Digital gateway

SMG-1016M

SMG1016M

Gateway equipment

SMG-1016M 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. It supports media-gateway functions, such as codec conversion, conferencing, DTMF signals receiving/generation, voice packet forming. E1 supporting number is 16, E1 media channel –up to 511, VoIP channel – 960 (under G.711 codec, packetization period is 20 ms and more).
Submodular construction provides easy capacity growth, and easy system modernization is achieved due to minimal module types number.
SMG-1016M provides a flexible and cost-effective platform that can evolve from TDM-IP to all IP.

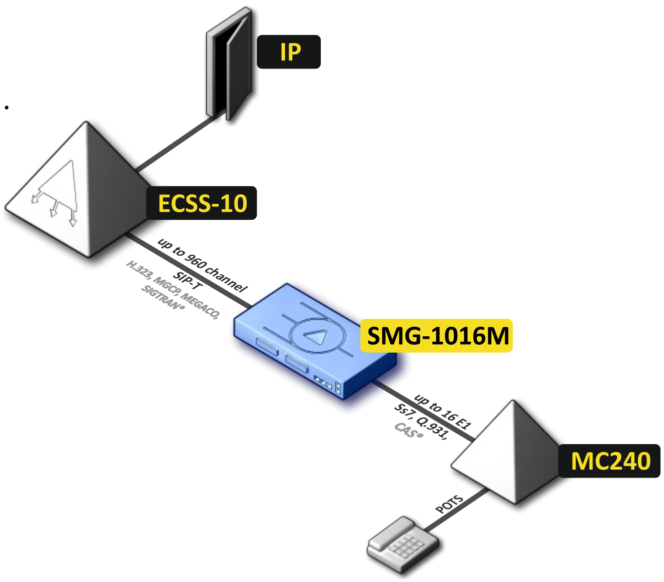
+ modular structure
+ up to 16 E1
+ up to 960 VoIP channels
+ SIP registrar

Voice services quality

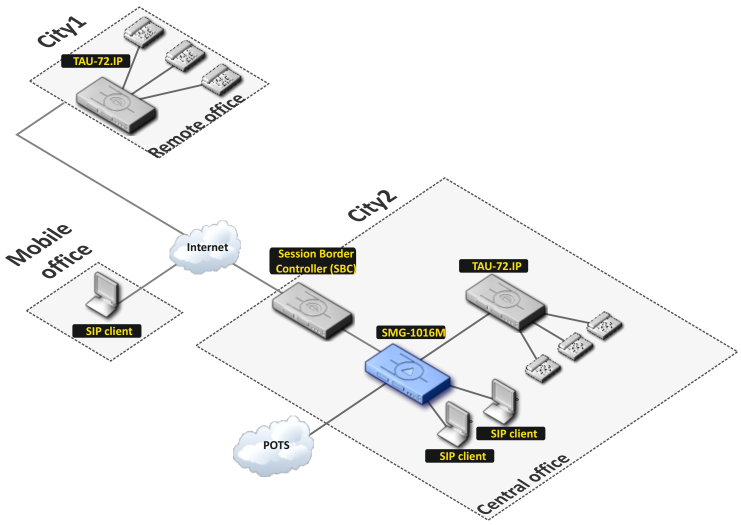
Integrated IP-PBX provides up to 2000 sip-subscriber registration. It supports QoS, adapter jitter-buffer and IP-telephony main codecs.

Application

There are many applications for SMG1016M. It can be used like protocol converter, local communication site, corporative network management device, transit-endpoint telecommunication node.



TDM to VoIP transition



Corporative network managed by SMG1016M

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Digital gateway

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Specification

Routing features

- Call routing and translation based on CdPN and CgPN
- Direct trunking (without routing table)
- Trunk group stop for maintenance
- Several numbering plans (Centrex)
- Pre-and post-routing digit translations
- Pre-routing number type, category, presentation and screening indicators translations

IP-telephony features

Voice codecs:

- G.711 (a-law, μ-law)
- G.729 (A/B)
- G.723.1
- G.726 - 32 Kbps

Fax:

- T.38 Real-Time Fax
- G.711 (a-law, μ-law) pass-through
- Fax/modem bypass
- V.152

Voice standards:

- Voice activity detection
- Echo cancellation: G.168 128ms tail length
- Comfort noise generation

QoS:

- Adaptive and fixed jitter buffer
- Configurable Type of Service (ToS) fields for packet prioritization

DTMF:

- Digit transmission inband, via RFC 2833 or SIP INFO

RADIUS-server:

- Authentification
- Accounting
- RFC 5090 RADIUS Extension for Digest Authentication

Flexibility

- SS7 Signaling: D-channel active/standby redundancy
- Voice connection activity control (RTP or RTCP)

TDM Signaling Protocols

- ISDN PRI (Q.931)
- QSIG for name transmission
- SS7 ISUP
- Q.699 (Interworking between ISDN and SS7)
- V5.2
- SORM

IP protocols

- SIP, SIP-T, SIP-I
- H.248/MEGACO*
- SIGTRAN (M2UA, M3UA, IUA)*
- MGCP*

I/O Interfaces

TDM Telephony:

- E1 (16 x RJ-45)

IP:

- 2 x 1000 Base-X (SFP)
- 3 x 10/100/1000 Base-T (RJ-45)

USB, 2 x e-SATA:

- Cacti reporting
- Storage of Accounting data
- Music on hold (MOH) server*
- External clock input

SIP/SIP-T/SIP-I Related Specifcations

- RFC 3261 SIP
- RFC 3262 Reliability of Provisional responses in SIP
- RFC 3263 Locating SIP servers for DNS
- RFC 4566 Session Description Protocol
- RFC 2976 SIP INFO (for DTMF transmission)
- RFC 3204 MIME Media Types for ISUP and QSIG (ISUP support)
- RFC 3264 SDP Offer/Answer Model
- RFC 3265 SIP Notify
- RFC 3311 SIP Update
- RFC 3323 Privacy Header
- RFC 3325 P-Asserted-Identity
- RFC 3326 SIP Reason Header
- RFC 3372 SIP for Telephones (SIP-T)
- RFC 3398 ISUP/SIP Mapping
- RFC 3515 SIP REFER
- RFC 3581 Symmetric Response Routing
- RFC 3666 SIP to PSTN Call Flows
- RFC 3891 SIP Replaces Header
- RFC 3892 SIP Referred-By Mechanism
- RFC 4028 SIP Session Timer
- RFC 5806 SIP Diversion Header
- 302 Responses support
- SIP OPTIONS Keep-Alive (Busy Out)
- Q.1912.5 SIP-I
- SIP, SIP-I and SIP-T Interaction
- SIP Back to Back User Agent
- SIP registrar (according to the customer request)
- NAT support (comedia mode)

*Currently not supported

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