HEAD acoustics

DATA SHEET

APPLICATION EXAMPLE INCLUDED

Code 7770

coreIP

*lab*CORE I/O Module – Voice over IP Reference Gateway

OVERVIEW

coreIP

Code 7770

*lab*CORE I/O Module – Voice over IP Reference Gateway

With *core*IP activated, *lab*CORE becomes a reference gateway for voice quality measurements of IP-based communication terminals using Voice over New Radio (VoNR), Voice over LTE (VoLTE), or Voice over IP (VoIP). *core*IP contains an integrated VoIP SIP client and supports Real-Time Transport Protocol (RTP) among other network protocols. Furthermore, *core*IP supports numerous voice and audio codecs. Additionally, special voice and audio codecs can be added as optional software extensions. Another optional feature provides the possibility for customized network impairments.

KEY FEATURES

Extending *lab*CORE into a full-featured VoIP/VoLTE/VoNR reference gateway

Support of various network protocols

Support of various voice and audio codecs

Optional upgrades for voice and audio codecs (AMR, EVS, Opus)

Optional upgrade for network impairment feature

Exact synchronization between audio signal and IP packets for repeatable measurement conditions

Configuration wizard for connecting to selected third-party radio testers

Establishing calls between *lab*CORE SIP client and device under test via IMS server of a radio tester

Receiving the RTP audio stream of an established call between radio tester and device under test for analyzing the audio data

APPLICATIONS

Performing voice quality measurements of IP-based communication terminals

DETAILS

DESCRIPTION

Supported Protocols

- SIP (Session Initiation Protocol, RFC 3261) via UDP, TCP, TLS
- RTP (Real-time Transport Protocol, RFC 3550), also usable without SIP
- Media Encryption by SRTP and ZRTP
- Firewall Policies NAT, STUN, or ICE
- IPv4 and IPv6

Supported Codecs

- G.711 (A-law, µ-law)
- G.722 (64 kbit/s (Mode 1))
- G.726, AAL2-G.726 (16 kbit/s, 24 kbit/s, 32 kbit/s, 40 kbit/s)
- G.729 Annex A and Annex B
- L16 (16 bit linear PCM at 8 kHz, 16 kHz, 32 kHz, 44.1 kHz, 48 kHz)
- GSM 06.10 Full Rate
- Speex at 8 kHz, 16 kHz, 32 kHz
- SILK at 8 kHz, 12 kHz, 16 kHz, 24 kHz

Advanced Features

- Codec payload type can be modified
- Format-specific parameters (fmtp) can be modified
- Static jitter buffer
  - Initial jitter buffer size can be defined
  - Current jitter buffer size can be reset to initial size
- Adjustable packet length depending on codec
- IP traffic can be monitored

GENERAL REQUIREMENTS

Hardware

*lab*CORE (Code 7700)
- Modular multi-channel hardware platform

Software

One of the following software applications:

ACQUA (Code 6810)
- Advanced Communication Quality Analysis Software, full license version

ACQUA Compact (Code 6860)
- Compact test system

SCOPE OF DELIVERY

*core*IP (Code 7770)
- *lab*CORE I/O Module – Voice over IP Reference Gateway

Initial equipping
- *core*IP software license key is stored on *lab*CORE during production

Retrofitting
- HEAD acoustics supplies the customer with the software license key

OPTIONS

*core*IP has various optional software extensions. It can be upgraded with additional voice and audio codecs and a network impairment feature.

*core*IP-IMP

*core*IP-IMP (Code 7771)
- *lab*CORE VoIP impairment option (*core*IP module required)

Software extension for simulating network impairments. *core*IP-IMP enables delaying or discarding specific RTP packets to simulate impairments like jitter, delay, and packet loss. The measurement conditions are reproducible, even with active Discontinued Transmission (DTX)/Silence Compression.

*core*IP-AMR

*core*IP-AMR (Code 7772)
- *lab*CORE VoIP AMR codec option (*core*IP module required)

Software extension providing the audio codecs: AMR-NB, AMR-WB (G.722.2), and GSM Enhanced Full Rate.

*core*IP-EVS

*core*IP-EVS (Code 7773)
- *lab*CORE VoIP EVS codec option (*core*IP module required)

Software extension providing the EVS audio codec in all specified bandwidths from narrowband to fullband with all bit rates and modes (incl. AMR-WB interoperable mode). *core*IP-EVS features static jitter buffer for accurate delay conditions instead of the adaptive jitter buffer from EVS.

*core*IP-OPUS

*core*IP-OPUS (Code 7774)
- *lab*CORE VoIP OPUS codec option (*core*IP module required)

Software extension providing the Opus audio codec in mono and stereo audio.

