AP-IP230 IP Phone

[Installation and Operation Guide]

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AddPac Technology

www.addpac.com

AP-IP230 IP Phone

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AP-IP230 IP Phone Document Revision

Version	Date	Change	Written By
Version 1.00	2012/1/27	Initial Released	AddPac Technology

Chapter 1. AP-IP230 Overview

AP-IP230 IP phone is designed to provide enhanced IP telephony functionality to meet the wide range of business user requirements. This high performance IP phone 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 5 Inch Color LCD with built-in touch screen, wide variety of feature keys, customizable soft-keys, two(2) ethernet ports, the latest QoS, public IP sharing. It supports not only the major VoIP signaling protocols concurrently such as SIP, H.323 but also G.711, G.726, G.729, G.723.1 voice codec, stereo audio in/out interfaces for external headset, internal speaker and MIC for high quality Speaker Phone, etc

New Paradigm for IP Telephony : Telephony + Broadcasting

AP-IP230 IP telephone combines AddPac's field proven VoIP technology and IP voice broadcasting technology. AP-IP230 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 level information display with 5 inch large size Color LCD mounted. Since AP-IP230 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 IPNext20, IPNext50, IPNext180, IPNext200, and IPNext500 on comprehensive IP-PBX system of AddPac Technology, 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 MS window based Smart Messenger.

Advanced IP Telephony Solution", Touch Screen based Speed-Dialing Service with User Presence Indication Features

AP-IP230 IP phone provides touch screen based 25 key buttons (5 column x 5 row) for speed-dialing and user presence service. User Information assigned each touch screen key button is easily registered using web based management. System operator can make several example user presence profiles for AP-IP230 user. AP-IP230 user select one among several candidate user presence files and can modify selected user presence profile for individual personal user presence

profile. User presence profiles provided via system operator can be upgraded via web based management, if user information is changed or deleted due to several reasons (for example, resign). Because user information is frequently changed, it is better to use default user presence profile provided by system. User presence information such as user busy (LAMP ON), ringing (LAMP BLINK) is individually displayed on LCD graphically of AP-IP230 Key Buttons.

Adapt to the Future Environment : Firmware Upgradeable Technology

Designed on programmable high performance RISC Integrated DSP Technology, AP-IP230 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-IP230 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-IP230 to offer voice broadcasting service, incorporated with in-house broadcasting system.

IP telephony with Outstanding Network Service Capability

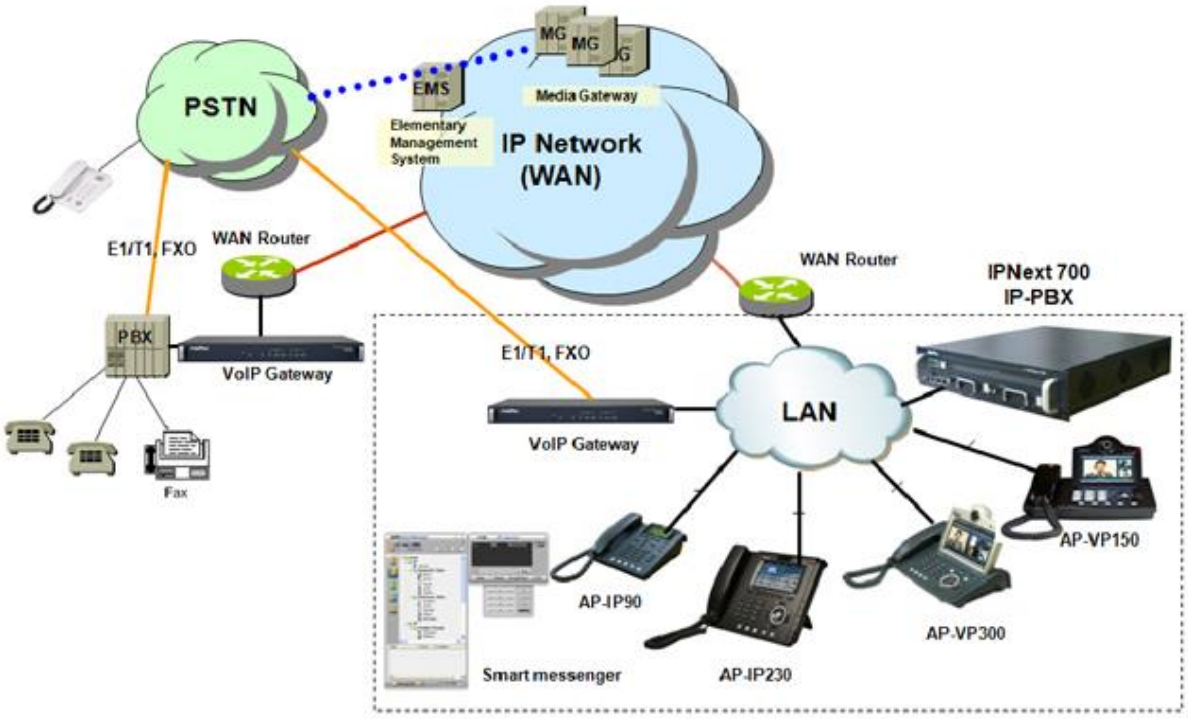
Not only IP telephony, AP-IP230 is an integrated, feature-rich network equipment delivering routing, NAT/PAT, DHCP Server/Relay, Public IP sharing and QoS. In today's mixed network of xDSL, Cable, FTTH, Metro Ethernet, Metro ATM, Leased line and dynamic IP environment, not only the ample network service features, but also high-end QoS (Quality of Service) and security features are requested. Based on two(2) 10/100Mbps Fast Ethernet ports, AP-IP230 offers integrated network and security service of LAN-to-LAN routing, bridge and NAT/PAT. Moreover, AP-IP230 supports H.323, SIP dual VoIP signaling protocols concurrently. So the customers easily migrate to different service providers' networks utilizing different VoIP signaling protocols.

Privacy and Encryption Features

AP-IP230 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.

AddPac's various VoIP gateway series, multi-service routers, video products 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-IP230 with which customers can

best optimize high performance, market strategy and budget for next-generation communication solution.



(Picture 1-1) AP-IP230 IP Phone Network Configuration

AP-IP230 IP Phone Hardware Specification

[Table 1-1] AP-IP230 IP Phone Hardware Specification Table

Category		Specification
Model		**AP-IP230**
Product		IP Phone (built in speaker)
Microprocessor		High performance RISC microprocessor
Digit and Key Button		User presence (25 user) + Touch screen based speed dialing
LCD Screen	Graphic LCD	5 inch color LCD + touch screen function
Memory	Boot Memory	512KB flash memory
	System Memory	8MB flash memory
	Main Memory	64MB high performance SDRAM
Audio Interface	Audio Input	1 port 3.5mm Male stereo in/out input
	Audio Output	1 Port 3.5mm Male Stereo out audio output
Ethernet Interface	LAN 0 access Ethernet port	1-port 10/100Mbps fast Ethernet RJ-45 connector
	LAN 1 access Ethernet port	1-port 10/100Mbps fast Ethernet RJ-45 connector
Power Supply	Power Input	External AC110~220V 50/60Hz, 5V, 3A power supply
Hardware	Structure, Material	ABS material compact phone hardware

AP-IP230 IP Phone Software Feature

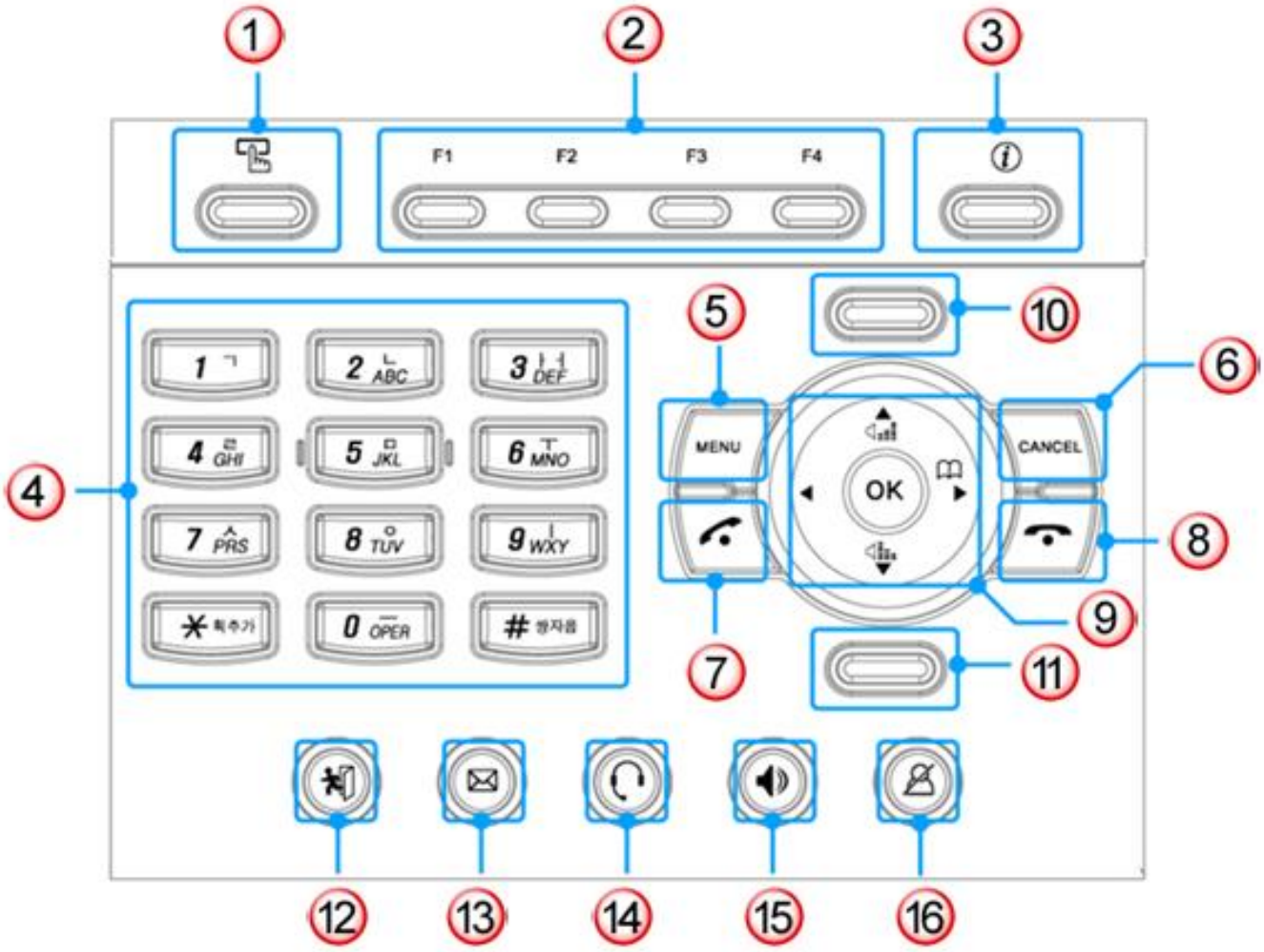
[Table 1-2] AP-IP230 Software Feature

Category	Specification
LAN Protocol	Static and IEEE 802.1Q VLAN Routing, RIP v1/v2, OSPF v2
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 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 IP230, Gatekeeper Support
	Enhanced QoS Management Features for Voice Traffics
IP-PBX Inter-working	SSCP AddPac Proprietary Protocol
IP-PBX Signaling Protocol	SIP Signaling Protocol between AddPac IP-PBX and IP Phone
Voice Mail	Voice Mail with IVR, Voice Mail Notification
Number & Call Routing	Basic Call, Music on Hold, Blind Transfer, Call Pickup, Consult Call, Switching Call, Consult Transfer, Call Waiting, Call Waiting Notify, Call Park, Call Pickup Remote, Hunt Group, Call Swapping, individual Call Park, Group Call Park, Call Forwarding , Unconditional, Busy, No Answer, Voice Mail, Etc.
Messenger Inter-working	MS Window based Smart Messenger Program
Conference	AddPac IP-PBX Audio MCU or External MCU Support
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

AP-IP230 IP Phone Front Configuration

It is a description of AP-IP230 IP Phone front panel. AP-IP230 IP Phone external is made of high quality ABS compact external network product. It is installed with major navigation keys to operate.



(Picture 1-2) AP-IP230 Front

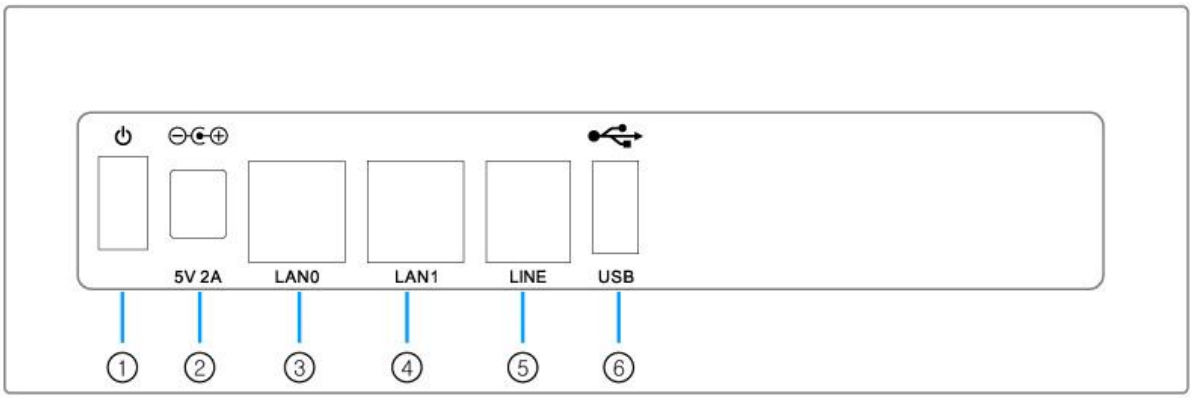
A below table is a description of AP-IP230 IP Phone front LED.

[Table 1-3] AP-IP230 IP Phone Front LED Description

No.	Button Name	Feature
(1)	Speed Dial	Brings out the Speed-Dial Menu
(2)	F1~F4	The soft keys which are displayed on the bottom of LCD screen and can be assigned with each different function such as Phonebook and Speed-Dial.
(3)	INFO	Displays the information on the top or bottom bar
(4)	Numeric Key	Used for Dialing and parameter setting in UI
(5)	Menu	Enter the UI Main Menu
(6)	Cancel	Move on to upper menu from current UI menu or cancel the current VoIP call
(7)	Call	Brings out the list of the recent calls Press to make a call after dialing
(8)	END	Ends the present call in progress
(9)	방향키, OK	Moves the direction in each UI menu
(10)		Not used
(11)		Not used
(12)	Absence	Used in Absence Mode
(13)	Voice Mail	Used in Voice Mail Mode
(14)	HDP Call	This KEY is used for VoIP call via Headphone Interface
(15)	SPK Call	The key is used for VoIP call via speaker phone. If this button is pressed, blue LAMP is turn on.
(16)	Privacy	Used for MUTE at conversation

AP-IP230 IP Phone Rear Configuration

AP-IP230 IP Phone real is composed of two 10/100Mbps fast Ethernet interface for WAN/LAN connection, FXO port for PSTN, and USB port.



(Picture 1-3) AP-IP230 Rear

[Table 1-4] explains the real panel of AP-IP230 IP Phone

[Table 1-4] AP-IP230 IP Phone Rear Interface Description

No.	Display	Description
(1)	SW	External Power ON/OFF switch.
(2)	DC 5V 3A	External Power Adaptor connector (DC 5V 2A)
(3)	LAN 0	10/100Mbps Fast Ethernet Interface for WAN such as ADSL, Leased Line, etc (RJ45)
(4)	LAN 1	10/100Mbps Fast Ethernet Interface for LAN (RJ45)
(5)	LINE	1-Port FXO PSTN Backup Interface
(6)	USB	This USB conforms to Standard 1.1. The maximum rate is 12Mbps, and the user is connected to the USB memory.

Chapter 2. AP-IP230 Preparation

Installation Requirement

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

- Ensure AP-IP230 IP Phone is in a dust-free environment before and after installation.
- Ensure AP-IP230 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 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-IP90 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.

Site Requirements

The AP-IP90 is ready for use where electronic products are used. However, a location with the following conditions is recommended for the 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

Requirements for Network Connection

This procedure is to follow EIA standards and other EMI regulations when you install the Gateway
The following section describes the Ethernet Cable, and the Console Cable which can be connected to the AP-IP90.

Required Tools and Equipment

Some cables and equipment are not included and your need to purchase them separately. Please prepare the following tools and equipment

LAN Cable & Console Port Cable

- RJ-45 to RJ-45
- RS232C console cable with RJ-45 connector (included in the box

Ethernet Port

The AP-IP90 has two RJ45 type of 10/100 BaseTX Ethernet ports on the rear side and LED for indicating the status of the port on the front. These ports are physically connected and using the direct cable you can connect to LAN0/ LAN1. Please use the standard cable and connector to access to LAN. You may refer to the cable details of Appendix in this guide

.

Console Port (Optional)

AP-IP90 has one RJ-45 type3 of RS-232C Female DCE Connector Interface. Through this port, you can perform the initial setup, monitoring and debugging the system. The cable and connector must be used. You may refer to the pin connection for RS-232 console cable of appendix in this guide.

Remove Packaging and Check Contents

Completely unpack all of the contents from the box and inspect each item for damage and ensure that you have all of the components listed below:

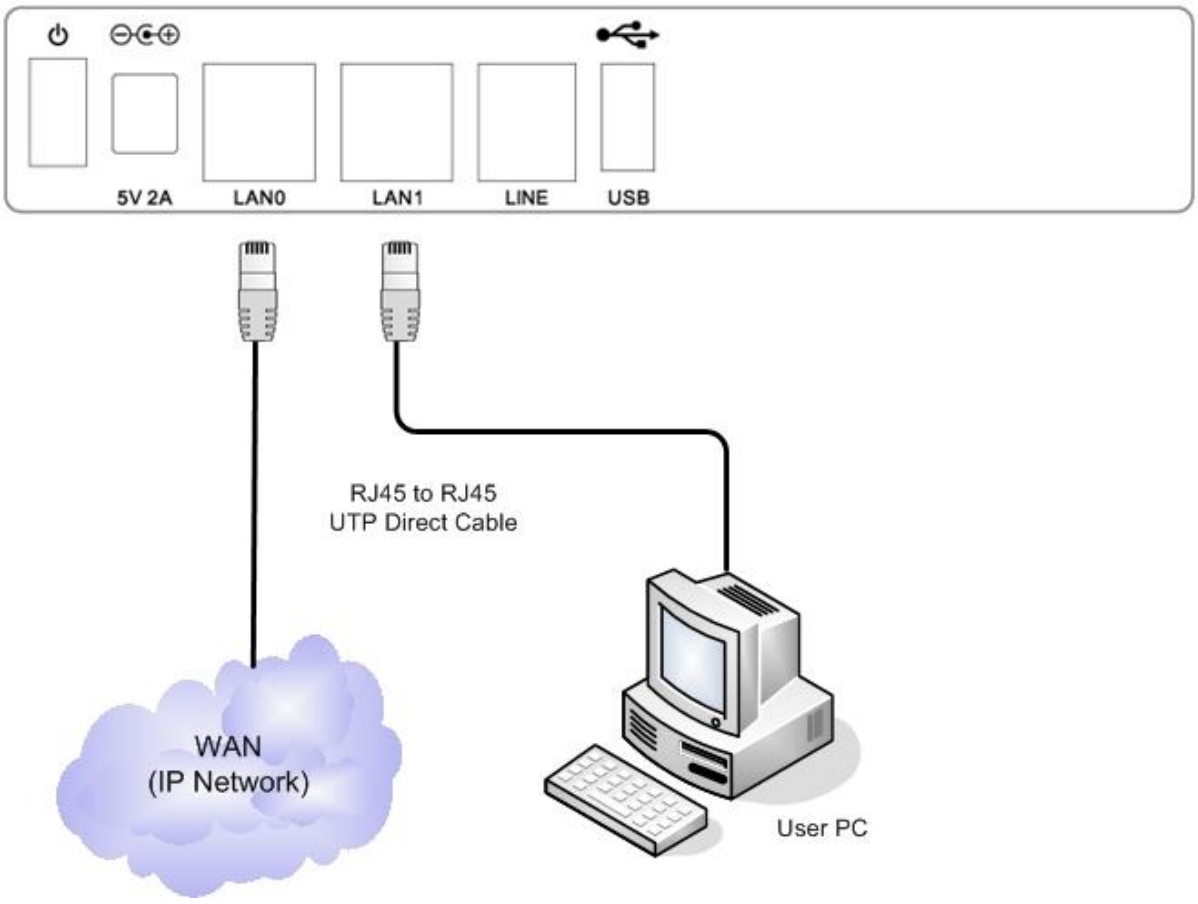
[Table 2-1] AP-IP230 IP Phone Contents

No	Name	Contents	Qt.
1	AP-IP230 IP Phone		1
2	LAN Cable (RJ45 to RJ45 Specification)		1
3	External Adapter (220V Power Code)		1

Chapter 3. AP-IP230 Installation

Ethernet Interface Connection

- Connect AP-IP230'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.



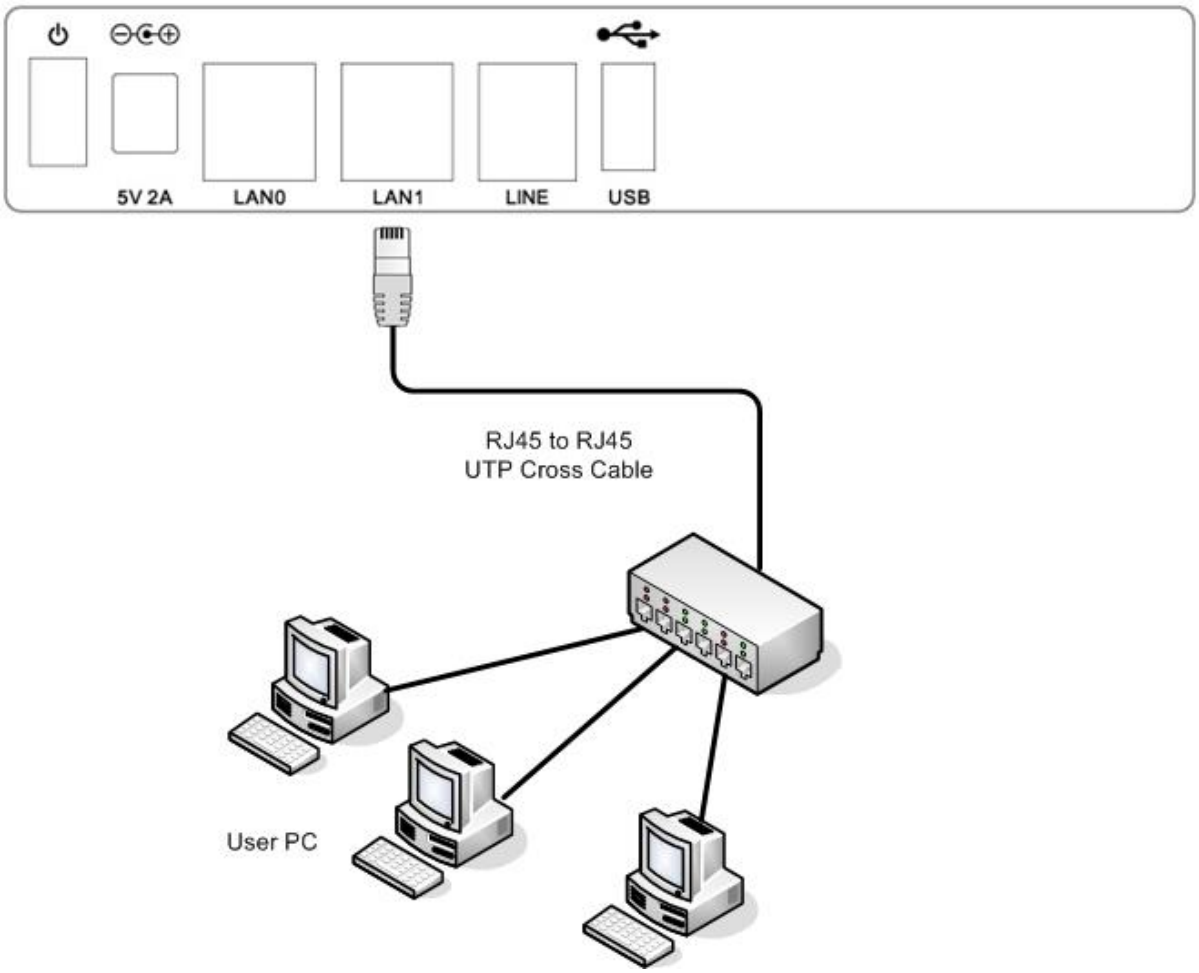
(Picture 3-1) AP-IP230 IP Phone WAN Interface Connection

• AP-IP230 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.

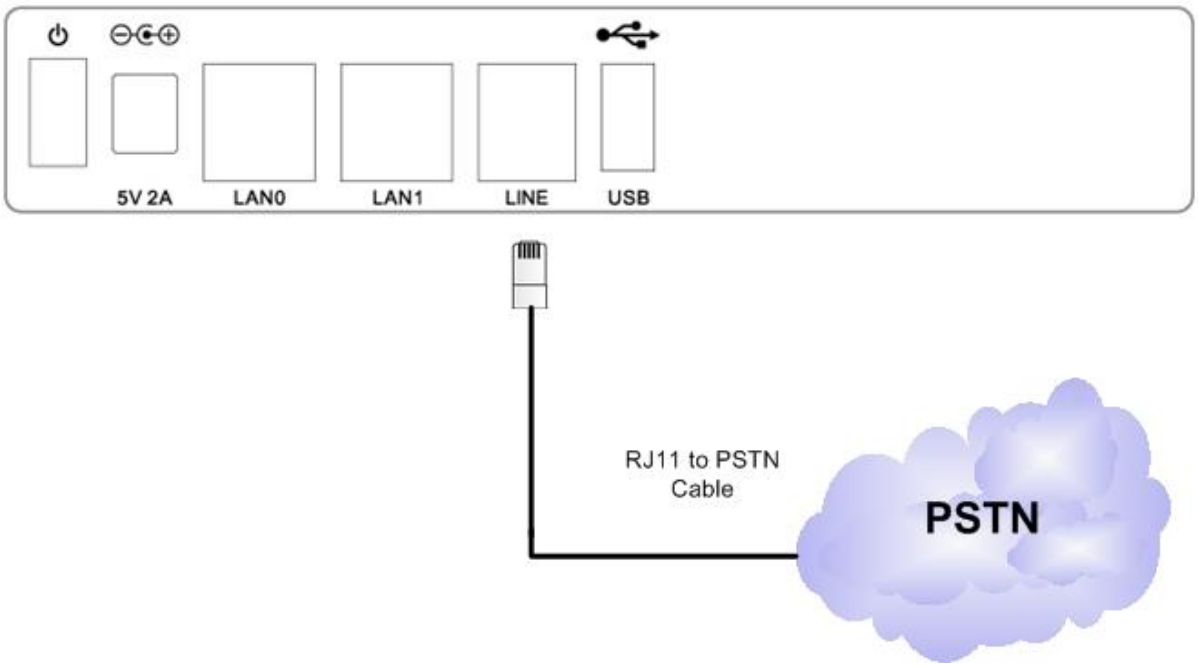
(Figure



(Picture 3-2) AP-IP230 IP Phone LAN Interface Connection

PSTN (FXO) Interface Connection

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.

