ELTEX
Integrated networking solutions

- **– IP PBX for up to 500 subscribers**
- **– Up to 100 calls simultaneously**
- **– Up to 4 E1 flows (RJ-48)**
- **– 4 LAN ports**
- **– Call center functions**
- **– Call recording[1]**

SMG-500 is an enterprise PBX for 500 subscribers with a full set of Value Added Services (VAS).

SMG-500

The basic configuration of the enterprise IP PBX SMG-500 is designed to connect up to 250 SIP subscribers and can be extended to connect up to 500 subscribers when purchasing the appropriate software[1]. The E1 ports and SIP trunks can be used for connection to PSTN. Analog phones are connected to SMG-500 via subscriber VoIP gateways, IP phones – directly via the company network. Call recordings and CDR files are stored on the SD card or USB drive. It is also possible to automatically upload files to external data storage or an FTP server.

Networking of separated offices

The SMG-500 allows clients to organize an enterprise telephone network between remote offices of the company with minimal costs. Landline phone numbers in all offices remain the same, as customers will continue calling the known numbers. Employees from different offices can call each other on short numbers absolutely free, thus reducing the cost of intercity and international calls.

Multiservice platform

The variety of services allows clients to create the most efficient individual call processing scenarios. The SMG-500 supports conference calls, call recording, multiple channels and interactive voice menu.

Functional compatibility

Strict compliance with modern protocols, recommendations and standards ensures 100% functional compatibility of SMG-500 with equipment from different vendors: digital PBX, IP PBX, Softswitch, VoIP gateways, SIP phones, SIP software clients, etc.

Smart IP network protection

The IP PBX SMG-500 has intelligent protection against unauthorized external connections of SIP subscribers (dynamic firewall, static firewall, white/black lists, etc.) and connections via http/https/telnet/ssh.

High quality voice processing

The high quality of voice processing is provided by the up-to-date hardware platform, support for main audio codecs used in VoIP networks (G.711, G.722, G.726, G.729), echo cancellation, silence detector, comfort noise generator, receiving and generating DTMF signals, as well as traffic prioritization mechanisms (QoS).

[1] *Optional.*

Features and capabilities

Interfaces
– 4 × E1 ports (RJ-48)
– 4 × Ethernet 10/100/1000BASE-T ports (RJ-45)
– 1 × USB 2.0; 1 x USB 3.0
– 1 SD card slot (SDHC)
– 1 COM port (RS-232, RJ-45)

VoIP protocols
– SIP, SIP-T/SIP-I
– H.323

Advanced SIP/SIP-T/SIP-I functions
– SIP, SIP-T/SIP-I interaction

Capacity and performance
– Up to 100 simultaneous calls
– Up to 4 E1 streams (RJ-48)
– RAM: 2 GB
– Maximum load intensity[1]:
  – 15 SIP-E1 calls per second
  – With SIP-CPS license: 45-50 SIP-SIP calls per second
  – Without SIP-CPS license: 40 SIP-SIP calls per second

TDM protocols
– SS7
– PRI (Q.931)
– Q.699 (PRI and SS7 interaction)

Voice codecs
– G.711 (a-law, μ-law)
– G.722
– G.726
– G.729 (A/B)

Fax transmission
– T.38 transit/offroad, G.711 (a-law, μ-law) pass-through

Voice standards
– VAD (Voice Activity Detector)
– CNG (Comfort Noise Generation)
– AEC (Acoustic Echo Cancellation, G.168 recommendation)

Functions
– Interactive Voice Response (IVR) system with graphic editor
– Direct Inward System Access (DISA)
– Call queue:
  – Various algorithms for choosing operators
  – Call distribution considering repeated client requests
– Reporting system by operators/groups of operators (processed calls, missed calls, average waiting time, etc.)
– Phone book:
  – Creating a phone book from the station subscribers list
  – Transferring a phone book to subscribers via LDAP
  – Obtaining a display name from the LDAP server
– Video processing:
  – Transmitting a video stream using Video Offroad, Video Transit modes

Call management
– Number modification before and after routing
– Call recording by parameters
– Routing by access category
– Subscriber lines restriction
– Subscriber service mode configuration
– Trunk group cut-off
– Direct connection of trunk groups
– Prefix for multiple trunk groups
– Limiting the number of simultaneous calls to the SIP interface
– Ingress load limiting (call per second) for a trunk group
– Interaction with the STUN server on a SIP server
– Routing by Called Party Number (CdPN) and/or Calling Party Number (CgPN)

Quality of Service (QoS)
– Diffserv assignment for SIP
– Diffserv assignment for RTP

DTMF
– Transmission via INBAND, RFC 2833, SIP INFO, SIP NOTIFY
– Ability to auto-detect the method of receiving DTMF

Value Added Services (VAS)
– Call Forward:
  – Call Forwarding on Out of Service (CFOS)
  – Call Forwarding on No Reply (CFNR)
  – Call Forwarding Unconditional (CFU)
  – Call Forwarding on Busy (CFB)
  – Call Forwarding on Time (CFT)
– Call Transfer
– Music on Hold (MOH)
– Call Hold
– SIP-forking support for SIP subscribers
– Voice Notification
– Call Hunt
– Call Pickup
– Call Parking
– Busy Lamp Field
– Subscriber registration status indicator (Presence)
– Message Waiting Indicator
– Add-on conference (CONF)
– Conference based on subscribers list
– 3-Way conference
– Intercom
– Paging Call
– Call Queue
– Call Back when the position in queue is reached[1]

[1] The values are specified for trunk mode operation (without registrations, subscriptions, or VAS use), with a load of one-second SIP-SIP calls. Only incoming call legs were taken into account in the calculation.
[2] Not supported in the current firmware version.

