RSBC600 Specifications

Physical specifications	Description
Chassis type	1U x 19 inch mental chassis
Dimension (H x W x D)	1.73 x 11.81 x 17.32 in. (44 x 300 x 440 mm)
Net weight	6.61 lb (3 kg)
Ethernet connector	4 x 10/100/1000Base-T RJ45 port, auto sensing
Configuration interface (CON)	RJ45, 1
SD card interface	1
LED	PWR, ETH, STU, ALM
Power consumption	18 W
Operating temperature	32 to 104°F (0 to 40°C)
Storage temperature	50 to 140°F (-10 to 60°C)
Operating humidity	10 to 90% RH (non-condensing)
Storage humidity	5 to 90% RH (non-condensing)
Power	
Input voltage	100 to 240 VAC, 50 to 60 Hz, 1A max.
Power adaptor	Chinese GB-2099, Euro CEE7/7, North American NEMA5-15
Hardware	
CPU	1 GHz
RAM	256 MB
Flash	32 MB
SD card	32 GB
Real time clock	support
Performance	
Call capacity	300 concurrent calls with TLS encryption and decryption. 600 concurrent calls without encryption and decryption
User registration	3000 SIP terminals with unencrypted or encrypted registration (within 5 minutes)
Packets transfer delay	＜30 microseconds
Protocol	
SIP (UDP)	Supports RFC3261/3262/3264/2976, RFC4028 (SESSION TIMER), RFC5009 (Early Media), RFC3966 , RFC2617, and RFC3323
TCP/UDP/IP	Support
MSRP	Support
RTP/RTCP	Support
HTTP/HTTPS	Support
ARP	Support
DNS	Support
NTP/SNTP	Support
SFTP	Support
ICMP	Support
Maintenance	
TR069	Support
Telnet	Support
SSH	Support
Web GUI based on HTTPS for management	Chinese and English
RS-232 console	Support

Remote upgrade	Support
Remote configuration and maintenance	Support
Export/Import data	Support
Auto provision	Support
Static route table configuration	Support
Static IP address	Support
Version info	Includes hardware and software version information
Terminal status	Registration and operation status
Export log	Support
Security for management	Password and authorized IP address
Logon levels	Two levels: administrator and operator
Functions	
Voice and video communication	Support
Encryption of signaling and media stream	Supports separate TLS/SRTP encryption/decryption for SIP signaling and media stream. Supports TLS/SRTP encryption. Both TLS and SRTP must be enabled, because SRTP must work with TLS.
Voice processing	RTP packet forward, firewall traversal, local/remote user NAT traversal
Multiple SIP servers	Connections to multiple (up to 5) SIP servers
Security policies	Access control through IP table, conceal of NAT address, encryption & decryption on signaling and/or voice media streams, blocking calls from untrusted devices, discarding unknown messages, security log file
Access security	Web access anti-cracking, password complexity check, Telnet/SSH enabling control, Web access whitelist, and Telnet access whitelist
Web/Telnet whitelist	Support up tp 30 whitelists which can be set as host IP address but not network segment.
Registration and authentication	Support registration and authentication of SIP terminals
QoS	Support IP TOS
Packet fragmentation	Support fragmentation of signaling packets, with an MTU of 1500 bytes.
Deployment location	UC2.2/UC1.0/IPT networking: Support deployment on the border between large and small networks and deployment behind the NAT (near-end NAT). UC2.0 networking: Supports deployment on the border between large and small networks but no support for deployment behind the NAT.
Log backup	Supports automatic dumping of operation and debugging log entries to an SD card.
Certification	
FCC	Support
CE	Support