AP-IP120 IP Phone

[Installation and Operation Guide]

March, 2009

AddPac Technology Co., Ltd.

www.addpac.com

AP-IP120 IP Phone

Note.

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Getting into AP-IP120 IP Phone Installation Guide

This chapter explains the AP-IP120 IP Phone installation guide.

[Contents of AP-IP120 Installation Guide]

The purpose of this guide is to assist the users to install the AP-IP120 IP Phone easily. This guide is composed of five 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-IP120 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-IP120 IP Phone and other basic information for the installation process.

- Chapter 4 『**How to Use AP-IP120**』 describes the UI operation of AP-IP120. 「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-IP120 IP phone.

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The revision history of AP-IP120 IP Phone installation guide is listed as to follow:

Revision History

Revision No.	Date	Contents	Written By
Version 1.00	March 4th, 2009	Initial Release	AddPac R&D Center

Chapter 1. Product Overview

AP-IP120 IP phone is designed to provide enhanced IP telephony functionality to meet the wide range of business user requirements. This cost effective IP phone optimally delivers rich featured voice telephony service on ordinary internet infrastructure as well as AddPac IP-PBX environment on local LAN as a fully featured IP extension for the complete AddPac VoIP solution.

Product Overview

New Paradigm for IP Telephony : Telephony + Broadcasting

AP-IP120 IP telephone combines AddPac's field proven VoIP technology and IP voice broadcasting technology. AP-IP120 is market-ready IP telephone which provides a full suit of remarkable functionality compared to other typical IP telephones. Apart from telephony service, it delivers IP voice broadcasting service supporting external MIC/Line-in, Line-out interface for various input/output devices such as headset, Amp or speaker. In addition, it provides high level information display with large size LCD mounted. Since AP-IP120 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 IPNext50, IPNext180, IPNext200, and IPNext500 on comprehensive IP-PBX system of AddPac Technology, it not only improves operation offering an wide variety of features such as Music on Hold, Coloring service, Call Transfer but also provides the easy-to-use, intelligent IP telephony service enhanced by MS window based Smart Messenger.

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

AP-IP120 IP telephone provides 12 key buttons (4 column x 3 row) for speed-dialing and user presence service. User Information assigned each key button is easily registered using web based management. System operator can make several example user presence profiles for AP-IP120 user. AP-IP120 user select one among several candidate user presence files and can modify selected user presence profile for individual personal user presence profile. User presence profiles provided via system operator can be upgraded via web based management, if user information is changed or deleted due to several reasons (for example, resign). Because user information is frequently changed, it is better to use default user presence profile provided by system. User presence information such as user busy(LAMP ON), ringing(LAMP BLINK) is individually displayed on LED of AP-IP120 Key

Buttons.

Adapt to the Future Environment : Firmware Upgradeable Technology

Designed on programmable high performance RISC Integrated DSP Technology, AP-IP120 is capable of adopting new capabilities and improvement by downloading firmware from website or with its auto-upgrade option as the customers' needs grow. Moreover, operators can download the latest protocol or service improvements as well as update firmware by checking the version and activating the auto-upgrade while AddPac's IP-PBX power on/booting sequence.

Compelling Supplementary Services: Extending Benefit of IP Telephony

AP-IP120 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-IP120 to offer voice broadcasting service, incorporated with in-house broadcasting system.

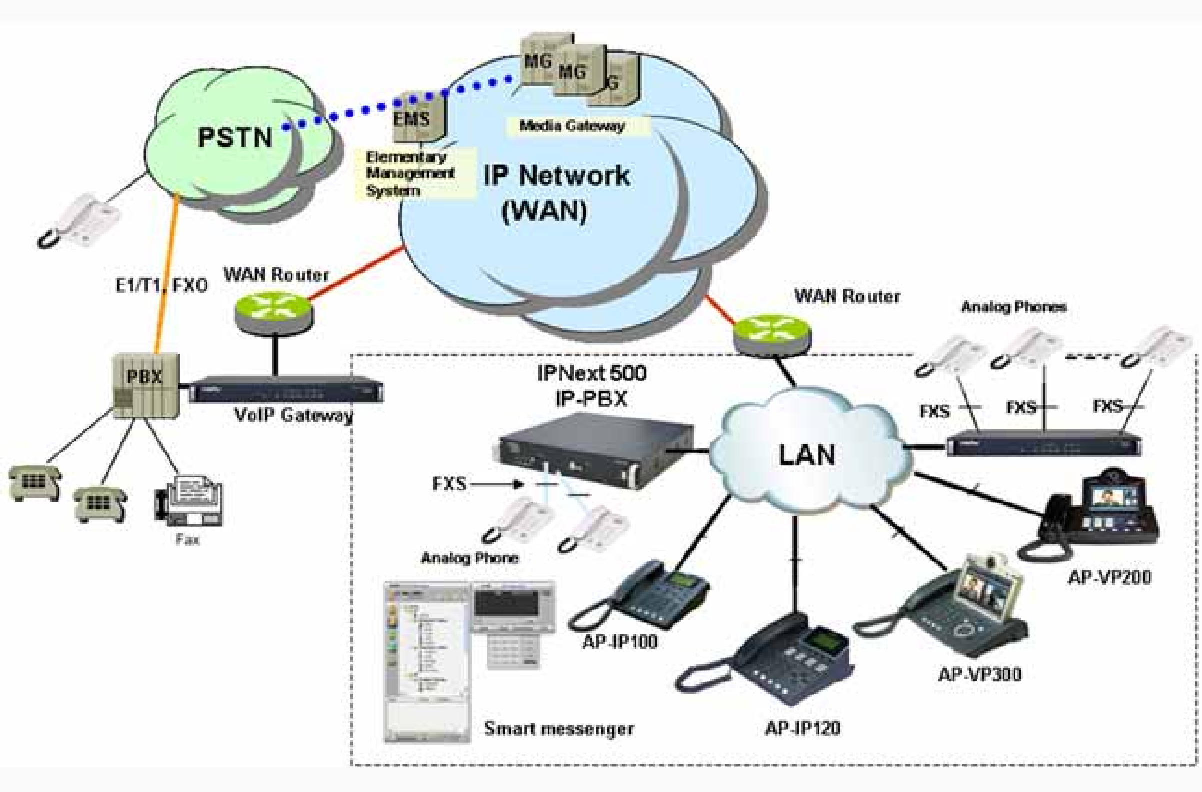
IP telephony with Outstanding Network Service Capability

Not only IP telephony, AP-IP120 is an integrated, feature-rich network equipment delivering routing, NAT/PAT, DHCP Server/Relay, Public IP sharing and QoS. In today's mixed network of xDSL, Cable, FTTH, Metro Ethernet, Metro ATM, Leased line and dynamic IP environment, not only the ample network service features, but also high-end QoS (Quality of Service) and security features are requested. Based on two (2) 10/100Mbps Fast Ethernet ports, AP-IP120 offers integrated network and security service of LAN-to-LAN routing, bridge and NAT/PAT. Moreover, AP-IP120 supports H.323, SIP dual VoIP signaling protocols concurrently. So the customers easily migrate to different service providers' networks utilizing different VoIP signaling protocols.

Privacy and Encryption Features

AP-IP120 brings the network security and service security as well. With the built-in CID (Caller ID Detection) feature, user is able to know who is calling before he answers and block the incoming call.

AddPac's various VoIP gateway series, multi-service routers, video products and comprehensive family of cutting-edge solutions have delivered high performance and stability to maximize customer satisfaction throughout the world. They provide high level of flexibility and scalability for each organization to find the solution that best fits their application needs and budget. With years of experience and industry-leading technology, AddPac provides AP-IP120 with which customers can best optimize high performance, market strategy and budget for next-generation communication solution.



(Figure 1-1) Network Configuration Diagram

Hardware Features

[Table 1-1] Hardware Specifications

Category		Specification
Model		**AP-IP120**
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, 12 Speed-Dial and Presence Indication Keys
LCD Display	Graphic LCD	4 Line Graphic LCD
Memory	Boot Memory	512Kbyte Flash Memory
	Flash Memory	4MB Flash Memory
	Main Memory	16MB High Speed SDRAM
Audio Interface	Input	One(1)-3.5mm Stereo-In Connector for Audio In
	Output	FXO Voice Interface
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
Physical Dimension		75 x 200 x 210mm
Weight		1.15Kg

Software Features

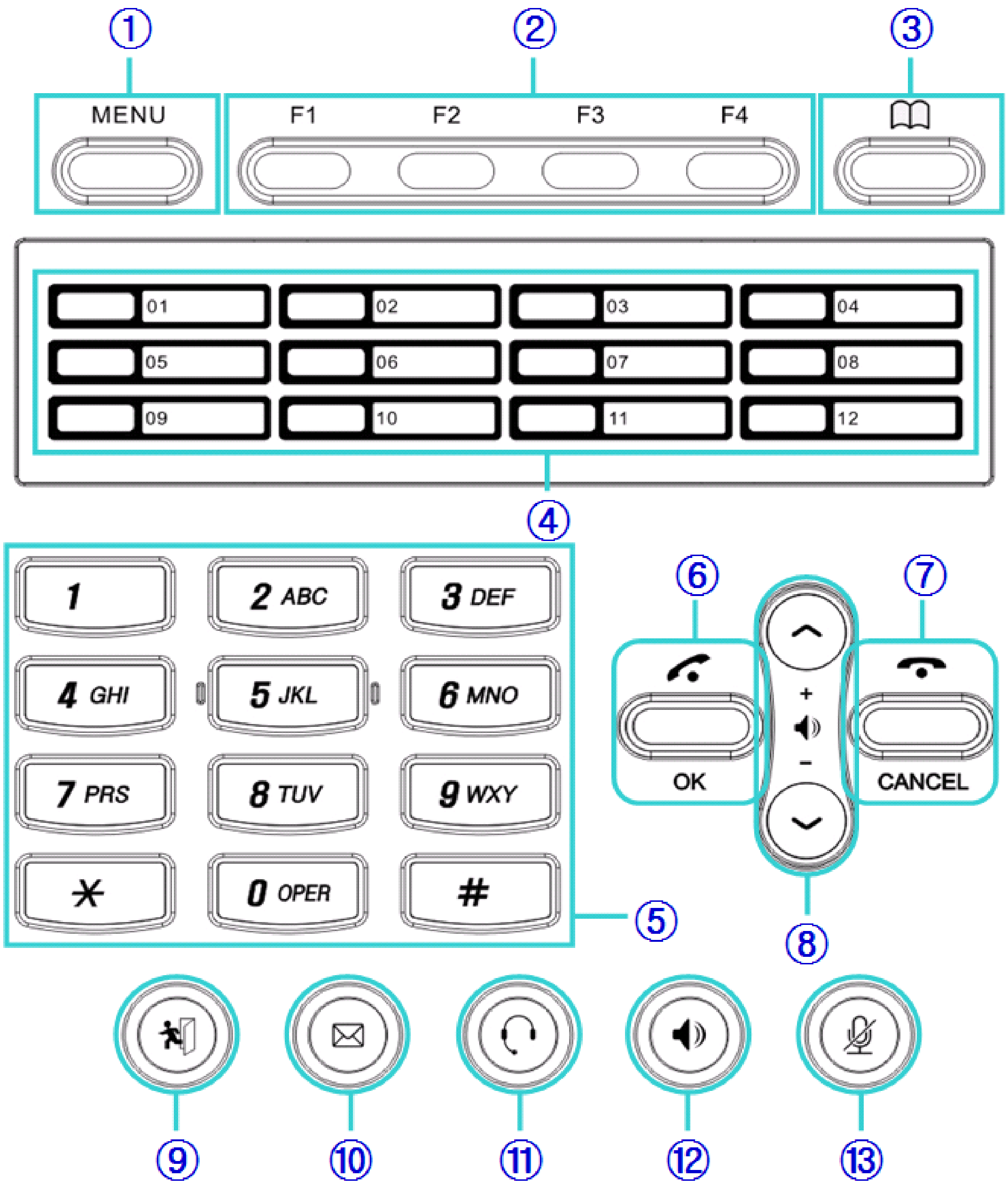
[Table 1-2] Software Specifications

Category	Specification
LAN Protocol	Static IPv4 Routing
WAN Protocol	Point-to-Point Protocol (PPPoE for ADSL), etc.
Audio Service & Signaling Protocol	Voice Codec - G.711, G.723.1, G.726, G.729, etc.
	SIP, and MGCP Triple Stack Support
	ITU-T H.235 Security Feature
	Voice Processing Features Supports - VAD, DTMF, CNG, G.168 and T.38 FAX Relay
	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 Features	Scalability	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

Physical Description-Front 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) Front View

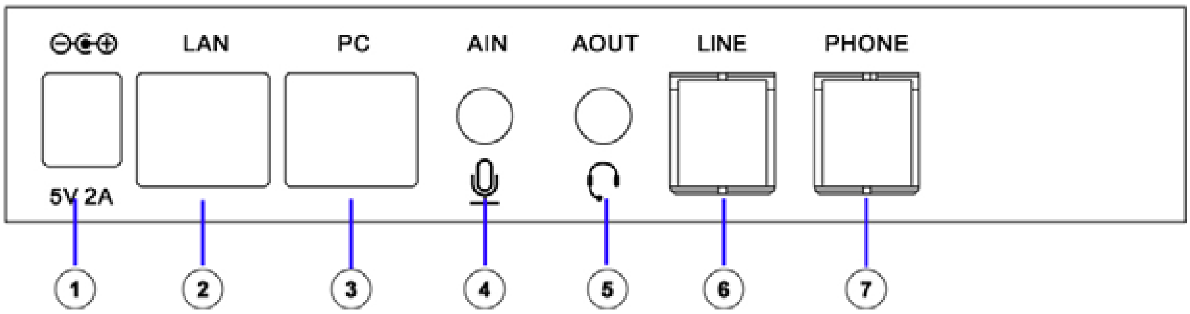
The following Table 1-3explains each button feature of AP-IP300 on the front side.

[Table 1-3] Description of the Button Features

No.	Button Names	Features
(1)	Menu	Enter the UI Main 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)	Recent Call	Brings out the recent call list
(4)	Speed Dial	Connects a call by using speed-dial which is set by the number in the Button List
(5)	Numeric Key	Used for Dialing and parameter setting in UI
(6)	OK/Send	Confirms each menu in UI or in stand-by state confirms a recent call or makes a call
(7)	Cancel/End	Moves on to upper menu from current UI menu or cancels or ends the current VoIP call
(8)	Volume/Navigation Key	Stand-by Mode: Ring Volume Menu Mode: Scrolls through text to make a selection On the Phone Mode: Adjusts output volume Volume Setting Mode : Adjusts Ring Volume , Input Volume , Output Volume
(9)	Absence	Takes an incoming call when the phone user is away from the phone
(10)	Voice Mail	Listens to the voice mail.
(11)	HDP Call	Activates a headset
(12)	SPK Call	Activates the built-in speaker phone
(13)	Privacy	Blocks the view on my side, so the other party cannot see you while you are placing a call

Physical Description-Rear View

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) Rear View

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

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

No.	Interface	Description
(1)	DC 5V 2A	External Power Adaptor connector (DC 5V 2A).
(2)	LAN	10/100Mbps Fast Ethernet Interface for WAN such as ADSL, Leased Line, etc (RJ45)
(3)	PC	10/100Mbps Fast Ethernet Interface for LAN (RJ45)
(4)	Audio Input	Input port for microphone (headset)
(5)	Audio Output	Output port for speaker (headset)
(6)	LINE	1-port PSTN backup interface (FXO) for connecting the line to central office
(7)	Phone	1-port FXS interface for analog phone

Chapter 2. Preparing for Installation

Installation Requirement

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

- Ensure AP-IP120 IP Phone is in a dust-free environment before and after installation.
- Ensure AP-IP120 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-IP120 IP Phone : the power supply and static electricity.

This section describes safety recommendations for each case.

- **Electrical Safety**

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

- **Prevention of Static Electricity**

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

Site Requirements

The AP-IP120 is ready for use where electronic products are used. However, a location with the following conditions is recommended for the maximum performance:

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

Requirements for Network Connection

This procedure is to follow EIA standards and other EMI regulations when you install the Gateway

The following section describes the Ethernet Cable, and the Console Cable which can be connected to the AP-IP120.

Required Tools and Equipment

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

LAN Cable & Console Port Cable

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

Ethernet Port

The AP-IP120 has two RJ45 type of 10/100 BaseTX Ethernet ports on the rear side and LED for indicating the status of the port on the front. These ports are physically connected and using the direct cable you can connect to LAN0/ LAN1. Please use the standard cable and connector to access to LAN. You may refer to the cable details of Appendix in this guide

.

Console Port (Optional)

AP-IP120 has one RJ-45 type3 of RS-232C Female DCE Connector Interface. Through this port, you can perform the initial setup, monitoring and debugging the system. The cable and connector must be used. You may refer to the pin connection for RS-232 console cable of appendix in this guide.

Package Contents

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

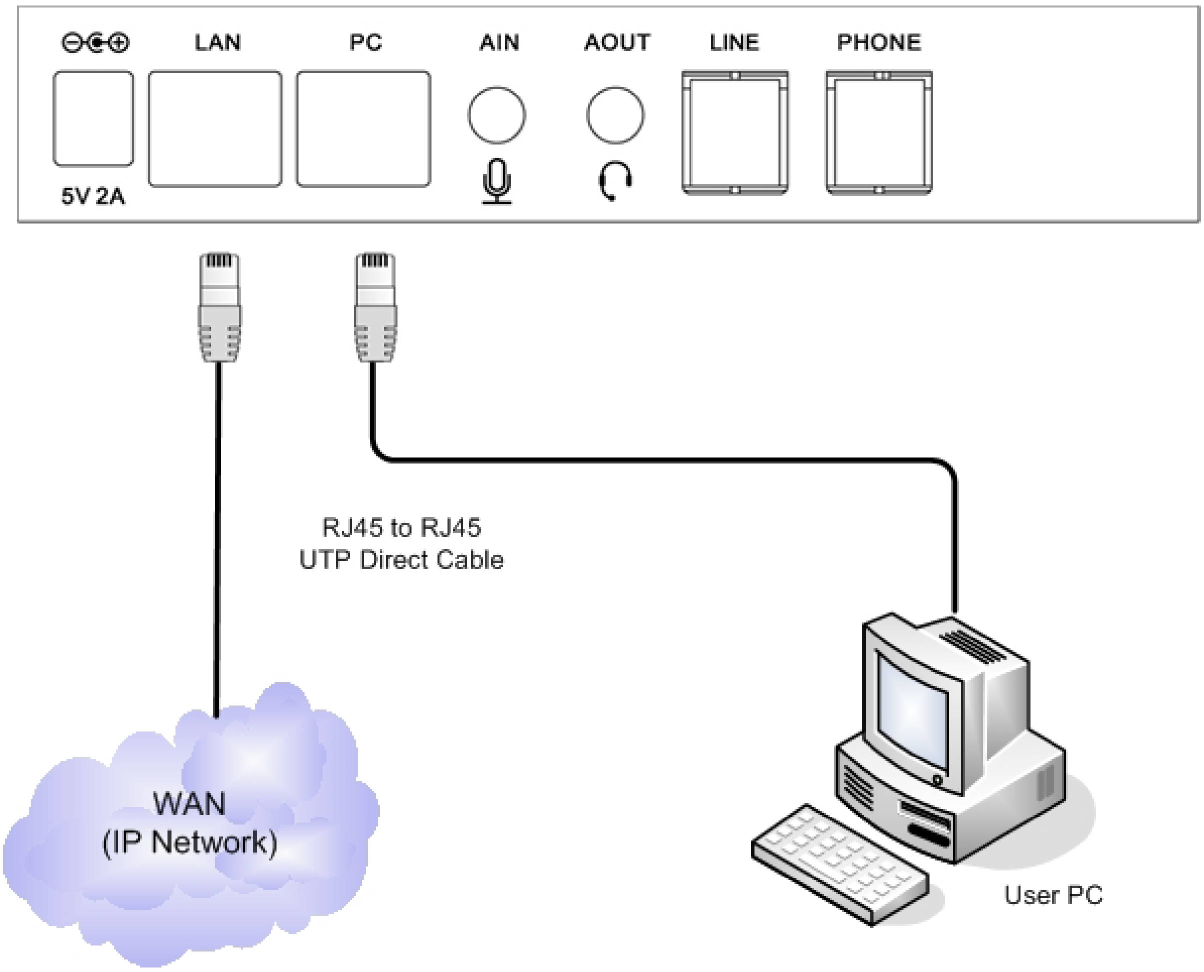
[Table 2-1] AP-IP120 Product Package

No	Name	Contents	Quantity
1	AP-IP120 Main Body		1
2	LAN Cable (RJ45 to RJ45)		1
3	External Power Adapter and Power Cable (220V Power Cord)		1

Chapter 3. **Installing**

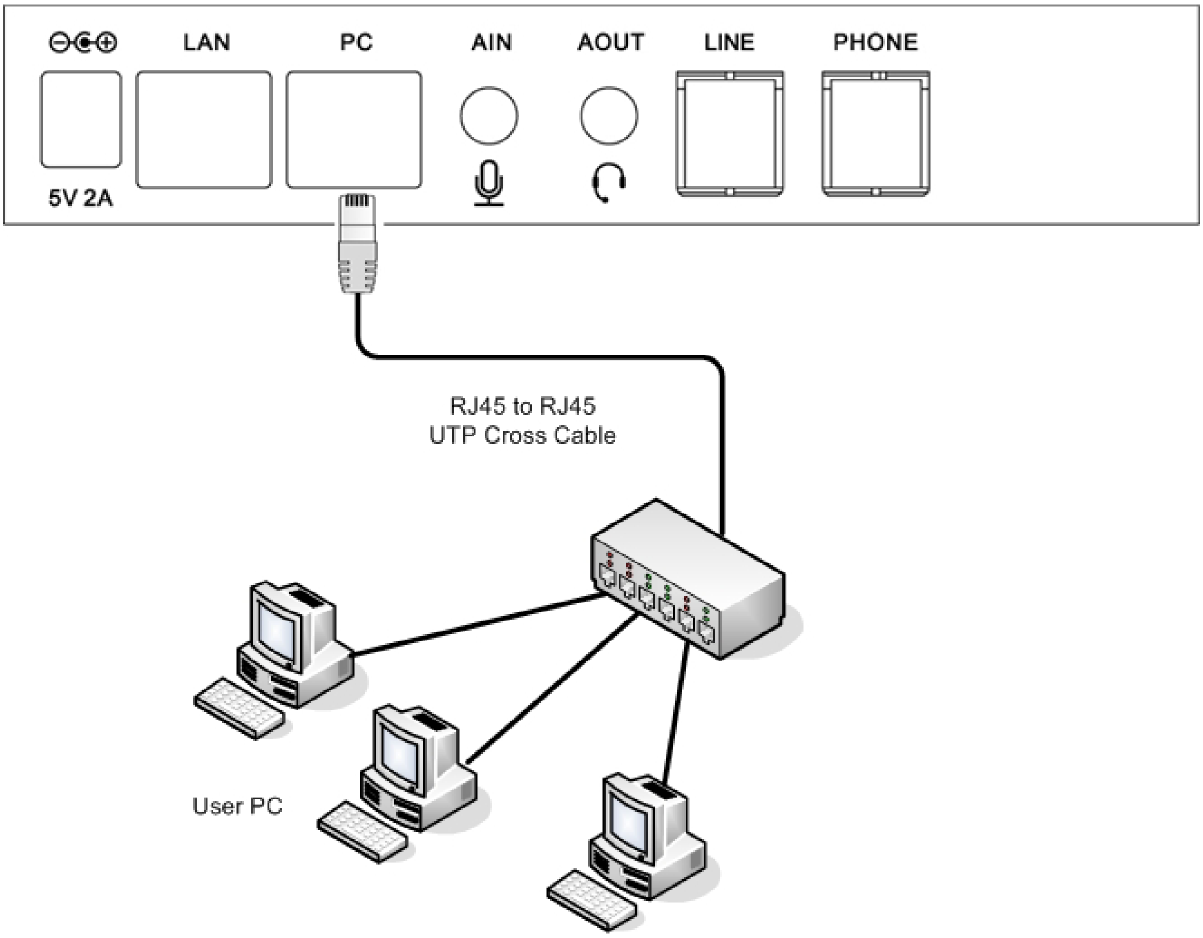
Connecting Ethernet

- A connection to Internet is made through the LAN interface of the WAN accessing equipment such as router, ADSL or cable modem. The LAN is connected by RJ45 standard of the UTP cable.
- Sometimes the cross-over cable is used to connect to the router, xDSL or cable modem
- Use Direct-Through cable for connecting to the hub



(Figure 3-1) Connecting WAN

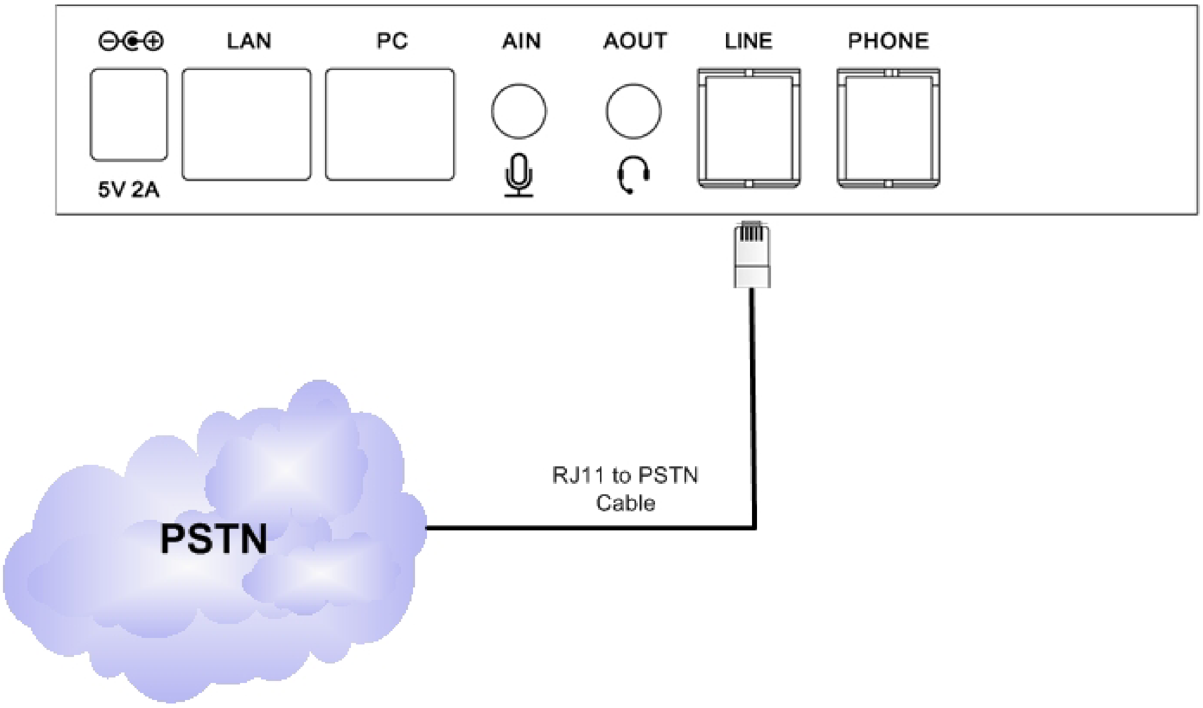
- When LAN1 is set to IP-Share mode, it is connected to PC on Direct-Through cable.
- Use Direct-Through cable to connect to PC directly
- User Cross-Over cable to connect to the hub directly



(Figure 3-2) Connecting LAN

Connecting PSTN Line (FXO-Optional)

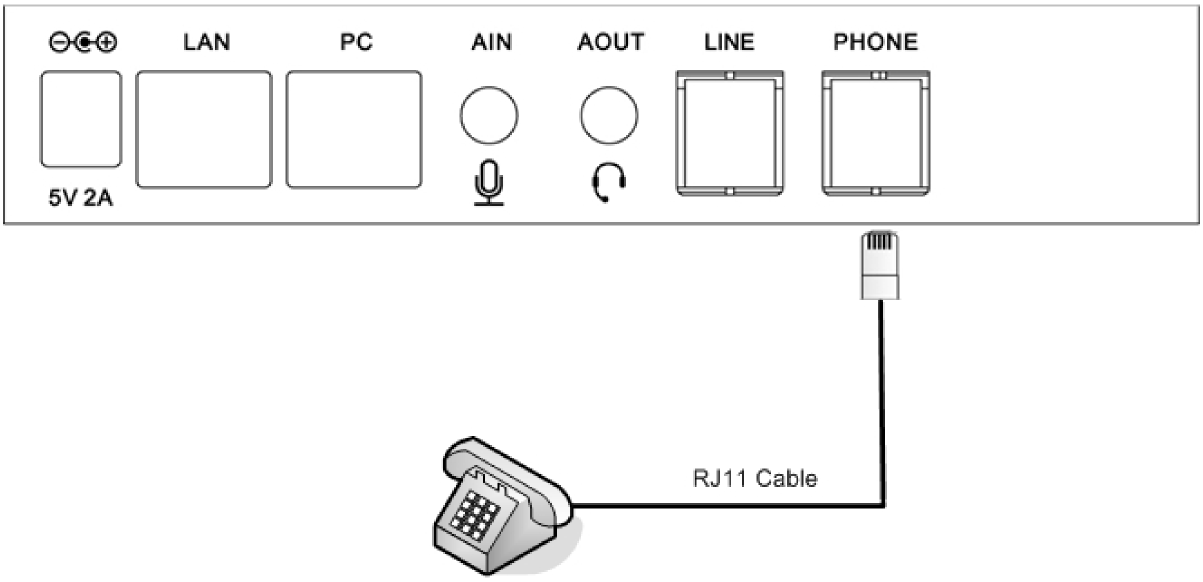
This PSTN port is used for connecting PSTN Central Office line in case of a network failure. The user may connect the PSTN cable to the PSTN port in order to implement PSTN backup. As shown in the figure below, connect the PSTN cable. **(available in AP-IP120E)**.



(Figure 3-3) Connecting PSTN

Connecting Analog Phone Line (FXS)

This PSTN port is used for connecting PSTN Central Office line in case of a network failure. The user may connect the PSTN cable to the PSTN port in order to implement PSTN backup. As shown in the figure below, connect the PSTN cable. **(available in AP-IP120B)**

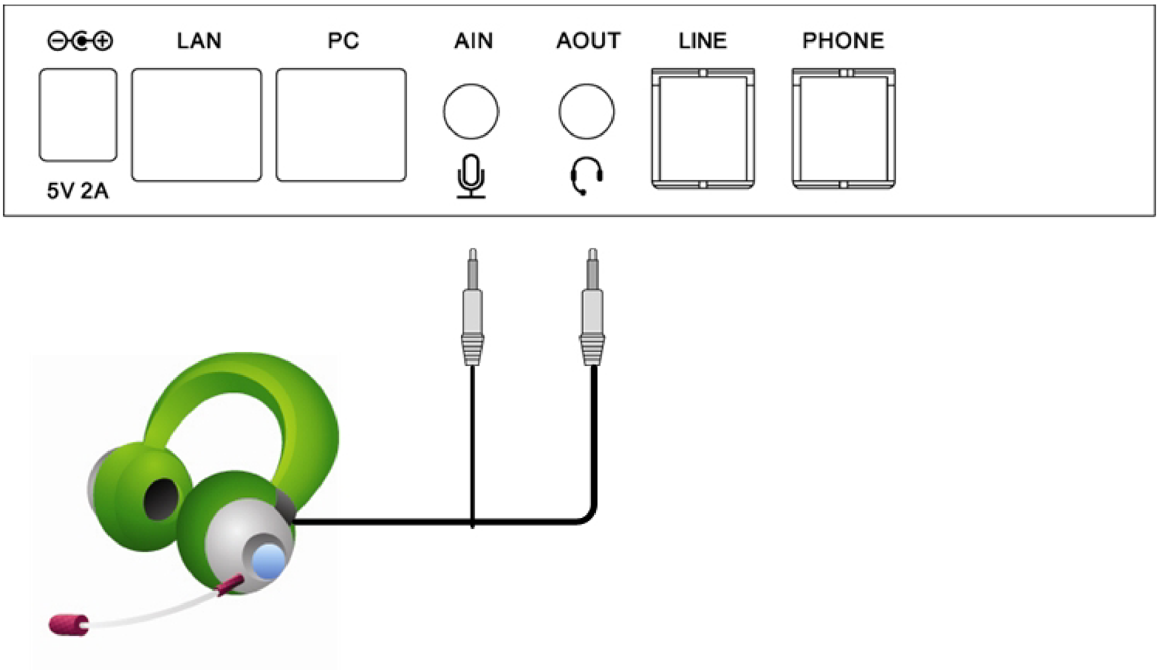


(Figure 3-4) Connecting PSTN

Connecting Audio-In/Out Interface

Audio-In/Out port located at left side of AP-IP120 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.



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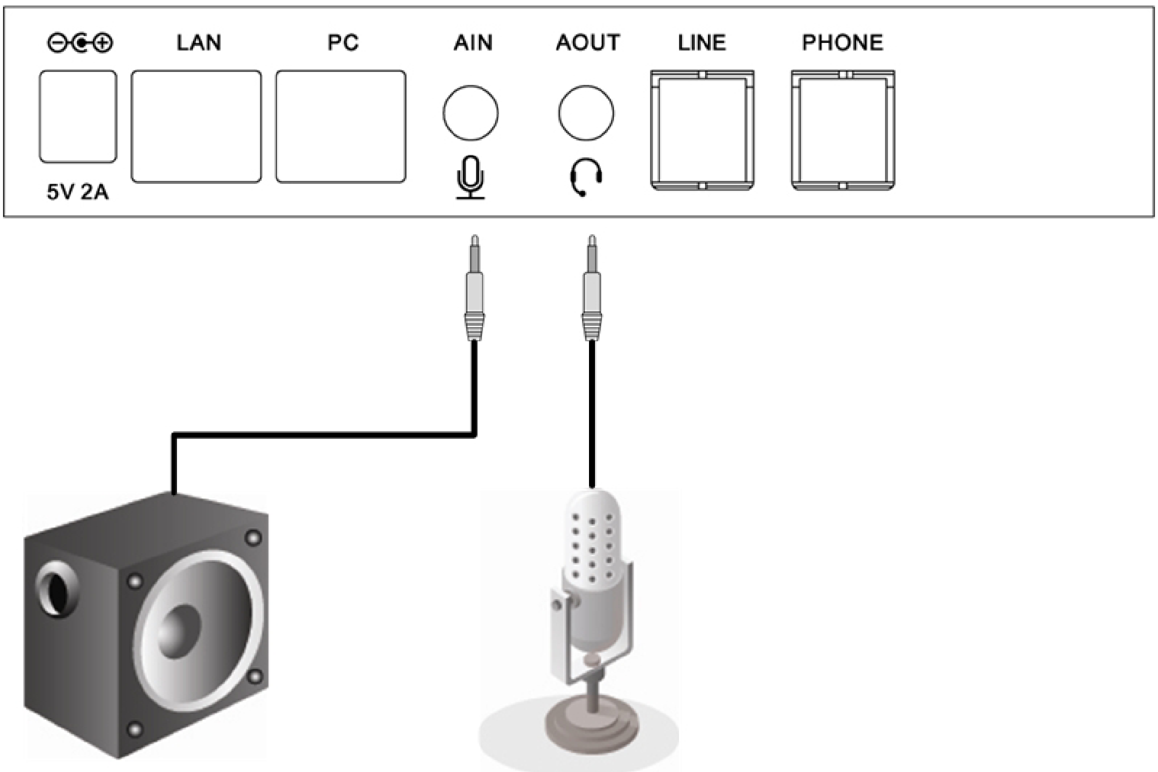
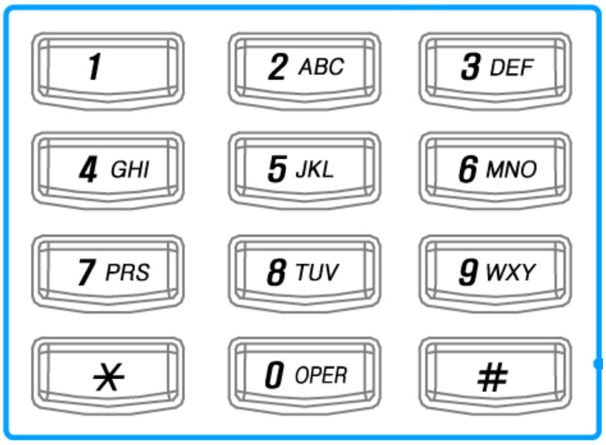


(Figure 3-5) Connecting External Audio IN/OUT Interface

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 Layout

[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 Call/Cancel Button on the Keypad

The functions of Call button is described in [Table 3-2].

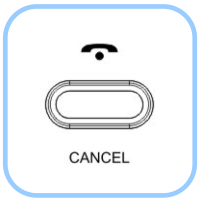
(Figure 3-7) Call Button on the Keypad

[Table 3-2] Using Call Button on the Keypad

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].



