– **Scalable 1U platform**
– **IP PBX for up to 2,000 subscribers with VAS support**
– **High quality voice processing**
– **Carrier-grade reliability**
– **Up to 768 VoIP channels**
– **Up to 16 streams E1**
– **Support for up to 2 embedded 8 GB SSDs**

The **SMG-1016M platform** can be used as a trunk gateway for connecting signal and media flows of TDM and VoIP networks, IP PBX with support for VAS functions, and can also act as a universal solution for building new generation infocommunication networks. The wide function set, strict compliance with requirements and standards, as well as carrier-grade reliability allow service providers and carriers to solve most part of their objectives on the basis of SMG-1016M.

Scalability

SMG-1016M is a beneficial investment in the future of your project due to its scalability. The gateway supports up to 16 E1 streams (SS7, PRI, V5.2) and up to 768 VoIP channels.

IP PBX with VAS support

Additional options for SMG-1016M gateway allow using it as a full-featured IP PBX for up to 2,000 SIP subscribers with support for a wide range of value added services. A programmable IP PBX module ECSS-10 is designed for fast deployment of a VoIP node with a minimum of capital expenses (CAPEX). ECSS-10 and SMG-1016M might be used as a PBX of any level.

Carrier-grade reliability

Uniform load distribution between submodules, redundant power supplies, as well as the use of modern technologies based on parallel computing provide a high level of fault tolerance of the SMG-1016M trunk gateway with automatic switching to a backup submodule in the event of any system submodule failure or the power source.

Functional compatibility

The strict compliance with requirements of up-to-date protocols, recommendations and standards provides functional compatibility of SMG-1016M with a variety of equipment: digital PBX, IP PBX, Softswitch, VoIP gateways, SIP phones, software SIP clients, etc.

Media streams transcoding

The hardware transcoding helps to negotiate media streams with different VoIP codecs which are used in up-to-date networks.

RADIUS routing

Intellectual call routing based on billing system responses via the RADIUS protocol allows creating flexible methods of call processing.

Intellectual protection of IP networks

The intellectual protection against unauthorized external SIP subscribers connection and connections via http/https/telnet/ssh is realized on the SMG-1016M (Dynamic Firewall, Static Firewall, black and white lists of IP addresses and subnets, etc.). For additional defense, SMG-1016M is compatible with session border controllers (e.g. SBC-1000) that are used as a firewall for VoIP networks.

Features and capabilities

Calls management
– Interaction with STUN-server on the SIP interface
– Routing based on called number (CdPN) and/or calling number (CgPN)
– Routing by the access category
– Number modifications before and after routing
– Call recording according to number mask and dialplan[1]
– Use of multiple dialplans
– Subscriber lines restriction
– Subscriber service mode settings
– Trunk group cut-off
– Call management via RADIUS[1]
– Direct forwarding for trunk groups
– Prefix for several trunk groups
– Interactive Voice Response (IVR)[1]
– Uploading/downloading of configuration as a single file
– Lines limiting for SIP interface
– Egress and ingress lines restrictions for a subscriber
– Ingress load limiting CPS (calls per seconds) for a trunk group

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

Video processing
– Video stream transmitting in the Video Offroad mode

Fax transmission
– T.38 Real-Time Fax, G.711 (a-law, μ-law) pass-through

Voice standards
– VAD (Voice Activity Detection)
– CNG (Comfort Noise Generation)
– AEC (Acoustic Echo Cancellation, G.168 recommendation)
– AGC (Automatic Gain Control)

Quality of service (QoS)
– Diffserv and 802.1p priorities assignment for SIP and RTP
– Dynamic and static jitter buffer
– Ingress/egress traffic rate limiting

DTMF
– INBAND, RFC 2833, SIP INFO, SIP NOTIFY transmission methods

Billing
– Billing data is recorded in CDR file. Simultaneously, CDR file is recorded to a local HDD and remote FTP server
– RADIUS Accounting
– Supported billing systems: Hydra Billing, LANBilling, PortaBilling, NetUP, BGBilling (possible integration with other systems)

