AP-IP300 IP Phone

[Installation and Operation Guide]

November, 2007

AddPac **Technology**

www.addpac.com

AP-IP300 IP Phone

Note.

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Getting into AP-IP100 IP Phone Installation Guide

This chapter explains the AP-IP300 IP phone installation guide.

[Contents of AP-IP300 Installation Guide]

The purpose of this guide is to assist the users to install the AP-IP200 IP Phone easily. This guide is composed of six chapters as to follow.

If you have a previous experience of using IP Phone, please refer to the chapters the user wants to know directly. But, if you have no experience of using IP Phone, it is highly recommended to thoroughly understand the manual before operation of this IP Phone.

- Chapter 1 『**Overview**』 provides an introduction to the hardware and software features of AP-IP300 and technical specification.

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

- Chapter 3 『**Installing**』 This chapter explains the procedures for installing the gateway. Installation involves the tasks of connecting cables, console to AP-IP100 IP Phone and other basic information for the installation process.

- Chapter 4 『**How to Use AP-IP300**』 describes the UI operation of AP-IP300. 「UI stands for 'User Interface', allows the user to change device settings through the screen.」

- Chapter 5 『Appendix』 provides the detailed cable specifications for AP-IP300 IP phone.

If you have any questions and need any technical support of AP-IP300, please contact AddPac Technology in the followings:

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The revision history of AP-IP300 IP Phone installation guide is listed as to follow:

Revision History of AP-IP300 IP Phone Installation Guide

Revision No.	Date	Contents	Written By
Version 1.00	July 25th, 2005	Initial Release	AddPac R&D Center

Chapter 1. Introduction

Introduction

AP-IP300 IP phone is designed to provide enhanced IP telephony functionality to meet the wide range of business user requirements. This IP telephone optimally delivers rich featured voice telephony service on ordinary internet infrastructure as well as AddPac IP-PBX environment on local LAN as a fully featured IP extension for the complete AddPac VoIP solution.

Product Overview

1. Emerging in a New Era of IP Communications

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 large LCD screen display with high resolution and color graphics, wide variety of feature keys, customizable hot-keys, two(2) Ethernet ports, the latest QoS, public IP sharing. It supports not only the major VoIP signaling protocols such as SIP, H.323, MGCP but also G.711, G.726 voice codec, stereo audio in/out interfaces for external Headset MIC. Etc..

2. New Paradigm for IP Telephony : Telephony + Broadcasting

AP-IP300 IP telephone combines AddPac's field proven VoIP technology and IP voice broadcasting technology. AP-IP300 is market-ready IP telephone which provides a full suit of remarkable functionality compared to other typical IP telephones. Apart from telephony service, it delivers IP voice broadcasting service supporting external MIC/Line-in, Line-out interface for various input/output devices such as headset, Amp or speaker. In addition, it provides high quality display with blue color LCD mounted. Since AP-IP300 supports diverse voice codecs according to bandwidth environment, it can be deployed anywhere on the internet, ensuring optimal voice quality by leveraging the latest QoS technology. Furthermore, installed along with IPNext500 and IPNext1000 on AddPac's comprehensive IP-PBX system, it not only improves operation offering an wide variety of features such as Music on Hold, Coloring service, Call Transfer but also provides the easy-to-use, intelligent IP telephony service enhanced by AddPac's unique PC-based User Agent.

3. Firmware Upgradeable Technology

Designed on programmable high performance RISC CPU and DSP, AP-IP300 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.

4. Compelling Supplementary Services : Extending Benefit of IP Telephony

AP-IP300 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-IP300 to offer voice broadcasting service, incorporated with in-house broadcasting system.

5. Seamless Stability and Service Consistency

AP-IP300 features 1-FXO port(optional) equipped avoiding operation failure caused by network error or proxy server/gatekeeper connection error. It supports both automatic and manual PSTN backup feature to maintain constant operation.

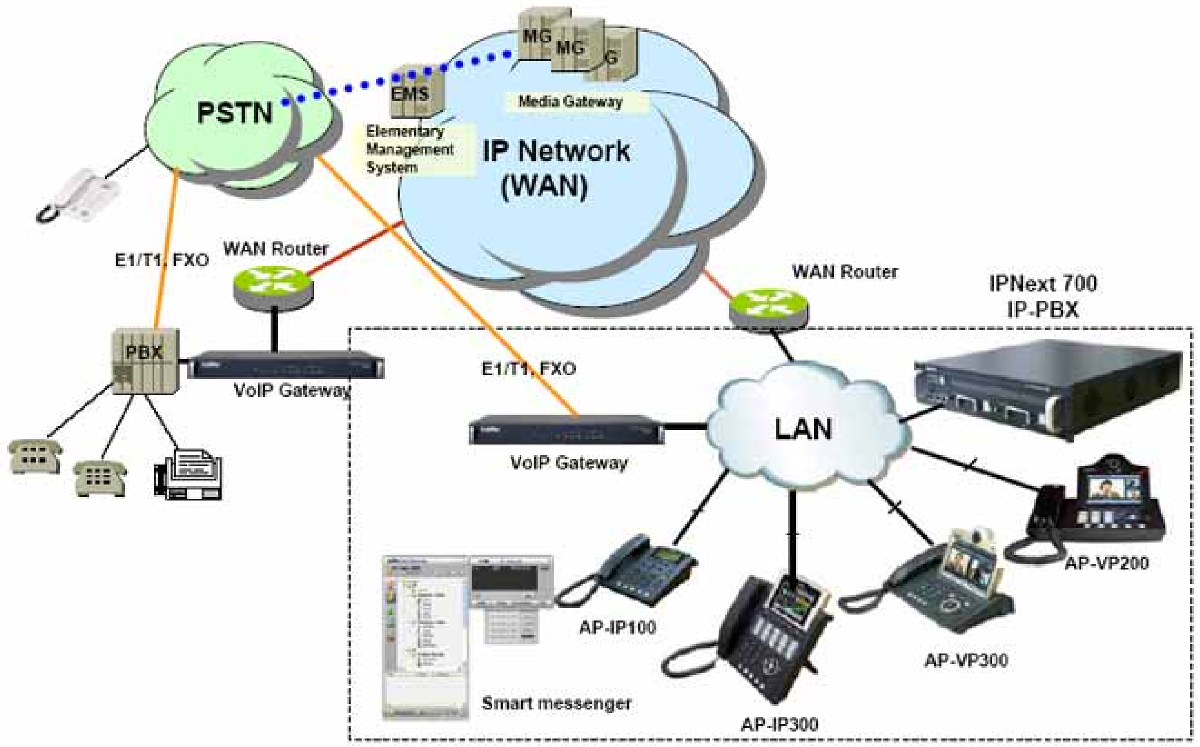
6. IP telephony with Outstanding Network Service Capability

Not only IP telephony, AP-IP300 is an integrated, feature-rich network equipment delivering routing, NAT/PAT, DHCP Server/Relay, Public IP sharing, VRRP and QoS. In today's mixed network of xDSL, Cable, FTTH, Metro Ethernet, Metro ATM, Leased line and dynamic IP environment, not only the ample network service features, but also high-end QoS (Quality of Service) and security features are requested. Based on two (2) 10/100Mbps Fast Ethernet ports, AP-IP300 offers integrated network and security service of LAN-to-LAN routing, bridge and NAT/PAT. Moreover, AP-IP300 supports H.323, SIP, MGCP signaling protocols concurrently. So the customers easily migrate to different service providers' networks utilizing different VoIP signaling protocols.

6. Privacy and Encryption Features

AP-IP300 brings the network security and service security as well. With the built-in CID (Caller ID Detection) feature, user is able to know who is calling before he answers and block the incoming call. Moreover, It supports SRTP protocol by encrypting exposed voice signal to avoid being fragile to hacking or wiretapping

AddPac's various VoIP gateway series, multi service routers and comprehensive family of cutting-edge solutions have delivered high performance and stability to maximize customer satisfaction throughout the world. They provide high level of flexibility and scalability for each organization to find the solution that best fits their application needs and budget. With years of experience and industry-leading technology, AddPac provides AP-IP300 with which customers can best optimize high performance, market strategy and budget for next-generation communication solution.



(Figure 1-1) Network Diagram of AP-IP300 IP Phone

AP-IP300 IP Phone Hardware Specification

[Table 1-1] List of Hardware Specification

Category		Specification
Model		**AP-IP300**
Product Category		IP Phone (Built-in Speaker Phone)
Microprocessor		High Performance RISC CPU Architecture
Digit and Key Buttopns		3 x 4 Standard Numeric Buttons, 17 Menu/ Function Key Speed Dialing Keys, 25 Speed-Dial and Presence Indication Keys
LCD Display	Graphic LCD	4.3” Color LCD
Memory	Boot Memory	512Kbyte Flash Memory
	Flash Memory	4/8Mbyte
	Main Memory	64Mbyte High Speed SDRAM
Audio Interface	Input	One(1)-3.5mm Stereo-In Connector for Audio In
	Output	One(1)-3.5mm Stereo-Out Connector for external speaker
Ethernet Interface	LAN0 Port	One(1) 10/100Mbps Fast Ethernet
	LAN1 Port	One(1) 10/100Mbps Fast Ethernet
Power Requirement	Poer	External Power VAC 110~220 VAC, 50/60Hz, 15Watt
Hardware Chassis	Composite, Material	ABS Material/Compact Phone Chassis

AP-IP300 IP Phone Software Specification

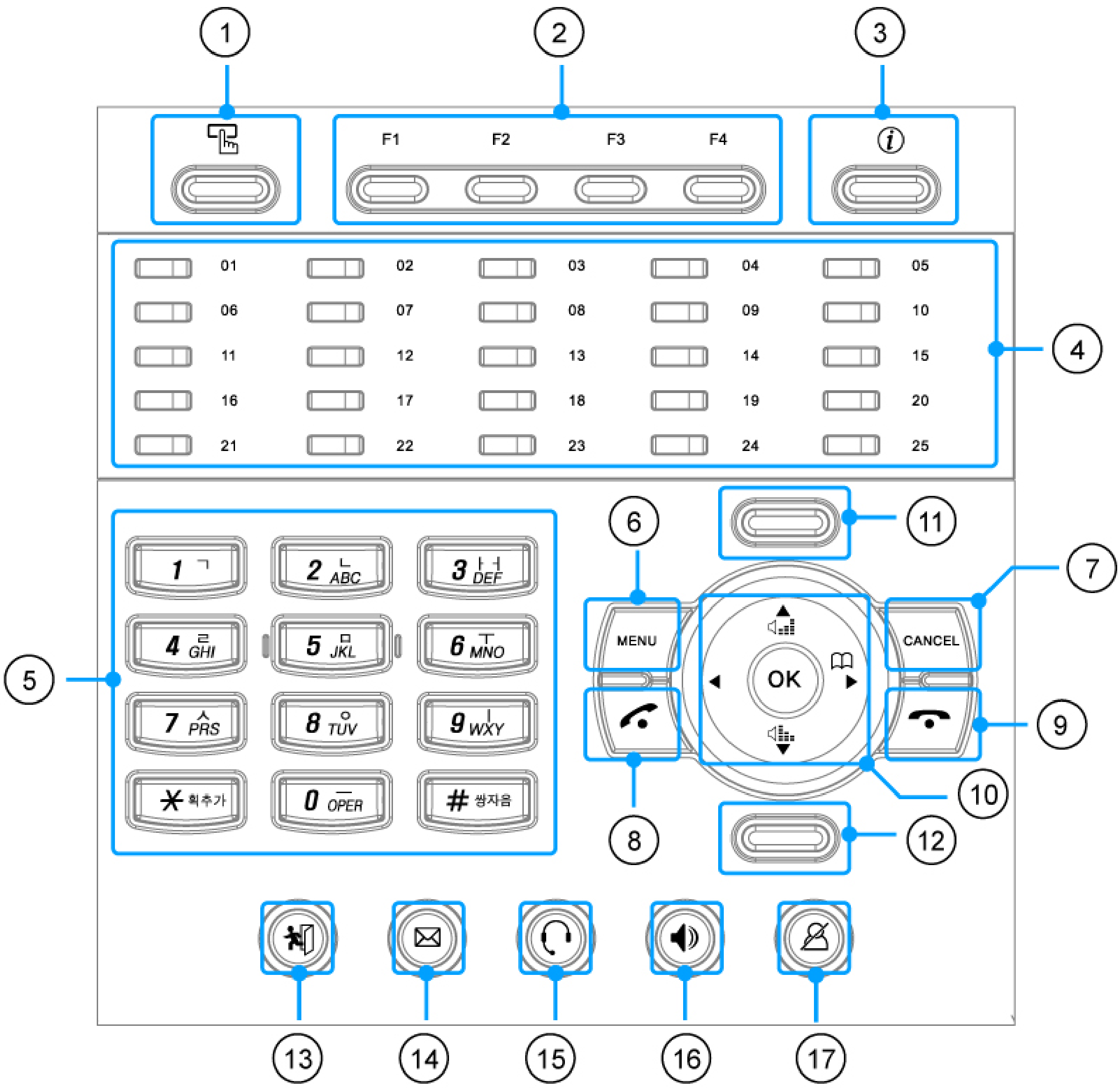
[Table 1-2] AP-IP300 Software Specification

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 IP300, 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

The Upper View

This chapter explains the front part's DIAL and FUNCTION KEY of AP-IP300 IP Phone. The external case is made of high degree of solidity ABS. Main key buttons are equipped on front part so that user can operate all the functions with these buttons.



(Figure 1-2) The Upper View for the Key Arrangement of AP-IP300

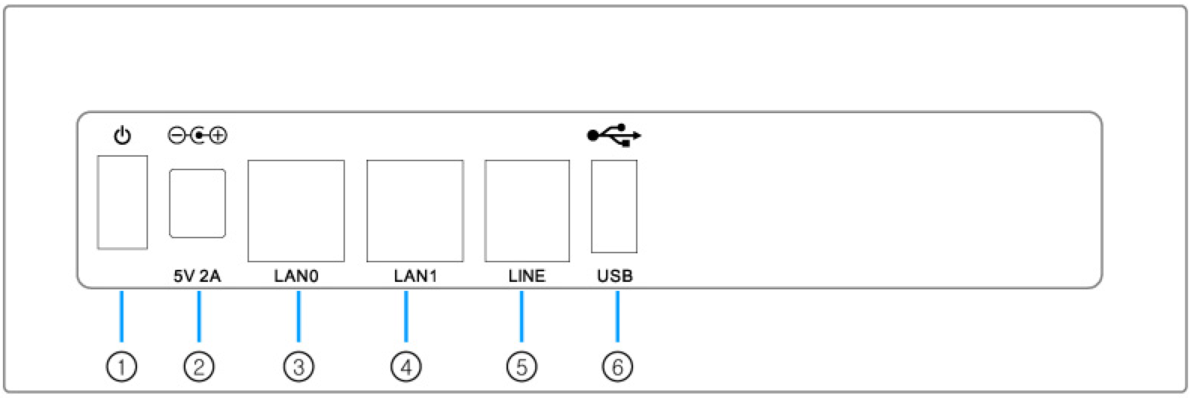
The following Table 1-3explains each button feature of AP-IP300 on the front side.

[Table 1-3] Description of the Button Features on the Front Side

No.	Button Names	Features
(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)	25 Speed Dialing	25 buttons for Speed Dialing and Presence Indication
(5)	Numeric Key	Used for Dialing and parameter setting in UI
(6)	Menu	Enter the UI Main Menu
(7)	Cancel	Move on to upper menu from current UI menu or cancel the current VoIP call
(8)	Call	Brings out the list of the recent calls Press to make a call after dialing
(9)	END	Ends the present call in progress
(10)	Navigation Key, OK	Moves the direction in each UI menu
(11)		Not used
(12)		Not used
(13)	Absence	Used at Absence Mode
(14)	Voice Mail	Used at Voice Mail Mode
(15)	HDP Call	This KEY is used for VoIP call via Headphone Interface
(16)	SPK Call	The key is used for VoIP call via speaker phone. If this button is pressed, blue LAMP is turn on.
(17)	Privacy	Used for MUTE at conversation

The Layout of the Rear Side

The rear side is composed of FXO PSTN backup interface, USB interface, power switch and connector and two (2) Fast Ethernet for WAN/LAN connections.



(Figure 1-3) The Rear Side

[Table 1-4] explains the interfaces on the rear side.

[1-4] Description of the Interfaces on the Rear Side

No.	Interface	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. Preparing for Installation

Installation Requirement

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

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

Electrical Requirement

There are two main sources of electrical problems with AP-IP300 IP Phone : the power supply and static electricity.

This section describes safety recommendations for each case.

- **Electrical Safety**

  ✓ In case of the occurrence of an electrical accident, operate at a position where immediate shut-off of power supply is possible.

  ✓ Switch the power off when installing or taking the cover off the equipment.

  ✓ Avoid operating the equipment alone at a potentially dangerous environment.

  ✓ Do not assume the power is switched off, but always confirm the power status.

  ✓ Be extremely cautious when operating in humidity or with an uncovered power extension cable.

- **Prevention of Static Electricity**

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

General Requirement

The AP-IP300 IP Phone is ready for use where electronic products are used. However, locations with the following conditions are recommended for maximum performance:

- A level and well ventilated location is recommended.

- Secure the equipment safely where intended to install.

- Avoid placing objects on top of the equipment.

- Install the equipment in a cool location avoiding direct sunlight.

- Maintain distance from flammable, chemical, or magnetic objects

Prerequisites for Installation

The user should consider the EMI standards and distance limitations (EIA recommendation) when installing the AP-IP300 IP Phone.
The following section describes the Ethernet cable and the RS-232C console cable AP-IP300 supports.

Prerequisites for Installation

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

Cable for LAN and Console port connection

RJ-45 to RJ-45 cable for LAN port (included in equipment packing box)

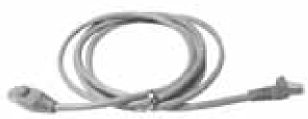
Ethernet port

AP-IP300 IP Phone has one RJ-45 type RS-232C connector interface in rear side. It can be used for AP-IP200 initial configuration, equipment monitoring and debugging. You must use a cable and a connector. Refer to cable specification in Appendix on RS-232C console cable pin specifications.

Unpacking and Verifying the Contents

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

[Table 2-1] The contents of AP-IP300 IP Phone in the Package Box

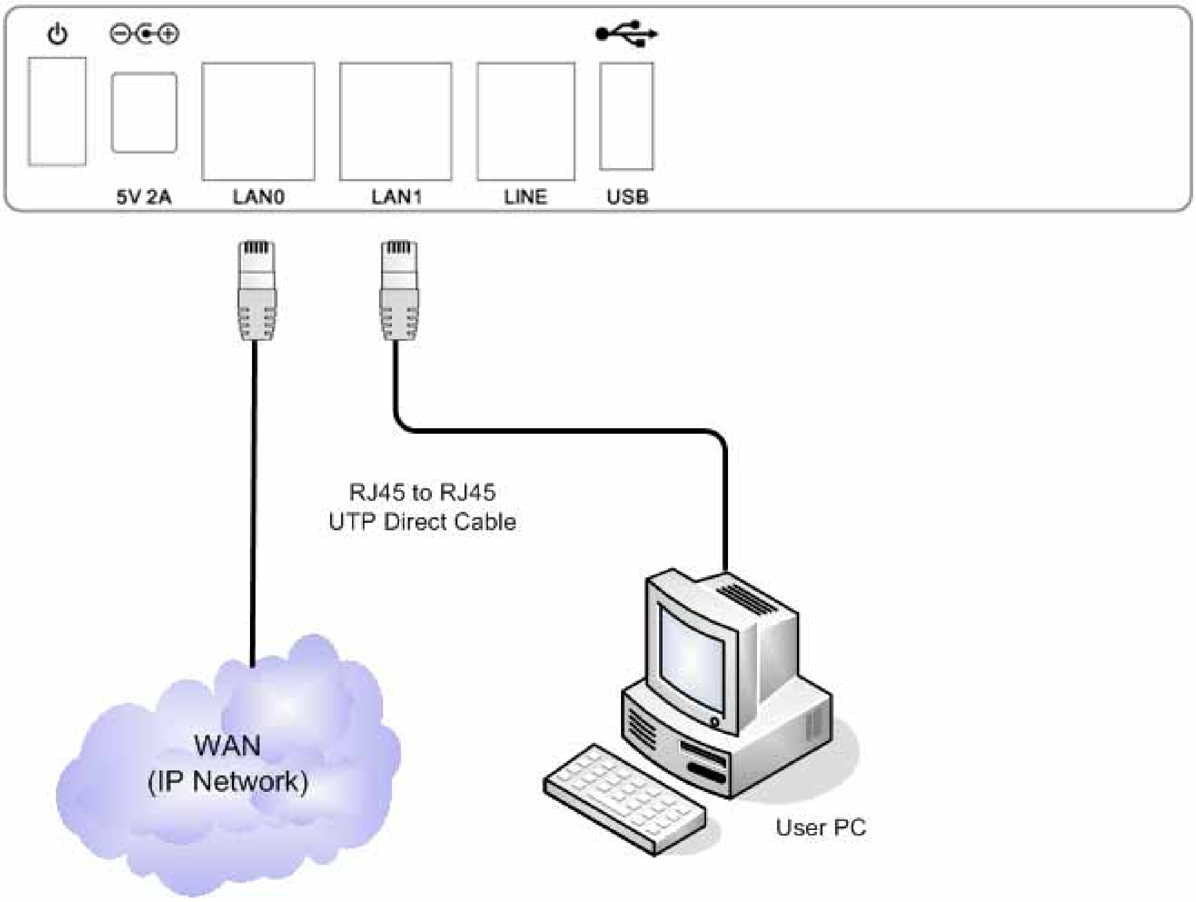
No	Items	Contents	Qunatity
1	AP-IP300 IP Phone Main Body		1
2	LAN cable (RJ45 to RJ45)		1
3	External Power Adaptor (220V Power Cord)		1

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

Chapter 3. Installation

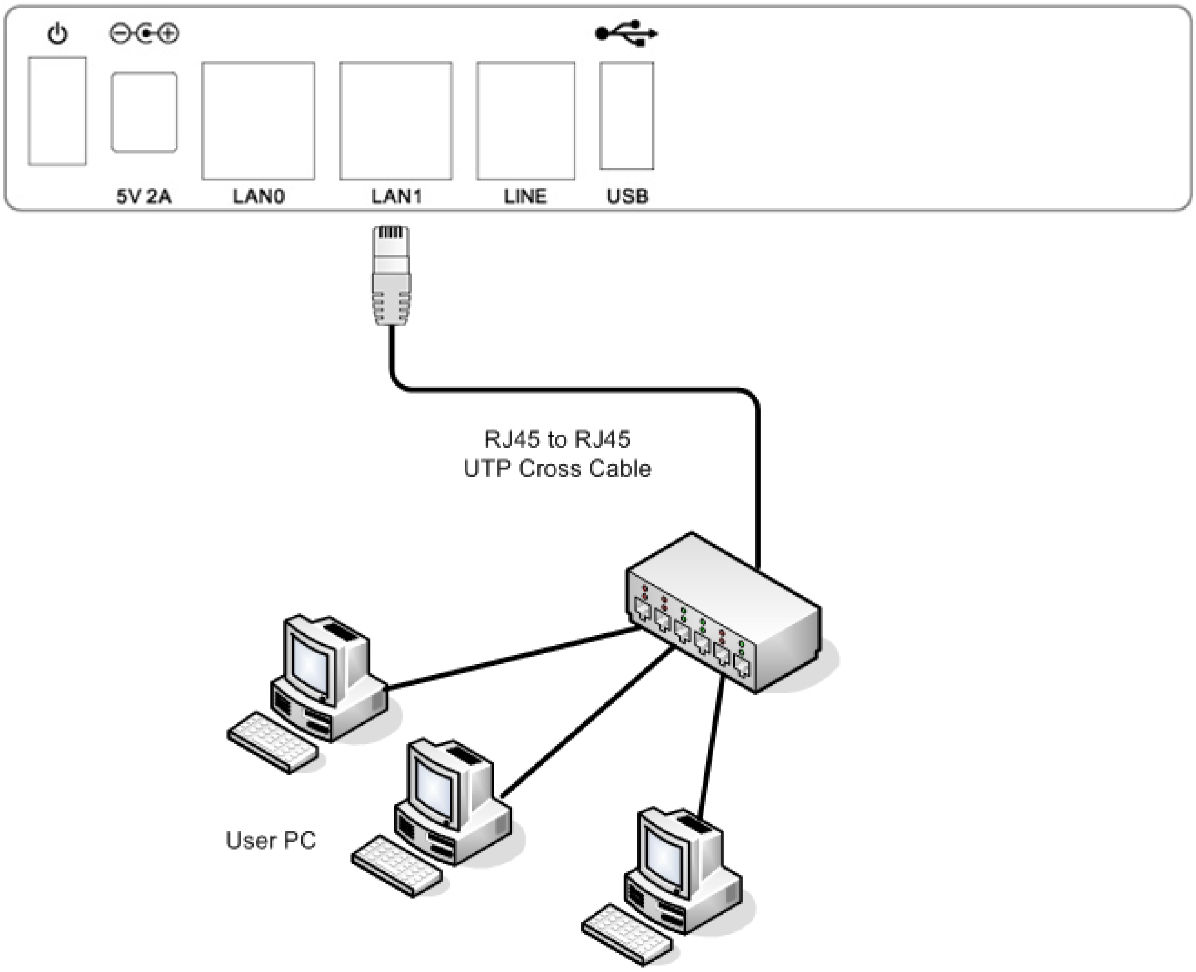
Connecting Ethernet Interface

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



(Figure 3-1) Connecting WAN Interface

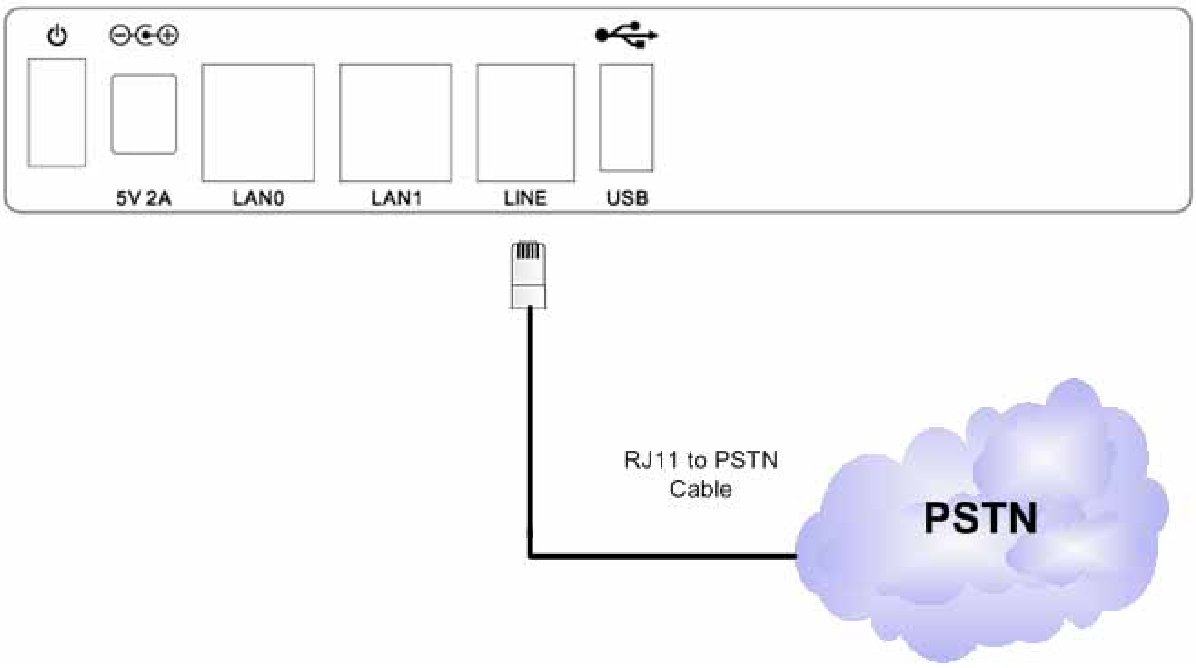
- AP-IP300 IP Phone's Fast Ethernet PC Interface is supposed to be connected into Desktop PC's LAN Port with Direct-Through cable in IP-Share mode and to be connected into HUB in NAT/PAT or Bridge mode.
- In case of connecting directly to Desktop PC's LAN Port, please use Direct-Through cable.
- In case of connecting directly to HUB, please use Cross-over cable.



