Integrated networking solutions

- **IP PBX for up to 500 subscribers**
- **Up to 100 simultaneous calls**
- **Up to 4 E1 flows (RJ-48)**
- **4 LAN ports**
- **Call center functionality**
- **Call recording**

SMG-500 – enterprise IP PBX for 500 subscribers with full Value Added Services (VAS) set.

SMG-500

Enterprise IP PBX SMG-500 connects up to 250 SIP subscribers in basic configuration and can be extended to 500 SIP subscriberswhen acquiring appropriate license[1].
The E1 ports and SIP trunks can be used for PSTN connection. Analogue phones are connected to SMG-500 via subscriber VoIP gateways, IP phones - directly through the company network. Call records and CDR files are stored on SD card or USB flash drive. It is also possible to upload files to external media and to FTP server automatically.

Aggregation of remote offices in a single network

SMG-500 allows clients to organize enterprise telephone network among remote offices with minimal expenses. In addition city numbers are stored – the clients will be able to call the known city numbers.
The co-workers from different offices are able to call each other for free using short numbers, thus saving on costs on intercity and international calls.

Multiservice platform

Variety of services allows creating of more effective individual scenarios for call processing. SMG-500 supports conference connection, call recording, multichanneling, interactive voice menu, etc.

Functional compatability

Strict compliance with up-to-date protocols requirements, recommendations and standards provides functional compatability with different vendors equipment: digital PBX, IP PBX, Softswitches, VoIP gateways, SIP phones, programmable SIP clients, etc.

Intellectual protection of IP networks

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

High-quality voice processing

The high quality of voice processing on SMG-500 is provided by the up-to-date hardware platform, support of main audio codecs used in VoIP networks (G.711, G.726, G.729), echo cancellation, silence detector, comfort noise generation, reception and generation of DTMF signals and prioritization mechanisms (QoS).

[1] *Optionally*

Features and capabilities

Interfaces
- 4 E1 (RJ-48) ports
- 4 ports of Ethernet 10/100/1000Base-T (RJ-45)
- 1 port of USB2.0, 1 port of USB3.0
- 1 slot for SD card (SDHC)
- 1 COM port (RS-232, RJ-45)

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

Advanced SIP/SIP-T/SIP-I features
- SIP and SIP-T/SIP-I interaction

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

Voice codecs
- G.711 (a-law, μ-law)
- G.726
- G.729 (A/B)
- OPUS[1]
- AMR[1]

Voice standards
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- AEC (echo cancellation, G.168 recommendation)

Functions
- Interactive voice response system (IVR) with graphic editor
- DISA - Direct Inward System Access
- Call queue:
  - Various algorithms for operators selection
  - Calls distribution mechanism takes into account repeated calls of clients
- Reporting system for operators/groups of operators (processed calls, missed calls, average timeout, etc.)

Call management
- Number modifications before and after routing
- Call recording according to parameters
- Subscriber lines restriction
- Subscriber service mode configuring
- Trunk group cut-off
- Direct connection of trunk groups
- Prefix for several trunk groups
- Limiting the number of simultaneous calls to a SIP interface
- Ingress load limiting (calls per seconds) for a trunk group
- Interaction with STUN server via SIP interface

Quality of Service (QoS)
- Diffserv assignment for SIP
- Diffserv assignment for RTP[1]

DTMF
- Transmission via INBAND, RFC 2833, SIP INFO, SIP NOTIFY methods

Billing
- Billing data is recorded to CDR file. CDR files are written on a local SD disk, USB flash or remote FTP server concurrently
- RADIUS Accounting
- Supported billing systems:
  - Hydra Billing
  - LANBilling
  - PortaBilling
  - NetUP
  - BGBilling

  Possible integration with other systems

Value Added Services
- Call Forwarding:
  - Call forwarding on 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[1]
- Paging Call[1]
- Call Queue
- Call Back when the position in queue is reached[1]
- Call Recording
- PIN Code Access
- Follow me
- Follow me on no response
- Do not disturb (DND) with white list
- Blacklist

Flexibility
- Uploading/downloading of configuration as a single file
- Multiple network interfaces creation for telephony (SIP, RTP) with different IP addresses
- Operation with multiple dial plans
- SS7 signal channel redundancy
- Voice activity control (by the presence of RTP or RTCP)[1]

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[1] *Not supported in the current firmware version(3.14.0)*

Integrated networking solutions

Features and capabilities

Security

– Black and white IP addresses lists for registration
– Attempts to access the 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 the device
– Access rights delimitation - admin/user
– Authentication of SIP subscribers
– RADIUS authentication (RFC 5090, Draft-Sterman)

Management and monitoring

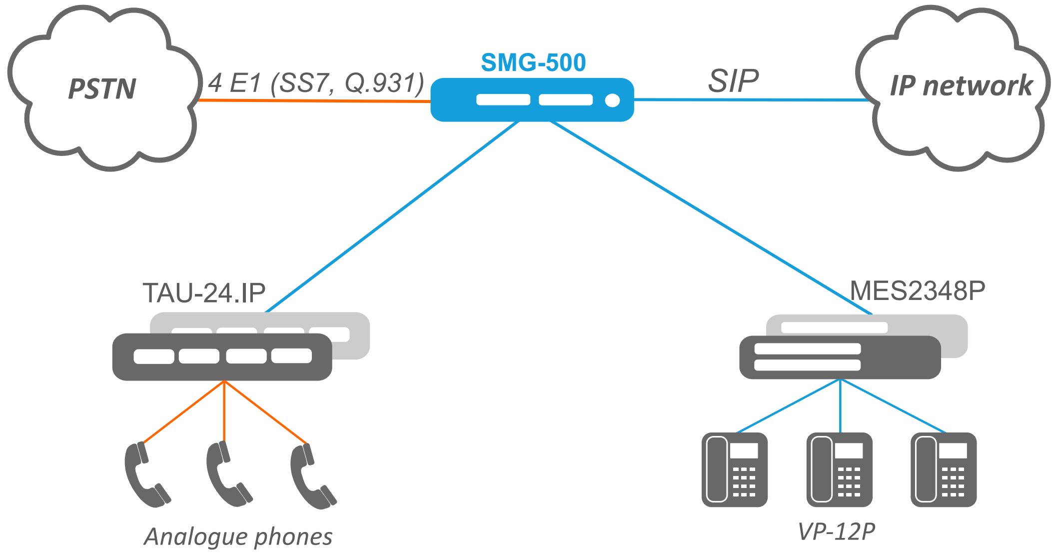
– E1 and VoIP channels monitoring in web interface
– Management of channels and SS7 links in web interface
– Alarm logging with opportunity to store entries on syslog server
– Tracings are stored on SD card/USB flash
– Emergency notification via SNMP[1]

Power supply

– AC network: 220V, 50 Hz
– Battery: 12V

[1] *Not supported in the current firmware version (3.1 40)*

Use case



Ordering information

Name	Description	Image
SMG-500	IP PBX SMG-500: 250 SIP subscribers (can be extended to 500), 4 ports of 10/100/1000Base-T (RJ-45), 1 port of USB2.0, 1 port of USB3.0, 4 E1 ports (RJ-48)	
SMG-500 modules		
SM-VP-M300	SM-VP-M300 submodule with support of up to 128 VoIP channels (G.711)	
C4E1	C4E1 submodule with support of up to 4 E1 flows	

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About EltexAlatau

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.