

- Scalable platform 1U
- Quad-Core ARMv7 Marvell Armada-XP 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 gateway SMG-2016 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 VAS support and a universal solution for infocommunication new generation networks (NGN). The wide function-set, strict compliance with requirements and standards and carrier class reliability allow service providers to solve most part of their objectives on the basis of SMG-2016.

Scalability

SMG-2016 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 768 VoIP channels.

Carrier class reliability

SMG-2016 provides high level of fault tolerance due to embedded state-of-the-art Quad-Core ARMv7 Marvell Armada-XP processor, uniform load distribution among submodules, power modules redundancy and usage of up-to-date technologies based on parallel computing. The gateway will switch to a backup submodule in case of a primary submodule fault.

Functional compatibility

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, programmable SIP clients, etc.

Media streams transcoding

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



IP PBX with VAS support

Additional options for SMG-2016 gateway allow using it as a full-featured IP PBX with up to 3000 SIP subscribers connection and support of a wide range of value added services. A programmable IP PBX module ECSS-10 is dedicated to fast deployment of a VoIP node with a minimum of capital expenses. ECSS-10 and SMG-2016 might be used as a PBX of any level.

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-2016 (Dynamic Firewall, Static Firewall, black and white lists of IP addresses and subnetworks, etc.). For additional defense, SMG-2016 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 customers 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) based routing
- Number modifications before and after routing
- Call recording according to number mask and dialplan¹
- Use of multiple dialplans
- Subscriber lines restriction
- Subscriber service mode settings
- Trunk group cut-off
- Call management via RADIUS¹
- Direct connection of 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
- Ingress/egress traffic rate limiting

DTMF

- INBAND, RFC 2833, SIP INFO, SIP NOTIFY transmission methods

Billing

- Billing data is recorded in CDR file. CDR files are kept on a local HDD and remote FTP server.
- 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 LE¹
- V5.2 AN¹

VoIP protocols

- SIP, SIP-T/SIP-I, SIP-Q
- H.323¹
- SIGTRAN (M2UA, IUA)¹
- H.248¹

Capacity and performance

- Up to **768** VoIP channels
- Up to **16** E1 streams (RJ-48)
- Maximum load intensity - 120 cps
- Quad-Core ARMv7 based Marvell Armada-XP 1.6 GHz
- RAM 4 GB

Interfaces

- 16 x E1 ports (RJ-48)
- 2 x 10/100/1000Base-T (RJ-45) / 1000Base-X (SFP) ports
- 2 x 10/100/1000Base-T (RJ-45) ports
- 2 slots for SATA HDD 2,5"

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
- Tracings are stored on HDD and USB storages
- Emergency notification through SNMP

Security

- Black and white IP addresses lists
- Attempts of access to device are logged
- Automatic blocking by IP address after unsuccessful login attempts or/and access via http/https/telnet/ssh
- List of permitted IP addresses for access to control of the device
- Access rights delimitation – admin/user
- Delimitation of rights to access calls records
- Control of opposite RTP stream's 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)

Redundancy

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

Advanced SIP/SIP-T/SIP-I functionality

- Registration and authentication of up to 3000 SIP subscribers¹
- VAS support for up to 3000 SIP subscribers¹
- 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

