We Focus · We Deliver

www.zycoo.com

About ZYCOO

ZYCOO is a leading designer and provider of IP Phone System around the world. Focus on small and medium enterprises' communication requirement, we are engaged in developing the IP Phone System, manufacture of VoIP device and providing VoIP communication and all related solutions.

We have outstanding R&D team with extensive experience on embedded device. ZYCOO developed four series IP PBX which apply to small and medium enterprises (SMEs) and small office&home offices(SOHOs). All series IP PBX are based on standard SIP, can be compatible with the global mainstream terminal.The first mini IP PBX was issued by ZYCOO.

ZYCOO's products and solutions are widely applied in more than 85 countries and regions. Meanwhile, we had developed and produced more than 10 customized embedded boards based on ARM and DSP chips for North American and European customers.

ZYCOO achieved ISO9001:2008 Certificate issued by Zhongjian Certification Co.,Ltd. This approve ZYCOO focus on the quality control, our production will be always comply with this standard.

Meanwhile, ZYCOO is awarded “Excellent Entrepreneurial Enterprise of China”, we expect more distribution channels to deliver our innovative technology and better service to the world wide, we are willing to adopt flexible ways to establish good business partnerships with all of our customers and friends all over the world.

WE FOCUS

We focus on the product quality, stability and service.

WE DELIVER

We deliver qualified product, strong technical and strategic support.

Our Achievements

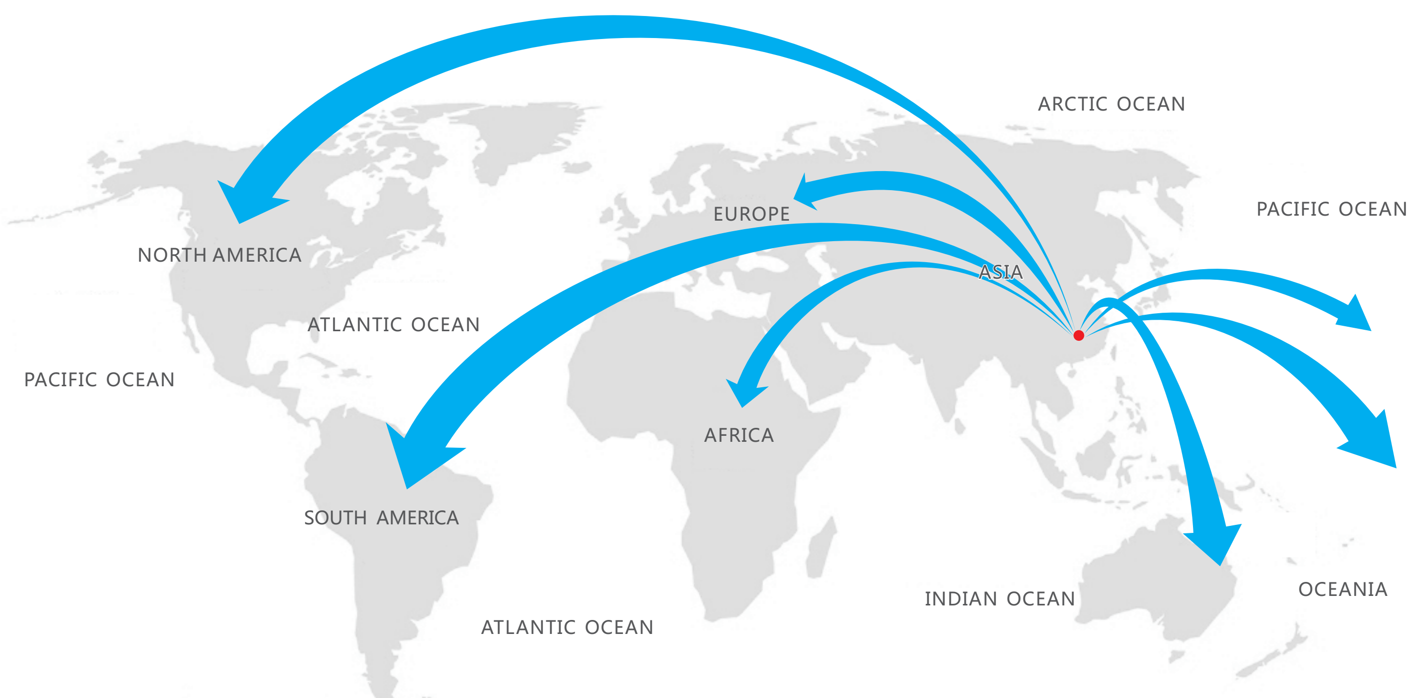
2010: Excellent Entrepreneurial Enterprise of China

2011: 2011 Internet Telephony Product Of The Year

2011: 2011 Communications Solutions Product Of The Year

Our Strategies

Nowadays, you can find that ZYCOO products are utilized in every country almost. The setup of Distribution channel is important to develop our brand and reputation. ZYCOO would like to share our talent and effort with our distribution partner, set up non-conflict relationship with them and achieve mutual business success.



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ZX20 Series IP Phone System

The ZX20 Series IP Phone System is the first mini IP PBX in the world released in 2010. Based on SIP, it supports 30 extension users.

It's the ideal IP Phone system for small businesses and home offices (SOHOs). It's easy to setup and use, highly cost-effective, and compact body takes minimal space.

Being more flexible, the ZX20 integrates 2 analog ports to support lifeline pass through, as well as PoE to promote productivity in your working environment.

	Model	FXS	FXO	SIP Extension
ZX20-A2	A202	0	2	30
	A211	1	1	30

Advantages:

Cost Savings

Free long distance call

After the setup of enterprise branch network, it's free to call within the enterprise.

Cheap VoIP long distance fee

Connected with VoIP Provider to enjoy the cheap long distance fee, it's very helpful for enterprise to set up the branches in different location.

Promote Working Efficiency

Promote communication efficiency

Auto Attendant, Call Transfer/Forward, IVR, Conference Call, etc., such features will help the enterprise to promote the working efficiency.

After registration to the ZX20 IP Phone system from soft phone installed in laptop or mobile phone, the staffs can utilize all the features of the IP Phone system wherever they are.

You can be out of the office without missing any call due to the "Follow Me" function.

Ease of management

Easy and friendly configuration interface.

Web-based local and remote management and maintenance.

