DIGITAL HEARING AIDS

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Contents

• Hearing and hearing loss
• Hearing aid types and processing constraints
• Dynamic-range compression
• Noise suppression
• Feedback cancellation
• Microphones and arrays
• List of additional areas
• Conclusions
Hearing and Hearing Loss
<table>
<thead>
<tr>
<th>Class</th>
<th>Loss, dB</th>
<th>Handicap</th>
<th>Handicap</th>
<th>Handicap</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal</td>
<td>-10 to 26</td>
<td>Difficulty hearing faint or distant speech</td>
<td>Understands speech at a distance of 3-5 feet</td>
<td>Conversation must be loud, difficulty in group or classroom</td>
</tr>
<tr>
<td>Mild</td>
<td>27-40</td>
<td></td>
<td></td>
<td>May hear a loud voice at 1 foot, may distinguish vowels but not consonants</td>
</tr>
<tr>
<td>Moderate</td>
<td>40-55</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Moderately-Severe</td>
<td>55-70</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Severe</td>
<td>70-90</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Profound</td>
<td>&gt;90</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
PLATE III  TRANSMISSION OF VIBRATIONS FROM DRUM THROUGH THE COCHLEA
Inner / Outer Hair Cells

- IHC is the transducer
- OHC enhanced motion:
  - to extent sensitivity for weak sounds
  - sharper filters
- OHC most vulnerable
  - aging
  - noise
Recruitment

• Reduced dynamic range in impaired ear
  - Healthy cochlea has instantaneous compression
  - Loss of active OHC gain mechanism in impairment
  - Reduced cochlear gain, increased filter bandwidth
  - Impaired ear is more linear than healthy ear
  - Recruitment = abnormal rapid increase in loudness
  - Loudness at 100 dB SPL approx equal

• Compression
  - Compensate for OHC damage
  - High face validity
  - Benefits in practice are mixed
Recruitment Example

Sound Pressure Level (dB)

Loudness in Assigned Numbers

Normal
Perceptual Auditory Filters
Simulated Firing Rate /da/, Normal Ear
Simulated Firing Rate /da/, Impaired Ear
Hearing Loss Conclusions

• Outer hair-cell damage
  - Shift in auditory threshold
  - Recruitment: Impaired system is more linear
  - Broader auditory filters

• Inner hair-cell damage
  - Shift in auditory threshold
  - “Dead regions” with no response

• Can not perceive low-intensity speech sounds

• Difficulty in noise and reverberation
Hearing Aid Types and Processing Constraints
Styles of Hearing Aids

- PL10
- PL10-B
- PL30-D
- PL40-D ITE
- PL50-D ITE
- PL70-DV BTE
- PL70-D BTE
- PL10 QC
- PL30-D ITC
- PL70-D BTE
ITE Hearing Aid

- Microphone
- Tube
- Battery Compartment
- Receiver
- Circuit
- Volume Control
- Ear Canal
- Vent
Digital Hearing Aid Sales, USA
Factors Affecting Power Use

• Number of Transistors
  – Transistor size decreasing over time
  – Circuit complexity increasing

• Word Size
  – Fixed point vs. floating point
  – Long digital word gives better sound quality
  – Adder and multiplier increase in complexity
  – Memory proportional to word size

• Clock Rate
  – Slow rate reduces battery drain
  – Constrains algorithm complexity

• Voltage
• Algorithms
Digital Processor Comparison

• Personal Computer
  – 6400 MIPS (Xbox 360)
  – 1 to 2 GB memory
  – Draws 20 W
  – Fits on circuit board

• Hearing Aid
  – 4 to 8 MIPS
  – 4 to 8 kW memory
  – Draws 1 mW
  – Fits inside hearing aid
Hardware Conclusions

• Hearing Aids Have Become Digital Devices
• Digital Advantages
  – Programmable processor
  – Algorithm complexity
  – Processing features
  – Fitting flexibility
• Digital Limitations
  – Power consumption and battery life
  – Circuit size
  – Cost of circuit and specialized DSP chip development
Dynamic-Range Compression
Compression Rules

• Processing steps
  - Detect signal level
  - Convert to dB SPL
  - Compute gain using input/output relationship
  - Multiply signal by gain

• Piece-wise linear function
  - Kneepoints where slope changes
  - Linear at low levels: Minimize amplification of noise
  - Compression limiting at high levels: Avoid UCL

