Iran
Communications
Industries

SMG-200

Enterprise IP PBX

- 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

Enterprise IP PBX SMG-200 connects up to 100 SIP subscribers in basic configuration and can be extended to 200 SIP subscribers when acquiring appropriate license.
16 RJ-11 ports can be used for analog phones connection as well as for connection to land lines. The LAN ports are intended for connection to carriers' networks via SIP trunks and for extension of FXS/FXO ports quantity via VoIP gateways (e.g. TAU-24 which has 24 FXS ports can be used). Call records and CDR files are written to SD card or USB flash drive. It is also possible to upload files to FTP server auto- matically.

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Networking of Separated Offices

SMG-200 allows clients to organize enterprise telephone network between company offices with minimal expenses. Landline numbers remain the same, hence clients will continue calling numbers they know.
Co-workers from different offices are able to call each other for free using short numbers, thus reducing costs for intercity and international calls.

Multiservice Platform

The variety of services allows creating more effective individual scenarios for call processing. SMG-200 has support for conference calling, call recording, multi-channeling and interactive voice menu.

Functional Compatibility

Strict compliance with the requirements of modern protocols, recommendations and standards provides functional compatibility of SMG-200 with different vendors' equipment: digital PBX, IP PBX, Softswitches, VoIP gateways, SIP phones, software SIP clients etc.

Intellectual Protection of IP Networks

SMG-200 has intellectual protection against unauthorized external connections of SIP subscribers (dynamic firewall, static firewall, white/black lists etc.) and connections via http/https/ ssh/telnet.

High-Quality Voice Processing

The high quality of voice processing on SMG-200 is provided by the up-to-date hardware platform, support for main audio codecs used in VoIP networks (G.711, G.726, G.729), echo cancellation, silence detector, comfort noise generation, prioritization mechanisms (QoS) and DTMF signals reception and generation.

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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, AMR

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

Quality of Service (QoS)
- Diffserv assignment for SIP and for RTP

Physical Specifications
- Dimensions (WxHxD), mm - 430 x 44 x 203
- Power supply: AC: 220V+-20%, 50 Hz; Lead-acid battery: 12V;

SMG-200 modules
- M8S: Subscriber set submodule M8S: 8 analog subscriber ports (FXS)
- M8O: PBX subscriber line submodule M8O: 8 analog ports (FXO)

Functions
- Interactive voice response system (IVR) with graphic editor
- DISA - Direct Inward System Access
- Call queue:
  - Various algorithms for choosing operators
  - Call distribution considering repeated calls from clients
- Reporting system for operators/groups of operators
(processed calls, missed calls, average waiting time etc.)
- Pulse and tone dialing
- Phone book:
  - Creating a phone book from station subscribers list
  - Transferring a phone book to subscribers via LDAP
  - Obtaining a display name from LDAP server
- Video processing:
  - Video transmission in the Video Offroad mode

Call Management
- Number modification before and after routing
- Call recording according to parameters
- Subscriber lines restriction
- Subscriber service mode configuration
- 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 second) for a trunk group
- Interaction with STUN server via SIP interfaces
- Dialing rules for FXO ports