KMG SBC 750

Media gateway

front

back

- Up to 3000 VoIP channels*
- Up to 1500 VoIP calls*
- Call classification in all the channels
- Route loyalty
- Inquiry on portability by webservice
- Redundant power supply source

Comparative table of call capacity

Quantity of calls	from 30 to 750	from 751 to 1500
Transcoding	YES	NO
Quantity of G.729 channels	from 60 to 1500	0
Quantity of G.711 channels	from 60 to 1500	from 1502 to 3000

** with transcoding deactivated*

The KMG SBC 750 is a media gateway which has function SBC of the type B2BUA of medium size. It is equipped with powerful signal processors which execute the conversion of media and protocols between networks. Developed for call center companies which work with SIP connection in their P.A.s, the KMG SBC 750 ensures secure connection between the local network and VoIP operator. It handles up to 750 VoIP calls with active transcoding or up to 1500 VoIP calls without transcoding.

KMG 30 VoIP license

The KMG SBC 750 leaves the factory with a KMG 30 VoIP license pre-installed, allowing up to 30 simultaneous calls. Another 24 KMG 30 VoIP licenses can also be installed. Furthermore, it has an internal configuration called "transcoding mode", which leaves the factory activated. In this mode, it is possible to increase the capacity of the gateway to up to 50 KMG 30 VoIP licenses, totaling 1500 simultaneous calls. The KMG SBC 750 has 8 network interfaces which can be configured to interconnect up to 8 different networks.

SIP trunking

It allows SIP connection, the ideal solution for companies and institutions with great requirement of communication through IP PBX and that seek quality of service, flexibility and accessible costs in the voice services.

Call routing system

The routing system allows the user to register routes with automatic overflow and fallback. The routes allow order of priority and modifications in the numbers of A and B, allowing a huge variety of routing combinations, including **creation of routes of lower cost**, **contingency** and **loyalty**. All the routing information can be stored and made available for analysis through CDR files with customized format.

Call classification

The call classification resource allows the gateway to identify if the call was intercepted by the operator or if the remote answerer is a cell phone voicemail, automatic answering or human answering. Passing to the dialer what was detected in this analysis, the best forwarding can be defined for each case.

(Optional item)

Features and Benefits

Support for trunks

- 8 gigabit network interfaces (100/1000 Mbps)
- Transcoding G711 (A-law and µ-law), G.729 and T-38

Hardware

- LCD 16x2
- Default module 1U for 19" rack
- Dimensions 437.8 (width) x 380 (length) x 44.7 mm (height)

Monitoring, diagnosis and administration by Web Interface

- Support for SNMP
- Viewing of channel status by Web
- Call counters by channel
- Channel occupation monitoring
- CDR in customizable format and download of files by FTP

Call classification

- Identification of call by human answering or intercepted by fax, voicemail, etc.
- Audit of call classification with recordings
- Allows actions to be created in accordance with call classification

Call routing

- Selection of routes based upon destination number, origin number or origin IP
- Creation of routes with prioritization
- Route loyalty (allows changing destination number)
- Configuration of alternate routes (by time, automatic overflow and fallback)
- Forwarding based upon inquiry on numerical portability
- Routing of calls by time
- Possibility of analysis through CDR files

Guarantees and Certifications

- Factory warranty: 1 year
- Anatel Certification
- Company certified by ISO 9001:2008