Perceptually Optimized Personal Sound Zones

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EE 395 Final Project

Introduction

Personal Sound Zones (PSZs)

- Localized sound field control (dereverberation)
- Loudspeaker arrays generate isolated audio programs in two individual zones via PSZ filtering

Poses a complex physical optimization problem - adaptive filtering to the rescue!

- Define optimal filtering solution
- Facilitate real-time implementation

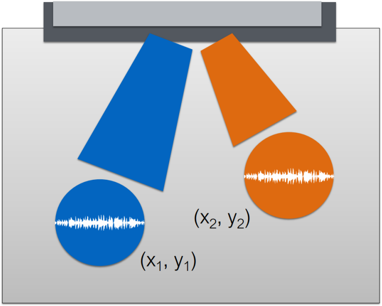
Perceptual optimization

- Account for psychoacoustic phenomena
- Enhance subjective performance measures

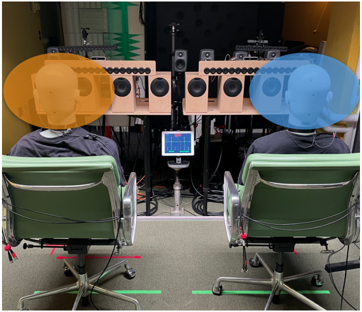
Personal Sound Zones

Applications

- Automotive cabins
- Individualized content by language or preference



(a) Bright and dark zones

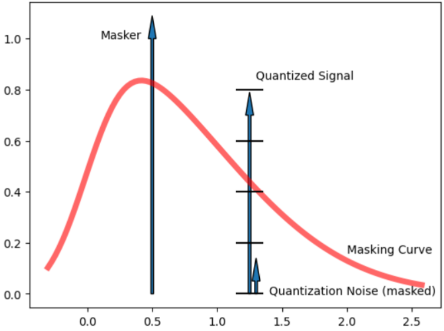


(b) Laboratory PSZ system

The *13 dB Miracle*

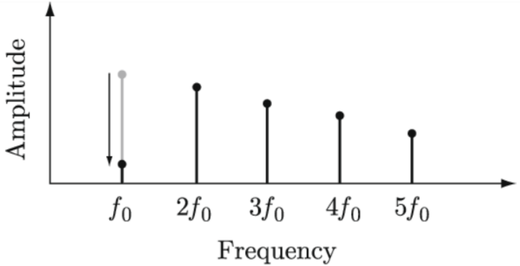
Psychoacoustic masking

- Backbone of audio compression and perceptual audio coding, i.e. MPEG-3
- Louder frequencies mask surrounding information - reproduction and quantization errors within masked regions are perceptually nonexistent
- 13 dB of masked noise nearly imperceptible!



Demo: white vs shaped noise

Missing-Fundamental Phenomenon



Pitch Perception

- Overtones or *harmonics* contribute to perceived pitch
- Removing $f_0 \rightarrow$ **virtual pitch**
- Nearly indistinguishable from pitch containing f_0!

Demo: true vs virtual pitch of a sawtooth wave, $f_0 = \{262, 440\}$ Hz

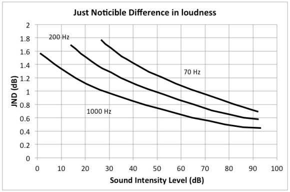
Just-Noticeable Difference (JND)

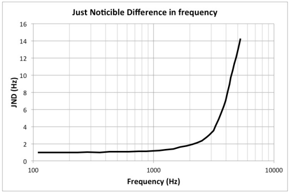
Monaural

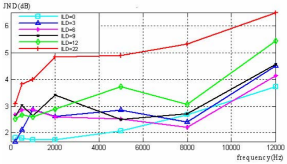
- JND-dB
- JND-Hz

Binaural

- JND-ILD (interaural level difference)
- JND-ITD (interaural time difference)





