

16-Port Multisim GSM Gateway Solution



AddPac 16-Port Multi-SIM GSM Gateway solution provides AP-GS3100 Front SIM type Multisim GSM Gateway and AP-GS916 Multisim GSM Gateway. AP-GS3100 GSM Gateway supports Front SIM Type Multi-SIM GSM VoIP Module like as AP-GSM4SF. This Front SIM Type GSM Multi-SIM VoIP Modules can be installed in next generation AddPac Front SIM Type GSM Gateway Hardware Platform such as AP-GS3200, AP-GS3100, etc. AP-GSM4S8F GSM VolP Module is a new Front SIM Type 4-Port GSM Module (8) SIM card slots per GSM Port) for GSM Call Termination Service. This GSM VoIP Module supports Quad-Band (850/900/1800/1900MHz) GSM frequency, Eight(8) SIM card slots per GSM port, Hot-Swap Switch Button, One(1) GSM Antenna Interface. This GSM module has thirty-two(32) GSM SIM card slots. Also this module supports the Voice DSP for GSM call termination service based on VoIP Technology. AP-GS916 Multi-SIM GSM Gateway supports various Multi-SIM GSM VoIP Modules like as AP-N1-GSM4S8, AP-N1-GSM4S4, etc. AP-N1-GSM4S8 GSM VoIP Module is a new

4-Port GSM Module (8 SIM card slots per GSM Port) for GSM Call Termination Service. This GSM VoIP Module supports Quad-Band (850/900/1800/1900MHz) GSM frequency, Eight(8) SIM card slots per GSM port, Hot-Swap Switch Button, One(1) GSM Antenna Interface. This GSM module has thirty-two(32) GSM SIM card slots internally. Also this module supports the Voice DSP for GSM call termination service based on VoIP Technology. AP-N1-GSM4S4 GSM VoIP Module is a new 4-Port GSM Module (4 SIM card slots per a GSM Port) for GSM Call Termination Service. This GSM VoIP Module supports Quad-Band (850/900/1800/1900MHz) GSM frequency, Four(4) SIM card slots per a GSM port, Hot-Swap Switch Button, one(1) GSM Antenna Interface. This GSM module has sixteen(16 = 4x4) GSM SIM card slots internally. Also this module supports the Voice DSP for GSM call termination service based on VoIP Technology. AddPac VTO service is a high performance VoIP Traffic Optimization service which can reduce bandwidth usage in VOIP call termination. This means that call termination service provider can deliver improved VoIP calling service with much lower VoIP traffic cost. Also, AddPac VTO service supports the VoIP antiblocking service features and can help to provide mobile VoIP termination service in locations where VoIP service is completely blocked. In addition to basic VoIP Traffic Optimization service features, AddPac VTO service supports real-time VoIP traffic monitoring service such as VoIP packet loss rate, up/down link status, etc in between VTO server and VTO client site each. This can help to provide real-time VTO technical support service AddPac VTO service is very easy to use because AddPac GSM VoIP gateway provides the VTO client service features internally besides GSM VoIP gateway service features. This means there is no need an external Linux server for VTO service and complex installation procedure. User just does new firmware upgrade for AddPac GSM gateway VTO client package. GSM VoIP gateway service concept supporting VTO client service is a new architecture and design concept. This concept reduces the hop count and enhances the VoIP QoS like as jitter,

delay. As a result, this architecture increases the ASR, ACD that are most important factors in call termination service area.

