

- Scalable platform 1U
- IP PBX for 3,000 subscribers with VAS support
- High-quality voice processing
- Carrier-grade reliability
- Up to 768 VoIP channels
- Up to 16 E1 streams (RJ-48)
- Support for 2 HDD SATA 2.5"
- Hardware redundancy



**Hybrid platform SMG-3016** 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-3016.

#### Scalability

SMG-3016 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-3016 gateway allow using it as a full-featured IP PBX for up to 3,000 SIP subscribers with support for a wide range of value added services. A program-mable 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-3016 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-3016 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-3016 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.

#### 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-3016 (Dynamic Firewall, Static Firewall, black and white lists of IP addresses and subnetworks, etc.). For additional defense, SMG-3016 is compatible with session border controllers (e.g. SBC-3000) that are used as a firewall for VoIP networks.

#### RADIUS routing

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

## 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<sup>1</sup>
- Use of multiple dialplans
- Subscriber lines restriction
- Subscriber service mode settings
- Trunk group cut-off
- Call management via RADIUS<sup>1</sup>
- Direct forwarding for trunk groups
- Prefix for several trunk groups
- Interactive Voice Response (IVR)<sup>1</sup>
- 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,  $\mu$ -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,  $\mu$ -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

### 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 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
- 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<sup>1</sup>

### VoIP protocols

- SIP, SIP-T/SIP-I, SIP-Q
- H.323<sup>1</sup>
- SIGTRAN (M2UA, IUA)<sup>2</sup>
- H.248<sup>2</sup>

### Capacity and performance

- Up to 768 VoIP channels
- Up to 16 E1 streams (RJ-48)
- Maximum load intensity — 120 cps
- RAM:
  - 2 GB
  - 8 GB — for PCB rev. B

### Interfaces

- 16 × E1 ports (RJ-48)
- 2 × 10/100/1000BASE-T ports (RJ-45)/1000BASE-X (SFP)
- 2 × 10/100/1000BASE-T ports (RJ-45)
- 2 × USB 2.0 ports
- 2 × slots for SATA HDD 2.5"

### 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
- Console port RS-232 (RJ-45)
- Allocated management port (OOB) 10/100/1000BASE-T (RJ-45)
- 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

<sup>1</sup> Optional.

<sup>2</sup> Not supported in the current firmware version.

## Features and capabilities (continued)

- 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
  - Authentication of subscribers on RADIUS server and SIP registrar
  - Digest authentication (RFC 5090, Draft-Sterman)
  - Digest authentication in RADIUS (RFC 5090, Draft-Sterman)
  - Checking the WEB user password reliability
  - The validity time of the WEB user password
- Advanced SIP/SIP-T/SIP-I functionality**
- Registration and authentication of up to 3,000 SIP subscribers<sup>1</sup>
  - VAS support for up to 3,000 SIP subscribers<sup>1</sup>
  - 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
- Redundancy**
- Operation in light redundancy mode 1+1
  - Automatic enabling reserve
  - Automatic synchronization of main redundant module settings
- Value added services<sup>1</sup>**
- 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 Parking
  - Voice mail
  - Call Hunt
  - Call Pickup
  - Busy Lamp Field
  - Subscriber registration status indicator (Presence)
  - 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)
  - Do Not Disturb (DND)
  - Blacklist
  - One Touch Record
  - Anonymous call
  - Reject anonymous calls
  - Reminder
  - Call Waiting
  - Do not disturb in the call group (CGDND)
  - Auto-dial
  - Auto-dial with callback

