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

SMG-1016M platform is used as a trunk gateway for interfacing of signal and media streams of TDM and VoIP networks. The gateway also might be used as an IP PBX with value added services (VAS) support and a multipurpose solution for infocommunication new generation networks (NGN). 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 provides the opportunity to evenly distribute investments for scaling throughout the entire project implementation period. 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 you to create 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 subnetworks, 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.

ELTEX
Integrated networking solutions

Features and capabilities

Calls management
– 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]
– Lines limiting for SIP interface
– Egress and ingress lines restrictions for a subscriber
– Ingress load limiting (calls per seconds) for a trunk group
– Interaction with the STUN server on the SIP interface

Voice codecs
– G.711 (a-law, µ-law), G.722, G.729 (A/B), G.723.1, G.726 (32 Kbps)

Video processing
– Video stream transmission in Video Offroad, Video Transit modes

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
– Outgoing/incoming traffic speed restriction

DTMF
– INBAND, RFC 2833, SIP INFO, SIP NOTIFY transmission methods
– Auto-detection of the DTMF receiving method

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

Flexibility
– Downloading and uploading configuration as a single file
– Downloading and uploading licenses as a single file
– Downloading and uploading subscriber settings in a single file
– Multiple network interfaces creation for telephony (SIP, RTP) with different IP addresses
– Operation with multiple dialplans
– 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[2]

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

Capacity and performance
– Up to 768 VoIP channels
– Up to 16 E1 streams (CENTRONICS-36)
– RAM: 512 MB
– Maximum load intensity[3]:
  – With SIP-CPS license: 15 SIP-E1 calls per second
  – With SIP-CPS license: 45 SIP-SIP calls per second
  – Without SIP-CPS license: 7 SIP-SIP calls per second

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

Phone book
– Obtaining the displayed name from the LDAP server

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

Security
– Black and white IP addresses lists
– Output to syslog of all attempts to access the device
– Automatic blocking by an IP address after unsuccessful login and/or by access attempts 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 for opposite RTP stream source IP address
– Digest authentication (RFC 5090, Draft-Sterman)
– Digest authentication in RADIUS (RFC 5090, Draft-Sterman)
– Verifying the reliability of the WEB user password
– WEB user password validity period
– TACACS+

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[1] *Optional.*
[2] *Not supported in the current firmware version.*
[3] *The values are specified for trunk mode operation (without registrations, subscriptions, or VAS use), with a load of one-second SIP-SIP calls. Only incoming call legs were taken into account in the calculation.*

Integrated networking solutions

Features and capabilities (continued)

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 (CFT)
- Call Transfer
- Music on Hold (MOH)
- Call Hold
- SIP-forking support for SIP subscribers
- Voice Notification
- Call Parking
- Voice mail
- Call Hunt
- Call Pickup
- Busy Lamp Field
- Subscriber registration status indicator (Presence)
- Message Waiting Indicator
- 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 Disturb (DND)
- Blacklist
- Anonymous call
- Reject anonymous calls
- Reminder
- Call Waiting
- Do not disturb in the call group (CGDND)
- Auto-dial
- Auto-dial with callback
- Chief secretary (only with VAS-ACG license)
- Current system time
- Interference in conversation

Advanced SIP/SIP-T/SIP-I functionality

- Registration and authentication of up to 2000 SIP subscribers[1]
- VAS support for 1000 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 a local servicing 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	
Power supply	DC: 36–72 V AC: 100–240 V, 47–63 Hz Power supply options: – 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	

