Objectives

- Direction provided by sensor (radar, optical)
- Beam width as small as possible.
- Bright spot pressure 70 to 90dBA at 1 m
- $V_d = 300^\circ/s$
- Limited aperture, limited number of sources
- No vertical directivity required

Array configuration

- Aperture $L$, influences beam width $BW$ (delay-and-sum beamformer)
  \[ BW = \frac{\theta}{\lambda} \]
- Spacing $d$, influences maximum frequency
  \[ \frac{\lambda}{2\pi d} \leq \Delta f \]
- Elements do not require equal spacing ⇒ if the number of the sources is limited then the worst-case beam width can be improved by non-uniform spacing
- Delay-and-sum directivity is the starting point ⇒ improved directivity with super-directive beamformers

Beamforming algorithms

- Use of measured transfer functions
  - No point sources
  - Coupling between sources
  - Complex geometry
- Delay-and-sum (DS) and 4 Super-directive beamformers:
  - Contrast Control (CC)
  - Acoustic Energy Difference (AED)
  - Least Squares (LS)
  - Sound Power Minimisation (SPM)
- Regularization
  - Reduced control signals
  - Stability
- Pressure constraint
- Ground reflections

Beam shape comparison

Influence of the algorithm

- Delay and sum
- Sound power minimisation

Uniform array geometry in both cases
Beam shape comparison

- Uniform array: -0.2860 -0.1716 -0.0572 0.0572 0.1716 0.2860 m → width: 0.572 m.
- Nonuniform array: -0.3718 -0.2002 -0.0572 0.0572 0.2002 0.3718 m → width: 0.7436 m.

Directivity

Transducer characterization (LMS)

- Maximum force, maximum sound pressure levels
- Power compression (heating of voice coil)
- Directivity
- Harmonic distortion

Maximum SPL with 6 element array

Source 1: 5cm loudspeaker type 1 in 0.07 L closed box
Source 2: 5cm loudspeaker type 2 in 0.07 L vented box tuned at 250 Hz

Finite Element acoustic model of the car front and environment (LMS)

Transfer function simulations by Bert van Genechten, LMS
Beamforming simulation by TNO

Influence of ground reflection (TNO, LMS)

- 6 sources, non-uniform array, 1.5 kHz max
- 6 sources, non-uniform array, 1.5 kHz max
- 6 sources, non-uniform array, 1.5 kHz max
- 6 sources, non-uniform array, 1.5 kHz max
- 6 sources, non-uniform array, 1.5 kHz max
- 6 sources, non-uniform array, 1.5 kHz max
Environmental conditions

Environmental conditions that may influence the performance of an acoustic beam-forming system:
- Temperature gradients
- Wind gradients
- Air turbulence
- Air density
- Sound speed
- Humidity
- Presence of reflecting objects
- Variation of ground impedance
- Reverberant level of the environment
- Free-field condition
- Other sound sources in the environment
- Level of incoherent background noise
- Presence of coherent/incoherent noise sources
- Time varying coherence

Examples of beam simulations

- Beam steering at an angle of 30 degrees
- Forward beam steering, Temperature: -10°C (20°C nominal)

Inertial mass shaker feasibility

- Transfer function from actuator current to pressure
- Impedance measurements on the front bumper
- Transfer functions from actuator current to bumper acceleration

Experimental validation

- Measurements in free-field, on asphalt
- 8-element uniform loudspeaker array for 3kHz max
- Using real-time implementation with Matlab/Simulink
- Measured transfer functions are used to determine the frequency domain control coefficients

Beamforming algorithms

- Transfer functions and coherence, 8 sources
  - 5 m distance
  - Sources 0.6 m above the ground
  - Microphones 1.75 m above the ground
Real-time results

- AED and CC lead to a reduction of the signal output if the coherence is reduced. SPM sustains the beam at more frequencies.
- Beam of SPM is slightly more narrow than for AED.
- AED has somewhat lower sidelobes, but precise tuning is required.
- Beam of DS is relatively wide, especially at low frequencies.
- SPM and AED give the best overall acoustic performance for this application.

Real-time implementation

- Latency requirement → Time-domain implementation.
- Required tracking speed is 300º/s.
- In case of a 30º beam width ⇒ deadline < 50ms.
- The control coefficients can be computed efficiently for each new beaming angle with the sound power minimization strategy.
  - The size of the matrix inversion that needs to be recomputed depends on the number of acoustic constraints, i.e., the number of beams, which is usually low.
  - Other matrix inverses are more complex, but do not have to be recomputed and therefore can be stored.
Real-time implementation on embedded platform

- Two real-time implementations:
  - Matlab/Simulink for laboratory tests and for measurement of the transfer functions
  - SHARC® floating-point DSP (1 GFLOP) for embedded systems using uploaded transfer functions
- On the embedded platform:
  - Sound power minimisation and delay-and-sum algorithms, real-time implementation validated with Matlab/Simulink version
  - A new set of FIR filters will be computed if a new direction is required
  - New 256-tap FIR filters will be available in less than 5ms

Array with uniform sound radiation

- Method: Sound pressure constraint at 5 different directions in the range -60° to +60°
- Method: Most efficient radiation patterns in the range -60° to +60°

Conclusions

- Five beamforming algorithms compared in real-time implementation
- A flexible and efficient implementation is possible with sound power minimization
- Sensitivity study based on finite element simulations of an acoustic array mounted at the front of a car for different environmental conditions
- A single array configuration can be used for both directional sound generation and uniform sound radiation