• Gain = output level - input level in dB
Compressor I/O Function

INPUT, dB SPL

OUTPUT dB SPL

LINEAR

COMPRESSOR CR:1

LOWER KNEE

COMPRESSION LIMITING

UPPER KNEE
Envelope Detection

- Track incoming signal level
- Response depends on sign of signal changes
  - Rapid response to increases in signal level (attack)
  - Slower response to decreases (release)
  - Defined by attack and release time constants
- Fast attack
  - Signals tend to increase more rapidly than decrease
  - Prevent over-amplification of large sudden increase
- Slow release
  - Rapid gain changes cause audible modulation
  - Hold gain relatively constant during syllables
Peak Detection

• Follow increase with fast time constant
• Follow decrease with slow time constant

\[
\begin{align*}
\text{if } |x(n)| &\geq d(n - 1) \\
n_d(n) &= \alpha d(n - 1) + (1 - \alpha)|x(n)| \\
\text{else} \\
n_d(n) &= \beta d(n - 1) \\
\text{end}
\end{align*}
\]
“Air” Male Talker
Peak Detector Output
Multichannel Compression

- Frequency analysis
  - Filter bank
  - FFT

- Filter bank
  - Auditory frequency spacing
  - Independent compression in each frequency band
  - Gain set in response to preceding signal level
  - Response to signal change depends on time constants
Multichannel Compressor

$X(f) \rightarrow Y(f)$

$V_1(f) \rightarrow$ COMP. GAIN
$V_2(f) \rightarrow$ COMP. GAIN
$V_k(f) \rightarrow$ COMP. GAIN

LOW-PASS FILTER
BAND-PASS FILTER
HIGH-PASS FILTER
Frequency Warping

- FFT problems
  - Uniform frequency spacing
  - Resolution at low frequencies is poor
  - Need long delay to get good low-frequency analysis

- Goals of frequency warping
  - Auditory frequency analysis
  - Reduced group delay
  - Reasonable computational requirements

- Want group delay < 10 msec
- Used in GN ReSound products
Warped Filter Structure

- Replace unit delays with all-pass filters
  - FIR filter sums outputs at different delays
  - Cascade of all-pass filters for cascade of unit delays
  - Warped FIR sums outputs of all-pass filters

- Effects of the all-pass filters
  - Low frequencies delayed from filter to filter
  - Separation between filtered samples at low frequencies > unit sample
  - Separation at high frequencies < unit sample
Group Delay of AP Filter

Normalized Frequency (Nyquist=1)

Group Delay, samples
Warped FIR Filter

\[ x(n) \rightarrow [1 - z^{-1}] \rightarrow [1 - z^{-1}] \rightarrow [1 - z^{-1}] \rightarrow y(n) \]

\[ b_0 \rightarrow b_1 \rightarrow b_2 \rightarrow b_3 \]

\[ A(z) \rightarrow \]
Group Delay Comparison

![Graph showing group delay comparison.](image)
Warped Compressor Example

- Input
- Sampling rate 22.05 kHz
- 16 bits

- 23-band compressor
- Att time=5 msec
- Rel time=70 msec
- CR=2:1 all bands
- Input at 65 dB SPL
Compression Conclusions

• Helps at low signal levels
• Compression ratio
  - Generally prefer CR < 2:1
  - Want lower CR as noise level increases
  - Comp preferred if residual dynamic range < 30 dB
• Number of Channels
  - No clear benefit to increasing the number of bands
  - One channel shows small benefit compared to multi
  - Compressor co-modulates noise to match speech
Noise Suppression
Spectral Subtraction

• Estimate clean power spectrum
  - Start with power spectrum of the noisy speech
  - Estimate the noise power spectrum
  - Clean spectrum approximated by noisy speech spectrum minus estimated noise spectrum
  - Goal: envelope of noisy speech matches clean speech

• Problems
  - Can not extract actual noise signal, only its statistics
  - Signal reconstructed using noisy phase
  - Gain changes from segment to segment, causing “musical noise”
Generic Noise Suppression

NOISE PARAMETER ESTIMATE

COMPUTE ATTEN.

