

SPAU 332

Hearing Aids I

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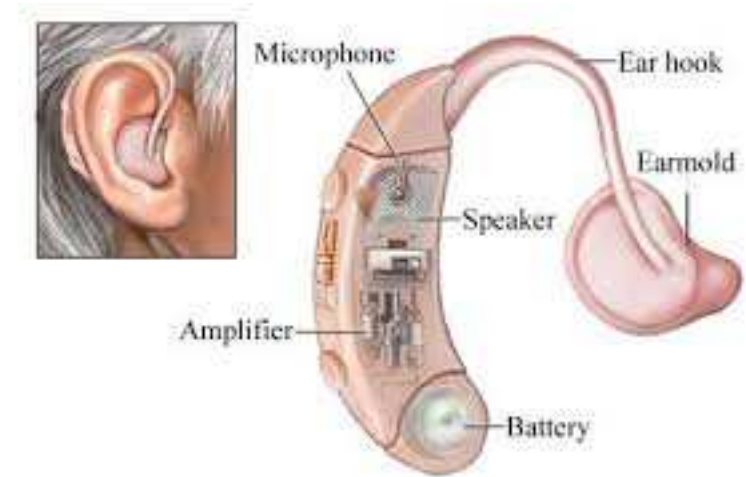
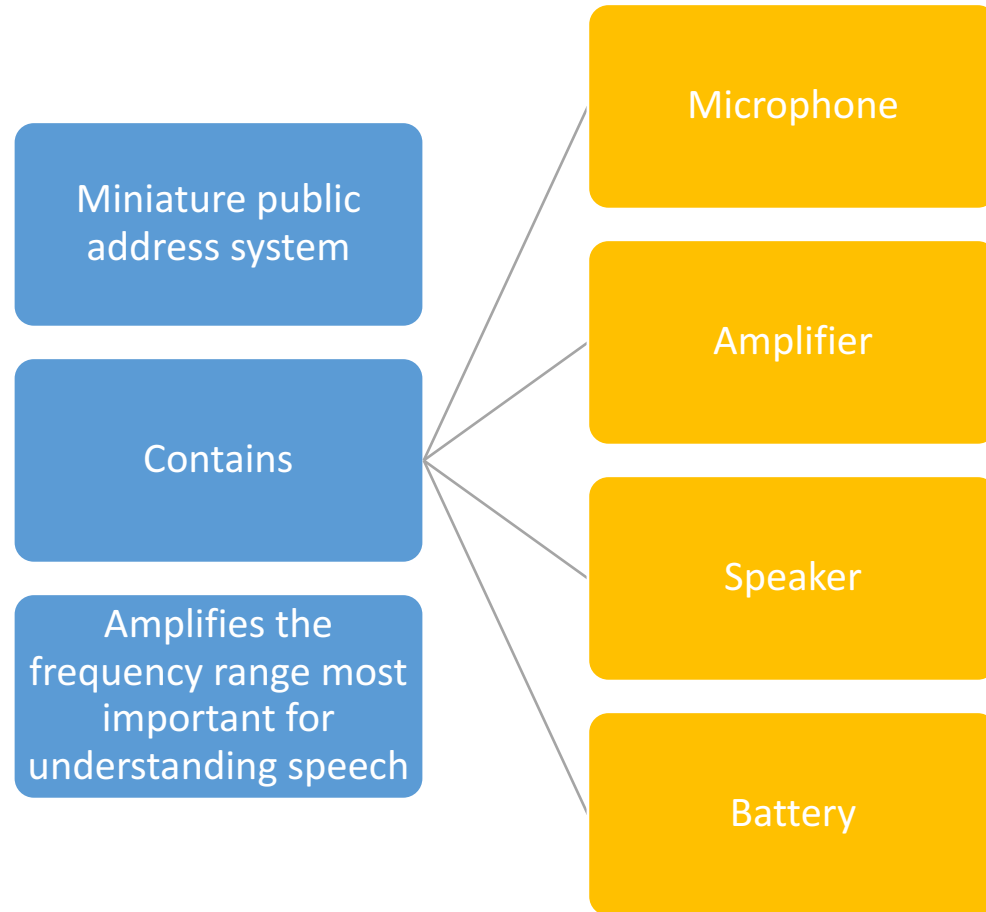


