KHOMP

Medium density media gateway with modular interfaces and SBC

Main features

- Up to one internal modular EBS and up to eight external modules, allowing for a great variety of modular solutions
- Integrated SBC with up to 480 VoIP sessions
- Up to 480 TDM channels (16 E1's)
- R2 and ISDN links (BNC or RJ45 connectors)
- Supports SS7/SIGTRAN and SIP-I
- Supports call classification
- Survivability and Register Authorization
- Standard 1U sizing for 19" racks

Applications

- Small to medium-sized call centers and VoIP carriers that need a low cost, easy to implement upgrade
- Connection between PSTN carrier and IP PBX
- Connection between VoIP carrier and digital IP PBX
- Connection between main office and branches with option of local survivability
- Integrated SBC with support for Register Authorization
- Control of telephone expenses and loyalty attributes for long-distance carriers
- Small-sized service providers requiring complete management of their IP telephony operations, including advanced features

Overview

The KMG 400 One is a product from the Khomp Media Gateways line. A low density device with modular interfaces and integrated SBC; the initial configuration can include 1 internal EBS module, allowing for the addition of up to 8 external telephony modules. These modules can be outfitted with E1/T1, GSM, FXO and/or FXS technologies, for a total of up to 480 TDM channels or VoIP calls . The KMG 400 One comes with 5 network ports, of which 4 can be used for connecting telephony modules, which can be connected to each other in a chain, allowing for a maximum of 8 modules.

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

Call capacity

The KMG 400 One has a capacity for up to 480 simultaneous calls, whether they are TDM or VoIP.

In the event of the use of transcoding of the standard VoIP G.711 codec to G.729 and G.722 codecs, this total capacity is cut in half, allowing for 240 simultaneous calls between any technologies (Any-to-Any).

For VoIP calls, there is also the option for Bridge***** mode configuration, with a capacity for up to 480 simultaneous calls, with the advantage of being able to use any audio or video codec.

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 400 One, with a customized format and RADIUS support.

Interconnection with SS7/SIGTRAN support and SIP-I

Multiple interconnection possibilities, through Signaling Point and Signaling Transfer Point (SP and STP), with SS7 and SIGTRAN signaling. In addition to support for the SIP-I protocol, enabling new expansion scenarios without the need to worry about TDM links. In this way, the KMG 400 One becomes the ideal equipment for operators, covering various scenarios with the possibility of future expansion.

Telephony modules (optional)

One of the features of the KMG 400 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 (H – for 3G): Module for applications that need GSM channels and advanced voice resources. This module has up to 16 GSM channels with a 3G six-band GSM interface with fallback to 2G, with 2 SIM cards per channel, one active and one in stand-by, in addition to 16 SIP channels for VoIP.
- 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)

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)

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.

Survivability - SAS (optional)

The Stand-Alone Survivability (SAS) feature ensures the continuity of telephone communications in case the IP PBX system becomes unavailable. When the KMG 400 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.

SIP trunking (optional)

The KMG 400 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.

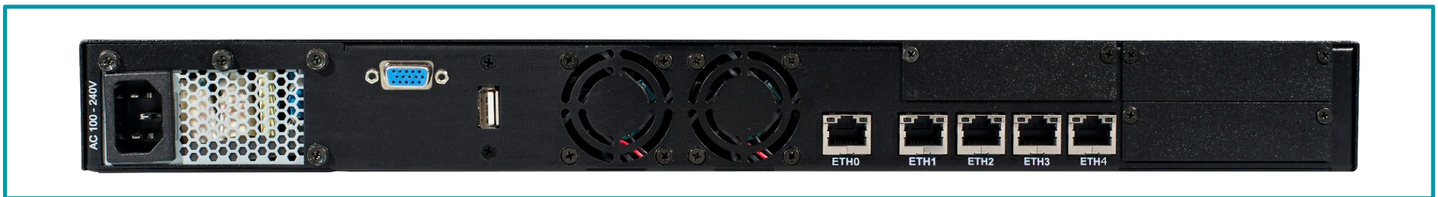
The KMG 400 One has 3 VoIP operation modes: In G.711 mode, it can make up to 480 VoIP calls. In transcode mode, the maximum capacity is 240 VoIP calls. And in bridge mode, the maximum capacity is 480 VoIP calls, with the advantage of being able to use any audio or video codec.

With this, a variety of other SBC and security resources are added to the device, allowing for interoperability among networks and protocols using its 5 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

Subtitle: *Front view*.



Subtitle: *Rear view*.

Technical specifications

> ⚠ Attention
> - Product hardware may be replaced without notice.
> - Replacement happens when raw material is not available on the market or when better hardware comes along.
> - When the hardware is replaced, the product will operate at the same potential as the previous configuration.

E1/T1 trunk support
- Network channels: 0–16 E1/T1 links
- Network protocols: ISDN and R2 digital (with up to 480 MFC signaling switches). 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)
- Signaling Point and Signaling Transfer Point (SP and STP) support in SS7 and SIGTRAN (Optional Licensing)
- 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

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)

Call Routing
- Lower Cost Routing (LCR)
- Routing based on the source number, destination 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 100 call attempts per second (CAPS)
- Up to 600 simultaneous records (shared resource between Survivability and Record 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

Call Admission Control
- Based on local resources
- Call rate limiting

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

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)

Physical/Environmental

- Full Range Power Supply
  - Input: 110–240 VAC, 50/60 Hz
  - Maximum power consumption: 150 W
- 5x gigabit network ports 10/100/1000 Mbps
- Dimensions: 488x395x45 mm
- Transport box dimensions: 580x510x110 mm
- Gross weight: 9.3 kg
- Net weight: 6 kg

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

NAT Traversal

- Interconnection of different networks
- External IP configuration
- STUN

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.

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
- Acoustic signaling treatment performed by hardware through DSPs
- Automatic fax tone detection (2100Hz) automatically enabling echo cancellation

Application model