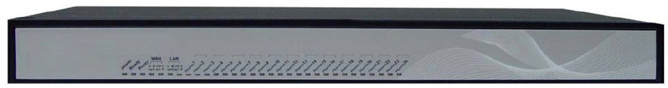
LotusGate 1024PS 24-Line FXS SIP IP Gateway

- Support up to 16 SIP proxy Servers
- Support 24 analog Phone sets at one IP address
- Support different SIP Trunk to each FXS line
- Auto HTTP Provision feature
- Flexible Routes Plan and Dial Plan
- Redundant Firmware Image



LotusGate 1024PS

Introduction

LotusGate 1024PS is an 24-line FXS gateway with SIP protocol IP device which allows to connect 24 sets of analog telephone to make or receive VoIP call over Internet or VPN network through Internet Telephony service provider. This device is suitable for office user through ITSP service provider to install at office or branch office to call between different offices. It can be installed at basement of apartment or dweller building to provide analog phone set to each house to make/receive telephony internet call via Telco or ITSP's broadband device and network.

To select up to 16 SIP service Accounts

LotusGate 1024PS is appropriate to use up to 16 VoIP Service Providers, IP Centrex service and IP-PBX within offices and remote branch offices. One of 16 SIP Servers (or ITSP Service provider or alternative IP-PBX) can be configured freely at each line (FXS port) to make or receive IP Call. It provides 16 service platforms to select lowest rate or different purpose according to your dial number or country or application.

19-inch, 1U chassis easy to install

LotusGate 1024PS is an 19-inch, 1U chassis and suitable to install at Relay Rack for vast lines installation at office, Telco and ITSP service provider. 1-WAN and 1-LAN with NAT feature together with RS-232 DB-9 local console port allows engineer to configure and maintain this device locally and remotely. It can be installed at outdoor electric pole and sidewalk. There are 24 ports RJ-11 line connectors to connect to telephone cable via existing Telecom MDF. No more new cable is needed.

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Flexible Dial plan and Route Plan Features

LotusGate 1024PS provides flexible Dial Plan from FXS to IP Trunk (SIP Softswitch). Dial Plan is to configure in what condition the digits can be sent out to IP network. The dial inter digit time before dialing is configurable to meet busy users or home user. Dial Rule is able to detect the prefix code and maximum digits reached and then dial out automatically. The Digit Manipulation (DM) allows you to configure matched prefix code, digits length, start and stop digit position to be replaced digits as well.

Routes Plan is to configure the incoming and outgoing call routes which you desire this call to go out or allow to income. For instance, IP incoming call may Ring to FXS port with Priority, Cyclic or Simultaneously ring . You can also configure IP incoming call by Matched prefix digits, Matched incoming FXS line number and Matched digit length. For FXS outgoing call routes, the hunting type supports Priority, Cyclic or Simultaneously ring and select which SIP trunk (SIP Proxy Server) to go. FXS outgoing call routes also support by Matched prefix digits, Matched incoming FXS line number and Matched digit length. Both direction supports No Answer time out and Backup Routes.

Specification

- **Interface:**
  - Ethernet port (RJ-45, 10/100 base-T)
  - 1-WAN port, connect to IP Network
  - 1-LAN port connect to PC with NAT
  - Support Bridge, NAT and Gateway mode
  - Telephony port (RJ-11 x 24 pcs)
  - RS-232 Console port, DB9 Male
  - AC power input Jack
  - AC Power ON/OFF Switch
  - LED Indicator for System, SIP and FXS status
- **IP Network connection**
  - IPv4 (RFC 791)
  - MAC Address (IEEE 802.3)
  - MAC Clone Setting
  - Vendor Class ID
  - IP/ICMP/ARP/RARP/SNTP
  - Static IP
  - DHCP Client (RFC 2131), WAN port
  - DHCP Server, LAN port
  - NAT Server (RFC 1631)
  - PPPoE Client
  - DDNS (DynDNS)
  - DNS Client
  - Firewall
  - URL Filter
  - IP Filter
  - MAC Address Filter
  - Application program Filter
  - Port Filter
  - Port Forwarding
  - Bandwidth Control (Download and Upload), Maximum Bandwidth and reserved bandwidth
  - UPnP Server at LAN port
  - Behind NAT, use DMZ for NAT traversal
  - SNTP with time zone and Daylight Saving

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- TCP/UDP (RFC 793/768)
- RTP/RTCP (RFC 1889/1890)
- IPV4 ICMP (RFC 792),
- TFTP Client
- VLAN Support 802.1Q, 802.1P
- VLAN ID Range : 2 to 4094
- VLAN Priority : 0 to 7
- QoS : DiffServ (RFC 2475), TOS (RFC791, 1394)

SIP Protocol :

- RFC3261 compliance
- Support up-to 16 SIP Server Register Accounts
- SIP UDP Protocol
- Support SIP compact Form
- Support SIP HOLD Type
- SIP Session Timer (RFC 4028)
- Configure SIP port and SIP QoS Type
- MD5 Digest Authentication (RFC2069/RFC2617)
- SIP PRACK (RFC3262)
- Early/Delay Media support
- Offer/Answer (RFC3265)
- Message Waiting Indication (RFC3842)
- Event Notification (RFC3265)
- REFER (RFC3515)
- Support Outbound Proxy
- SIP Proxy Keep Alive time setup
- Support Primary and Secondary SIP Server
- Support STUN NAT Traversal
- Support “rport” parameter (RFC 3581)

