WIRELESS WISDOM

RTX8663

IP DECT BASE
W. OPTIONAL DSP AUDIO MODULE
PRODUCT INFORMATION SHEET

The RTX8663 is a patent protected scalable multi-cell IP DECT Base Station for Small, Medium and Large Enterprise. The optional DSP Audio Module for RTX8663 enables additional audio codecs for the RTX8663 base stations.

RTX 8663 IP DECT BASE STATION

The RTX8663 is a scalable multi-cell IP DECT Base Station for Small, Medium and Large Enterprise. The multi cell system offers a unique redundant solution that ensure the best up time. The system is scalable from 1 base, and 30 user and up to 4000 bases and 16000 users, the scalability is achieved by used patent protect technology. The system easily integrates towards a cloud solution that ensures easy maintenance, high security, and flexibility.

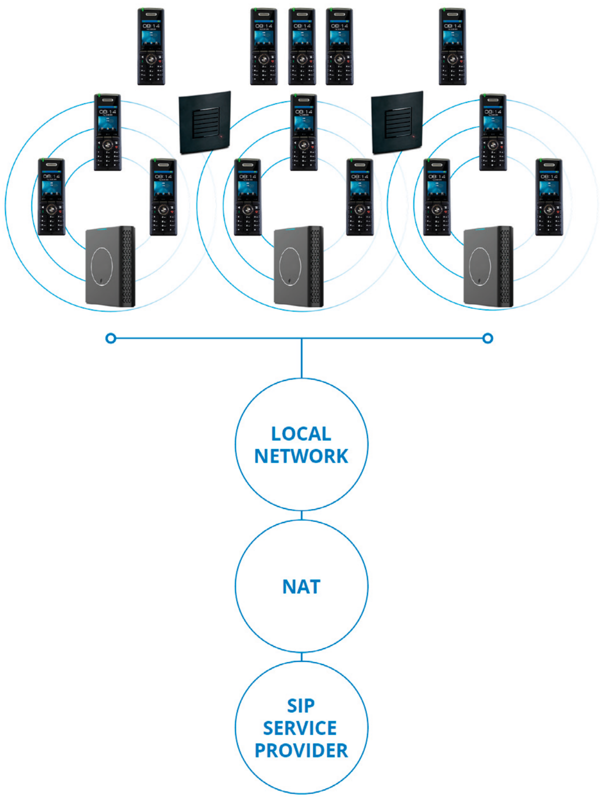

Figure 1: RTX8663 setup scenario:
3 RTX8863 bases, 2 repeaters and 14 handsets

FEATURES

- Wideband and narrowband audio support
- 12 slot radio with up to 10 voice channels active
  - 10 audio channels using G.726 / G711 codec
  - 10 audio channels using G729AB / BV32
  - 8 audio channels using OPUS
  - 5 CAT-iq wideband audio channels using G.722
- Worldwide radio power levels / frequency bands
- Scalable system from 1 to 4000 bases in same network
- 16000 subscriptions (max 16000 handsets)
- Power over Ethernet or DC supply
- IEEE1588 Lan and/or Air synchronization
- Support software download to wireless terminals
- LED status indication
- LDAP and/or XML phonebook support
- Seamless handover
- Repeater support
- Auto/Remote provisioning
- Supports all RTX devices

OPTIONAL DSP AUDIO MODULE

The DSP Audio Module for RTX8663 enables additional audio codecs for the RTX8663 base stations.

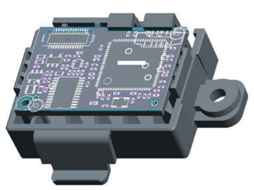

Figure 2: DSP Audio Module for RTX8663

CODEC SUPPORT

The DSP modules adds support for the following audio codecs:

VOIP codecs

- G.729 with Annex A and optional B (NB, 8 kbit/s)
- BroadVoice32 (BV32) (WB, 32 kbit/s)
- Opus (NB/WB, 10-24 kbit/s)

DECT codecs:

- BroadVoice32 (BV32) (WB, 32 kbit/s, 1 DECT slot)

