AP-IP200 IP Phone Installation Guide

[Data, Voice, Video & IP Telephony Solution]

August, 2005

AddPac Technology Co., Ltd.

Technical Sales Division

www.addpac.com

AP-IP200 IP Phone

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Getting Into AP-IP200 IP Phone Installation Guide

This chapter explains the AP-IP200 IP phone installation guide.

[Contents of AP-IP200 Installation Guide]

This guide assists users to install the AP-IP200 IP Phone easily. This guide is composed of six chapters as following. If user has previous experience to use IP Phone, please refer to the chapters user want to know directly. But, if user has no experience to use IP Phone, it is highly recommended to thoroughly understand the manual before operation of this IP Phone.

- Chapter 1 『AP-IP200 overview 』 provides an introduction to the hardware and software features of AP-IP200 and technical specification.

- Chapter 2 『Before Installation 』 explains the installation environment and cable requirements along with recommendations for safe operation of the equipment.

- Chapter 3 『AP-IP200 Installation』 explains how to connect cables, audio devices such as Headset, Earphone, MIC and basic installation information.

- Chapter 4 『AP-IP200 UI Operating Guide』 explains the AP-IP200 UI(User Interface) operating guide using Keypad.

- Chapter 5 『AP-IP200 Network Configuration』 explains the detailed network configuration through AP-IP200 CLI(command line interface) command (RS-232C).

- Chapter 6 『Appendix』 provides the detailed LAN cable and RS-232C console cable pin assignment specifications for AP-VP300 video phone and warranty.

For technical support about AP-IP200 IP Phone, contact the sales department of AddPac Technology Co. Ltd.

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The revision history of AP-IP200 IP Phone installation guide is as follows.

<Table 1-1> Revision History of AP-IP200 IP Phone Installation Guide

Revision No.	Date	Contents	Written By
Version 1.00	July 20th, 2005	Initial Released	AddPac R&D Center

Chapter 1. AP-IP200 Overview

AP-IP200 IP phone is designed to provide enhanced IP telephony functionality to meet the wide range of business user requirements. This IP telephone optimally delivers rich featured voice telephony service on ordinary internet infrastructure as well as AddPac IP-PBX environment on local LAN as a fully featured IP extension for the complete AddPac VoIP solution.

This new and versatile IP telephone brings the integrated solution for the IP based voice communication and the broadcasting feature to maximize business potentials. It provides the advanced IP telephony device features such as LCD screen, wide variety of feature keys, customizable hot-keys, two(2) ethernet ports, the latest QoS, public IP sharing. It supports not only the major VoIP signaling protocols such as SIP, H.323, MGCP but also G.711, G.726 voice codec, stereo audio in/out interfaces for external Headset MIC. Etc

New Paradigm for IP Telephony: Telephony + Broadcasting

AP-IP200 IP telephone combines AddPac's field proven VoIP technology and IP voice broadcasting technology. AP-IP200 is market-ready IP telephone which provides a full suit of remarkable functionality compared to other typical IP telephones. Apart from telephony service, it delivers IP voice broadcasting service supporting external MIC/Line-in, Line-out interface for various input/output devices such as headset, Amp or speaker. In addition, it provides high quality display with compact LCD mounted. Since AP-IP200 supports diverse voice codecs according to bandwidth environment, it can be deployed anywhere on the internet, ensuring optimal voice quality by leveraging the latest QoS technology. Furthermore, installed along with IPNext500 and IPNext1000 on AddPac's comprehensive IP-PBX system, it not only improves operation offering an wide variety of features such as Music on Hold, Coloring service, Call Transfer but also provides the easy-to-use, intelligent IP telephony service enhanced by AddPac's unique PC-based User Agent.

Adapt to the Future Environment : Firmware Upgradeable Technology

Designed on programmable high performance RISC CPU and DSP, AP-IP200 is capable of adopting new capabilities and improvement by downloading firmware from website or with its auto-upgrade option as the customers' needs grow. Moreover, operators can download the latest protocol or service improvements as well as update firmware by checking the version and activating the auto-upgrade while AddPac's IP-PBX power on/booting sequence.

Compelling Supplementary Services: Extending Benefit of IP Telephony

AP-IP200 delivers not only fully featured IP telephony services, but also various supplementary services to users. It features advanced phone directory, voice mail, CID(Caller ID), call transfer on site or at a remote site. One of its greatest services is IP broadcasting feature which enables AP-IP200 to offer voice broadcasting service, incorporated with in-house broadcasting system.

Seamless Stability and Service Consistency

AP-IP200 features 1-FXO port (optional) equipped avoiding operation failure caused by network error or proxy server/gatekeeper connection error. It supports both automatic and manual PSTN backup feature to maintain constant operation.

IP telephony with Outstanding Network Service Capability

Not only IP telephony, AP-IP200 is an integrated, feature-rich network equipment delivering routing, NAT/PAT, DHCP Server/Relay, Public IP sharing, VRRP and QoS. In today's mixed network of xDSL, Cable, FTTH, Metro Ethernet, Metro ATM, Leased line and dynamic IP environment, not only the ample network service features, but also high-end QoS (Quality of Service) and security features are requested. Based on two (2) 10/100Mbps Fast Ethernet ports, AP-IP200 offers integrated network and security service of LAN-to-LAN routing, bridge and NAT/PAT. Moreover, AP-IP200 supports H.323, SIP, MGCP signaling protocols concurrently. So the customers easily migrate to different service providers' networks utilizing different VoIP signaling protocols.

Privacy and Encryption Features

AP-IP200 brings the network security and service security as well. With the built-in CID (Caller ID Detection) feature, user is able to know who is calling before he answers and block the incoming call. Moreover, It supports SRTP protocol by encrypting exposed voice signal to avoid being fragile to hacking or wiretapping.

AddPac's various VoIP gateway series, multi service routers and comprehensive family of cutting-edge solutions have delivered high performance and stability to maximize customer satisfaction throughout the world. They provide high level of flexibility and scalability for each organization to find the solution that best fits their application needs and budget. With years of experience and industry-leading technology, AddPac provides AP-IP200 with which customers can best optimize high performance, market strategy and budget for next-generation communication solution.

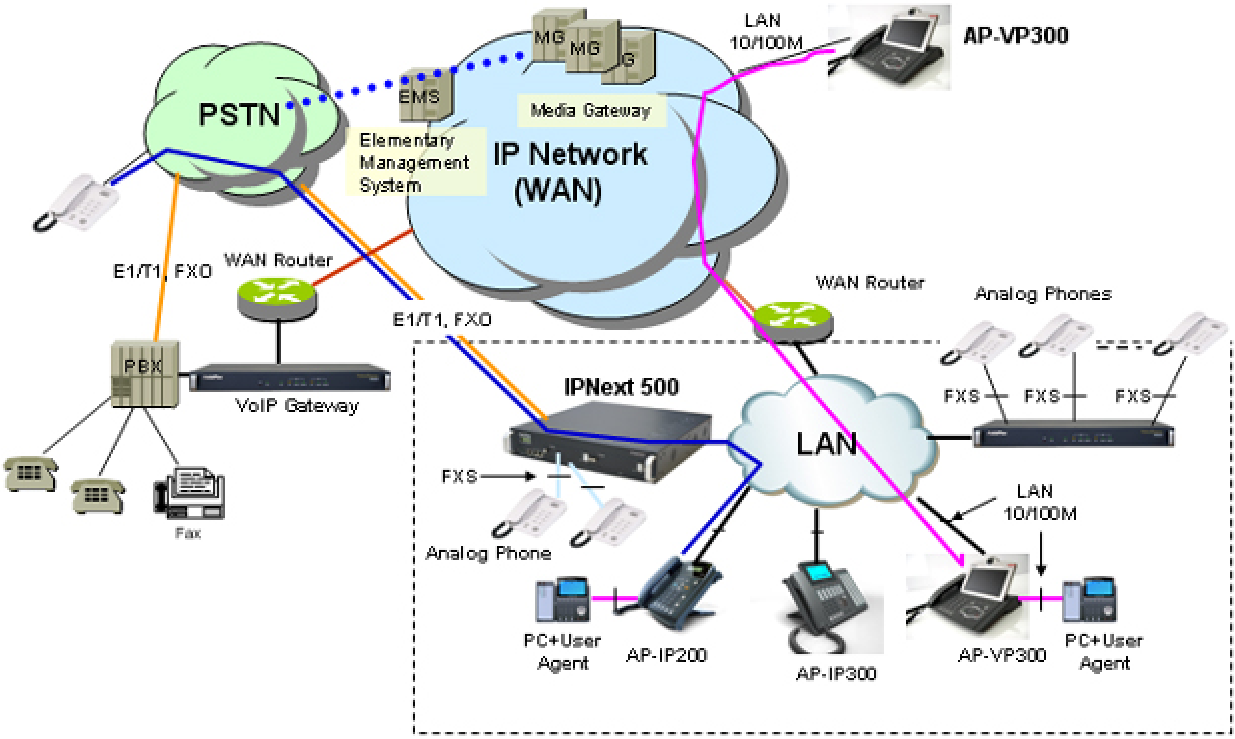

Figure 1-1 AP-IP200 IP Phone Network Diagram

AP-IP200 IP Phone Hardware Specification

[Table 1-1] AP-IP200 IP Phone HW Specification

Category		Specification
Model		**AP-IP200**
Product category		IP Phone (Speaker Phone, Mute, Headset Interface)
Microprocessor		High Performance RISC Microprocessor
Digit and KEY Button		3 x 4 Standard Button and 17 Menu/Function KEY
LCD Display	Graphic LCD	2 Line Graphic LCD
Memory	System Memory	4MB Flash Memory
	Main Memory	16/32MB High Speed SDRAM 16M : APOS 1.0 Stack : IPv4 Support only 32M : APOS 2.0 Stack : IPv4/IPv6 Support
Audio Interface for Headset	Audio Input	1-Port 3.5mm Male Stereo-In Audio Input for Mic, etc.
	Audio Output	1-Port 3.5mm Male Stereo-Out Audio Output for Earphone, etc.
PSTN Backup (Option)	FXO VoIP Interface	1-Port RJ-11 Connector
Ethernet Interface	LAN 0 Port for WAN	1-Port 10/100Mbps Fast Ethernet RJ-45 Connector
	LAN 1 Port for PC	1-Port 10/100Mbps Fast Ethernet RJ-45 Connector
Console Interface	RS-232C Serial Port	1-Port RJ-45 Connector
Power Supply	External Power Supply	External AC110~220V 50/60Hz, 5V, 2A Power Supply
	Power over Ethernet(Option)	PoE(Power over Ethernet) Support via LAN Port
Hardware Chassis	Material	ABS Material/Compact Phone Chassis

AP-IP200 IP Phone Software Specification

[Table 1-2] AP-IP200 S/W Specification

Category	S/W Specification
LAN Protocol	Static and IEEE 802.1Q VLAN Routing,
WAN Protocol	Point-to-Point Protocol (PPPoE for ADSL), etc.
Audio Service & Signaling Protocol	Voice Codec - G.711, G.723.1, G.726, G.729, etc.
	H.323, SIP, and MGCP Triple Concurrent Stack Support
	ITU-T H.323 v3 VoIP Protocol with ITU-T H.235 Security Feature
	Voice Processing Features Supports - VAD, DTMF, CNG, G.168 and T.38 FAX Relay
	ITU-T H.323 Gateway, Gatekeeper Support
	Enhanced QoS Management Features for Voice Traffics
Network Management	Standard SNMP Agent (MIB v2) Support
	Traffic Queuing and Frame-Relay Flow Control
	Remote Management using Console, Rlogin, Telnet
	Web based Managements using HTTP Server Interface
Security Functions	Standard & Extended IP Access List
	Access Control and Data Protections
	Enable/Disable for Specific Protocols
	Multi-Level User Account Management
	Auto-disconnect for Telnet/Console Sessions
	PPP User Authentication Supports → Password Authentication Protocol(PAP) → Challenge Handshake Authentication Protocol (CHAP)
Operation & Management	System Performance Analysis for Process, CPU, Connection I/F
	Configuration Backup & Restore for APOS Managements
	Debugging, System Auditing, and Diagnostics Support
	System Booting and Auto-rebooting with Watchdog Feature
	System Managements with Data Logging
	IP Traffic Statistics with Accounting
Other Scalability Features	DHCP Server & Relay Functions
	Network Address Translation (NAT) Function
	Port Address Translation (PAT) Function
	Transparent Bridging (IEEE Standard) Function

	→ Spanning Tree Bridging Protocol Support → Remote Bridging Support → Concurrent Routing and Bridging Support
	Cisco Style Command Line Interface(CLI)
	Network time Protocol(NTP) Support

[Table 1-3] AP-IP200 Network Protocol Specification

Category	S/W Specification
Basic Network Protocols	ARP, IPv4, IPv6, TCP, UDP, ICMP, ICMPv6, SCTP, IGMP, MLD
Routing Protocol	IPv4 : Static
	IPv6 : Static
Service Protocol	FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
	CDP (Cisco Discovery Protocol)
	DNS Resolver , DDNS(nsupdate)
	Bridge
	Syslog
	IPv4/IPv6 policy control (QoS)
	VPDN (Virtual Private Dial-up Network : L2TP Server)
IPv4/IPv6 Interworking	NAT/PAT for IPv4
	IP connect (formerly ip-share) and device cascade for IPv4
	IP/IP, IP/GRE tunneling
	NAT-PT
	6 to 4, Autoconfig tunneling
IPv4 Address Configuration	Fixed (Static)
	DHCP
	PPPoE
IPv6 Address Configuration	Fixed (Static)
	EUI-64
	Autoconfig (Neighbor Advertisement and Solicitation)
Miscellaneous	Standard & Extended IPv4/IPv6 Access List
	Multi-level User Account Management
	IP accounting
	fsh (Embedded file system shell)

	STUN Client

AP-IP200 IP Phone Front Part

This chapter explains the front part's DIAL and FUNCTION KEY of AP-IP200 IP Phone. AP-IP200 IP Phone's external CASE is made of high degree of solidity ABS. Main key buttons are equipped on front part so that user can operate all the functions with these buttons. The headset interface such as Mic Jack., Ear phone Jack is located at lower left side of AP-IP200. The external MIC for speaker phone is located at lower right side of AP-IP200.

Figure 1-2 shows the KEY arrangement diagram of AP-IP200 front part.

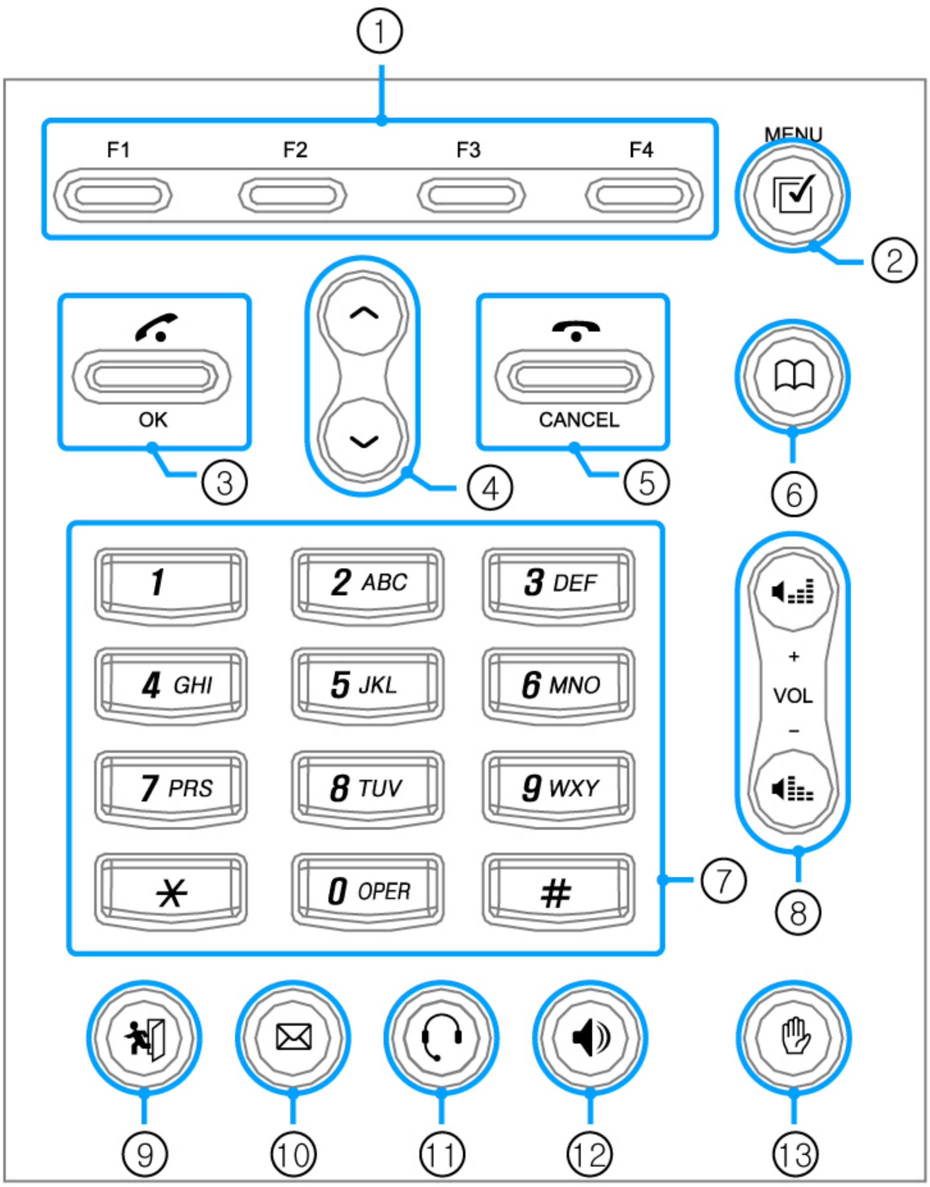

Figure 1-2 AP-IP200 Front Part KEY arrangement diagram

Following Table 1-4 explains the KEY button's role of AP-IP200 front side each.

[Table 1-4] The KEY button's role of AP-IP200 IP Phone Front Part

No.	KEY button	Function
(1)	F1~F4	Function key for AP-IP200 UI(User Interface) on Compact LCD Example F1: Backspace KEY, F2: Space KEY at Edit Mode in Phonebook, Speed Dial UI. Used at HookFlash button for call transfer at conversation.
(2)	MENU	Enter the UI Main Menu
(3)	OK	Confirm the menu setting, Check the Recent Call at on hook and Dialing
(4)	Direction key	Direction keys for menu change in UI.
(5)	CANCEL	Move on to upper menu from current UI menu or cancel the current VoIP call.
(6)	Phonebook	Used for jumping Phonebook menu in UI.
(7)	Numeric Key	Used for Dialing and parameter setting in UI.
(8)	Volume	On-hook mode: Adjust Ring Volume At conversation: Adjust Output Volume Volume Setting: Adjust Ring Volume, Input Volume, Output Volume.
(9)	Absence	Used at Absence Mode
(10)	Voice Mail	Used at Voice Mail Mode
(11)	HDP Call	This KEY is used for VoIP call via Headphone Interface. If this button is pressed, blue LAMP is turn on.
(12)	SPK Call	The key is used for VoIP call via speaker phone. If this button is pressed, blue LAMP is turn on.
(13)	Privacy	Used for MUTE at conversation.

AP-IP200 IP Phone Rear Part

Rear part consists of FXO PSTN backup interface, Handset connector interface, RS-232C RJ45 interface for Command Line Interface, power switch and connector including two (2) Fast Ethernet for WAN/LAN interfaces.

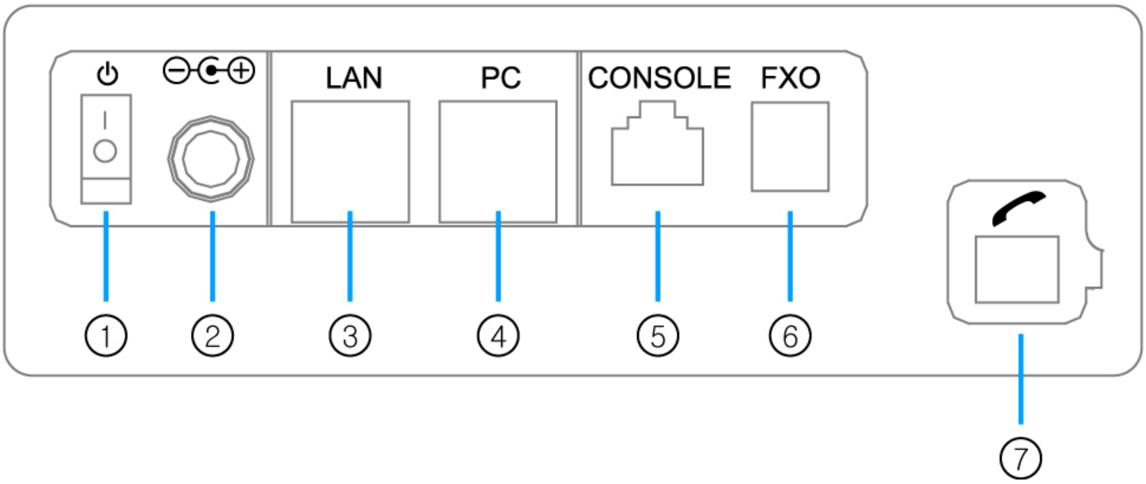

Figure 1-3 AP-IP200 Rear Part Connector Diagram

Table 1-5 explains the AP-IP200 rear part connector interface.

[Table 1-5] AP-IP200 IP Phone Rear Part Interface

No.	Interface	Description
(1)	Switch	External Power ON/OFF switch
(2)	DC 5V 2A	External Power Adaptor connector (DC 5V 2A)
(3)	LAN	10/100Mbps Fast Ethernet Interface for WAN such as ADSL, Leased Line, etc (RJ45)
(4)	PC	10/100Mbps Fast Ethernet Interface for LAN (RJ45)
(5)	CONSOLE	RS232C interface for CLI network management (RJ45)
(6)	FXO	1-Port FXO PSTN Backup Interface
(7)	Handset Connector	RJ11 Handset Connector Interface

Chapter 2. Before Installation

Installation Requirement

The followings are the recommendation for safe operation of the equipment.

- Ensure AP-IP200 IP Phone is in a dust-free environment before and after installation.
- Ensure AP-IP200 IP Phone upper part is empty on a flat and safe surface.
- To prevent accidents, avoid ties, scarf, sleeves, and any other loose clothing from entangling with the chassis.
- Avoid any actions that may lead to the malfunction of the equipment or the operator.

Electrical Requirement

There are two main sources of electrical problems with AP-IP200 IP Phone : the power supply and static electricity.

This section describes safety recommendations for each case. .

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

- **Prevention of Static Electricity**
  - ✓ The main chip-set of the Videophone is very delicate and misuse may result in static electrical damage.

General Requirement

The AP-IP200 IP Phone is ready for use where electronic products are used. However, locations with the following conditions are recommended for maximum performance.

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

Prerequisites for AP-IP200 Installation

User should consider the EMI standards and distance limitations (EIA recommendation) when installing the AP-IP200 IP Phone.
The following section describes the Ethernet cable and the RS-232C console cable AP-IP200 supports.

Prerequisites for Installation

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

Cable for LAN and Console port connection

RJ-45 to RJ-45 cable for LAN port

RS-232C console cable with RJ-45 connector (included in equipment packing box)

Ethernet port

AP-IP200 IP Phone has two(2) RJ45 Fast Ethernet ports in rear side. In case that you want to connect LAN using these ports, please use right cable and connector. See the cable specification in Appendix for Ethernet cable pin specifications.

Console port

AP-VP200 IP Phone has one RJ-45 type RS-232C connector interface in rear side. It can be used for AP-IP200 initial configuration, equipment monitoring and debugging. You must use a cable and a connector. Refer to cable specification in Appendix on RS-232C console cable pin specifications.

