IP PBX

IP02

The IP02 is a complete Asterisk Appliance with two FXO/FXS modules. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT functions. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications. Targeting for SOHO user and SMB market with an easy to use graphical interface, IP02 provides a cost-saving solution on their telecommunication/data needs. With IP02, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

FEATURES

Open Source Asterisk

uClinux Operation System

Configurable IVR menu

Over 50-100 available SIP/IAX2 extensions for SMB use

20 concurrent calls

Voicemail to Email

Call Forward, Call Waiting, Call Transfer (Blind Transfer/ Attender Transfer), Call Pickup/Call Parking, Call Queues, Ring Group, Call Detail Record Call Routing

www.houyuanhk.com
Conference Room

Password Protect for Conference Room

Follow Me

Music On Hold

Skype for SIP

SIP Trunk, IAX2 Trunk, PSTN Analog Trunk

Configure via WEB interface

Codec: G.711u/a, G.729, GSM, Speex, G.726

Full SSH access

OSLEC (Open Source Line Echo Canceller)

Support VPN

Support N2N

**SPECIFICATION**

**Hardware**

CPU: 400MHz Blackfin 532 Chip
Flash: 256 MB
SDRAM: 64MB
LEDs x 6
Programmable Reset Button

www.houyuanhk.com
Module configuration
Slots for modules : 1
Dual port FXO: 210X
Dual port FXS :210S
Dual port FXO/FXS: 210XS

Interface
2 X RJ45 ports
1 X Power port
1 X UART interface
2 X RJ11 ports (FXS/FXO interchangeable)

Electrical
Power Input:DC 12V/1000mA

IP04

The IP04 is a complete Asterisk Appliance with four FXO or FXS module. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT functions. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications.

Targeting for SOHO user and SMB market with an easy to use graphical interface, IP04 provides a cost-saving solution on their telecommunication data needs. With IP04, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

FEATURES

Open Source Asterisk

uClinitx Operation System

www.houyuanhk.com
Configurable IVR menu

Over 50-100 available SIP/IAX2 extensions for SMB use

20 concurrent calls

Voicemail to Email

Call Forward, Call Waiting, Call Transfer (Blind Transfer/Attender Transfer), Call Pickup/Call Parking, Call Queues, Ring Group, Call Detail Record Call Routing

Conference Room

Password Protect for Conference Room

Follow Me

Music On Hold

Skype for SIP

SIP Trunk, IAX2 Trunk, PSTN Analog Trunk

Configure via WEB interface

Codec: G.711u/a, G.729, GSM, Speex, G.726

Full SSH access
**SPECIFICATION**

**Hardware**
- CPU: 400MHz Blackfin 532 Chip
- Flash: 256 MB
- SDRAM: 64MB
- LEDs x 8
- Programmable Reset Button

**Module configuration**
- Slots for modules: 4
- Single port FXO: 110X
- Single port FXS: 110S

**Interface**
- 1 X RJ45 port
- 1 X Power port
- 1 X UART interface
- 1 X MMC/SD slot
- 4 X RJ11 ports (FXS/FXO interchangeable)

**Electrical**
- Power Input: DC 12V/3000 MA

**IP08**
The IP08 is a complete Asterisk Appliance with eight FXO/FXS ports. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT functions. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications. Targeting for SOHO user and SMB market with an easy to use graphical interface, IP08 provides a cost-saving solution on their telecommunication data needs. With IP08, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

FEATURES

- Open Source Asterisk
- uClinux Operation System
- Configurable IVR menu
- Over 50-100 available SIP/IAX2 extensions for SMB use
- 20 concurrent calls
- Voicemail to Email
- Call Forward, Call Waiting, Call Transfer (Blind Transfer/ Attender Transfer), Call Pickup/Call Parking, Call Queues, Ring Group, Call Detail Record Call Routing
- Conference Room
- Password Protect for Conference Room
- Follow Me
- Music On Hold
- Skype for SIP

www.houyuanhk.com
SIP Trunk, IAX2 Trunk, PSTN Analog Trunk

Configure via WEB interface

Codec: G.711u/a, G.729, GSM, Speex, G.726

Full SSH access

OSLEC (Open Source Line Echo Canceller)

