feature, use the no form of this command.

out-barred-group *tag*

no out-barred-group

3.10.4.24.1. Syntax

Keyword / Argument	Description
This command has no arguments or key words.	

3.10.4.24.2. Command Default

disable

3.10.4.24.3. Command Modes

Dial peer configuration

3.10.4.24.4. Usage Guidelines

One way to limit the outbound called party number is to generate new dial-peer and use shutdown command. Another way of doing this is using dialpattern-group for a called party number.

3.10.4.24.5. Examples

The following example shows how to limit an outbound call for a cell phone:

```
dial-peer voice 1000 voip
destination-pattern .T
session target ras
dtmf-relay h245-alphanumeric
out-barred-group 0
!
dialpattern-group 0
```

```
pattern 0 01[16789].T
```

3.10.5. sip-ua (SIP User Agent) Configuration Commands

3.10.5.1. call-transfer-mode

During SIP server connection and VoIP call, this feature transfers a call connected to one party presently to another. To return to default, use the no form of this command.

call-transfer-mode attended

no call-transfer-mode

3.10.5.1.1. Syntax

Keyword / Argument	Description
basic	Same as no call-transfer-mode at default
attended	Change transfer method to attended mode

3.10.5.1.2. Command Default

call-transfer-mode basic

3.10.5.1.3. Command Modes

SIP configuration

3.10.5.1.4. Usage Guidelines

If you want to call-transfer while you are in telephone conversation on one VoIP call, press hook-flash button on the phone.

At this time, GW2 (Transferee) user is place on call hold. GW1 (transferor) listens to a dial tone, then dial the telephone number of GW3 (target). The user listens to trying tone at this time. The operation method afterwards can be different for each call-transfer mode.

First, at the basic mode, as the call is transferred to GW3 user, GW1 user's role is finished. GW1 user hook-on the telephone.

Differently, at the attended mode, after GW1 is transferred to GW and when GW3 hook-off, GW1 and GW3 can have telephone conversation. Of course, it would be still same after GW1 hook-on after GW1 is transferred to GW3.

3.10.5.1.5. Examples

The following example changes call-transfer-mode from SIP configuration.

(* basic mode at default)

```
sip-ua
    call-transfer-mode attended
```

3.10.5.2. conference-server

Conference call allows having a telephone conversation with more than 3 phone users on one call. The AddPac VoiceFinder Series does not support conference call by themselves, but with a conference server.

To interoperate with conference server, use **conference-server**. To disable this configuration, use the no form of this command.

conference-server *STRING* <0-65535>

no conference-server

3.10.5.2.1. Syntax

Keyword / Argument	Description
string	Enter ID(name) to SIP Conference
<0-65535>	Specify a port to communicate with SIP conference server

3.10.5.2.2. Command Default

No default behavior or value

3.10.5.2.3. Command Modes

SIP configuration

3.10.5.2.4. Usage Guidelines

To use conference call feature, configure dial-peer call-transfer h and dial-peer-call-hold h first.

Use hook-flash-button on the telephone for this feature.

Different from call-transfer, you need to press hook-flash button twice for conference call. Basically, it takes 500 ms (0.5 sec) to recognize hook-flash button from the AddPac gateway

If you think 500ms (0.5 sec) is too short, you can change hook-flash detect timeout value when hook-flash duration time of PBX is more than 500ms.

3.10.5.2.5. Examples

The following example configures conference server from SIP configuration:

```
sip-ua
   conference-server mcu.addpac.com 5060
```

3.10.5.3. enable-ping

Use this command to enable-ping. To disable this feature, use the no form of this command.

enable-ping *ENTITY-NAME*

no enable-ping

3.10.5.3.1. Syntax

Keyword / Argument	Description
entity-name	Firewall/NAT Server-Entity

3.10.5.3.2. Command Default

No default behavior or value.

3.10.5.3.3. Command Modes

SIP configuration

3.10.5.3.4. Usage Guidelines

When the gateway operates under PAT/NAT or firewall environment, incoming VoIP call can be problematic.

The cause of the problem is due the gateway, located in the private environment, exchanges packets with the public network outside through PAT/NAT.

However, the network equipment located in the public network does not recognize IP and port of the gateway, so incoming VoIP call can not be processed.

For incoming and outgoing VoIP calls to be possible in the private environment, set up the public IP address of PAT/NAT server to the gateway and configure PAT/NAT server with the static mapping of VoIP port information of the gateway.

To take complement measures of this problem, Nortel's enable-ping is used for incoming outgoing of normal communication with Nortel's SIP proxy.

*This feature is Nortel-specific.

3.10.5.3.5. Examples

The following example configures enable-ping message to be sent out every 45 seconds

```
sip-ua
enable-ping 211.110.10.1
timeout tsipping 45
```

3.10.5.4. media-channel

Media channels are RTP/RTPC path for VoIP communication. In other words, media channel can be divided by transmit channel and receive channel for VoIP communication.

This feature configures a point of time to enable transmit channel.

Use this command to set up and enable media channel type.

media-channel {early| late}

no media-channel

3.10.5.4.1. Syntax

Keyword / Argument	Description
Early	Opens transmit channel when 18X SDP is received after INVITE is transmitted
late	Does not open transmit channel, even 18X SDP is received after INVITE is transmitted, opens it only when 200 OK is received

3.10.5.4.2. Command Default

No default behavior or value

3.10.5.4.3. Command Modes

SIP configuration

3.10.5.4.4. Usage Guidelines

When the gateway operates under PAT/NAT environment, VoIP equipment from the other side provide ring back tone(Color Ring included) by inband (RTP), ringback tone fails to reach the gateway under the private environment . The cause is same as enable-ping.

To take compliment measures of this problem, the gateway under the private environment transmits INVITE, transmit channel opens when to receive 18X SDP, a port table is created in PAT/NAT server, inband right back tone can be heard as a result.

The first type is the default mode. Transmit cannel opens to only 183 progress SDP after INVITE is transmitted. Transmit channel does not open even 180 SDP is received.

The second type is early mode. Transmit channel opens when 18X SDP is received after INVITE is transmitted.

18X SDP means whether SDP presents in 180 ring message or in 183 progress message.

The third type is late mode. Transmit channel does not open even when 18X SDP is received after INVITE is transmitted.

3.10.5.4.5. Examples

The following example sets up media-channel early in SIP-UA configuration mode.

```
sip-ua
medial-channel early
```

3.10.5.5. min-se

To inform the proxy server that the gateway can support session timer, use min-se. To reset to the default, use the **no** form of this command.

min-se *sec*

no min-se

3.10.5.5.1. Syntax

Keyword / Argument	Description
sec	Sets up a frequency of session timer. Ranges from 60 to 86,400 seconds

3.10.5.5.2. Command Default

1,800 seconds

3.10.5.5.3. Command Modes

SIP-UA Configuration Mode

3.10.5.5.4. Usage Guidelines

Most of SIP gateways use UDP for signaling and voice packet transmission as well the AddPac VoiceFinder Gateway Series.

Generally, SIP proxy server is used accounting and call routing of SIP telephone network.

Some unexpected failure may take a place in internet or power after session is formed between the terminals (SIP-UA). SIP sever can send re-invite messages periodically to check a call status SIP-UA and proxy server.

Session timer forms session and sends re-invite from the terminal (SIP-UA) to proxy, proxy to the terminal (SIP-UA). If no call-clear (bye) message is sent and no re-invite is received in a certain time period, the proxy determines something wrong with the equipment and send BYE (call-clear) messages to both ends.

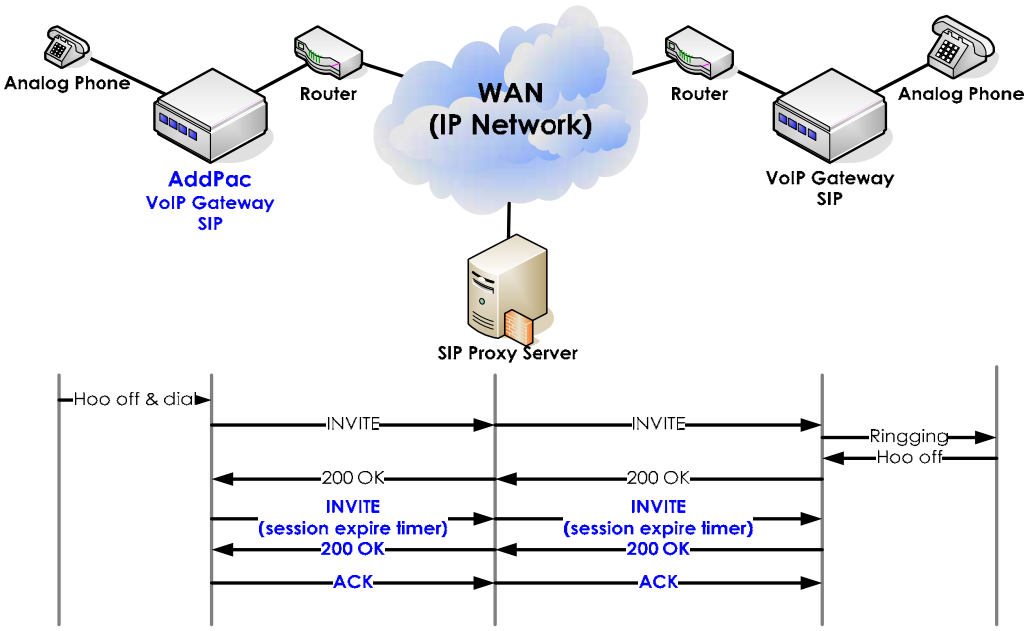


Figure 3.12 Basic SIP Network Diagram

3.10.5.5.5. Examples

The following example sets session timer to 180. When session timer is set more than 180, it belongs to the gateway settings. The timer value is sent by INVITE message. If the setting between sever and UA is different, the server notifies 422 message as a timer value that the server intends to use. UA takes this message and operate at the setting value. Therefore, even a timer value is determined by this command, when 422 message is received from the server, this value takes a priority.

```
sip-ua
  min-se 180
```

3.10.5.6. register

To register a gateway to SIP Proxy server, use register command in SIP-UA configuration mode.

To un register the gateway from the proxy server, use the no form of this command.

register { e164 | gateway}

no register

3.10.5.6.1. Syntax

Keyword / Argument	Description
e164	Tries registration by using an assigned number to each dial peer. Includes e164 or user-id information in request URI of REGISTER message To: <sip:**2000@proxy.com**>;tag=4240c200a4
gateway	Tries registration to the gateway without user-id. Use the gateway a trunk-gateway or media gateway Request URI does not include e164 or user-id information. To: <sip:**proxy.com**>;tag=4240c200a4

3.10.5.6.2. Command Default

Disabled

3.10.5.6.3. Command Modes

SIP-UA Configuration Mode

3.10.5.6.4. Usage Guidelines

Generally registration is required for using SIP proxy to make VoIP calls. The concept of registration if same as gateway register of H.323.

Register e164 tries authentication for each user-id (destination [telephone number])

If 10 dial peers are set up, register e164 tries 10 times.

If sip-username and sip-password are set up in SIP-UA configuration mode, each dial peer tries with the same username and password for authentication.

If there is no configuration of sip-username and sip-password and each dial peer is configured with the same command, then each user name and password configured with each peer tries for authentication.

To register the gateway to trunk like trunk-gateway or media gateway, use register gateway

command. The most of these equipment are operated by network service provide, they do not authenticate user-id.

3.10.5.6.5. Examples

The following commands peform registration by each user-id to SIP proxy server:

```
sip-ua
  register e164
```

3.10.5.7. rel1xx

To enable all Session Initiation Protocol (SIP) provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint, use the **rel1xx** command in SIP configuration mode. To reset to the default, use the **no** form of this command.

rel1xx {require| supported}

no rel1xx

3.10.5.7.1. Syntax

Keyword / Argument	Description
supported	Supports reliable provisional responses.
require	Requires reliable provisional responses.

3.10.5.7.2. Command Default

rel1xx supported

3.10.5.7.3. Command Modes

SIP configuration

3.10.5.7.4. Usage Guidelines

The basic call proceeding is listed as to follow:

1. Send 1xxresponse for the invite
2. Then send ack. If you want to set up a policy to receive ack, the function can be used

The ack is supported for 1xx response but it is not for 100 trying

3.10.5.7.5. Examples

The following example sets to Reliable Provisional Response require.

```
sip-ua
rel1xx require
```

3.10.5.8. remove-all-binding

To enable this feature use remove-all-binding. To disable, use the no form of this command.

remove-all-binding enable

no remove-all-binding

3.10.5.8.1. Syntax

Keyword / Argument	Description
enable	remove-all-binding 기능을 enable 시킵니다.

3.10.5.8.2. Command Default

Disabled

3.10.5.8.3. Command Modes

SIP configuration

3.10.5.8.4. Usage Guidelines

Sometimes SIP-UA fails to send registration cancel message to proxy server and reboots due to administrator's mistake and other reasons, while it operates normally. If SIP-UA still fails to process the registration cancel message after the reboot, the message can be verified in a particular site of the proxy server.

To prepare for a case like this, configure the settings to send a message to delete all the registration information before the registration message is sent and after rebooting. retry registration by sending the registration message again.

3.10.5.8.5. Examples

The following example enables remove-all binding command

```
sip-ua
remove-all-binding enable
```

3.10.5.9. retrycounter

To set a counter for retransmit when no reply is found for all type of SIP request message

(register, invite, 200 OK and others) to be transmitted, use **retry-counter**. To set retry-counter to default, use the no form of this command.

retry-counter *counter*

no retry-counter

3.10.5.9.1. Syntax

Keyword / Argument	Description
Counter	Sets counter. Ranges from 3 to 10 times and adjustment is possible

3.10.5.9.2. Command Default

10 times

3.10.5.9.3. Command Modes

SIP-UA Configuration Mode

3.10.5.9.4. Usage Guidelines

Retry transmission frequency and timeout parameter are correlated together. timeout * retry count for each message is the expected maximum time until a call drops by no reply of peer side equipment.

3.10.5.9.5. Examples

The following example sets retry counter value to 3 times or to the default (10).

```
retry-counter  3
no retry-counter
```

3.10.5.10. remote-party-id

To apply From field to user-name instead of E.164 defined by destination-pattern when to send

INVITE message, use the remote-party-id command in SIPUA configuration mode.

3.10.5.10.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.5.10.2. Command Default

disabled

3.10.5.10.3. Command Modes

SIP configuration

3.10.5.10.4. Usage Guidelines

Caller-id represents user-id of From field.
However, when to interoperate with Nortel's IP phone and trunk gateway, the outgoing number is processed from SIP proxy to digit.

Because SIP proxy is not able to process From fields, remote-party-id as an option field is used to generate/detect (display) the caller-id.

This feature is Nortel-specific and used to interoperate with VoIP equipment.

The AddPac VoiceFinder Gateway Series generates remote-party-id field from FXO, E&M, and E1/T1 type except the module of FXS type.

An attention is required for the fact that remote-party-is is not generated even FXS type of module is enabled.