Compatibility

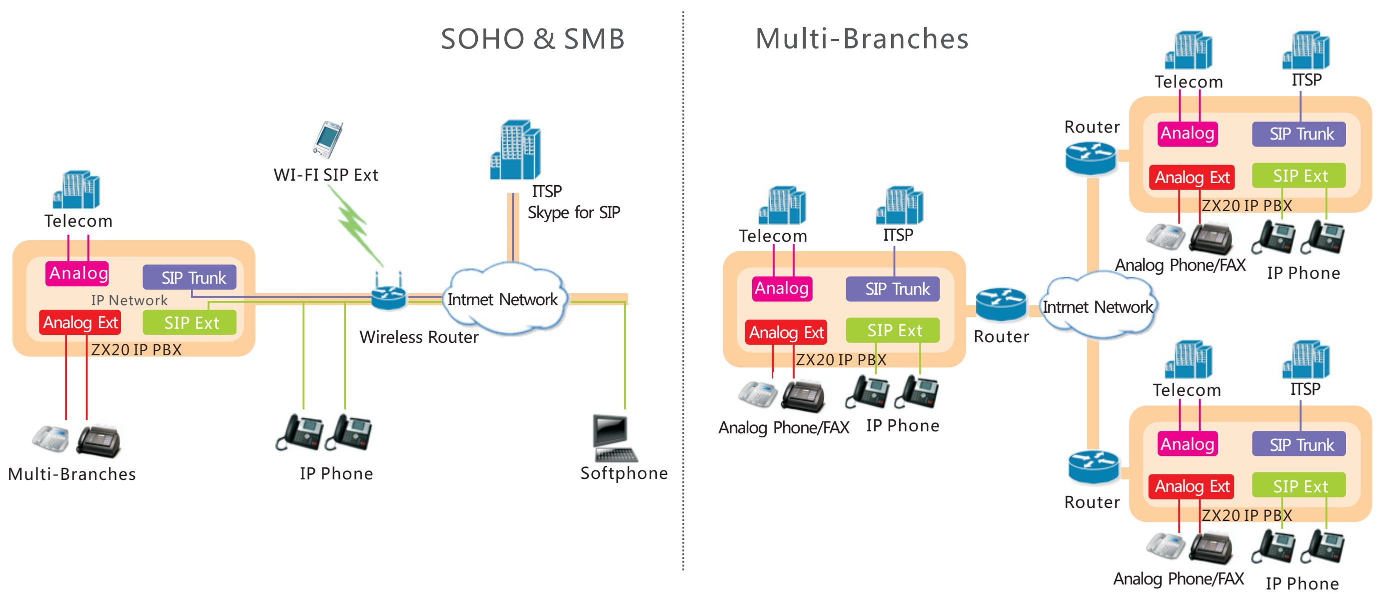
Compatible with mainstream SIP terminal

Based on open-standard SIP, it's compatible with mainstream SIP terminal and you can choose the cost-effective SIP terminal as you like.

Specifications

Item	Specification
PBX Features	Auto Attendant Busy Lamp Field (BLF) Call Recording Conference Room(3 rooms) Direct Inward Dialing Number (DID) DISA Feature Code FXO To FXS Lifeline Interactive Voice Response (IVR) IP Phone Provisioning Multi-languages Phone Book Skype For SIP T.38 Fax Voicemail To Email VoIP Trunk(10 trunks)
System Capacity	30 Extensions | 256MB Onboard Flash | Recording Storage(GSM):2100 mins
Call Features	Caller ID Call Transfer/Parking/Forward/Hold/Waiting Call Queue Conference Call Music On Hold Paging and Intercom Ring Group Video Call
Codec & Protocol	Audio Codec: G.711(a-law, u-law),G.729,G.726,GSM,Speex Video Codec(Pass through): H.261, H.263, H.263+, H.264 Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band
Network Features	DHCP Server DHCP/Static IP Assignment VPN(N2N/L2TP) Client NAT(Network Address Translation) DDNS Client (dyndns.org/no-ip.org/ zoneedit.com) PoE(Power Over Ethernet)
Hardware Interface	1 WAN Interface 2 Analog Interfaces (RJ11) 1 Power Interface 1 Reset Button LED Indicator Power Input : AC 100~240V Power Output : DC 12V 1A
Environment Requirement	Temperature: -10 °C -45 °C | Storage temperature: -30 °C -65 °C | Humidity: 10-80% no dew

VoIP Solution



ZYCOO
Zycoo Co.,Ltd
Add : 7F/ B7, Tianfu Software Park, Chengdu, China. 610041 Tel : +86 28 85337096 Fax : +86 28 85337096 ext:800
www.zycoo.com E-Mail : zycoo@zycoo.com

ZX50 Series IP Phone System

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The ZX50 Series IP Phone System is entry level system designed for SOHO and SMEs with less than 100 employees. It can be hybrid with Analog/GSM/E1/T1/BRI interface to offer a seamlessly-integrated solution for the up-to-date telecommunication needs.

ZX50-A4

ZX50-A8

ZX50-AG42

ZX50-G4

ZX50-AE41

ZX50-B4

Advantages:

Cost Savings

Free long distance call

After the setup of enterprise branch network, it's free to call within the enterprise.

Cheap VoIP long distance fee

Connected with VoIP Provider to enjoy the cheap long distance fee, it's very helpful for enterprise to set up the branches in different location.

Promote Working Efficiency

Promote communication efficiency

Auto Attendant, Call Transfer/Forward, IVR, Conference Call, etc., such features will help the enterprise to promote the working efficiency.

After registration to the ZX50 IP Phone system from soft phone installed in laptop or mobile phone, the staffs can utilize all the features of the IP Phone system wherever they are.

You can be out of the office without missing any call due to the "Follow Me" function.

Ease of management

Easy and friendly configuration interface.

Web-based local and remote management and maintenance.

Compatibility

Compatible with mainstream SIP terminal

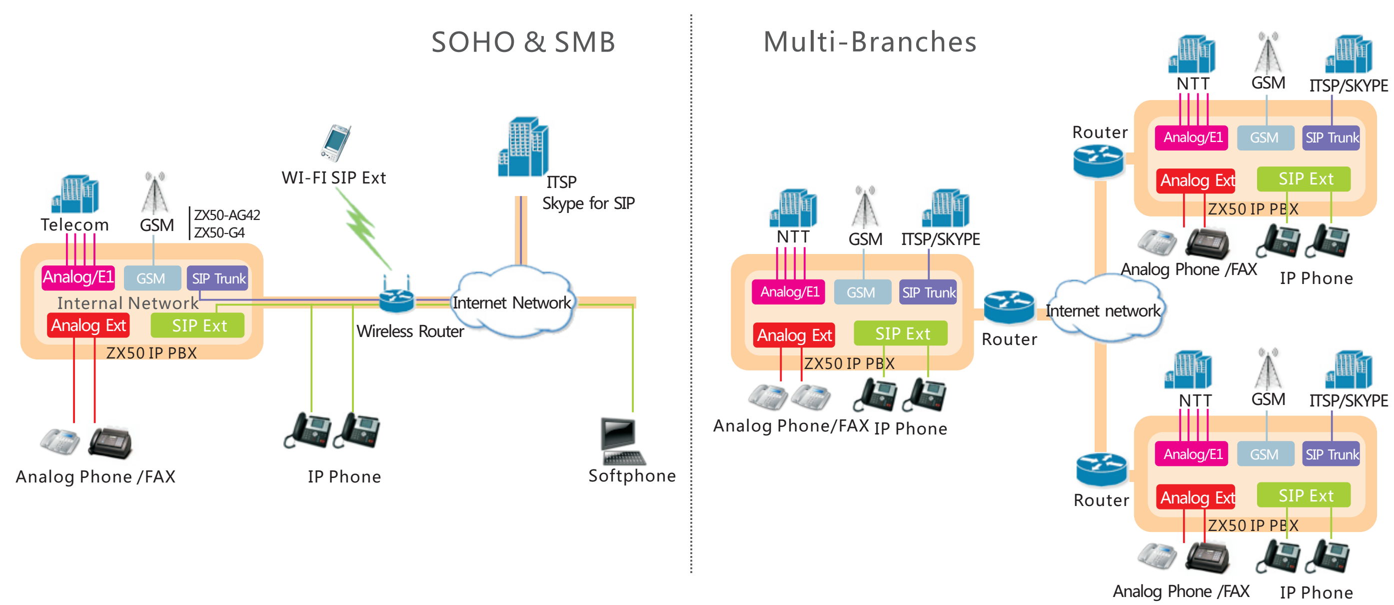
Based on open-standard SIP, it's compatible with mainstream SIP terminal and you can choose the cost-effective SIP terminal as you like.

