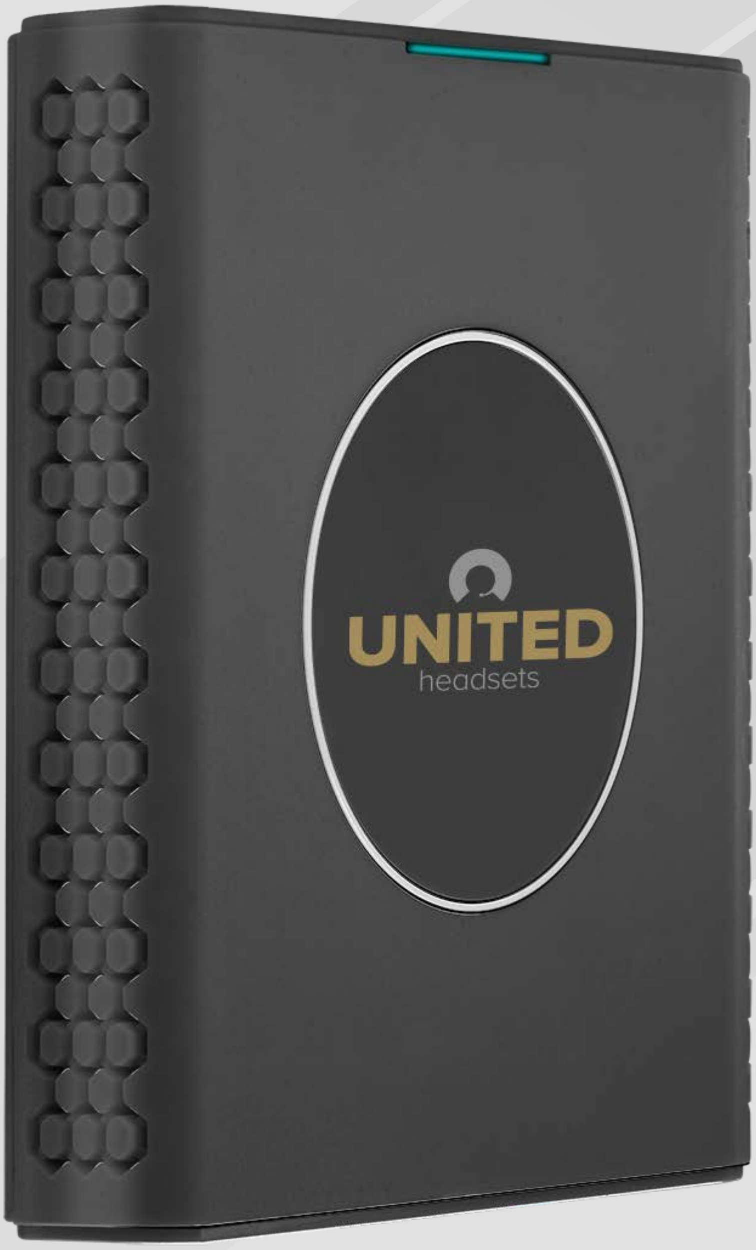

UH-RBASE

IP DECT BASE

The UH-RBASE is a patent protected scalable multi-cell IP DECT Base Station for Small, Medium and Large Enterprise.

UH-RBASE IP DECT BASE STATION

The UH-RBASE 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.

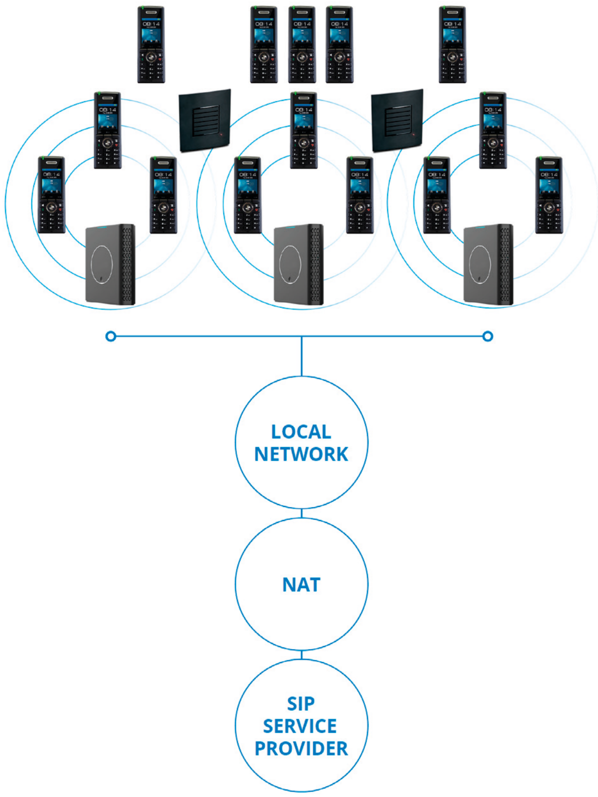

Figure 1: UH-RBASE setup scenario: 3 UH-RBASE bases, 2 Repeaters and 14 Handsets

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 United Headsets devices

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

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