Flexibility
– Multiple network interfaces creation for telephony (SIP, RTP) with different IP addresses
– Operation with multiple numbering plans
– Signal SS7 channel redundancy
– Voice activity control (by the presence of RTP or RTCP)
– Individual routing for streams of a single SS7 linkset

TDM protocols
– SS7
– PRI (Q.931)
– Q.699 (PRI and SS7 interaction)
– V5.2 LE[1]
– V5.2 AN[1]

VoIP protocols
– SIP, SIP-T/SIP-I, SIP-Q
– H.323[1]
– SIGTRAN (M2UA, IUA)[2]
– H.248[2]

Capacity
– Up to 768 VoIP channels
– Up to 16 E1 streams (CENTRONICS-36)
– Maximum load intensity 14 cps

Interfaces
– 2 × 1000BASE-X ports (2 slots for SFP modules)
– 3 × 10/100/1000BASE-T ports (RJ-45)
– E1 (2 × CENTRONICS-36 connectors)
– 1 USB 2.0 port
– 1 console port (RS-232)
– 2 × SATA ports (for SSD storage modules)

Phone book
– Retrieving Display Name from LDAP server

Management and monitoring
– E1 and VoIP channels monitoring in web interface
– Management of channels and SS7 links in web interface
– Alarm logging with the opportunity to save entries to syslog server
– Storing traces on SSD and USB drives
– Emergency notification through SNMP
– Automatically enable logging after the gateway restart
– Monitoring of web interface active user sessions

[1] *Optional.*
[2] *Not supported in the current firmware version.*

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Features and capabilities (continued)

Security

– Black and white IP addresses lists
– Logging of all access attempts to the device
– Automatic blocking by IP address after unsuccessful login attempts and/or access via http/https/telnet/ssh
– List of permitted IP addresses for access to control the device
– Access rights delimitation – admin/user
– Delimitation of access rights to calls records
– Control of opposite RTP stream source IP address
– Digest authentication (RFC 5090, Draft-Sterman)
– Digest authentication in RADIUS (RFC 5090, Draft-Sterman)

Value added services[1]

– Call Forwarding
  – Call Forwarding Out of Service (CFOS)
  – Call Forwarding on No Reply (CFNR)
  – Call Forwarding Unconditional (CFU)
  – Call Forwarding on Busy (CFB)
  – Forwarding by day of week and time of day
– Call Transfer
– Music on Hold (MOH)
– Call Hold
– SIP-forking support for SIP subscribers
– Voice Notification
– Call Hunt
– Call Pickup
– Call Parking
– Busy Lamp Field
– Conference add-on (CONF)
– Conference for a list of subscribers
– 3-Way conference
– Intercom
– Paging
– Outgoing calls restrictions (Out Calls Restrict)
– Egress communication by password (RBP)
– Password activation (PWD ACT)
– Password reset (PWD)
– Voice mail
– One Touch Record
– Do Not Distrurb (DND)
– Blacklist
– Anonymous call
– Reject anonymous calls
– Reminder

Advanced SIP/SIP-T/SIP-I functionality

– Registration and authentication of 2,000 SIP subscribers[1]
– VAS support for 1,000 SIP subscribers[1]
– SIP and SIP-T/SIP-I interaction
– Trunking and subscriber registration of SIP trunks
– Transit registration of subscribers on SIP trunk with switching to local service mode in case of server unavailability

Physical specifications and environmental parameters

Operating temperature range	from 0 to +40 °C	
Relative humidity	up to 80 %	
Noise level	from 44 to 60 dB	
Supply voltage	DC: 36–72 V AC: 100–240 V, 47–63 Hz Power options: – 1 AC/DC power supply; – 2 hot-swappable AC/DC power supplies.	
Power modules	DC, power module PM100-48/12 100 W	AC, power module PM160-220/12 160 W
Power consumption	no more than 50 W	
Dimensions (W × H × D)	430 × 45 × 260 mm	
Form factor	19", 1U	
Weight	3.2 kg	

