

AddPac IP/Video Phone Series

Paging, Intercom & BLF(Busy Lamp Field)
function for Asterisk PBX



AddPac

AddPac Technology

2011, Sales and Marketing

Contents

- Paging & Intercom Service with Asterisk PBX
- SIP Protocol for Paging and Intercom Service
- Asterisk PBX Intercom Configuration
- Asterisk PBX Paging Configuration
- Paging & Intercom Service Message Flow Example
- BLF Service Diagram
- BLF IP Phone Example
- BLF Protocols
- BLF Message Flow Example
- IP Phone Comparison Table
- IP Video Phone Comparison Table
- AddPac IP Phones

Paging and Intercom with Asterisk PBX

- Paging and Intercom

- ✓ On legacy phone systems you can find the following kinds of paging:
 - Dial a code to connect to a separate overhead paging and announcement system (like in an airport)
 - Dial a code and connect directly to a built-in one-way announcement speaker on one or more phones.
 - Dial a code and connect directly to a built-in two-way announcement and talkback function on one or more phones
- ✓ Asterisk IP-PBX supports Paging and Intercom functions as follows.
 - **New in Asterisk 1.2:** The new dial-plan command Page utilizes MeetMe to page one or more phones.
 - **New in Asterisk 1.8:** A new RTP engine and channel driver have been added which supports Multicast RTP.
The channel driver can be used with the Page application to perform multicast RTP paging.
The dial string format is:
MulticastRTP/<type>/<destination>/<control address>
- ✓ All AddPac's IP-Phone series & VP-Phone series support Paging and Intercom function.

SIP Protocol for Paging and Intercom

- SIP MIME Header for Paging and Intercom

Call-Info: <uri>;answer-after=0

Alert-Info: Ring Answer

AddPac IP-Phone will automatically answered a call with a ring when the call contains SIP header “Call-Info: answer-after=0”.

And when the call hung up by the remote party, the phone will automatically on hook without alerting user with disconnect busy tones.

FreePBX Intercom Configuration

FreePBX 2.9.0.7 on 172.16.60.177

Admin Reports Panel Recordings Help

Setup Tools

Admin

- FreePBX System Status
- Module Admin
- Basic
- SIPSTATION
- DAHDI
- Digium Addons
- Extensions
- Fax Configuration
- Feature Codes**
- General Settings
- Outbound Routes
- Trunks
- Voicemail Admin
- Extension Settings
- Administrators
- Inbound Call Control
- Inbound Routes
- Zap Channel DIDs
- Announcements
- Blacklist
- Call Flow Control
- CallerID Lookup Sources
- Directory
- Follow Me
- IVR
- Queue Priorities
- Queues

Feature Code Admin

	Use Default?	Feature Status
Blacklist		
Blacklist a number	+30 <input checked="" type="checkbox"/>	Enabled
Blacklist the last caller	+32 <input checked="" type="checkbox"/>	Enabled
Remove a number from the blacklist	+31 <input checked="" type="checkbox"/>	Enabled
Feature Codes - Select 'Feature codes' Menu for Paging and Intercom.		
Call Forward All Activate	+72 <input checked="" type="checkbox"/>	Enabled
Call Forward All Deactivate	+73 <input checked="" type="checkbox"/>	Enabled
Call Forward All Prompting Deactivate	+74 <input checked="" type="checkbox"/>	Enabled
Call Forward No Answer/Unavailable Deactivate	+53 <input checked="" type="checkbox"/>	Enabled
Call Forward Toggle	+740 <input checked="" type="checkbox"/>	Enabled
Paging and Intercom		
Intercom prefix	+80 <input checked="" type="checkbox"/>	Enabled
User Intercom Allow	+54 <input checked="" type="checkbox"/>	Enabled
User Intercom Disallow	+55 <input checked="" type="checkbox"/>	Enabled

FreePBX Paging Configuration

FreePBX
FreePBX 2.9.0.7 on 172.16.60.177

Admin Reports Panel Recordings Help

Setup | Top

Admin

FreePBX System

Module Administration

Basic

SIPSTATION

DAHDI

Digium Addons

Extensions

Internal Options & Configuration

Callback

Conferences

DISA

Languages

Misc Applications

Misc Destinations

Music on Hold

PIN Sets

Paging and Intercom

Parking Lot

Phone Restart

System Recordings

Voicemail Blasting

Queue Priorities

Queues

Paging and Intercom
- Select 'Paging and Intercom' Menu for Paging and Intercom group list.

