RTX9431

BASE STATION

PRODUCT SHEET

The RTX9431 is the Next Generation Single Cell Base Station for small and medium businesses and enterprises. It can easily be upgraded to a dual cell that extends range and call capacities.

RTX9431 SINGLE CELL BASE STATION

The Next Generation Single Cell Base Station RTX9431 can easily be upgraded to a dual cell that extends range and call capacities. The dual cell uses the same patent-protected tech as the RTX8663 Multi Cell system. The Dual Cell offers 100% security and plug and play of the second base.

Figure 1: RTX9431 Base Station

FEATURES

- IP DECT solution for small office environment
- Up to two RTX9431 in one system
- Supports 1 – 20 users
- Mix between traditional DECT and CAT-iq wideband audio
- Air synchronization
- 6 repeaters support for coverage extension
- Up-to 10 simultaneous calls
- Full security
- Power through standard 5V adaptor and PoE
- Supports Local / Hosted IP PBX or SIP trunk lines
- Wideband audio (HDSP) basic and extended
- 12 slot radio with up to 10 active voice channels
  - 10 audio channels using G.726 / G711 codec
  - 4 audio channels using G.729
- Worldwide radio power levels / frequency bands
- TR069
- LLDP-MED
- Build in Certificates

A COMPLETE SOLUTION

A single Cell DECT solution with the RTX9431 is a perfect fit for small businesses and enterprises. The RTX9431 base stations provide on-site mobility for users who must be contactable wherever they are within the office.

This solution can easily be upgraded to a dual cell supporting up to two RTX9431 in one system. The two base stations can be combined into one system to ensure a larger coverage area, up to 20 calls per system, 8 G729 calls, and seamless handover.

With this solution, it is possible to set up with up to 12 repeaters, which can extend the range even more. For larger businesses and enterprises with a need for more devices and even more range, the RTX8663 Multi Cell DECT solution is recommended.

SECURITY

The audio is encrypted (SRTP), the HTTPS server is enabled (TLS 1.2), and there is build in certificates (50).

DECT communication is encrypted, and all communication between the bases is encrypted with secure LDAP and secure communication to broadwork (XSI).

PLUG AND PLAY

This base station solution has Plug and Play of the 2. base station, with zero touch installation. It has LLDP-MED for easy VLAN deployment, TR069 for standard deployment, remote configuration, and firmware update.

The solution ensures a large range of terminals with Handsets, Headsets, RTX8200 Location Gateway (BTLE Beacon) and a DECT Repeater.

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
- 4 audio channels using G.729 calls
- 5 CAT-iq wideband audio channels using G.722
- RFC3711 SRTP

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
- System monitoring
- Realtime sanity checks and system statistics
- 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 (Optional)

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)

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TECHNICAL SPECIFICATIONS

MECHANICS	• Housing: IP20 • Dimensions: 109x93x24mm (HxWxD) • Temperature Range: -10° to +50°
OTHER	• LED status indication • Firmware update

APPROVALS	• EN 301406 (TBR6)
	• IEC62368-1:2014
	  - EN 62368-1
	  - CSA 62368-1-14
	  - UL 62368-1
	• EN 301489 (EMC, ESD)
	• FCC part 15D, conducted & radiated
	• RSS213
	• EN 50385

TECHNICAL SPECIFICATIONS	• 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