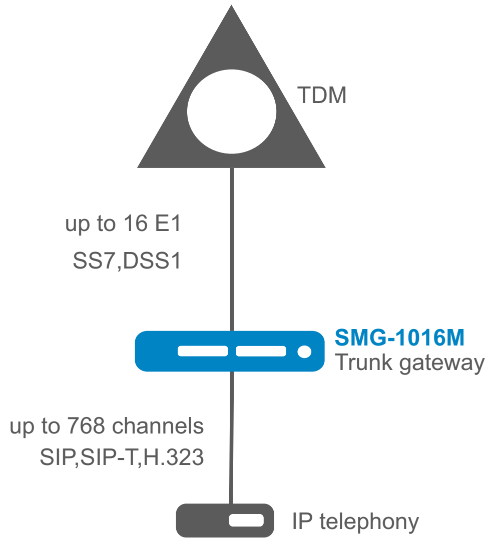
[1] *Optional.*

Use cases

Protocols converter

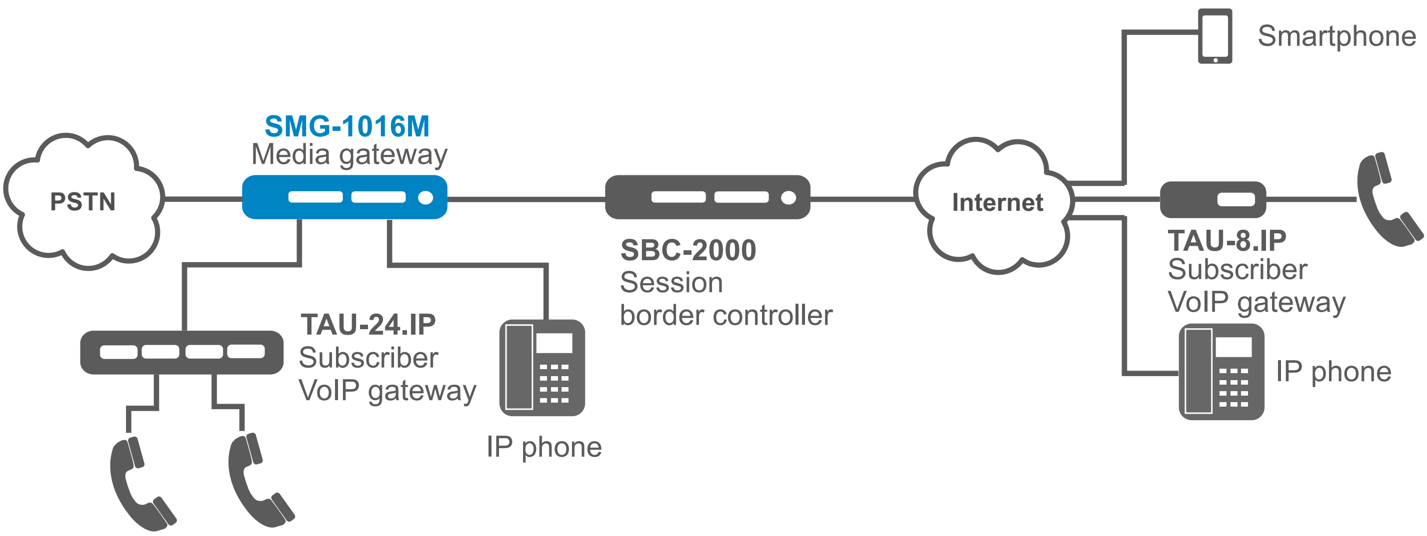
The wide range of supported TDM and VoIP protocols allows using SMG-1016M for signal and media streams negotiation in different directions:

- VoIP <-> VoIP
- VoIP <-> TDM
- TDM <-> VoIP
- TDM <-> TDM



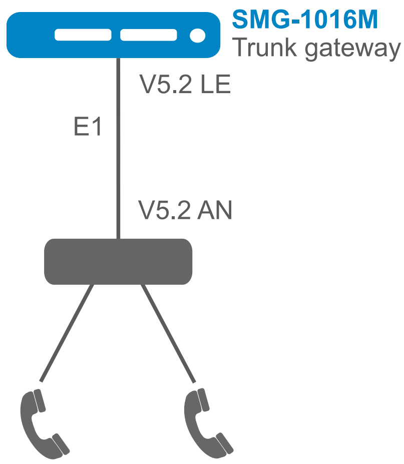
Local communication node

The additional options of software module IP PBX ECSS-10 (SMG1-PBX-2000 and SMG1-VAS-500) allow using SMG-1016M with up to 2000 subscribers connection at the initial stage of local communication node as a full-function PBX with support of VAS. In case of extending of the capacity, there is an opportunity of migration to a full-featured server solution ECSS-10 with the support of a multi-layer redundancy and flexible scalability of the elements.



Outstation via V5.2 protocol

The additional options of software module IP PBX ECSS-10 (SMG1-V5.2LE, SMG1-VAS-500) allow organizing outstation via V5.2 protocol and service up to 2,000 subscribers with support of a full VAS set. Equipment of any manufacturer that supports V5.2 might be used as an outstation.



Ordering information

Name	Description
SMG-1016M	SMG-1016M digital gateway chassis: 4 slots for C4E1 submodules, 6 slots for SM-VP-M300 submodules, 2 slots for PM160-220/12 and PM100-48/12 power modules
SMG-1016M modules	
SM-VP-M300	SM-VP-M300 submodule with support of 128 VoIP channels (G.711)
C4E1	C4E1 submodule with support of up to 4 E1 streams
SSD-8Gb	Embedded SSD for SMG-1016M equipment, 8 GB, Form factor: 44×30 mm, 22P/90D
PM160-220/12	PM160-220/12 power module, 220 V AC, 160 W
PM100-48/12	PM100-48/12 power module, 48 V DC, 100 W
Cables	
UTP-18-X	UTP-18-X cable: 18-pair cable, X meters length, terminated with CENTRONICS-36 connectors (X=4, 6, 12, 20, 30)
SMG-1016M options	
SMG1-PBX-2000	Activation of ECSS-10 module for 2,000 SIP registrations with BLF function on the SMG-1016M digital gateway
SMG1-VAS-500	Extension for SMG1-PBX-2000 option: activation of standard VAS set for 500 subscribers of digital gateway SMG-1016M
SMG1-H323	Activation of H.323 (without Gatekeeper function) on the SMG-1016M digital gateway
SMG1-RCM	Activation of Radius CallManagement functionality on the SMG-1016M digital gateway
SMG1-VNI-40	Extension of VLAN interfaces to 40 on the SMG-1016M digital gateway
SMG1-REC	Activation of Call Recording functionality on the SMG-1016M digital gateway
SMG1-CORP	Activation of ECSS-10 module for 500 SIP registrations with VAS on the SMG-1016M digital gateway
SMG1-VNS	Activation of Voice Notification System (VNS) functionality on the SMG-1016M digital gateway
SMG1-AUTH-CALL	Activation of "Authorization by callback" functionality
SMG1-IVR	Activation of IVR functionality
SMG1-V5.2LE	Organization of outstation via V5.2LE on the SMG-1016M digital gateway
SMG1-V5.2AN	Organization of outstation via V5.2AN on the SMG-1016M digital gateway
Discounted option sets for SMG-1016M	
SMG1-SP2	"PBX+VAS" set, includes 3 options for one gateway SMG-1016M: 1×SMG1-PBX-2000 and 2×SMG1-VAS-500
SMG1-SP4	"Triple" set, includes 3 options for one SMG-1016M: SMG1-H323, SMG1-RCM and SMG1-VNI-40

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About ELTEX

ELTEX company is a leading Russian developer and manufacturer of communications equipment with 30 years of history. Complete solutions and their seamless integrability into the Customer's infrastructure are the priority growth areas of the company.