Delete Group 6000

Modify Paging Group

Paging Extension: 6000

Group Description: Lab Broadcast

Device List:

- 4005 - knkim
- 4006 - bgchoi
- 4008 - 4008
- 4009 - 4009

Force if busy: No Yes Whisper

Duplex:

Default Page Group:

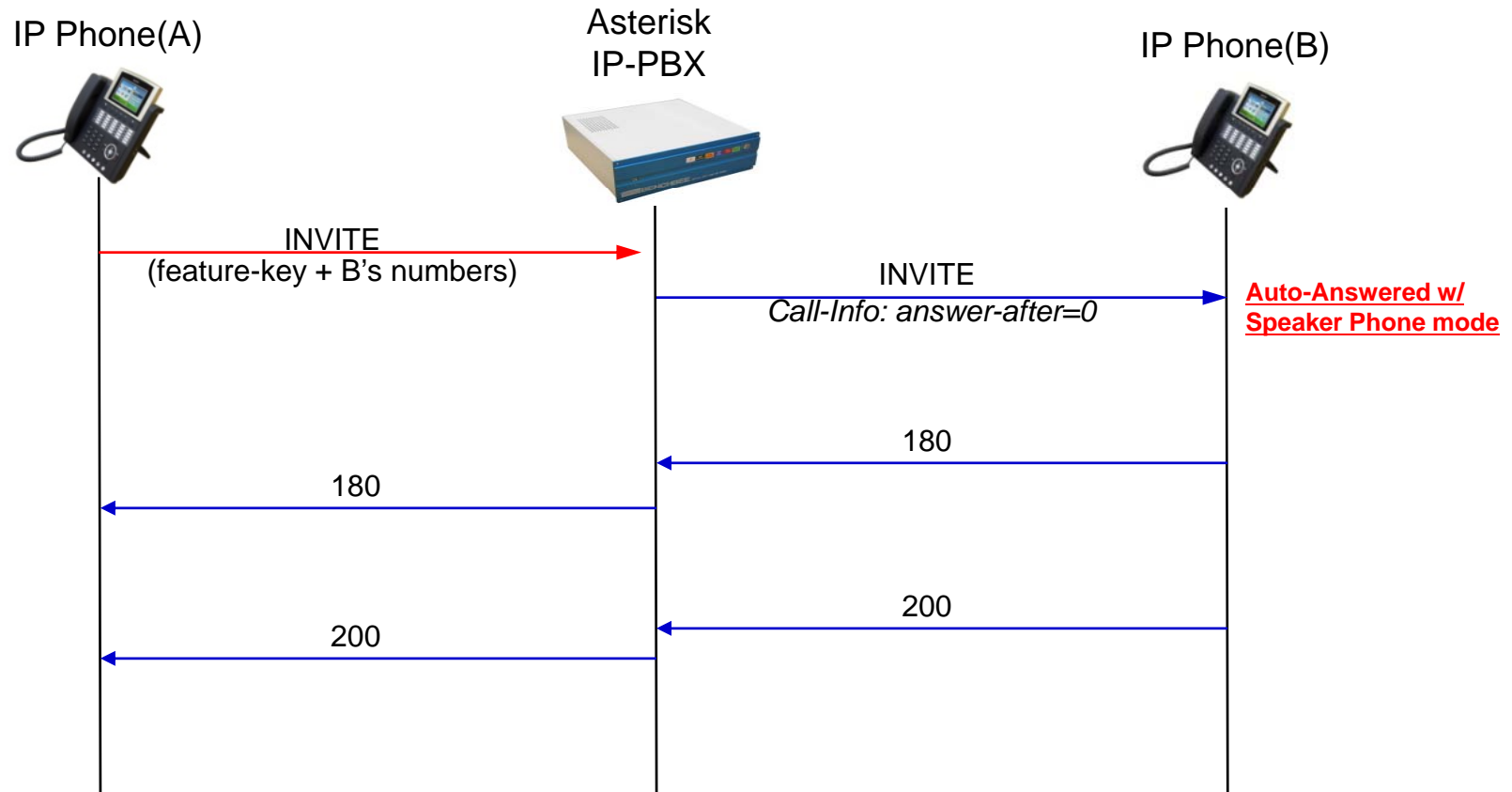
Submit Changes

Paging and Intercom

- Paging Extension numbers.
- Description
- Device List (Multi-Select)
- busy option
- Duplex option (1-way or 2-way)
- default page group

FreePBX Let Freedom Ring™
FreePBX is a registered trademark of Bandwidth.com
FreePBX 2.9.0.7 is licensed under GPL

Paging and Intercom Message Flow Example



BLF Service Diagram



BLF IP-Phone Example (1/2)



BLF IP-Phone Example (2/2)

- **BLF (Busy Lamp Field)**

The Busy Lamp Field (BLF) feature allows you to program a key that monitors whether or not another user is on a call.

The BLF Key also acts as a Speed Dial key to the monitored user's number, and as a Call Pickup key on behalf of the monitored user.

The appearance of the LED indicator for the BLF key is as follows:

Line Status	BLF Indicator	Description
Idle (On hook)	Off	The user being monitored is not on a call, nor dialing a call. The BLF key can be used as a speed dial key to the monitored user.
Off hook/Outgoing call	Solid Red	The monitored user is in the process of placing a call. The BLF key can be used as a speed dial key to the monitored user.
Incoming call	Flashing Red	A call is incoming for the monitored user. Press the BLF key to pick up the call on behalf of the monitored user.
Connected call	Solid Red	The incoming call has been answered by the monitored user. The BLF key can be used as a speed dial key to the monitored user.

BLF Protocols

- **RFC 4235 – Dialog Event Package**

RFC 4235 defines the mechanism for an endpoint to subscribe to the state of any dialog involving another endpoint. RFC 4235 is an instantiation of the SIP Specific Event Notification Framework (RFC 3265) and defines the dialog event package. It also defines the behavior of the device (subscriber) and the Application Server in the context of subscriptions to the dialog event package.