	Model	FXS	FXO	GSM	E1	BRI
ZX50-A4	A404		4			
	A422	2	2			
ZX50-A8	A808		8			
	A826	2	6			
ZX50-AG42	AG4204		4	2		
	AG4222	2	2	2		
ZX50-G4	G4			4		
ZX50-AE41	AE4104		4		1	
	AE4122	2	2		1	
ZX50-B4	B4					4

Specifications

Item	Specification	
PBX Features	Auto Attendant Blacklist Busy Lamp Field (BLF) Call Recording Conference Room(3 rooms) Direct Inward Dialing Number (DID) DISA Feature Code	FXO To FXS Lifeline Interactive Voice Response (IVR) IP Phone Provisioning Multi-languages(Chinese/English/Portuguese) Phone Book Skype For SIP Voicemail To Email VoIP Trunk(10 trunks)
System Capacity	100Extensions | 1GB Onboard Flash | Recording Storage(GSM):8500mins	
Call Features	Caller ID Call Transfer/Parking/Forward/Hold/Waiting Call Queue Conference Call	Music On Hold Paging and Intercom Ring Group Video Call
Codec & Protocol	Audio Codec: G.711(a-law, u-law),G.729,G.726,GSM,Speex Video Codec(Pass through): H.261, H.263, H.263+, H.264 Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band	
Network Features	DHCP Server DHCP/Static IP Assignment VPN(N2N/L2TP) Client NAT(Network Address Translation) DDNS Client (dyndns.org/no-ip.org/ zoneedit.com	
Hardware Interface	2 Ethernet Interfaces(WAN/LAN) 1 Power Interface 1 Reset Button 4-8 Analog Interfaces (ZX50-A4/A8) 1 E1/T1 Interface (ZX50-AE41)	2/4 GSM Interfacess (ZX50-AG42/G4) 4 BRI Interfaces (ZX50-B4) LED Indicators Power Input : AC 100~240V Power Output : DC 12V 2A
Environment Requirement	Temperature: -10 °C -45 °C | Storage temperature: -30 °C -65 °C | Humidity: 10-80% no dew	

VoIP Solution



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ZX60 Series IP Phone System

The ZX60 Series IP Phone system is developed based on embedded DSP with 32 analog ports. It's applicable for enterprises with less than 60 employees.

It looks like a gateway, but it's more than a gateway by providing advanced IP PBX features, such as visual voice mail, music on hold, and auto attendant, etc.

With 32 FXS analog ports connected to analog phone at most, it can help you save the cost for purchasing IP Phones. With ZYCOO VoIP solutions, SMEs can take advantage of the VoIP services and quickly deploy VoIP networks to connect with multiple branch locations over the Internet without any change on the current equipment or dial plan.

2011 Communications Solutions Product Of The Year

Model		A32284	A32248
Trunk	Analog	4	8
	SIP	10	10
Extension	Analog	28	24
	SIP	30	30

Advantages:

Cost Savings

Free long distance call

After the setup of enterprise branch network, it's free to call within the enterprise.

Cheap VoIP long distance fee

Connected with VoIP Provider to enjoy the cheap long distance fee, it's very helpful for enterprise to set up the branches in different location.

Promote Working Efficiency

Promote communication efficiency

Auto Attendant, Call Transfer/Forward, IVR, Conference Call, etc., such features will help the enterprise to promote the working efficiency.

After registration to the ZX60 IP Phone system from soft phone installed in laptop or mobile phone, the staffs can utilize all the features of the IP Phone system wherever they are.

You can be out of the office without missing any call due to the "Follow Me" function.

Ease of management

Easy and friendly configuration interface.

Web-based local and remote management and maintenance.

Compatibility

Compatible with mainstream SIP terminal

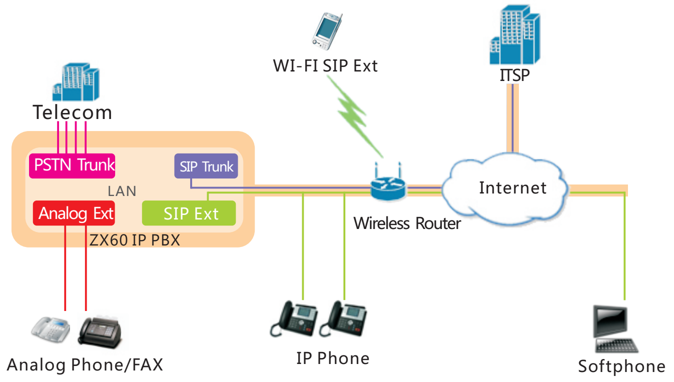
Based on open-standard SIP, it's compatible with mainstream SIP terminal and you can choose the cost-effective SIP terminal as you like.

