Dialogic® IMG 1004 Integrated Media Gateway

Dialogic® IMG 1004 Integrated Media Gateway is a carrier-ready VoIP gateway that supports both media and signaling in a single chassis. It allows service providers to add new telephony services quickly, and gives them a clear migration path to an all-IP network. The IMG 1004 brings the value proposition of an integrated media gateway into the low-density (1-4 span) gateway market.

The IMG 1004 provides any-to-any voice network connectivity and can work in tandem with Dialogic's IMG 1010 to deliver SIP services into legacy SS7, PRI, and CAS networks, as well as IP-to-IP transcoding and multimedia border element functions, such as SIP mediation for network edge applications. Its compact 1U design, integrated SS7 links termination and CICs, GUI-based management, and software licensing for in-service capacity expansion make the IMG 1004 an excellent option for VoIP.

The IMG 1004 also features the Dialogic® Programmable Protocol Language (PPL), which allows rapid implementation of ISDN PRI variants and other signaling changes.

Features	Benefits
Simultaneous support for PRI, CAS, and SS7 signaling, along with SIP and H.323	Provides a flexible, cost-effective platform that can evolve from TDM-IP to all IP
SS7 signaling, call routing, call translation, and IP transcoding supported in a single chassis	Can reduce complexity and administrative overhead for VoIP services, and allows on-the-fly voice coder conversion
Supports multimedia border element capabilities, including SIP mediation, topology hiding, and media transcoding	Facilitates efficient operations between incompatible network elements in a service provider infrastructure
Supports up to 128 channels in a 1U chassis	Allows easy scalability to 128 channels in a small footprint
Wireline and wireless support, including ENUM and DNS	Enables fast connection time and low phone charges because callers can connect to each other directly without using the PSTN
Carrier-ready design uses dual Ethernet network interfaces	Provides high reliability and service availability
Works with load balancers	Optimizes distribution of SIP traffic and improves scalability and fault tolerance

Technical Specifications

Routing Features

Call routing and translation based on ANI, DNIS, Generic Number (call routing only supported), and Nature of Address (NOA), Time of Day, Day of Week/Year

Algorithms include percentage-based routing and disposition by Cause Code

Pre- and post-routing digit translations with wildcard support

Multiple routing algorithms per trunk group or groups of trunks for IP-to-TDM and IP-to-IP, both A-law and µ-law conversions

Pre-call announcement (branding)

IP Bearer Features

Coder support: AMR, iLBC, G.711, G.723.1, G.729 A/B, G.729 E/G, GSM-FR, G.726, RFC 4040 clear channel

Echo cancellation: G.168 128 ms tail length

Voice activity detection

Comfort noise generation

T.38 Real Time Fax

Fax/modem bypass

Digit transmission via RFC 2833 (SIP and H.323) or H.245 UII (H.323)

Symmetric NAT Traversal

Secure RTP media (for SIP)

OAM&P

Centralized Element Management System

GUI-based system allows monitoring and provisioning of up to 32 gateways

Node wizard for simplified configuration

Centralized routing engine simultaneously configures gateways in the network

Radius (billing, authentication, prepaid)

Local time zone support and Network Time Protocol (NTP)

SNMP

MIBs: MIB-2, Interface, Alarms, DS0, DS1

Cacti reporting

Power Requirements

120 - 240 VAC 50/60 Hz with voltage range (90 V to 240 V)

Power consumption: 30 W

Physical Specifications

Dimensions: 1.72 in. high (43.7 mm) x 17.25 in. wide (438.2 mm) x 11.00 in. deep (279.0 mm)

Weight: 9.1 lb (4.12 kg)

Resiliency

Local termination of ISUP links and IP backhaul to IMG 1010 signaling node

Redundant Element Management System servers

Graceful software upgrade over multiple IMG 1004s

Graceful busy out per trunk group

Virtual IP addresses for SIP load balancing (via third party server)

Call Release due to media inactivity timeouts

Capacity

32 - 128 TDM channels per 1U shelf (scalable from 1 E1/ 1 T1 to 4 E1 / 4 T1)

