- **Scalable platform 1U**
- **Quad-Core ARMv8 Marvell Armada 7040 Processor**
- **IP PBX for 3 000 subscribers with VAS support**
- **High-quality voice processing**
- **Carrier class 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 and carrier class reliability allow service providers and carriers to solve most part of their objectives on the basis of SMG-3016.

Scalability

SMG-3016 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-3016 gateway allow using it as a full-featured IP PBX for up to 3000 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. ECSS-10 and SMG-3016 might be used as a PBX of any level.

Functional compatibility

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

Carrier class reliability

SMG-3016 provides high level of fault tolerance due to embedded state-of-the-art Quad-Core ARMv8 Marvell Armada 7040 processor, uniform load distribution among submodules, usage of up-to-date technologies based on parallel computing and power modules redundancy. In case of a primary submodule fault, the gateway switches to a backup submodule.

Media streams transcoding

The hardware transcoding based on Mindspeed Technologies media codecs 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-1000) 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
– Interaction with STUN-server on the SIP interface
– Routing based on called number (CdPN) or calling number (CgPN)
– 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 (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)

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 (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

Billing
– Billing data is recorded as a CDR file that is kept on a local HDD and remote FTP server simultaneously.
– RADIUS Accounting
– Supported billing systems: Hydra Billing, LANBilling, PortaBilling, NetUP, BGBilling (there is an opportunity of integration with other systems)

Flexibility
– 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[2]
– 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 (RJ-48)
– Maximum load intensity — 120 cps
– Quad-Core ARMv8 Marvell Armada 7040 1.4 GHz
– RAM 2 GB

Interfaces
– 16 E1 ports (RJ-48)
– 2 ports of 10/100/1000Base-T (RJ-45) / 1000Base-X (SFP)
– 2 ports of 10/100/1000Base-T (RJ-45) ports
– 2 USB2.0 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
– Tracings are stored on HDD and USB storages
– Emergency notification through SNMP
– Console port RS-232 (RJ-45)
– Allocated management port (OOB) 10/100/1000BASE-T (RJ-45)

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[1] *Optional*
[2] *Not supported in the current firmware version — 3.15.0*

Features and capabilities

Security

– Black and white IP addresses lists
– Attempts of accessing the device are logged
– Automatic blocking by an IP address after unsuccessful login or/and 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 rights to access calls records
– Control for opposite RTP stream's source IP address
– Authentication of subscribers on RADIUS server and SIP registar
– Digest authentication (RFC 5090, Draft-Sterman)
– Digest authentication in RADIUS (RFC 5090, Draft-Sterman)

Advanced SIP/SIP-T/SIP-I functionality

– Registration and authentication of up to 3000 SIP subscribers[1]
– VAS support for up to 3000 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

Redundancy[2]

– Operation in warm redundancy mode 1+1
– The system switches the redundant part on automatically
– Automatic synchronization of main redundant module settings

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)
– Call Transfer
– Music on Hold (MOH)
– Call Hold
– Call Hunt
– Call Pickup
– Busy Lamp Field
– Conference add-on (CONF)
– Conference for a list of subscribers
– 3-Way conference
– Intercom
– Paging
– Outgoing calls restrictions
– Egress communication by password (RBP)
– Password activation (PWD ACT)
– Password reset (PWD)
– Do not disturb
– Blacklist

[1] *Optional*
[2] *Not supported in the current firmware version — 3.15.0*

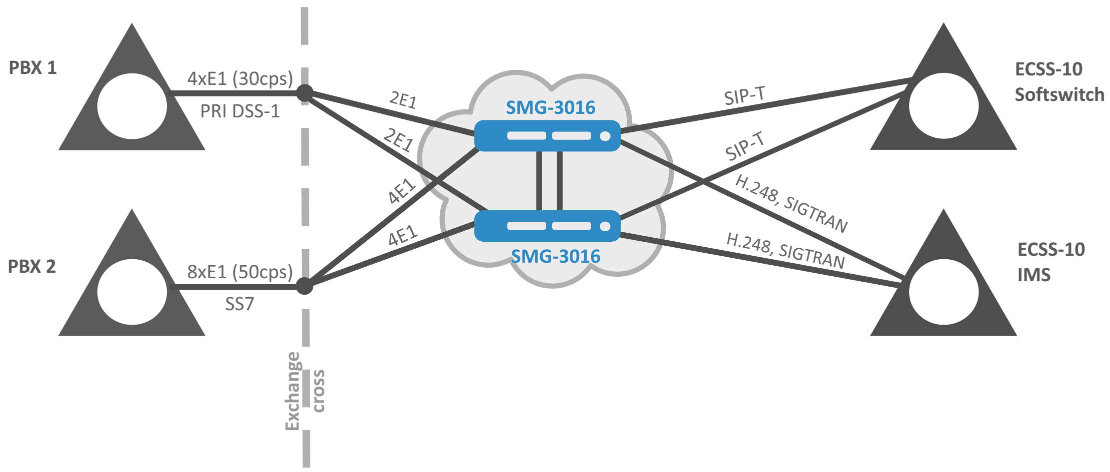
Physical specifications and environmental parameters

