- Scalable 1U platform
- IP PBX for 2,000 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"

Hybrid platform SMG-3116 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 requirments 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-3116.

Scalability

SMG-3116 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-3116 gateway allow using it as a full-featured IP PBX for up to 2,000 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-3116 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-3116 trunk gateway with automatic switching to a backup submodule in the event of any system submodule or the power source failure.

Functional compatibility

The strict compliance with requirements of up-to-date protocols, recommendations and standards provides functional compatibility of SMG-3116 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-3116 (Dynamic Firewall, Static Firewall, black and white lists of IP addresses and subnetworks, etc.). For additional defense, SMG-3116 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 creating 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[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]
– 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, µ-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, 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
– 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

TTDM protocols
– SS7
– PRI (Q.931)
– Q.699 (PRI and SS7 interaction)
– V5.2 LE[1]

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 — 14 cps

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
– 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

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

Features and capabilities (continued)

Advanced SIP/SIP-T/SIP-I functionality

- Registration and authentication of up to 2,000 SIP subscribers[1]
- VAS support for up to 2,000 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

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 (CFT)
- 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

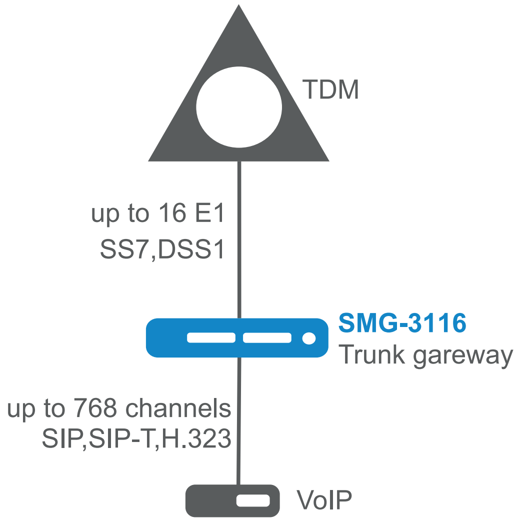
Physical specifications and environmental parameters