On the other hand, all types of modules of the VoiceFinder Gateway recognize caller-id by referencing the field when the gateways receive INVITE message and remote-party-id presents.

3.10.5.10.5. Examples

The following example generates remote-party-id field in INVITE message.

```
sip-ua
remote-party-id
```

3.10.5.11. response

These feature responses to 180 ringing/ 183 progress when they are received to SIP protocol

platform. To reset and return to the initial setting, use the no form of this command.

response {alert| progress| default} {with-sdp|without-sdp|<cr>}

no response

3.10.5.11.1. Syntax

Keyword / Argument	Description
alert	180 ringing - with-sdp: includes and sdp - without-sdp: doe not includes sdp and delivers - <cr>: 180 at default same as without-sdp
progress	Set up to response 183 Progress 응답하도록 설정합니다. - with-sdp: includes and sdp - without-sdp: doe not includes sdp and delivers - <cr>: 180 at default same as without-sdp

3.10.5.11.2. Command Default

Disabled

3.10.5.11.3. Command Modes

SIP configuration

3.10.5.11.4. Usage Guidelines

Generally, SIP protocol does not include SDP but 183 progress does. Recently, this feature is added to support SIP-Server/TrunkGW. Therefore, this feature is applied depending on environmental settings.

3.10.5.11.5. Examples

The following example sends the response including SDP when 180 is sent after INVITE message is received.

```
sip-ua
response alert with-sdp
```

3.10.5.12. route-by-auxiliary

To enable route-by-auxiliary, use this command. To disable this feature, use no form of this

command.

route-by-auxiliary

no route-by-auxiliary

3.10.5.12.1. Syntax

Keyword / Argument	Description
This command has not arguments or keywords.	

3.10.5.12.2. Command Default

Disables

3.10.5.12.3. Command Modes

SIP configuration

3.10.5.12.4. Usage Guidelines

In other words, when the user id of request URI is 1000@X.X.X.X, a call arrives to a designated port with destination-pattern 1000.
Enable this feature if you want to route by referencing the user-id in the field of the initial INVITE.

3.10.5.12.5. Examples

The following example sets up route-by-auxiliary.

```
sip-ua
route-by-auxiliary
```

3.10.5.13. set-local-domain

To set URL of gateway SIP message to a specific domain and enable a function to set to local

domain of SIP URL, use this command.

To disable this feature, use the no form of this command.

set-local-domain {domain name}

no set-local-domain

3.10.5.13.1. Syntax

Keyword / Argument	Description
string	Enter a domain name to be applied to SIP URL

3.10.5.13.2. Command Default

Disabled

3.10.5.13.3. Command Modes

SIP configuration

3.10.5.13.4. Usage Guidelines

When to use URL as a domain, it must be registered to SIP-server for generating URL as a domain to other SIP messages. If SIP-server is not registered to SIP server, then URL is generated as IP address for communicating with peer to peer.

3.10.5.13.5. Examples

The following example sets the domain name, which is used for SIP message, to sip.addpac.com.

```
sip-ua
set-local-domain sip.addpac.com
```

3.10.5.14. set-local-host

To generate sip url as a host name of the gateway.

To disable this feature, use the no form of this command.

set-local-host

no set-local-host

3.10.5.14.1. Syntax

Keyword / Argument	Description
This command has no arguments or key words.	

3.10.5.14.2. Command Default

Disabled

3.10.5.14.3. Command Modes

SIP configuration

3.10.5.14.4. Usage Guidelines

Generally the following URL uses IP addresses of the gateway:

REGISTER sip:172.17.201.15 SIP/2.0

Via: SIP/2.0/UDP 172.17.201.51:5060;branch=z9hG4bK3064de00a41

From: <sip:9000@172.17.201.15>;tag=3064de00a4

To: sip:9000@172.17.201.15

Call-ID: 30150c64-f149-de63-8000-0002a400380b@172.17.201.51

CSeq: 1 REGISTER

Date: Sat, 11 Mar 2023 05:44:16 GMT

User-Agent: AddPac SIP Gateway

Contact: <sip:9000@172.17.201.51>;expires=60

Expires: 60

Content-Length: 0

Max-Forwards: 70

To enable set-local-hose, use the following URL as host-name.

(*To change host name, use hostname command in global configuration mode.)

REGISTER sip:172.17.201.15 SIP/2.0

Via: SIP/2.0/UDP 172.17.201.51:5060;branch=z9hG4bK3064de00a41

From: <sip:9000@AP100-hostname>;tag=3064de00a4

To: sip:9000@AP100-hostname

Call-ID: 30150c64-f149-de63-8000-0002a400380b@172.17.201.51

CSeq: 1 REGISTER

Date: Sat, 11 Mar 2023 05:44:16 GMT

User-Agent: AddPac SIP Gateway

Contact: <sip:9000@172.17.201.51>;expires=60

Expires: 60

Content-Length: 0

Max-Forwards: 70

3.10.5.14.5. Examples

The following example sets up to use hostname (AP100-hostname) which has been set to the gateway.

```
AP100-hostname(config)# sip-ua
AP100-hostname(config-sip-ua)# set-local-host
```

3.10.5.15. signaling-port

When a gateway wants to change UDP source port number for SIP signaling randomly, use signaling-port command. To set UDP5060 at default, use the no form of this command.

signaling-port *port-number*

no signaling-port

3.10.5.15.1. Syntax

Keyword / Argument	Description
port-number	Set up UDP listen port which is used for signaling

3.10.5.15.2. Command Default

UDP 5060

3.10.5.15.3. Command Modes

SIP-UA Configuration Mode

3.10.5.15.4. Usage Guidelines

To change SIP Signaling UDP listen port randomly, use this command.

As the outgoing call is changed to sip signaling UDP port number, the incoming call is changed to listen port. When the gateway is installed and operated in firewall network, only a particular port is allowed to use this command. SIP REGISTER message include this signaling port information, so the proxy server is able to recognize.

Pay attention avoid a conflict with the port which has been assigned with a specific feature or protocol already

3.10.5.15.5. Examples

The following example changes a signaling port to 5620(UDP).

```
sip-ua
  signaling-port 5620
```

3.10.5.16. force-forwarding

To enable SIP force-forwarding, use this command. To set to default, use the no form of this command.

force-forwarding response-code {403|404}

no force-forwarding response-code {403|404}

3.10.5.16.1. Syntax

Keyword / Argument	Description
{403|404}	Enables the feature - 403: When 403(Forbidden) is received set up for forwarding - 404: When 403404(Not Found) is received set up for forwarding

3.10.5.16.2. Command Default

Disabled

3.10.5.16.3. Command Modes

SIP configuration

3.10.5.16.4. Usage Guidelines

When SIP protocol is used for a call try and 403(Forbidden), 404(Not Found), the call terminated without condition.

If 403/404 is received for INVITE by using this command, then chooses next VoIP peer to set the call to be processed (The matching voip peer should be more than 2, so a normal operation can be carried out)

3.10.5.16.5. Examples

The following example configures voip call though voip-peer, when 403(forbidden is received).

```
sip-ua
force-forwarding response-code 403


dial-peer voice 1000 voip
destination-pattern T
```

```
session target sip-server
session protocol sip
preference 1

dial-peer voice 1001 voip
destination-pattern T
session target 211.110.11.1
session protocol sip
preference 2
```

3.10.5.17. sip-server

To configure a network address for the Session Initiation Protocol (SIP) server interface, use the **sip-server** command in SIP UA configuration mode. To remove a network address configured for SIP, use the **no** form of this command.

sip-server *ip-addr [port] [priority*

no sip-server *ip-addr*

3.10.5.17.1. Syntax

Keyword / Argument	Description
IP-address	Send SIP REGISTER Message to the specified address
Port	Designates UDP port of SIP Proxy server optionally. The default value is 5060.
Priority	Specifies many alternate SIP proxy servers in a priority. Ranges from 0~24, as the value becomes lower, higher the precedence. The default is 128.
SIP Server ID	Specifies IP of SIP server. This is a simple description which does not affect any SIP message field.

3.10.5.17.2. Command Default

No default behavior or value.

3.10.5.17.3. Command Modes

SIP-UA Configuration Mode

3.10.5.17.4. Usage Guidelines

SIP User Agent is registered SIP proxy server as RAS (Registration, Admission, and Status) server and receives number and accounting services. AP1000 gateway is able to assign more than one proxy server to the gateway. A list of proxy server can be displayed by **show sip** command. When more than one proxy server is specified, registration is tries by using REGISTER message to the proxy server in an order. Only one proxy server can be registered at same time. If receiving message with the proxy server is failed, the gateway retries registration to the proxy server in a priority order.

SIP Signaling port address based on SIP RFC 2543/3261 is 5060.

3.10.5.17.5. Examples

The following example shows the setup of proxy server assigned with 192.7.5.1 IP address.

```
sip-server 192.7.5.1
```

The following example sets a priority order o to the proxy server assigned with 192.7.5.1 IP address:

```
sip-server 192.7.5.1 6000 0
```

The following example sets priority order 0 and id 1 to the proxy server assigned with 192.7.5.1 IP address and port 6000.

```
sip-server 192.7.5.1 6000 0 1
```

3.10.5.18. sip-username

To register the gateway to SIP proxy server and to specify user name and user password for user authentication, use **sip-user name** command in SIP-UA configuration mode. To delete sip-username, use the no form of this command.

sip-username *string*

no sip-username *string*

3.10.5.18.1. Syntax

Keyword / Argument	Description
User name	Used for user authentication in REGISTER process

3.10.5.18.2. Command Default

No default value or behavior

3.10.5.18.3. Command Modes

SIP-UA Configuration Mode

3.10.5.18.4. Usage Guidelines

When to register and go through registration process, 401 unauthorized message is receive as a response to REGISTER. At this time, the gateway tries REGISTER with authentication by authorization key, which is generated by the assigned user name and password.

3.10.5.18.5. Examples

The following example sets the username to addpac:

```
sip-username addpac
```

3.10.5.19. sip-password

To register the gateway to SIP proxy server and to specify user name and user password for user authentication, use **sip-user password** command in SIP-UA configuration mode. To delete sip-password, use the no form of this command.

3.10.5.19.1. Syntax

Keyword / Argument	Description
password	Used for user authentication in REGISTER process

3.10.5.19.2. Command Default

No default behavior or value

3.10.5.19.3. Command Modes

SIP-UA Configuration Mode

3.10.5.19.4. Usage Guidelines

When to register and go through registration process, 401 unauthorized message is receive as a response to REGISTER. At this time, the gateway tries REGISTER with authentication by authorization key, which is generated by the assigned user name and password.

3.10.5.19.5. Examples

The following example sets to the password to addpac.

```
sip-password addpac
```

3.10.5.20. srv

To enable SRV DNS query, use this command. To reset to the default, use the **no** form of this

command

srv enable

no srv

3.10.5.20.1. Syntax

Keyword / Argument	Description
enable	Enable the feature

3.10.5.20.2. Command Default

Disabled

3.10.5.20.3. Command Modes

SIP-UA Configuration

3.10.5.20.4. Usage Guidelines

Register the following DNS namesever when to enable srv DNS query.

(config)# dnshost nameserver *IP-Address*

3.10.5.20.5. Examples

Configuring DNS SRV to be Enabled

Step	Commands	Description
1	# **config**	Change to APOS command configuration mode
2	(config)# **sip-ua**	Enter SIP UA configuration mode
3	(config-sip-ua)# **srv enable**	Set up DNS SRV
4	(config-sip-ua)# **sip-server voip.addpac.com**	Set SIP server IP to DNS server domain
5	(config-sip-ua)# **sip user-name Addpac**	Register SIP user name
6	(config-sip-ua)# **sip password 1234**	Register SIP password
7	(config-sip-ua)# **register e164**	Register E.164SIP Serve
8	(config-sip-ua)# **exit**	Exit SIP UA configuration mode
9	(config)# **exit**	Exit configuration mode

Disbaling SIP srv

Step	Commands	Description
1	(config-sip-ua)# **no srv**	Disable dns srv

3.10.5.21. timeout

To set up SIP signaling related timeout parameter, use timeout command SIP-UA configuration mode. To reset to default, use the no form of this command.

3.10.5.21.1. Syntax

Keyword / Argument	**Description**
tretry	Timeout for response message to SIP Request message (such as REGISTER, INVITE)
treg	REGISTER message retransmit cycle when reject is received for REGISTER message
tregtry	REGISTER message retransmission cycle after it is registered
texpires	time out starting from INVITE transmission and to receiving 200 OK (connect)

3.10.5.21.2. Command Default

tretry : 500ms

treg : 60 sec

tregtry : 20 sec

texpires : 180 sec

3.10.5.21.3. Command Modes

SIP-UA Configuration Mode

3.10.5.21.4. Usage Guidelines

Timeout or retransmit cycle is organically related to retry-counter

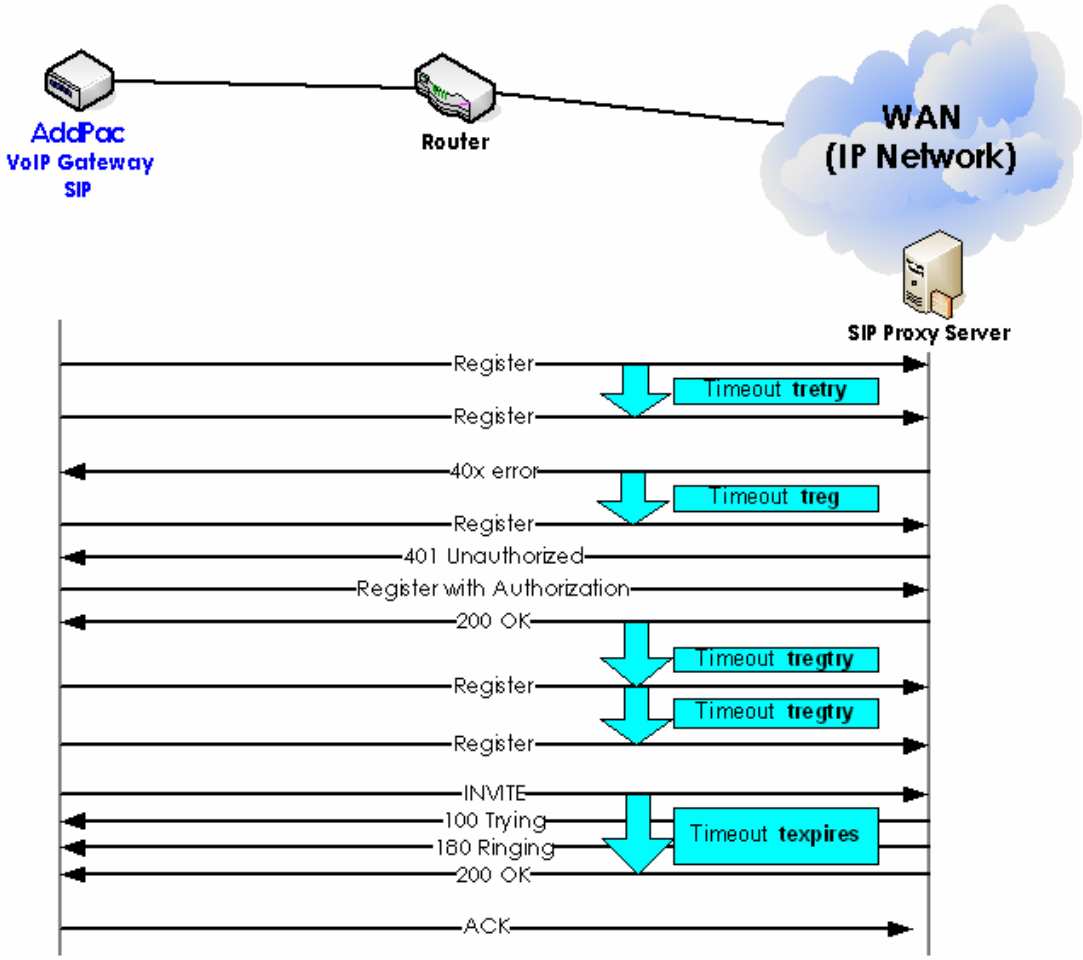


Figure 3.13 SIP Timer

3.10.5.21.5. Examples

The following example sets each parameter to 100ms, 30, 40, 50 seconds:

```
tretry 100

treg 30

tregtry 40

texpires 50
```

