Transaural Audio - The reproduction of binaural signals over loudspeakers

Fabio Kaiser

Outline

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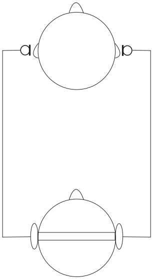
Binaural technology

- 3D immersive audio technology
- Provides interaural cues for sound localization
- Reproduction with headphones

⇒ Why not loudspeakers?

⇒ Crosstalk!!!



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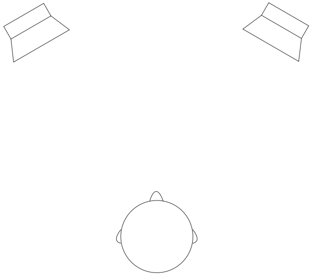

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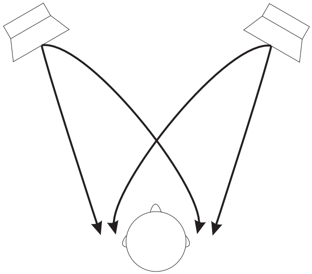

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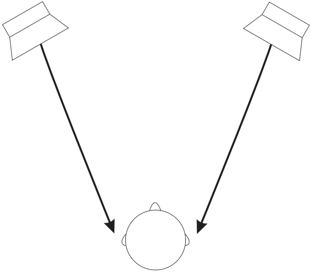
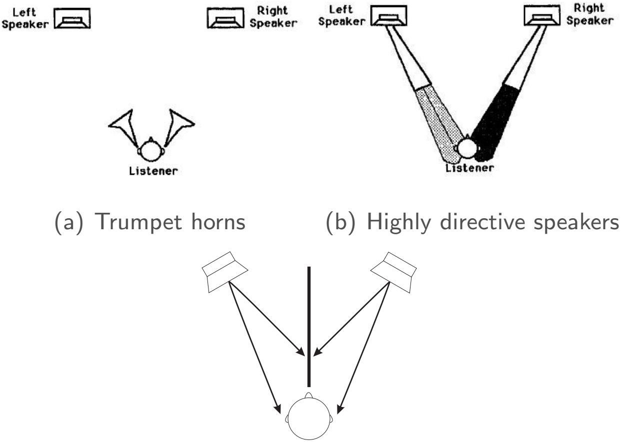


Acoustical means for crosstalk cancellation



(a) Trumpet horns

(b) Highly directive speakers

(c) Physical barrier

Fig. 1: A few suggestions for acoustical CTC [Boc86]

This work

$\Rightarrow$ Signal processing techniques for crosstalk cancellation

Introduction to transaural audio

- Virtual headphones

⇒ Crosstalk cancellation (CTC) + direct path cancellation

- History
  - Bauer 1961 [Bau61]
  - Atal and Schroeder 1973 - First implementation [Sch73]
  - Cooper and Bauck 1989 - Shuffler filters [CB89]
  - Kirkeby et al. 1998 - Stereo Dipole [KN98]
  - Takeuchi et al. 2000 - Optimal Source Distribution (OSD) [TN00]

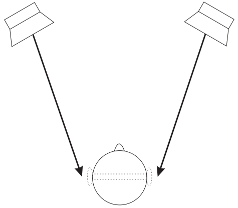
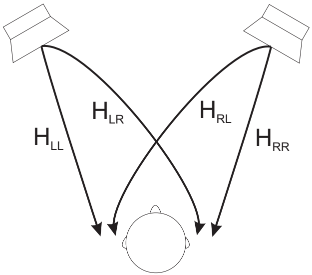


Fig. 2: Virtual headphones

Problem formulation (1)

- Matrix formulation (frequency domain)
- The head transfer matrix **H** contains the acoustic paths inherent to the plant



$$\mathbf{e} = \mathbf{H}\mathbf{y} \tag{1}$$

$$\mathbf{y} = \begin{bmatrix} y_L \\ y_R \end{bmatrix} \quad , \quad \mathbf{e} = \begin{bmatrix} e_L \\ e_R \end{bmatrix} \quad , \quad \mathbf{H} = \begin{bmatrix} H_{LL} & H_{RL} \\ H_{LR} & H_{RR} \end{bmatrix} \tag{2}$$

- **y**...loudspeakers signals
- **e**...ear signals

Problem formulation (2)