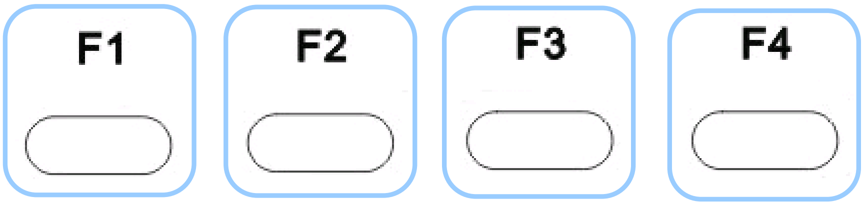
(Figure 3-8) Cancel Button on the Keypad

[Table 3-3] Using Cancel Button on the Keypad

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.
Move the view mode	Moves the menu option from from the present to the upper by one level on User Interface

Using the Soft-key Interworking with 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 3-9) Function Keys Layout

[Table 3-4] Available Softkey Features in On Hook State

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] Available Softkey Features in Off Hook State

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)	Allows you to pick up incoming calls within their own group

[Table 3-6] Available Softkey Features in Busy State

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] Available Softkey Features in Hold State

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

[Table 3-8] Available Softkey Features in Incoming Call State

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

[Table 3-9] Available Softkey Features in Outgoing Call State

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

[Table 3-10] Available Softkey Features for Voice Mail

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

[Table 3-11] Available Softkey Features in Call Transfer

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] Available Softkey Features for Conference

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] Available Softkey Features for Conference Participant

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)

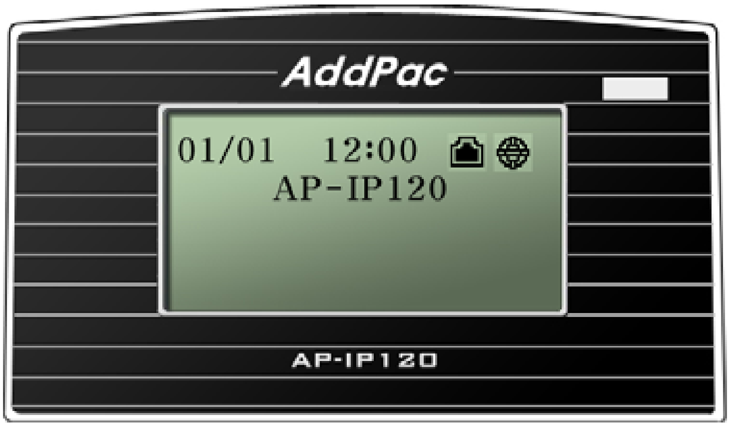
[Table 3-14] Conference Participants

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)

Chapter 4. Using AP-IP120

Default Screen

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



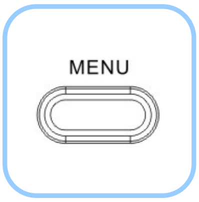
(Figure 4-1) Default Screen

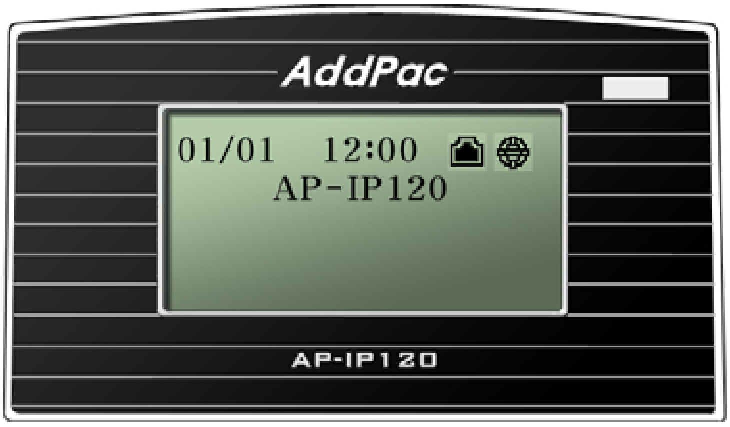
[Table 4-1] Description of Default Screen

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	Display Name	Display the name of the device (System Setup -> 1.Display Name)

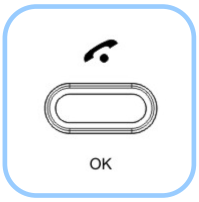
Phone Book Menu Category

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.





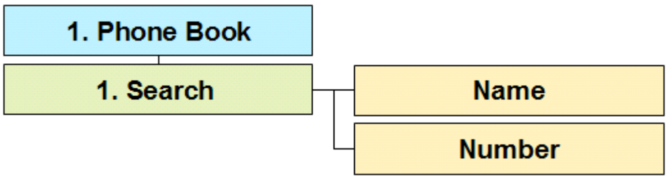
(Figure 4-2) Main Screen

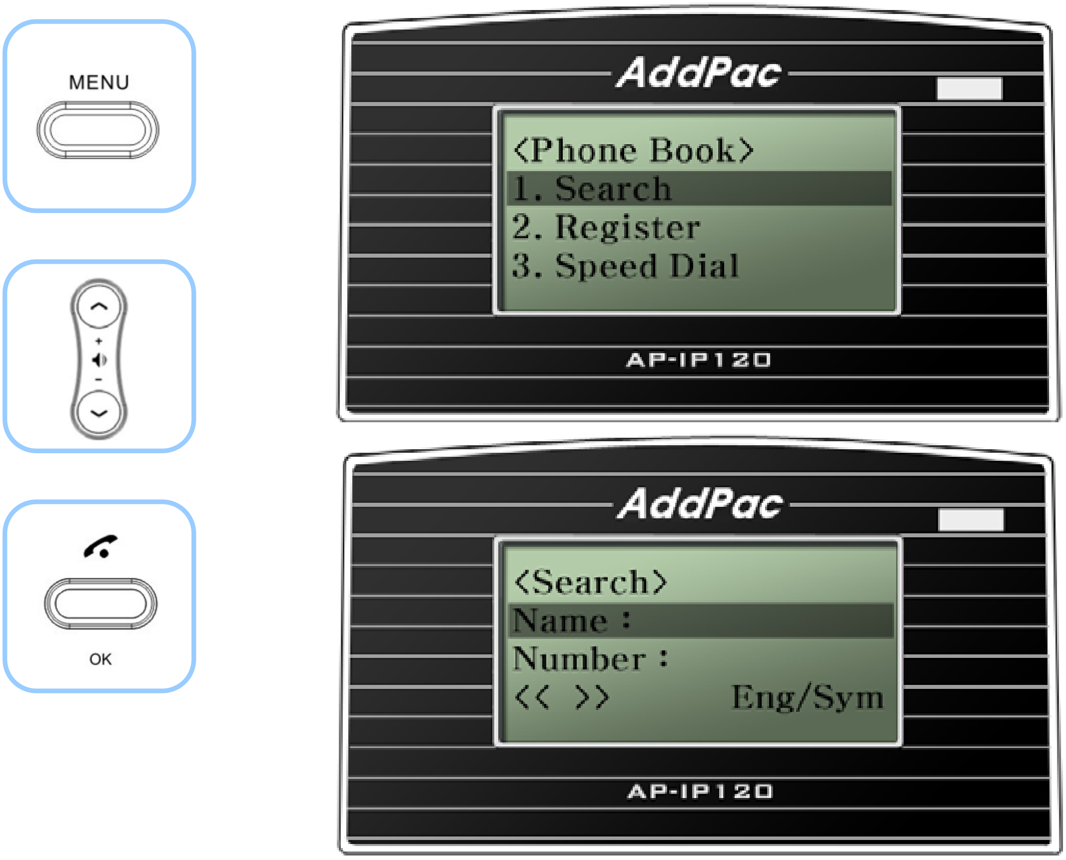


(Figure 4-3) Phone Book Menu Section in Main Menu

Phonebook – Search

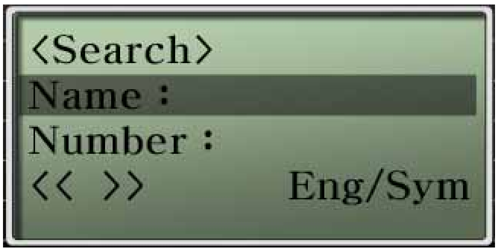
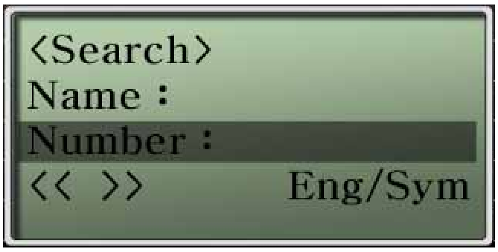
The Phonebook uses the registered name, phone number and speed dial number to search the phone number.





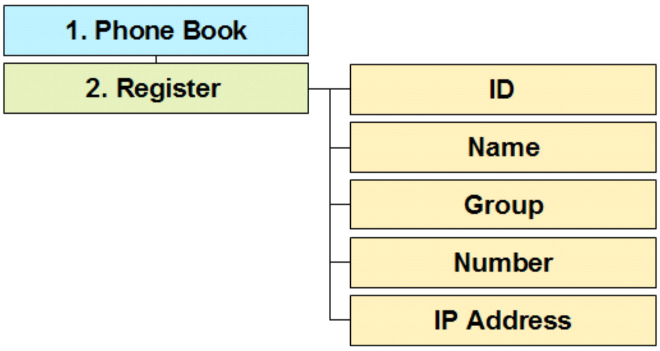
(Figure 4-4) Search Menu Section in Phone Book Menu Section

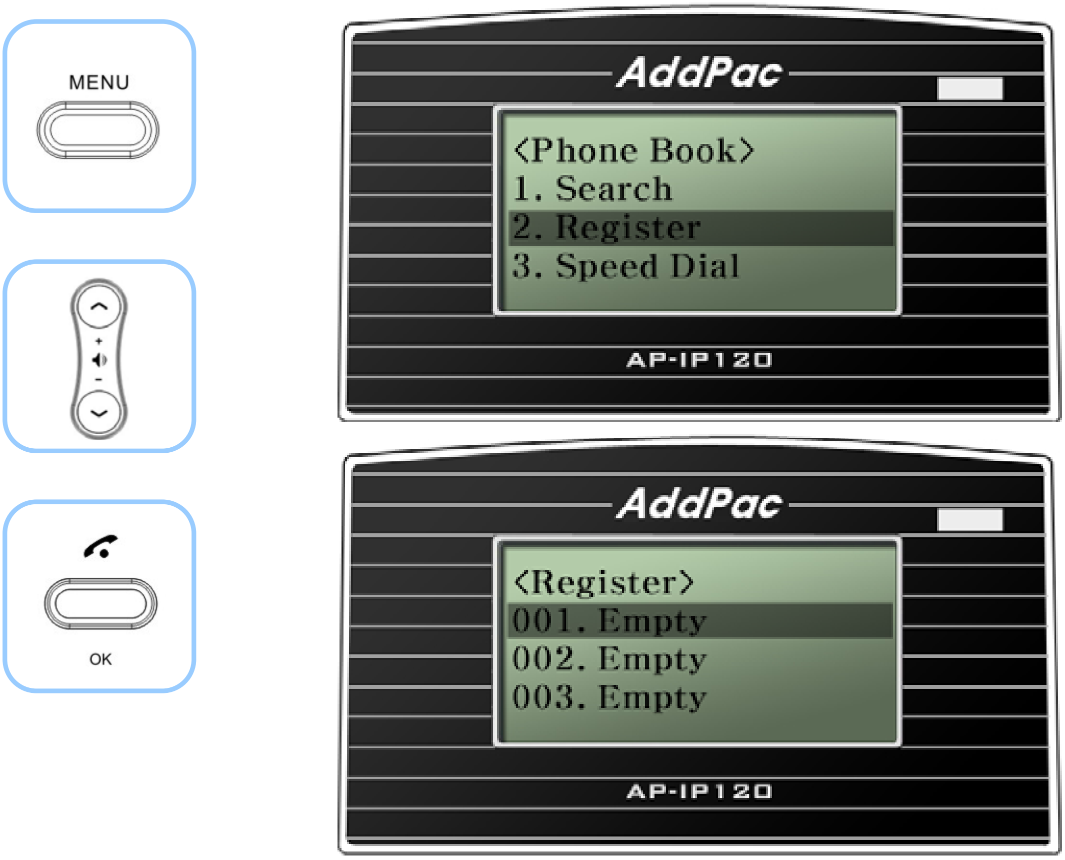
[Table 4-2] Description of Search Menu Section

Category	Description
	Search by the name which has been saved in the Phone Book previously. If the Name has more than 2 fields that are same as the search word, all of them are displayed. F1: Backspace F2: Space F4: Change the text
	Search by the number in the Phone Book *(Multi-Search with name and phone number is possible)

Phone Book – Registration

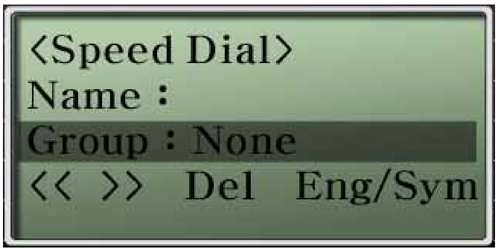
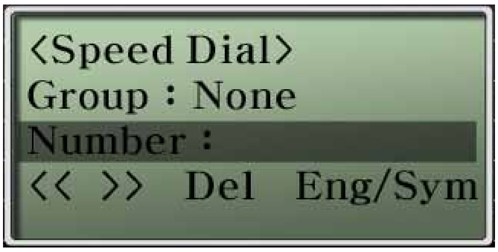
This option allows you to register a new telephone number to the Phone Book. You can use this registered number for other option of search and speed dial.





(Figure 4-5) Register Menu Section

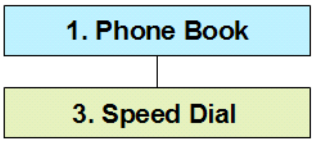
[Table 4-3] Description of Register Menu Section

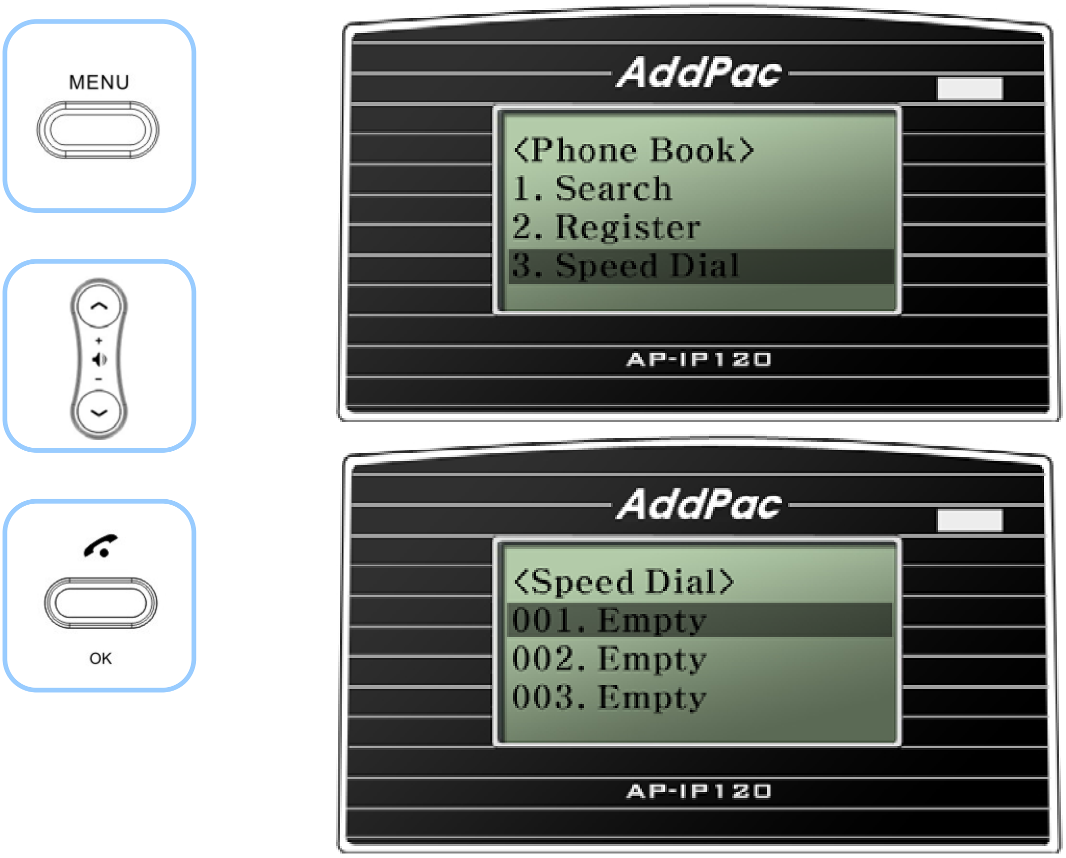
Registration Parameters	Description
	Enter the index number for the speed dial.
	Register a name of the other party to Phone Book
	Enter the other party's group in Phone Book.
	Enter the other party's number in Phone Book.
	Enter the other party's IP address in Phone Book.

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

Phonebook – Speed Dial

The telephone numbers are listed per LCD display by simply using name. User can quickly find and make a call to others on the list in this menu. You can also press Call button by using a saved phone number.





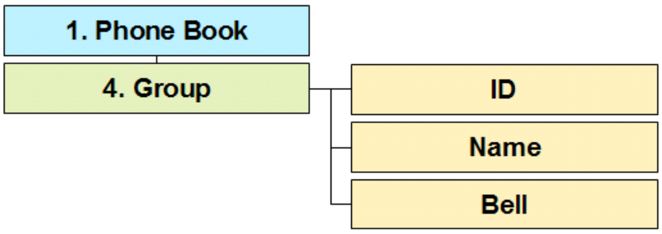
(Figure 4-6) Speed Dial Menu Section

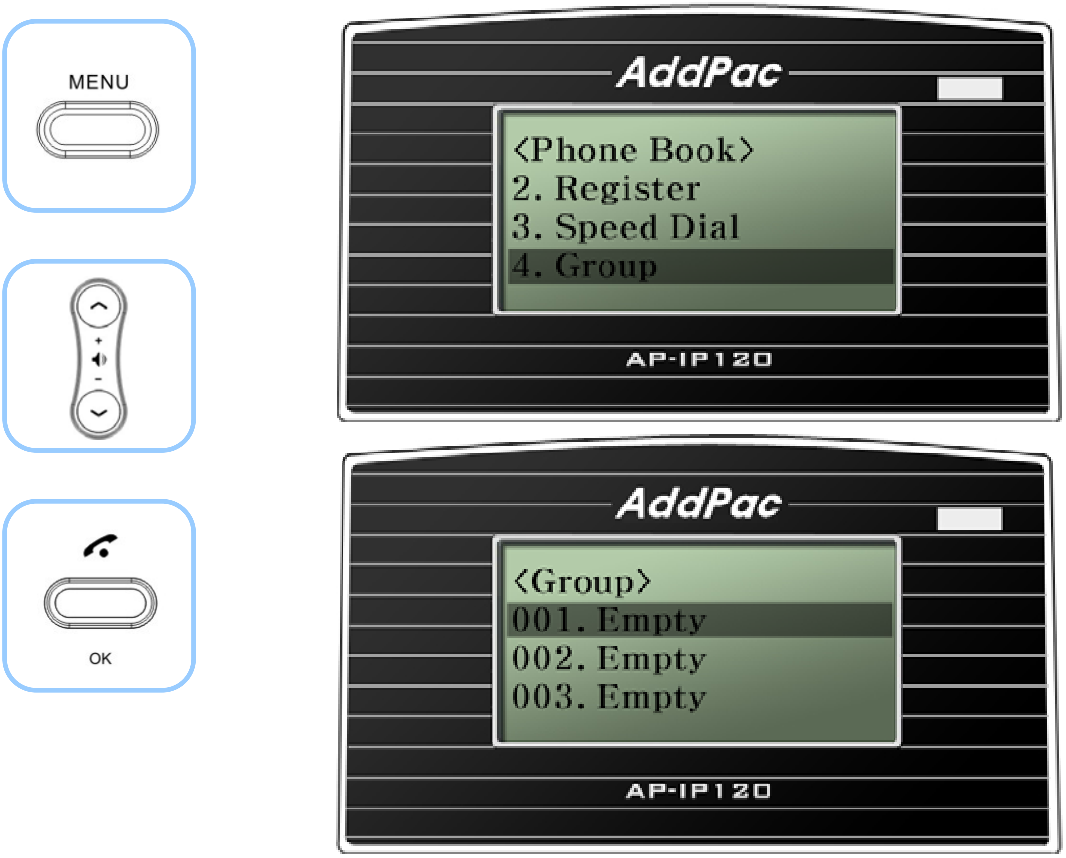
[Table 4-4] Description of Speed Dial Menu Section

Option	Description
①	Display the index number in Phonebook
②	Display the name in Phonebook
③	Display the Dial number in Phonebook

Phonebook – Group

You can use Group Menu Section to store, manage and use the phone number by group. You can differentiate the bell sound for each group.





(Figure 4-7) Group Menu Section in Phone Book

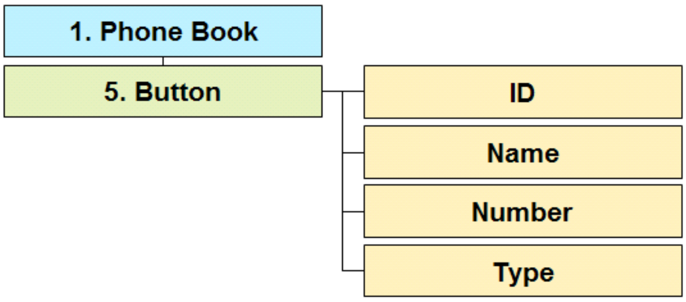
[Table 4-5] Description of Group Menu Section

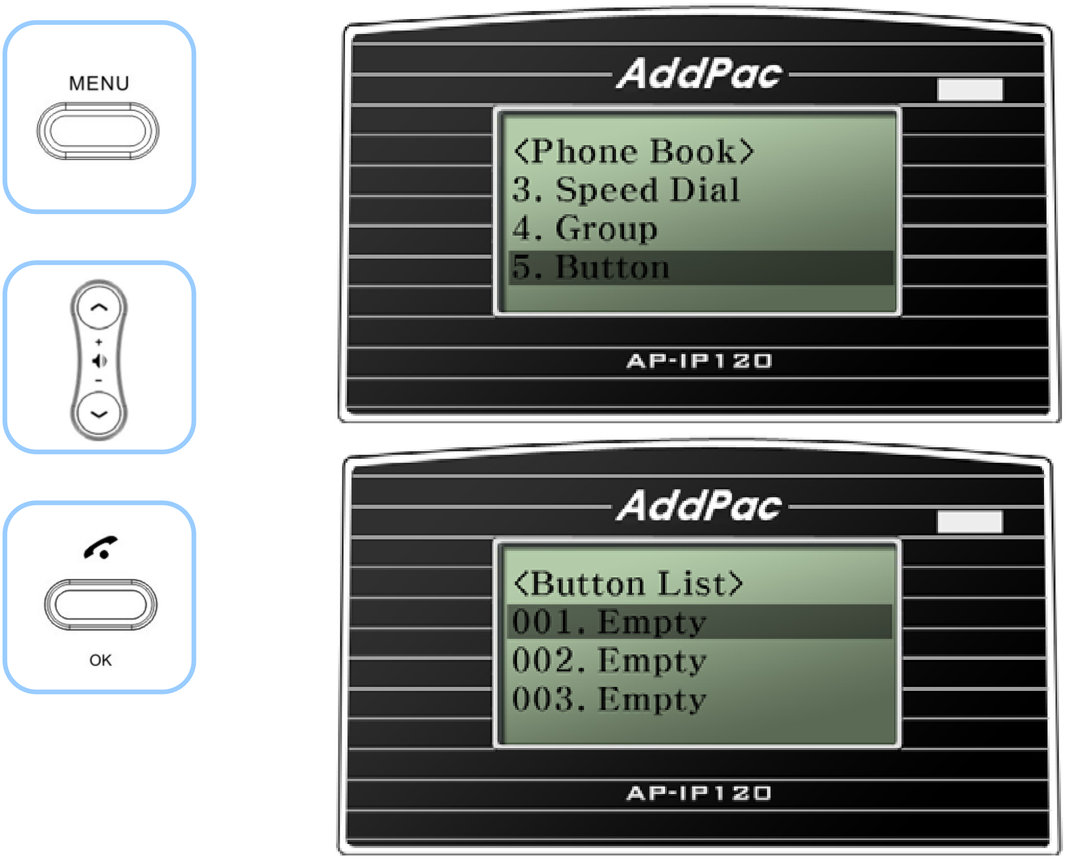
Option	Description
	Select a group number to be used in Phone Book.
	Enter a group name in Phone Book.
	Make a selection of bell sound for each group.

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

Phonebook – Button List

Button List Menu Section specifies Speed Button List. The list takes the speed dial list which is configured to the server by using Button Profile Menu Section from the presence server.





(Figure 4-8) Button List Menu Section in Phone Book

[Table 4-6] Description of Button List Menu Section

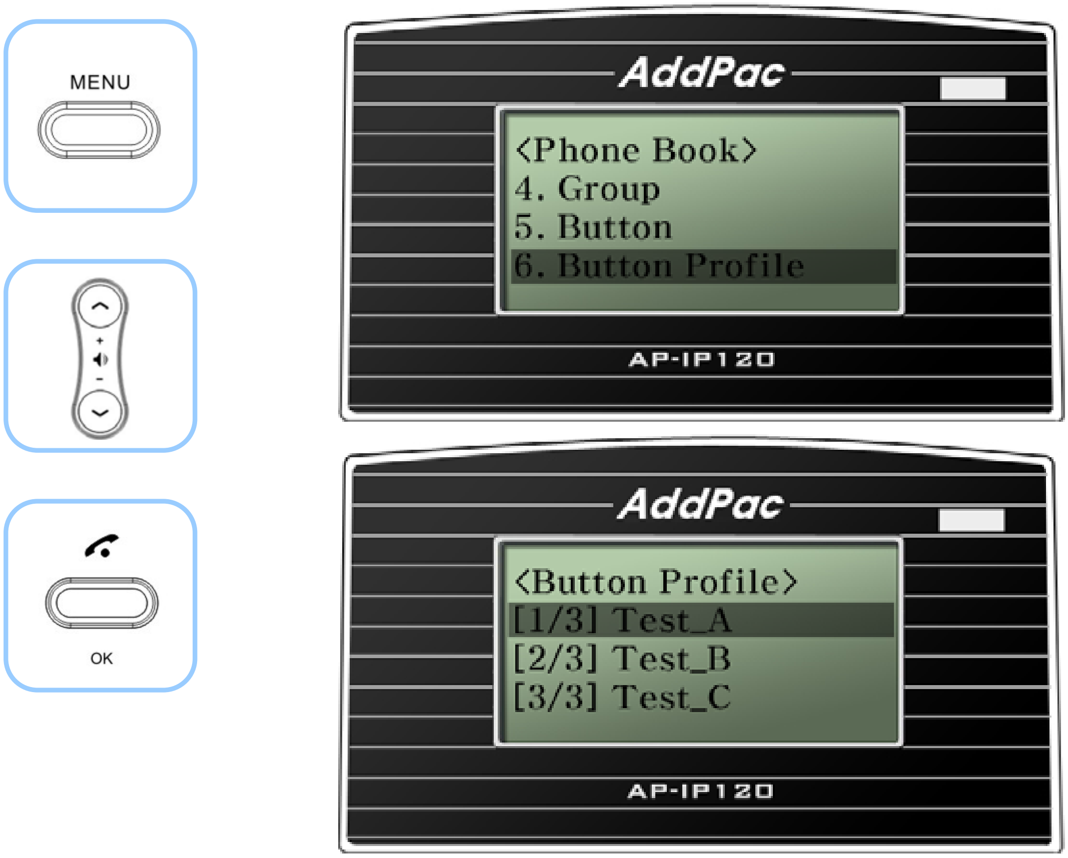
Category	Description
Name	Set a name for the Speed Button.
Number	Set a number for the Speed Button
Type	Set a type of icon for the

Phonebook – 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.

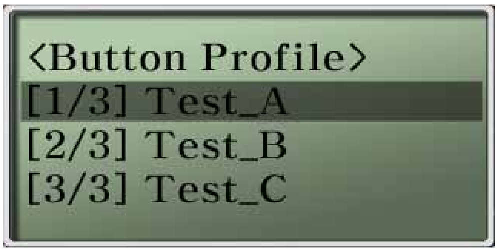
1. Phone Book

6. Button Profile



(Figure 4-9) Button Profile Menu Section

[Table 4-7] Description of Button Profile Menu Section

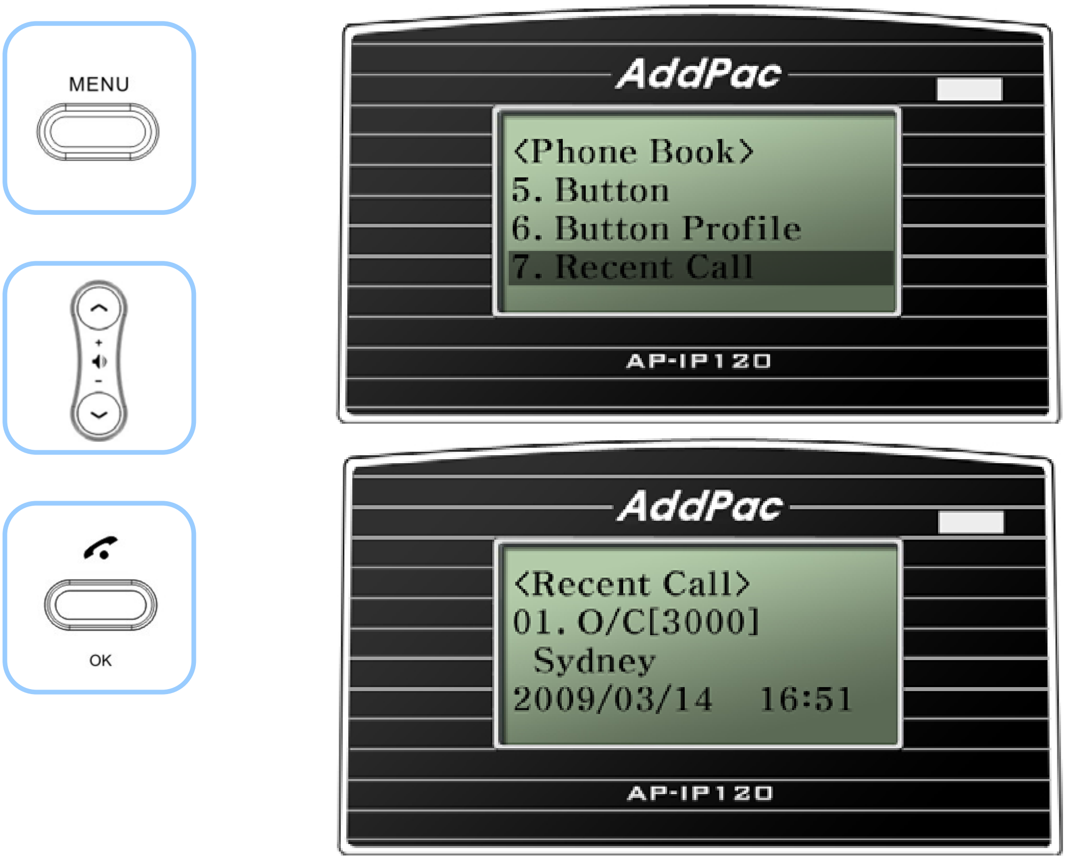
Category	Description
	Select a button profile in Speed Dial List which has been set to Presence Server

Phonebook – Recent Call

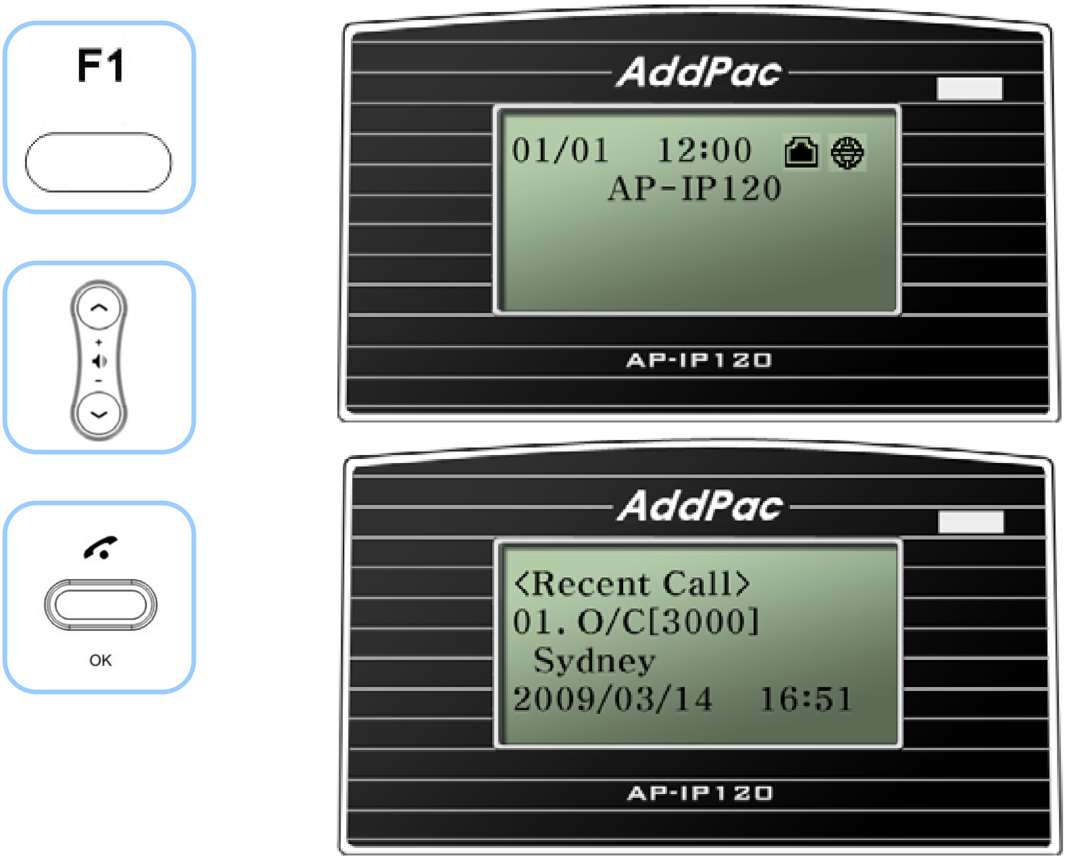
The recent call displays a call log of the user for incoming and outgoing calls. This feature enables the user to check any incoming call which has been arrived during one's absence, calls back by using the number of the incoming call and save the number of the incoming call. 2 different ways of (Figure 4-10) and (Figure 4-11) are used to retrieving Recent Call.

1. Phone Book

7. Recent Call



(Figure 4-10) Recent Call Menu Section 1



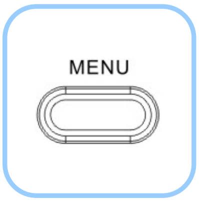
(Figure 4-11) Recent Call Menu Section 2

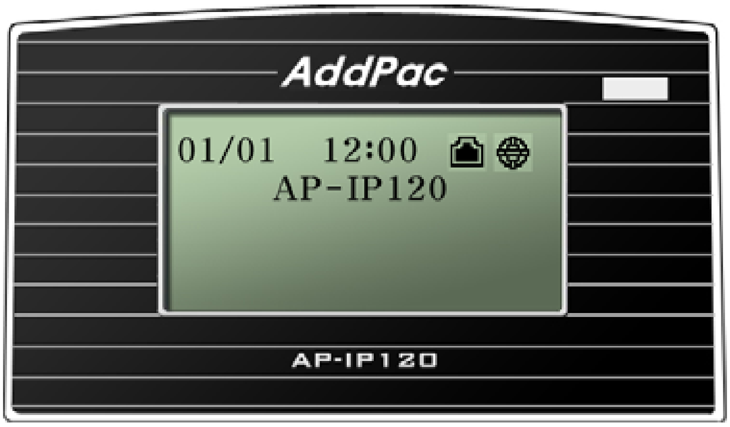
[Table 4-8] Description of Call Menu Option 2

Option	Description
①	01. => Display the index number in Recent Call [O/C] ⇒ Outgoing Call / Connected [O/N] ⇒ Outgoing Call / Not Connected [I /C] ⇒ Incoming Call / Connected [I /C] ⇒ Incoming Call / Not Connected [3000] => Display the other party's call information (phone number)
②	Display the other party's call information (name)
③	Display the call duration of incoming/ outgoing call

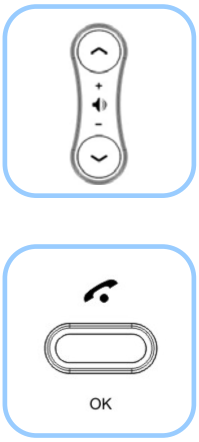
Tool Box Menu Category

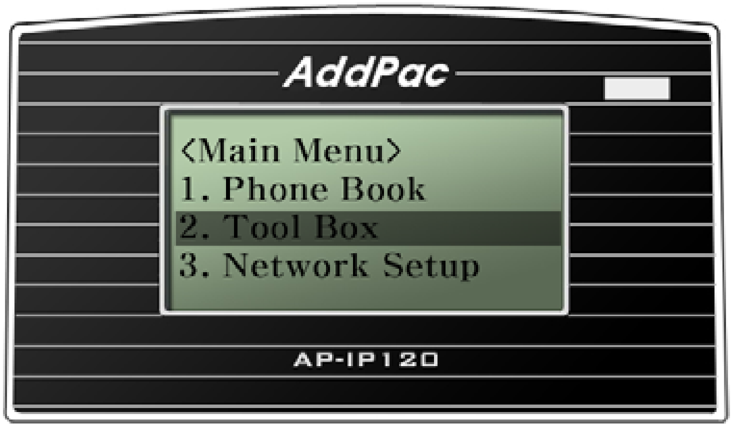
Tool Box menu consists of date/time setting, configuration saving, initialization to factory default mode and language selection.