## Physical specifications and environmental parameters

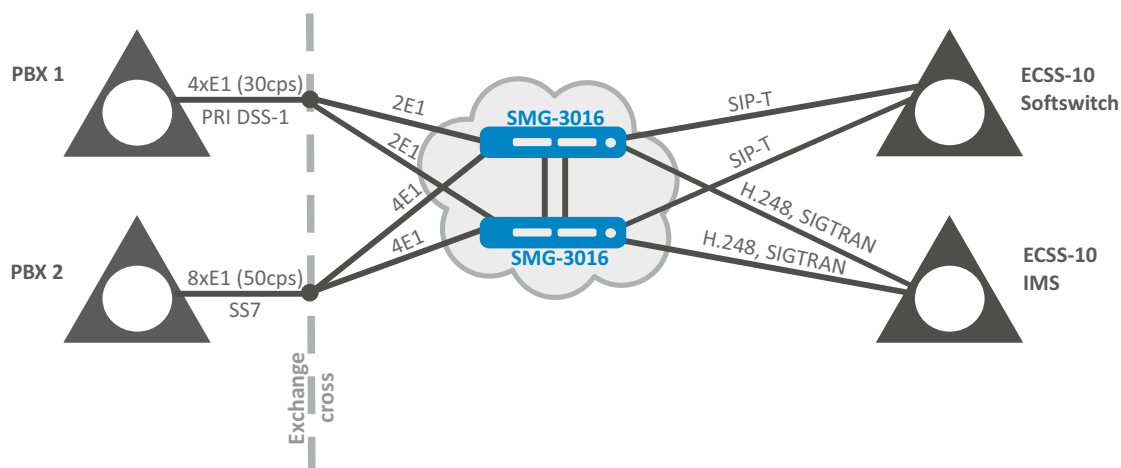
Operating temperature range	from 0 to +40 °C	
Relative humidity	up to 80 %	
Noise level	from 44 to 60 dB	
Power supply	AC: 100–240 V, 47–63 Hz DC: 36–72 V Power supply options: – AC/DC power supply; – 2 hot-swappable AC/DC power supplies.	
Power modules	AC, power module PM160-220/12 160 W	DC, power module PM100-48/12 100 W
Power consumption	up to 50 W	
Dimensions (W × H × D)	430 × 45 × 340 mm	
Form factor	19", 1U	
Weight	5.3 kg	

<sup>1</sup> Optional.

## Application diagrams

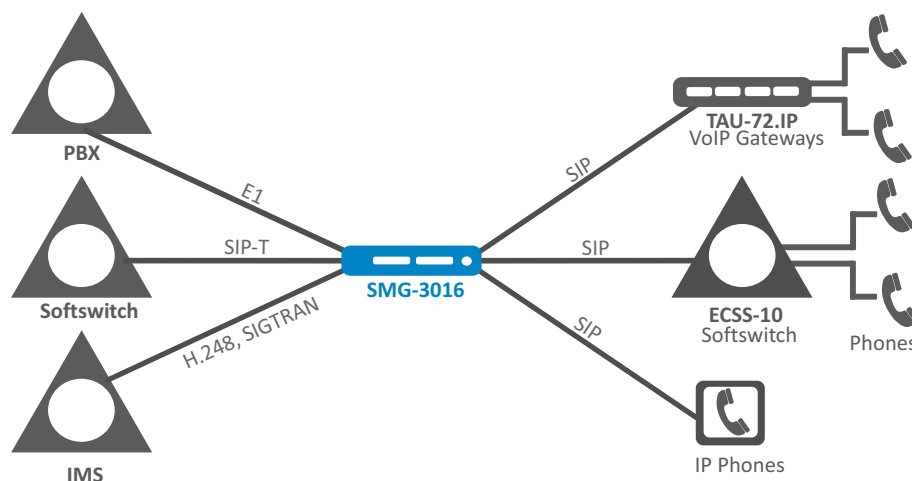
### High-load transit nodes

The high performance and hot-swap capability allow using SMG-3016 at nodes with a high load intensity. Redundancy of TDM connections is implemented due to E1 streams duplication, while VoIP connection redundancy is performed by switching to the available SMG-3016 gateway.



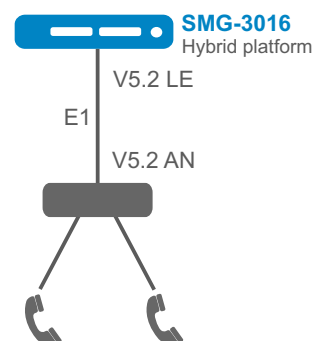
### Terminal network node

The trunk gateway SMG-3016 might be used for organization of a single node for connection of PSTN to several electronic PBX as well as for subscribers connection via VoIP gateways (e. g. TAU-72.IP).