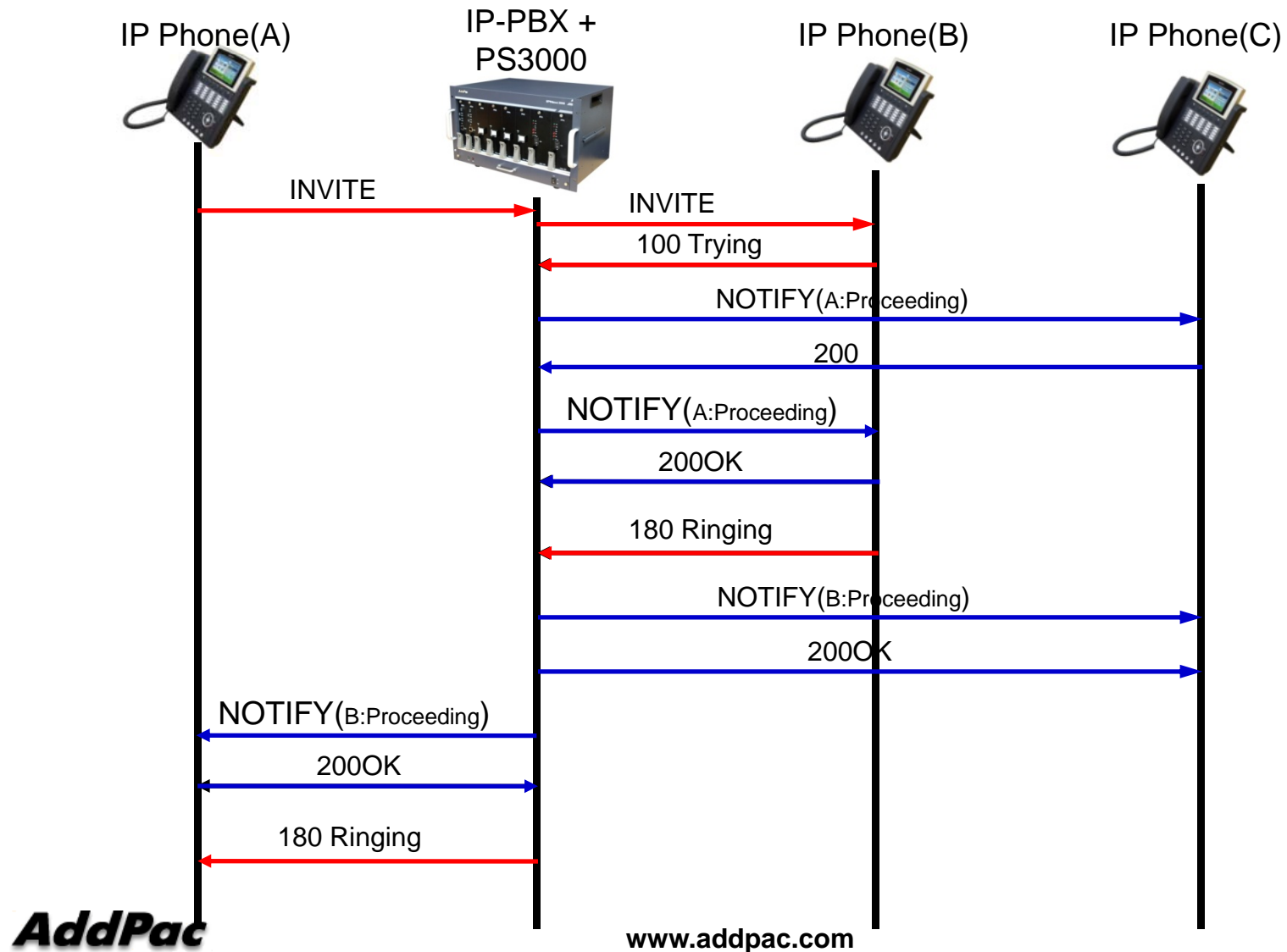
- **RFC 4662– Resource Lists**

RFC 4662 defines an aggregating mechanism that allows for subscribing and notifying for a list of resources. This is useful in the context of the Attendant Console application whether the device needs to subscribe to a large number of resources.

- **Compatibility**

- AddPac IP-PBX
- Asterisk IP-PBX
- BroadWorks IP-PBX







BLF Message Flow Example



IP Phone Comparison Table

	AP-IP300	AP-IP230	AP-IP160	AP-IP120	AP-IP90
					
LCD Size	4.3 Inch Color LCD	5 Inch Color LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD
Touch Screen	N/A	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	16 Key with Presence LED	12 Key with Presence LED	N/A
Voice Codec	G.711/G.726/G.729/G.723	G.711/G.726/G.729/G.723	G.711/G.726/G.729/G.723	G.711/G.726/G.729/G.723	G.711/G.726/G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support
LAN Port	2	2	2	2	2
PoE(Optional)	Support	Support	Support	Support	Possible
FXO(Optional)	Support	Support	Support	Support	Support

IP Video Phone Comparison Table

	AP-VP500	AP-VP300N	AP-VP280	AP-VP250	AP-VP230	AP-VP120
						
LCD Size	12.1 Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	4.3Inch Touch Screen	5Inch Touch Screen	4.3Inch
Camera	CCD	CCD	CMOS	CMOS	CMOS	CMOS
Video Codec	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Video MCU	N/A	N/A	N/A	N/A	N/A	N/A
Voice MCU	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party
LAN Port	2	2	2	2	2	2
PoE	N/A	Support	N/A	Support	Support	Support



AP-IP300 IP Phone

IP Phone

AP-IP300 Premium IP Phone

Main Features

- 4.3 Inch Color LCD Display
- 25 Speed-Dial Keys with User Presence LED
- 4 Soft Key for Call Control
- Various Function Keys
- H.323/SIP Concurrent VoIP Signaling Stack Embedded
- High-performance Voice Codec Support
 - G.711/G.726/G.729/G.723, etc
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Quality Speaker Phone Features (Acoustic Echo Canceller)
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism

IP Phone

AP-IP300 Premium IP Phone

RISC
CPU

High-end
DSP

Hardware Specification

- RISC Microprocessor Computing Power
- 4.3 Color LCD
- 25 Speed Dial Keys with User Presence LED
- High-end Programmable DSP Hardware Architecture
- High quality Audio and Voice Interface
 - Stereo Audio Input & Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
 - One(1) USB 1.0 Interface
- PSTN Interface
 - One(1) FXO(RJ11) interface
- Power Supply
 - *External DC adaptor (5V)*
 - *Power Switch*
 - *PoE (Power over Ethernet)*

IP Phone

AP-IP300 Premium IP Phone

RISC
CPU

High-end
DSP

Hardware Specification



AddPac

www.addpac.com

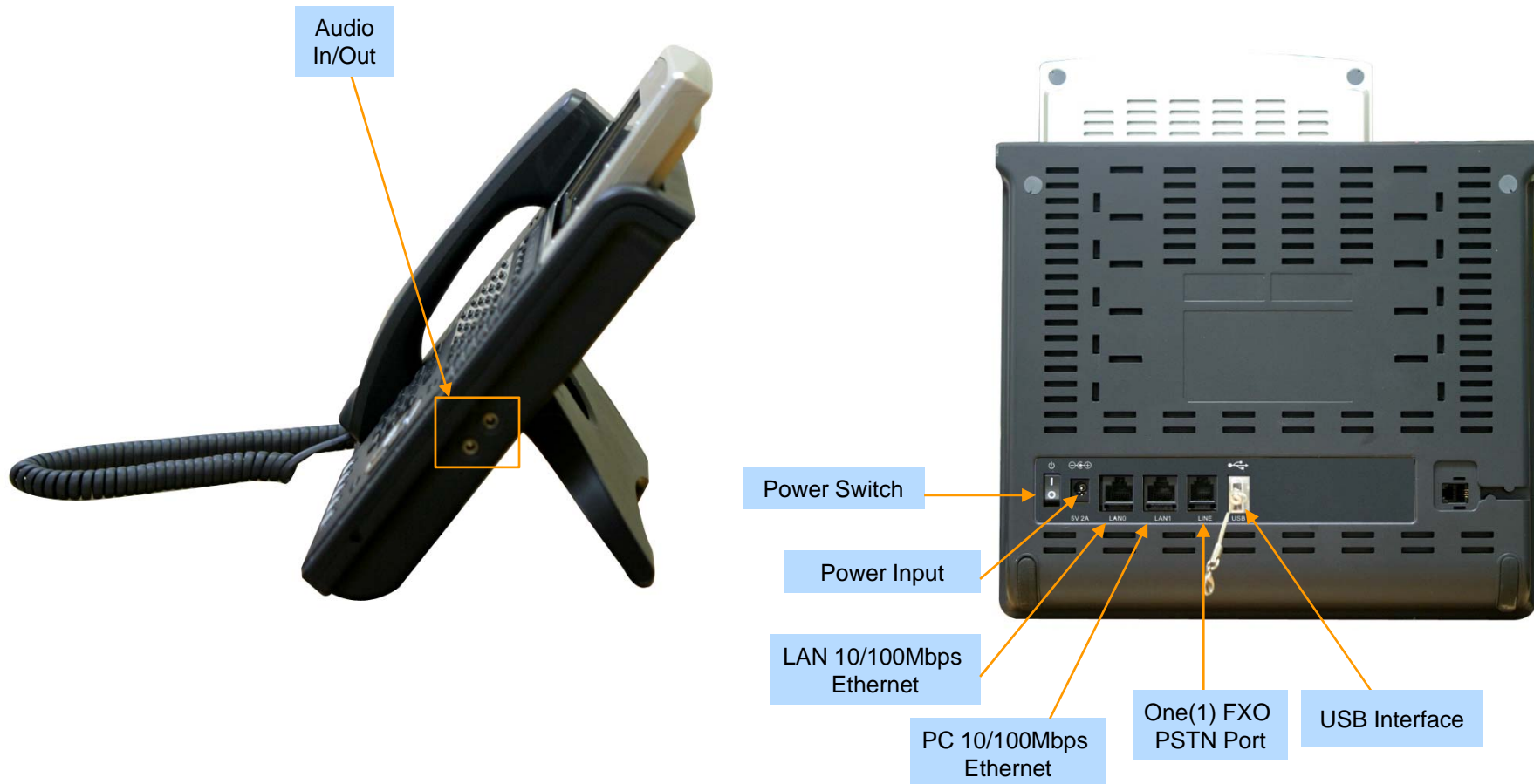
IP Phone

AP-IP300 Premium IP Phone

RISC
CPU

High-end
DSP

Hardware Specification





AP-IP230 IP Phone

IP Phone

AP-IP230 Premium IP Phone

Main Features

- 5 Inch Color LCD Display, Touch Screen
- Touch Screen based 25 Speed-Dial Keys
- 4 Soft Key for Call Control
- Various Function Keys
- H.323/SIP Concurrent VoIP Signaling Stack Embedded
- High-performance Voice Codec Support
 - G.711/G.726/G.729/G.723, etc
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Quality Speaker Phone Features (Acoustic Echo Canceller)
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism

IP Phone

AP-IP230 Premium IP Phone

RISC
CPU

High-end
DSP

Hardware Specification

- RISC Microprocessor Computing Power
- 5 Inch Color LCD with Touch Screen
- High-end Programmable DSP Hardware Architecture
- High quality Audio and Voice Interface
 - Stereo Audio Input & Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
 - One(1) USB 1.0 Interface
- PSTN Interface
 - One(1) FXO(RJ11) interface
- Power Supply
 - *External DC adaptor (5V)*
 - *Power Switch*
 - *PoE (Power over Ethernet)*