3.10.5.22. user-register

To register the gateway to SIP proxy server bye164 and to REGISTER request URI by username instead of e164, use user-register command with register e164 in SIP-UA configuration mode.

user-register

no user-register

3.10.5.22.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.5.22.2. Command Default

Disables

3.10.5.22.3. Command Modes

SIP-UA Configuration Mode

3.10.5.22.4. Usage Guidelines

When user-register command is set and the gateway is registered to e164, request URI replaces e164 with user name. At this time, the user name is not same as the one, which is set in SIP-UA configuration mode, uses an assigned value to POTS peer, If the user-name is not set in POTS peer, the gateway uses the value (e164) which is set to destination pattern at default.

```
Example 1) Registering SIP-UA to e164 (no user-register)
!
dnshost nameserver 172.17.1.254
dial-peer voice 0 pots
        destination-pattern 2000
        port 0/0
        user-name jinyoung
!
```

```
dial-peer voice 1 pots
        destination-pattern 2001
        port 0/1
        !
sip-ua
        sip-username addpac
        sip-password 1234
        sip-server proxy.com
        register e164
```

REGISTER message in response to the configuration

```
REGISTER sip:1.1.1.2 SIP/2.0
  Via: SIP/2.0/UDP 172.19.1.200:5060;branch=z9hG4bK4240c200a4567
  From: <sip:2000@proxy.com>;tag=4240c200a4
  To: sip:2000@proxy.com
  Call-ID: 42da4a40-6a77-c297-8000-0002a400380d@172.19.1.200
  CSeq: 567 REGISTER
  Date: Mon, 08 Mar 2004 14:48:23 GMT
  User-Agent: AddPac SIP Gateway
  Contact: sip:2000@172.19.1.200
  Expires: 60
  Content-Length: 0
  Max-Forwards: 70register e164

REGISTER sip:1.1.1.2 SIP/2.0
  Via: SIP/2.0/UDP 172.19.1.200:5060;branch=z2hG4bK4240c200a4567
  From: <sip:2001@proxy.com>;tag=4240c200a4
  To: sip:2001@proxy.com
  Call-ID: 42da4a40-6a77-c297-8000-0002a400380d@172.19.1.200
  CSeq: 568 REGISTER
  Date: Mon, 08 Mar 2004 14:48:23 GMT
  User-Agent: AddPac SIP Gateway
  Contact: sip:2001@172.19.1.200
  Expires: 60
  Content-Length: 0
```

```
  Max-Forwards: 70register e164


Example 2) Registering e164 SIP-UA (user-register)

!

dnshost nameserver 172.17.1.254

dial-peer voice 0 pots

        destination-pattern 2000

        port 0/0

        user-name jinyoung


dial-peer voice 1 pots

        destination-pattern 2001

        port 0/1

        !

sip-ua

        sip-username addpac

        sip-password 1234

        sip-server proxy.com

        register e164

        user-register


Register message

REGISTER sip:proxy.com SIP/2.0

  Via: SIP/2.0/UDP 172.19.1.200:5060;branch=z9hG4bK4240c200a4572

  From: sip:addpac@ proxy.com;tag=4240c200a4

  To: sip:addpac@ proxy.com

  Call-ID: 42da4a40-6a77-c297-8000-0002a400380d@172.19.1.200

  CSeq: 572 REGISTER

  Date: Mon, 08 Mar 2004 16:24:20 GMT

  User-Agent: AddPac SIP Gateway

  Contact: sip:addpac@172.19.1.200

  Expires: 60

  Content-Length: 0

  Max-Forwards: 70
```

```
REGISTER sip:proxy.com SIP/2.0
  Via: SIP/2.0/UDP 172.19.1.200:5060;branch=z1hG4bK4240c200a4572
  From: sip:2001@ proxy.com;tag=4140c200a4
  To: sip:2001@ proxy.com
  Call-ID: 42da4a40-6a77-c297-8000-0002a400380d@172.19.1.200
  CSeq: 572 REGISTER
  Date: Mon, 08 Mar 2004 16:24:20 GMT
  User-Agent: AddPac SIP Gateway
  Contact: sip:2001@172.19.1.200
  Expires: 60
  Content-Length: 0
  Max-Forwards: 70
```

3.10.5.22.5. Examples

The following example enables user-register.

```
user-register
```

3.10.5.23. hook-flash-info-ignore

To enable hook-flash not to send SIP info message to SIP server. To disable this feature use the no form of this command:

hook-flash-info-ignore

no hook-flash-info-ignore

3.10.5.23.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.5.23.2. Command Default

Disabled

3.10.5.23.3. Command Modes

SIP-UA Configuration Mode

3.10.5.23.4. Usage Guidelines

SIP-UA Configuration Mode

3.10.5.23.5. Examples

The following example enables hook-flash-info-ignore.

```
hook-flash-info-ignore
```

3.10.6. Gateway, Voice Service, Voice Class and Rule Configuration Commands

3.10.6.1. announcement

To enable announcement, use this command. To disable this feature, use the no form of this command.

announcement

no announcement

3.10.6.1.1. Syntax

Keyword / Argument	**Description**
This command has no arguments or keywords.	

3.10.6.1.2. Command Default

Disabled

3.10.6.1.3. Command Modes

Voice service voip configuration

3.10.6.1.4. Usage Guidelines

When announcement is enabled, you can hear a voice announcement of password entry and PSTN reroute, abnormal call termination. Some gateway product models are not supported with this version depending on version of operational system.

3.10.6.1.5. Examples

The following example enables announcement.

```
voice service voip
  announcement
```

3.10.6.2. busyout monitor

To place a voice port into the busyout monitor state, enter the **busyout monitor** command in voice-port configuration mode. To remove the busyout monitor state from the voice port, use the **no** form of this command.

busyout monitor {callagent| gatekeeper| sip-server| voip-interface}

no busyout monitor

3.10.6.2.1. Syntax

Keyword / Argument	Description
callagent	Monitors the binding state with MGC
gatekeeper	Monitors the binding state with gatekeeper
sip-server	Monitors the binding state with proxy
voip-interface	Monitors the up/down state of VoIP interface

3.10.6.2.2. Command Default

The voice port does not monitor any interfaces.

3.10.6.2.3. Command Modes

Voice Service voip configuration

3.10.6.2.4. Usage Guidelines

The gateway provides busyout monitor service for the configuration of which PSTN transfers automatically, when a call can not be delivered due to a network failure or an abnormal state of communication with Gatekeeper/SIP-Proxy/MGC.

The related setting of callagent/gatekeer/sip-server/voip-interface can be configured repeatedly

3.10.6.2.5. Examples

The following example monitors the binding state of the voip-interface with the gatekeeper.

```
voice service voip
busyout monitor gatekeeper
busyout monitor voip-interface
```

3.10.6.3. codec preference

To place a voice port into the busyout monitor state, enter the **busyout monitor** command in voice-port configuration mode. To remove the busyout monitor state from the voice port, use the **no** form of this command.

busyout monitor {callagent| gatekeeper| sip-server| voip-interface}

no busyout monitor

3.10.6.3.1. Syntax

Keyword / Argument	Description
callagent	Monitors the binding state with MGC
gatekeeper	Monitors the binding state with gatekeeper
sip-server	Monitors the binding state with proxy
voip-interface	Monitors the up/down state of VoIP interface

3.10.6.3.2. Command Default

The voice port does not monitor any interfaces.

3.10.6.3.3. Command Modes

Voice Service voip configuration

3.10.6.3.4. Usage Guidelines

The gateway provides busyout monitor service for the configuration of which PSTN transfers automatically, when a call can not be delivered due to a network failure or an abnormal state of communication with Gatekeeper/SIP-Proxy/MGC.

The related setting of callagent/gatekeer/sip-server/voip-interface can be configured repeatedly

3.10.6.3.5. Examples

The following example monitors the binding state of the voip-interface with the gatekeeper.

```
voice service voip
busyout monitor gatekeeper
busyout monitor voip-interface
```

3.10.6.4. counter

To set a value of counter parameter for VoIP, use **counter** command in voice service configuration. To return to default, use the no form of this command.

counter { cras } *value*

no counter { cras }

3.10.6.4.1. Syntax

Keyword / Argument	Description
cras *value*	RAS message retransmit counter for gatekeeper. Ranges from 1 to 5 and the default value is 5

3.10.6.4.2. Command Default

Counter is set to 5 at default

3.10.6.4.3. Command Modes

Voice-Service configuration mode

3.10.6.4.4. Usage Guidelines

This command sets a part of global voice-service configuration for VoIP service.

cras counter retransmits a message if there is no message is received during **timeout tra,** after RAS message for gatekeeper such as GRQ, RRQ, ARQ, DRQ are transmitted.

3.10.6.4.5. Examples

The following example tries RAS message twice.

```
voice service voip
  counter cras 2
```

3.10.6.5. discovery

To enable GRQ (Gatekeeper Request) message transmission, use this command. To disable this feature, use the no form of this command.

discovery

no discovery

3.10.6.5.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.6.5.2. Command Default

Enabled

3.10.6.5.3. Command Modes

Gateway configuration

3.10.6.5.4. Usage Guidelines

When the gateway is registered for the first time and this feature is enabled, send GRQ first, then receive GCF and send RRQ. When this feature is disabled, it does not send GRQ and send RRQ directly.

3.10.6.5.5. Examples

The following example disables discovery.

```
gateway
no discovery
```

3.10.6.6. fax protocol

To specify the global default fax protocol to be used for all 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

3.10.6.6.1. Syntax

Keyword / Argument	Description
t38	The fax protocol of ITU-T T.38 standard
Inband-t38	The fax protocol deviated from T.38 Transmits T.38 with RTP payload. The fax transmission method of COMMWORKS(3COM)
bypass	Passes the fax protocol through clean voice channel (G.711) transparently.
redundancy	Configures redundancy for T.38
value	Ranges from 0 to 5. The default is 0.

3.10.6.6.2. Command Default

T.38 fax protocol

3.10.6.6.3. Command Modes

Voice-service configuration

3.10.6.6.4. Usage Guidelines

Use **fax protocol t38** command to configure t.38 fax relay. This keyword enables T.38 fax relay protocol packet. Redundancy is used to send T.38 fax packet as an optional parameter.

This option of inband-t38 must be chosen, when you use Commworks' (formerly 3Com) equipment.

3.10.6.6.5. Examples

The following example configures t.38 fax protocol for VoIP starting from global configuration mode.

```
voip service voip
fax protocol t38
```

3.10.6.7. fax rate

To set rate for transmitting fax for the dial-peer, use **fax rate** command in voice-service configuration mode. To reset the dial peer for voice call, use the no form of this command.

fax rate { 2400 | 4800 |7200 | 9600 | 12000 | 14400 |disable }

no fax rate

3.10.6.7.1. Syntax

Keyword / Argument	Description
2400	Set fax rate to 2400bps
4800	Set fax rate to 4800bps
7200	Set fax rate to 7200bps
9600	Set fax rate to FAX 9600bps
12000	Set fax rate to FAX 12000bps
14400	Set fax rate to FAX 14400bps
disable	Disables fax rate

3.10.6.7.2. Command Default

9600 bps

3.10.6.7.3. Command Modes

Voice-Service configuration

3.10.6.7.4. Usage Guidelines

To set fax transmission speed for all dial peers. Use **fax rate** command.

This command uses a value for only fax transmission speed and has no effect fax quality. 높은

The fax with high speed is transmitted as fast as 14400bps and even faster but it takes a significantly large bandwidth. The fax with low transmission speed is transmitted as slow as 2400bps, it takes a relatively small bandwidth.

This command is applicable only with T.38 fax relay. If the value is set to fax rate disable then T.38 fax relay does not work.

If the actual fax rate is set to 9600bps, the actual fax rate can be negotiated as 9600bps, even this command sets T.38 to 1440 bps.

3.10.6.7.5. Examples

The following example sets a fax rate to 9600bps for the fax transmitted by using dial peer:

```
voice service voip

fax rate 9600
```

3.10.6.8. force-h245address-at-setup

To enable including H.245 address information for its procedure in SETUP message from sending side when to start a call to H.245 no tunneling, use this command. Not to include H.245 address information, use the no form of this command.

force-h245address-at-setup

no force-h245address-at-setup

3.10.6.8.1. Syntax

Keyword / Argument	Description
This command had no arguments or keywords.	

3.10.6.8.2. Command Default

Enabled

3.10.6.8.3. Command Modes

Voice Service configuration

3.10.6.8.4. Usage Guidelines

To take a definite interoperation with another gateway on the other side, use this command.

3.10.6.8.5. Examples

The following example does not include h245address information in SETUP message:

```
voice service voip

  no force-h245address-at-setup
```

3.10.6.9. force-starth245

To set up TCP connection for H.245 procedure to startH245, when a call completed to H.245 no tunneling, use this command. Not use startH245 procedure for no tunneling, use the no form of this command.

force-starth245

no force-starth245

3.10.6.9.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.6.9.2. Command Default

Disables

3.10.6.9.3. Command Modes

Voice Service configuration

3.10.6.9.4. Usage Guidelines

To take a definite interoperation with another gateway on the other side, use this command. When a call is competed by no tunneling and TCP is not connected, startH245 based procedure can be carried out.

3.10.6.9.5. Examples

The following example enables startH245 procedure.

```
voice service voip

  force-starth245
```

3.10.6.10. 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 reset to the default, use the **no** form of this command.

h323 call start { fast | slow | preferred-slow }

no h323 call start

3.10.6.10.1. 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	After this setup, proceed slow start (normal start procedure) when to send a call, receive a call from the other side, depending on mode, proceed with fast start to slow start procedure

3.10.6.10.2. Command Default

Fast

3.10.6.10.3. Command Modes

Voice-Service configuration

3.10.6.10.4. Usage Guidelines

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.

