Dialogic® 1000 Media Gateway Series by Sangoma

The Dialogic® 1000 Media Gateway Series (DMG1000 Gateways) allows for a well-planned, phased migration to an IP network, making the gateways a smart solution for enterprises looking to enhance their legacy PBX equipment with new VoIP access and applications. Connected between a PBX or a digital handset and a LAN or WAN, the DMG1000 Gateways convert proprietary digital PBX messages into a format suitable for transmission over IP networks.

Features	Benefits
Suitable for small to medium enterprises and easy to install, configure, and maintain. **Compatible with a variety of popular PBX manufacturers including Avaya, NEC, Nortel, and Siemens**	Protects investment in legacy telecommunications equipment and allows a controlled migration to IP technology
Support for IP load balancing and IP fault tolerance	Allows the ability for inbound (TDM-to-IP) calls to round-robin between available media servers and automatically routes calls away from unresponsive media or proxy servers
Seamless interoperability with Dialogic® PowerMedia™ HMP Software	Provides the options for customers to build enhanced applications on top of base gateway and PBX functions
Supports configuration via serial, telnet, and a web browser including context-sensitive help	Easy to install, configure, debug, and maintain
IP security features include TLS, SRTP, and HTTPS	Enables secure communications for SIP messages via TLS, for media stream via SRTP, and for web interface via HTTPS

Applications

- Centralized VoIP and FoIP application servers, including IP-based voice mail and unified messaging
- IVR and announcements
- IP PBX
- VoIP extension to branch offices
- Contact centers

Specific PBX digital network interface gateway units are compatible with the PBXs listed in Table 1. Units are specified by product code for convenient ordering.

Manufacturer	Models	Software Version	Product Code
Avaya	DEFINITY G3 S8100, S8300, S8700, and S8710	Version 3 or greater Communications Manager SW V2.0 or greater	DMG1008DNIW
	Legend	Release 7.0 or greater	DMG1008LSW
	Magix	Release 2.0 or greater	DMG1008DNIW
NEC	2000 IPS 2400 IMG 2400 IMX 2400 IPX	Release 8.2 or greater Release 7400 or greater Release 5200 Dec. 92 1b or greater Release V.17 issue 3.46.001 or greater	DMG1008DNIW
Nortel	Meridian 1 – Option 11, 21, 21A, 51, 61, 71, and 81 Meridian SL1 – Generic X11 Nortel Communication Server – 1000E, 1000M, and 1000S	Release 15 or greater and options 19 and 46 are required Release 15 or greater and options 19 and 46 are required Release V3.0 or greater	DMG1008DNIW
	Norstar 8X24 Norstar MICS	DR5 Release 1.2 or greater Release 4.5 or greater	
Siemens	Hicom 300E CS	Release 9006.4 or greater (**Note:** North American software load only)	DMG1008DNIW
	Hicom 300E	Release 2.0 or greater (**Note:** EU software load only)	DMG1008DNIW or DMG1008LSW
Various	Including Alcatel, Avaya, Ericsson, Fujitsu, Mitel, Siemens, etc., through analog port and/or serial port integration		DMG1008LSW or DMG1004LSW

Cables are not included. Each unit requires one Ethernet cable per unit and one RJ-11 cable per PBX channel.

Table 1. PBX Digital Network Interface PBX Compatibility

Functional Description

The DMG1000 Gateways each contain eight digital PBX emulation interfaces and a 10/100 Base-T Ethernet connection for connecting to a LAN. An analog loop start unit designed for voice mail and unified messaging applications is also available to connect to PBXs that do not have an appropriate digital interface. The analog loop start unit supports integration via in-band signaling (DTMF or FSK) or serial protocols (SMDI, MCI, and MD-110).

The DMG1000 Gateways provide a simple, cost-effective transition to voice and data convergence for enterprises with PBXs. Connected externally, they offer an IP solution that works with current legacy equipment. They support SIP-based applications as well as T.38 for fax transmissions over IP (FoIP).

