An Integrated Approach to Hearing Aid Algorithm Design for Enhancement of Audibility, Intelligibility and Comfort

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Introduction

Hearing Aid Algorithm Goals

- Audibility: maximize normal loudness restoration
- Intelligibility: maximize signal-to-noise ratio
- Comfort: minimize perceptual distortion

Signal Processing Goal

Integrated approach to gain control algorithm

Processing model

Analyze received signal into time-frequency grid

$$X(t,f) = S(t,f) + N(t,f)$$

Process signal by adaptive (real) gain

$$Y(t, f) = G(t, f)X(t, f)$$

Prior Art

 Bottom-up approach: ad hoc network of gain modules (for, e.g., noise reduction, compression, de-reverberation, feedback suppression etc.)

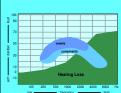
Problem

- Signal distortion vs. residual noise trade-off is uncontrolled
- Hearing aids patients are not satisfied (20% of patients do not wear their hearing aids)

'Normal Loudness' Restoration

Problem

 Increased hearing threshold leads to dynamic range reduction and loudness distortion





Restore relative loudness to normal

$$C(\mathbf{G}_{\!\scriptscriptstyle fr}) = \! \left[\frac{(\mathbf{G}_{\!\scriptscriptstyle fr} + \mathbf{P}_{\!\scriptscriptstyle x}) - \mathbf{P}_{\!\scriptscriptstyle t}^{i}}{\mathbf{P}_{\!\scriptscriptstyle u}^{i} - \mathbf{P}_{\!\scriptscriptstyle t}^{i}} - \frac{\mathbf{P}_{\!\scriptscriptstyle x} - \mathbf{P}_{\!\scriptscriptstyle t}^{n}}{\mathbf{P}_{\!\scriptscriptstyle u}^{n} - \mathbf{P}_{\!\scriptscriptstyle t}^{n}} \right]^{\!2} \qquad \text{(where } \mathbf{G}_{\!\scriptscriptstyle t}\mathbf{P}_{\!\scriptscriptstyle t}$$

leads to
$$G_{lr} = \alpha \left(\frac{P_t^n}{P_x}\right)^{\frac{\beta}{2}}$$
 where $\alpha \equiv \sqrt{\frac{P_t^i}{P_t^i}}$, $\beta \equiv \frac{\frac{P_u^i}{P_t^i}}{P_t^i} \log \frac{P_u^n P_t^i}{P_u^i P_t^n}$

• G_{lr} implements compressive amplification

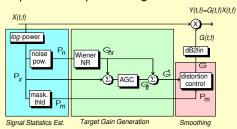
Improving Signal-to-Noise Ratio

- Estimate 1 (global) *SNR*(*t,t*), where *S* refers to target signal (usually speech) and *N* integrates (power of) everything else, including all sources of noise, feedback and reverberation.
- · Compute a (motivated) gain, e.g., Wiener gain

$$G_{nr} = \frac{P_s}{P_s + \hat{P}_n} = 1 - \frac{\hat{P}_n}{P_x}$$
 if s, n uncorrelated

Integration and Comfort

• **Cost integration**: loudness restoration of "cleaned-up" signal (target gain G^*), subject to "principle of least processing"



target gain
$$G^* = \alpha \left(\frac{P_t^n}{G_{nr}^2 P_x}\right)^{\beta/2} \times G_{nr} = G_{lr}G_{nr}^{1-\beta} > G_{lr}G_{nr}$$

Principle of least processing (Wolfe, ICASSP 2003)

$$C(G) = \left[GX - \left(1 - \frac{P_m}{P_x} \right) S - \frac{P_m}{P_x} X \right]^2 \Rightarrow G = \left(1 - \frac{P_m}{P_x} \right) G^* + \frac{P_m}{P_x} \cdot 1$$

Total Integrated Gain

$$G = \frac{P_m}{P_x} + \alpha \left(1 - \frac{P_m}{P_x}\right) \left(1 - \frac{P_n}{P_x}\right)^{1-\beta} \left(\frac{P_t^n}{P_x}\right)^{\beta/2}$$

- Needs relative power estimates \hat{P}_m/P_x , \hat{P}_n/P_x , (P_t^n/P_x) and patient signature (α, β)
- to do: formal listening tests