3.10.6.10.5. Examples

The following example selects Slow Connect procedures for the gateway:

```
voice service voip
h323 call start slow
```

3.10.6.11. inband-ringback-tone

To deliver ringback tone when fast connect is open and a terminal at receiving user side is in alerting state, use this command

inband-ringback-tone

no inband-ringback-tone

3.10.6.11.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.6.11.2. Command Default

Disabled

3.10.6.11.3. Command Modes

Voice-Service configuration

3.10.6.11.4. Usage Guidelines

To set up inband-ringback-tone when virtual ringback tone is not used by the gateway on sender's side, ringback tone should be carried with inband (opened RTP channel by fast start), even ALERT message is received, because virtual ringback tone is not used.

3.10.6.11.5. Examples

The following example sets up inband-ringback tone.

```
voice service voip
  inband-ringback-tone
```

3.10.6.12. local-ringback-tone

To setup timing for generating a virtual ringback tone from sending side. For not using ringback tone, use the no form of this command.

local-ringback-tone [early]

no local-ringback-tone

3.10.6.12.1. Syntax

Keyword / Argument	Description
Early	Generates ringback tone after SETUO message is sent out at sender side.
Alert	Generate ringback tone after alert message is received at sender side
<cr>	(default). Generate ringback tone at the point for receiving either inband or alert message at sender side

3.10.6.12.2. Command Default

Enabled

3.10.6.12.3. Command Modes

Voice-service configuration

3.10.6.12.4. Usage Guidelines

Ringback tone can be heard after a call is sent out from sender's side by processing audio packer sent by in-band (RTP channel opened by fast start) from the other side and when ALERT message is received.

When the system is set at default, ringback tone is generated and heard by receiving ALERT message and announcement through in-band or tone is received. When the other party progress indicator information elements are configured with "inband announcement available, "the ringback tone inside is not generated.

If you want to generate a virtual ring tone right after sending SETUP message, set up **local-ringback-early**. If you want to hear only announcement to tone, which is delivered by in-band, set up **no local-ringback-tone**?

3.10.6.12.5. Examples

The following example sets up a virtual ringback tone.

```
voice service voip
  local-ringback-tone  early
```

3.10.6.13. minimize-voip-ports

To setup timing for generating a virtual ringback tone from sending side. For not using ringback tone, use the no form of this command.

local-ringback-tone [early]

no local-ringback-tone

3.10.6.13.1. Syntax

Keyword / Argument	Description
Early	Generates ringback tone after SETUO message is sent out at sender side.
Alert	Generate ringback tone after alert message is received at sender side
<cr>	(default). Generate ringback tone at the point for receiving either inband or alert message at sender side

3.10.6.13.2. Command Default

Enabled

3.10.6.13.3. Command Modes

Voice-service configuration

3.10.6.13.4. Usage Guidelines

Ringback tone can be heard after a call is sent out from sender's side by processing audio packer sent by in-band (RTP channel opened by fast start) from the other side and when ALERT message is received.

When the system is set at default, ringback tone is generated and heard by receiving ALERT message and announcement through in-band or tone is received. When the other party progress indicator information elements are configured with "inband announcement available ," the ringback tone inside is not generated.

If you want to generate a virtual ring tone right after sending SETUP message, set up **local-ringback-early**. If you want to hear only announcement to tone, which is delivered by in-band, set up **no local-ringback-tone**?

3.10.6.13.5. Examples

The following example sets up a virtual ringback tone.

```
voice service voip

  local-ringback-tone  early
```

3.10.6.14. max-frame

To set audio frame size in RTP packet, use this command. . To reset to default, use the no form of this command.

max-frame { g726 | g729 | g711 | g7231 } *value*

no max-frame { g726 | g729 | g711 | g7231 }

3.10.6.14.1. Syntax

Keyword / Argument	Description
g726	Sets a frame size of G.726 codec
g729	Sets a frame size of G.729 codec
g711	Sets a frame size of G.711 codec
g7231	Sets a frame size of G.7231 codec
value	Displays frame size. The value ranges from 1 to 8

3.10.6.14.2. Command Default

The default for G.7231 is 1. Every 30 msec, audio frame is generated and RTP packet is transmitted.

The default for G.729/G.711/G.726 is 2. Every 10 msec, audio frame is generated and RTP packet is transmitted every 20 msec.

3.10.6.14.3. Command Modes

Voice service configuration

3.10.6.14.4. Usage Guidelines

Refer to ‘Error! The referenced original document can not be found’ for the relation between audio frame and RTP packet.

If internet environment is not good, increase max-frame-size. First increase max-frame-size to 2 for G.7231 and 2 for G.729. If bandwidth still short, then increase max-frame to 3 for G.7231 and 6~8 for G.729.

As a disadvantage, increasing max-frame causes to increase delay.

This command is useful for cable network with relatively low upstream traffic environment. When this command is used with quos-control in this network interface command, voice

quality can be improved significantly.

3.10.6.14.5. Examples

The following example sets quos-control and max-frame in cable network:

```
interface ethernet 0 0
   qos-control 128 100
voice service voip
   max-frame g729 4
   max-frame g7231 2
```

3.10.6.15. gkip

To specify a gatekeeper to be registered to the gateway, use gkip in gateway configuration mode. To delete a particular gatekeeper from gatekeeper list, use the no form of this command.

gkip *ip-addr [port] [priority]*

no gkip *ip-addr*

3.10.6.15.1. Syntax

Keyword / Argument	Description
ip-addr	Transmits gatekeeper discovery message to the specified address
port	(Optional) Specifies UDP port of gatekeeper. The default is 1719.
priority	(Optional) Specifies a priority to many alternate gatekeeper. The value ranges from 0 to 254. Lower the number, higher the priority is. The default priority is 128

3.10.6.15.2. Command Default

128 in priority

3.10.6.15.3. Command Modes

Gateway configuration

3.10.6.15.4. Usage Guidelines

Registering to gatekeeper, the AddPac Voice Finder Gateway can take number and billing services. The AP1000 Gateway specify up to 10 gatekeepers. When more than one gatekeeper is specified, the gateway registers the gatekeeper in a priority order by using GRQ message. Only one gatekeeper can registered at the same time. When registration with the gatekeeper or receiving message fails, the gateway attempts to register the gatekeeper in a priority order.

Another way of specifying the gateway is to use alternategk list in the received message from the registered gatekeeper. For the reference, The public gatekeeper multicast IP address complying with H.323 is 224.0.41 and port is 1718.

3.10.6.15.5. Examples

The following example sets up the gatekeeper with 192.7.5.1:

```
gkip 192.7.5.1
```

The following example sets the gatekeeper with the multicast ip address 224.0.1.41 and priority to 0:

```
gkip 224.0.1.41 1718 0
```

3.10.6.16. h323-id

To register an H.323 proxy alias with a gatekeeper, use the **h323 h323-id** command in interface configuration mode.

h323-id *h323_id*

3.10.6.16.1. Syntax

Keyword / Argument	Description
h323-id	Name of the proxy. It is recommended that this name be a fully qualified e-mail ID, with the domain name being the same as that of its gatekeeper

3.10.6.16.2. Command Default

voip.*ip_address*

3.10.6.16.3. Command Modes

Gateway configuration

3.10.6.16.4. Examples

The following example registers a gateway to GW13@addpac.com.

```
gateway
gkip 211.238.1.1
h323-id GW13@addpac.com
```

3.10.6.17. lightweight-irr

To enable transmitting Information Request Response (IRR) message as a brief form of information, use this command. To disable this feature, use the no form of this command.

lightweight-irr

no lightweight-irr

3.10.6.17.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords	

3.10.6.17.2. Command Default

Disabled

3.10.6.17.3. Command Modes

Gateway configuration

3.10.6.17.4. Usage Guidelines

The gateway sends IRR in response to IRQ message sent from the gatekeeper. The usage purpose of IRR message is to check the gateway status and the message can contain much information at default, but if IRR cycle is short and only brief information is needed, then it can contain only the necessary information.

3.10.6.17.5. Examples

The following example send only the necessary information on IRR message.

```
gateway
  lightweight-irr
```

3.10.6.18. h323 call channel

To open voice channel prior to CONNECT and when H323 call start is process in normal (slow) call start procedure instead of fast connect, use h323 call channel early command in voice service configuration. To reset and return to the default selection order, use the no form of this command.

h323 call channel { early | late }

no h323 call channel

3.10.6.18.1. Syntax

Keyword / Argument	Description
early	Opens the voice channel prior to CONNECT in normal (slow) call start
late	Opens the voice channel after CONNECT in normal (slow) call start

3.10.6.18.2. Command Default

Late

3.10.6.18.3. Command Modes

Voice service configuration

3.10.6.18.4. Usage Guidelines

This command set up a part of global voice service for VoIP call. Logical channel (voice channel), based on H.245 procedure can be open before and after a point of time, which the other party delivers CONNECT by hook off, when the gateway or a remote gateway sets h323 call start to normal.

The call channel is set to late at default. In this process, the other party receives CONNECT message after hook off, then opening voice channel procedure starts. In this process, the form end of speech can be cut off. To avoid such a problem from happening, use h323 call channel early to open the voice channel before hook off.

3.10.6.18.5. Examples

The following example sets a voice channel to open early:

```
voice service voip
  h323 call channel early
```

3.10.6.19. h323 call response

To specify other messages besides alert message after CALL PROCEEDING is connected in response to Q.931 SETUP, use h323 call response command in voice configuration mode. To return to the default, use the no form of this command.

h323 call response { alert| progress | none}

no h323 call response

3.10.6.19.1. Syntax

Keyword / Argument	Description
alert	Send alert message as a reply
progress	Send progress message as a reply
none	Send CONNECT message as a rely after call proceeding

3.10.6.19.2. Command Default

Alert

3.10.6.19.3. Command Modes

Voice configuration

3.10.6.19.4. Usage Guidelines

This command takes a part of configuration of global voice service for VoIP service. When the gateway operates on receiver side, the following process takes a place:

- SETUP message is received
- CALL PROCEEDING message is sent
- The user hooks off the phone
- Whether to send ALERT or PROGRESS or not depending on this command
- CONNECT message is sent

  Except some special circumstances, it is recommended to set this setting to default.

3.10.6.19.5. Examples

The following example sets up a reply to PROGRESS message:

```
voice service voip
  h323 call response progress
```

3.10.6.20. max-digits

FXO port로의 발신한 Call에 대한 보안을 위하여 특정 User Class에 대한 발신 To secure an outgoing call to FXO by setting a limit of the maximum number or digits to the outgoing call for a particular user class, use max-digits command. To set to the default value of 0 which means no limit, use the no form of this command.

max-digits *number*

no max-digits

3.10.6.20.1. Syntax

Keyword / Argument	Description
number	Maximum number of digits for an outgoing call

3.10.6.20.2. Command Default

0 (no limit)

3.10.6.20.3. Command Modes

User class configuration

3.10.6.20.4. Examples

The following example sets the maxim number of digit for user class 1.

```
voice class user 1
max-digits 10
```

3.10.6.21. password

To secure an outgoing call to FXO by setting a limit of the maximum number or digits to the outgoing call for a particular user class, use password command. To set to null at default, use the no form of this command. When password is set to null, the outgoing call to FXO is not checked for security. However, if any of password digits is set in the registered user class, the outgoing call is checked for security.

password *string*

no password

3.10.6.21.1. Syntax

Keyword / Argument	Description
string	Security code based on IA5 text sequence and organized with binar4y coded decimal

3.10.6.21.2. Command Default

The default value is enabled with null string

3.10.6.21.3. Command Modes

User class configuration

3.10.6.21.4. Examples

The following example sets password 1234 to user class 1:

```
voice class user 1
password 1234
```

3.10.6.22. public-ip

To assign a public IP number mapping to the private IP number of the gateway under a static NAT/ PAT environment, use this command. To disable this feature, use the no form of this command.

public-ip *addr*

no public-ip

3.10.6.22.1. Syntax

Keyword / Argument	Description
addr	For instance, this is a type of public IP number such as 211.238.72.3

3.10.6.22.2. Command Default

Disabled

3.10.6.22.3. Command Modes

Gateway configuration

3.10.6.22.4. Usage Guidelines

When the gate way located in a private network of a company, a private IP is assigned to VoIP interface. When the gateway communicates with a gatekeeper located in PSTN, the gateway must be assigned with static NAT or PAT and the public IP address can be specified by using this command.

3.10.6.22.5. Examples

The following example assigns a public IP.

```
gateway
  public-ip xxx.xxx.xxx.xxx
```

3.10.6.23. register

To register H.323 voice over IP gateway to a gatekeeper, use **register** command. To deregister the gateway from the gatekeeper, use the no form of this command.

register

no register

3.10.6.23.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.6.23.2. Command Default

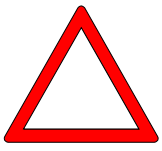
Disabled

3.10.6.23.3. Command Modes

Gateway configuration

3.10.6.23.4. Usage Guidelines

To enable H.323 VoIP gateway feature, use **registe**r command. When the gateway is enables, the gateway tries to find a gatekeeper by using H.323 RAS GRQ (or RRQTo deregister the gateway from the gatekeeper by using H.323 RAS URQ message, use **no register** (no gateway in global configuration mode) command.



If you want to use a script file to register and change the number for the gateway which is already registered and is in operation, use no register command first to deregister from the gatekeeper, then load a configuration or state clearly no register (no gateway) in the beginning of script file. If you do not use this command, the gatekeeper can be over-flooded with messages for renewing the updated information of the gateway.

3.10.6.23.5. Examples

The following example specifies registration:

```
gateway
  register
```

3.10.6.24. signaling-port

To change q931 signaling port (default TCP 1720) which is used in H.323, use **signaling-port** command. Since TCP port 1720, which is used in Q931, is a well know port. To change this port, only the gateways with the same settings can be set up. Therefore, If you use this set up, a general gateway loose its interoperability and it requires an attention.

signaling-port port-number

no signaling-port

3.10.6.24.1. Syntax

Keyword / Argument	Description
Signaling-port	Sets up TCP port Save the settings after changing to the other port then reboot.

3.10.6.24.2. Command Default

TCP port 1720

3.10.6.24.3. Command Modes

Gateway configuration

3.10.6.24.4. Usage Guidelines

Use this command to change Q931 signaling port, which is not in use.

3.10.6.24.5. Examples

The following example changes signaling port to 1004(TCP).

```
gateway
 signaling-port 1004
```

3.10.6.25. rule

To apply translation rule to the calling or called party number of the inbound or outbound call, use **rule** command in translation configuration mode. 설정된 rule을 제거하기 위하여는 위 To delete the rule with configured settings, use the no form of this command.

rule *tag input-matched-pattern substituted-pattern*

no rule *tag*

3.10.6.25.1. Syntax

Keyword / Argument	Description
tag	Only an identifier specifies rule in rule set. Valid entry ranges from 0 to 65535
input-matched-pattern	Input digits for pattern matching. Valid character entry is 0-9#*[].T
substituted-pattern	pattern matching The pattern to be changed when pattern matching succeeds. Valid character entry is 0-9#*[].T

3.10.6.25.2. Command Default

No default behavior or value.

