A Novel Structure to Compensate for Frequency-Dependent Loudness Recruitment of Sensorineural Hearing Loss

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The Problem

I need your help.

What?

I need your help.

What?

I need your help.

My pleasure, but you don't have to shout.

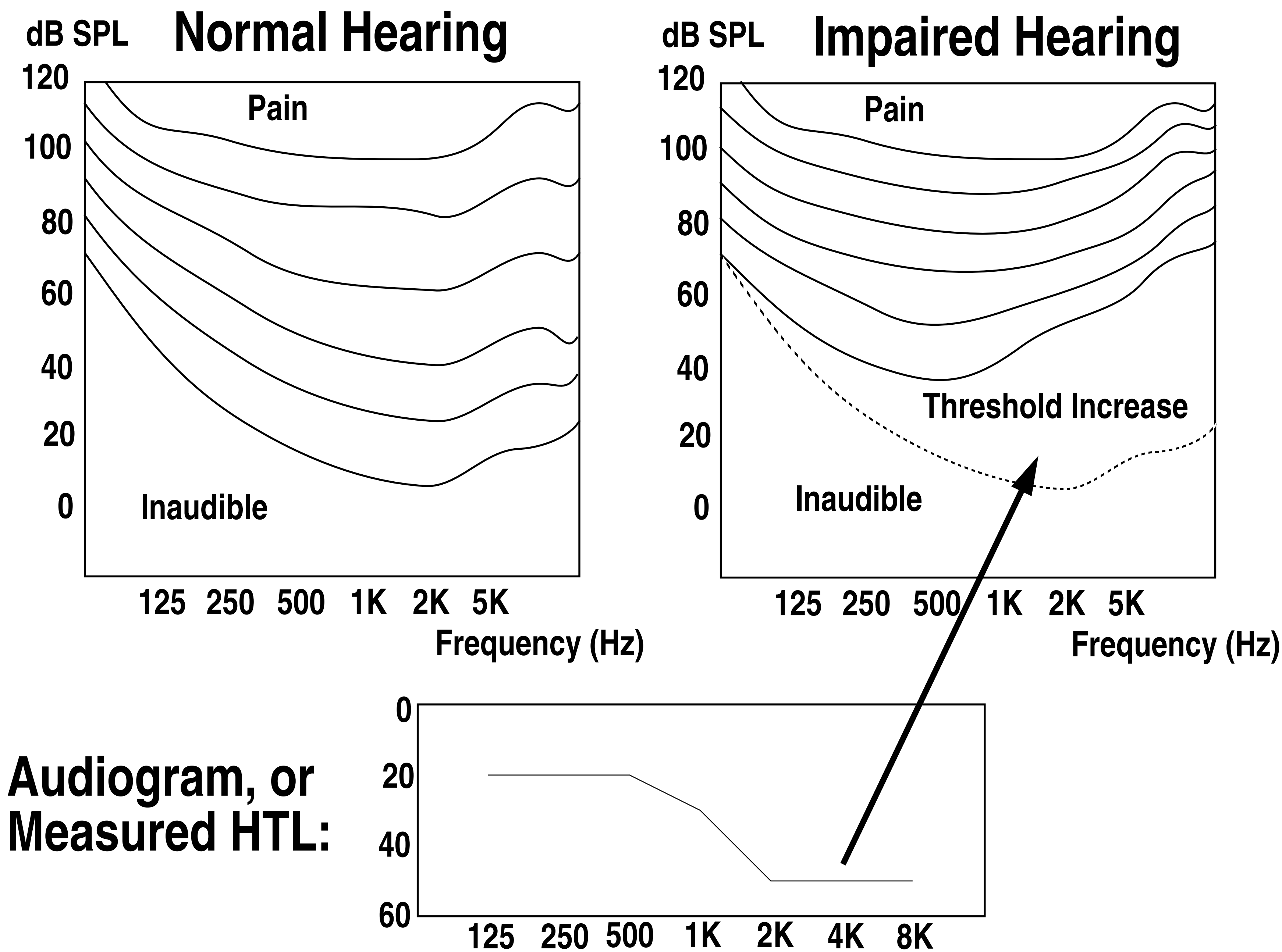
- **Quiet sounds are inaudible, but loud signals are still uncomfortably loud.**
- **The amount of hearing loss is frequency-dependent.**

A Direct Solution

- **Understand Loudness Recruitment in terms of the <u>differences between constant loudness contours</u> of healthy and impaired hearing.**

- **Define a structure which provides a <u>continuum of transfer functions</u> to compensate for these differences.**