TECHNICAL SPECIFICATIONS

DECT	• Frequency band: 1880 MHz – 1930 MHz (DECT) - 1880 – 1900 MHz (10 carriers) Europe - 1910 – 1930 MHz (10 carriers) Latam - 1920 – 1930 MHz (5 carriers) US and customized frequency bands • Four power levels (14, 17, 20 and 24 dBm) • Seamless handover using connection handover • Wideband voice (HDSP) Basic • Interoperability, Phase I (CAT-iq 1.0) • Authentication / encryption of base and handset • DECT ULE • CAT-iq 2.0 security
AUDIO	• 10 audio channels using G.726 / G711 codec • 5 CAT-iq wideband audio channels using G.722 • RFC3711 SRTP WITH DSP MODULE • 10 audio channels using G.729 • 10 wideband audio channels using BV32 • 8 audio channels using OPUS
ANTENNAS	• Internal omni-directional antennas • Range: Indoor: 50 m • Range: Outdoor: 300 m • Fast antenna diversity switching
ADDITIONAL FEATURES	• TR069 • Zero Touch Installation • Syslog (RFC3164 and RFC5424) • LDAP(s) • UASCTA • CTI • Picture CLIP • Push to Talk (< 250ms latency) • Intercom • System monitoring • Realtime sanity checks • Build-in Certificates • Actions URL's • Priority of Emergency Calls • Pairing of headset and handset • GDPR Compliant • SIP Redundancy • Dialplan • Text messaging • Alarm Server support
PBX COMPLIANCE	• Asterisk • 3CX • Centile • Metaswitch • BroadSoft - DMS - XSI - Shared call appearance
POWER SUPPLY	• Power over Ethernet (PoE): 37-57V - IEEE802.3af (Class 2) • DC power input: 5VDC@2A
NETWORK	• TFTP, HTTP, HTTPS for remote configuration and firmware download • DHCP options 66 and custom • HTTPS or HTTP embedded web server • IPv6 • TLS 1.2 • SNTP • LLDP-MED • IEEE 802.1Q VLAN • TOS / QOS • 802.11
ETHERNET	• Connector: RJ 45 • 10/100 Base-T interface (IEEE802.3)
MECHANICS	• Housing: IP20 • Dimensions: 144x140x35mm (HxWxD) • Temperature Range: -10° to +50°
OTHER	• LED status indication • Firmware update
APPROVALS	• EN 301406 (TBR6) • EN 30176 - 2 (TBR10) • EN 60950 - 1 (Safety) - IEC60950 -1 - CSA c-CSA-us or UL60950 - 1 • EN 301489 (EMC, ESD) • FCC part 15D, conducted & radiated • RSS213 • EN 62311 (SAR)

TECHNICAL SPECIFICATIONS

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- RFC2327 SDP: Session Description Protocol
- RFC2387 The MIME Multipart / Related Content-type
- RFC2396 Uniform Resource Identifiers (URI): Generic Syntax
- RFC2543 Session Initiation Protocol (HOLD Option)
- RFC2833 In-band DTMF
- RFC2976 The SIP INFO method
- RFC3261 SIP 2.0
- RFC3262 Reliability of Provisional Responses in the Session Initiation Protocol (PRACK)
- RFC3263 Locating SIP Servers (DNS SRV, redundant server support)
- RFC3264 Offer/Answer Model with SDP
- RFC3265 Specific Event Notification
- RFC3310 Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)
- RFC3311 The Session Initiation Protocol UPDATE Method
- RFC3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC3325 P-Asserted Identity
- RFC3326 The Reason Header Field for the SIP
- RFC3361 Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers
- RFC3420 Internet Media Type message/sipfrag
- RFC3489 STUN
- RFC3515 REFER: Call Transfer
- RFC3550 RTP: A Transport Protocol for Real-Time Application
- RFC3581 An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
- RFC 3665 Basic Call Flow Examples
- RFC3680 A Session Initiation Protocol (SIP) Event Package for Registrations
- RFC3711 HTTP Digest and HTTPS (server/client)
- RFC3842 Message Waiting Indication
- RFC3891 Replace header support
- RFC3892 SIP Referred-By Mechanism
- RFC3960 Early Media and Ringing Tone Generation
- RFC3966 The tel URI for Telephone Numbers
- RFC4028 Session Timers in the Session Initiation Protocol (SIP)
- RFC4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC4244 An Extension to the Session Initiation Protocol (SIP) for Request History Information
- RFC4475 SIP Torture Test Messages
- RFC4566 SDP: Session Description Protocol
- RFC4662 A Session Initiation Protocol (SIP) Event Notification Extension for Resource Lists
- RFC4733 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC4749 RTP Payload Format for the G.729.1 Audio Codec
- RFC4961 Symmetric RTP / RTP Control Protocol (RTCP)
- RFC5393 Addressing an Amplification Vulnerability in Session Initiation Protocol (SIP) Forking Proxies
- RFC5630 The Use of the SIPS URI Scheme in the Session Initiation Protocol (SIP)
- RFC5939 SDP Capability Negotiation
- RFC5954 Essential Correction for IPv6 ABNF and URI Comparison in RFC 3261
- RFC6035 SIP Package for Voice Quality Reporting Event
- RFC2806 URLs for Telephone Calls
- RFC3551 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC3840 Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)
- RFC1889 RTP: A Transport Protocol for Real-Time Applications
- RFC1890 RTP Profile for Audio and Video Conferences with Minimal Control
- RFC2474 Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers
- RFC791 Internet protocol
- RFC1349 Type of Service in the Internet Protocol Suite
- RFC6176 Prohibiting Secure Sockets Layer (SSL) Version 2.0