Dialogic® BorderNet™ 500 Gateways

Datasheet

Dialogic® BorderNet™ 500 Gateways are turnkey appliances that can enable the rapid deployment of new SIP-based communications services to enterprise customers by providing a flexible means to deliver SIP services from public IP networks to private enterprise IP networks and their resident communications systems.

BorderNet 500 Gateways supply any-to-any connectivity and call routing for connection to SIP trunks or PSTN trunks and virtually any on-premise PBX, including IP-PBXs, hybrid PBXs, and legacy TDM PBXs, along with integrated enterprise Session Border Control (SBC) features. SBC features include Network Address Translation (NAT) traversal, network-edge security, and a wide variety of SIP controls for interoperability. By defining a distinct and secure demarcation point or border for SIP services between public and private networks, the SIP service can become both more manageable and reliable.

Features	Benefits
Any-to-any connectivity and call routing	Provides flexibility in connecting to a wide variety of services and equipment, including SIP trunks, PSTN trunks, and legacy, hybrid, and IP PBXs
Extensive interoperability testing with SIP service providers and PBX manufacturers	Delivers a high degree of confidence that BorderNet 500 Gateways will work effectively with a wide variety of vendor interfaces and equipment
Robust SIP security features	Creates a secure demarcation point for an enterprise at the network edge to fend off malicious outside threats
Built-In SIP Proxy to enable firewall and NAT traversal	Allows an enterprise to connect to a SIP trunk or SIP service
Detailed call quality statistics	Enhances the ability to troubleshoot voice quality issues
Optional software modules	Allows an enterprise to tailor its network edge solution to user needs with added QoS, enhanced security, remote access, and primary SIP endpoint registration
T.38 Fax over IP (FoIP) at V.34 speeds	Includes high speed, reliable FoIP that reduces expenses by decreasing the time needed to transmit/receive fax messages

Benefits from Security Technology

With BorderNet 500 Gateways, businesses can benefit from the productivity and cost-savings of VoIP and other IP-based communications technology without scrapping their previous investment in security technology.

BorderNet 500 Gateways work seamlessly with existing firewalls to allow SIP traffic through the enterprise network edge. While traditional firewalls normally block SIP traffic — including mission-critical applications such as VoIP — BorderNet 500 Gateways resolve this problem by working with existing security solutions.

BorderNet 500 Gateways also provide a wide variety of SIP security features, including Back-to-Back User Agent (B2BUA) network elements, deep packet inspection, NAT traversal, SRTP, TLS, and HTTPS. These features can provide a secure demarcation point for the premise's network edge, nullifying threats from denial of service, SPAM, and SPIT attacks and other SIP security concerns.

Reduced Costs with SIP Trunks

An increasing number of service providers are offering SIP trunks, which deliver voice connections or "sessions" over a broadband internet connection. For some enterprises, SIP trunks allow exceptional cost savings since these enterprises can eliminate PSTN trunks completely for standard connectivity. When a more resilient environment is needed, enterprises can benefit from significant savings by reducing the number of PSTN trunks and only using them for failover routing in case of SIP trunk failure.

SIP trunk connectivity requires SIP and data traffic to traverse the enterprise firewall. Using a built-in SIP proxy, BorderNet 500 Gateways enable firewall and NAT traversal for SIP trunking and can deliver advanced security for SIP communications, including those entering the enterprise via a SIP trunk.

BorderNet 500 Gateways can also streamline compatibility between an on-premise PBX and internet telephony services, allowing customers to enjoy the benefits of SIP trunking and/or hosted SIP services without expensive forklift or software upgrades to existing on-premise telephony equipment.

Tested for Interoperability

Many enterprises today continue to use legacy PBX equipment, but also want to cut communications costs and improve productivity by implementing a VoIP solution. Because these enterprises have very diverse PBX equipment, solution providers need an easily managed gateway that can handle a wide range of legacy equipment.