Comparing Constant Loudness Contours



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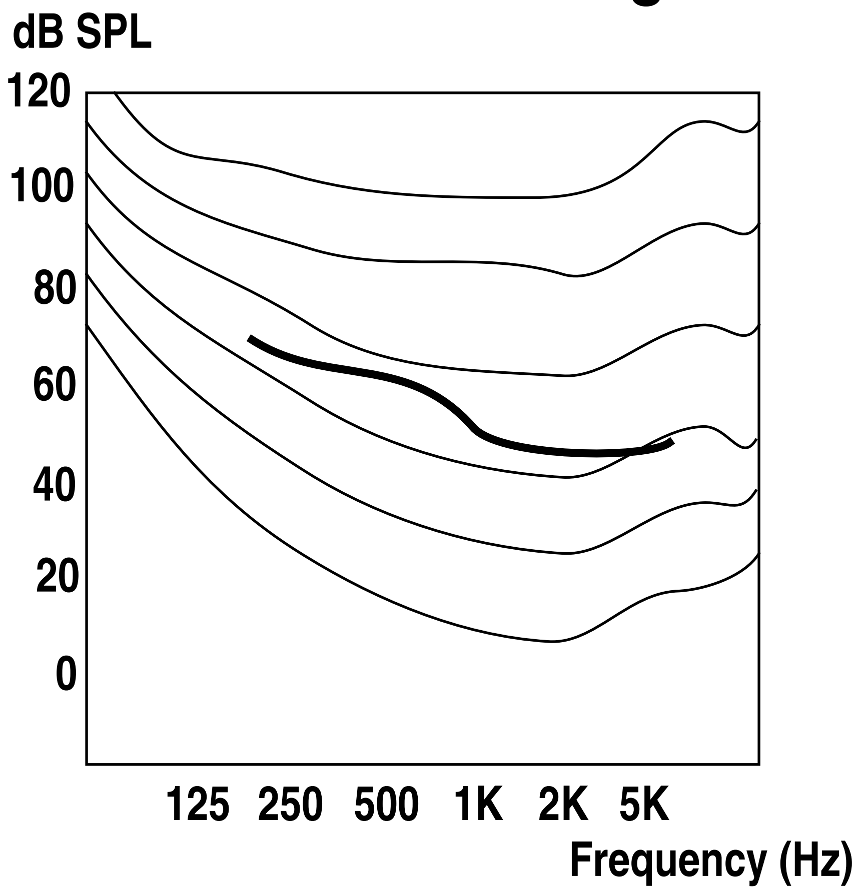
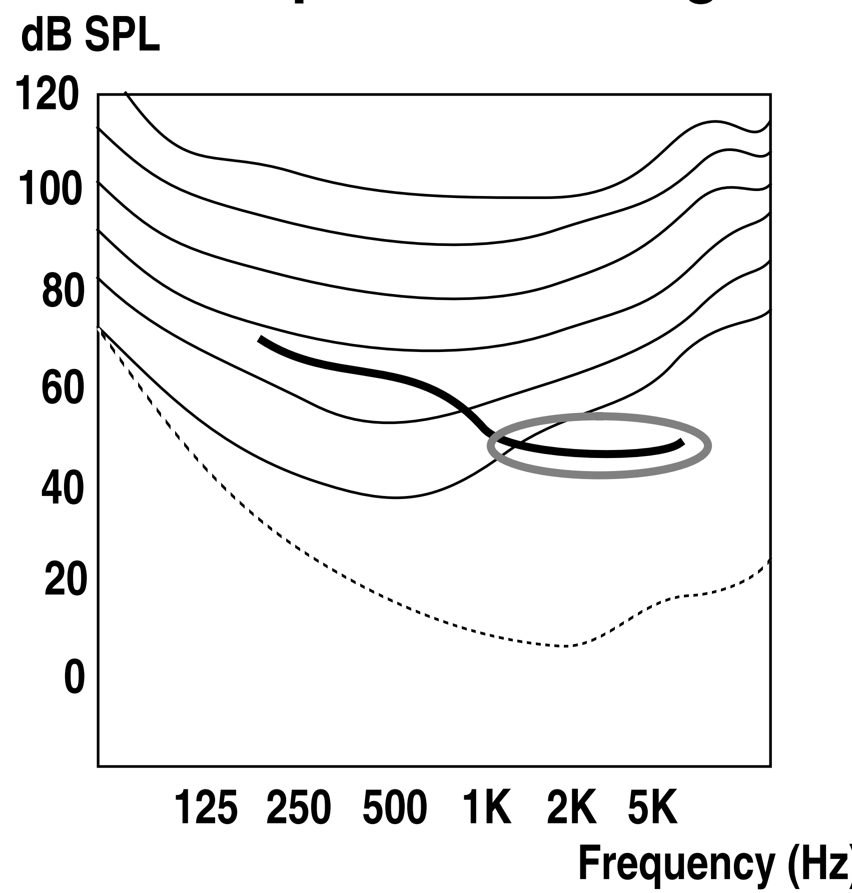
Lost Speech Information

Normal Hearing



Impaired Hearing



Average Speech Spectrum (Byrne, and Dillon 1986)