(Figure 3-2) Connecting LAN Interface

Connecting PSTN (FXO) interface

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

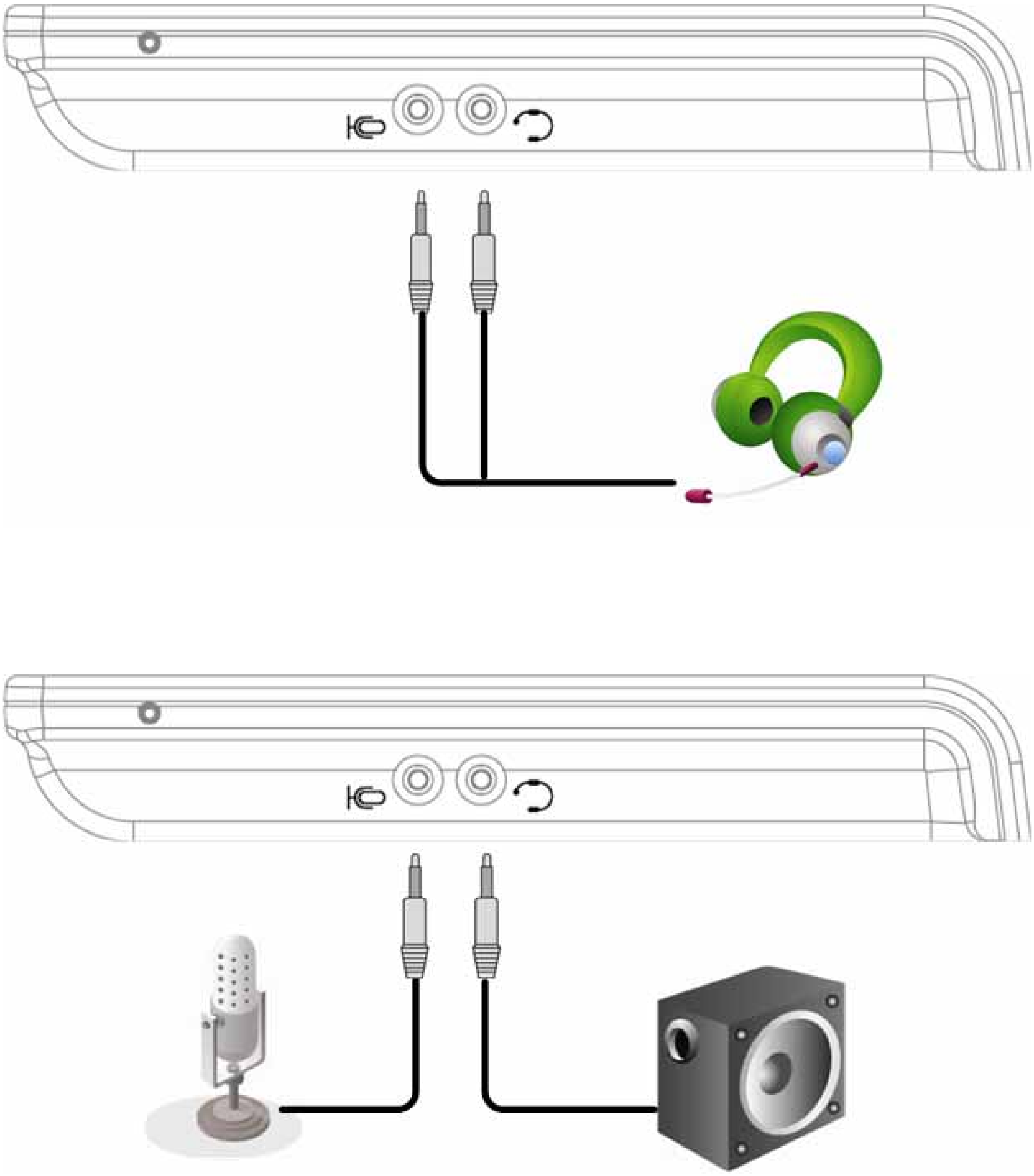


(Figure 3-3) AP-IP300 IP Phone PSTN Interface Connection

Connecting Audio-In/Out Interface for Headset

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.

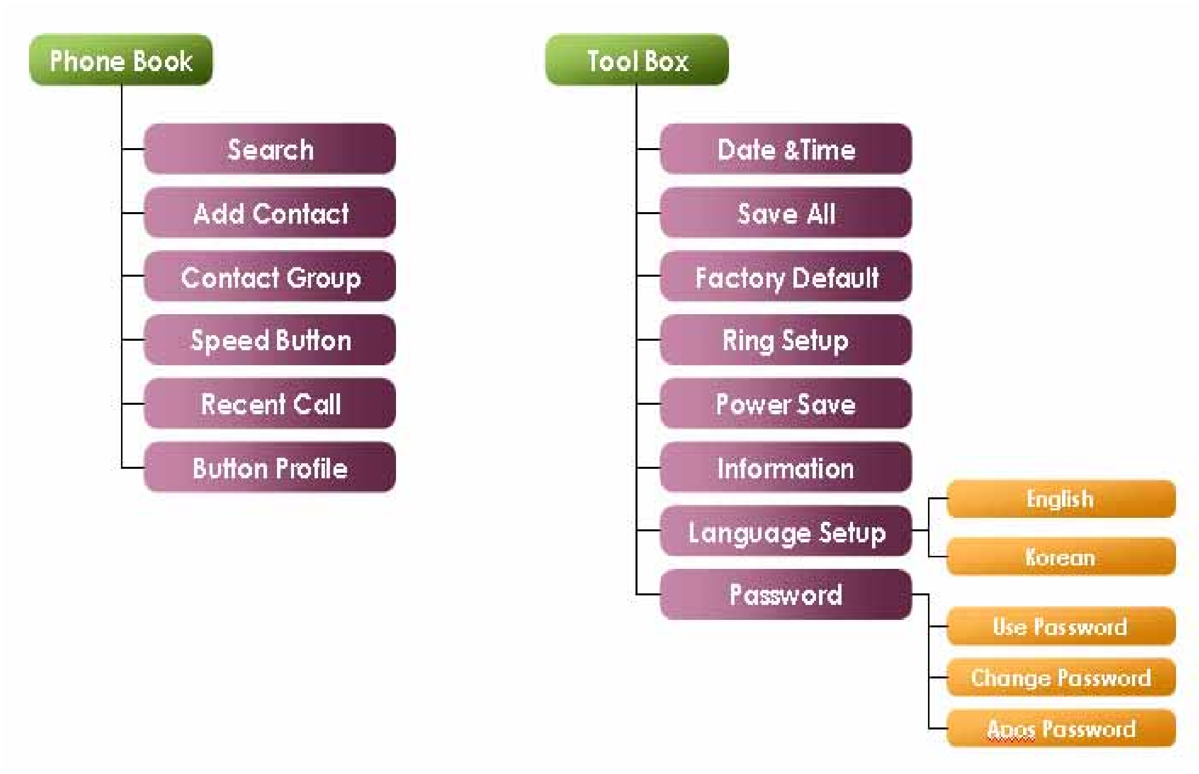
Connect this port to MIC system or External Speaker System using '3.5mm stereo jack' cable.

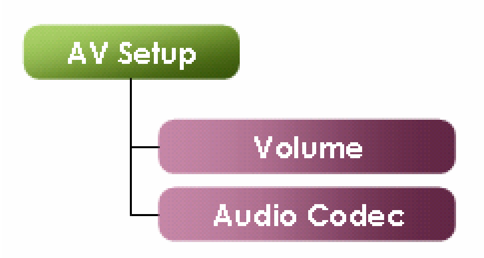


(Figure 3-4) External Audio IN/OUT Interface Diagram of AP-IP300 IP Phone

Structural Diagram of User Interface Menu

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.

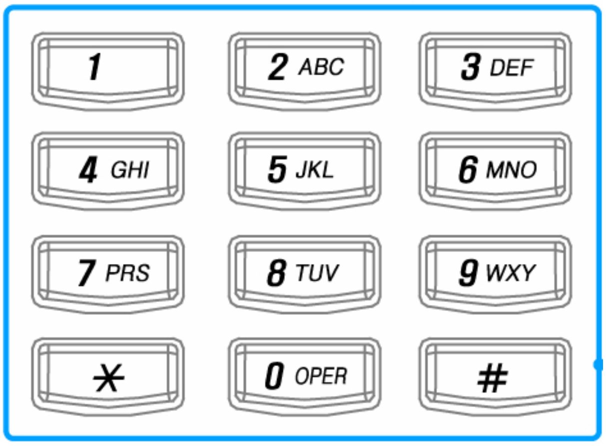




(Figure 3-5) Structural Diagram of UI Menu

Using the Dial Pad Buttons

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



(Figure 3-6) Dial Pad Buttons

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

Dial Pad Buttons	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	BackSpace
F2	Space	Space

**** F2(Space) Function:** 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

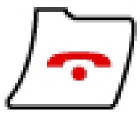
The functions of Call button is described in Table 3-2 below:

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

Functions	Descriptions
Retrieving the Recent Incoming Call	When you just 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 a 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.
Taking a Call	After all the settings are entered to apply, 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)

The END button is used for the purposes described in [Table 3-3].

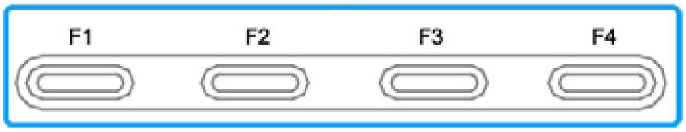


[Table 3-3] Using End Button

Functions	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 Softkeys Supported by SSCP

Softkey functions are supported by SSCP and change depending on the status of the phone (for example, when you are on a call or the phone is not in use) which are shown at the bottom of the LCD screen and they are interconnected with the 4 buttons of the softkey. When more than 4 softkeys present, you can see more softkeys by pressing F4 ('More') in the next screen.



(Figure 3-7) Layout of Softkeys

[Table 3-4] When the Phone is on the Hook

No.	Function	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	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] When the Phone is off the Hook

No.	Function	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)	Ends a call

[Table 3-6] When You are Busy on Line

No	Function	Description
1	Hold	Places a call on hold
2	EndC (End Call)	Ends a 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.
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] While a Call is Placed on Hold

No	Function	Description
1	Resu (Resume)	Returns on a call from hold status
2	NewC (New Call)	Connects to a new phone call
3	Tran(Transfer)	Transfers a call

[Take 3-8] When the Phone Rings

No	Function	Description
1	Answ (Answer)	Takes an incoming call

[Table 3-9] When the Phone Rings

No	Function	Description
1	EndC (End Call)	Ends an outgoing call

[Table 3-10] On Voice Mail Screen

No	Function	Description
1	EndC (End Call)	Disconnects Voice Mail

[Table 3-11] While a Call is Being Transferred

No	Function	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

No	Function	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] Conference Conference Host

No	Function	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 4-prty), VP350MCU(video 4-party), VC2000(video 4-party), MC1000(video 16-party)

[Table 3-14] Conference Participants

No	Function	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 for Looking up the Configured Settings

```
IP300# show run
Building configuration...

Current configuration:
!
hostname IP300
!
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--
```

* Configuring IP Addresses and Default Route Settings

```
IP300# configure terminal
IP300(config)#
IP300(config)# interface FastEthernet 0/0    → Fast Ethernet Interface 0 Port
IP300(config-if)# ip address 172.20.103.1 255.255.0.0    → IP address setting
IP300(config-if)# VOIP_INTERFACE_DOWN
VOIP_INTERFACE_DOWN
VOIP_INTERFACE_UP : (172.20.103.1)

IP300(config-if)# exit
IP300(config)#
IP300(config)# ip route 0.0.0.0 0.0.0.0 172.20.1.1    → default router
IP300(config)#
IP300(config)# end                        → end of configuration
IP300#
IP300# write                              → saving the setting
Proceed with write? [confirm]y            → confirm
Building configuration...
[OK] Configuration saved to flash:/apos.cfg
IP300#
```

* After the network configuration is finished, you may perform the ping test from AP-IP300 to the default router

```
IP300# 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
IP300#
```

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

Chapter 4. Using AP-IP300

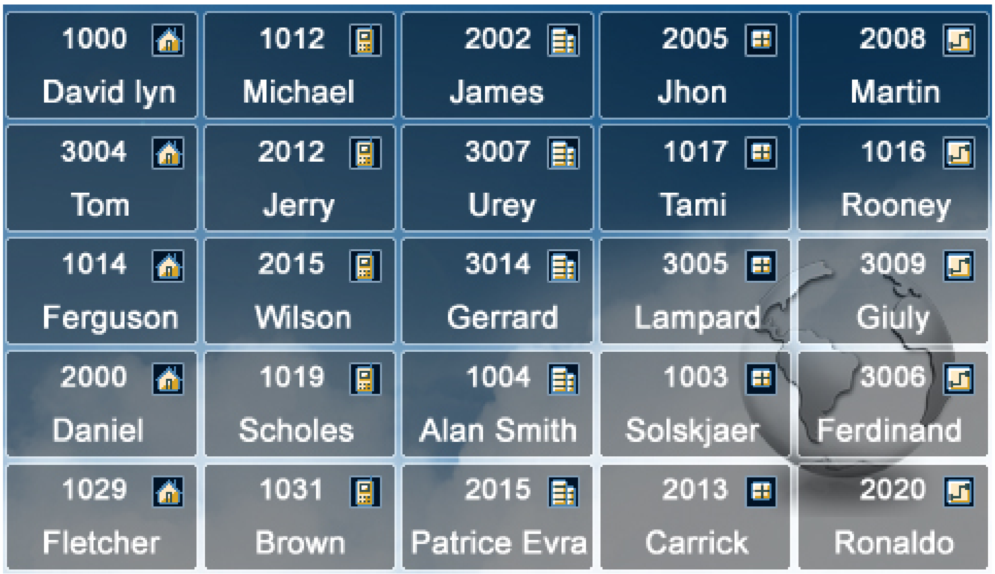
Default Screen

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

(Figure 4-1) Default Screen

[Table 4-1] Description of the Default Screen

No.	Description	
①	Date and Time	Display the present date & time.When you are on a call, it displays the real “connection time” (SSCP takes the clock souce from AddPac IP-PBX 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)



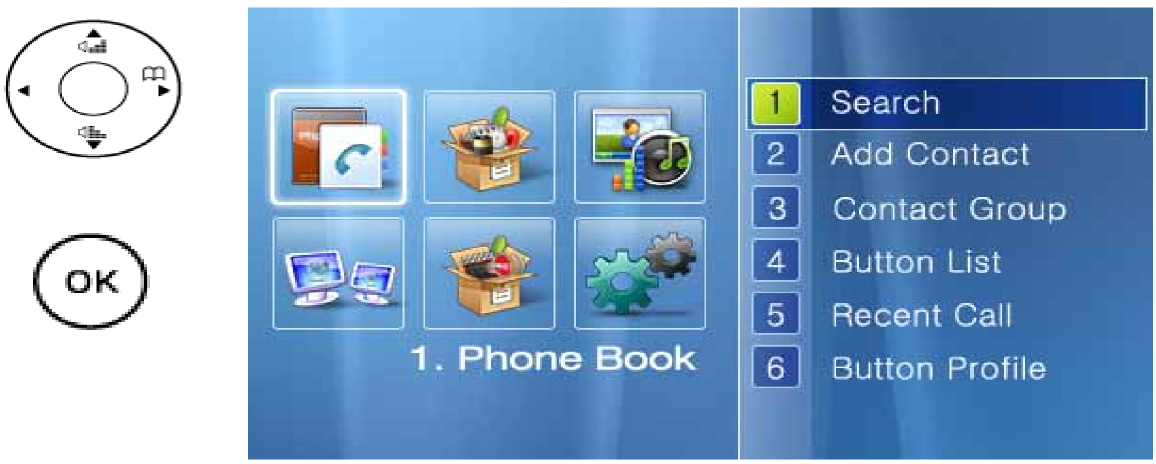
(Figure 4-2) Basic Screen Layout

< 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. >

Phone Book 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.

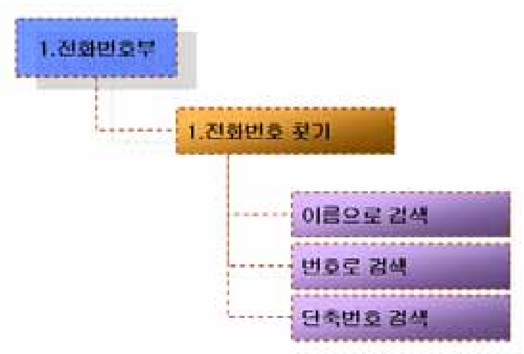
(Figure 4-3) Main Screen

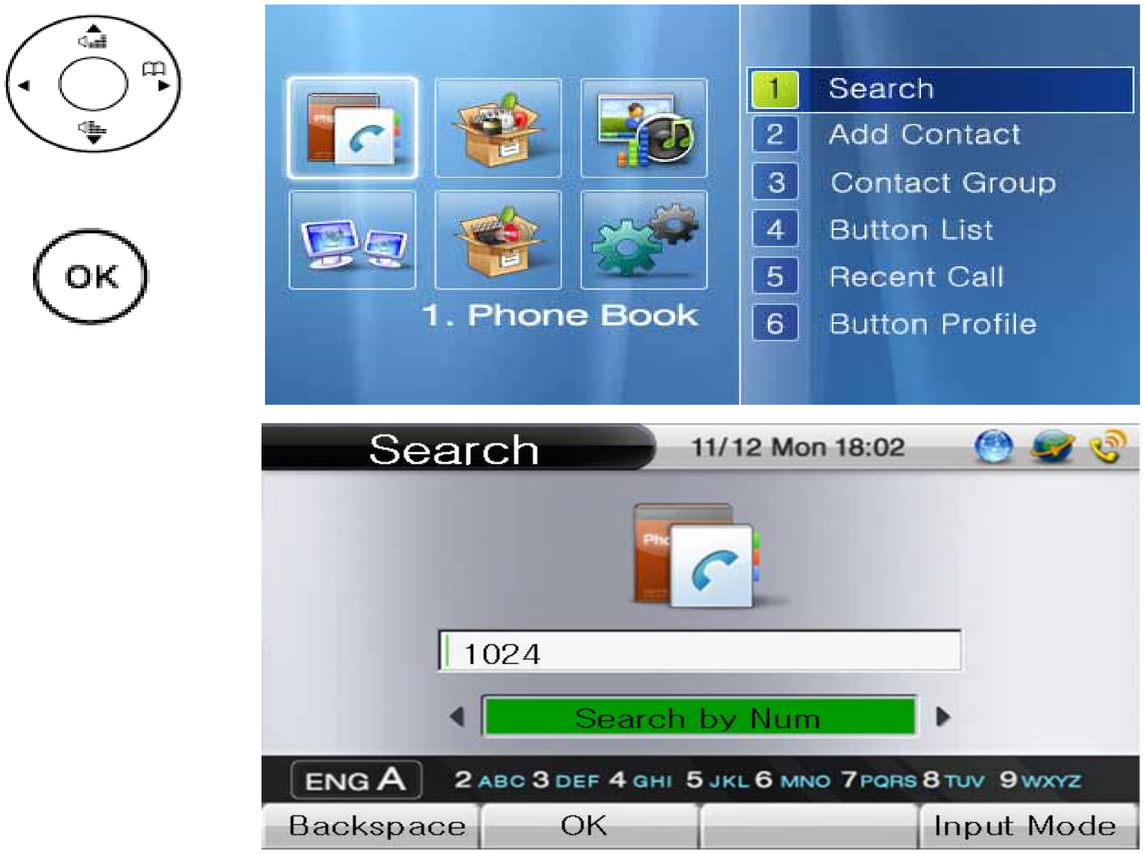


(Figure 4-4) Phonebook Menu Screen

Phonebook – Searching a Phone Number

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



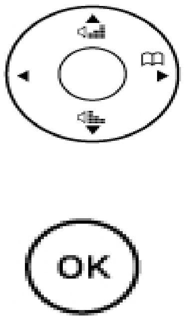


(Figure 4-5) Search Menu Screen in the Phonebook

[Table 4-2] Description of Search Menu Screen in the Phonebook

Phonebook Search Menu	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: Erase F2: Complete F4: Change the text
	Searches by the numbers which have been saved previously F1: Erase F2: Complete F4: Change the text
	Searches by the speed dial numbers which have been saved previously F1: Erase F2: Complete F4: Change the text

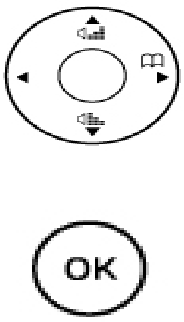
(Figure 4-6) When OK is pressed on the dial pad (it searches all the number which have been saved previously)



(Figure 4-7) when F1 (modify) is pressed

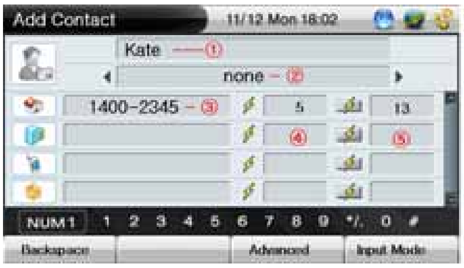
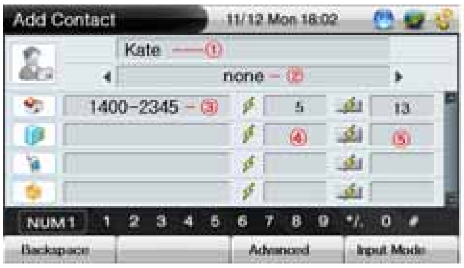
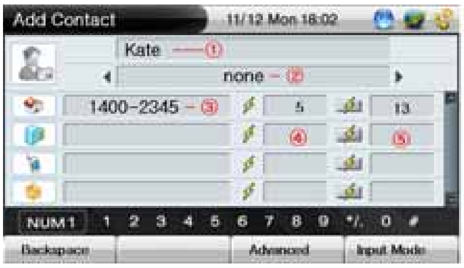
Phonebook – Registration

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.



(Figure 4-8) Registration Menu Screen in the Phonebook

[Table 4-3] Description of Registration Menu Screen in the Phonebook

Registration Menu	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.

(Figure 4-9) Advanced Registration Setup Screen

(Figure 4-10) Advanced Setup for Registration Screen

[Table 4-4] Description of Advanced Setup for Registration Screen

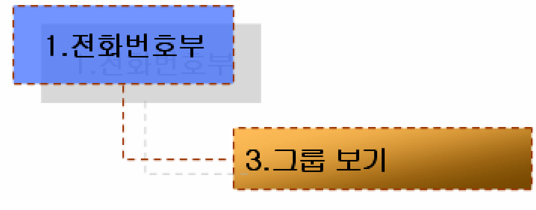
Registration	Description
	1. Enter a phone number

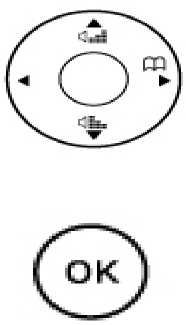
2. Select a type of audio codec.

