KHOMP

Media gateway with modular interfaces and SBC

Main Features

- Modular composition: Up to 16 external modules to be used with E1/T1, FXS, FXO, or GSM technologies.
- Integrated SBC with up to 2,000 VoIP sessions
- Up to 2,000 TDM channels (up to 64 E1's)
- Supports SS7/SIGTRAN and SIP-I
- Support for call classification
- Survivability and Register Authorization
- High availability in active/inactive mode

Applications

- Recommended for busy organizations with high volume of calls who need an end-to-end management solution for their telephony operations
- Call classification and intelligent routing using the Analytics solution
- TDM Gateway: high-capacity VoIP with up to 64 E1's
- Integrated SBC with support for Register Authorization
- Interconnection with support for SS7/SIGTRAN and SIP-I
- Control of expenses with telephony and loyalty attributes of long-distance carriers
- Intelligent centralized management

Overview

The KMG 3200 One is a product from the Khomp Media Gateway line. A device with great capacity for simultaneous calls, supporting up to 64 E1/T1 links or 2,000 TDM channels, which can also be used by GSM, FXO, and/or FXS technologies. It is ideal for reliable network structures requiring maximum voice quality. It has 13 network ports that can be used for interconnection of separate networks, or for connection of telephony modules.

Telephony modules can be connected to each other in a chain in up to a maximum of 16 modules.

It has advanced routing resources and B2BUA-type SBC security. It also has features that include call classification, local survivability, high availability and intelligent channel monitoring in real time.

Call Capacity

The KMG 3200 One has a capacity for up to 2,000 simultaneous calls, whether they are TDM or VoIP.

In the event that transcoding of the standard VoIP G.711 codec to G.729 and G.722 codecs is performed, this total capacity is cut in half, resulting in 1,000 simultaneous calls to any technology (Any-to-Any).

For VoIP calls, there is also the option of configuration in Bridge* mode, with a capacity of 2,000 simultaneous calls, with the added advantage of being able to use any audio or video codec.

**In this mode, it is not possible to use the Analytics call classification feature.*

Call Routing

Obtain greater control of costs related to telephony charges through configuration of routes by prefix or in accordance with loyalty attributes, making it possible to direct calls to the carriers that offer the best cost benefit for each call, leading to lower call charges.

Register call routes with automatic transbording by time or retry; order routes by priority and change the A and B numbers, as necessary, thus providing a wide array of combinations, including the creation of lower cost routes, contingency and balancing.

Route failover is another critical capability for organizations that cannot afford telephony services downtime in their networks. It is implemented using the routes together with the monitoring of the destination server for the VoIP call. If the VoIP server doesn't respond to the commands sent by the monitoring resource, the KMG ignores the route and searches for another compatible route.

Moreover, it uses routing scripts to facilitate compliance with different scenarios. All routing information can be stored and made available for analysis through the CDR files generated by KMG 3200 One, with a customized format and RADIUS support.

Interconnection with SS7/SIGTRAN Support and SIP-I

Multiple possibilities for interconnection though support of SS7 and SIGTRAN protocols. In addition to new support for the SIP-I protocol, it makes new expansion scenarios possible without the need to worry about TDM connections. For this reason, the KMG 3200 One is the ideal device for carriers addressing diverse scenarios, with the possibility for future expansion.

Telephony Modules (optional items)

One of the features of the KMG 3200 ONE is modularization, which allows it to be set up according to the business model to be deployed, simultaneously accepting E1/T1, FXS, FXO, and GSM interfaces. Get more details on the external telephony modules:

• KMG GSM – 160 Module: Module for applications requiring GSM channels and advanced voice resources. This module has up to 16 GSM channels with 2G quad-band interface, with 2 SIM cards per channel, one active and one in stand-by, as well as 16 VoIP SIP channels.

• KMG GSM – 160 Module (H – for 3G): Similar to the GSM – 160 module, but with a six-band GSM 3G interface and fallback to 2G.

• KMG FXS 240 Module: Module for applications that need an analog extension interface. This module has 24 analog FXS channels and 24 VoIP SIP channels, as well as PBX protocols such as call transfer, second line and alternate call answering.

• KMG FXO 120 Module: Module for applications that need analog trunking. This module can have 4, 8, or 12 analog FXO channels, being 1 SIP channel for each analog channel for VoIP. It has PBX protocols that allow for Flash generation and detection.