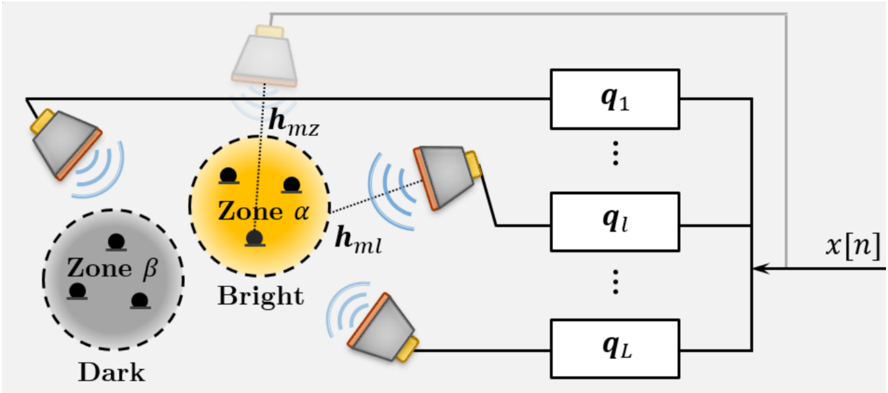


Static PSZ Formulation

Zone control point m, time index n, L loudspeakers, J FIR coefficients

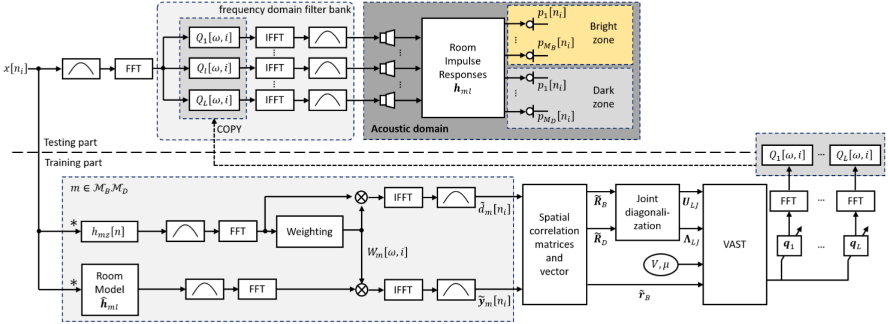
Input signal	$x[n]$
Room impulse responses (RIRs)	$\boldsymbol{h}_{ml}$
Uncontrolled pressure	$\boldsymbol{y}_{ml}[n] = \boldsymbol{X}[n]\boldsymbol{h}_{ml}$
Optimal $L \times J$ FIR PSZ filter matrix	$\boldsymbol{q}$
Optimal pressure	$p_m[n] = \sum_{l=1}^{L} \boldsymbol{y}_{ml}^T[n]\boldsymbol{q}_l = \boldsymbol{y}_m^T[n]\boldsymbol{q}$
Desired signal	$d_m[n] = \begin{cases} (h_{mz} * x)[n], & BZ \\ 0, & DZ \end{cases}$
Error (VAST)	$\varepsilon_m[n] = d_m[n] - p_m[n]$
Error (AP-VAST)	$(\varepsilon_m * w_m)[n] = \tilde{\varepsilon}_m[n] = \tilde{d}_m[n] - \tilde{p}_m[n]$

where $w_m[n]$ is the time-varying inverse of the psychoacoustic masking filter at point m and time n!


$\boldsymbol{q}_1$
$\boldsymbol{q}_l$
$\boldsymbol{q}_L$
$\boldsymbol{h}_{mz}$
$\boldsymbol{h}_{ml}$
$x[n]$
Zone α
Bright
Zone β
Dark

AP-VAST

Adaptive and perceptually optimized variable-span tradeoff (AP-VAST) filtering

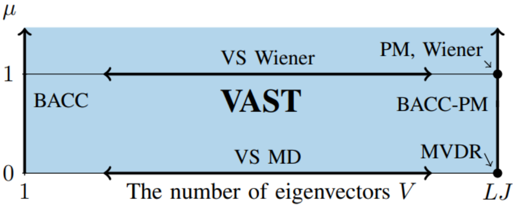


Optimal Solution

- Sparing you the details...

$$\boldsymbol{q}_o(V,\mu) = \boldsymbol{U}_V \boldsymbol{a}_o(V,\mu) = \sum_{v=1}^{V} \frac{\boldsymbol{u}_v^T \tilde{\boldsymbol{r}}_B}{\lambda_v + \mu} \boldsymbol{u}_v, \quad 1 \leq V \leq LJ$$