Support VPN

Support N2N

**SPECIFICATION**

**Hardware**

CPU: 400MHz Blackfin 533 Chip
Flash: 256 MB
SDRAM: 64MB
LEDs x 14
Programmable Reset Button

**Module configuration**

Slots for modules :4
Dual port FXO: 210X
Dual port FXS : 210S
Dual port FXO/FXS: 210XS

**Interface**

2 X RJ45 ports
1 X Power port
1 X UART interface
1 X MMC/SD slot
1 X USB port
8 X RJ11 ports (FXS/FXO interchangeable)

**Electrical**

Power Input:DC 12V/3000 mA

www.houyuanhk.com
Asterisk Cards

400P

Features
-The Asterisk card 400P is a half-length PCI 2.2-compliant card that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.
-It's a PCI Card for Asterisk, TrixBox and other Open Source Telephony projects with 4 modules.
-The card is fully compatible with all analog Digium and other similar analog cards and modules with no changes to the drivers.
-FXO modules are used to plug existing analog telephone lines into your phone system.

With the Asterisk card 400P, Open Source Asterisk PBX software and a standard PC, users can create a Small Office Home Office (SOHO) telephony environment which includes all the sophisticated features of a high-end PBX/Voicemail platform.

Support 1-4 FXO/FXS with PCI Interface

Support Asterisk, Freeswitch, Dahdi, Zaptel

Support Trixbox, Elastix, Askozia

Application: PBX/Voicemail/IVR/Call Center/Call Park/Call Pickup/Call Transfer/Call Forward/Caller ID/Call Waiting/Call Conference

Scalable and Effective SOHO Solution, Target Applications
-
Small Office Home Office (SOHO) applications
-Gateway Termination to Analog Telephones
-Add Inexpensive Analog Phones to Existing PBXs
-Wireless Point-to-Point Applications between Asterisk Servers
Services and Features
- Caller ID and Call Waiting Caller ID
- ADSI Telephones
- PCI Half-length Slot
- RJ-11C Connector

Modules
Motherboard: 400P
FXO module: 1-4 pcs
FXS module: 1-4 pcs
Hardware Requirement
500-Mhz Pentium III or better with 64MB RAM
Available PCI Slot 5v

410P

Features
-The Asterisk PCI card 410P is a half-length PCI 2.2-compliant card that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.
- It's a PCI Card for Asterisk, TrixBox, Elastix and other Open Source Telephony projects with 4 modules.
- The card is fully compatible with all analog Digium and other similar analog cards and modules with no
changes to the drivers.

- FXO modules are used to plug existing analog telephone lines into your phone system.

With the Asterisk card 410P, Open Source Asterisk PBX software and a standard PC, users can create a Small Office Home Office (SOHO) telephony environment which includes all the sophisticated features of a high-end PBX/Voicemail platform.

**Support 1-4 FXO/FXS with PCI Interface**

**Support Asterisk, Freeswitch, Dahdi, Zaptel**

**Support Trixbox, Elastix, Askozia**

**Application:** PBX/Voicemail/IVR/Call Center/Call Park/Call Pickup/Call Transfer/Call Forward/Caller ID/Call Waiting/Call Conference

### Scalable and Effective SOHO Solution, Target Applications

- Small Office Home Office (SOHO) applications
- Gateway Termination to Analog Telephones
- Add Inexpensive Analog Phones to Existing PBXs
- Wireless Point-to-Point Applications between Asterisk Servers

### Services and Features

- Caller ID and Call Waiting Caller ID
- ADSI Telephones
- PCI Half-length Slot
- RJ-11C Connector

### Modules

- Motherboard: 410P
- FXO module: 3 pcs
- FXS module: 1 pcs

### Hardware Requirement

- 500-Mhz Pentium III or better with 64MB RAM
- Available PCI Slot 5v

---

**X16P**

www.houyuanhk.com
X16P Asterisk card is the telephony PCI card that support up to sixteen FXO and FXS ports. Using X16P analog card, open source Asterisk PBX and stand alone PC, users can create their SOHO telephony solution which include all the sophisticated features of traditional PBX, and extend features such as voicemail in IP PBX. The FXO and FXS modules are interchangeable to suit various requirements.