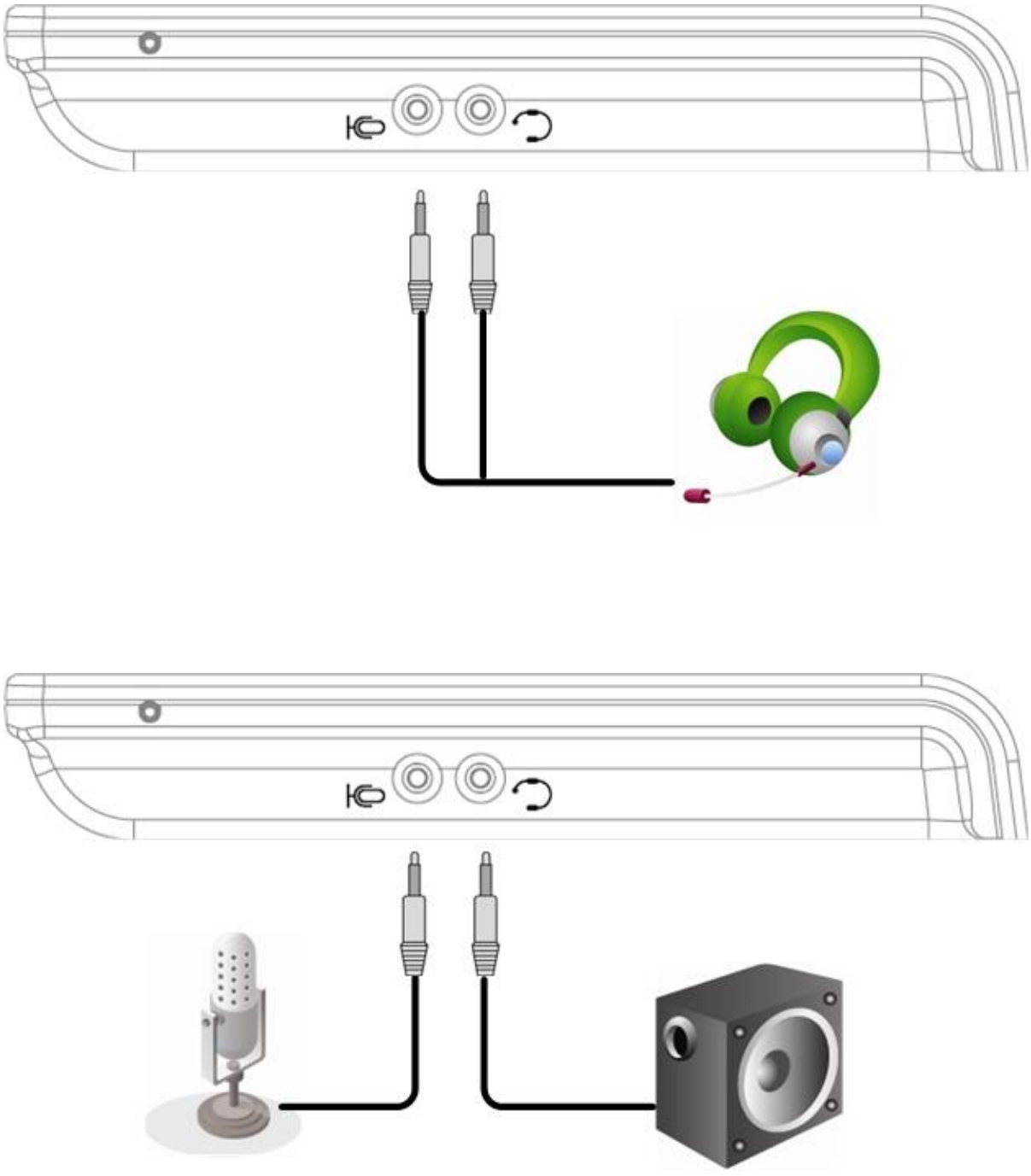


(Picture 3-3) AP-IP230 IP Phone PSTN Interface Connection

Audio In/Out Interface Connection

Audio-In/Out port located at left side of AP-IP300 IP Phone is for audio devices such as MIC, Speaker System or Headset Device etc.

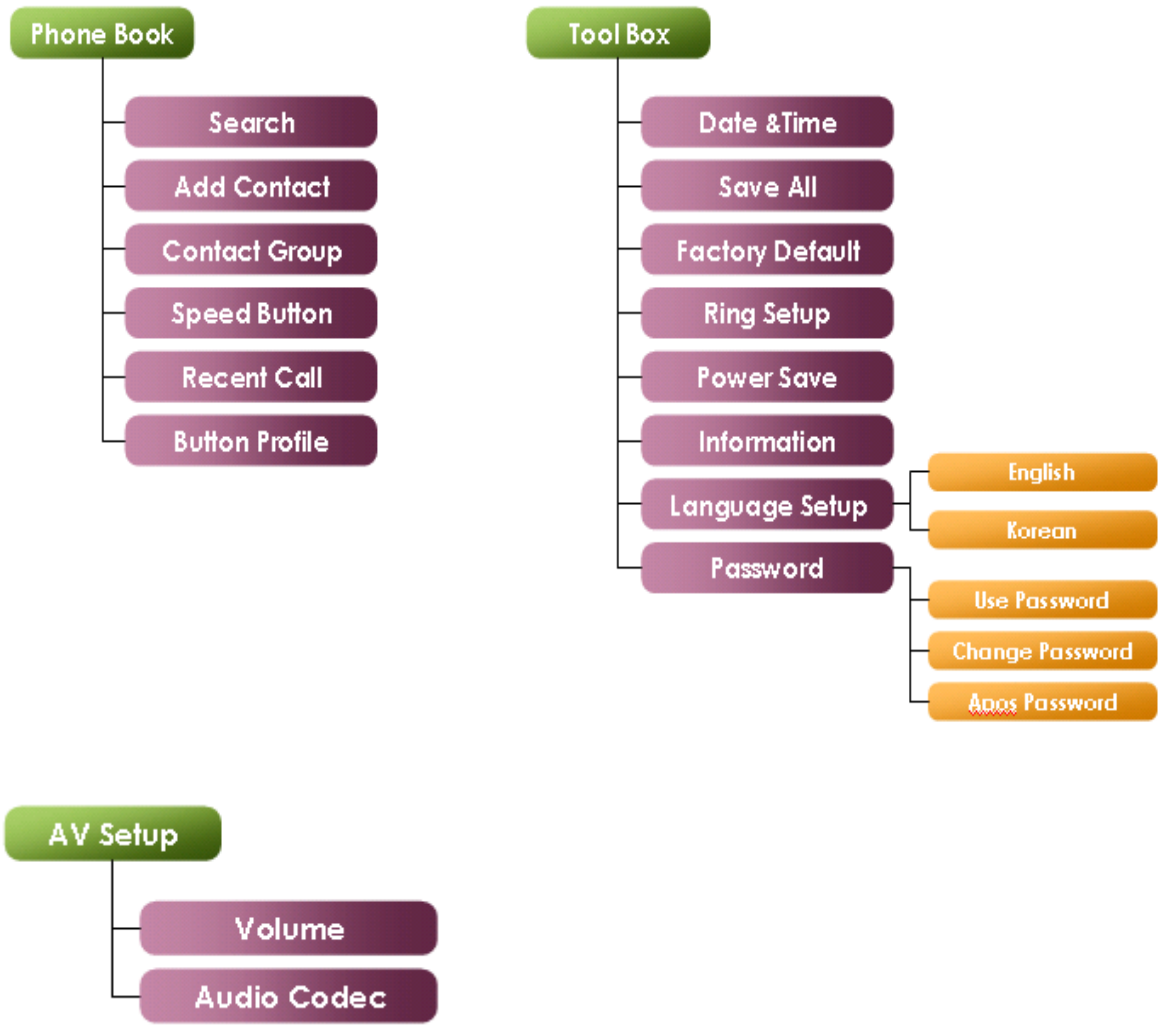
Connect this port to MIC system or External Speaker System using '3.5mm stereo jack' cable.



(Picture 3-4) AP-IP230 IP Phone Audio In/Out Interface Connection

AP-IP230 IP Phone User Interface Menu Configuration

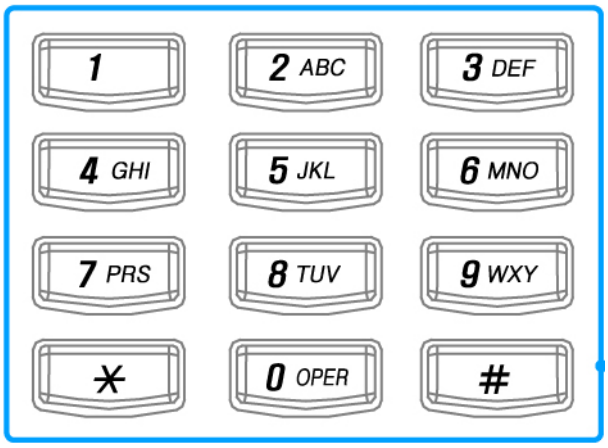
When you press the Menu key, you can see all the lists of options as they are shown in Figure 3-5. You can use the Menu key even while you are having a conversation on the phone.



(Picture 3-5) AP-IP230 Menu Configuration

Using Dial Pad

You can enter the characters by using the dial pad buttons in the Menu options:



(Picture 3-6) Key Pad Configuration Screen

[Table 3-1] The Characters Associated with Dial Pad Buttons

Dial Pad Button	Characters	Description
1	1 < > & ( )	The characters can be changed in the order as you press the same button consistently.
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	Back Space
F2	Space	Space

** F2(Space)⇒ If you are entering the different characters by pressing the same button consistently, you can use F2 button to enter the second character after entering the first one or you may wait 2 seconds to enter the second character after entering the first one.

Ex1) Enter ‘Apple’

Step1 Press 2 button five times then press 7 button twice

Step 2 Hit F2 key then press 7 button twice

Step 3 Press 5 button four times then press 3 button three times “

Ex2) Enter ‘2005/09/14’

⇒ Press the buttons in the order 2, 0, F2, 0, 5, # twice, 0, 9, # twice, 1, 4

Ex3) Enter ‘2aB’ ⇒ 2 , F2 , 2 twice, F2 , 2 six times

Using Send/End Button

The functions of Call button is described in Table 3-2 below:

[Table 3-2] Using Call Button on the Dial Pad

Configuration	Description
Retrieving the Recent Incoming Call	When you press the Call button and leaving the phone is on the hook, the recent incoming calls are listed. When you select one of the calls as to highlight, you can make a call by pressing the call button again.
Placing Call	When the phone is on the hook, you can make a call by just pressing the numeric buttons on the dial pad. Also the speed dial and recent call features of the Call button allows you to make a call very easily.
Receiving Call	After all the settings are applied, you can press the button and use it as to confirm

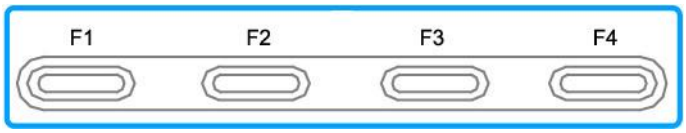
* You must press OK button to apply all the settings that you have done in the Menu options. If you want to keep the settings after restart, the settings must be saved to Tool Box-Save (reference to Tool Box Menu)

AP-IP230key pad End button is used for [Table 3-3] purpose.

[Table 3-3] Using Key Pad End Button

Category	Description
Hang Off	The END button works as to hang off the phone while you are in conversation
Drop Call	When you make a call by pressing the Call button, you can use the END button to drop the call.

Using SSCP Soft-Key



(Picture 3-7) Key Pad Function Key Configuration Screen

[Table 3-4] On-Hook while Pressing F1~F4

Step	Command	Description
1	Redi (Redial)	Dials the same number as the last time you made a call to that number again
2	Pick (Pickup)	Allows you to answer calls that come in on a directory number other than their own
3	PTT	Place a call to PTT(Push-To-Talk) Room
4	CCBS	When you make a call to the other party, he/she can be on a call already and the line is busy. This function enables the phone to call back automatically after he/she completes the call.

[Table 3-5] Off-Hook

Step	Command	Description
1	Redi (Redial)	Dials the same number as the last time you made a call to that number again
2	Pick (Pickup)	Allows you to answer calls that come in on a directory number other than their own
3	GPik (Group-Pickup)	Allows you to pick up incoming calls within their own group
4	EndC (End Call)	End call

[Table 3-6] Busy Status

Step	Command	Description
1	Hold	Places a call on hold
2	EndC (End Call)	End call
3	Tran(Transfer)	Transfers a call to the other extension
4	Park	Allows you to place an incoming call on hold by pressing Park button, then you can see the Park number on the LCD screen. You can move to the other desired place and then make a call by dialing the Park number to be connected. (Must be setup in SMM)
5	GPik (Group-Pickup)	WhenGoupPark send an announcement messages to all the phones in a group, anybody in the group can pickup the call to be connected (requires SMM configuration)
6	Conf (Conference)	Allows you to have a conference call (This is possible only when IP-PBX has the audio MCU module or the external MCU device is registered)
7	AddP (Add Party)	Allows you to add the conference party on by one as to invite (This is possible only when IP-PBX has the audio MCU module or4 the external MCU device is registered)
8	More	The 4 soft key can be displayed on a screen and press ‘More’ to see more softkeys.

[Table 3-7] Hold Status

Step	Command	Description
1	Resu (Resume)	Resume
2	NewC (New Call)	Connects to a new phone call
3	Tran(Transfer)	Call transfer

[Table 3-8] When the phone rings

Step	Command	Description
1	Answ (Answer)	Pickup incoming call

[Table 3-9] When the Phone Rings

Step	Command	Description
1	EndC (End Call)	End outgoing call

[Table 3-10] On Voice Main Screen

Step	Command	Description
1	EndC (End Call)	Disconnect voice mail

[Table 3-11] While Call is Being Transferred

Step	Command	Description
1	EndC (End Call)	Ends a new call which is currently on line, without call transfer and returns to the original held call
2	Tran(Transfer)	Connects a new call, which is currently on line, to the original held call.

[Table 3-12] Conferencing

Step	Command	Description
1	EndC (End Call)	Ends a call on line without establishment of conference and returns to the original held call for 1:1 communication
2	Join	Connects the third party

[Table 3-13] Conferencing Host

Step	Command	Description
1	AddP (Add Party)	Adds more parties to 3-party conferencing (depending on the capacity of MCU, the number of conferencing party is limited)
2	Info (Party Info)	Information of the present held conferencing participants
3	EndC (End Call)	Ends the conference in progress (Ends all the terminals in the conference)

* Conference Max participants :

IP-PBX (audio 4자), VP350MCU(video 4자), VC2000(video 4자), MC1000(video 16자)

[Table 3-14] Conference Participants

Step	Command	Description
1	Info (Party Info)	Information of the present held conferencing participants
2	EndC (End Call)	Exits from the conference in progress (Ends a call)

Basic CLI Command for Network Setup

* CLI Command to View Configuration Status

```
IP230# show run
Building configuration...

Current configuration:
!
hostname IP230
!
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 Setup

```
IP230# configure terminal
IP230(config)#
IP230(config)# interface FastEthernet 0/0    → Fast Ethernet Interface 0 Port
IP230(config-if)# ip address 172.20.103.1 255.255.0.0    → IP address configuration
IP230(config-if)# VOIP_INTERFACE_DOWN
VOIP_INTERFACE_DOWN
VOIP_INTERFACE_UP : (172.20.103.1)

IP230(config-if)# exit
IP230(config)#
IP230(config)# ip route 0.0.0.0 0.0.0.0 172.20.1.1    → Default route configuration
IP230(config)#
IP230(config)# end                                    → Go to general mode
IP230#
IP230# write                                          → Configuration save
Proceed with write? [confirm]y                        →
Building configuration...
[OK] Configuration saved to flash:/apos.cfg
IP230#
```

* After configuration, star Ping test to the end of default router.

