Yeastar

Easy Communication

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MyPBX

Hybrid IP-PBX for Your Business

Easy to use

Easy to deploy and manage via Web-based configuration interface

Robust all-in-one features

Deliver enterprise-class communication features and functionality to SMBs

Modular technology

Customizable combination of FXO, FXS, BRI, GSM/CDMA, and UMTS (WCDMA) modules

Match your IP phone

Perfect interoperability with major IP Phones

Speak your language

Multiple languages Web GUI and voice prompts

No future licensing fees

Expand your system to maximum capacity without license fees

COMMUNICATION IS EASY AND EFFICIENT WITH MyPBX SOHO and MyPBX Standard

SOHO

Capacity

- 32 Users
- 15 Concurrent Calls
- Up to 4 FXS Ports
- Up to 4 FXO Ports
- Up to 4 BRI Ports

Rich Features Sets

- IVR
- Queues
- Conferencing
- Voicemail
- And More

Small and Simple

- Compact Design
- Flexible Modular Technology
- User-friendly Web Interface
- Very Low Power Consumption

Standard

Capacity

- 100 Users
- 25 Concurrent Calls
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Ports

All-in-one IP-PBX

- Basic Call Handling
- Advanced IP-PBX Features
- Fully Featured

  No Licensing Fees

Sleek and Sophisticated

- Meet the Highest Standards
- Telephony Interfaces Custom
- Intuitive Web-based Management
- Green with Small Foot Print

EVERYTHING YOU NEED IN IP-PBX IDEAL VOIP SOLUTION

MyPBX U Series

U Series

MyPBX U Series delivers a comprehensive suite of telephony features and functionalities for small and medium sized business. Engineered to support up to 500 users and as many as 80 concurrent calls, MyPBX U Series boasts expandability, flexibility, and affordability.

U100

- 100 Users
- 25 Concurrent Calls
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Channels

U200

- 200 Users
- 50 Concurrent Calls
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Channels

U300

- 300 Users
- 50 Concurrent Calls
- 1 E1/T1/PRI Port
- 2 FXS Ports

U500

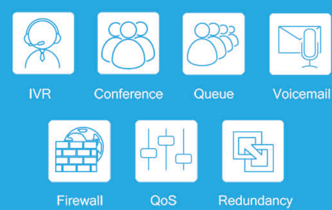
- 500 Users
- 80 Concurrent Calls
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Channels

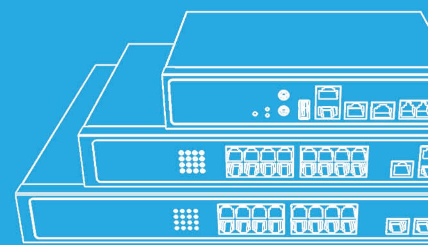
U510

- 500 Users
- 80 Concurrent calls
- 1 E1/T1/PRI Port
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Channels

U520

- 500 Users
- 80 Concurrent calls
- 2 E1/T1/PRI Ports
- Up to 16 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 8 GSM/CDMA/UMTS Channels





Add-ons

Immense Possibilities

Add-ons attend to various additional requirements for MyPBX U Series and add new features or functionality to your installation of U Series.

Call Recording

Call Recording add-on automatically detects every conversation on the phone and records all inbound and outbound calls. Monitor the conversation for various purposes required by your business.

- Record according to call type (inbound, outbound, callback, internal, external), extensions or trunks.
- Manage call recording in the independent management portal.
- Log in the portal to download recording files. Share the files so computers in the same network can visit the files.

Billing System

Billing System can charge and recharge extensions and conduct analysis on the basis of statistics. Both prepaid and postpaid are supported. Flexible rate settings and detailed records make it extremely easy for enterprise to monitor charges, spot misuse, and enhance efficiency.

- Account Management
- Rate Settings
- Call Logs and Statistics

Hotel Module

Hotel Module integrates rich IP-PBX features with professional hospitality features and empowers users to intuitively manage the booking and check-in and check-out of customers, offer mini-bar service, and run personalized billing reports, and more daily operational tasks in hotels.

- Check-in & Check-out
- Booking
- Room Groups
- Set Do Not Disturb
- Wake-up Calls
- Mini Bar
- Billing Report
- Customers List

MyPBX Client

MyPBX Client is an application coordinated with MyPBX U Series that connects your PC and IP phones. you can perform various operations including check extension status, manage contacts, voicemails, and CDR of extensions, send instant messages between extensions, initiate a conference call, and create a call task, etc.