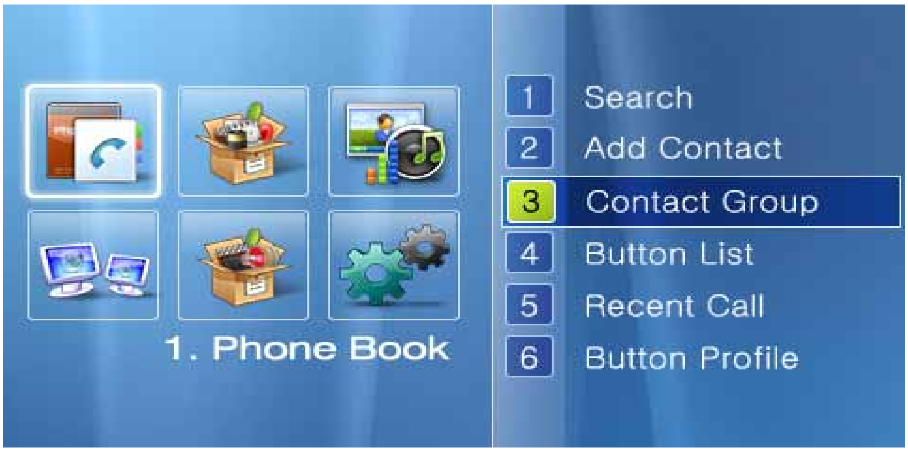
3. Enter an IP address

Phonebook- Contact Group

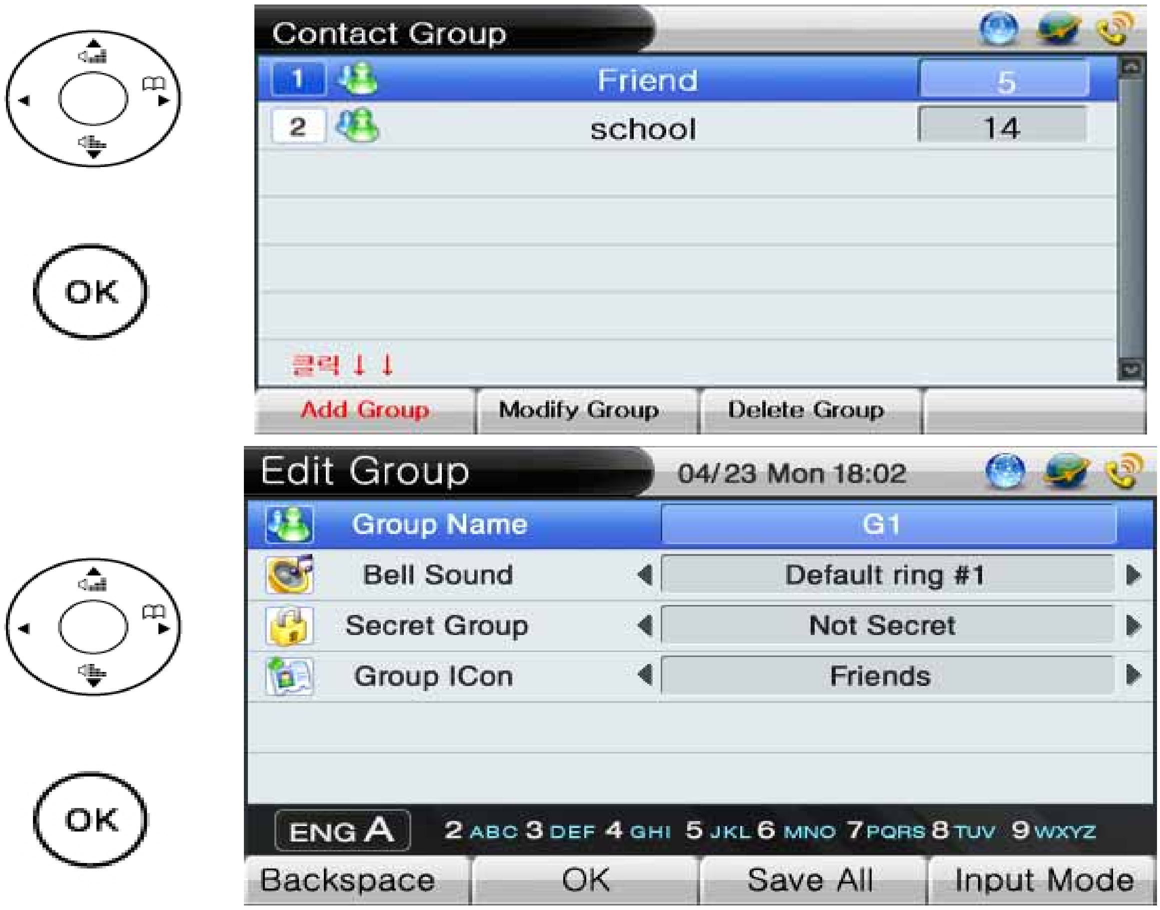
The user can specify a group during registration to the phonebook. The specified group can be set with an icon, a bell sound and secret group. The added group can be applied right after the phonebook registration.







(Figure 4-11) Contact Group



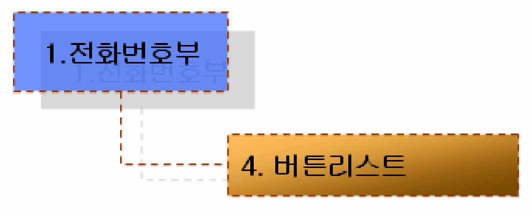
(Figure 4-12) When the softkey of Add Group or Modify Group (when the existing group presents) is pressed

[Table 4-5] Edit Group

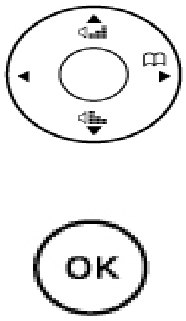
Category	Description
Group Name	Enter a group name
Bell Sond	The bell sound for the specified group
Secret Group	Locks the group so others can not see
Group Icon	Sets an icon for the specifies group

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.



(Figure 4-13) Button List Screen



(Figure 4-14) Editing Screen in the Button List

[Table 4-6] Description of the Editing Screen in the Button List

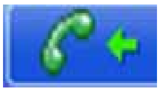
Category	Description
Name	Set a name for the Speed Button.
Number	Set a number for the Speed Button
Type	Set a type of icon for the

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.

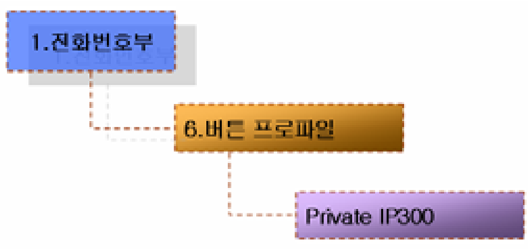
(Figure 4-15) Recent Call Screen

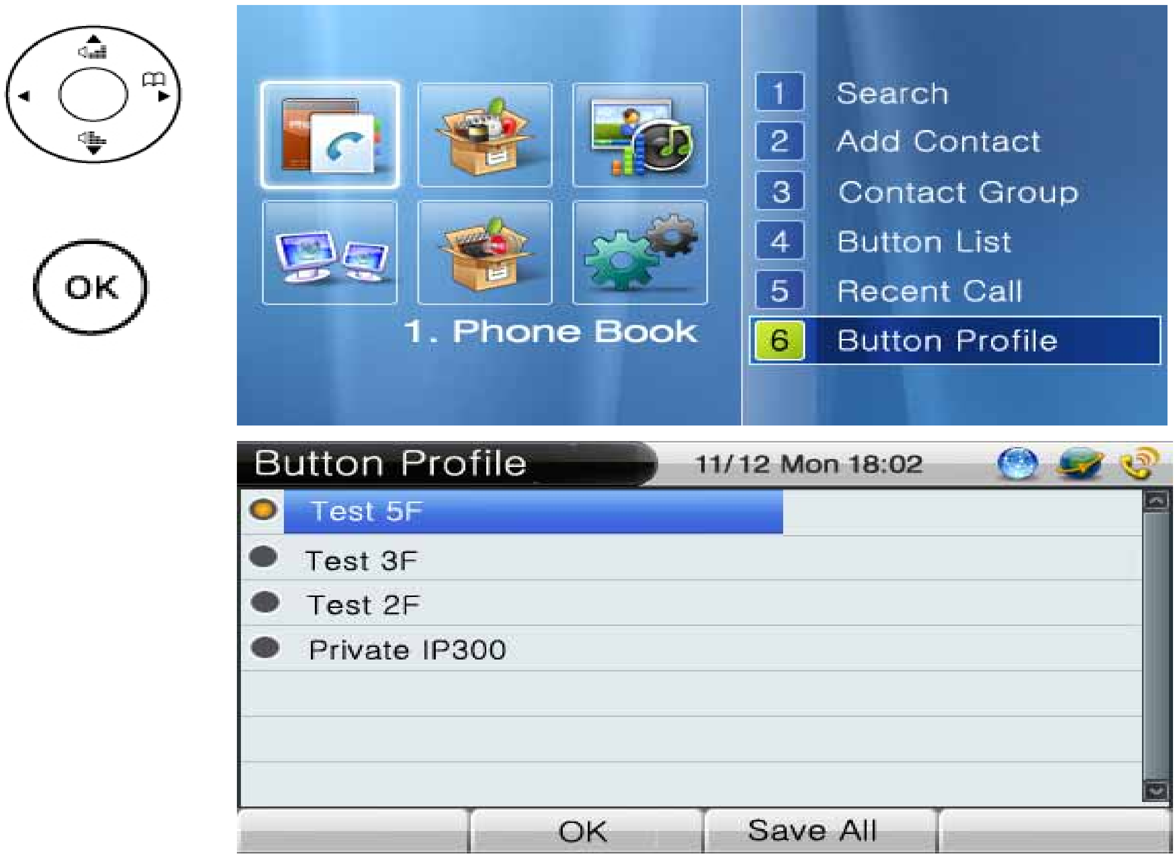
[Table 4-7] Description of the Recent Call Screen

Category	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 taken for placing or receiving a call
④ Delete	Delete a recent call log
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 set up [4. Network and Call Setup – 10. Presence] first.





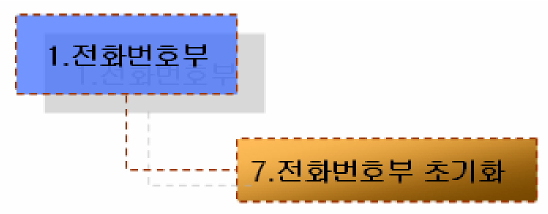
(Figure 4-16) Button Profile Menu Screen

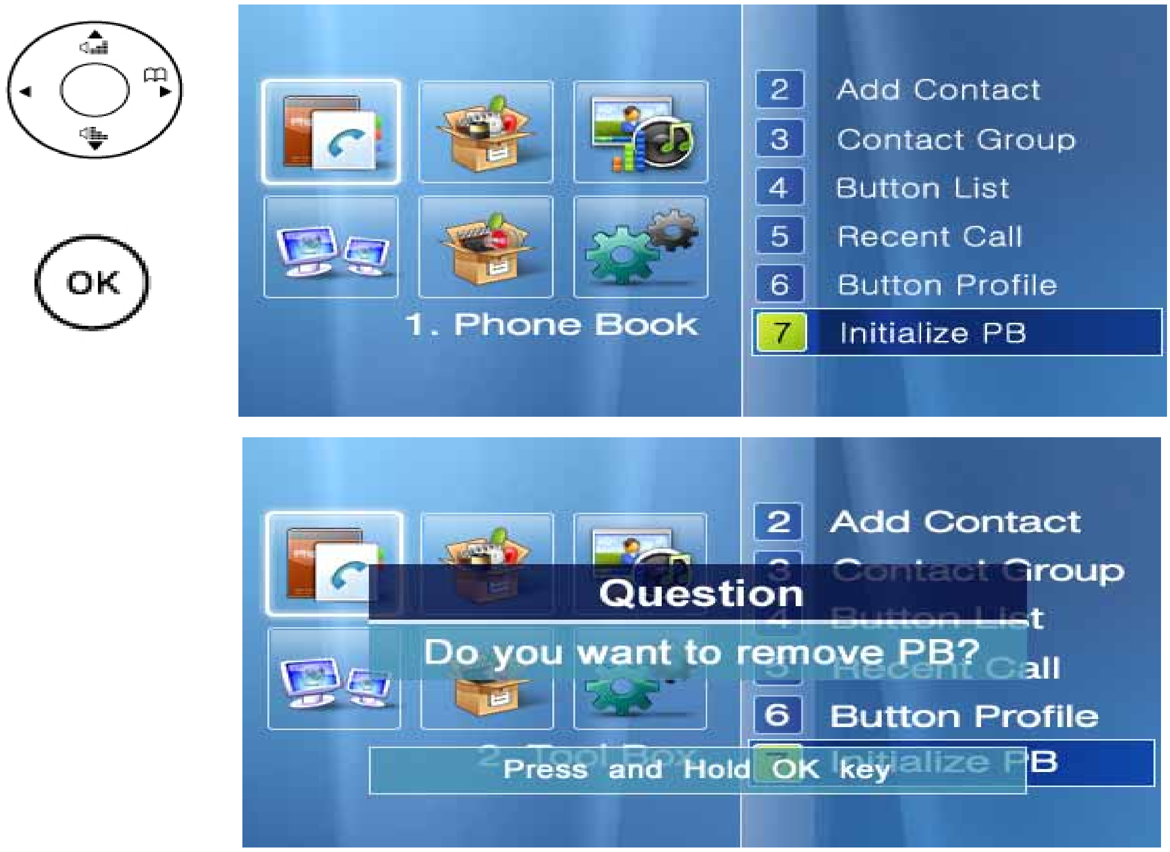
[Table 4-8] Description of Button Profile Menu Screen

Category	Description
Private IP300	Assign a phone number to the Speed Button directly. This profile is displayed when AP-IP300 is not registered to the Presence Server
Test 5F	Receive the profile form the Presence Server

Phonebook – Initialize PB

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.

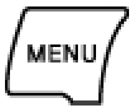




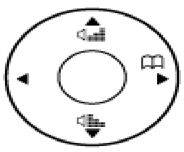
(Figure 4-17) Initialize PB Menu Screen

Tool Box Menu

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



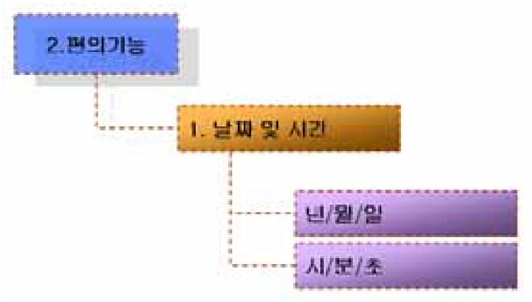
(Figure 4-18) Main Screen

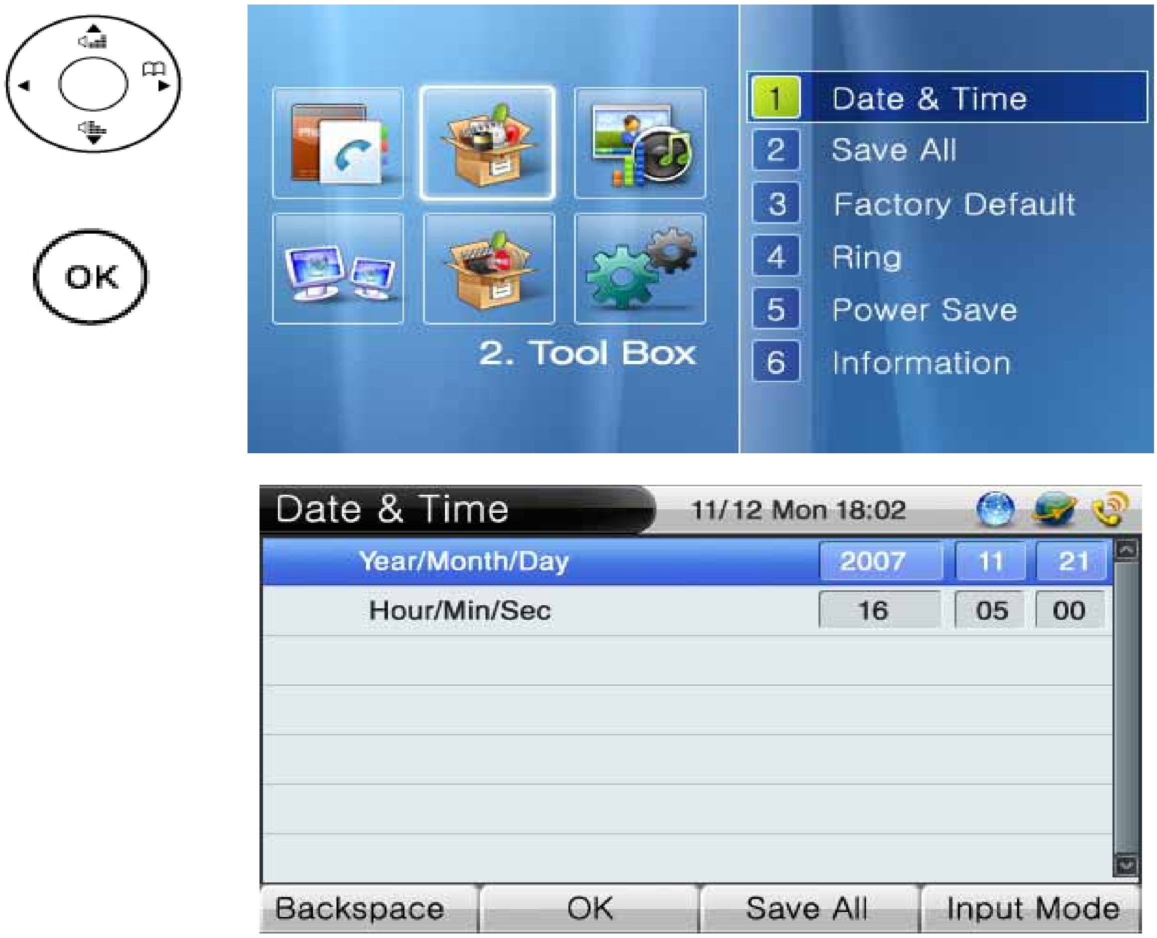


(Figure 4-19) Too Box Menu Screen

Tool Box – Date & Time

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





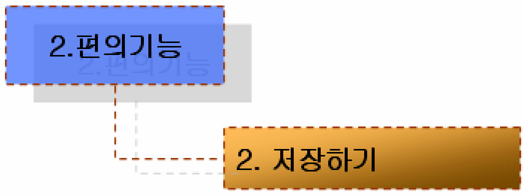
(Figure 4-20) Date & Time Menu Screen

[Table 4-9] Description of Date & Time Menu Screen

Category	Description
Year	Enter the present 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

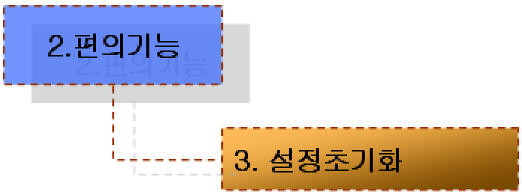
This UI menu saves the settings which the user has entered in UI. Once the settings are saved, values are preserved even after rebooting.

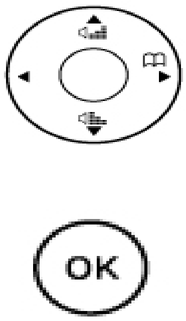


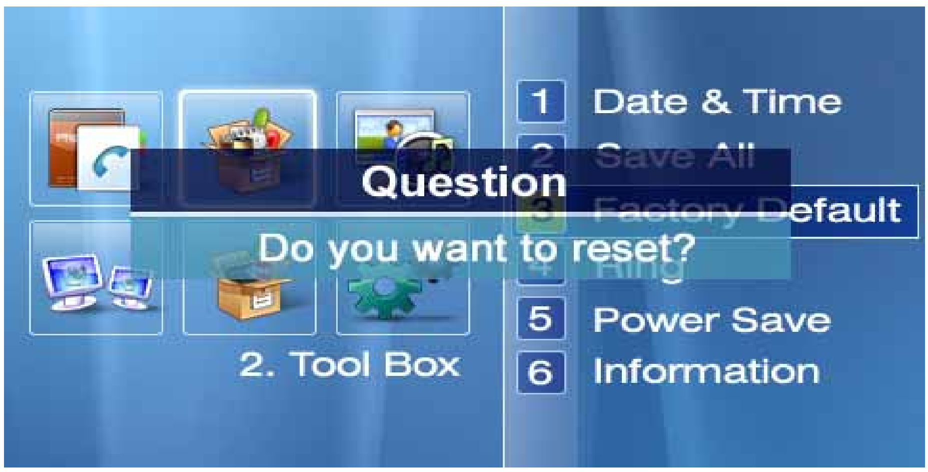
(Figure 4-21) Save All Menu Screen

Tool Box – Factory Default

The Factory Default deletes all the configured settings of AP-IP300 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.



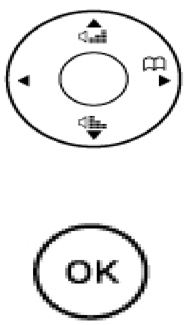




(Figure 4-22) Factory Default Menu Screen

Tool Box – **Ring**

You can set the ringer up to 8 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 7 different kinds of sound, except the mute, by using F1(Play). Also the volume can be adjusted.



(Figure 4-23) Ring Screen

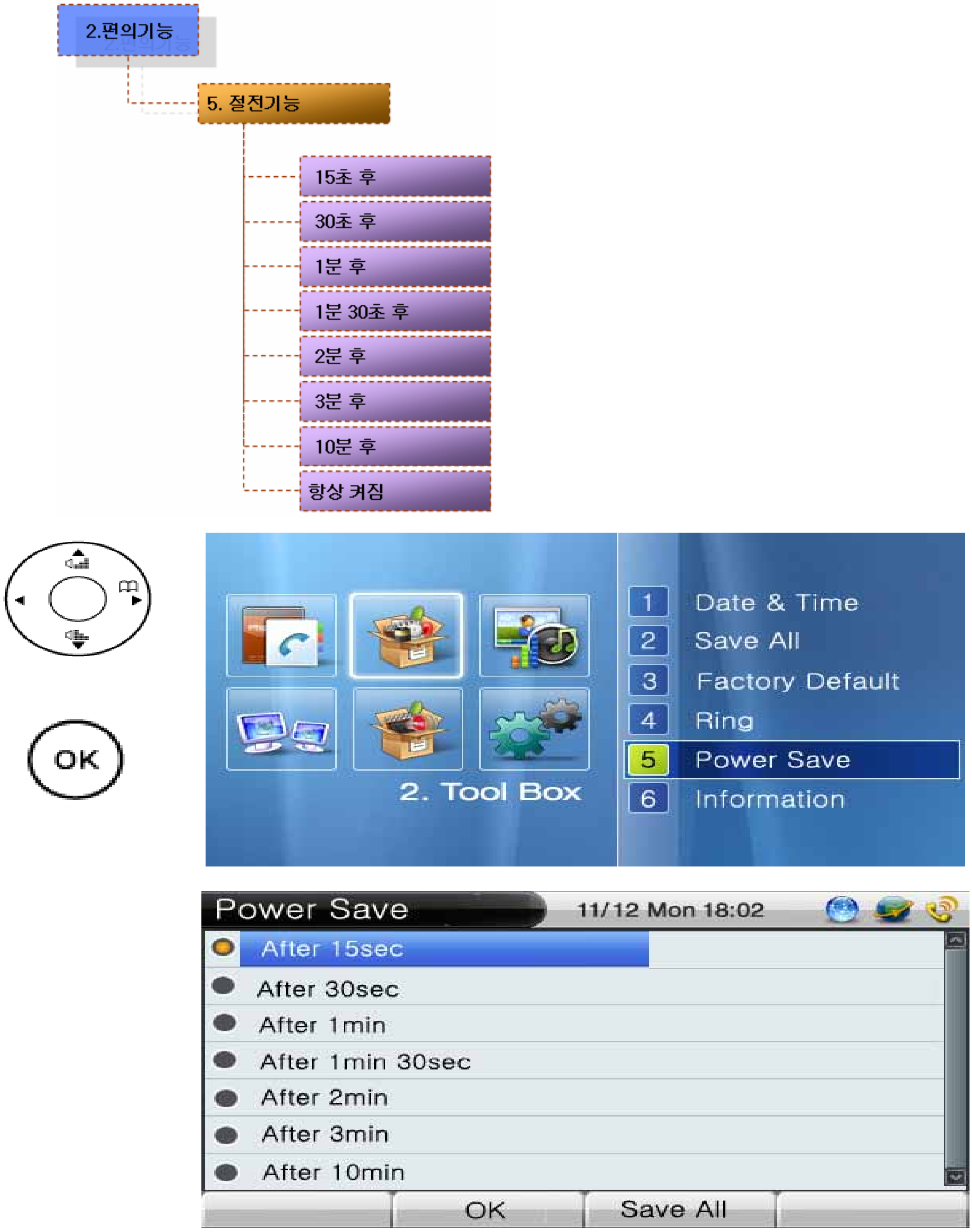
[Table 4-10] Description of Ring Screen

Category	**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**

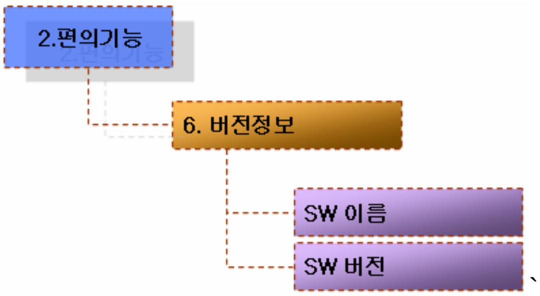
The Power Save turns off the LCD automatically in a specified time. The LCD can be turned on again by pressing any button on the key pad from the state of Power Save. This setting is recommended for preserving the life time of LCD and maintaining its quality.