3.10.6.25.3. Command Modes

Translation rule configuration

3.10.6.25.4. Usage Guidelines

This command is used to apply translation rule to the calling or called party number of inbound or outbound call.

substituted-pattern translates the fixed digit (the digit excluding wildcard) of *input-matched-pattern* to a string of substituted-pattern.

substituted-pattern can be divided by 2 formats.

In first case, *substituted-pattern* is configured only with IA5 texts (0-9#), the fixed digit part of input-matched-pattern is translated into the string part of *substituted-pattern*, the rest of digits, except the fixed digits of the called (or calling) party number, are to added next to the end.

In next case, *substituted-pattern* uses '%' to enable configuration of the number by

substituting each digit of the called (or calling) party number to %xx variable.

substituted-pattern is only configured with '.'or 'T', the called(or calling) -party-number is to be configured with the digits except the fixed patter of *input-matched-pattern*.

3.10.6.25.5. Examples

The following example expands the number with 5554123 to 140855554123.

rule 0 55541 14085555541

The following example does not translate for the number with 5551 but it translates 551234 to 14085551234

rule 0 555.. 1408555

The following example translates the number with 1251234 to 14085551234 and 3551234 to 14085551234.

rule 0 [1-3][25]5.. 1408555

The following example translates the number with 5551234 to 4441234.

rule 0 555.. 444%04%05%06%07%08%09%10%11%12

. The following example translates all the numbers with 55512, 5551234, 555123456 to 444.

rule 0 555.. 444%99

The following example translates the number with 5551234 to 3334.

rule 0 555.. 111

rule 1 55512 222

rule 2 555[0-9][0-9][0-9] 333

The following example translates the number with 5551234 to 1234

rule 0 555 .

rule 0 555 T

The following example translates the number with 5551234 to 95551234.

rule 0 . 9

rule 0 T 9

3.10.6.26. security password

To configure secure token with a gatekeeper, use security password command. If this password is enables, the gateway adds crypto token and send it to the gateway. This crypto token registers the gateway to the gatekeeper by MD5 Hashed Token. When a call is allowed, it should have been enabled already. To disable security between the gateway and gatekeeper, use the no form of this command.

security password *string*

no security password

3.10.6.26.1. Syntax

Keyword / Argument	설명
string.	Security Code based on ASCII tests

3.10.6.26.2. Command Default

Disables

3.10.6.26.3. Command Modes

Gateway configuration

3.10.6.26.4. Examples

The following example sets the password to “okok1234”

```
gateway
security password okok1234
```

3.10.6.27. acf-dest-info

When the gateway transmits ARQ to the gatekeeper in general, ACF message transmit destination information with ARQ as it is. However, the gatekeeper needs to change the destination information by different settings. ACF information was used to be ignored previously, but SETUP message is to be delivered basin on this information in this case. To delete the application of this command, use the no form of this command.

acf-dest-info

no acf-dest-info

3.10.6.27.1. Syntax

Keyword / Argument	Description
This command has no arguments or keyword.	

3.10.6.27.2. Command Default

Enabled

3.10.6.27.3. Command Modes

Gateway configuration

3.10.6.28. security permit-FXO

Security must consider for an outgoing call going through FXO port of this system to PSTN or PABX. When security permit-FXO feature is disabled, the call originated from an unauthorized user in a remote side is to be dropped.

To allow all the calls directing toward FXO when session-target is set to RAS of the gatekeeper, use this command. To enable security, use the no form of this command. Then all the calls with deregistered IP address to VoIP peer are not allowed.

security permit-FXO

no security permit-FXO

3.10.6.28.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.6.28.2. Command Default

Permit all the calls

3.10.6.28.3. Command Modes

Voice service configuration

3.10.6.28.4. Usage Guidelines

The reason, that the security is needed for the incoming call on FXO, is that there can be a misuse of unauthorized remote user by the direct call attempt, which is possible through this FXO port, and the indirect call attempt to PSTN through an extension of PBX is also possible. The gateway provided 2 type of security system which is described in the following advantages and disadvantages:

security permit-FXO is simple because the remote user doe not need to enter a password. On the other hand, all the IP addresses of VoIP peer must be registered and can not be registered together with a gatekeeper and can not perform the call limit to classify the registered peer.

Voice class user many be inconvenient in a way as for the user to enter password digits, but the

security can be stronger and classification of call limit is possible.

3.10.6.28.5. Examples

The following example allows all the calls directing toward FXO:

```
voice service voip

security permit-FXO
```

3.10.6.29. security type (Secure VoIP gateway Specific)

To specify security type for security call, use this command

security type {none | des | 3des}

3.10.6.29.1. Syntax

Keyword / Argument	Description
none	disable security feature
des	set security algorithm to DES
3des	set security algorithm to triple-DES

3.10.6.29.2. Command Default

No default behavior or value.

3.10.6.29.3. Command Modes

Voice service configuration

3.10.6.29.4. Usage Guidelines

Generally VoIP communication delivers voice by using Real Time Protocol (RTP). However, RTP packet is week for security. Voice message can be intercepted by using a voice analyzer and personal or business secrets can be exposed. To secure against this problem, Security Real Time Protocol (SRTP) is used to make this interception impossible.

If the other party's equipment does not provide security, this feature sends the voice packet without security, so it can be interoperated.

3.10.6.29.5. Examples

The following example specifies security type to 3des:

```
voice service voip

security type 3des
```

3.10.6.30. security module (Secure VoIP gateway Specific)

This command enables or disables security feature

security module {enable | disable}

3.10.6.30.1. Syntax

Keyword / Argument	Description
enable	Enable communication with security
disable	Disable communication with security

3.10.6.30.2. Command Default

No default behavior or value.

3.10.6.30.3. Command Modes

Voice service configuration

3.10.6.30.4. Usage Guidelines

If this feature is not enabled after setting up security type, you can communicate with security. So this feature must be enabled.

3.10.6.30.5. Examples

The following example enables communication with security:

```
voice service voip

security module enable
```

3.10.6.31. timeout

To set VoIP related timer parameters, use timeout command in voice service configuration. To return to the default state, use the no form of this command.

timeout { tinit | tring | t301 | t303 | tras | tttl | tidt | treg | treg2 | tohd | tpoll | tterm } *value*

no timeout tinit | tring | t301 | t303 | tras | tttl | tidt | treg | treg2 | tohd | tpoll | tterm }

3.10.6.31.1. Syntax

Keyword / Argument	Description
tinit *value*	First digit entry to voice port after hook off. The value ranges from 1 to 600, The default is 10.
tring *value*	Time out value of ring generation to voice port. The value ranges from 5 to 600, The default is 30. The unit is second.
t301 *value*	Time out value starting from the point Q.931 message received until CONNECT message is received. The value ranges from 5 to 600. The default is 180. The unit is second
t303 *value*	Time out value starting from the point Q.931 message received until CONNECT message is received. The value ranges from 5 to 60. The default is 8. The unit is second
tras *value*	The time out value starting from the point that RAS message is received until the message is replied. The value ranges from 2 to 30. The default is 6. The unit is second.
tttl *value*	Time out value for RAS time-to-live. The default is 60. The unit is second. The value is renewed by gatekeeper.
tidt *value*	Inter-digit time out for digit entry to voice port. The value ranges from 1 to 600. The default is 10. The unit is second.
treg *value*	Timeout for registration retry to gatekeeper. The value ranges from 10 to 600. The default is 30. The unit is second.
treg2 *value*	When registration to gatekeeper fails (RRJ), it is not switched to back up gatekeeper. Timeout value for registration retry to the main gatekeeper. The registration value ranges from 10 to 86400. The default value is 120. The unit is second.
tohd *value*	The time out value for on hook to voice port. The value ranges from 10 to 30. The default is 0. The unit is second
tpoll *value*	Polling timeout value. The value ranges from 10~86400. The default is 180. The unit is second.
tterm *value*	Time out value for call duration of voice port. The value ranges from 10~86400. The default is infinite. The unit is second.

3.10.6.31.2. Command Default

Refer to the table above for the default values

3.10.6.31.3. Command Modes

Voice service configuration

3.10.6.31.4. Usage Guidelines

The command sets up a part of global voice service configuration for VoIP service.

The timeout at default is set to the proper values. The defaults are basically recommended

3.10.6.31.5. Examples

The following example sets RAS message timeout value to 3 seconds:

```
voice service voip
  timeout tras 3
```

3.10.6.32. translate-voip-incoming

To apply translation rule to all the inbound VoIP calls, use this command. To delete application of translation rule, use the no form of this command.

translate-voip-incoming { called-number | calling-number } *tag*

no translate-voip-incoming { called-number | calling-number }

3.10.6.32.1. Syntax

Keyword / Argument	Description
called-number	Applies translation rule to the inbound called party number
calling-number	Applied translation rule to the inbound calling party number
tag	References to rule set. The value ranges from 0 to 65535

3.10.6.32.2. Command Default

Not to apply translation rule.

3.10.6.32.3. Command Modes

Voice service configuration

3.10.6.32.4. Usage Guidelines

This command applies the configured number translation by using translation-rule to all the inbound VoIP calls from network

3.10.6.32.5. Examples

The following example creates translation rule set 10 and applies it to the calling party number of VoIP inbound call. If the calling party number is 93450, then it is translated to 9563450.

```
translation-rule 10
        rule 0  9  956
        rule 1  8  878
voice sevice voip
        translate-voip-incoming calling-number 10
```

3.10.6.33. voice-confirmed-connect

.No to send connect message to the other party when the end user is not able to listen to voice even FXO port of the gateway is connected. To delete application of this command, use the no form of this command.

voice-confirmed-connect

no voice-confirmed-connect

3.10.6.33.1. Syntax

Keyword / Argument	Description
The command has no arguments or keywords.	

3.10.6.33.2. Command Default

Disabled

3.10.6.33.3. Command Modes

Voice service configuration

3.10.6.33.4. Usage Guidelines

When FXO port is connected to PBX extension and the subscriber does take the call, connect message is not sent to sender side and billing is not included.

3.10.6.33.5. Examples

The following example sets up voice-confirmed-connect:

```
voce service voip
   voice-confirmed-connect
```

3.10.6.34. accept-fst-at-connect

When H.323 cal is originated, its own ringback tone is played to the user side (FXS or FXO) regardless of receiving alerting message.

When this feature is set up, OLC channels is not open for not sending from the remote equipment to inband (RTP).

To delete application of this command, use the no form of this command.

accept-fst-at-connect

no accept-fst-at-connect

3.10.6.34.1. Syntax

Keyword / Argument	Description
This command has no arguments or key words.	

3.10.6.34.2. Command Default

Disabled

3.10.6.34.3. Command Modes

Voice service configuration

3.10.6.34.4. Usage Guidelines

To provide ringback tone to inband in general, repose message is used for progress message. However, some equipment delivers a small number of RTP packets or silence to inband after delivering response message to alerting.

In this case, the gateway plays its own ringback tone and tries to play the packets coming from RTP. However, ringback tone can not be heard if it is silence and a very little RTP packet is transmitted

By using this feature, ringback tone can heard if RTP packet, which comes prior to connect, is ignored

3.10.6.34.5. Examples

The following example sets up accept-fst-at-connect.

```
voice service voip
   accept-fst-at-connect
```

3.10.6.35. Resource Threshold (RAI)

To configure a gateway to report H.323 resource availability to its gatekeeper, use the **resource threshold** command in gateway configuration mode. To disable gateway resource-level reporting, use the **no** form of this command.

resource threshold *high-thresold | low-threshold*

no resource threshold

3.10.6.35.1. Syntax

Keyword / Argument	Description
This command has no arguments and keywords.	

3.10.6.35.2. Command Default

Disabled

3.10.6.35.3. Command Modes

Voice service configuration

3.10.6.35.4. Usage Guidelines

This command defines the resource load levels that trigger RAI messages and limit a number of calls to a proper level. When a call reaches a high threshold value, the gatekeeper stops sending any more calls to the gateway by setting "**OutOfResources"** field of RAI message to **"TRUE".** After the call is terminated and reaches to the low-threshold value, setting "OutOfResources" field of RAI message to false allows the gatekeeper to receive or the gateway to send more calls.

3.10.6.35.5. Examples

The following example configures accept-fst-at-connect:

```
voce service voip
  accept-fst-at-connect
```

3.10.7. Other Commands

3.10.7.1. clear h323 call

To disconnect all active calls or a specific call, from a remote user, by force, **use clear h323 call** command.

clear h323 call { all | *local_call_ID* **}**

3.10.7.1.1. Syntax

Keyword / Argument	Description
all	Disconnect all enabled h323 call to the remote user
local_call_ID	Specify and disconnect a call by a particular local call identification number

3.10.7.1.2. Command Default

No default behavior or value.

3.10.7.1.3. Command Modes

Administrator's command

3.10.7.1.4. Usage Guidelines

To disconnect a specific call by force (interrupting all the active calls to gatekeeper), use this command with call-id number specifies a particular call. To find the local call-id number for the particular call, use **show call active** all command.

3.10.7.1.5. Examples

The following example disconnects all the active h323 calls.

```
clear h323 call all
```

3.10.7.2. clear voice-port

To terminate a call for a particular voice port, use clear voice-port in administrator's command.

If the port is not specified, all the calls of this system are terminated.

clear voice-port *port]*

3.10.7.2.1. Syntax

Keyword / Argument	Description
port	Argument to specify a port with a call to be terminated.

3.10.7.2.2. Command Default

No default behavior or value.

3.10.7.2.3. Command Modes

Administrator's command

3.10.7.2.4. Usage Guidelines

None

3.10.7.2.5. Examples

The following example terminates all the active calls of VoIP gateway system.

```
clear voice port
```

3.10.7.3. show call active

To display call information for voice calls or fax transmit in progress, use the **show call active** command.

show call active { all/summary }

3.10.7.3.1. Syntax

Keyword / Argument	Description
all	Displays detailed information for all the active calls
summary	Displays a summarized information for all the active calls

3.10.7.3.2. Command Default

No default behavior or value.

3.10.7.3.3. Command Modes

Administrator's command

3.10.7.3.4. Usage Guidelines

Use show call active command to display the active call table. This command displays call time, dial peer, call configuration and connection and other status and static information.

3.10.7.3.5. Examples

The following example displays the summarized call information:

```
show call active summary
```

3.10.7.4. show call history

To display the call history table for voice calls and fax transmits, use the **show call history**command in administrator's command mode.

show call history { all } { last *number* **}**

3.10.7.4.1. Syntax

Keyword / Argument	Description
all	Displays all the information of call history
last	Displays the last call history in a number range (optional)
number	Specifies a number of call history to be displayed in descending time order

3.10.7.4.2. Command Default

No default behavior or value.