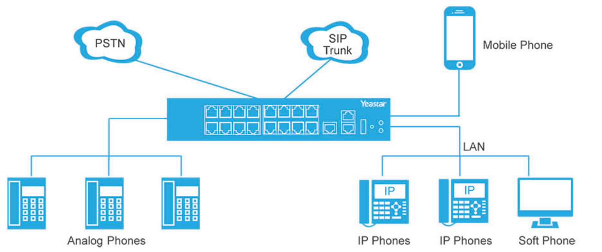
- Presence
- Contacts List
- Instant Messaging
- Click to Call
- Handle Calls
- Call Popup
- Visual Voicemail
- Call Task
- Conferencing
- Call Logs

MyPBX Specifications & Features

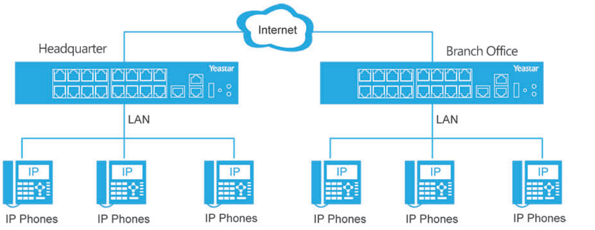
Items		SOHO	Standard	U100	U200	U300	U500/U510/U520
Users		32	100	100	200	300	500
Concurrent Calls (Max)		15	25	25	50	50	80
Voicemail		3000 min	3000 min	Default 3000 min (upgradable)			
Hard Disk		—	—	—	—	—	Support 2.5 inch SATA2
USB		1	1	1	1	1	1
Telephone Interface	Analog Ports (Max)	4	16	16	16	2	16
	GSM/CDMA Ports (Max)	—	8	8	8	—	8
	UMTS Ports (Max)	—	8	8	8	—	8
	BRI Ports (Max)	4	8	8	8	—	8
	E1/T1/J1	—	—	—	—	1	0/1/2
Network Interface	LAN	1(10/100Mbps)					1(10/100/1000Mbps)
	WAN	1(10/100Mbps)					1(10/100/1000Mbps)
VoIP	Protocol	SIP(RFC3261), IAX2					
	Transport Protocol	UDP, TCP, TLS, SRTP					
	Management Protocol	—	—	TR-069	TR-069	TR-069	TR-069
	Audio Codec	G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, SPEEX, ADPCM					
	Video Codec	H.261, H.263, H.263P, H.264, MPEG4					
	DTMF	Inband, RFC2833, SIP INFO					
Network	Connection Type	Static IP, PPPoE, DHCP, OpenVPN					
	Others	Firewall, VLAN, DDNS, QoS, DHCP Server					
Recording	One Touch Record (Manual)	Support	Support	Support	Support	Support	Support
	Call Recording (Auto)	—	—	On demand	On demand	On demand	By default
Redundancy		—	Support	Support	Support	Support	Support
MyTwins		Support	Support	Support	Support	Support	Support
Audio In/Out		—	—	Support	Support	Support	—
Call Back		Support	Support	Support	Support	Support	Support
FAX (T.38)		Support	Support	Support	Support	Support	Support
SMS to Mail/Mail to SMS		—	Support	Support	Support	—	Support
Add-ons		—	—	Support	Support	Support	Support

MyPBX CAN BE TAILORED TO YOUR REQUIREMENTS

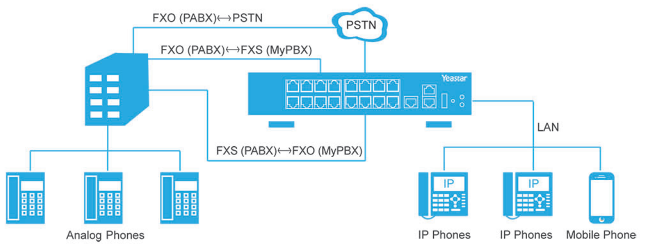
• VOIP SOLUTION FOR SMBS



• MULTI-SITE & BRANCH OFFICES



• COMBINE MYPBX WITH EXISTING TRADITIONAL PBX



VoIP
TA/TG/TE/TB
Gateways
for Enterprises and
Service Providers
BEST GATEWAYS

Analog VoIP Gateways
TA Series

Yeastar TA Analog VoIP Gateways are cutting-edge products that connect legacy telephones, fax machines and PBX systems with IP telephony networks and IP-based PBX systems. Featuring rich functionalities and easy configuration, TA is ideal for small and medium enterprises that wish to integrate a traditional phone system into IP-based system.

Models / Ports / Items	TA400	TA800	TA1600	TA2400	TA3200	TA410	TA810	TA1610
	4 FXS	8 FXS	16 FXS	24 FXS	32 FXS	4 FXO	8 FXO	16 FXO
Protocol	SIP, IAX2							
Transport	UDP, TCP, TLS, SRTP							
Codec	G.711 (alaw/ulaw), G.722, G.723, G.726, G.729A/B, iLBC, GSM, ADPCM							
DTMF	RFC2833, SIP Info, In-band							
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer, VAD, CNG, PLC							
Ethernet	1 10/100BASE-T Ethernet							

Enormous Benefits

Carrier-grade Quality

High-performance Analog VoIP Gateways with powerful features and great reliability. Fully compliant with SIP and IAX2, TA Series also features flexible calling rules and high-quality voice calls. It simplifies the integration of legacy phone system and IP-based system.

Easy and Flexible

An easy-to-use Web GUI allows intuitive configuration; LED present system and port status visually. Desktop and wall-mount installation of TA400/800/410/810 enables flexible deployment; TA1600/2400/3200 boasts 1U form factor.

