DATASHEET

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Grandstream Voice Gateway

Grandstream Voice Gateway

- T1/E1/J1 interface
- Two PSTN Trunk FX0 ports. Two analog FXS ports
- Gigabit network ports with PoE+; Integrated NAT router
- Zero configuration provisioning of endpoints
- SRTP, TLS and HTTPS encryption
- Built-in Call recording server; recordings accessed via web user interface
- LDAP and XML phonebooks, flexible dial plan
- H.264, H.263 and H.263+ codecs
- Voicemail and fax forwarding to email

Description

The Grandstream Voice Gateway creates an easily manageable on premise anchor to your communications network, and it is designed to bring leading edge voice, video, data, and mobility features to enterprises, small and medium businesses, retail and residential environments in an easy-to-manage fashion. This enterprise-grade IP PBX comes equipped with a suite of advanced call handling and network data features, all with no licensing and no fees. Its scalability offers deployments that can support up to 2000 users, E1, T1 and J1.

The Grandstream Voice Gateway allows businesses to unify multiple communication technologies, such as voice, video, surveillance, data tools, and facilities access management into one common platform that can be managed and accessed remotely. With features such as customizable call-routing, multi-level IVRs, call queues, auto-attendant, call detail records, multi-site peering, SIP video support, voicemail/fax forwarding to email and more, the Vingtor-Stentofon PBX-6510 delivers complete unified communications.

Specifications

GENERAL

Power Supply	Input: 100 ~ 240VAC, 50/60Hz; Output: DC+12V, 1.5A
Weight	2.165 kg
Dimensions	440mm(L) x 185mm(W) x 44mm(H)
Environmental	Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing); Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing)
Mounting	Rack mount & Desktop
Analog Telephone FXS Ports	2 RJ11 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	2 RJ11 ports (both with lifeline capability in case of power outage)
T1/E1/J1 Interface	1 RJ45 port
Network Interfaces	Dual Gigabit ports (switched or routed) with PoE+
NAT Router	Yes (user configurable)
Peripheral Ports	USB, SD
LED Indicators	Power 1/2, PoE, USB, SD, T1/E1/J1, FXS 1/2, FXO 1/2, LAN, WAN
LCD Display	128x32 dot matrix graphic LCD with DOWN and OK buttons
Reset Switch	Yes, long press for factory reset and short press for reboot
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, iLBC, GSM,

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	AAL2-G.726-32, ADPCM; T.38
Video Codecs	H.264, H.263, H263+
QoS	Layer 3 QoS, Layer 2 QoS
DTMF Methods	In Audio, RFC2833, and SIP INFO
Digital Signaling	TPRI, SS7, MFC/R2, RBS (pending)
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE mDNS), eventlist between local and remote trunk
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending)
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone
Media Encryption	SRTP, TLS, HTTPS, SSH
Advanced Defense	Fail2ban, alert events, Whitelist, Blacklist, strong password based access control
Multi-Language Support	English/Simplified Chinese/Traditional Chinese/Spanish/French/Portuguese/German/Russian/Italian/ Polish/Czech for Web UI; Customizable IVR/voice prompts for English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic; Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT
	Yes. with enable/disable

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Polarity Reversal/Wink	option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ work-load, in-queue announcement
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response)
Maximum Call Capacity	Up to 2000 registered SIP endpoints, up to 200 concurrent calls
Conference Bridges	Up to 8 bridges, up to 64 simultaneous conference attendees
Call Features	Call park, call forward, call transfer, DND, DISA, ring group, pickup group, blacklist, paging/intercom etc.
Compliance	FCC: Part 15 (CFR 47) Class B, Part 68; CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, TBR21, RoHS; RCM: AS/NZS CISPR 22, AS/NZS CISPR 24, AS/NZS 60950, AS/ACIF S002; ITU-T K.21 (Basic Level); UL 60950 (power adapter); T1: TIA-968-B Section 5.2.4; E1: TBR4/TBR12/TBR13, E1: AS/ACIF

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