- up to 16 E1 flows (RJ-48)
- up to 768 VoIP channels
- IP-PBX for 3000 SIP users with VAS and LI support
- scalable 1U platform

SMG-2016 platform can be used as a trunking gateway for converting signal and media flows of TDM and VoIP networks, IP-PBX with supporting VAS functions as well as be used as a universal solution for building new-generation research communication networks. Wide functionality, strict compliance with the standards and high reliability of the operator's class help to handle a greater number of tasks coming up with the operators and service providers on the basis of SMG-2016.

Scalability

SMG-2016 provides a possibility of uniform distribution of investments for scaling within the entire period of project implementation. The gateway supports up to 16 E1 flows (SS7, PRI) and up to 768 VoIP channels.

Operator's class reliability

The up-to-date Marvell processor , uniform load distribution between sub-modules, backed-up power supplies, ensure high level of SMG-2016 platform fail-safe behaviour with automatic switchover to a standby element in case of failure of any system sub-module as well as power module.

Functional compatibility

Strict conformity to the requirements of up-to-date protocols, recommendations and standards ensure 100% functional compatibility of SMG-2016 with different equipment: digital PBX, IP-PBX, Softswitches, VoIP gateways, SIP phones, software SIP clients, etc.

Transcoding media flows

The hardware transcoding based on MediaCodecs Mindspeed Technologies helps coordinate media flows with different VoIP codecs.

RADIUS routing

Intellectual call routing based on the billing system responses according to RADIUS protocol will help build flexible rules for calls processing.

Intellectual protection of IP networks

The intellectual protection against unauthorized external connections of SIP users (fail2ban, iptables, white/blacklists, etc.) also via protocols http/telnet/ssh has been implemented in trunking gateway SMG-2016. In order to provide additional protection when connecting to public IP networks, a compatibility with the session border controllers (e.g., SBC-1000) performing the functions of internetwork screens for VoIP networks.

IP-PBX with VAS support

Additional options for SMG-2016 gateway allow using it as a full-function IP-PBX up to 3 000 SIP users with supporting a wide set of Value Added Services. The IP-PBX ECSS-10 software module is intended for fast deployment of VoIP communications node with minimal capital expenses (CAPEX).

FUNCTIONAL CAPABILITIES

Calls management
– Interaction with STUN-server via SIP interface
– Routing by phone number of CdPN and CgPN
– Phone modification before and after routing
– Recording conversations by parameters[1]
– Several numeration plans
– Limitation of lines' amount
– Service mode setting
– Switching off trunk group
– Call management via RADIUS[1]
– Direct bypass of trunk-group
– SORM support[1]
– Prefix for several trunk-groups
– Interactive voice menu (IVR)[1]
– Configuration upload and download by one file

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

Fax
– T.38 Real-Time Fax, G.711 (a-law, μ-law) pass-through

Quality enchancement
– VAD (Voice Activity Detection)
– CNG (Comfort Noise Generation)
– AEC (Acoustic Echo Cancellation, G.168)

Quality of Service (QOS)
– Sets DiffServ Valueand 802.1p priority for SIP and RTP packets
– Dynamic and static programmable jitter buffer
– Based on bandwidth

DTMF
– INBAND, RFC 2833, SIP INFO

Billing
– Recording of billing information to CDR file, parallel recording of CDR file to HDD-disc and remote FTP-server
– Radius Accounting
– Different billing system support: Hydra Billing, LANBilling, PortaBilling, NetUP, BGBilling and other system integration

Flexability
– Configuration loading/uploading to one file
– Several network interfaces creation for telephony (SIP, RTP) with different IP-adresses
– Work with several numeration plans
– Signal channel SS7 reservation
– Control of telephone connection duration (RTP or RTCP presence)

TDM protocols
– Ss7
– PRI (Q.931)
– Q.699
– V5.2 LE[2]
– V5.2 AN[2]

VoIP control protocols
– SIP, SIP-T/SIP-I
– H.323[1]
– SIGTRAN[2]
– H.248[2]

Capacity
– up to **768** voice channels
– up to **16** E1 trunks (CENTRONICS-36 connector)
– Allowed calls per second - 14 cps

Interfaces
– 2 x 1000Base-X ports (SFP)
– 3 x 10/100/1000Base-T ports (RJ-45)
– E1 (2 x CENTRONICS-36 connectors)
– 2 x SATA ports (SSD)

Monitoring and management
– Monitoring of E1 channel and VoIP in web-interface
– Alarm logging with saving logs to syslog-server
– Keeping tracings on SSD-storage
– Informing about accidents, errors by SNMP

Security
– Black and Write lists of IP-adresses for registration
– Demonstration of all attempts to get access
– Automatic IP-Address locking after unsuccessful registration attempts and/or access via protocols http/telnet/ssh
– List of permitted IP-adresses for management access
– Accesses rights admin/user
– IP-address source control of RTP-flow
– Authentication on RADIUS-server and SIP registar
– Digest-autorisation

Expanded function SIP/SIP-T/SIP-I
– Registration and authentication to 3000 SIP users[1]
– Value Added Services support for 3000 users[1]
– Interaction between SIP and SIP/T/SIP-I
– SIP-trunks registration
– Upper Registration

Value Added Services (VAS)[1]
– Call Forwarding
– Call Forwarding Outside System (CFOS)
– Call Forwarding No Reply (CFNR)
– Call Forwarding Unconditional (CFU)
– Call Forwarding Busy (CFB)
– Call Transfer
– Music on Hold (MOH)
– Call Hold
– Call Hunt
– Call Pickup
– Interactive voice response

[1]Optional
[2]Isn't supported in 3.5.0 software current version

Ordering information

Product name	Description	Figure
SMG-2016	Trunking Gateway SMG-2016: 4 slots for submodules M4E1, 6 slots for submodules SM-VP-M300, 2 slots for power modules PM160-220/12 and PM75-48/12	
	Modules for platform SMG-2016	
SM-VP-M300	Submodule SM-VP-M300 with support up to 128 VoIP channels (G.711)	
M4E1	Submodule M4E1 with support up to 4 E1 flows	
PM160-220/12	Power module PM160-220/12, 220V AC, 160W	
PM75-48/12	Power module PM75-48/12, 48V DC, 75W	
	Options for SMG-1016M	
SMG2-PBX-3000	SMG2-PBX-3000 Option for ECSS-10 activation for 3000 SIP-registrations with BLF support on SMG-2016	
SMG2-VAS-1000	SMG2-PBX-3000 expanded Option: SMG2-VAS-1000 Option for VAS activation of 1000 users on SMG-2016	
SMG2-CORP-1000	SMG2-CORP-1000 expanded Option: SMG2-CORP Option for 1000 SIP-registrations with VAS on SMG-2016	
SMG2-H323	SMG2-H323 Option for H.323 protocol (without function Gatekeeper) on SMG-1016M	
SMG2-RCM	SMG2-RCM Option for Radius CallManagement activation on SMG-2016	
SMG2-VNI-40	SMG2-VNI-40 Option for expanding of VLAN-interfeces amounts on SMG-2016	
SMG2-REC	SMG2-REC Option for CallRecording on SMG-2016	
SMG2-CORP	SMG2-CORP Option for ECSS-10 module activation for 1000 SIP-registrations on SMG-2016	

About company

Eltex company is leading Russian developer and manufacturer of telecommunications equipment with 20 years of history. Integrity of solutions and seamless integration capability into Customer infrastructure is priority area of company development.

Contact us

+7 (383) 274 48 48
+7 (383) 274 48 49

eltex@eltex.nsk.ru

www.eltex.nsk.ru