3.10.7.4.3. Command Modes

Administrator's command mode

3.10.7.4.4. Usage Guidelines

This command displays a call-history table that contains call continuation time, call setup time and information of called and calling party

3.10.7.4.5. Examples

The following example displays the last 10 call history:

```
show call history all last 10
```

3.10.7.5. show clear-down-tone

To see clear-down-tone class information, use show clear-down-tone command. If a number is not assigned, all clear-down-tone class are displayed

show clear-down-tone

3.10.7.5.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.7.5.2. Command Default

No default behavior or value.

3.10.7.5.3. Command Modes

Administrator's command

3.10.7.5.4. Usage Guidelines

This command uses voice class clear-down-tone command. No only the user's clear-down-tone, but the entire clear-down-tone provided by the system is displayed.

3.10.7.5.5. Examples

The following example displays all clear-down-tone class:

```
show clear-down-tone
```

3.10.7.6. show codec-class

To display codec class information, use **show codec-class** command in administrator's command. Without specification of option number, all codec class is displayed.

show codec-class *[number]*

3.10.7.6.1. Syntax

Keyword / Argument	Description
number	(Optional) Specified coded class tag number to be displayed.

3.10.7.6.2. Command Default

No default behavior or value.

3.10.7.6.3. Command Modes

Administrator's command

3.10.7.6.4. Examples

The following example displays all codec classes.

```
show codec-class
```

3.10.7.7. show dial-peer

administrator's command mode. If no number is specified, the command displays all the dial-peers.

show dial-peer {voice |pots| voip} [*number* **|** ***summary]***

3.10.7.7.1. Syntax

Keyword / Argument	Description
voice	Displays POTS and VoIP Dial-peer
pots	Displays POTS Dial-peer
voip	Displays VoIP Dial-peer
number	(Optional)Specifies Dial-peer tag number to be displayed
summary	(Optional) Displays a summarized information only

3.10.7.7.2. Command Default

No default behavior or value

3.10.7.7.3. Command Modes

Administrator's command

3.10.7.7.4. Usage Guidelines

Use this command to display the configuration for all VoIP and POTS dial peers configured for a gateway. To show configuration information for only one specific dial peer, use the *number* argument to
identify the dial peer.

3.10.7.7.5. Examples

The following example displays all dial peer information of a single system

```
show dial-peer voice
```

3.10.7.8. show dialplan number

To display which outgoing dial peer is reached when a particular telephone number is dialed, use the **show dialplan number** command in administrator's command mode.

show dialplan number *dial_string*

3.10.7.8.1. Syntax

Keyword / Argument	Description
dial_string	A paticular destination pattern(telephone number)

3.10.7.8.2. Command Default

No default behavior or value

3.10.7.8.3. Command Modes

Administrator's mode

3.10.7.8.4. Usage Guidelines

Use this command to test whether the dial plan configuration is valid and working as expected.

3.10.7.8.5. Examples

The following example displays all the dial peers matching with the telephone number 4441234:

```
show dialplan number 4441234
```

3.10.7.9. show dialplan port

To display the related information of which POTS dial peers is matched with voice port, use show dialplan port command.

show dialplan port *voice-port*

3.10.7.9.1. Syntax

Keyword / Argument	Description
voice_port	Specifies voice_port location(Port Number)

3.10.7.9.2. Command Default

No default behavior or value.

3.10.7.9.3. Command Modes

Administrator's command

3.10.7.9.4. Usage Guidelines

To determine which POTS dial peer is matched , use show dialplan number command as a trouble shooting tool.

3.10.7.9.5. Examples

The following example displays all the dial peers matched with voice port 2:

```
show dialplan port 2
```

3.10.7.10. show gateway

To display the current status of the gateway, use the **show gateway** command

show gateway

3.10.7.10.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.7.10.2. Command Default

No default behavior or value

3.10.7.10.3. Command Modes

Administrators command

3.10.7.10.4. Usage Guidelines

This command displays IP address, registration status, registered names of a gatekeeper and a number of dial-peers, voice ports of a gateway.

3.10.7.10.5. Examples

The following example displays the information of the gateway.

```
show gateway
```

3.10.7.11. show num-exp

To display the current status of the gateway, use the **show gateway** command

show num-exp

3.10.7.11.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.7.11.2. Command Default

No default behavior and value

3.10.7.11.3. Command Modes

Administrator's command

3.10.7.11.4. Usage Guidelines

This command does not display wildcard, if a user creates a number expansion by using wildcard(*).

3.10.7.11.5. Examples

The following example displays a number expansion of the system:

```
show num-exp
```

3.10.7.12. show translation-rule

To display the contents of the rules that have been configured for a specific translation name, use the **show translation-rule** command

show translation-rule *[tag]* *[dial_string]*

3.10.7.12.1. Syntax

Keyword / Argument	Description
tag	Specifies a particular rule set. Without specification, all the translation rules are displayed.
dial_string	When you enter a particular destination pattern (telephone number), the rule displays the result of the application

3.10.7.12.2. Command Default

No default behavior and value

3.10.7.12.3. Command Modes

Administrator's command

3.10.7.12.4. Usage Guidelines

Use this command to test whether the translation-rule configuration is valid and working as expected.

3.10.7.12.5. Examples

The following is the sample output of this command applied with a telephone number 4441234:

```
show translation-rule 10 4441234
```

3.10.7.13. show user-class

To display user class information, use show user-class in administrator's command

show user-class

3.10.7.13.1. Syntax

Keyword / Argument	Description
This command has not arguments or keywords.	

3.10.7.13.2. Command Default

No default behavior or value.

3.10.7.13.3. Command Modes

Administrator's command

3.10.7.13.4. Usage Guidelines

This command displays the maximum digit information for entering tag and password of user class.

3.10.7.13.5. Examples

The following information displays user class information of the system:

```
show user-class
```

3.10.7.14. show voice port

To display configuration information about a specific voice port, use the **show voice port** command. If port information is not specified, all the available voice ports of the system would be displayed.

show voice port [summary | *port* **]**

3.10.7.14.1. Syntax

Keyword / Argument	Description
summary	(Optional) Output displays a summary of all voice ports.
port	(Optional) Specifies a port number to be displayed.

3.10.7.14.2. Command Default

No default behavior or value

3.10.7.14.3. Command Modes

Administrator's command

3.10.7.14.4. Usage Guidelines

This command can be used in voice port configuration mode.

3.10.7.14.5. Examples

The following example a summary of all voice ports of the gateway:

```
show voice port summary
```

3.10.7.15. show voip-interface

To see output of all the currently specified VoIP interfaces Use **show voip-interface** command.

show voip-interface

3.10.7.15.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords.	

3.10.7.15.2. Command Default

No default behavior or value.

3.10.7.15.3. Command Modes

Administrator's command

3.10.7.15.4. Usage Guidelines

Displays VoIP interface currently in service

3.10.7.15.5. Examples

The following example displayed VoIP information of the system.

```
show voip-interface
```

3.10.7.16. debug voip call

To trace VoIP related events, use debug voip call command in administrator's command

debug voip call

no debug voip call

3.10.7.16.1. Syntax

Keyword / Argument	Description
This command has no arguments or keywords,	

3.10.7.16.2. Command Default

No default behavior or value.

3.10.7.16.3. Command Modes

Administrator's command

3.10.7.16.4. Usage Guidelines

This command traces Q.931 event and H.245 event and user interface event and display to the console port. This command can slow down the performance of the system. Therefore, this feature must be disabled normally.

3.10.7.16.5. Examples

The following example displays by tracing VoIP calls:

```
debug voip call
```

The following example turns off tracing enabled voice calls in the system:

```
undebug voip call
```

3.10.7.17. debug voip

To trace ASN.1 of VoIP related event, use debug voip command.

debug voip { h225-asn1 | h245-asn1 | ras-asn1 }

no debug voip { h225-asn1 | h245-asn1 | ras-asn1 }

3.10.7.17.1. Syntax

Keyword / Argument	Description
h225-asn1.	H.225 ASN.1 Event
h245-asn1	Traces H.245 ASN.1 Event
ras-asn1	Traces RAS ASN.1 Event

3.10.7.17.2. Command Default

No default behavior or value.

3.10.7.17.3. Command Modes

Administrator's command

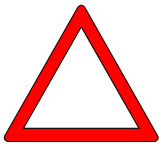
3.10.7.17.4. Usage Guidelines

This command traces the event of H.225 ASN.1, H.245 ASN.1 and RAS ASN.1 and displays them on the console port. This command can slow down performance of the system. Therefore, this feature must be disabled normally.

Information

VoIP related messages and call tracing can be displayed on the console port. To see tracing by telnet terminal from a remote location, use debug-port command in global command. Message tracing is operated only one remote terminal, then the terminal, to which debug-port command s used, is operated. If telnet is disconnected, tracing operates automatically on the console. To display tracing on the console at default, use no debug-port command.

Caution



The message tracing with debug command give a lot of load to the gateway, this feature must be disabled normally. When terminal is finished after tracing from telnet, without no debug or undebug command, tracing can go through the console, so caution is required.

3.10.7.17.5. Examples

The following example traces H.225 ASN.1 event for a voice call of the system:

```
debug voip h225-asn1
```

The following example turns off the enables debugging feature of H.245 ASN.1 to the system:

```
undebug voip h245-asn1
```

Appendix A H.323 Call Termination Cause Codes

The following table describes call termination cause code which is mapped with Q.931 cause or H.225 cause.

To trace the call cause code, use **show call history** command

AP1100 Call Termination Cause Code	Call Termination hosts	Call Termination Cause	Configuration Code for Transmit
RemoteNoBandwidth	remote side	For the cause transmitting RELCOM* noBandwidth(H225) NoCircuitChannelAvailable (Q931:34)	For the cause receiving RELCOM H225 destinationRejection
RemoteGatekeeperResourceUnavailable	remote side	For the cause transmitting RELCOM gatekeeperResources(H225) ResourceUnavailable (Q931:47)	For the cause receiving RELCOM H225 destinationRejection
RemoteUnreachableDestination	remote side	For the cause transmitting RELCOM unreachableDestination (H225) NoRouteToDestination (Q931: 3)	For the cause receiving RELCOM H225 destinationRejection
RemoteCallClear	remote side	For the cause transmitting RELCOM destinationRejection (H225) NormalCallClearing (Q931: 16)	For the cause receiving RELCOM H225 destinationRejection
RemoteIncompatibleDestination	remote side	For the cause transmitting RELCOM invalidRevision (H225) IncompatibleDestination (Q931: 88)	For the cause receiving RELCOM H225 destinationRejection
RemoteNoPermission	remote side	For the cause transmitting RELCOM noPermission (H225) InterworkingUnspecified (Q931: 127)	For the cause receiving RELCOM H225 destinationRejection
RemoteUnreachableGatekeeper	remote side	For the cause transmitting RELCOM unreachableGatekeeper (H225) NetworkOutOfOrder (Q931: 38)	For the cause receiving RELCOM H225 destinationRejection
RemoteResourceUnavailable	remote side	For the cause transmitting RELCOM gatewayResources (H225) SwitchingEquipmentCongestion (Q931: 42)	For the cause receiving RELCOM H225 destinationRejection
RemoteInvalidNumber	remote side	For the cause transmitting RELCOM badFormatAddress (H225) InvalidNumberFormat (Q931: 28)	For the cause receiving RELCOM H225 destinationRejection
RemoteAdaptiveBusy	remote side	For the cause transmitting RELCOM adaptiveBusy (H225) TemporaryFailure (Q931: 41)	For the cause receiving RELCOM H225 destinationRejection
RemoteUserBusy	remote side	For the cause transmitting RELCOM inConf (H225) UserBusy (Q931: 17)	For the cause receiving RELCOM H225 destinationRejection
RemoteUnknown	remote side	For the cause trasmitting RELCOM undefinedReason (H225) NormalUnspecified (Q931: 31) 또는 unspecified reason from remote side	For the cause receiving RELCOM H225 destinationRejection
RemoteCallDeflection	remote side	For the cause trasmitting RELCOM	For the cause receiving

		facilityCallDeflection (H225)	RELCOM H225 destinationRejection
RemoteSecurityDenial	remote side	For the cause trasmitting RELCOM securityDenied (H225)	For the cause receiving RELCOM H225 destinationRejection
RemoteCalledPartyNotRegistered	remote side	For the cause trasmitting RELCOM calledPartyNotRegistered (H225) SubscriberAbsent (Q931: 20)	For the cause receiving RELCOM H225 destinationRejection
RemoteCallerNotRegistered	remote side	For the cause trasmitting RELCOM callerNotRegistered (H225)	For the cause receiving RELCOM H225 destinationRejection
GkCalledPartyNotRegistered	gatekeeper	Gatekeeper ARJ ** cause calledPartyNotRegistered	For the cause receiving RELCOM H225 alledPartyNotRegistered
GkInvalidPermission	gatekeeper	Gatekeeper ARJ cause invalidPermission	For the cause receiving RELCOM H225 noPermission
GkRequestDenied	gatekeeper	Gatekeeper ARJ cause requestDenied	For the cause receiving RELCOM H225 noPermission
GkUndefinedReason	gatekeeper	Gatekeeper ARJ cause undefinedReason	For the cause receiving RELCOM H225 undefinedReason
GkCallerNotRegistered	gatekeeper	Gatekeeper ARJ cause callerNotRegistered	For the cause receiving RELCOM H225 callerNotRegistered
GkRouteCallToGatekeeper	gatekeeper	Gatekeeper ARJ cause routeCallToGatekeeper	RELCOM 송신 시 cause를 H225 unreachableGatekeeper
GkInvalidEndpointIdentifier	gatekeeper	Gatekeeper ARJ cause invalidEndpointIdentifier	For the cause receiving RELCOM H225 undefinedReason
GkResourceUnavailable	gatekeeper	Gatekeeper ARJ cause resourceUnavailable	For the cause receiving RELCOM H225 gatekeeperResources
GkSecurityDenial	gatekeeper	Gatekeeper ARJ cause securityDenial	For the cause receiving RELCOM H225 securityDenied
GkQosControlNotSupported	gatekeeper	Gatekeeper ARJ cause qosControlNotSupported	For the cause receiving RELCOM H225 gatekeeperResources
GkIncompleteAddress	gatekeeper	Gatekeeper ARJ cause incompleteAddress	For the cause receiving RELCOM H225 badFormatAddress
GkAliasesInconsistent	gatekeeper	Gatekeeper ARJ cause aliasesInconsistent	For the cause receiving RELCOM H225 undefinedReason
GkDisengageRequested	gatekeeper	Gatekeeper DRQ	For the cause receiving RELCOM H225 undefinedReason

