RSBC10000-S SESSION BORDER CONTROLLER

MAX10,000 USERS

Max concurrency
20000 routes

Transcoding：12500　　CPS：1000

SX10000-S Session Border Controller is tailor-made for real-time VoIP communication.

It is used to connect VoIP equipments in different IP domains, to realize interoperability and traffic convergence of hetero-IP sessions, integrated management of VoIP services, to provide communication security, and to cope with the problems brought about by the evolution of enterprise voice networks, such as mobile office, branch networking, and multi-network convergence.

Product Features

1.Deployment

The RSBC10000-S session border controller supports virtualization and general server deployment

Hardware Platform

Users can install it on mainstream hardware server platforms in the market, general hardware servers based on the X86 architecture such as Huawei, Lenovo, Wave, etc.

Private Cloud

Support deploy on private clouds,VMware vSphere, OpenStack, FusionCompute, KVM, etc.

Public Cloud

Support deploy on Azure, AWS, GCP, Ali Cloud, Huawei Cloud, Tencent Cloud, Tianyi Cloud, etc.

2.Telecom-grade High Performance

RSBC10000-S Session Border Controller has a packet forwarding delay of less than 30 milliseconds, which ensures high quality voice calls.

Max registered users 100000, Max trial calls per sec 1000 CPS

Max concurrency 20000 routes, Max transcodes is 12500 routes

which can satisfy the communication requirements of medium and high end telecommunication service providers and medium and large enterprise users.

3.Telecom-grade High Scecurity

Core business systems inevitably encounter security threats when traversing public networks. Internet DoS/DDoS, UDP/TCP-Flood attacks are becoming more and more prevalent, consuming system resources with abnormal service requests, resulting in system resource depletion and the inability to respond to normal business requests. As a network edge device, the RSBC10000-S SBC provides total network security for organizations. Provide both local and off-site dual hot standby solutions for highly reliable disaster recovery.

4.Integration & Interoperability

Interconnection between corporate headquarters and branch offices, interconnection between companies and carriers

The RSBC10000-S SBC is compatible with the mainstream signaling and media protocols in the current market, eliminating the differences between networks from different manufacturers, different solutions, and different times.

5. Highly Compatible

Supports multiple protocols: IPv4/IPv6, UDP/TCP, RTP/RTCP/MSRP, WS/WSS, TLS/SRTP, SIP/IMS, HTTP/HTTPS, NTP/SNTP, etc., with strong compatibility.

6.Multi-Features

The RSBC10000-S is a SIP/IMS interfacing solution designed for medium to large telecom providers and enterprises. It offers standard SBC functions, AI capabilities, rapid response, and supports customized header fields, codec conversion, and third-party platform connections like NewLync and Lark. Additional features include protocol conversion, call recording, voice quality monitoring, trunk redundancy, load balancing, and support for state secret algorithms

7. Easy to Mangement

User-friendly graphical, Web-based operation interface, easy to use. Supports remote upgrade and remote maintenance, convenient for daily maintenance.

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Performance

Call Concurrency	Maximum 20000 (unencrypted)/15000 (encrypted) concurrent calls
Transcoding Concurrency	Supports up to 12500 channels of encoding and decoding conversion
Call CPS	1000 completed calls per second
Registered Users	Maximum 100000 Registered Users

Call Handling

- Calling and Caller Number Transformation
- Route selection based on calling/caller number prefixes
- Select the corresponding call routing according to the time slot policy
- Trunking redundancy and load balancing
- Heterogeneous network routing redundancy/load balancing

Security

- Black/white list based access control
- DOS/DDOS Attack Defense
- UDP/TCP-Flood Attack Defense
- Policy-based SIP Attack Defense
- TLS, SRTP/DTLS-SRTP Encrypted Sessions
- Black/white list of calling and receiving numbers
- Policy-based call frequency restriction
- Local and off-site dual-server disaster recovery deployment

Protocols

Networking	IPv4/IPv6, SSH, HTTP/HTTPS, DNS, ARP, NTP/SNTP, ICMP
Media	UDP/TCP, RTP/RTCP/MSRP, TLS/SRTP
Call CPS	1000 completed calls per second

- Terminal agent/forwarding registration and authentication
- SIP V2, IMS, WebRTC, WS/WSS
- QoS (IP TOS)
- SIP B2BUA
- Signaling message fragmentation
- NAT travel through

Management & Maintenance

- Web-based visual configuration and maintenance management (Simplified Chinese/Traditional Chinese/English)
- Command line based configuration and maintenance management (SSH)
- Configure data backup/recovery
- Web-based firmware upgrade/rollback
- Built-in network data capture function
- Multi-level log management; system logs, operation logs, call logs
- Unified management by New Rock UMS, third-party network management system, supports SNMP, TR-069 (TR- 104, TR-106)
- Device/system status monitoring, phone/email/message fault alerts

Voice

Audio and video codecs	G.711 (A-Law / U- Law), G.723.1, G.729a/b, iLBC, G.722, G.722.1, G.722.2, AMR, OPUS, VP8, VP9, SILK, H.264, and H.265, etc;
Voice Quality	802.1p/QVLANtagging, QOS, TOS, DiffServ, Voice Attenuation (VAD), Comfort Noise Generation (CNG), Echo Cancellation (AEC), Fixed/Dynamic Voice Gain Control, Jitter Buffer
Fax	G.711 Transparent Fax, T.38 Fax, compatible with Class G3 fax machines up to 33.6kbps (Transparent)
DTMF mode	RFC2833, SIP-INFO, INBAND

- NGN and IMS Protocol Conversion
- Call Recording