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Integrated Networking Solutions

- 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 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 3,000 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-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

– 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 (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 (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

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 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 (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

[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 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)

Advanced SIP/SIP-T/SIP-I functionality

– Registration and authentication of up to 3,000 SIP subscribers[1]
– VAS support for up to 3,000 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 a local servicing in case of server unavailability

Redundancy[2]

– Operation in light redundancy mode 1+1
– Automatic enabling reserve
– 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)
  – 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
– 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

Physical specifications and environmental parameters

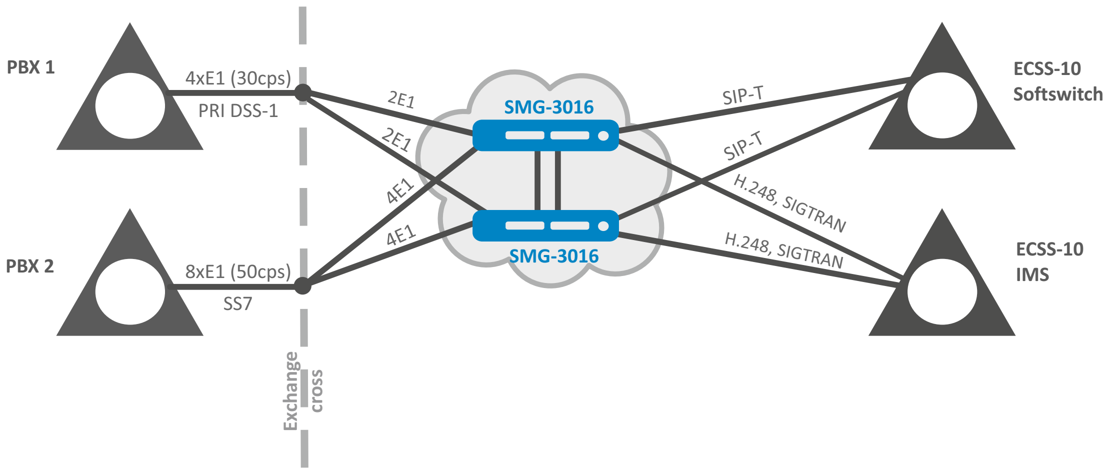
Operating temperature range	from 0 to +40 °C	
Relative humidity	up to 80 %	
Noise level	from 44 to 60 dB	
Power supply	AC: 100–240 V, 47–63 Hz DC: 36–72 V Power supply options: – AC/DC power supply; – 2 hot-swappable AC/DC power supplies.	
Power modules	AC, power module PM160-220/12 160 W	DC, power module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

[1] *Optional.*
[2] *Supported from 3.19.0 firmware version.*

Application diagrams

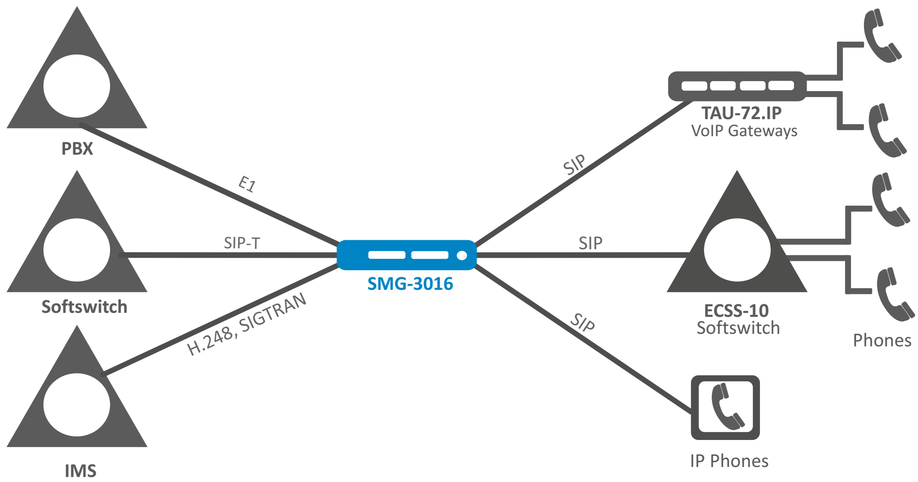
High-load transit nodes

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.



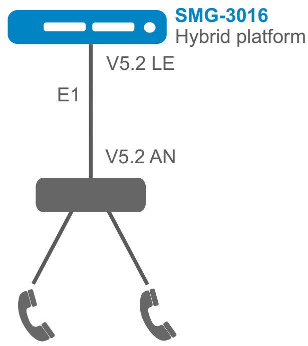
Terminal network node

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.



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Ordering information

Name	Description
SMG-3016	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
SMG-3016 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
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
SMG-3016 options	
SMG3-PBX-3000	Activation of ECSS-10 module for 3,000 SIP registrations with BLF support on the SMG-3016 digital gateway
SMG-CORP-1000	Activation of ECSS-10 module for 1,000 SIP registrations with VAS on the SMG-3016 digital gateway
SMG3-VAS-1000	Extension of SMG3-PBX-3000 option: activation of standard VAS set for 1000 subscribers on the SMG-3016 digital gateway
SMG3-REC	Activation of Call Recording functionality on the SMG-3016 digital gateway
SMG3-VNS	Activation of Voice Notification System (VNS) functionality on the SMG-3016 digital gateway
SMG1-AUTH-CALL	Activation of "Authorization by callback" functionality
SMG3-H323	Activation of H.323 (without Gatekeeper) on the SMG-3016 digital gateway
SMG3-RCM	Activation of Radius Call Management functionality on the SMG-3016 digital gateway
SMG3-VNI-40	Extension of VLAN interfaces to 40 on the SMG-3016 digital gateway
SMG1-IVR	Activation of IVR functionality
SMG3-RESERVE	Activation of redundancy by IP in master-slave mode on the SMG-3016 platform
SMG3-RESERVE-E1	SMG3-RESERVE-E1 option to activate E1 reservation on the SMG-3016 platform
SMG3-V5.2LE	Organization of an outstation V5.2LE on the digital gateway SMG-3016
Discounted option sets for SMG-3016	
SMG3-SP2	"PBX+VAS" set, includes 2 options for one gateway SMG-3016: 1×SMG3-PBX-3000 and 1×SMG3-VAS-1000
SMG3-SP4	"Triple" set, includes 3 options for one SMG-3016: SMG3-H323, SMG3-RCM and SMG3-VNI-40
SMG3-SP5	"VAS-3000" includes 3 options for one SMG-3016: 3×SMG3-VAS-1000
SMG3-SP7	"VAS-2000" includes 2 options for one SMG-3016: 2×SMG3-VAS-1000

Contact us

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

eltex@eltex-co.ru

www.eltex-co.com

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.