Unpacking

Before unpacking, check for external damage of the packaging box.
If no external damage has been found, confirm if the following items are enclosed

[Table 2-1] The contents of AP-IP200 IP Phone package box

No	Items	Contents	Q'ty
1	AP-IP200 IP Phone Main Body		1
2	LAN cable (RJ45 to RJ45)		1
3	Console port cable (RJ45 to DB9)		1
4	External power Adaptor (Free Voltage, DC 5V, 2A)		1

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

Chapter 3. AP-IP200 Installation

• Connecting Ethernet Interface

- Connect AP-IP200's LAN interface to LAN interface of WAN equipment (Router or ADSL/Cable modem) with RJ45 UTP cable.
- There might be some cases of direct connection to router or modem with cross-over cable.
- Please use direct-through cable to connect to HUB.

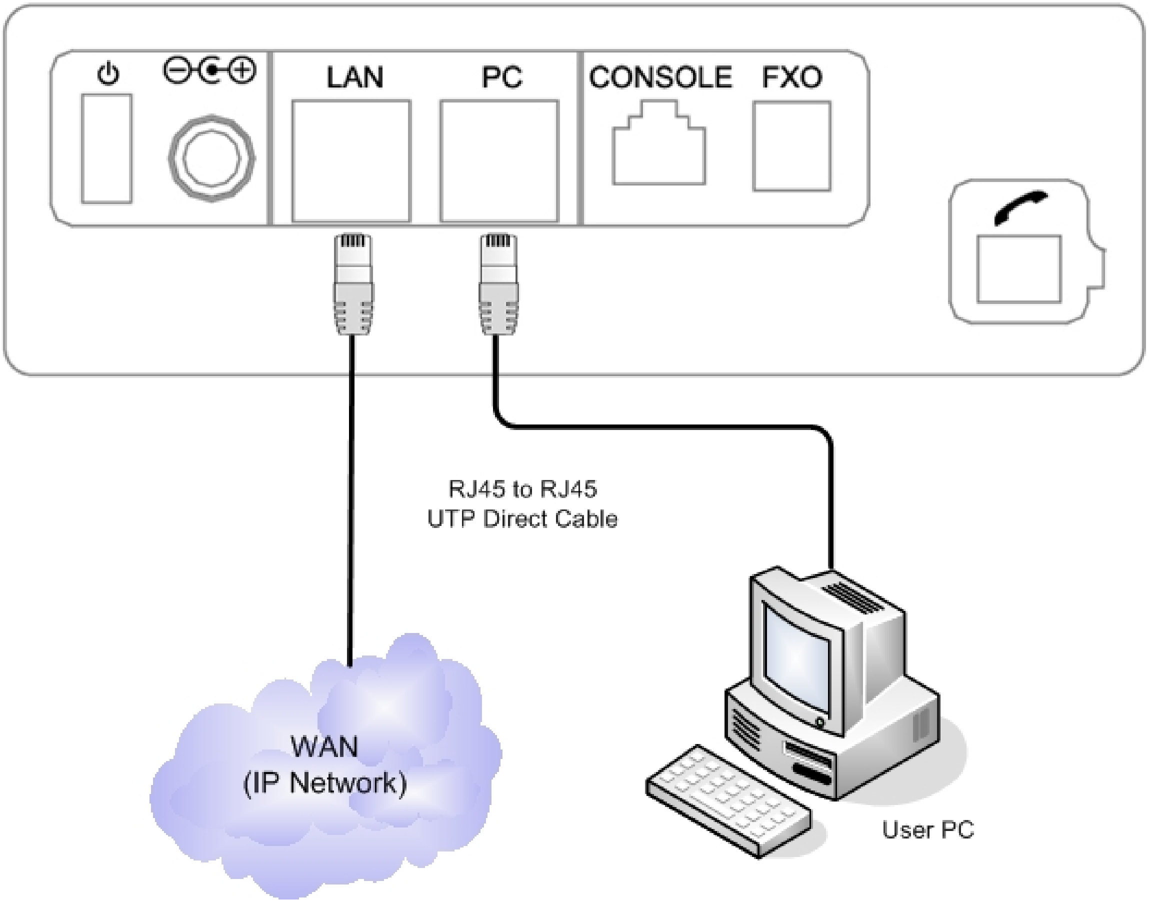

Figure 3-1 Network Interfaced Diagram of AP-IP200 IP Phone

- AP-IP200 IP Phone's Fast Ethernet PC Interface is supposed to be connected into Desktop PC's LAN Port with Direct-Through cable in IP-Share mode and to be connected into HUB in NAT/PAT or Bridge mode.
- In case of connecting directly to Desktop PC's LAN Port, please use Direct-Through cable.
- In case of connecting directly to HUB, please use Cross-over cable.

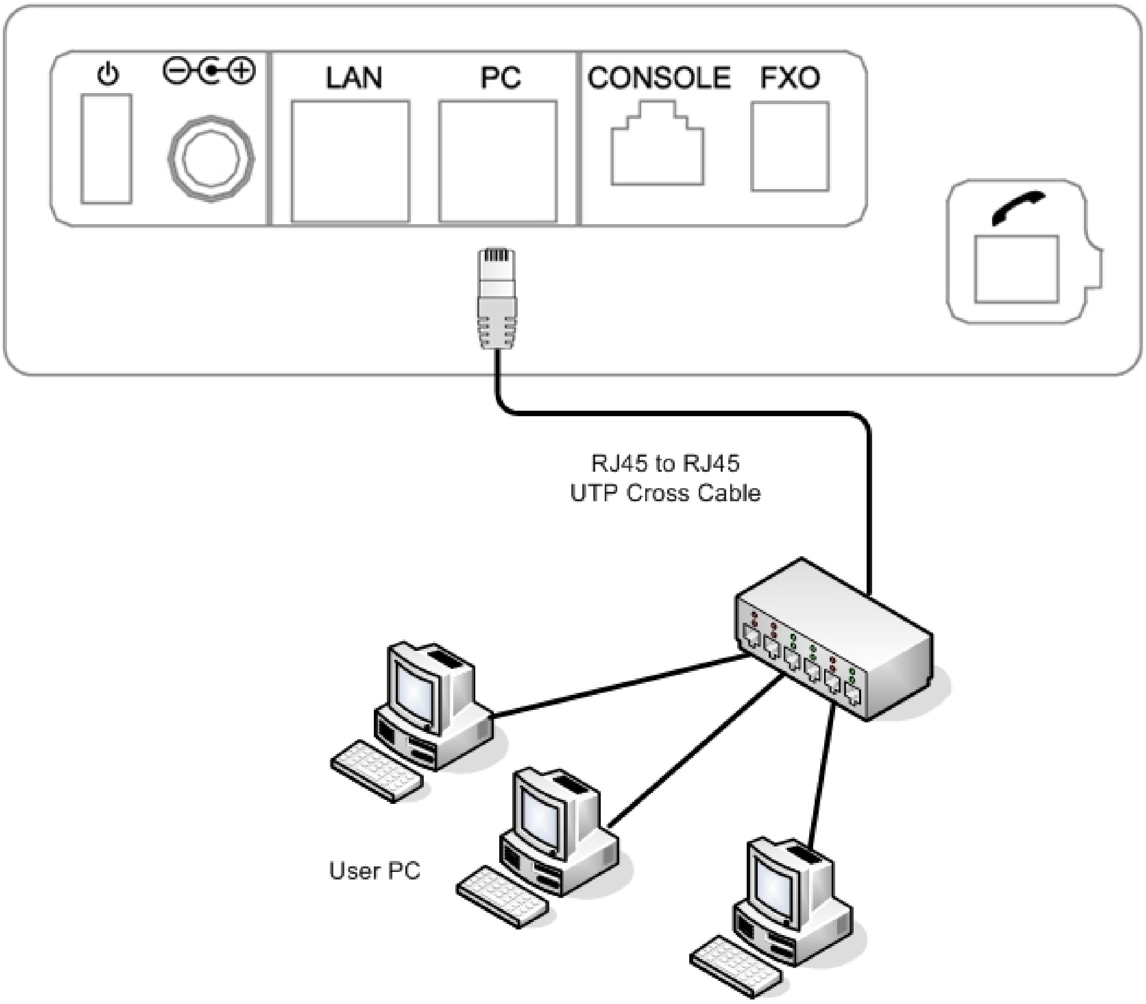

Figure 3-2 LAN Interface Diagram of AP-IP200 IP Phone

Connecting PSTN (FXO) interface

The FXO PSTN interface port is available when PSTN access-line is used or impossible to make a VoIP call due to network problem. PSTN backup is implemented by connecting PSTN access-line to PSTN port, illustrated as following figure.

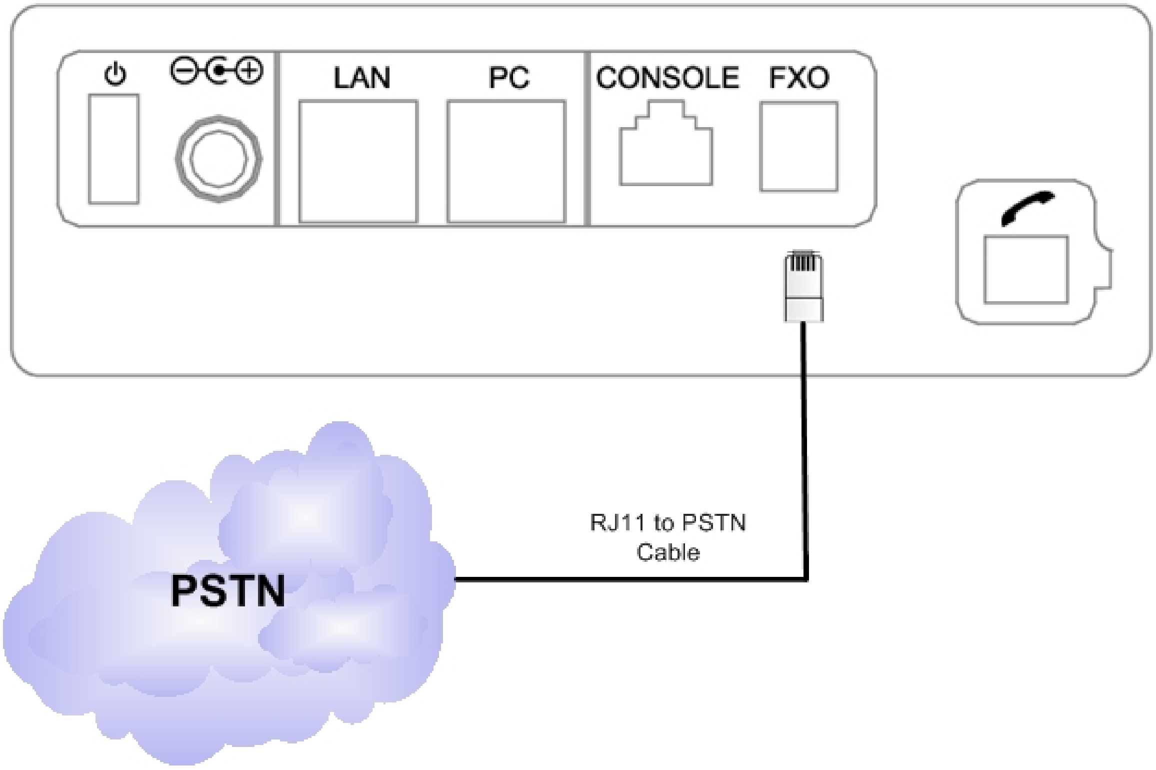

Figure 3-3 PSTN FXO Interface Diagram of AP-IP200 IP Phone

Connecting Audio-In/Out Interface for Headset

Audio-In/Out port located at left side of AP-IP200 IP Phone is for audio devices such as MIC, Speaker System or Headset Device etc.

Connect this port to MIC system or External Speaker System using '3.5mm stereo jack' cable.

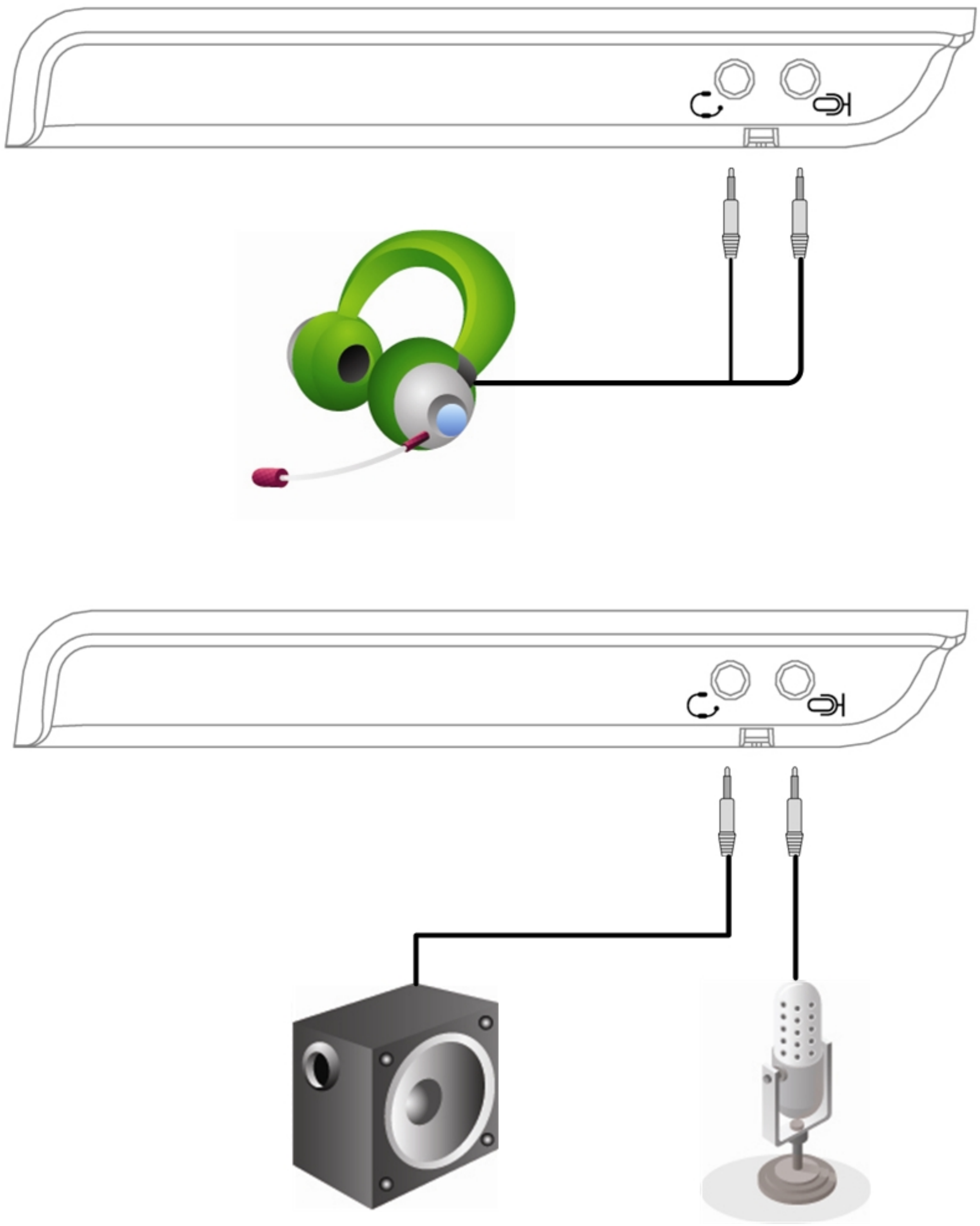

Figure 3-4 External Audio IN/OUT Interface Diagram of AP-IP200 IP Phone

Chapter 4. AP-IP200 UI Operating Guide

AP-IP200 IP Phone Initial Information Display

If power-on booting procedure or warm start reset procedure is done, LCD panel of AP-IP200 displays the message shown as figure 4-1. Table 4.1 explains what is the LCD display message and upper, right side's red color LED of AP-IP200 front side.

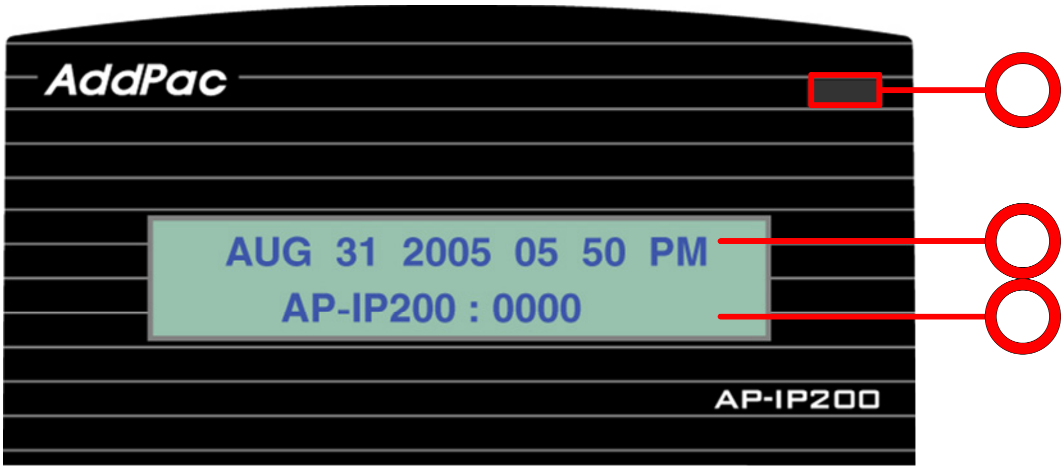

Figure 4-1 LCD Display Message AP-IP200 IP Phone

[Table 4-1] Upper LCD Panel's LED and Message Display of AP-IP200 IP Phone

No.	Description	
①	Red color LED	LED ON: Hook Off (Handset Pickup, Speaker Phone Mode, and Headset Mode) and Conversation Mode. LED OFF : Idle state (On Hook) LED Blinking: Red color LED is blinking at absence call. If User Hook-Off, checks the absence call, LED blinking is disappear.
②	Date & Time	Display the date & time of current. At conversation, display the real time "connection time".

③	Display Name	Display the Nick Name of this device. Change the Nick Name using following UI menu setting (System Setup -> 1.Display Name)

PhoneBook Menu

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Phone Book is a phone number directory in which user can search by name and number, register phone number. It also has call log and speed dial menu.

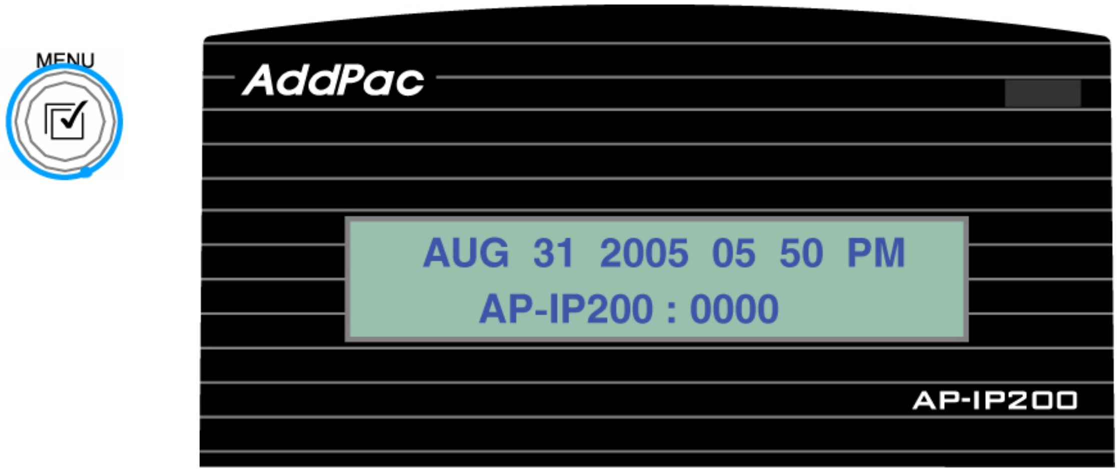


Figure 4-2 Main Display

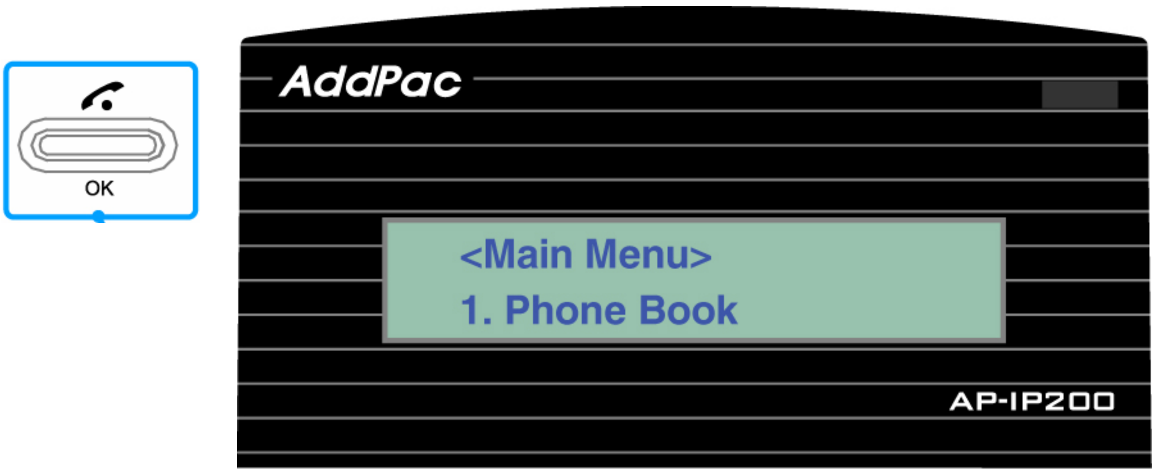


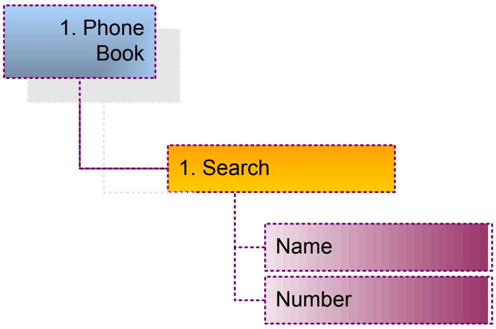
Figure 4-3 Phonebook Menu Display

Phonebook — Search by Name

Search phone number using name registered in Phone Book. User can find right number by inputting name with KEY button and make a VoIP call by pushing call button.

Phonebook — Search by Dial Number

Search phone number throughout Phone Book. User can input telephone number using KEY button, search right person and make a VoIP call by pushing call button.



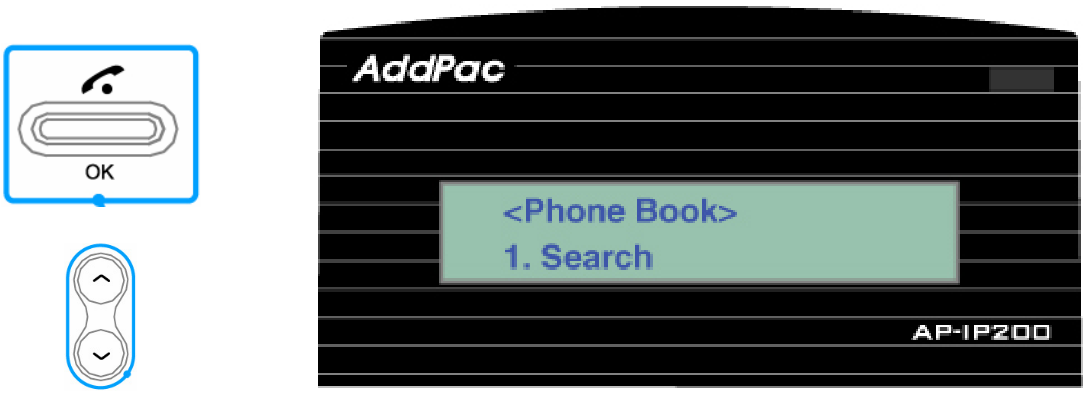


Figure 4-4 Phonebook Search Menu Display

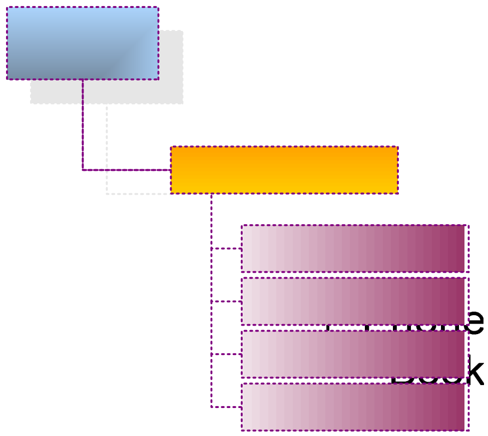
Phonebook Search Menu	Description
	The search by NAME is necessary for searching registered name throughout Phone Book with saved name before. Therefore, cursor automatically moves to the right category in accordance with inputting the letter. If there are over 2 field which the first letter of input name matches the registered user name,

	matched user name is displayed on LCD and then user can subsequently input the second letter for search. Besides, user can make a call or modify information using selected person's information.
<Phone Search> Number:	A LCD display message when user select Main >> Phone Book >> Number. Number menu has search function to search throughout PhoneBook with saved number before. Therefore, cursor automatically moves to the right category in accordance with the letter inputted using KEY button. If there are over 2 field which the first letter of input number matches the registered user dial number, , matched user dial number is displayed on LCD so then user can subsequently input the second letter for search. Besides, user can make a call or modify information using selected person's information. Please refer to search by name description for displayed user's information.