Audio Codec :

- G.711 A-law/µ-law, G.729A, G.723.1 (6.3K, 5.3K), GSM-FR Full Rate (13kbps)
- Select voice codec priority : Local or Remote
- Configure RTP port and RTP QoS Type
- Silence Suppression
- VAD/CNG
- LEC : Line Echo Canceller
- Max Echo Tail Length (G.168): 32, 64 and 128ms
- Packet Loss Compensation
- Input (Encode) Gain setup
- Output (Decode) Gain setup
- In-band/out of band DTMF (RFC4733, RFC2833 / SIP INFO)
- Adaptive/Configurable Jitter Buffer
- G.168 Acoustic Echo Cancellation
- Dialing Plan with drop, replace, Insert dialing digits
- Select First digit and Inter digit timeout duration (Sec)
- Selectable Call Progress Tone
- Support Specified Line Calling

Call Features :

- Caller ID display DTMF (before/after 1st ring) and FSK (before 1st ring), ETSI and Bellcore
- DTMF Caller ID start and stop BIT(A to F) configurable
- Polarity Reversal before Caller ID or not
- Tone Generation: Ring, Ring Back, Dial, Busy, call waiting, ROH, Warning, Holding, Stutter dial tone and disconnect tone
- Configure Tone Frequency, Cadence, Level and Cycle
- Global Country Based Tone Specification
- NAT Traversal support STUN, UPnP and Behind NAT

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- Out-Band DTMF : RFC2833 and SIP Info
- Configure DTMF send ON and OFF time
- Configure DTMF detect Min. ON and OFF time
- DTMF Relay Volume
- Flash Time transmit via SIP Info (Enable or Disable)
- Message Waiting Indication (Stutter Tone Notice)
- Speed Dialing
- Call Waiting/Switching between Calls
- Call Forward (Busy, Unconditional, No Answer)
- No Answer Time out
- Block Anonymous Call
- Hot Line
- Call Hold
- Call Transfer
- Flexible Dial Plan
- First digit and Inter digit time out timer setup
- Manual SEND digits selection key
- Retrieve transfer call from 3rd party by dial Code (default: *#)
- Dial Rule : Prefix and Maximum digits, digit position to replace
- Digit Manipulation (Drop and Replace Rule)
- Extension Hunting
- Support SIP Trunk up to 16 SIP Servers
- Import and Export SIP Trunk configuration
- Outgoing SIP Caller ID Selection
- Accept desired SIP Proxy incoming calls Only
- Flexible Routing Plan
- Prefix Match and Length
- Matched FXS Line (port) number
- Incoming call type: FXS or VoIP
- Priority Ring
- Cyclic Ring
- Simultaneous Ring
- Programmable Hunting Cycle
- Backup Routing with Digit Manipulation
- Default Routing
- T.38 FAX: ECM, Redundant depth, Volume
- FAX Relay : T.38 or Disable
- Retrieve Voice Mail from IP-PBX
- FXS Caller ID Mode : Transparent or Inhibit
- SIP Caller ID Mode : Transparent or Inhibit
- Support Peer to Peer Dialing
- Flash Time Detection: range from 80 to 800 ms
- ON-HOOK Voltage -48Vdc
- Ring Sine Wave frequency : 10 to 70 HZ
- Ring Level : 10 to 95 Vrms
- Ring REN : 3
- Configure Ring ON time : 0 to 8000 ms
- Configure Ring OFF time : 0 to 8000 ms
- Configure Ring Cadence, Frequency and Voltage
- Support Polarity reversal for Billing
- Service Up to 1 Kilo-meter distance to analog telephone set
- Generate Current Drop Time (Open Loop Disconnect time)

MANAGEMENT :

- Administrative Telnet CLI and HTTP, HTTPS with desired Port number
- Enable/Disable HTTPS and Telnet Service
- http provision through MAC address
- RS-232 console cable to configure WAN/LAN IP Address
- Multilingual Web User Interface
- 3 Levels of User Access Right with Password protection (Administrator, Supervisor and User)

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- HTTP/HTTPS Service Access limitation from WAN port
- Provides System Status Logs
- Status display: Network, Line, SIP Trunk status
- Diagnostics (debug through Syslog Event Notice)
- Debug in real time by Telnet
- 8 Debug Level: Emergency, Alert, Critical. Error, Warning, Notice, Information, Debug
- Auto Provision via HTTP Server
- Support SNMP V2/Trap
- Configuration Backup/Restore
- Dual Firmware Image Backup
- Reset to factory Default

** Support LotusCom proprietary encryption protocol at SIP Signal and Voice codec during transmitting to IP network in order to Anti ISP block of VoIP call. This feature only be available with LotusCom SIP server or IP-PBX3020IPv6

Environmental :

- Actual Dimension: 44(W)×4.4(H)×26.2(D) CM
- 19-inch, 1U chassis with Relay Rack Mount Bracket
- Weight: 4.3kg (One unit with packing)
- Operating Temp. & Humidity
  - □ Temp.: 0°C~45°C (32°F~113°F)
  - □ Humidity: 10%~90% relative humidity, non-condensing
- Power Input: AC100V to 240V, 50/60Hz

Packing Accessories

- LotusGate 1024PS gateway x 1 pcs
- Relay Rack Mount Bracket x 2 pcs
- AC Power cable x 1 pcs
- CD User Manual x 1 pcs

Warranty

- One year

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