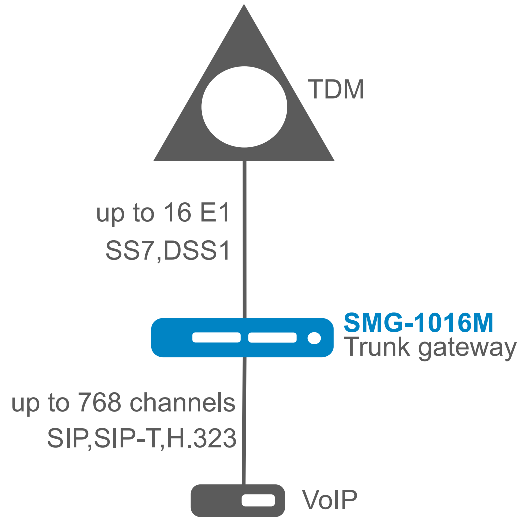
[1] *Optional.*

Application diagrams

Protocol converter

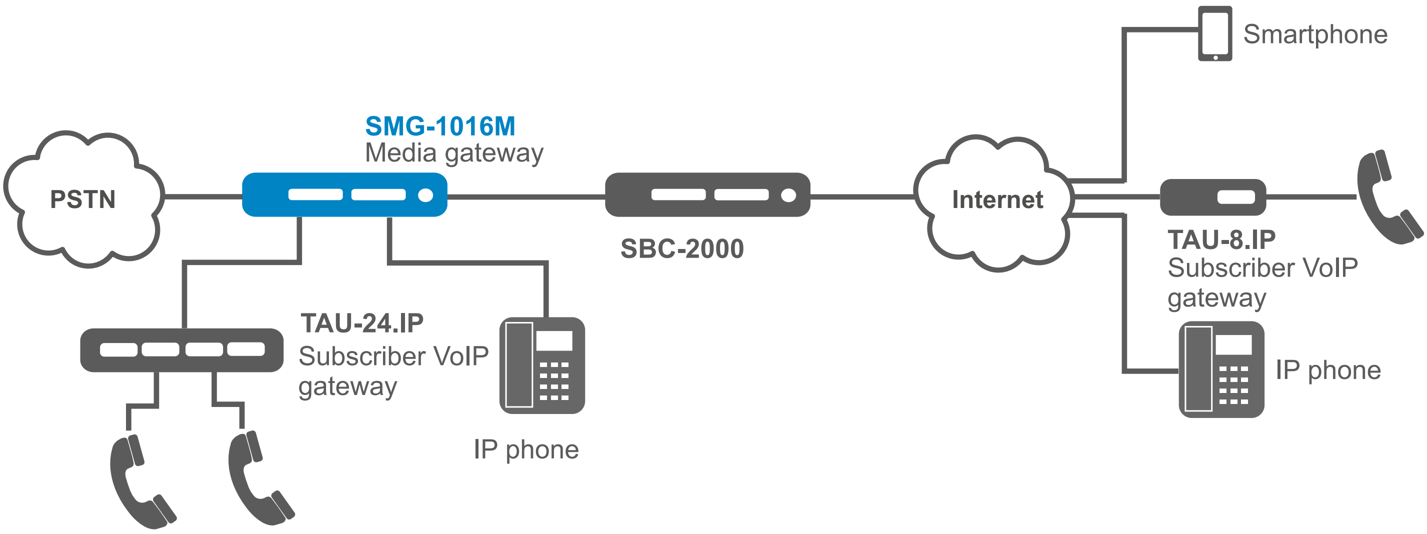
A wide range of supported TDM and VoIP protocols allows the SMG-1016M to be used for signaling and media stream coordination in various directions:

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



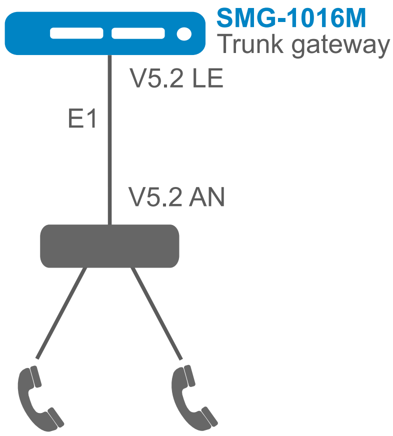
Local exchange

Enabling additional options of the ECSS-10 IP PBX software module (SMG1-PBX-2000, SMG1-VAS-500) makes it possible to use the SMG-1016M at the initial stage of building a local communication node with a capacity of up to 2,000 SIP subscribers as a fully functional PBX supporting the basic set of supplementary services. As the platform's capacity grows and the list of provided services needs to be expanded, the SMG-1016M can be migrated to a full-featured ECSS-10 softswitch server solution with multi-level redundancy support and flexible scaling of all components.