(Figure 4-12) Main Screen

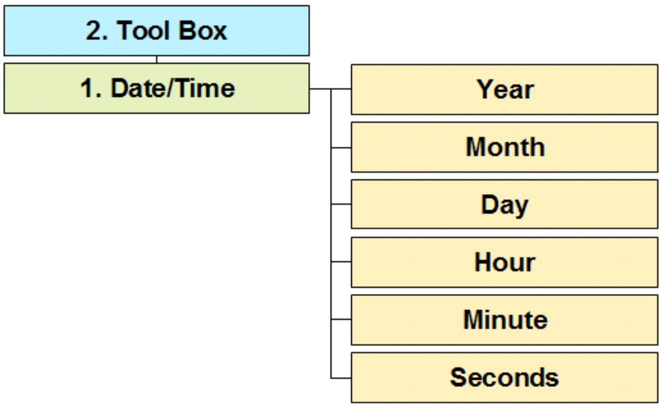


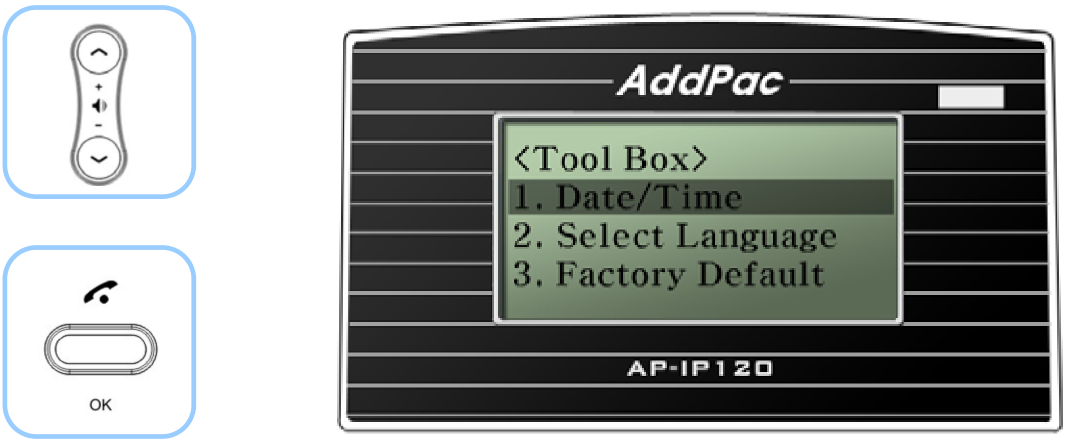


(Figure 4-13) Too Box Menu Section

Tool Box – Date & Time

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



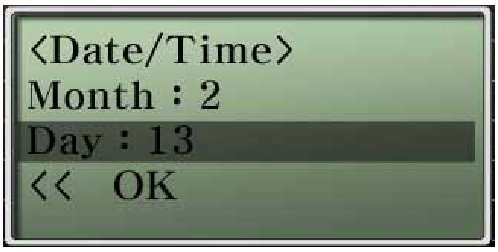


(Figure 4-14) Date & Time Menu Section

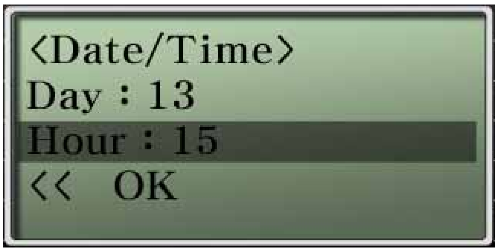
[Table 4-9] Description of Date & Time Menu Section

Menu Option	Description
	Enter the present year

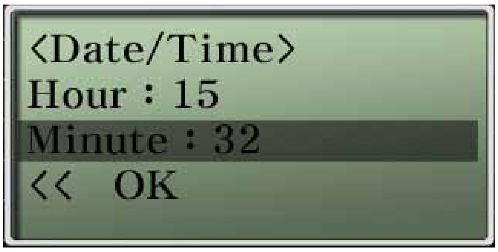
Enter the present month



Enter the present date



Enter the present hour

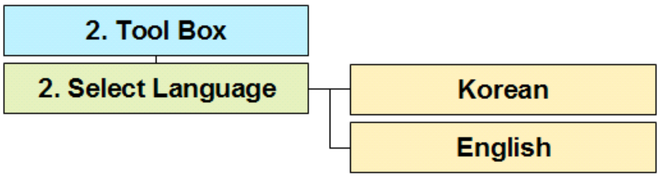


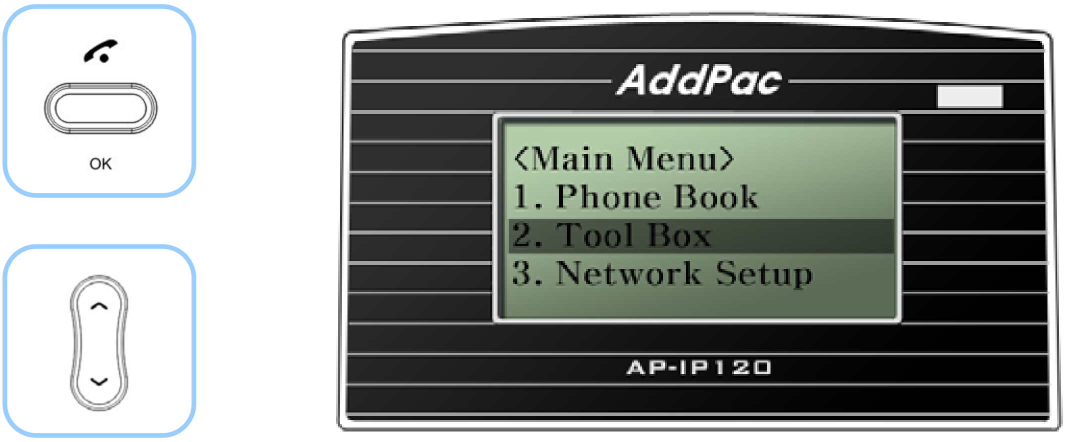
Enter the present minute

Enter the present second (Press OK to complete entry)

Tool Box – Language

You can select a language of your choice. You can choose English or Korean.





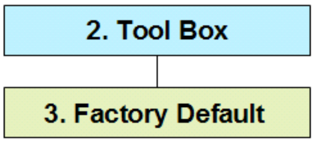
(Figure 4-15) Language Setup Menu Section

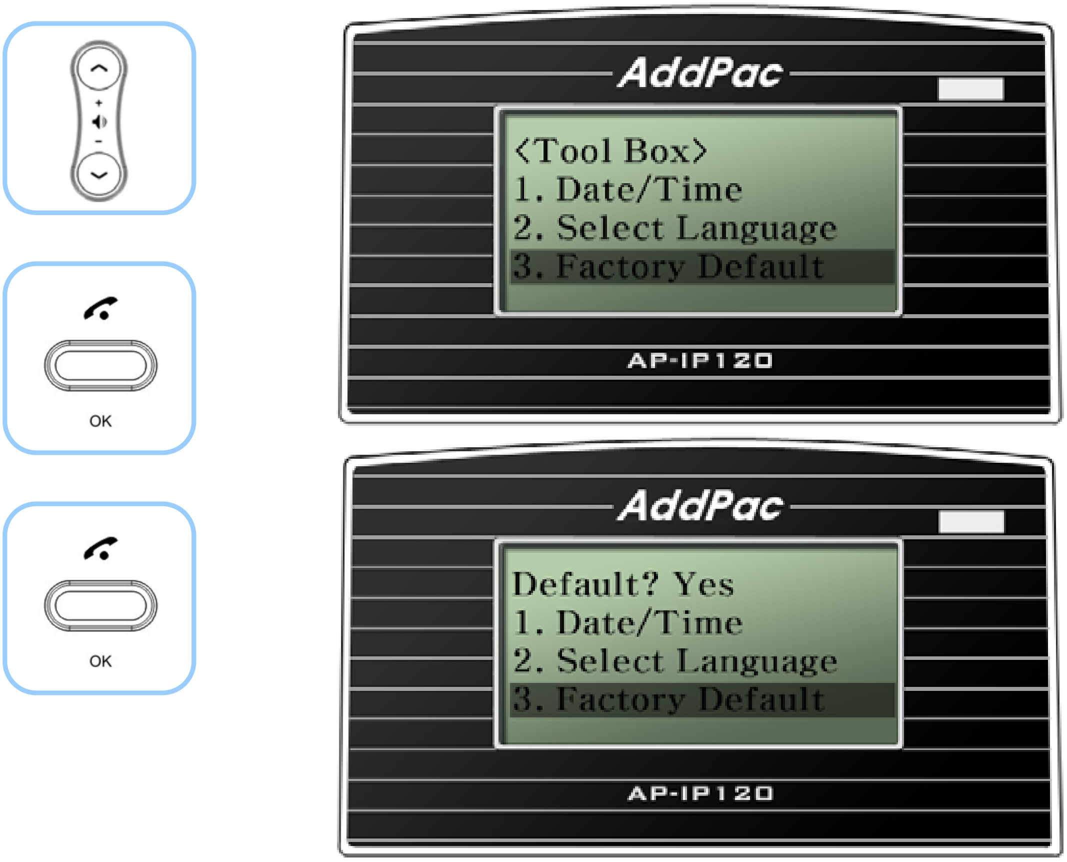
[Table 4-10] Description of Language Setup Menu Section

Option	Description
	Make a selection of English (default)
	Make a selection of English

Tool Box – Factory Default

The Factory Default deletes all the configured settings of AP-IP120 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.

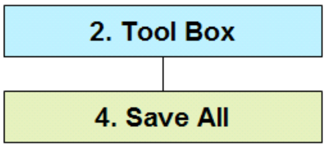


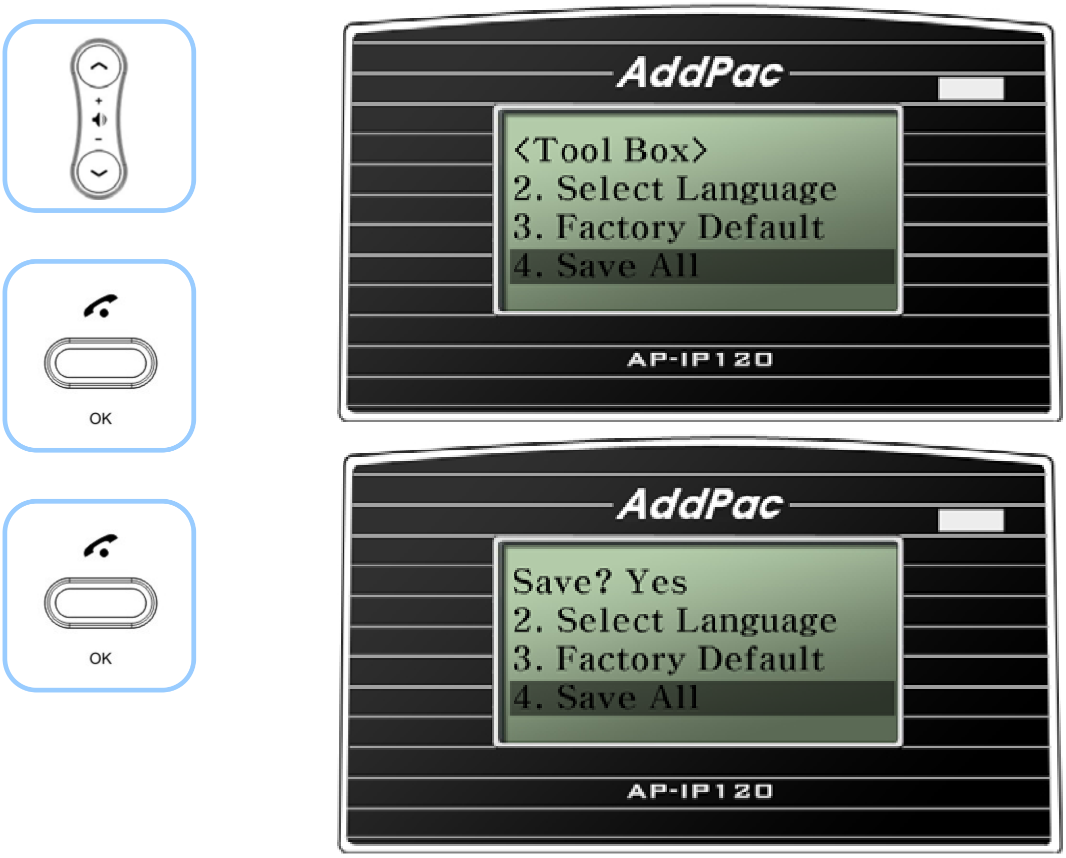


(Figure 4-16) Factory Default Menu Section

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.

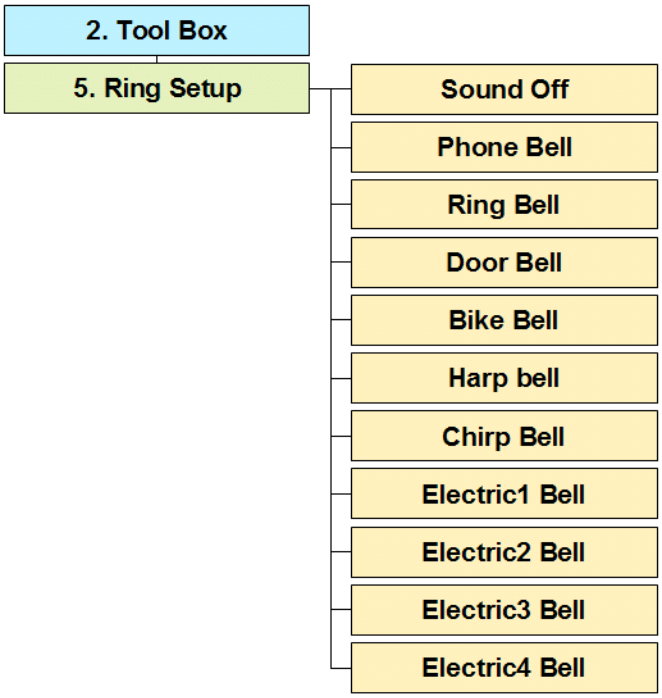


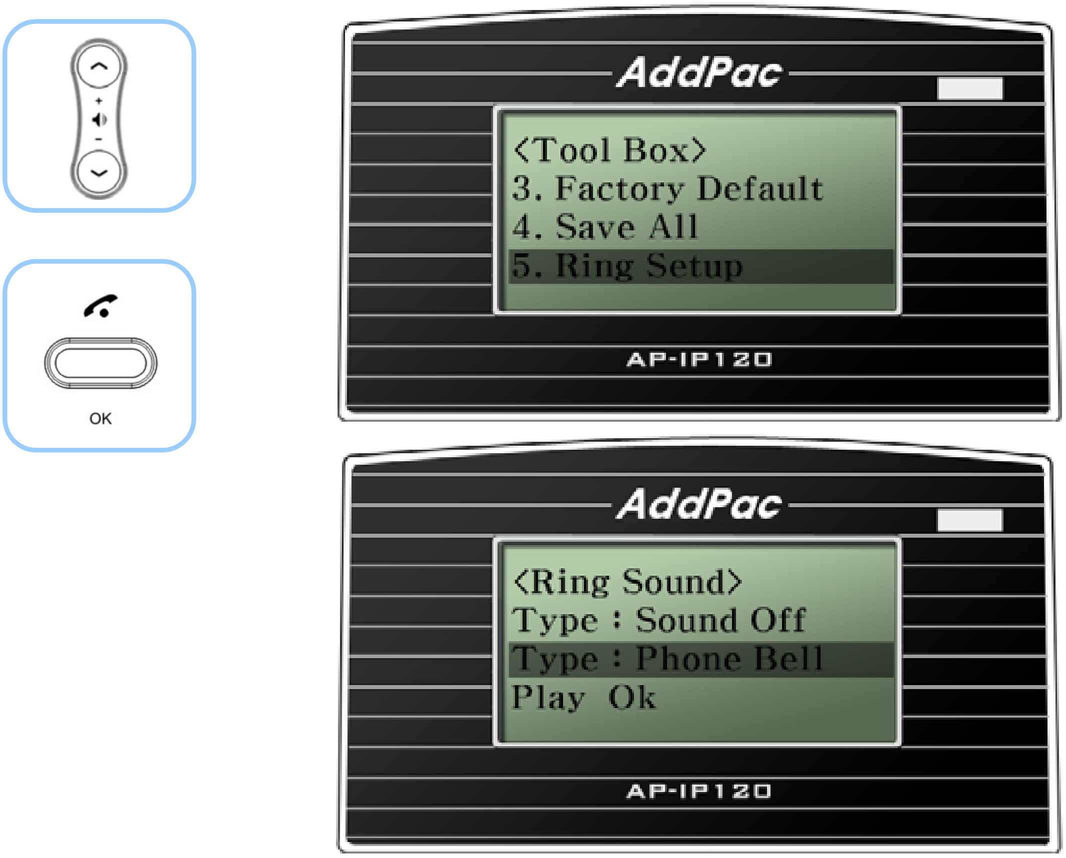


(Figure 4-17) Save All Menu Section

Tool Box – Ring Setup

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.



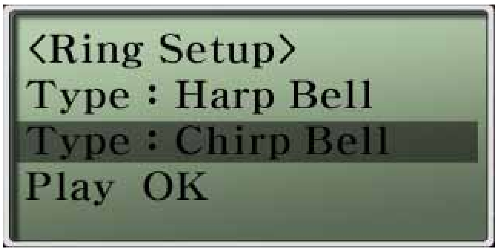


(Figure 4-18) Ring Setup Menu Section

[Table 4-11] Description of Ring Setup Menu Option

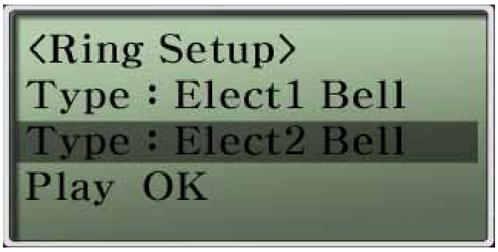
Menu Option	Description
	Set to mute
	digital phone
	analog phone
	door bell
	bicycle bell

harp bell

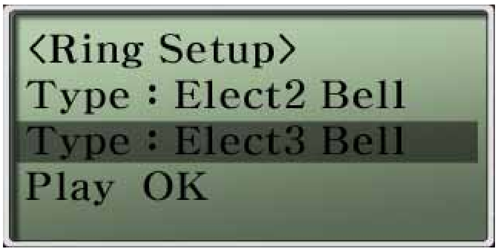


chirp Bell

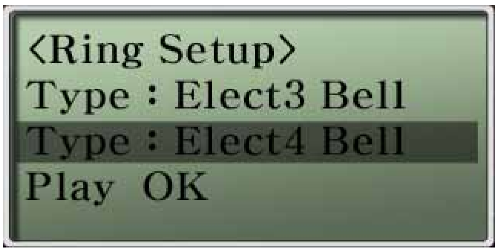
Electric Bell 1



Electric Bell 1



Electric Bell 1



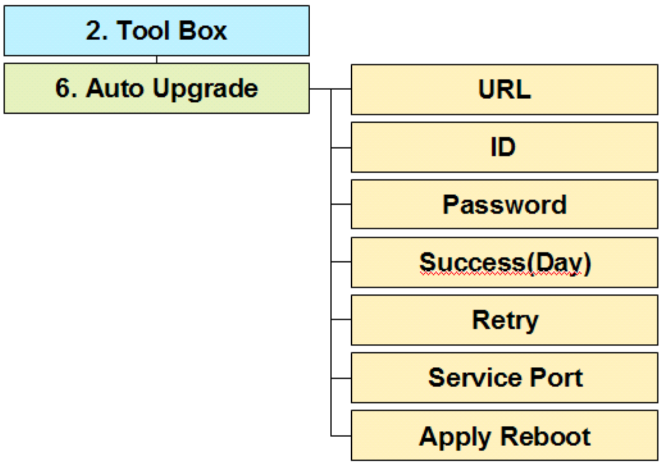
Electric Bell 1

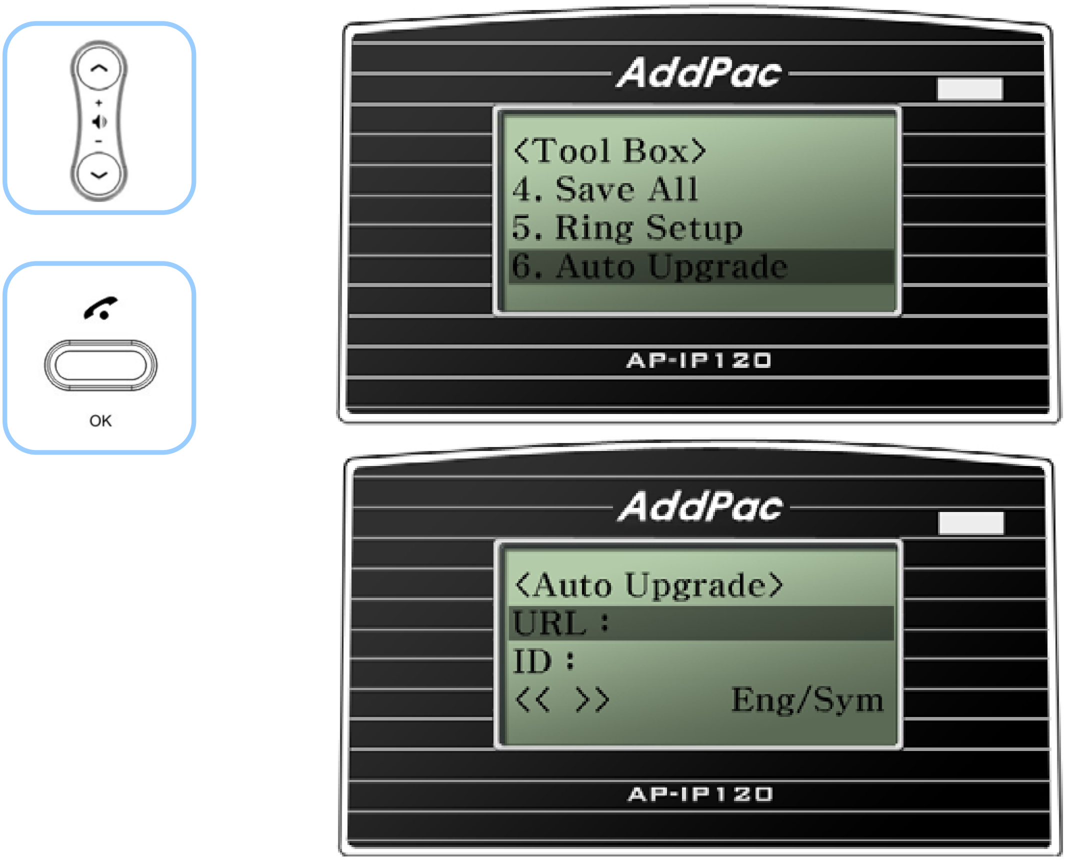
* Play(F1) => Listening to a bell sound in advance

Tool Box – 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.

.

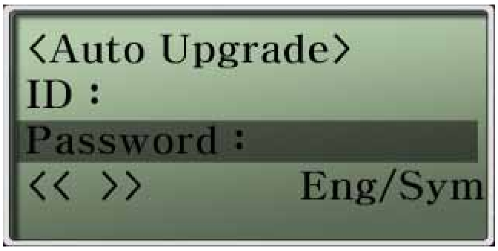




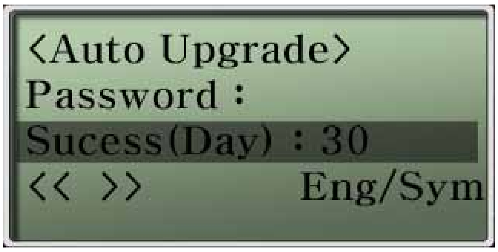
(Figure 4-19) Auto Upgrade Menu Section

[Table 4-12] Description of Auto Upgrade Menu Option

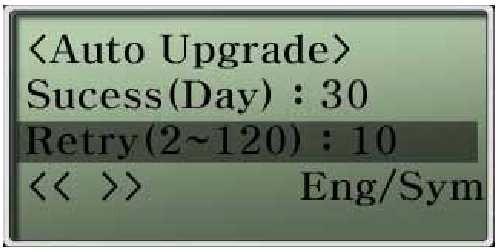
Menu Sections	Description
	Enter URL of the Auto Upgrade server Ex)down.addpac.com/apos/IP120/packing.lst
	Enter the ID for an authorized access to the Auto Upgrade server Ex) addpac



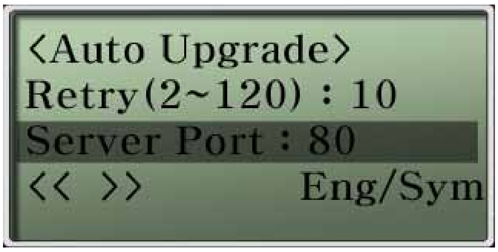
Enter the password for an authorized access to the Auto Upgrade server
Ex) addpac



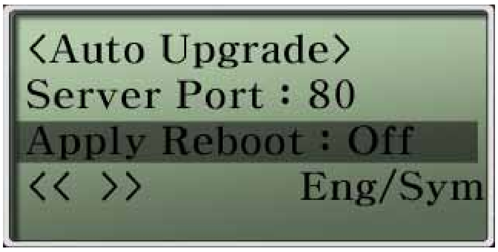
The succeeded Auto Upgrade can be kept in a record for a certain time. The basic default value is set to 30 days.



The failed Auto Upgrade can be kept in a record for a certain time. The basic default value is set to 10 minutes.



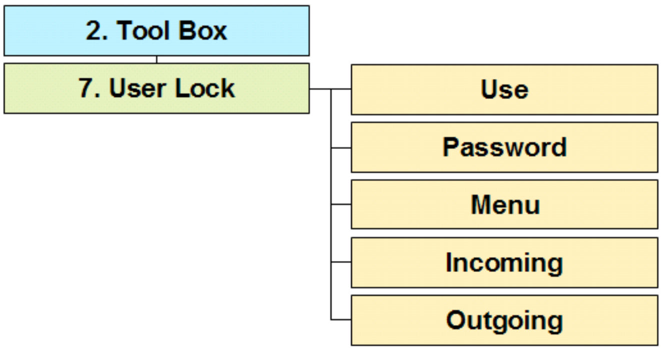
Enter a Port of the Auto Upgrade Server. The default value is set to 80 for HTTP.

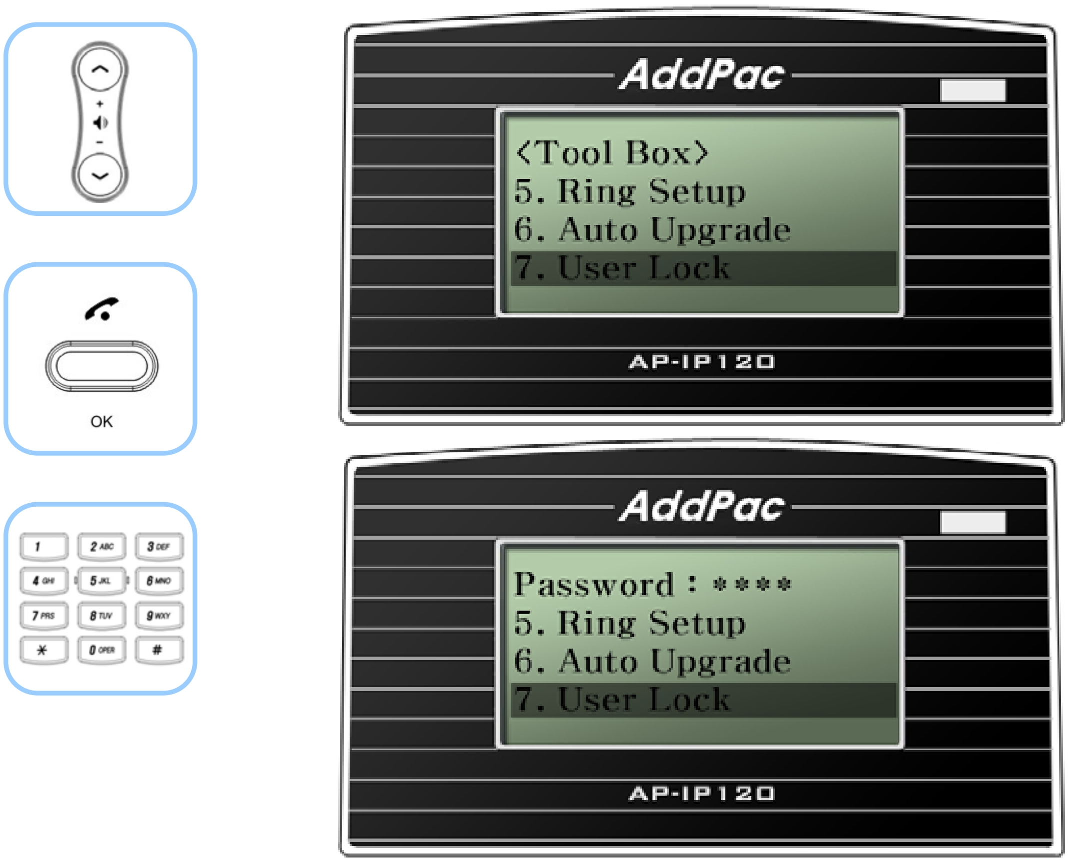


Select whether to apply the settings of the Auto Upgrade after rebooting or not
Ex) On/Off
(using the numeric button for On/Off)

Tool Box - User Lock

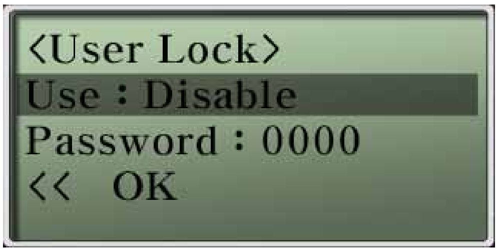
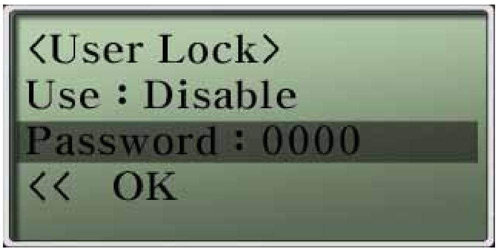
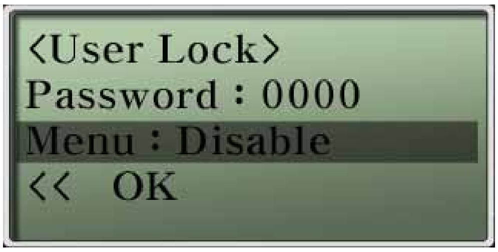
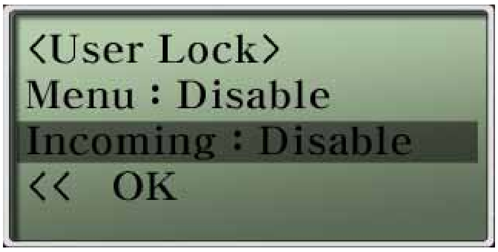
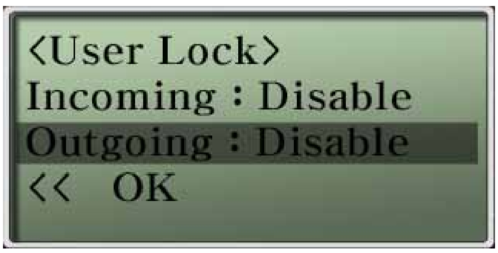
User Lock menu component configures the settings to block the entry to a particular menu section and changes the password. '0000' is set at the default. **Keeping the same password is recommended.**





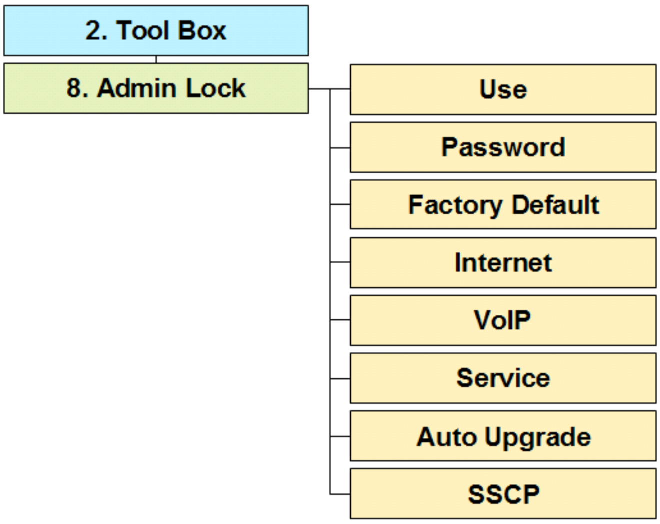
(Figure 4-20) User Lock Menu Section

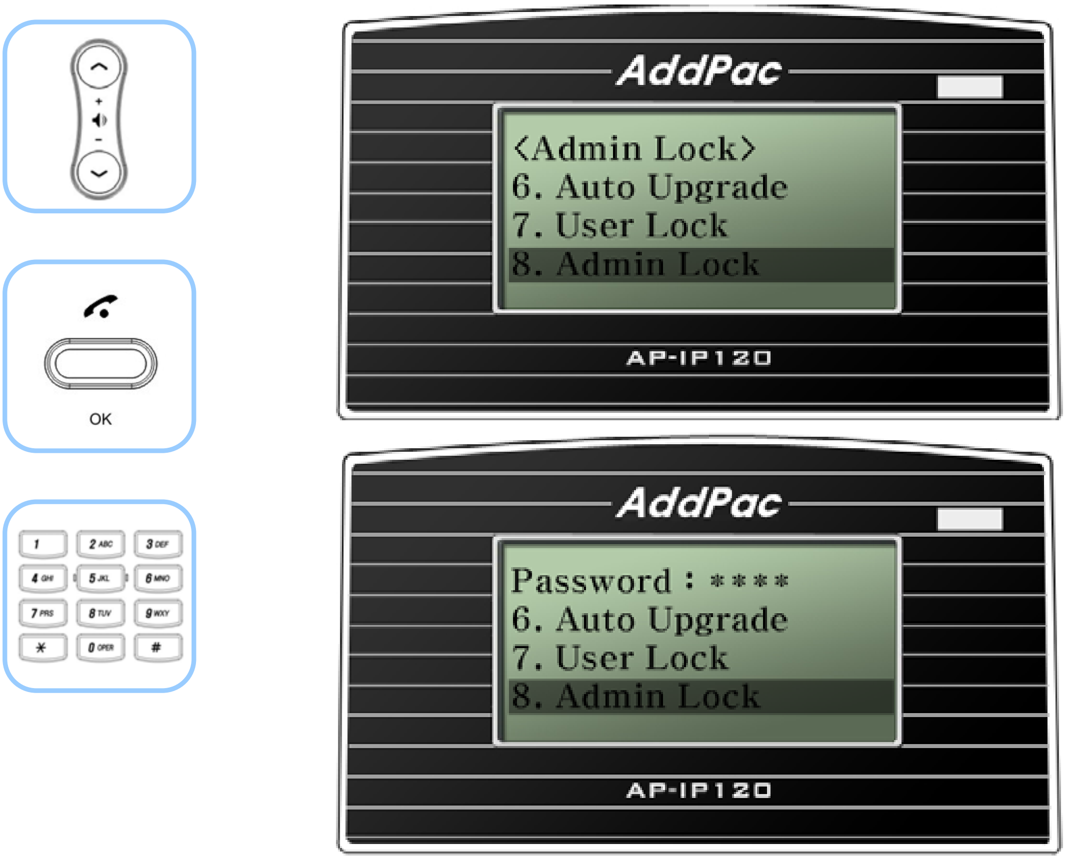
[Table 4-13] Description of User Lock Menu Option

Option	Description
	Enable/Disable(Default : Enable)
	Replaces the old password with the new one (Default : 0000)
	Selects the menu section to be locked by password. The available menu sections are Incoming Lock, Outgoing Lock, and Menu Lock. (Default : Disable)
	Specifically locks the incoming call (Default : Disable)
	Specifically lock the outgoing call (Default : Disable)

Tool Box - Admin Lock

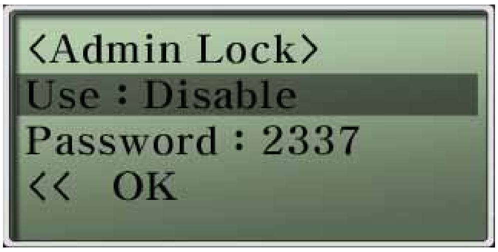
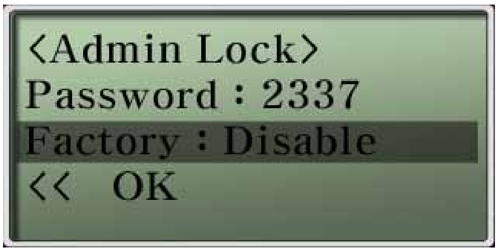
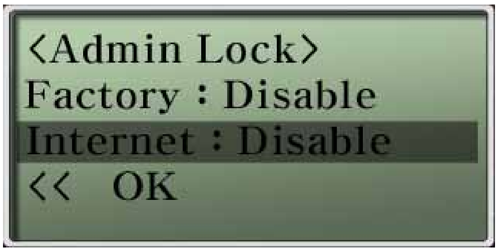
Admin Lock menu component configures the settings to block the entry to a particular menu and changes the password. ‘2337’ is set at the default. **Keeping the same password is recommended.**