Features and capabilities (continued)

– Call Recording
– PINCodeAccess
– Follow me
– Follow me on No Response
– Do not disturb (DND) with whitelist
– Blacklist
– Intervention
– Voice mail
– One Touch Record
– Speed Dial for FXS ports
– Anonymous call
– Reject anonymous calls
– Reminder
– Call Waiting
– Do not disturb in the call group (CGDND)
– Auto-dial
– Auto-dial with callback
– Chief secretary (only with VAS-ACG license)
– Current system time

Flexibility

– Uploading/downloading configuration as a single file
– Uploading/downloading license as a single file
– Downloading and uploading subscriber settings in a single file
– Creating multiple network interfaces for telephony (SIP, RTP) with different IP addresses
– Operation with multiple dial plans
– Voice activity control (by the presence of RTP or RTCP)[1]

Management and monitoring

– E1 and VoIP flow channels monitoring via web interface
– Channels and SS7 signal links management via web interface
– Alarm logging with the option of storing entries on the syslog server
– Storing traces on SD card/USB storage device
– Alarm reporting via SNMP
– Automatic logging activation after gateway restart
– Monitoring of active sessions of the web interface users
– Notification of device status and failures via email[1]

Billing

– Billing data is recorded to CDR file. Simultaneously, CDR file is recorded to a local SD disk, USB storage device or remote FTP server
– RADIUS Accounting
– Supported billing systems:
  – Hydra Billing
  – LANBilling
  – PortaBilling
  – NetUP
  – BGBilling
– Integration with other systems

Security

– Black and white IP address lists for registration
– Logging of all access attempts to the device
– Automatic IP blocking after unsuccessful login attempts or/and access via http/https/telnet/ssh
– List of IP addresses allowed to manage the device
– Multilevel web interface access permission
– SIP subscribers authentification
– RADIUS authorization (RFC 5090, Draft-Sterman)
– Checking the WEB user password reliability
– The validity time of the WEB user password

Physical specifications and ambient parameters

Operating temperature	From 0 to +40 °C
Relative humidity	Up to 80 %
Power supply	AC: 220 V+-20 %, 50 Hz; Lead-acid battery: 12 V; – Battery charge current: 1.6+-0.1 A; – Low battery voltage threshold indication: 11 V; – Threshold voltage for battery deep discharge protection: 10-10.5 V.
Maximum power consumption	Up to 40 W during battery charge, up to 20 W without battery charge
Dimensions (W × H × D)	430 × 44 × 203 mm
Form factor	19", 1U
Weight	2.35 kg

[1] Not supported in the current firmware version.

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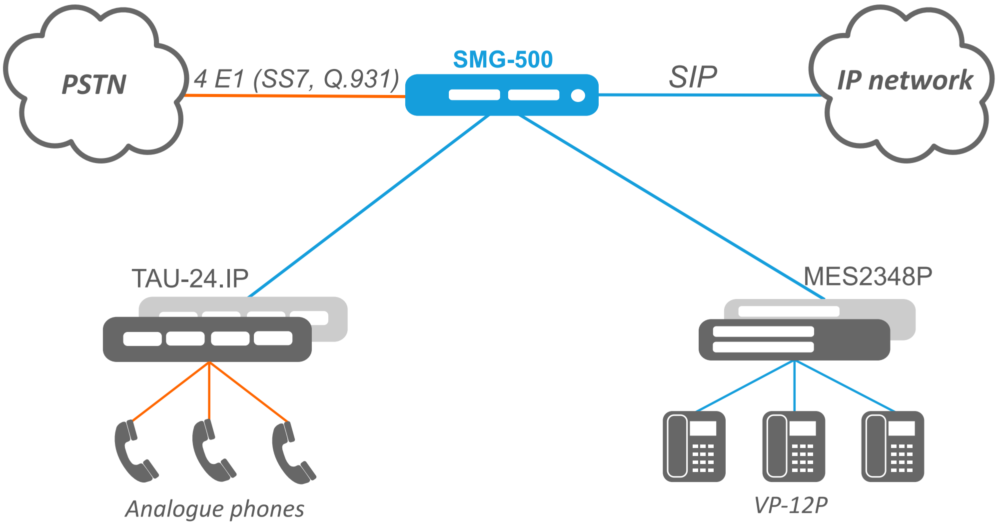
Use case



Ordering information

Name	Description
SMG-500	IP PBX SMG-500: 250 SIP subscribers (can be extended to 500), 4 × 10/100/1000BASE-T ports (RJ-45), 1 × USB 2.0; 1 × USB 3.0, 4 × E1 ports (RJ-48)
SMG-500 modules	
SM-VP-M300	SM-VP-M300 submodule: up to 128 VoIP channels (G.711)
C4E1	C4E1 submodule: up to 4 E1 streams

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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.