```
IP230# 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
IP230#
```

If Ping test in properly operating, network configuration input is completed in order to call.

Chapter 4. Using AP-IP230

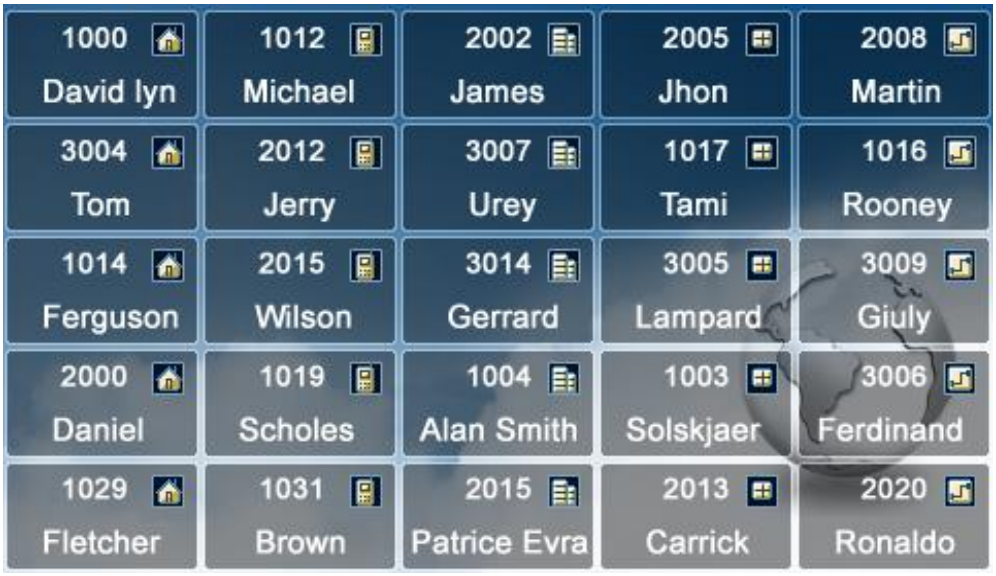
AP-IP230 IP Phone Initial Screen

Once the start-up operation is completed, the default screen is organized as it is shown in Figure 4-1.

(Picture 4-1) AP-IP230 IP Phone Basic Screen Configuration

[Table 4-1] AP-IP230 IP Phone Basic Screen Description

No.	Description	
①		Display the present date & time.When you are on a call, it displays the real "connection time" (SSCP takes the clock source from AddPac IPPBX and it automatically sets the time)
②	Name	Display the name of the device (System Setup -> User Information)
③	Number	Display the number on the default screen (System Setup -> User information)



(Picture 4-2) AP-IP230 IP Phone Basic Screen Layout

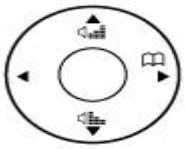
< Reference: The Speed Button Map can take the speed button information for the Presence Server after registering to the server or the user can create one's own speed button map. The screen can be changed by the Speed Button Key on the upper right side. >

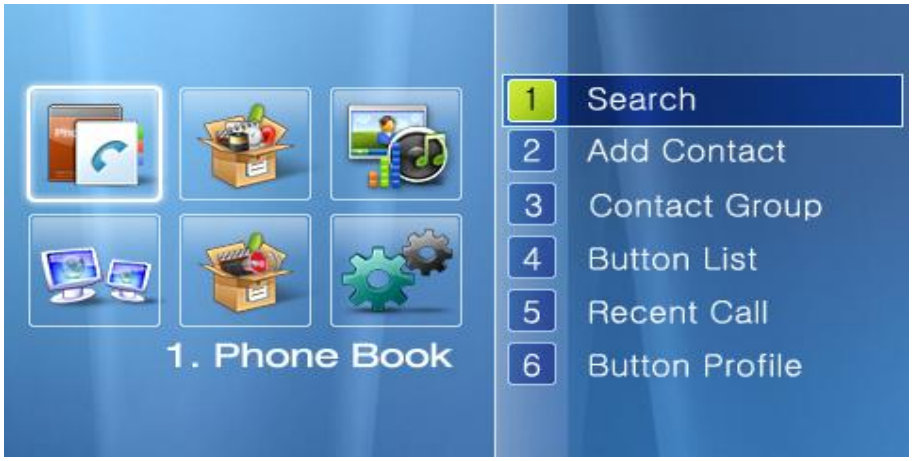
Phonebook Menu

The Phone Book is a directory in which user can search by name and number and has the functions including phone number registration, recent call history, group lookup, button list, the default setting. It also has call log and speed dial menu.

(

(Picture 4-3) Main Screen

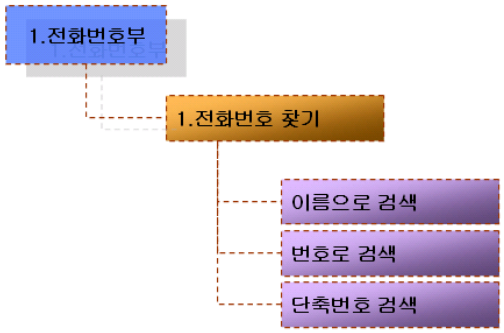




(Picture 4-4) Phonebook Menu Screen

Phonebook – Search Phone Number

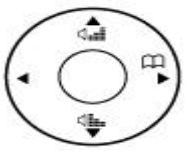
The Phonebook uses the registered name, phone number and speed dial number to search the phone number.



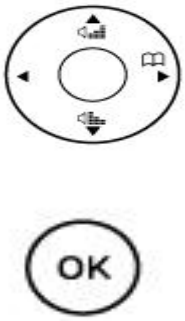
(Picture 4-5) Phonebook Search Menu Screen

[Table 4-2] AP-IP230 IP Phone Phonebook Search Screen Description

No.	Description
	The search by NAME looks for the registered name throughout the Phone book with the previously saved name. Therefore, cursor automatically moves to the right category in accordance with inputting the letter. If more than 2 same fields are found in the name including the letter for the search word, all the names with this filed are to be displayed. F1: Delete F2: Complete F4: Change the text
	Searches by the numbers which have been saved previously F1: Delete F2: Complete F4: Change the text
	Searches by the speed dial numbers which have been saved previously F1: Delete F2: Complete F4: Change the text



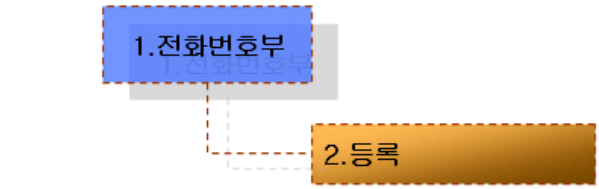
(Picture 4-6) When OK is pressed on the dial pad (it searches all the number which was saved previously)

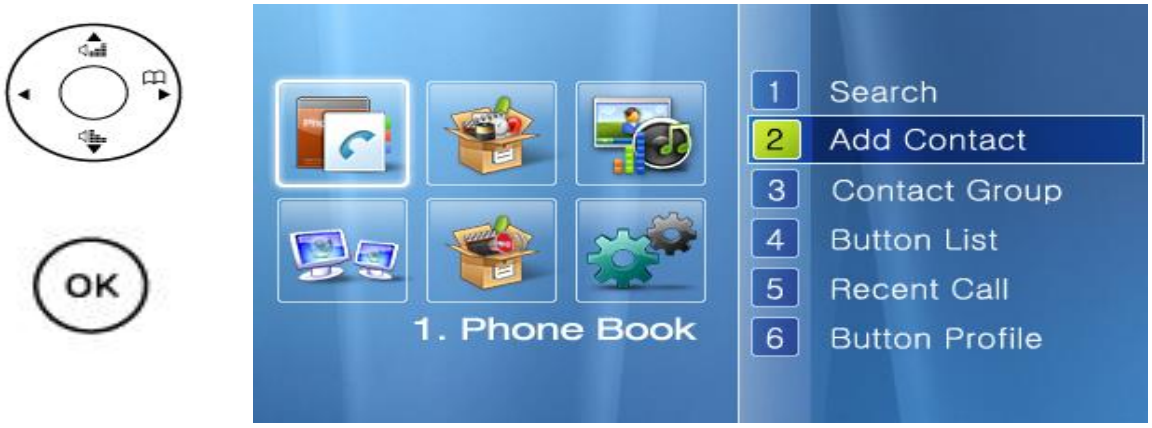


(Picture 4-7) When F1 (Modify) is Pressed

Phonebook – Registration

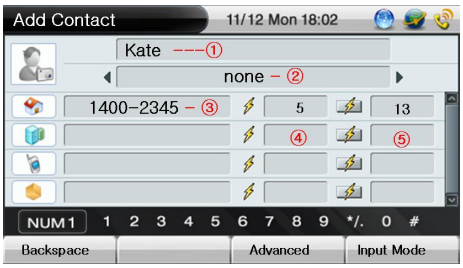
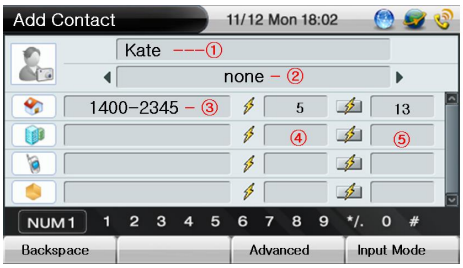
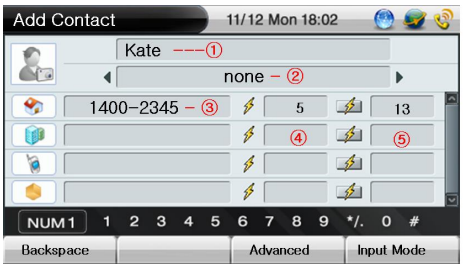
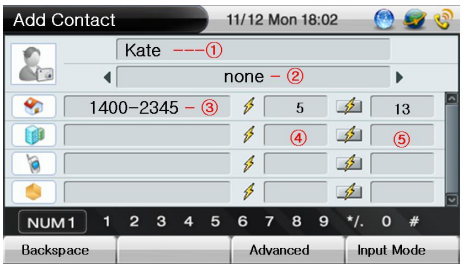
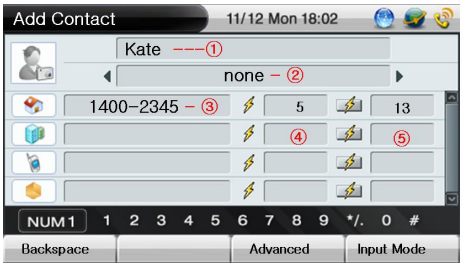
The registration menu takes a new phone number. The user can enter a name, telephone number, speed dial, speed button, IP address or codec information in the Phone Book. The entered phone numbers can be used for speed dial, search and speed button.



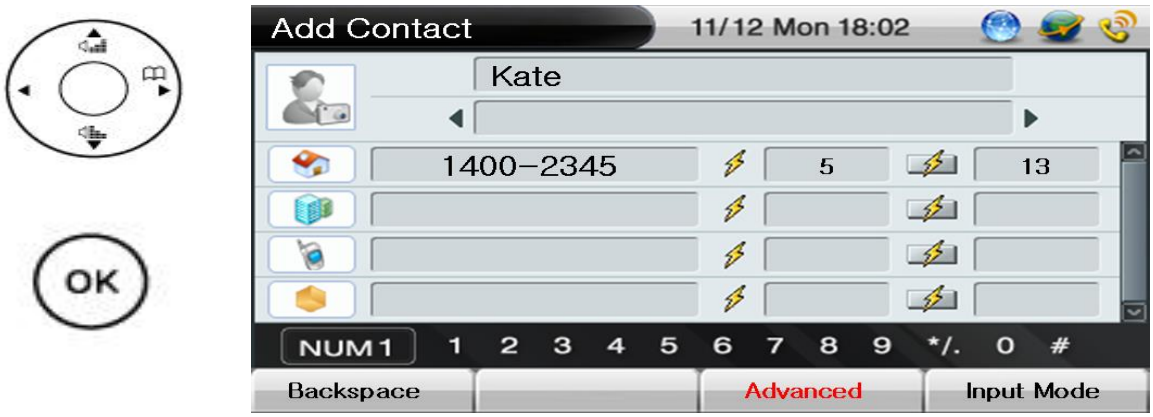


(Picture 4-8) Phonebook Registration Menu Screen

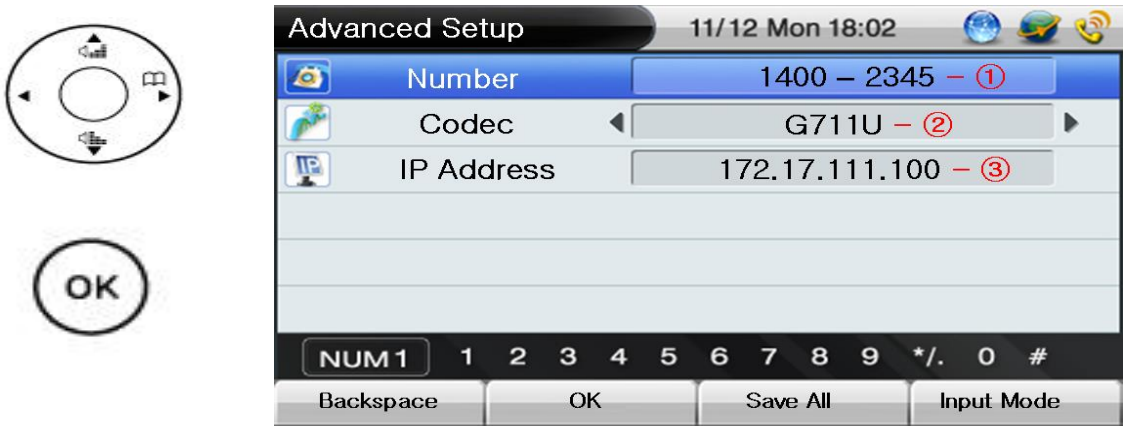
[Table 4-3] AP-IP230 IP Phone Phonebook Registration Screen Description

No.	Description
	1. Enter a new name in the Phonebook
	2. Select the group to which the number to registered
	3. Enter a phone number
	4. Enter a speed button number ranging from 1 to 25
	5. Enter a speed dial number in the phonebook ranging from 1 to 25

* If user wants to apply the registered value in the menu, the 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.



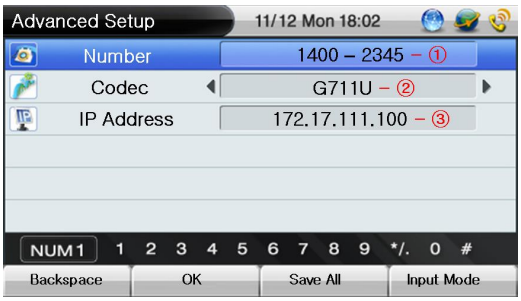
(Picture 4-9) Advanced Registration Setup Screen



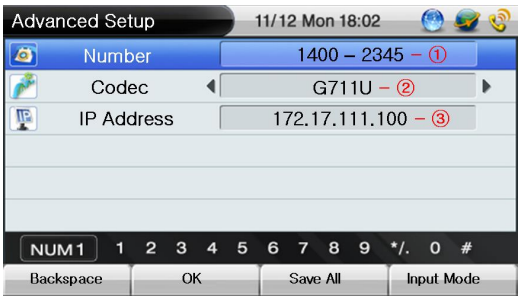
(Picture 4-10) Advanced Setup for Registration Screen

[Table 4-4] Description of Advanced Setup for AP-IP230 IP Phone Phonebook Registration

No.	Description
	1. Enter a phone number



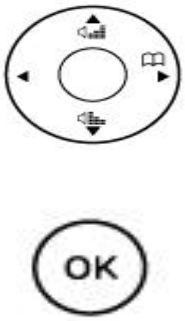
2. Select a type of audio codec.

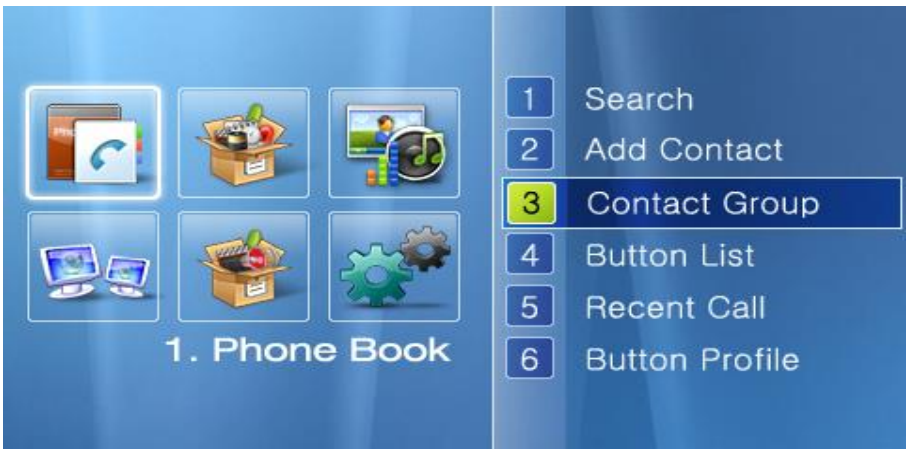


3. Enter an IP address

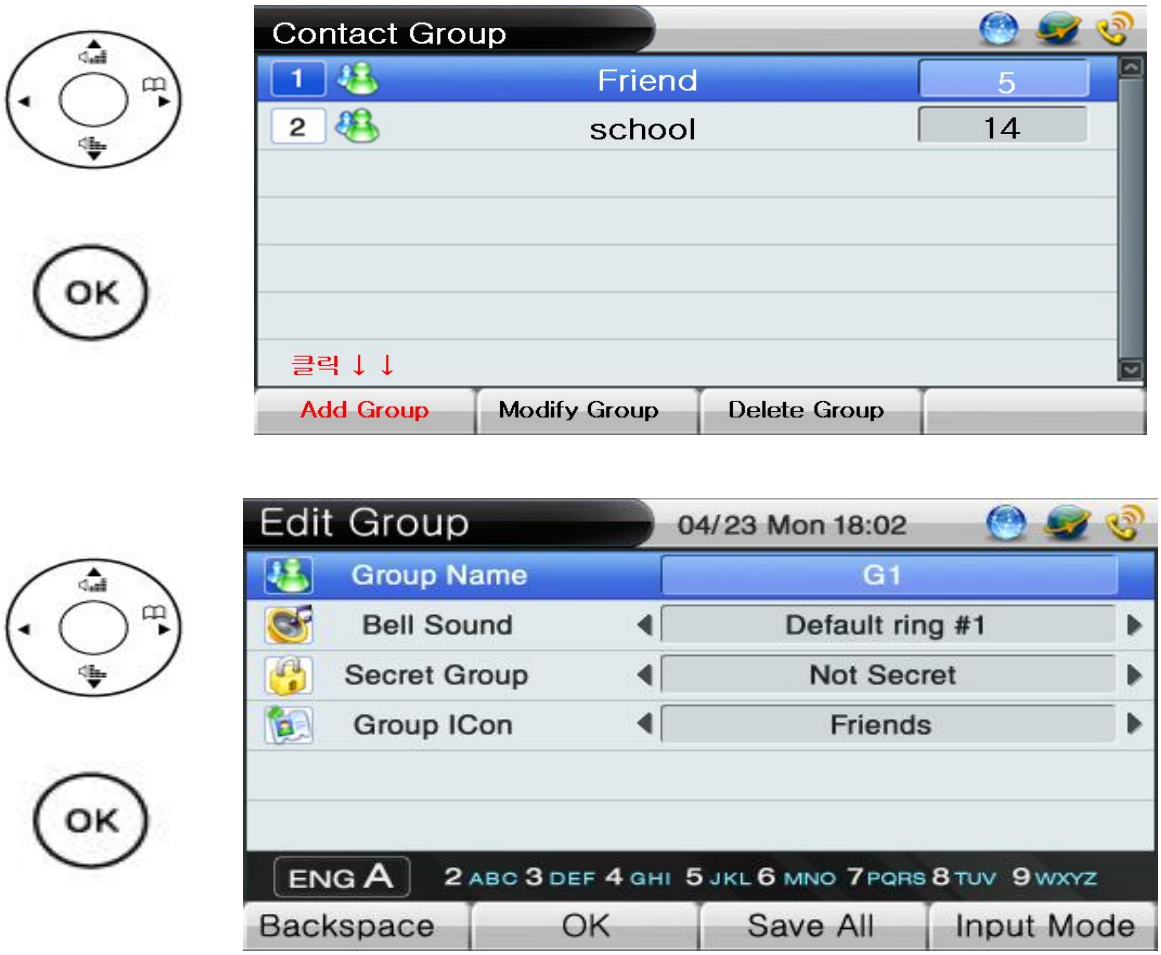
Phonebook – View Group

The user can specify a group during registration to the phonebook. The specified group can be set with an icon, a bell sound and secret group. The added group can be applied right after the phonebook registration.





(Picture 4-11) Phonebook Group View Screen



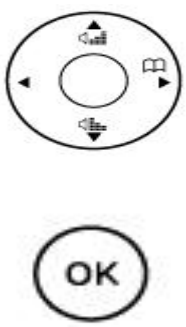
(Picture 4-12) Screen for Group Add, Group Edit

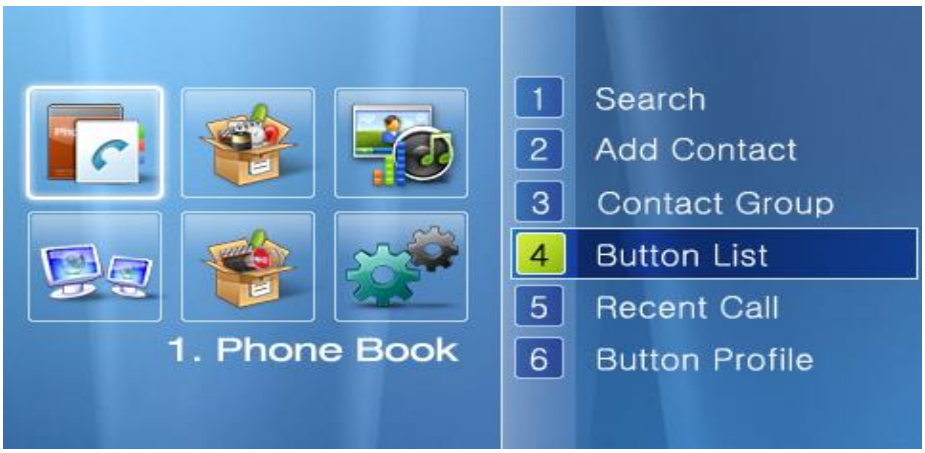
[Table 4-5] AP-IP230 IP Phone Group Edit

No.	Description
Group Name	Enter a group name
Group Bell	The bell sound for the specified group
Secret	Locks the group so others cannot see
Group Icon	Setup an icon for the specifies group

Phonebook – Button List

The Button list is laid out with the names of speed buttons and phone numbers. By using the saved list, the outgoing call can be carried out by Send button, OK button of the keypad and 25 speed buttons. Also editing and deleting of the speed button is possible on the Button List screen.





(Picture 4-13) Phonebook Button List Menu Screen

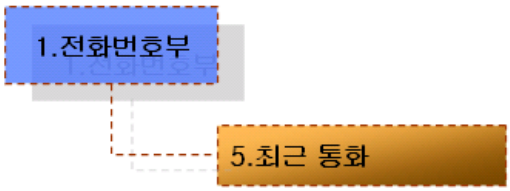
(Picture 4-14) Phonebook Button List Edit Screen

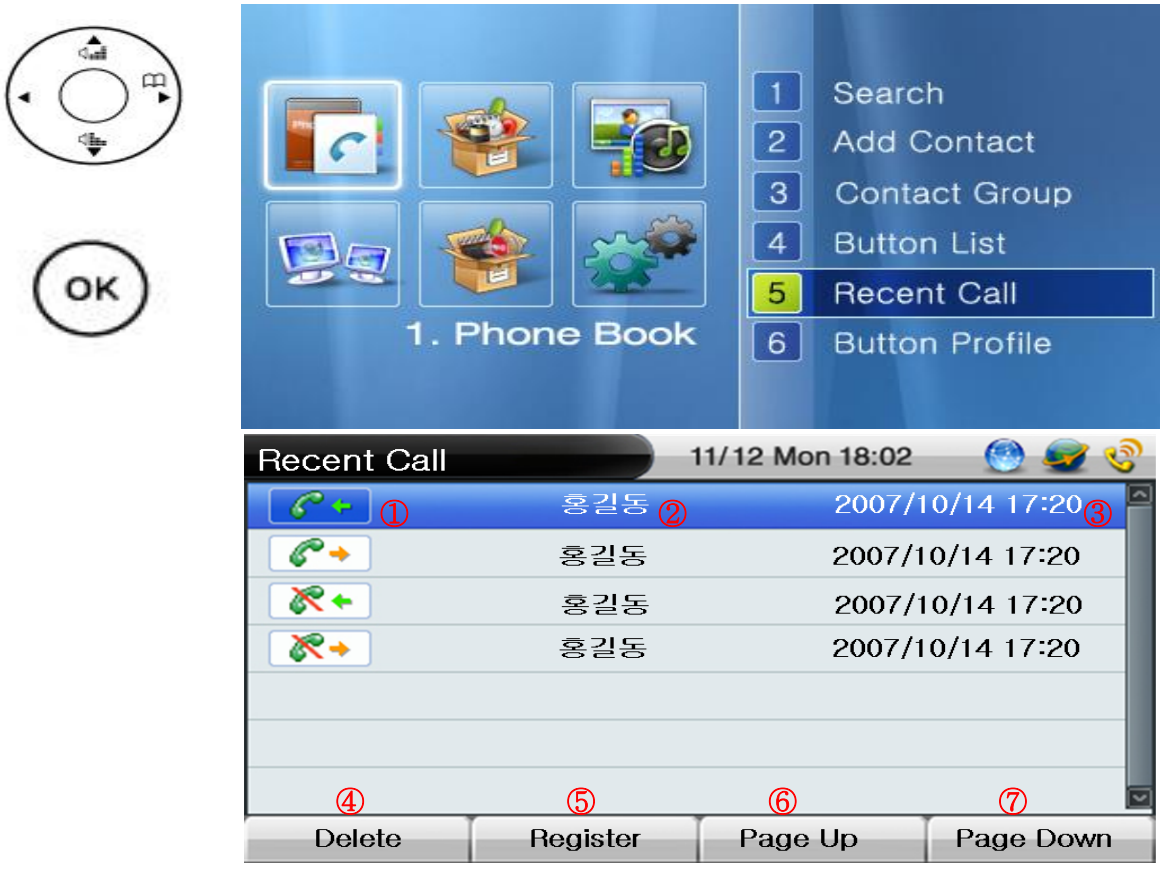
[Table 4-6] AP-IP230 IP Phone Phonebook Button List Edit Screen Description

No.	Description
Name	Setup a name for the Speed Button
Number	Setup a number for the Speed Button
Type	Setup a type of icon for the

Phonebook – Recent Call

The recent call displays a call log of the user for incoming and outgoing calls. This feature enables the user to check any incoming call which has been arrived during one's absence, calls back by using the number of the incoming call and save the number of the incoming calls.





(Picture 4-15) Phonebook Recent Call Menu Screen

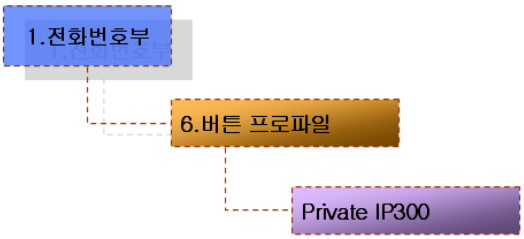
[Table 4-7] AP-IP230 IP Phone Phonebook Recent Call Screen Description

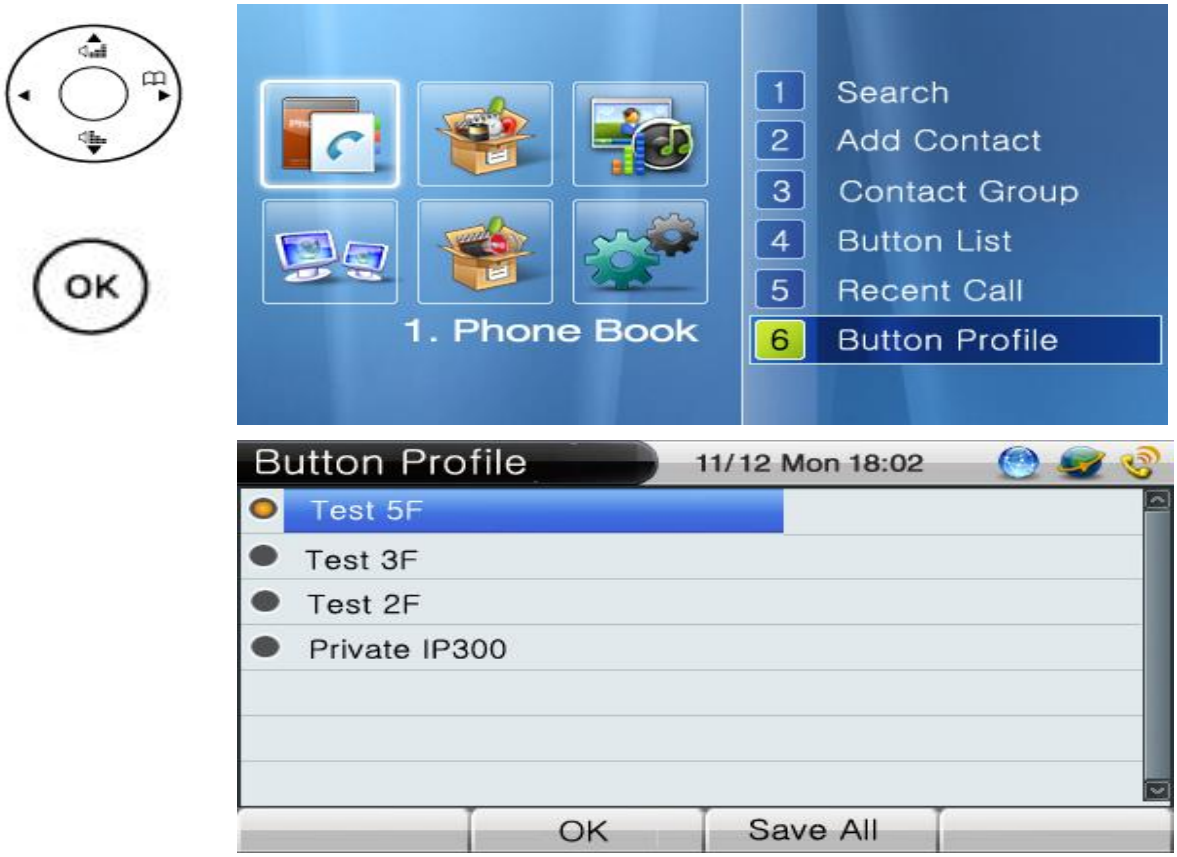
No.	Description	
① Incoming/Outgoing		Displays an incoming call
		Displays that the incoming call has not been answered

		Displays an outgoing call
		Displays that the outgoing call has not been answered
② Remote Information		Display the call number for placing a call to the other party directly. This call information is displayed by H.323 protocol (H.323 ID) and SIP (URL)
③ Call Duration		The time that takes for placing or receiving a call
④ Delete		Delete a recent call history
⑤ Register		Save the session
⑥ Page backward		Move to the next page
⑦ Page Forward		Move to the previous page

Phonebook – Button Profile

The Button Profile can interoperate with the Presence Server only. The user can choose the Button Profile form the Speed Button list which has been provided from the server. In order to interoperate with the Presence Server, you need to setup 4. Network and Call Setup – 10. Presence first.





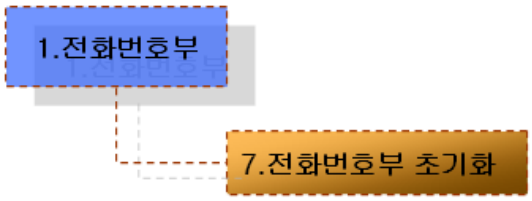
(Picture 4-16) Phonebook Button Profile Menu Screen

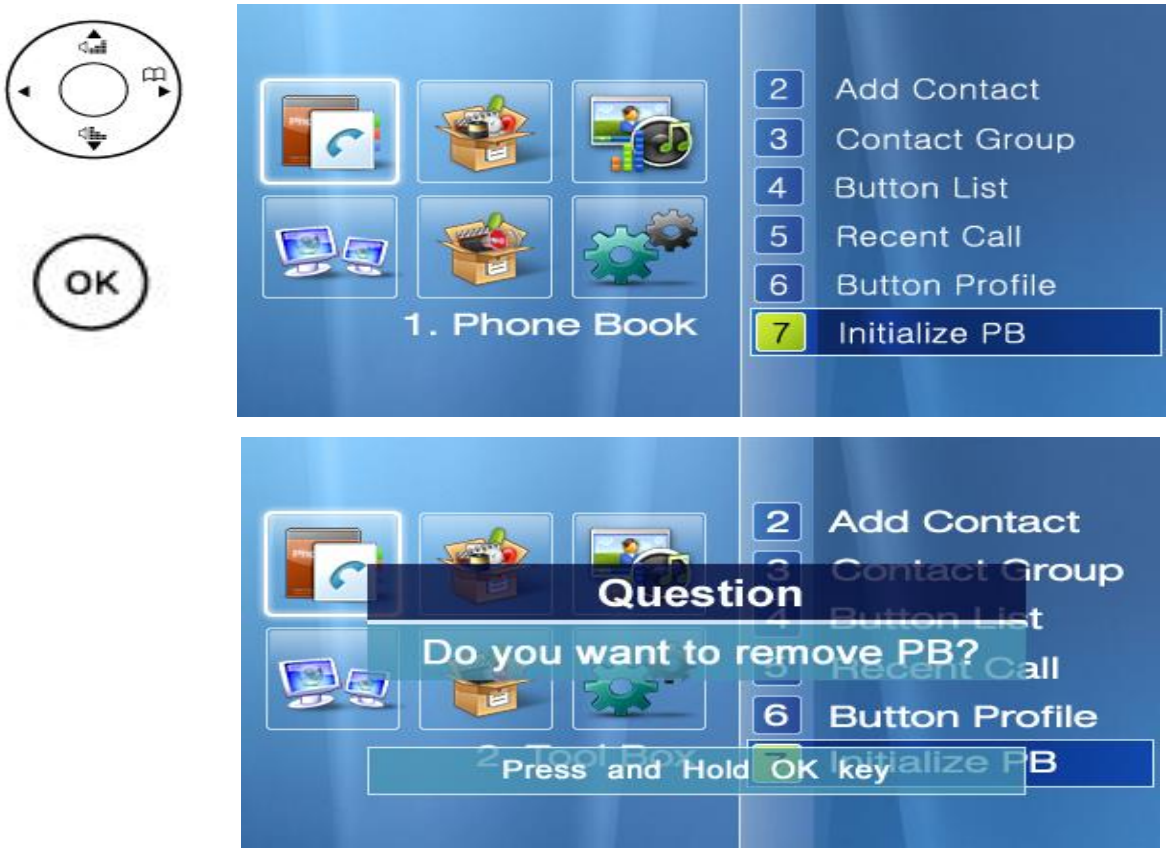
[Table 4-8] AP-IP230 IP Phone Phonebook Button Profile Screen Description

No.	Description
Private IP230	Assign a phone number to the Speed Button directly. This profile is displayed when AP-IP230 is not registered to the Presence Server
Test 5F	Receive the profile form the Presence Server

Phonebook – Phonebook Initialization

The default mode initialization feature deletes all the configured settings of AP-IP300 and all the content of Phonebook, Speed Button numbers and recent cal. This command reboots the system automatically.



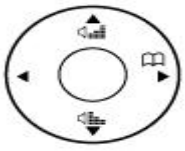


(Picture 4-17) Phonebook Initialization Menu Screen

Tool Box Menu

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

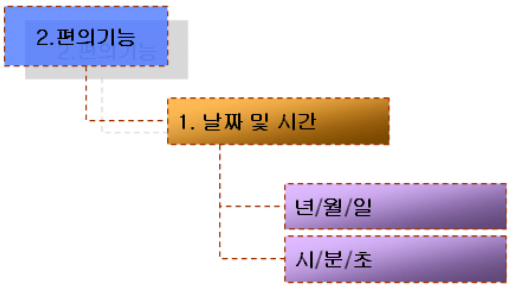
(Picture 4-18) Main Screen

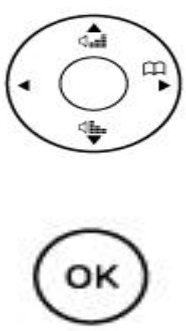


(Picture 4-19) Tool Box Menu Screen

Tool Box ─ Date & Time

The user can setup the date and time. Press F3 to save





(Picture 4-20) Date/Time Enter, Change Menu Screen

[Table 4-9] AP-IP230 IP Phone Date/Time Screen Description

No.	Description
Year	Enter present system year.
Month	Enter the present month
Date	Enter the present date
Hour	Enter the present hour
Minute	Enter the present minute
Second	Enter the present second

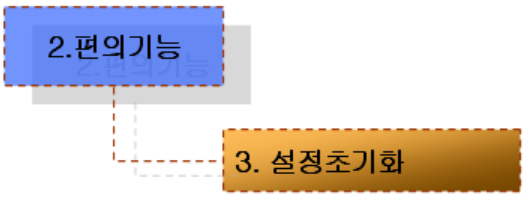
Tool Box – Save All

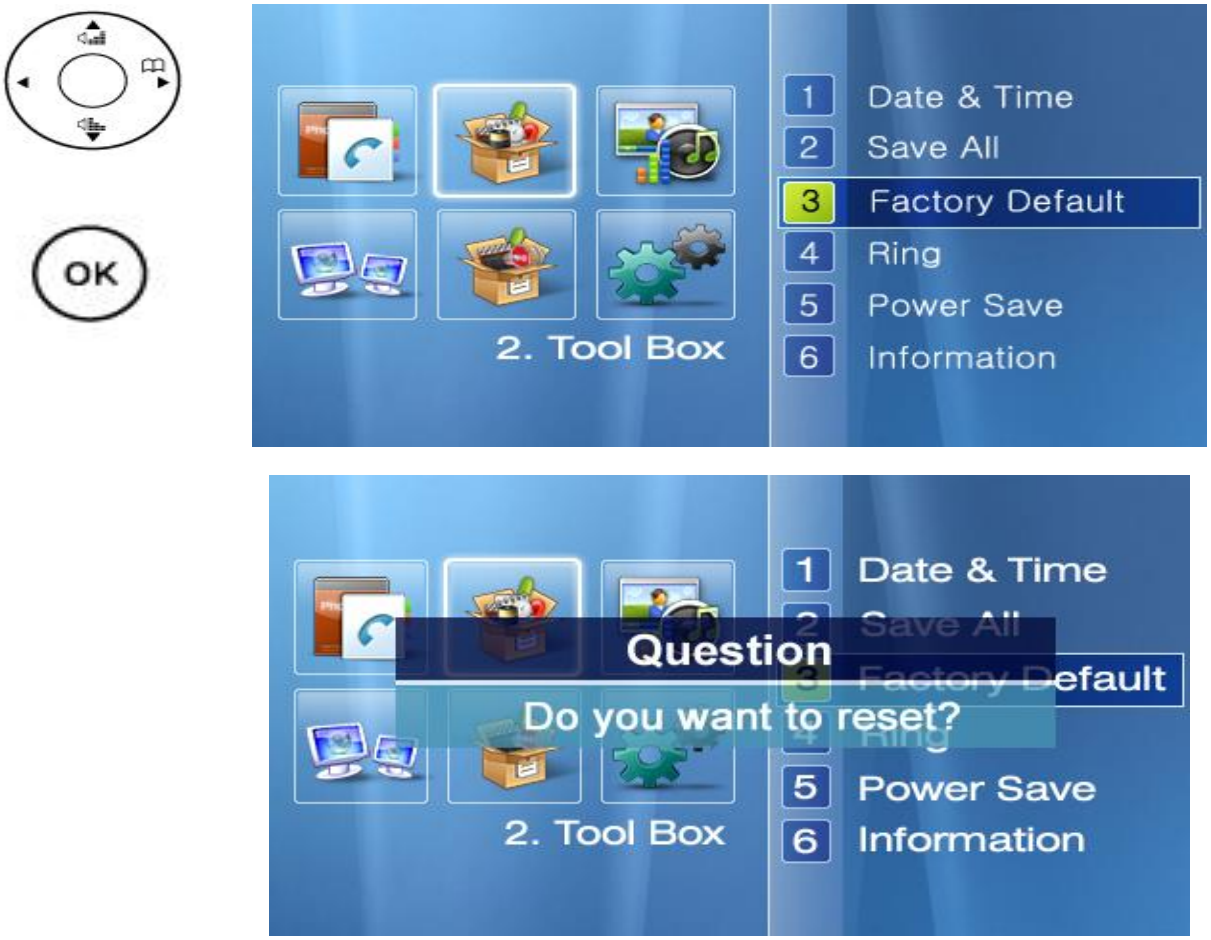
This UI menu saves the settings which the user has entered in UI. Once the settings are saved, values are preserved even after rebooting.

(Picture 4-21) Save Menu Screen

Tool Box – Factory Default

The Factory Default deletes all the configured settings of AP-IP230 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 for some inevitable circumstances.

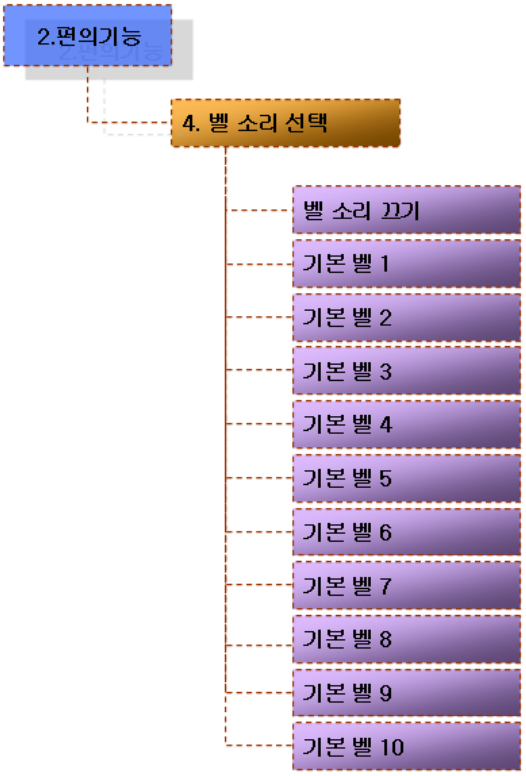


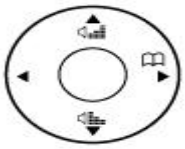


(Picture 4-22) Factory Default Menu Screen

Tool Box- Select Ring Sound

You can set the ringer up to 10 different kinds of sound including mute on the integrated speaker, in the ringer settings. The user can choose the sound that one likes after hearing the 9 different kinds of sound, except the mute, by using F1(Play). Also the volume can be adjusted.





(Picture 4-23) Bell Sound Configuration Menu Screen

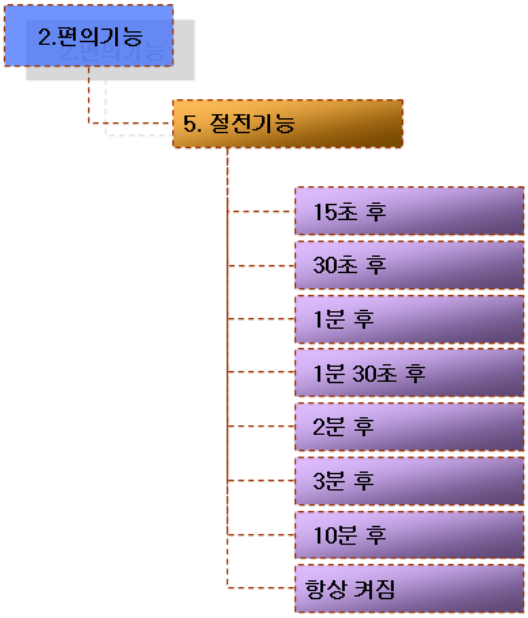
[Table 4-10] AP-IP230 IP Phone Bell Sound Screen Description

No.	Description