Phonebook – Registration

Register new number. User can register name, telephone number, IP address or domain in Phone Book.

Figure 4-5 is a LCD display message when user selects Main Menu >> Phone Book >> Registration. When user wants to input new number in Phone Book, use this menu. Input name, phone number, IP address using KEY button. Saved information is automatically registered in Speed Dial.



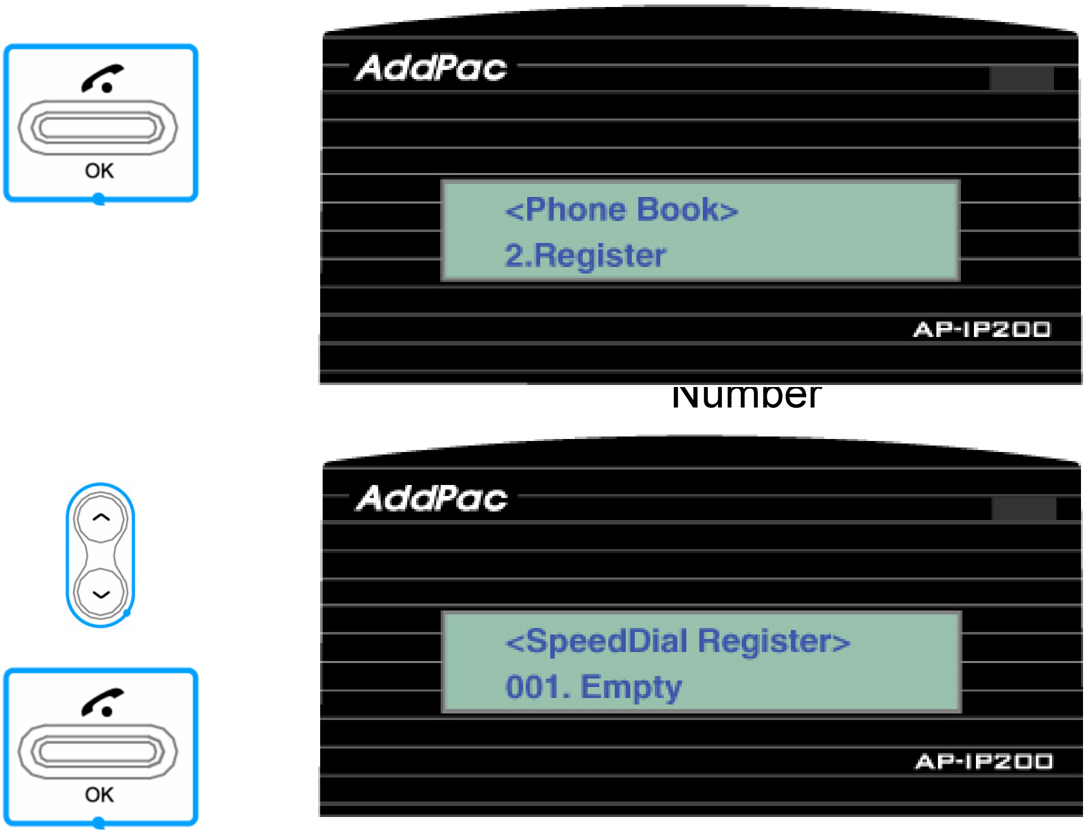


Figure 4-5 Phonebook Registration Menu

Phonebook Registration Menu	Description
<Speed Dial Edit> Speed Dial ID:001	Indicates index number of phone number in Phone Book. This index number is for sorting in Speed Dial Index list.

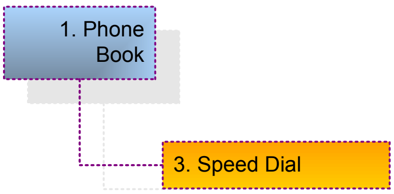
<Speed Dial Edit> Name:	Input peer's name for registering in Phonebook
<Speed Dial Edit> Number:	Input peer's dial number for registering in Phonebook.
<Speed Dial Edit> IP:	Input peer's IP address for registering in Phonebook

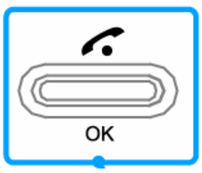
*

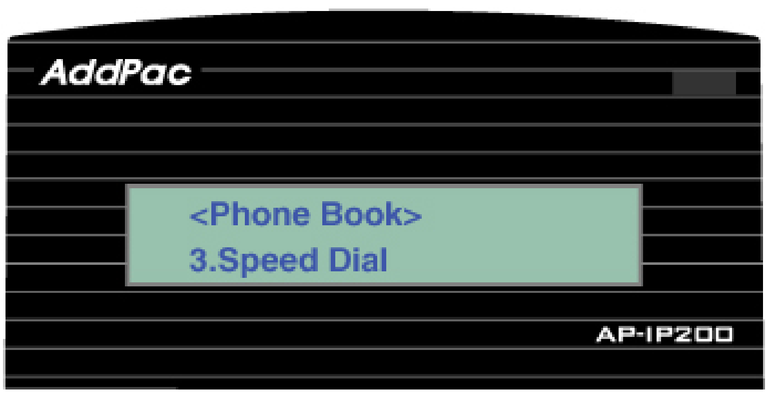
If user wants to apply the registered value In Menu, user should press the OK button. And if user wants to maintain the applied value after reboot, user should press the OK button at ToolBox-SaveAll Menu.(refer to Tool Box Menu)

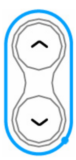
Phonebook – Speed Dial

Telephone numbers are listed per LCD display by simply using name. User can quickly find and make a call to others on the list by this menu.









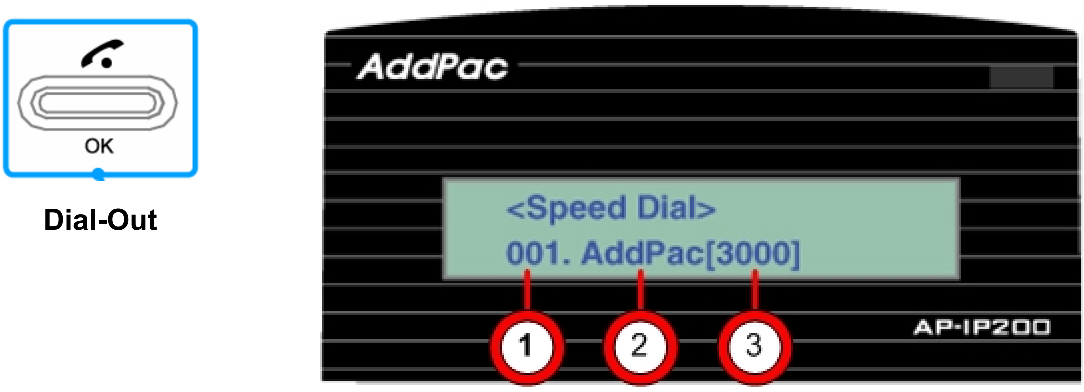
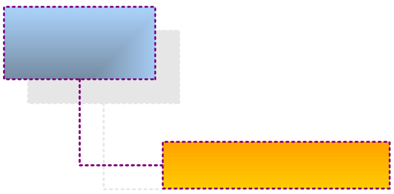


Figure 4-6 Phonebook Speed Dial Menu

Phone book Speed Dial	Description
①	Display the index number in Phonebook
②	Display the name in Phonebook
③	Display the Dial number in Phonebook

Phonebook — Recent Call

Recent Call Menu shows recent call log. Call record provides number, name, and IP address of counter parties. It's possible for user to make a VoIP call by OK call button. Two (2) recent call dialing method are available shown as Figure 4-7, 4-8.



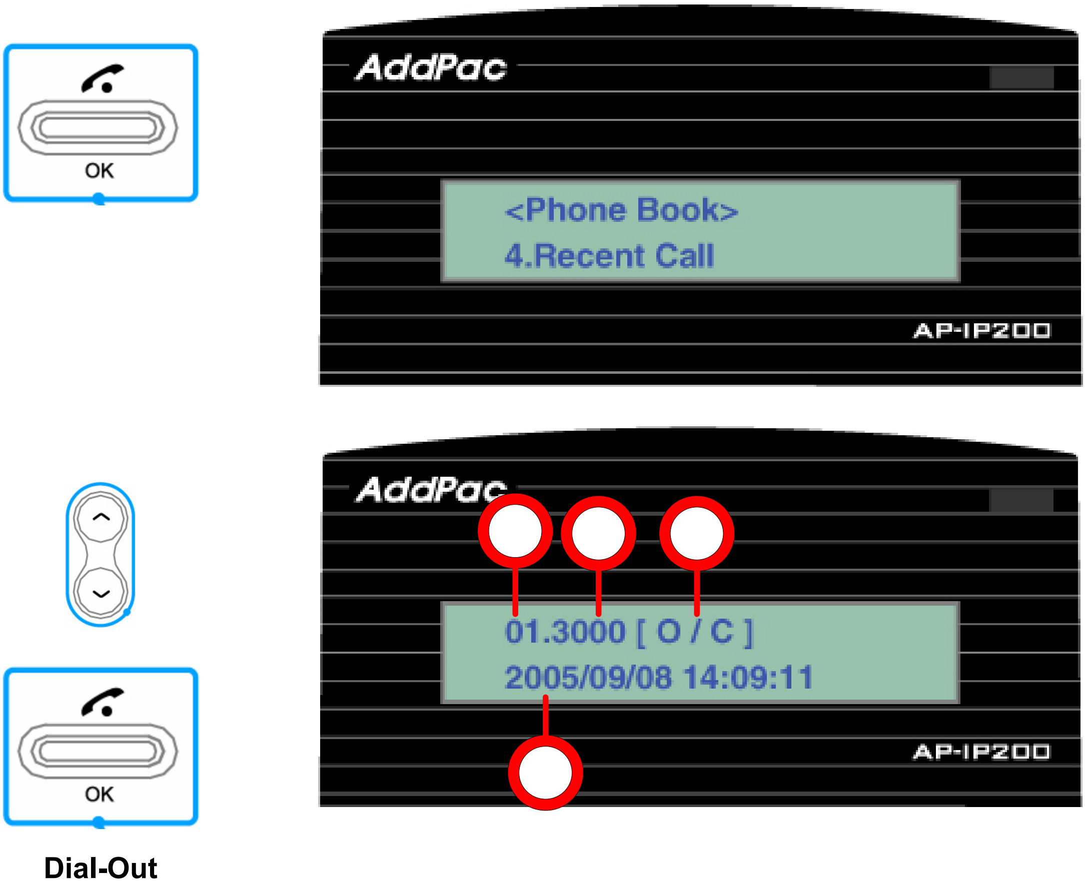
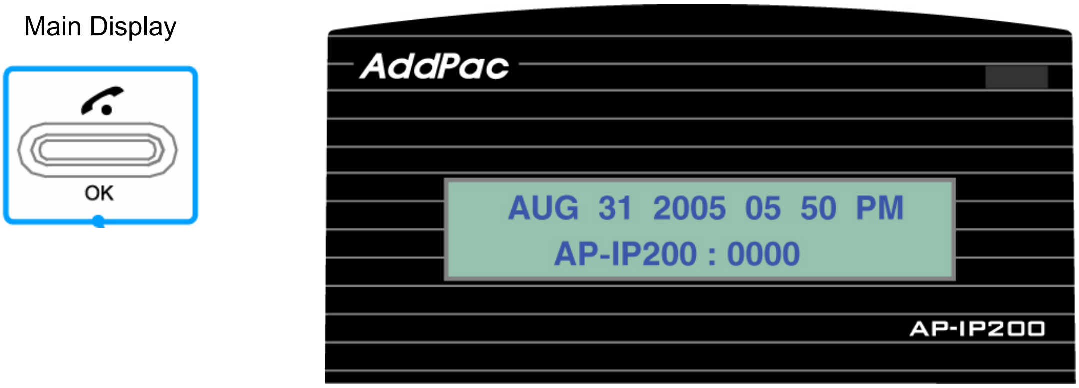
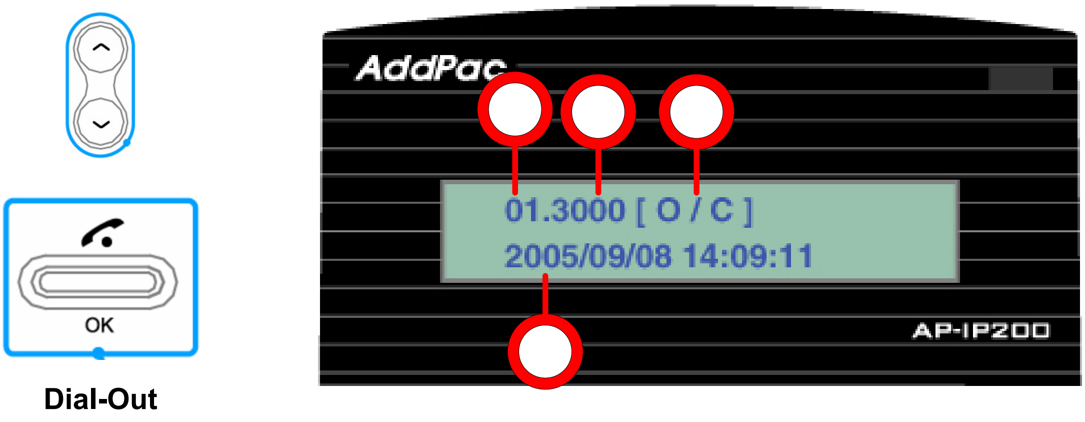


Figure 4-7 Phonebook Recent Call Menu 1

1



4



1

Figure 4-8 Phonebook Recent Call Menu 2

Phonebook recent call menu	Description
①	Display the index number in Recent Call History
②	Display the name and dial number of counter party in Recent Call History
③	[O/C] ⇒ Dial-Out / Connection [O/N] ⇒ Dial-Out /Connection Fail [I /C] ⇒ Dial-In/ Connection [I /C] ⇒ Dial-In/Connection Fail
④	Display Dial-in/Out conversation connection time

4

ToolBox Menu

Tool Box menu consists of date/time setting, configuration saving, initialization for factory default mode and language selection.

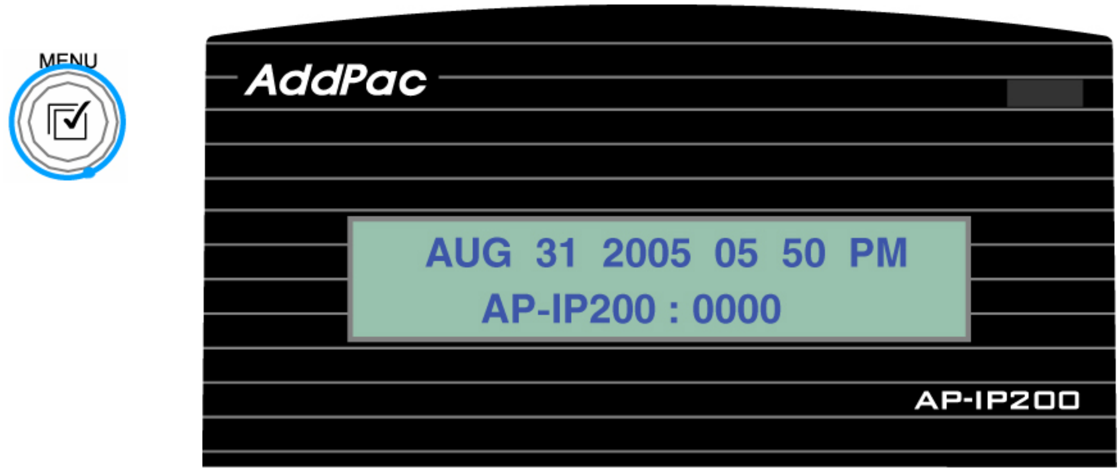


Figure 4-9 Main Menu

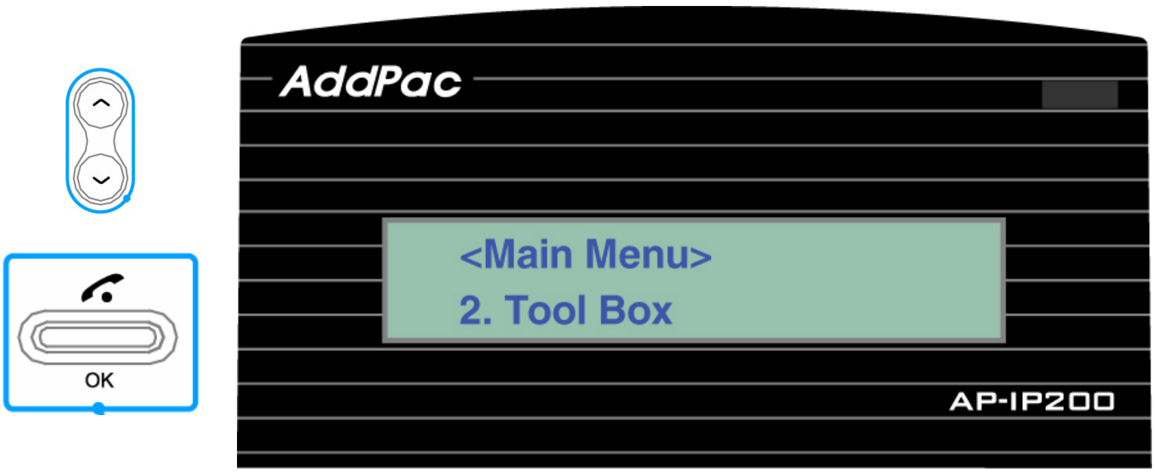
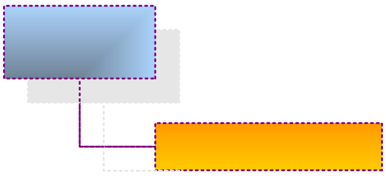


Figure 4-10 ToolBox Main Menu

ToolBox — date & time



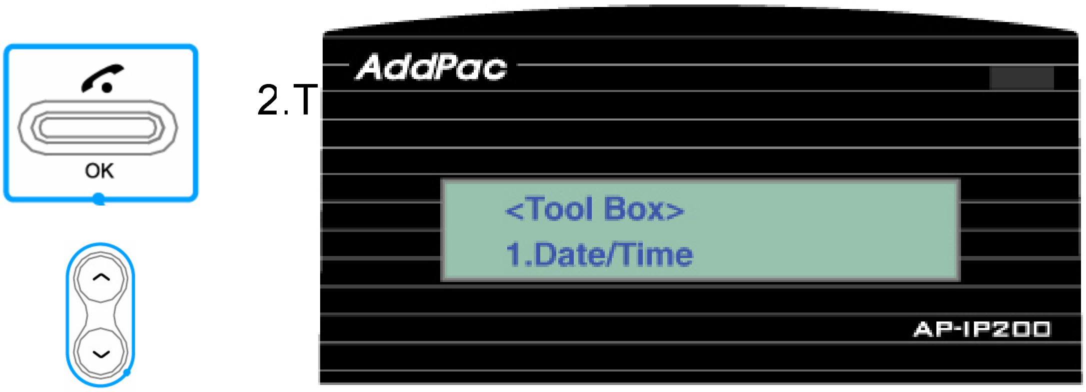
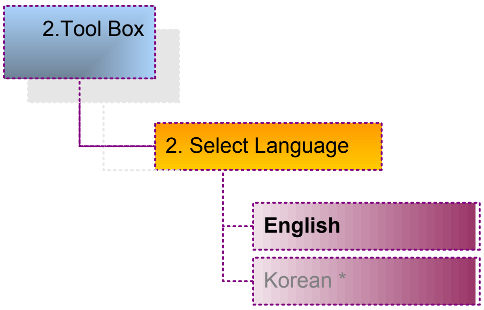


Figure 4-11 Data/Time ToolBox Menu

Date/Time ToolBox Menu	Description
<Date/Time Setting> Year : 2005	Input and modify Year setting of AP-IP200.
<Date/Time Setting> Month : 9	Input and modify Month setting of AP-IP200.
<Date/Time Setting> Day :13	Input and modify Day setting of AP-IP200.
<Date/Time Setting> Hour :10	Input and modify Hour setting of AP-IP200.
<Date/Time Setting> Minute : 31	Input and modify Minute setting of AP-IP200.
<Date/Time Setting> Second : 16	Input and modify Second setting of AP-IP200. (If OK button is pressed, date/time input procedure is finished and input

	data/time is saved)

ToolBox -Language



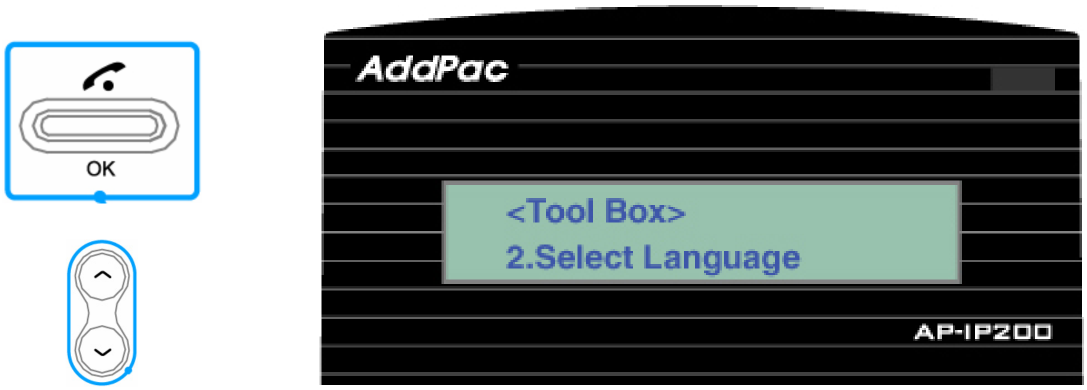
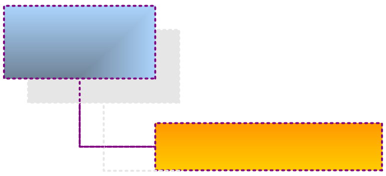


Figure 4-12 ToolBox Language Selection Menu

Language Selection	Description
<Language Setup> Language : English	English Language Mode
<Language Setup> Language : Korean	Other languages including KOREAN (*will be supported)

ToolBox — Factory Default Mode Initialization

Factory Default mode initialization feature deletes all the configuration of AP-IP200 and all the content on phone book and recent call menu. This command reboots the system automatically. This command is not recommended to use except in inevitable case.