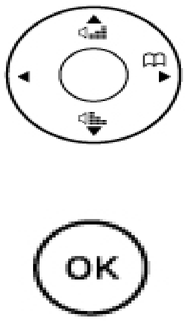


(Figure 4-24) Power Save Menu Screen

Tool Box – Version Information

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





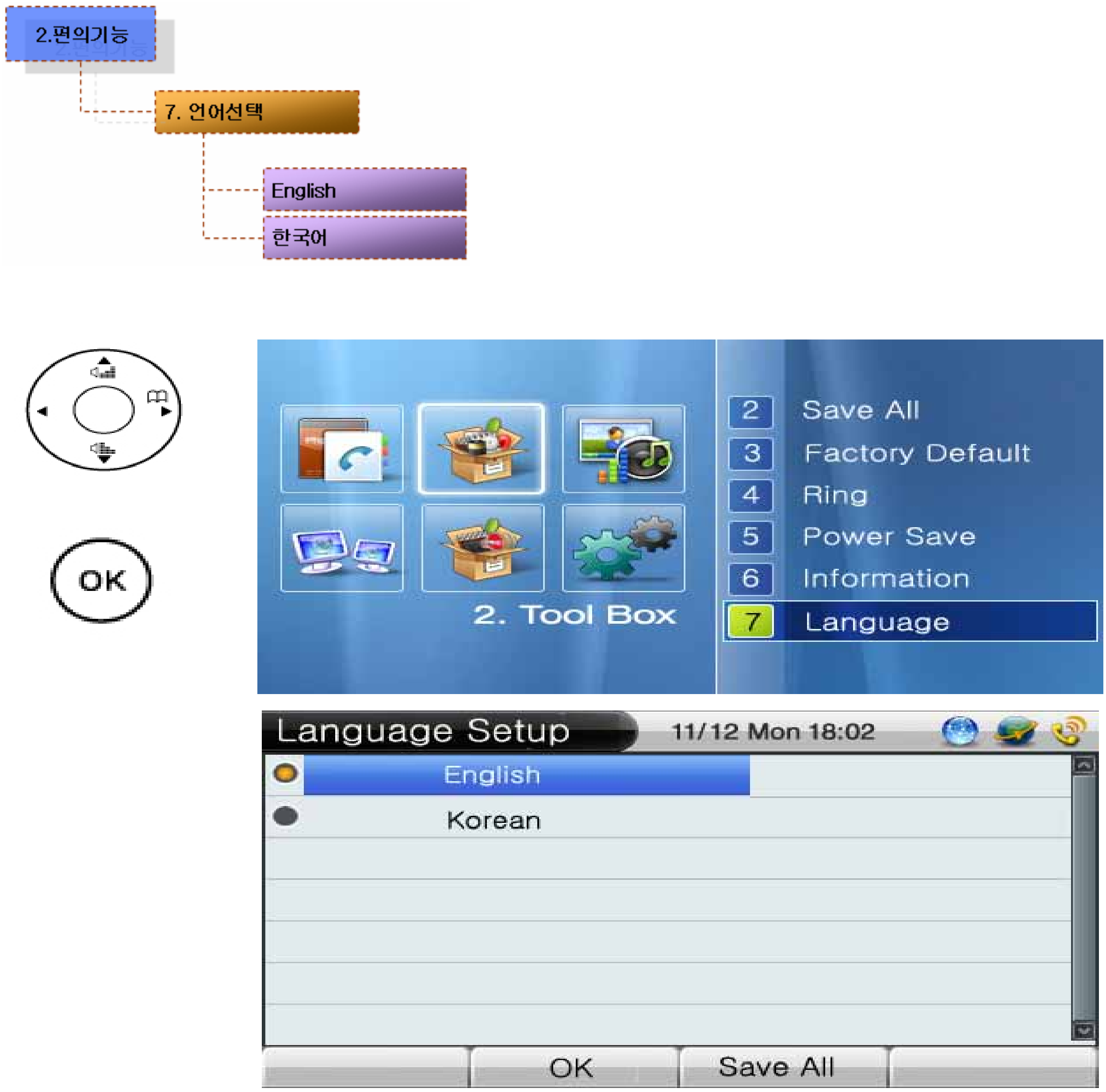
(Figure 4-25) Information Menu Screen

[Table 4-11] Description of Information Menu Screen

Category	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 – Language

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



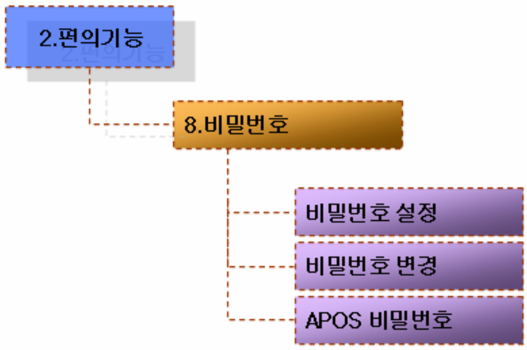
(Figure 4-26) Language Menu Screen

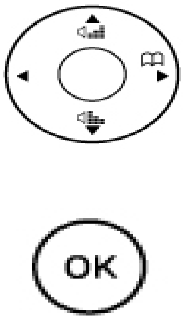
[Table 4-12] Description of Language Menu Screen

Category	Description
English	English Language Mode
Korea	Korean Langage Mode

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.





(Figure 4-27) Password Menu Screen

[Table 4-13] Description of the Password Menu Screen

Category	Description
Use Password	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.

AV Setup Menu

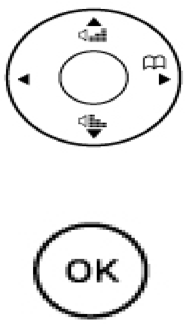
AV Setup consists of Volume and Audio Codec.

(Figure 4-28) Main Menu

(Figure 4-29) AV Setup Menu Screen

AV Setup – Volume

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



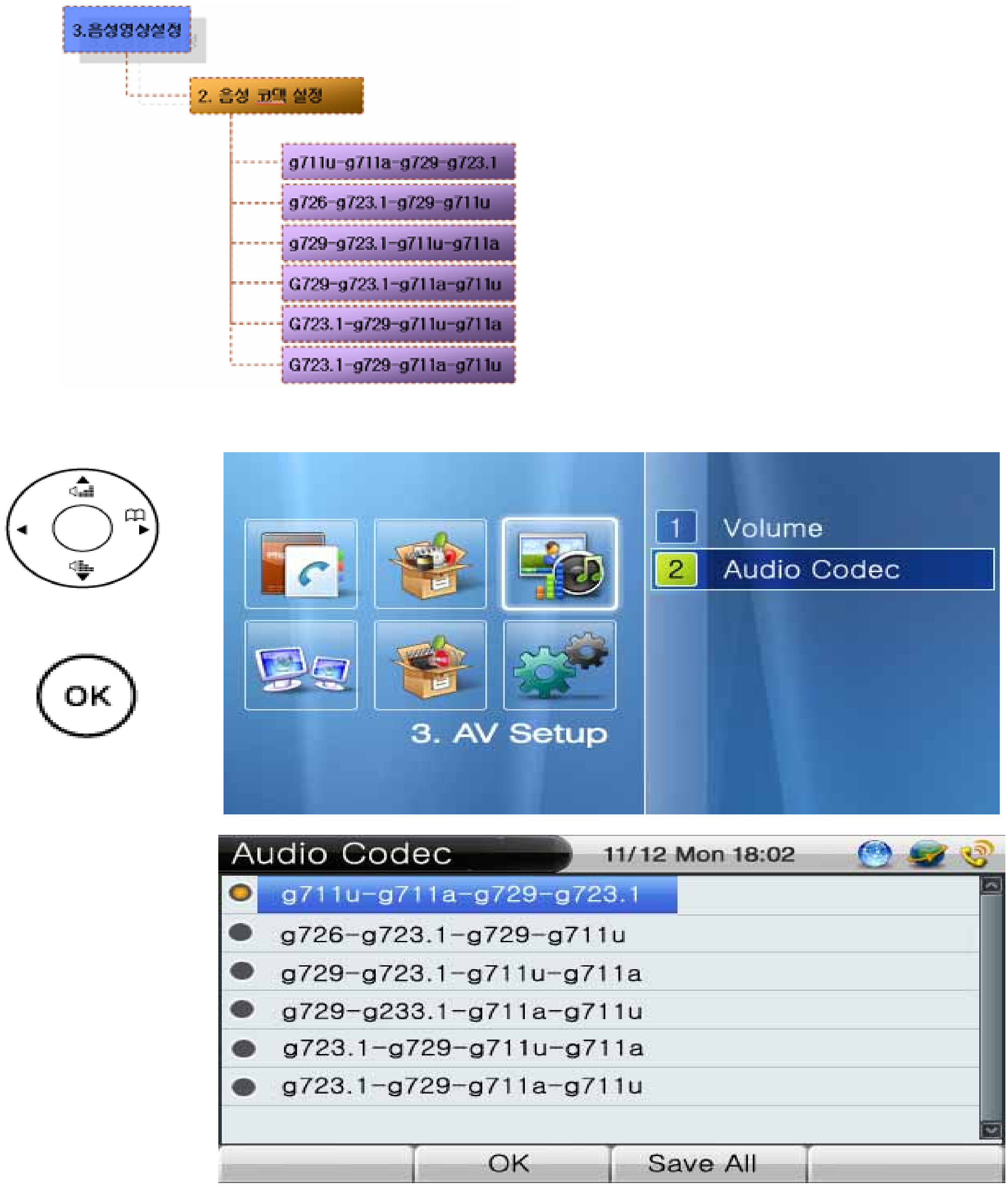
(Figure 4-30) Volume Setup Menu Screen

[Table 4-14] Description of Volume Setup Screen

Category	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

AV Setup – 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.



(Figure 4-31) Audio Codec Menu Screen

Network & Call 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-IP200. This menu cannot be skipped for optimized environment.

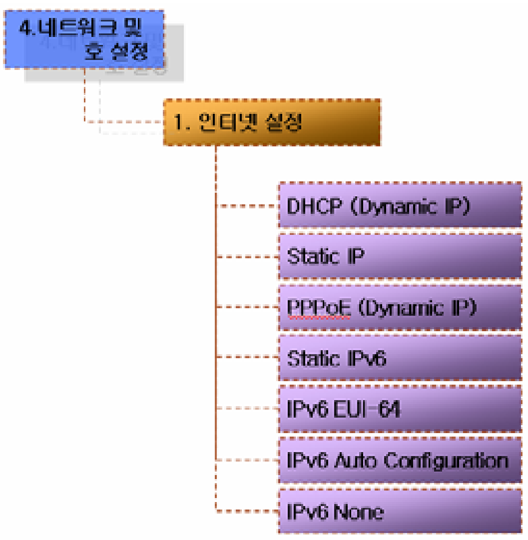
(Figure 4-32) Main Screen

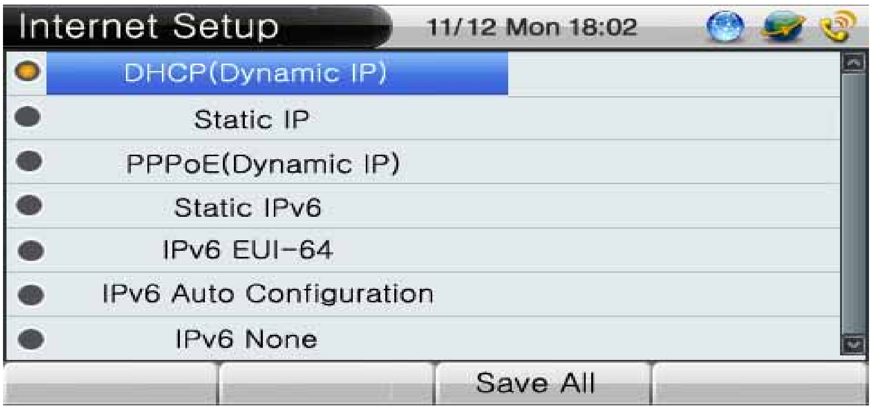
(Figure 4-33) Network & Call Menu Screen

Network & Call – Internet Setup

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-IP300 are DHCP, static IPv4, PPPoE, and IPv6 etc.

The following figure shows the UI command tree structure in Network & Call Menu.





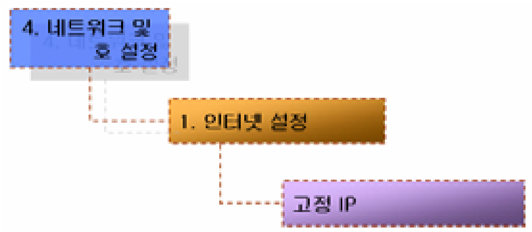
(Figure 4-34) Internet Setup Menu Screen

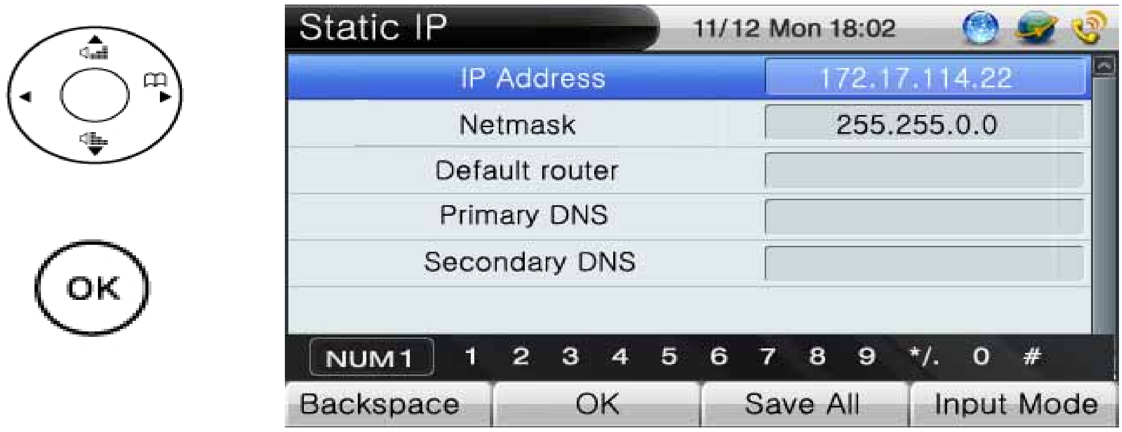
[Table 4-15] Description of Internet Setup Menu Screen

Category	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 (Dynamic IP)	Configures IP address manually and build WAN interface such as static IP ADSL, E1/T1 leased line.
Static IPv6	Configures IPv6 address manually and WAN interface
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 & Call – Internet Setup – IP Static IP

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





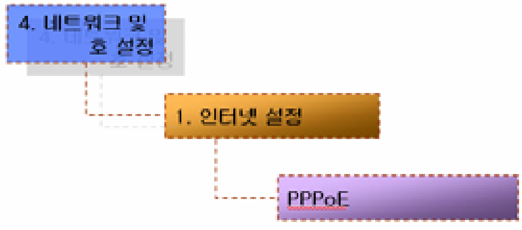
(Figure 4-35) Static IP Menu Screen

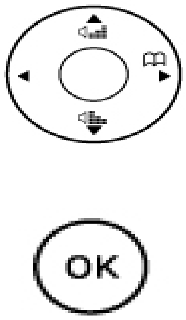
[Table 4-16] Description of Static IP Menu Screen

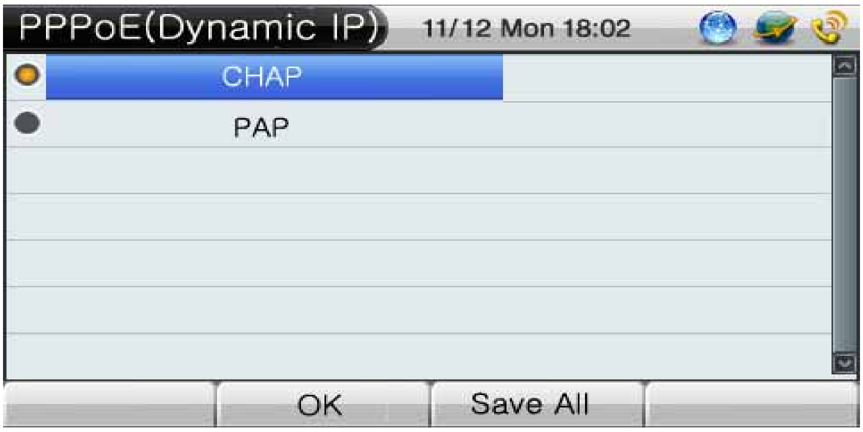
Category	Description
IP Address	IP Address Entry Ex> 172.20.1.100
Net mask	Net Mask Entry Ex> 255.255.0.0
Primary Router	Primary Router Entry Ex> 172.20.1.1
Primary DNS	First DNS Entry (same as IPv6) Ex> 168.126.63.1
Secondary DNS	Secondary DNS Entry (optional)

Network & Call – 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.







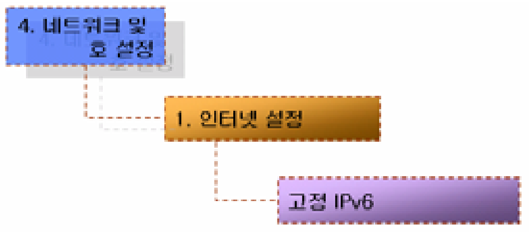
(Figure 4-36) PPPoE Menu Screen

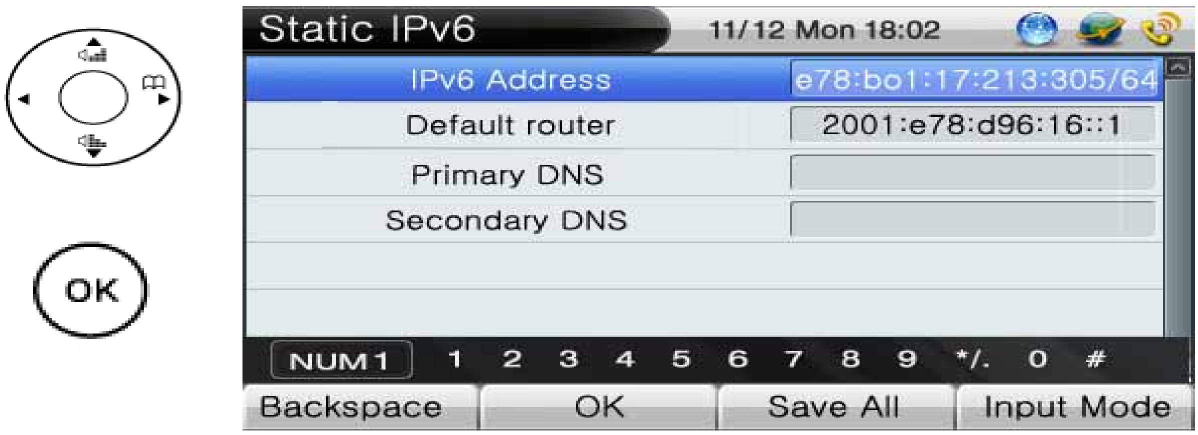
[Table 4-17] Description of PPPPoE Menu Screen

Category	Description
CHAP	Authentication Mode – CHAP
PAP	Authentication Mode – PAP

Network & Call – Internet Setup – Static IPv6

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





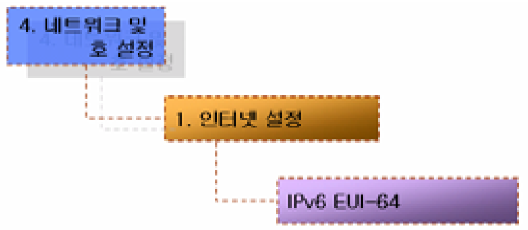
(Figure 4-37)Static IPv6 Menu Screen

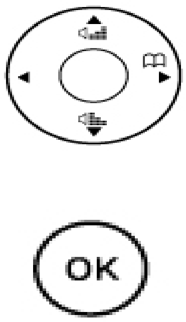
[Table 4-18] Description of Static IPv6 Menu Screen

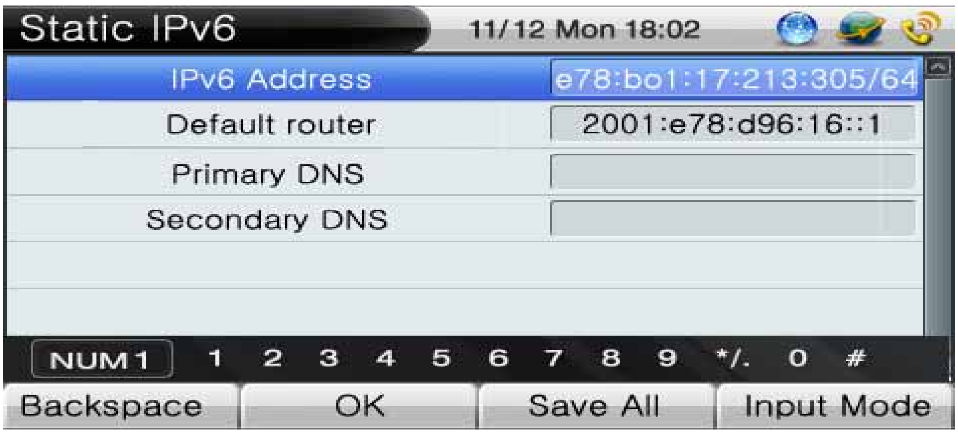
Category	Description
IPv6 Address	IPv6 Address Entry Ex> 2001:e78:b01:17:114::10/64
Primary Router	Primary IPv6 Router Address Entry Ex> 2001:e78:b01:17:114::1
Primary DNS	Primary DNS Entry Ex> 168.126.63.1
Secondary DNS	Secondary DNS Entry (optional)

Network & Call – Internet Setup – 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.







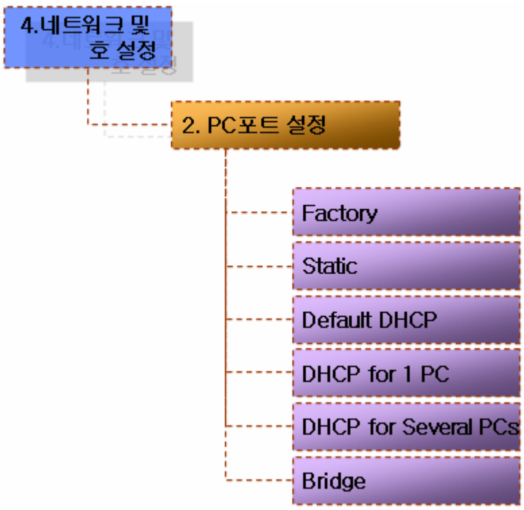
(Figure 4-38) IPv6 EUI-64 Menu Screen

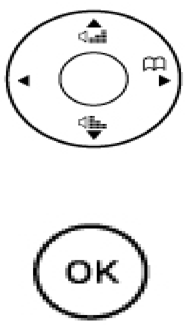
[Table 4-19] Description of IPv6 EUI-64 Menu Screen

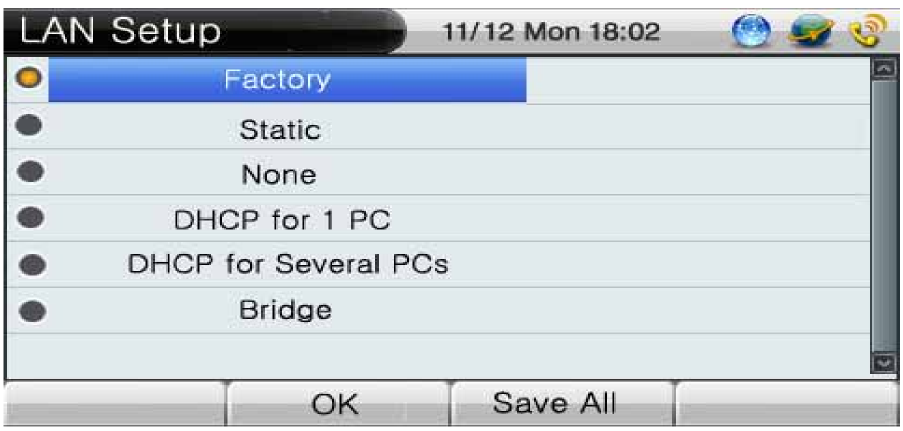
Category	Description
IPv6 Address	IPv6 Address Entry Ex> 2001:e78:b01:17:114::10/64
Primary Router	Primary IPv6 Router Address Entry Ex> 2001:e78:b01:17:114::1
Primary DNS	Primary DNS Entry Ex> 168.126.63.1
Secondary DNS	Secondary DNS Entry (optional)

Network & Call – LAN Setup

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







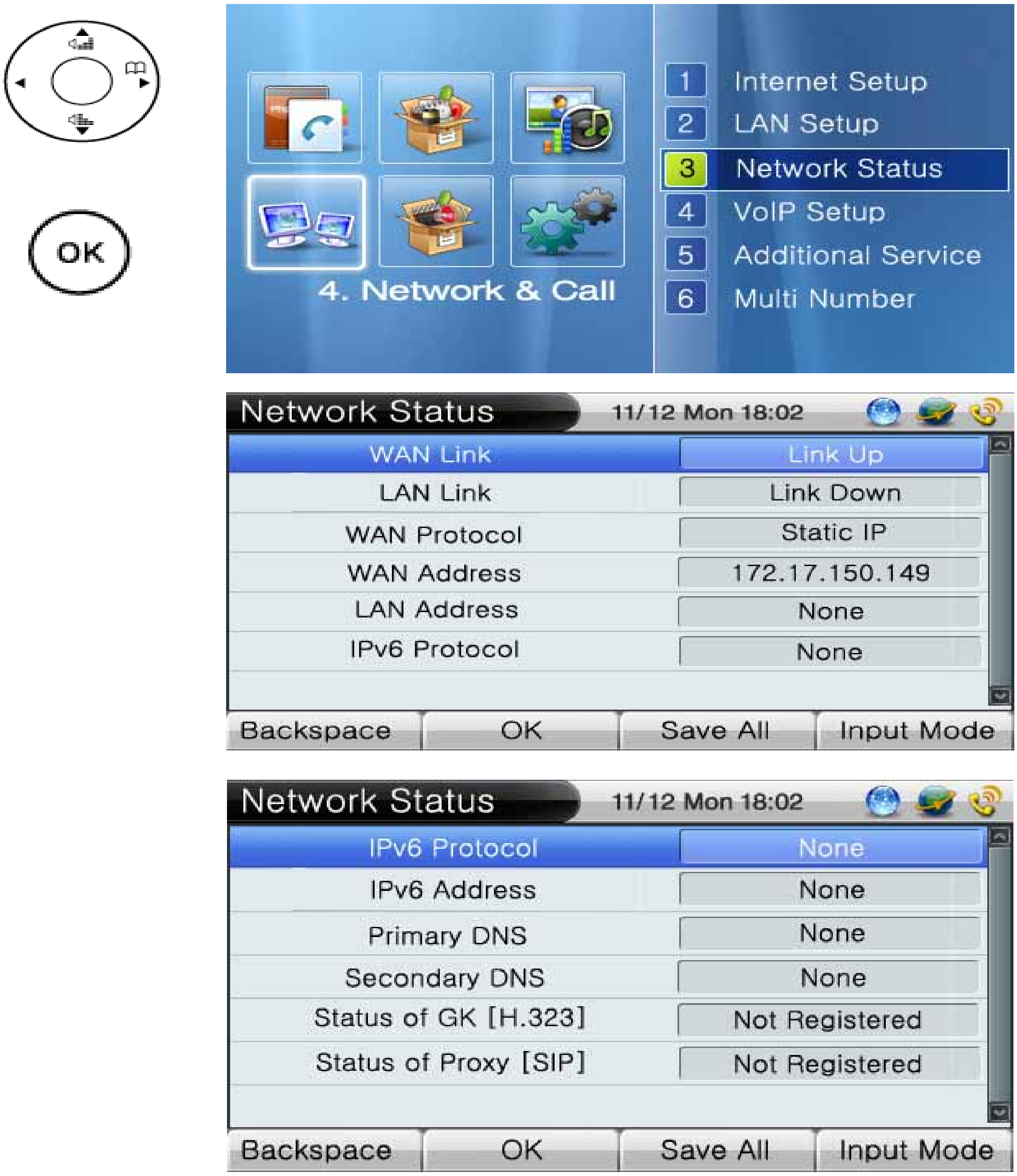
(Figure 4-39) LAN Setup Menu Screen

[Table 4-20] Description of LAN Setup Menu Screen

Category	Description
Factory	Set LAN to the 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 & Call – **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.