Hearing Aid Components and Principles

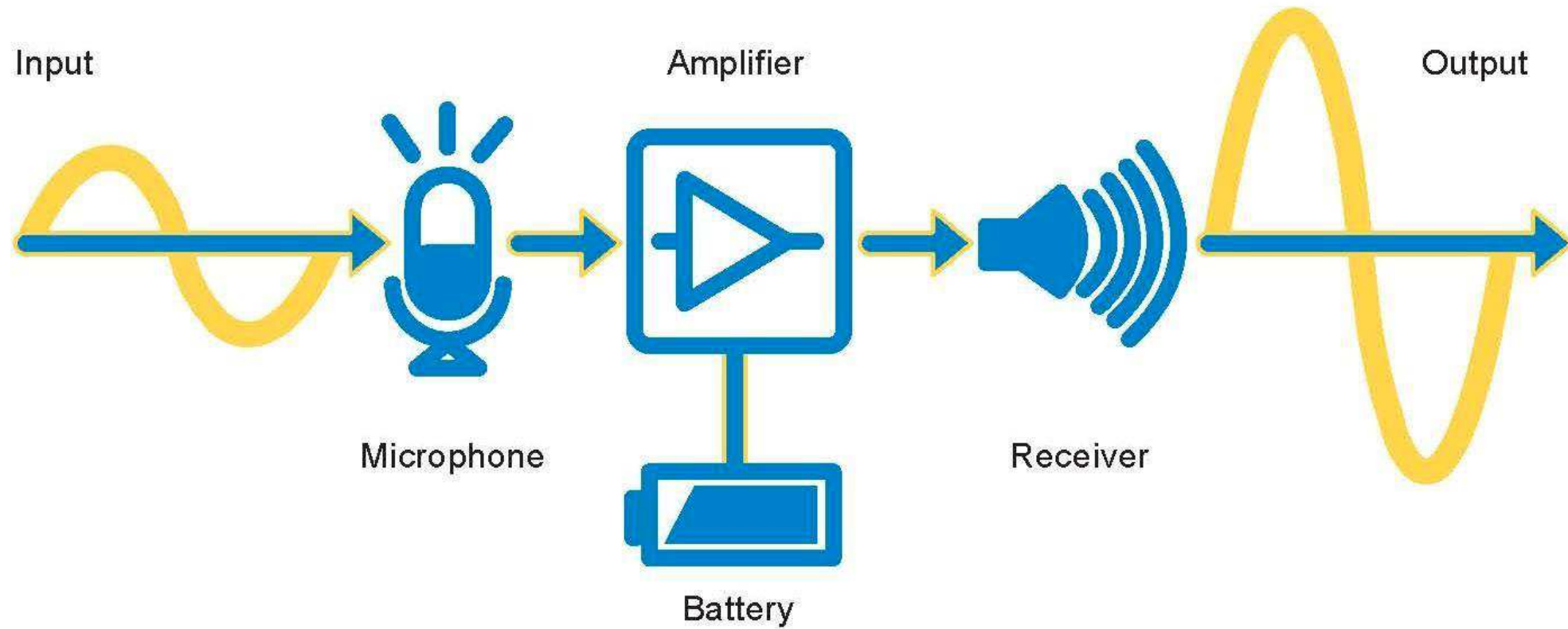
**Aim of today's
lecture:**

Demonstrate
understanding of
hearing instrument