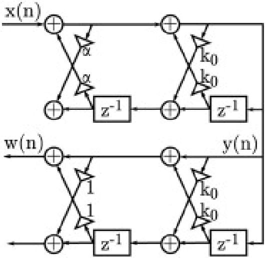
- Manipulating number of eigenvalues/vectors V and Lagrange multiplier μ provides many other existing adaptive solutions
- AP-VAST provides a 20% increase in subjective performance compared to the standard VAST approach!
- **Demo:** Pressure-matching (PM) vs AP-VAST



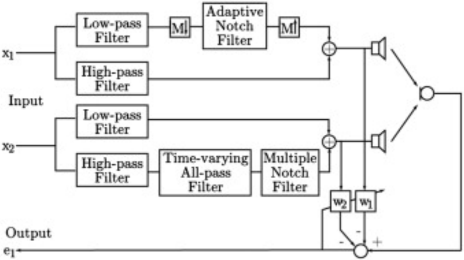
Adaptive Multichannel Decorrelation

Comb filtering effect due to linear loudspeaker array geometry causes audible distortion

- Solution: decorrelate adjacent multichannel signals by adaptive notch filtering and time-varying all-pass filtering



(a) Adaptive 2nd-order IIR notch filter



(b) Decorrelation algorithm block diagram

1) 2nd-order f_0-tracking lattice notch filter:

$$H(z) = \frac{W(z)}{X(z)} = \frac{1 + 2k_0[n]z^{-1} + z^{-2}}{1 + k_o[n](1+\alpha)z^{-1} + \alpha z^{-2}}, \quad W(z) = \sum_{k=-\infty}^{\infty} w[k]z^{-k} \tag{1}$$

$$k_0[n] = \frac{2}{1 + e^{-\mathbf{g_0}[n]}} \tag{2}$$

$$\nabla_{\mathbf{g_0}} \left(\sum_{k=0}^{n} \lambda^{n-k} w^2[k] \right) = 0, \quad 0 < \lambda < 1 \tag{3}$$

$$\hat{f}_0[n] = \frac{f_s}{2\pi M} \cos^{-1}(-k_0[n]) \tag{4}$$

2) Time-varying 2nd-order all-pass filter:

$$A(z) = \frac{k_0^2[n] - 2k_0[n]z^{-1} - z^{-2}}{1 - 2k_0[n]z^{-1} + k_0^2[n]z^{-2}}$$

- Perturbs group delay across frequency range within JND-ITD threshold ($< 40\ \mu$s)
- Preserves magnitude response (all-pass)
- Imperceptible phase distortion decorrelates adjacent channels!

3) Adaptive polynomial multiple-notch filter:

$$M(z) = \frac{\prod_{m=M_{min}}^{M_{max}} (1 - e^{j\omega_m(n)} z^{-1})}{\prod_{m=M_{min}}^{M_{max}} (1 - e^{j\omega_m(n)} \rho z^{-1})}$$

$$\omega_m(n) = 2\pi m \hat{f}_0[n], \quad 1 \leq M_{min} < m < M_{max} \leq \left\lfloor \frac{f_s}{\hat{f}_0[n]} \right\rfloor$$

- Create the lowest frequency notch within the high frequency range due to perceptual insensitivity of phase and magnitude distortion at increasingly higher frequencies

Decorrelation Performance (MSC)

Between two channels x_i and x_j, define the Magnitude Square Coherence (MSC):

$$\mathrm{MSC}_{i,j}(f) = \frac{|S_{ij}(f)|^2}{S_{ii}(f)S_{jj}(f)}$$

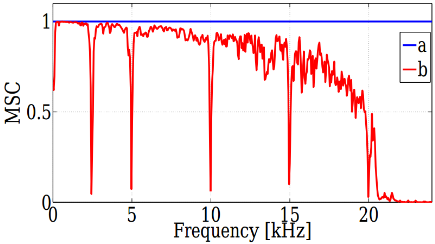


Figure: Magnitude square coherence before (a) and after (b) adaptive decorrelation method for $f_0 = 2.5$ kHz and harmonics 5 kHz, 10 kHz, 15 kHz, and 20 kHz

Conclusions

- Background, motivation, and formulation of perceptually motivated adaptive filtering algorithms for PSZs
- Possible to improve on the subjective performance of existing filtering techniques without only considering optimization of physical parameters
- Able to efficiently improve performance through imperceptible filtering operations
- Techniques can be integrated within deep neural network (DNN) methods for PSZ filter generation
- Extends to scenarios involving head-tracking to further improve listening performance of moving zones and head movements

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