Specifications

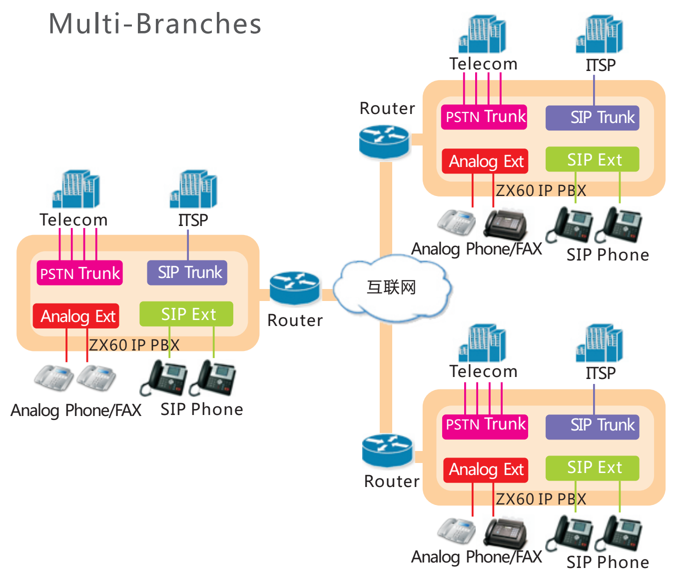
Item	Specification	
PBX Features	Auto Attendant Busy Lamp Field (BLF) Call Recording Conference Room(3 rooms) Direct Inward Dialing Number (DID) DISA Feature Code FXO To FXS Lifeline	Interactive Voice Response (IVR) IP Phone Provisioning Multi-languages(Chinese/English/Portuguese) Phone Book Skype For SIP T.38 Fax Voicemail To Email VoIP Trunk(10 trunks)
System Capacity	32Analog Extension | 30 IP Extension | 1GB Onboard Flash | Recording Storage(GSM):8500 mins	
Call Features	Caller ID Call Transfer/Parking/Forward/Hold/Waiting Call Queue Conference Call	Music On Hold Paging and Intercom Ring Group Video Call
Codec & Protocol	Audio Codec: G.711(a-law, u-law),G.729,G.726,GSM,Speex Video Codec(Pass through): H.261, H.263, H.263+, H.264 Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band	
Network Features	DHCP Server DHCP/Static IP Assignment VPN(N2N/L2TP) Client NAT(Network Address Translation) DDNS Client (dyndns.org/no-ip.org/ zoneedit.com) PoE(Power Over Ethernet)	
Hardware Interface	2 Ethernet Interfaces(WAN/LAN) 1 Power Port 1 Reset Button	1 Console Interface 32 Analog Interfaces LED Indicators Power Input : AC 100~240V
Environment Requirement	Temperature: -10 °C -45 °C | Storage temperature: -30 °C -65 °C | Humidity: 10-80% no dew	

VoIP Solution

SOHO & SMB



Multi-Branches



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ZX100 Series IP Phone System

The ZX100 is enterprise-class IP Phone system, applicable for enterprises up to 500 employees. With 1U rack-mountable housing case, it's convenient to set up. Its strong hardware design provides you a solid, unified platform for voice and data network communications. You can choose 16 analog channels, or 1 E1/T1 channel based on your local condition, and it's easy for you to deploy your networks through its friendly configuration in management panel.

ZX100-A16

ZX100-A16

ZX100-E1

ZX100-E1

	Model	FXS	FXO	E1
ZX100-A16	A16016	0	16	0
	A16214	2	14	0
	A16412	4	12	0
	A16610	6	10	0
	A16808	8	8	0
ZX100-E1	E1	0	0	1

Advantages:

Cost Savings

Free long distance call

After the setup of enterprise branch network, it's free to call within the enterprise.

Cheap VoIP long distance fee

Connected with VoIP Provider to enjoy the cheap long distance fee, it's very helpful for enterprise to set up the branches in different location.

Promote Working Efficiency

Promote communication efficiency

Auto Attendant, Call Transfer/Forward, IVR, Conference Call, etc., such features will help the enterprise to promote the working efficiency.

After registration to the ZX100 IP Phone system from soft phone installed in laptop or mobile phone, the staffs can utilize all the features of the IP Phone system wherever they are.

You can be out of the office without missing any call due to the ″Follow Me″ function.

Ease of management

Easy and friendly configuration interface.

Web-based local and remote management and maintenance.

Compatibility

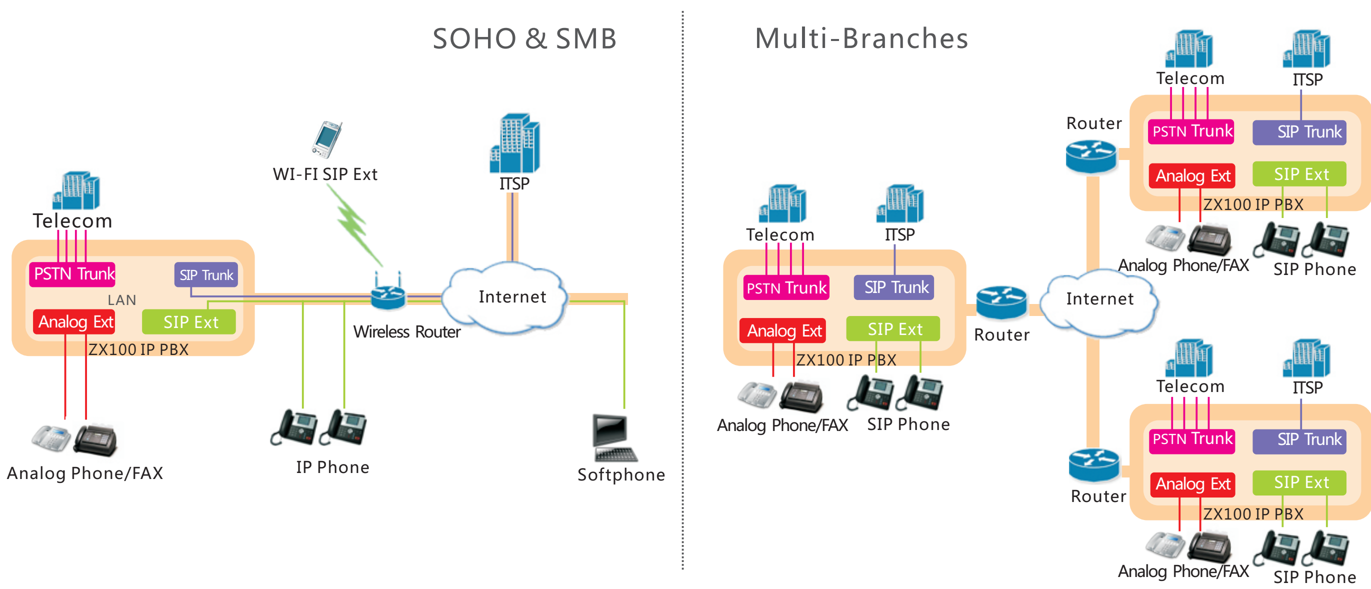
Compatible with mainstream SIP terminal

Based on open-standard SIP, it's compatible with mainstream SIP terminal and you can choose the cost-effective SIP terminal as you like.