Lost Speech Information

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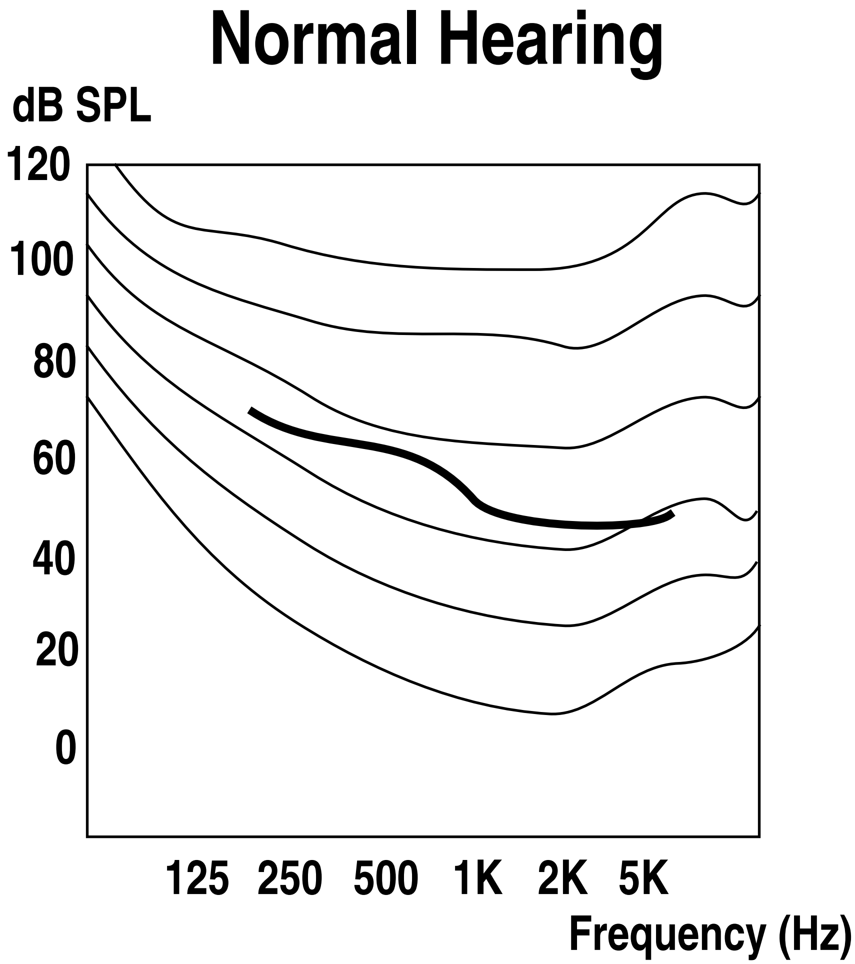
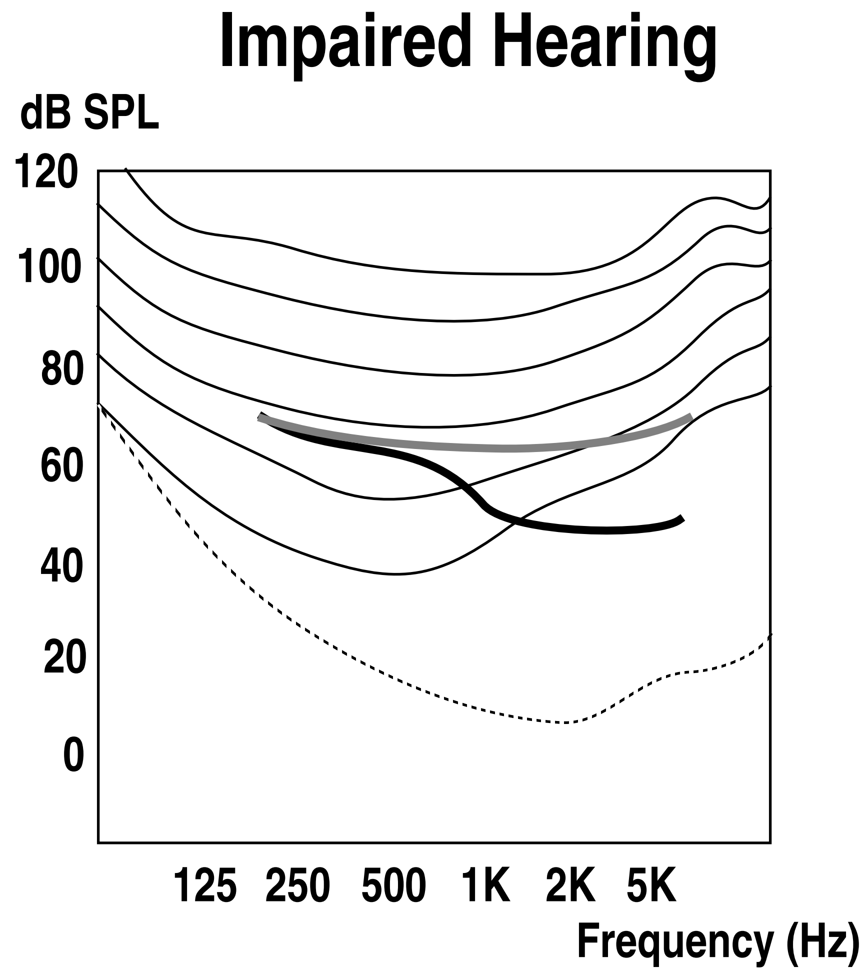
Average Speech Spectrum After Correction





Average Speech Spectrum

Approximate Spectrum after NAL Correction (Byrne 1986)

