Grandstream
Innovative IP Voice & Video

GXE502X All-In-1 IP-PBX

GXE502x IP-PBX appliance is the next generation in converged business communication systems. It's a powerful all-in-one voice + fax + video + data communication solution for the small to medium sized business (SMB), especially companies with sub-30 seat per location. Designed from ground up to support distributed IP communications, intelligent unified messaging, advanced application integration, and popular PBX features, the GXE502x product family also optimally integrates legacy PSTN trunk and telephone interfaces for fail-safe hybrid communication needs in all circumstances, including situations of power or network loss. The GXE502x IP-PBX raises the level of modern business communication systems for SMB to a new height of innovation, quality, reliability, feature integration, user productivity, ease of deployment & administration, and affordability.

Key Feature Highlights

- Integrated high performance data router with advanced voice/video QoS support
- Integrated legacy PSTN trunks, analog phone/FAX ports, & unlimited SIP trunk options
- Integrated session border controller (SBC) for NAT/firewall traversal and secure telecommuting
- Integrated conference bridges that allow any combination of IP or PSTN calls using any codecs (built-in transcoding)
- Unified messaging including voicemail-to-email, fax-to-email, and video-to-email (pending)
- Power and network failure survivability and recovery; Integrated PoE (802.3af)
- Support true & local emergency call routing in all circumstances
- Automated detection and provisioning of IP phones, video phones, ATA and other endpoints for easy deployment
- Rich PBX features such as presence, shared line appearance, call park & pickup, call queue, ACD, intercom & paging, ring group, customizable auto-attendant & IVR, personal music-on-hold, branch office system peering
- Hardware accelerated encryption engine to ensure strongest security protection using SRTP and TLS
- Personal Web portal to manage individual phone/call setting, personal greeting, new or saved voice/fax/video messages for each extension user
- Flexible dial plan, call routing and call recording (pending)

GXE502X Technical Specifications

Network Interfaces
- 1xWAN and 1xLAN 10M/100M auto-sending Ethernet ports (RJ45), with integrated PoE(802.3af) on the LAN port

PSTN Trunk Line, Phone/Fax & Peripheral Interfaces
- 4 (GXE5024) or 8 (GXE5028) FXO ports and 2xFXS ports (RJ11)
- 3 REN, Bellcore Type 1 & 2, ETSI, BT, NTT DTMF based caller ID, Polarity reversal/wink supported
- 1xAudio Input, 1xAudio Output, 1xUSB

LED Indicators & Universal Power Supply
- LED: Power, Ready, USB, VoIP, WAN, LAN, Line, Phone
- Power: 100V-240V 50-60Hz input; 12VDC 1.25A output

Dimension, Weight, and Environmental
- 265 mm X 175mm X 40.5 mm; 0.75kg
- Operational: 32-104°F or 0-40°C; Storage: 14-140°F or -10-60°C
- Humidity: 10- 90%, non-condensing

Compliance
- FCC Part 68 & 15B; CE: EN55022, EN55024, TBR21, EN60950;
- C-Tick: AS/NZS CISPR22, CISPR24; A-Tick: AS-ACIF S002, AS/NZS60950; UL (power supply)

Power & Network Loss Survivability and Emergency Call
- 2 PSTN fail-over life lines in case of power outage;
- IP network disruption fail-over to PSTN line and recovery
- Support true & local emergency (911) call routing and E911 over IP

Communication & Security Protocols
- TCP/UDP/IP, RTP/RTCP, ICMP, ARP/RARP, DNS, DDNS, DHCP, NTP, TFTP, TELNET, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS/SIPS

Voice/Video/Fax-over-IP Capabilities & QoS
- G.711, G.723.1, G.729A/B/E, G.726, iLBC, H.264, H.263/H.263+
- Carrier grade G.168 compliant line echo cancellation, T.38 Group 3 fax relay, multiple DTMF methods (in-audio, RFC2833, SIP INFO)
- Layer 2 (802.1p/Q) and layer 3 (DiffServ, ToS) QoS

System Management & Endpoint Provisioning
- HTTP/HTTPS, TELNET, Syslog, TR-069 (pending)
- Auto provisioning of Grandstream IP phone & ATA, plug-n-play