Ring sound off	Set to mute
Default Ring #1	The ordinary digital phone sound
Default Ring #2	The ordinary analog phone sound
Default Ring #3	The ordinary door bell sound
Default Ring #4	The ordinary bicycle bell sound
Default Ring #5	The harp bell sound
Default Ring #6	The chirp bell sound
Default Ring #7	The electronic bell sound

Play(F1) => Preview the ringer sound

Tool Box- Power Save

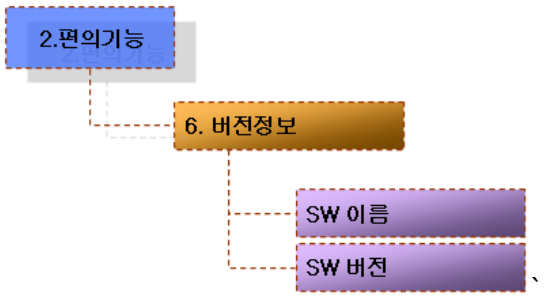
The Power Save turns off the LCD automatically in a specified time. The LCD can be turned on again by pressing any button on the key pad from the state of Power Save. This setting is recommended for preserving the life time of LCD and maintaining its quality.

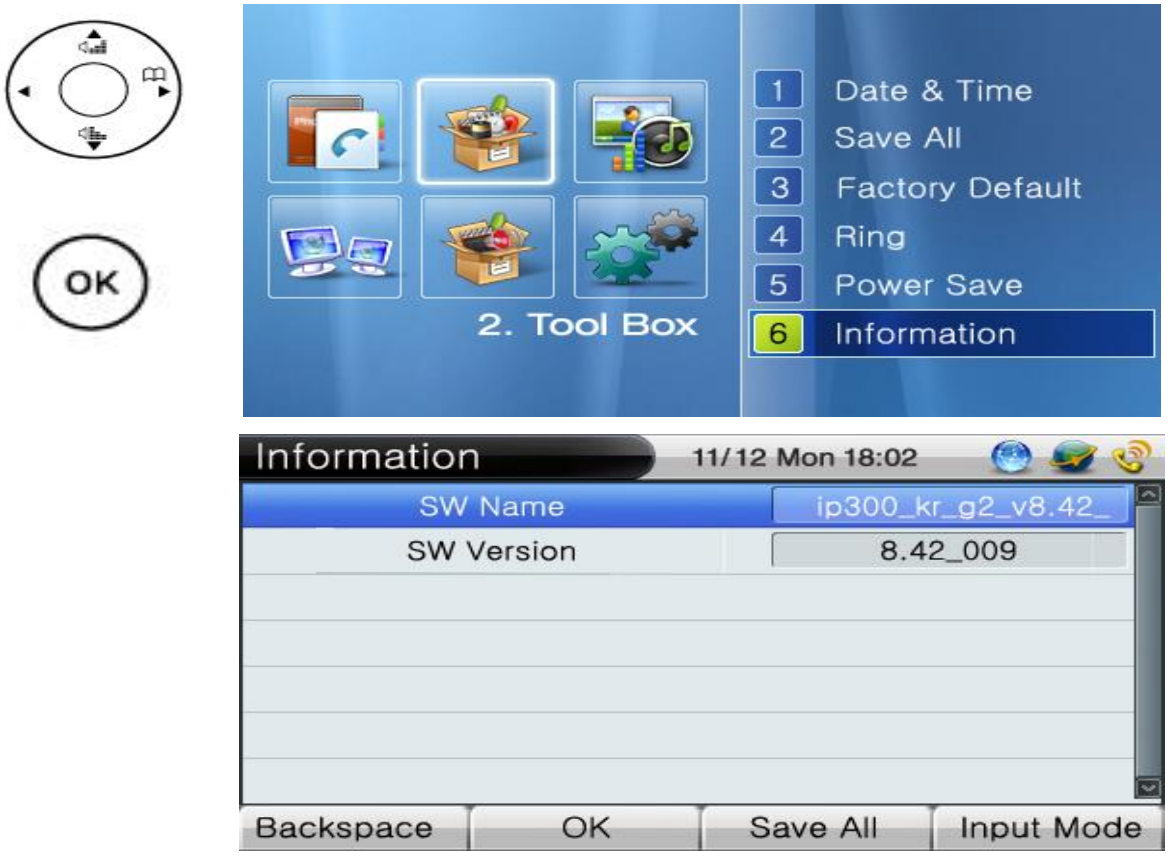


(Picture 4-24) Power Save Menu Screen

Tool Box- Version Information

This option allows you to verify the version of the software running at this present time.





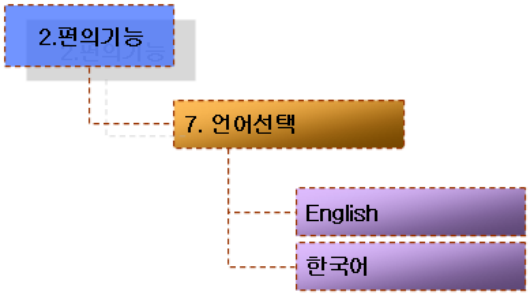
(Picture 4-25) Version Information Menu Screen

[Table 4-11] AP-IP230 IP Phone Version Information Screen Description

No.	Description
SW Name	This is the name of the firmware running at this present time.
SW Version	This is the version running at this present time

Tool Box – Select Language

This option allows you to verify the language being used at this present time.



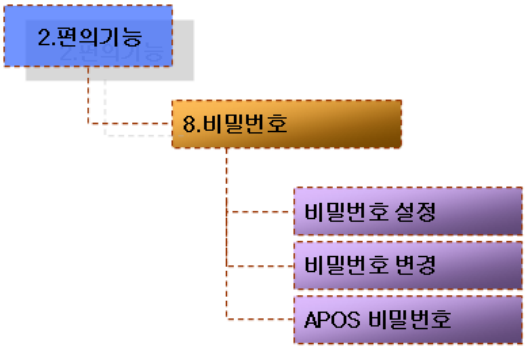
(Picture 4-26) Language Configuration Menu Screen

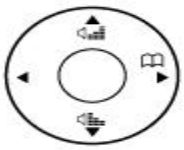
[Table 4-12] AP-IP230 IP Phone Language Configuration Screen Description

No.	Description
English	English setup mode.
Korean	Korean setup mode (G2 version-RUSSIAN)

Tool Box- Password

The Password blocks the access to a particular menu and it can be changed. The default password is 2337. Changing APOS password is not recommended as possible.





(Picture 4-27) Password Configuration Menu Screen

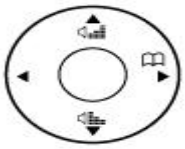
[Table 4-13] AP-IP230 IP Phone Password Configuration Screen Description

No.	Description
Password Setup	Set/ Cancel the password (default : cancel) The password can be set to [Factory Default] [Internet Setup] [VoIP Setup] [Message] [Personal Information Setup] [Speed Dial Number Setup] menus.
Change Password	Change the password
APOS Password	Change APOS password. It is not recommended to change this password as much as possible.

Voice/Video Configuration Menu

AV Setup consists of Volume and Audio Codec.

(Picture 4-28) Main Screen



(Picture 4-29) Network Setup Menu Screen

Voice/Video Configuration – Volume

The Volume menu consists of Bell Volume, Input/Output Volume Adjustment and External Microphone.

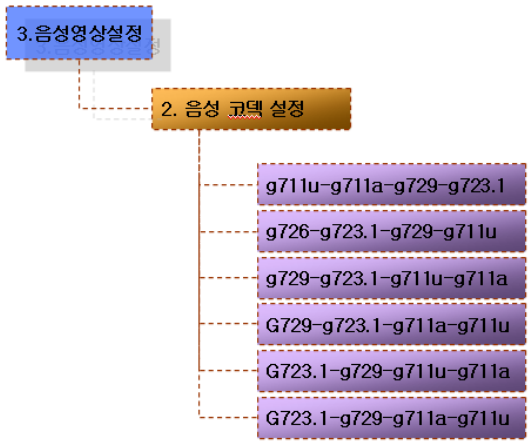
(Picture 4-30) Volume Setup Menu Screen

[Table 4-14] AP-IP230 IP Phone Volume Configuration Screen Description

No.	Description
Bell Sound Volume	Adjust the bell sound volume. The default is set to 5.
Input Volume	Adjust the input volume of the speaker phone and sender/receiver. The default is set to 5.
Output Volume	Adjust the output volume of the speaker phone and sender/receiver. The default is set to 5.
External Microphone Boost	Select Boost, when the audio input is set to MIC. The default is set to cancel

Voice/Video Configuration – Audio Codec

The Audio Codec determines a type of voice codec. You can choose the options of G.711[PCM] and G726, G.729, and G.723.1 on UI, basing on the priority level which can be suitable to your network settings.

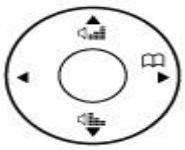


(Picture 4-31) Voice Codec Configuration Menu Screen

Network and Call Setup Menu

The Network & Call consists of WAN, LAN interface setting, SIP/H.323 signaling, FTP service, QoS, call options etc. The user should know this network setup menu for efficient usage of AP-IP230. This menu cannot be skipped for optimized environment..

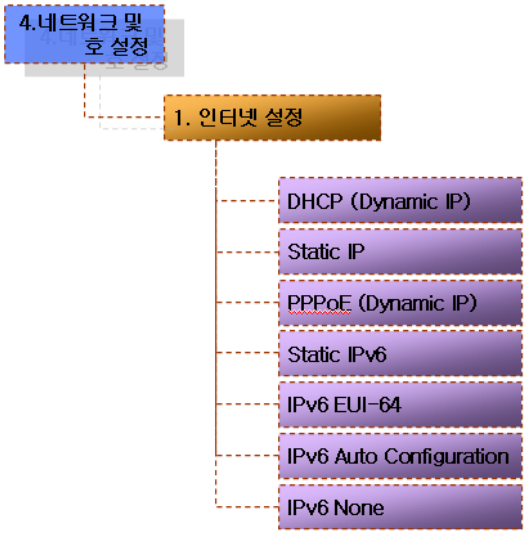
(Picture 4-32) Main Screen

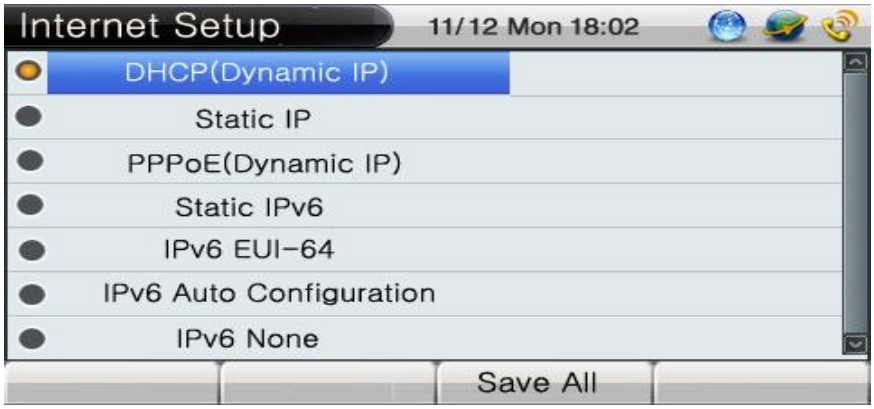


(Picture 4-33) Network Setup Menu Screen

Network and Call Setup – Internet Configuration

The Internet Menu has functions related to WAN interface for Internet connection. As there are various network environments, the user has to configure pursuant to his or her own network environment. The WAN protocols supported by AP-IP230 are DHCP, static IPv4, PPPoE, and IPv6 etc. The following figure shows the UI command tree structure in Network & Call Menu..





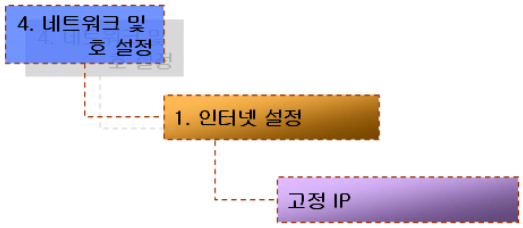
(Picture 4-34) Internet Configuration Menu Screen

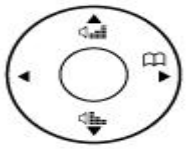
[Table 4-15] AP-IP230 IP Phone Internet Configuration Screen Description

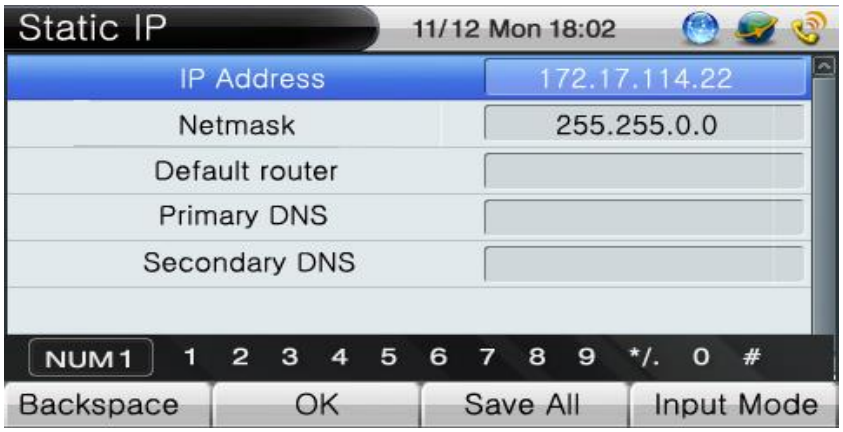
No.	Description
DHCP	Takes a dynamic IP address from DHCP server such as cable modem, VDSL, IP-ADSL.
Static IP	Configures IP address manually and build WAN interface such as static IP ADSL, E1/T1 leased line.
PPPoE	WAN protocol to receive dynamic IP address from PPP server.
Static IPv6	Configures IP address manually and build WAN interface such as static IP ADSL, E1/T1 leased line.
IPv6 EUI-64	Configured with company_id(24-bit) basing on the standard of IEEE Registration Authority and extension id(40-bit) basin on the same standard.
IPv6 Auto Configuration	Configured with WAN interface taking the dynamic IPv6 address from DHCP server
IPv6 None	The relevant settings can be cancelled while IPv6 is in use

Network and Call Setup – Internet Setup – Static IP

This menu configures WAN interface such as static IP ADSL, E1/T1 leased line.







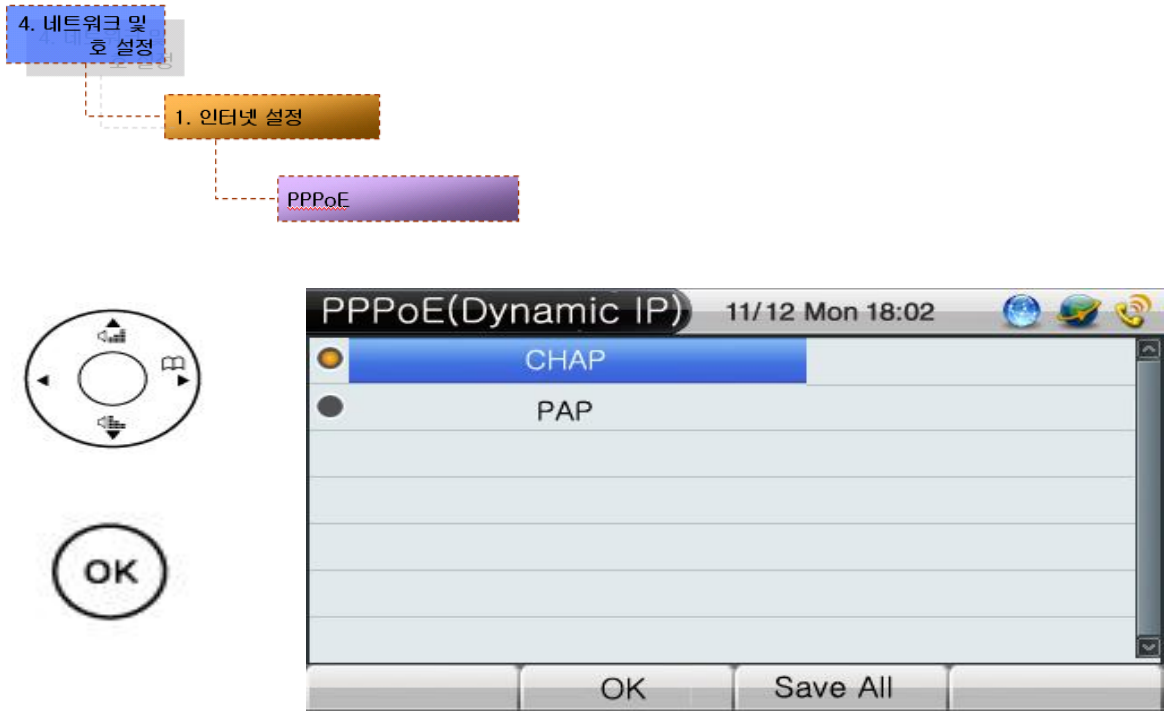
(Picture 4-35) Static IP Setup Menu Screen

[Table 4-16] AP-IP230 IP Phone Static IP Configuration Screen Description

No.	Description
IP Address	Enter IP address Ex> 172.20.1.100
Net Mask	Enter net mask Ex> 255.255.0.0
Primary Router	Enter primary router Ex> 172.20.1.1
Primary DNS	Enter first DNS (Apply to IPv6) Ex> 168.126.63.1
Secondary DNS	Enter secondary DNS (optional)

Network and Call Setup – Internet Setup – PPPoE

This is the WAN protocol which takes a dynamic IP address from the PPP Server. ADSL is one of the typical applications in which PPPoE is used.



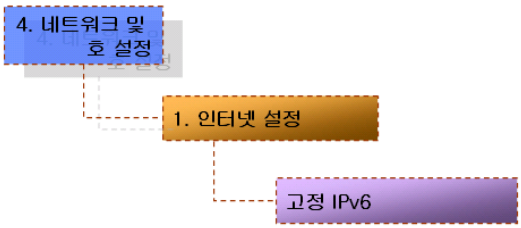
(Picture 4-36) PPPoE Configuration Menu Screen

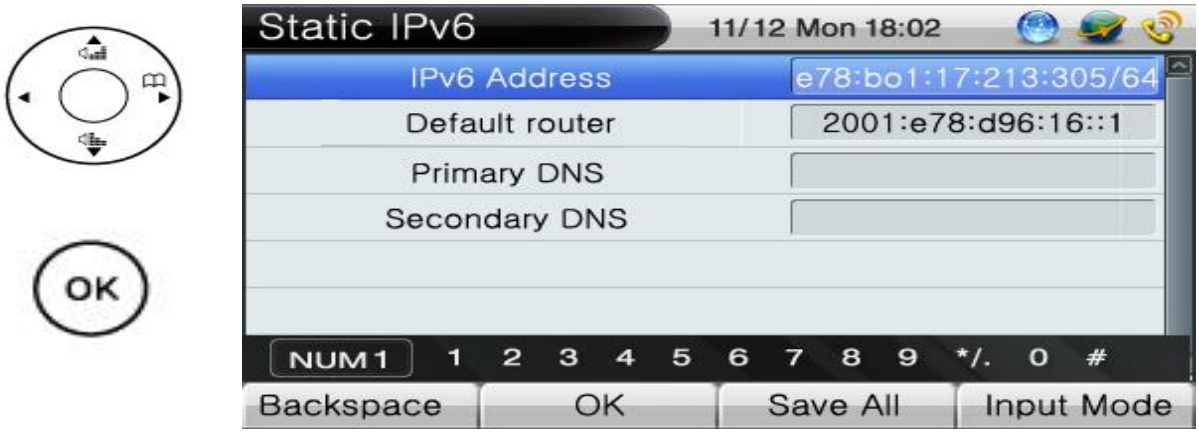
[Table 4-17] Description of AP-IP230 IP Phone PPPoE Configuration Screen

No.	Description
CHAP	Authentication Mode – CHAP
PAP	Authentication Mode – PAP

Network and Call Setup – Internet Configuration – Static IPv6

This menu configures WAN interface such as static IP ADSL, E1/T1 leased line.





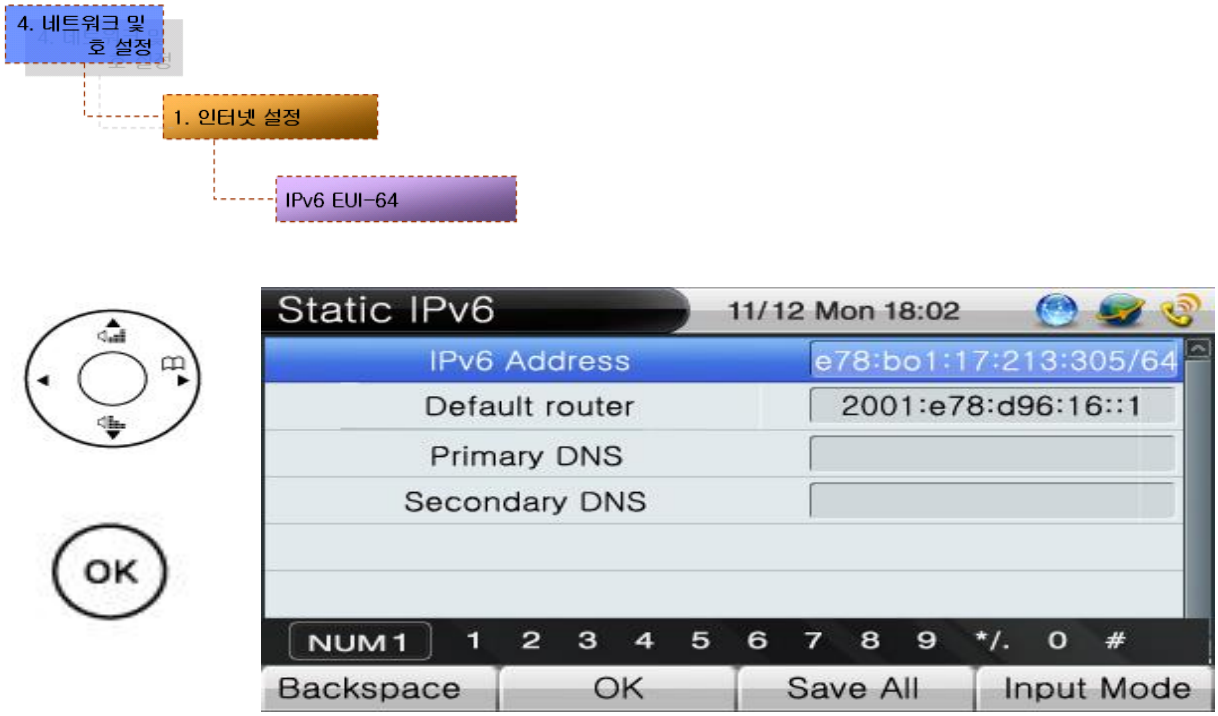
(Picture 4-37) IPv6 Configuration Menu Screen

[Table 4-18] AP-IP230 IP Phone IPv6 Configuration Screen Description

No.	Description
IPv6 Address	Enter IPv6 address Ex> 2001:e78:b01:17:114::10/64
Primary Router	Enter Primary IPv6 router address Ex> 2001:e78:b01:17:114::1
Primary DNS	Enter primary DNS Ex> 168.126.63.1
Secondary DNS	Enter secondary DNS (optional)

Network and Call Setup – Internet Configuration – IPv6 EUI-64

This EUI-64 IPv6 address scheme configures company_id(24-bit) basing on the standard of IEEE Registration Authority and extension id (40-bit) basin on the same standard.



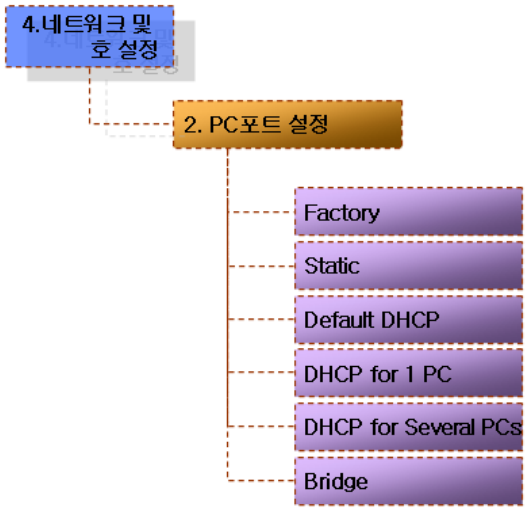
(Picture 4-38) IPv6 EUI-64 Configuration Menu Screen

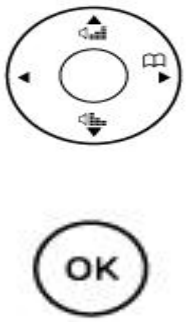
[Table 4-19] AP-IP230 IP Phone IPv6 EUI-64 Configuration Screen Description

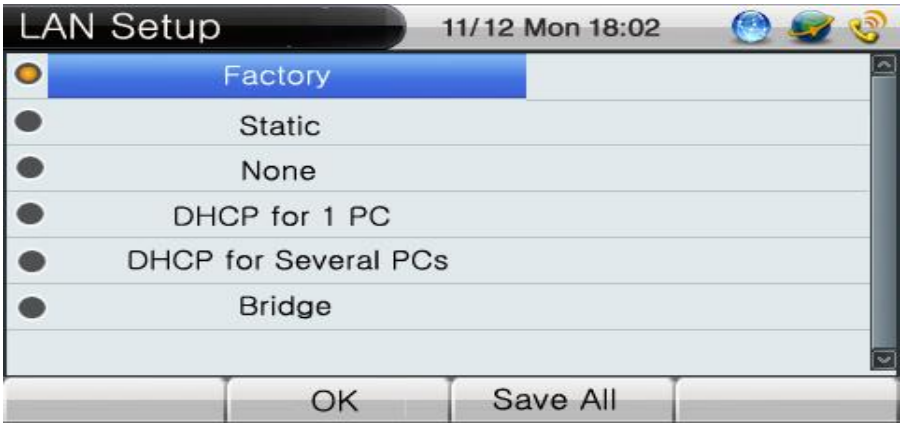
No.	Description
IPv6 Address	Enter IP address Ex> 2001:e78:b01:17:114::10/64
Primary Router	Enter primary gateway Ex> 2001:e78:b01:17:114::1
Primary DNS	Enter primary DNS Ex> 168.126.63.1
Secondary DNS	Enter secondary DNS (optional)

Network and Call Setup – PC Port Configuration

This LAN menu is used for protocol setting of AP-IP230 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-IP230'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.







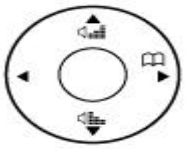
(Picture 4-39) PC Port Configuration Menu Screen

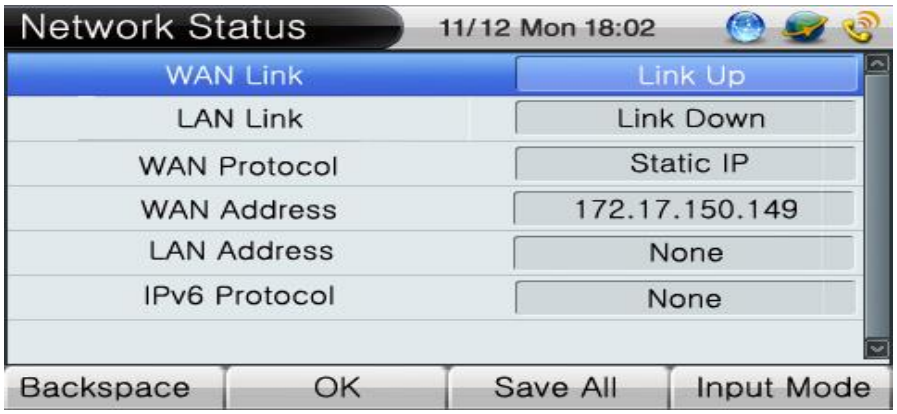
[Table 4-20] AP-IP230 IP Phone PC Port Configuration Screen Description

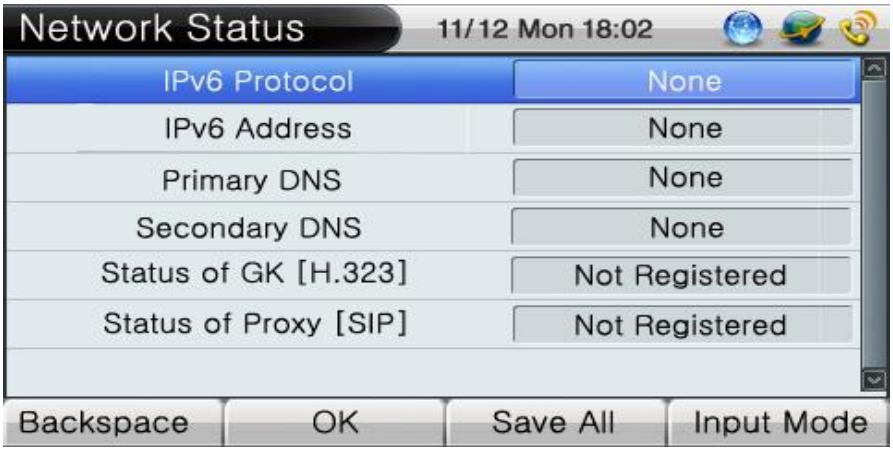
No.	Description
Factory	Set LAN factory default mode. (default : 192.168.10.1)
Static	Configure LAN (the user sets the configuration)
None	Disable LAN setup (Press ok button to select this option)
DHCP for 1 PC	In DHCP protocol mode for single PC, for sharing public same IP address of AP-IP300's WAN interface and LAN interface connected to PC, AddPac proprietary public IP-Share mechanism is used
DHCP for Several PCs	DHCP for multiple PC are similar to general IP sharer which links two (2) PCs or more.
Bridge	Configures LAN settings with the bridge mode

Network and Call Setup – Network Status

This menu displays the current network status of Link Status, IPv4 Protocol, IPv4 address, LAN address, IPv6 Protocol, IPv6 address, DNS, SIP Proxy Server, GK[H.323] Registration Status at a glance.







(Picture 4-40) Network Status Menu Screen

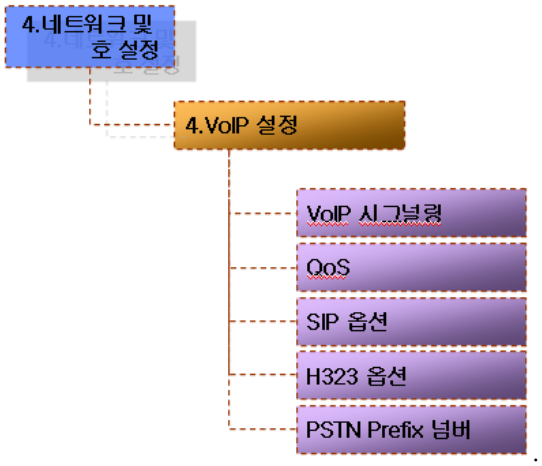
[Table 4-21] AP-IP230 IP Phone Network Status Configuration Screen Description

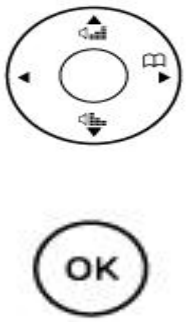
No.	Description
WAN Link	Display whether LAN0 interface link is up/down
LAN Link	Display whether LAN1(PC) interface link is up/down
WAN Protocol	Display WAN IPv4 protocol (DHCP, Static IPv4, PPPoE)
WAN Address	Display WAN IPv4 address
LAN Address	Display AN IPv4 address
IPv6 Protocol	Display WAN IPv6 protocol

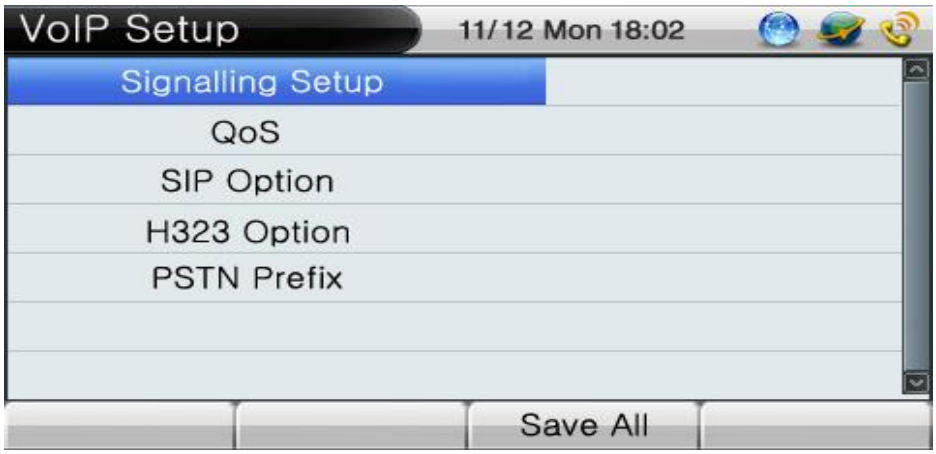
IPv6 Address	Display WAN IPv6 address
Primary DNS	Display primary Domain Name Server
Secondary DNS	Display secondary Domain Name Server