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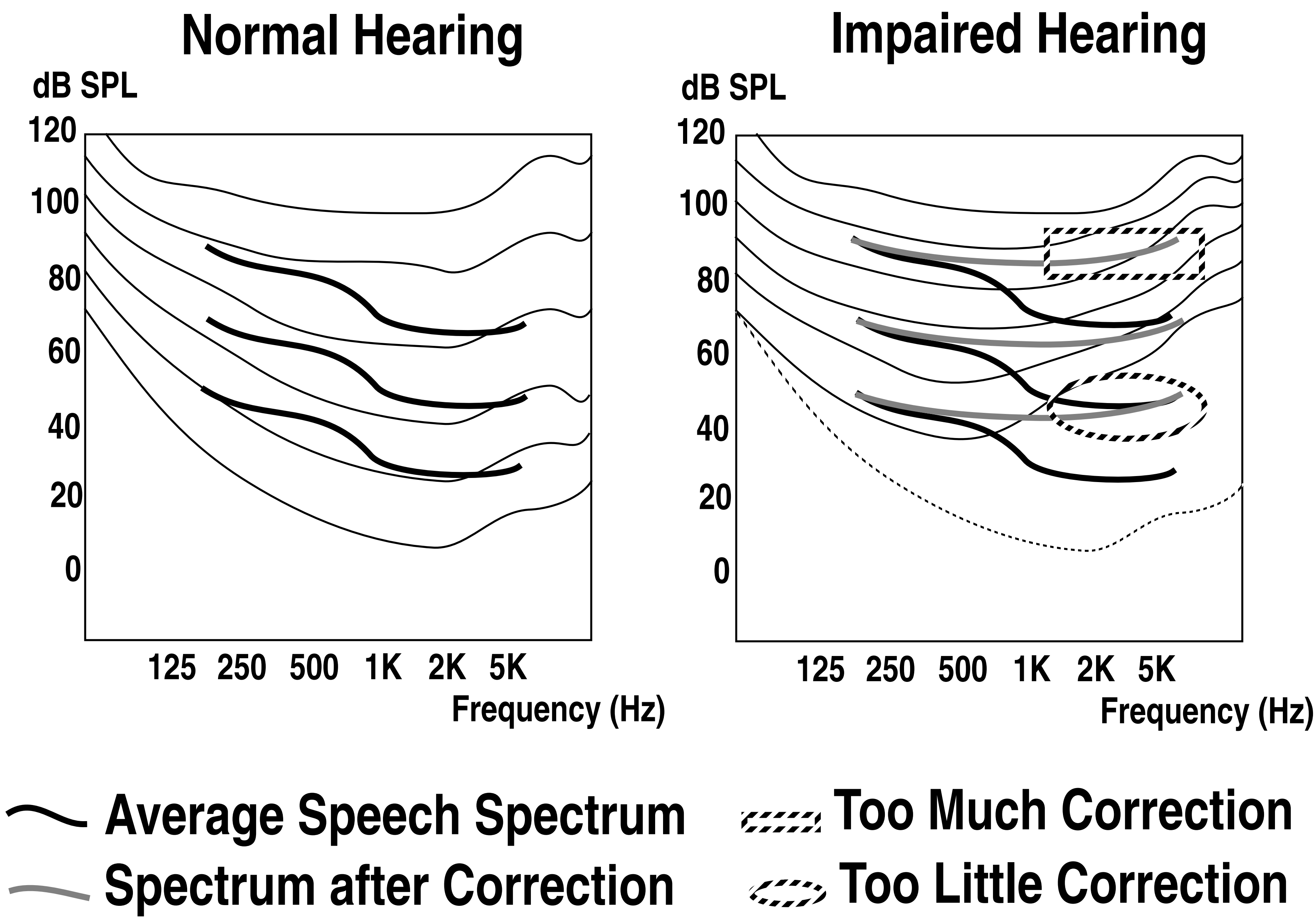
Average Speech at Different Levels



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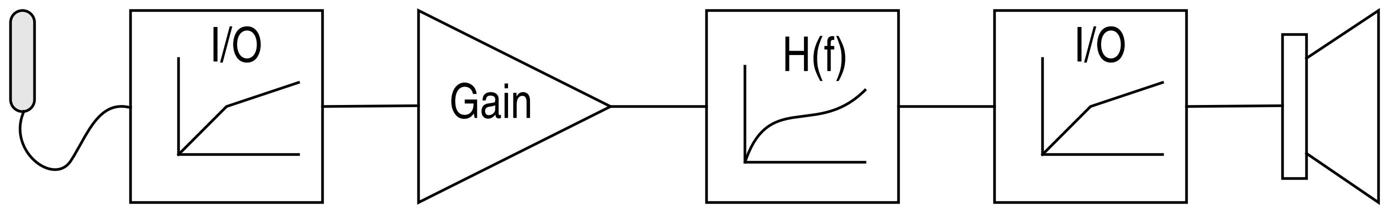
System Goals

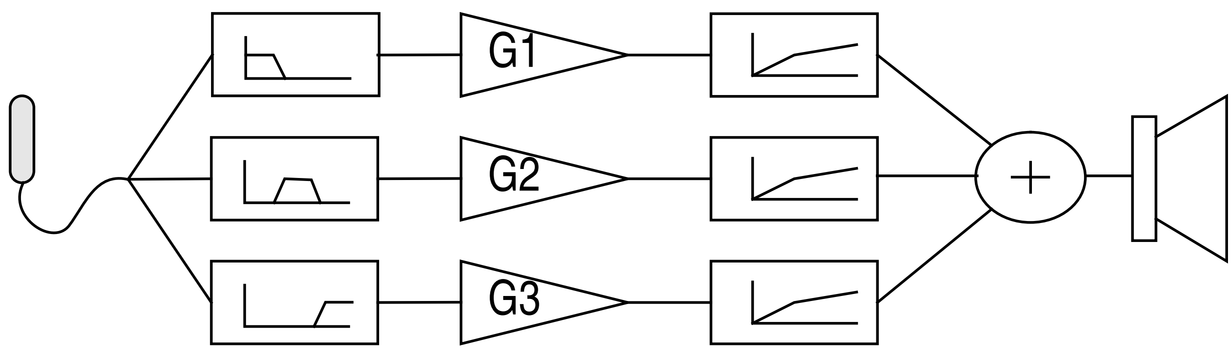
- Precise prescription of smoothly varying frequency responses as a function of signal input intensity.
- Extremely low complexity suitable for mass implementation and wide-ranging prescriptions.
- Strong performance in naturally (dynamically) noisy environments.