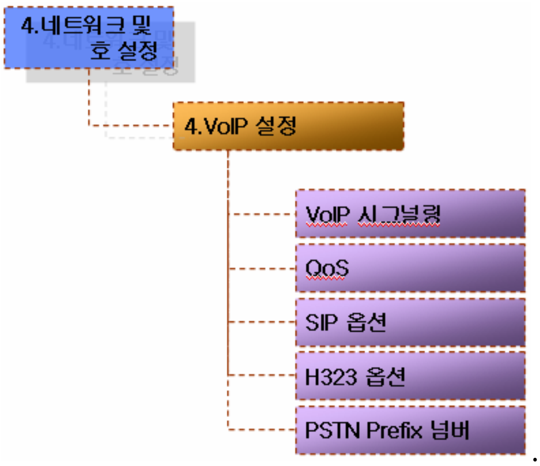
(Figure 4-40) Network Status Menu Screen

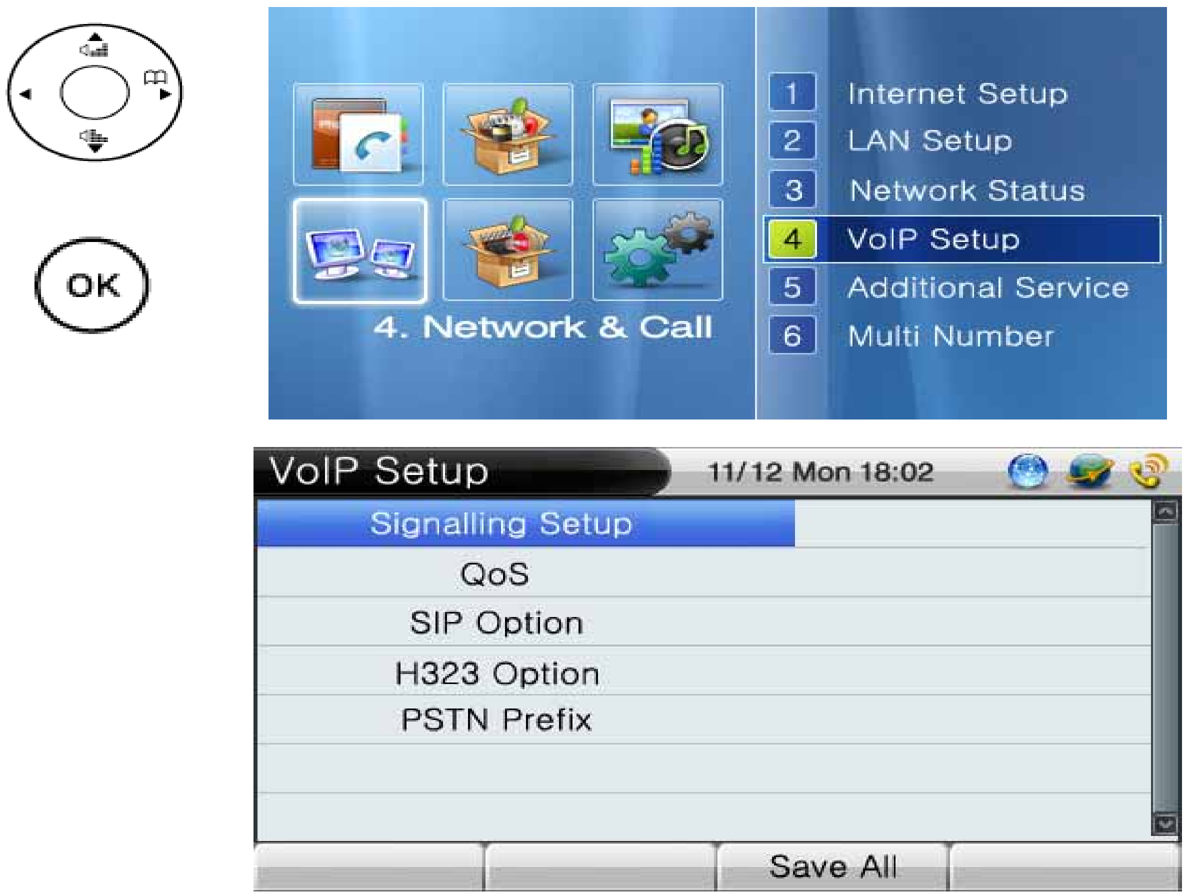
[Table 4-21] Description of the Network Status Menu Screen

Category	Description
WAN Link	Displays whether the link is up/ down of LAN0 interface
LAN Link	Displays whether the link is up/ down of LAN1 (PC) interface
WAN Protocol	Displays WAN IPv4 protocol (DHCP, Static IPv4, PPPoE)
WAN Address	Displays WAN IPv4 address
LAN Address	Displays LAN IPv4 address Table시
IPv6 Protocol	Displays WAN IPv6 protocol
IPv6 Address	Displays WAN IPv6 address
Primary DNS	Displays primary Domain Name Server
Secondary DNS	Displays the secondary Domain Name Server
Status of GK [H.323]	Displays the status of the gatekeeper
Status of SIP Proxy	Displays the status of the proxy server

Network & Call – VoIP Setup

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.

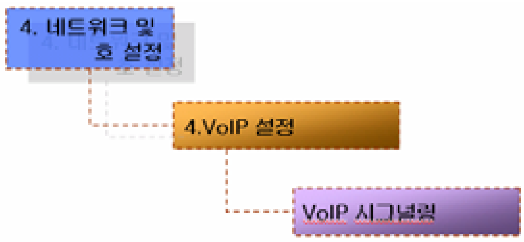


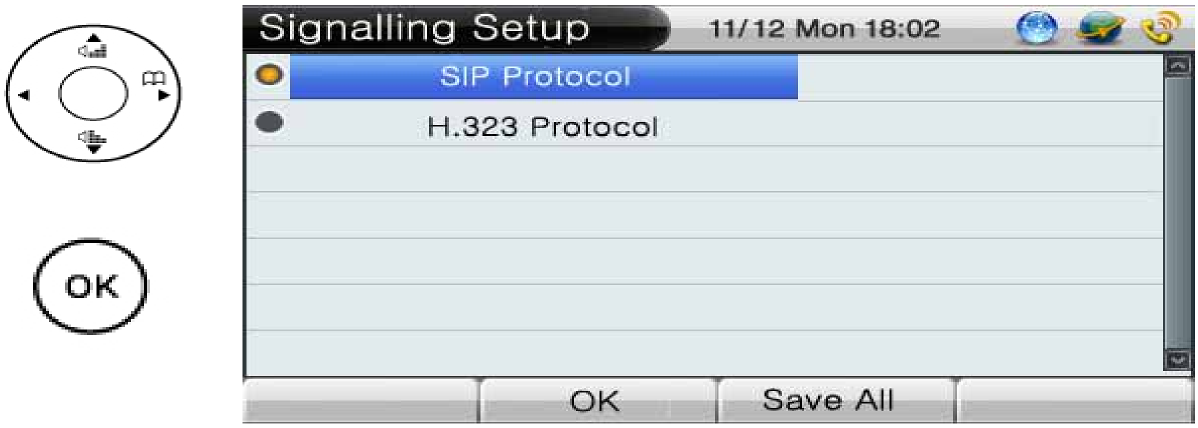


(Figure 4-41) VoIP Setup Menu Screen

Network & Call – **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.





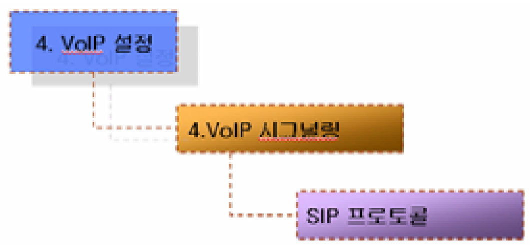
(Figure 4-42) VoIP Signaling Setup Menu Screen

[Table 4-22] Description of VoIP Signaling Setup Menu Screen

Category	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 & Call – **VoIP Setup - 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.



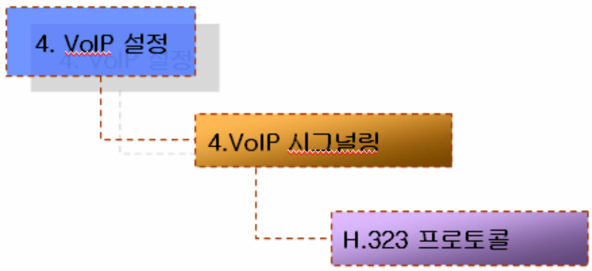
(Figure 4-43) SIP Protocol Menu Screen

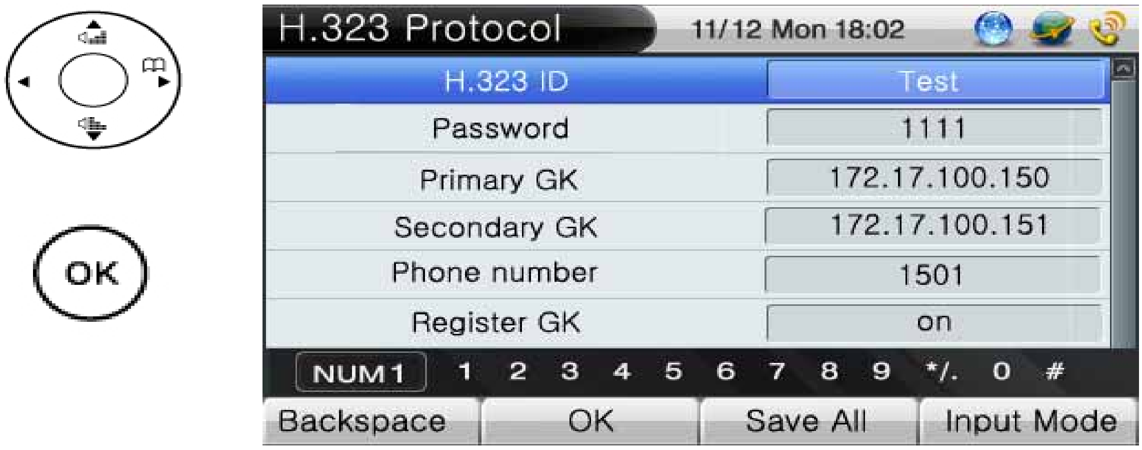
[Table 4-23] Description of SIP Protocol Menu Screen

Category	Description
User Name	Enter a username for SIP server registration
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 the user’s E.164 number
Register e.164	Use the key pad or numeric key to register E.164 to SIP server

Network & Call – **VoIP Setup - 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.





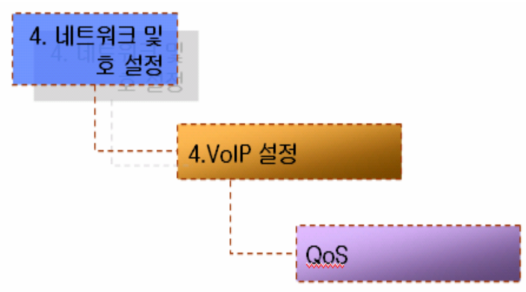
(Figure 4-44) H.323 Protocol Menu Screen

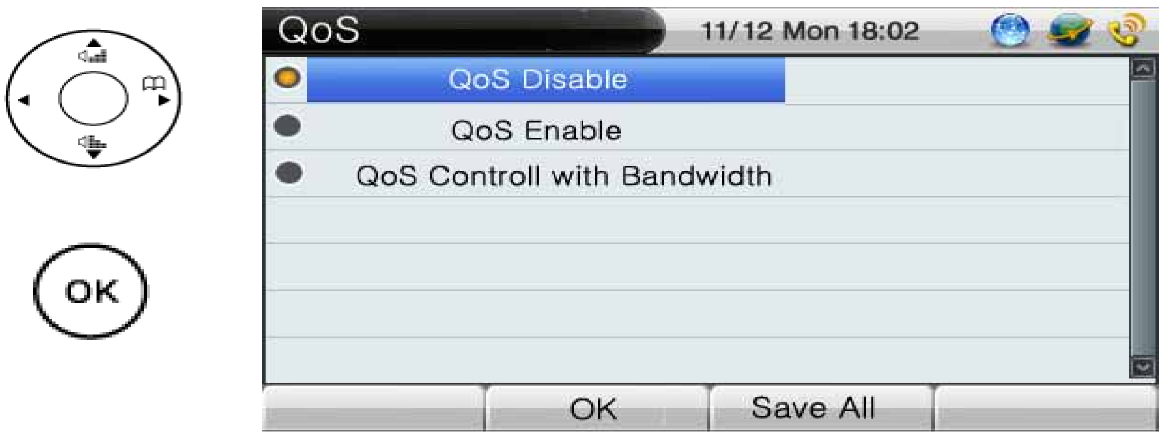
[Table 4-24] Description of H.323 Protocol Menu Screen

Category	**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 – **VoIP Setup – 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.





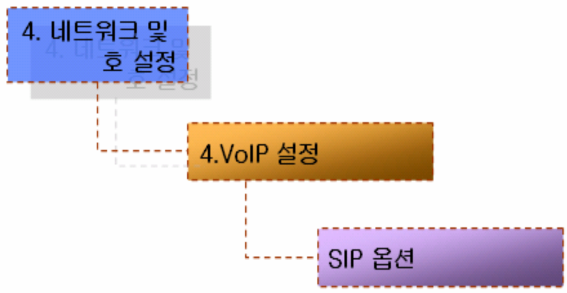
(Figure 4-45) QoS Screen

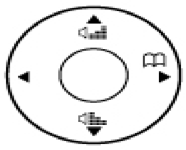
[Table 4-25] Description of QoS Screen

Category	**Description**
QoS Disable	Disable QoS
QoS Enable	Enable QoS
QoSBandwidth	QoS function is for WAN interface and cannot be applied to LAN interface. Range of value covers 48Kbps~4Mbps

Network and Call – **VoIP Setup** – **SIP Options**

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





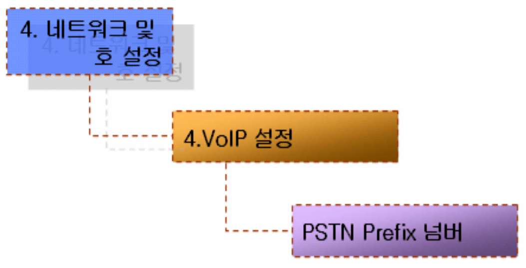
(Figure 4-46) SIP Options Menu Screen

[Table 4-26] Description of SIP Options

SIP Options	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-IP300 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-IP300 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 & Call – VoIP Setup– **PSTN Prefix**

When user wants to access the FXO interface for PSTN backup, this prefix number is used as PSTN access code. Additionally, AP-IP300 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.



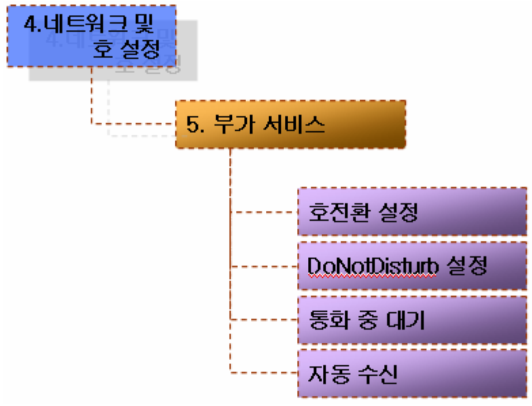
(Figure 4-47) PSTN Prefix Menu

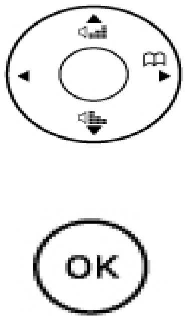
[Table 4-27] Description of PSTN Prefix Menu

PSTN prefix	Description
Number Input	PSTN prefix number is an access code for PSTN FXO interface, default value is #.

Network & Call – Additional Service

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

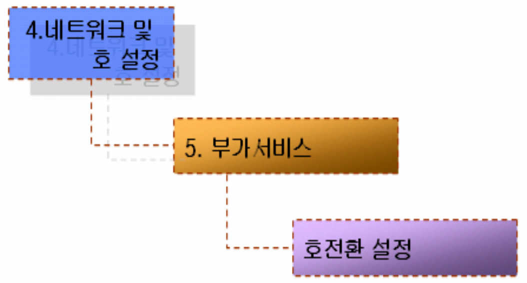


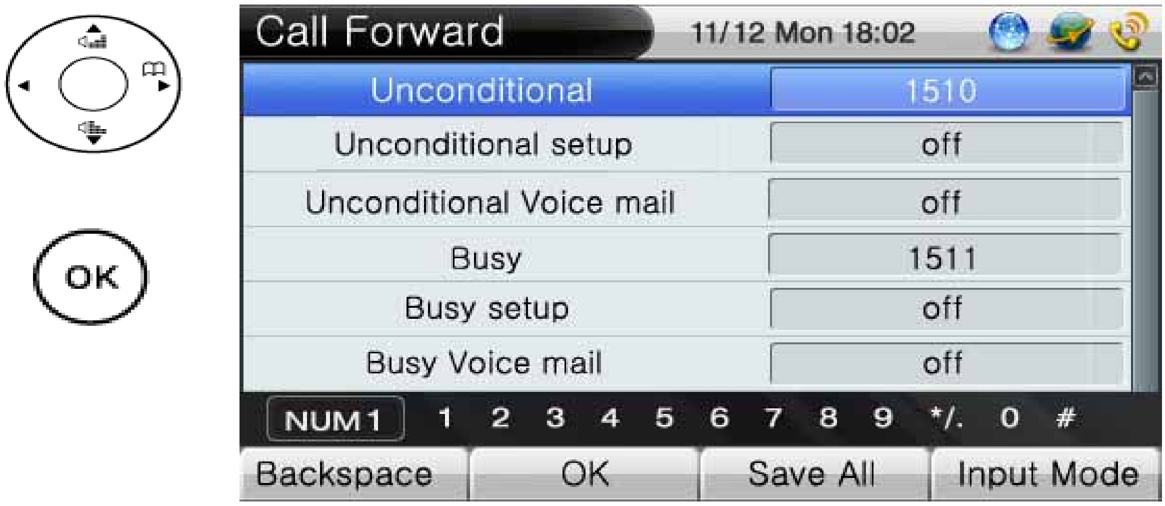


(Figure 4-48) Additional Service Menu Screen

Network & Call – 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.





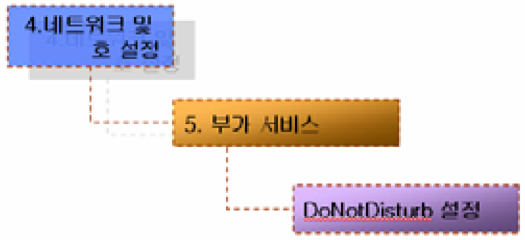
(Figure 4-49) Call Forward Menu Screen

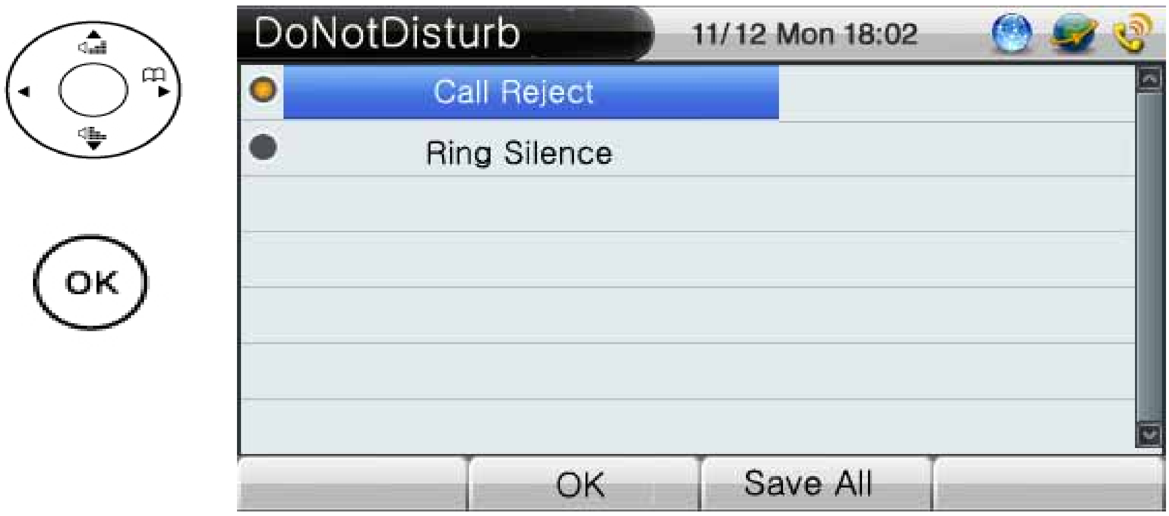
[Table 4-28] Description of Call Forward Menu Screen

Category	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 & Call – 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.





(Figure 4-50) DND Menu Screen

[Table 4-29] Description of DND Menu Screen

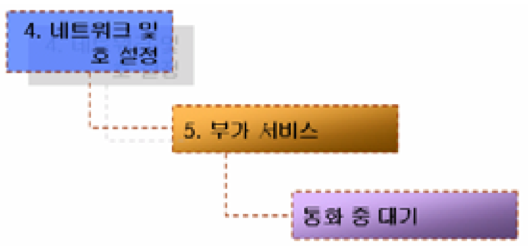
Category	Description
Call Reject	Set the mode to Call Reject
Ring Silence	Set the mode to Ring Silence

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

(Figure 4-51) DND Menu Screen

Network & Call – Additional Service – Call Wait Setup

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.



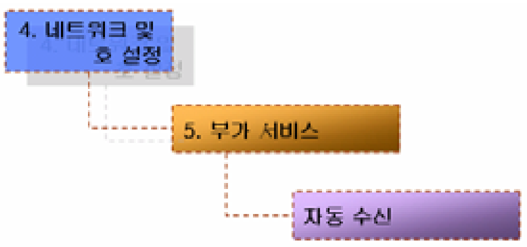
(Figure 4-52) Call Wait Setup Menu Screen

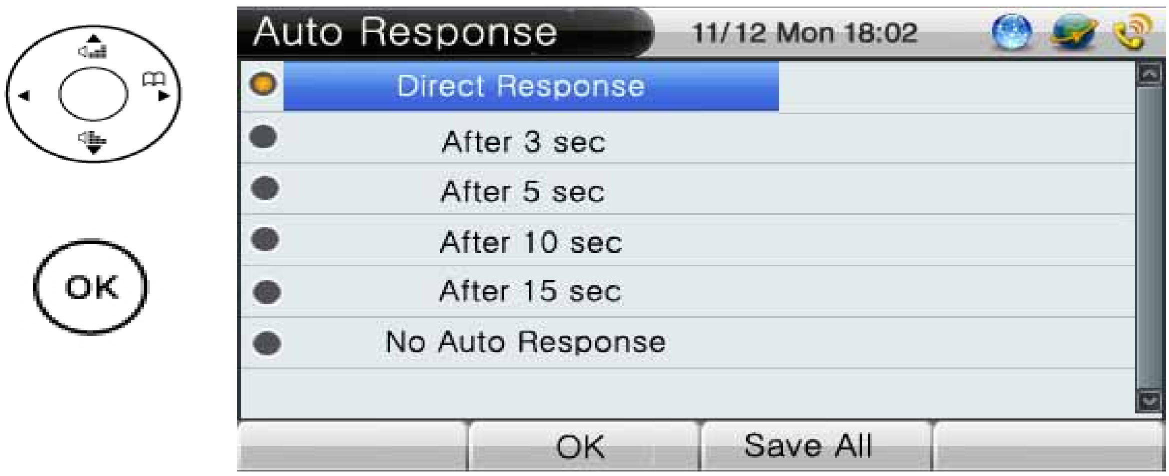
[Table 4-30] Description of Call Wait Setup Menu Screen

Category	Description
Call Wait Off	Disable Call Wait
Call Wait On	Enable Call Wait

Network & Call – Additional Service – Auto Response

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.