Features

- Support 1-16 FXO/FXS with PCI Interface
- Support Asterisk, Freeswitch, Dahdi, Zaptel
- Support Trixbox, Elastix, Askzia
- Application: PBX/Voicemail/IVR/Call Center/Call Park/Call Pickup/Call Transfer/Call Forward/Caller ID/Call Waiting/Call Conference

Scalable and Effective SOHO Solution, Target Applications
- Small Office Home Office (SOHO) applications
- Gateway Termination to Analog Telephones
- Add Inexpensive Analog Phones to Existing PBXs
- Wireless Point-to-Point Applications between Asterisk Servers

Services and Features
- Caller ID and Call Waiting Caller ID

www.houyuanhk.com
- ADSI Telephones
- PCI Half-length Slot
- RJ-11C Connector

Configuration
Motherboard: AX1600P
Single port FXS module: AX210S
Single port FXO module: AX210X

Hardware requirement
500-Mhz Pentium III
64MB RAM
3.3V or 5V PCI 2.2 slot

PCI card dimension
264mm (length) × 121mm (height)

IP PHONE
620P

620P VoIP phone is Houyuan business IP phone terminal with Broadcom solution which adopts multiple voice control protocols and voice compression codec to directly convert analog voice into IP packet for
internet transport, thus effectively using the existing bandwidth to provide PSTN quality voice service. 620P IP phone supports SIP and IAX2 protocol, offering two 10/100Mbps Ethernet interface with built in router. It is compatible with various softswitch systems and VoIP voice gateways to provide broadband IP voice service.

**Features**

- High performance Broadcom solution to support crystal clear voice quality
- SIP /IAX2 support
- Back light on LCD display
- 2 SIP/IAX lines
- G.722 high definition voice codec support
- Elastix certified
- Dual RJ45 port (WAN+PC) with Router/Switch function
- Full duplex speaker with HiFI voice quality
- VLAN QoS
- L2TP VPN and Open VPN available
- Wall mounted structure
- Independent RJ9 headset support
Customized right tone for different regions

Auto provision of configuration by Mac address

Power over Ethernet

**VoIP**

- SIP (SIP RFC3261, RFC 2543), IAX2 support
- 2 SIP lines + 1 IAX2 line
- Redundancy sip server capable
- Hotline
- Call Forward, Call transfer, Call hold, Call waiting
- 3-way Talking, Pickup, Join call, Redial, Unredial
- Call Park, vport, click to dial
- DND (Do Not Disturb)
- Black List, Limit List
- E.164 dial plan and customized dial rules

**Voice**

- Tone generation and Local DTMF re-generation according with ITU-T
- G.711 (A-law or u-law), G722, G723, G726, and G729
- AGC (Auto Gain Control)
- AEC (Auto Echo Cancellation)
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- G.165 compliant 96 ms echo cancellation

**Networking**

- Static/Dynamic WAN-IP-Addressing
- NAT, Firewall
- Support L2TP and OpenVPN
- Support DMZ
- WAN support Primary and Alter function
- Qos support Diffserv
- Support VLAN
- DHCP client and server
- Support PPPoE, (used for ADSL, cable modem connecting)

**Management**

- Web, telnet and keypad management

www.houyuanhk.com
Adjustable user password and super password
Upgrade firmware through HTTP, FTP or TFTP
Telnet remote management
Upload/download configuration file
Auto-provisioning (firmware and configuration)
Safe mode provide reliability
Phone book, maximum 500 entries

**Hardware**
- 128 x64 Dot matrix
- Support multi language (LCD support Latin language system, web support all languages) and easy dynamic switch between different languages
- Wall mounting
- Full duplex speaker
- MWI : Mono-color x 1
- Line keys: Bi-Color x 2
- Voicemail LED: Mono-Color x 1
- Headset LED: Mono-Color x 1
- Mute LED: Mono-Color x 1
- Speaker LED: Mono Color x 1

**Interfaces**
- WAN * 1
- LAN * 1
- RJ9 handset Jack *1
- RJ9 headset Jack * 1
- Power jack * 1

**Compliance Certificates**
- CE: EN55024 EN55022
- FCC: part 15
- ROHS in EU