Typical Approaches

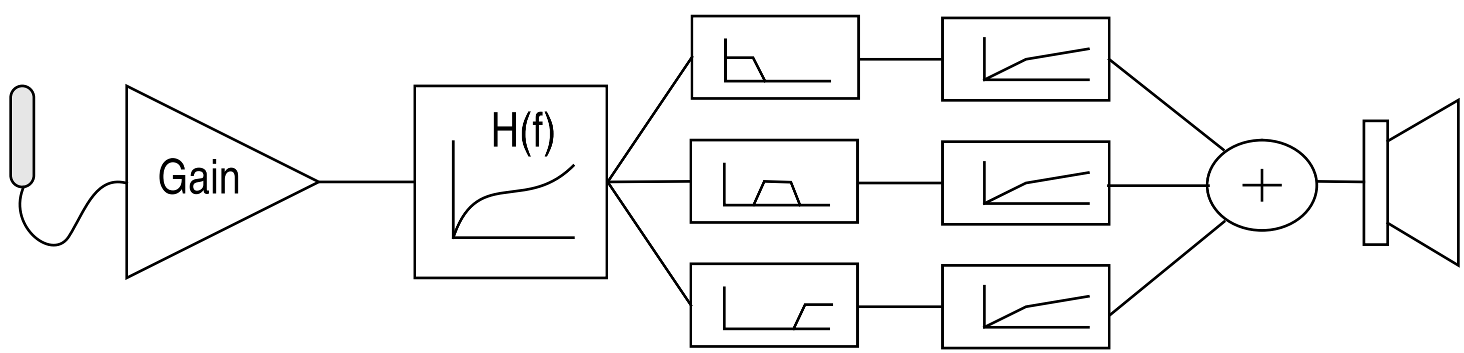
Linear Filter, Compression



Multi-Band Compression



Single Filter, with MBC



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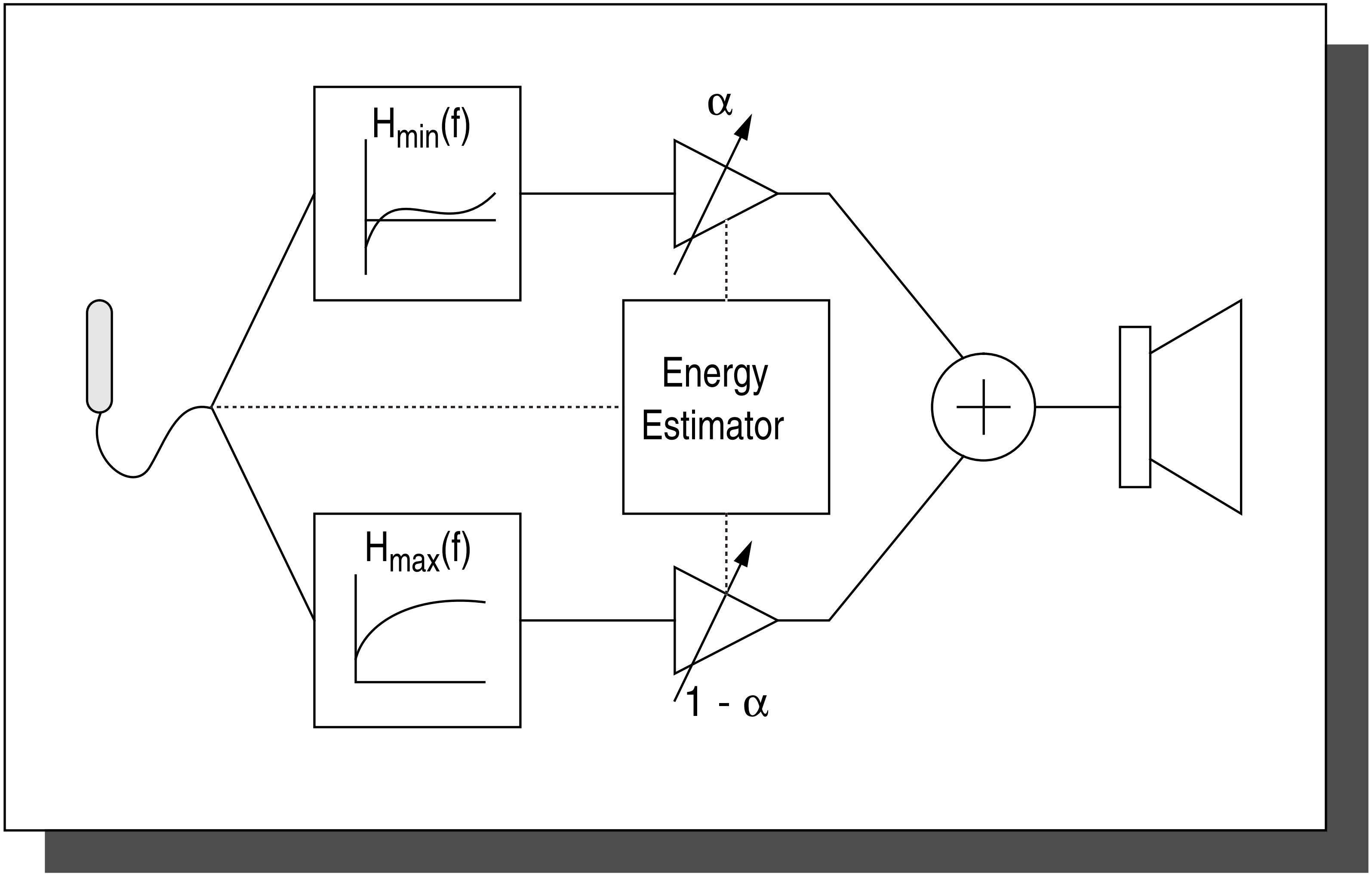
Proposed Dynamic Mixing Structure



$$y[n] = \alpha(x[n]*h_{min}[n]) + (1 - \alpha)(x[n]*h_{max}[n])$$

Parameters: $H_{min}(f)$, $H_{max}(f)$, Mixing Control Algorithm

System Parameters

- **$H_{max}(f)$ - The maximum correction, suitable for the quietest inputs.**
- **$H_{min}(f)$ - The minimum correction, suitable for the loudest inputs.**
- **Mix Control Start Point - Above this level the output of $H_{max}(f)$ is gradually attenuated, and that from $H_{min}(f)$ is gradually increased.**
- **Mix Control Slope - How much the mixture changes from $H_{max}(f)$ to $H_{min}(f)$ with increased input level.**
- **Dynamic Parameters - Window length for energy estimation, and how quickly the system changes.**

