Techroutes NETWORK

DATASHEET

TS-300

Session Border Controller

Product Overview

TS-300 SBC provides rich SIP-based services such as safe network access, robust security, system interworking, flexible session routing & policy management, QoS, media transcoding and media processing for small-and-medium telecommunication operators. With distributed multi-core processor, rear panel for non-blocking gigabit switching data network, embedded Linux operating system, TS-300 delivers high capability while achieves low power dissipation. It supports 300 concurrent SIP sessions and transcodes up to 100 concurrent calls, and allows encrypted sessions via TLS and SRTP. The session border controller supports transcoding of G.729, G.723, G.711, G.726, iLBC, AMR and OPUS.

Besides, TS-300 offers two E1 ports that are compatible with PBX to meet customers' networking needs.

Salient Features

- Supports up to 300 SIP sessions and 100 transcoding sessions
- SIP trunks & flexible routing rules for accessing IMS
- Embedded VoIP firewall, prevention of DoS and DDos attacks
- Bandwidth limitation and dynamic white list & black list
- QoS, static route, NAT traversal
- Import & export of remote upgrade and configuration data
- Encrypted sessions
- User-friendly web interface, multiple management ways

Physical Interfaces

- Ethernet Ports: 5* 10/100/1000M Base-T Ethernet ports
- E1/T1 Ports: 2* E1/T1, RJ48C
- 1* USB
- 1* TF Card Slot
- Serial Console: 1* RS232, 115200bps, RJ45
- LTE Uplink (Under Development): 1* SIM Card Slot

VoIP

- SIP 2.0 compliant, UDP, TCP, TLS
- SIP trunk (Peer to peer)
- SIP trunk (Access)
- SIP Registrations
- B2BUA (Back-to-Back User Agent)
- SIP Request rate limiting
- SIP registration rate limiting
- SIP registration scan attack detection
- SIP call scan attack detection
- SIP anti-attack
- SIP Header manipulation
- SIP malformed packet protection
- Multiple Soft-switches supported
- QoS (ToS, DSCP)
- NAT Traversal

Capabilities

- Concurrent Calls: Supports 300 SIP sessions at maximum
- Transcoding : Supports 100 transcoding calls
- CPS for Call: 20 calls per second at maximum
- Registrations: Maximum SIP registrations: 3000
- CPS for Registration: 20 Registration per second
- SIP Trunks: 128 SIP Trunks at maximum

Media Capabilities

- Voice, FAX support
- Codecs: G.729, G.723, G.711, iLBC
- RTP Transcoding
- Pass-through fax
- No RTP detection
- One-way audio detection
- RTP/RTCP
- RTCP statistics reports
- DTMF: RFC2833, SIP Info, INBAND
- Silence Suppression
- Comfort Noise
- Voice Activity Detection
- Echo Cancellation
- Adaptive Dynamic Buffer

VOIP

Security
- Prevention of DoS and DDos attacks
- Control of access policies
- Policy-based anti-attacks
- Call Security with TLS
- White List & Black List
- Access Rule List
- Embedded VoIP Firewall

Call Control
- Dynamic load balancing and call routing
- Flexible Routing Engine
- Call routing base on prefixes
- Call routing base on caller/called number regular express
- Call routing base on time profile
- Call routing base on SIP URI
- Call routing base on SIP method
- Call routing base on endpoint
- Caller/ Called number Manipulation

Maintenance
- Web-bases GUI for Configurations
- Configuration Restore/Backup
- HTTP Firmware Upgrade
- CDR Report and Export
- Ping and Tracert
- Network Capture
- System log
- Statistics and Reports
- Multiple language support
- Centralized management system
- Remote Web and Telnet

Environmental
- Power Supply: DC 12V
- Power Consumption: 10w
- Operating Temperature. 0 ℃~ 45 ℃
- Storage Temperature: -20 ℃~80 ℃
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 226×146×39mm
- Unit Weight: 0.85 kg

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VOIP