IP Phone

AP-IP230 Premium IP Phone

Hardware Specification

RISC
CPU

High-end
DSP

5 Inch
LCD with Touch
Screen



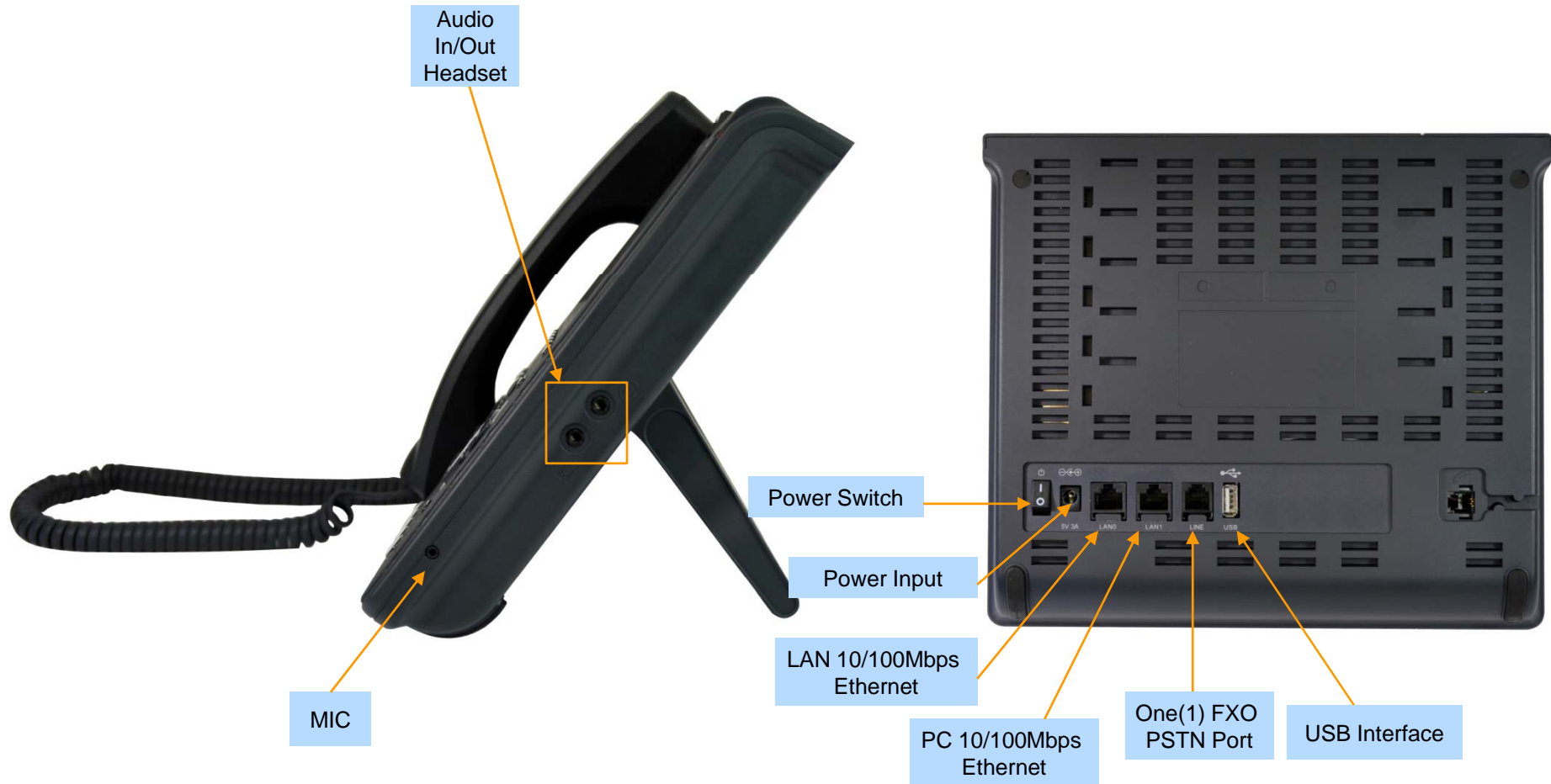
IP Phone

AP-IP230 Premium IP Phone

RISC
CPU

High-end
DSP

Hardware Specification





AP-IP160 IP Phone

IP Phone

AP-IP160 IP Phone

Main Features

- 4 Text Line Graphic LCD Display
- 16 Speed-Dial Keys with User Presence LED
- 4 Soft Key for Call Control
- Various Function Keys
- H.323/SIP Concurrent VoIP Signaling Stack Embedded
- High-performance Voice Codec Support
 - G.711/G.726/G.729/G.723, etc
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Quality Speaker Phone Features (Acoustic Echo Canceller)
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism

IP Phone

AP-IP160 IP Phone

RISC
CPU

High-end
DSP

Hardware Specification

- RISC Microprocessor Computing Power
- 4 Text Line Graphic LCD
- Navigation Key for Menu Search
- High-end Programmable DSP Hardware Architecture
- High quality Audio and Voice Interface
 - Stereo Audio Input & Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
- PSTN Interface
 - One(1) FXO(RJ11) interface
- Power Supply
 - *External DC adaptor (5V)*
 - *Power Switch*
 - *PoE (Power over Ethernet)*