### Outstation via V5.2 protocol

The additional options of IP PBX software module ECSS-10 (SMG3-V5.2LE, SMG3-VAS-1000) allow clients to organize outstation via V5.2 protocol and service up to 3,000 subscribers with support for a full VAS set. Equipment of any manufacturer that supports V5.2AN might be used as an outstation.




## Ordering information

Name	Description
<b>SMG-3016</b>	SMG-3016 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
<b>SMG-3016 modules</b>	
<b>SM-VP-M300</b>	SM-VP-M300 submodule with support for up to 128 VoIP channels (G.711)
<b>C4E1</b>	C4E1 submodule with support for up to 4 E1 streams
<b>PM160-220/12</b>	PM160-220/12 power module, 220 V AC, 160W
<b>PM100-48/12</b>	PM100-48/12 power module, 48 V DC, 100 W
<b>SMG-3016 options</b>	
<b>SMG3-PBX-3000</b>	Activation of ECSS-10 module for 3,000 SIP registrations with BLF support on the SMG-3016 digital gateway
<b>SMG3-CORP-1000</b>	Activation of ECSS-10 module for 1,000 SIP registrations with VAS on the SMG-3016 digital gateway
<b>SMG3-VAS-1000</b>	Extension of SMG3-PBX-3000 option: activation of standard VAS set for 1000 subscribers on the SMG-3016 digital gateway
<b>SMG3-REC</b>	Activation of Call Recording functionality on the SMG-3016 digital gateway
<b>SMG3-VNS</b>	Activation of Voice Notification System (VNS) functionality on the SMG-3016 digital gateway
<b>SMG3-VNS-EXT</b>	Activation of the functionality of the voice notification system with an expanded number of objects (notification tasks, lists of numbers, voice messages)
<b>SMG3-AUTH-CALL</b>	Activation of "Authorization by callback" functionality
<b>SMG3-H323</b>	Activation of H.323 (without Gatekeeper) on the SMG-3016 digital gateway
<b>SMG3-H323-EXT</b>	Activation of H.323 local Gatekeeper functionality
<b>SMG3-H323-GK</b>	Activation of H323 local Gatekeeper functionality on the SMG-3016 digital gateway
<b>SMG3-RCM</b>	Activation of Radius Call Management functionality on the SMG-3016 digital gateway
<b>SMG3-VNI-40</b>	Extension of VLAN interfaces to 40 on the SMG-3016 digital gateway
<b>SMG3-IVR</b>	Activation of IVR functionality
<b>SMG3-RESERVE</b>	Activation of redundancy by IP in master-slave mode on the SMG-3016 platform
<b>SMG3-RESERVE-E1</b>	SMG3-RESERVE-E1 option to activate E1 redundancy on the SMG-3016 platform
<b>SMG3-V5.2LE</b>	Organization of an outstation V5.2LE on the digital gateway SMG-3016
<b>SMG3-MSR</b>	Activation of the software Media Server (MSR) functionality
<b>Discounted option sets for SMG-3016</b>	
<b>SMG3-SP2</b>	"PBX+VAS" set, includes 2 options for one gateway SMG-3016: 1×SMG3-PBX-3000 and 1×SMG3-VAS-1000
<b>SMG3-SP4</b>	"Triple" set, includes 3 options for one SMG-3016: SMG3-H323, SMG3-RCM and SMG3-VNI-40
<b>SMG3-SP5</b>	"VAS-3000" includes 3 options for one SMG-3016: 3×SMG3-VAS-1000
<b>SMG3-SP7</b>	"VAS-2000" includes 2 options for one SMG-3016: 2×SMG3-VAS-1000

### Contact us

### About Eltex



+7 (383) 274 10 01  
+7 (383) 274 48 48



eltex@eltex-co.ru



www.eltex-co.com

**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.