The gateway function within BorderNet 500 Gateways has been tested for interoperability with legacy PBXs from leading vendors such as Alcatel, Avaya, Mitel, NEC, Nortel, and Siemens. Additionally, their SBC function has been tested with service providers such as AT&T, Level 3, Broadvox, and many others. The rigorous interoperability testing performed with BorderNet 500 Gateways allows solution providers to focus on customer application needs rather than integration efforts.

BorderNet 500 Gateways also allow the customization of SIP protocol messages through header manipulation, advanced routing capabilities, and B2BUA. This permits the connection of disparate SIP streams from different vendors, enabling interoperability between equipment that would otherwise not be able to communicate with each other.

Handling Fax

Fax is often overlooked when gateways are considered, especially when a SIP trunk deployment is planned. Depending on the model chosen, BorderNet 500 Gateways provide various types of fax and Fax over IP (FoIP) functionality, including up to 120 channels of T.38 FoIP at V.34 (33.6 kbps) speed.

In the future, Dialogic plans to support T.38 fax on all BorderNet 500 Gateways. Fax functionality supported as of November 2010 is described in Table 1.

Model	Fax Functionality Supported
BN500IP	G.711 and T.38 passthrough
BN508BRI	G.711 and T.38 passthrough; translation between TDM T.30 fax and T.38 FoIP; full density V.34
BN501PRI	G.711 passthrough (T.38 support is planned for a future release)
BN504PRI	G.711 and T.38 passthrough; translation between TDM T.30 fax and T.38 FoIP; full density V.17 (full density V.34 is available via a separately purchased fax license)

Table 1. Current Fax Functionality Supported on Dialogic® BorderNet™ 500 Gateways

Providing Flexibility with Add-on Modules

Dialogic offers several add-on software modules that allow BorderNet 500 Gateways to address specific needs. These modules include:

- **Quality of Service (QoS) Module** — Sets priorities for different types of data and allocates bandwidth for various purposes (for example, to give priority to VoIP)
- **Remote SIP Connectivity Module** — Extends SIP capabilities to employees working remotely; manages remote NAT traversal from a central firewall, and includes a STUN (Simple Traversal of UDP through NAT) server that allows NAT clients (for example, a computer behind a firewall) to set up phone calls to a VoIP provider hosted outside of the local network
- **VoIP Survival Module** — Enables redundancy in a SIP-based IP-PBX environment for secure hosted VoIP services
- **Enhanced Security Module** — Provides encryption and intrusion detection and prevention for SIP
- **SIP Registrar Module** — Allows SIP endpoint registration on BorderNet 500 Gateways for survivability routing

Built with Robust Building Blocks

BorderNet 500 Gateways contain field-proven Dialogic® Diva® Media Boards and Dialogic® Diva® SIPcontrol™ Software, in addition to Ingate SIParator software. These elements are integrated in a rugged, compact 1U rack-mount server chassis.

BorderNet 500 Gateways also offer four-Gigabit Ethernet interfaces for IP network connectivity, as well as a variety of PSTN and PBX interface options, including ISDN BRI and E1/T1 interfaces.

In addition, BorderNet 500 Gateways can be configured to support up to 120 TDM channels or 150 SIP-to-SIP sessions or a combination.