Gateway unit features include:

- **Voice over Internet Protocol (VoIP)** – Supports SIP per RFC 3261. Uses Real-time Transport Protocol/Real-Time Control Protocol (RTP/RTCP) for delivery of voice over the LAN or WAN
- **IP security** — Supports TLS for SIP messages, SRTP for media stream, and HTTPS for web interface
- **Enhanced voice processing** – Supports a variety of compression algorithms, including G.711 A-law and μ-law, G.723.1, and G.729AB
- **T.38 Fax over Internet Protocol (FoIP)** – Emulation units transcode fax from T.30 fax protocol, supporting V.17, V.21, V.27, and V.29 modulation schemes, to T.38 for transmission over a packet network
- **Hot swap** – Allows gateway units to be added or removed without affecting other gateway units
- **Web server interface** – Each gateway unit is delivered with a web server interface, allowing configuration and software upgrades via a web browser

Configurations

The DMG1000 Gateways can be used to connect IP telephones to a legacy PBX, integrate network-hosted applications with the PBX, extend the PBX to branch offices, and integrate various voice and call processing capabilities in an enterprise LAN or WAN environment. Using exclusive PBX network interfaces (emulating), these media gateway appliances provide exceptional IP to PBX integration capabilities to protect an investment in legacy telecom equipment.

Figures 1 and 2 provide sample configurations.

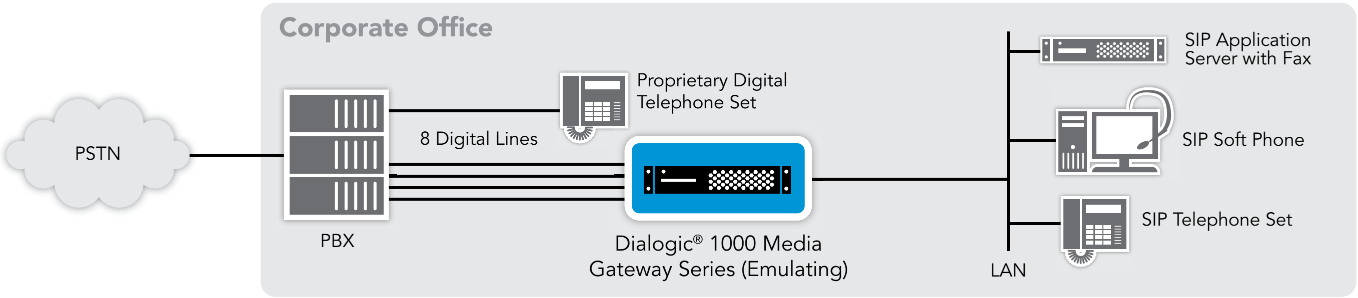


Figure 1. IP-Enabled PBX in Communication with SIP Devices over a LAN

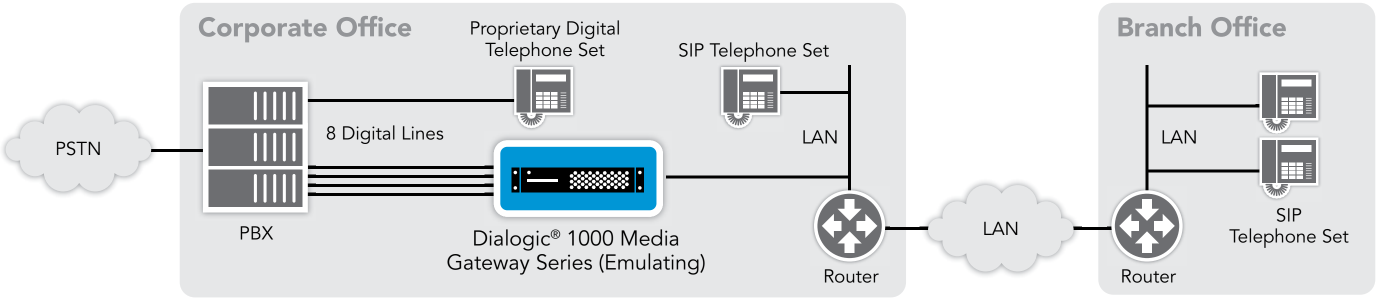


Figure 2. IP-Enabled PBX in Communication with SIP Devices at a Branch Office over a WAN

Call Routing

The DMG1000 Gateways route calls from the switched network to a VoIP destination on the IP network. Conversely, the DMG1000 Gateways route calls from the IP network through a switch port to a destination telephone number on the switched network. The DMG1000 Gateways support the following call routing options:

- User-configurable list of VoIP servers
- IP load balancing
- IP fault tolerance
- TDM-to-TDM

Physical Description

Figure 3 shows the LEDs on the front panel of a DMG1000 Gateway, which reflects the status of the unit, Ethernet, and PBX telephony ports.