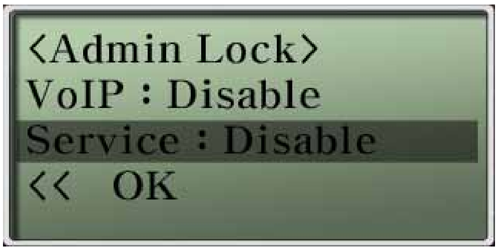




(Figure 4-21) Admin Lock Menu Section

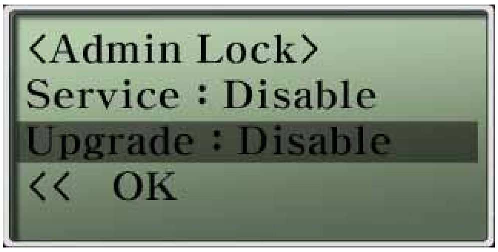
[Table 4-14] Description of Admin Lock Menu Option

Option	Description
	Enable/Disable(Default : Enable)
	Replaces the old password with the new one (Default : 2337)
	Set a password for Factory Default (Default : Disable)
	Set a password for Internet Setup (Default : Disable)
	Set a password for VoIP Setup (Default : Disable)



Set a password for Service Port Setup

(Default : Disable)

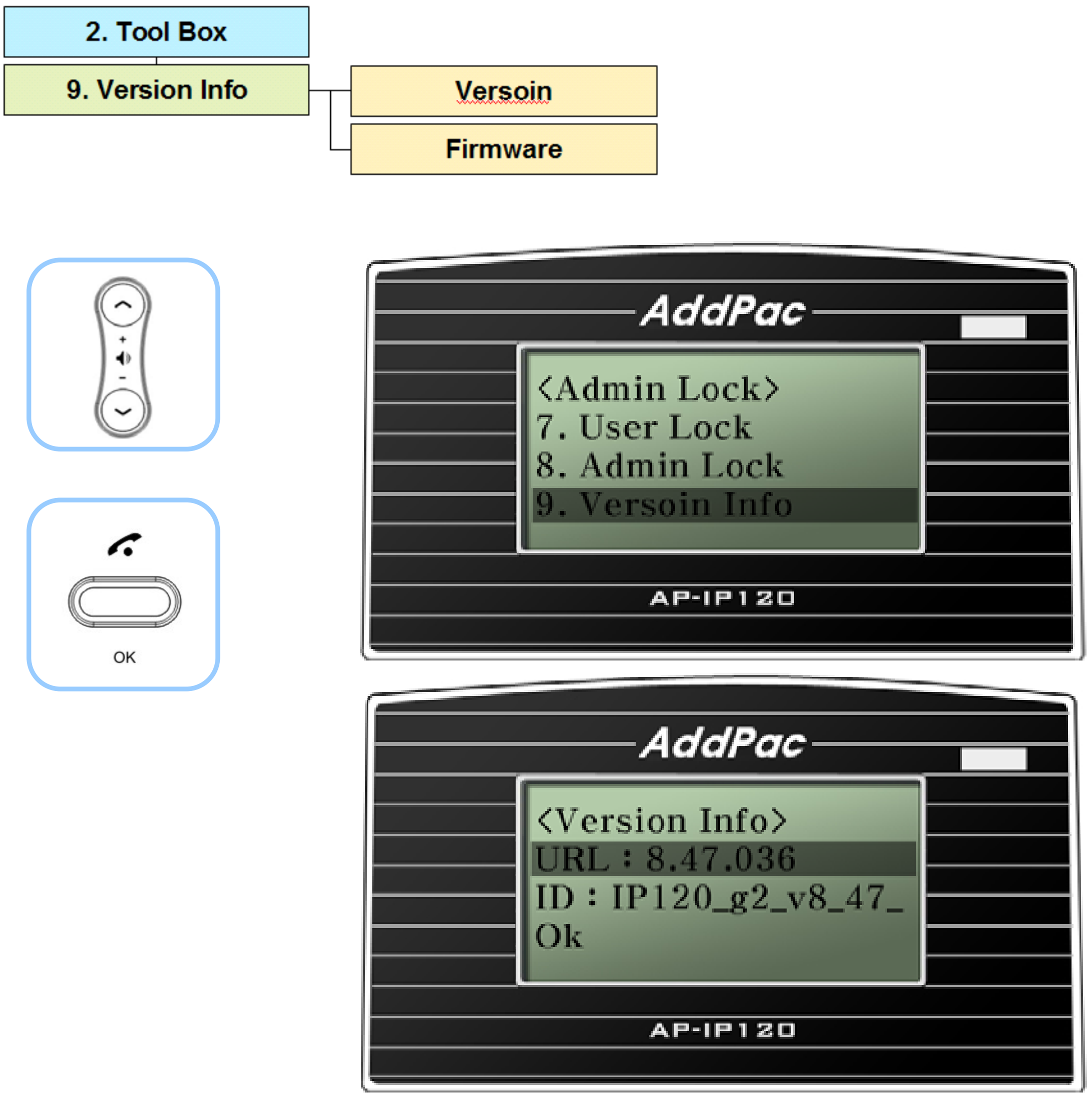


Set a password for Auto Upgrade

(Default : Disable)

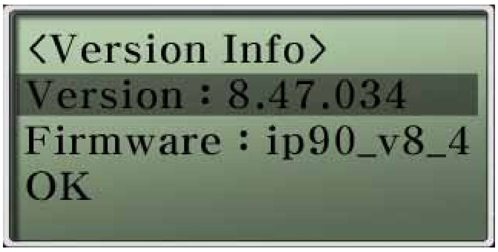
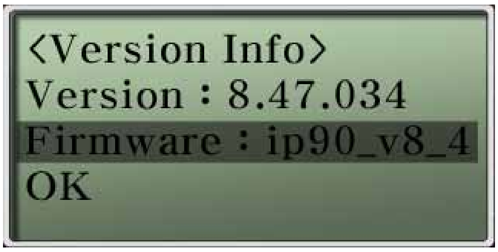
Tool Box-Version Info

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



(Figure 4-22) Version Info Menu Section

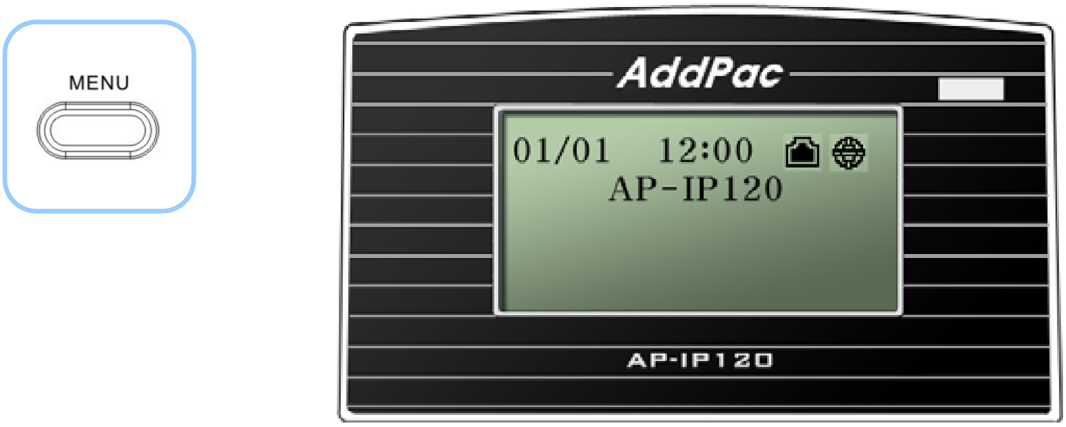
[Table 4-15] Description of Version Info Menu Option

Menu Detail	Description
	This is the version running at this present time
	This is the name of the firmware running at this present time.

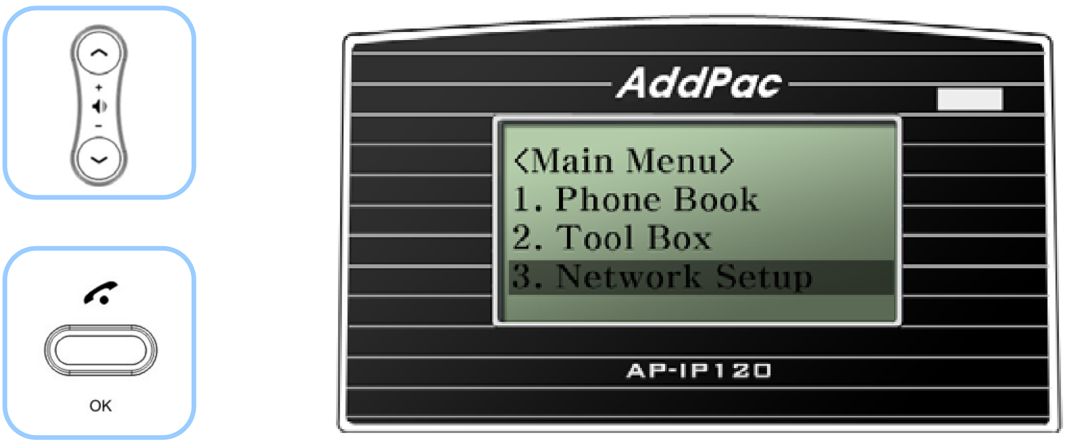
Network Setup Menu Category

The network setup of AP-IP120 is organized in details of WAN, LAN interface setting, SIP/H.323 signaling, FTP service, QoS for guaranteeing VoIP quality, call options and others.

This menu option is very important and the user must be familiar with them for using AP-IP120.



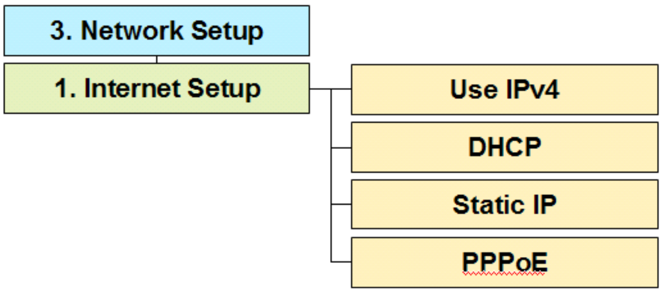
(Figure 4-23) Main Screen



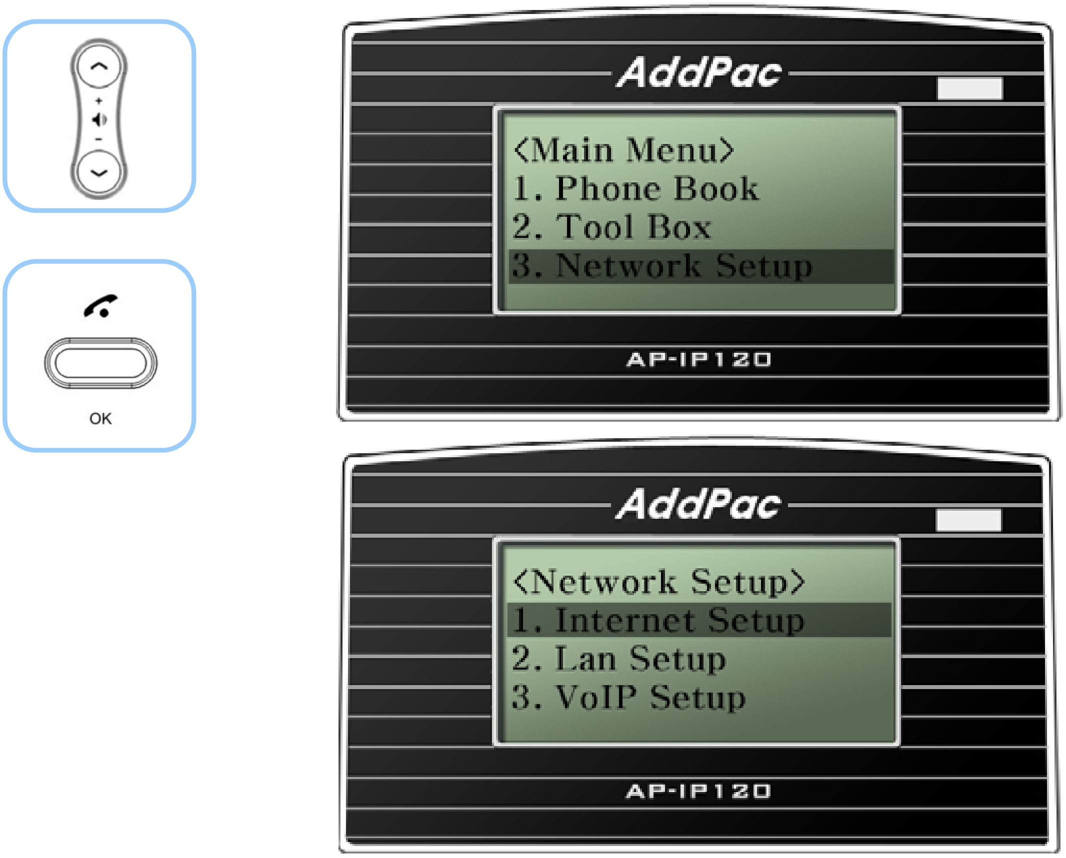
(Figure 4-24) Network Setup Menu Category

Network Setup – Internet

This internet menu has the optional features for the WAN interface. Since there are many different network settings, the user has to choose the features which are most suitable to the given. These WAN protocols supported by AP-IP120 are DHCP, static IPv4, PPPoE, and IPv6 etc.

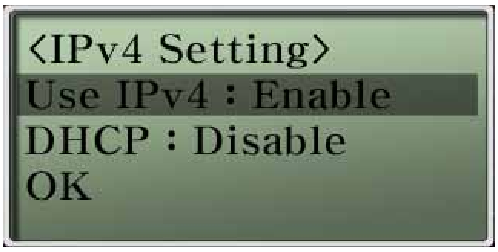
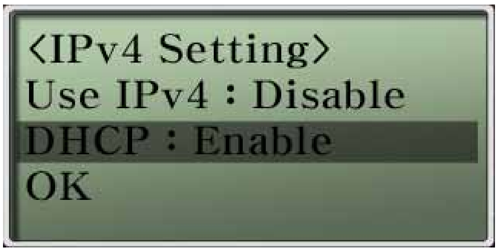
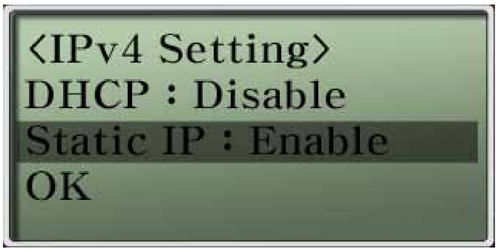
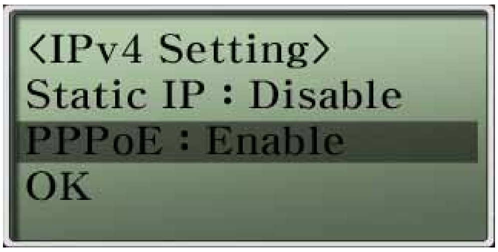


Internet – IPv4 Setting



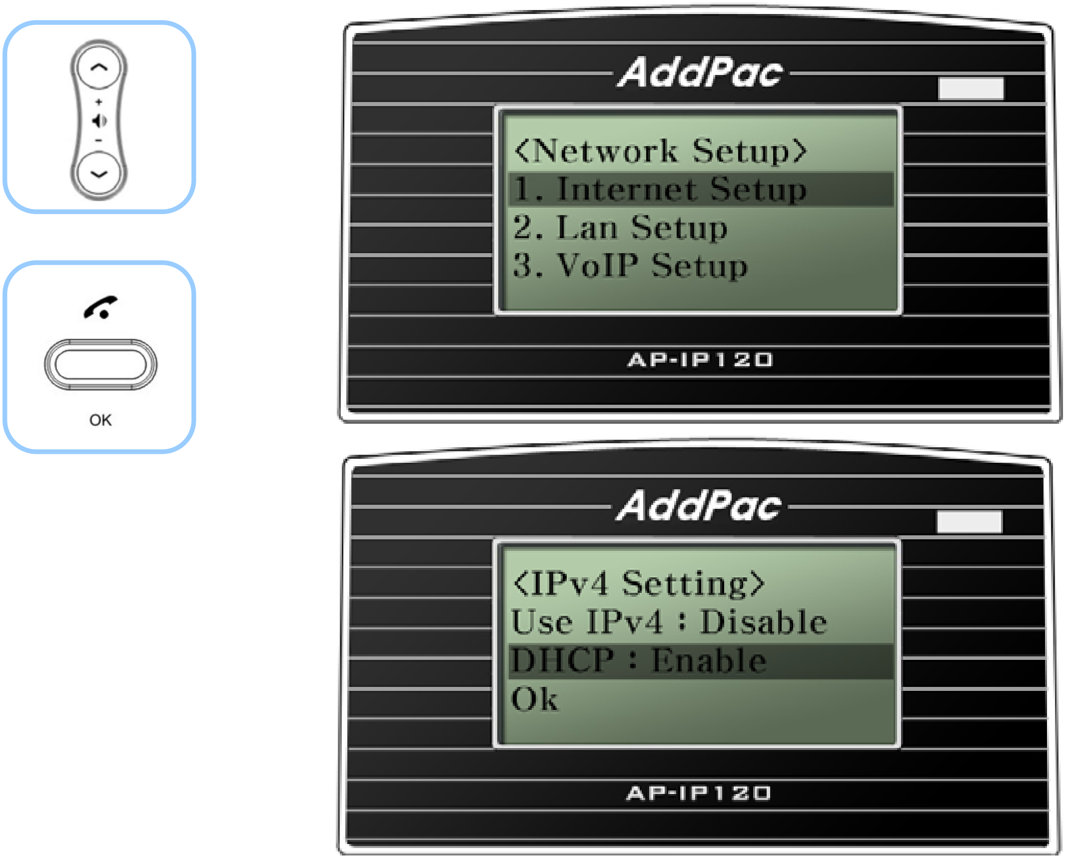
(Figure 4-25) IPv4 Setup Menu Section

[Table 4-16] Description of IPv4 Setting Menu Option

Menu Detail	Description
	Enable/ Disable IPv4
	Enable DHCP server to take a dynamic IP address fromWAN protocol such as cable , VDSL, ADSL modems
	Set IP address manually and build WAN interface such as static IP ADSL, E1/T1 leased line.
	Getting a dynamic IP address from PPP server such as ADSL.

IPv4 Setting– DHCP

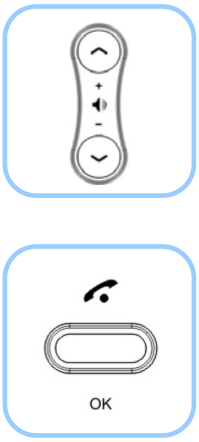
If user enables DHCP protocol in network setup internet menu, user gets a dynamic IP address from DHCP server such as cable modem, VDSL, IP-ADSL.

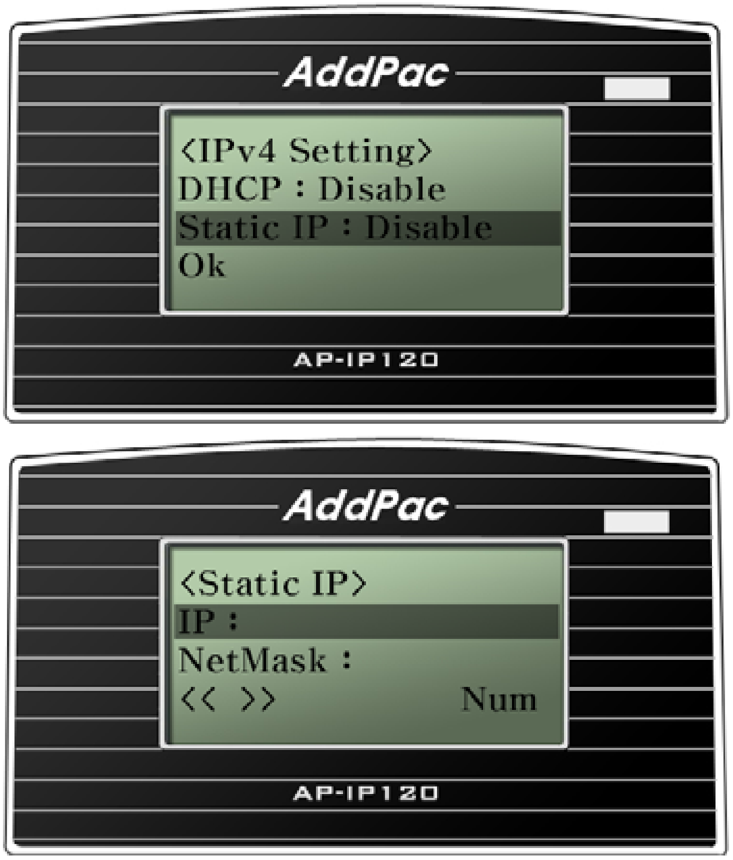


(Figure 4-26) IPv4 DHCP Setting Menu Detail

IPv4 Setting – Static IP

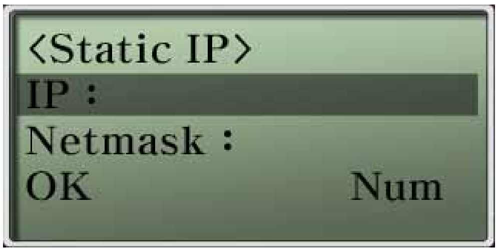
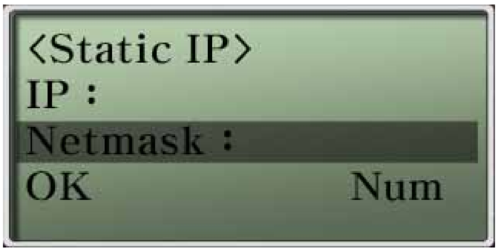
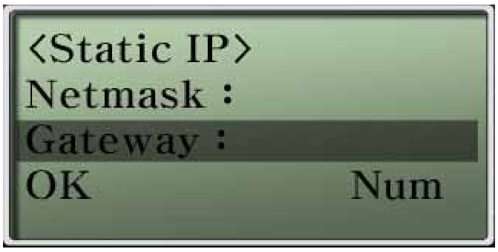
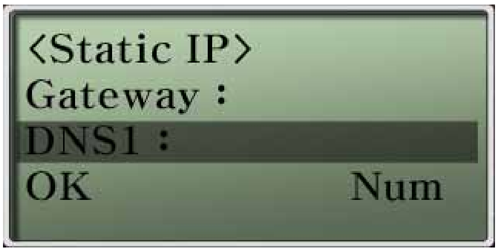
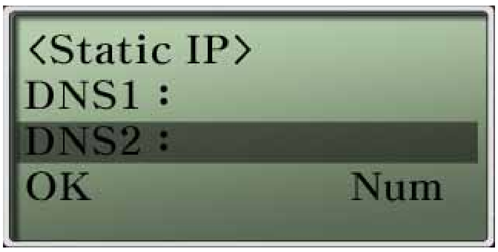
If you want to use static IP address, you may find this menu helpful to set a static IPv4 address manually and to configure WAN interface such as static IP ADSL, E1/T1 leased line, etc..





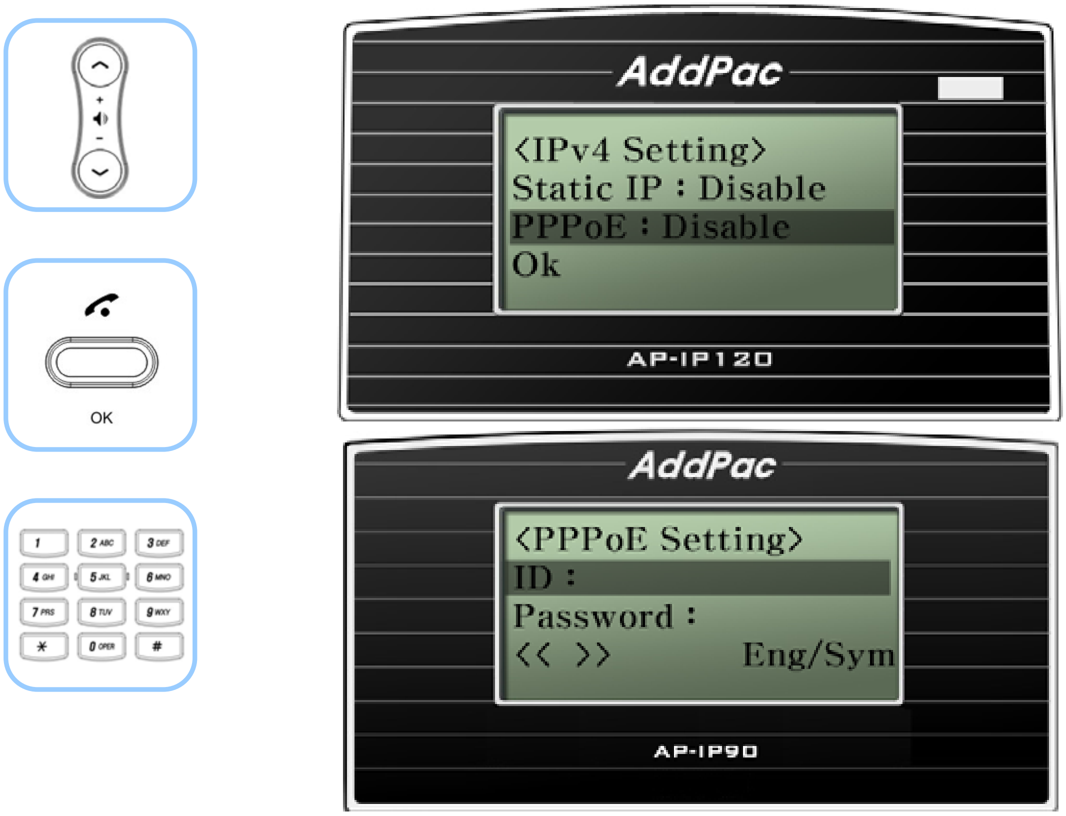
(Figure 4-27) Static IP Menu Detail

[Table 4-17] Description of Static IP Menu Option

Menu Options	Description
	IP Address Setting Ex> 172.20.1.100
	Network Mask Setting Ex> 255.255.0.0
	Default Router Setting Ex> 172.20.1.1
	First DNS setting (this menu can be applied at IPv6 mode) Ex> 168.126.63.1
	Second DNS Setting (Option) (this menu can be applied at IPv6 mode) Ex> 168.126.63.1

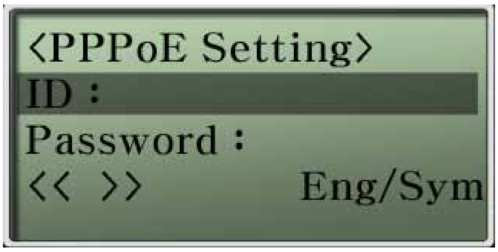
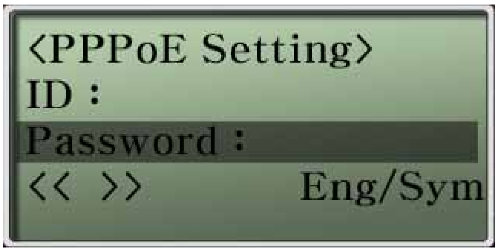
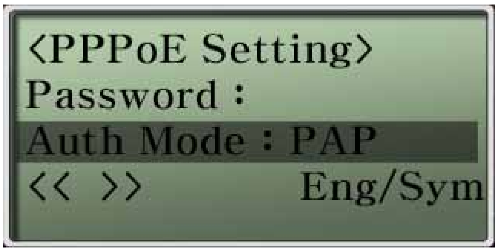
IPv4 Setting – PPPoE

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 4-28) PPPoE Setting Menu Detail

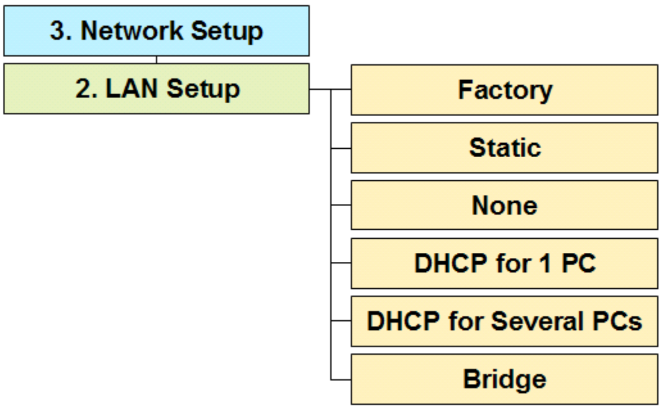
[Table 4-18] Description of PPPoE Setting Menu Detail

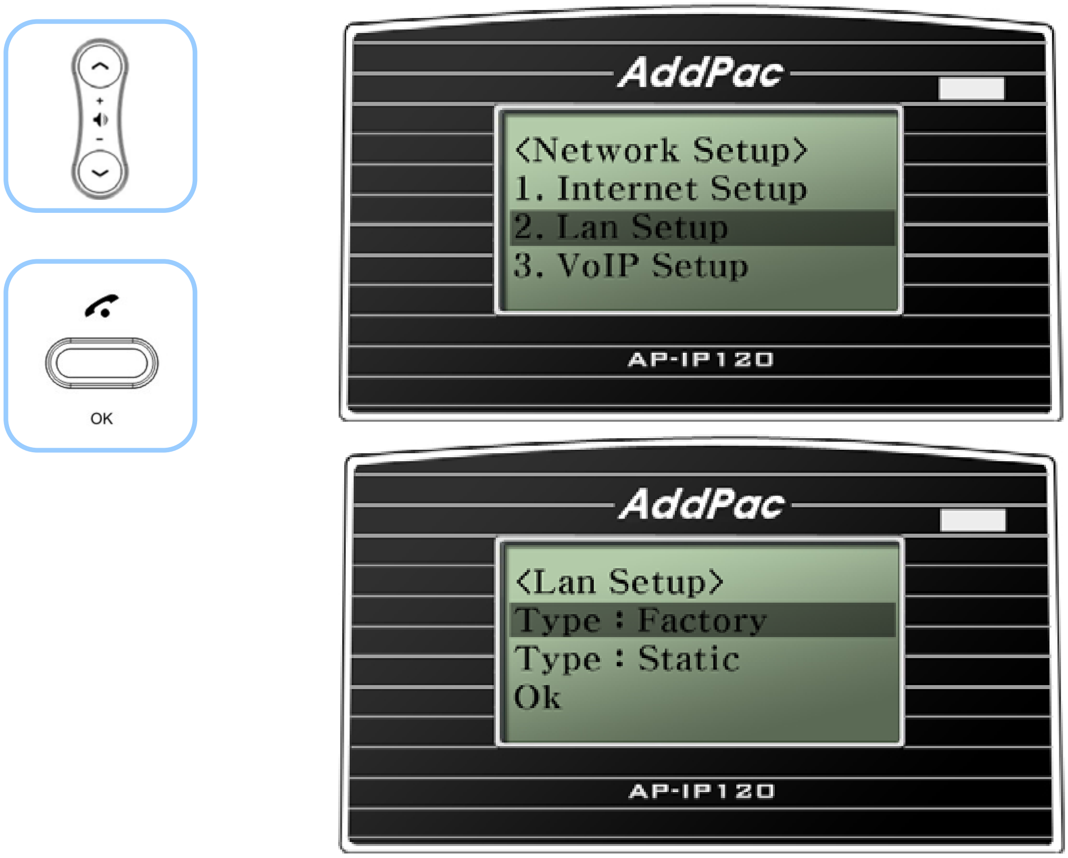
Menu Detail	Description
	Enter a user name Ex> Addpac
	Enter a user password Ex> 1234
	Enter an authentication mode, “PAP/CHAP” using alphanumeric key

* Press F4 key to change the input mode from the number to English or English to number.

Network Setup– LAN

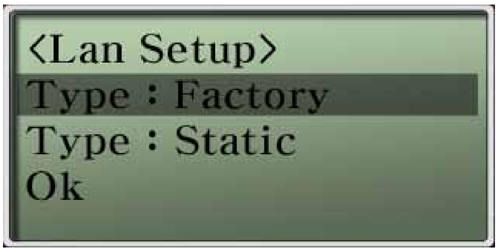
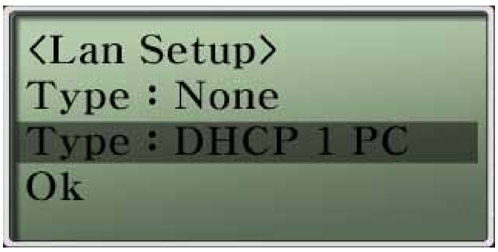
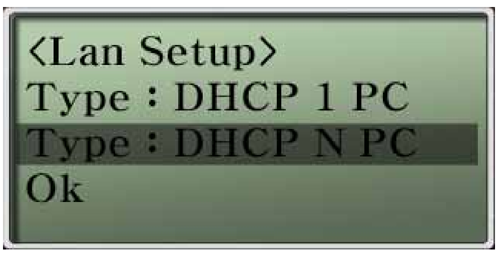
This LAN menu option is used for protocol setting of AP-IP120 second LAN interface which is used to connect PC or Ethernet Hub. None, DHCP for single (1) PC, DHCP for multiple PC are available as protocols for second fast ethernet LAN port.

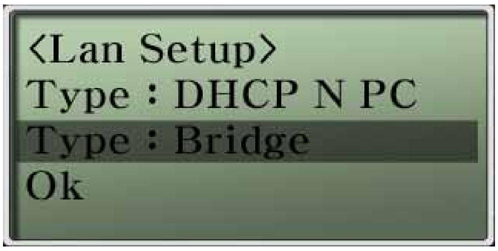




(Figure 4-29) Lan Setup Menu Section

[Table 4-19] Description of LAN Setup Menu Option

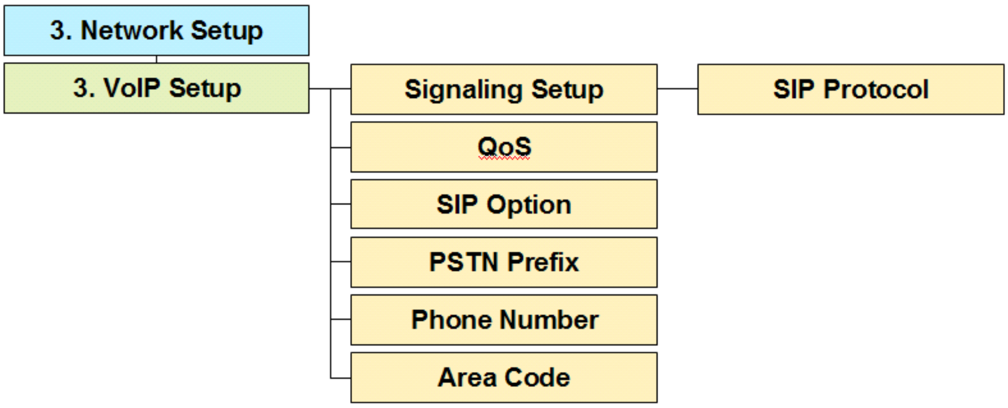
Menu Option	Description
	Set LAN to the factory default mode. The default is set to NAT.
	Configure LAN (the user sets the configuration)
	Disable LAN Setup (Press OK button to select this option))
	Share the same IP address of the PC connected to LAN1 of AP-IP120
	Use this option to connect more than 2 PC to the router for IP sharing



Configure Lan Setup to Bridge mode.

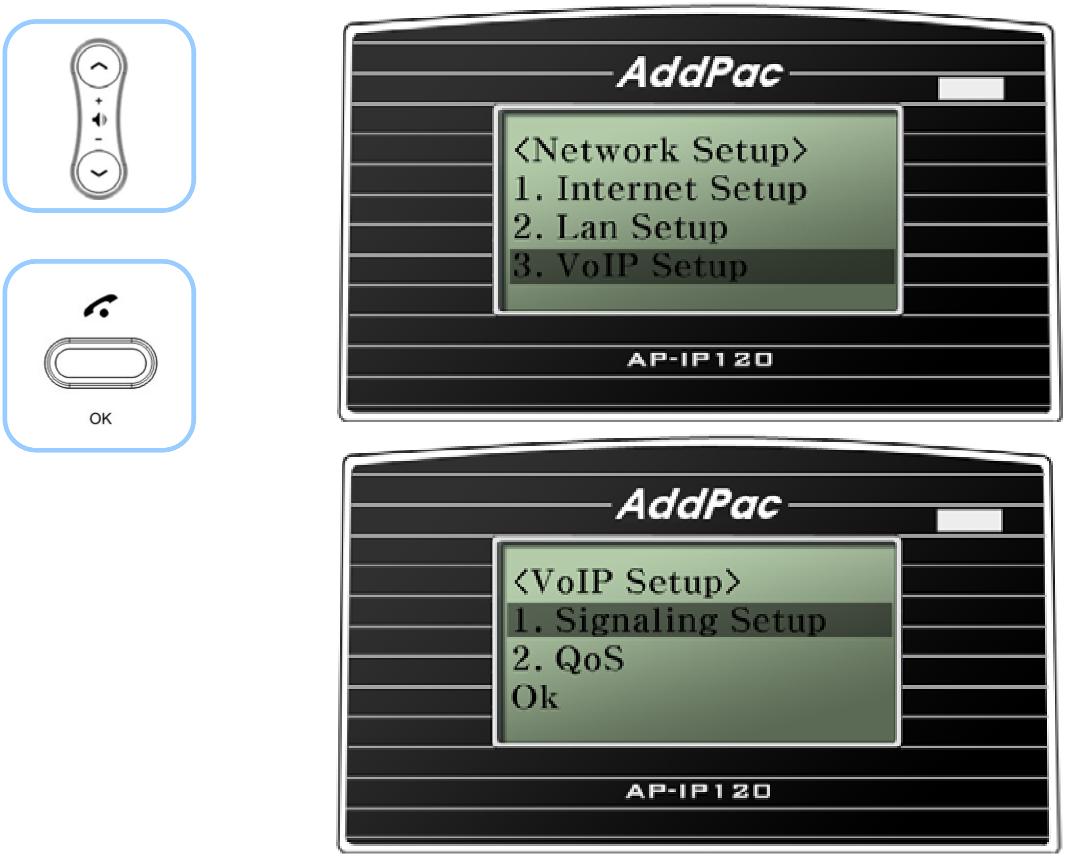
Network Setup– VoIP Setup

This VoIP setup menu is used for SIP/H.323 VoIP signaling inter-working parameter setting between SIP server, Gatekeeper and AP-IP120 IP Phone..