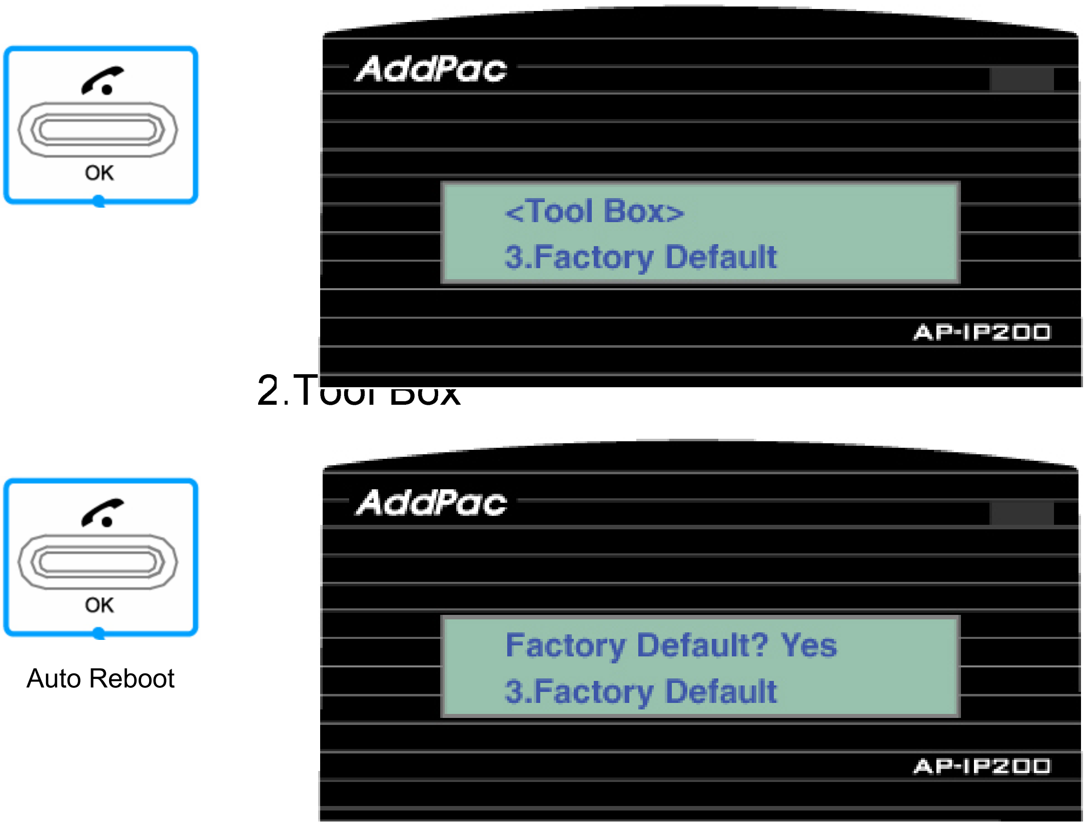


Figure 4-13 Factory Default Menu

Tool Box– Save All

This UI menu is for saving values which user inputs in UI. Once being saved, values are preserved even after rebooting.

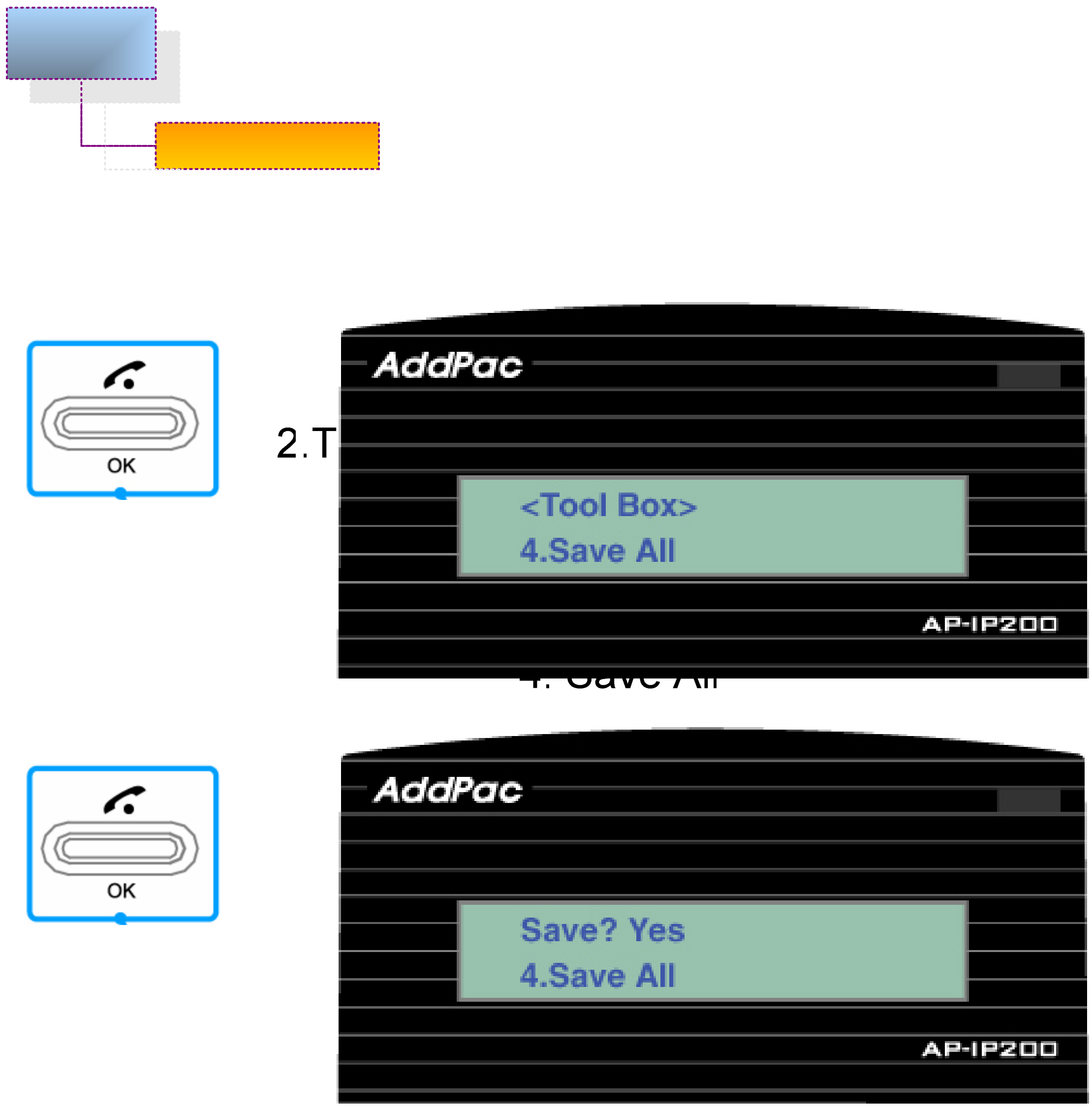


Figure 4-14 ToolBox Save All Menu

Network Setup Menu

The network setup of AP-IP200 consists of WAN, LAN interface setting, SIP/H.323 signaling, FTP service, QoS for guaranteeing VoIP quality, call options etc. User should know this network setup menu for efficient usage of AP-IP200. This menu cannot be skipped for optimized environment.

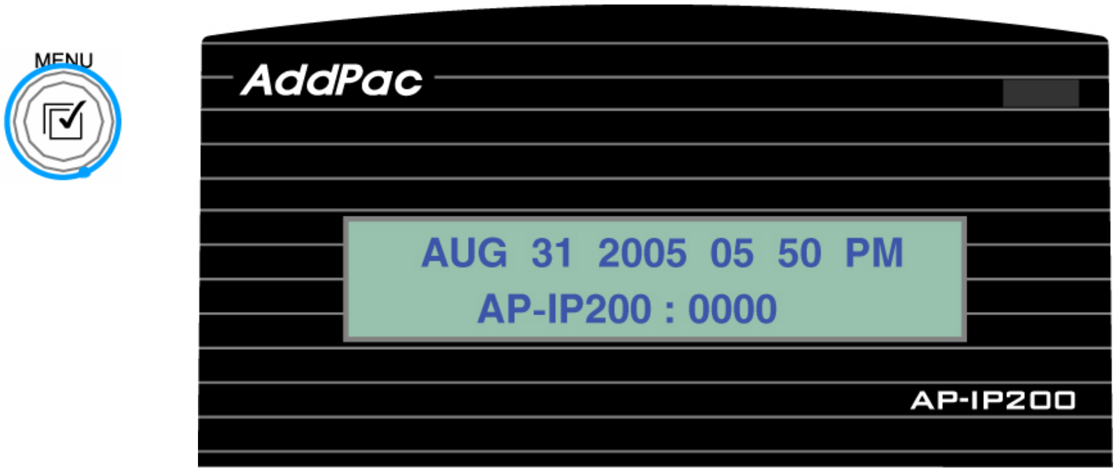


Figure 4-15 Main Menu

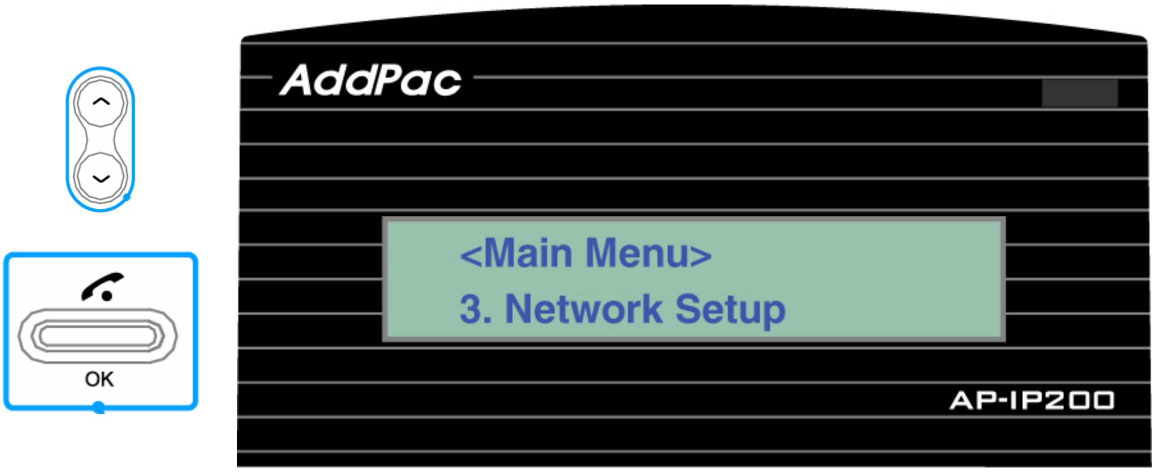
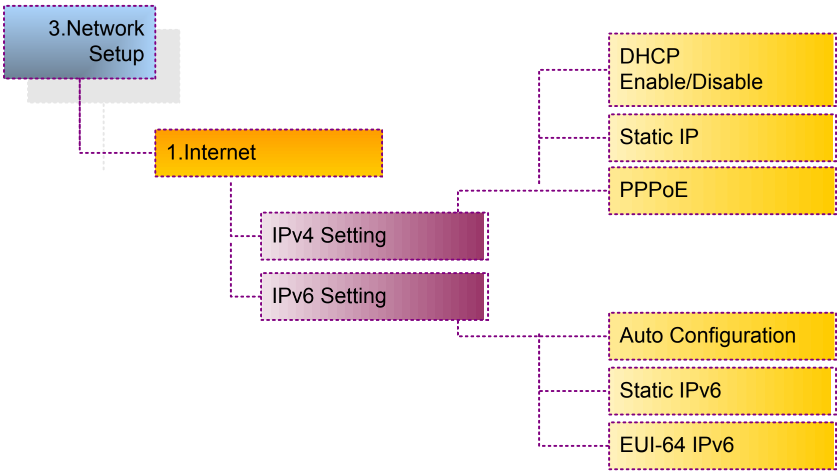


Figure 4-16 Network Setup Main Menu

Network Setup – Internet

Internet Menu has functions in relevance to AP-IP200's fast ethernet LAN interface for Internet connection. As there are various network environments, user has to configure pursuant to his or her own network environment. WAN protocols supported by AP-IP200 are DHCP, static IPv4, PPPoE, and IPv6 etc.
Following figure shows the UI command tree structure in Network Setup Menu.



Internet — IPv4 Setting

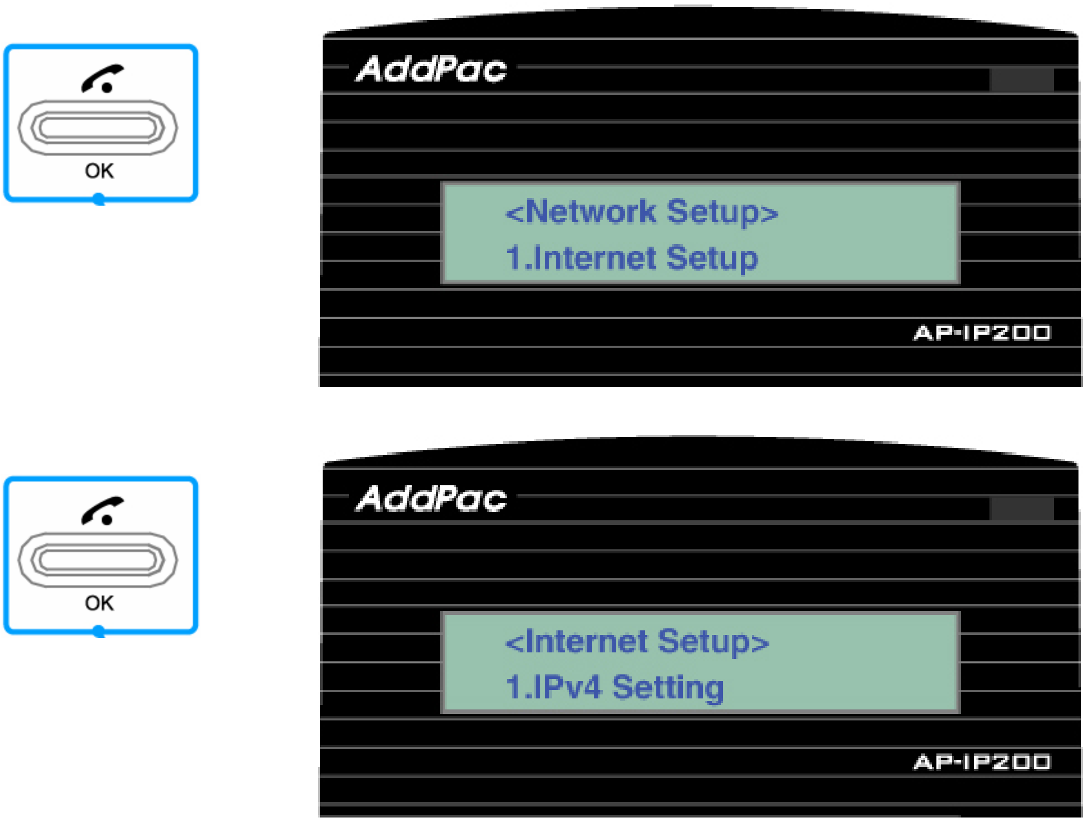


Figure 4-17 IPv4 protocol setting Menu

IPv4 protocol setting menu	Description
<IPv4 Setting> DHCP : Enable	Getting a dynamic IP address from DHCP server such as cable modem, VDSL, IP-ADSL.
<IPv4 Setting> Static IP : Enable	Set IP address manually and build WAN interface such as static IP ADSL, E1/T1 leased line.
<IPv4 Setting> PPPOE : Disable	Getting a dynamic IP address from PPP server such as ADSL.

IPv4 Setting— DHCP

If user enables DHCP protocol in network setup internet menu, user gets a dynamic IP address from DHCP server such as cable modem, VDSL, IP-ADSL.

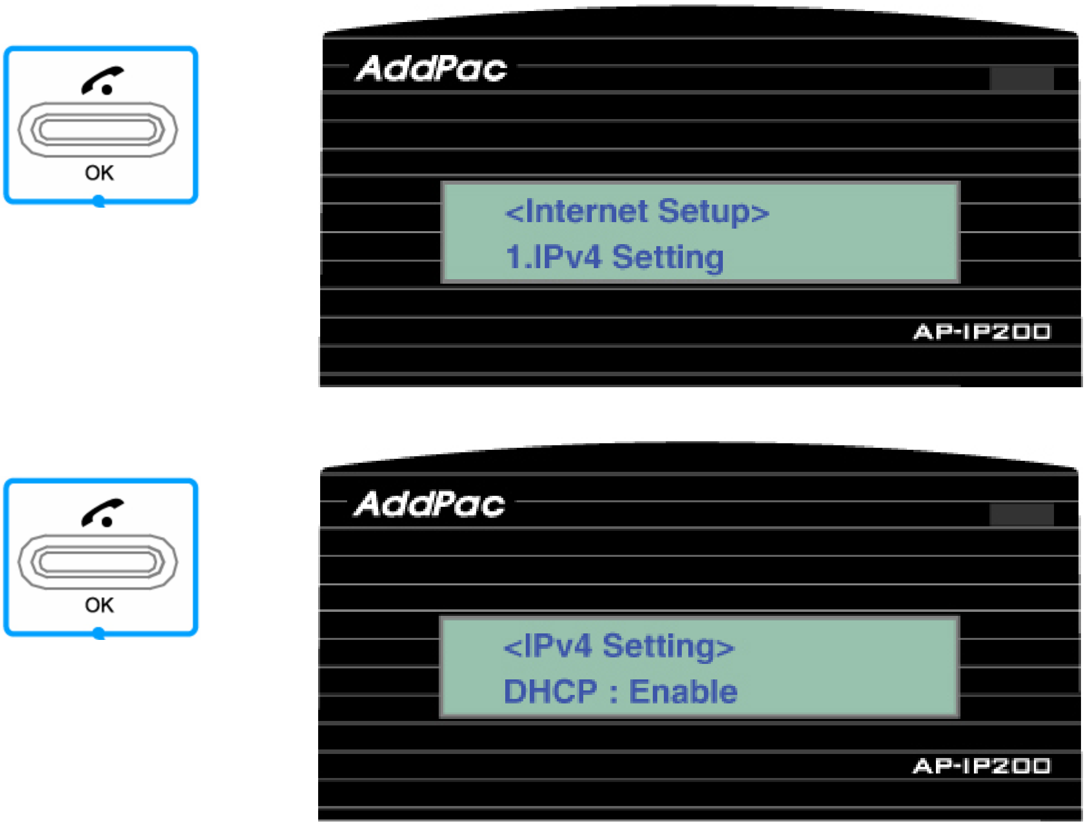


Figure 4-18 IPv4 DHCP Setting Menu

IPv4 Setting— Static IP

If user wants to use static IP address, this menu helps to set a static IPv4 address manually and build WAN interface such as static IP ADSL, E1/T1 leased line, Metro ethernet, ATM, etc.

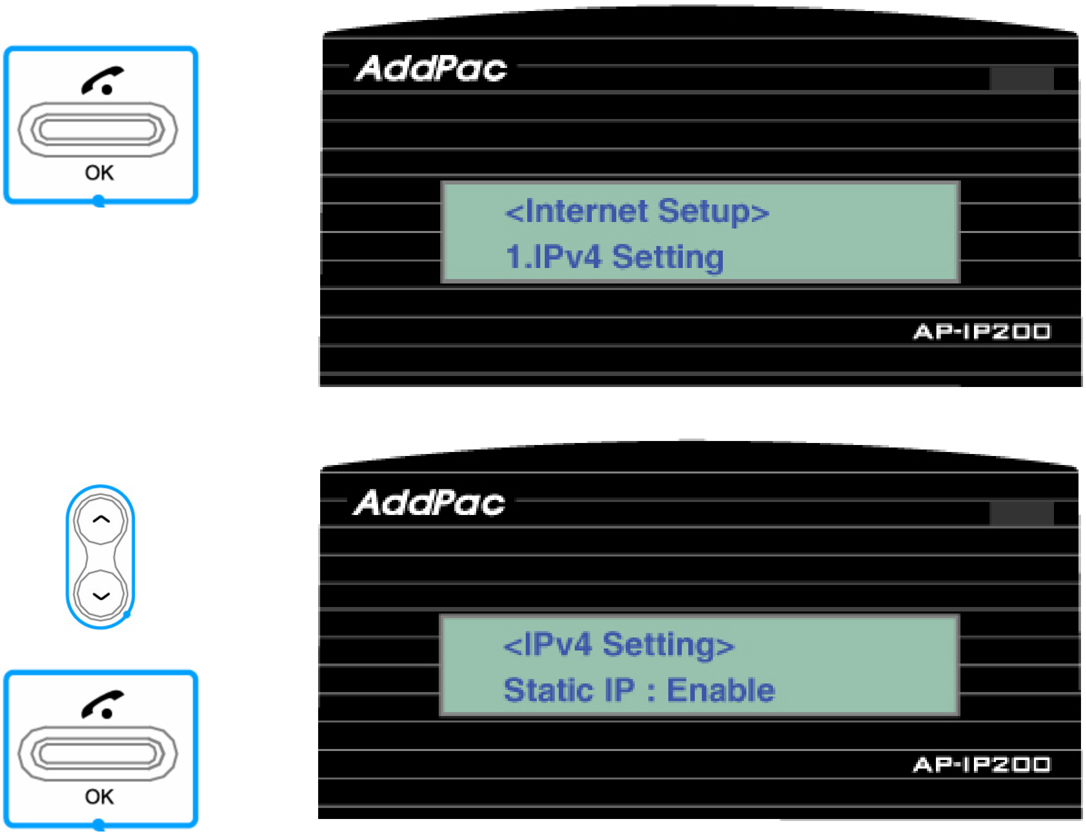


Figure 4-19 Static IP Address Setting Menu

Static IP Address Setting	Description
<Static IP Setting> IP Addr:	IP Address Setting Ex> 172.20.1.100
<Static IP Setting> Netmask:	Network Mask Setting Ex> 255.255.0.0
<Static IP Setting> Gateway:	Default Router Setting Ex> 172.20.1.1

<Static IP Setting> DNS1:	First DNS setting (this menu can be applied at IPv6 mode) Ex> 168.126.63.1
<Static IP Setting> DNS2:	Second DNS Setting (Option) (this menu can be applied at IPv6 mode) Ex> 168.126.63.1

If user wants to apply this input value In Menu after setting, user should press the OK button. And if user wants to maintain the applied value after reboot, user should press the OK button at ToolBox-SaveAll Menu.(refer to Tool Box Menu)

IPv4 Setting— PPPoE

If user wants to get a dynamic IP address from PPP server, user enables the PPPoE protocol in IPv4 internet protocol setting menu to get a dynamic IP address from PPP server such as ADSL, etc.

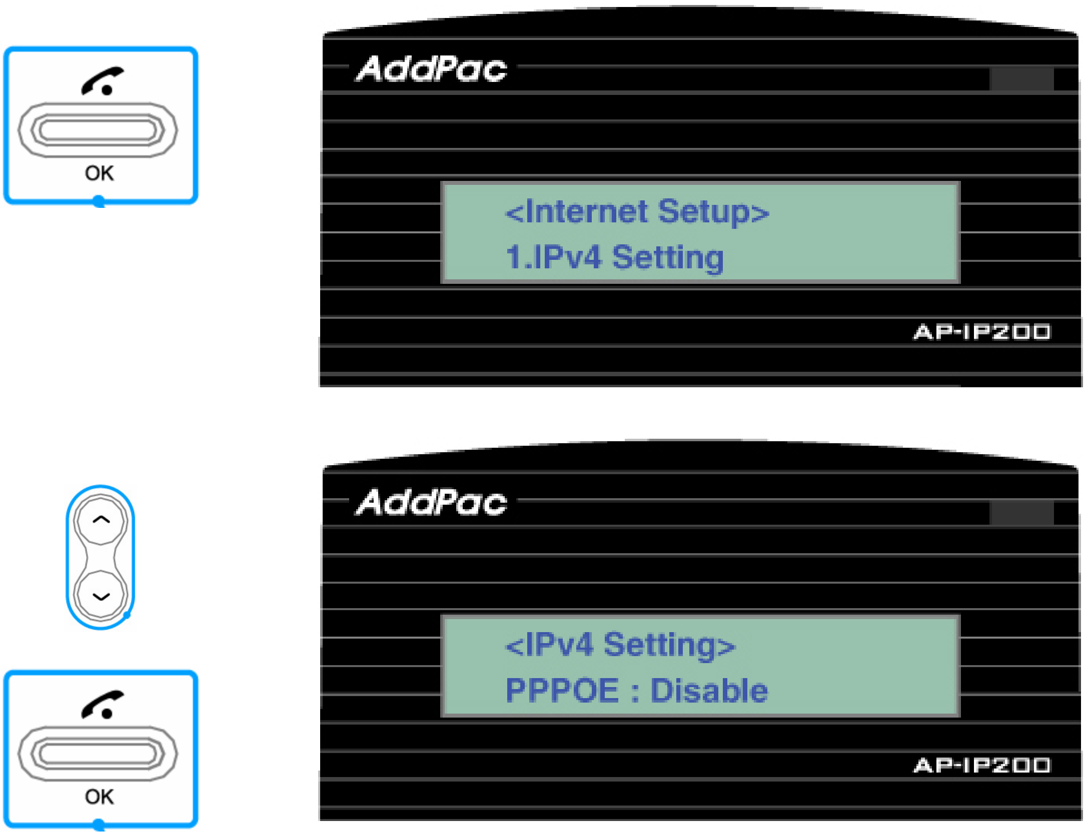


Figure 4-20 PPPoE Setting Menu

PPPoE Parameter Setting	Description
<PPPOE Setting> ID :	Input User Name Ex> Addpac
<PPPOE Setting> Password :	Input User Password Ex> 1234
<PPPOE Setting> Auth Mode : PAP	Input Auth. Method “PAP/CHAP” using Alphanumeric KEY.