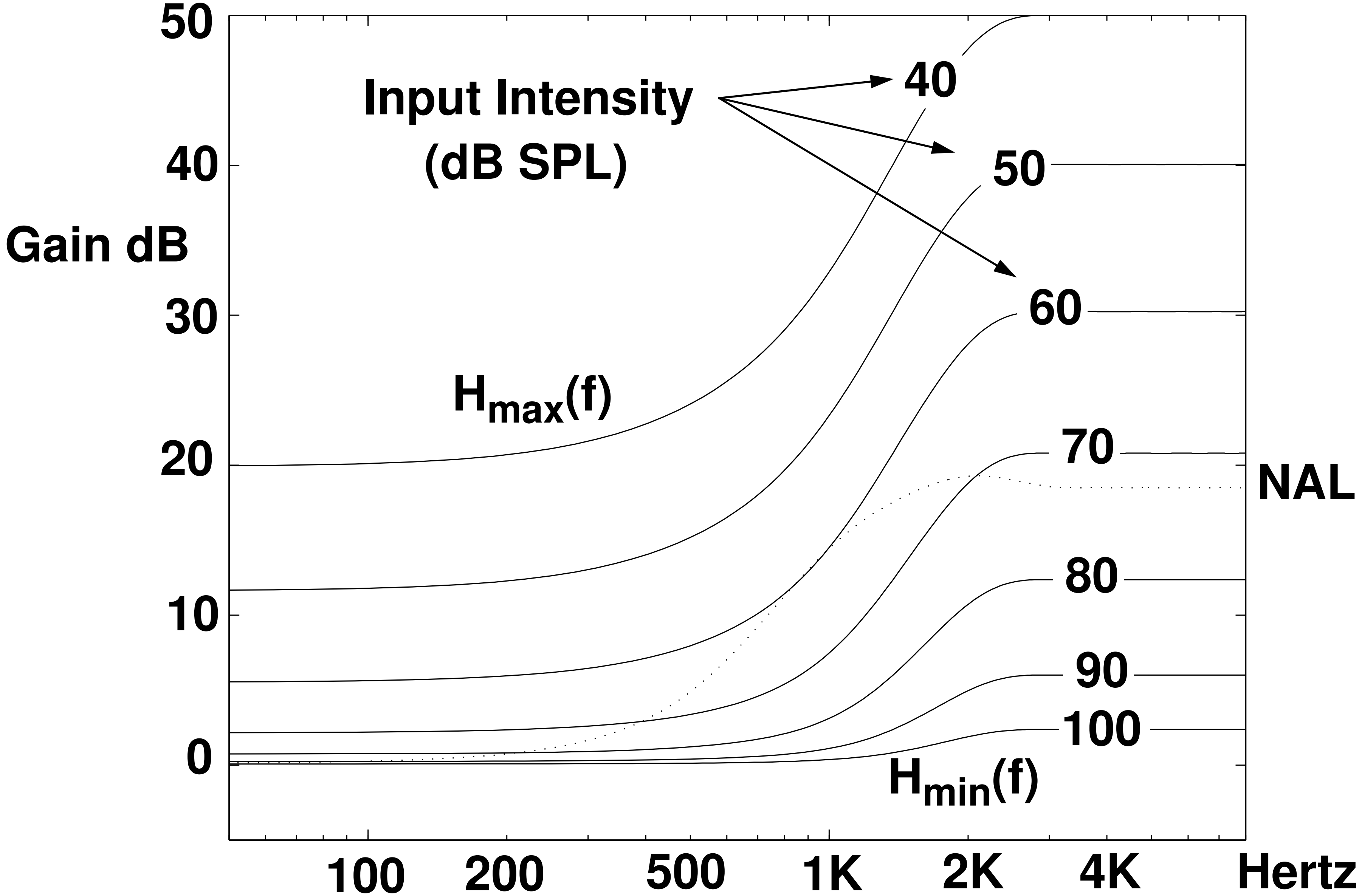
Have we Met our Goals?

- Allows for direct prescription a continuum of transfer functions as a function of input intensity.
- A more precise mapping of constant loudness contours and dynamic adaptation may provide more useful spectral information in natural noise.
- The structure is a low-complexity digital realization using 2 FIR filters with potentially very few (64-32) taps.
- Low complexity and relatively few system parameters simplify prescription process.

Parameter Selection: Some Possibilities

- **Flat System** - Maps normal loudness contours to those of the impaired listener: $H_{max}(f)$ is the threshold increase, $H_{min}(f)$ is flat, and mix control parameters are chosen to approximate the NAL standard at 70 dB input. (Mix Start 40 dB, Slope 1.0)
- **Speech Corrected System** - Starts with $H_{max}(f)$ at 0.6 (instead of 0.31) times the threshold increase, and intersects the NAL standard at 70 dB input. (Mix Start 50 dB, Slope 0.75)
- **Hybrid Evaluated** - Starts with the Flat System and adjusts the $H_{max}(f)$ and $H_{min}(f)$ so that the 70 dB input more exactly matches the NAL standard. (Mix Start 40 dB, Slope 1.0)

"Flat" Dynamic Frequency Response



Different input intensities cause different transfer functions.