IP Phone

AP-IP160 IP Phone

RISC
CPU

High-end
DSP

Hardware Specification



AddPac

www.addpac.com

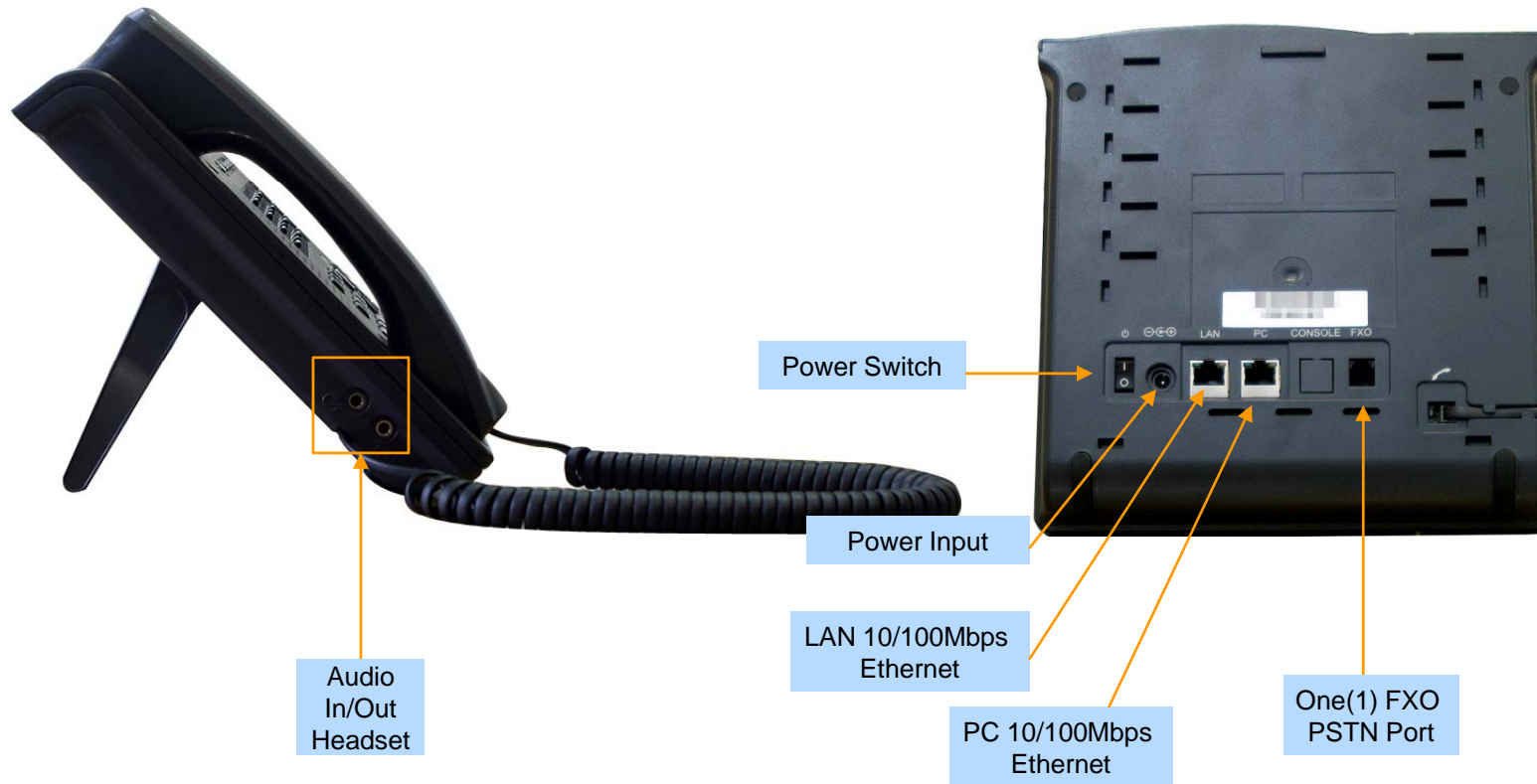
IP Phone

AP-IP160 IP Phone

RISC
CPU

High-end
DSP

Hardware Specification





AP-IP120 IP Phone

IP Phone

AP-IP120 IP Phone

Main Features

- 4 Text Line Graphic LCD Display
- 12 Speed-Dial Keys with User Presence LED
- 4 Soft Key for Call Control
- Various Function Keys
- H.323/SIP Concurrent VoIP Signaling Stack Embedded
- High-performance Voice Codec Support
 - G.711/G.726/G.729/G.723, etc
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Quality Speaker Phone Features (Acoustic Echo Canceller)
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism

IP Phone

AP-IP120 IP Phone

RISC
CPU

High-end
DSP

Hardware Specification

- RISC Microprocessor Computing Power
- 4 Text Line Graphic LCD
- Navigation Key for Menu Search
- High-end Programmable DSP Hardware Architecture
- High quality Audio and Voice Interface
 - Stereo Audio Input & Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
- PSTN Interface
 - One(1) FXO(RJ11) interface
- Power Supply
 - *External DC adaptor (5V)*
 - *PoE (Power over Ethernet)*

IP Phone

AP-IP120 IP Phone

RISC
CPU

High-end
DSP

Hardware Specification





Thank you!

AddPac Technology Co., Ltd.
Sales and Marketing

Phone +82.2.568.3848 (KOREA)

FAX +82.2.568.3847 (KOREA)

E-mail : sales@addpac.com