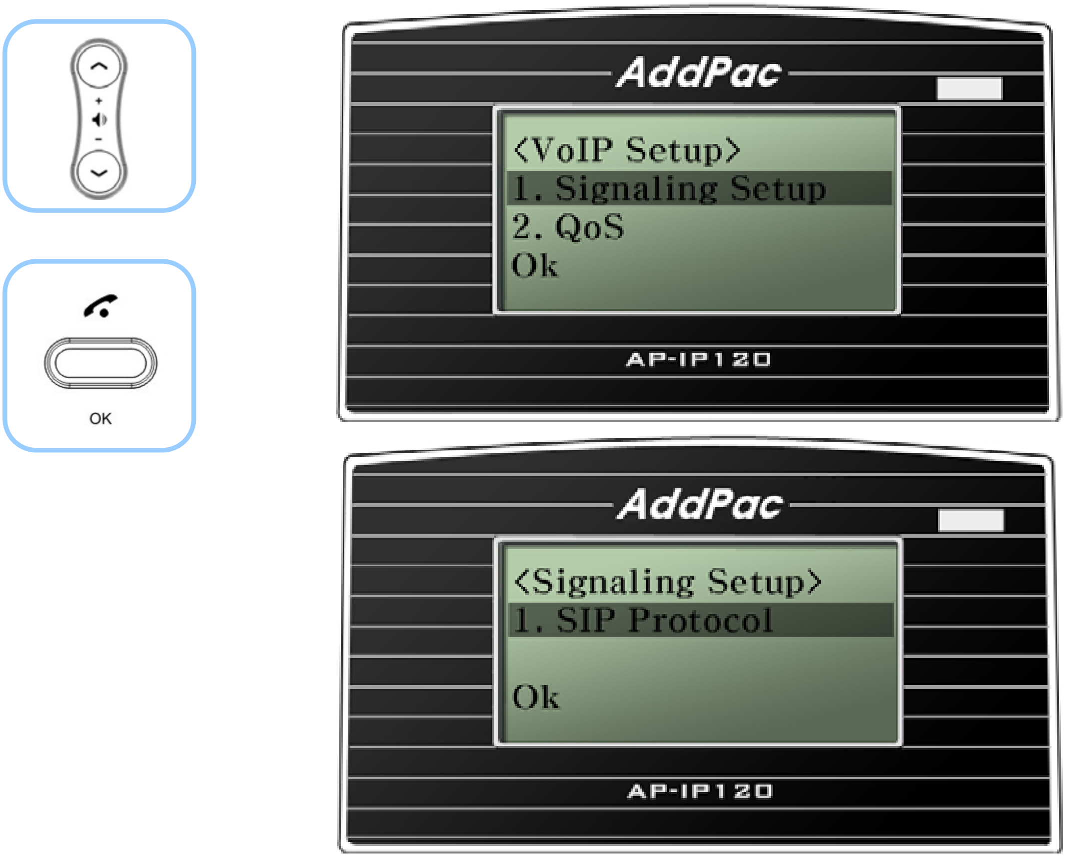
VoIP Setup – Signaling Setup

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.



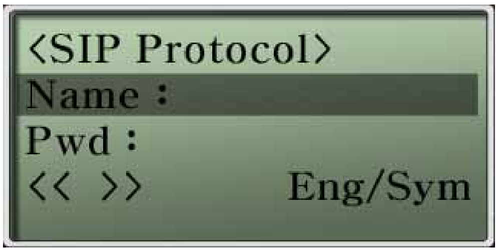
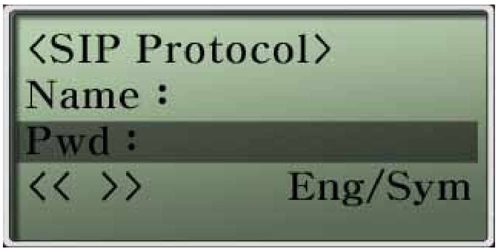
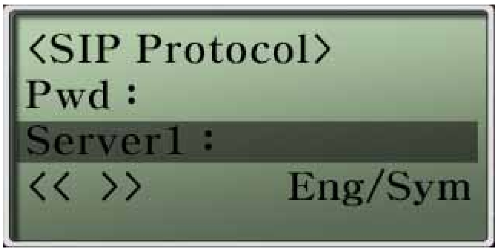
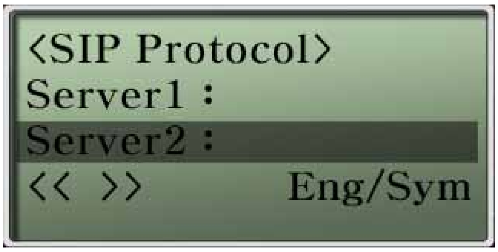
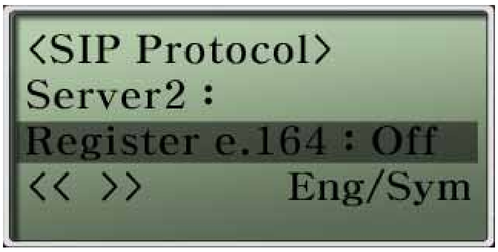
(Figure 4-30) Signaling Setup Menu Option

Signaling Setup – SIP Protocol



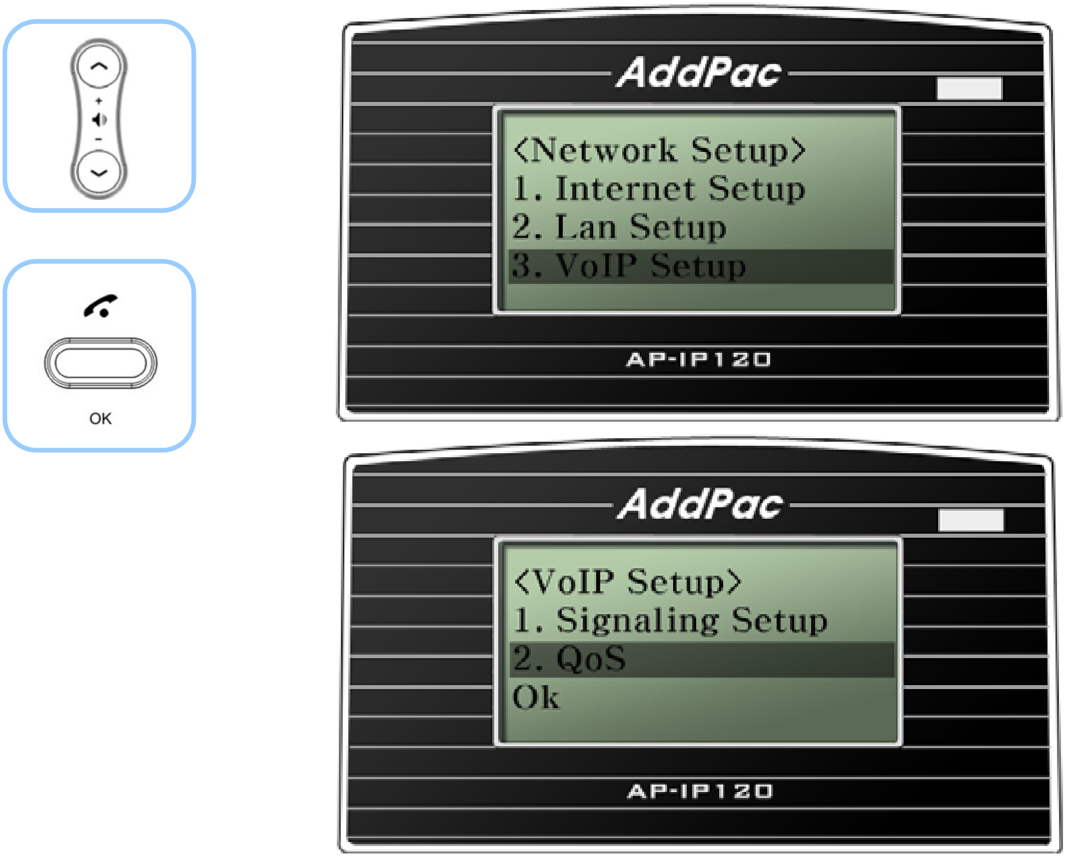
(Figure 4-31) SIP Protocol Menu Detail

[Table 4-20] Description of SIP Protocol Menu Detail

Menu Detail	Description
	Enter a username for SIP server registration
	Enter a user password for SIP server registration.
	Enter a primary server IP address or domain of SIP server.
	Enter a secondary server IP address or domain of SIP server.
	Select ON/OFF mode whether E.164 address is registered in SIP server or not

VoIP Setup– QoS

QoS sets a limit to transmitting voice packet within the bandwidth which is guaranteed by the network capacity.



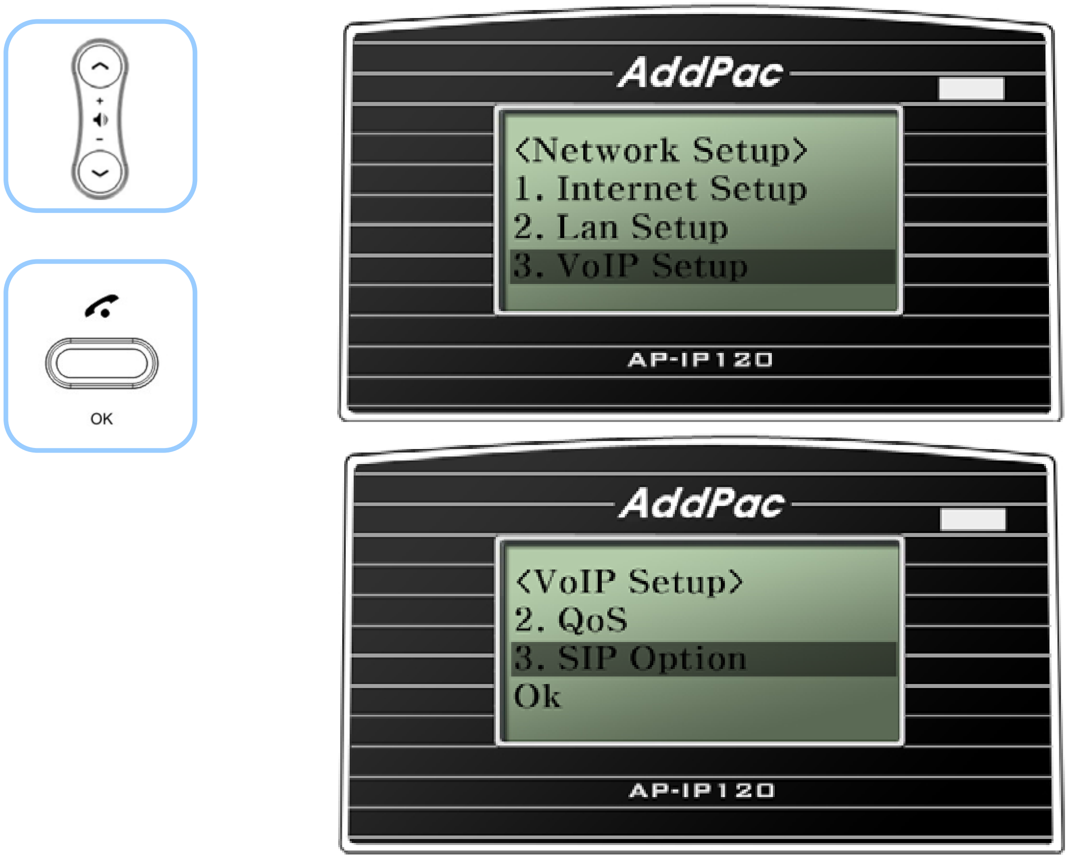
(Figure 4-32) QoS Menu Option

[Table 4-21] Description of QoS Menu Option

Menu Detail	Description
	Disable QoS
	Enable QoS
	Set QoS to WAN interface in the range of 48Kbps~4Mbps

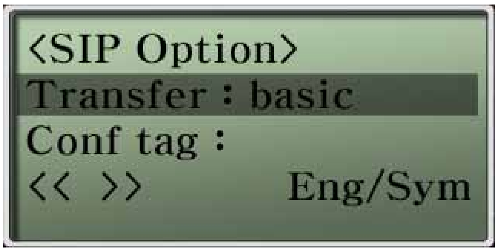
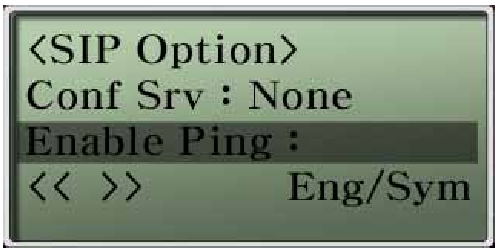
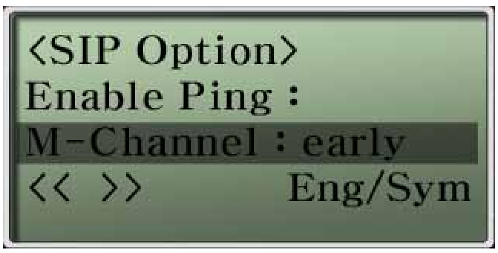
VoIP Setup–SIP Options

These menu options are to set up other supplementary functions and options of SIP protocol in addition. These options need to be setup basing on the network settings.

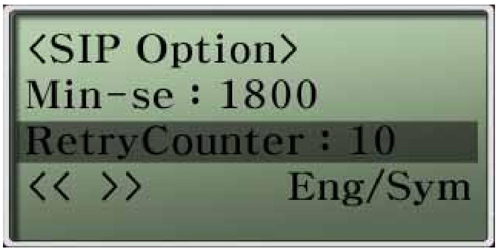


(Figure 4-33) SIP Option Menu Option

[Table 4-22] Description of SIP Option Menu Option

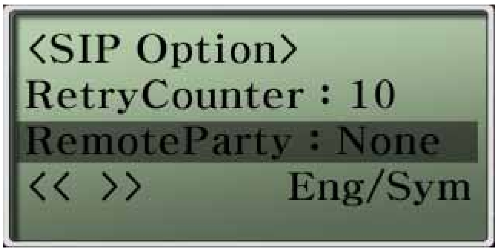
Menu Option	Description
	Select the call-transfer mode. basic/attend.
	Enter VoIP Tag for conference service
	Enter conference service name
	Enter firewall address to check the public IP address when AP-IP120 is used under NAT/Firewall network environment.
	Transfer RTP Session information to listen Inband Ringbacktone of Public network under NAT/Firewall environment.

Set Session Timer



SIP UA Retry Counter sets SIP INVITE re-transmission count when AP-IP120 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-IP120 transfer next INVITE message.

Default is 10 and usually set as '3.'

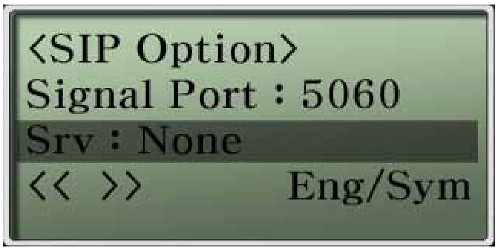


When user-name is not number but characters, apply to register message.

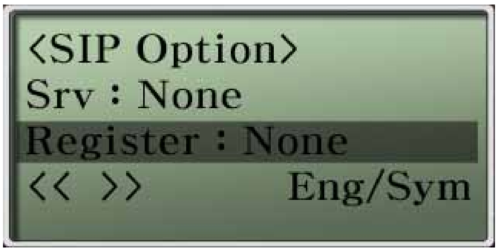
When called party is not number but characters, this option is used.

Transfer From/To field within SIP message to designated domain not to IP address.

The default is set to 5060 and this setting can be changed



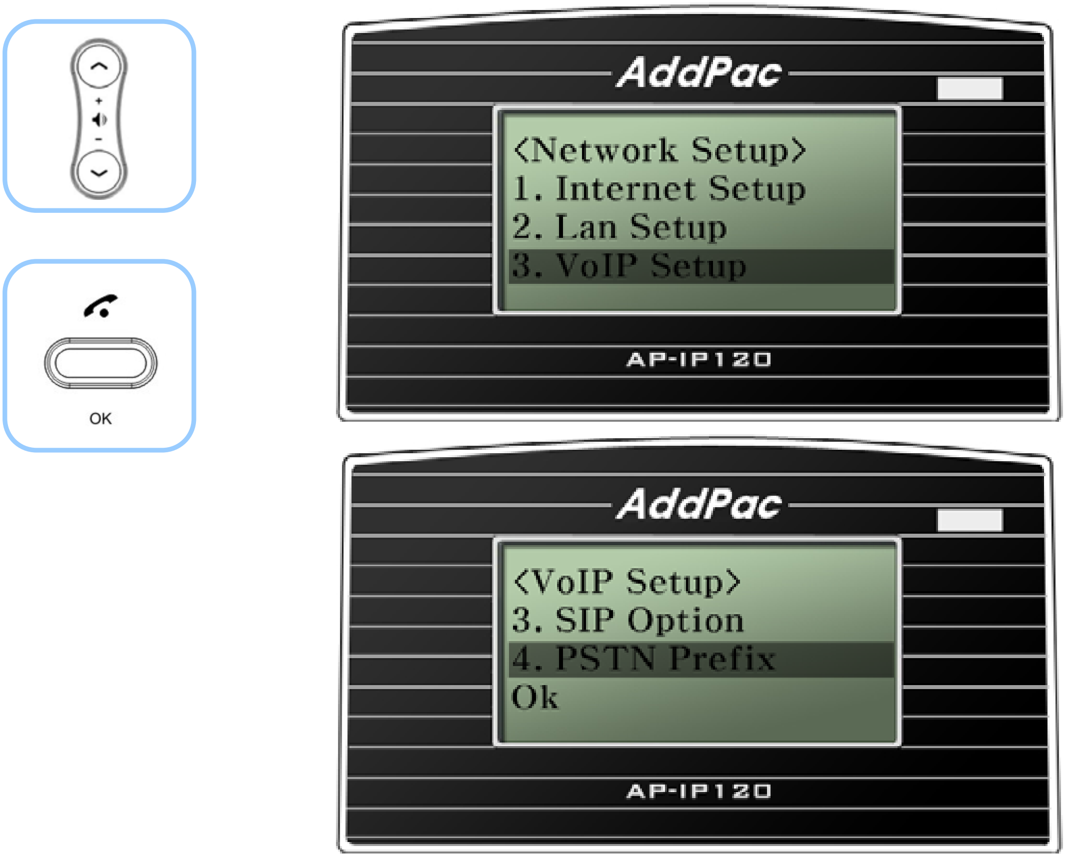
Set the DNS SRV.



When user-name is not number but characters, this option is used to register SIP server.

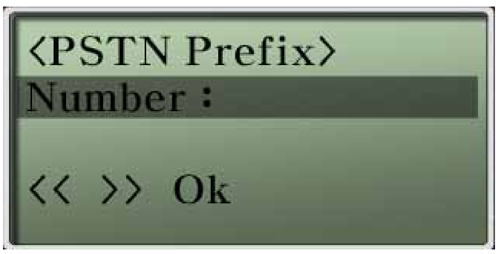
VoIP Setup– PSTN Prefix

When user wants to access the FXO interface for PSTN backup, this prefix number is used as PSTN access code. Additionally, AP-IP120 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 accident.



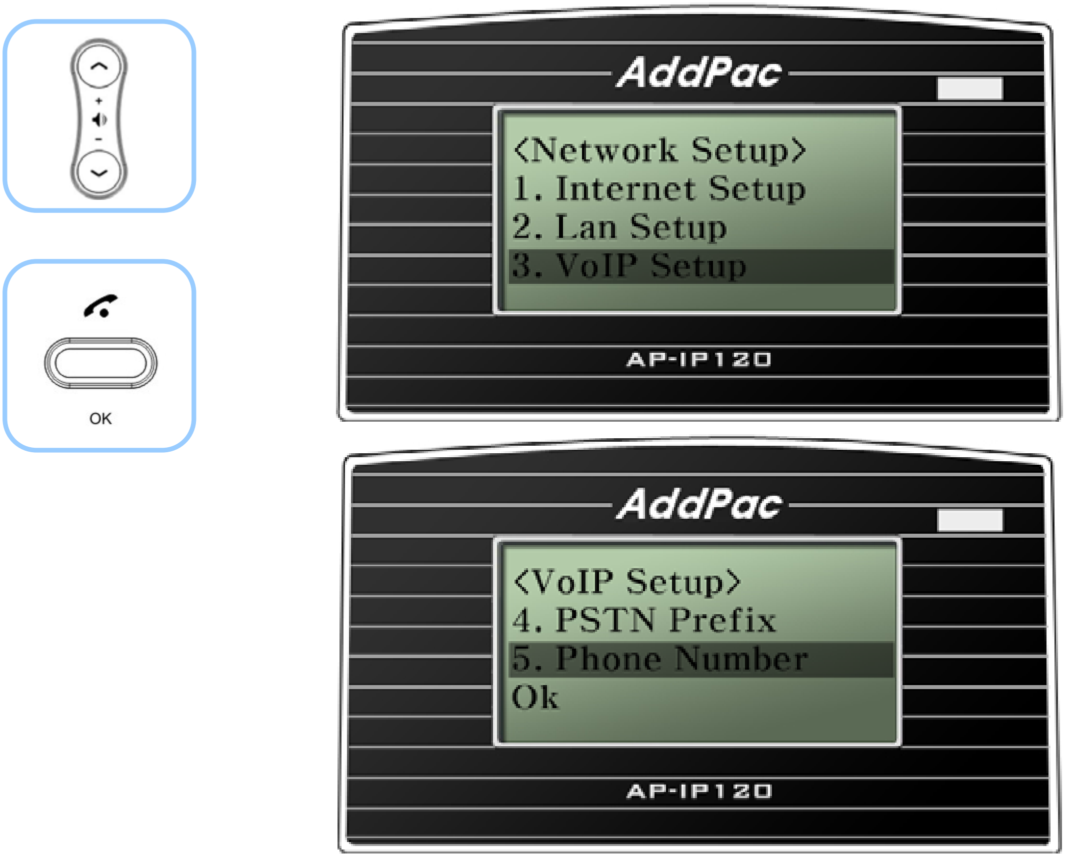
(Figure 4-34) PSTN Prefix Menu Option

[Table 4-23] Description of PSTN Menu Option

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

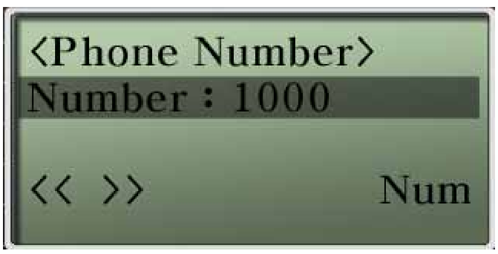
VoIP Setup– Phone Number

This menu is used for E.164 AP-IP120 Number Assignment.



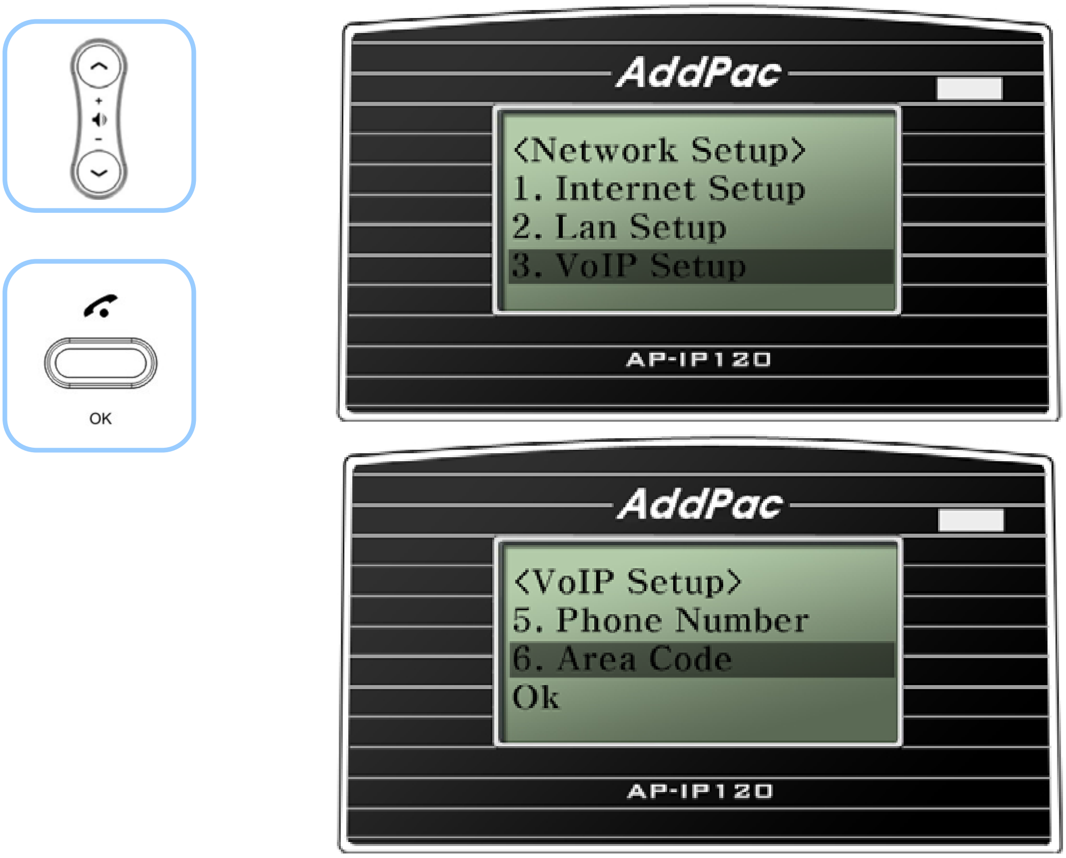
(Figure 4-35) Phone Number Menu Option

[Table 4-24] Description of VoIP Menu Option

Menu Detail	Description
	Enter the E.164 Phone Number.

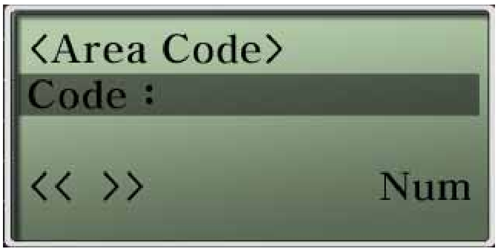
VoIP Setup – Area Code

In this menu option, the user can specify an area code.



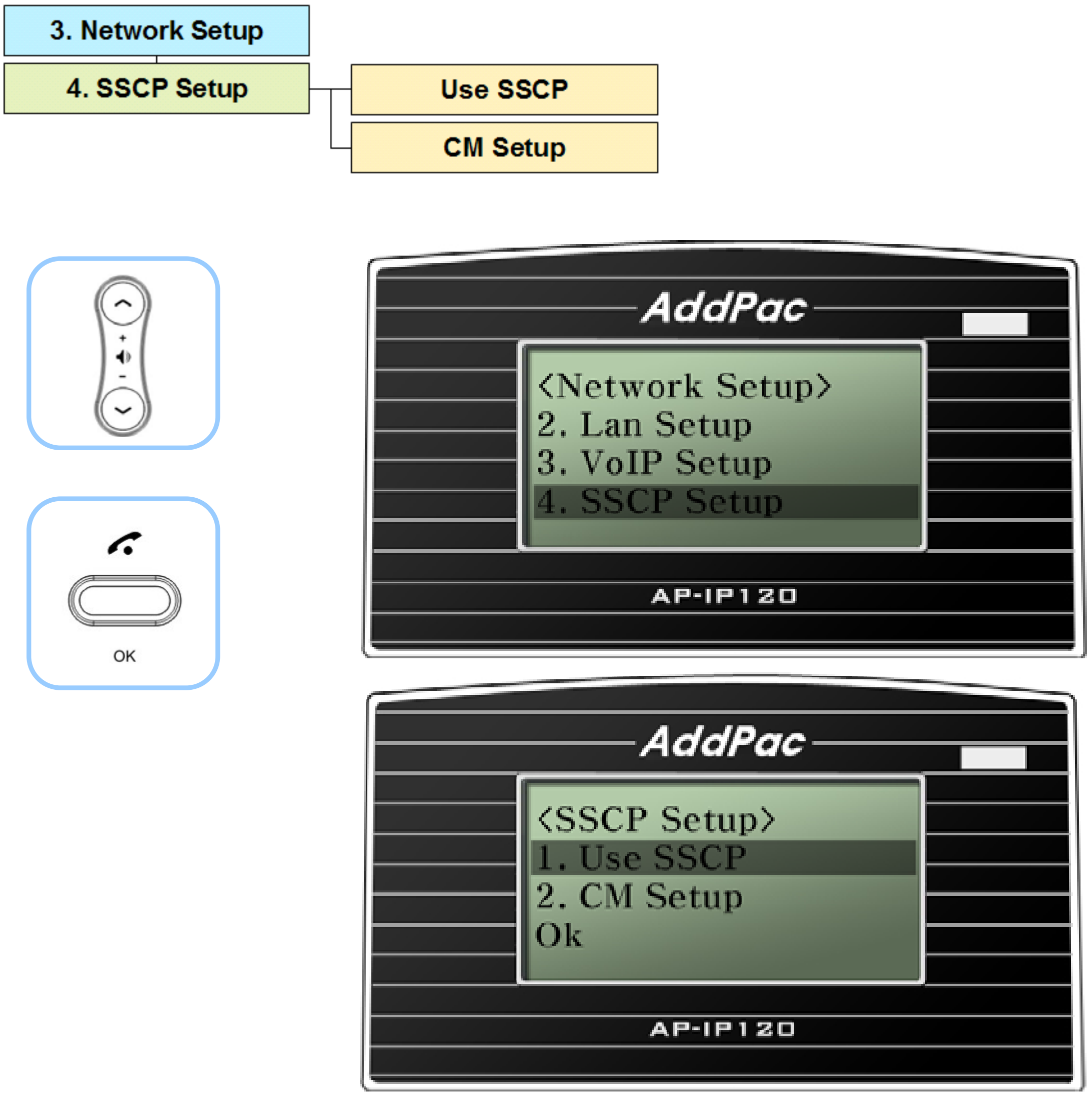
(Figure 4-36) Area Code Menu Option

[Table 4-25] Description of Area Code Menu Option

Menu Detail	Description
	Enter an area code to be used

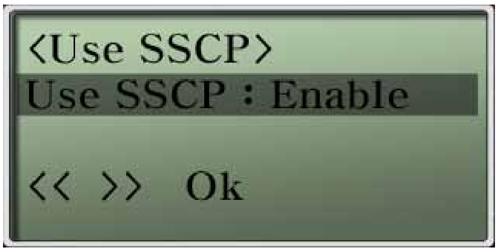
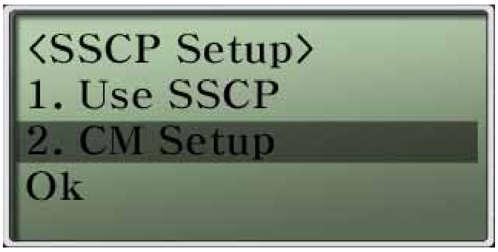
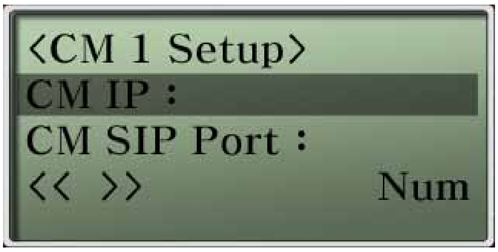
Network Setup – SSCP (Smart Service Control Protocol)

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 4-37) SSCP Setup Menu Section

[Table 4-26] Description of SSCP Setup Menu Option

Menu Option	Description
	Either enable or disable the setting mode of SSCP
	Make a selection of Call-Manager Setup
	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)
	Set IP of Call Manager server
	Set SIP port of the Call Manager server (the default is set 5060)

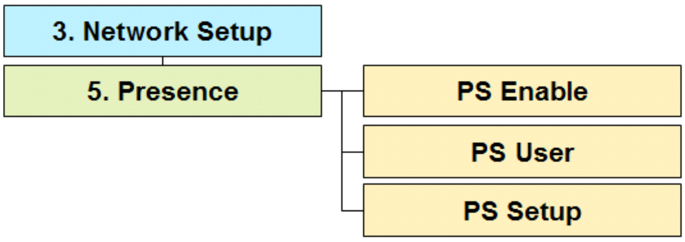
Network Setup– Presence

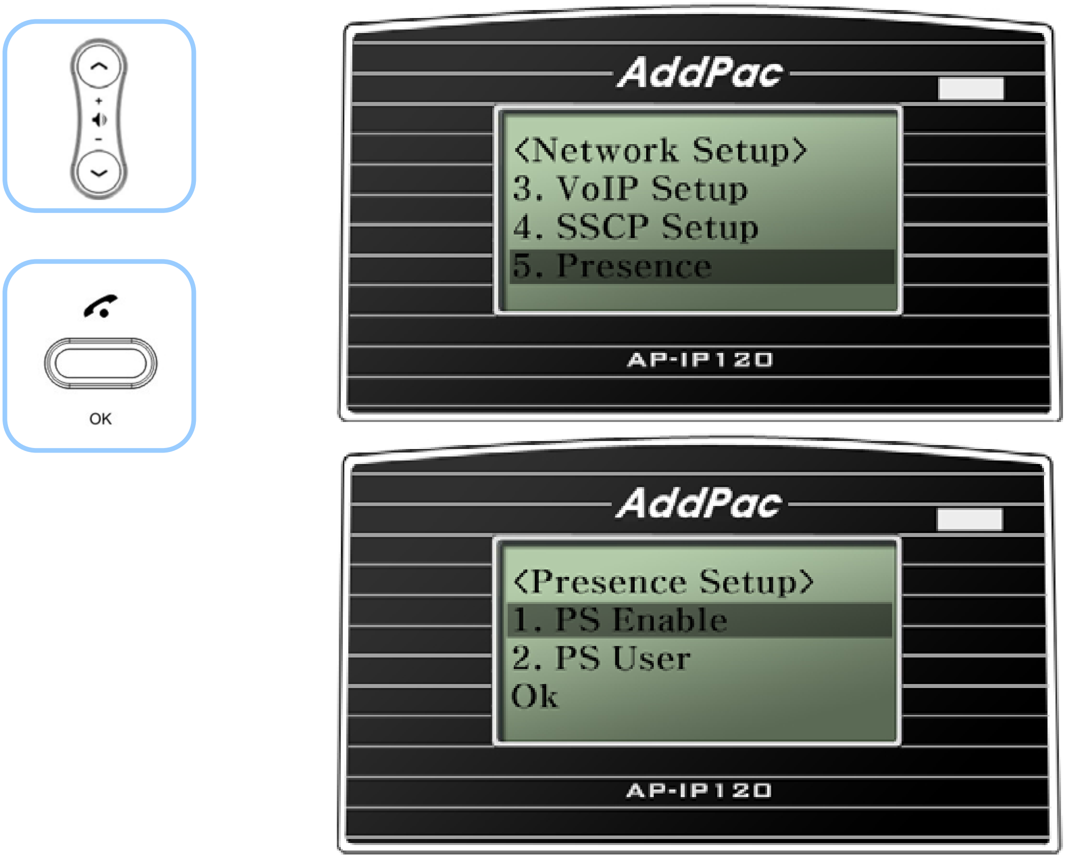
Presence menu section provides the user presence service from Smart Multimedia Manager which operates in MS Window-based PC Platform environment as to interwork with the AddPac IP-Next PBX.

Interworking with Presence Server (integrated in IP-PBX), the user presence can be used by Smart Messenger which is PC-based and Speed Button List in the terminal.

Smart Messenger interacts with the AddPac Next Generation terminals (IP Phone and Video Phone) and provides various services such as messenger service, phonebook, user presence, Unified Message Service (Voice Mail, Short Message), Call Control and Forward Setup.

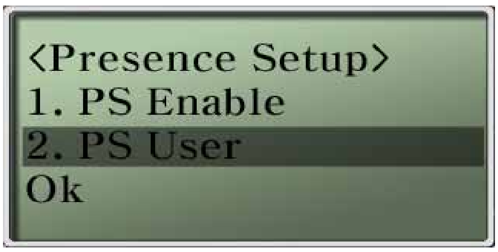
Smart Messenger operates, in the basis of Smart Service Control Protocol (SSCP), which is the AddPac proprietary protocol, between the AddPac IP-PBX and IP terminals.

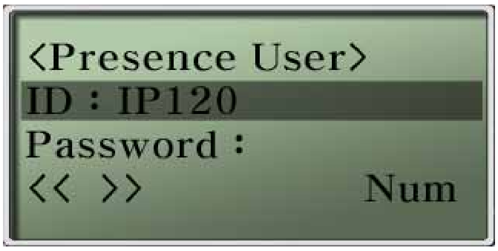




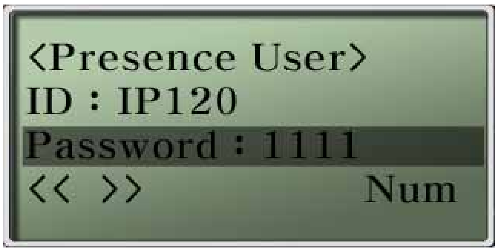
(Figure 4-38) Presence Menu Section

[Table 4-27] Description of Presence Menu Option

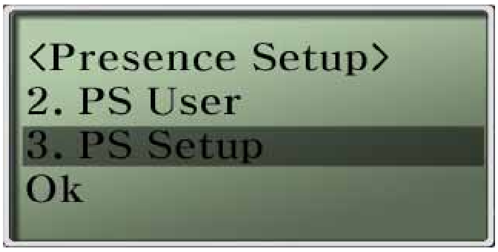
Menu Option	Description
	Enable or disable the presence server Setup (Enable/Disable)
	Presence User정보를 입력하는 메뉴입니다.