LocalCallClear	local side	Hang on the local voice port normally	For the cause receiving RELCOM H225 destinationRejection
LocalResourceUnavailable	local side	Lacking ing local resource (example: exceeding the maximum possible number of calls)	For the cause receiving RELCOM H225 gatewayResources
LocalPortBusy	local side	The local voice port is in busy condition	For the cause receiving RELCOM H225 inConf
LocalPortNoConnect	local side	No response from the local voice port(ringing timer expired)	For the cause receiving RELCOM H225 destinationRejection
LocalPortShutdowned	local side	The local voice port is in shudown condition	For the cause receiving RELCOM H225 unreachableDestination
LocalPeerShutdowned	local side	The local dial peer is in shutdown condition	For the cause receiving RELCOM H225 unreachableDestination
LocalInterdigitTimerExpired	local side	The local inter-digit timer is expired	No relevance is found
LocalSecurityDenial	local side	Call termination by the security	For the cause receiving RELCOM H225 securityDenial
LocalInvalidGatekeeperRoute	local side	The transport pass, which a local gateway received from a gatekeeper, is not normal	For the cause receiving RELCOM H225 unreachableGatekeeper
LocalUnreachableGatekeeper	local side	The call canot be processed because the gateway fails to register to the gatekeeper	For the cause receiving RELCOM H225 unreachableGatekeeper
LocalUnreachableDestination	local side	The local gateway fails to connect to the other gateway	No relevance is found
LocalNoAnswerFromDestination	local side	The local gateway fails to receive the first message from toher gateway(T303 Expired)	No relevance is found
LocalNoConnectFromDestination	local side	The local gateway fails to send CONNECT message to the local gateway (T301 Expired)	For the cause receiving RELCOM H225 destinationRejection
LocalUnknown	local side	unknown reason of the local side	For the cause receiving RELCOM H225 undefinedReason
LocalProtocolError	local side	The local side determines protocol and message erroro	For the cause receiving RELCOM H225 undefinedReason
LocalInvalidNumber	local side	invalid number The local side determines an invlid number	For the cause receiving RELCOM H225 badFormatAddress
LocalT38FaxError	local side	The local side determines T.38 fax error	For the cause receiving RELCOM H225 undefinedReason
LocalManagement	local side	The call is terminated by an administrator in the local side	For the cause receiving RELCOM

			H225 undefinedReason
LocalUnavailableDestination	local side	The call is terminated due to the invalid destination (example: FXO – FXO call, H323 – H323 call)	For the cause receiving RELCOM H225 undefinedReason
LocalAbortedDestination	local side	Disconnected with the other gateway in the local side	No relevance is found
LocalCapabilityNegotiationFail	local side	Fails to process capability negotiation with the other gateway from the local side	For the cause receiving RELCOM H225 undefinedReason

*RELCOM : Q.931 Release Complete message

**ARJ : H.225 Admission Reject message

The following table displays H.225 and Q.931 mapping explicated in H.323 of ITU-T recommendations.

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 B References

SIP

RFC References

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2976 The SIP INFO Method. S. Donovan. October 2000. (Format: TXT=17736 bytes) (Status: PROPOSED STANDARD)

3261 SIP: Session Initiation Protocol. J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler. June 2002. (Format: TXT=647976 bytes) (Obsoletes RFC2543) (Updated by RFC3265) (Status: PROPOSED STANDARD)

3264 An Offer/Answer Model with Session Description Protocol (SDP). J. Rosenberg, H. Schulzrinne. June 2002. (Format: TXT=60854 bytes) (Obsoletes RFC2543) (Status: PROPOSED STANDARD)

3265 Session Initiation Protocol (SIP)-Specific Event Notification. A. B. Roach. June 2002. (Format: TXT=89005 bytes) (Obsoletes RFC2543) (Updates RFC3261) (Status: PROPOSED STANDARD)

3311 The Session Initiation Protocol (SIP) UPDATE Method. J. Rosenberg. October 2002. (Format: TXT=28125 bytes) (Status: PROPOSED STANDARD)

3420 Internet Media Type message/sipfrag. R. Sparks. November 2002. (Format: TXT=14745 bytes) (Status: PROPOSED STANDARD)

3515 The Session Initiation Protocol (SIP) Refer Method. R. Sparks. April 2003. (Format: TXT=47788 bytes) (Status: PROPOSED STANDARD)

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H.323

ITU-T Recommendation H.225.0

- Call signalling protocols and media stream packetization for packet-based multimedia communication systems

ITU-T Recommendation H.245

- Control protocol for multimedia communication

ITU-T Recommendation H.323

- Packet-based multimedia communications systems

ITU-T Recommendation Q.931

- ISDN user-network interface layer 3 specification for basic call control

MGCP

RFC References

3264 An Offer/Answer Model with Session Description Protocol (SDP). J. Rosenberg, H. Schulzrinne. June 2002. (Format: TXT=60854 bytes) (Obsoletes RFC2543) (Status: PROPOSED STANDARD)

3435 Media Gateway Control Protocol (MGCP) Version 1.0. F. Andreasen, B. Foster. January 2003. (Format: TXT=467084 bytes) (Obsoletes RFC2705) (Updated by RFC3661) (Status: INFORMATIONAL)

Others

IEEE 802.1Q VLAN

0791 Internet Protocol. J. Postel. Sep-01-1981. (Format: TXT=97779 bytes) (Obsoletes RFC0760) (Updated by RFC1349) (Also STD0005) (Status: STANDARD)

0793 Transmission Control Protocol. J. Postel. Sep-01-1981. (Format: TXT=172710 bytes) (Updated by RFC3168) (Also STD0007) (Status: STANDARD)

0868 Time Protocol. J. Postel, K. Harrenstien. May-01-1983. (Format: TXT=3140 bytes) (Also STD0026) (Status: STANDARD)

1058 Routing Information Protocol. C.L. Hedrick. Jun-01-1988. (Format: TXT=93285 bytes) (Updated by RFC1388, RFC1723) (Status: HISTORIC)

1157 Simple Network Management Protocol (SNMP). J.D. Case, M. Fedor, M.L. Schoffstall, J. Davin. May-01-1990. (Format: TXT=74894 bytes) (Obsoletes RFC1098) (Also STD0015) (Status: HISTORIC)

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1305 Network Time Protocol (Version 3) Specification, Implementation. D. Mills. March 1992. (Format: TXT=307085, PDF=442493 bytes) (Obsoletes RFC0958, RFC1059, RFC1119) (Status: DRAFT STANDARD)

1583 OSPF Version 2. J. Moy. March 1994. (Format: TXT=532636, PS=990794, PDF=465711 bytes) (Obsoletes RFC1247) (Obsoleted by RFC2178) (Status: DRAFT STANDARD)

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1662 PPP in HDLC-like Framing. W. Simpson, Ed.. July 1994. (Format: TXT=48058 bytes) (Obsoletes RFC1549) (Also STD0051) (Status: STANDARD)

1723 RIP Version 2 - Carrying Additional Information. G. Malkin. November 1994. (Format: TXT=18597 bytes) (Obsoletes RFC1388) (Obsoleted by RFC2453) (Updates RFC1058) (Also STD0056) (Status: STANDARD)

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Appendix C. Cable Specifications

This Appendix provides information about the Pin-out specifications of the following cables used with the AP6800 VoiceFinder Gateway .

- Console Port Signal and Pin-out(RJ-45 to DB9)
- Pin-out for UTP Cable (RJ-45 to RJ-45)

[Console Port Signal and Pin-out]

To connect console port of the router and the PC installed with terminal emulation software, use RJ-45 to D89 (Female DTE connector) type:

Console Port (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 C.1 Signal and Pinout of Console Port

[Pinout of Ethernet Cable Assemble(RJ-45 to RJ-45)]

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".

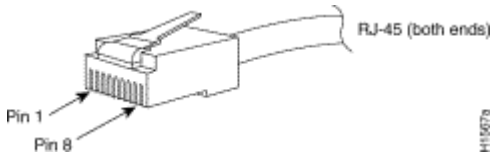

Figure C.1 10Base-T RJ-45 Connector

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

Table C.2 Signal and Pinout Serial Ethernet Cable

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.

Appendix D. Abbreviation and Glossary

Glossary and Abbreviation	Definition
ADSL	Stands for Asymmetric Digital Subscriber Line. If you use ADSL, the central office will be connected to each home directly in a 1:1 method. In a down-link where data is transferred downward from the central office to the users, high-speed data communications of at least 1.5 Mb can be made. On the contrary, in an up-link from the users to the central office, communications are made very slowly. Thus, this service is called an asymmetrical service not a symmetrical service.
AP-VPMS	Stands for VoIP Plug & Play Management Software. This integrated management software developed by AddPac Technology enables VoIP products to be installed in a GUI environment, be monitored in real-time, or to be upgraded. This software also enables network administration.
API	Stands for Application Programming Interface. API is a function call legend standard that defines service interfaces.
APOS	Stands for AddPac Internetworking Operation System. This is an operating system that supports the network products developed by AddPac Technology.
ATM	Stands for Asynchronous Transfer Mode. This is an international cell relay standard for providing a variety of services such as voice, video, and data in the form of a cell of a fixed length (53 bytes). If you use a fixed-length cell, cell processing will be performed in the hardware; thus, transmission delay can be reduced. ATM is designed to make use of high-speed transmission media such as E3, SONET, and T3.
ATM High-Speed National Network	This network has been commercialized by the Korean government since 1993. The high-speed national network designed for governmental offices provides data services (transport network services) and Internet services. Data services are categorized into ATM, dedicated lines, packet exchange, and frame relay services. Internet services are categorized into Internet multi-services provided through ATM connection circuits and simple Internet services.
ATM Forum	This is an international organization founded by Cisco Systems, NET/ADAPTIVE, Northern Telecom, and Sprint in 1991 to reach the agreement of a standard for ATM technologies. ATM Forum expands the formal standards developed by ANSI and ITU-T and the agreements on the implementation of technologies.
Authentication	Operation of verifying the identification of a person or a process. This is a security

	feature.
BNC Connector	This is a standard connector used to connect IEEE 802.3 10Base-2 coaxial cables to Media Access Unit (MAU).
Boot Loader	This is a chip installed into a printed circuit board used to send executable boot commands to a network device.
Bps	Stands Bits per second. Typically called bps. Refer to bit rate.
Cable Modem	This device converts analog signals to digital signals in order to enable the Internet through a cable network. Since telephone networks are made of copper wires and cable networks are made of coaxial and optical cables, the bandwidth of cable networks are much wider than that of telephone networks. However, the modulation/demodulation technology, which converts digital to analog and vice versa, is required for cable networks when data is transferred.
Call Center	Call Center is a central place where calls from customers and other people are processed systematically. Computer automation is implemented in Call Center to some degree. Typically, Call Center processes many calls simultaneously, categorizes calls, connects the calls to personnel, and records calling logs automatically. Call Center is typically used for mail order catalog firms, telemarketing firms, customer centers for PC products, and large enterprises that sell products or provide services.
Caller ID	Caller ID is a call service that enables the phone number of the caller to be sent to the recipient. To see the phone number, a digital reader should be installed into the phone.
Category 5 cabling	One of the five-level UTP cable connection methods specified by the EIA/TIA-586 standard. Category 5 cabling enables data to be transferred at a rate of up to 100Mbps.
CBR	Stands for Constant Bit Rate. The ATM network QoS class CBR defined by ATM Forum is used for a connection device that is based on a precise clock processing method to ensure untwisted data transfer.
CES	Stands for Circuit Emulation Service. This service allows you to multiplex multiple line emulation streams for voice and video with packet data through a single high-speed ATM link without using a separate ATM access multiplexer.
Checksum	This is a method for checking the integrity of transferred data. Checksum is an integer calculated from the octet sequence obtained by a series of operations. This value is calculated by the recipient again for verification.
Coaxial cable	This coaxial cable is made of an external cylinder-type conductor that wraps an internal wire conductor. **Examples** of the coaxial cables used for LAN include 50Ω cables used for digital signal processing and 75Ω cables used for high-speed

	digital signal processing.
CODEC	Stands for COder-DECoder. CODEC is: 1. A built-in circuit device that converts analog signals to digital bit streams and vice versa based on a pulse code modulation method; 2. A DSP software algorithm that compresses or decompresses voice or audio signals over Voice over IP, Voice over Frame Relay, or Voice over ATM.
Console	A DTE interface through which a command enters a host
CoS	Stands for Class of Service. CoS refers to the standard method that enables a higher-level protocol to make a lower-level protocol process messages. For the SNA lower-level area routing, CoS is used to determine the optional path for lower-level area nodes to set a given session. CoS consists of a virtual path number and a transmission priority field. Also called ToS
Decryption	Decryption means restoring data to the original non-encrypted state by applying the encryption algorithm to the encrypted data in reverse.
DHCP	Stands for Dynamic Host Configuration Protocol. DHCP has a mechanism that reassigns an IP address dynamically in order for the host to recycle unnecessary IP addresses.
DNS	Stands for Domain Name Server. This is a server system used for the Internet to convert the name of a network node name to an address.
DS-3	Stands for Digital Signal level 3. This is a frame processing standard used to transmit digital signals at a rate of T3 (44.736Mbps).
DSP	Stands for Digital Signal Processor. This is a dedicated processor that processes only digital signals. DSP is used as a sub-processor for voice processing in NEXT.
DTMF	Stands for Dual Tone MultiFrequency. Two voice-band tones are simultaneously used for dialing (just like touch tones).
E&M	Stands for either recEive and transmit or Ear and Mouth. Typically, this is a trunking device used for switch-to-switch or switch-to-network two-way communications. The analog E&M interface of Cisco is a RJ-48 connector for PBX trunk lines. E&M is available for E1/T1 digital interfaces.
E1	This is a wide area digital transmission technique used mainly in Europe. E1 enables data transfer at a rate of 2,048Mbps. E1 can be lent by regular service providers for a private use.
Encryption	Encryption means that a specific algorithm is applied to data in order to convert data to a form that unauthorized users cannot identify.
Ethernet	Baseband LAN standard initiated by Xerox Corporation and co-developed by Xerox,

	Intel, and DEC. CSMA/CD is used for Ethernet networks, which operate through a variety of cables at a rate of 10Mbps. Ethernet is similar to the IEEE 802.3 standard. Refer to 10Base-2, 10Base5, 10Base-F, 10Base-T, 10Broad-36, Fast Ethernet, and IEEE 802.3.
FAX	Abbreviation of Facsimile. FAX refers to the transmission of scanned texts or images to a printer or an output device connected to another phone number by using a telephone line. Once the original document is read by a facsimile, the facsimile treats the document as a fixed graphic image, and converts it to bitmap. In this digital form, data is transferred in the form of an electrical signal through a phone system. The receiving facsimile restores the data to a encoded image, and prints it on a sheet of paper.
Frame	Logical group of data transferred to a data link layer unit through a transmission medium. From frames, the header and trailer that include user data are important. Headers and trailers are used for synchronization and error control. Cells, datagrams, messages, packets, and segments are used to describe logical data groups in various layers of OSI or based on various technologies.
Frame-Relay	This is an industry-standard switching-type data link layer protocol that processes multiple virtual lines in inter-connected devices by using the HDLC encapsulation. Frame-Relay is more efficient than X.25.
FTP	Stands for File Transfer Protocol. FTP, which is an application protocol, is part of the TCP/IP protocol stack used for file transfer between network nodes. FTP is defined in RFC 959.
FXO	Stands for Foreign Exchange Office. The FXO interface is connected to the switching center of Public Switched Telephone Network (PSTN), and is provided by a regular phone. The FXO interface of Cisco is a station interface of the switching center or PBX on PSTN, and is a RJ-11 connector for analog connection devices.
FXS	Stands for Foreign Exchange Station. The FXS interface is directly connected to a standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX.
G.711	This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation.