*core*IP-WebRTC

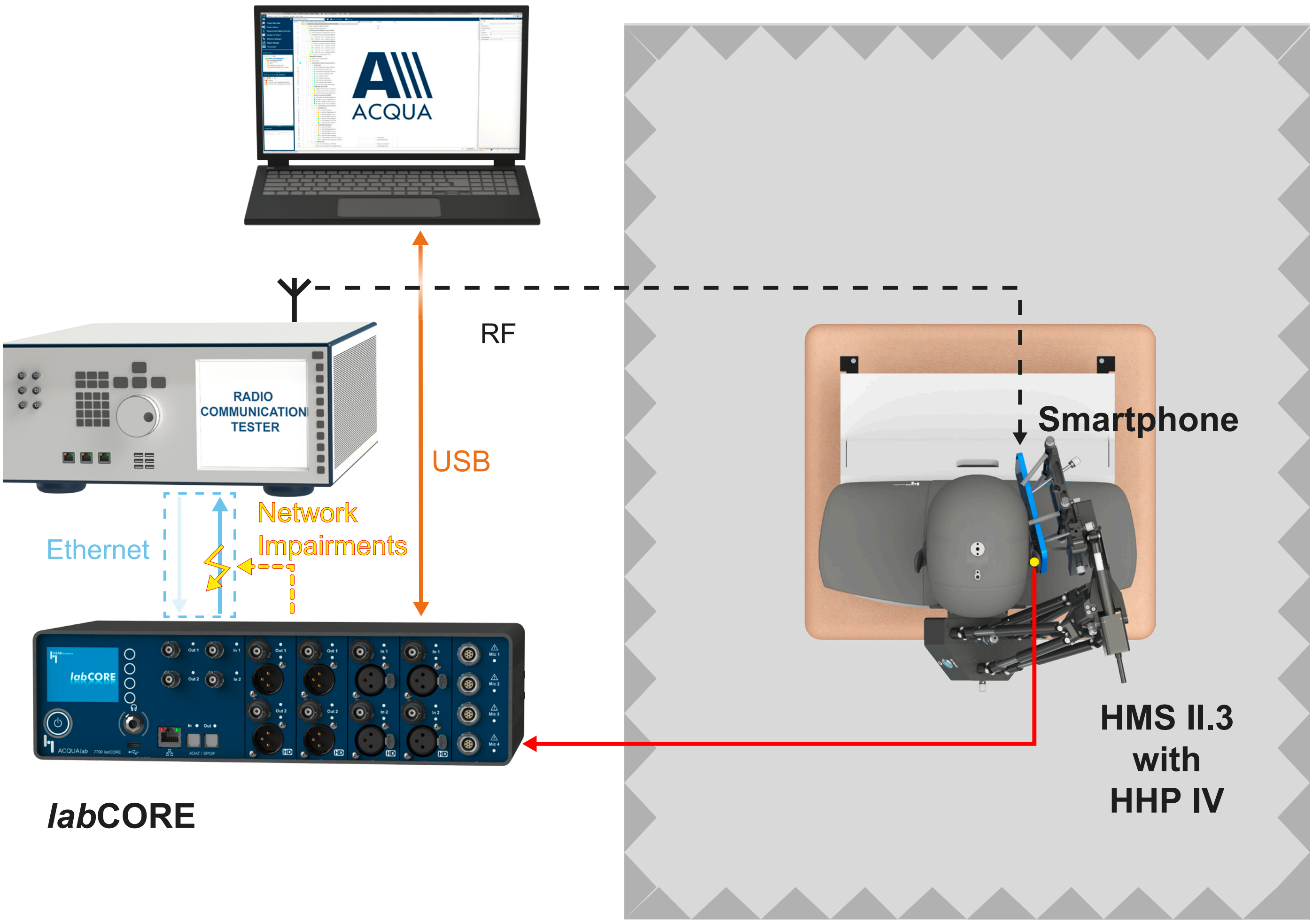
*core*IP-WebRTC (Code 7779)
- *lab*CORE WebRTC option (*core*IP module required)

Software extension that adds the functionality of WebRTC (Web Real-Time Communication) to *lab*CORE. It enables the ability to establish calls with, e.g., browser phones via a SIP server.

IN PRACTICE

Handset: VoNR Measurement with Packet Loss, Jitter, or Delay

The handset is clamped into HHP IV and connects via a packet-switched connection to a radio tester. *lab*CORE is equipped with *core*IP, *core*IP-EVS, and *core*IP-IMP. *core*IP is the basis for transmitting a digital audio signal as voice packets. *core*IP-EVS is necessary for encoding the audio signal according to the EVS codec. The EVS codec is also applied by the handset. *core*IP-IMP impairs the RTP packets of the audio signal for challenging the packet loss concealment of the handset. *lab*CORE transmits the impaired RTP packets with the encoded audio signal via the simulated 5G NR network of the radio tester to the handset. The handset processes the impaired RTP packets with the encoded audio signal and plays it back through its earpiece. *lab*CORE receives the signal from HMS II.3 and forwards it to ACQUA for recording. ACQUA analyses and assesses the recorded signal.



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