Specifications

Item	Specification
PBX Features	Auto Attendant Busy Lamp Field (BLF) Call Recording Direct Inward Dialing Number (DID) DISA Feature Code Interactive Voice Response (IVR) IP Phone Provisioning Phone Book Skype For SIP T.38 Fax Voicemail To Email VoIP Trunk
System Capacity	500Extensions | 2GB SDRAM | 320GB Hard Disk
Call Features	Caller ID Call Transfer/Parking/Forward/Hold/Waiting Call Queue Conference Call Music On Hold Paging and Intercom Ring Group Video Call
Codec & Protocol	Audio Codec: G.711(a-law, u-law),G.729,G.726,GSM,Speex Video Codec(Pass through): H.261,H.263,H.263+,H.264 Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band
Network Features	DHCP Server DHCP/Static IP Assignment VPN(N2N/L2TP) Client NAT(Network Address Translation) DDNS Client (dyndns.org/no-ip.org/ zoneedit.com)
Hardware Interface	2 Ethernet Interfaces(WAN/LAN) 1 Power Interface Analog Interfaces (ZX100-A16) 1 E1/T1 Interfaces (ZX100-E1) LED Indicator Power Input : AC 100~240V
Environment Requirement	Temperature: -10 °C -45 °C | Storage temperature: -30 °C -65 °C | Humidity: 10-80% no dew

VoIP Solution



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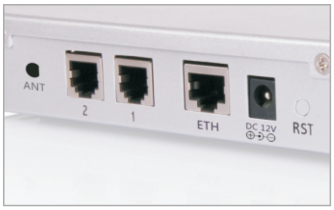
CooVox–U20 Mobile PBX

CooVox-U20 IP Phone system is the second generation of Mini IP Phone system released from ZYCOO, meanwhile it's the first Mobile IP Phone system with the ability to use the GSM or UMTS (3G) module to connect to the mobile network for data transport.

CooVox-U20 is a next-generation smart IP PBX system designed and optimized for today' s business telecommunication needs.



	Model	FXS	FXO
CooVox-U20	A202	0	2
	A211	1	1

GSM or UMTS(3G) is optional for configuration.

Highlights:

-No license fee.

-Fax to Email/Email to Fax.

-Voicemail to Email.

-Monitors and Whispers.

-3G(UMTS) Module allows working in the mobile/remote work sites.

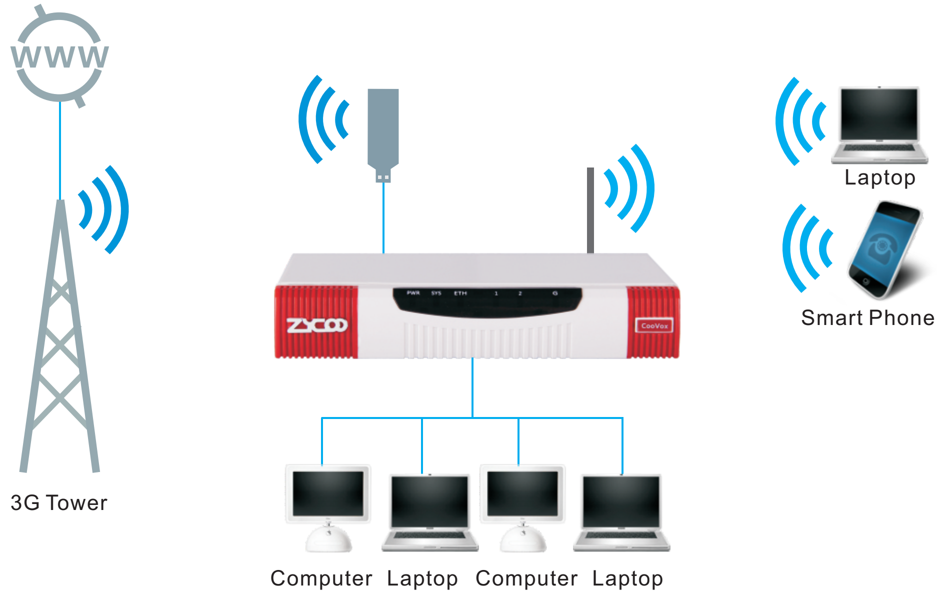
-HD voice codec G.722 for perfect voice quality.

-Strong security features protect your system from hacking(intrusion detection, firewall.)

Specifications

Item	Specification	
PBX Features	Black List Distinctive Ringtone DND/DNIS Feature Codes/Follow Me Ringgroup/Pin Sets T.38 FAX(Pass-through) Time based rule Virtual FAX	BLF (Busy Lamp Field) CDR (Call Detailed Record) DID (Direct Inward Dialing Number) DISA (Direct Inward System Access) IP Phone Provisioning Flash Operator Panel Mobility Extension Smart DID/Speed Dial
System Capacity	30 Extensions | DualCore 500Mhz DSP | 128MB DDR2 | 4GB SD Card (factory default)	
Call Features	Call Back/Call Forward Call Hold/Call Paging and Intercom Call Park/Dial by Name Video Calls	Call Routing/Blind Transfer Attend Transfer/Call Waiting Caller ID/Music on Hold/ Transfer Three-way Conference
Codec & Protocol	Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX Video Codec: H.261/ H.263 / H.263+ / H.264	Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band
Network Features	DDNS Client DHCP Server IEEE802.1Q of VLAN IP Assignment (PPPoE / DHCP / Static) IPv4 / IPv6	SNMP v1/v2 Manual Configuration of Static Route Table Trouble Shooting (Ping, Traceroute) VPN Client (N2N / L2TP/PPTP/OpenVPN) VPN Server(PPTP/L2TP/OpenVPN Server 10)
Hardware Interface	1 Reset Button 1 Power Interface (Output: DC 12V⎓ 1A) 1 Ethernet Interface 2 Analog Ports (FXO/FXS)	1 GSM Antenna Interface 1 UMTS Interface for 3G Data (onboard) LED indicators
Environment	Working Temperature: -10 °C -45 °C | Storage Temperature: -30 °C -65 °C | Humidity: 10-80% No Dew	

Solution



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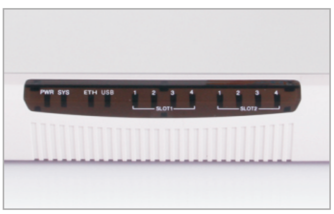
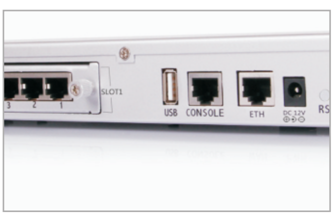
CooVox–U50 IP PBX

CooVox-U50 IP Phone system is the ideal solution for small and medium business with up to 100 extensions. Adopting an innovative modular design, which means it is very convenient to add telephony ports to expand the Phone system. CooVox-U50 supports industry standard SIP trunks as well as analog PSTN trunks(FXO module), mobile GSM trunks(GSM module), analog stations (FXS module), as well as digital trunks (E1/T1 module).





U50 Slot / U50 Module	Slot1	Slot2
4FXS	✓	✓
4FXO	✓	✓
2FXOS	✓	✓
2GSM	✓	✓
4GSM	✓	✓
4BRI	✓	✗
1E1/T1(PRI/R2)	✓	✗

Highlights:

-Modular design; plug and play.

-No license fee.

-Fax to Email/Email to Fax.

-Desktop Call Management.

-Remote Backup service.