Major Feature

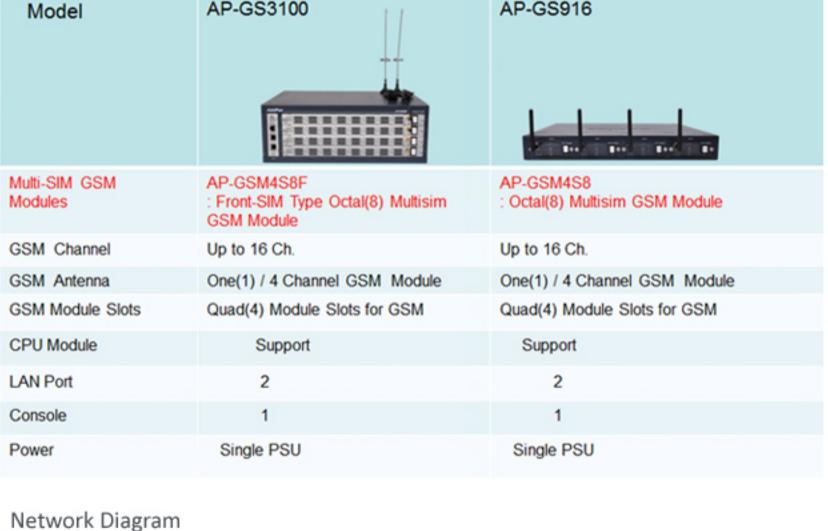
Overall Features

- AddPac VTO(VoIP Traffic Optimzer) Service Client: VTO Plug & Play Service, Anti-VOIP, Bandwidth Compression, VoIP Traffic Monitoring (Packet loss, round-trip time), etc
- Renew the Main Window
- Mobile Statistics Display using Graph Chart (ASR, ACD, etc)
- Several Enhancements for ASR Improvement (zombie call, etc)
- > Human Behavior Call Pattern Modeling Algorithm for Anti-SIM Block
- IMEI change, BTS Selection, etc.
- > Powerful RISC Microprocessor + DSP Architectures > High Performance GSM Modular Architecture
- Two(2) 10/100Mbps Fast Ethernet Interfaces for WAN, LAN
- Maximum 16-Port GSM Interface Support
- 1-Port RS-232C Console Interface for CLI
- SIP protocol support compliant with IETF RFC3261 (or RFC2543)
- H.323 and SIP Dual Stack APOS Internetworking Software to provide Scalability, Functionality,
- Stability and QoS Control for AddPac VoIP Gateway
- Static, Default IP-Routing and IEEE 802.1Q VLAN Routing Protocols
- Enhanced QoS Managements for Voice Traffic SNMP MIB v2 for Network Management Features
- Essential Scalability Features such as DHCP Server & Relay, NAT/PAT, IEEE Transparent Bridging, IP Accounting, and Debugging/ Diagnostics, etc. > Remote Software Upgrade using FTP & TFTP

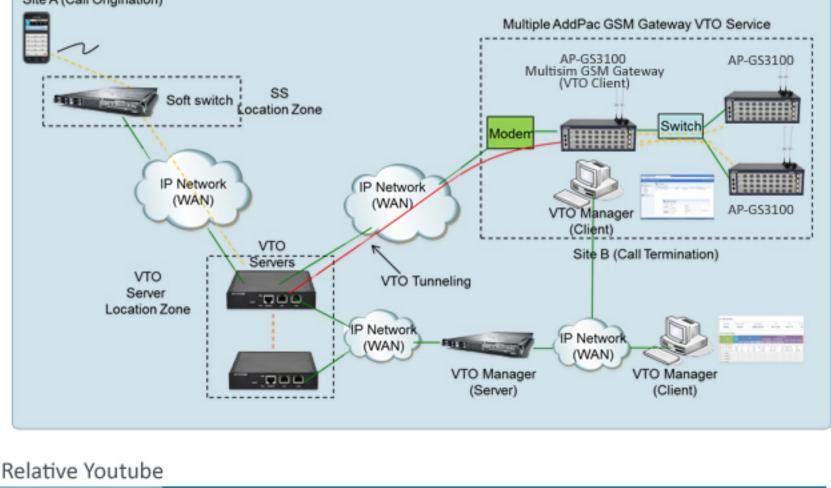
Octal(8) Multi-SIM GSM VoIP Module



AP-GS3100



Site A (Call Origination)









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