

Overview

A Session Border Controller(SBC) is a SIP/IP network core element that deployed for next generation networks (NGN) to support peer/access connection by SIP/IMS trunk and provide interworking between incompatible signaling messages or media flow from endpoint devices and SIP applications. In particular SBC supports SIP/IMS network connection, sip endpoint remote registration, NAT traversal, signaling modification, media control, flexible call routing, codec transcoding, billing and QoS management.

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DINSTAR SBC8000 is a new software-based SBC. It provides very rich features for medium and large enterprise SIP network connection and supports various security mechanisms, call routing based on control policies, carriergrade high availability(HA) and debug tools for service providers and telecom operators.

SBC8000 can be installed on VM/cloud-based system or third-party hardware appliances. Users can install SBC8000 in X86, ARM architecture and any cloud-based platforms for many different SIP implementations with advantages of VM and cloud platforms.

Key Features

- Up to 10,000 concurrent call sessions, 5,000 media transcoding and 100,000 SIP registrations
- Operate on physical server, virtual machine and public cloud
- Intelligent bandwidth limit and dynamic blacklist
- Cross-network, NAT traversal and high availability(HA)
- SIP over TLS, SRTP
- Compatible with different codecs: G.711A/U, G.723.1, G.729A/B, iLBC, AMR, OPUS
- Flexible call routing
- Broad compatibility with IMS network
- VoIP firewall, anti-attacks and core network protection
- Call recording



Deployment Platforms

Hardware Architecture: X86/ARM Virtual Machine: VMware, Fusionsphere, FusionComputer, KVM Cloud Deployment: Alibaba Cloud, Amazon Cloud, Baidu Cloud, Huawei Cloud, Telecom Cloud. etc.

Capabilities

Concurrent Calls Up to 10,000 SIP sessions Transcoding Supports 5,000 transcoding calls CPS for Call 800 calls per second at maximum Registrations Up to 100,000 SIP registrations CPS for Registration 800 Registration per second

Media Capabilities

Codecs: G.711a/µ, G.723, G.729A/B, iLBC, G.726, AMR, OPUS Silence Suppression Voice Activity Detection(VAD) Comfort Noise Generator(CNG) Echo Cancellation: G.168 with up to 128ms RTP/RTCP Voice Interrupt Protection Adaptive Dynamic Buffer Adjustable Gain Control Automatic Gain Control (AGC) FAX: T.38, Pass-through

Call Control Features

Dynamic Load Balancing and Call Routing Flexible Routing Engine Routing Based on Caller/Called Prefixes Regular Express Call Routing Base on Time Profile Call Routing Base on SIP URI Call Routing Base on SIP Method Caller/ Called Number Manipulation

VoIP

SIP 2.0 Compliant, UDP, TCP, TLS, SIP Trunk (Peer to peer) SIP Trunk (Access) SIP Proxy Registrations: Up to 3,000 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 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 Message format detection and processing **UDP-Flood Anti-attacks TCP-Flood Anti-attacks** Call Security with TLS/SRTP Whitelist & Blacklist Access Control List Built-in VoIP Firewall

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 NTP Multiple languages support SNMP Remote Web and Telnet

Shenzhen Dinstar Co.,Ltd.

www.dinstar.com

DTMF: RFC2833/Signal/Inband

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



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.