components

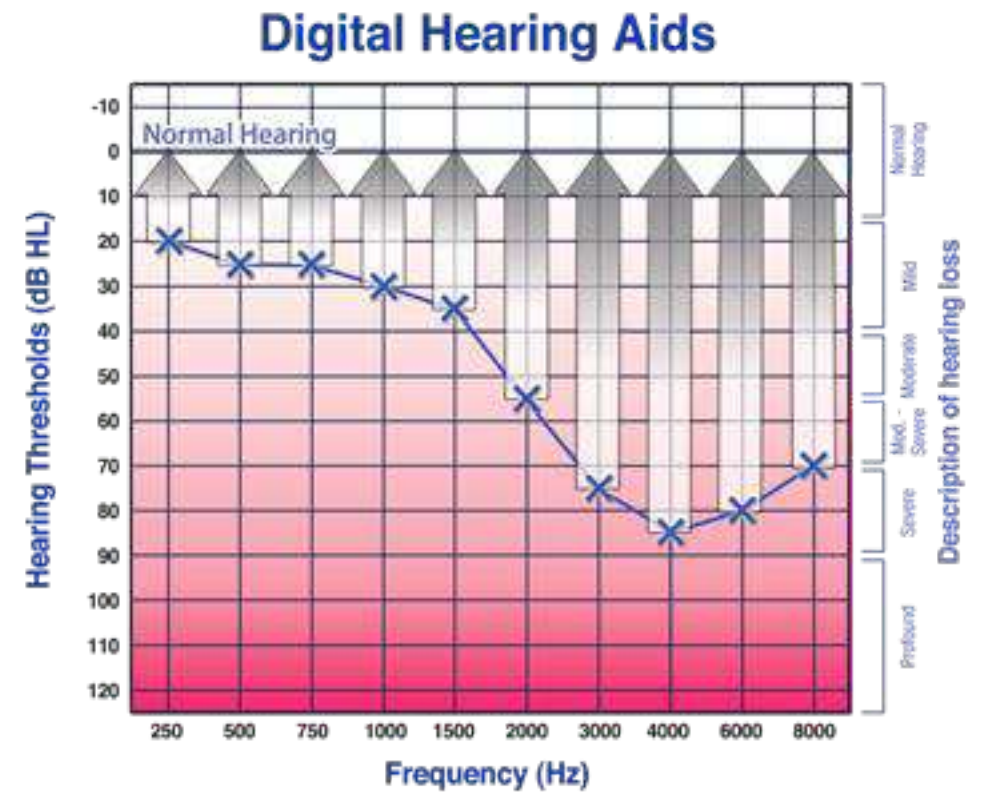
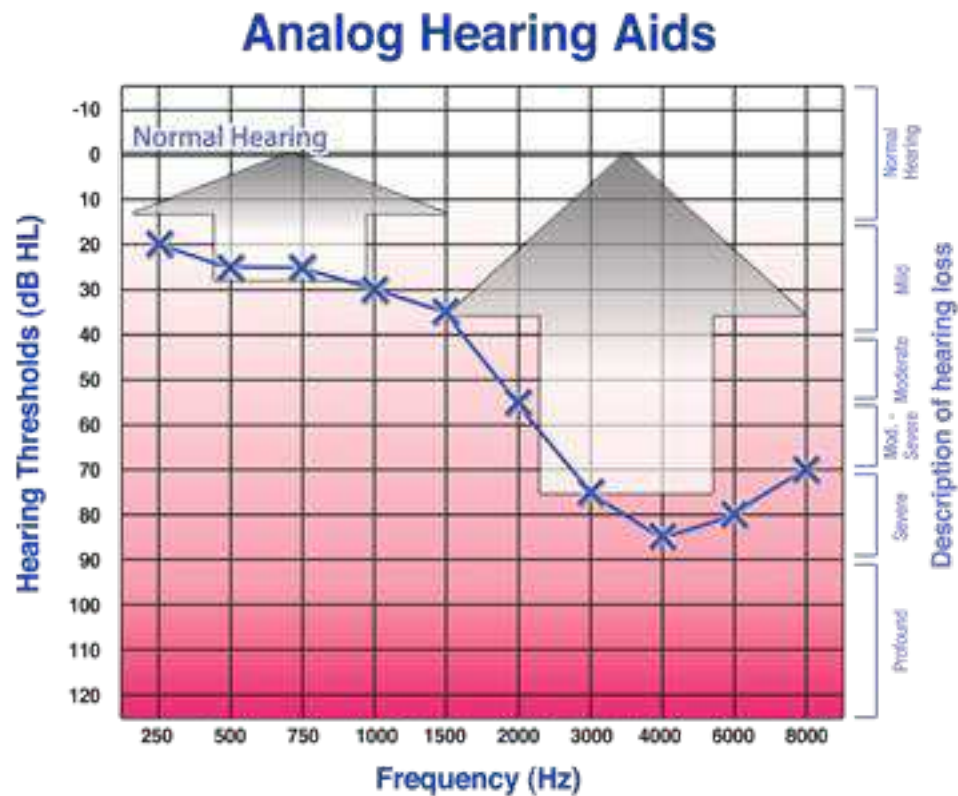
What hearing instruments do



What is amplification?



