Integrated Networking Solutions

- IP PBX for up to 200 subscribers
- Up to 50 simultaneous calls
- Up to 16 FXS/FXO ports
- 4 LAN ports
- Call center functionality
- Call recording

SMG-200 – enterprise IP PBX for 200 subscribers with full Value Added Services (VAS) set.

SMG-200

Enterprise IP PBX SMG-200 is a telecom grade device that connects up to 100 SIP subscribers in basic configuration and can be extended to 200 SIP subscribers when acquiring appropriate license[1].

16 RJ-11 ports can be used for analogue phones connection as well as for connection to land lines. The LAN ports are dedicated to carriers connection via SIP trunks and for extension of FXS/FXO ports quantity via VoIP gateways (e.g. you can use TAU-24 which has 24 FXS ports). Call records and CDR files are stored on SD card or USB flash drive. It is also possible to upload files to FTP server automatically.

Aggregation of remote offices in a single network

SMG-200 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 more effective individual scenarios for call processing. SMG-200 supports conference connection, call recording, multichanneling, interactive voice response, 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-200 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-200 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
– 16 FXS/FXO RJ-11 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

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.)
– Pulse dialing and DTMF

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

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

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
– Voice activity control (by the presence of RTP or RTCP)[1]

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
– Calling Line Identification Presentation (CLIP) in FSK formats (ITU-T V.23; Bell 202), DTMF, «Russian CLI»
– Caller ID and time of a call issuing in FSK mode
– Caller Line Identification Restriction (CLIR) (for FXS ports)

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

Features and capabilities

Management and monitoring
– Alarm logging with opportunity to store entries on syslog server
– Tracings are stored on SD card/USB flash
– Emergency notification via SNMP[1]

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)

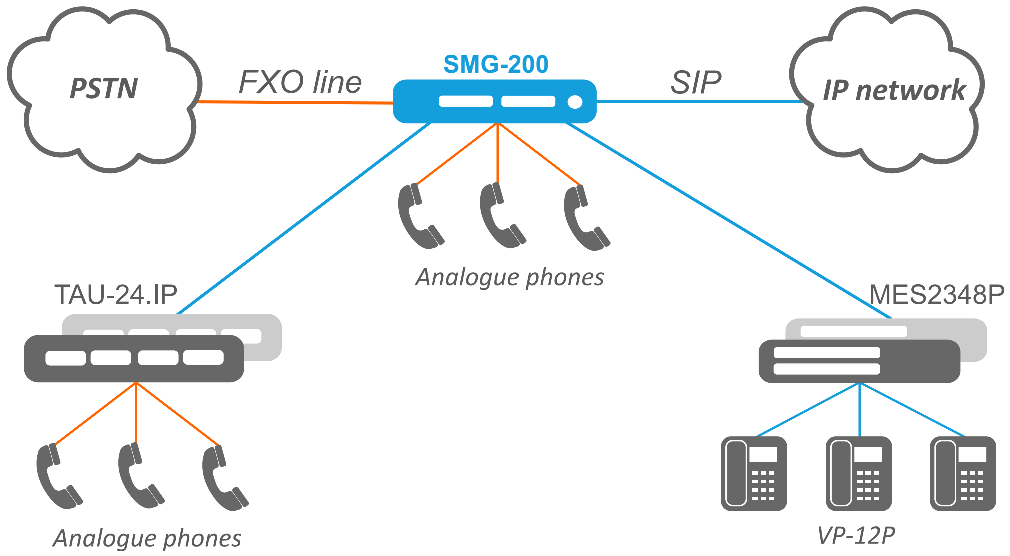
Quality of Service (QoS)
– Diffserv assignment for SIP
– Diffserv assignment for RTP[1]

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

Power supply
– AC network: 220V, 50 Hz
– Battery: 12V
– Class I protection

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

Use Case



Ordering information

Name	Description	Image
SMG-200	IP PBX SMG-200: 100 SIP subscribers (can be extended to 200), 4 ports of 10/100/1000Base-T (RJ-45), 1 port of USB2.0, 1 port of USB3.0, 16 FXS/FXO ports	
SMG-200 modules		
M8S	Subscriber set submodule M8S: 8 analogue subscriber ports (FXS)	
M8O	PBX subscriber line submodule M8O: 8 analogue ports (FXO)	

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About Eltex

Eltex company is a leading Russian developer and manufacturer of telecommunication equipment with 25 years of history. Integrity of solutions and seamless integration capability into Customer infrastructure is a priority area of company development.