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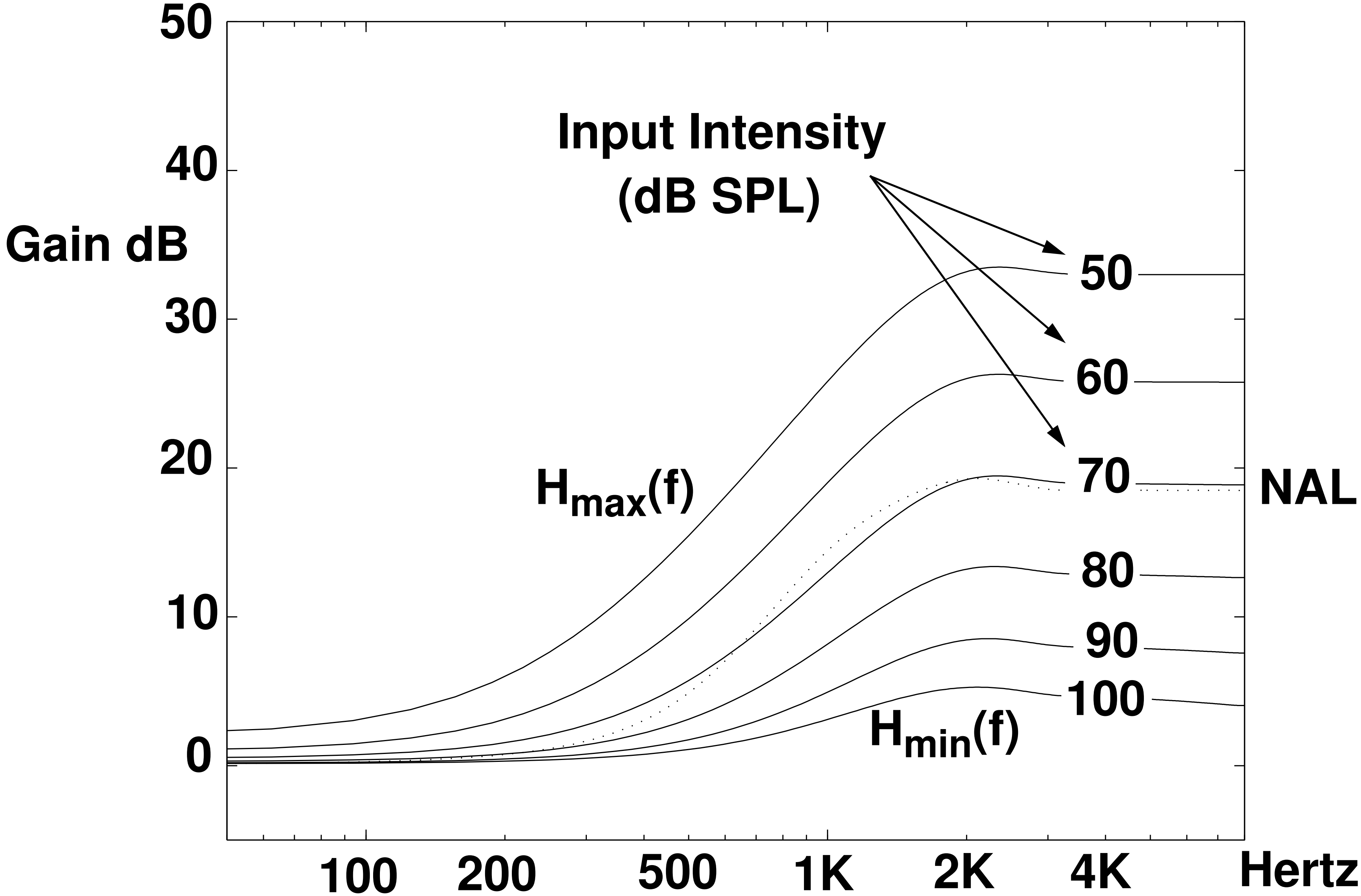
“Speech Corrected” Dynamic Response



This response is a closer match to the NAL standard at 70dB.