If user wants to apply this input value In Menu after setting, user should press the OK button. And if user wants to maintain the applied value after reboot, user should press the OK button at ToolBox-SaveAll Menu.(refer to Tool Box Menu)

Internet– IPv6 Setting

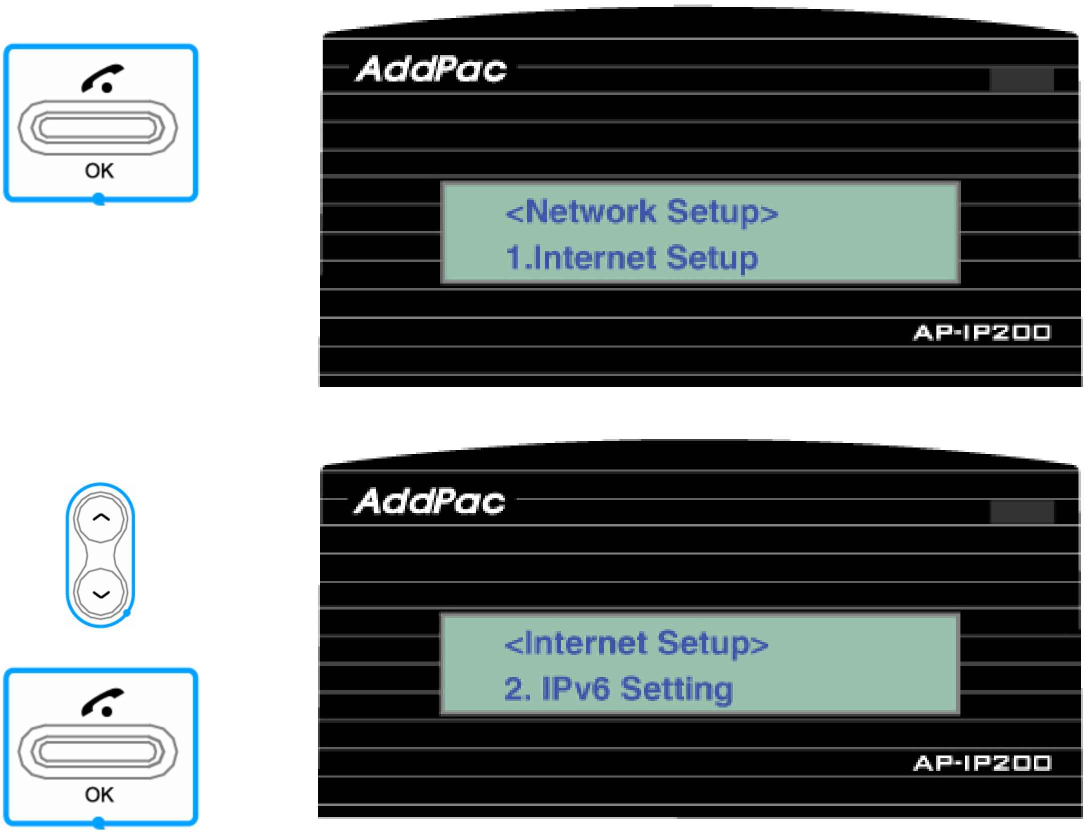


Figure 4-21 IPv6 Setting Menu

IP version 6 setting	Description
<IPv6 Setting> IPv6 Auto : Enable	Getting a dynamic IPv6 address from IPv6 server.
<IPv6 Setting> Static IPv6 : Disable	Set IP version6 address manually and build WAN interface such as static IP address E1/T1 leased line.
<IPv6 Setting> IPv6 EUI64 : Disable	This EUI-64 IPv6 address scheme use the Network ID(64bit) and Host ID(64bit). User sets Network ID by KEY input. System MAC address is used for 64bit

	Host ID.

IPv6 Setting– Auto Configuration

This auto configuration mode enables to get a dynamic IPv6 address from IPv6 server.

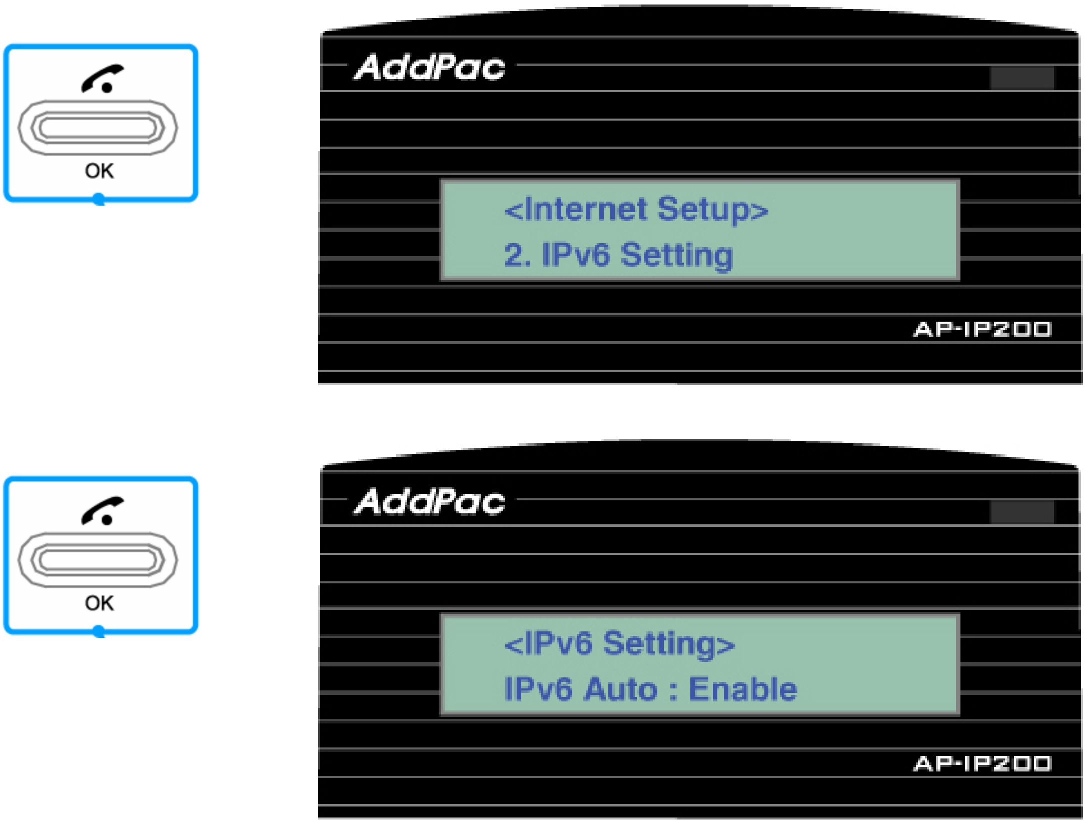


Figure 4-22 IPv6 Auto Configuration Setting Menu

IPv6 Setting— Static IPv6 Address

If user wants to use static IP version 6 address, this menu helps to set a static IP version 6 address manually and build WAN interface such as static IPv6 E1/T1 leased line, etc.

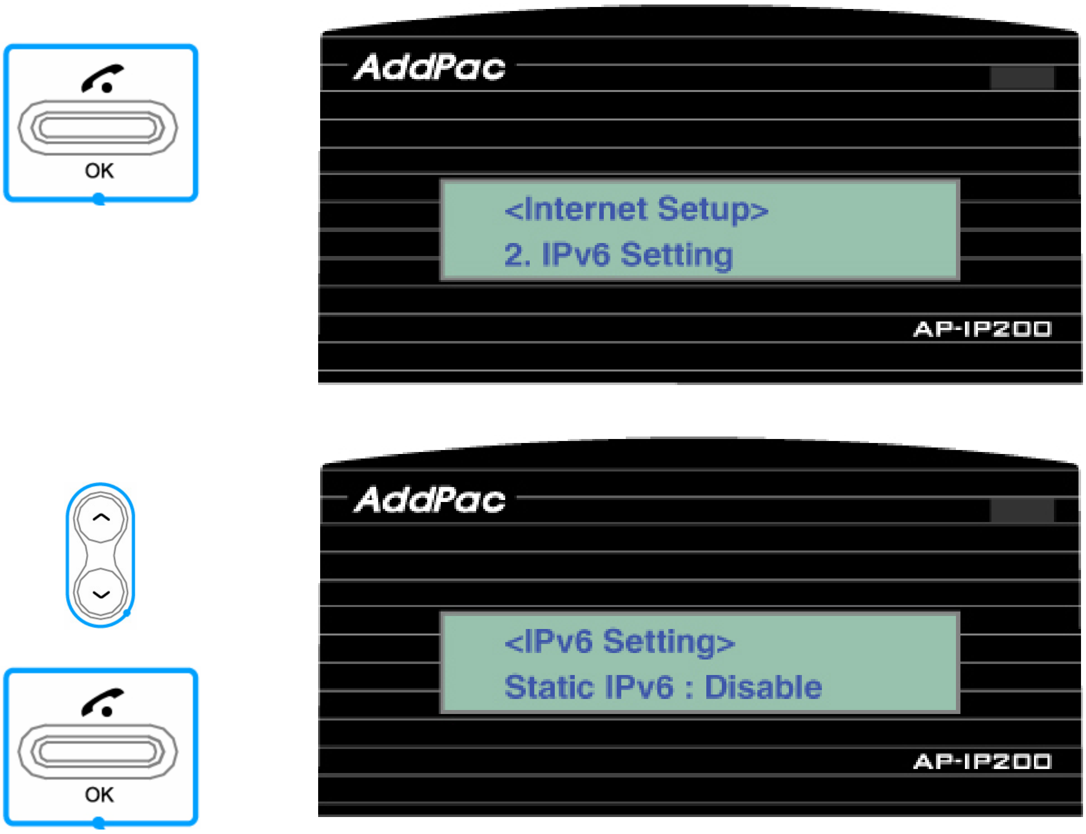


Figure 4-23 Static IPv6 Address Setting Menu

Static IP version6 Address Setting	Description
<Static IPv6 Setting> IPv6 Addr:	Input IPv6 Address Ex> 2001:230:20c:16:103::1/64
<Static IPv6 Setting> Gateway:	Input default Ipv6 router address Ex> 2001:230:20c:16::1
<Static IPv6 Setting> DNS1:	Input first DNS server address Ex> 2001:230:20c:20::1 Ex> 168.126.63.1

<Static IPv6 Setting> DNS2:	Input second DNS server address Ex> 2001:230:20c:20::2

IPv6 Setting— EUI-64 IPv6

This EUI-64 IPv6 address scheme use the address combination of Network ID(64bit) and Host ID(64bit). User sets Network ID by KEY input. System MAC address is used for 64bit Host ID.

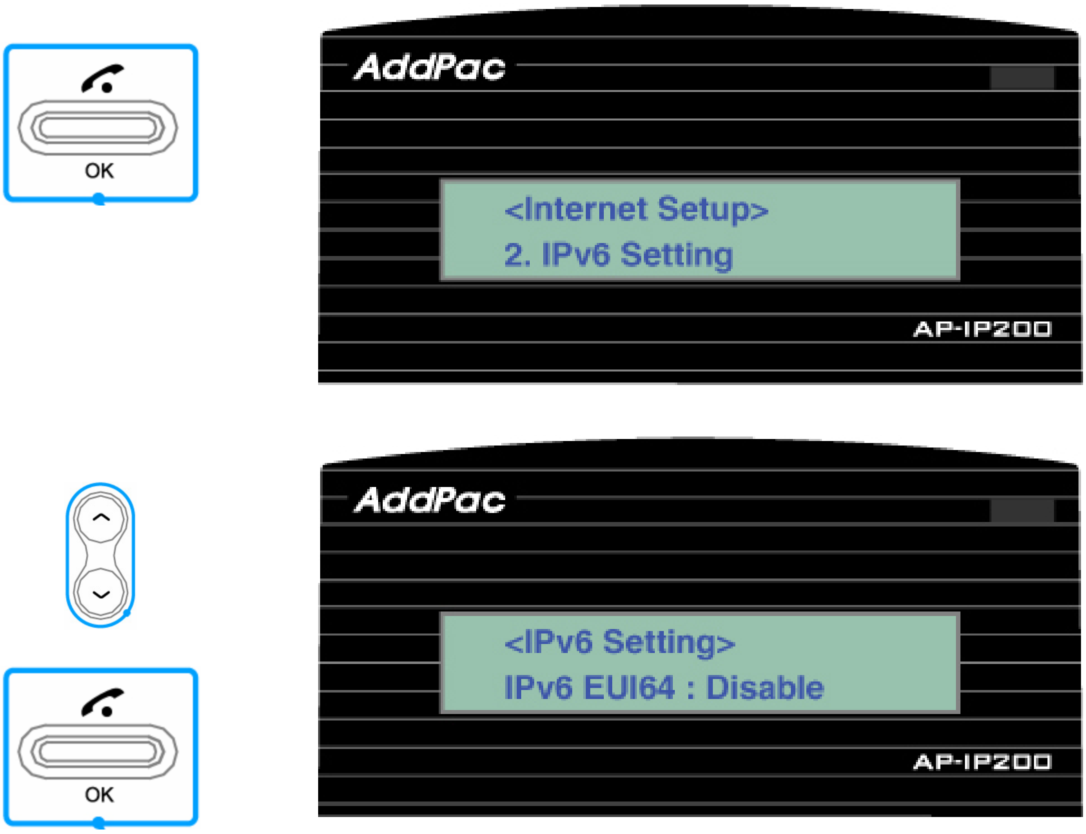


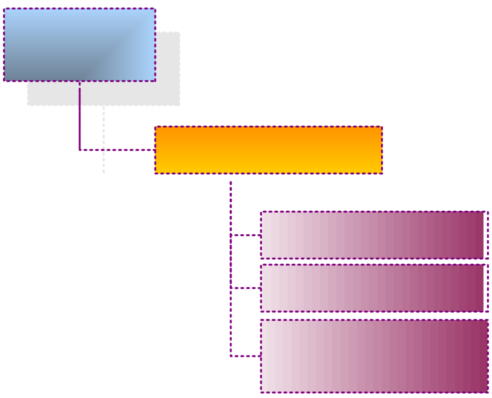
Figure 4-24 EUI-64 IPv6 Setting Menu

EUI-64 IPv6 Address Setting	Description
<IPv6 EUI Setting> IPv6 EUI64 :	Input IPv6 Address Ex> 2001:230:20c:16::1/64
<IPv6 EUI Setting> Gateway :	Input IPv6 Default Router Address Ex> 2001:230:20c:16::1

<IPv6 EUI Setting> DNS1 :	Input First DNS Server Address Ex> 2001:230:20c:20::1 Ex> 168.126.63.1
<IPv6 EUI Setting> DNS2 :	Input Second DNS Server Address (Option) Ex> 2001:230:20c:20::2

Network Setup— LAN

This LAN menu is used for protocol setting of AP-IP200 second LAN interface which is used to connect PC or Ethernet Hub. None, DHCP for single (1) PC, DHCP for multiple PC are available as protocols for second fast ethernet LAN port. In DHCP protocol mode for single PC, for sharing public same IP address of AP-IP200's WAN interface and LAN interface connected to PC, AddPac proprietary public IP-Share mechanism is used. DHCP for multiple PC are similar to general IP sharer which links two (2) PCs or more.



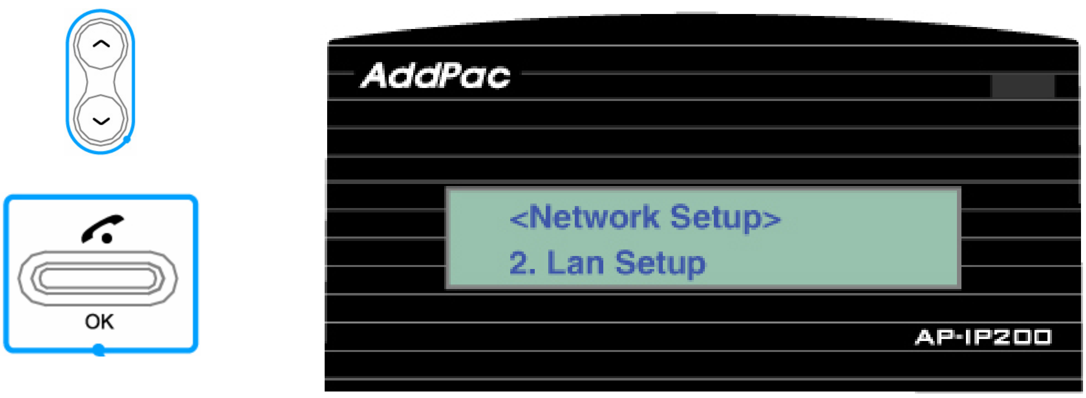
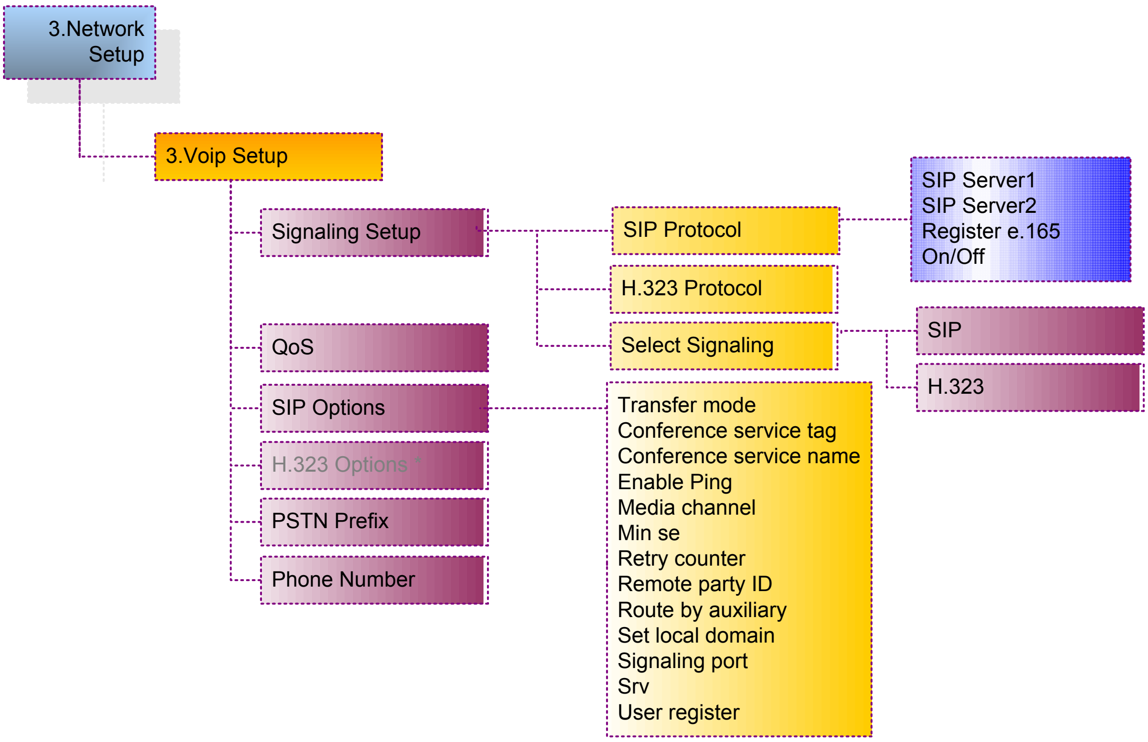


Figure 4-25 LAN Setup Menu

LAN Setup Menu	Description
<Lan Setup> NONE : Enable	Disable the LAN Setup function (If select, use OK button)
<Lan Setup> DHCP 1 PC : Enable	In DHCP protocol mode for single PC, for sharing public same IP address of AP-IP200's WAN interface and LAN interface connected to PC, AddPac proprietary public IP-Share mechanism is used
<Lan Setup> DHCP N PC : Enable	DHCP for multiple PC are similar to general IP sharer which links two (2) PCs or more.

Network Setup— VoIP

This VoIP setup menu is used for SIP/H.323 VoIP signaling interworking parameter setting between SIP server, Gatekeeper and AP-IP200 IP Phone. Following figure shows the UI menu tree structure for VoIP parameter setup.



