Yeastar TA Series Analog VoIP Gateways



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. TA helps them to preserve previous investment on legacy telephone system and reduce communication costs significantly with the true benefits of VoIP.

Benefits

- 4/8/16/24/32 FXS ports or 4/8/16 FXO ports
- Fully compliant with SIP and IAX2
- Flexible calling rules
- Reliable fax performance with T.38
- Provides high-quality voice compression with industry standard codecs

- Line echo cancellation for 8, 16, 32, 64
 or 128 ms echo delays
- Web-based GUI for easy configuration and management
- Excellent interoperability with a wide range of legacy and IP equipment

	TA400/800	TA1600/2400/3200	TA410/810	TA1610	
Interfaces					
Ports	4/8 FXS Ports	16/24/32 FXS Ports	4/8 FXO Ports	16 FXO Ports	
Telephony Interfaces	RJ11	RJ11 and RJ21 50-pin	RJ11	RJ11 and RJ21 50-pin	
		Telco Connectors		Telco Connectors	
Network Interfaces	1 10/100BASE-T Ethernet				
Indicators	Ports Status and Activity LEDs				
Media Processing					
Protocol	SIP (RFC3261), IAX2				
Transport	UDP, TCP, TLS, SRTP				
Codec	G.711 (alaw/ulaw), G.722, G.723, G.726, G.729A/B, GSM, ADPCM				
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer, VAD, CNG, PLC				
DTMF Mode	RFC2833, SIP Info, In-band			T00	
Fax	T.38 and Pass-through		T.30		
QoS	DiffServ, ToS, 802.1 P/Q VLAN tagging				
Telephony Features	2 0 f	Attanded and Dind Transfer			
	3-way Conference, Attended and Blind Transfer, Call Forward, Caller ID Display, Call Waiting,		Programmable AC Impedance, Hangup		
Call Handling			Detection, Answer Detection, Caller ID		
Call Harldling		CDR, Direct Caller ID Calling, Distinctive Ringtone, Do Not Disturb, Flexible Dialing Plan,		Detection, Answer Detection, Caller 15	
		Hotline, MWI, Speed Dial		5000000	
Security/ IP Services					
Firewall	Built-in Firewall, IP Blacklist, Attack Alert				
Network	DHCP, DDNS, OpenVPN, Static Route, VLAN				
Nat Traversal	Static NAT, STUN				
Network Protocol	FTP, TFTP, HTTPS, SSH				
Management					
Protocol	RADIUS, TR-069	SNMP, RADIUS, TR-069	TR-069	TR-069	
Provisioning	Web GUI, Auto Provisioning, Backup and Restore, Firmware Upgrade by HTTP/TFTP				
Configuration	Remote Management, Packet Capture, System Logs,				
	Voice Menu for FXS Gateways Basic Configuration, API				
Connectivity	EVOL OU EVOLUTION OF THE PROPERTY OF THE PROPE				
Signaling	FXS Loop Start, FXS Kewl Start FXO Loop Start, FXO Kewl Start				
Caller ID	BELL202, ETSI (V23), NTT (V23-Japan), and DTMF-based CID				
Disconnect Methods	Busy Tone, Polarity Reversal 3REN, up to 1.5 km on 24 AWG wire —				
Short & Long Haul	3REN, up to 1	.5 km on 24 AVVG wire			
Physical				A O 400 040V	
Power	12V, 1A	AC 100-240V (12V 5A)	12V, 1A	AC 100-240V	
Dimensions				(12V 5.5A)	
$(L \times W \times H) \text{ (mm)}$	200 × 137 × 25	440 × 250 × 44	200 × 137 × 25	440 × 250 × 44	
Operation	0°C to 40°C,	0°C to 45°C,	0001 10		
Temperature	32°F to 104°F	32°F to 113°F	0°C to 40	°C, 32°F to 104°F	
Storage	-20°C to 65°C, -4°F to 149°F				
Temperature					
Humidity	10-90% non-condensing				
Mounting	Desktop,	Desktop,	Desktop,	Desktop,	
-	Wall-mount	Rack-mount	Wall-mount	Rack-mount	
Compatibility Softswitch and	Interenerable with a wide range of IDDDV and astrovitables				
IP-PBX	Interoperable with a wide range of IPPBX and softswitches Certified by BroadSoft				
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Yeastar TG Series

VoIP GSM/3G/4G Gateway for Cost Reduction





Yeastar TG is a series of VoIP GSM/3G/4G gateway connecting GSM or 3G WCDMA or 4G LTE network to VoIP network directly and can support two-way communication: GSM/3G/TG to VoIP or VoIP to GSM/3G/4G. It is the best solution ever to connect IP-based telephone systems, and softswitches to GSM/3G/4G network; and also the best fallback solution when landline goes down.

