REX8000 IP-PBX

REX8000

Permitted IP Extension(s)	10,000
Permitted IP trunk(s)	10,000
Concurrent calls	800 (with call recording)
Call processing capacity	37,000 busy-hour call completions (BHCC)
Dimensions (H × W × D)	Individual unit: 44mm × 442mm × 420mm (1U) Dual unit stack: 88mm × 442mm × 420mm (2U)

REX8000 IP-PBX can provide a converged solution package of telecommunication (including audio calls, video calls, high-speed fax, telephone conferencing, call recording, and mobile extensions).
It satisfies the demand for a distributed network of over 10,000 extensions throughout government agencies and business groups, both of which require at least 1,000 extensions per site.

REX8000 supports access for 8,000 users with 800 concurrent calls, demonstrating its powerful call processing capacity. This outstanding performance even extends to its stability, reliability, security, expandability, configurability, and more.

REX8000 meets various telephony requirements, including deployment at a single site of large- and medium-size enterprises, as well as a multi-site telephone network of large corporations and government agencies via internet/VPN. This is made possible by supporting direct connection with carriers via SIP trunk (including IMS), or connection with the carriers or other audio private networks via digital trunk ① or analog trunk②, as well as connection to analog telephones, fax machines, POS , IP phones, video phones, softphones on PCs/cellphones/PADs.

Features

Multi-functional with powerful interface

- VoIP operations, such as audio, video, fax, conferencing, and recording
- Support for SIP trunking (including IMS), digital trunking ①, and analog trunking ②
- Various add-on services, including applications for call recording management, telephone conferencing, centralized equipment management, PMS middleware*, and attendant console system*
- Connection with third-party application systems and secondary development based on API*

Multiple redundancy/backup methods and security protection mechanism

- Dual system hot backup with switching time less than 5 seconds
- 1+1 redundancy for power supply/network port/main control card, N+1 redundancy for media resource card with auto load balancing
- Encrypted signaling, media, and data transmission
- Access whitelist, external user authorization, long-distance call restrictions, and more

High performance and high stability

- Distributed architecture that configures an integrated media resource card
- Highly responsive concurrent call processing
- Auto load balancing

Flexible deployment, easy operation, and simple maintenance

- Centralized single-site deployment and multi-site distributed network
- Centralized equipment management to ensure efficiency operation and maintenance
- Graphical configuration interface
- Detailed alert report via telephone/email ③

①Externally connected to Redstone digital VoIP gateway (1/2/4 30B+D interfaces per unit)
②Externally connected to Redstone analog trunking gateway (2 to 96 FXO ports)
③Paired with the Redstone Remote Device Management System
* Available since the first quarter of 2020

Redstone Systems

REX8000 IP-PBX

Specifications

Protocols

Call control **Network**	SIP/UDP and SIP/TCP (RFC3261), IMS (3GPP) SSHv2, HTTP, HTTPS (TLS 1.2), DHCP client, DNS (A/SRV record), STUN

Media Processing

Codec	G.711a/u, G.729a/b, G.722, G.722.2
DTMF	In-band audio, RFC2833, SIP-INFO
Fax over IP	T.38, G.711 pass-through T.38 compliant Group 3 Fax Relay Maximum fax rate of 33,600 bps (pass-through)
Voice-quality enhancement	Jitter buffer, QoS

Voice

Smart auto attendant/ Receptionist	Business/non-business hours/holiday, call queuing, attendant group, multilingual/multi-level IVR, auto attendant profiles, VIP service, DID
Dialing	Call barring (5 levels), outgoing route selection, speed dial, emergency call, black/white list for outbound calling, hotline (immediate/delay), least cost routing, automatic route selection
Call settings	Call holding, call parking, call waiting, 3-way calling, call forking, do not disturb, call barge, silent monitoring, secretary, call transfer,call forward, sharing extension, hunt group, etc.
DISA	Authorization with calling party number, authorization triggered by feature access code
Call recording	Through Call Recording Management System
SIP trunk	IMS, multiple SIP servers, Skype Connect

Security

User-defined ports	SIP port, RTP port, HTTP/HTTPS port for Web GUI access
Access list	IP addresses allowed to access HTTP/HTTPS/SSH services; SIP IP address filtering
Authentication	IEEE 802.1X with TLS 1.2 for IP phone devices; local or external authentication via LDAPS
Management security	SSHv2 for secure management sessions; HTTPS with TLS 1.2 for secure web management; local or external LDAPS for secure directory access
Encryption	SIP signaling and media encryption; DTLS 1.2 and SRTP with AES-128/AES-256 encryption and SHA-2 certificate authentication; TLS 1.2 and SRTP with AES-128/AES-256 encryption; encrypted configuration file import/export; encrypted password/PIN
IP phone protection	Prohibition on outgoing dialing by IP extensions on public networks; User-Agent authentication; registration password cracking protection
Login to Web GUI	Prohibition on login from public IP addresses; login password cracking protection
Security level	Three levels of security settings
Call restrict	Restrict total/single call duration, concurrent calls and call frequency for long-distance calls
Intrusion prevention	ACL-based traffic filtering

High Availability

Dual system hot backup	Switching time is less than 5 seconds
Component redundancy	Main control board/Power supply/Network port: 1+1 redundancy Media source board: N+1 redundancy

Provisioning, Administration and Maintenance

Device management	Redstone Remote Device Management System, TR-069 management (TR-069, TR-104, and TR-106), SNMP
Application interface	API (XML/HTTP)
Log	8-level log management, FTP backup
Data capture	Network packet capture by port mirroring
Configuration file	import/export/backup
Status and alarm	Status/performance monitoring, Detailed alert report via telephone/email
Version	Upgrade, rollback

Others

Multi-site voice	Internal calls between extensions, call forward, call transfer, outbound trunk sharing, three-way calling, attendant on remote site
QoS	DiffServ, TOS, 802.1p/Q VLAN tagging
Internal storage	Voicemail, history logs, IVR audio files(uploaded by users), system audio files management, configuration file backup, version rollback

Supported add-on software

Call recording management system	Comprehensive call recording search, score rating, labeling, statement generation, and other managerial operation, authorization management, and file encryption
Telephone conferencing system	10 conferences with 128 participants
Attendant console system	Fast and convenient help desk platform that features queue calls, call transfer, and extension status reporting
Billing system	Call monitoring and billing management
DockPMS middleware	Connectivity to Fidelio, OPERA, and other hotel PMSs, providing check-in/check-out, wakeup call, and other features
RLink softphone app	Available for iOS and Android, which enables corporate mobile extensions through the phones of individual employees via WiFi or 4G network

Hardware specifications

Single-unit frame	1 main control card, 1 fan module, 1/2 power supply module, and 4 idle slots
Media resource card	A single card supports 1,000 IP extensions, 125 concurrent calls and is hot swappable. A single-unit hosts a maximum of 4 media resource cards.
Network port	RJ45, 2×10/100/1000 Base-T, auto negotiation
CON port	RJ45
Internal storage	64GB
Dimensions (H × W × D)	Individual unit: 44 mm × 442 mm × 420 mm (1U) Dual unit stack: 88 mm × 442 mm × 420 mm (2U)
Weight(Net)	7.5kg(a fully equipped single unit)
AC power	100 to 240 VAC, 50/60 Hz, 3A maximum
Operating	Temperature: 0 to 40°C, Humidity: 10% to 90% RH (non-condensing)
Storage	Temperature: -40 to 70°C, Humidity: 5% to 90% RH (non-condensing)

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