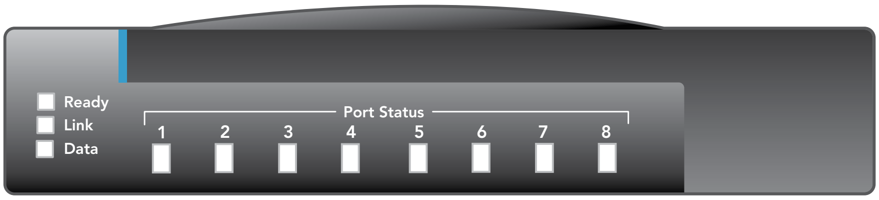


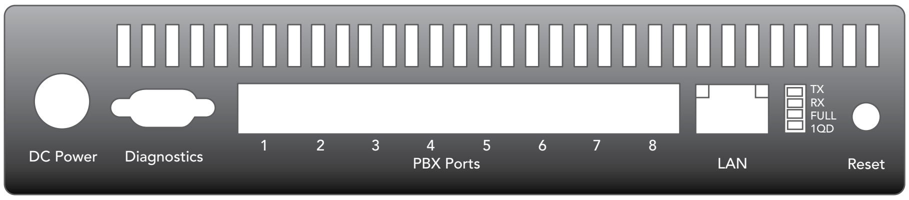
Figure 3. DMG1000 Gateway Front Panel

Ready — Shows overall unit status

Link — Shows the unit's Ethernet status

Data — Shows the unit's Ethernet RTP activity

Port Status 1–8 — Shows the unit's PBX link status for each TDM port



The back panel (Figure 4) contains both interfaces and indicators.

Figure 4. DMG1000 Gateway Rear Panel

Interfaces

- DC power
- Serial port for diagnostics or serial protocol support
- 8 telephony ports
- Ethernet port
- Reset switch

Status Indicators

- 10/100Base-T
- Full/half duplex
- TX/RX traffic
- Ethernet link state
- Ethernet collision

Technical Specifications

PBX Interface

Number of ports	8 port analog units, and 8 port Digital PBX emulation units
	Use multiple gateway units for higher port counts
Connectors	8 shielded female RJ-45 jacks

Network Interface

	10/100 Base-T Ethernet LAN port
Connector	1 shielded female RJ-45 jack for LAN

VoIP Protocols

SIP per RFC 3261

RTP/RTCP for delivery of voice

FoIP Protocol

T.38 FoIP	Emulation units transcode fax from T.30 fax protocol, supporting V.17, V.21, V.27, and V.29 modulation schemes, to T.38 for transmission over a packet network

Voice Support

G.711 -Law and A-Law, G.723.1, G.729AB

Silence suppression with comfort noise

G.168 automatic echo cancellation

Call Progress Analysis (CPA), including Positive Voice Detection, Positive Answering Machine Detection (PAMD), DTMF detection, and fax tone detection

Quality of Service

Type of Service (ToS)

IP precedence

Configuration and Management

SNMP v1	Read-only for alarm reporting
Web GUI	With context-sensitive Help facility
Telnet	
BOOTP client and TFTP client	Built-in

Call Routing

User configuration list of VoIP endpoints

IP load balancing

IP fault tolerance

Supports configuration of a backup SIP proxy server

IP Security

TLS for SIP messages

SRTP for media stream

HTTPS for web interface

Power Requirements

Line voltage	90 VAC to 264 VAC
Frequency	47 Hz to 63 Hz

Physical Dimensions

Length	10 in. (25.4 cm)
Width	9.5 in. (24.1 cm)
Height	2.1 in. (5.3 cm)
Weight	Approximately 2.5 lbs. (1.13 kg)

Environmental Requirements

Operating temperature	32°F to 122°F (0°C to 40°C)
Non-operating temperature	–4°F to 158°F (–20°C to 70°C)

Approvals, Compliance and Warranty

Country-specific safety and telecom approvals	https://portal.sangoma.com/
Warranty Information	https://www.sangoma.com/warranties

Ordering Information

Please see the Models tab for these products

ABOUT SANGOMA

Sangoma Technologies Corporation is a trusted leader in delivering globally scalable Voice-Over-IP telephony systems, both on-site and cloud-based. As the communication landscape evolves and businesses invest in new strategies to provide effective communications, Sangoma Technologies is your trusted partner; delivering Unified Communications solutions for SMBs, Enterprises, OEMs, Carriers, and service providers.

Founded in 1984, Sangoma Technologies Corporation is publicly traded on the TSX Venture Exchange (TSX VENTURE: STC).

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