What is amplification?



Hearing aid batteries

- **Most commonly zinc/air**

- Relatively high energy density
- Inexpensive materials
- Constant voltage rate over a relatively long time
- Very low self-discharge rate under sealed condition
- Relatively long-lasting under low power conditions
- Proven technology
- Once the seal is broken self-discharge starts irrevocably and at a high rate
- Not rechargeable



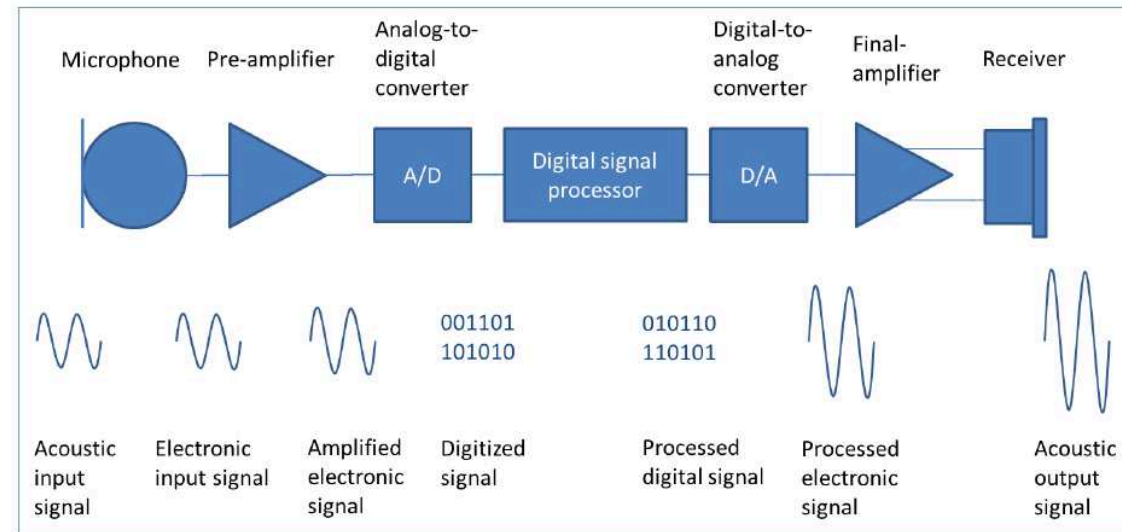
Hearing aid batteries

- **Rechargeable batteries**
 - What are the advantages of rechargeable batteries compared to disposable?

Transducer

- Any device that changes energy from one form to another

Digital hearing aid components



Microphones



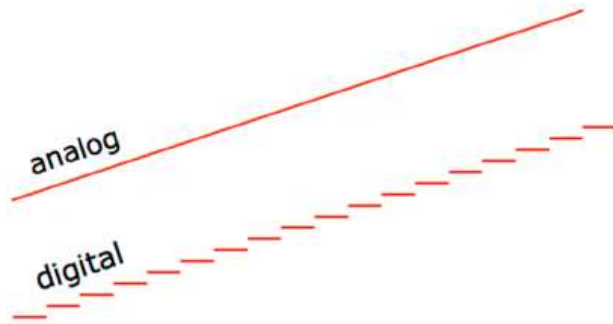
- Input transducer
 - Convert acoustical sound to electrical energy
- Have internal noise due to components of the electrical circuit
- Wind striking the microphone causes noise
- Easily damaged by debris

Receivers



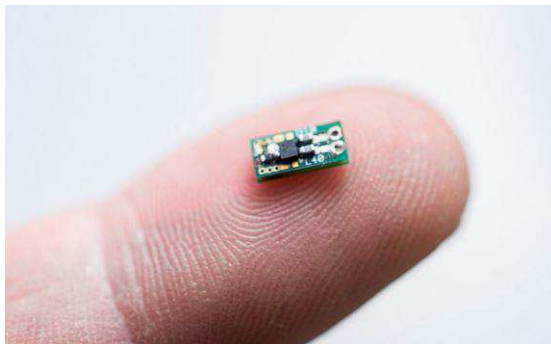
- Output transducer
- Change the amplified electrical signal into acoustic form
- The size of the receiver determines its output
- The receiver is a major consumer of the hearing aid battery
- Easily damaged by debris
- Easily damaged by dropping (may continue to work but could be distorting)
- Receiver vibrations can lead to vibratory feedback due to proximity to other components