FREQUENCY ANALYSIS

RECOMBINE BANDS
Noise Estimation

• Spectral subtract needs accurate noise estimate
• Voice activity detection
  - Monitor signal segment by segment
  - Determine if segment is speech or noise
  - Update noise statistics during noise segment
  - Hold statistics constant during speech segment
• Continuous noise estimation
  - Adjust noise statistics on every segment
  - Update rate based on probability that segment is noise
  - Reduced computation better match to hearing aid
Hirsch-Ehrlicher Algorithm

- **Incoming noisy signal:** $|X(k,m)|$
- **Noise estimate:** $|N(k,m)|$
- **Update the noise estimate**

\[
\text{if } |X(k,m)| > b|N(k,m-1)| \\
|N(k,m)| = |N(k,m-1)| \\
\text{else} \\
|N(k,m)| = a|N(k,m-1)| + (1 - a)|X(k,m)| \\
\text{end}
\]
Histogram Estimation

• Sample each noisy speech segment
• Assign segment to closest histogram bin
  - Histogram bins contain count of number of segments at that intensity level
  - Decay contents of all bins
  - Increment contents of assigned bin
• Find peak of noise distribution
  - Model as two Gaussian distributions
  - Assume noise lower distribution
  - Search for peak below the mean
Log-Level Histogram
Gaussian Fit to Histogram

![Gaussian Fit to Histogram](image-url)
Noise Estimation, SNR=20dB
Wiener Filter Calculation

• Noisy signal $x(n) = s(n) + d(n)$, output $y(n)$

• Min MSE between output and clean input

$$\epsilon = \sum_{n} [s(n) - y(n)]^2$$

• Optimal filter:

$$G(f) = \frac{|S(f)|^2}{|S(f)|^2 + |D(f)|^2}$$

• Approx from noisy speech and estimated noise:

$$G(f) \approx \frac{|X(f)|^2 - |N(f)|^2}{|X(f)|^2}$$
Adaptive Wiener Filter

- Time-varying signal and noise estimates
  - Signal spectrum: Replace $X(f)$ with $X(k,m)$
  - Noise spectrum: Replace $N(f)$ with $N(k,m)$

- Time-varying gain $G_W(k,m)$ in band $k$, block $m$

$$G_W(k,m) = \frac{|X(k,m)|^2 - |N(k,m)|^2}{|X(k,m)|^2} = 1 - \frac{|N(k,m)|^2}{|X(k,m)|^2}$$
Spectral Subtraction Gain

![Graph showing spectral subtraction gain with lines for power and Wiener magnitude.](image)
TMK and Related Algorithms

• Gain based on signal and estimated noise

\[ G(k, m) = \frac{1}{1 + \nu \left( \frac{|N(k, m)|}{|X(k, m)|} \right)^\gamma} \]

• Parameters
  - Increasing \( \nu \) increases maximum attenuation
  - Increasing \( \gamma \) increases function slope with SNR
TMK Spectral Subtraction

![Graph showing SNR vs Gain for different models (i), (ii), (iii), (iv), (v).]
Spectral Subtraction Example

- Input
- Sampling rate 22.05 kHz
- 16 bits
- Stationary speech-shaped noise
- SNR=10dB

- 23-band warped system
- TMK algorithm
- 12-dB maximum attenuation
Noise Suppression Conclusions

- Little improvement in intelligibility
  - Depends on algorithm
  - TMK and Ephraim-Malah most effective
- Some improvement in quality
  - TMK shows benefit for NH and HI listeners
  - Strongest effect in range of 10 to 20 dB SNR
- Greatest benefit in stationary noise
Feedback Cancellation
The Problem of Feedback

• Primary sources of feedback
  - Acoustic from vent and leaks
  - Mechanical from receiver vibration

• Effect on hearing aid
  - Instability causes "whistle" or high-frequency tone
  - Ringing after sound stops
  - Hearing aid amplifier saturates => distortion

• Maximum stable gain
  - Maximum gain for operation without whistles
  - Problem most acute at high frequencies
Algorithm Goals

• Increased gain
  - Achieve fitting targets
  - Improved speech intelligibility

• No processing artifacts
  - Hearing aid always stable
  - No whistles due to feedback
  - No chirps, momentary tones, or audible gain changes

• Computationally efficient
Feedback System

**Feedback System Diagram**

- **Input**: $X(f)$
- **Mic**: $M(f)$
- **S(f)**
- **Hearing Aid Processing**: $H(f)$
- **Amp**: $A(f)$
- **Receiver**: $R(f)$
- **Output**: $Y(f)$

**Paths**

- **Acoustic + Mechanical Feedback Path**: $B(f)$
- **Acoustic Feed-forward Path**: $C(f)$

**Symbols**

- $F(f)$
- $U(f)$
System Output

- Output = Processed + Direct

\[ Y = X \frac{C + H[MAR]}{1 - BC - H[MARB]} \]

