DATASHEET

GX-7100/GX-7500

Nuera Session Border Controllers

Overview

Nuera GX-7100/GX-7500 Session Border Controllers (SBCs) are cutting-edge, mid-capacity solutions designed for enterprises and service providers. They ensure service assurance, robust security, and reliable connectivity across diverse VoIP networks. These SBCs seamlessly connect UC Platforms and Voice Services to any SIP trunking provider, delivering exceptional performance in any SIP-to-SIP environment.

The GX-7100 and GX-7500 SBCs are ideal for service providers and large organizations, including contact centers, private cloud centers, hosted services, and government institutions, where security, reliability, and high performance are critical.

Up to 10,000 SBC Sessions | Pure IP SBC | 1+1 High Availability | Certified SBC for Teams Direct Routing

Comprehensive interoperability
Proven interoperability with SIP trunks, UC and CC platforms and IP cloud services

Flexible licensing
Various licensing options for easy and cost-effective scalability regardless of enterprise size

Enhanced security
Comprehensive perimeter defense against cyber, DoS, and DDoS attacks, eavesdropping, fraud, and service theft

Superior voice quality
Ensure crystal-clear voice communications with advanced optimization and monitoring capabilities

High resiliency
Maintain uninterrupted service with 1+1 high availability and local branch survivability

GX-7100/GX-7500 SPECIFICATIONS

Capacities	GX-7100	GX-7500
Signaling Sessions	1,000	10,000
Registered Users	10,000	50,000
Transcoding	1,000	1,000
Media Sessions	1,000	10,000
RTP/SRTP Sessions	1,000	10,000
Network Interfaces		
Ethernet Ports Configuration	8 x 100/1000/2500 Base-T RJ45 ports (Copper) 4 x 10G Base-X SFP+ ports (Fiber)	
Security		
Access Control	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)	
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching	
Encryption/Authentication	TLS, DTLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest	
Privacy	Automatic topology hiding, user privacy	
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces	
Authentication	Microsoft Azure Active Directory (Azure AD) , LDAP	
STIR/SHAEKN	STIR/SHAKEN support. Interworking with STI-AS/VS	
Interoperability		
SIP B2BUA	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode	
SIP Interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer and more	
Registration and Authentication	SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication	
Transport Mediation	Mediation between SIP over UDP/TCP/TLS/WebSocket/SCTP, IPv4/IPv6, RTP/SRTP (SDES/DTLS)	
Header Manipulation	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions	
Number Manipulations	Ingress and egress digit manipulation	
Transcoding and Vocoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729A/B, GSM-FR, AMR-NB, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB	
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion	
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, DTLS, RTP multiplexing, secure RTCP with feedback	
NAT	Local and far-end NAT traversal for support of remote workers, ICE full and lite support (RFC 8445)	
Voice Quality and SLA		
Call Admission Control	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions	
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS	
Standalone Survivability	Maintains local calls in the event of WAN failure.	
Voice Monitoring and Enhancement	Transrating, RTCP-XR, acoustic echo cancellation, replacing voice profile due to impairment detection, fixed and dynamic voice gain control, packet loss concealment, dynamic programmable jitter buffer, silence suppression/comfort noise generation, RTP redundancy, broken connection detection	
Direct Media	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption	
High Availability	SBC high availability with two-box redundancy, active calls preserved	
Test Agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs	
SIP Call Handling		
Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth	
Querying External Databases	Destinations based on customized queries of ENUM, LDAP, HTTP server (REST API)	
Available Destinations	Configured SIP peers, registered users, IP address, request URI	
SBC Media Types	Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)	
Recording Solutions	Lawful Interception (LI)[1], SIPREC for both audio and video sessions	
Management		
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC), Session Detail Records (SDRs)	
Multi Tenancy	Advanced multi-tenant SBC partitioning	
Comprehensive System Management	Integrated Baseboard Management Controller (BMC) for advanced out-of-band monitoring, maintenance and control	
Physical/Environmental		
Dimensions (LxWxH) (mm)	436.2 X 454.3 X 44.2	
Mounting	19” mount; Rack Height 1U	
Power	320W Redundant/non-redundant PSUs	
Operating Temperature	0°C ~ 40°C	

[1] Requires a dedicated software build

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