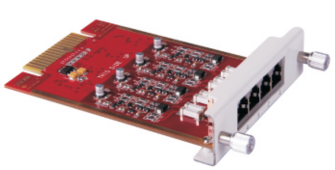
-Disaster Recovery.

-HD voice codec G.722 for perfect voice quality.

Specifications

Item	Specification	
PBX Features	Black List Distinctive Ringtone DND/DNIS Feature Codes/Follow Me Ringgroup/Pin Sets T.38 FAX(Pass-through) Time based rule Virtual FAX	BLF (Busy Lamp Field) CDR (Call Detailed Report) DID (Direct Inward Dialing Number) DISA (Direct Inward System Access) IP Phone Provisioning Flash Operator Panel Mobility Extension Smart DID/Speed Dial
System Capacity	100 Extensions | DualCore 500Mhz DSP | 256MB DDR2 | 4GB SD Card (factory default)	
Call Features	Call Back/Call Forward Call Hold/Call Paging and Intercom Call Park/Dial by Name Video Calls	Call Routing/Blind Transfer Attend Transfer/Call Waiting Caller ID/Music on Hold/ Transfer Three-way Conference
Codec & Protocol	Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX Video Codec: H.261/ H.263 / H.263+ / H.264	Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band
Network Features	DDNS Client DHCP Server IEEE802.1Q of VLAN IP Assignment (PPPoE / DHCP / Static) IPv4 / IPv6	SNMP v1/v2 Manual Configuration of Static Route Table Trouble Shooting (Ping, Traceroute) VPN Client (N2N / L2TP/PPTP/OpenVPN) VPN Server(PPTP/L2TP/OpenVPN Server 10)
Hardware Interface	1 Reset Button 1 Power Interface (Output: DC 12V ⎓ 2A) 1 Ethernet Interface 1 Console Interface 1 USB Interface	1 Hardware Echo Cancellation Interface 1 UMTS Interface for 3G Data (onboard) 1 SLOT 1 for Analog/ GSM/ E1/T1(PRI, R2) /BRI Module Card 1 SLOT 2 for Analog/ GSM Module Card Only
Environment	Working Temperature: -10 °C -45 °C | Storage Temperature: -30 °C -65 °C | Humidity: 10-80% No Dew	

Module Card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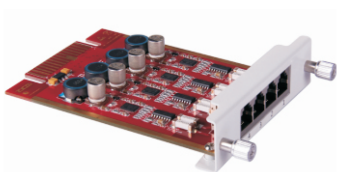
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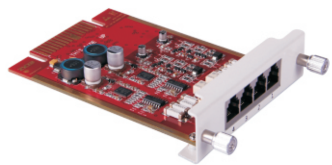
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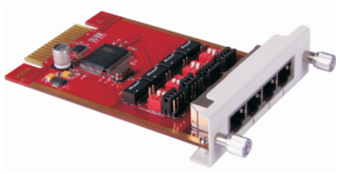
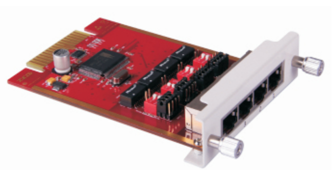

4BRI


2GSM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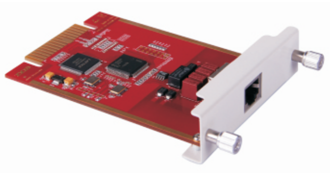
1E1/T1

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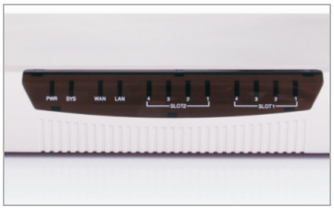
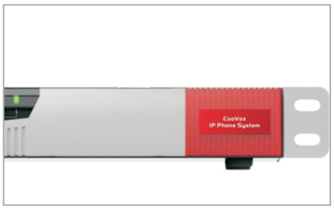
CooVox–U100 IP PBX

The CooVox-U100 IP Phone system is the ideal solution for business with up to 500 extensions and up to 80 concurrent calls. Adopting an innovative modular design, means it is very convenient to add telephony ports to expand the phone system or basic rate interface (BRI module).

CooVox-U100 is a next-generation Hybrid IP PBX system designed and optimized for today' s business telecommunication needs. CooVox-U100 is an affordable, scalable, interoperable, secure and reliable solution which not only provides traditional PBX functions such as automated attendant and voicemail, but also many advanced features including remote extensions, remote office connection, conference bridge, call recording, call detail records (CDR), automatic call distribution (ACD), unified messaging (voice mail to email), and many more features.





U100 Slot / U100 Module	Slot1	Slot2
4FXS	✓	✓
4FXO	✓	✓
2FXOS	✓	✓
2GSM	✓	✓
4GSM	✓	✓
4BRI	✓	✓
1E1/T1(PRI/R2)	✓	✓

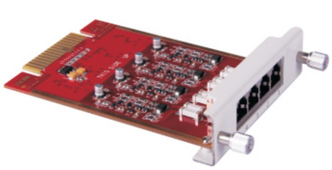
Highlights:

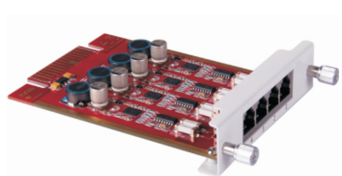
-Modular design; plug and play.
-500 IP Phone Registrations/Extensions and around 80 concurrent call legs.
-No license fee.
-Fax to Email/Email to Fax.
-Intrusion Detection.
-Remote Backup service.
-HD voice codec G.722 for perfect voice quality.
-Easy management through console port.
-USB port for expanded memory or 3G network connectivity.
-Hardware Echo Cacellation for perfect and effective communication. (Optional)

Specifications

Item	Specification	
PBX Features	Black List Conference Room(3) Distinctive Ringtone DND/DNIS Feature Codes/Follow Me Ringgroup/Pin Sets T.38 FAX(Pass-through) Time based rule/Virtual FAX	BLF (Busy Lamp Field)/CDR (Call Detailed Record) DID (Direct Inward Dialing Number) DISA (Direct Inward System Access) IP Phone Provisioning/Flash Operation Panel Mobility Extension SIP Register with UDP/TCP/TLS Smart DID/Speed Dial Spy/SRTP
System Capacity	500 Extensions | DualCore 1.86 Ghz ATOM | 2GB DDR3 | 500GB hard disk or 16GB SSD (factory default)	
Call Features	Call Back/Call Forward Call Hold/Call Paging and Intercom Call Park/Dial by Name Video Calls	Call Routing/Blind Transfer Attend Transfer/Call Waiting Caller ID/Music on Hold/ Transfer Three-way Conference
Codec & Protocol	Audio Codec: G.722/ G.711-Ulaw/ G.711-Alaw/ G.726/ G.729/ GSM/ SPEEX Video Codec: H.261/ H.263 / H.263+ / H.264	Protocol: SIP (RFC3261) , IAX2 DTMF: RFC2833, SIP INFO, In-band
Network Features	DDNS Client DHCP Server IEEE802.1Q of VLAN IP Assignment (PPPoE / DHCP / Static) IPv4 / IPv6	SNMP v1/v2 Manual Configuration of Static Route Table Trouble Shooting (Ping, Traceroute) VPN Client (N2N / L2TP/PPTP/OpenVPN) VPN Server(PPTP/L2TP/OpenVPN Server 10)
Hardware Interface	1 Power Interface(Input: AC 100-240V) 1 Power Switch 2 Ethernet Interfaces 1 VGA Interface 2 Audio Interfaces	2 USB Interfaces (USB2.0) 2 Hardware Echo Cancellation Interface onboard 1 UMTS Interface onboard for 3G Data 1 SLOT 1 for any ZYCOO Module Card 1 SLOT 2 for any ZYCOO Module Card
Environment	Working Temperature: -10 °C -45 °C | Storage Temperature: -30 °C -65 °C | Humidity: 10-80% No Dew	