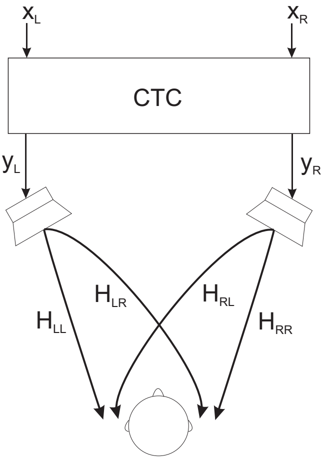
- Process input signals $\mathbf{x} = \begin{bmatrix} x_L \\ x_R \end{bmatrix}$
- Cancellation matrix **C** cancel for the head transfer matrix **H**

$$\mathbf{e} = \mathbf{HCx} \tag{3}$$

$$\mathbf{C} = \mathbf{H}^{-1} = \frac{1}{D} \begin{bmatrix} H_{RR} & -H_{RL} \\ -H_{LR} & H_{LL} \end{bmatrix} \tag{4}$$

$$D = H_{LL} H_{RR} - H_{LR} H_{RL} \tag{5}$$

⇒ **Inversion problem**



Design questions

- Static vs. dynamic
- Two-channel vs. Multi-channel
- Location of transducers
- Matched vs. mismatched plant
- Implementation method and computational complexity

Outline

Inversion of non-minimum phase filters - Theory (1)

- Unilateral z-transform of a discrete-time signal $h[n]$ (FIR)

$$H(z) = \sum_{n=0}^{\infty} \frac{h[n]}{z^n} \tag{6}$$

- The direct inverse is

$$G(z) = \frac{1}{H(z)} \tag{7}$$

- If one zero of $H(z)$ lies outside the unit circle

$\Rightarrow$ $G(z)$ is instable

Inversion of non-minimum phase filters - Theory (2)

- Head-related impulse response (HRIR)
  - Finite impulse response (FIR)
  - Typically 128-512 samples
- Mixed phase characteristics
  - Non-minimum phase part due to reflections in the pinna
  - Sound energy focused to the ear canal after direct sound arrived

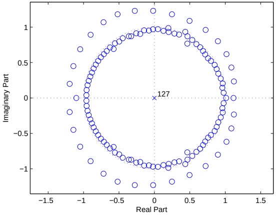


Fig. 3: Zero-pole plot of a HRIR ($\phi = 30°, \theta = 0°$, length is 128 at 44.1kHz)

Inversion of mixed phase filters - Theory (3)

- Choose region of convergence (ROC) to include unit circle
- Inverse $g(n)$ then is
  - Stable
  - Two-sided
  - Infinite

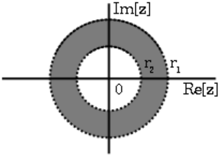


Fig. 4: ROC of a two-sided sequence

- Window $g(n)$ to obtain a FIR filter

$$\tilde{g}[n] = g[n] \cdot w[n], w[n] = \begin{cases} 1, L < n < M \\ 0, otherwise \end{cases} \quad (8)$$

$\Rightarrow$ Maximal energy inside window

- Delay if $\tilde{g}[n]$ is anti-causal

Inversion using the DFT

- Discrete Fourier transform
- Inversion in the frequency domain
- Regularization by limitation of magnitude
- Procedure
  - Compute $H(k)$, the DFT of $h[n]$
  - Invert $H(k)$
  - Limit the resulting magnitude response
  - Compute the inverse DFT
  - Delay result if necessary
- Performance is determined by DFT size (time-aliasing)
- Limitation yields poles with shorter decay time

Least-squares technique (1)

- Error function $e[n]$

$$e[n] = \delta[n] - \tilde{\delta}[n] \tag{9}$$

- $\delta[n]$...desired output
- $\tilde{\delta}[n]$...actual output
- Minimize squared error

$$J = \frac{1}{L}\sum_{n=0}^{L} e^2[n] = \frac{1}{L}\sum_{n=0}^{L}(\delta[n] - \tilde{\delta}[n])^2 \tag{10}$$

Least-squares technique (2)