System Capacity, Performance & Expandability
- Integrated SIP proxy server that supports up to 100 user extensions and 50+ simultaneous calls
- 512MByte NAND Flash storage that allows up to 150 hours of voicemail, 10,000 pages of fax messages, & 4 hours of video mail
- Support unlimited external PSTN trunk or FXS gateways
- Support unlimited SIP trunk accounts
- Support unlimited peering with remote offices

Conference Bridge
- 2 (GXE5024) or 4 (GXE5028) password protected conference bridges allowing up to 12 (GXE5024) or 20 (GXE5028) simultaneous participants from either PSTN trunk or IP using any voice codec

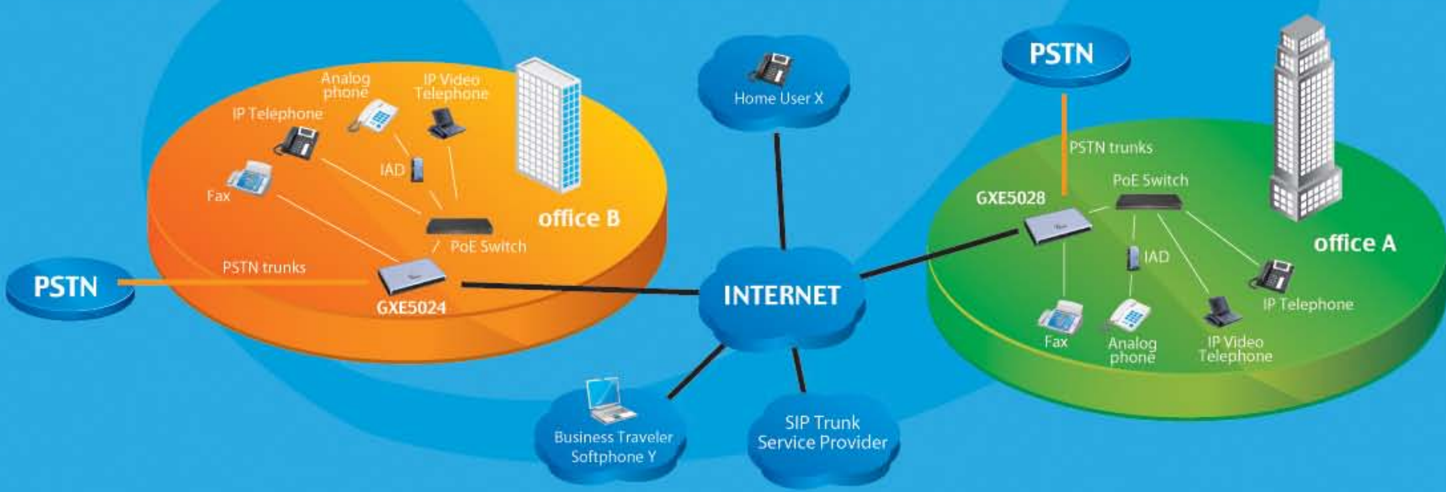
Unified Messaging
- Voicemail-to-email (.wav), fax-to-email (PDF), and video-to-email (pending)
- Companion PC tool to allow print-to-fax from any Windows applications
- Personal Web portal to manage phone/call setting, personal greeting, new or saved voice/fax/video messages

Auto-Attendant
- Support unlimited configurable interactive voice responses (IVR) applications with easy-to-use interface and multi-language voice prompt
- Flexible time and condition control to allow virtually unlimited playing modes (business hour, lunch or night mode, holiday, etc)

Router, NAT/Firewall Traversal and Telecommuting
- Integrated high performance NAT router
- Integrated Session Border Controller (SBC) that supports automated NAT traversal for remote endpoints from behind NAT/firewall
- Support dynamic DNS to allow telecommuting & peering without reliance on fixed IP address

Key Telephony Features
- Call park/pick-up, transfer, hold, shared line appearance, presence, hunt/ring group, intercom & paging, personalized music-on-hold, flexible dial plan and call routing control, call recording

Call Center & Other Applications Support
- Configurable call queues, automatic call distribution (ACD) based on agent skills/availability/busy level, in-queue announcement
- Integration with CRM and other 3rd party applications (pending)



VoIPon www.voipon.co.uk sales@voipon.co.uk Tel: +44 (0)1245 808195 Fax: +44 (0)1245 808299