Digital amplifiers



- Analog-to-digital converter
 - Digitizes electrical waveform
 - Samples at discrete points in time
- Digital amplifier
 - Able to manipulate information at speed
 - Allows for:
 - Less internal noise
 - Less distortion
 - Great shaping flexibility of incoming sound
 - Ability to perform changes in the frequency response e.g. noise suppression, feedback management
- Digital-to-analog converter
 - Converts digital waveform back to analog output

Digital signal processing (DSP)



- Amplifier performs series of very fast calculations
- Chip technology
 - Electronics on integrated circuit board
 - New chip every two to three years
 - All major manufacturers use one chip for an entire line of products

DSP Cont.

Smaller components (e.g. tiny microchip) that can handle complex signals

Consumes less power

Less internal noises

Multi-frequency bands processing

Increasingly complex directional mic systems

Potential for improvements in background noise

Improved solution to feedback problems

More precise processing that produces better signal quality

Analog Vs. Digital HA's

- Analog: Old hearing aids technology.
- The difference b/w digital and analog HAs is in the way that they process sound and the individual benefits that they offer
- Analog HAs use conventional electronics (analogue circuit) to convert sound into electrical signals that are amplified
- The electrical current is analogous to the acoustic sound pressure

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Analog HA's

Components are of bigger size

Have Less than 3 frequency bands

Poor quality of output signal

Signal processing is limited to amplification and frequency shaping

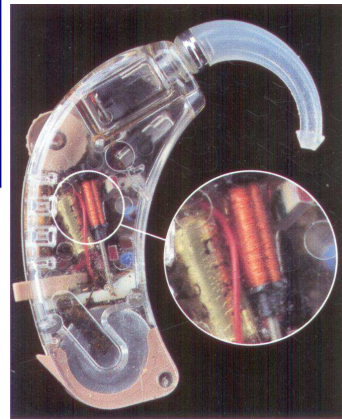
Lower prices on average

Easier to set up

What are Digital and Analog Signals? - Definition & Explanation

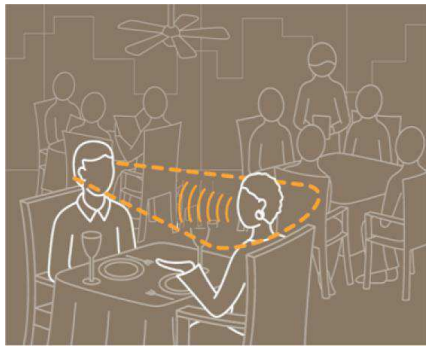
- <https://www.youtube.com/watch?v=btgAUdbj85E>
- <https://study.com/academy/lesson/what-are-digital-and-analog-signals-definition-lesson-quiz.html>

Telecoil



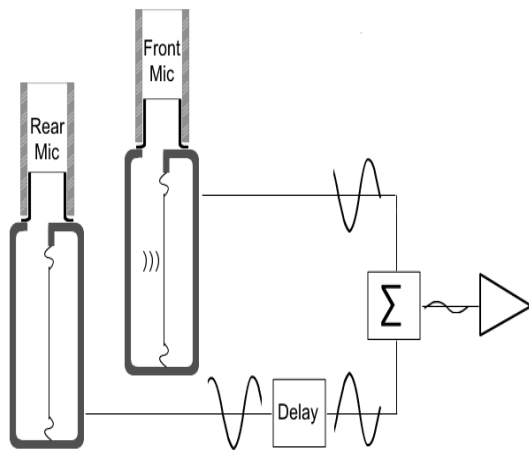
- Induction loop facilities found in facilities
 - E.g. public theatres, banks, places of worship
- Improves signal-to-noise ratio
- Prone to interference
- Another type of input transducer
- Uses electromagnetic energy present around telephone
- Bringing the phone close to the aid allows the magnetic signal from the telephone to pass directly into the hearing aid
- Using a magnetic connection (rather than an airborne signal) eliminates feedback from the signal