- Write deconvolution in matrix form

$$\mathbf{d} = \mathbf{H}\tilde{\mathbf{g}} \tag{11}$$

- $\mathbf{d}$... desired output (with delay) of length L
- $\mathbf{H}$...Toeplitz matrix of size $L \times M$
- $\tilde{\mathbf{g}}$...inverse filter of length M
- The error function is then

$$\mathbf{e} = \mathbf{d} - \mathbf{H}\tilde{\mathbf{g}} \tag{12}$$

- Minimize squared error $||e||^2 = e^T e$
- Set gradient to zero
- This yields

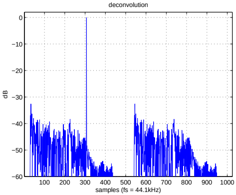
$$\tilde{\mathbf{g}} = (\mathbf{H}^T\mathbf{H})^{-1}\mathbf{H}^T\mathbf{d} \tag{13}$$

Minimum phase inversion

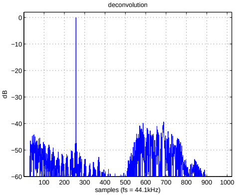
- HRIR is of minimum phase
  - Except for a nearly frequency independent delay
- The delay is ignorable for the inversion (Schroeder)
- Procedure
  - Invert the magnitude response only
  - Compute the minimum phase by using the Hilbert transform of the log-magnitude spectrum

$$arg[H(j\omega)] = -\mathcal{H}\left\{log(|H(j\omega)|)\right\} \tag{14}$$

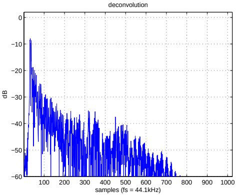
Comparison



(a) DFT inversion



(b) Least-squares



(c) Min phase inversion

Fig. 5: Deconvolution using a filter length of 512 samples.

Outline

Implementation of CTC

Problem

How to implement the crosstalk cancellation matrix **C**?

- First Atal and Schroeder 1973
- Assumed a perfectly symmetric loudspeaker setup
- **H** can be rewritten to

$$\mathbf{H} = \begin{bmatrix} S & A \\ A & S \end{bmatrix}, \mathbf{C} = \frac{1}{S^2 - A^2} \begin{bmatrix} S & -A \\ -A & S \end{bmatrix} \tag{15}$$

- S...ipsilateral HRTF
- A...contralateral HRTF

Implementation of CTC

- Reordering leads to their implementation

$$\mathbf{C} = \frac{\frac{1}{S}}{1 - (\frac{A}{S})^2} \begin{bmatrix} 1 & -\frac{A}{S} \\ -\frac{A}{S} & 1 \end{bmatrix} \qquad (16)$$

- The capital $C = -\frac{A}{S}$, not to confuse with the inverse head transfer matrix $\mathbf{C}$

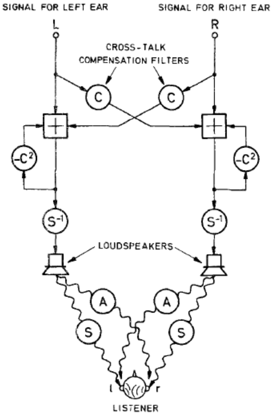


Fig. 6: The crosstalk canceler by Atal and Schroeder [Sch73]

General topology

- Direct implementation of

$$\mathbf{C} = \mathbf{H}^{-1} = \frac{1}{D}\begin{bmatrix} H_{RR} & -H_{RL} \\ -H_{LR} & H_{LL} \end{bmatrix} \tag{17}$$

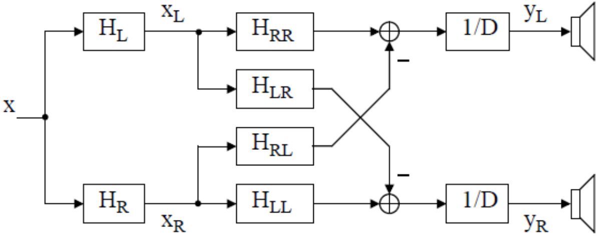


Fig. 7: Block diagram of the general topology including the binaural synthesis stage [Gar97].