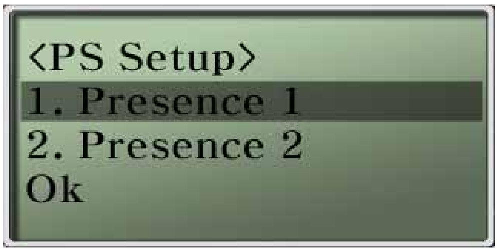
Presence Server에 등록 된 ID를 입력합니다.



Presence Server에 등록 된 ID에 해당하는 Password를 입력합니다.



Presence Server를 설정하는 메뉴입니다.



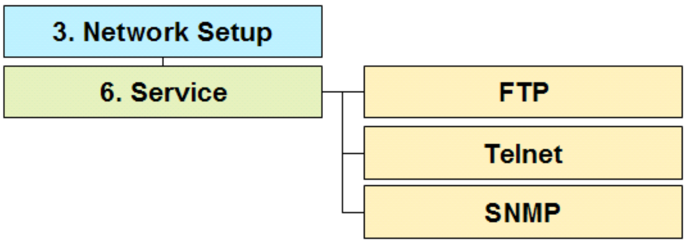
등록 할 Presence Server를 설정합니다.

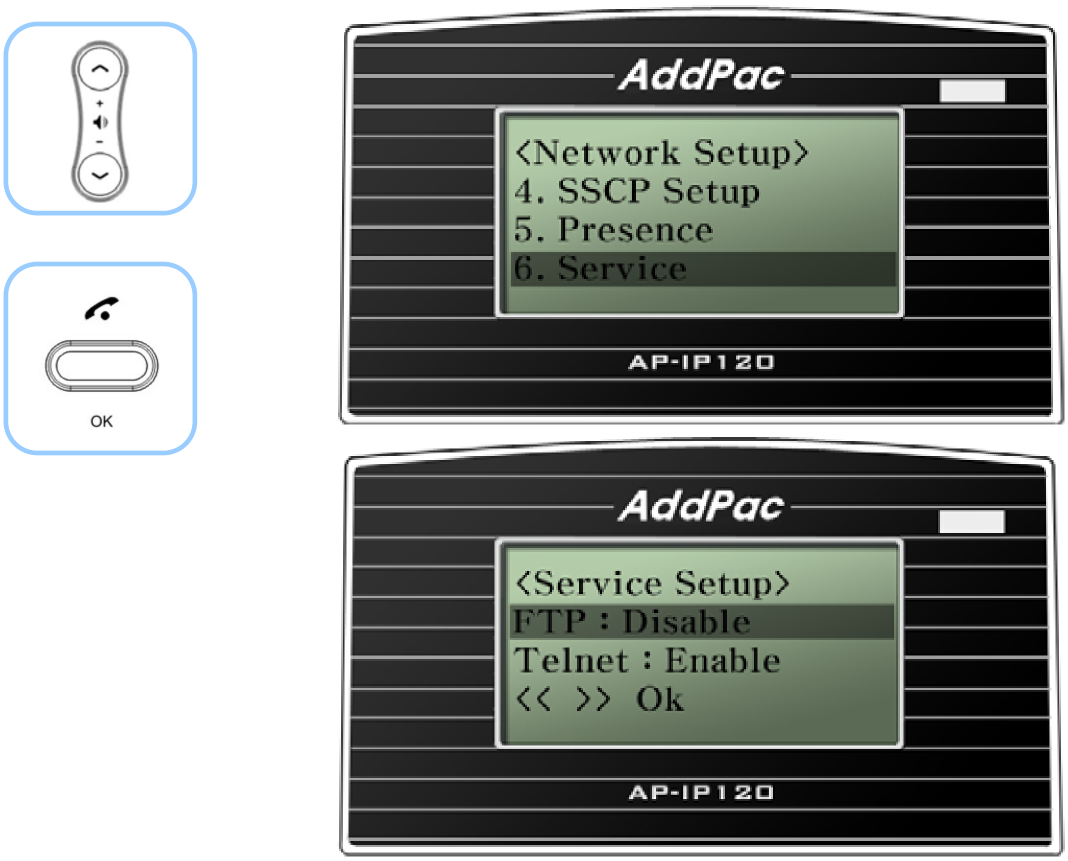
선택한 Presence Server의 IP, Port정보를 입력합니다.

Network Setup – Service

This menu enables or disables FTP, TELNET, TFTP, SNMP protocol service of AP-IP120.

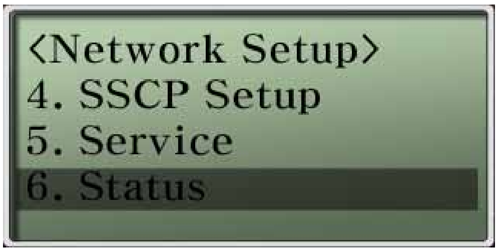
You can use FTP to access to AP-IP300 from a remote location and Telnet is used for changing all kinds of information and monitoring and SNMP is also used to access to AP-IP120 from a remote location.





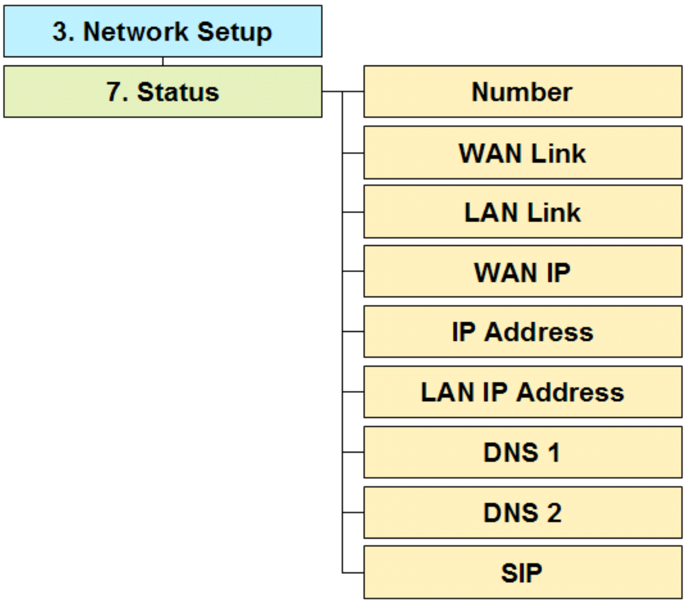
(Figure 4-39) Service Setup Menu Option

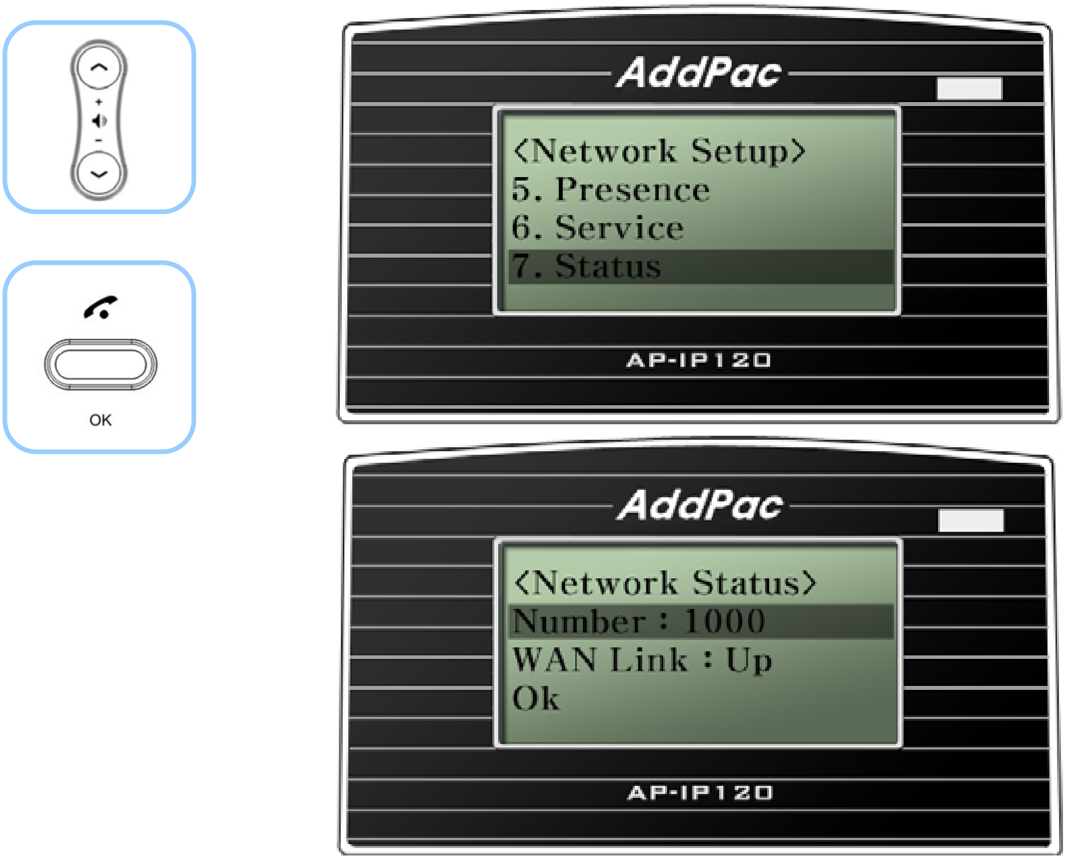
[Table 4-28] Description of Service Setup Menu Option

Menu Option	Description
	Actives/Deactivates the FTP service protocol. Default is enable mode (activating FTP service). Default port number is 21.
	Actives/Deactivates the TELNET service protocol. Default is enable mode (activating TELNET service). Default port number is 23.
	Actives/Deactivates the SNMP service protocol. Default is enable mode (activating SNMP service). Default port number is 161

Network Setup – Status

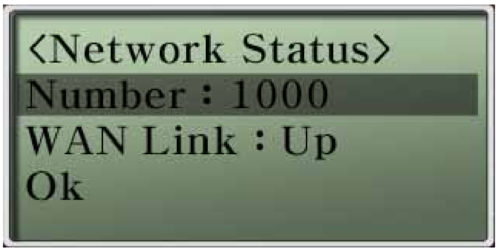
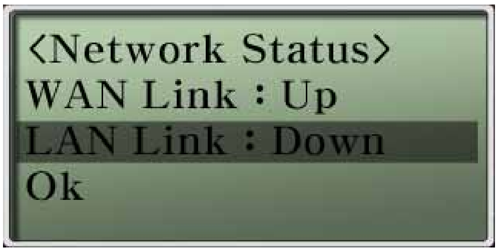
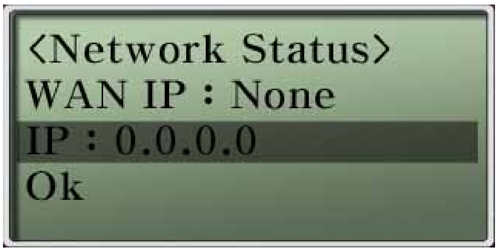
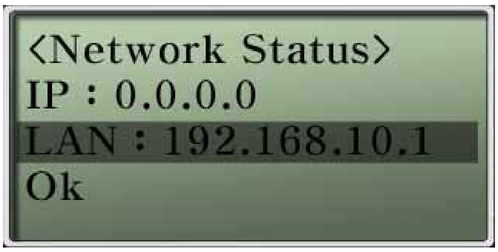
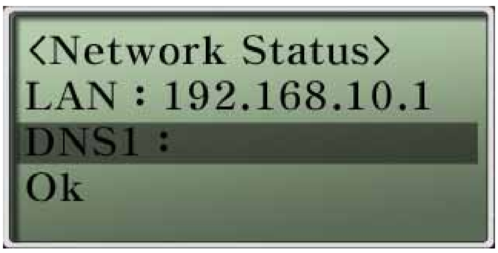
This setup allows you to see the current status of the network on the screen such as Link Status , IPv4 Protocol, IPv4 address, Lan address, DNS, SIP Proxy Server, status of registration to GK[H.323], at a glance.

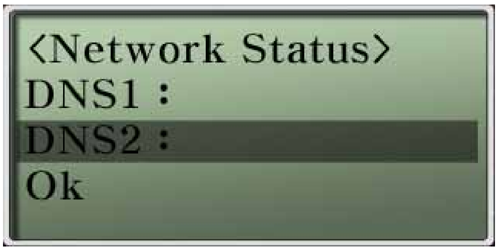




(Figure 4-40) Status Menu Option

[Table 4-29] Description of Network Status Menu Detail

Menu Detail	Description
	Displays the phone number
	Indicates the status of the link: whether the link of LAN0 (WAN interface) is up or down
	Indicates the status of the link: whether the link of LAN1 (PC interface) is up or down
	Displays WAN IPv4 address T
	PC LAN Port(LAN 1)의 address Table시
	Displays Domain Name Server 1

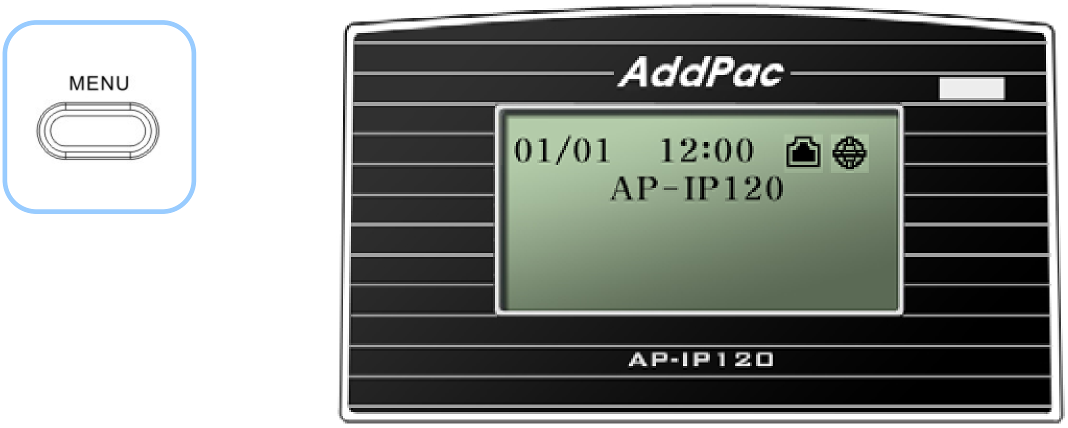


Domain Name Server 2 Table시

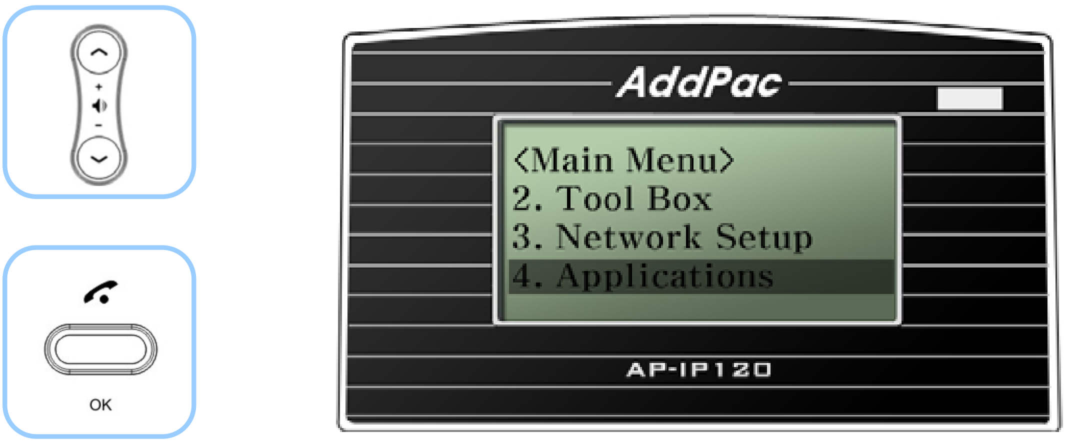
Displays the registration status of SIP-Proxy Server
(Registered, Not registered)

Applications Menu Category

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).



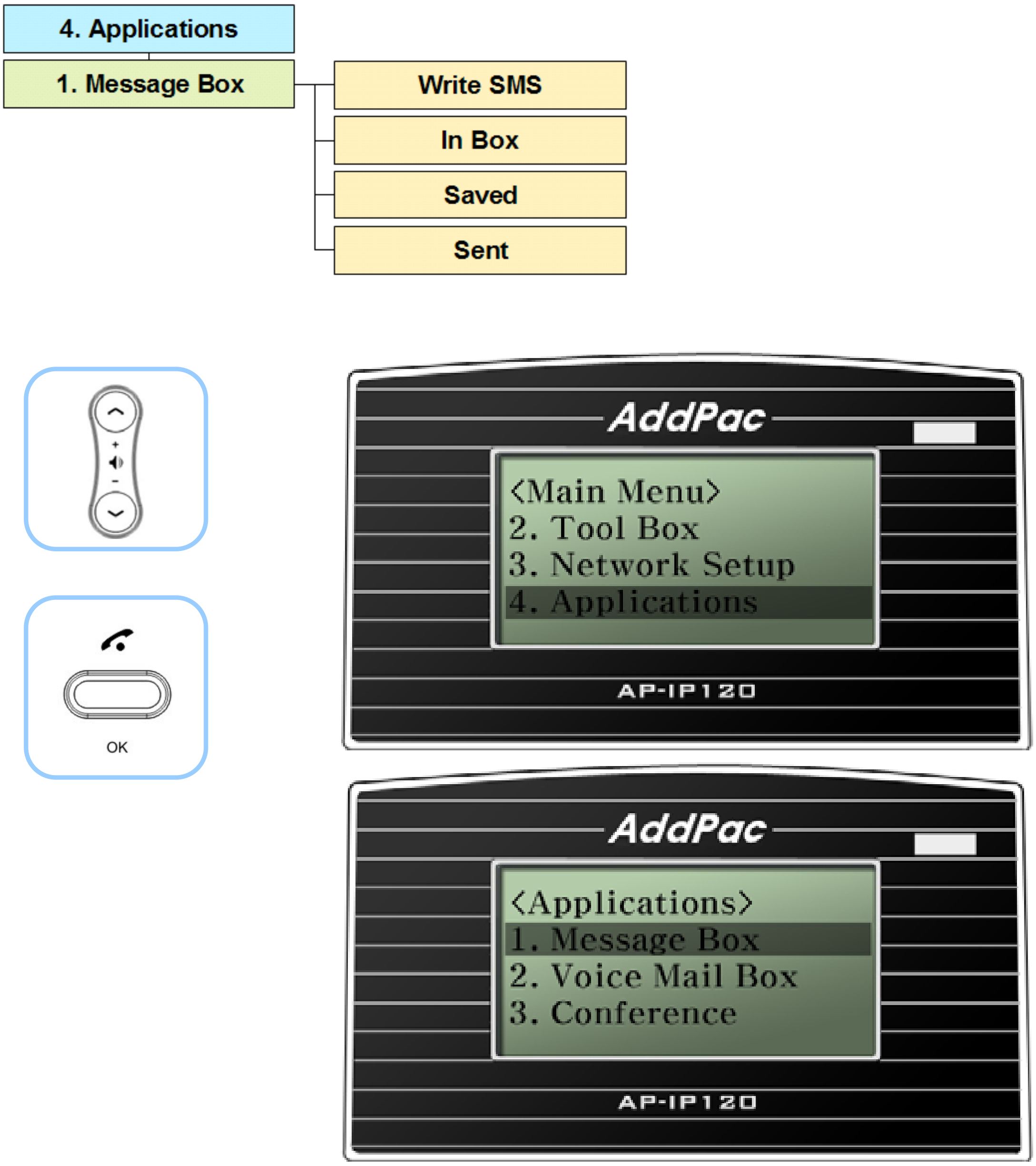
(Figure 4-41) Main Screen



(Figure 4-42) Applications Menu Option

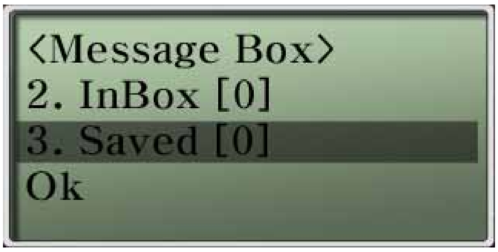
Applications – Message Box

The Message Box allows you to transmit and verify SMS text messages between one terminal and the other, when it is connected to SSCP.

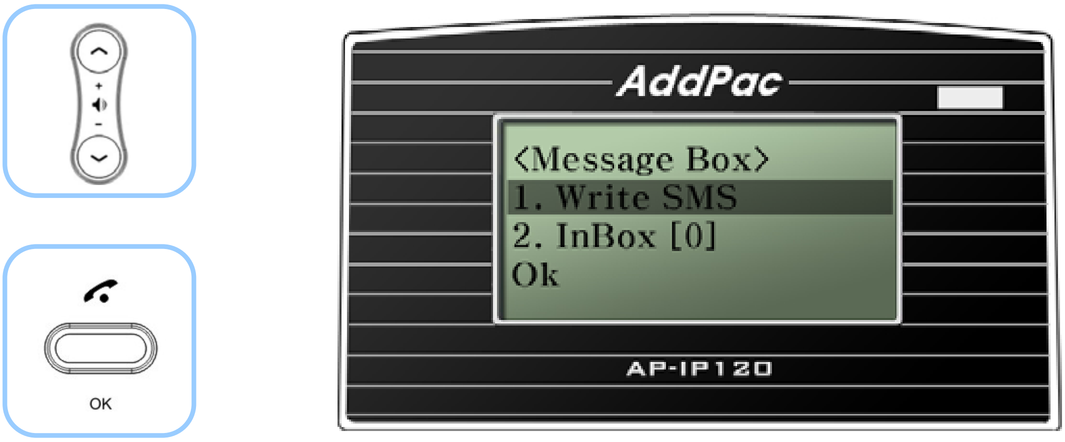


(Figure 4-43) Message Box Menu Section

[Table 4-30] Description of Message Box Menu Option

Menu Option	Description
	Write SMS messages to be sent
	Store the SMS messages in the box which have been received
	Save the message in the box which have been sent
	Store the messages in the box which have been sent

Write SMS



(Figure 4-44) Write SMS Menu Option

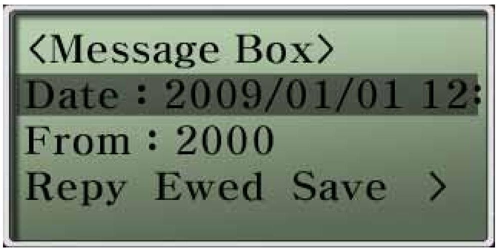
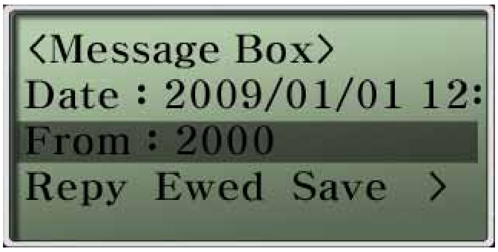
[Table 4-31] Description of Write Message Menu Detail

Menu Detail	Description
	Enter the messages in this field. (F4 : Changing from the alphabetic characters to the numeric and the numeric to the alphabetic)
	Enter the sender's telephone number to be displayed on the receiver's phone.
	Enter the receiver's telephone number: transmitting 10 different numbers simultaneously at maximum

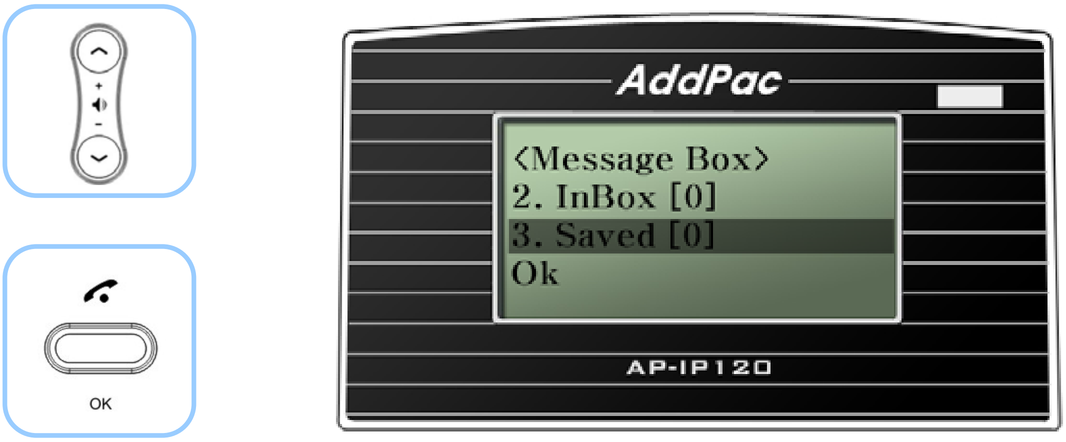
InBox

(Figure 4-45) SMS Inbox Menu Option

[Table 4-32] Description of SMS Inbox Menu Option

Menu Option	Description
	[1/3] : The first message of the 3 messages received and stored Seoul: the sender's name 1003: the sender's number Hello~~ : the content of the message Read: to read the selected message Save: to save the selected message Dele: to delete the selected message
	The date and time of the message received Repl: to reply the message to the sender Fwrd: to forward the message to the other Save: to save the selected message Dele: to delete the selected message
	The sender's telephone number
	The content of the message

Saved



(Figure 4-46) SMS Saved Menu Option

[Table 4-33] Description of Saved Menu Detail

Menu Option	Description
	[1/2] : the first message of the 2 messages received and stored Seoul: the name of the sender 1003: the sender's telephone number Hello~~ : the content of the message Read: to read the selected message Dele: to delete the selected message

Sent

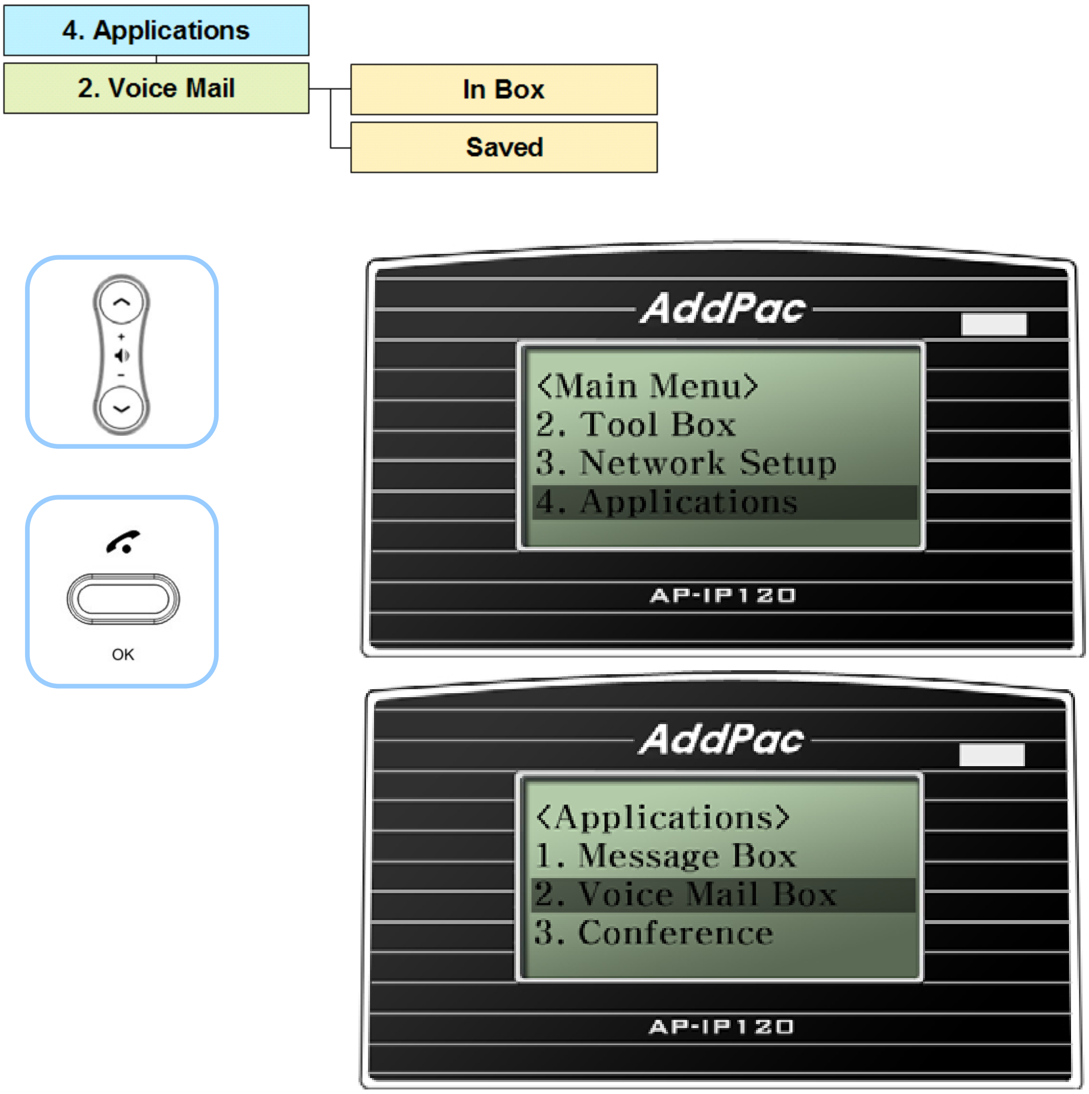
(Figure 4-47) Sent Menu Option

[Table 4-34] Description of Sent Menu Option

Menu Detail	Description
	[1/2] : the first message of the 2 messages sent and stored Seoul: the sender's name 1003: the sender's telephone number Hello~~ : the content of the message Read: to read the selected message Dele: to delete the selected message

Application – Voice Mail Box

Voice Mail Box enables you to check the voice message as it interworks with SSCP.



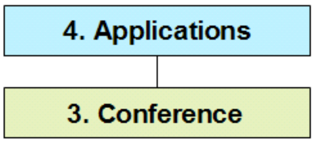
(Figure 4-48) Voice Mail Box Menu Section

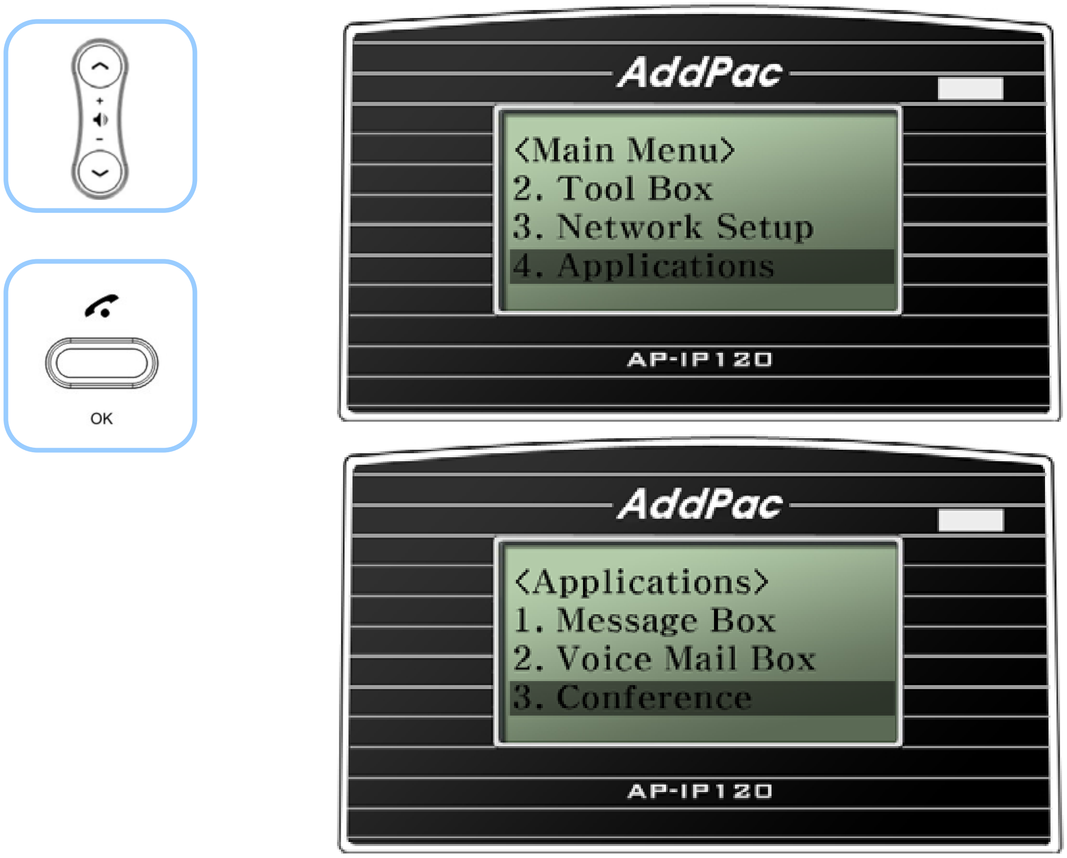
[Table 4-35] Description of Voice Mail Box Menu Option

Menu Option	Description
	This is the box to where the sent voice messages are kept
	This is where to keep the voice message which is saved.