VoIP Setting— Signaling Setup

This menu is used for VoIP signaling setup such as H.323, SIP protocol

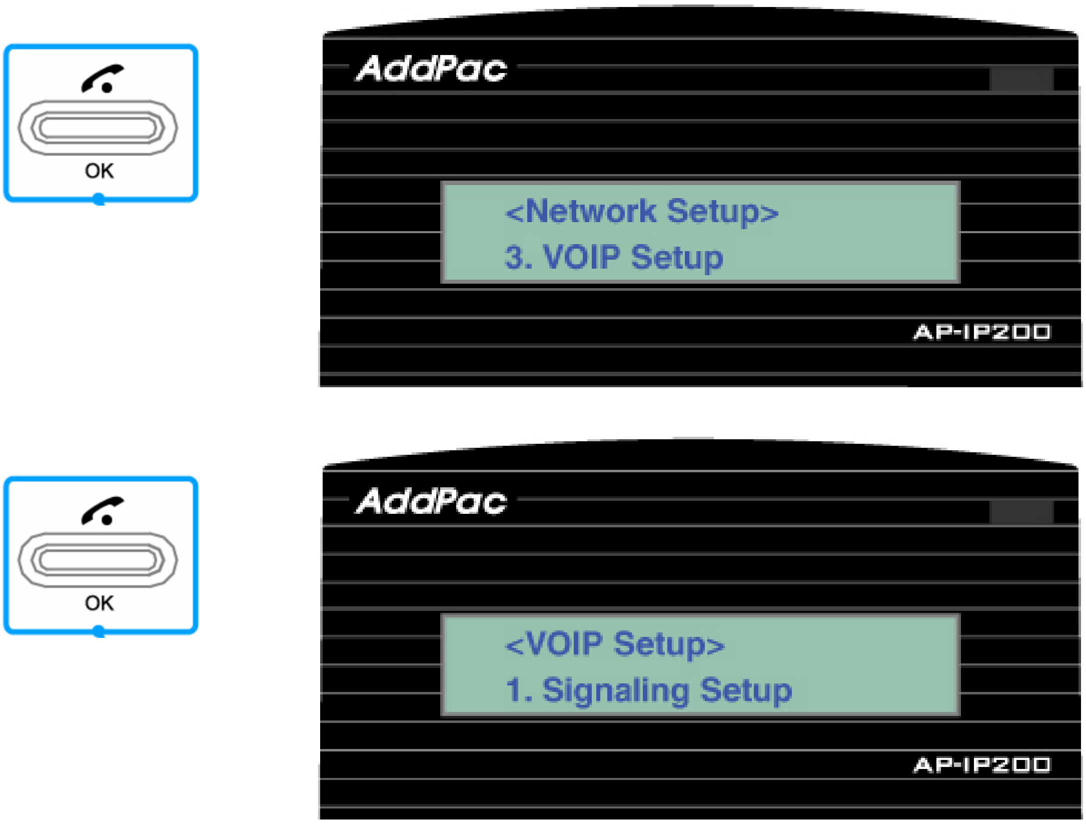


Figure 4-26 VoIP Signaling Setup Menu

VoIP Signaling Setup	Description
<Signaling Setup> 1. SIP Protocol	SIP parameter setup menu for SIP proxy server interworking
<Signaling Setup> 2. H.323 Protocol	H.323 parameter setup menu for H.323 Gatekeeper interworking
<Signaling Setup> 3. Select Signaling	Select one among SIP or H.323 VoIP signaling (User should select VoIP signaling used in this menu)

Signaling Setup— SIP Protocol

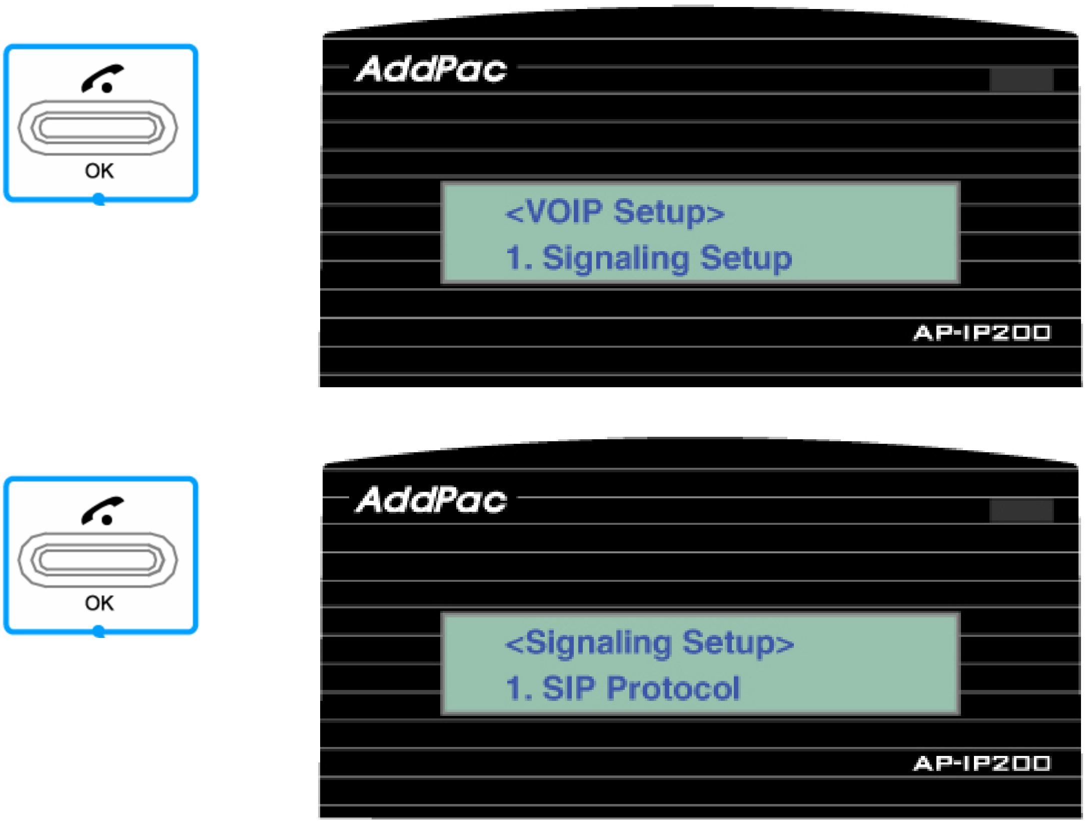


Figure 4-27 SIP Protocol Parameter Setup Menu

SIP Protocol Parameter	Description
<SIP Protocol> ID:	Input username for SIP server registration
<SIP Protocol> Password:	Input user password for SIP server registration.
<SIP Protocol> SIP Server1:	Input primary server IP address or domain of SIP server.
<SIP Protocol> SIP Server2:	Input secondary server IP address or domain of SIP server.

<SIP Protocol> Register e.164:off	Select ON/OFF mode whether E.164 address is registered in SIP server or not

Signaling Setup— H.323 Protocol

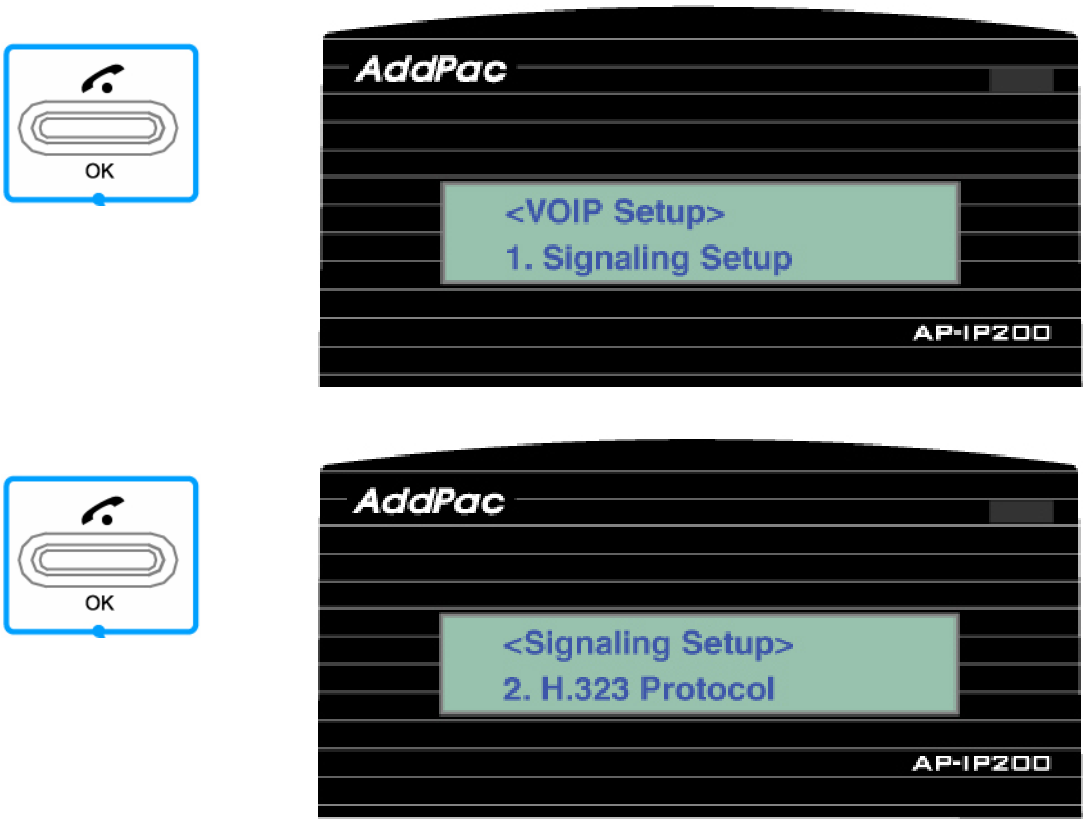


Figure 4-28 H.323 Protocol Setup Menu

H.323 Protocol Setup	Description
<H.323 Protocol> ID:	Input H.323 ID for Gatekeeper registration
<H.323 Protocol> Password:	Input H.323 password for Gatekeeper registration, if authentication is needed.
<H.323 Protocol> H.323 GK1:	Input primary Gatekeeper IP address

<H.323 Protocol> H.323 GK2:	Input secondary Gatekeeper IP address
<H.323 Protocol> Register GK :off	Select ON/OFF whether registered in Gatekeeper or not. (ON / OFF)

Signaling Setup– Select Signaling

This menu is used for selecting one VoIP signaling among SIP / H.323 Protocol.

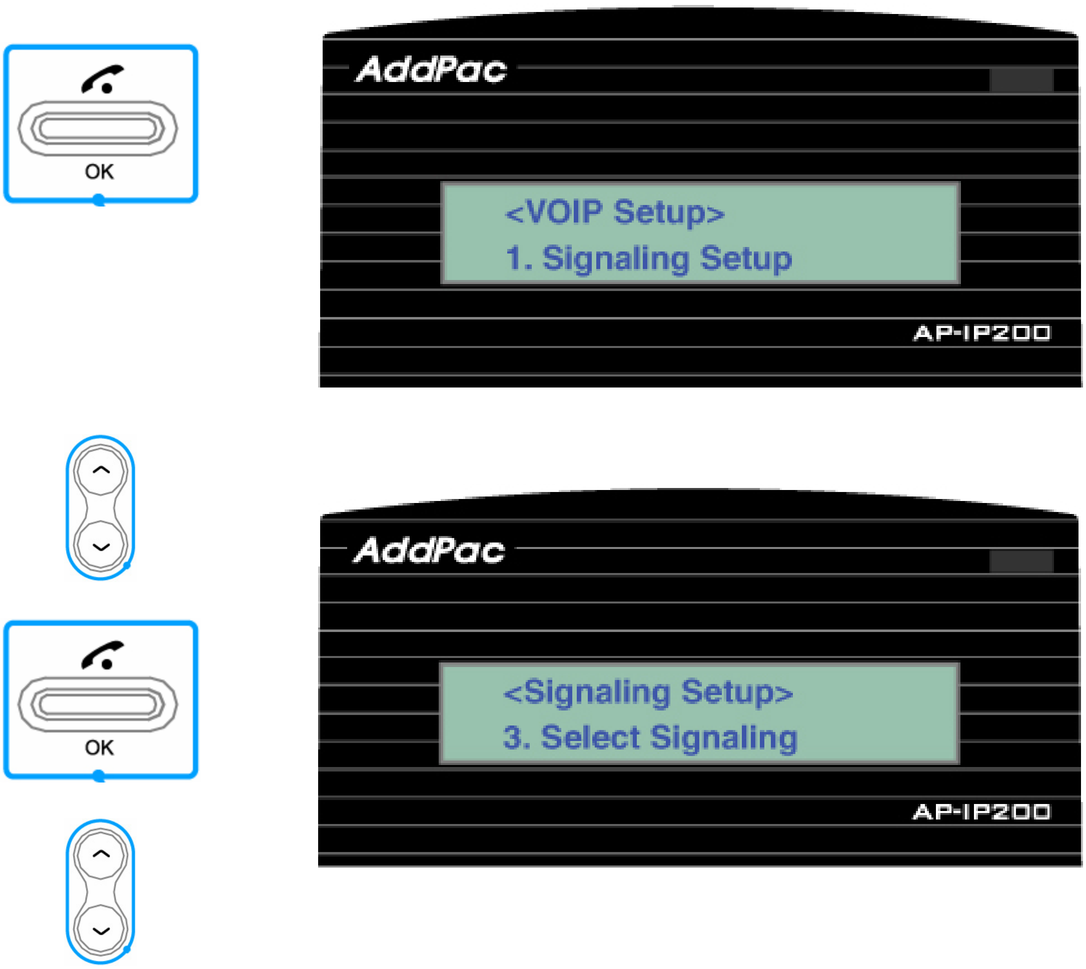
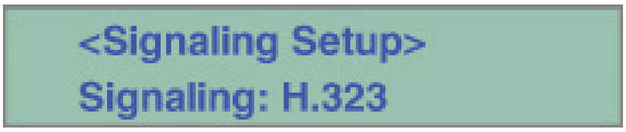


Figure 4-29 Select Signaling Menu

Signaling Setup	Description
<Signaling Setup> Signaling: SIP	SIP protocol choice (Pressed the OK button after moving

	using upper and lower KEY)
	H.323 Protocol choice (Pressed the OK button after moving using upper and lower KEY)

VoIP Setup— QoS

AP-IP200 transfers compressed VoIP data such as G.729, G.711 voice packet. Network condition like jitter, packet loss considerably affects high quality voice stream like G.711. Sometimes, it can be a major reason of voice quality degradation problems.
QoS enables transferring range of voice packet within the limit of bandwidth. User has to configure QoS matched with voice packet bandwidth. QoS function is for WAN interface and cannot be applied to LAN interface.

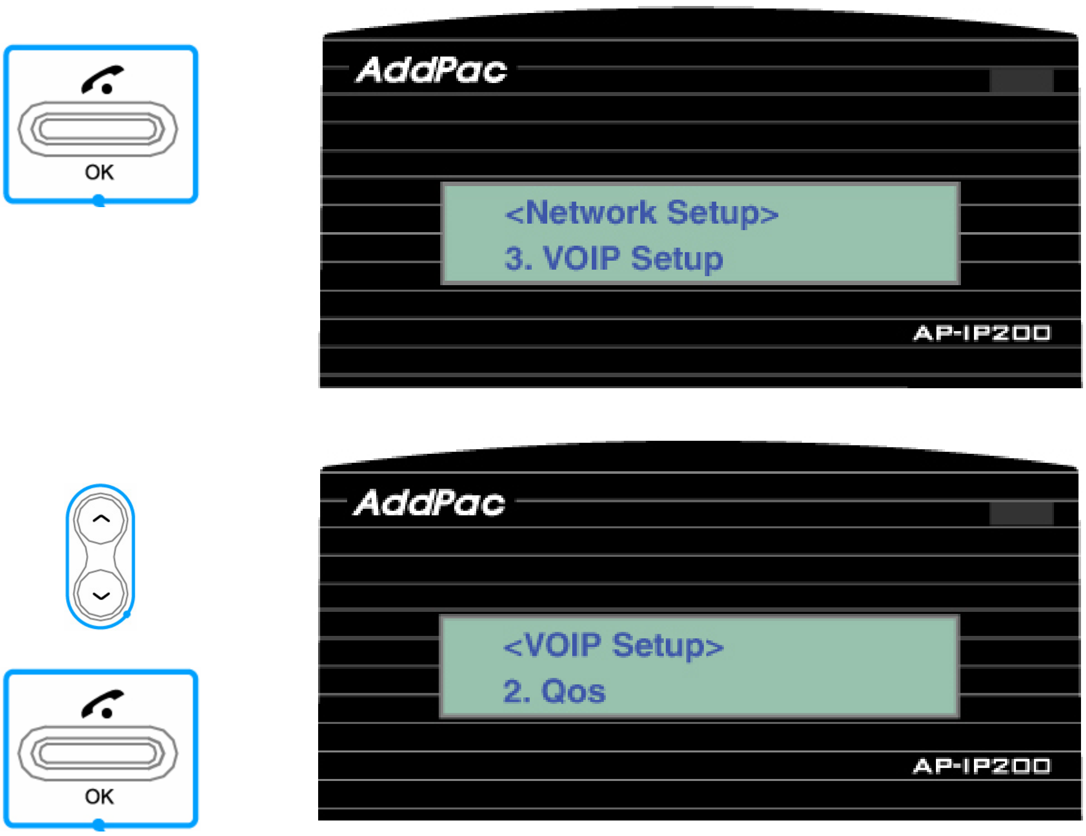


Figure 4-30 QoS Setting Menu

QoS	Description

<Qos Setup> Qos(48-4096):	QoS function is for WAN interface and cannot be applied to LAN interface. Range of value covers 48Kbps~4Mbps

VoIP Setup—SIP Options

This menu is for additional functions and options of SIP protocol. The SIP options are to be updated continuously.

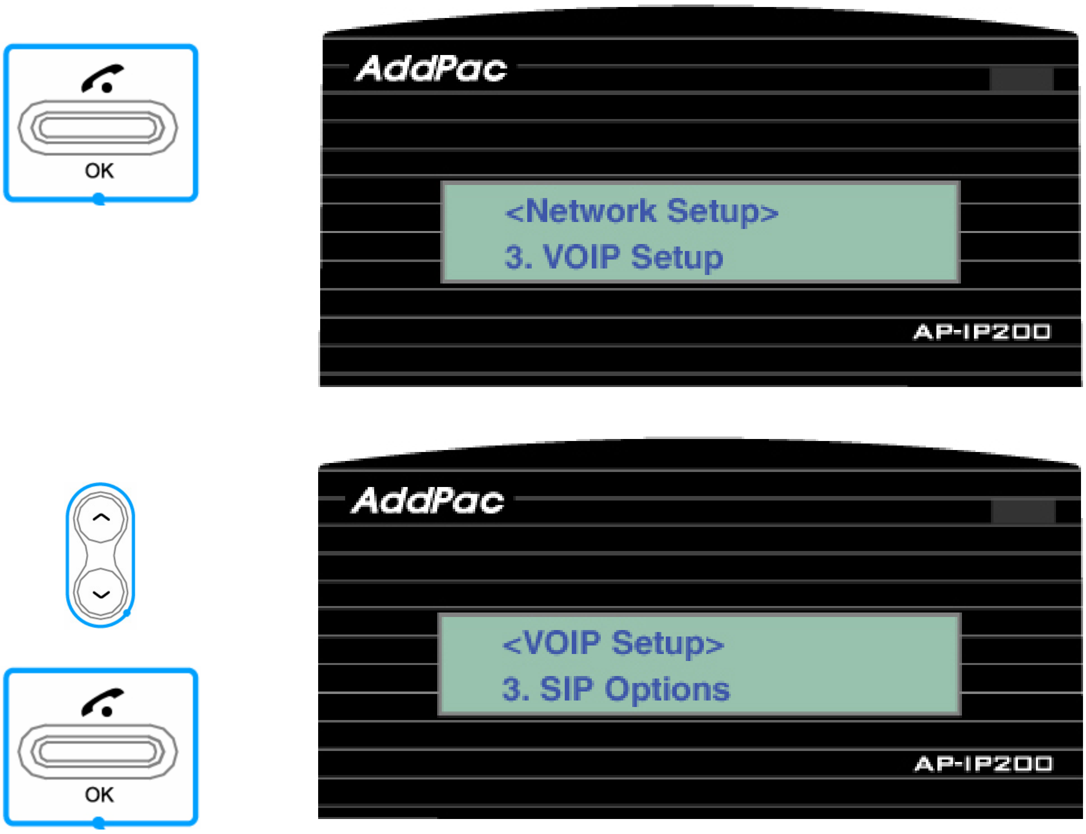


Figure 4-31 SIP Options Setting Menu

SIP Options	Description
<SIP Options> Transfer mode : basic	Select the call-transfer mode. basic/attend.

<SIP Options> Conf tag:	Enter VoIP Tag for conference service
<SIP Options> Conf srv:	Enter conference service name
<SIP Options> Enable ping:	Enter firewall address to check the public IP address when AP-IP200 is used under NAT/Firewall network environment.
<SIP Options> Media channel : early	Transfer RTP Session information to listen Inband Ringbacktone of Public network under NAT/Firewall environment.
<SIP Options> Min se : 1800	Set Session Timer
<SIP Options> Retry counter :10	SIP UA Retry Counter sets SIP INVITE re-transmission count when AP-IP200 is dial-out. When there is fault on network or network quality is not good, Trying message of INVITE message will be delayed. In this case AP-IP200 transfer next INVITE message. Default is 10 and usually set as ‘3.’
<SIP Options> Remote Party : none	When user-name is not number but characters, apply to register message.
<SIP Options> Route by aux : none	When called party is not number but characters, this option is used.
<SIP Options> Local domain	Transfer From/To field within SIP message to designated domain not to IP address.
<SIP Options> Signal port : 5060	Default is 5060 and this value is changeable.

<SIP Options> Srv : none	Set the DNS SRV.
<SIP Options> User register : none	When user-name is not number but characters, this option is used to register SIP server.

VoIP Setup— PSTN Prefix

When user wants to access the FXO interface for PSTN backup, this prefix number is used as PSTN access code. Additionally, AP-IP200 IP phone supports the PSTN back-up service when VoIP service is impossible due to network failure or VoIP call service is interrupted by exception.

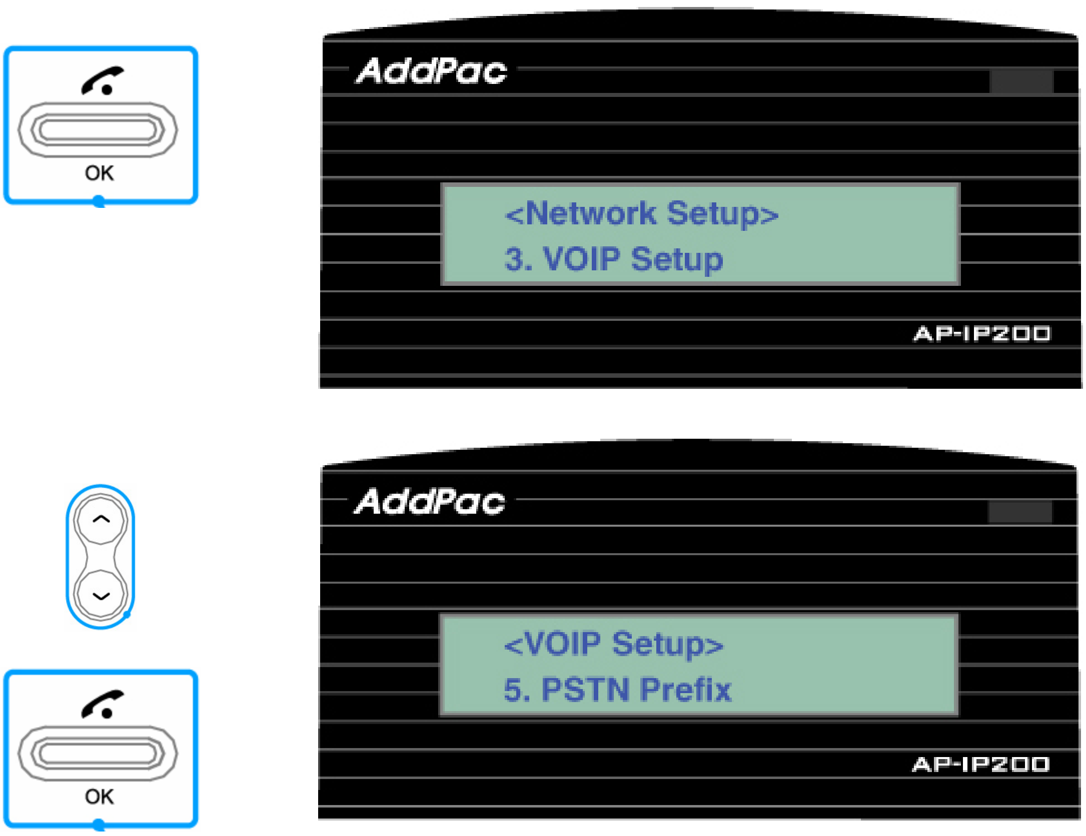


Figure 4-32 PSTN Prefix Menu

PSTN prefix	Description

<PSTN Prefix> Number:#	PSTN prefix number is an access code for PSTN FXO interface, default value is #.

VoIP Setup— Phone Number

This menu is used for E.164 AP-IP200 Number Assignment.

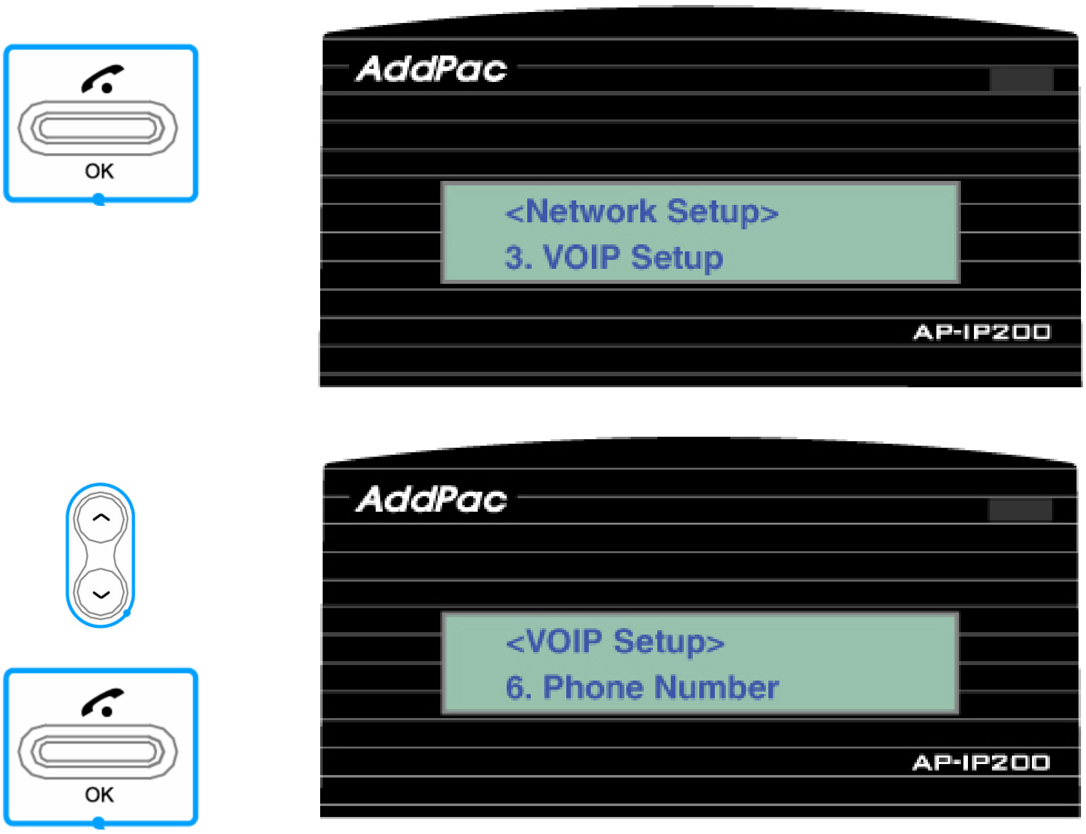


Figure 4-33 Phone Number Setting Menu

Phone Number Setup	Description
<Phone Number Setup> Number:	Enter the E.164 Phone Number.