GK Register Status	Display gate keeper register status
SIP Proxy Register Status	Display SIP proxy server register status

Network and Call Setup – VoIP Configuration

This VoIP setup menu is used for interoperating with SIP server or Gatekeeper on H.323 and SIP basis and adjusting E.164, PSTN number and QoS.



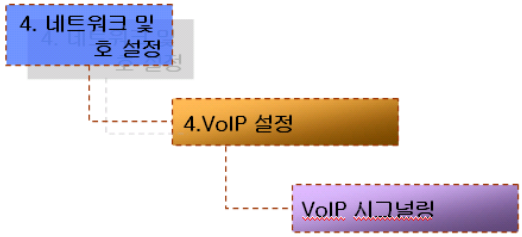


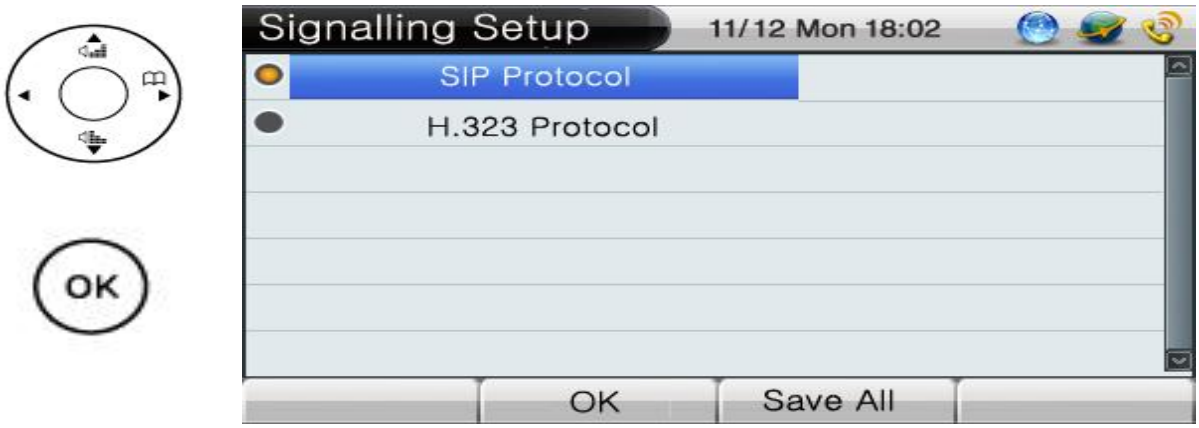


(Picture 4-41) VoIP Configuration Menu Screen

Network and Call Setup – VoIP Setup – VoIP Signaling

This menu is used for VoIP signaling setup such as H.323, SIP protocol. There are 2 different ways: connecting directly to VoIP network and connecting indirectly through SIP proxy server. Each way needs the different optional settings.





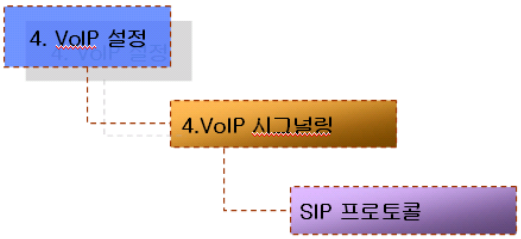
(Picture 4-42) VoIP Signaling Configuration Menu Screen

[Table 4-22] AP-IP230 IP Phone VoIP Signaling Configuration Screen Description

No.	Description
SIP Protocol	SIP parameter setup menu for SIP proxy server interworking
H.323 Protocol	H.323 parameter setup menu for H.323 Gatekeeper interworking

Network and Call Setup – VoIP Configuration – VoIP Signaling– SIP Protocol

This menu is used for configuring SIP protocol. There are 2 different ways: connecting directly to VoIP network and connecting indirectly through SIP proxy server. Each way needs the different optional settings.



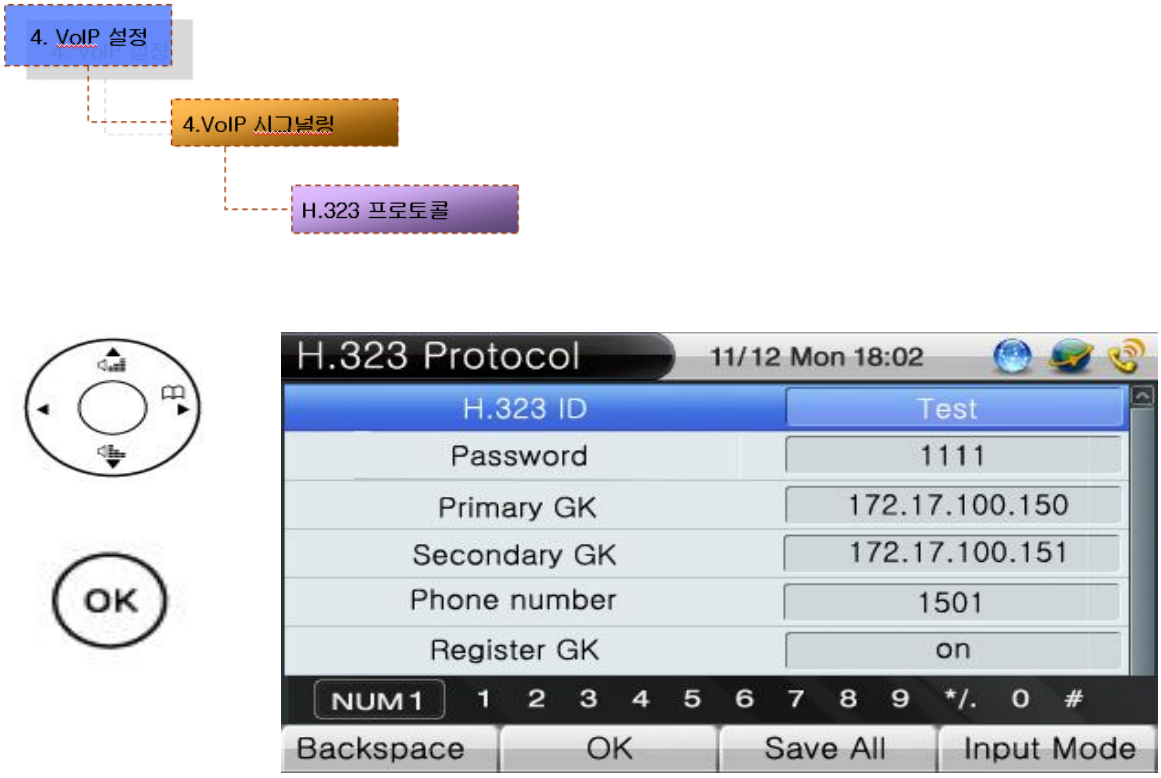
(Picture 4-43) SIP Protocol Configuration Menu Screen

[Table 4-23] AP-IP230 IP Phone SIP Protocol Configuration Screen Description

No.	Description
User Name	Enter a username for SIP server registration
User Password	Enter a password for SIP server registration
Primary Server	Enter the primary server IP address or domain of SIP server
Secondary Server	Enter the secondary server IP address or domain of SIP server
Phone Number	Enter user phone number
e.164 Register	Use the key pad or numeric key to register E.164 to SIP server

Network and Call Setup – VoIP Configuration – VoIP Signaling– H.323 Protocol

This menu is used for configuring H.323 protocdol. There are 2 different ways: connecting directly to VoIP network and connecting indirectly through SIP proxy server. Each way needs the different optional settings.



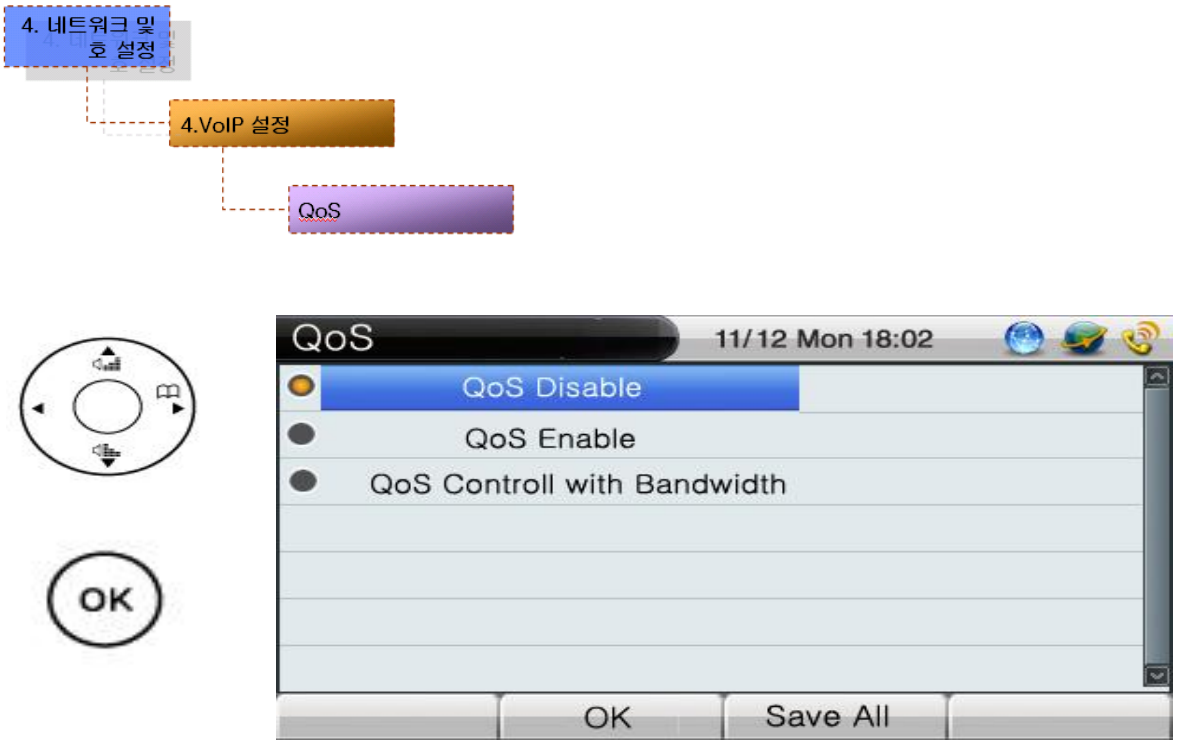
(Picture 4-44) H.323 Protocol Configuration Menu Screen

[Table 4-24] AP-IP230 IP Phone H.323 Protocol Configuration Screen Description

No.	Description
H.323 ID	Enter a H.323 ID for Gatekeeper registration
H.323 Password	Enter a H.323 password for Gatekeeper registration, if authentication is needed
Primary GK	Enter a primary Gatekeeper IP address
Secondary GK	Enter the secondary Gatekeeper IP address
Phone Number	Enter the user's E.164 number
Register GK	Use the key pad or numeric key to register E.164 to SIP server (on/off)

Network and Call Setup – VoIP Configuration – QoS

QoS enables transferring range of voice packet within a bandwidth limit. The user has to calculate the bandwidth, then to apply it to QoS.



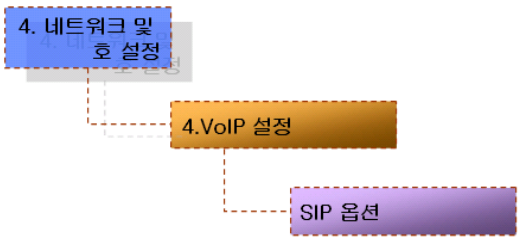
(Picture 4-45) QoS Configuration Screen

[Table 4-25] AP-IP230 IP Phone QoS Configuration Screen Description

No.	Description
QoS Deactivate	Deactivate QoS configuration
QoS Activate	Activate QoS configuration
Bandwidth Configuration	QoS function is for WAN interface and cannot be applied to LAN interface. Range of value covers 48Kbps~4Mbps

Network and Call Setup – VoIP Configuration – SIP Option

This menu is used for configuring additional features and options of SIP protocol. These optional settings are dependent on the network configuration.



(Picture 4-46) SIP Options Configuration Screen

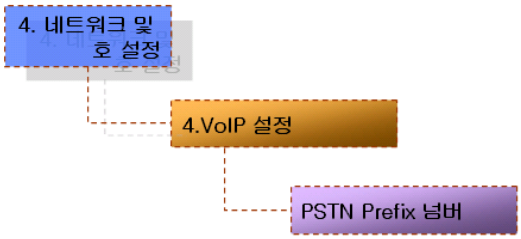
[Table 4-26] AP-IP230 IP Phone SIP Options Configuration Screen Description

No.	Description
Call Transfer Mode	Select the call-transfer mode. basic/attend.
Conference Service Tag	Enter a VoIP Tag for conference service
Conference Service Name	Enter a name for the conference service
Enable Ping	Enter firewall address to check the public IP address when AP-

	IP230 is used under NAT/Firewall network environment.
Media Channel	Transfer RTP Session information to listen Inband Ringbacktone of Public network under NAT/Firewall environment.
Minimum Second	Set Session Timer
Retry Counter	SIP UA Retry Counter sets SIP INVITE re-transmission count when AP-IP230 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-IP300 transfer next INVITE message. The default is set to 10.
Remote Party ID	When the user-name is not numeric but character, apply to register message.
Route by Auxiliary	When the called party is not number but characters, this option is used.
Set Local Domain	Transfer From/To field within SIP message to designated domain not to IP address.
Signaling Port	The default is 5060 and this value is changeable.
Srv	Set the DNS SRV.
User Register	When the user-name is not numeric but character, this option is used to register SIP server.

Network and Call Setup – VoIP Configuration – PSTN Prefix

When user wants to access the FXO interface for PSTN backup, this prefix number is used as PSTN access code. Additionally, AP-IP230 IP phone supports the PSTN back-up service when VoIP service is impossible due to network failure or VoIP call service is interrupted by an exception.



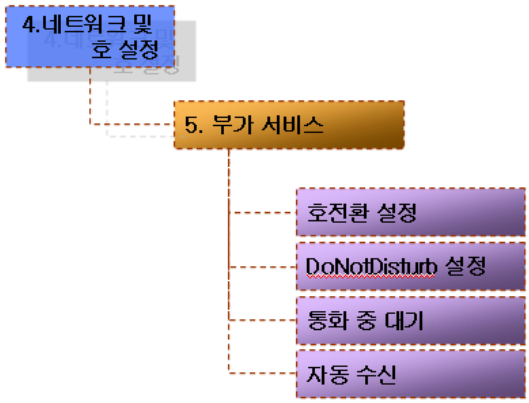
(Picture 4-47) PSTN Prefix Configuration Description

[Table 4-27] AP-IP230 IP Phone PSTN Prefix Configuration Screen Description

No.	Description
Number Input	PSTN prefix number is an access code for PSTN FXO interface, default value is #.

Network and Call Setup – Additional Service

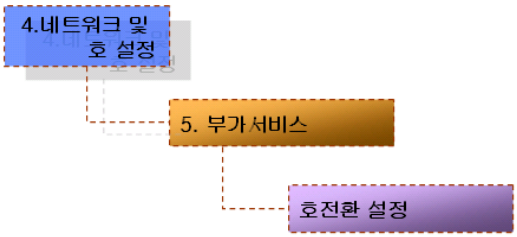
The Additional Service menu sets up Call Transfer, DND, Call Wait and Auto Response.

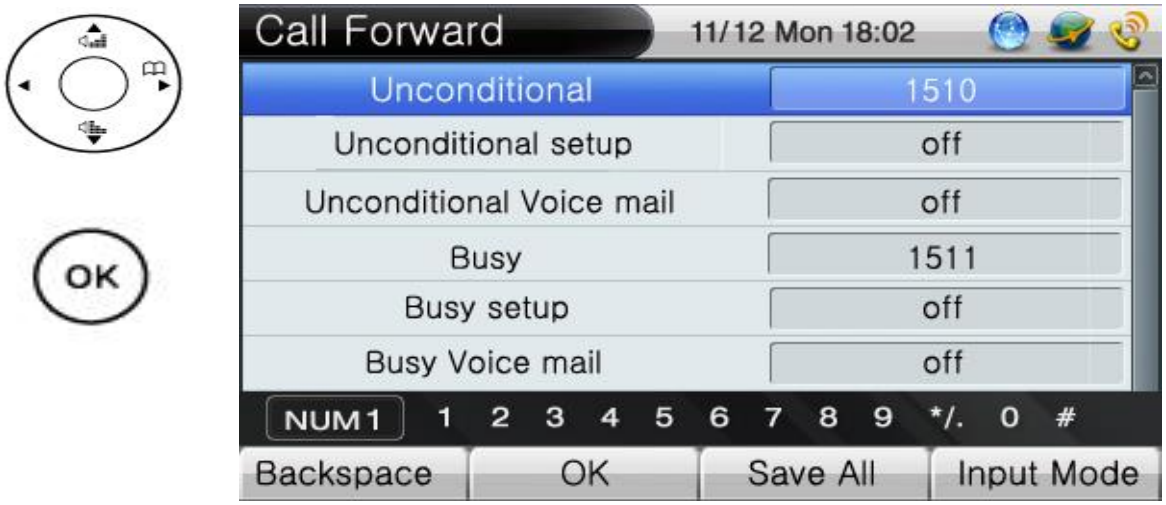


(Picture 4-48) Additional Service Configuration Screen

Network and Call Setup – Additional Service – Call Forward

This is the menu sets up the call forward when the user is busy on line or unable to answer the call or forward a call unconditionally. When a call is forwarded, you can set the call to a specific number or voice mail. If you set it to the both, the voice message is applied.





(Picture 4-49) Call Forward Configuration Menu Screen

[Table 4-28] AP-IP230 IP Phone Call Forward Configuration Screen Description

No.	Description
Unconditional	Enter the number to be forwarded to no matter what (Call Forwarding Unconditional)
Unconditional Setup	Enable or Disable Call Forwarding Unconditional (the default setting: Disable)
Unconditional Voice Mail	Disable or Enable the Call Forwarding to be connected to Voice Mail when there is no answer (the default setting: disable)
Busy	Enter the number to be forwarded to when the line is busy
Busy Setup	Disable or Enable the Call forwarding when the line is busy (the default setting : Disable)
Busy Voice Mail	Set the Call Forwarding to be connected to Voice Mail when the line is

	busy (the default setting : Disable)
No Answer	Enter the number to be forward to when there is not answer
No Answer	Enable or Disable Call Forwarding when there is no answer (the default setting: Disable)
No Answer Voice Mail	Disable or Enable the Call Forwarding to be connected to Voice Mail when there is no answer (the default setting: disable)

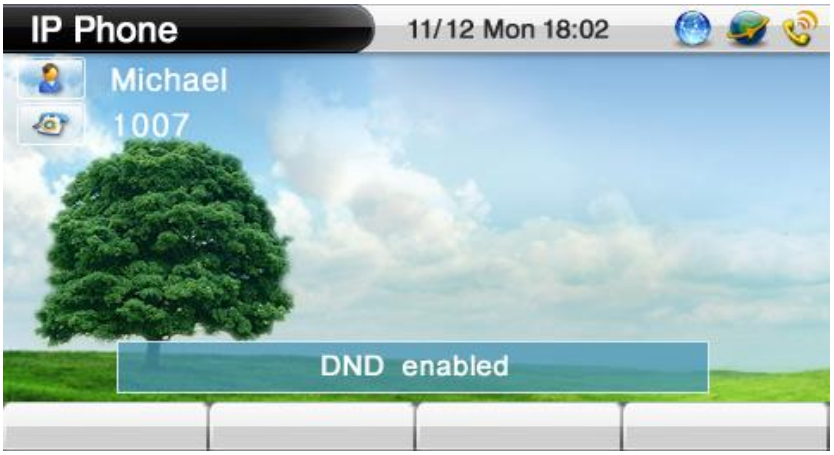
Network and Call Setup – Additional Service – DND(Do Not Disturb)

Do Not Disturb (DND) features allows you to turn off the ringer (Ring Silence) for an incoming call or to reject the call (Call Reject. You may hold pressing the leave of absence button of the IP-Phone for more than 2 seconds to enable or disable this function. The Call Reject can work only in the SSCP mode.

(Picture 4-50) DND Configuration Menu Screen

[Table 4-29] AP-IP230 IP Phone DND Configuration Menu Screen Description

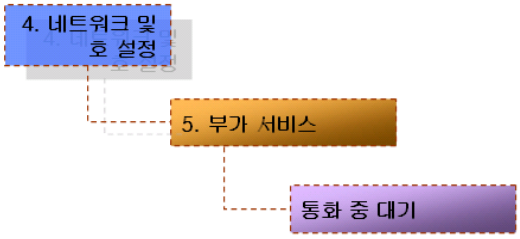
No.	Description
Call Reject	Set the mode to Call Reject
Ring Silence	Set the mode to Ring Silence

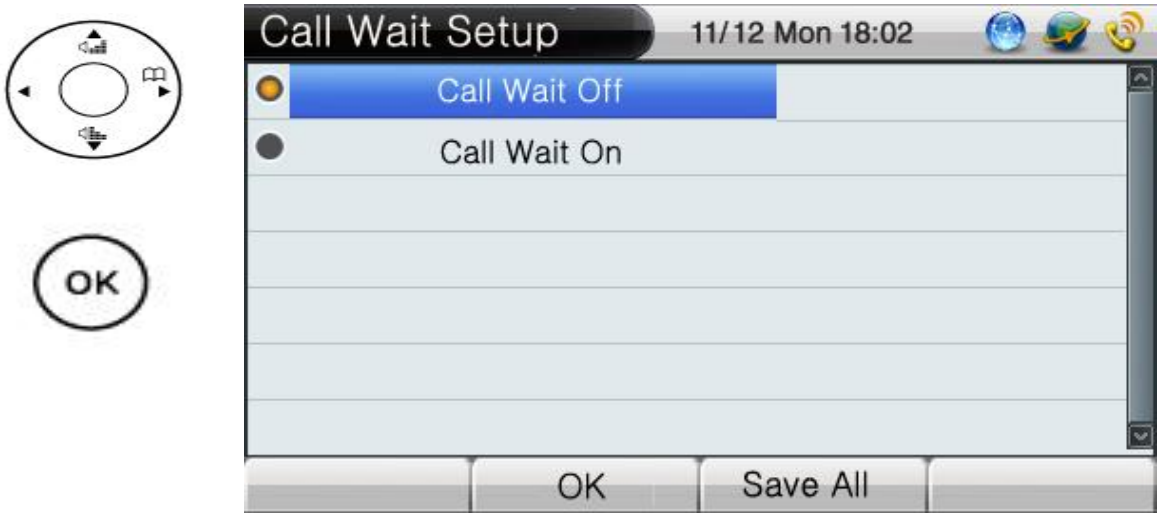
 Hold more than 2 seconds	
[DND] =>Enable DND	
 Hold more than 2 seconds	
Disable DND	

(Picture 4-51) DND Configuration Screen

Network and Call Setup – Additional Service – Call Waiting

Call Wait feature enables you to receive a second incoming call with on the same line without disconnecting the first call. This call feature allows you to receive an auditory call alert while you are on the first call. You can place the first on Hold and wait and connect to the second call. You can even return to the first call after you finish conversation with the second call.





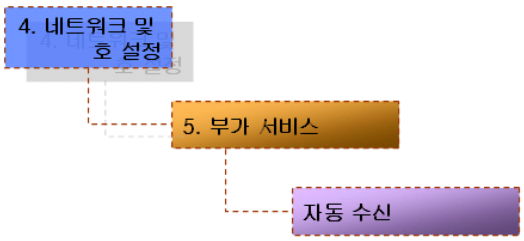
(Picture 4-52) Call Waiting Configuration Menu Screen

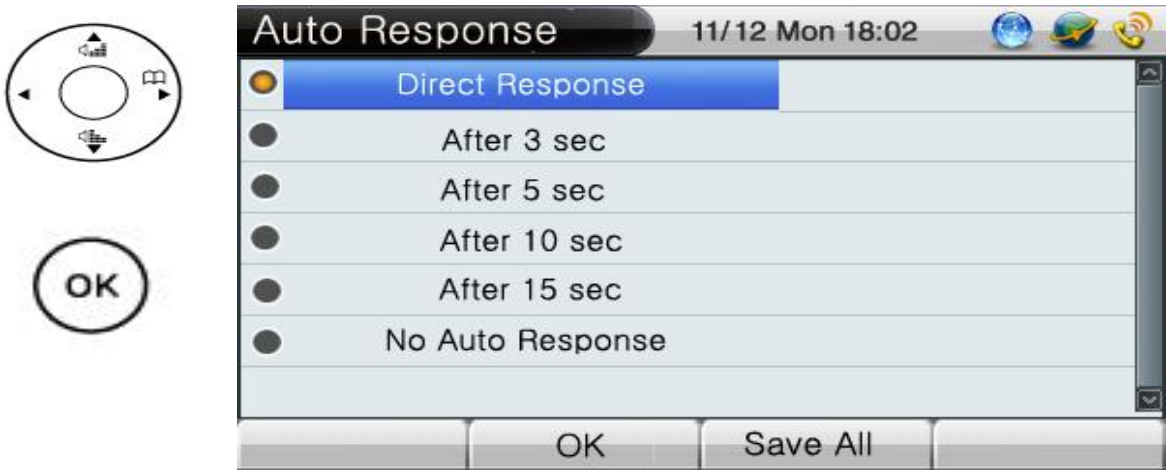
[Table 4-30] AP-IP230 IP Phone Call Waiting Configuration Menu Screen Description

No	Description
Call Waiting Disable	Disable Call Waiting
Call Waiting Enable	Enable Call Waiting

Network and Call Setup – Additional Service – Auto Answering

This feature allows your telephone to answer a call automatically and you do not have to pick up the phone. You can set the interval of answering a call selectively: 3, 5, 10 or 15 seconds.





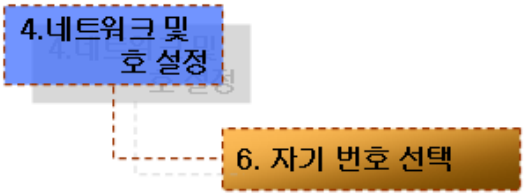
(Picture 4-53) Auto Answering Configuration Menu Screen

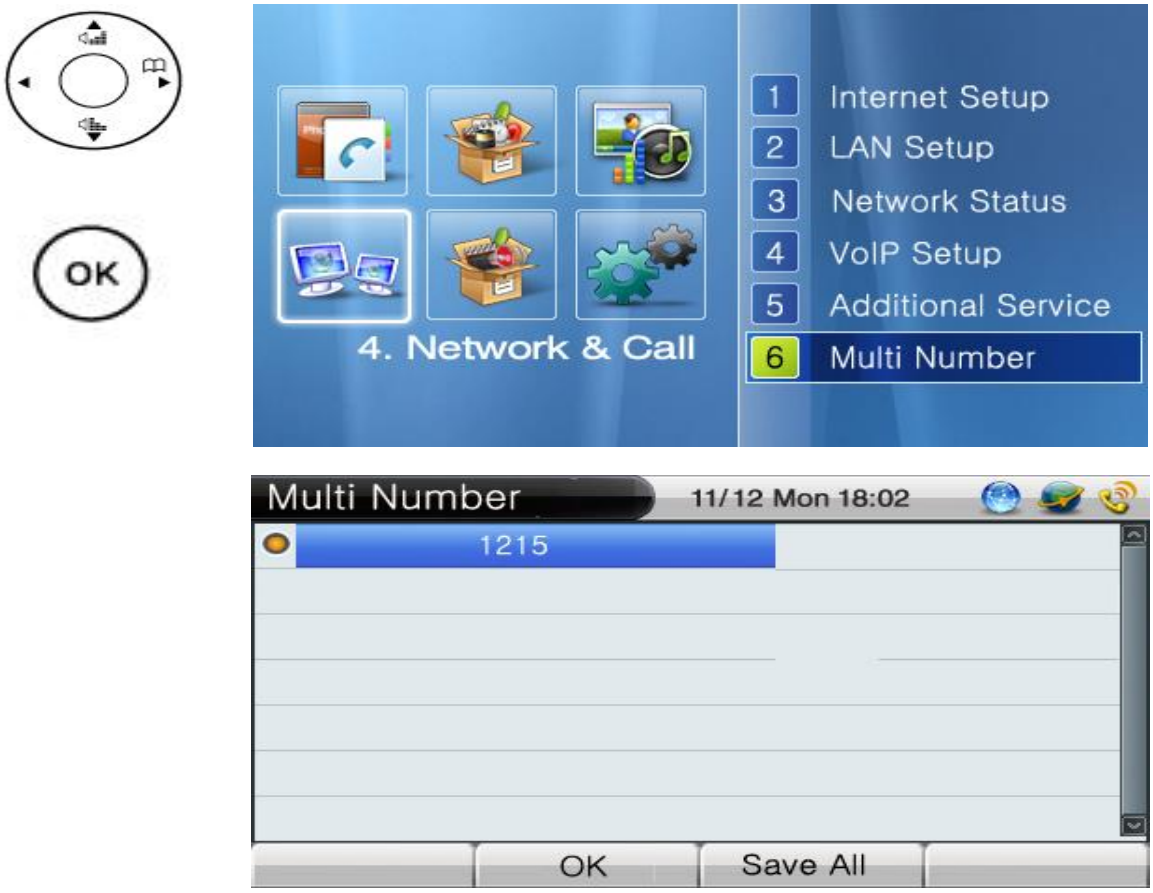
[Table 4-31] AP-IP230 IP Phone Auto Answering Configuration Menu Screen

No.	Description
Direct Response	Take a call at first ring
After 3 sec	Set the mode to Auto Answer to reply on 3 seconds after the bell rings.
After 5 sec	Set the mode to Auto Answer to reply on 5 seconds after the bell rings.
After 10 sec	Set the mode to Auto Answer to reply on 10 seconds after the bell rings.