Outstation via V5.2 protocol

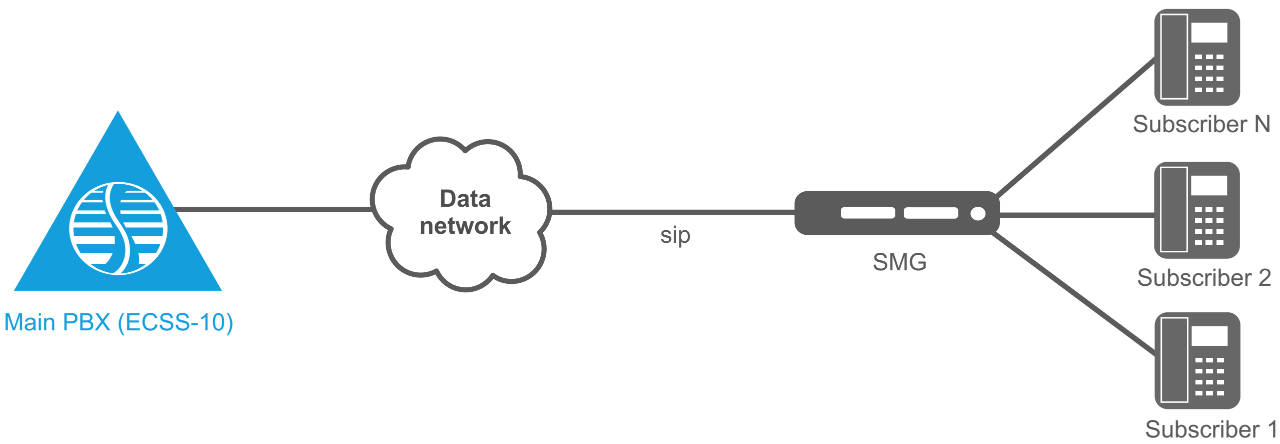
The additional options of IP PBX software module ECSS-10 (SMG1-V5.2LE, SMG1-VAS-500) allow clients to organize outstation via V5.2 protocol and service up to 2,000 subscribers with support for a full VAS set. Equipment of any manufacturer that supports V5.2AN might be used as an outstation.



Application diagrams

Transit registration

There are several subscribers connected at a remote site (e.g., a company branch). Under normal operating conditions, calls are handled by the central ECSS-10 PBX (which also provides call waiting, call recording, etc.). However, if the connection to the PBX is lost, subscribers at the remote site lose telephone service. It is necessary to ensure that subscribers retain telephone service in the event of a communication failure with the central PBX. To achieve this, the SMG IP PBX installed at the remote site can be configured for transit registration mode, in which subscribers will register with the main PBX when it is available, and with the SMG itself when the main PBX is unavailable. In this case, the SMG acts as a "survivor server," ensuring call routing. Another SMG can serve as a replacement for the central ECSS-10 PBX.



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 for up to 128 VoIP channels (G.711)
C4E1	C4E1 submodule with support for up to 4 E1 streams
SSD-8Gb	Internal SSD drive for the SMG-1016M, 8 GB, form factor: 44×30 mm, 22P/90D
PM160-220/12	PM160-220/12 power module, 220 V AC, 160W
PM100-48/12	PM100-48/12 power module, 48 V DC, 100 W
Cabels	
UTP-18-X	UTP-18-X cable: 18-pair cable X meters long, 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 support on the SMG-1016M digital gateway
SMG1-VAS-500	Extension of SMG3-PBX-2000 option: activation of standard VAS set for 500 subscribers on the SMG-1016M digital gateway
SMG1-H323	Activation of H.323 (without Gatekeeper) on the SMG-1016M digital gateway
SMG1-RCM	Activation of Radius Call Management 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 an outstation V5.2LE on the digital gateway SMG-1016M
SMG1-V5.2AN[1]	Organization of an outstation V5.2AN on the digital gateway SMG-1016M
SMG1-SIP-CPS	Unblocking the limit on the number of calls per second (SIP)
SMG1-VAS-ACG	Activation of the chief secretary feature

[1] *V5.2AN requires specialized software to operate.*

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About Eltex

Eltex Enterprise is a leading Russian developer and manufacturer of communication 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.