Trusted Compatibility

Tested and certified with Elastix, BroadSoft. Interoperable with 3CX, Lync Server, Asterisk, FreePBX, Freeswitch, and a wide range of legacy and IP equipment.

VoIP GSM Gateways

GSM/CDMA/UMTS Channels

TG Series

Yeastar TG Series connect GSM/CDMA/UMTS network to VoIP network directly and support two-way communication: GSM/CDMA/UMTS to VoIP and VoIP to GSM/CDMA/UMTS. It is the best solution ever to connect IP-based telephone systems, and softswithces to GSM/CDMA/UMTS network; and also the best fallback solution when landline goes down.

Models / Channels / Items	TG100	TG200	TG400	TG800	TG1600
	1	2	4	8	16
Protocol	SIP, IAX2				
Transport	UDP, TCP, TLS, SRTP				
GSM Frequency	850/900/1800/1900MHz				
CDMA Frequency	800MHz				
UMTS Frequency	850/1900Mhz, 850/2100Mhz, 900/2100Mhz				
Integrated Antenna Splitter (4 in 1)	–	–	Support	Support	Support
Ethernet	1 10/100BASE-T Ethernet				

Varied Applications

Mobile Trunkings

TG adds GSM/CDMA/UMTS trunkings for business making high numbers of calls to mobile networks. It transforms fixed-to-mobile calls to mobile-to-mobile calls. The GSM trunking is also a low-cost alternative to landlines in instances where fixed line communication is not available.

Bulk SMS Service

TG bulk SMS feature is a great tool in conducting mobile marketing. When enterprises need to introduce special offers, manage customer relations, send holiday wishes, etc, they can use bulk SMS to achieve all these with a low cost.

Backup Mobile Trunks

TG can be a perfect backup for IP-PBX that has limited ability in backup solution, enabling your workforce to carry on their business when landline goes down.

E1/T1/PRI VoIP Gateways
TE Series

PRI

VoIP

BRI

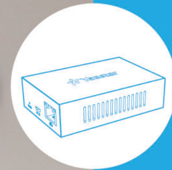
Yeastar TE100/TE200 is a single port or dual port VoIP E1/ T1 /J1 gateway (VoIP to E1/T1/J1, and E1/T1/J1 to VoIP) that supports up to 30 or 60 concurrent calls. Yeastar TE Series offers SMBs cost-effective additions to a legacy telephone system to bring the true benefits of VoIP.

Models / Ports / Items	TE100	TE200
Ports	1 Port (PRI, MFC R2, SS7, E&M)	2 Ports (PRI, MFC R2, SS7, E&M)
Protocol	SIP	
Transport	UDP, TCP, TLS, SRTP	
Codec	G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, ADPCM, Speex	
DTMF	RFC2833, SIP Info, In-band	
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer	
Ethernet	Dual 10/100BASE-T Ethernet	Dual 10/100/1000BASE-T Ethernet

BRI VoIP Gateways
TB Series

Yeastar TB200/400 is a compact and reliable standalone BRI VoIP gateway (BRI-VoIP/VoIP-BRI) offering 2 or 4 BRI ports for companies using ISDN BRI lines an easy, cost-effective and flexible integration into any VoIP system or enabling any IP PBX to be connected to the public ISDN network.

Models / Ports / Items	TB200	TB400
Ports	2 BRI Ports	4 BRI Ports
Protocol	SIP	
Transport	UDP, TCP, TLS, SRTP	
Codec	G.711 (alaw/ulaw), G.723, G.726, G.729A, GSM, iLBC, ADPCM	
DTMF	RFC2833, SIP Info, In-band	
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer	
Ethernet	1 10/100BASE-T Ethernet	



Cost-effective Analog Telephone Adapter

For teleworkers and small business

- Connect analog phones and fax machines to VoIP network
- Protect investment in existing analog telephones
- Migrate to IP voice with the easy-to-install ATA

Analog Telephone Adapter
TA100 & TA200

Yeastar TA100/200 provides 1 or 2 analog interfaces for residential users and small business to convert existing analog equipment to IP-based networks cost effectively. Yeastar TA100/200 is ideal for small business to achieve quick and easy connection in various network environments.

Items / Models	TA100	TA200
Ports	1 FXS	2 FXS
Protocol	SIP	
Transport	UDP, TCP, TLS, SRTP	
Codec	G.711 (alaw/ulaw), G.729A/B	
DTMF	RFC2833, SIP Info, In-band	
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer	
Ethernet	1 10/100BASE-T Ethernet	

VoIP ATA Benefits

Miniature and USB Powered

Small form factor that can save space when placing on the desk. Micro USB port equipped, TA100/200 can be directly connected to and powered by the USB port of PC and IP-PBX.

Rich Calling Features

Connect up to 2 analog phones/faxes and benefit from features like call waiting, call transfer, call conference, hotline, MWI, and more.

High-quality Voice Call

Fully compliant with SIP and support industry standard codecs and line echo cancellation for 32, 64 or 128 ms echo delays for crystal clear voice calls.

Simple Management

Auto Provision by FTP, TFTP, HTTP. Manageable from any Web browser with intuitive Web GUI. Also provide Voice Menu for basic configurations.