After 15 sec	Set the mode to Auto Answer to reply on 15 seconds after the bell rings.
No Auto Response	Disable Auto Response

Network and Call Setup – Select Multi Number

The Multi Number allows you to set the native number for the Outbound Call, as to select the one number among many numbers that have been assigned. You can take many numbers of incoming calls, but you can send only the predetermined number of the outgoing call at the default setting.





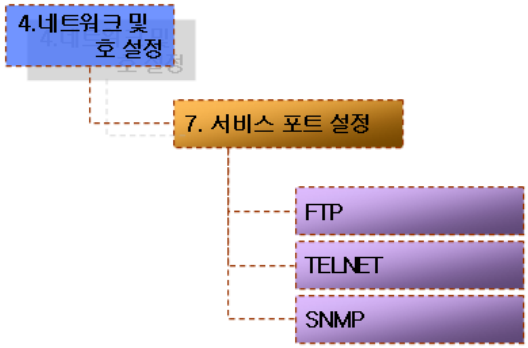
(Picture 4-54) Multi Number Menu Screen

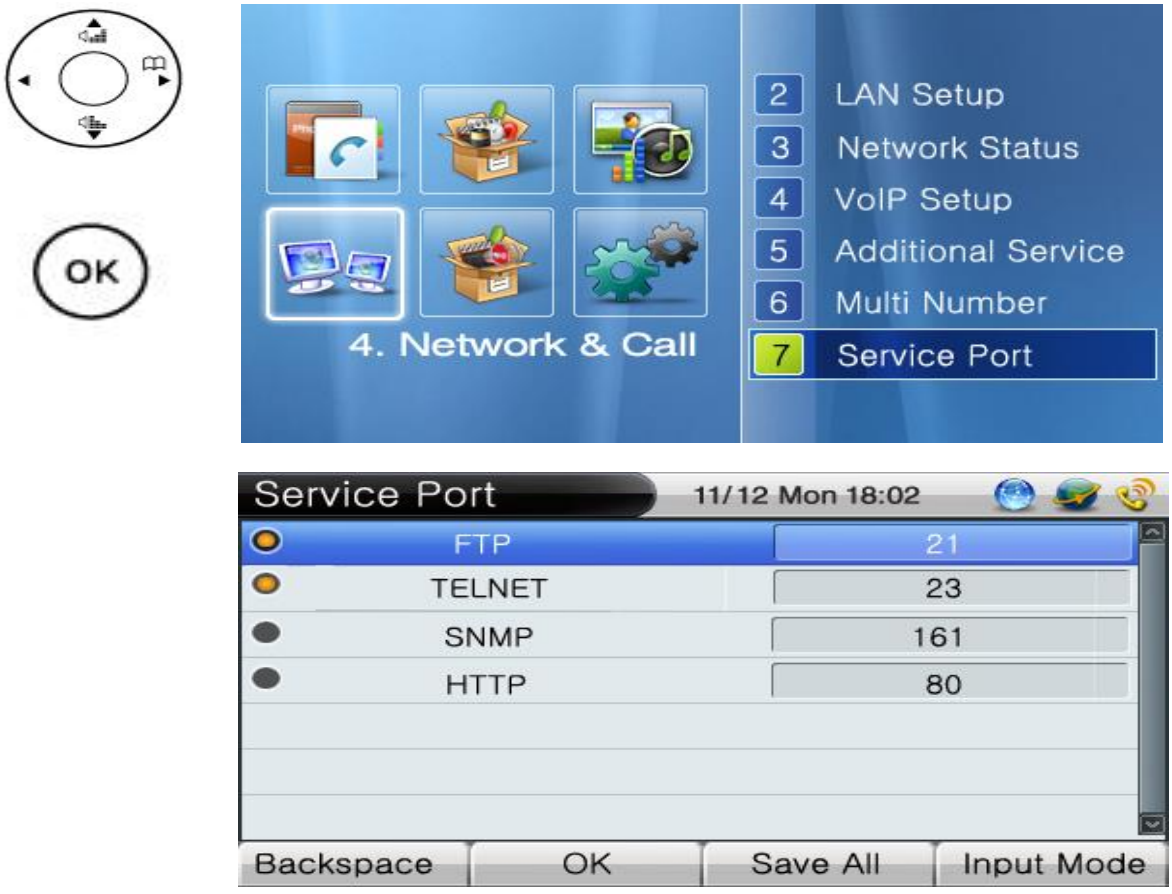
[Table 4-32] AP-IP230 IP Phone Multi Number Selection Screen Description

No.	Description
Multi Number	This screen shows an example for assigning the phone number of 1215

Network and Call Setup – Service Port Configuration

This menu activates or deactivates FTP, TELNET, TFTP, SNMP protocol service of AP-IP230. You can use FTP to access to AP-IP230 from a remote location and Telnet is used for changing all kinds of information and monitoring and SNMP is also used to access to AP-IP230 from a remote location.





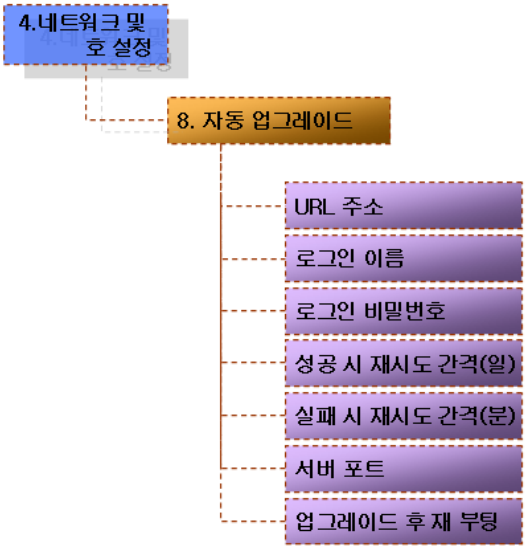
(Picture 4-55) Service Port Configuration Menu Screen

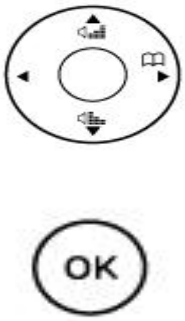
[Table 4-33] AP-IP230 IP Phone Service Port Configuration Screen Description

No.	Description
FTP	Actives/Deactivates the FTP service protocol. Default is enable mode (activating FTP service). Default port number is 21.
TELNET	Actives/Deactivates the TELNET service protocol. Default is enable mode (activating TELNET service). Default port number is 23.
SNMP	Actives/Deactivates the SNMP service protocol. Default is enable mode (activating SNMP service). Default port number is 161
HTTP	Enable or disable HTTP service The default is set to disable. The default port number is set to 80

Network and Call Setup – Auto Upgrade

Whenever a new feature is added, the software (firmware) of the IP phone needs to be upgraded. One of the ways of doing this upgrade is download the new software by using a network transmission protocol such as ftp which is capable of transmitting a large files. This Auto Upgrade enables the phone to access a particular server and to compare the version of OS and Configuration. Then it determines to download the firmware.





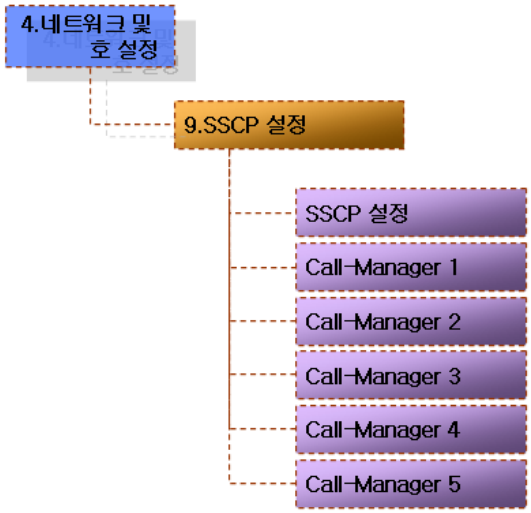
(Picture 4-56) Auto Upgrade Configuration Menu Screen

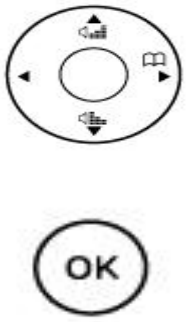
[Table 4-34] AP-IP230 IP Phone Auto Upgrade Configuration Screen Description

No.	Description
URL Address	Enter URL of the Auto Upgrade server Ex)down.addpac.com/apos/IP230/packing.lst
Login Name	Enter the ID for an authorized access to the Auto Upgrade server Ex) addpac
Login Password	Enter the password for an authorized access to the Auto Upgrade server Ex) addpac
Interval Success (day)	The succeeded Auto Upgrade can be kept in a record for a certain time. The basic default value is set to 30 days.
Interval Retry (Min)	The failed Auto Upgrade can be kept in a record for a certain time. The basic default value is set to 10 minutes.
Server Port	Enter a Port of the Auto Upgrade Server. The default value is set to 80 for HTTP.
Apply Reboot	Select whether to apply the settings of the Auto Upgrade after rebooting or not Ex) On/Off (using the numeric button for On/Off)

Network and Call Setup – SSCP

SSCP Smart Service Control Portocol) is the AddPac proprietary protocol operates between the AddPac IP-PBX systems and IP terminals. The IP-PBX systems support many different call features, through SSCP, in addition to the basic call features of the IP Phone itself. The IP terminals take these call features supported by the IP-PBX, then it display these features on its softkeys. These call features include Redial, GroupPark, GroupPickup, NewCall, CCBS, Park, Pickup, Transfer, Hold, AddParty, Conference.





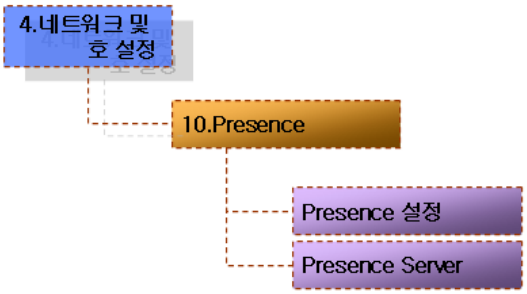
(Picture 4-57) SSCP Configuration Menu Screen

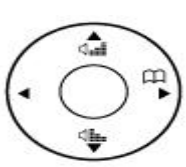
[Table 4-35] AP-IP230 IP Phone SSCP Configuration Screen Description

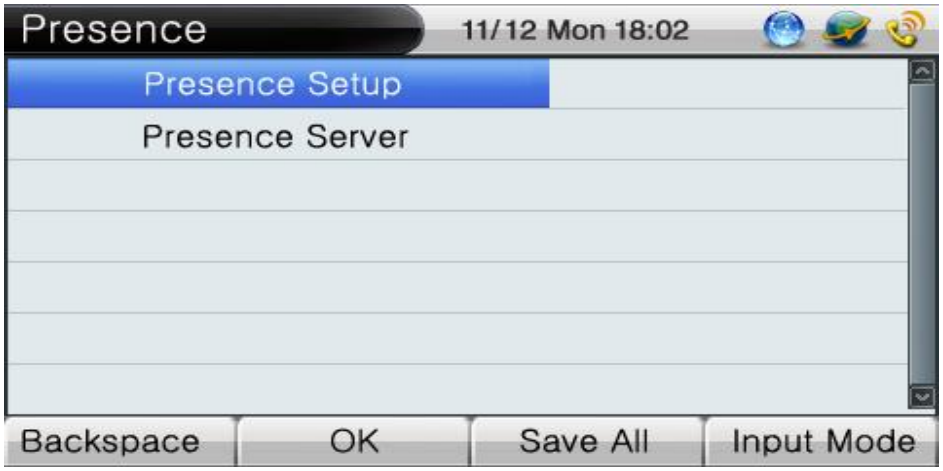
No.	Description
SSCP Configuration	Either enable or disable the setting mode of SSCP (On/Off)
Call-manager 1 ~ 5	Configure the Call Manager server: 5 servers can be configured at maximum. In case of redundancy, 2 Call Manager server (Call Manager 1 and Call Manager 2 are to be configured)

Network and Call Setup – Presence.

When AP-IP230 is connected with Presence Server, the IP300 can take the Speed Button Key from the server. The LED of each speed button key is changed on real time basis, so the present status of the user can be informed. To be connected with Presence Server, you should know the address of the server, port number, ID and password.



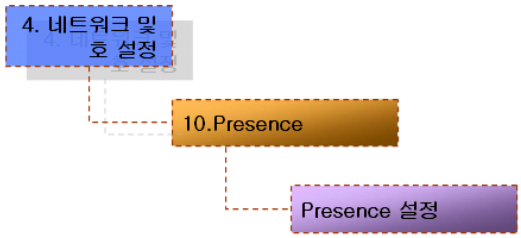


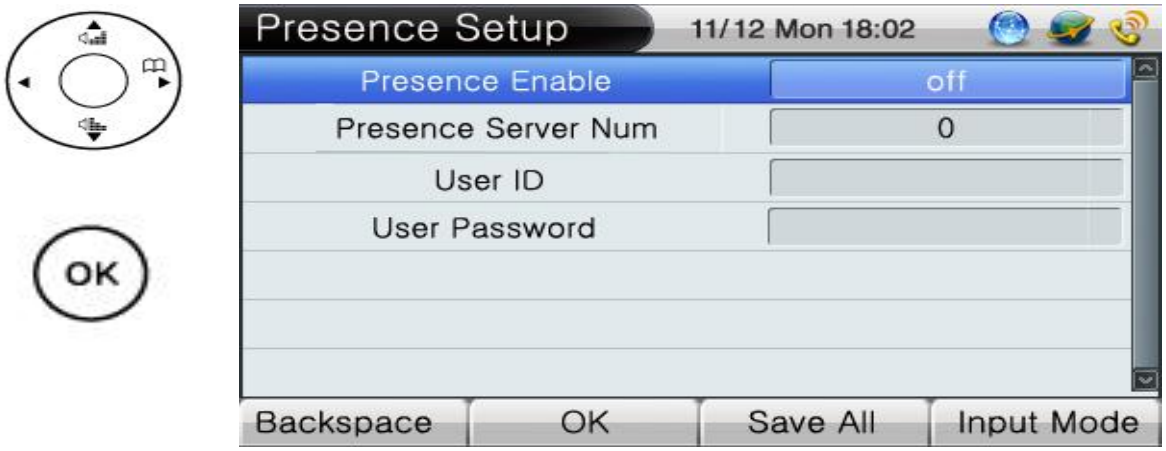


(Picture 4-58) Presence Menu Screen

Network and Call Setup – Presence – Presence Configuration

In order to register to Presence Server, you may enter an ID and password. When you enter one server address and port, the Presence Server Number is indicated as 1. And then you may enter the ID and password which have been registered to the IP-PBX.





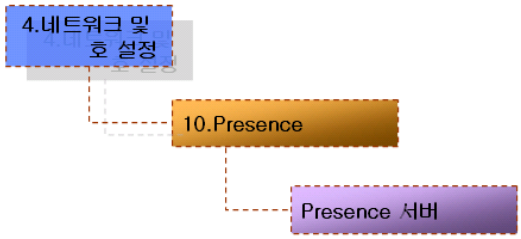
(Picture 4-59) Presence Configuration Menu Screen

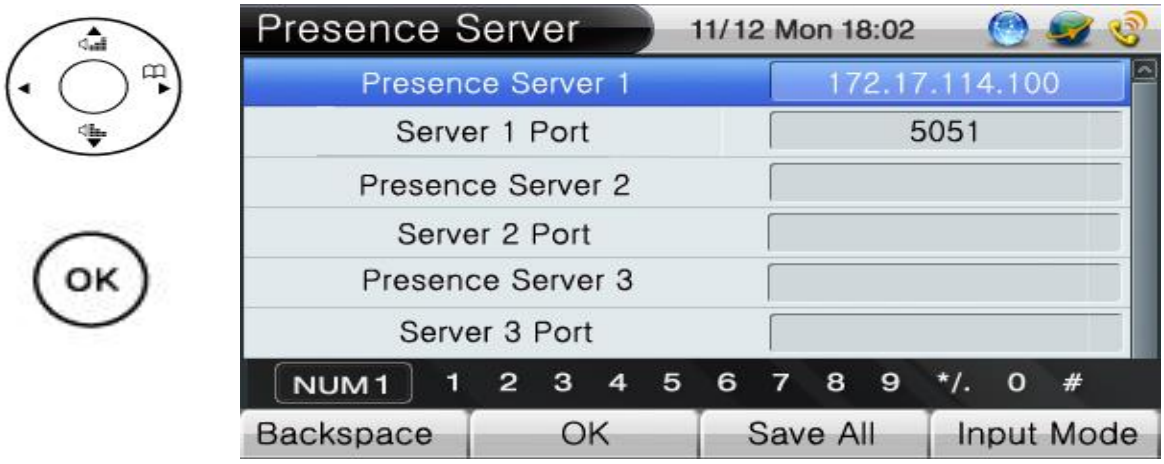
[Table 4-36] AP-IP230 IP Phone Presence Configuration Screen Description

No.	Description
Presence	Enable or disable the Presence Setup
Presence Server Number	When you enter one server address and port, the Presence Server Number is indicated as 1
ID	An ID to be registered to the Presence Server (same as the one registered to SMM)
Password	A password to be registered to the Presence Server (same as the one registered to SMM)

Network and Call Setup – Presence – Presence Server

You can enter an IP and port to register to Presence Server. The default port number is 5051 and the server address supports both IPv4 / IPv6.





(Picture 4-60) Presence Server Configuration Menu Screen

[Table 4-37] AP-IP230 IP Phone Presence Server Configuration Screen Description

No.	Description
Server Address	Enter the IP address to be used to Presence Server at default
Server Port Number	Enter the port number to be used to Presence Server at default (Default : 5051).

Application Program Menu

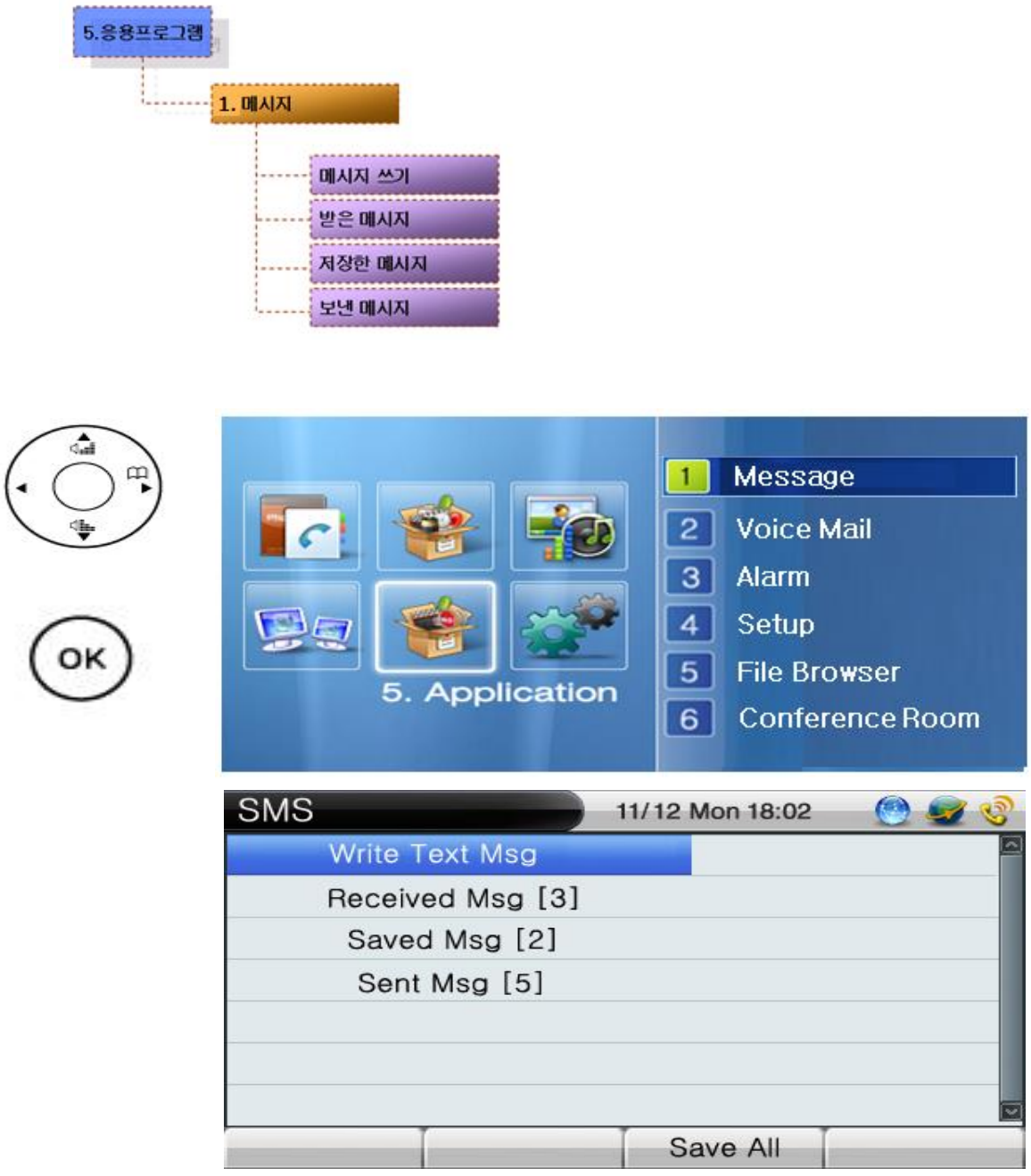
Applications composed of a group of the call features including Message Box, Voice Mail Box, Conference. You can use Message Box and Voice Box only when they are connected to and supported by SSCP. You can make Conference calls on when they are connected to and supported by Multipoint Control Unit (MCU). For remote broadcasting, you may need the AddPac Broadcasting **Equipment (AP3120) and Broadcasting management program** (e-MBMS Server).

(Picture 4-61) Main Screen

(Picture 4-62) Application Menu Screen

Application Program – Message

Message function can be used when interworking with AddPac IP-PBX and SSCP (Smart Service Control Protocol)



(Picture 4-63) Message Configuration Menu Screen

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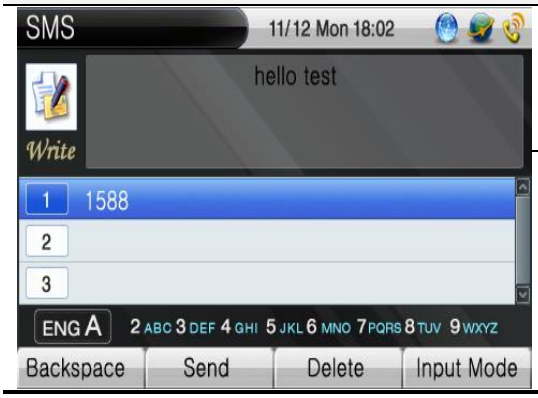
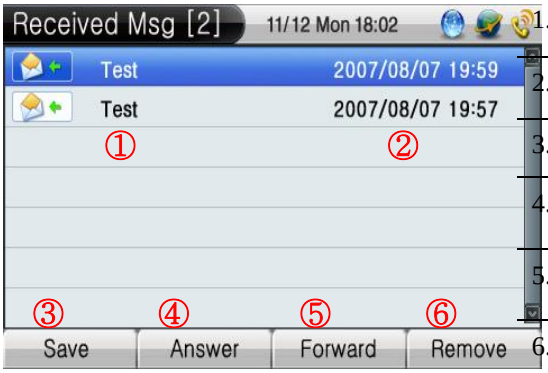
[Table 4-38] AP-IP230 IP Phone Message Configuration Screen Description

No.	Description
Write Message	A menu for write/send SMS message
Received Message	Received SMS message box
Saved Massage	Saved SMS message box
Sent Massage	Sent SMS message box

SMS

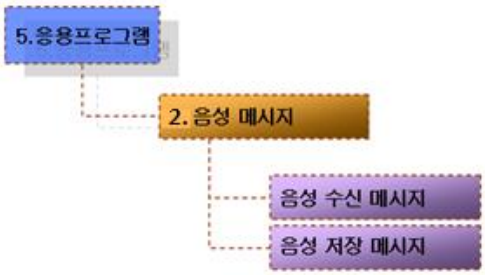
Category	Description
	Enter SMS contents. (F4 : English, Number change) . Enter sender phone number (Max 9)

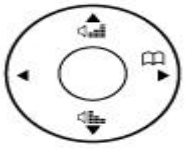
Received SMS Message

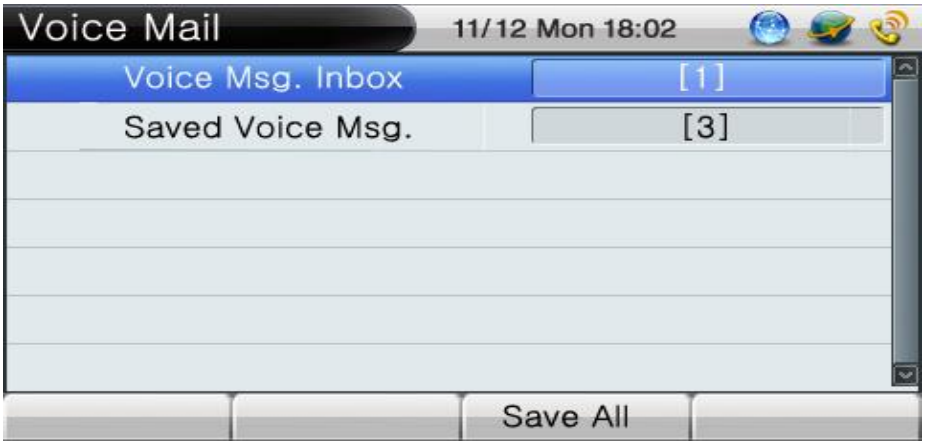
Category	Description
	1. Received SMS message 2. Received SMS message date/time 3. Received SMS message save 4. Resend 5. Send received message to third party 6. Delete received SMS message

Application Program – Voice Message

Voice Mail enables you to check the voice messages by pressing Play key. You should know the user's password which has been registered to IP-PBX to listen to the voice messages.







(Picture 4-64) Voice Message Configuration Menu Screen

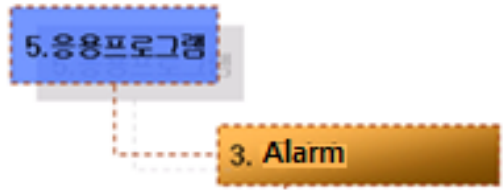
[Table 4-39] AP-IP230 IP Phone Voice Message Configuration Menu Screen

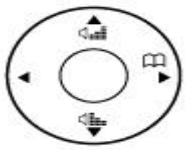
No.	Description
Voice Msg. Inbox	This is the box to keep the sent voice messages are saved
Saved Voice Msg.	This is where save the voice messages

(Picture 4-65) Saved Voice Message Screen

Application Program – Alarm

Alarm function supports alarm function by playing the sound of bell.

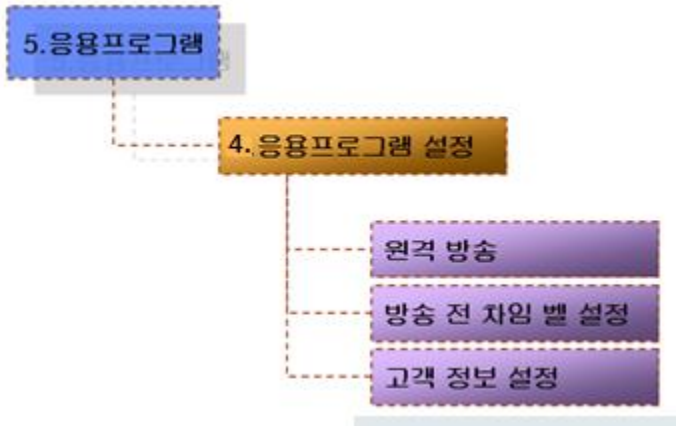


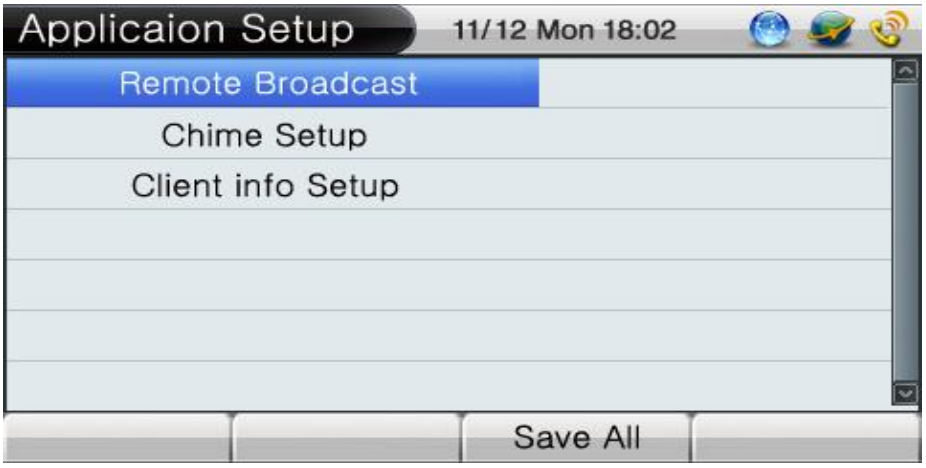


(Picture 4-66) Alarm Configuration Screen

Application Program - Application Program Configuration

Voice broadcasting is composed of remote broadcasting setup and view customer information. To use remote broadcasting, AddPac broadcasting device (AP3120) and broadcasting management program (e-MBMS server) are required.





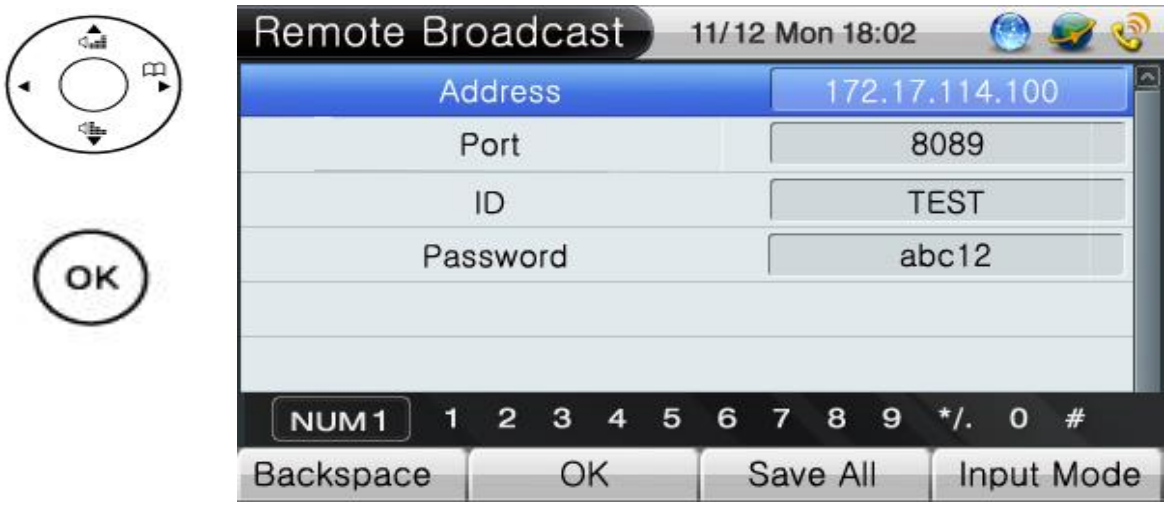
(Picture 4-67) Application Program Configuration Menu Screen

[Table 4-40] AP-IP230 IP Phone Application Program Screen Description

No.	Description
Remote Broadcasting	Setup IP, port, ID, and password
Setup Chime Bell	Select Chime Bell Setup/Cancel
Customer Information	Setup IP, port for customer information

Application Program - Application Program Configuration – Remote Broadcasting

It is a menu to setup e-MBMS server address, port, user ID, and password. To user remote broadcasting, you may use AddPac broadcasting device (AP3120) and broadcasting management program (e-MBMS server)