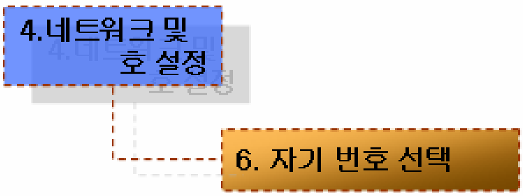
(Figure 4-53) Auto Response Menu Screen

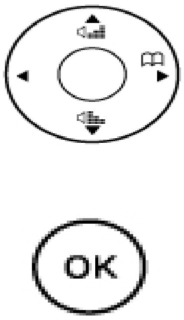
[Table 4-31] Description of Auto Response Menu Screen

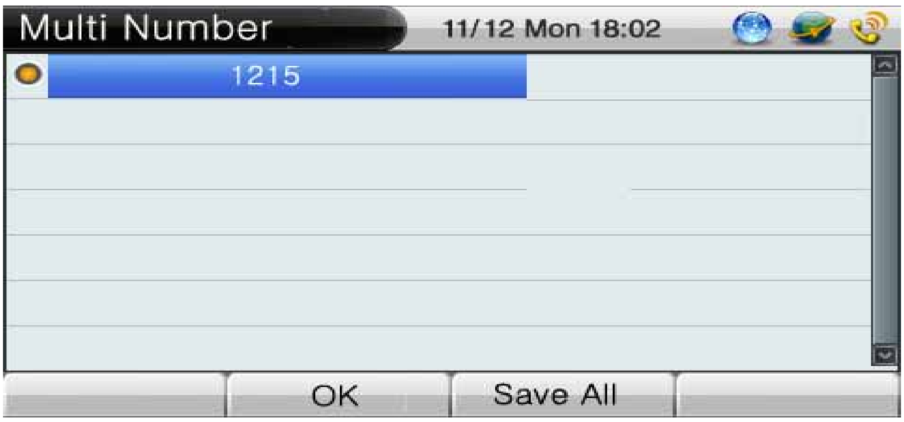
Category	Description
Direct Response	Take a call at the first ring
After 3 sec	Set the mode to Auto Answer to reply on 3 seconds after the bell rings.
After 5 sec	Set the mode to Auto Answer to reply on 5 seconds after the bell rings.
After 10 sec	Set the mode to Auto Answer to reply on 10 seconds after 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 & Call – Additional Service – 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.







(Figure 4-54) Multi Number Menu Screen

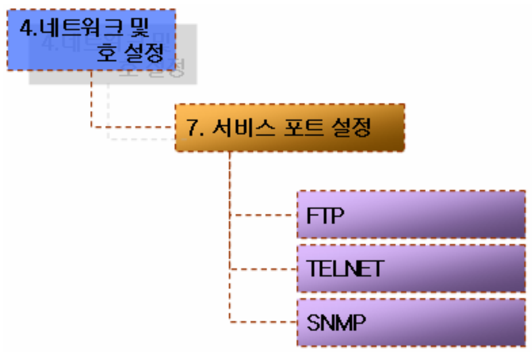
[Table 4-32] Description of the Multi Number Menu Screen

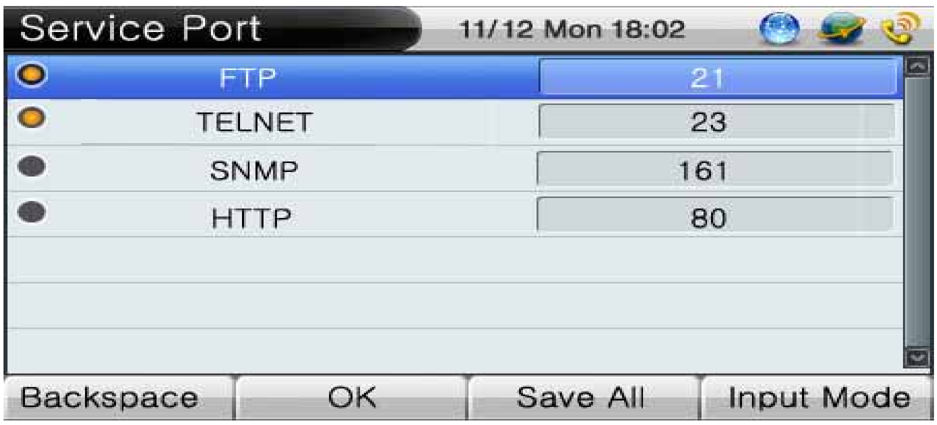
Category	Description
Multi Number	This screen shows an example for assigning the phone number of 1215

Network & Call – Additional Service – Service Port

This menu activates or deactivates FTP, TELNET, TFTP, SNMP protocol service of AP-IP300.

You can use FTP to access to AP-IP300 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-IP300 from a remote location.





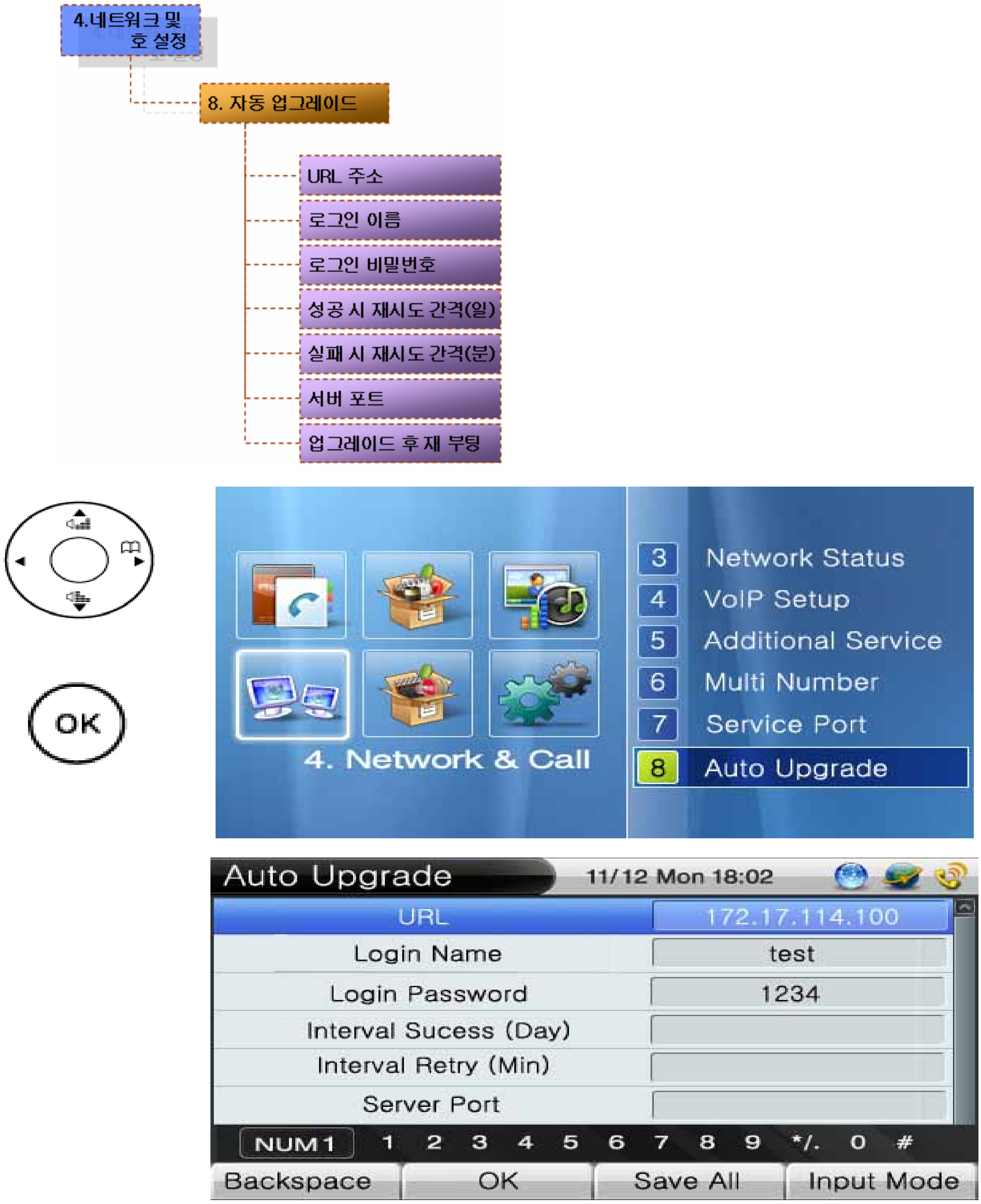
(Figure 4-55) Service Port Menu Screen

[Table 4-33] Description of Service Port Menu Screen

Category	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 & Call – Additional Service – 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.



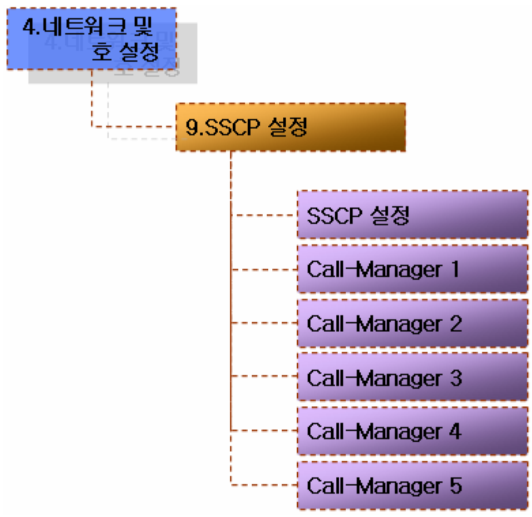
(Figure 4-56) Auto Upgrade Menu Screen

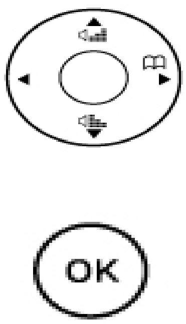
[Table 4-34] Description of Auto Upgrade Menu Screen

Category	Description
URL	Enter URL of the Auto Upgrade server Ex)down.addpac.com/apos/IP300/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 (분)	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 & Call – Additional Service – SSCP Setup

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.





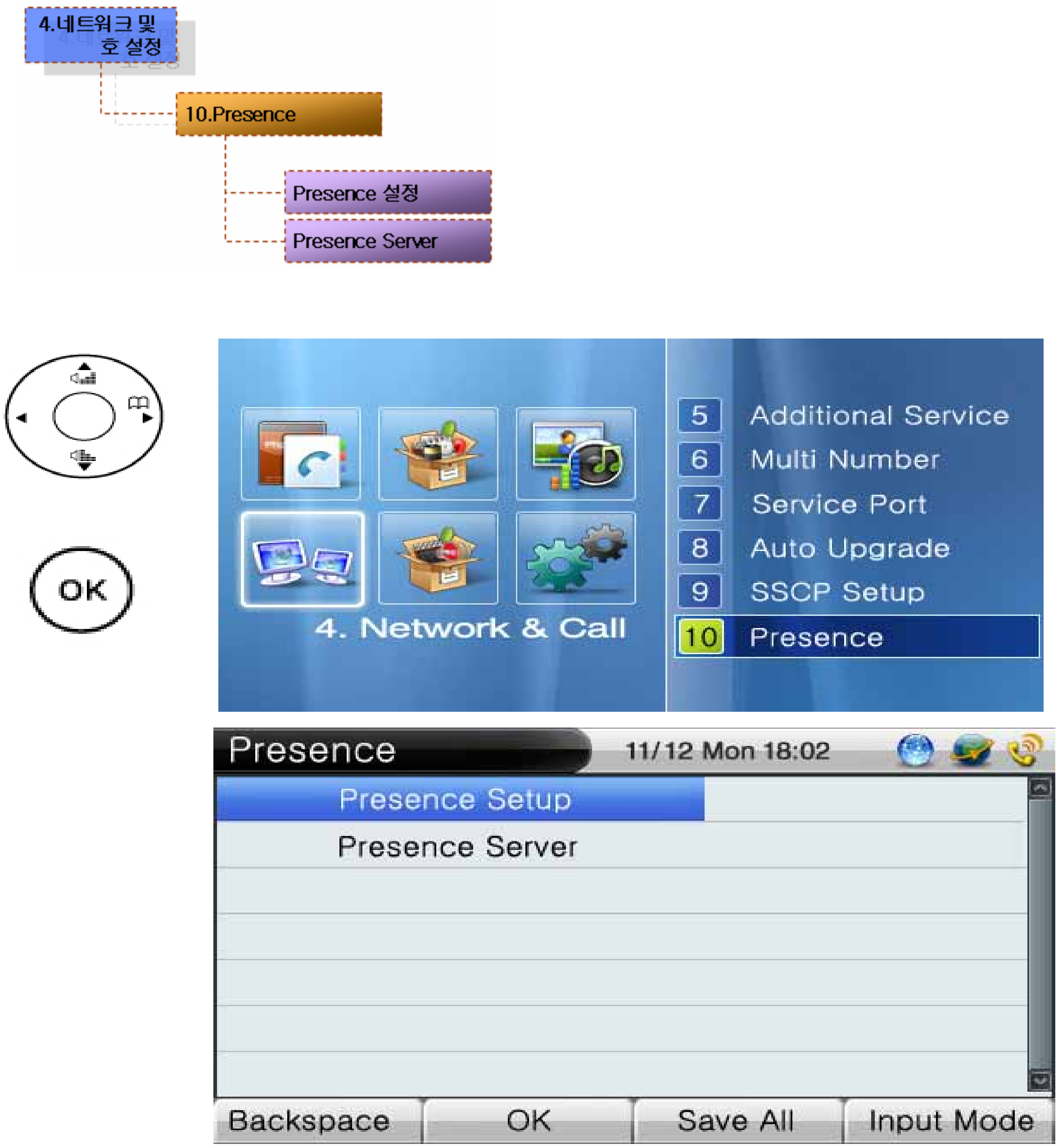
(Figure 4-57) SSCP Setup Menu Screen

[Table 4-35] Description of SSCP Setup Screen

Category	Description
SSCP Setup	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 & Call – Presence

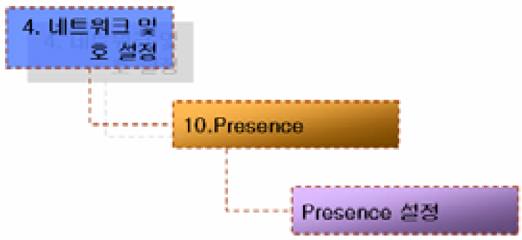
When AP-IP300 is connected with Presence Server, the IP300 can take the Speed Button Key from the server. The LED of each speed button key is changed on real time basis, so the present status of the user can be informed. To be connected with Presence Server, you should know the address of the server,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

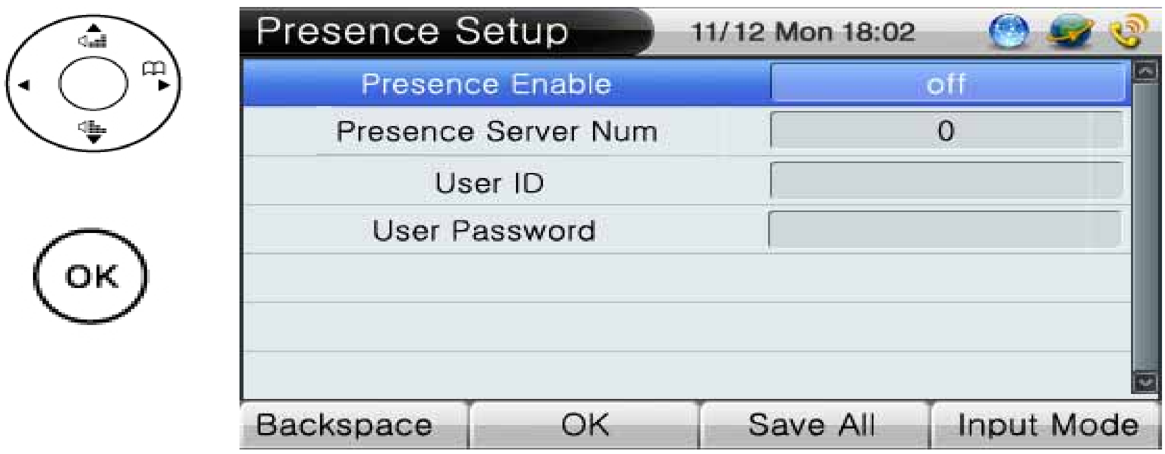


(Figure 4-58) Presence Menu Screen

Network & Call – Presence – Presence

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.





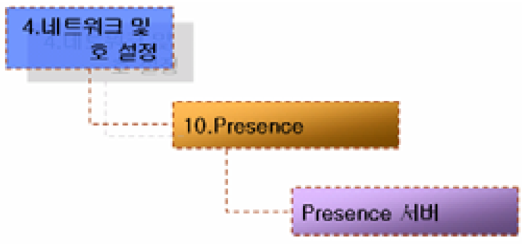
(Figure 4-59) Presence Setup Menu Screen

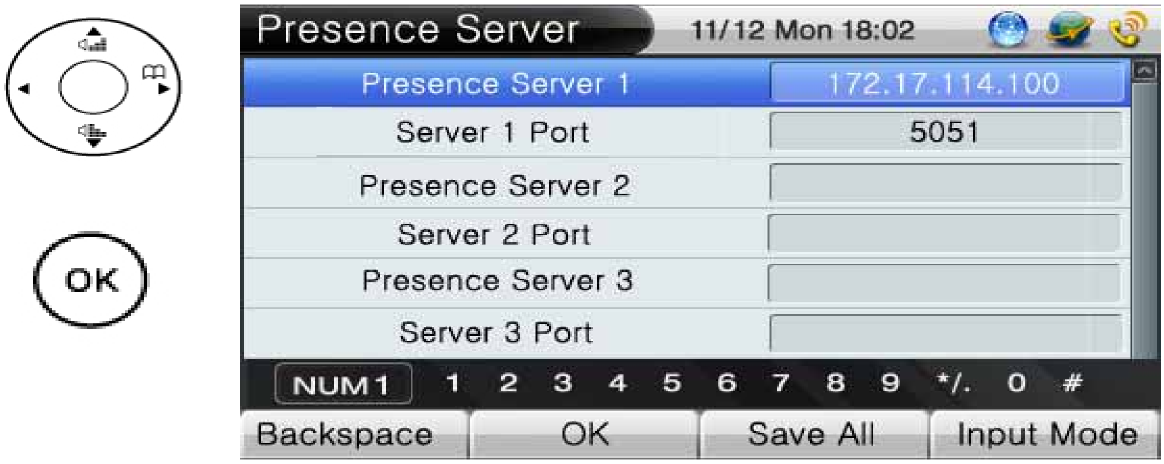
[Table 4-36] Description of Presence Setup Menu Screen

Category	Description
Presence Enable	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 & Call – 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.





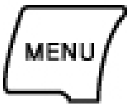
(Figure 4-60) Presence Server Menu Screen

[Table 4-37] Description of Presence Server Menu Screen

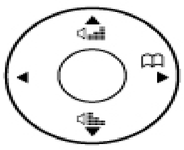
Category	Description
Server Address	Enter the IP address to be used to Presence Server at default
Server Port Number	Enter the port number to be used to Presence Server at default (Default : 5051).

Application

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 Multi-point Control Unit (MCU). For remote broadcasting, you may need the AddPac Broadcasting Equipment (AP3120) and Broadcasting management program (e-MBMS Server).



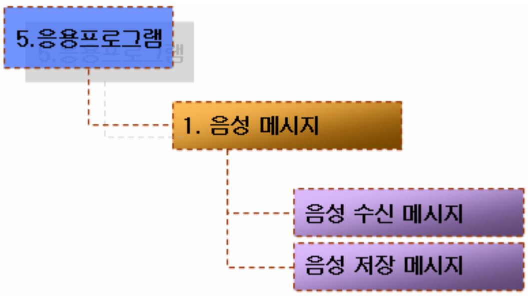
(Figure 4-61) Main Screen



(Figure 4-62) Application Menu Screen

Application – Voice Mail

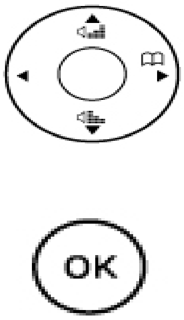
Voice Mail enables you to check the voice messages by pressing Play key. You should know the user's password which has been registered to IP-PBX to listen to the voice messages.



(Figure 4-63) Voice Mail Menu Screen

[Table 4-38] Description of Voice Mail Menu Screen

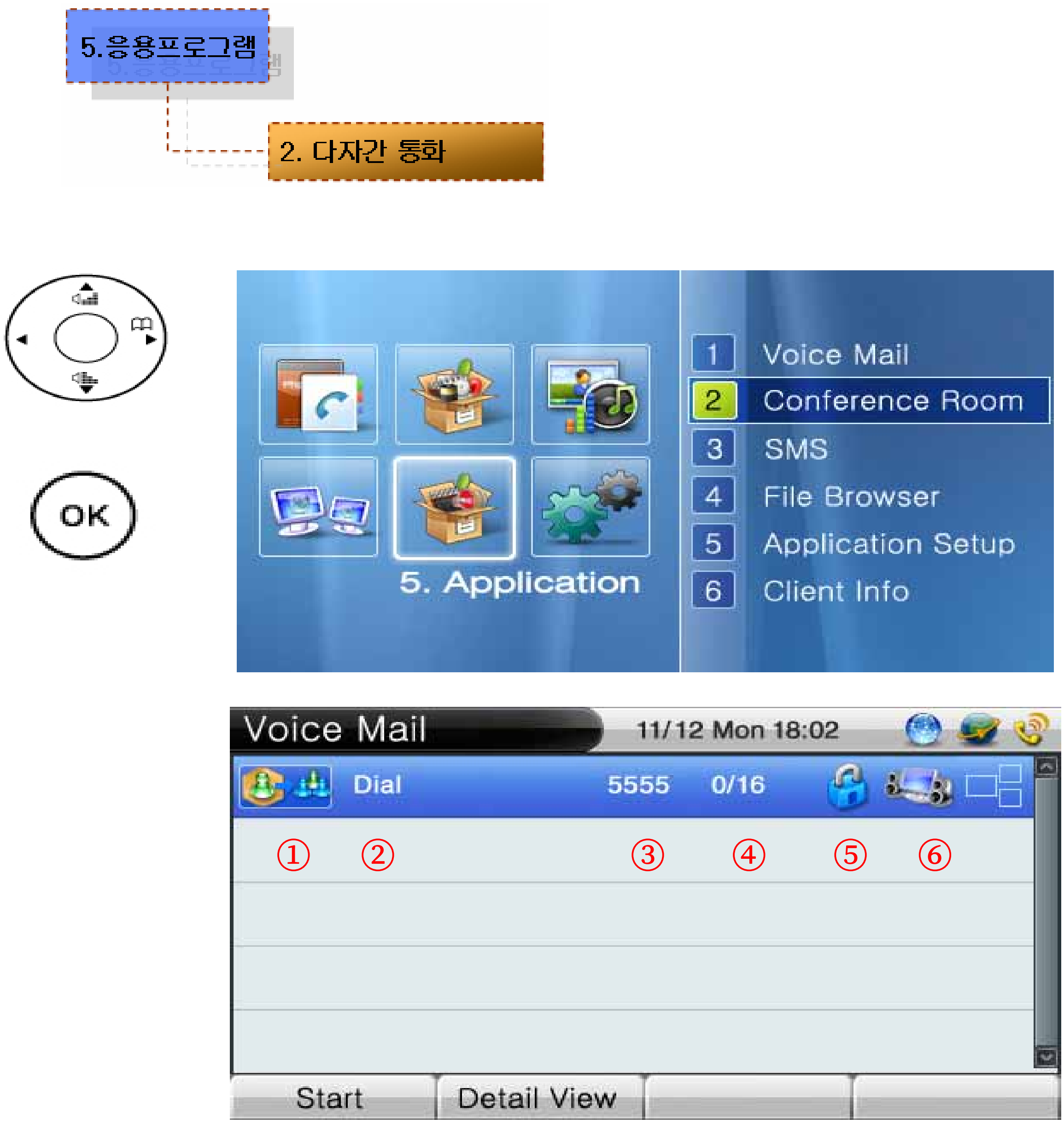
Category	Description
Voice Msg. Inbox	This is the box to keep the sent voice messages are kept
Saved Voice Msg.	This is where save the voice messages



(Figure 4-64) Saved Voice Message Screen

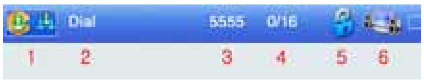
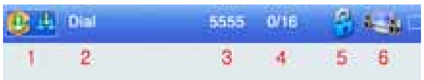
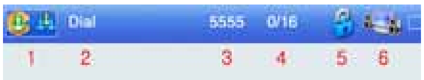
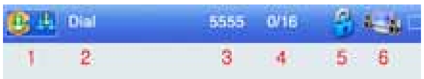
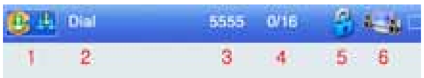
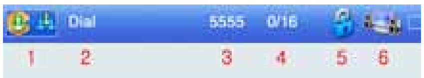
Application – Conference

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



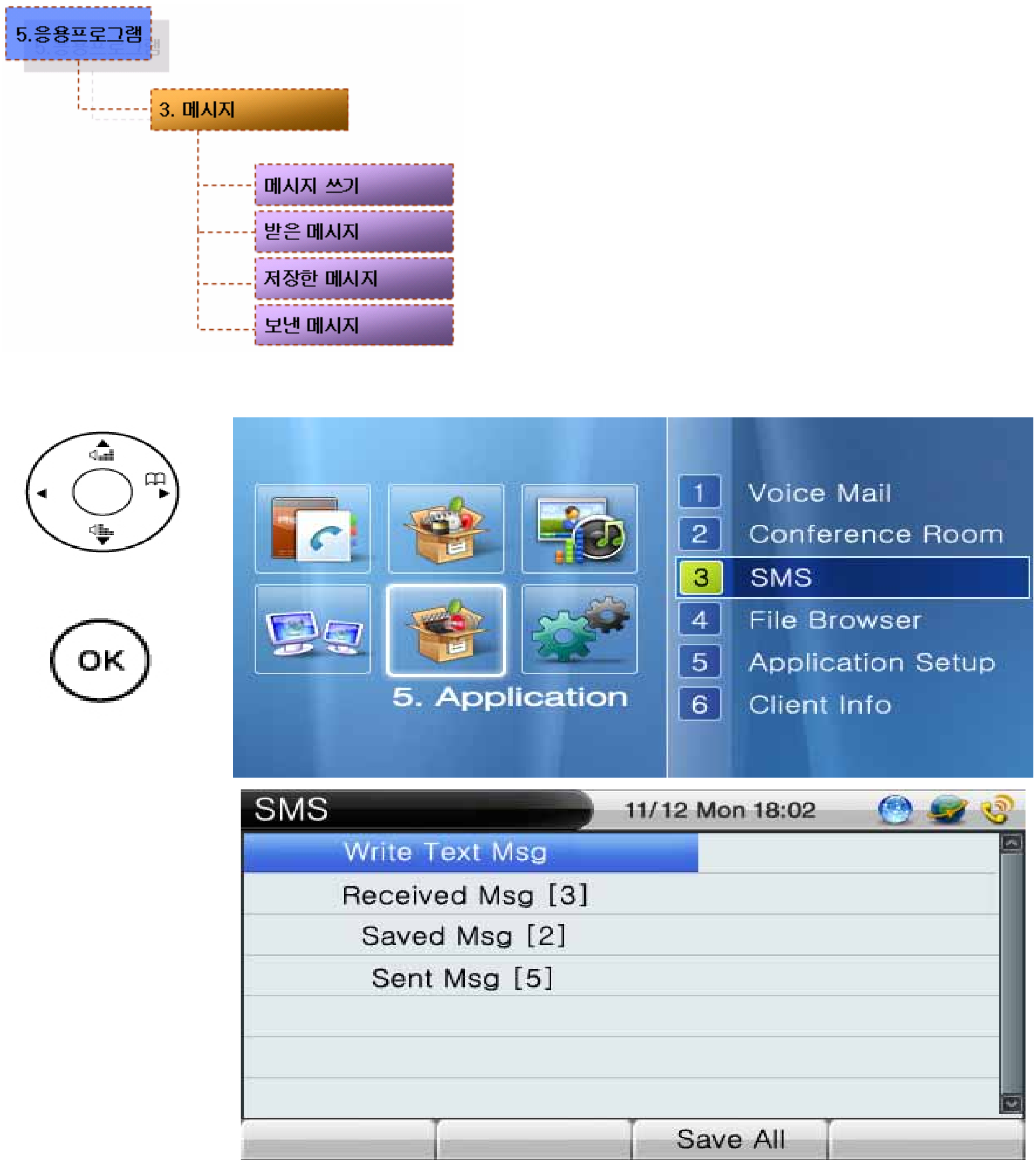
(Figure 4-65) Conference Room Menu Screen

[Table 4-39] Description of the Conference Room Menu Screen

Category	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 – SMS

The Message allows you to transmit and verify SMS text messages between one terminal and the other, when it is connected to SSCP. The messages can be sent to 9 recipients at same time. The received messages are kept as pop-up notice in the desktop area.



