The NEC UNIVERGE BX500 Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The BX500 connects IP-PBXs to any SIP trunking service provider, scaling up to 60 concurrent SBC sessions. It offers superior performance in connecting any SIP to SIP environment, legacy TDM-based PBX systems to IP networks, and IP-PBXs to the PSTN.

Vast mediation capabilities and proven interoperability
The BX500 supports a wide range of voice coders and is capable of providing SIP normalization, fax handling, gain control and numerous additional media processing features. It offers certified interoperability with leading unified communications solutions and SIP trunking providers.

Security
The BX500 provides robust protection for the IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability
The BX500 offers and maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities (including PSTN fallback with E911) result in minimum communications downtime.

Applications
- SIP trunking
- Hosted PBX & UC as a Service
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems

www.nec-enterprise.com
Specifications

**Capacities**
- **Max. Signaling/Media Sessions**: 60
- **Max. Registered Users**: 200
- **Max. SRTP/RTP Sessions**: 60

**Telephony Interfaces**
- **Analog**: Up to 4 FXS ports
- **Digital**: 1-4 BRI ports, network S/T interfaces, NT or TE termination
- **Clock source**: 5 ppm High Precision

**Network Interfaces**
- **Ethernet**: 4GE interfaces configured in 1+1 redundancy or as individual ports

**Security**
- **Access Control**: DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting
- **VoIP Firewall**: RTSP pinhole management, rogue RTSP detection and prevention, SIP message policy, advanced RTSP latching
- **Encryption/Authentication**: TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
- **Privacy**: Topology hiding, user privacy
- **Traffic Separation**: VLAN/physical interface separation for multiple media, control and OAMP interfaces
- **Introspection Detection**: Detection and prevention of VoIP attacks, theft of service and unauthorized access

**Interoperability**
- **SIP B2BUA**: Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
- **SIP Interworking**: 3xx redirect, REFER, PRAK, session timer, early media, call hold, delayed offer
- **Registration and Authentication**: User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
- **Transport Mediation**: SIP over UDP/TCP/TLS, IPv4/IPv6, RTSP/RTSP (SDES)
- **Message Manipulation**: Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
- **URI and Name Manipulations**: URI user and host name manipulations, ingress and egress digit manipulation
- **Vocoders**: Coder normalization including coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB, AMR-WB (G722.2), SILK-NB/WB, Opus-NB/WB
- **Signal Conversion**: DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion

**Voice Quality and SLA**
- **Call Admission Control**: Based on bandwidth, session establishment rate, number of connections/registrations
- **Packet marking**: 802.1p/Q VLAN tagging, DiffServ, TOS
- **Standalone Survivability**: Maintains local calls in the event of WAN failure
- **Impairment Mitigation**: Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort, Noise Generation, RTP redundancy, broken connection detection
- **Voice Enhancement**: Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control
- **Direct Media (No Media Anchoring)**: Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
- **Voice Quality Monitoring**: RTCP-XR
- **Quality of Experience**: Access control and media quality enhancements based on QoE and bandwidth utilization
- **Test agent**: Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs

**SIP Routing**
- **Routing Methods**: Request URL, IP address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API
- **Advanced Routing Criteria**: QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
- **Routing Features**: Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization
- **SIPRec**: IETF standard SIP recording interface

**OAM&P**
- **OMM&I**: Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS

**Physical / Environmental**
- **Dimensions**: 51 x 296 x 160 mm (2 x 11.65 x 6.3 in.) (HxWxD)
- **Weight**: 670 g

**About NEC Corporation** - NEC Corporation is a leader in the integration of IT and network technologies that benefit businesses and people around the world. By providing a combination of products and solutions that cross utilize the company’s experience and global resources, NEC’s advanced technologies meet the complex and ever-changing needs of its customers. NEC brings more than 100 years of expertise in technological innovation to empower people, businesses and society. For more information, visit NEC at http://www.nec.com

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