Module Card


4FXO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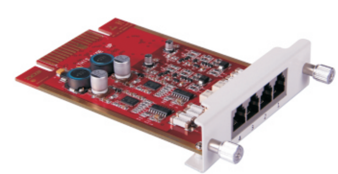
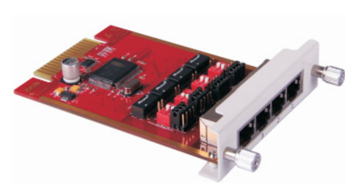

2FXOS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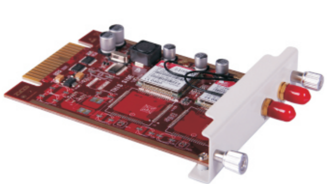
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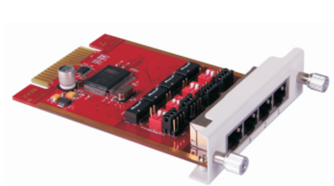

4GSM


1E1/T1

ZYCOO
Zycoo Co.,Ltd
Add : 7F/ B7, Tianfu Software Park, Chengdu, China. 610041 Tel : +86 28 85337096 Fax : +86 28 85337096 ext:800
www.zycoo.com E-Mail : zycoo@zycoo.com

We Focus • We Deliver

CooFone-D30/D30P

CooFone-D30/D30P is entry level IP Phone which is designed for junior employees or middle managers specially for promoting productivity and efficiency in daily working environment.

This IP Phone is provided with reasonable price at great quality and working performance with HD voice, friendly user-interface and auto provisioning. And PoE is taken as another option (D30P) for users in case it’ s necessary.

Highlights:

-High Cost-effective IP Phone with 2 SIP lines.

-HD voice: HD Codec(G.722) , HD Speaker, HD handset.

-D30P supports PoE.

-XML Phonebook, XML Browser.

Power Adapter

Item		CooFone-D30/D30P SIP Phone
Power Adapter (Input/Output)		Input: AC 100-240V Output: DC 5V/1A
Port	WAN	10/100Base-T RJ-45 for WAN
	LAN	10/100Base-T RJ-45 for PC
	Headset	N/A
	EXT	N/A
Power Consumption		Idle:2.5W/Active:2.8W
LCD Size		128X48pixel
Operation Temperature		0-40°C
Relative Humidity		10-65%
Main Chipset		Broadcom
SDRAM		128Mbits
Flash		32Mbits

Inner Box Meas

Inner Box Meas	250mm × 205mm × 60mm
Gross Weight/Unit	0.99kg
CTN Meas	430mm × 325mm × 275mm
Qty/CTN	10 units
Gross Weight/CTN	10kg

Phone Features

-SIP authentication (none, basic, Md5)
-Call forward/ Caller ID
-Call transfer (blind/attended)
-Call holding/waiting
-9 kinds of ring types and 3 user-defined music ring
-Multiple road call waiting in line
-Soft keys/ Function keys programmable
-Phonebook 500 records
-SMS and Speed Dial
-3 way conferencing
-Multi-language realizes localization
-Flexible dial map
-Empty calling no. reject service
-Black list /white list
-No disturb
-Incoming calls /outgoing calls /missing calls(100 records for each)

Advanced Features

-XML phonebook/browser
-Code synchronization via IP PBX/IMS
-Click to dial via web phone book
-Support action URL/active URI
-Voice codec setting for each SIP line
-Hands-free headset ringing choice

IP Phone System Integration

-Conference call park
-Call pickup
-Paging and intercom
-Redial and unredial
-Click to dial
-Secondary dialing automatically
-CLIR (reject the anonymous call)
-CLIP (make a call with anonymous)
-Dial without register

Codec & Voice

-Wideband codec: G.722.1
-Narrowband codec: G.711a/u, G.723.1 high/low, G.729a/b, G.726
-Voice Gain Setting, VAD, CNG
-Echo cancellation: G.168
-Hands-free can support 96ms
-Full duplex hands-free speakerphone

We Focus · We Deliver

CooFone–D60

CooFone-D60 is advanced and feature-rich IP Phone for office business. Programmable Softkeys and function keys allows you to define more features easily and enhance your productivity. With HD codec G722, and echo cancellation, you can enjoy the wonderful voice quality in the constant communication environment.

The reasonable price with default PoE supported, as well as expandable key module(optional), remotable auto provisioning and software upgrading, this IP Phone will provide you the valuable working experience.

Highlights:

-High Cost-effective IP Phone with 4 SIP lines and 6 SIP accounts maximumly.

-HD voice: HD Codec(G.722) , HD Speaker, HD handset.

-4 Softkeys and 8 DSS keys are programmable.

-PoE default.

Power Adapter

Item		CooFone-D60 SIP Phone
Power Adapter (Input/Output)		Input: AC 100-240V Output: DC 5V/1A
Port	WAN	10/100Base-T RJ-45 for WAN
	LAN	10/100Base-T RJ-45 for PC
	Headset	RJ9 Jack
	Handset	RJ9 Jack
	EXT	RJ11 for expansion module
Power Consumption		Idle:2.5W/Active:2.8W
LCD Size		80*43mm/128X48pixel
Operation Temperature		0-40°C
Relative Humidity		10-65%
Main Chipset		Broadcom
SDRAM		128Mbits
Flash		32Mbits

Inner Box Meas

Inner Box Meas	290mm × 260mm × 60mm
Gross Weight/Unit	1.2kg
CTN Meas	540mm × 330mm × 310mm
Qty/CTN	10 units
Gross Weight/CTN	13.15kg