(Figure 4-66) SMS Menu Screen

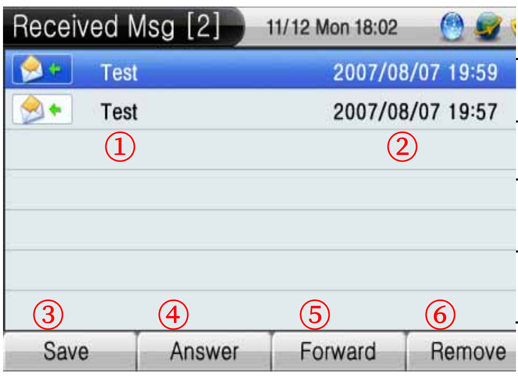
[Table 4-40] Description of SMS Menu Screen

Category	Description
Write Text Message	Write SMS messages to be sent
Received Message	Store the SMS messages in the box which have been received
Saved Message	Save the message in the box which have been sent
Sent Message	Store the messages in the box which have been sent

Write and Send SMS

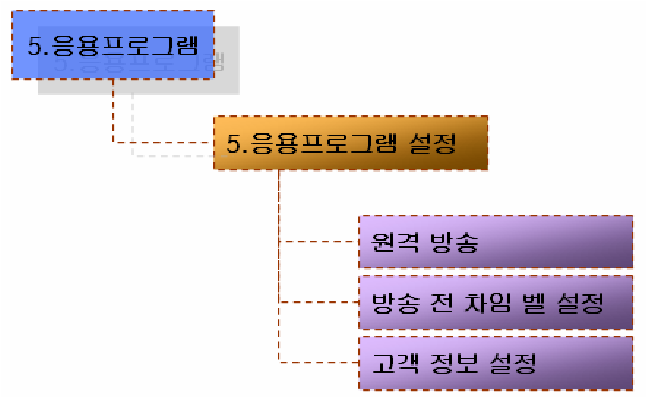
Category	Description
	Enter the messages in this field. (F4 : Changing from the alphabetic characters to the numeric and the numeric to the alphabetic) * When Korean is selected for the language, the texts change from Korean to English, English to number in order
	Enter the sender's telephone number to be displayed on the receiver's phone. Up to 9 messages can be transmitted at same time

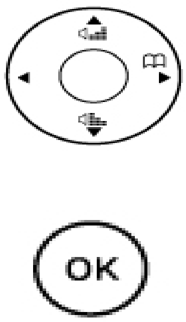
Received Message Box

Category	Description
	1. the content of the message
	2. The date and time of the message received
	3. Saving the received SMS message in the box
	4. Answering the message and send it to the sender
	5. Forwarding the message to the third party
	6. Erasing the SMS message

Application – Application Setup

The Application Setup menu consists of Remote Broadcast, Chime and Client Information. To use Remote Broadcast, you need the AddPac Broadcasting Equipment (AP3120) and broadcasting management program (e-MBMS Server). To preview Client Information, you need to save the client information and interoperate with the server.





(Figure 4-67) Application Setup Menu Screen

[Table 4-41] Description of Application Setup Menu Screen

Category	Description
Remote Broadcast	Set up the IP, Port, ID, Password for the remote broadcast
Chime Setup	Select Chime bell on/off prior to the broadcasting
Client Information Setup	Set up the IP, port for Client Information

Application - Application – Remote Broadcast

The menu sets up e-MBMS Server address and Port, User ID, Password for the Remote Broadcast. The Remote Broacast can be used in the environment installed with the AddPac Broadcasting Equipment (AP3120) and Broadcasting Management Program (e-MBMS Server).

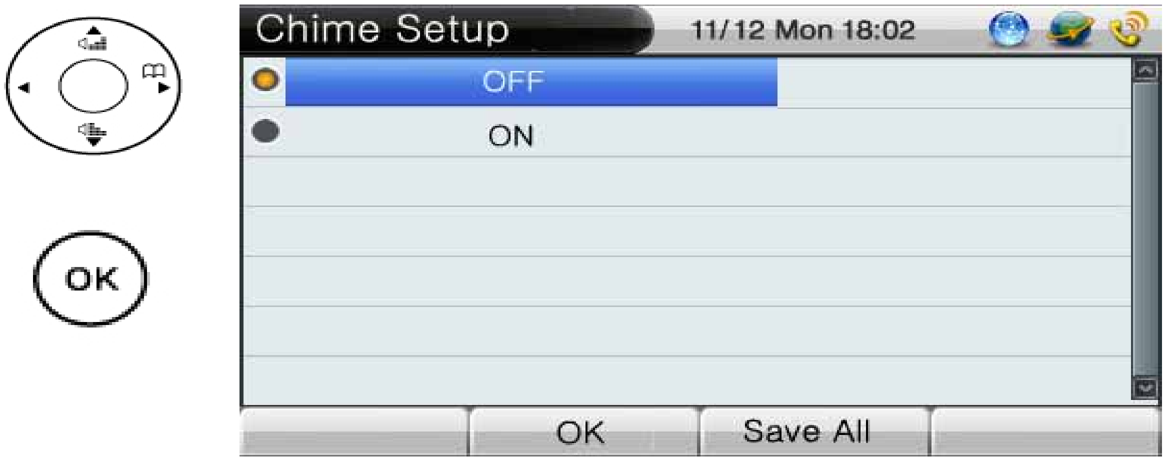
(Figure 4-68) Remote Broadcast Menu Screen

[Table 4-42] Description of Remote Broadcast Menu Screen

Category	Description
Address	Enter the IP address of e-MBMS Server
Port	Enter the port number of e-MBMS Server (Default: 8089)
ID	Enter the ID to access to e-MBMS Server
Password	Enter the password to access to the e-MBMS Server

Application - Application Setup – Chime Setup

This menu sets up or cancels the Chime Bell for starting up the Remote Broadcast.



(Figure 4-69) Remote Broadcast Menu Screen

[Table 4-43] Description of Remote Broadcast Menu Screen

Category	Description
Off	Not to transmit the Chime Bell for starting the Remote Broadcast
On	Transmit the Chime Bell for starting the Remote Broadcast

Application - Application Setup – Client Info Setup

When to take a call, this menu displays a clients information, so you can check the sender's name, resident's ID and home address. From Client Info Setup menu you can set up the IP, port, enabling viewing the client's information, and Auto CRM pop-up.

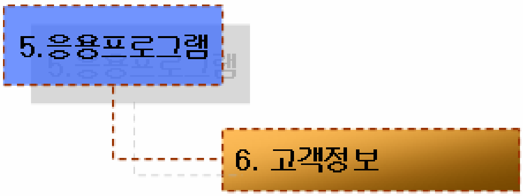
(Figure 4-70) Client Info Setup Menu Screen

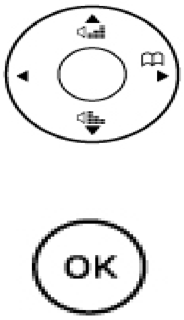
[Table 4-44] Description of Client Info Setup Menu Screen

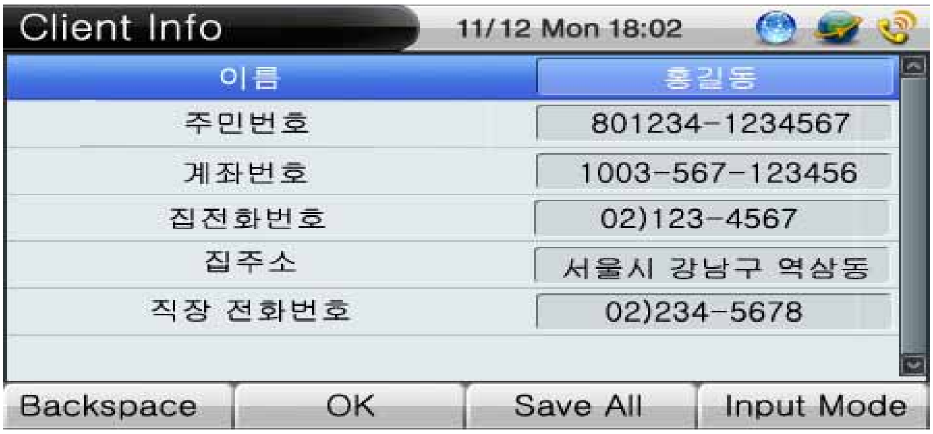
Category	Description
Address	Enter the IP address of the CRM (Client info) server
Port	Enter the port of the CRM (Client info) server
Use CRM	View the Client Info on/off
Use Auto CRM Popup	Automatic Pop-up of Client Info on/off

Application – Client Info

When to take a call, this menu displays a clients information, so you can check the client's name, resident's ID and home address.





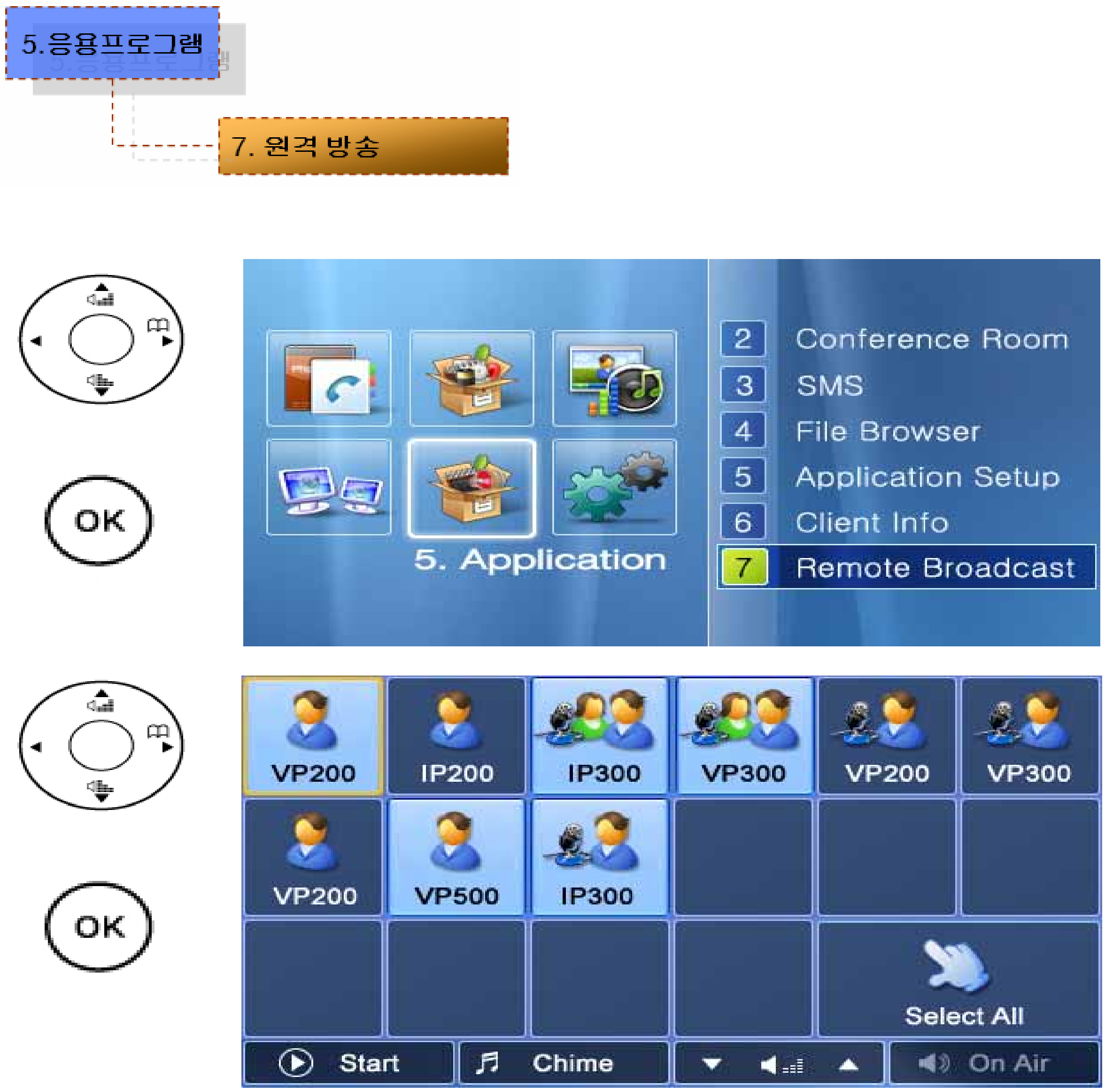


(Figure 4-71) Client Info Menu Screen

Application – Remote Broadcast

The Remote Broadcast Menu enables the user to check the broadcasting status of each terminal. Through this menu, the user can select the terminals to be participating in broadcasting and control start and end, Chime Bell setup, volume adjustment and transmission

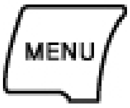
Use the navigation and OK keys to select the broadcasting terminal and then press F1 to start broadcasting (Press F2 to transmit Chime Bell, press F3 to adjust the volume of broadcasting)



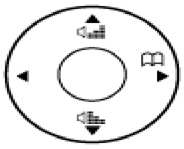
(Figure 4-72) Remote Broadcast Screen

System Setup Menu

The System Setup Menu can set up the name and number to be displayed on the desktop area and disable or enable the speed dial number to send out a call.



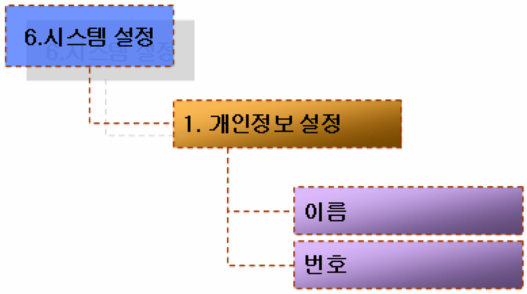
(Figure 4-73) Main Screen

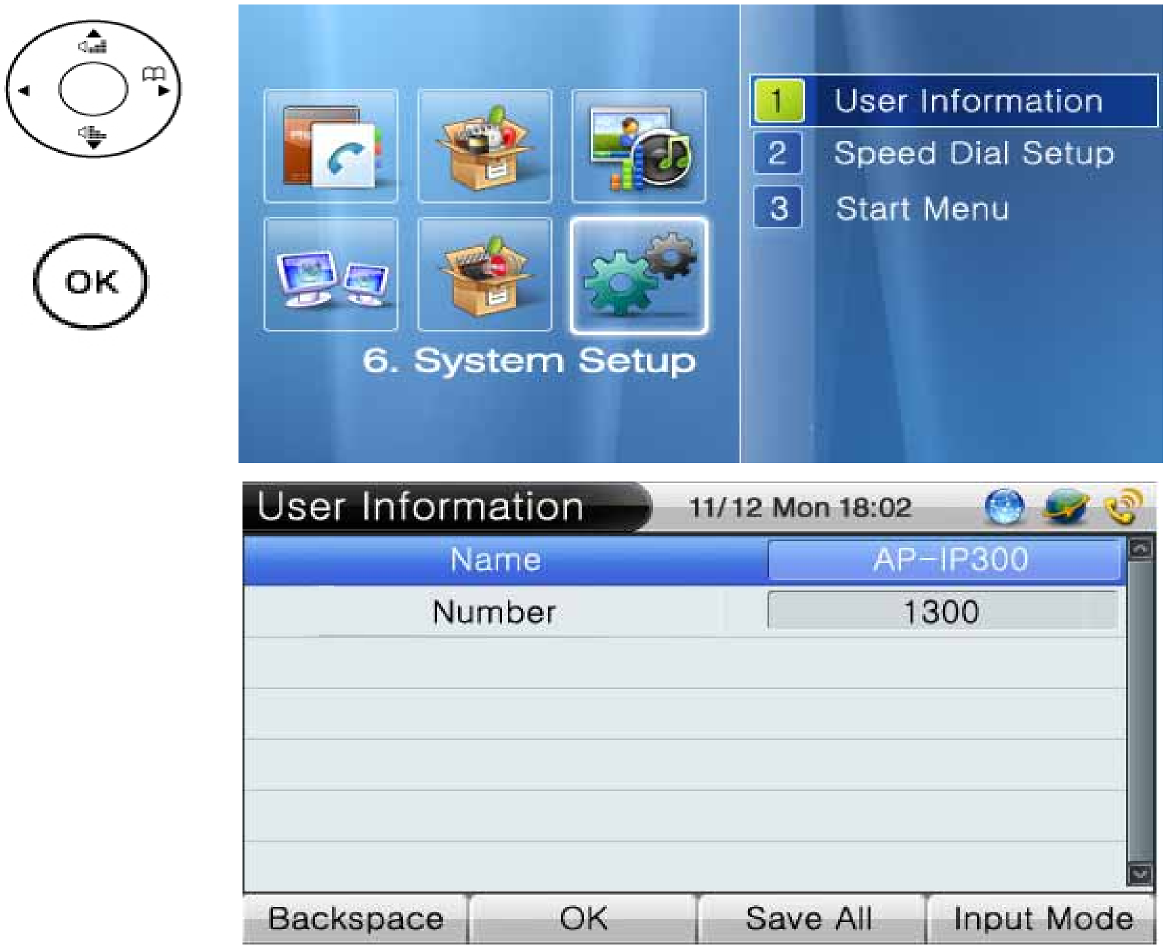


(Figure 4-74) System Setup Menu Screen

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 have been set up, the name and number are displayed on the desktop area, which are taken from the IP-PBX.





(Figure 4-75) User Information Menu Screen

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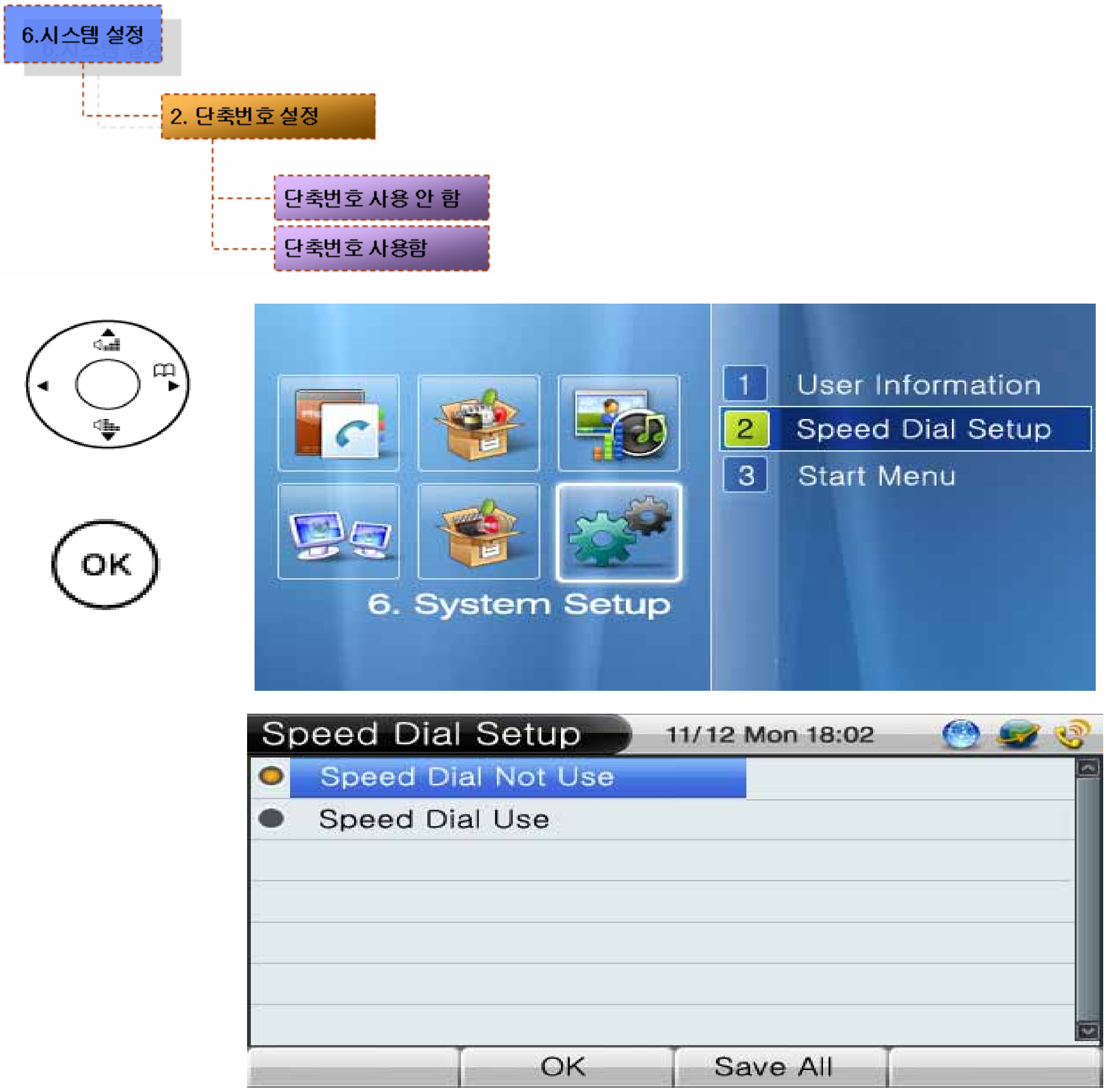
[Table 4-45] Description of User Information Menu Screen

Category	Description
Name	Enter the name to be displayed on the desktop area
Number	Enter the number to be displayed on the desktop area

System Setup – Speed Dial Setup

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.



(Figure 4-76) Speed Dial Setup Menu Screen

[Table 4-46] Description of Speed Dial Setup Menu Screen

Category	**Description**
Speed Dial Not Use	Not using the Speed Dial
Speed Dial Use	Using the Speed Dial

System Setup – 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.

(Figure 4-77) Start Menu Screen

[Table 4-47] Description of Start Menu Screen

Category	Description
Idle	Using the default screen
Speed Button	Using the Speed Button Map

Chapter 5. Testing Operation

Booting Procedure and Operating Bases

The things that you have to know, prior to turning on the power of AP-IP300, are the booting procedure.