- Approximate solution for \(|BC| \ll 1\)

\[ Y \approx X \frac{C + H[MAR]}{1 - H[MARB]} \]

- Stability guaranteed for

\[ |H[MARB]| < 1 \]
Effects of ITE Vent Feedback
Adaptive Feedback Cancellation

• Model feedback path
  - Adaptive model to minimize system power
  - Subtract model output from microphone signal

• Why adaptive?
  - Feedback path can change over time
  - Track changes

• Limitations
  - Resolution of the feedback path model
  - Ability to track rapid changes
  - Cancellation of tonal inputs
  - Room reverberation
Feedback Path Model

- Electroacoustics vary slowly
  - Microphone
  - Amplifier
  - Receiver

- Acoustics can change rapidly
  - Telephone handset
  - Hat
  - Earmold shift in the ear canal (jaw movement)

- Model = Fixed in series with adaptive
  - Fixed delay + IIR filter for computational efficiency
  - Short adaptive FIR filter to model changes
System Output

- Output = Processed + Direct

\[ Y = X \frac{C + H[WC + MAR]}{[1 - BC] - H[MARB - W[1 - BC]]} \]

- Approximate solution for \( |BC| << 1 \)

\[ Y \approx X \frac{C + H[WC + MAR]}{1 - H[MARB - W]} \]

- Stability guaranteed for

\[ |H[MARB - W]| < 1 \]
LMS Adaptation

- Adapt FIR filter coefficients \( \{g_k\} \)
- Minimize error \( \varepsilon(n) = e^2(n) \)
- Compare error with processed
  - Error = microphone output - feedback path model
  - Processed = HA output => delay & fixed filter

- Weight update

\[
g_k(n + 1) = g_k(n) + \frac{2\mu}{\sigma_d^2} e(n)d(n - k)
\]

- Proportional to error
- Trade-off between fast time constant and accuracy
Filtered-X Algorithm

• Objective
  - LMS adaptation works best for white noise input
  - Track changes in the feedback path
  - Prevent response to sinusoidal inputs

• Approach
  - Goal is to decorrelate (whiten) the inputs to the LMS cross-correlation
  - Filter inputs to remove sinusoids
  - Fixed filter based on average signal spectrum
  - Adaptive filter to detect and remove sinusoids
Fig 7.15 Adaptive Filtered-X
Feedback Conclusions

• Feedback cancellation
  - Model the feedback path
  - Subtract model output from microphone signal
  - LMS adaptive filter update
  - Filtered-X approach most effective

• Processing effectiveness
  - Generally get 10 - 15 dB headroom increase
  - Increased gain gives increased audibility
  - Increased stability improves sound quality

• Problems remain
  - LMS tries to cancel tones and narrowband inputs
  - Reverberation
  - Clients don't like initialization probe signal
  - Linear model affected by distortion (e.g. power aids)
Microphones and Arrays
Omnidirectional Microphone

• Single Surface
  - Measure pressure at a point in space
  - Proportional to membrane displacement

• Microphone Technology
  - Moving coil
  - Electret
  - Silicon
Directional Microphone

- Measure pressure at two nearby points
  - Pressure difference
  - Proportional to velocity

- Implementation
  - Pressure difference across a single membrane
  - Difference between adjacent omnidirectional mics

- Control the directional pattern
  - Microphone separation
  - Time delay of rear microphone
  - Relative amplitude of rear microphone
Spatial Signal Processing

\[ y(t) = \frac{x_1(t) + x_2(t)}{2} \]

\[ x_1(t) \quad \text{FRONT MIC} \]

\[ x_2(t) \quad \text{REAR MIC} \]

GAIN \( b \)

DELAY \( \tau \)

\[ \text{DIRECTION OF ARRIVAL} \quad \theta \]

\[ \text{SEPARATION} \quad d \]
# Directional Responses

The directional response is \( R(\theta) = a + (1-a)\cos(\theta) \).

