Sangoma IMG 2020

Integrated Media Gateway

Datasheet

The Sangoma IMG 2020 Integrated Media Gateway connects and secures sessions across IP and mixed network boundaries to support the seamless delivery of services. It connects IP and hybrid networks via high-density optical, telephony and Ethernet links in a compact 1U form factor. It also transforms media and signaling to support efficient and reliable voice, fax and multimedia sessions for mobile, fixed and cloud-based applications. The combination of IP multimedia and TDM gateway functionality in one chassis offers potentially significant reductions in CAPEX and OPEX when compared to less integrated alternatives. Along with providing a broad range of session performance scalability, the IMG 2020 handles signaling and media in a single chassis and can deliver SIP services into SS7, SIGTRAN, PRI, and SIP-I networks. It also provides basic IP session control and security features to help service providers deliver multimedia services with features that include Denial of Service (DOS) protection, IPv6 to IPv4 interworking, SIP mediation, SIP-to-H.323 interworking, SIP back-to-back user agent (B2BUA), SIP trunking support, and IP-to-IP transcoding of voice, mobile HD voice, fax and tones. The IMG 2020 is part of a line of gateway solutions that help service providers and enterprises energize their networks with a better way to interconnect and deliver services through ease-of-use and low total cost of ownership.

√ Scalable from 50 to 2250 simultaneous SIP sessions with multimedia transcoding, and 128 to 2016 channels of SS7 signaling

- Scalable IP/TDM connectivity provides high performance in a small footprint to help lower OPEX and CAPEX

√ Combined IP/TDM gateway features on one platform

- Integrated multimedia gateway features facilitate TDM and IP interworking to provide service delivery flexibility and automated failover between domains

√ Any-to-any signaling and media support

- Support for SS7, SIP signaling, and IPv6 and IPv4 interworking along with voice transcoding provides a cost-effective platform to help service providers evolve from a TDM to an all-IP environment

√ SIP profiler, web-based user interface and offline configuration

- Easy-to-use service configuration and management tools can help accelerate service deployment and simplify platform management

√ Integrated encryption and transcoding support for voice, tones and faxing

- Eliminates the need to add separate hardware to support both security and transcoding requirements, helping to reduce CAPEX and number of platforms deployed

√ Carrier class solution

- Carrier class design and features provide high availability, reliable throughput and enhanced service delivery

Technical Specifications

Routing Features

- Call routing and translation based on ANI, DNIS, Generic Number (only translation is supported), Nature of Address (NOA)
- 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 and both A-law and µ-law conversions
- Pre-call announcement (branding)

IP Bearer Features

- Voice activity detection and packet loss concealment
- Comfort noise generation
- T.38 real-time fax, T.38 – G.711 interworking
- Fax/modem bypass
- Digit transmission via RFC 2833 (SIP)
- G.711 tones, SIP INFO, RFC 2833 interworking
- Hosted NAT
- VLAN tagging
- Secure RTP (SRTP) to RTP interworking (SIP audio media only)

Coder Support

- AMR-NB, AMR-WB, G.711, G.723.1, G.729 A/B, G.726, G.722, GSM-FR, GSM-EFR, iLBC, RFC 4040 Clear Channel

Echo Cancellation

- G.168 128 ms tail length

OAM&P

- Web User Interface (WebUI) supports configuration via browser
- Sangoma Multi-Node Element Management System — Enables monitoring and provisioning of up to six (6) nodes via web browser
- Offline Configuration software utility
- Trace Server software for logging
- Centralized routing engine simultaneously configures gateways in the network
- Radius (billing, authentication, prepaid)
- Local time zone support and Network Time

Protocol (NTP)

- Configuration tracking and reporting by user SNMP MIBs
- MIB-2, Interface, Alarms, Private Call Reporting and System Statistics, Private Alarms, DS0, DS1, DS3, and OC3

Power Requirements

AC Power Supply Range

Note: AC power supply will operate at frequencies between 47 Hz and 63 Hz

- 100 — 132 VAC (115 VAC nominal)
- 180 — 264 VAC (230 VAC nominal) DC Power Supply Range
- -36 to -60 VDC (-48 VDC nominal)

Power Consumption

No DSP Modules

- Typical: 90 Watts | Maximum: 120 Watts 1 DSP Module
- Typical: 1100 Watts | Maximum: 145 Watts

2 DSP Module

- Typical: 130 Watts | Maximum: 170 Watts

3 DSP Module

- Typical: 150 Watts | Maximum: 195 Watts

4 DSP Module

- Typical: 170 Watts | Maximum: 220 Watts

Environment

Operating Temperature Range

- 0 to +50 °C, 95% relative humidity non-condensing

Storage Temperature Range

- -10 to +75 °C, 95% relative humidity non-condensing

Physical Specifications

Dimensions

- 1.72 in (43.7 mm) high, 16.97 in (431 mm) wide, 19.67 in (499.6 mm) deep

Weight

- 24 lb (10.9 kg)

Maintenance

Field replaceable items

- Fan filter (available in 10-packs)
- Power supplies
- OC-3/STM-1 optical module
- Motherboard tray
- Up to four (4) DSP modules

Resiliency

- SS7 signaling: 1+1 active/standby redundancy
- Smart IP probing
- Automated failover (Ethernet links)
- Failover via automatic protection switching (optical links)
- Graceful out of service per node and channel group
- Virtual IP addresses for SIP load balancing (via ▪
- Sangoma PowerVille™ LB - Load Balancer)
- Call release due to media inactivity timeouts
- Dual, hot swappable, AC/DC power supplies