G.723.1	This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP, and ensures high flexibility for system designers. This standard is specified under the G-series ITU-T standard.
G.726	This standard specifies ADPCM coding performed at a rate of 40Kbps, 32Kbps, 24Kbps, or 16Kbps. If the PBX network is configured to support ADPCM, you can exchange ADPCM encoding voice with packet voice networks, PSTN, or PBX networks. This standard is specified under the ITU-T standard of G-series recommendation.
G.728	This standard specifies variations that ensure low delay of CELP voice compression performed at 16Kbps. The CELP voice coding should be converted to a public telephony format for transmission over either PSTN or PSTN. This standard is specified under the ITU-T standard of G-series recommendation, and defines the CELP compression that encodes G.729 voice to a stream of 8Kbps. G.728 has two variations (G.729 and G.729 Annex A), and the variations are different in terms of calculation complexity. The two variations have voice quality similar to ADPCM of 32Kbps. G.728 is specified under the ITU-T standard of G-series recommendation.
Gatekeeper	This is the component of the H.323 video conference system that analyzes a caller ID, controls access authorization, and manages the subnet bandwidth. A gatekeeper is H.323 entity that provides the features that enable address conversion and LAN access control to the H.323 terminal and gateway on LAN. Gatekeepers can provide other services such as bandwidth control and search for a gateway to the H.323 terminal and gateway. This device manages a device registry on a multimedia network. The devices are registered with the gatekeeper, and they request the gatekeeper to authorize a call.
H.225	This ITU standard is applied to the session setting and packetization of H.225.0. H.225.0 specifies a variety of protocols such as RAS, Q.931, and RTP.
H.245	This ITU standard is applied to H.245 endpoints control.
H.323	This standard is an extension of the ITU-T standard H.320 that enables voice conferences over LAN or another packet switching network as well as video transmission over the Internet.
HBD3	This is a type of line codes used for E1.

HDLC	Stands for High-Level Data Link Control. HDLC is a transmission protocol used in the data link layer, which is the second layer of the 7-layer OSI model. HDLC is used in the X.25 packet switching network. Data consists of frames in HDLC, and frames are transmitted through a network. The destination verifies if the frames have been successfully transmitted. The HDLC protocol includes data for controlling data flow and troubleshooting errors in a data frame.
Hookflash	This is a short on-hook duration of a device such as phones during a call. Hookflash means that a phone attempts to make a dial tone recall through PBX. This is usually used to perform call transfer.
HTTP	Stands for Hypertext Transfer Protocol. This protocol enables a Web browser or a Web server to transfer files such as text files and graphic files.
IPSec	Stands for Internet Protocol Security protocol. IPSec is a still developing standard for the security of networks or the packet processing layer of network communications. In the previous security techniques, security has been included in the application layers of a communication model. IPSec is particularly useful for the implementation of remote user access through dial-up access to Virtual Private Networks (VPN) and regular private networks. The main advantage of IPSec is that security can be ensured without replacing an individual user PC with a new one. Cisco takes the initiative of suggesting IPSec as the standard, and has embedded support to this feature into its network router.
IPv6	IPv6 is the latest IP, and has been embedded into part of IP support into many products including the operating systems of PC. IPv6 is called IP Next Generation (IPng), that is the next-generation IP. IPv6 is the formal IETF standard. IPv6 is designed as an evolutional version of the currently used IP version 4. Network hosts or intermediate nodes that adopt either IPv4 or IPv6 can process any packets formulized by either IPv4 or IPv6; thus, the users and service provides can upgrade their IP to IPv6 individually without collaboration.
ISP	Stands for Internet Service Provider. ISP refers to service providers that provide Internet access services, Web site construction and Web hosting services to individuals or enterprises. ISP has devices and communication lines required for Internet access, and large ISPs have their own high-speed dedicated lines in order to provide services that have better quality and are less dependent on telephone network service providers to their customers. The large nationwide ISPs of the U.S. are AT&T WorldNet, IBM Global Network, MCI, Netcom, UUNet, and PSINet. Those of Korea are INet, Channeli, Netsgo, and Netian. The users access the Internet through online service

	providers. The main online service providers of the U.S. are America Online and Compuserve, and those of Korea are Chollian, Unitel, and Hitel.
ITU-T	Stands for International Telecommunication Union Telecommunication Standardization Sector. This is an international organization that develops global standards on communication technologies. ITU-T performs the previous tasks of CCITT.
IVR	Stands for Interactive Voice Response. IVR refers to a system that provides data in the form of recorded messages through phone lines as a response to user input in the form of human voice or mainly DTMF signal processing. **Examples** are banks that allow you to check balance by using a phone or automated stock quotations system.
LAN	Stands for Local Area Network. This is a low-error, high-speed data network that covers relatively small geographical areas of up to several thousand meters. LAN inter-connects workstations, peripherals, terminals, and other devices in a building or a geographically limited area. The LAN standard specifies a cable connection and signal processing method in the physical layer and data link layer of the OSI model. Reference: MAN, WAN.
Link	This is a network communication channel configured with lines or a transmission path between the transmitter and receiver and related devices. A link mainly refers to WAN connections, and is sometimes called a line or a transmission link.
Loopback Test	This test is performed as follows: Transmit a signal or return it to the transmitter at a location on the communication path. This loopback test is usually performed to test the availability of network interfaces.
MAC Address	Stands for Media Access Control Address. This is a standard data link layer address required for any and all ports and devices connected to LAN. Other devices on a network use this address to locate a specific port within the network and to create or update a routing table and data structure. A MAC address is 6 bytes long, and is managed by IEEE. A MAC address is called as a hardware address, a MAC-layer address, or a physical address. Compare to: Network Address
MAN	Stands for Metropolitan-Area Network. This network covers the entire area of a large city. The operation area of MAN is geographically larger than that of LAN; however, is smaller than that of WAN. Compare to: LAN, WAN.
MGCP	MGCP, which is also known as H.248 or Megaco, is a standard protocol required to operate signals required during a multimedia conference or to manage sessions. This protocol defines a method of communications between the media gateway that

	converts the data format required for a circuit switching network to the one required for a packet switching network and the media gateway control device. MGCP may be used to set up, manage, and complete calls among multiple endpoints. Megaco and H.248 are the improved version of MGCP.
NAT	Stands for Network Address Translation. NAT is a mechanism for reducing the need for globally unique IP addresses. NAT allows you to access the Internet as an organization whose address is not globally unique converts the address to an address space where the address can be globally routed. NAT is also called Network Address Translator.
NTP	Stands for Network Time Protocol. NTP, which is built based on TCP, sets a local time accurately based on a wireless clock and an atomic clock on the Internet. NTP can synchronize a distributed clock in the unit of milliseconds for a long time.
PABX	Stands for Private Automatic Branch eXchange. PABX is a switch for phones used at enterprises. PABX is used in Europe, while PBX is used in the U.S.
Packet	A packet is a group of logical data that contains user data and a header where control data is contained. A packet mainly refers to the unit of network layer data.
PBX	Stands for Private Branch eXchange. PBX, which is located in a subscriber building, is a digital or analog phone switchboard used to connect private networks to public phone networks.
PING	Stands for Packet INternet Groper. ICMP echo-processes a response between messages. PING is used for an IP network to test the accessibility of network devices.
Point to Point Connection	One of the two basic connection types. In ATM, the point to point connection may be either a one-way connection or a two-way connection between two ATM end systems.
Pont to Multipoint Connection	One of the two basic connection types. In ATM, the point to multipoint connection is a one-way connection method that enables a transmitting end-system (root node) to be connected to multiple receiving end-systems (riff). Compare to: Point to Point Connection
POTS	Stands for Plain Old Telephone Service. Reference: PSTN.
PPP	Stands for Point-to-Point Protocol. This protocol is the advanced version of SLIP that enables a router-to-router connection or a host-to-network connection through synchronous or asynchronous lines. SLIP is designed to be used on an IP, while PPP is used along with network layer protocols such as IP, IPX, and ARA. PPP has a bulletin board security mechanism such as CHAP and PAP. PPP has two sub-protocols, LCP and NCP. Reference: CHAP, LCP, NCP, PAP, and SLIP

Protocol Stack	This is a collection of communication protocols that inter-work with one another and that process communications in part or all of the seven layers of the OSI reference model. All protocol stacks are not related to each layer of the OSI model, and one protocol of a stack can process multiple layers at one time. TCP/IP is a typical protocol stack.
PSTN	Stands for Public Switched Telephone Network. PSTN is a general term that refers to various telephone networks and services used worldwide. PSTN is also called POTS.
PVC	Stands for either Permanent Virtual Circuit or Permanent Virtual Connection. PVC is a virtual circuit installed permanently. PVC allows you to reduce a bandwidth for setting up or releasing a circuit when a specific virtual circuit must always exist. As an ATM term, PVC is called Permanent Virtual Connection.
Q.931 Signaling	This is an ITU standard that specifies ISDN signal processing methods. The H.225.0 standard uses a variation of Q.931 to set up or disconnect the session of H.323.
QoS	Stands for Quality of Service. QoS is the criterion of measuring the performance (e.g. transmission quality and service availability) of a transmission system.
RAM	Stands for Random-Access Memory. RAM is a volatile memory that can be read or written by a microprocessor.
RAS	RAS refers to a protocol for registration, connect acknowledgement, and status protocol. RAS is used for H.323 to find or have a conversation with a gateway.
RISC	Stands for Reduced Instruction Set Computing.
Router	This is a network layer device that determines the optional route to which network traffic is delivered by using one or more metrics. A router forwards packets from a network to another network based on the network layer information. A router is sometimes called a gateway. (A gateway in this meaning is getting older.) Compare to: Gateway; Reference: Relay
RS-232	This is a frequently used physical layer interface, and is known as EIA/TIA-232 nowadays.
RTCP	Stands for RTP Control Protocol. This protocol monitors the QoS of IPv6 RTP connections, and transfers data on sessions in operation. Reference: Real-Time Transport Protocol (RTP)
RTP	1. Stands for Routing Table Protocol. This VINES routing protocol based on RIP distributes network topology data, and helps the VINES server that searches for adjoining clients, servers, and routers. A delay time is used as a routing metric. Reference: SRTP

	2. Stands for Rapid Transport Protocol. RTP provides facing and error recovery services to the APPN data when the data passes the APPN network. RTP allows you to check error recovery and flow control synthetically. RTP does not recover but prevents traffic congestion. 3. Stands for Real-Time Transport Protocol. This is one of the IPv6 protocols. RTP is designed to enable the synthetic network transmission feature in the application that transfers real-time data such as audio, video, and simulation data through multicast or unicast network services. RTP enables the real-time application to identify a payload type, specify a sequence number, perform time-stamping, and to monitor a transmission procedure.
SIP	Stands for Session Initiation Protocol. SIP is an application layer control protocol based on very simple texts, and allows more than one user to make, correct, or complete a session. **Examples** of sessions include remote conferences, phones, meetings, event notifications, and instant messaging on the Internet. SIP is independent to lower-level packet protocols (e.g. TCP, UDP, ATM, and X.25).
SmartViewer	This is software that allows you to monitor AP-GK1000, AP-GK2000, and AP-GK3000, which are the gatekeeper series of AddPac Technology, in a Graphical User Environment (GUI) environment in real-time and to search or manage statistical data.
SNMP	Stands for Simple Network Management Protocol. This is a network management protocol almost dedicated to TCP/IP networks. SNMP monitors and controls network devices, and manages setup, collection of statistical data, operation performance, and security features. Reference: SGMP and SNMP2
T1	This is the facility of a digital WAN service provider. T1 uses the AMI or B8ZS coding method to transfer DS-1 format data at a rate of 1.544Mbps over a phone switching network. Compare to: E1; Reference: AMI, B8ZS, DS-1
TCP/IP	Stands for Transmission Control Protocol/Internet Protocol. TCP/IP is a general name of the protocol suites developed in the seventies by DoD of the U.S. to help build a global inter-network. TCP and IP are two of the best known protocol suites. Reference: IP and TCAP
Telco	Stands for Telephone Company. Telco refers to a telephone service provider. Typically, Telco means individual local telephone service providers such as Bell, and sometimes includes long distance telephone service providers.
Telnet	This is a standard terminal emulation protocol included in the TCP/IP protocol stacks.

	Telnet is used to connect remote terminals. Telnet allows you to log into a remote system and to use the resources like they are connected to a local system. Telnet is defined in RFC 854.
VCI	Stands for Virtual Channel Identifier. VCI refers to a 16-bit field in the header of an ATM cell. VCI as well as VPI allows you to identify the next receiver of a cell while the cell is being delivered to the receiver through a series of ATM switches. The ATM switches use the VPI/VCI field to identify the next network VCI that the cell should pass to reach the receiver, which is the final destination. The features of VCI are similar to those of DLCI.
VDSL	Stands for Very-high-data-rate Digital Subscriber Line. VDSL is one of the four DSL technologies. VDSL provides downstream of 13 Mbps to 52 Mbps and upstream of 1.5Mbps to 2.3Mbps through a pair of twisted copper wires. The operation range of VDSL is limited to 1,000ft to 4,500ft (304.8m to 1,372m). Compare to: ADSL, HDSL, and SDSL
VoATM	Stands for Voice Over ATM. VoATM enables a router to deliver voice traffic (e.g. phone calling or facsimile) over an ATM network. Voice traffic is encapsulated in a specific AAL encapsulation method for multiplexed voice when voice traffic is sent in ATM.
VoFR	Stands for Voice Over Frame Relay. VoFR enables a router to deliver voice traffic (e.g. phone calling or facsimile) over a frame relay network. When voice traffic is sent through frame relay, the voice traffic is encapsulated after being decomposed into segments by using the FRF.12 encapsulation technique to pass the frame relay network.
VoHDLC	Stands for Voice over HDLC. Voice over HDLC enables a router to deliver live voice traffic (e.g. phone calling and facsimile) to another router through a serial line.
VoIP	Stands for Voice over IP. VoIP is a capability that enables normal telephony voice of the same features, reliability, and voice quality as POTS to deliver over the IP-based Internet. VoIP enables a router to deliver voice traffic (e.g. phone calling and facsimile) over an IP network. Over VoIP, DSP decomposes voice signals into frames, and a pair of the decomposed frames is grouped. Then, the grouped frames are saved in a voice packet. The voice packet is forwarded by using an IP under the ITU-T standard, H.323.
VPN	Stands for Virtual Private Network. VPN allows you to encrypt entire traffic that moves from a network to another network so that IP traffic can safely move over a

	public TCP/IP network. On VPN, all data is encrypted in an IP level by using the 'tunneling' technique.
WAN	Stands for Wide-Area Network. WAN is a data communication network that provides services to the users in a wide area and that uses transmission services provided by regular service providers. **Examples** of WAN include frame relay, SMDS, and X.25. Compare to: LAN and MAN