Phone Features
-Call forward/ Caller ID
-Call transfer (blind/attended)
-Call holding/waiting
-Flexible dial map
-Empty calling no. reject service
-Black list/ White list
-BLF/ No disturb/ Speed dial/ SMS
-4 line keys defined as multi line with screen display, or used as SIP line keys
-Softkeys/ Function keys programmable
-8 DSS keys
-IAX2 line key
-SIP authentication (none, basic, Md5)
-Phonebook 500 records
-9 kinds of ring type and 3 user-defined music ring
-Incoming calls /outgoing calls /missing calls(100 records for each)

Advanced Features
-XML phonebook/browser
-Code synchronization via IP PBX/IMS
-Click to dial via web phone book
-Action URL/Active URI
-Voice codec setting for each SIP line
-Hands-free headset ringing choice
-Signal tone parameters customized
-Voice codec setting for each SIP line
-Keypad lock, and emergency call during the keypad lock.
-Ring play via headset or speaker setting
-Code synchronization via IP PBX/IMS
-Click to dial via web phone book/Group listening

IP Phone System Integration
-Conference call park
-Call pickup
-Paging and intercom
-Redial and unredial
-Click to dial
-Secondary dialing automatically
-CLIR (reject the anonymous call)
-CLIP (make a call with anonymous)
-Dial without register

Codec & Voice
-Wideband codec: G.722.1
-Narrowband codec: G.711a/u, G.723.1 high/low, G.729a/b, G.726
-Echo cancellation: G.168
-Compliance in LEC, additional acoustic echo cancellation(AEC) can reach 96ms max

ZYCOO
Zycoo Co.,Ltd
Add : 7F/ B7, Tianfu Software Park, Chengdu, China. 610041 Tel : +86 28 85337096 Fax : +86 28 85337096 ext:800
www.zycoo.com E-Mail : zycoo@zycoo.com

We Focus · We Deliver

CEP-26

CEP-26 Expansion Module has been designed to improve the flexibility and productivity of advanced CooFone-D60 IP Phone.

It adopts 26 programmable DSS keys for strenthening the IP Phone features to meet the requirements of executive and administrative staff who require high call coverage and flexibility from communication systems. Three(3) colors LEDs of each DSS key is helpful to indicate the working status. It allows daisy-chain up to five(5) modules, so you can use 130 programmable keys, call coverage and management is even more easy and efficient to handle.

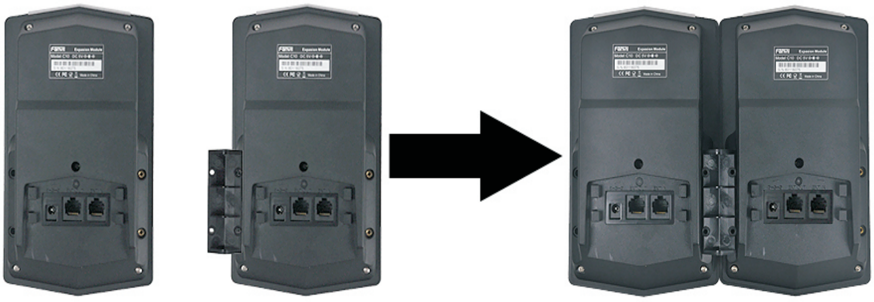
Physical Features

-Power Adapter: Input- AC 100~240V
Output- 5V,1A
-2x RJ11 ports for power/data transport
-Operating Temp.: 0-40℃
-Operating humidity: 10~90%
-Dimensions: 205*125*40mm / 205*125*135mm (with base)
-Net Weight: 320g

Main Features

-26 programmable DSS keys.
-Maximum 5 modules connection, 130 DSS keys.
-3 colors (Red/Green/Orange) LED lights to indicate status.
-Supports for IP-PBX functions, such as BLF/BLA and Intercom.
-Programmable features: speed dial, DND, intercom, call pickup, call hold, call transfer, call forward and Voicemail, etc.
-Powered by host IP Phone when two expansion modules(<2) are connected.

How to connect



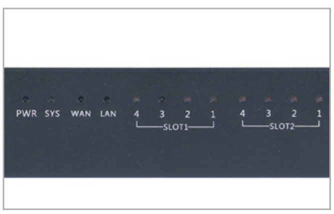
Zycoo Co.,Ltd
Add : 7F/ B7, Tianfu Software Park, Chengdu, China. 610041 Tel : +86 28 85337096 Fax : +86 28 85337096 ext:800
www.zycoo.com E-Mail : zycoo@zycoo.com

We Focus • We Deliver

Asterisk Appliance P2

The Asterisk Appliance P2 is the ideal development platform which is used to deploy telephony system based on Asterisk, Elastix in the SMB(Small and Medium Business) markets.

Adopting modular design, the Asterisk Appliance P2 supports various combination of telephony interfaces with up to two modules, including FXO, FXS, GSM, ISDN BRI, E1/T1. This device is designed for enterprise with up to 500 users.



Highlights:

-Linux operating system and x86 architechture allows direct development.

-Any applications based on Asterisk is available for development, including IVR, Call Center,

-Call Detail Report(CDR), etc.

-Optional 3G(UMTS) module can provide wireless data to enterprise working environment and remote/mobile work sites.

-Easy management through VGA interface

-19", 1U chassis to allow easy rack mounting.

-2 External USB can be used for storage: voicemail, IVR, etc., and emergency system recovery(future).

-Audio line-in/line-out for music source or speaker output.

-2 hardware echo cancellation modules work for smooth communication.

Software

Asterisk Appliance P2 is completely development platform for integrators and developers specially. Tools, utilities and applications of P2 are all openly available and provided as open source code to support your overall solution. Linux-based OS, whatever Debian or CentOS can be installed as you want, as well as the application and related components, that means you can develop it from zero.

Technical Specifications

Processor	ATOM D2550 Dual Core 1.86GHz
RAM	DDRIII 2GB
Storage	500GB 2.5"SATA or 16GB SSD (optional)
Ethernet	WAN/LAN(10/100/1000Mbps)
USB	2 External USB2.0
VGA	1 VGA port
Echo Cancellation	2 Echo cancellation module onboard(optional)
Audio	Audio In/Out port
ANT	3G Access Antenna

Telephony Interfaces

Expansion Telephony Module Quantity	2(Slot1/2,each module for one slot)
Module Type	4FXS,4FXO,2FXOS(2FXO+2FXS),4GSM,2GSM,4BRI,1E1/T1
Capacity	-Up to 8FXS -Up to 8FXO -Up to 8GSM (SIMs) -Up to 2E1/T1 -Up to 8ISDN BRI (16 channels)

Power Requirements

Input Voltage	AC 100-240V 47-63Hz
Max Input Power	81W

Environment Requirements

Working Temperature	-10 °C -45 °C
Storage Temperature	-30 °C -65 °C
Relative Humidity	10-80% No Dew

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Zycoo Co.,Ltd
Add : 7F/B7,Tianfu Software park,Chengdu,China
Tel:+86 28-8533-7096
Fax:+86 28-8533-7096 ext:800
Web : www.zycoo.com
E-mail : zycoo@zycoo.com