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Rear Mic Delay ( \tau )</th>
<th>( a )</th>
<th>DI, dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>Omnidirectional</td>
<td>--</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>Cosine</td>
<td>0</td>
<td>0</td>
<td>4.8</td>
</tr>
<tr>
<td>Hypercardioid</td>
<td>( d/3c )</td>
<td>0.25</td>
<td>6.0</td>
</tr>
<tr>
<td>Supercardioid</td>
<td>( 2d/3c )</td>
<td>0.4</td>
<td>5.7</td>
</tr>
<tr>
<td>Cardioid</td>
<td>( d/c )</td>
<td>0.5</td>
<td>4.8</td>
</tr>
</tbody>
</table>
Hypercardioid Pattern
Supercardioid Pattern
Cardioid Pattern
Adaptive Microphone Arrays

• Start with 2 microphones
  - Extension of directional microphone technology
  - Can be built into hearing-aid case

• Multi-microphone arrays
  - Combine outputs from multiple microphones
  - Improved directivity (DI) compared to 2-mic array
  - Fixed combination of outputs
  - Adaptive combination of outputs
Cardioids with Adaptive Gain

\[ x_1(n) \xrightarrow{\text{SUM}} c_1(n) \xrightarrow{\text{SUM}} y(n) \]

\[ x_2(n) \xrightarrow{\text{SUM}} c_2(n) \rightarrow \text{GAIN } b \]

\[ + \quad + \quad + \]

\[ - \quad - \quad - \]

\[ \text{FRONT MIC} \quad \text{REAR MIC} \]

\[ \text{DELAY d/c sec} \quad \text{DELAY d/c sec} \]

\[ d \]
Adaptive Gain

• Combine mics to give cardioid responses
  - One response faces forward, null at 180 deg
  - Second response faces rear, null at 0 deg

• Cardioid outputs $c_1(n)$ and $c_2(n)$
  - Also give directional patterns
    \[ c_1(\theta) = \frac{1}{2}[1 + \cos(\theta)] \quad c_2(\theta) = \frac{1}{2}[1 - \cos(\theta)] \]
  - Front output $c_1(n)$ primarily desired signal
  - Rear output $c_2(n)$ primarily interference
Combined Cardioid Outputs

- Sum cardioid signals: \( y(n) = c_1(n) - bc_2(n) \)
- Sum directional patterns: \( y(\theta) = c_1(\theta) - bc_2(\theta) \)
- Null direction depends on \( b \)
  - Set \( b=0 \): front cardioid pattern
    \[ y(n) = c_1(n) \quad y(\theta) = 1/2[1 + \cos(\theta)] \]
  - Set \( b=1 \): cosine pattern
    \[ y(n) = c_1(n) - c_2(n) \]
    \[ y(\theta) = c_1(\theta) - c_2(\theta) = 1/2[1 + \cos(\theta)] - 1/2[1 - \cos(\theta)] = \cos(\theta) \]
- Adjust \( 0 \leq b \leq 1 \) to steer null 180 to 90 deg
LMS Adaptive Update

- Adjust $b$ based on rear signal and output
  \[ b(n + 1) = b(n) + \frac{\mu}{\sigma_y^2} c_2(n)y(n) \]

- Interference from side
  - $c_2(n)$ similar to $y(n)$, interference in both signals
  - $b$ is driven towards 1, cancel interference at 90 deg

- Interference from rear
  - Average of $c_2(n)$ times $y(n)$ tends negative
  - $b$ is driven negative
  - Pattern moves towards forward cardioid
Microphone Conclusions

- Summary from Walden et al (2003):
  - Directional mics test better than real-world results
  - Environment often limits directional benefit
  - Benefit should be expected only when the signal source is in front of and relatively near the listener and the interference is spatially separated
  - Omnidirectional mode should be the default
Additional Processing Areas
Additional Areas, I

• Electro-acoustic interactions
  - Head and external ear modify microphone input
  - Vent reduces system low-frequency response
  - Occlusion effect: own voice louder

• Wind noise
  - Wind flow over microphone generates turbulence
  - Directional mics especially sensitive

• Spectral enhancement
  - Broader auditory filters reduce spectral contrast
  - Modify short-time spectrum to increase contrast
Additional Areas, II

• Sound classification
  - Change parameters for different environments
  - Identify environments to set optimum processing

• Binaural processing
  - Link the hearing aids at the two ears
  - Low-power short-range communication link
  - Synchronize device parameter settings
  - Compare signals for compression and noise suppression
Conclusions
Conclusions

• Hearing aid = perceptual engineering
  - Designs apply criteria such as MMSE or Max Likelihood
  - These criteria ignore perception
  - Need to consider the human listener as well

• Problems remain
  - Have not solved the basic problem of speech intelligibility in noise
  - Most algorithms are not effective despite having a strong intuitive appeal
• April 2008
• Plural Publishing, San Diego