*

Network Setup— Service

This menu activates or deactivates FTP, TELNET, TFTP, SNMP protocol service of AP-IP200.

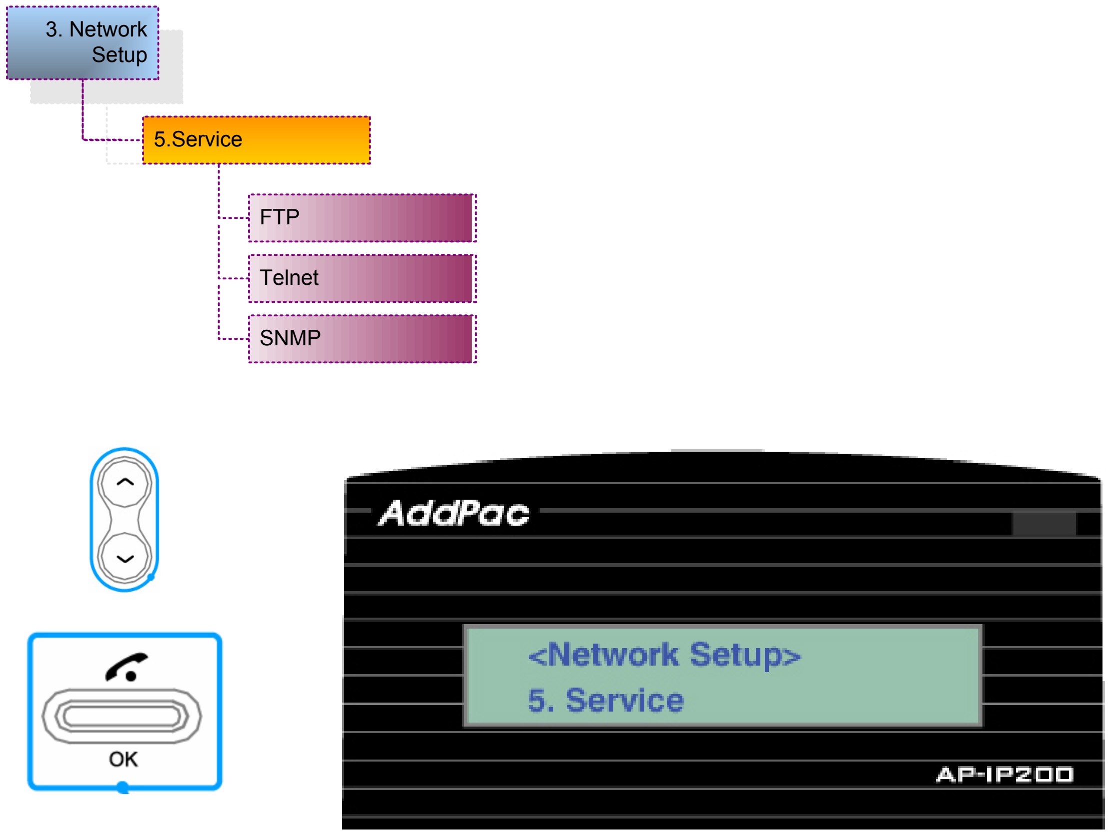


Figure 4-34 Service Setup Menu

Service Setup	Description
<Service Setup> FTP : Enable	Actives/Deactivates the FTP service protocol. Default is enable mode (activating FTP service). Default port number is 21.
<Service Setup> Telnet : Enable	Actives/Deactivates the TELNET service protocol. Default is enable mode (activating TELNET service). Default port number is 23.

Actives/Deactivates the SNMP service protocol. Default is enable mode (activating SNMP service). Default port number is 161

*

System Setup Menu

The system setup includes the Display Name, Ring volume control, input/output volume control.

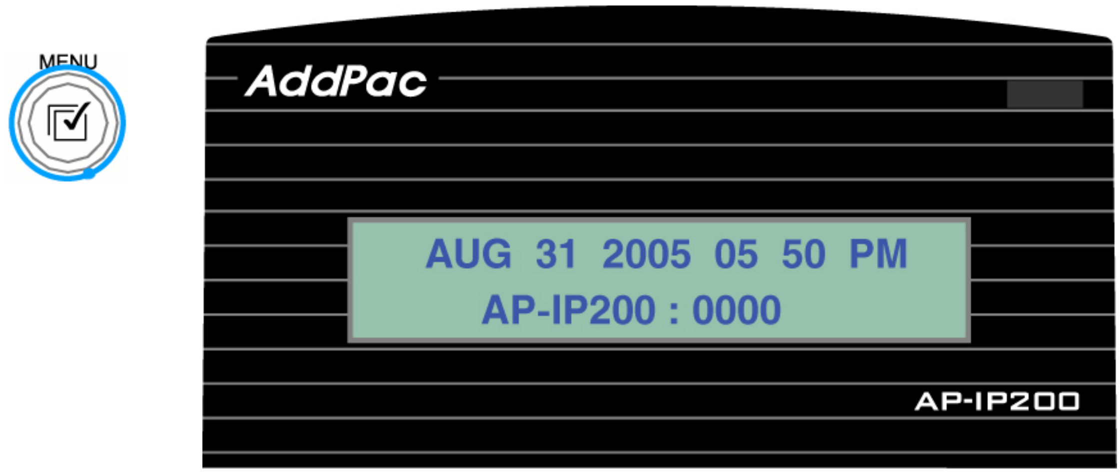


Figure 4-35 Main Menu

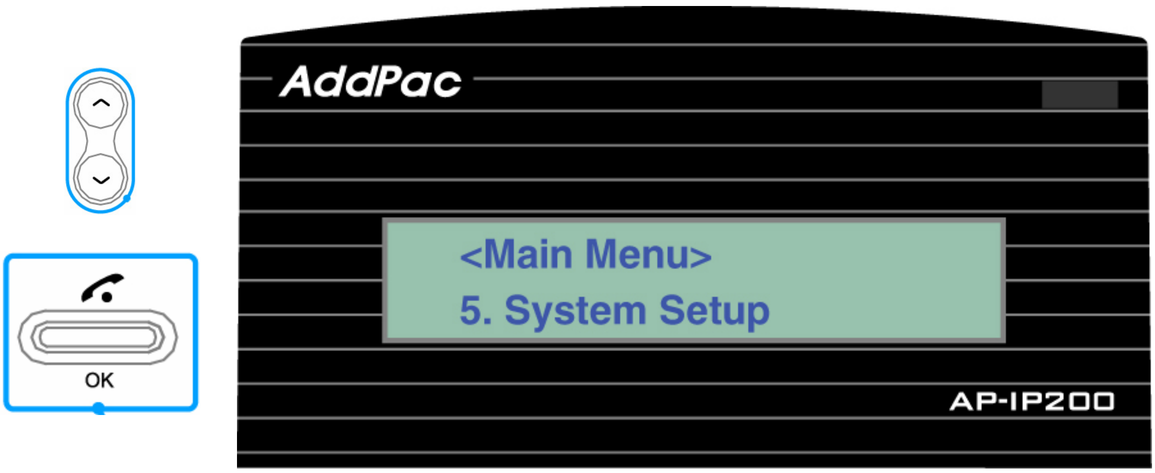
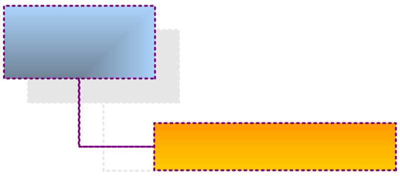


Figure 4-36 System Setup Menu

System Setup – Display Name

This is Name displayed on LCD panel. Default name is AP-IP200. If user wants to change default name, change or modify the default name at Display Name in System Setup Menu.



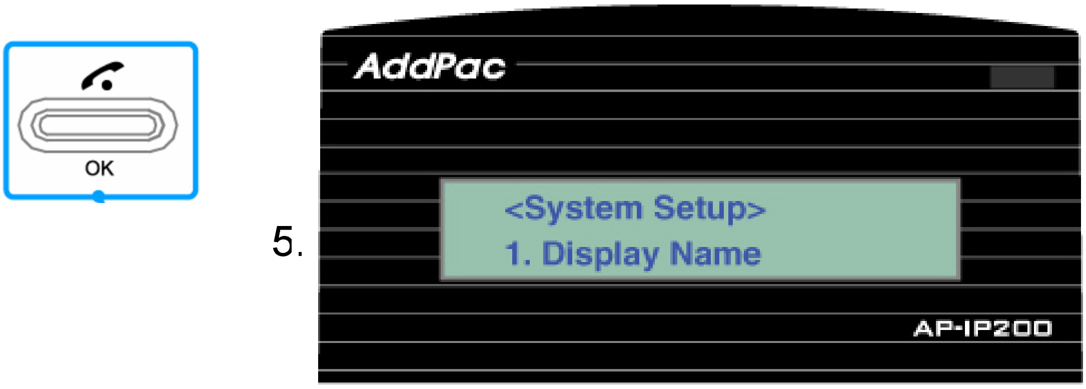


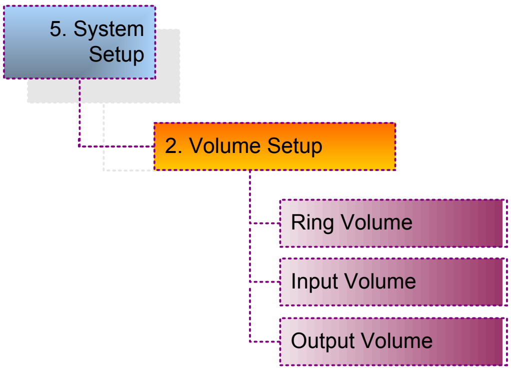
Figure 4-37 Display Name Menu

1. Display Name

Display Name	Description.
	Enter the Name you want (ex: Phone number, room number, office name). Default is AP-IP200 : 0000

System Setup — Volume Control

This volume menu controls the Ring volume, Input/Output volume (Speaker volume, Handset volume) of AP-IP200 IP Phone.



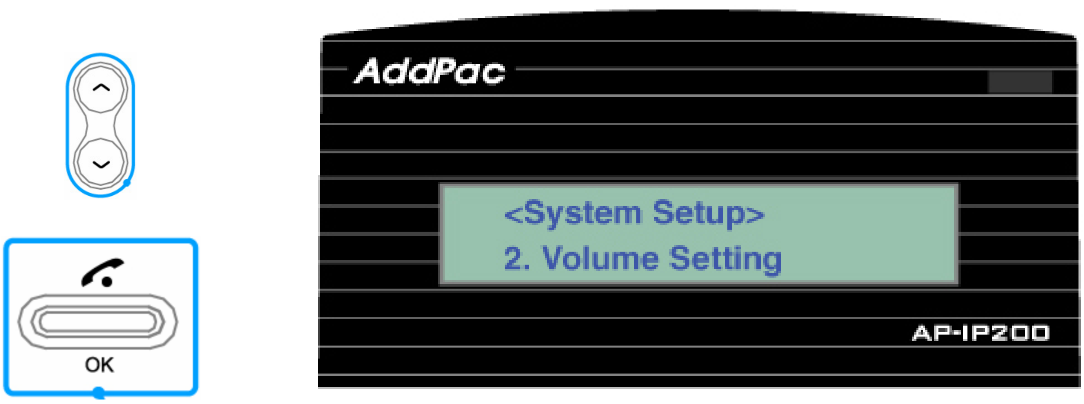
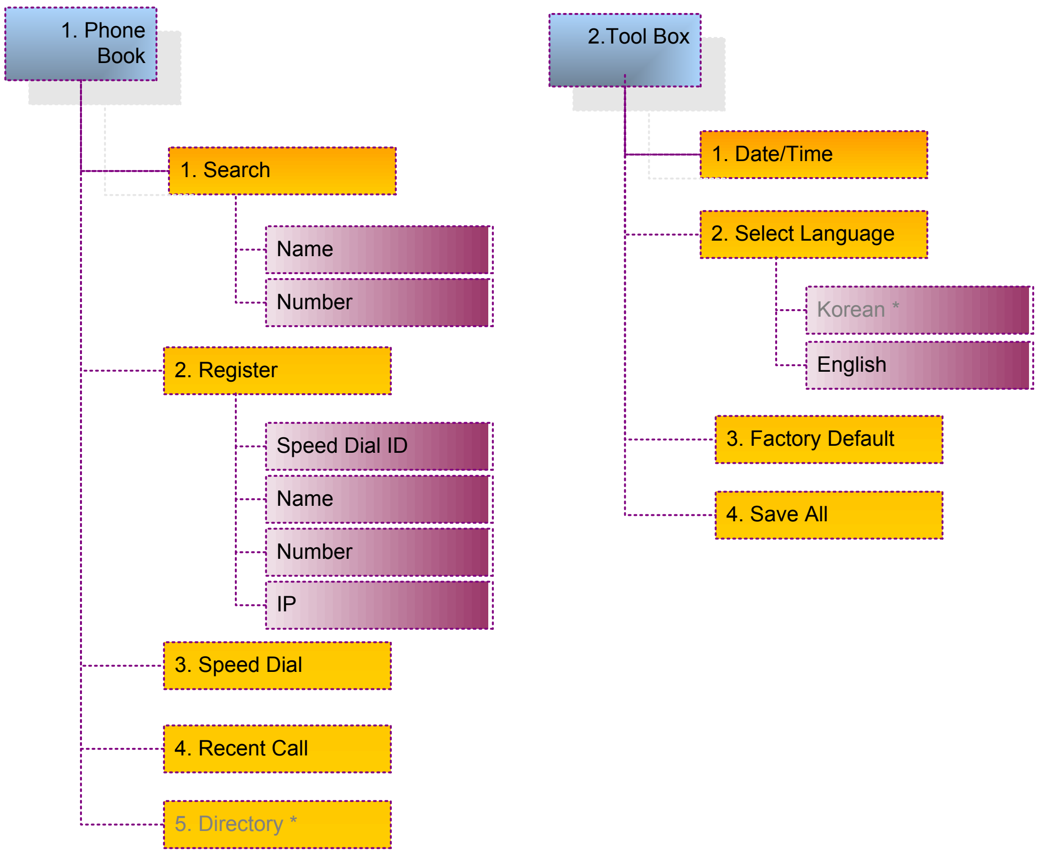


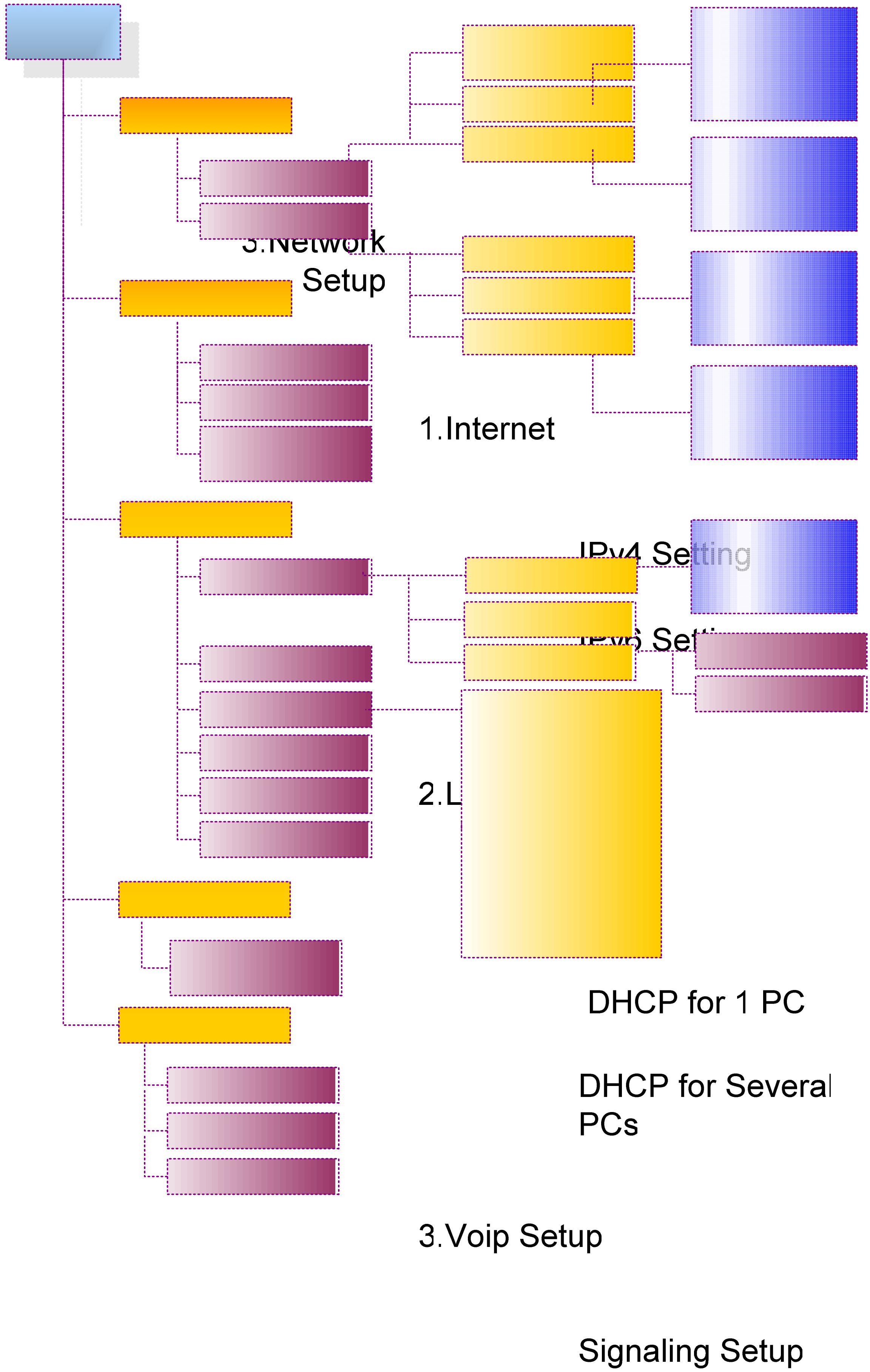
Figure 4-38 Volume control Menu

Volume Control	Description
<Volume Setup> Ring Volume : 4	Select the ring volume of internal speaker. Default ring volume level is 4.
<Volume Setup> Input Volume : 4	Select the input volume of Speaker Phone or Handset MIC. Default input volume level is 4.
<Volume Setup> Output Volume : 4	Select the output volume of Speaker Phone or Handset Speaker. Default input volume level is 4.

AP-IP200 IP Phone User Interface Menu Tree

If user presses the MENU button, user can see the all MENU tree shown as Figure 4-39. Also, user uses the MENU KEY at conversation mode.




3.Network
Setup
1.Internet
IPv4 Setting
IPv6 Set
2.L
DHCP for 1 PC
DHCP for Several
PCs
3.Voip Setup
Signaling Setup

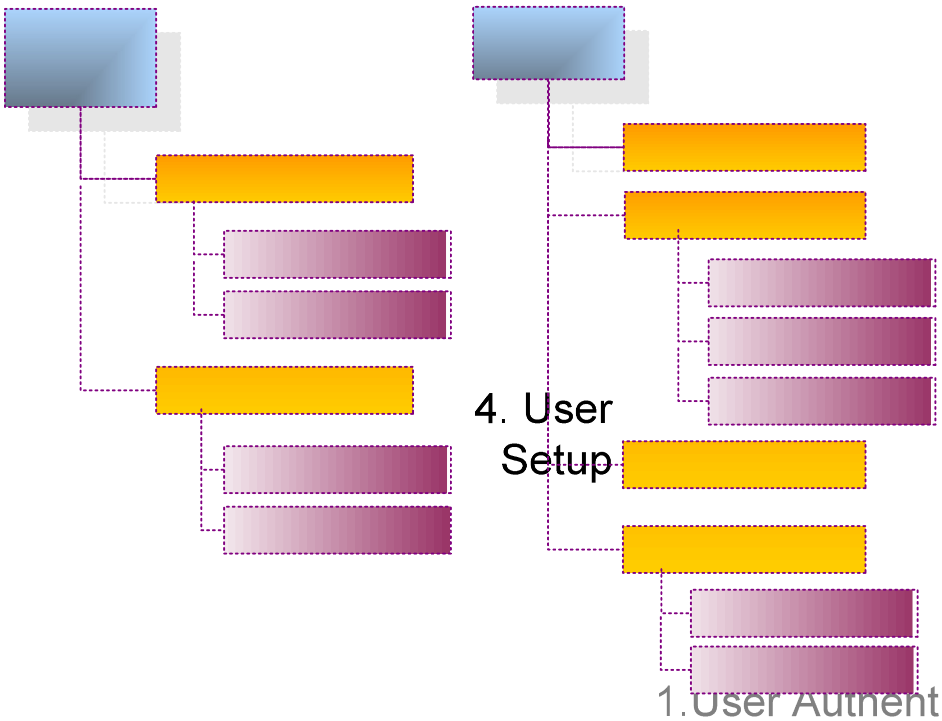


Figure 4-39 AP-IP200 Menu Tree

ID

Password

2.Voice Mail *

New Message

Saved Message

KEY Input Method

This parts explains the KEY input method and available characters used in AP-IP200

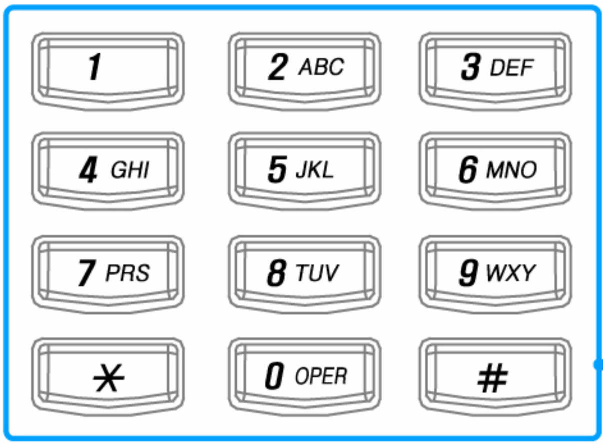


Figure 4-40 KEY PAD Diagram

[Table 4-2] Available Characters and KEY Input Procedures

KEYs	Characters	Description
1	**1 < > & ( )**	If pressed continuously, displayed character is changed.
2	**2 a b c A B C**	"
3	**3 d e f D E F**	"
4	**4 g h I G H I**	"
5	**5 j k l J K L**	"
6	**6 m n o M N O**	"
7	**7 p r s P R S**	"
8	**8 t u v T U V**	"
9	**9 w x y z W X Y Z**	"
0	**0 ~ = _ – ^**	"
*****	**. : * [ ] ; ?**	"
#	**# / ! @ $ % **	"
F1	**Back Space**	BackSpace Function
F2	**Space**	Space Function

**** F2(Space) function** ⇒ if user want to input same character continuously after first character, user should input the second character after F2 Key or waiting 2 second.

Ex1) Apple Input ⇒ “ 2 5time + 7 2 time + F2 + 7 2time + 5 4time + 3 3time “

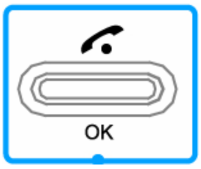
Ex2) 2005/09/14 Input ⇒ “ 2 + 0 + F2 + 0 +5 + # 2 time + 0 + 9 + # 2time + 1 + 4

Ex3) 2aB Input ⇒ “ 2 + F2 + 2 2time + F2 + 2 6time”

Call/Cancel Button

This Call button is used for various call scenarios shown as Table 4-3.

