DINSTAR

SBC3000 Pro Session Border Controller

Overview

Dinstar SBC3000 Pro provides rich SIP-based services such as safe network access, robust security, system interconnectivity, 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 as well as embedded Linux operating system, the session border controller delivers high capability while achieves low power dissipation.

It is able to process up to 5000 concurrent SIP sessions and transcode 1500 concurrent calls. Meanwhile, it supports dual hot-swappable power supplies, and allows encrypted sessions via TLS and SRTP. Apart form traditional codecs like G.729, G.723, G.711 and G.726, SBC3000 Pro also supports the transcoding of iLBC, AMR and OPUS.

Key Features

- Supports up to 5000 concurrent SIP sessions
- Support WebRTC2SIP to turn WebRTC client into a phone with audio capability
- Support transcoding of 1500 calls on media and fax
- Triple anti-attacking, embedded VoIP firewall, prevention of DoS and DDoS attacks
- Bandwidth limitation and dynamic white list & black list
- VLAN, IPSec, QoS, static route, NAT traversal
- Import & export of remote upgrade and configuration data
- User-friendly web interface, multiple management ways
- 1+1 high availability with two SBCs, dual hot-swappable power supplies

Physical Interfaces

- MCU (Main Control Unit): 1
- MFU (Main Function Unit): 4
- Ethernet Ports:
  8* 10/100/1000M Base-T Ethernet ports
- Serial Console
- 1* USB Port

Media Capabilities

- Voice, FAX support
- Codecs: G.729, G.723, G.711, iLBC, OPUS, G.722, AMR
- RTP Transcoding
- Pass-through fax
- No RTP detection
- One-way audio detection
- RTP/RTCP, SRTP
- RTCP statistics reports
- DTMF: RFC2833/SIP Info/INBAND
- Silence Suppression
- SIP Recording (SIPREC)
- Comfort Noise
- Voice Activity Detection
- Echo Cancellation
- Adaptive Dynamic Buffer

Environmental

- Dual Power Supply:
  AC 100-240V, 50/60Hz
- Power Consumption: 70W
- Operating Temperature: 0 ℃ ~ 45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H):
  437×320×44mm
- Unit Weight: 6 kg

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

Security

- Prevention of DoS and DDoS attacks
- Control of access policies
- Policy-based anti-attacks
- Call Security with TLS/SRTP
- 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
- TLS mutual authentication

Capabilities

- Concurrent Calls:
  Supports 5000 SIP sessions at maximum
- Transcoding:
  Supports 1500 transcoding calls
- CPS for Call:
  500 calls per second at maximum
- Registrations:
  Maximum SIP registrations: 20000
- CPS for Registration:
  500 Registration per second
- SIP Trunk:
  Maximum of 1024 SIP trunks
- SIPREC (Concurrency):
  Supports up to 2000 recording calls
- SRTP (Concurrency):
  Supports Up to 3000 encrypted calls
- WebRTC (Voice):
  Up to 2000 concurrent calls

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
- Port Mapping
- Network port binding
- Static Routing
- Remote capture
- Signaling tracking

Shenzhen Dinstar Co.,Ltd.

Floor 18, Building 7A, Vanke Cloud City Phase 1, Xingke 1st Street, Xili Sub-district, Nanshan District, Shenzhen, P.R. China 518000

www.dinstar.com
sales@dinstar.com

DINSTAR

About Us

Founded in 2011 in Shenzhen, DINSTAR is a leading global provider of IP Unified Communication products including VoIP Gateways, IP PBXs, IP Phones and SBCs, we have been delivering more agile, efficient and affordable communication solutions and unparalleled communication experiences to our customers with our reliable, innovative and future-proof products for years. Through our value-added distributors and resellers worldwide, now DINSTAR serves telecom operators, service providers, system integrators, enterprises, SMBs and OEM partners in over 100 countries.