Shuffler topology (1)

- Cooper and Bauck 1989
- Reformulation of **H**

Lemma

A $n \times n$ *matrix* **A** *is diagonalizable if it can be written as*

$$\mathbf{A} = \mathbf{P}\mathbf{D}\mathbf{P}^{-1} \tag{18}$$

- **D**...$n \times n$ diagonal matrix containing the Eigenvalues of **A**
- **P**...$n \times n$ matrix containing the corresponding Eigenvectors

Shuffler topology (2)

- Assume $\mathbf{H}$ to be symmetric
- Diagonalize $\mathbf{H}$
- The inverse can be written to

$$\mathbf{H}^{-1} = \frac{1}{2}\begin{bmatrix}1 & 1\\ 1 & -1\end{bmatrix}\begin{bmatrix}\frac{1}{S+A} & 1\\ 1 & \frac{1}{S-A}\end{bmatrix}\begin{bmatrix}1 & 1\\ 1 & -1\end{bmatrix}. \tag{19}$$

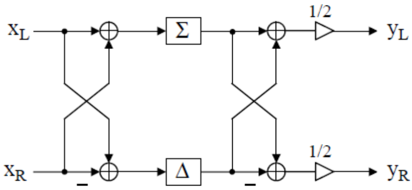


Fig. 8: Shuffler topology. Here Σ denotes the sum filter and Δ the difference filter [Gar97].

Shuffler topology (3)

- Two filters are of joint-minimum phase
  - Excess phase is equal
  - Excess phase is a frequency independent delay
- Sum and difference filter
  - Excess phases ≈ the same
  - Excess phase ≈ frequency independent delay

⇒ Excess phase can be ommited

- Invert the minimum phase part only

Asymmetric shuffler form

- Vandernoot PhD 2001
- Topology is equivalent to shuffler form

$$
\begin{aligned}
H_{LL} &= \frac{H_{RR} + H_{RL}}{D} \\
H_{LR} &= \frac{H_{LR} + H_{LL}}{D} \\
H_{RL} &= \frac{H_{RR} - H_{RL}}{D} \\
H_{RR} &= \frac{-H_{LR} + H_{LL}}{D}
\end{aligned}
\tag{20}
$$

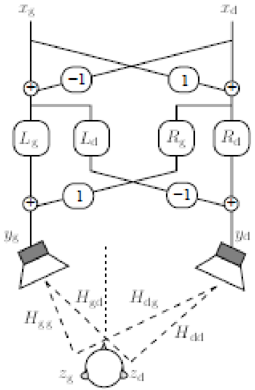


Fig. 9: Asymmetric shuffler topology [Van01].

Fast least-squares deconvolution (1)

- Kirkeby et al. 1998 [KNHOB98]
- Based on the least-squares approximation in frequency domain
- Cost function
  - Performance error
  - Effort penalty

$$J(z) = e^H(z)e(z) + \beta v^H(z)v(z) \tag{21}$$

- β...regularization parameter
- Finding the minimum of the cost function yields

$$\mathbf{C}(z) = \left[\mathbf{H}^T(z^{-1})\mathbf{H}(z) + \beta\mathbf{I}\right]^{-1}\mathbf{H}^T(z^{-1})\mathbf{A}(z) \tag{22}$$

Fast least-squares deconvolution (2)

- Using the fast fourier transform (FFT)

$$\mathbf{C}(k) = \left[\mathbf{H}^T(k)\mathbf{H}(k) + \beta\mathbf{I}\right]^{-1}\mathbf{H}^T(k)\mathbf{A}(k) \qquad (23)$$

- k...k-th frequncy index
- Implementation summery
  - Calculate the N-point FFT of the relevant HRIRs
  - Calculate **C** by applying Eq.(23)
  - Calculate the N-point inverse FFT of **C**
  - Introduce a modeling delay by doing a cyclic shift of m samples of the resulted IRs. A good value for m is suggested by N/2.
- β has to be set appropriately
- $\beta = 0.0001$ is suggested
- Topology is the direct implementation of **C**

Recursive form (1)

- Iwahara and Mori 1978
- Very intuitive
- Derived by Step-by-step analysis of the crosstalk process
- Interaural transfer function $ITF = \frac{H_c}{H_i}$
  - Predicts crosstalk
- Compensates for higher-order crosstalk

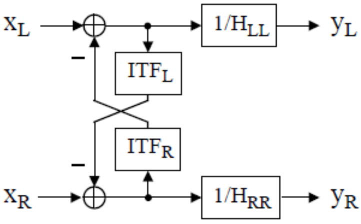


Fig. 10: Block diagram of the recursive topology [Gar97].