Value added services¹

- 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

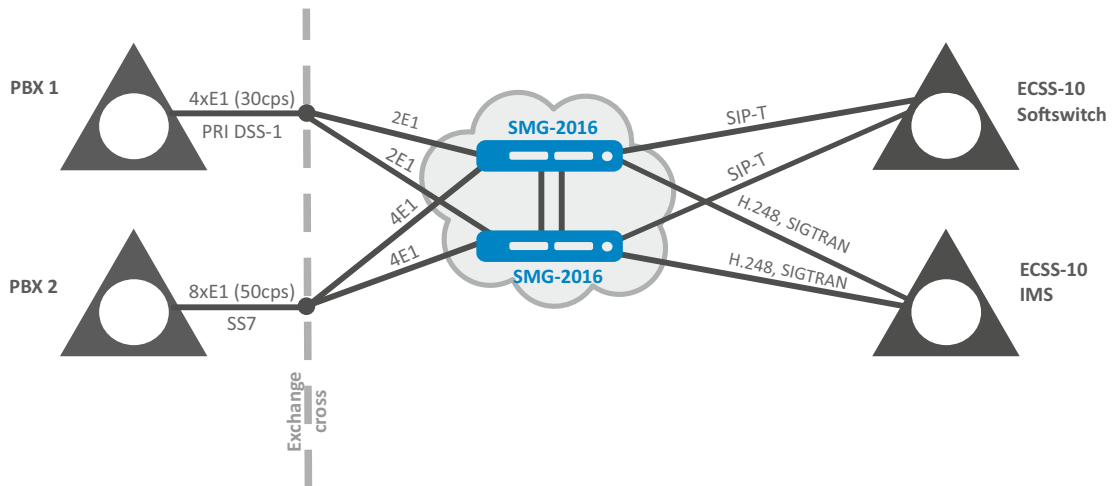
¹Optional

Current firmware version 3.14.0

Application diagrams

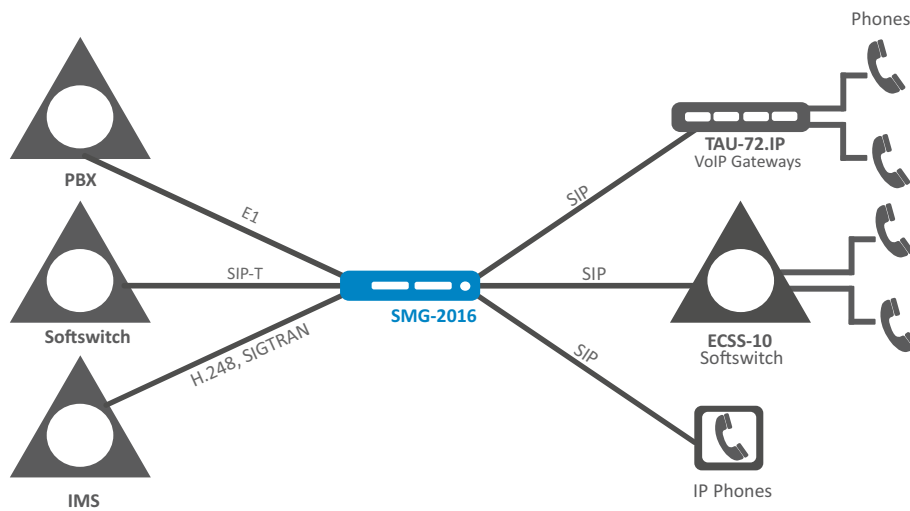
High-load transit nodes

The high performance and hot swap capability allow using SMG-2016 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-2016 gateway.



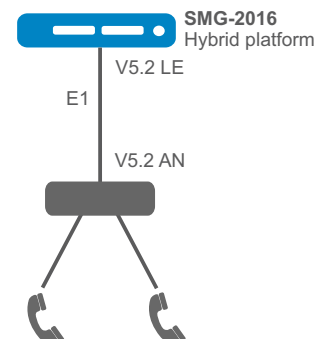
Terminal network node

The trunk gateway SMG-2016 might be used for organization of a single node for connection of PSTN to several electronic PBX as well as subscribers connection via VoIP gateways (e.g. TAU-72.IP).




Outstation via V5.2 protocol

The additional options of software module IP PBX ECSS-10 (SMG2-V5.2LE, SMG1-VAS-1000) allow organizing outstation via V5.2 protocol and service up to 3000 subscribers with support of a full VAS set. Equipment of any manufacturer that supports V5.2AN might be used as an outstation.



Ordering information

Name	Description	Image
SMG-2016	Digital gateway chassis SMG-2016: 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-2016		
SM-VP-M300	SM-VP-M300 submodule with support of 128 VoIP channels (G.711)	
C4E1	C4E1 submodule with support of 4 E1 streams	
PM160-220/12	PM160-220/12 power module, 220 VAC, 150W	
PM100-48/12	PM100-48/12 power module, 48 VDC, 100W	
Options for SMG-2016		
SMG2-PBX-3000	Activation of ECSS-10 module for 3000 SIP registrations with BLE function support on the digital gateway SMG-2016	
SMG2-VAS-1000	Extension of SMG2-PBX-3000 option: activation of standard VAS set for 1000 subscribers on the digital gateway SMG-2016	
SMG2-H323	Activation of H.323 (without Gatekeeper) on the digital gateway SMG-2016	
SMG2-RCM	Activation of Radius Call Management functionality on the digital gateway SMG-2016	
SMG2-VNI-40	Extension of the number of VLAN interfaces to 40 on the digital gateway SMG-2016	
SMG2-REC	Activation of Call Recording functionality on the digital gateway SMG-2016	
SMG2-CORP	Activation of ECSS-10 module for 1000 SIP registrations with VAS support on the digital gateway SMG-2016	
SMG2-V5.2LE	Organization of an outstation V5.2LE on the digital gateway SMG-2016	
SMG2-V5.2AN	Organization of an outstation V5.2AN on the digital gateway SMG-2016	
SMG2-RESERVE	Activation of redundancy via IP in master-slave mode on SMG-2016	
Discounted option sets for SMG-2016		
SMG2-SP2	"PBX+VAS" set, includes 2 options for one gateway SMG-2016: 1xSMG2-PBX-3000 and 1xSMG2-VAS-1000	
SMG2-SP4	"Triple" set, includes 3 options for one SMG-2016: SMG2-H323, SMG2-RCM and SMG2-VNI-40	

Contact Us

About EltexAlatau


+7 (727) 220 76 10


post@eltexalatau.kz


www.eltexalatau.kz

EltexAlatau company is one of the first communication equipment manufacturers in Kazakhstan established in 2012. The main focus of the enterprise is a set of solutions and the opportunity of their seamless connection to the customer's infrastructure.