Applications – Conference

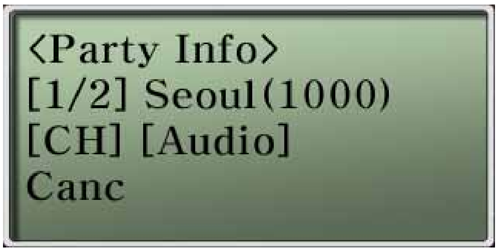
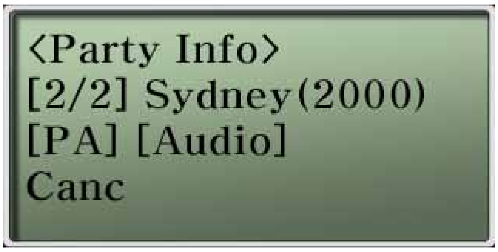
This feature enables you to see the list of connections can be made for a conference call at the present time and you can join the conference by just pressing call button. There are 4 different ways of participating in conference call: Ad Hoc, Dial-Out, Ad Hoc Dial-Out, Meet-Me and the conference parties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





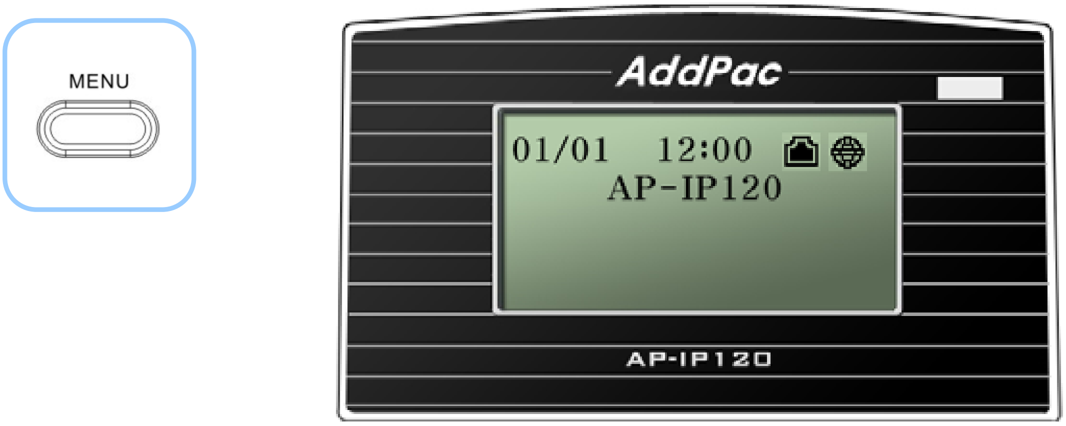
(Figure 4-49) Conference Menu Section

[Table 4-36] Description of Conference Menu Option

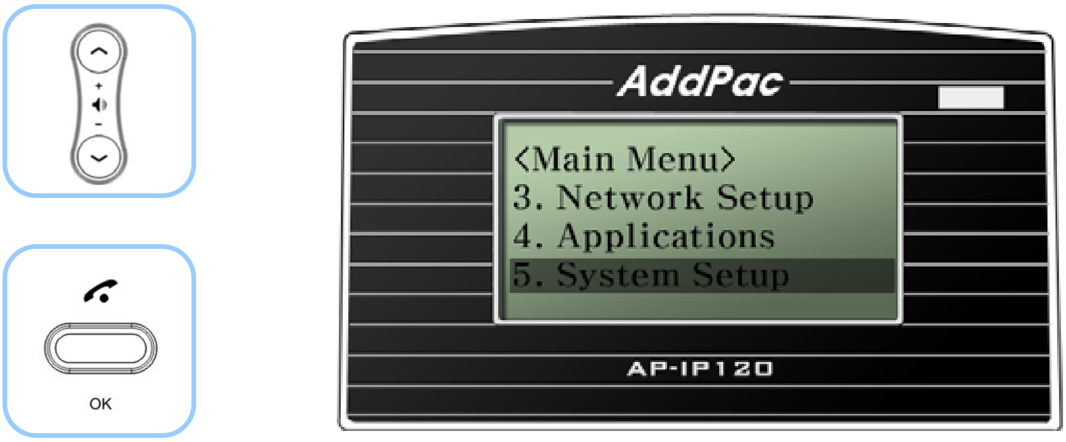
Menu Option	Description
	[1/2]: the first conference room in the list of two conference rooms [DO]: Dial-Out Conference Meet-Me: the name of the conference room [2/4]: the number of conference party in the room [Sec: O] locked for Secret Room [Sec: X] unlocked for Secret Room [V]: Voice Begin: Start the conference View: to see the information on the conference participants
	Displays the information on the conference party [1/2]: the first of the 2 parties Seoul: the name of the party (1000): the telephone number of the party [CH]: the host of the conference(Chair)
	Displays the information on the conference participants [2/2]: the second of the 2 participants Sydney: the name of the participant (2000): the telephone number of the participant [PA]: Participant

System Setup Menu Category

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



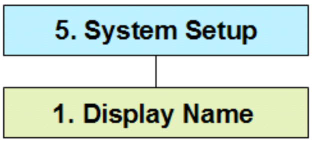
(Figure 4-50) Main Screen

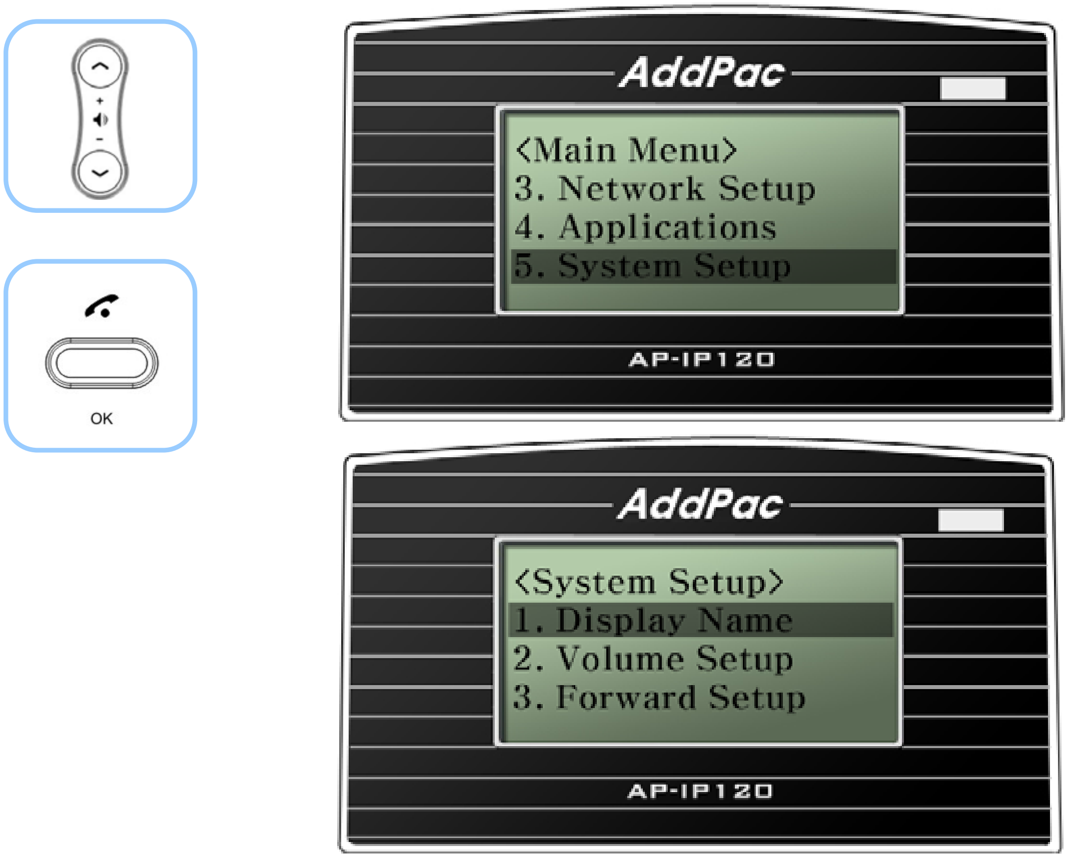


(Figure 4-51) System Setup Menu Category

System Setup – Display Name

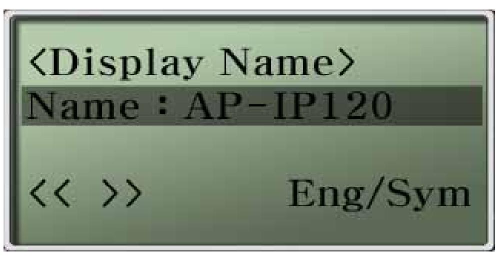
This is Name displayed on LCD panel. Default name is AP-IP120. If user wants to change default name, change or modify the default name at Display Name in System Setup Menu.





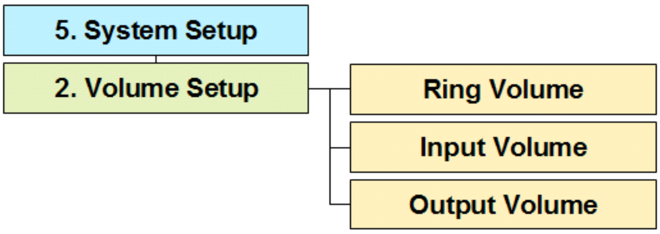
(Figure 4-52) Display Name Menu Section

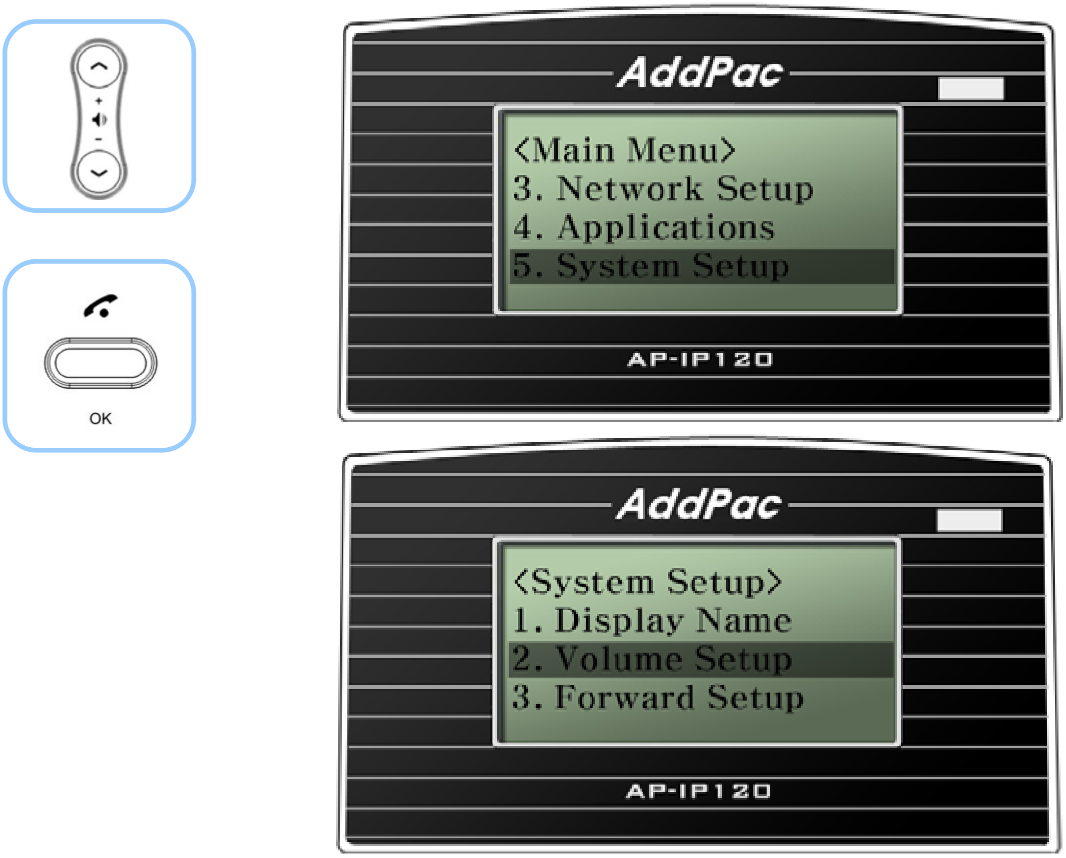
[Table 4-37] Description of Display Name Menu Option

Menu Option	Description
	Enter the Name you want (ex: Phone number, room number, office name). AP-IP120 is set at default

System Setup – Volume Control

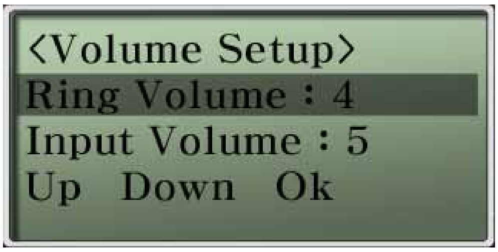
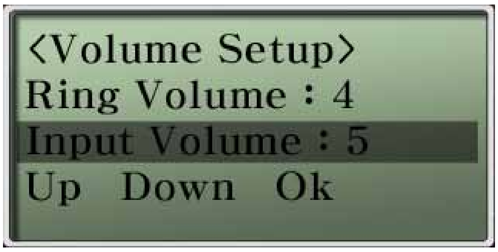
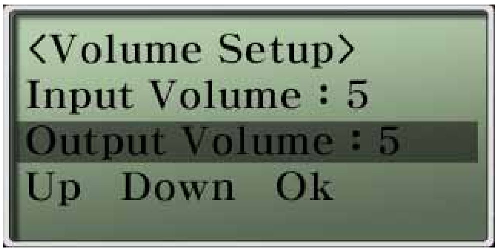
This volume menu controls the Ring volume, Input/Output volume (Speaker volume, Handset volume) of AP-IP120 IP Phone.





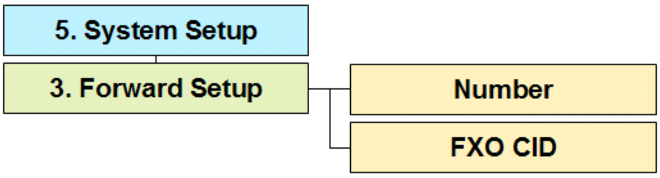
(Figure 4-53) Volume Setup Menu Section

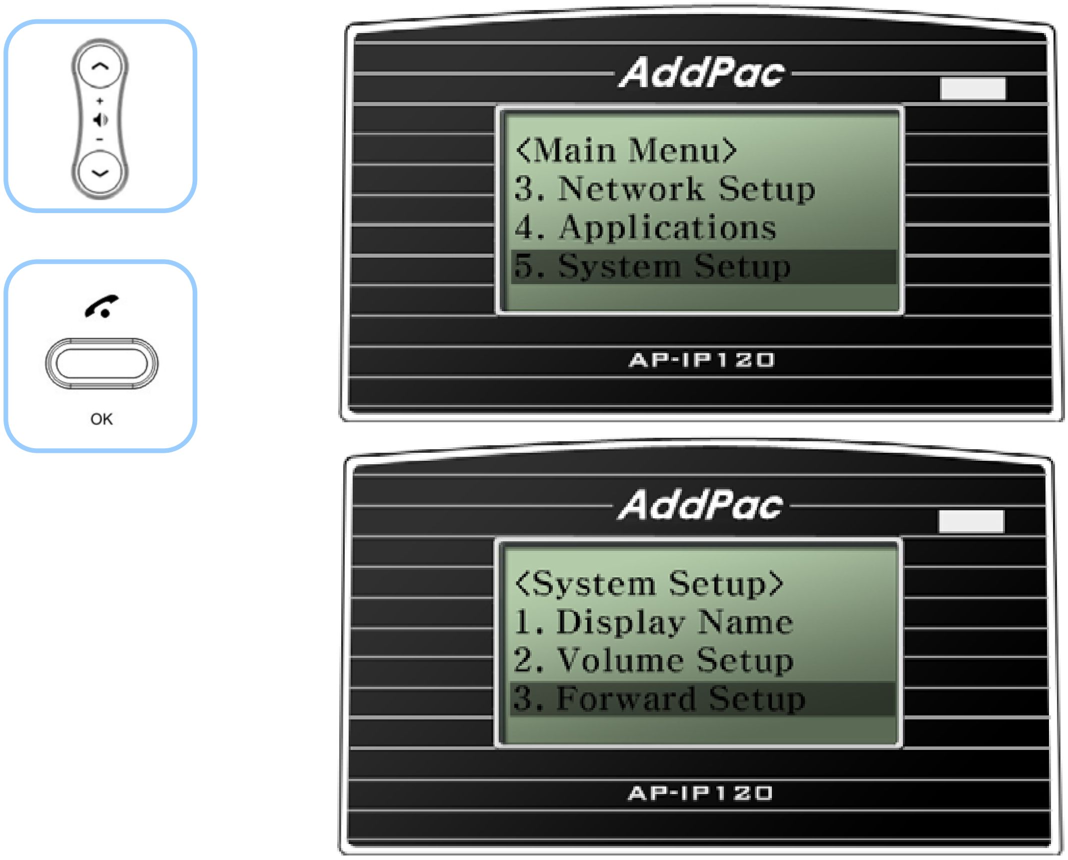
[Table 4-38] Description of Volume Setup Menu Option

Menu Option	Description
	Select the ring volume of internal speaker. Default ring volume level is 4.
	Select the input volume of Speaker Phone or Handset MIC. The default is set to the level 4.
	Select the output volume of Speaker Phone or Handset Speaker. The default is set to the volume level 4.

System Setup – Forward Setup (SIP)

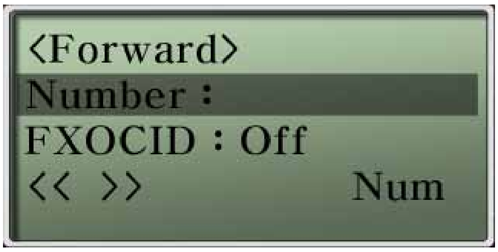
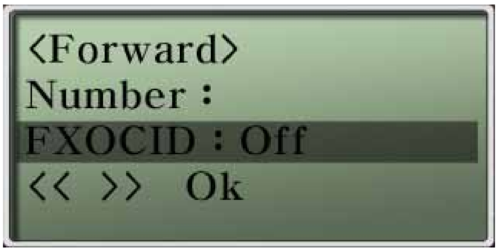
During the user's to take a call, he/she uses this menu option to configure the incoming call arriving on FXO to be transferred to VoIP call (during the one's absence, this forwarding feature is not supported for the incoming VoIP call).





(Figure 4-54) Forward Setup Menu Option

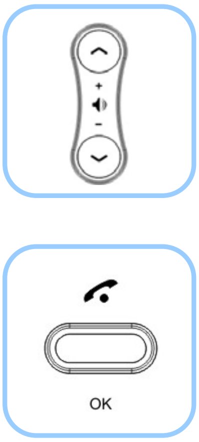
[Table 4-39] Description of System Setup Menu Detail

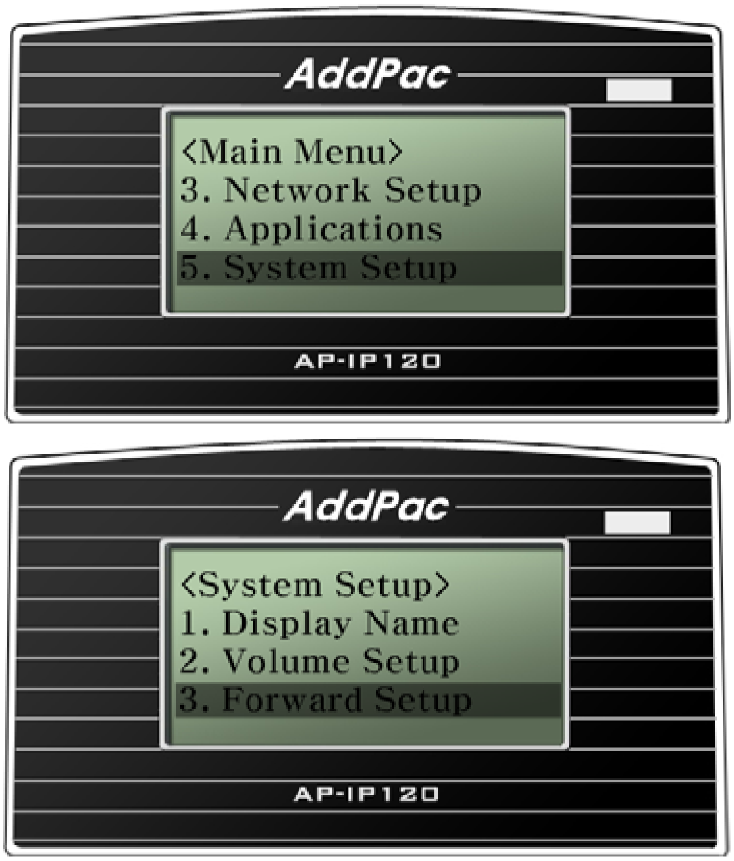
Menu Detail	Description
	Set the number to be forwarded
	Determine which number to be forwarded: FXOCID: On - the number of the incoming call on FXO is to be forwarded (the default setting) FXOCID: Off – the number of the incoming call on Speech Port (the predetermined number of AP-IP120)

System Setup – Forward Setup (SSCP)

This is the menu option to configure the call forward to redirect no matter what, either the user in the conversation on call or no response to the call.

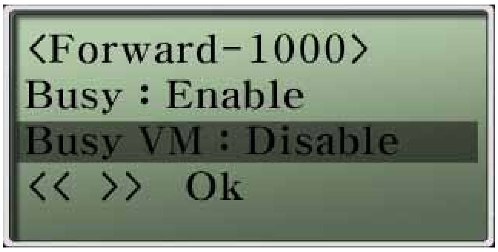
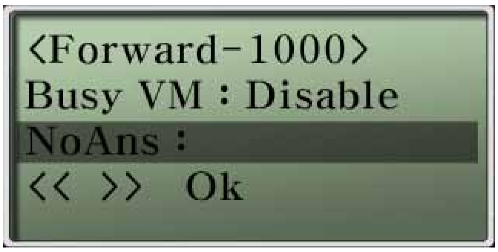
- **5. System Setup**
  - **3. Forward Setup**
    - **Busy**
    - **Busy Setup**
    - **Busy Voice Mail**
    - **No Answer**
    - **No Answer Setup**
    - **No answer Voice Mail**
    - **Unconditional**
    - **Unconditional Setup**
    - **Uncon, Voice Mail**

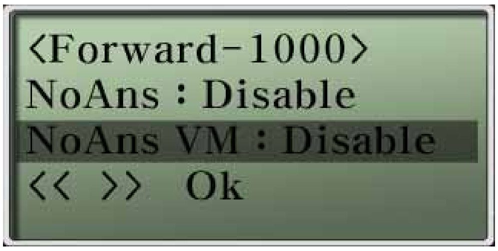




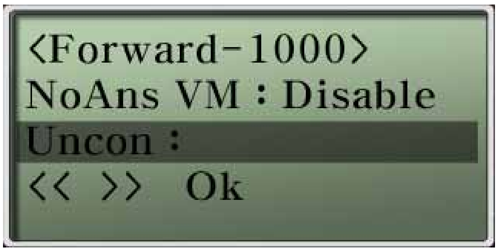
(Figure 4-55) Forward Setup Menu Option

[Table 4-40] Description of Forward Setup Menu Detail

Menu Detail	Description
	Enter the number to be forwarded to when the line is busy
	Disable or Enable the Call forwarding when the line is busy (the default setting : Disable)
	Set the Call Forwarding to be connected to Voice Mail when the line is busy (the default setting : Disable)
	Enter the number to be forward to when there is not answer
	Enable or Disable Call Forwarding when there is no answer (the default setting: Disable)

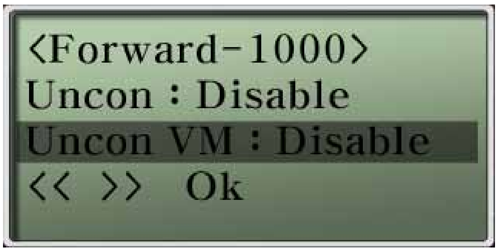


Disable or Enable the Call Forwarding to be connected to Voice Mail when there is no answer
(the default setting: disable)



Enter the number to be forwarded to no matter what (Call Forwarding Unconditional)

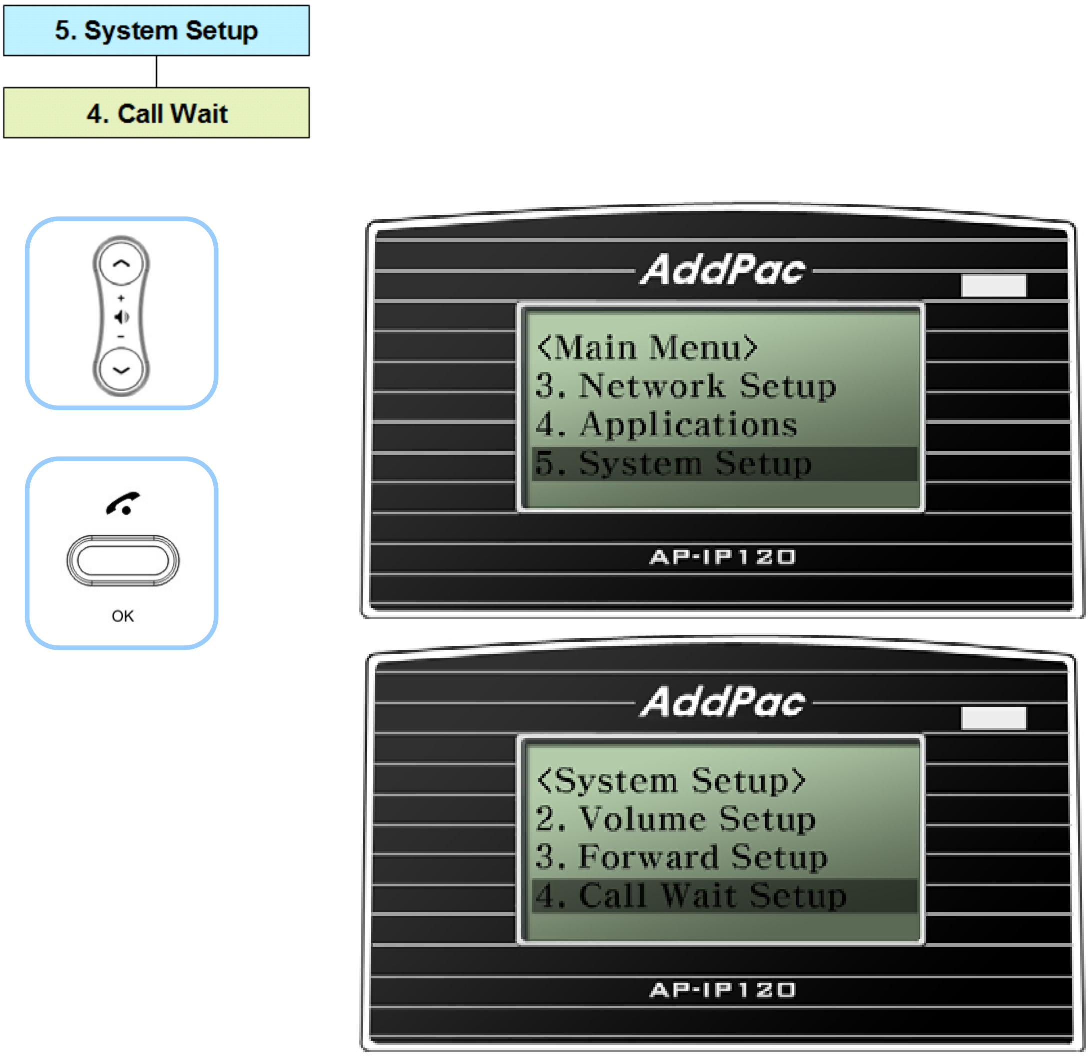
Enable or Disable Call Forwarding Unconditional (the default setting: Disable)



Disable or Enable Call Forward Unconditional to be connected to Voice Mail
(the default setting : Disable)

System Setup – Call Wait Setup (SSCP)

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.



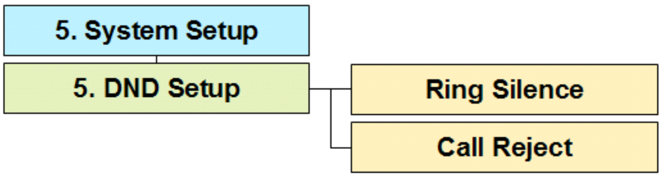
(Figure 4-56) Call Wait Setup Menu Option

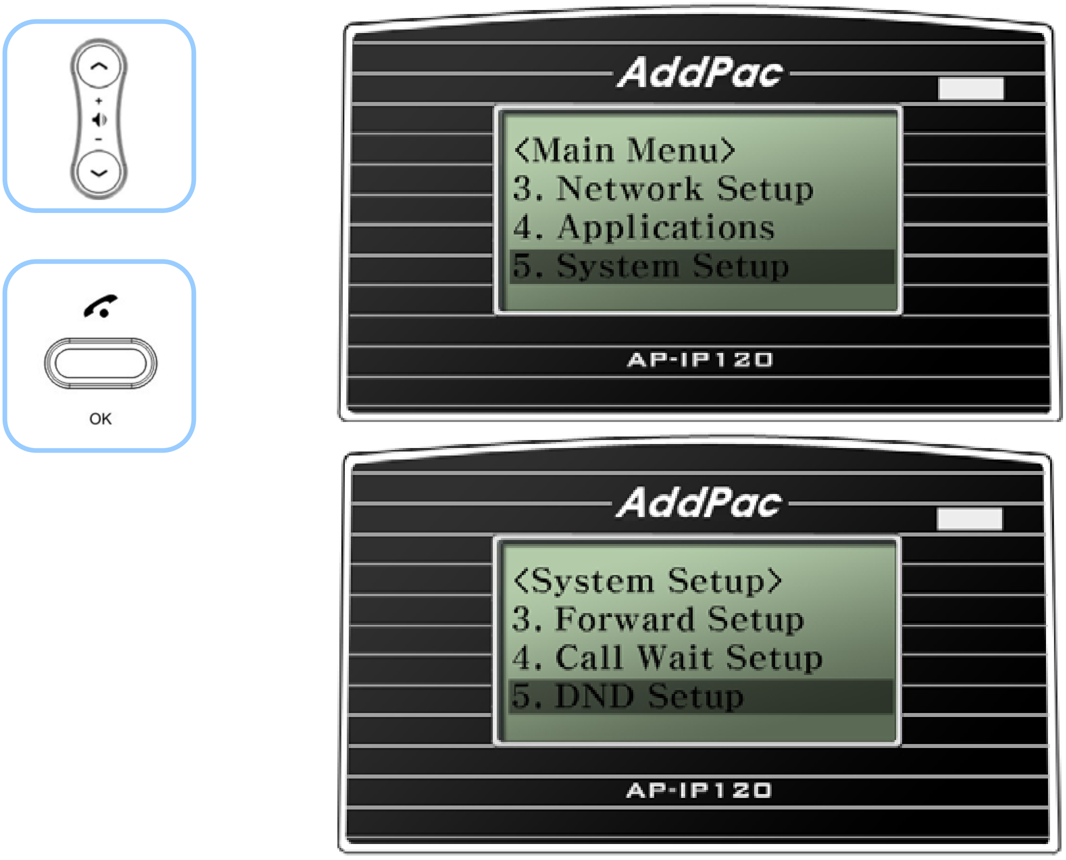
[Table 4-41] Call Wait Setup Menu Detail

Menu Detail	Description
	Disable or Enable Call Wait (Default : Disable)

System Configuration – DND(Do Not Disturb) Setup

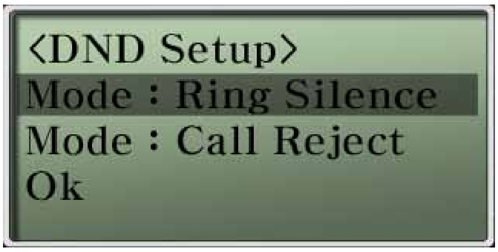
Do Not Disturb (DND) features allows you to turn off the ringer (Ring Silence) for an incoming call or to reject the call (Call Reject. You may hold pressing the leave of absence button of the IP-Phone for more than 2 seconds to enable or disable this function. The Call Reject can work only in the SSCP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





(Figure 4-57) DND Setup Menu Section

[Table 4-42] Description of DND Setup Menu Option (1)

Menu Option	Description
	Set the mode to Ring Silence
	Set the mode to Call Reject

[Table 4-43] Description of DND Setup Menu Option (2)

Hold pressing the button for more than 2 seconds

[DND]
=>DND is enabled

Hold pressing the button for more than 2 seconds

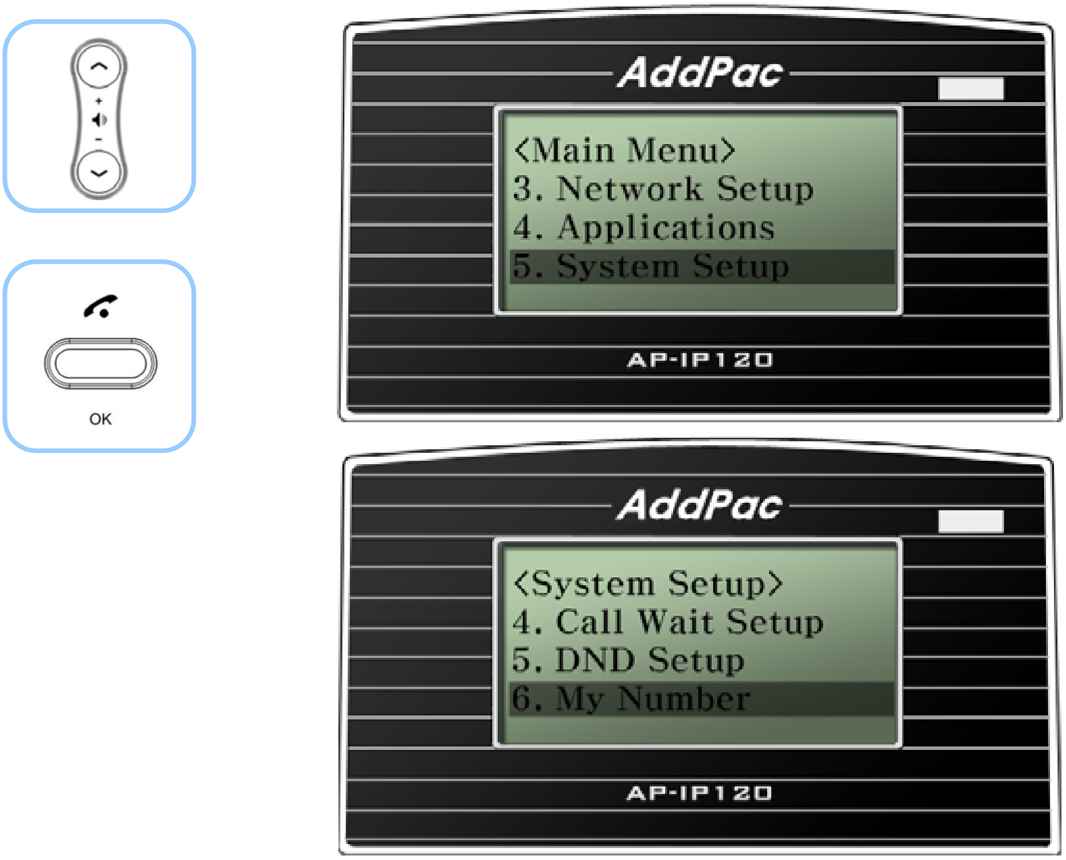
DND is disabled

System Setup – My Number

My 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.

5. System Setup

6. My Number



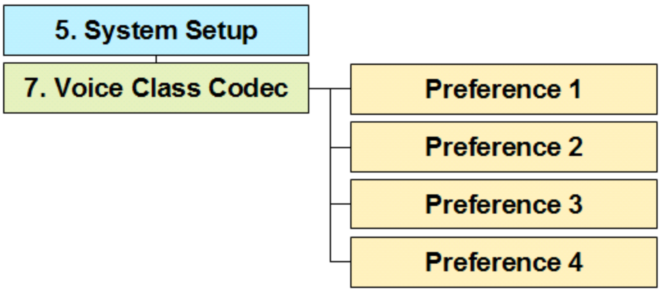
(Figure 4-58) My Number Menu Section

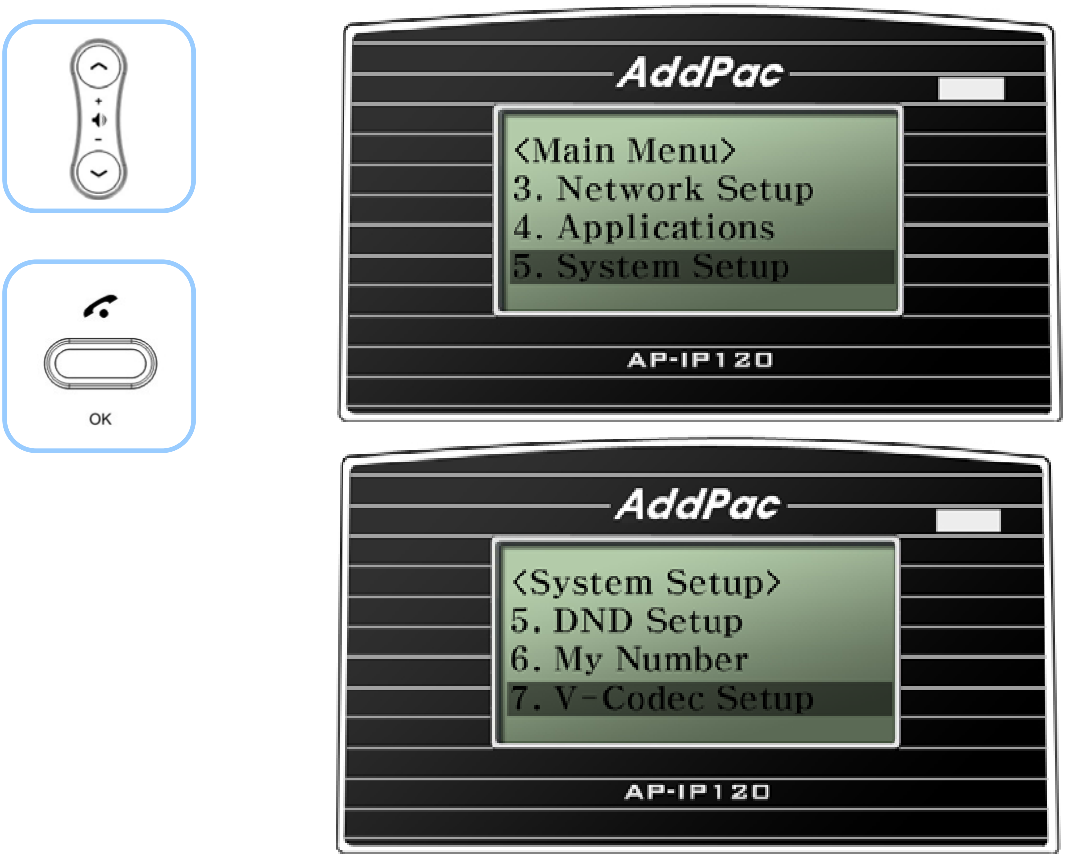
[Table 4-44] Description of My Number Menu Option

Menu Option	Description
	This is an example for assignment of the 2 telephone numbers. The default setting is set to 1000.

System Setup – Voice Class Codec

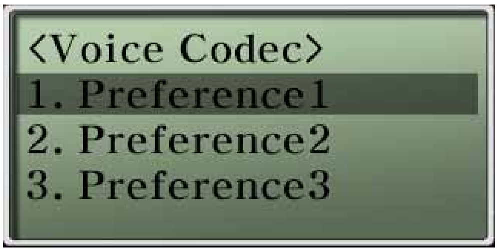
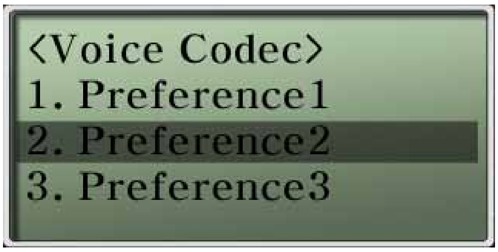
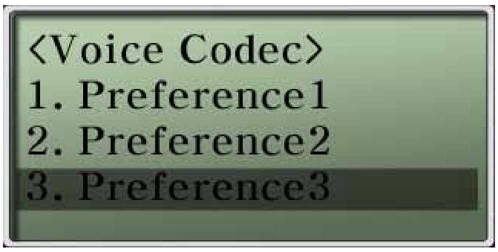
The audio codec of AP-IP120 determines the voice codec. You can choose the options of G.711[PCM] and G726, G.729, and G.723.1 on OSD, basing on the priority level which can be suitable to your network settings.