Recursive form (2)

- Feedback loop is a power series of ITFs

$$1 + ITF(\omega) + ITF(\omega)^2 + ITF(\omega)^3 + \cdots = \frac{1}{1 - ITF(\omega)} \tag{24}$$

- Instable if absolute values of ITF are
  - Equal to one or
  - Close to one (due to numerical inaccuracies)
- Commercial system *Ambiophonics*
  - Avoids instabilities by band-limitation (250Hz-6kHz)

Summary

- Quantitative comparison

	General	Shuffler	A. shuffler	Fast LS	Recursive
Convolutions	4-6	2	4	4	4
Asymmetry	+	-	+	+	+

Outline

Objective evaluation

- Objective analysis of the conditioning of the problem
- Objective comparison of different implementations
  - Free-field simulation
- HRTF data
  - IRCAM LISTEN database
  - Diffuse field equalized
  - 512 samples

Condition number

- Linear equation system $b = \mathbf{A}x$
  - Well-conditioned $\Rightarrow$ small condition number
  - Ill-conditioned $\Rightarrow$ high condition number
- Measures the sensitivity to numerical errors
- Ratio of the largest to smallest singular value of a matrix

$$\kappa(A) = \frac{\sigma_{max}(A)}{\sigma_{min}(A)} \tag{25}$$

- $\kappa(A) = \infty$

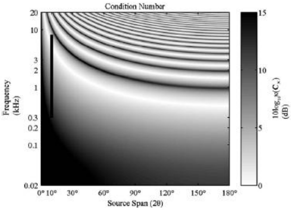
$\Rightarrow$ System is singular

- $\kappa(A) = 1$

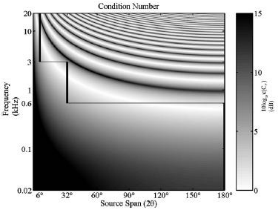
$\Rightarrow$ Perfect inversion is possible

Conditioning of the head transfer matrix

- Ward and Elko 1999 [WE99]
- Condition Number
  - Frequency dependent
  - Source span dependent
- Takeuchi 2000
  - Optimal source distribution (OSD)



(a) Stereo dipole [NR05]



(b) OSD

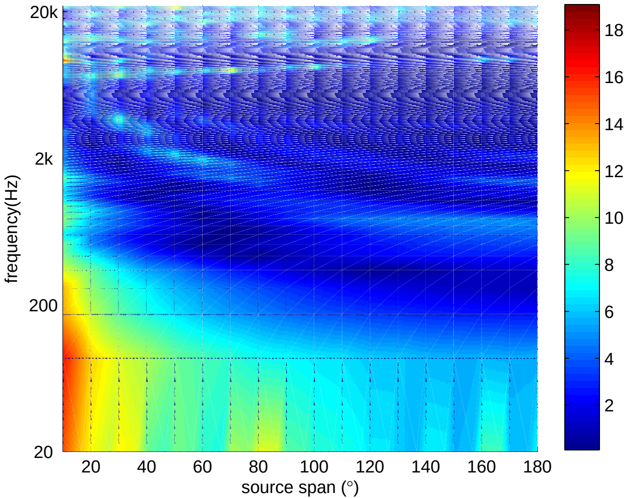


Fig. 11: Condition Numbers as a function of frequency and source span of a measured set of HRTFs. The colorbar indicates $10log_{10}(\kappa(C))$.

Average best-fit

- Frequency average of condition number
- Find source span with smallest average

- Advantage
  - Conventional transducer system
  - Providing the smallest errors
- Disadvantage
  - Local errors at high frequencies
  - Individual differences

Average best-fit

- Source spans in between $130^\circ - 180^\circ$

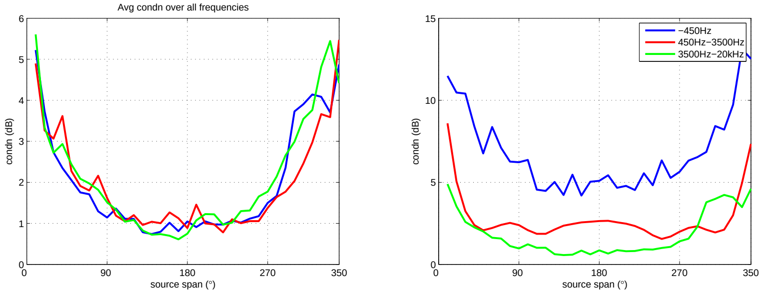


Fig. 12: Frequency average of the condition number. Left, of three different head transfer matrices (subjects). Right, of one head transfer matrix (subject) divided into three sub-bands.