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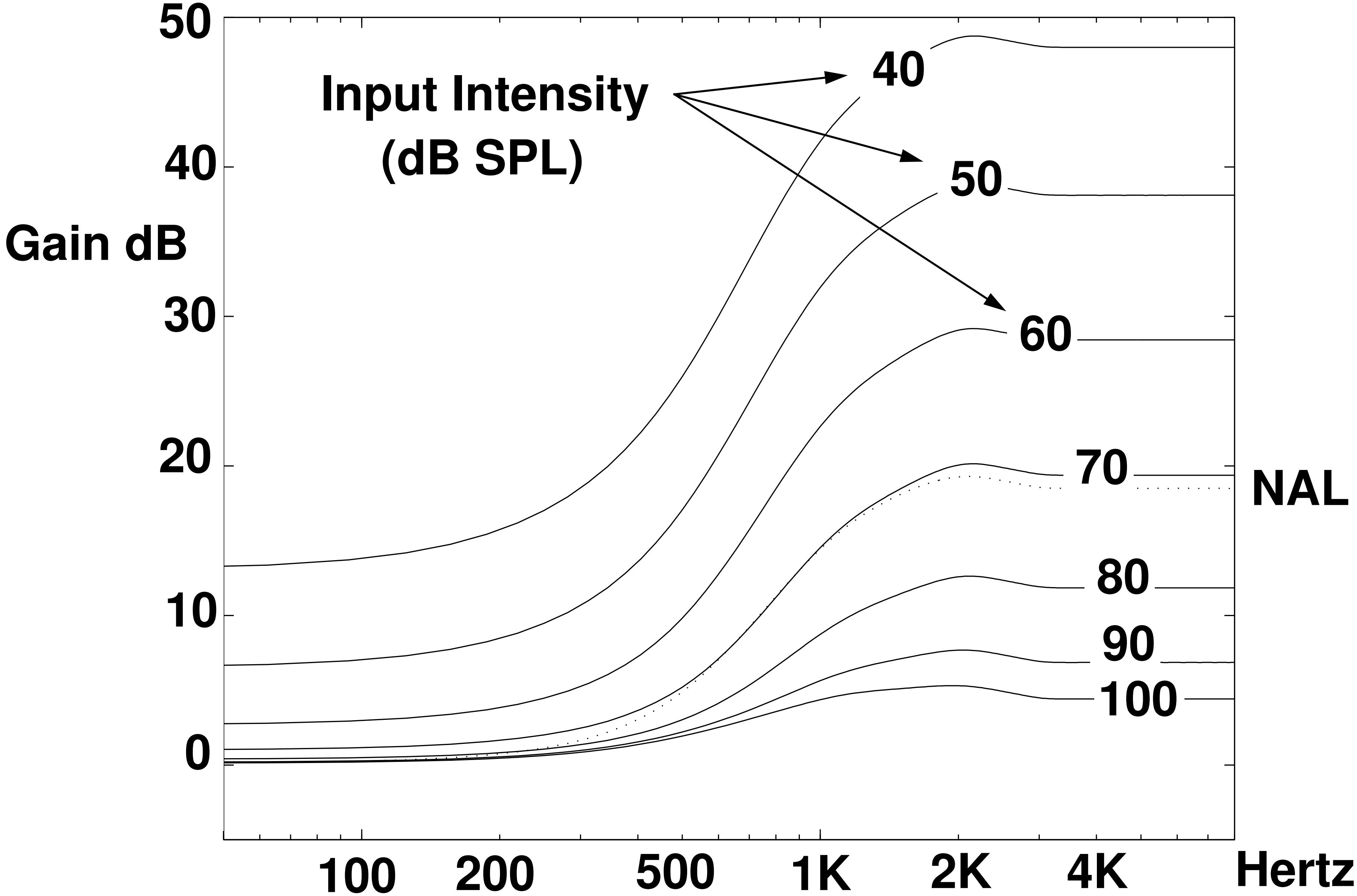
Dynamic Response Evaluated



Start with the threshold, intersect the NAL standard at 70 dB.

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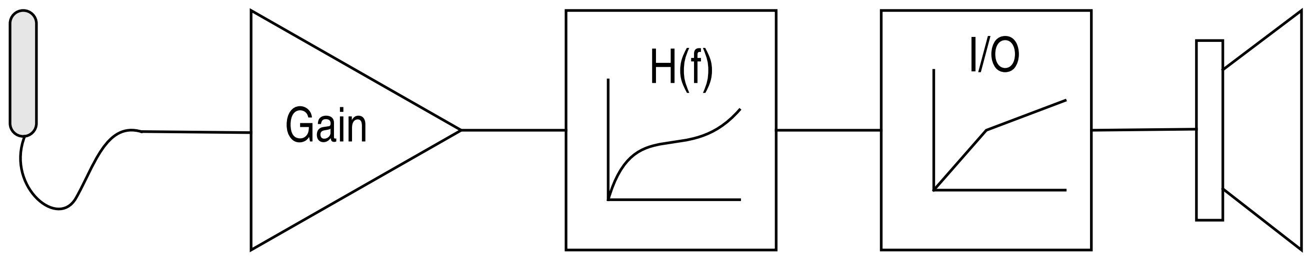
System Evaluation

- **Mask normal listeners with shaped noise:**
  - **Noise shaping is adjusted until tested threshold increase is within 5 dB of simulated moderate loss.**
  - **A training period is included.**
  - **This is a simplified approximation of a hearing loss.**
- **"HINT" sentences (Nilsson, et al. 1994) recorded in naturally noisy environments:**
  - **Recordings were made driving in a car, and at the beach on a fairly windy day. Recordings are stereo.**
  - **Natural noise creates a dynamic (more realistic) segmental SNR. (~50-75% intelligible to normals)**
  - **Different levels fully exercise algorithms' dynamic characteristics.**

Baseline System for Comparison



- **Linear filter system with very slight output compression:**
  - **Filter specification from NAL standard (0.31HTL + off-sets)**
  - **Compression parameters: threshold 85 dB SPL, ratio 3:1, attack 2ms, release 40 ms.**

Results

	Linear	Mixture	%Improvement
Subject 1	59%	72%	22%
Subject 2	47%	57%	21%
Subject 3	67%	76%	13%

Percentage of Words Correct
46 sentences: 255 words

(Unaided, healthy hearing at roughly 80% correct)

Conclusion: A Simple Algorithm for a Better Match

- Using a relatively simple structure, this algorithm allows direct control of a continuum of transfer functions that vary with input intensity.

- Direct control of the continuum of transfer functions at least marginally improves recognition performance for masked listeners with naturally (dynamically) noisy input over a wide dynamic range.

Future Work

- **Improve specification of system parameters:**
  - **Formalize $H_{min}(f)$, $H_{max}(f)$, Mix Start, and Mix Slope prescription from HTLs, based on impaired constant loudness contours and experimental results.**
- **Fine tune dynamic parameters:**
  - **How quickly should the algorithm adjust in each direction? How should these dynamic parameters vary with the hearing loss?**
- **Further Evaluation:**
  - **Evaluate with hearing-impaired subjects and compare to other, adaptive structures.**
- **Investigate hybridization with Multi-Channel AGC for severe losses.**