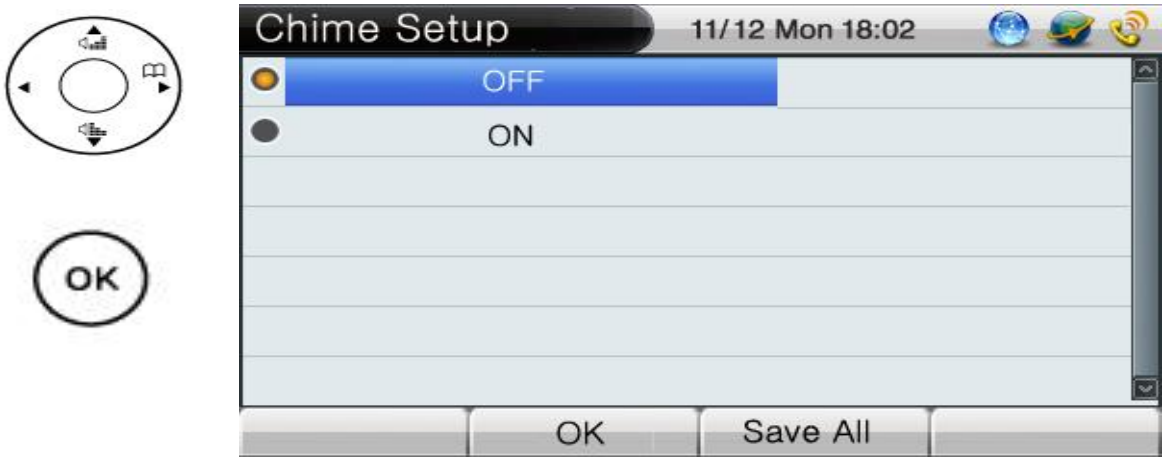
(Picture 4-68) Remote Broadcasting Configuration Menu Screen

[Table 4-41] AP-IP230 IP Phone Remote Broadcasting Configuration Description

No.	Description
Address	Enter e-MBMS server IP address
Port	Enter e-MBMS server port number (Default: 8089)
ID	Enter e-MBMS server ID
Password	Enter e-MBMS server password

Application Program - Application Program Configuration – Chime Bell Setup

It is a menu to setup/cancel chime bell.



(Picture 4-69) Remote Broadcasting Configuration Menu Screen

[Table 4-42] AP-IP230 IP Phone Remote Broadcasting Configuration Description

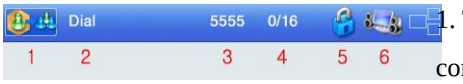
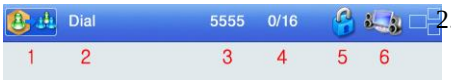
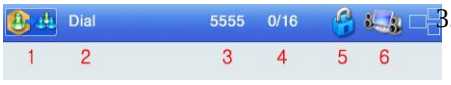
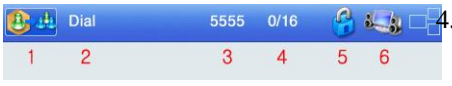
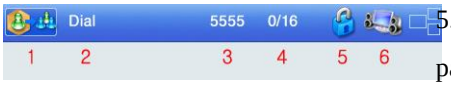
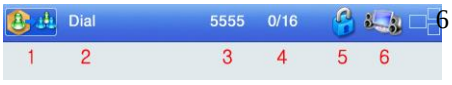
No.	Description
Chime Bell Off	Do not send chime bell when starting remote broadcasting
Chime Bell On	Send chime bell when starting remote broadcasting

Application Program – Conferencing

This feature enables you to see the list of connections can be made for a conference call at the present time and you can join the conference by just pressing call button. There are 4 different ways of participating in conference call: Ad Hoc, Dial-Out, Ad Hoc Dial-Out, Meet-Me and the conference parties can be classified by each of their ranks: Chair, Operator, Participant, Audience

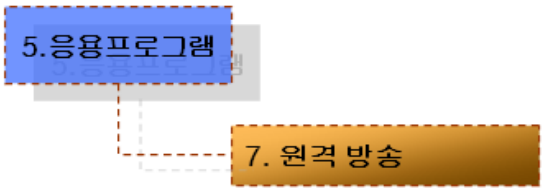
(Picture 4-70) Conferencing Configuration Menu Screen

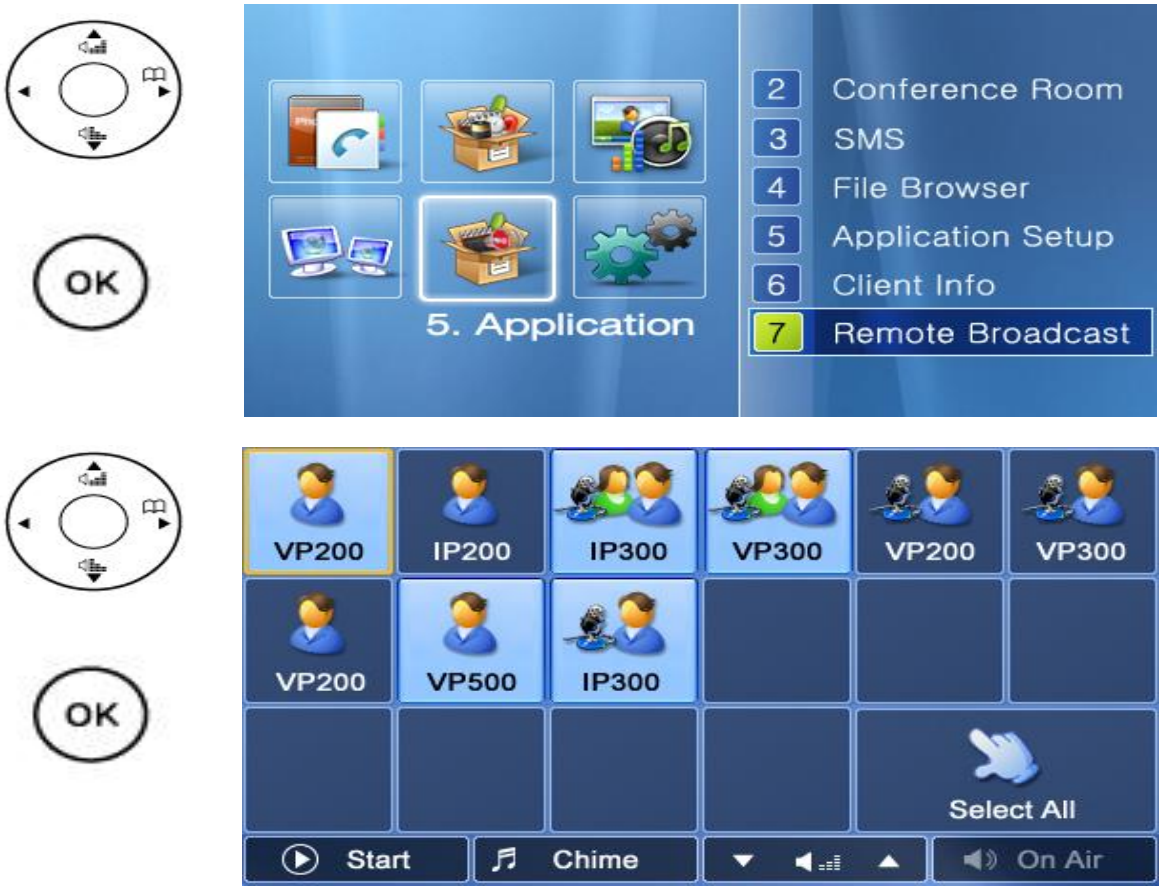
[Table 4-43] AP-IP230 IP Phone Conferencing Configuration Screen Description

No.	Description
	1. The icon for the conference room represents the Dial-out conference
	2. the name of the conference room
	3. The Speed Dial Number for the Conference
	4. The conference with 16 participants
	5. locked for Secret Room: only the user who knows the password can enter
	6 . The media type is set to video

Application Program – Remote Broadcasting

Remote broadcasting screen allows checking the status of server and accessed terminals. To start broadcasting, select broadcasting with navigation key and ok key, then start broadcasting with F1 button.



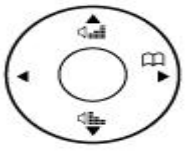


(Picture 4-71) Remote Broadcasting Participants Screen

System Configuration Menu

System configuration menu let you setup displayed name/number on the screen. It is composed of activation/deactivation to use speed dial.

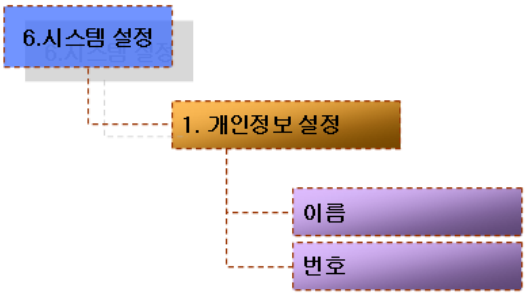
(Picture 4-72) Main Screen

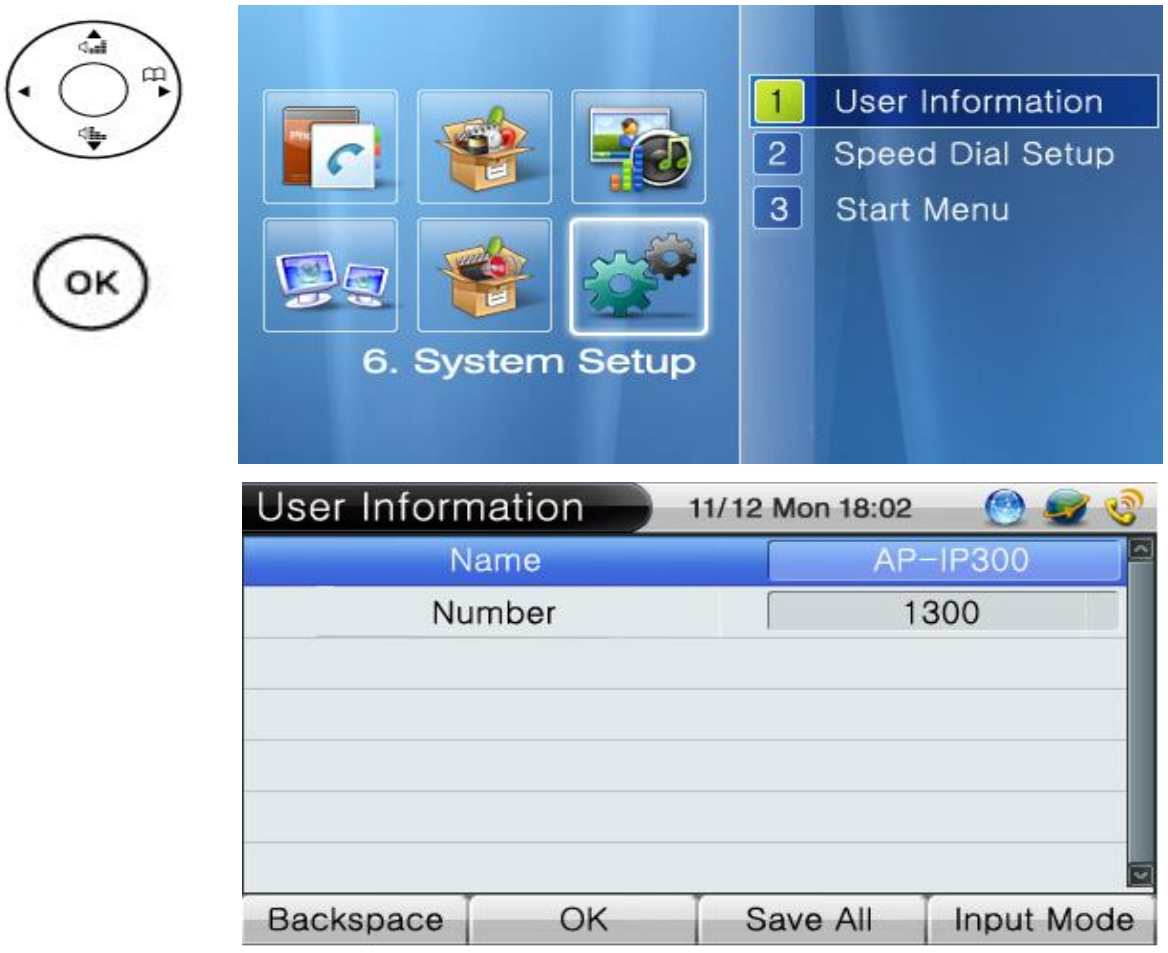


(Picture 4-73) Menu Screen

System Setup – User Information

This menu sets up the name and number that the user wants on the desktop area. Even if the name and number has been set up, the name and number are displayed on the desktop area, which are taken from the IP-PBX.





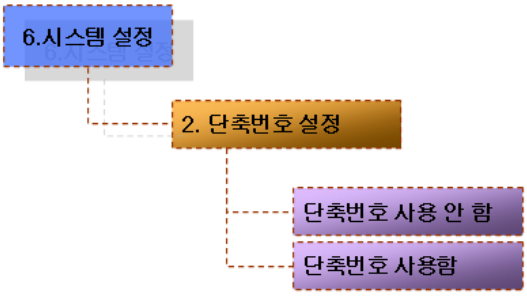
(Picture 4-74) User Information Configuration Menu Screen

[Table 4-44] AP-IP230 IP Phone User Information Screen Description

No.	Description
Name	Enter the name to be displayed on the desktop area
Number	Enter the number to be displayed on the desktop area

System Configuration – Speed Dial

The Speed Dial Setup menu allows the user to place a call by pressing the speed dial number which has been assigned with the outgoing call number. In order to use this feature, you may enable the Speed Dial and the Speed Dial Number should be assigned to the Phonebook.



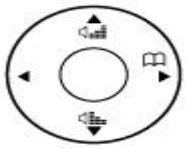
(Picture 4-75) Speed Dial Configuration Menu Screen

[Table 4-45] AP-IP230 IP Phone Speed Dial Configuration Screen Description

No.	Description
Speed Dial Not Use	Not using the Speed Dial
Speed Dial Use	Using the Speed Dial

System Configuration – Start Menu

The default screen (Tree) and Speed Button Map can be used selectively. Instead of the default screen, the Speed Button key can be used on the upper left side of the keypad to change the desktop screen.



(Picture 4-76) Start Menu Description Menu Screen

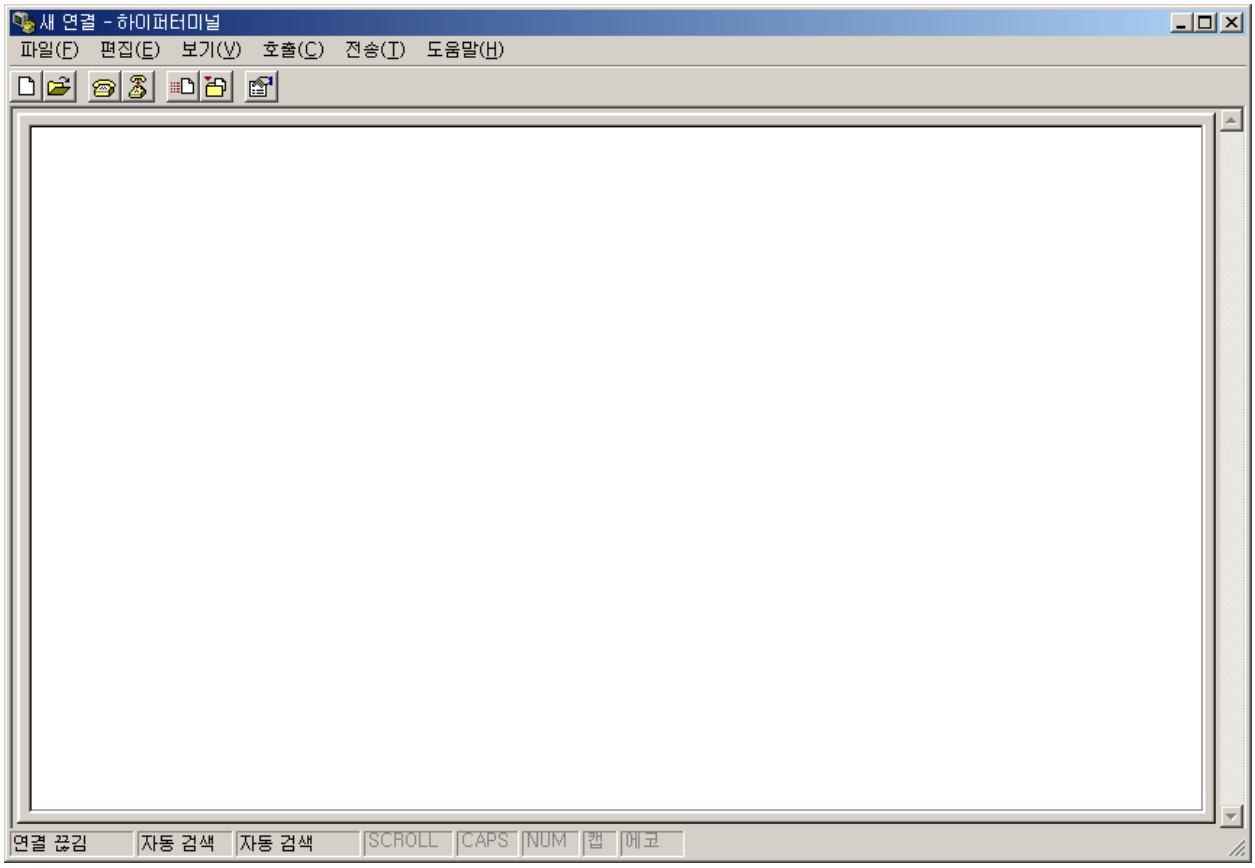
[Table 4-46] AP-IP230 IP Phone Start Menu Configuration Screen Description

No.	Description
Idle	Using the default screen
Speed Button	Using the Speed Button Map

Chapter 5. AP-IP230 Testing Operation

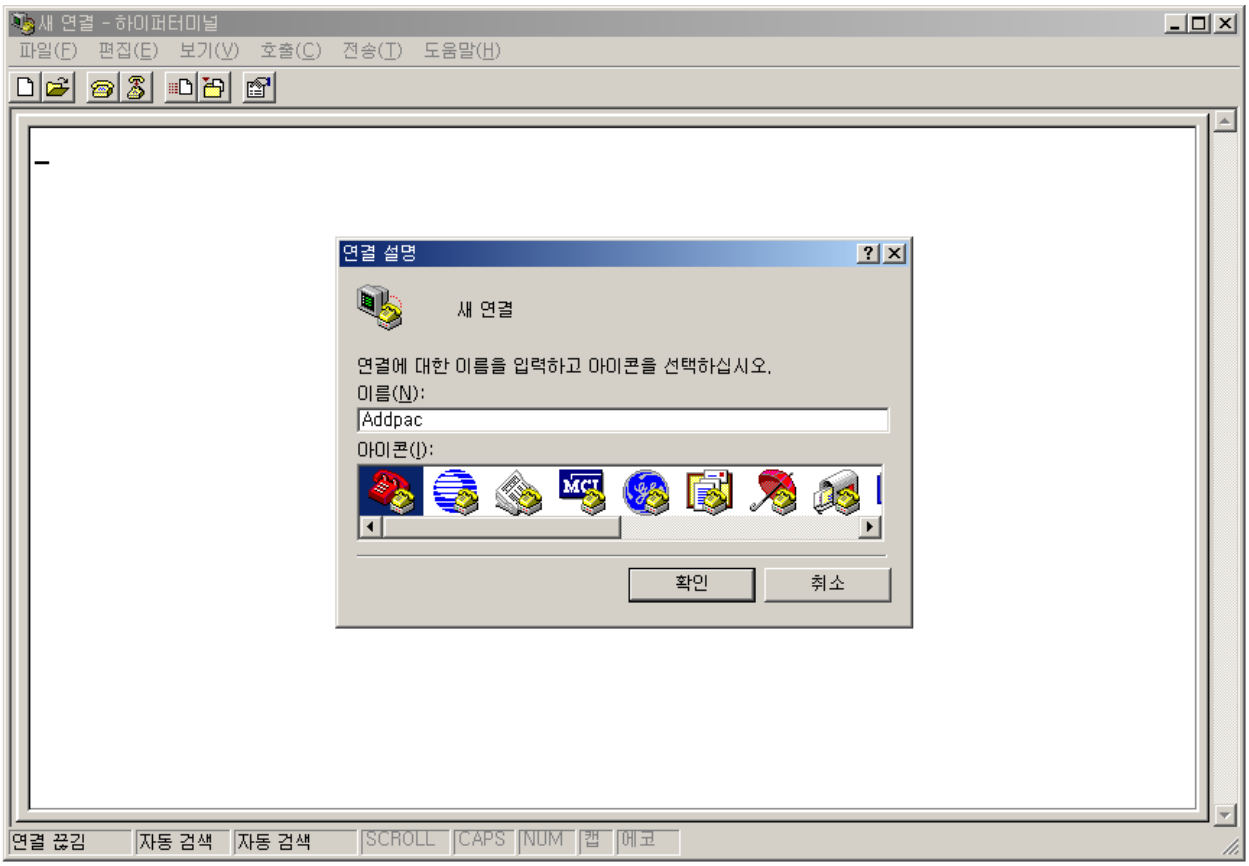
Using HyperTerminal for the Console

Terminal Emulator Application must be installed for using the PC for the console terminal. Hyper-Terminal Application is used for MS-Windows.



(Picture 5-1) MS-Windows Terminal Emulator HyperTerminal

After HyperTerminal is performed, determine the name of the new connection. The user can decide the name of the connection.

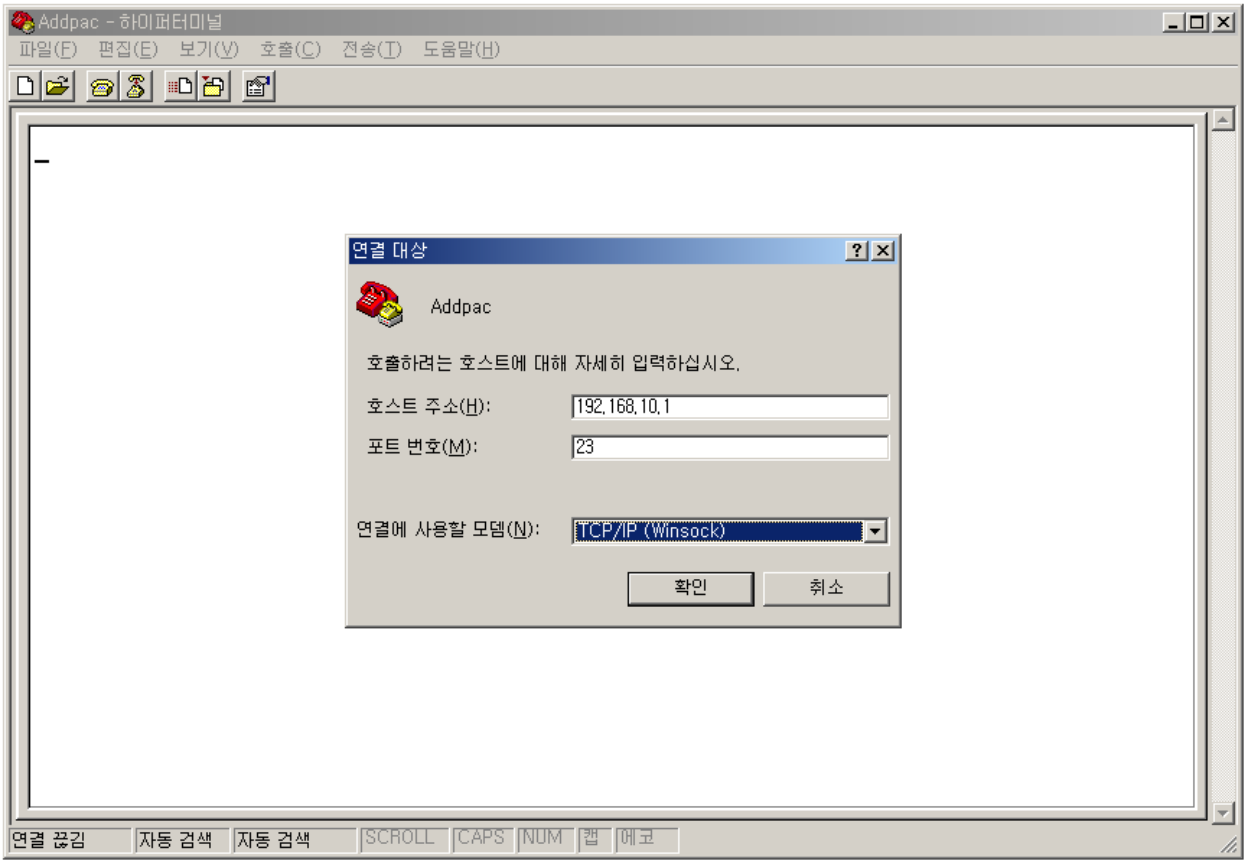


(Picture 5-2) Entering the Name of the Connection in HyperTerminal

Select the interface of which the console cable is connected

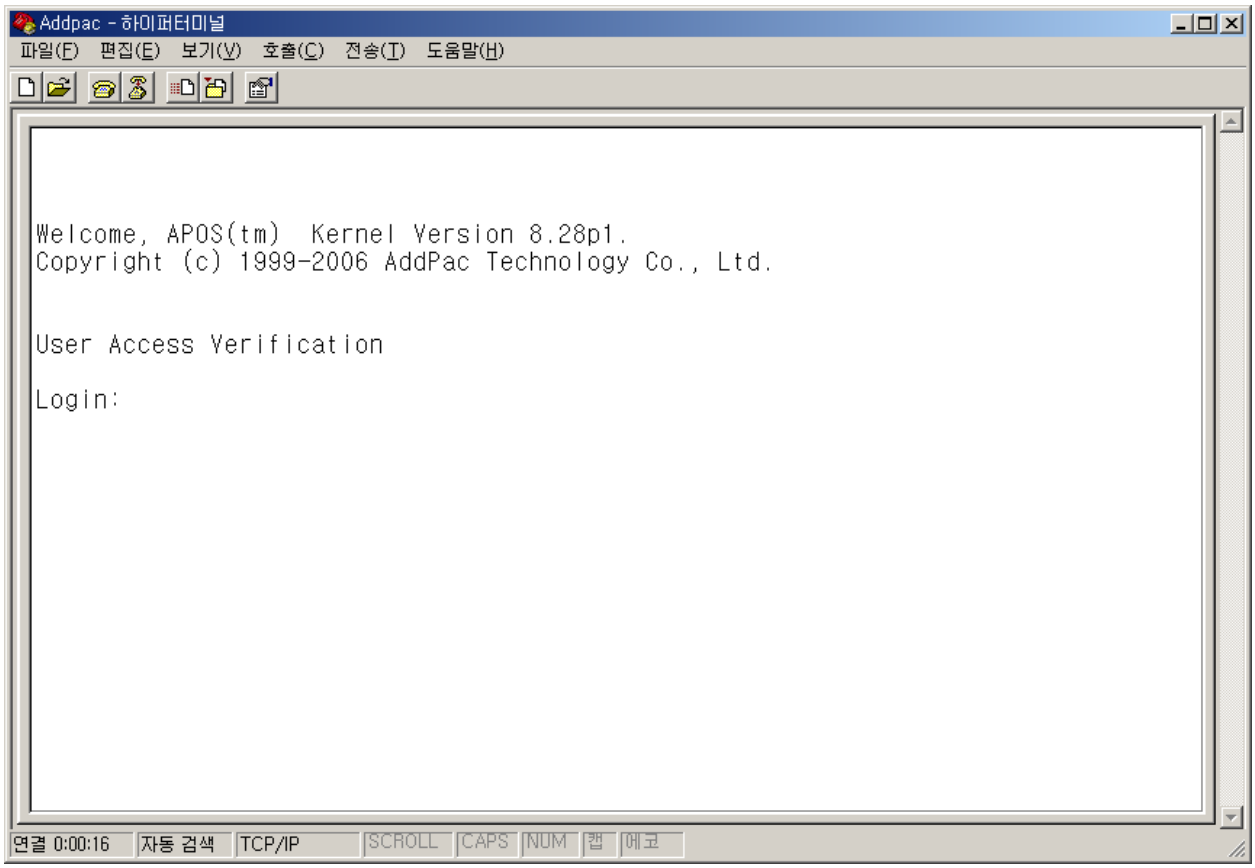
Since AP-IP230 does not support Console Interface, the IP Address of LAN1 interface is used to connect PC as it is shown in the following figure: (LAN1 default ip address : 192.168.10.1)

Access after the IP address of 192.168.10.100 for the PC is set.



(Picture 5-3) Access to Telnet Using TCP/IP

After completing the setup, you can see the login screen of AP-IP230 in the following Figure



(Picture 5-4) AP-IP230 IP Phone login Screen

When login message is displayed at the end, enter login basic value “root” and password value “router” to complete login. After login procedure is completed, prompt “IP230>” will be displayed in console terminal.

APOS Commands

NOTE	All the product lines of AddPac Technology are imbedded with APOS AddPac Operating System). Therefore, all the basic settings of CLI (Command Line Interface) are same.

The commands are used for the following types of modes:
User Mode: placing limitations on the system or providing an access for the data communication
Management Mode: checking the status of the system configuration or the debugging functions of the system
Configuration Mode: changing the settings or creating new settings

• You do not have to enter the entire command. Entering partial command is acceptable as to enter just ‘sh’ or ‘sho’ and it is recognized as ‘show’ automatically.

• If you made an error of entering the system commands, on-line help function provides the list all the possible commands.

• More function provides the additional screen to display all the remaining messages which are missed out from one screen.

• All the possible commands and their descriptions are executed in that particular mode by providing Help and ‘?’ functions.
History provides a list commands which have been used previously. By using the number of the prompt, you can enter the commands easily when you need to reenter them.

• The structure of the system commands are divided into 3 types of modes and the commands used are different to each other. The commands used for the each mode is described in the followings.

General User Mode Commands

These are the functions that all the types of the users who logged in the system
The prompt for the general user can be indicated as 'IP-230>'.

[Table 5-1] General User Mode Command

Command	Description
enable	Change to the administrator mode
exit	Move to the lower case from the current
help	Display the list of APOS help
quit	Same as exit
show	See the status of the system operation and configuration
terminal	Determine the number of lines to be executed on the terminal at once
who	Indicate the user accessing vtv
whoami	Indicate the current status of connection

Management Mode Command

These are the types of commands that the administrator, whoever logged in the system, can use. To get into configuration of this system mode, the user must be logged in as an administrator. In this mode, the commands can be same as General User mode such as 'show', but more information can be shown depending on the option.
The prompt of Management Mode can be indicates as 'IP230#'.

[Table 5-2] Manager Mode Command

Command	Description
auto-upgrade	Configure to upgrade the image by using Http
clear	Reset the interface counter, statistics
clock	Set the present time, date, year
configure	Move to configuration mode
copy	Copy running config to startup conifg
debug	Debug the overall system
disable	Mover to General User Mode
disconnect	Close VTY connection
dnsquery	Test DNS Query
dnsrv	Test DNS SRV Record
end	Move to Management Mode
erase	Delete config file
exit	Move to the last mode
fsh	Get into File Shell
help	Indicate APOS help
no	Delete the present setup
nsupdate	Transmit the updated information to Name Server
ntpdate	Bring the time from ntp server
ping	Test network connection
quit	Move to the last mode
reboot	Reboot the system
show	Show the status of the system operation and the status of configuration
terminal	Set the number of lines to be terminated at once
tftp	Transmit a file to tftp server

traceroute	Test the path for IPv4 routing
who	Indicate the users connected to vty
whoami	Indicate a type of connection established at the present time
write	Save the configuration in operation process

Basic Configuration

Password Configuration

After a connection is established to the console, the user can only have the basic show command. To gain more privilege to access, the user has to enter enable mode. When the general use enter enable mode, he/she gains the all the privilege to change the system configuration. Therefore it is important to set the password, so only the administrator can enter to configure the settings.

[Table 5-3] Password Configuration

