vicPURE voice

Speech Enhancement for Communication Products

In vehicles, conference systems and door communication, robust speech transmission as well as crystal-clear voice quality are essential for effective communication between end customers, operators and control centre. voice INTER connect's speech enhancement and multimedia processing framework vicPURE not only offers the most comprehensive set of algorithms on the market (including speech enhancement, echo cancellation, beamforming and noise suppression), but is also available on numerous embedded platforms and runs in real-time with minimal resource requirements.
For a quick evaluation of performance, we offer our customers our dedicated OEM hardware platforms or close-to-product system designs.
Our team of experts will accompany each customer project individually with their comprehensive experience and adhering only to the highest standard of voice quality: from requirements engineering via design and development to validation and series release.

PRODUCT FEATURES

- Speech enhancement adapted to environment and speaker
- Crystal-clear, full-duplex hands-free communication
- Robust, high-performing noise cancellation
- Integrated beamforming for digital directional microphones
- Adaptive speaker tracking also for dynamic or far-field applications
- Complete support for integration and parametrisation
- Extensive multimedia features
- Low power requirements and power consumption on embedded platforms

APPLICATIONS

- Infotainment and driver assistance for cars and commercial vehicles
- Intercom and door communication
- Medical intercom and nurse call
- Conference technology and professional audio
- Control centre technology
- Emergency call and communication systems
- Passenger information systems
- Audio sensors for mobile robots

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Technical Details

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FUNCTIONS

- Support for different microphone array topologies (e.g. distributed, circular, and line arrays, as well as 3-dimensional where applicable)
- Dynamic speaker localisation and separation
- Spatial noise reduction, >20 dB SNR gain depending on application
- Full-duplex echo cancellation, with attenuation >50 dB
- Adaptive noise reduction, with SNR gain up to 20 dB
- Automatic volume control
- Acoustic feedback cancellation
- Reliable, continuous adaptation and convergence, even in the presence of strong noise, reflections and room reverberation
- Available for different signal bandwidths from telephone to multimedia quality

COMPATIBILITY

- Support of various embedded platforms (e.g. ADSP Blackfin & SHARC; TI C6000 & OMAP; NXP i.MX 6 to 9; ARM Cortex M)
- Support of PCs with Windows, Linux, Mac OS, on request also sound API integration (Linux ALSA, Windows MME, Mac OS CoreAudio)
- Mobile devices running iOS, Android, Embedded Linux

RESOURCES REQUIREMENT (EXAMPLES)

	ARM Cortex M4F	**ADI SHARC SC584**
Microphones	3	8
Beamformer	3	adaptiv
Processor load	140 MCPS	400 MCPS
Latency	6 ms	16 ms
Memory requ.	50 kB	200 kB
Noise Reduction	Ja	Ja
Additional	Source loca-lization for 2 simultaneous speakers	Full-duplex AEC, EQ, limiter, compressor

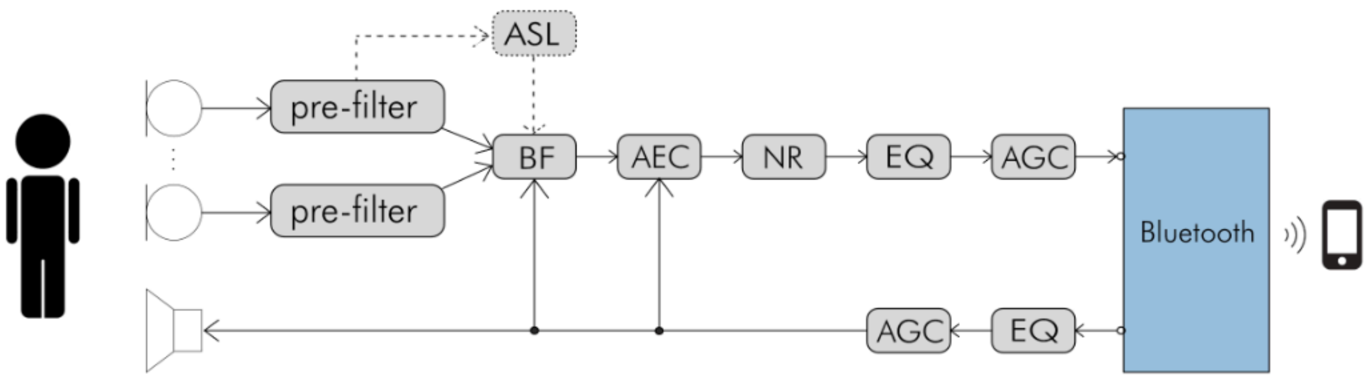

Figure: Typical processing scheme for handsfree communication

SUPPORT SERVICES

- Application-specific microphone array design (number, arrangement, directivity of microphones, microphone selection)
- Mechanical integration design (sound channels, sealing, structure-borne noise)
- Application-specific algorithm simulation for pre-assessment and optimisation
- Acoustic measurements and objective evaluation of attenuation values and speech quality
- Performance optimisation on target hardware
- Manufacturing support (EOL tests, automatic calibration)

FIND OUT MORE

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