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	

[1] *Optional.*

Application diagrams

Protocol converter

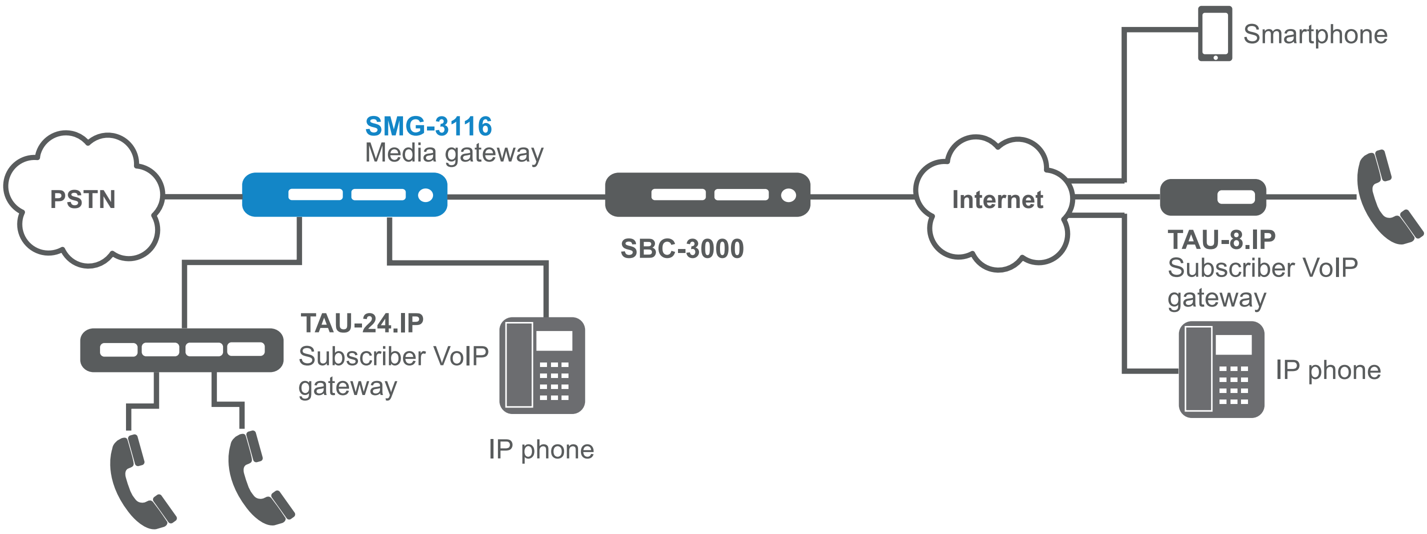


A wide range of supported TDM and VoIP protocols allows the SMG-3116 to interwork signaling and media streams in various directions:

- VoIP <-> VoIP
- VoIP <-> TDM
- TDM <-> VoIP
- TDM <-> TDM

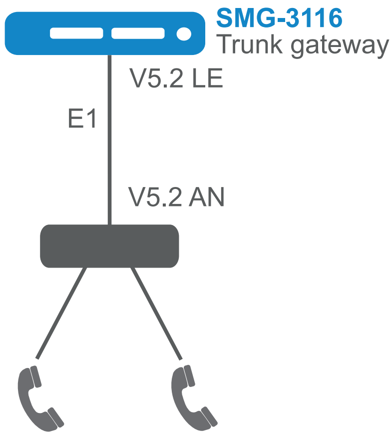
Local exchange

Enabling additional options of the ECSS-10 IP PBX software module (SMG1-PBX-2000, SMG1-VAS-500) makes it possible to use the SMG-3116 at the initial stage of building a local communication node with a capacity of up to 2,000 SIP subscribers as a fully functional PBX supporting the basic set of supplementary services. As the platform's capacity grows and the list of provided services needs to be expanded, the SMG-3116 can be migrated to a full-featured ECSS-10 softswitch server solution with multi-level redundancy support and flexible scaling of all components.



Outstation via V5.2 protocol

The additional options of IP PBX software module ECSS-10 (SMG1-V5.2LE, SMG1-VAS-500) allow clients to organize outstation via V5.2 protocol and service up to 2,000 subscribers with support for a full VAS set. Equipment of any manufacturer that supports V5.2AN might be used as an outstation.



Integrated networking solutions

Ordering information

Name	Description
SMG-3116	SMG-3116 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-3116 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-3116 options	
SMG1-PBX-2000	Activation of ECSS-10 module for 2,000 SIP registrations with BLF support on the SMG-3116 digital gateway
SMG1-VAS-500	Extension of SMG3-PBX-2000 option: activation of standard VAS set for 500 subscribers on the SMG-3116 digital gateway
SMG1-H323	Activation of H.323 (without Gatekeeper) on the SMG-3116 digital gateway
SMG1-RCM	Activation of Radius Call Management functionality on the SMG-3116 digital gateway
SMG1-VNI-40	Extension of VLAN interfaces to 40 on the SMG-3116 digital gateway
SMG1-REC	Activation of Call Recording functionality on the SMG-3116 digital gateway
SMG1-CORP	Activation of ECSS-10 module for 500 SIP registrations with VAS on the SMG-3116 digital gateway
SMG1-VNS	Activation of Voice Notification System (VNS) functionality on the SMG-3116 digital gateway
SMG1-AUTH-CALL	Activation of "Authorization by callback" functionality
SMG1-IVR	Activation of IVR functionality
SMG1-V5.2LE	Organization of an outstation V5.2LE on the digital gateway SMG-3116
SMG1-V5.2AN	Organization of an outstation V5.2AN on the digital gateway SMG-3116

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About Eltex

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.