Benefits



Mobile networks

Link up PBX systems with GSM or 3G WCDMA or 4G LTE networks



SMS

Send and receive SMS messages via Web interface and support bulk SMS



Easy to install

Everything can be easily set up in the Web based management



Cost saving

Save phone bills tremendously with mobile-to-mobile calls and SIP trunks



Back up

Work as a cost-effective backup when the landline network goes down



Easy to integrate

High compatibility with mainstream IP-PBX and softswitch

Interface

- TG100:1 GSM or 3G Channel*
- TG200: 2 GSM or 3G or 4G Channels*
- TG400: 4 GSM or 3G or 4G Channels*
- TG800: 8 GSM or 3G or 4G Channels*
- TG1600: 16 GSM 3G or 4G Channels*
- TG100, TG200, TG400:
 - 1 10/100 Mbps Ethernet Interface
- TG800, TG1600:
 - 2 10/100 Mbps Ethernet Interfaces
- Antenna Splitter (4 in 1 antenna): TG400, TG800, TG1600

Voice Processing

- Protocol: SIP, IAX2
- Transport: UDP, TCP, TLS, SRTP
- Voice Codec: G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, ADPCM, Speex
- Echo Cancellation: ITU-T G.168 LEC
- DTMF Mode: RFC2833, SIP INFO, In-band
- SIP Server for IP Phones Included
- SIP Response Code Switch
- Call Progress Tone Generation
- 1 Stage/2 Stage Dial

System Features

- Mobile Ports Groups
- VoIP Trunk Group
- Incoming /Outgoing Routing rules
- White List and Black List
- Hotline
- AutoCLIP
- Call Duration Limitation
- Call Transfer
- Call Waiting
- Call Back
- Call Detail Record (CDR)
- Call Status Display
- NTP
- IP Blacklist
- Network Attack Alert
- Packet Capture
- System Logs

Mobile Features

- GSM Frequency: 850/900/1800/1900MHz
- WCDMA Frequency: 850/1900MHz, 850/2100MHz, 900/2100MHz
- 4G LTE Band: depends on the module type
- VoLTE if the Carrier Supports
- SMS Sending and Receiving
- Send Bulk SMS
- SMS Center
- Gain Adjustment
- USSD
- PIN Modify
- Carrier Selection: Auto/Manual
- Talk Time Alarm
- Caller ID/CLIR
- Open API for SMS and USSD

Network & Management

- Network Protocol: FTP, TFTP, HTTP, SSH
- NAT Traversal: Static NAT, STUN
- Static Route
- OpenVPN
- Firewall
- VLAN
- QoS/ToS
- DDNS
- · Configure Backup/Restore
- Firmware Upgrade by HTTP/TFTP
- Web based Configuration

Physical

Model	Size (mm)	Power Supply
TG100	110x70x24	DC 12V, 1A
TG200	213x160x44	AC 100-240V
TG400	213x160x44	AC 100-240V
TG800	340x210x44	AC 100-240V
TG1600	440x280x44	AC 100-240V

- Operation Range: 0°C to 40°C, 32°F to 104°F
- Storage Range: -20°C to 65°C, -4°F to 149°F
- Humidity: 10-90% non-condensing

^{*}Depends on which type of network you order.

Yeastar TE Series E1/T1/BRI VoIP Gateway



Yeastar TE100/200 offers SMBs cost effective additions to legacy telephone systems to bring the true benefits of VoIP. TE100/200 is a single 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. It's designed to bridge the gap between E1/T1/J1 and VoIP networks. Integrating TE100/200 into an existing network will allow inexpensive communication via SIP trunking. Also, it could connect VoIP systems with E1/T1/J1 service from legacy carriers.

Benefits

- Software configurable E1/T1/J1 ports
- Support PRI, MFC R2, SS7, E&M
- Easy-to-navigate GUI
- Cost-effective call routing
- Simple installation and management

- Interoperable with Elastix and Skype for Business
- Compatible with various ISDN PBX, IP-PBX,
 and softswitch
- Low power consumption for your green office

Physical Interfaces

- E1/T1/J1 port: 1 or 2
- Dual Base-T connectors
- TE100: 10/100 Mbps
- TE200: 10/100/1000 Mbps
- USB port:1

Voice Processing

- Protocol: SIP(RFC3261)
- DTMF: RFC2833, SIP Info, In-band.
- Codec: G711a, G711u, G.722, G.726, G.729a, GSM, Speex, ADPCM
- Transport: UDP, TCP, TLS, SRTP
- Echo Cancellation: ITU-T G.168 LEC

Telephony

- Blacklist
- · Call detail records
- Call duration limitation
- Caller ID
- Caller ID prefix
- Call routing rules
- SIP trunk support
- SIP peer mode
- SIP Registrar for IP phones
- T.38 fax
- Trunk group

Signalling

- Support PRI, MFC R2, SS7, E&M
- PRI switch types: Era ISDN, nation, Q.SIG
- CAS: MFC R2 (Argentina, Brazil, China, Czech Republic, Colombia, Ecuador, Indonesia, ITU, Mexico, Philippines, Venezuela)
- SS7: ITU, ANSI, China

Internet

- DDNS
- IP Blacklist
- NAT Traversal
- QoS/ToS
- PPPoE
- Static Route

System

- Backup and restore
- Custom prompts
- E1/T1/J1 diagnostics
- Firmware upgrade by HTTP/TFTP
- Packet Capture Tool
- System logs
- Security: Firewall, TLS, SRTP, HTTPS
- Web based configuration

Applications

- Connect legacy PBX to VoIP services
- Connect IP-PBX to ISDN trunkings
- ISDN connectivity for Contact Center
- Connect legacy PBX to IP-PBX at the branch office
- Interconnect multiple PABX to realize free inter-office calls

Physical

- Dimensions: 213x160x44 mm
- Power Supply: AC 100-240V (12V 2A)
- Operation Range: 0°C to 40°C, 32°F to 104°F
- Storage Range: 20°C to 65°C, -4°F to
- Humidity: 10-90% non-condensing