• KMG Modular Module: Module that integrates the GSM, FXS, FXO, E1/T1, and VoIP interfaces in a single hardware. The interfaces can be purchased according to the needs of the application, making it possible to combine three of the following interfaces: 1x or 2x E1/T1 links, 8x FXS channels, 4x FXO channels, 1x or 2x GSM channels. Each interface has the same performance characteristics and features as the modules described above, but they are all combined in a single device.

For more modular options, consult the product manual.

E1/T1 Bypass for the solution security (optional item)

The E1/T1 bypass provides contingency to products with these links. Installed inside the device, it physically switches link 1 to link 2, performing transfer from one E1/T1 link to another in case of server failure.

Call Monitoring: INSIGHT (optional item)

Effective dashboard monitoring, in real time, with intelligent management of calls made by the Gateway, giving the number of calls, average duration of calls, and hang up causes, besides issuing warnings based on predefined parameters that keep operating performance high.

Call Classification: Analytics (optional item)

Analytics is a call classifying resource from Khomp. It includes a powerful call classification algorithm that determines whether the call was intercepted by the carrier, or if the remote answer comes from a cellular answering service, and also if the answering action was automatic or human. This improves the performance of the calls made and reduces operating costs. It is based on standards that are previously registered in the system and specific behaviors of audio and signaling on the call.

After identification, Analytics checks the values configured in the gateway, and then performs a previously configured action that may be hanging up the call with the corresponding cause, which can be customized. It can also issue a notification via SIP Info, with the result of the answering analysis. Analytics operates in all calls simultaneously, regardless of the number of interfaces in operation on the same gateway, even if the calls are TDM, GSM or VoIP. And it can be configured for different locations within the same device.

For each type of interface, Analytics must be acquired through modular licenses, according to the solution needed. The Analytics modules available for the KMG 3200 One are:

•KMG Analytics v1 - Single Voip Session: Analytics License for 30 VoIP calls
•KMG Analytics v1 - 16 GSM: Analytics License for 16 GSM calls
•KMG Analytics v1 - 1E1/T1: Analytics License for 1 E1/T1 link

Survivability - SAS

The Stand-Alone Survivability (SAS) feature ensures the continuity of telephone communications in case the IP PBX system becomes unavailable. When the KMG 3200 One has an installed SAS license applied, it assumes the basic functions of the IP PBX system, such as: making and receiving calls between extensions, making external calls, and transferring of calls. This way, communication isn’t compromised while you are waiting for your IP PBX to be available again.

High Availability

The KMG 3200 has an integrated High Availability system based on the concept of active/inactive devices (1+1), with automatic replication of the configurations. In case of an active device failure, it automatically switches to the inactive device, which takes over the network addresses and routing tasks, becoming the active device. This prevents prolonged downtime caused by hardware failure or replacement/servicing of the active gateway.

SIP Trunking (optional item)

The KMG 3200 One allows you to make calls using the SIP connection. It is an ideal solution for companies and institutions with a great demand for communication through IP exchanges that also seek quality of service, flexibility and affordable costs for voice services.

The KMG 3200 One has 3 VoIP operation modes: In G.711 mode, it can make up to 2000 VoIP calls. In transcode mode, the maximum capacity is 1000 VoIP calls. And in bridge mode, the maximum capacity is 2000 VoIP calls, with the advantage of being able to use any audio or video codec.

The use of the Analytics feature (a separate license is needed) is only available in the G.711 and transcode modes.

With this, a variety of other SBC and security resources are added to the device, allowing for interoperability among networks and protocols using its 13 network interfaces, as well as NAT traversal and other resources provided by Register Authorization (a separate license is needed).

Find out more on Khomp SBC resources from our commercial consultants.