Range of operating temperatures	from 0 to +40°C	
Relative humidity	80% max.	
Noise level	from 44 to 60 dB	
Power supply	AC: 220V+-20%, 50 Hz DC: -48V+30%-20% Power supply options: -1 AC/DC power supply; -2 hot-swappable AC/DC power supplies.	
Источники питания	AC, power module PM160-220/12 160W	DC, power module PM100-48/12 100W
Max. power consumption	50W	
Dimensions (W x H x D)	430 x 45 x 340 mm	
Constructive	19", 1U	
Nett weight	5,3 kg	

Application diagrams

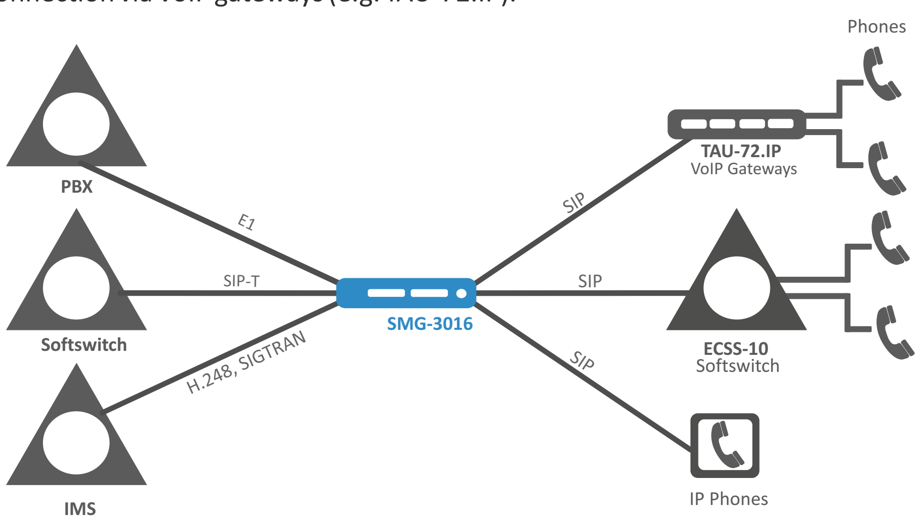
High-load transit nodes

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.



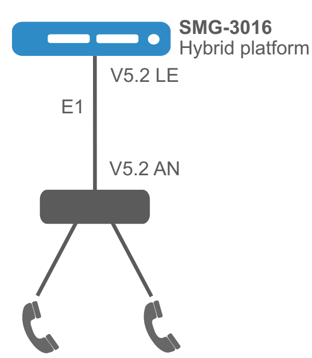
Terminal network node

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 3000 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	Image
SMG-3016	Digital gateway chassis SMG-3016: 4 slots for C4E1 submodules, 6 slots for SM-VP-M300 submodules, 2 slots for PM160-220/12 and PM100-48/12 power modules	
Modules of SMG-3016		
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	
PM160-220/12	PM160-220/12 power module, 220 VAC, 160W	
PM100-48/12	PM100-48/12 power module, 48 VDC, 100W	
Options for SMG-3016		
SMG3-PBX-3000	Activation of ECSS-10 module for 3000 SIP registrations with BLF support on the digital gateway SMG-3016	
SMG3-VAS-1000	Extension of SMG3-PBX-3000 option: activation of standard VAS set for 1000 subscribers on the digital gateway SMG-3016	
SMG3-H323	Activation of H.323 (without Gatekeeper) on the digital gateway SMG-3016	
SMG3-RCM	Activation of Radius Call Management functionality on the digital gateway SMG-3016	
SMG3-VNI-40	Extension of number of VLAN interfaces to 40 on the digital gateway SMG-3016	
SMG3-REC	Activation of Call Recording functionality on the digital gateway SMG-3016	
SMG3-CORP	Activation of ECSS-10 module for 1000 SIP registrations with VAS support on the digital gateway SMG-3016	
SMG3-V5.2LE	Organization of an outstation V5.2LE on the digital gateway SMG-3016	
SMG3-V5.2AN	Organization of an outstation V5.2AN on the digital gateway SMG-3016	
SMG3-RESERVE	Activation of redundancy via IP in master-slave mode on SMG-3016	
Discounted option sets for SMG-3016		
SMG3-SP2	"PBX+VAS" set, includes 2 options for one gateway SMG-3016: 1xSMG3-PBX-3000 and 1xSMG3-VAS-1000	
SMG3-SP4	"Triple" set, includes 3 options for one SMG-3016: SMG3-H323, SMG3-RCM and SMG3-VNI-40	

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About Eltex

Eltex company is a leading Russian developer and manufacturer of telecommunication equipment with 25 years of history. Integrity of solutions and seamless integration capability into Customer infrastructure is a priority area of company development.