- AP-IP300 checks its basic operation of its interface, memory and CPU through a self-testing procedure

- After the Boot Loader is started, the IP100 looks for the appropriate software image file. At the default configuration, AP-IP300 is set to load the software in the Flash Memory

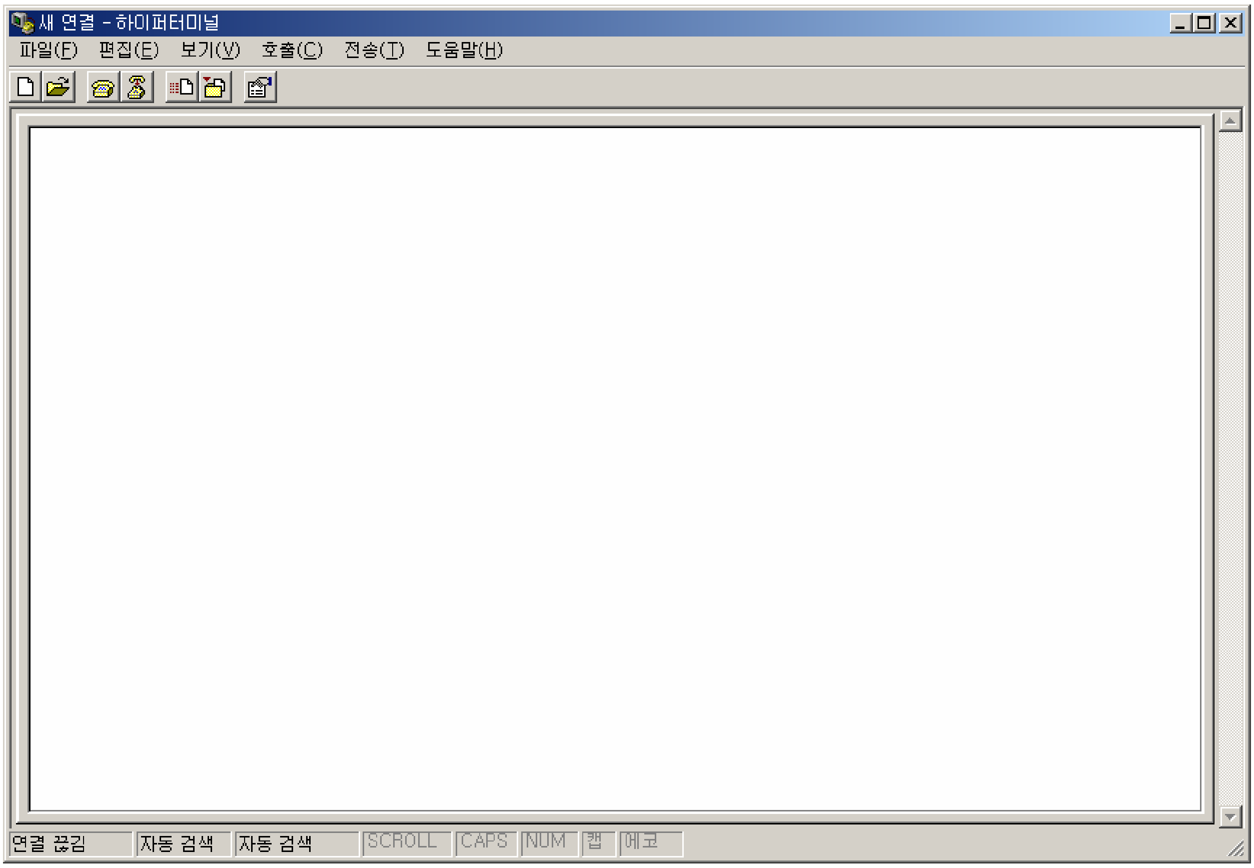
- If the IP100 fails to find the appropriate software image file, the Boot Loader stands by until the appropriate file is found to be downloaded from the appropriate system, at the booting mode.

- Once the software is downloaded, the IP-100 is to operate basing on the information of the configuration which is saved. If there is no information of the configuration, the IP100 is to operate with the initial setups. The operator needs to set up the related functions for the network operation

When the power is 110voltage, you have to use the power cable of 110. Since the AddPac AP-IP300 IPO Phone can recognize which is 110 or 220 voltage automatically, all you need to do is to use the suitable type of the power cable. There is no need for an additional setups separately.

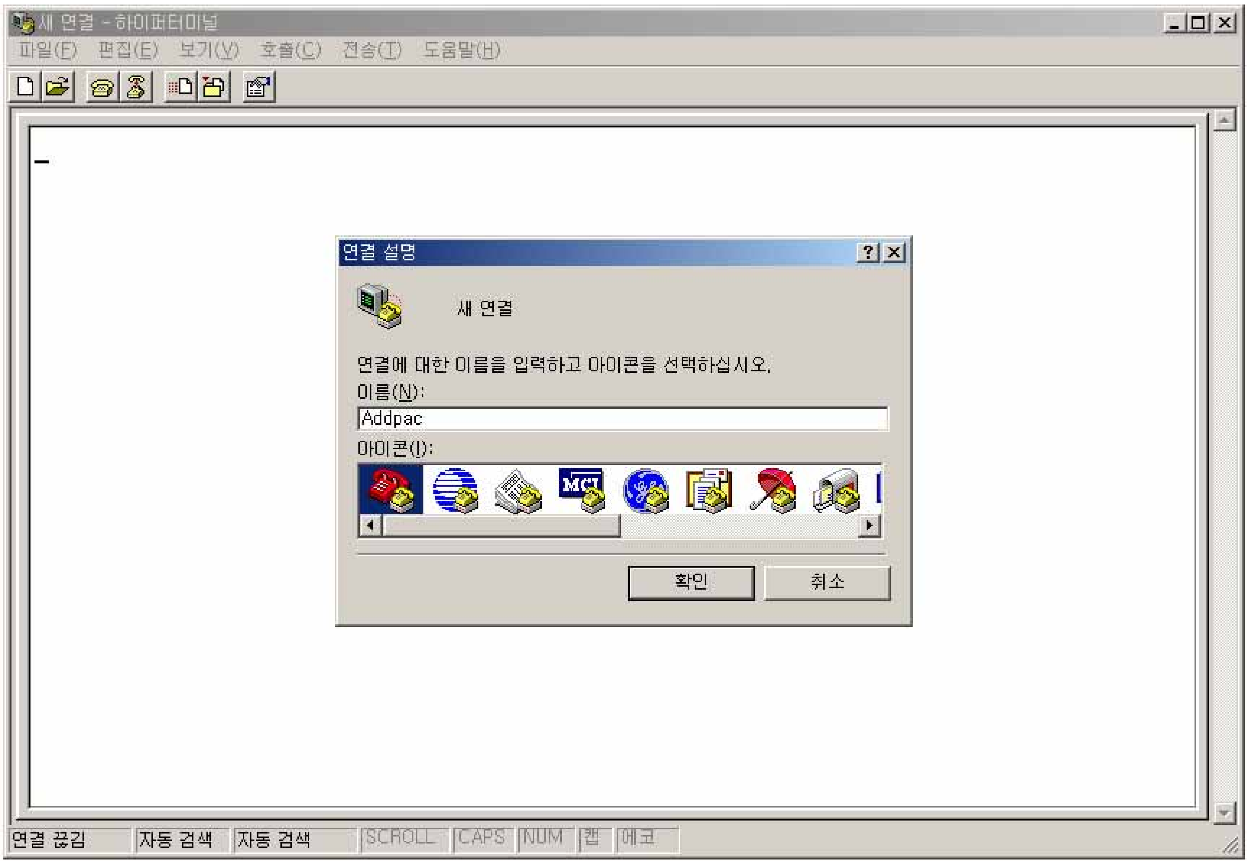
Using the Hyper-Terminal for the Console

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



(Figure 5-1) Terminal Emulator HyperTerminal of MS-Windows

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



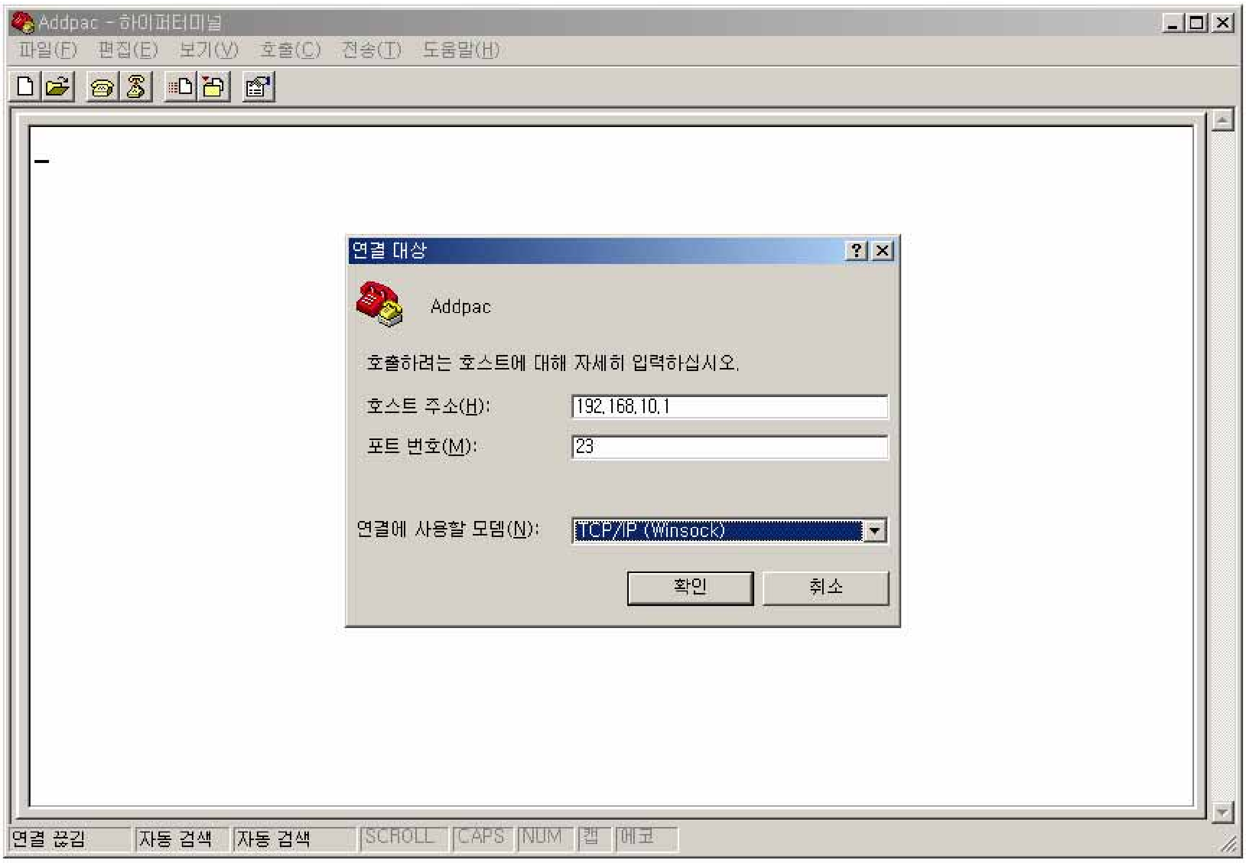
(Figure 5-2) Entering the name of the connection in HyperTerminal

Select the interface of which the console cable is connected

Since AP-IP300 does not support Console Interface, the IP Address of LAN1 interface is used to connect PC as it is shown in the following figure:

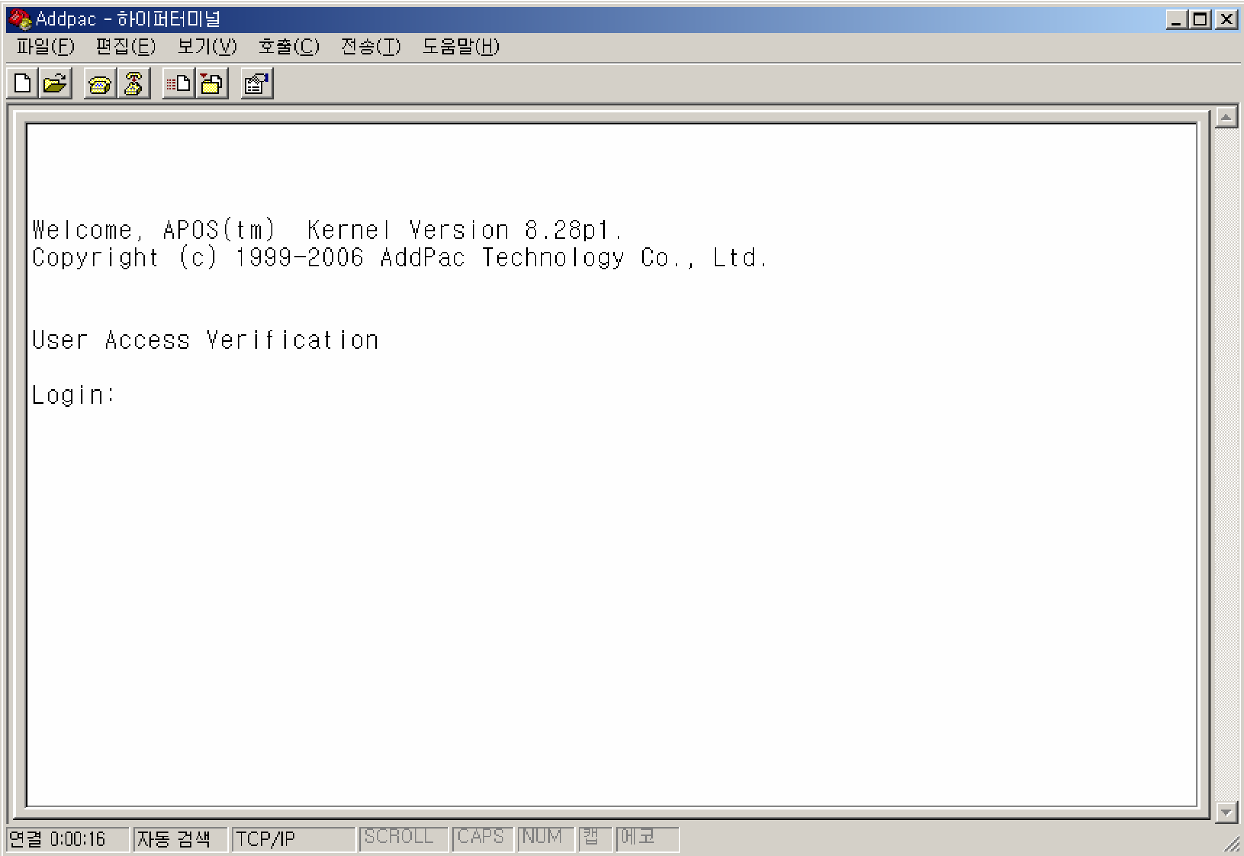
(LAN1 default ip address : 192.168.10.1)

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



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

After completing the setup, you can see the login screen of AP-IP300 in the following Figure



(Figure 5-4) Login Screen

You may see the 2 types of prompt: ‘AP-IP300>’ and ‘AP-IP300#’. The prompt with starting with ‘>’ means that the user, who logged in, has the least privilege of ‘admin’ only. He/she is not allowed to change the settings of the IP Phone. The prompt with ‘#’ means that the user, who logged in, had the privilege of ‘admin’. He/She is allowed to use all the functions of the IP Phone.

When you logged in with Admin, you can change all the settings of the IP Phone, Therefore it is recommended to change the account password of the initial value for the security reason.

Using 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-300>’.

[Table 5-1] Commands for the General User Mode

Commands	Description
enable	Change to the administrator mode
exit	Mover 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 Commands

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 'IP300#'.

[Table 5-2] anagement Mode Commands

Commands	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

Configuring Password

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 Setup

AP-IP300# **configure terminal**
AP-IP300(config)#
AP-IP300(config)# **enable password {password}**
AP-IP300(config)#

Configuring Host Name

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] Configuring Host Name

AP-IP300# **configure terminal**
AP-IP300(config)#
AP-IP300(config)# **hostname {name}**
AP-IP300(config)#

User Administration

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

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

[Table 5-5] User Administration

AP-IP300# **configure terminal**
AP-IP300(config)#
AP-IP300(config)# **username {ID} password {password} {administrator | operator |**

user}
AP-IP300(config)#

Configuring FXS/FXO Port

* Check show run first

[Table 5-6] Configuring FXS/FXO Port

```
IP300# show run

Building configuration...
Current configuration:
version 8_42_003
hostname IP300
!
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

--- 중략 -----
!
! Voice port configuration.
!
! SPEECH
voice-port 0/0
!
  ! FXS                        => If both FXS and FXO ports present, choose FXO
voice-port 0/1                 => checking voice-port FXS/FXO (port number 0/1)!
! Pots peer configuration.
```

```
!
dial-peer voice 0 pots
 destination-pattern 1004
 port 0/0
!

IP300#
IP300# con t
```

IP300(config)# **dial-peer voice 1 pots** ***=> FXS/FXO port dial peer configuration***

IP300(config-dialpeer-pots-1)# **destination-pattern 1014** ***Assigning the port numbers of FXS/FXO***

IP300(config-dialpeer-pots-1)# **port 0/1** ***=> Assigning the port that has been checked from FXS/FXO voice-port***

IP300# **show run** ***(checking the configuration)***

```
Building configuration...
Current configuration:
version 8_42_003
hostname IP300
!
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

--- 중략 -----
!
```

```
! 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. Emergency 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-IP100 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 doe not have IP routing function. Therefore, PC and LAN1 of the IP300 which are used for accessing by TELNET/FTP must be connected directly.

Entering the Boot Loader mode

Since AP-IP100 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-IP100 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-IP100 and PC are connected directly to each other, the user can access to AP-IP100 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.

Initialize APOS Settings

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 Kernal 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-IP100 IP Phone allows FTP access, which is supported by the binary code, to transmit APOS image file.

APOS image of AP-IP100 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-IP100_g2_v8_41_015.bin

D:\>
D:\> ftp 172.17.201.88
Connected to 172.17.201.88.
220 IP100 FTP server (Version 8.41.015) 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-IP100_g2_v8_41_015.bin
200 PORT command successful.
150 Opening BINARY mode data connection for ' AP-IP100_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-IP200 IP Phone.

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

[RS-232C Console Port Signal & Pinout]

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

[Table 7-1] The signal and Pinout specification

RS-232C Console Port (DTE)	RJ-45	DB-9	Console Port (PC)
Signal	RJ-45 Pin	DB-9 Pin	Signal
RTS	1	8	CTS
DTR	2	6	DSR
TxD	3	2	RxD
GND	4	5	GND
GND	5	5	GND
RxD	6	3	TxD
DSR	7	4	DTR
CTS	8	7	RTS

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

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

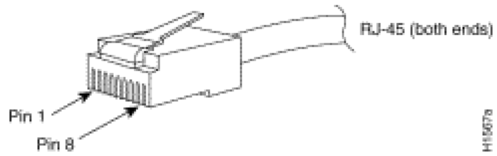


Figure 7-1 100Base-TX RJ-45 Connector

[Table 7-2] Signal and Pinout of Direct Ethernet Cable

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

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

Acronyms and Glossary

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

	cable modem can be used to achieve more bandwidth. Cable network also requires modularization and demutualization process while sending the data.
Call Center	A call center is a central place where customer and other telephone calls are handled by an organization, usually with some amount of computer automation. Typically, a call center has the ability to handle a considerable volume of calls at the same time, to screen calls and forward them to someone qualified to handle them, and to log calls. Call centers are used by mail-order catalog organizations, telemarketing companies, computer product help desks, and any large organization that uses the telephone to sell or service products and services.
Caller ID	A feature that displays the name and/or number of the calling party on the phone's display when an incoming call is received. Virtually all digital phones - as well as many analog phones - have this capability. While typically only the number is received, most phones will display the name, if the number matches an entry in the phone's built-in phone book.
Category 5 cabling	unshielded twisted pair (UTP) cabling. An Ethernet network operating at 10 Mbits/second (10BASE-T) will often tolerate low quality cables, but at 100 Mbits/second (10BASE-Tx) the cable must be rated as Category 5, or Cat 5 or Cat V, by the Electronic Industry Association (EIA).
CBR	Constant Bit Rate. A data transmission that can be represented by a non-varying, or continuous, stream of bits or cell payloads. Applications such as voice circuits generate CBR traffic patterns. CBR is an ATM service type in which the ATM network guarantees to meet the transmitter's bandwidth and Quality of Service requirements
CES	An acronym for Circuit Emulation Service. enables users to multiplex or to concentrate multiple circuit emulation streams for voice and video with packet data on a single, high-speed ATM link without a separate ATM access multiplexer.
Checksum	A computed value which is dependent upon the contents of a packet. This value is sent along with the packet when it is transmitted. The receiving system computes a new checksum based upon the received data and compares this value with the one sent with the packet. If the two values are the same, the receiver has a high degree of confidence that the data was received correctly.
Coaxial cable	A cable with a single inner conductor with foam insulation and braided shield. There are two types of this cable; 50Ω cable for digital signaling process and 75Ω cable for analog signal process and high speed digital signal process.
CODEC	An acronym for COder-DECoder 1. Built-in circuit device for coding/decoding

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

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

	startup and request admission to a call from the gatekeeper.
H.225	An International Telecommunication Union (ITU-T) standard for H.225.0 session control and packetization. It defines various protocols of RAS, Q.931, RTP and etc.
H.245	An International Telecommunication Union (ITU-T) standard for H.245 end-point control.
H.323	An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing.
HBD3	Line code type of E1 line.
HDLC	An acronym for High-Level Data Link Control. A transmission protocol for the Data Link Layer. In HDLC, data is organized into a unit (called a frame) and sent across a network to a destination that verifies its successful arrival. Variations of HDLC are also used for the public networks that use the X.25 communications protocol and for frame relay, a protocol used in both and wide area network, public and private.
Hookflash	Short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone is attempting to perform a dial-tone recall from a PBX. Hookflash is often used to perform call transfer.
HTTP	An acronym for Hypertext Transfer Protocol. A file transfer protocol used by web browser or web server for transmitting text or graphic files.
IPSec	Internet Protocol Security protocol, a framework for a set of protocols for security at the network or packet processing layer of network communication. Earlier security approaches have inserted security at the Application layer of the communications model. IPsec is said to be especially useful for implementing virtual private networks and for remote user access through dial-up connection to private networks. A big advantage of IPsec is that security arrangements can be handled without requiring changes to individual user computers. Cisco has been a leader in proposing IPsec as a standard (or combination of standards and technologies) and has included support for it in its network routers.
IPv6	IPv6 (Internet Protocol Version 6) is the latest level of the Internet Protocol (IP) and is now included as part of IP support in many products including the major computer operating systems. IPv6 has also been called "IPng" (IP Next Generation). Formally, IPv6 is a set of specifications from the Internet Engineering Task Force (IETF). IPv6 was designed as an evolutionary set of improvements to the current IP Version 4. Network hosts and intermediate nodes with either IPv4 or IPv6 can handle packets formatted for either level of the Internet Protocol. Users and service providers can update to IPv6

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

	LANs but smaller than WANs. Compare with: LAN, WAN.
MGCP	MGCP, also known as H.248 and Megaco, is a standard protocol for handling the signaling and session management needed during a multimedia conference. The protocol defines a means of communication between a media gateway, which converts data from the format required for a circuit-switched network to that required for a packet-switched network and the media gateway controller. MGCP can be used to set up, maintain, and terminate calls between multiple endpoints. Megaco and H.248 refer to an enhanced version of MGCP
NAT	NAT (Network Address Translation) is the translation of an Internet Protocol address (IP address) used within one network to a different IP address known within another network. One network is designated the inside network and the other is the outside.
NTP	Network Time Protocol (NTP) is a protocol that is used to synchronize computer clock times in a network of computers. In common with similar protocols, NTP uses Coordinated Universal Time (UTC) to synchronize computer clock times to a millisecond, and sometimes to a fraction of a millisecond.
PABX	Private Automatic Branch Exchange. A telephone switch for use inside a corporation. It connects offices (internal extensions) with each other and provides access (typically by dialing an access number such as 9) to the public telephone network PABX is the preferred term in Europe, PBX is used in the USA.
Packet	Packets contain a source and destination address as well as the actual message. Packets also known as Datagrams.
PBX	A PBX (private branch exchange) is a telephone system within an enterprise that switches calls between enterprise users on local lines while allowing all users to share a certain number of external phone lines.
PING	Packet INternet Groper, a packet (small message) sent to test the validity / availability of an IP address on a network
Point to Point Connection	Basic connection type. In ATM, point to point connection is half duplex connection between two ATM end systems or full duplex connection.
Pont to Multipoint Connection	Basic connection type. In ATM, point to multipoint connection is half duplex connection among one sending end system (root node) and multiple receiving end system. Compare with: point-to-point connection.
POTS	Plain Old Telephone Service. Compare with: PSTN.
PPP	The most popular method for transporting IP packets over a serial link between the user and the ISP. Developed in 1994 by the IETF and superseding the SLIP protocol, PPP establishes the session between the

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

	2. Rapid Transport Protocol. Provides pacing and error recovery for APPN data as it crosses the APPN network. With RTP, error recovery and flow control are done end-to-end rather than at every node. RTP prevents congestion rather than reacts to it. 3. Real-Time Transport Protocol. Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time-stamping, and delivery monitoring to real-time applications.
SIP	The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality. Like HTTP or SMTP, SIP works in the Application layer of the Open Systems Interconnection (OSI) communications model. The Application layer is the level responsible for ensuring that communication is possible. SIP can establish multimedia sessions or Internet telephony calls, and modify, or terminate them. The protocol can also invite participants to unicast or multicast sessions that do not necessarily involve the initiator. Because the SIP supports name mapping and redirection services, it makes it possible for users to initiate and receive communications and services from any location, and for networks to identify the users whatever they are. SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol, such as UDP, SCTP, or TCP. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination. The Session Initiation Protocol is specified in IETF Request for Comments [RFC] 2543.
SmartViewer	The real-time monitoring, statistical data search and management GUI based software developed by AddPac Technology for AP-GK1000, AP-GK2000, AP-GK3000 models.
SNMP	Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security. Refer to: SGMP, SNMP2.
T1	A TDM physical transmission standard consisting of two twisted wire pairs and

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

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

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