Product Images

Front View

Rear View

Technical Specifications

E1/T1 trunk support

- Network channels: up to 64 E1/T1 links
- Network protocols: ISDN and R2 digital. It's possible to configure different protocols on each link.
- PBX protocols: EL7, Line Side, LC and QSIG (SSCT and CT)
- Connector options:
  - BNC coaxial (75 Ohms)
  - RJ45 (120 Ohms)
- 30 SIP channels for each E1/T1 link (G.711)
- SS7 and SIGTRAN (optional license)
- SIP-I support

System status

- System status via web
- Status of trunks and channels via web
- Detailed diagnosis of the E1/T1 links
- SNMP support

Operation interfaces

- Configuration, monitoring, management, and diagnostics via web
- Control of access and registration of changes made by the user on the web interface
- Generation of signaling and system logs
- Analysis of call log integrated into the interface (R2/ISDN)
- Capture of packets via web
- Serial interface (RS-232 DB-9 connector)

Traffic control

- Ability to limit the number of simultaneous calls per network

Supported codecs

- G.711 A-law and μ-law, native to the system, for all interfaces
- G.729A, G722, GSM, DVI, T-38; in transcoding
- VoIP bridge for any codec, including video codecs (without support for call classification)

NAT Traversal

- Interconnection of different networks
- External IP configuration
- STUN

Call Routing

- Lower Cost Routing (LCR)
- Routing based on the caller number, called number, time of day, and priority
- Route fidelization (ability to change the called number)
- Allows queries to the portability database
- Fallback in case of route failure
- Failover retry based on the cause of the failure
- Script routing
- Load balancing
- Route profile
- Up to 500 call attempts per second(CAPS)
- Up to 2000 simultaneous records (Resource shared between Survival and Register Authorization)

VoIP features

- Handling of called number (to) and caller number (from)
- NAP (Network Access Points) monitoring or Keep Alive (sends UDP packets to the router to indicate that the port is in use, without impacting the bandwidth)
- SIP Proxy Fallback
- DTMF sending mode selection: In band, Out band – RTP (RFC 2833) or Out band – SIP Info
- Adding, removing, and retransmitting headers
- Transcoding (conversion between the G.711, G.729 and G.722 codecs)

Survivability - SAS

- Forwarding of incoming and outgoing calls
- Transfer with and without consultation
- Automatic proxy fallback

QoS (quality control)

- DiffServ - RFC 4594 4 (traffic classification and management)
- VLAN Tagging

Interoperability

- Fax Interoperation (T.38 with fallback to G.711)
- IPv4 to IPv6
- RTP with conversion between UDP, TCP and SRTP (SDES and DTLS)
- SIP junction
- Microsoft Teams Direct Routing. Beta phase, interoperability with and without Media Bypass.

Call Register

- Generation of CDR with configurable format
- Channel use monitoring
- Call counters per channel
- Option for download in CSV format (compatible with Microsoft Excel)
- Automatic export via FTP
- RADIUS protocol used for Accounting (billing) purposes

Telephony modules

FXS

- Network channels: 24 analog FXS channels
- PBX protocols: transfer, second line, hold and conference
- Configurable ring cadences
- Compatible with FOP (Flash Operator Panel)

FXO

- Network channels: 4, 8, or 12 analog channels
- Modularity: 3 x 4 lines
- PBX protocols: generation and detection of flash
- Line impedance configurable for 900 Ohms or 600 Ohms

GSM

- Modular with up to 16 GSM interfaces
- Capacity for 2 SIM cards per channel, one active and another in stand-by
- Allows for different carriers on the same module
- 3G Six Band: 800/850/900/1700/1900/2100 MHz
- 2G Quad Band: 850/900/1800/1900 MHz
- SIM card size: mini-SIM (2FF)

Warranties and certifications

- Total warranty (legal + Khomp warranty): 1 year
    - Legal warranty: 90 days
    - Khomp warranty: 9 months
- Anatel (Brazilian National Telecommunications Agency) Certification
- ISO 9001 certified

Call Answering Classification – Analytics

- Combined voice and signaling analysis
- Actions configurable per classification result
- Classification audit
- Pre & Post call answering analysis
- Fax and Voicemail detection within the standard range 600 Hz / 450 ms - 2300 Hz / 450 ms

Security

- Access via HTTP or HTTPS
- Fraud prevention: call blocking by called number and caller number
- Protection against DoS/DDoS attacks
- Network topology hiding
- SIP TLS and SRTP protocols (SDES and DTLS)
- Access Control List – ACL (Whitelisting and Blacklisting)
- Protection against malformed packets
- Register Authorization*