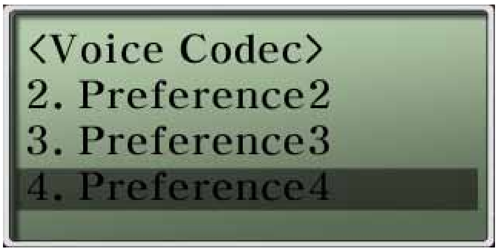


(Figure 4-59) V-Codec Menu Section

[Table 4-45] Description of Voice Codec Menu Option

<table>
<tr><th>Menu Option</th><th>Description</th></tr>
<tr><td>
</td><td>Select Preference 1 of voice codec</td></tr>
<tr><td>
</td><td>Specify the voice codec to Preference 1.
(Default : G711ullaw)</td></tr>
<tr><td>
</td><td>Select Preference 2 of voice codec.</td></tr>
<tr><td>
</td><td>Specify the voice codec to Preference 2.
(Default : G711allaw)</td></tr>
<tr><td>
</td><td>Select Preference 3 of voice codec</td></tr>
</table>

Specify the voice codec to Preference 3.
(Default : G729)

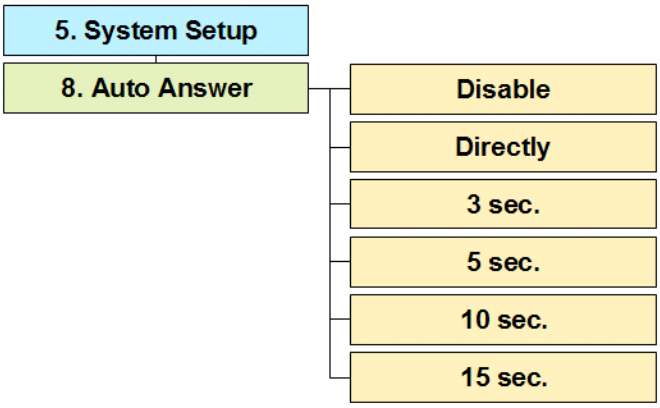


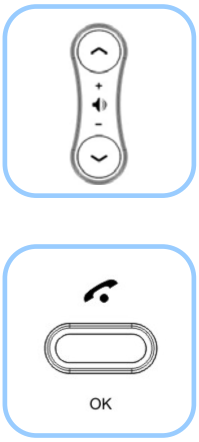
Select Preference 3 of voice codec.

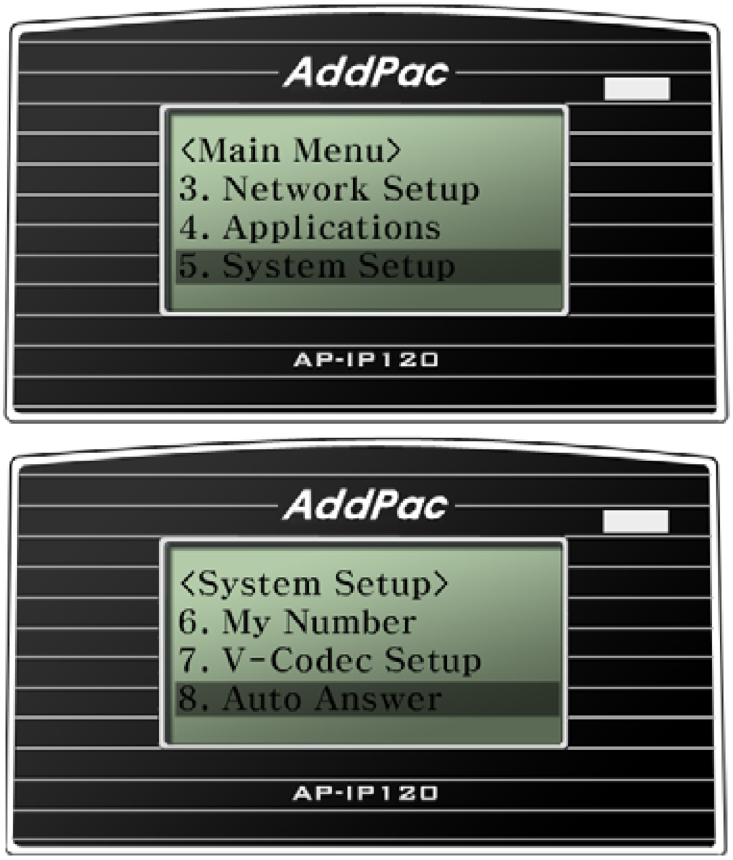
Specify the voice codec to Preference 3.
(Default : G7231r63)

System Setup – Auto Answer

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.

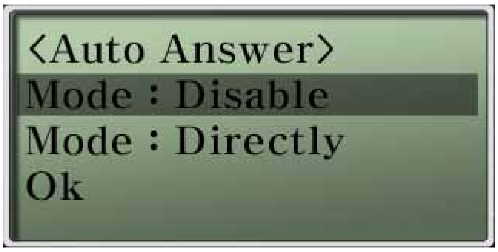
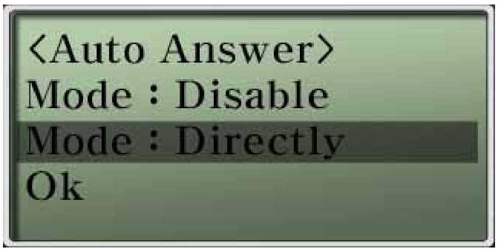
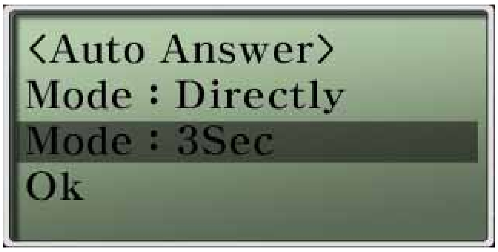
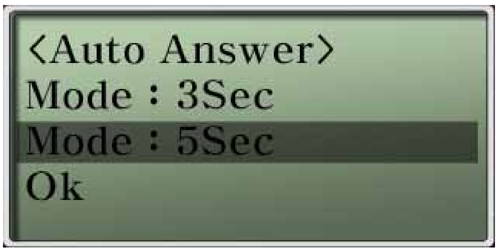
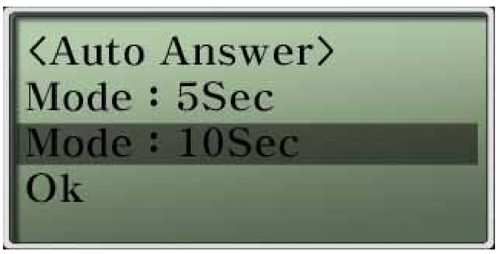


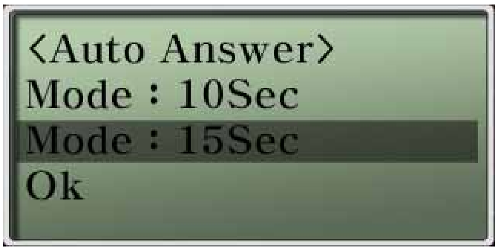




(Figure 4-60) Auto Answer Menu Section

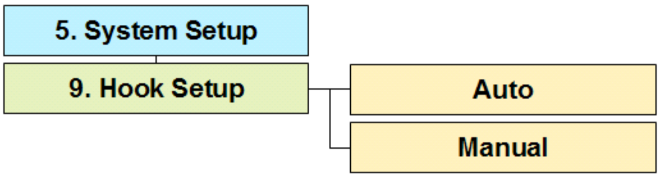
[Table 4-46] Description of Auto Answer Menu Option

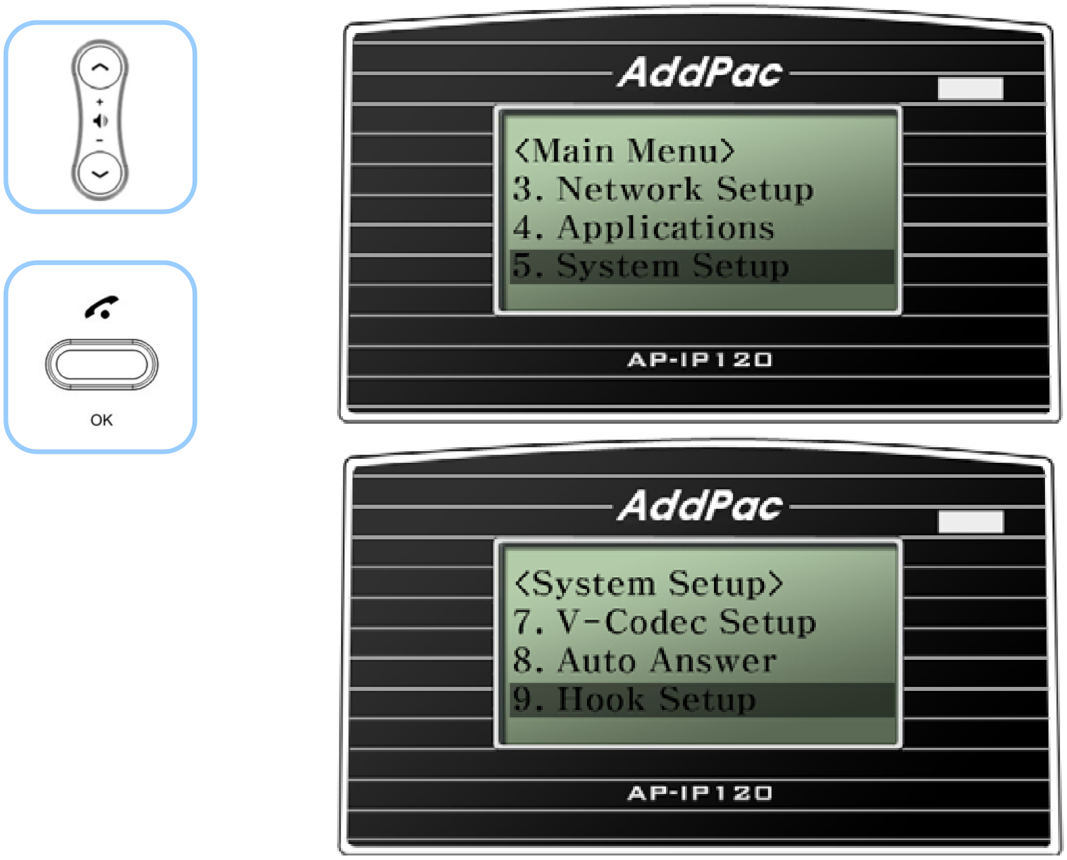
Menu Option	Description
	Disable Auto Answer
	Receive the call at the first ring
	Set the mode to Auto Answer to reply on 3 seconds after the bell rings.
	Set the mode to Auto Answer to reply on 5 seconds after the bell rings.
	Set the mode to Auto Answer to reply on 10 seconds after the bell rings.

	Set the mode to Auto Answer to reply on 15 seconds after the bell rings.

System Setup – Hook Setup

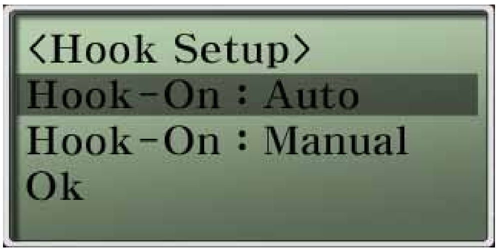
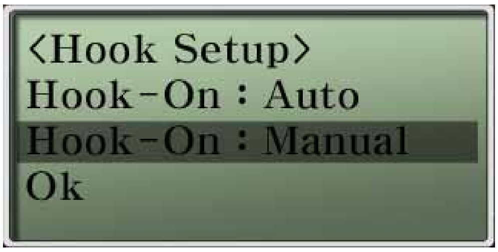
Hook is the feature that allows you to disconnect the phone even when the other part did not hang up the line. This feature is supported by SSCP.





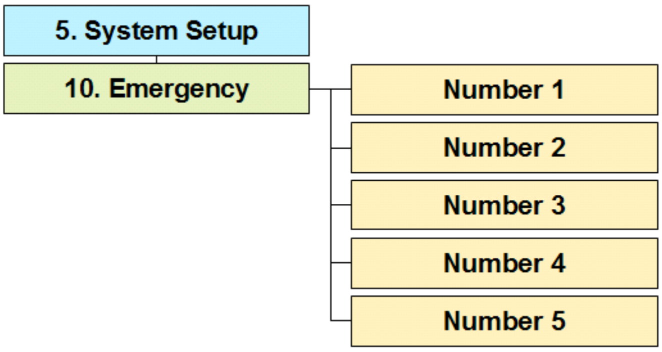
(Figure 4-61) Hook Setup Menu Section

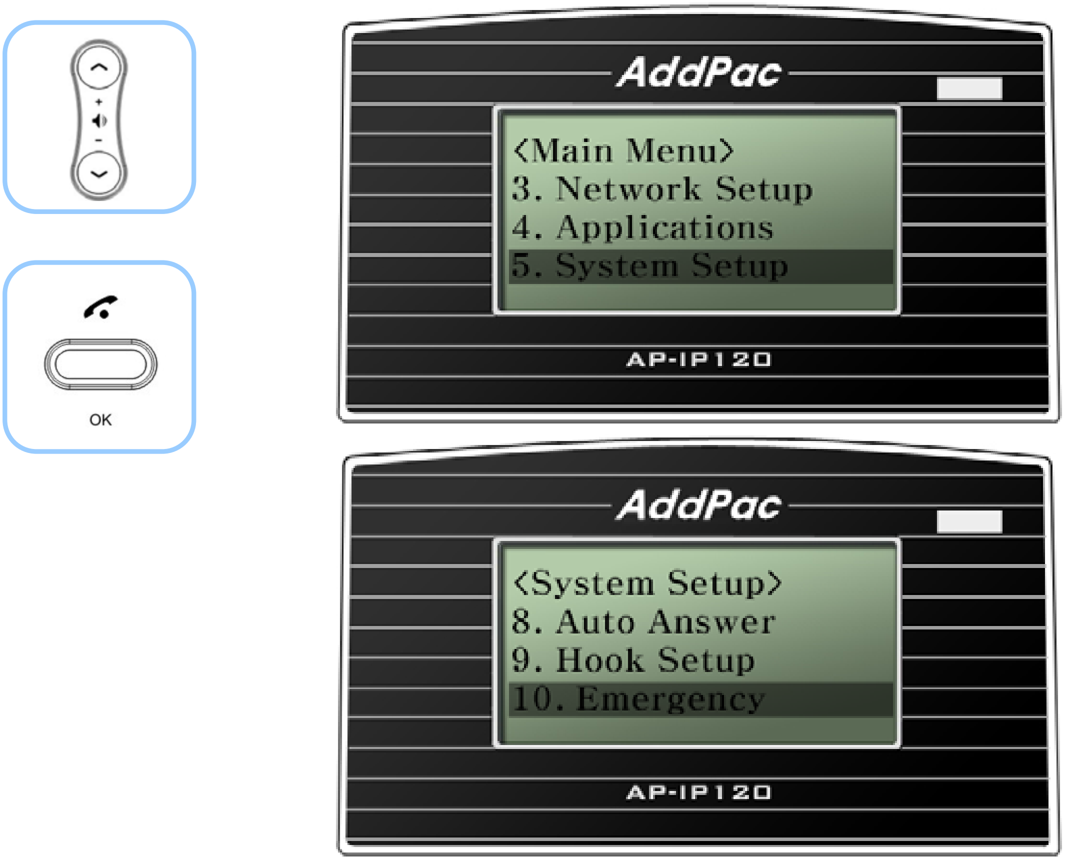
[Table 4-47] Description Hook Setup Menu Option

Menu Option	Description
	Set Hook-On to manual (the default setting)
	Set Hook-On to Automatic

System Setup – Emergency

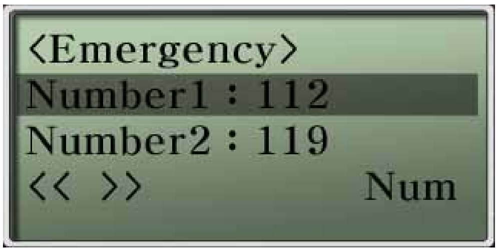
If a terminal limits an outgoing call due to Administrator/ User Lock, the terminal can be configured to take an emergency call. For the phone number specified in Emergency menu section, the outgoing call can be sent out regardless of lock status.





(Figure 4-62) Emergency Menu Section

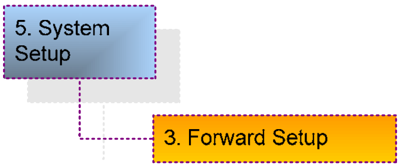
[Table 4-48] Description of Emergency Menu Option

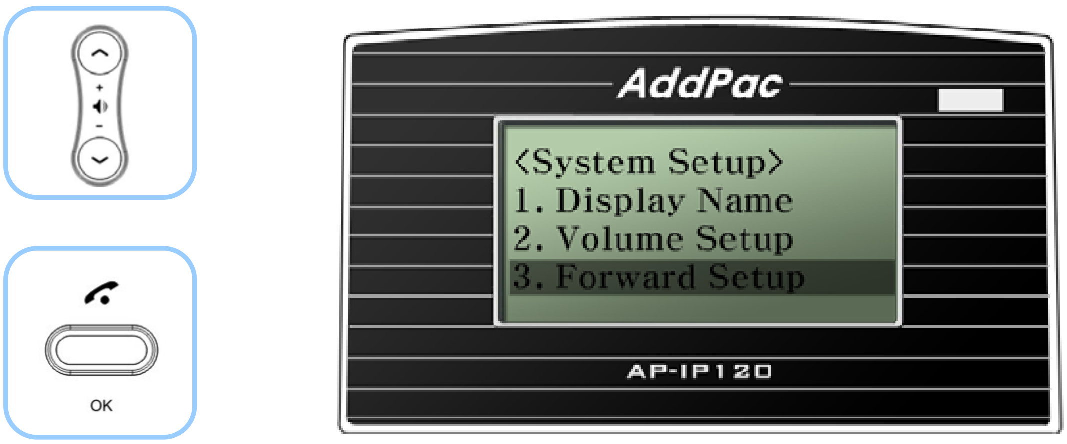
Menu Option	Description
	Specifying a phone number for emergency

Chapter 5. Additional Features for FXO

Forward Setup (SIP)

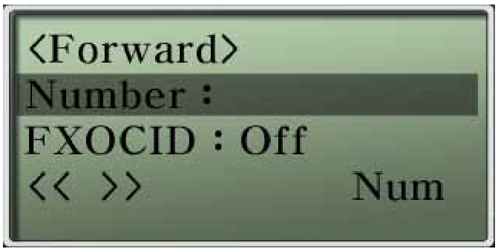
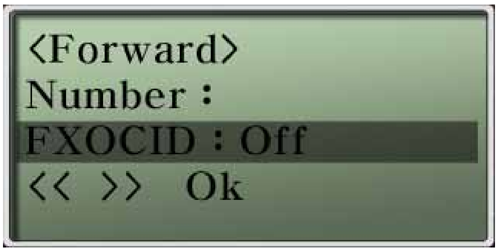
This feature allows you to set to change a incoming call on FXO to VoIP, when you are available to take the call.





(Figure 5-1) Forward Setup Menu Section

[Table 5-1] Description of Forward Setup Menu Option

Menu Option	Description
	Set the telephone number to where the call is to be forwarded
	Determine whether to assign the number of Speech port (the predetermined number of AP-IP120) or the number of the incoming call on FXO (The default setting is set to On), when the call is forwarded.

Call Waiting (SIP)

For any incoming call arriving to FXO can be placed on hold while the phone user is on another VoIP call. While the phone user is on PSTN call, any incoming VoIP call can be placed on hold as well. Call-waiting needs to be configured on speech port and call-hold h also needs to be configured at the configuration mode. (**Configuration on CLI screen after connecting to Telnet**)

[Table 5-2] Configuration Settings for Call Waiting

IP120#
IP120# **con t**
IP120(config)#
IP120(config)# **dial-peer voice 0 pots** **=> Speech Port Mode**
IP120(config-dialpeer-pots-0)#
IP120(config-dialpeer-pots-0)# **call-waiting** **=> Set Speech Port Call Waiting**
IP120(config-dialpeer-pots-0)#
IP120(config-dialpeer-pots-0)# **exit**
IP120(config)#
IP120(config)# **dial-pee call-hold h** **=> Set call-hold h from configuration mode**
IP120(config)#
IP120(config)# **show run** **=> Verifying the configuration**
Building configuration...

Current configuration:
version 8.41.014
hostname IP120
!
username root password router administrator
!

------ 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
 call-waiting
!
! Voip peer configuration.
!
dial-peer voice 1001 voip
 destination-pattern T
 session target sip-server
 session protocol sip
 voice-class codec 0
 dtmf-relay dual-mode
 vad
 huntstop
!
dial-peer voice 1002 voip
 destination-pattern T
 session target ras
 dtmf-relay h245-alphanumeric
 vad
 preference 1
 huntstop
!
!
!
dial-peer call-hold h
!
```

Chapter 6. Testing Operation

Booting Procedure and Operating Bases

The things that you have to know, prior to turning on the power of AP-IP120, are the booting procedure.

- AP-IP120 checks its basic operation of its interface, memory and CPU through a self-testing procedure

- After the Boot Loader is started, the IP120 looks for the appropriate software image file. At the default configuration, AP-IP120 is set to load the software in the Flash Memory

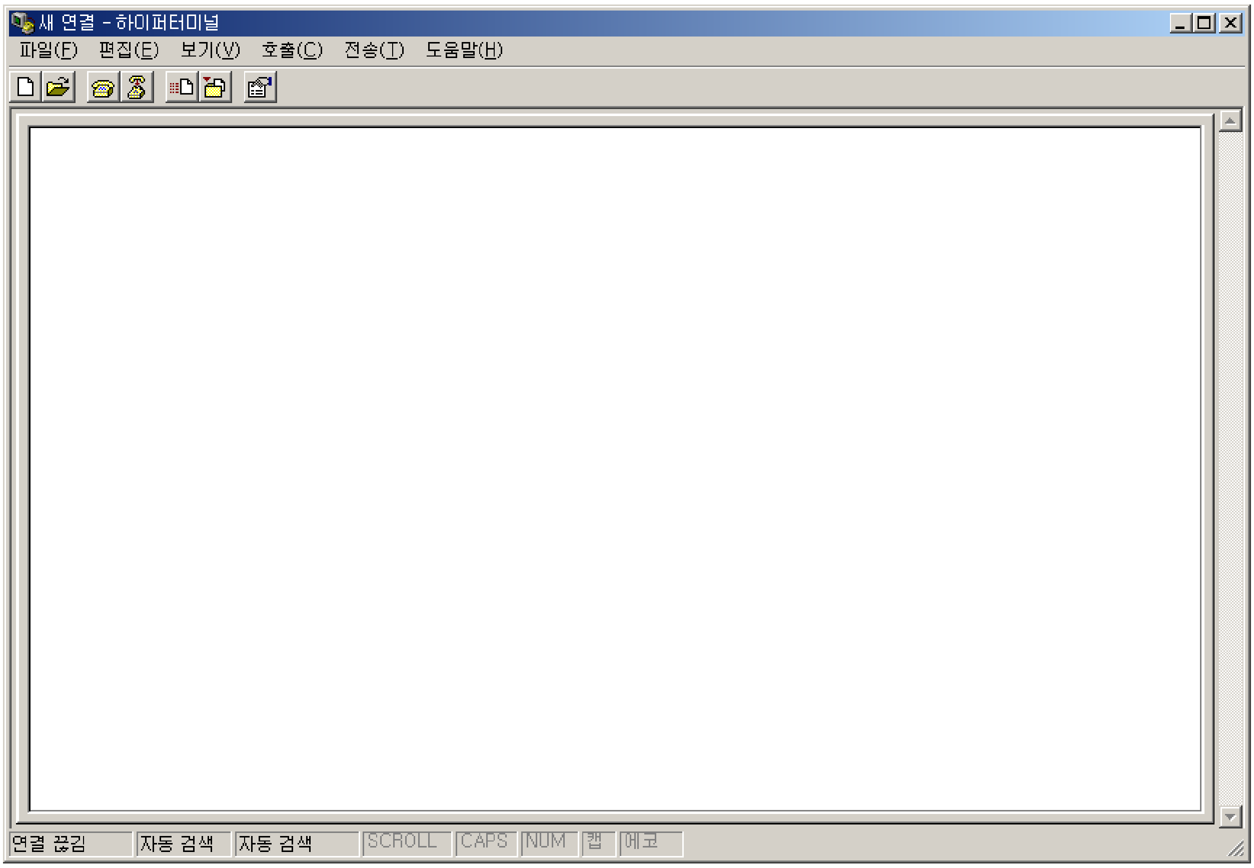
- If the IP120 fails to find the appropriate software image file, the Boot Loader stands by until the appropriate file is found to be downloaded from the appropriate system, at the booting mode.

- 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 IP120 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-IP120 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.

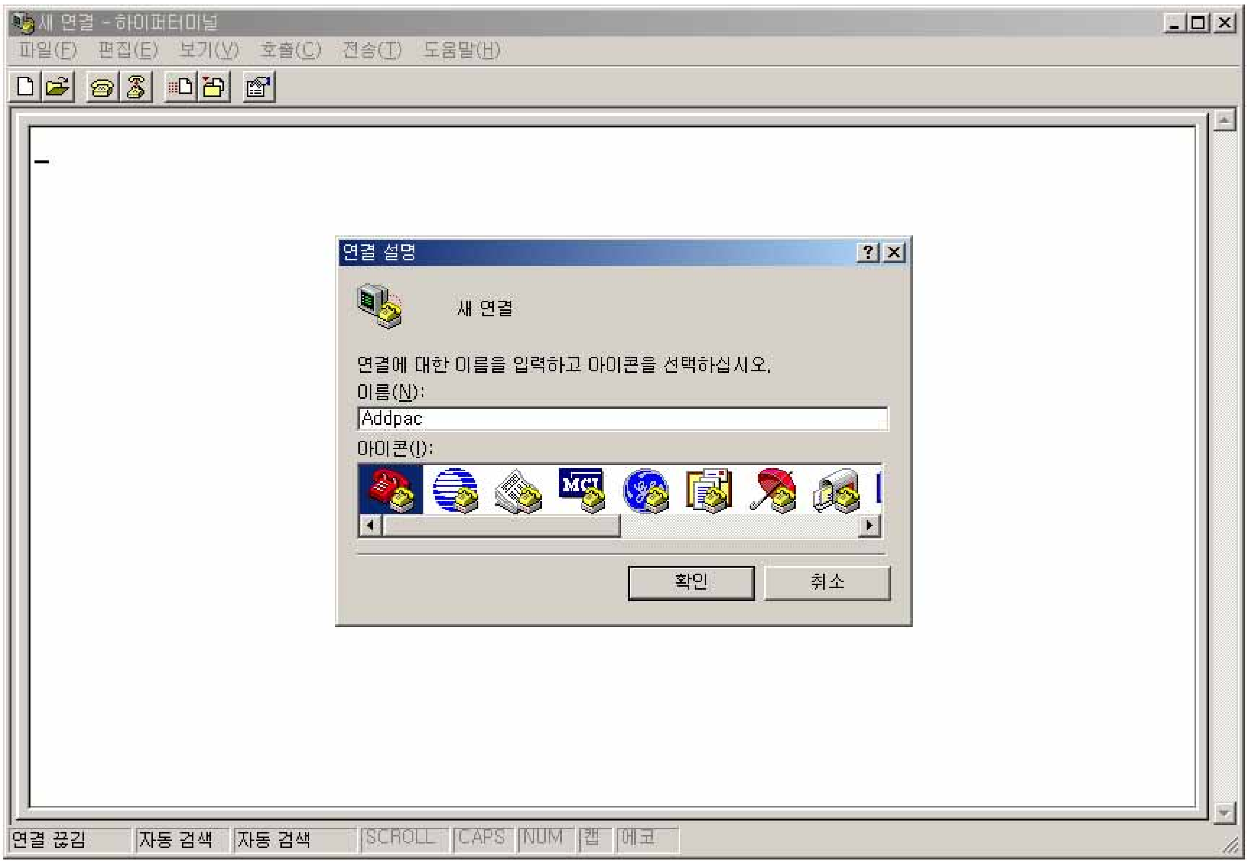
Using HyperTerminal for the Console Terminal

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



(Figure 6-1) Terminal Emulator HyperTerminal on MS-Windows

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



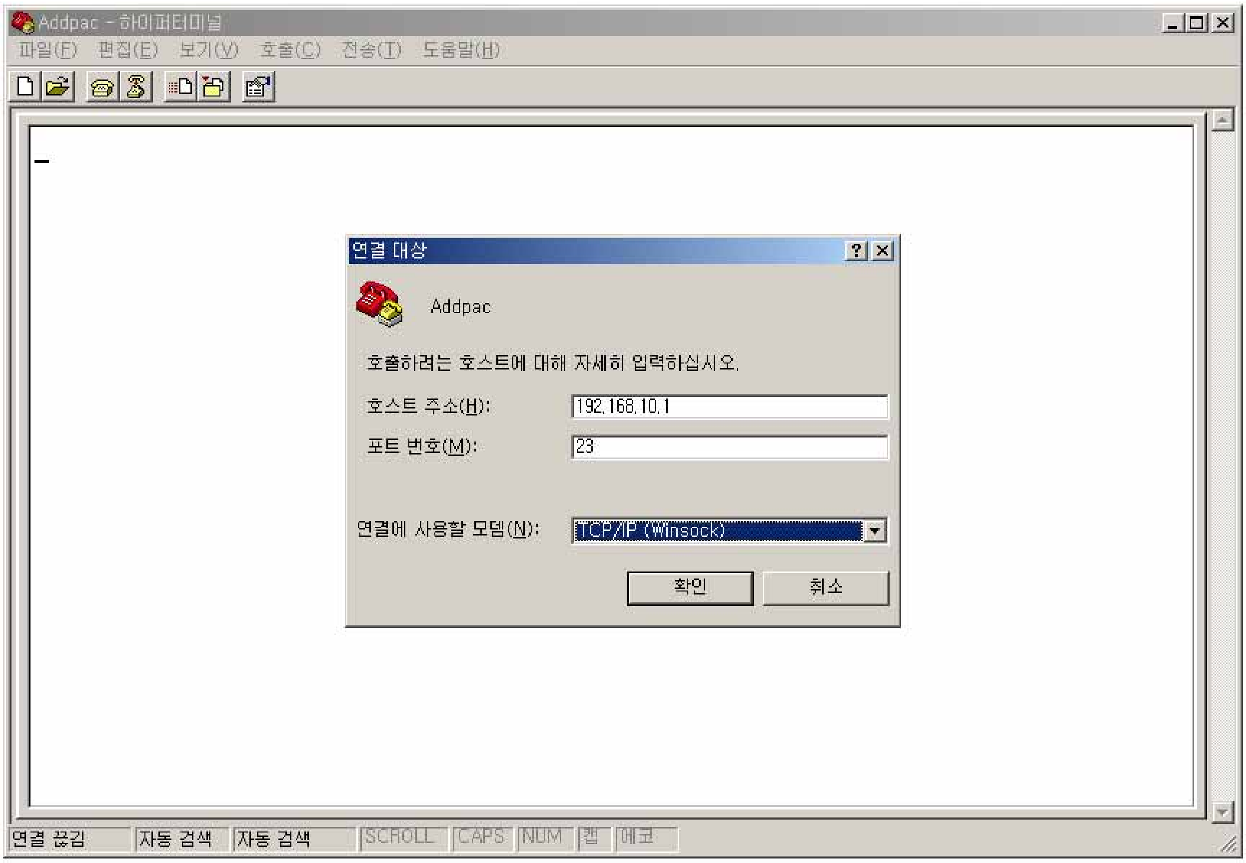
(Figure 6-2) Name Entry for HyperTerminal Connection

Select the interface of which the console cable is connected

Since AP-IP120 does not support Console Interface, the IP Address of LAN1 interface is used to connect PC as it is shown in the following figure:

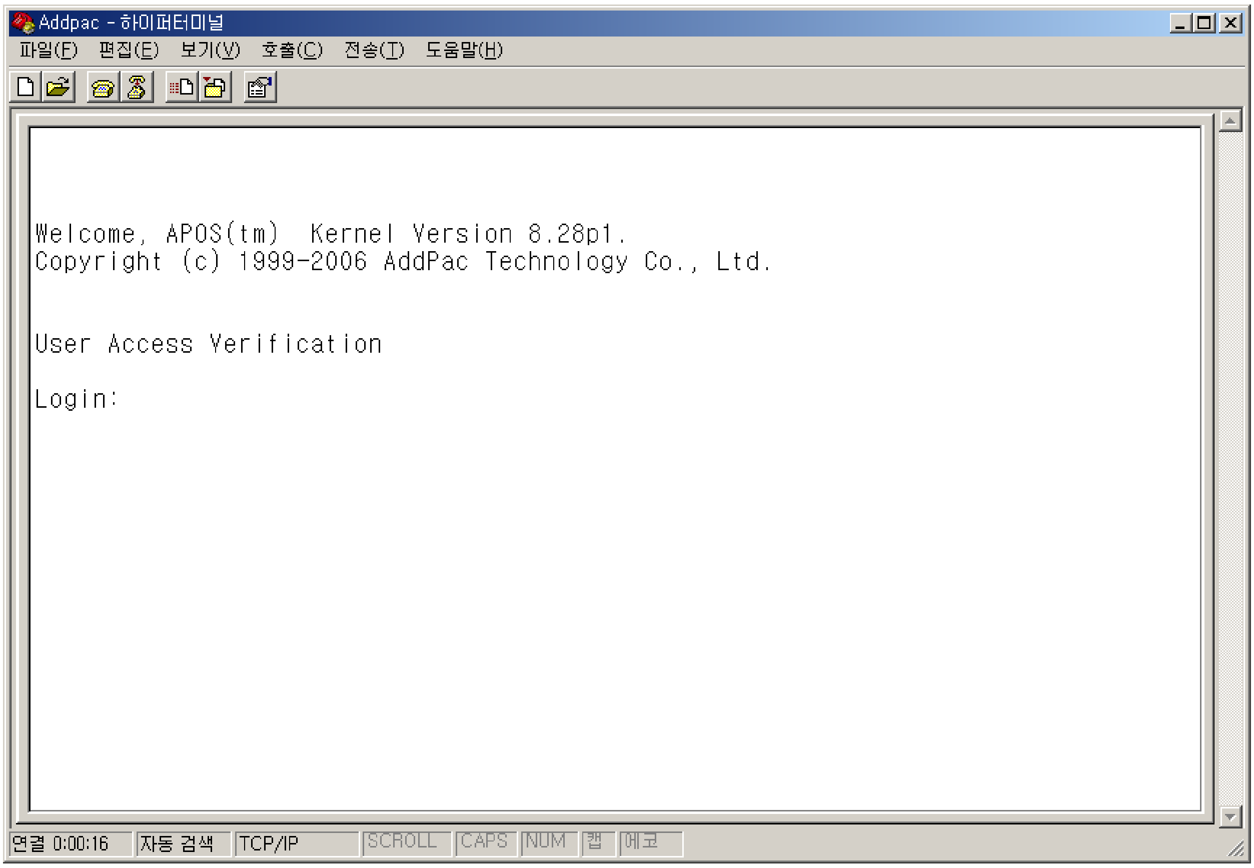
(LAN1 default ip address : 192.168.10.1)

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



(Figure 6-3) Access to Telnet by Using TCP/IP

After completing the setup, you can see the login screen of AP-IP120 in the following Figure



(Figure 6-4) Login Screen

You may see the 2 types of prompt: 'AP-IP120>' and 'AP-IP120#'. 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.

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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-120>’.

[Table 6-1] 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 ‘IP120#’.

[Table 6-2] Management 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 6-3] Password Setup

AP-IP120# **configure terminal**
AP-IP120(config)#
AP-IP120(config)# **enable password {password}**
AP-IP120(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 6-4] Configuring Host Name

AP-IP120# **configure terminal**
AP-IP120(config)#
AP-IP120(config)# **hostname {name}**
AP-IP120(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 6-5] User Administration

```
AP-IP120# configure terminal
AP-IP120(config)#
AP-IP120(config)# username {ID} password {password} {administrator | operator | user}
AP-IP120(config)#
```

Configuring FXS/FXO Port

* Configuring after the Display of show run

[Table 6-6] Configuring FXS/FXO Port

```
IP120# show run

Building configuration...
Current configuration:
version 8.41.015
hostname IP120
!
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
!
```

IP120#

IP120# con t

IP120(config)# **dial-peer voice 1 pots** ***=> FXS/FXO port dial peer configuration***

IP120(config-dialpeer-pots-1)# **destination-pattern 1014** *=> **Assigning the port numbers of FXS/FXO***

IP120(config-dialpeer-pots-1)# **port 0/1** ***=> Assigning the port that has been checked***

IP120# **show run** ***(checking the configuration)***

```
Building configuration...
Current configuration:
version 8.41.015
hostname IP120
!
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 7. 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-IP120 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 IP120 which are used for accessing by TELNET/FTP must be connected directly.

Entering the mode of Boot Loader

Since AP-IP120 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-IP120 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-IP120 and PC are connected directly to each other, the user can access to AP-IP120 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-IP120 IP Phone allows FTP access, which is supported by the binary code, to transmit APOS image file.

APOS image of AP-IP120 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-IP120_g2_v8_41_015.bin

D:\>
D:\> ftp 172.17.201.88
Connected to 172.17.201.88.
220 IP120 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-IP120_g2_v8_41_015.bin
200 PORT command successful.
90 Opening BINARY mode data connection for ' AP-
IP120_g2_v8_41_015.bin '.
226 Transfer complete.
ftp> bye
221 Goodbye.
D:\>
```

Chapter 8. Appendix

This Appendix provides information about the Pinout specifications of the following cables used with AP-IP120 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.

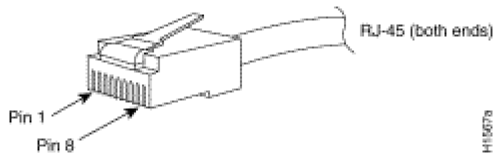
[Table 8-1] The signal and Pinout specification

[Table 8-2] Console Port Pinout

콘솔 포트 (DTE)	RJ-45	DB-9	콘솔 디바이스(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 8-1) 100Base-TX RJ-45 Connector

[Table 8-3] 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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