Technical Specifications

Ports per unit	Dialogic® BorderNet™ 500IP Gateway: 25 SIP-to-SIP sessions (no TDM interfaces) Dialogic® BorderNet™ 508BRI Gateway: 4-port ISDN BRI (8 channels) Dialogic® BorderNet™ 501PRI Gateway: 1-span T1/E1 (24/30 channels) Dialogic® BorderNet™ 504PRI Gateway: 4-span T1/E1 (96/120 channels)
Server Type	Nexcom NSA 3110
Processor	E1500 Celeron, 2.2 GHz
Memory	1GB RAM1066-DIMM DDR3
Hard disk subsystem	Hitachi 500GB (24X7 rated)
Network interface	4x 10/100/1000 Base-T Ethernet ports
Protocol support	ISDN BRI: DSS1 (Euro-ISDN), NI-1, 5ESS, 1TR6, INS Net 64, VN3, CT1, QSIG E1 ISDN: ETSI-DSS1 (EuroISDN), INS-1500 (Japan), QSIG E1 CAS: MFR2 T1 ISDN: NI-1, 4ESS, 5ESS, DMS100, QSIG T1 CAS: RBS
VoIP services	SIP methods: ACK, BYE, INVITE, NOTIFY, REFER, CANCEL, OPTIONS, REGISTER Configurable IP transport layer UDP or TCP Number normalization and manipulation of Called/Calling/Redirected Number Call Routing based on Called/Calling/Redirected Number, PSTN Interface, and/or SIP Peer Call Hold/Retrieve (for example, Re-Invite mapping towards ISDN) PSTN-side Call Transfer (REFER points to PSTN) Call Diversion Message Waiting Activation/Deactivation Call Redirection via 302 Moved Temporarily Simplified Number Normalization based on PSTN connection parameters Number Manipulation using Regular Expressions Configurable Cause Code Mapping Clear Channel Fax Clear Channel Modem
FoIP (T.38) services	T.30 Fax Group 3 up to 33.6 kbps using T.38 real-time FoIP Fax compression MH, MR, MMR Error Correction Mode (ECM)

Technical Specifications *(continued)*

Additional SIP features	SIP Proxy and Registrar
SIP Connect Compliant Security	TLS and SSL authentication SRTP (Secure Real-time Transport Protocol) SIPS (Secure SIP) Supported ciphers: DH, ADH, AES (128-256 bits), 3DES (64 bits), DES (64 bits), RC4 (64 bytes), RC4 (256 bytes), MD5, SHA1
Reliability	Load balancing and failover on PSTN side Load balancing and failover on SIP side (optionally uses OPTIONS for keep-alive check) Alive check for active calls on SIP side via SIP session timer (RFC4028)
Call routing	TDM-to-TDM TDM-to-SIP SIP-to-TDM SIP-to-SIP
Media processing features	DTMF generation and recognition (in-band) DTMF relay, RFC2833 Echo Cancellation as per G.168 standard with up to 256 ms echo tail (depending on media gateway interface) Voice Activity Detection and Comfort Noise Generation
IP Media CODEC features	IP Real-time Transport Protocol (RTP) RTP profile name RTP/AVP RTP event (RFC2833) for DTMF, fax, and modem tones G.711 CODEC, 64 kbps (64 kbps, A-law, μ-law) G.726 (16, 24, 32, and 40 kbps) G.729 CODEC (requires additional license from Dialogic) GSM full rate CODEC iLBC CODEC Comfort Noise (RFC3389) Configurable packetization time between 20 ms and 200 ms (iLBC only between 20 ms and 30 ms)
Management	Configuration via web GUI (HTTP or HTTPS) or CLI SNMP for monitoring Logging to PCAP file, SYSLOG Radius interface

Technical Specifications *(continued)*

Physical dimensions	Height: 44 mm (1U) Width: 426 mm Depth: 365 mm
Power supply	200 W ATX Supply

Approvals, Compliance, and Warranty

Hazardous substances	RoHS compliance information at http://www.dialogic.com/rohs
Country-specific approvals	Global product approvals database at http://www.dialogic.com/declarations
Warranty	Warranty information at http://www.dialogic.com/warranties

Ordering Information

Dialogic® BorderNet 500 Gateways and Related Products	Order Code	Description		
			Channel Density	
		Product Type	TDM Ports	SIP-to-SIP Sessions
BN500IP	306-422	Gateway	0	25
BN508BRI	306-424	Gateway	4	8
BN501PRI	306-425	Gateway	30	30
BN504PRI	306-426	Gateway	120	120
BN500SWRCM	M01-201-01	Remote SIP Connectivity Module	N/A	N/A
BN500SWQOSM	M01-202-01	QoS Module	N/A	N/A
BN500SWESM	M01-203-01	Enhanced Security Module	N/A	N/A
BN500SWVSM	M01-204-01	VoIP Survivability Module	N/A	N/A
BN500SWSRM	M01-205-99	SIP Registrar Module	N/A	N/A
BN500SWASTC	M01-206-01	Additional SIP Traversal Channel	0	1

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