Capacity

- 128 - 768 TDM channels per 1U shelf with Rear I/O Type 1 (scalable from 4 E1/5 T1 to 24 E1/T1)
- 672 - 2016 TDM channels per 1U shelf with Rear I/O Type 2 (supports either Optical 0C3 interface or 3 DS3s)
- 100 — 4500 VoIP channels per 1U shelf
- 50 to 2250 VoIP sessions

I/O Interfaces — Rear I/O Type 1 — T1/E1 Telephony — T1 and E1

- 24 T1/E1 for timing (BITS clock), signaling and bearer traffic (T1 — 100 ohms and E1 — 120 ohms

Clock Sync

- Stratum-3 via T1/E1 interface

IP Interfaces

- LAN IP
  - Dual redundant 100/1000 Base-T Ethernet for control; 2 — 100/1000 Base-T Ethernet Aux ports (reserved for later use)

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- WAN IP
  - 4 - 100/1000 Base-T Ethernet for VoIP payload and signaling

I/O Interfaces — Rear I/O Type 2 — High Density

Telephony — T1 and E1, OC3/STM-1, and DS3

- 1 to 3 DS3 + 4 - T1/E1 for timing (BITS clock), signaling and bearer traffic
- 1 OC3/STM-1 with Automatic Protection Switching (APS) + 4 T1/E1 for timing (BITS clock), signaling, and bearer traffic (T1 — 100 ohms and E1 — 120 ohms)

Clock Sync

- Stratum-3 via T1/E1 interface or OC-3/ STM-1 interface

IP Interfaces

- LAN IP
  - Dual redundant 100/1000 Base-T Ethernet for control; 2 - 100/1000 Base-T Ethernet Aux ports (reserved for later use)
- WAN IP
  - 4 — 100/1000 Base-T Ethernet for VoIP payload and signaling (additional 4 reserved for later use)
- Optical Transceiver
  - Hot plug LC connector type SFP modules (1310 nm 15 KM)

TDM Signaling Protocols

- ISDN PRI (FAS and NFAS): NI2, Euro ISDN, DMS 250, 5ESS, JATE/Japan INS-NET1500, ISDN Net 5
- Q.699 ISDN to SS7 mapping
- ISDN/SS7 UUI mapping to SIP
- SS7/C7 ISUP: ITU, ETSI and ANSI variants supported through the Sangoma® Programmable Protocol Language (PPL); JT-ISUP with TTC, PTC 331
- SS7 TCAP for message-waiting-indication (MWI) and Caller Name (CNAM) service
- 64 SS7 links in standalone configuration
- 128 SS7 links in redundant configuration
- A-links and F-Links supported
- E1 to DS3 mapping for third-party multiplexor compatibility
- ISDN call transfer and bridging via Explicit Call Transfer, Two B Channel Transfer, and Release Link Trunking (initiated via SIP REFER)
- ISUP call transfer and bridging via Explicit Call

Transfer (initiated via SIP REFER)

- Delayed ANM for ISUP (triggered by third-party SIP call transfers)
- ISDN and ISUP Multilevel Precedence and Preemption (MLPP)

IP Protocols

- H.323
- H.323 v2
- H.323 RAS, H.245, and H.225
- H.323 Tunneling
- H.246 Annex C – ISDN User Part Function — H.225.0 Interworking

Core SIP Specifications & Notable Extensions

- RFC 3261 SIP Basic
- RFC 3262 SIP PRACK
- RFC 3263 Locating SIP servers for DNS lookup SRV and A records (partial support)
- RFC 3264 SDP Offer/Answer Model
- RFC 3265 SIP Subscribe/Notify

Notable SIP Extensions – Partial List

- RFC 2246 Transport Layer Security (TLS) for SIP
- RFC 3372 SIP for Telephones (SIP-T)
- RFC 3398 ISUP/SIP Mapping
- RFC 3711 SRTP (for SIP)
- RFC 3966 –Tel URI
- RFC 5806 – SIP Diversion Header
- RFC 6140 – Bulk SIP Registration
- RFC 6157 – IPV6 Transition in SIP
- RFC 7433 – SIP User to User Information (UUI)
- ITU-T Q.1912.5 - IP and ISUP interworking
- 3GPP 29.163 – SS7 to SIP interworking (partial)

SIGTRAN

- RFC 3332 — M3UA Adaption Layer
- M3UA Application Server
- M3UA Signaling Gateway for TCAP/ SCCP

QoS

- Adaptive jitter buffer
- Packet loss compensation
- Configurable Type of Service (ToS) fields for packet prioritization and routing

Approvals, Compliance and Warranty

Country Approvals

- https://portal.sangoma.com/

Warranty Information

- https://www.sangoma.com/warranties

Rear I/O Type 1 — T1/E1

No DSP Modules

- 148000 hours

1 DSP Module

- 121000 hours

2 DSP Module

- 103000 hours

3 DSP Module

- 89000 hours

4 DSP Module

- 79000 hours

Rear I/O Type 1 — T1/E1

No DSP Modules

- 162000 hours

1 DSP Module

- 130000 hours

2 DSP Module

- 109000 hours

3 DSP Module

- 94000 hours

4 DSP Module

- 83000 hours

100 Renfrew Drive, Suite 100, Markham ON L3R 9R6 Canada | +1 (256) 428-6000 or 1 (877) 344 4861 (toll free in N. America)