Comparison of implementations - Setup (1)

- Free-field simulation in Matlab
- Measure amount of CTC
- Symmetric source spans only
  - $\pm 5°$, $\pm 30°$ and $\pm 65°$
- Evaluate the different topologies
- Input to system

$$x_L(t) = \begin{cases} 1 & \text{if } t = 0 \\ 0 & \text{otherwise} \end{cases}$$
$$x_R(t) = 0 \tag{26}$$

$\Rightarrow$ Impulse response of system $\mathbf{y}$

- Convolution with plant HRIRs

$\Rightarrow$ Ear signals $\mathbf{e}$

Comparison of implementations - Setup (2)

- Topologies compared
  - General topology (blue)
  - Symmetric shuffler topology (red)
  - Fast least squares deconvolution ($\beta = 0.0001$) (green)
- Asymmetric shuffler topology
  - Output equivalent to general topology
- Recursive topology
  - Instable
- Filter computation
  - Frequency domain
  - 1024-point FFT

Results

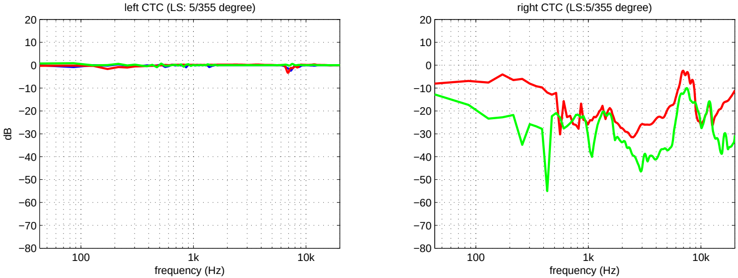
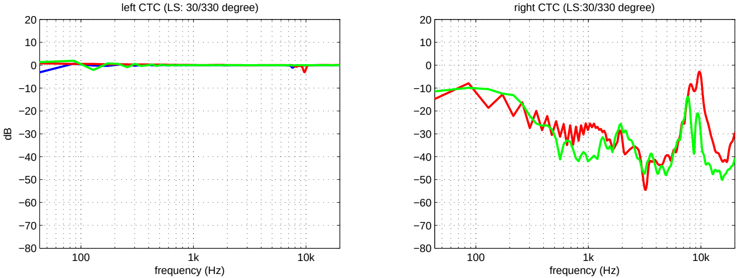


Fig. 13: Magnitude spectra of the acoustic paths between left speaker/left ear (left plot) and left speaker/right ear (right plot). Source span: $\pm5^\circ$ (first), $\pm30^\circ$ (second) and $\pm65^\circ$ (third).

Results



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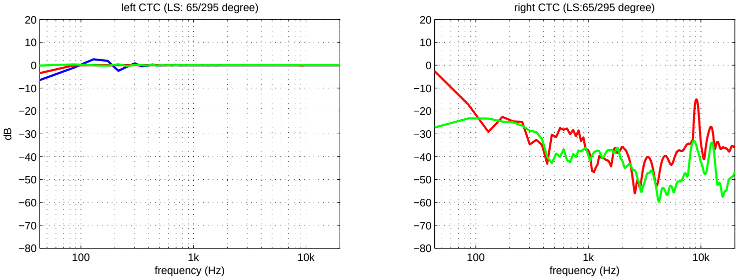

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Outline

Conclusions and prospects (1)

- Design questions
  - Two-channel static transaural system

Position of loudspeakers

- OSD
- Average best-fit
- Multi-channel approach [HBC07]

Implementation methods/topologies

- General topology (direct implementation)
- A couple of alternative methods exist
- Adaptive approach [NHE92]
- Dynamic approach [Len06]

Conclusions and prospect (2)

Matched vs. mismatched CTC filters

- Mismatched filters deteriorate performance [ACB+07]
- Plant identification

Influence of reflections

- Severe early reflections lead to distortion [Sae04]

Main challenges

- Robustness against movements
- Sound coloration

Thank you for your attention!

Questions?

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