VoIP Framework

Comfortable audio and video transmission in digital quality and with numerous convenience functions - the VoIP framework vicMEDIA is ideally suited for various communication tasks in buildings, vehicles, intercom, industrial communication and control center technology.
voice INTER connect offers not only ready-made software solutions but also customer-specific extensions and adaptations for communication tasks. These include, for example, digital audio and video connections, announcements for individual devices or equipment groups, voice communication and conference calls.
Extensive additional modules such as equalizers, compressors, automatic gain control, acoustic echo cancellation or noise reduction ensure optimum audio quality, high interference robustness and intelligibility of the communication system.
Upon request, our specialists can fit vicMEDIA into your application. We are also glad to advise you on product optimization and parameterization of your application.

Product Features

- Object-oriented communication stack
- Digital audio and video transmission in telephone and multimedia quality
- Ready for various communication tasks: announcements for devices or equipment groups, voice communication and conference calls
- Modules for intelligibility enhancement (ANR, AGC, compressor, AEC)
- Comfortable control and parameter interface
- Test environment and QoS

Applications

- Building communication
- On-board communication (tram, bus, train, ship, plane)
- Control rooms
- Conference systems

vicMEDIA

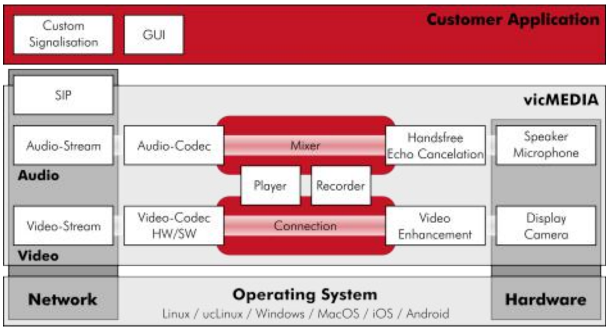
SIP-based Audio and Video Transmission.

voice INTER connect

Technical Data

Functions

- Calls and convenience features corresponding to SIP specificiation
  - Call mute, hold/resume, conference calls
  - Several, simultaneously active calls
  - Priority setting and waiting lists
  - Operation with and without SIP server possible
- Public announcement (speech, music) via multicast and broadcast



Specification

- SIP calls (RFC 3261 and others): instant messaging (RFC 3428), SDP (RFC 4566), presence notification, DTMF (RFC 2833 or SIP INFO)
- Audio codecs: G.711, G.722, GSM, iLBC, Speex, G.729 (optional integration of additional codecs possible)
- Video codecs: MJPEG, H.264, integration of additional codecs possible, support for hardware acceleration (i.MX6)
- Media transport: RTP/RTCP, SRTP/TLS (for secured connections), multicast, broadcast, packet loss concealment (for interrupted connections), adaptive jitter buffer (for latency optimization)
- Echo cancellation (vicHSES) already pre-integrated: Acoustic echo cancellation (full-duplex), line echo cancellation (full-duplex), loss control, noise reduction, limiter, compressor, equalizer, noise gate

Compatibility

- Due to abstraction layer largely independent of OS and hardware. Supported are:
  - Processors: x64/x86, ARM, Blackfin, MIPs, PowerPC
  - OS: Windows, Linux/ucLinux, MacOS, iOS, Android
- Flexible audio / hardware support:
  - OS-specific APIs: Linux ALSA, Windows WMME, MacOS CoreAudio
  - Customised driver, e.g. bidirectional audio streaming over RS485
- Support of hardware accelerators, e.g. Freescale i.MX6 VPU/IPU (codec, transformation)
- Flexible application:
  - Server for VoIP and audio streaming functions
  - Direct and remote monitoring of customer application via TCP socket
  - Integration in customer application as library
- Based on PJSIP-Framework and Gstreamer
- Transmission of customer specific information with RTP-stream possible, e.g. control information
- Integration and test support with QoS-tools
- Easy expandible through object oriented architecture

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