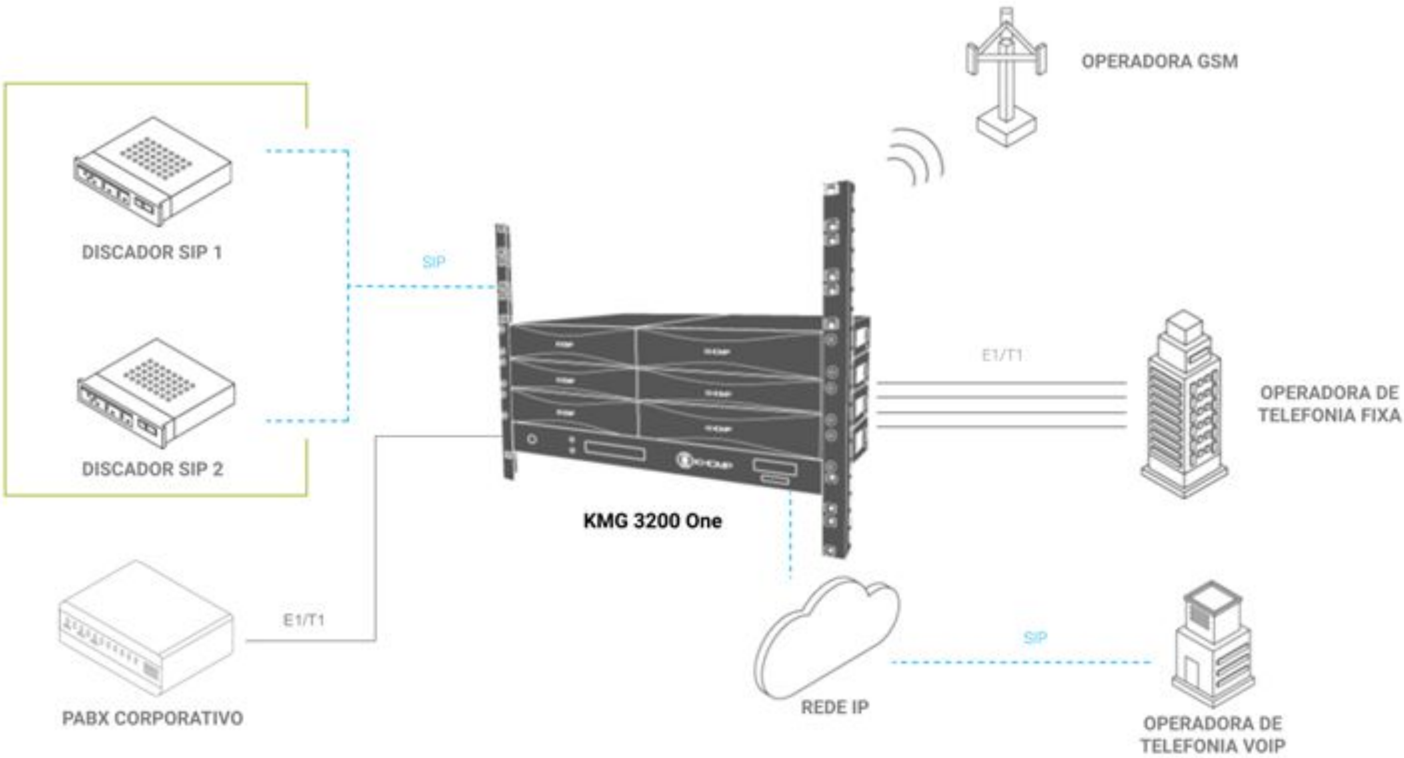
Other features

- Provisioning (settings export and import)
- History and restoration of changes to settings via web
- Remote terminal with advanced CLI (Command Line Interface)
- TR-069 Support
- Support ITU-T G.165 and G.168 standards
- High Availability (1+1)
- Acoustic signaling treatment performed by hardware through DSPs
- Automatic fax tone detection (2100Hz) automatically enabling echo cancellation

Physical/Environmental

- Redundant power supply
    - Input: 110–240 VAC – 50/60 Hz
    - Maximum power consumption: 150 W
- 13 Gigabit network interfaces 10/100/1000 Mbps
- 6.3" x 0.8" LCD display
- Standard 1U Module for 19" rack
- Telephony modules use 1U for every 2 modules
- Dimensions: 17.24" x 1.75 mm x 14.97" mm
- Approximate weight (without packaging): 16.53 lb

Application Model



KHOMP
Rua Joe Collaço 253, Florianópolis
+55 (48) 3722.2900
comercial@khomp.com