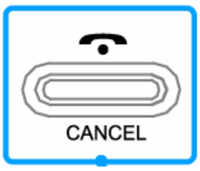
.



[Table 4-3] Key PAD Call Button Usage

Call Button Usage	Description
Recent Call	If user presses the Call button at idle mode, recent call history is listed. If user presses the Call button once again after searching the recent call history, user can dial-out the peer’s phone number.
Dial-Out	If user presses the Call button after inputting dial number at idle mode, AP-IP200 starts dial-out signal to counter part user. Also, this Call button starts the dial-out signal to remote peer in Speed-Dial and Recent Cal Menu.
Call Hook-off	When VoIP call is coming from remote user, if user presses the this Call button, user can receive VoIP call via Speaker Phone function.
Confirm (OK)	If user want to apply after parameter setting in all Menu Tree, This Call button is used as a OK button

This Cancel button is used for various call cancel /on-hook shown as Table 4-4.



[Table 4-4] Key PAD Cancel Button Usage

Cancel Button Usage	Description
On-Hook	The Cancel button cancels the call at conversation mode.

Call Cancel	While the VoIP call is proceeding using Call button, if user presses the Cancel Button, Dial-out Call proceeding is cancelled.
Menu Tree level change (Move to One(1) Upper Level)	If user presses the Cancel button in UI menu tree, Menu tree is changed from current level to one (1) upper level.

Chapter 5. AP-IP200 Network Configuration

Basic CLI Command for Network Setup

＊ CLI command for current network status

```
IP200# show run
Building configuration...

Current configuration:
!
hostname IP200
!
username root password router administrator
!
!
interface Loopback0
 ip address 127.0.0.1 255.0.0.0
!
interface FastEthernet0/0
 ip address 172.20.103.100 255.255.0.0
 speed auto
!
interface FastEthernet0/1
 no ip address
 speed auto
!
 --More--
```

* IP address and default router setting

```
IP200# configure terminal
IP200(config)#
IP200(config)# interface FastEthernet 0/0          → Fast ethernet Interface 0 Port
IP200(config-if)# ip address 172.20.103.1 255.255.0.0    → IP address setting
IP200(config-if)# VOIP_INTERFACE_DOWN
VOIP_INTERFACE_DOWN
VOIP_INTERFACE_UP : (172.20.103.1)

IP200(config-if)# exit
IP200(config)#
IP200(config)# ip route 0.0.0.0 0.0.0.0 172.20.1.1    → default router
IP200(config)#
IP200(config)# end                                 → end of configuration
IP200#
IP200# write                                       → flash writing for parameter save
Proceed with write? [confirm]y                     → confirm
Building configuration...
[OK] Configuration saved to flash:/apos.cfg
IP200#
```

* If finished network configuration, start the ping test to check

```
IP200# ping -c 5 172.20.1.1
PING 172.20.1.1 (172.20.1.1): 56 data bytes
64 bytes from 172.20.1.1: icmp_seq=0 ttl=255 time=0 ms
64 bytes from 172.20.1.1: icmp_seq=1 ttl=255 time=5 ms
64 bytes from 172.20.1.1: icmp_seq=2 ttl=255 time=5 ms
64 bytes from 172.20.1.1: icmp_seq=3 ttl=255 time=5 ms
64 bytes from 172.20.1.1: icmp_seq=4 ttl=255 time=5 ms
--- 172.20.1.1 ping statistics ---
5 packets transmitted, 5 packets received, 0% packet loss'
round-trip min/avg/max = 0/4/5 ms
IP200#
```

If ping test from AP-IP200 to default router is OK, network configuration setup is finished.

Chapter 6. Appendix

This Appendix provides information about the Pinout specifications of the following cables used with AP-IP200 IP Phone.

- Console Port Signal and Pinout (RJ-45 to DB9)
- Ethernet UTP Cable Assemble (RJ-45 to RJ-45) Pinout

[RS-232C Console Port Signal & Pinout]

In order to connect RS-232C console port with the Terminal Emulating PC, the RJ-45 to DB9 (Female DTE Connector) cable is used. The transferred signal and Pinout specifications are enlisted in the following table.

[Table 6-1] The signal and Pinout specification

RS-232C Console Port (DTE)	RJ-45	DB-9	Console Port (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

[UTP Cable (RJ-45 to RJ-45) Pinout Specification]

In order to connect the LAN port of this equipment with other equipments (i.e. HUB), the RJ-45 to RJ-45 Ethernet Cable is used. The RJ-45 Connector Pin sequence is provided below and the signal and Pinout specifications are enlisted at the below table.

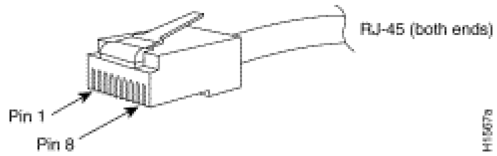


Figure 6-1 100Base-TX RJ-45 Connector

[Table 6-2] Signal and Pinout of Direct Ethernet Cable

RJ-45	Signal	Direction	RJ-45 PIN
1	**Tx +**	→	1
2	**Tx -**	→	2
3	**Rx +**	←	3
4	-	-	4
5	-	-	5
6	**Rx -**	←	6
7	-	-	7
8	-	-	8

1. These specifications are for ethernet direct cables connecting this equipment and HUB.
2. For IP Phone to IP Phone or IP Phone to PC connection, the Cross Cable must be used.

Glossary

Terms	Definition & Description
ADSL	An acronym for Asymmetric Digital Subscriber Line, ADSL is a method of transmitting data over traditional copper telephone lines. Data can be downloaded at speeds of up to 1.544 Megabits per second and uploaded at speeds of 128 Kilobits per second (asymmetric).
AP-VPMS	An acronym for VoIP Plug & Play Management Software. AddPac Technology developed integrated management software for VoIP product remote installation, real-time monitoring, network management on Graphic User Interface (GUI).
API	An acronym for Application Programming Interface, an Interface which is used for accessing an application or a service from a program.
APOS	An acronym for AddPac Internetworking Operation System, AddPac Technology developed operating system for network devices.
ATM	An acronym for Asynchronous Transfer Mode. It an International Cell Relay standard sending various service such as voice, video and data as fixed size (53bytes) cells. With the fixed size cells, the cell processing is mainly done by hardware, so the transmission delay is significantly reduced. ATM is designed for high transmission media such as E3, SONET, T3.
ATM Information Super-highway	Starting from '1993, ATM information Super-highway was established to offer data service and internet service to public offices by the Korean government. Data service includes ATM, Dedicated line, packet switching, Frame relay and Internet service includes Internet compound service and internet service via ATM access lines.
ATM Forum	Establish by Cisco Systems, NET/ADAPTIVE, Northern Telecom, Sprint in '1991 for the development and acceleration of ATM technology star nards. It encompasses the standard by ANSI and ITU-T, and further develops the agreed terms of ATM standard.
Authentication	Authentication ensures that digital data transmissions are delivered to the intended receiver. Authentication also assures the receiver of the integrity of the message and its source (where or whom it came from).
BNC Connector	A standard connector connecting IEEE 802.3 10Base-2 coaxial cable to MAU(Media Access Unit).
Boot Loader	The built-in chip on the printed circuit board generating booting command of network equipment.
Bps	Bits per second. Refer to: bit rate.

Cable Modem	A modem designed to operate over cable TV lines. Because the coaxial cable used by cable TV provides much greater bandwidth than telephone lines, a cable modem can be used to achieve more bandwidth. Cable network also requires modularization and demutualization process while sending the data.
Call Center	A call center is a central place where customer and other telephone calls are handled by an organization, usually with some amount of computer automation. Typically, a call center has the ability to handle a considerable volume of calls at the same time, to screen calls and forward them to someone qualified to handle them, and to log calls. Call centers are used by mail-order catalog organizations, telemarketing companies, computer product help desks, and any large organization that uses the telephone to sell or service products and services.
Caller ID	A feature that displays the name and/or number of the calling party on the phone's display when an incoming call is received. Virtually all digital phones - as well as many analog phones - have this capability. While typically only the number is received, most phones will display the name, if the number matches an entry in the phone's built-in phone book.
Category 5 cabling	unshielded twisted pair (UTP) cabling. An Ethernet network operating at 10 Mbits/second (10BASE-T) will often tolerate low quality cables, but at 100 Mbits/second (10BASE-Tx) the cable must be rated as Category 5, or Cat 5 or Cat V, by the Electronic Industry Association (EIA).
CBR	Constant Bit Rate. A data transmission that can be represented by a non-varying, or continuous, stream of bits or cell payloads. Applications such as voice circuits generate CBR traffic patterns. CBR is an ATM service type in which the ATM network guarantees to meet the transmitter's bandwidth and Quality of Service requirements
CES	An acronym for Circuit Emulation Service. enables users to multiplex or to concentrate multiple circuit emulation streams for voice and video with packet data on a single, high-speed ATM link without a separate ATM access multiplexer.
Checksum	A computed value which is dependent upon the contents of a packet. This value is sent along with the packet when it is transmitted. The receiving system computes a new checksum based upon the received data and compares this value with the one sent with the packet. If the two values are the same, the receiver has a high degree of confidence that the data was received correctly.
Coaxial cable	A cable with a single inner conductor with foam insulation and braided shield. There are two types of this cable; 50Ω cable for digital signaling process and

	75Ω cable for analog signal process and high speed digital signal process.
CODEC	An acronym for COder-DECoder 1. Built-in circuit device for coding/decoding of analog signal to bit stream with Pulse Code Modulation method. 2. DSP software algorithm for compressing/ decompressing voice or audio signal
Console	DTE interface whether the command is delivered to the host.
CoS	Class of Service (CoS) is a way of managing traffic in a network by grouping similar types of traffic (for example, e-mail, streaming video, voice, large document file transfer) together and treating each type as a class with its own level of service priority. Unlike Quality of Service (QoS) traffic management, Class of Service technologies do not guarantee a level of service in terms of bandwidth and delivery time; they offer a "best-effort."
Decryption	The process of converting encrypted data back into its original form, so it can be understood.
DHCP	Dynamic Host Configuration Protocol. A protocol which allows a host to obtain configuration information, such as its IP address and the default router from a server. This simplifies network administration because the software keeps track of IP addresses. With DHCP device can have a different IP address every time it connects to the network
DNS	Domain Name Server, an Internet service that translates domain names into IP addresses.
DS-3	Digital signal level 3, A line capable of delivering 44.7 Mbps (44,700 Kbps) in both directions
DSP	Digital Signal Processor. Dedicated microprocessor for digital signal process.
DTMF	Dual Tone Multi-Frequency. Using two types of voice-band tones for dialing.
E&M	An acronym for recEive and transmit or ear and mouth. E&M interface uses a RJ-48 telephone cable to connect remote calls from an IP network to PBX trunk lines (tie lines) for local distribution. It is a signaling technique for two-wire and four-wire telephone and trunk interfaces.
E1	The basic building block for European multi-megabit data rates, with a bandwidth of 2.048Mbps.
Encryption	the manipulation of a packet's data in order to prevent any but the intended recipient from reading that data.
Ethernet	Broadband LAN standard initiated by Xerox Corporation and co-developed by Intel and DEC. Utilizing CSMA/CD and the various cables of 10Mbps are used. It is similar to IEEE 802.3. Refer to: 10Base-2, 10Base5, 10Base-F, 10Base-T, 10Broad-36, Fast Ethernet, IEEE 802.3.
FAX	Short for "FACSimile." In essence, a fax machine sends an electronic "facsimile" or copy of the document. An optical scanner in the machine scans

	the document and the resulting bit stream is then sent to the receiving machine via telephone line. The transmission and the reproduction at a distance of still pictures printed matter and similar documented material
Frame	data that is transmitted between network points as a unit complete with addressing and necessary protocol control information. A frame is usually transmitted serial bit by bit and contains a header field and a trailer field that "frame" the data. (Some control frames contain no data.)
Frame-Relay	Switching type Data Link Layer Protocol. Using HDLC capsule, process multi-number of virtual circuits between devices.
FTP	an acronym for File Transfer Protocol, a very common method of transferring one or more files from one computer to another. Defined at RFC 959.
FXO	Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone.
FXS	Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone.
G.711	Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs.
G.723.1	Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This CODEC has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility.
G.726	Describes ADPCM coding at 40, 32, 24 and 16 kbps. ADPCM encoded voice can be interchanged between packet voice, PSTN, and PBX networks if the PBX networks are configured to support ADPCM. Described in the ITU-T standard in its G-series recommendations.
G.728	Describes a 16 kbps low-delay variation of CELP voice compression. CELP voice coding must be translated into a public telephony format for delivery to or through the PSTN. Described in the ITU-T standard in its G-series recommendations..
Gatekeeper	The component of an H.323 conferencing system that performs call address resolution, admission control, and subnet bandwidth management. H.323 entity on a LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper can provide other services to the H.323 terminals and gateways, such as bandwidth

	management and locating gateways. A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup and request admission to a call from the gatekeeper.
H.225	An International Telecommunication Union (ITU-T) standard for H.225.0 session control and packetization. It defines various protocols of RAS, Q.931, RTP and etc.
H.245	An International Telecommunication Union (ITU-T) standard for H.245 end-point control.
H.323	An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing.
HBD3	Line code type of E1 line.
HDLC	An acronym for High-Level Data Link Control. A transmission protocol for the Data Link Layer. In HDLC, data is organized into a unit (called a frame) and sent across a network to a destination that verifies its successful arrival. Variations of HDLC are also used for the public networks that use the X.25 communications protocol and for frame relay, a protocol used in both and wide area network, public and private.
Hookflash	Short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone is attempting to perform a dial-tone recall from a PBX. Hookflash is often used to perform call transfer.
HTTP	An acronym for Hypertext Transfer Protocol. A file transfer protocol used by web browser or web server for transmitting text or graphic files.
IPSec	Internet Protocol Security protocol, a framework for a set of protocols for security at the network or packet processing layer of network communication. Earlier security approaches have inserted security at the Application layer of the communications model. IPsec is said to be especially useful for implementing virtual private networks and for remote user access through dial-up connection to private networks. A big advantage of IPsec is that security arrangements can be handled without requiring changes to individual user computers. Cisco has been a leader in proposing IPsec as a standard (or combination of standards and technologies) and has included support for it in its network routers.
IPv6	IPv6 (Internet Protocol Version 6) is the latest level of the Internet Protocol (IP) and is now included as part of IP support in many products including the major computer operating systems. IPv6 has also been called "IPng" (IP Next Generation). Formally, IPv6 is a set of specifications from the Internet Engineering Task Force (IETF). IPv6 was designed as an evolutionary set of improvements to the current IP Version 4. Network hosts and intermediate

	nodes with either IPv4 or IPv6 can handle packets formatted for either level of the Internet Protocol. Users and service providers can update to IPv6 independently without having to coordinate with each other.
ISP	An ISP (Internet service provider) is a company that provides individuals and other companies access to the Internet and other related services such as Web site building and virtual hosting. An ISP has the equipment and the telecommunication line access required to have a point-of-presence on the Internet for the geographic area served. The larger ISPs have their own high-speed leased lines so that they are less dependent on the telecommunication providers and can provide better service to their customers. Among the largest national and regional ISPs are AT&T WorldNet, IBM Global Network, MCI, Netcom, UUNet, and PSINet.
ITU-T	The ITU-T (for Telecommunication Standardization Sector of the International Telecommunications Union) is the primary international body for fostering cooperative standards for telecommunications equipment and systems. It was formerly known as the CCITT. It is located in Geneva, Switzerland
IVR	Interactive Voice Response (IVR) is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media. IVR is usually part of a larger application that includes database access. Common IVR applications include: Bank and stock account balances and transfers.
LAN	A local area network is a group of computers and associated devices that share a common communications line and typically share the resources of a single processor or server within a small geographic area (for example, within an office building). LAN standard defines cable connection and signal processing on Physical Layer and Data Link Layer.
Link	Network communication channels consisting of sending and receiving devices, circuits, transmission path. Usually refer to WAN connection. Referred as Line, or transmission link.
Loopback test	A loopback test is a test in which a signal in sent from a communications device and returned (looped back) to it as a way to determine whether the device is working right or as a way to pin down a failing node in a network.
MAC Address	Standardized data link layer address that is required for every port or device that connects to a LAN. Other devices in the network use these addresses to locate specific ports in the network and to create and update routing tables and data structures. MAC addresses are 6 bytes long and are controlled by the IEEE. Also known as a hardware address, MAC-layer address, and

	physical address. Compare with network address.
MAN	A data network designed for a town or city. MANs are considered larger than LANs but smaller than WANs. Compare with: LAN, WAN.
MGCP	MGCP, also known as H.248 and Megaco, is a standard protocol for handling the signaling and session management needed during a multimedia conference. The protocol defines a means of communication between a media gateway, which converts data from the format required for a circuit-switched network to that required for a packet-switched network and the media gateway controller. MGCP can be used to set up, maintain, and terminate calls between multiple endpoints. Megaco and H.248 refer to an enhanced version of MGCP
NAT	NAT (Network Address Translation) is the translation of an Internet Protocol address (IP address) used within one network to a different IP address known within another network. One network is designated the inside network and the other is the outside.
NTP	Network Time Protocol (NTP) is a protocol that is used to synchronize computer clock times in a network of computers. In common with similar protocols, NTP uses Coordinated Universal Time (UTC) to synchronize computer clock times to a millisecond, and sometimes to a fraction of a millisecond.
PABX	Private Automatic Branch Exchange. A telephone switch for use inside a corporation. It connects offices (internal extensions) with each other and provides access (typically by dialing an access number such as 9) to the public telephone network PABX is the preferred term in Europe, PBX is used in the USA.
Packet	Packets contain a source and destination address as well as the actual message. Packets also known as Datagrams.
PBX	A PBX (private branch exchange) is a telephone system within an enterprise that switches calls between enterprise users on local lines while allowing all users to share a certain number of external phone lines.
PING	Packet INternet Groper, a packet (small message) sent to test the validity / availability of an IP address on a network
Point to Point Connection	Basic connection type. In ATM, point to point connection is half duplex connection between two ATM end systems or full duplex connection.
Pont to Multipoint Connection	Basic connection type. In ATM, point to multipoint connection is half duplex connection among one sending end system (root node) and multiple receiving end system. Compare with: point-to-point connection.
POTS	Plain Old Telephone Service. Compare with: PSTN.
PPP	The most popular method for transporting IP packets over a serial link

	between the user and the ISP. Developed in 1994 by the IETF and superseding the SLIP protocol, PPP establishes the session between the user's computer and the ISP using its own Link Control Protocol (LCP). PPP supports PAP, CHAP and other authentication protocols as well as compression and encryption.
Protocol Stack	Any set of communication protocols, such as TCP/IP, that consists of two or more layers of software and hardware. It's called a stack because each layer builds on the functionality in the layer below
PSTN	Public Switched Telephone Network – term for the entire, world-wide telephone network. Sometimes refers to as POTS.
PVC	Permanent Virtual Circuit or permanent virtual connection. A continuously available communications path that connects two fixed end points.
Q.931 Signaling	ITU-T specification for network layer of ISDN. Q.931 uses out-of-band signaling on the D-channel to control calls.
QoS	This refers to the assumption that data transmission rates, error rates, and other characteristics can be measured, improved, and to some degree, guaranteed in advance. Basically, QoS describes a collective measure of the level of service a provider delivers to its customers or subscribers.
RAM	Random-Access Memory, a non-retentive memory, whose contents get lost after a switch-off or reset. Application programs run in the random access memory and data is stored and processed.
RAS	Registration Admission Status protocol. The communication protocol used to convey registration, admission and status messages between H.323 endpoints and the gatekeeper.
RISC	Reduced Instruction Set Computing
Router	On the Internet, a router is a device or, in some cases, software in a computer, that determines the next network point to which a packet should be forwarded toward its destination. The router is connected to at least two networks and decides which way to send each information packet based on its current understanding of the state of the networks it is connected to. A router is located at any gateway (where one network meets another), including each Internet point-of-presence. A router is often included as part of a network switch. Compare with: gateway. Refer to: relay.
RS-232	Most common Physical Layer interface. Known as EIA/TIA-232.
RTCP	Real-time Control Protocol (RTCP) is a companion protocol of RTP that is used to maintain quality of service. Refer to: RTP(Real-Time Transport Protocol).
RTP	1. Routing Table Protocol, VINES routing protocol based on RIP. Distributes

	network topology, and aids VINES servers in finding neighboring clients, servers, and routers. Uses delay as a routing metric. Refer to: SRTP. 2. Rapid Transport Protocol. Provides pacing and error recovery for APPN data as it crosses the APPN network. With RTP, error recovery and flow control are done end-to-end rather than at every node. RTP prevents congestion rather than reacts to it. 3. Real-Time Transport Protocol. Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time-stamping, and delivery monitoring to real-time applications.
SIP	The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality. Like HTTP or SMTP, SIP works in the Application layer of the Open Systems Interconnection (OSI) communications model. The Application layer is the level responsible for ensuring that communication is possible. SIP can establish multimedia sessions or Internet telephony calls, and modify, or terminate them. The protocol can also invite participants to unicast or multicast sessions that do not necessarily involve the initiator. Because the SIP supports name mapping and redirection services, it makes it possible for users to initiate and receive communications and services from any location, and for networks to identify the users whatever they are. SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol, such as UDP, SCTP, or TCP. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination. The Session Initiation Protocol is specified in IETF Request for Comments [RFC] 2543.
SmartViewer	The real-time monitoring, statistical data search and management GUI based software developed by AddPac Technology for AP-GK1000, AP-GK2000, AP-GK3000 models.
SNMP	Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics

	collection, performance, and security. Refer to: SGMP, SNMP2.
T1	A TDM physical transmission standard consisting of two twisted wire pairs and related equipment capable of carrying a 1.544 Mbps DS-1 signal. Term often used interchangeably with DS-1. Refer to: AMI, B8ZS, DS-1.
TCP/IP	Transmission Control Protocol/Internet Protocol, The protocol suit developed by DoD (USA) in 1970s for the worldwide inter-network development. TCP & IP is the most well known protocols of the suite. Refer to: IP, TCAP.
Telco	Telephone Company, referring to the company offering telephone service to customers. Typically, it refers to an individual company such as Bell operating company offering local telephone service, however, sometimes local telephony service providers are included.
Telnet	Standard Terminal Emulation program covered by TCP/IP protocol stack. Used for remote terminal connection. Via Telnet, users can log-in to the system and operate the resources as working on the local system. Defined on RFC 854.
VCI	the address or label of a VC; a value stored in a field in the ATM cell header that identifies an individual virtual channel to which the cell belongs. VCI values may be different for each data link hop of an ATM virtual connection.
VDSL	New DSL technology that accepts bandwidths of up to 27 Mbps over relatively short distances. VDSL, in the process of being standardized, allows symmetric or asymmetric throughputs that are much higher than other xDSL standards (up to 27 Mbps when downloading and 3 Mbps when uploading under asymmetric or 14 Mbps in symmetric), as well as the simultaneous transport of ISDN (Numeris) services but with much shorter ranges that do not exceed 900 m to 1 km. In practice, this technique may require the deployment of optical remotes and the setting up of active equipment in the local loop. Compare with: ADSL, HDSL, SDSL.
VoATM	Voice Over ATM. Voice over ATM enables an ATM switch to carry voice traffic (for example, telephone calls and faxes) over an ATM network. When sending voice traffic over ATM, the voice traffic is encapsulated using AAL1/AAL2 ATM packets.
VoFR	Voice Over Frame Relay. Voice over Frame Relay enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network using FRF.12 encapsulation.
VoHDLC	Voice Over HDLC. Voice over HDLC enables a router to carry live voice traffic (for example, telephone calls and faxes) back-to-back to a second router over

	a serial line.
VoIP	VoIP (Voice delivered using the Internet Protocol) is a term used in IP telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.
VPN	Virtual Private Network, VPN allows IP traffic to travel securely over a public TCP/IP network by encrypting all traffic from one network to another. A VPN uses "tunneling" to encrypt all information at the IP level.
WAN	A network that covers a large geographical area. Typical WAN technologies include point-to-point, X.25 and frame relay. Compare with: LAN, MAN.

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