```
AP-IP230# configure terminal
AP-IP230(config)#
AP-IP230(config)# enable password {password}
AP-IP230(config)#
```

Host Name Configuration

When the user is connected to telnet or console, he/she can change a name of prompt in the setting of CLI. Naming the host becomes more important when many devices are connected to telnet to be administered. It would be more convenient to use the words representing location as a name.

[Table 5-4] Host Name Configuration

```
AP-IP230# configure terminal
AP-IP230(config)#
AP-IP230(config)# hostname {name}
AP-IP230(config)#
```

User Administration

The user account is used for connecting telnet, FTP, Samba.

The user account and password must be known to the administrator only. If they are exposed to any other, the product cannot be operated properly.

[Table 5-5] User Administration

AP-IP230# **configure terminal** AP-IP230(config)# AP-IP230(config)# **username {ID} password {password} {administrator | operator | user}** AP-IP230(config)#

FXS/FXO Port Configuration

* Check show run first

[Table 5-6] FXS/FXO Port Configuration

```
IP230# show run

Building configuration...
Current configuration:
version 8_42_003
hostname IP230
!
username root password router administrator
!
interface Loopback0
 ip address 127.0.0.1 255.0.0.0
!
interface FastEthernet0/0
 ip address 172.17.201.88 255.255.0.0
 ip nat outside
 speed auto
 no qos-control
!
interface FastEthernet0/1
 ip address 192.168.10.1 255.255.255.0
 ip nat inside
 speed auto
 no qos-control

--- omit -----
!
! Voice port configuration.
!
! SPEECH
voice-port 0/0
!
! FXS                    => Display as FXO when FXS exist vice versa
```

```
voice-port 0/1              => Check FXS/FXO voice-port ( 0/1 port )
!
! Pots peer configuration.
!
dial-peer voice 0 pots
 destination-pattern 1004
 port 0/0
!

IP230#
IP230# con t
IP230(config)# dial-peer voice 1 pots                    =>Setup FXS/FXO port dial peer
IP230(config-dialpeer-pots-1)# destination-pattern 1014   => Setup FXS/FXO port number
IP230(config-dialpeer-pots-1)# port 0/1
IP230# show run   (Check setup value)

Building configuration...
Current configuration:
version 8_42_003
hostname IP230
!
username root password router administrator
!
interface Loopback0
 ip address 127.0.0.1 255.0.0.0
!
interface FastEthernet0/0
 ip address 172.17.201.88 255.255.0.0
 ip nat outside
 speed auto
 no qos-control
!
interface FastEthernet0/1
 ip address 192.168.10.1 255.255.255.0
 ip nat inside
 speed auto
 no qos-control

--- omit -----
```

```
!
! Voice port configuration.
!
! SPEECH
voice-port 0/0
!
! FXS
voice-port 0/1
!
! Pots peer configuration.
!
dial-peer voice 0 pots
 destination-pattern 1004
 port 0/0
!
dial-peer voice 1
 destination-pattern 1014
 port 0/1
```

Chapter 6. AP-IP230 Recovery

The entire AddPac VoiIP product line has 2 different zones. One is to store APOS and Boot Loader is the other. The functions of Boot Loader can be used in the followings:

1. Loss of the password for the root account
2. Damage or erase of the software in APOS image

You can recover the Default IP by resetting APOS settings in case you lost or change the Default IP(192.168.10.1) AP-IP230 which can be accessed by TELNET, FTP. For damaged or erased APOS image can be recovered and used normally again by downloading the image at the mode of Boot Loader.

NOTE	Boot Loader of the IP Phone does not have IP routing function. Therefore, PC and LAN1 of the IP230 which are used for accessing by TELNET/FTP must be connected directly.

Entering the Boot Loader Mode

Since AP-IP230 does not have console interface, it is not possible to enter the mode of Boot Loader by using ‘ctrl+x’, ctrl+c’ , which is possible for APOS with presence of console such as ‘send break’., during the booting process.
During the booting process, AP-IP230 checks the basic operation of CPU, memory and interface. Then it waits for about 3 seconds for the user to make an access. In this status, you can see the LED on the front side is beginning to be turned on one after another
While LAN1 interface of AP-IP230 and PC are connected directly to each other, the user can access to AP-IP230 when the LED is turned on one after another.
In general, TELNET is used for an access to check the password or resetting the APOS settings. To download APOS image, the user can access to FTP server (to get into the mode of Boot Loader, enter ‘root’ is for the ID and ‘router’ for the password.

APOS Configuration Initialization

When the user lost the default IP address of the IP Phone (192.168.10.1) that enables TELNET and FTP access, after making a change, the default IP can be recovered by initializing APOS settings (Please be cautious when you initializes APOS configuration, all the existing settings of configuration are to be erased.)

You can initialize APOS settings by TELNET access.

```
D:\>
D:\> telnet 192.168.10.1

Welcome,  APOS™ Boot Kernel Version 5.0.10.
Copyright (c) 1999-2005 AddPac Technology Co., Ltd.

User Access Verification

Login: root
Password:
Booter>
Booter> enable
Booter#
Booter # erase apos-config
Do you want to ERASE configuration ? [y|n] y
Erasing configuration....done
Booter#
```

Downloading APOS Image File in Boot Loader Mode

The AddPac AP-IP230 IP Phone allows FTP access, which is supported by the binary code, to transmit APOS image file.

APOS image of AP-IP230 can be downloaded from PC by using FTP.

```
D:\ >dir
2006-05-15   05:21p       <DIR>              .
2006-05-15   05:21p       <DIR>              ..
2006-05-15   05:21p             1,775,360        AP-IP230_g2_v8_41_015.bin

D:\>
D:\> ftp 172.17.201.88
Connected to 172.17.201.88.
220 IP230 FTP server (Version 8_42_003) ready.
User (172.17.201.88:(none)): root
331 Password required for root.
Password:
230 User root logged in ok.
ftp>
ftp> bin
200 Type set to I.
ftp>
ftp> put    AP-IP230_g2_v8_41_015.bin
200 PORT command successful.
150 Opening BINARY mode data connection for ' AP-IP230_g2_v8_41_015.bin '.
226 Transfer complete.
ftp> bye
221 Goodbye.
D:\>
```

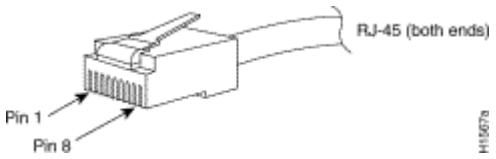
Chapter 7. Appendix

This Appendix provides information about the Pinout specifications of the following cables used with AP-IP230 IP Phone.

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

[UTP Cable (RJ-45 to RJ-45) Pin Out]

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.



(Picture 7-1) 100Base-TX RJ-45 Connector

[Table 7-1] Serial Ethernet Cable Signal and Pin Out

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.

Abbreviation and Terminology

Terms	Definition and 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	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 nonvarying, 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 twowire 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 endpoint 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 dialup 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 highspeed

	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, APGK3000 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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