32 - 128 VoIP channels per 1U shelf

I/O Interfaces

Telephony: T1 and E1

IP: 2 - Fast Ethernet for control, signaling, and media

T1/E1s for timing (BITS clock) and signaling

Loop timing via any telephony port

TDM Signaling Protocols

ISDN PRI (FAS and NFAS): NI2, Euro ISDN, DMS 250, 5ESS

T1/E1 CAS (FGB, FGD and MFR2)

Q.699 ISDN to SS7 mapping

ISDN UUI mapping to SIP and H.323

SS7/C7 ISUP: ITU and ANSI variants supported through the Dialogic® Programmable Protocol Language (PPL)

SS7 ISUP CICs and Links Termination (connects to SS7 stack on IMG 1010)

Up to 8 SS7 links and 128 CICs per IMG 1004

ISDN call transfer and bridging via Explicit Call Transfer, Two B Channel Transfer, and Release Link Trunking (initiated via SIP REFER)

Delayed ANM for ISUP (triggered by third-party SIP call transfers)

IP Protocols

H.323

H.323 v2

H.323 Keep Alive

SIP and Related Specifications

RFC 2246 Transport Layer Security (TLS) for SIP

RFC 2327 Session Description Protocol (SDP)

RFC 2976 SIP Info for digit transmission (#,*) and interworking DTMF

RFC 3204 MIME Media Types for ISUP and QSIG (ISUP only supported)

RFC 3261 SIP Basic

RFC 3262 SIP PRACK

RFC 3263 Locating SIP servers for DNS lookup SRV and A records

RFC 3264 SDP Offer/Answer Model

RFC 3265 SIP Subscribe/Notify

RFC 3311 SIP Update

RFC 3323 Privacy Header Field (partial support)

RFC 3325 Asserted Identity

RFC 3326 SIP Reason Header

RFC 3372 SIP for Telephones (SIP-T)

RFC 3398 ISUP/SIP Mapping

RFC 3515 SIP REFER

RFC 3578 ISUP Overlap Signaling to SIP

RFC 3581 Symmetric Response Routing

RFC 3666 SIP to PSTN Call Flows

RFC 3711 SRTP (for SIP)

RFC 3725 Third-Party Call Control for SIP

RFC 3764 ENUM for SIP Address of Record

RFC 3891 SIP Replaces Header

RFC 3892 SIP Referred-By Mechanism

RFC 4028 SIP Session Timer

RFC 4244 SIP History info (for call diversion)

RFC 4412 Communications Resource Priority for SIP (partial support)

RFC 4568 SDP Security Descriptions for Media Streams

RFC 4904 SIP tgrp (trunk group) parameter

SIP 3xx Gateway Responses and 302 Initiate

SIP Diversion Header

SIP Trunk Group IDs (OTG, DTG)

SIP Coder Negotiation

SIP Busy Out

ITU-T Q.1912.5 — SIP and ISUP Interworking (includes SIP-I) and Overlap signaling (SIP to SIP ISUP)

SIP Mediation (SIP to SIP)

SIP to SIP-I/SIP-T

QoS

Adaptive jitter buffer

Packet loss compensation

Configurable Type of Service (ToS) fields for packet prioritization and routing

Approvals and Compliance

For information about global approvals, visit www.dialogic.com/declarations or contact your Dialogic sales representative. For information about RoHS compliance visit www.dialogic.com/rohs or contact your Dialogic sales representative.

EMC/EMI

USA/Canada: FCC 47 CFR Part 15, ICES-003

European Union: EN55022: 2006/A1:2007, EN55024: 1998/A1:2001/A2:2003, EN 300 386 V1.4.1 (2008)

Australia/New Zealand: AS/NZS CISPR 22:2006

Japan: VCCI

Safety

USA/Canada: UL/CSA 60950-1 – 2nd Edition (2007)

European Union: EN60950-1:2006

Australia/New Zealand: AS/NZS 60950.1:2003

CB Scheme

International CB Scheme IEC 60950-1 2nd Edition

Telecom Approvals

USA/Canada: FCC Part 68 (TIA-968-A)/IC CS-03

European Union: TBRs 4, 12, 13

Australia: AS/ACIF S-016 and AS/ACIF S-038

Japan: JATE Green Book

Reliability/Warranty

Warranty information at www.dialogic.com/warranties

Estimated MTBF per Telecordia Method 1:
AC power: 160,627 hours

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