Directional microphones



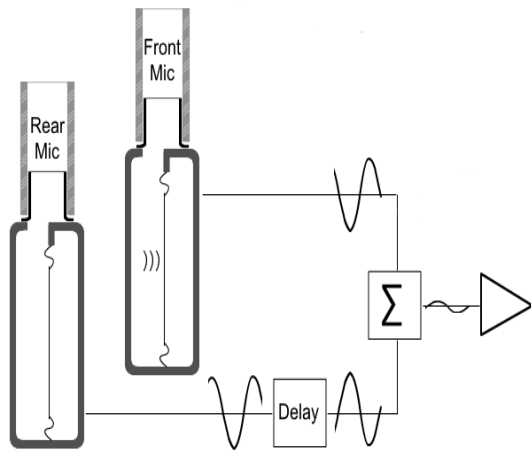
- Inability to hear speech in the presence of background noise is major reason hearing aids are rejected
- Directional microphones depend on noise being spatially separated from speech
- Reduce sound coming from behind rather than increasing sound from in front
- Improve signal-to-noise ratio (SNR)

Directional microphones



- Two omnidirectional microphones
- Delay from rear microphone is created electronically
- Electrical signals from front and rear microphones cancel each other when input is from the rear
- Sound enters the microphones where the acoustic energy is converted to electrical energy

Directional microphones



- The two signals are sent through an electrical network where a time delay is applied to the rear microphone signal
- The two signals are subtracted to produce directivity
- When both microphones are active, a directional pattern is achieved
- When an omnidirectional condition is desired, the rear microphone is shut off

Directional microphones

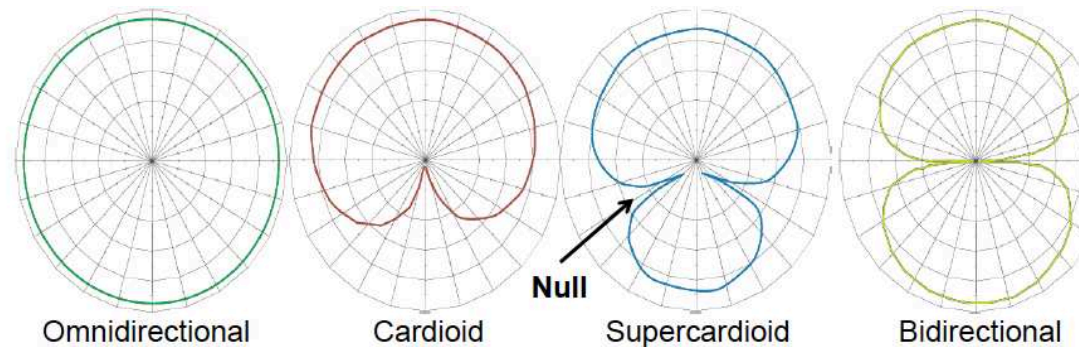
- A variety of spatial directional patterns can be achieved by changing the time delay applied to the signals
- Directivity index (DI)
 - Standard for measurement of effectiveness of directional microphones
- The greater the DI, the more effective the separation of signal and noise
- Current directional microphones have a DI of up to 6 dB
- DI are represented using polar plots

Directional microphones – polar plots

- Plot of output intensity for a 360-degree pattern for sound arriving at the microphone
- Constructed by measuring the output of the hearing aid at several points within an imaginary sphere around the microphone

Directional microphones – polar plots

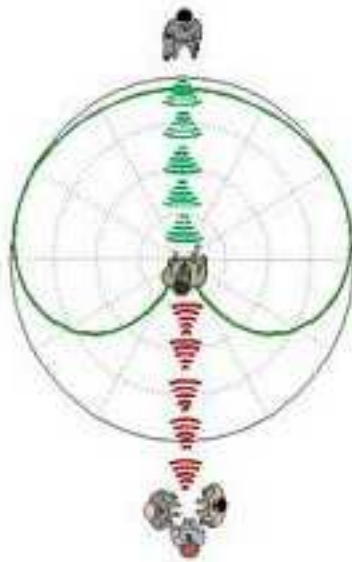
- Fixed directional
 - Nulls are always at the same angle of the pattern
 - Common directional polar plots:



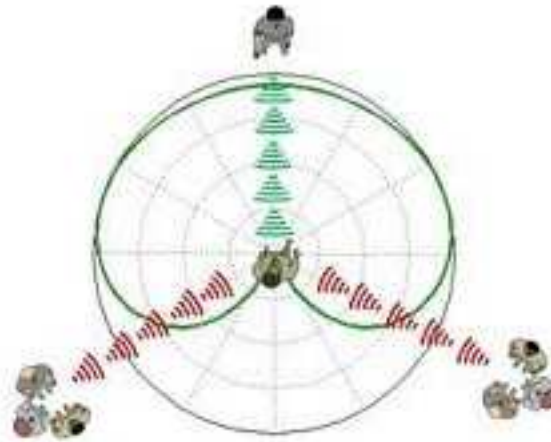
Directional microphones

- Directionality is most effective when the signal of interest is in front of the listener and within about two meters of them
- Beyond this distance, directional microphones do not provide significant benefits
- Directional microphones work best when the noise and signal of interest are spatially separated (coming from different directions)

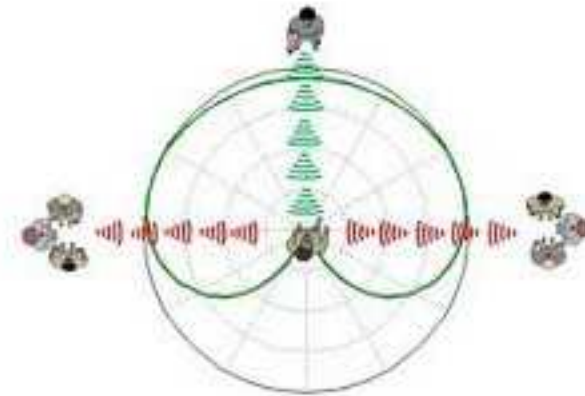
Directional microphones



Attenuation of noise due to the null position will lead to large improvement in signal-to-noise ratio



Less improvement in signal-to-noise ratio



No improvement in signal-to-noise ratio

Directional microphones

- With fixed directionality users have to manually switch programs in noisy environments
- But
 - Some users do not switch between settings
 - Some users do not know when to switch
 - Some users do not want to manually switch

Automatic switching

- Hearing aid automatically changes from an omnidirectional setting to a directional setting depending on the environment
- Switching algorithm depends on environmental classification systems within the hearing aids
 - Analyses acoustic scene and decides which microphone mode would be most beneficial

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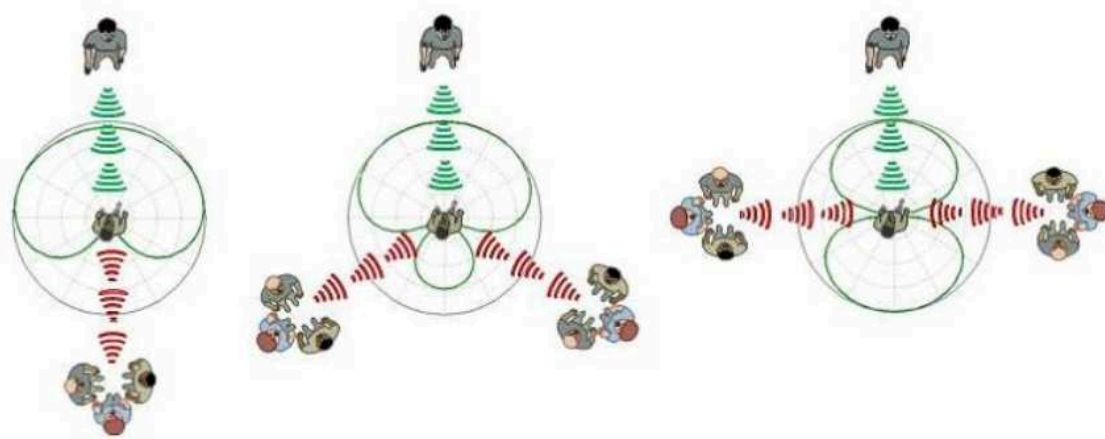
Adaptive directionality

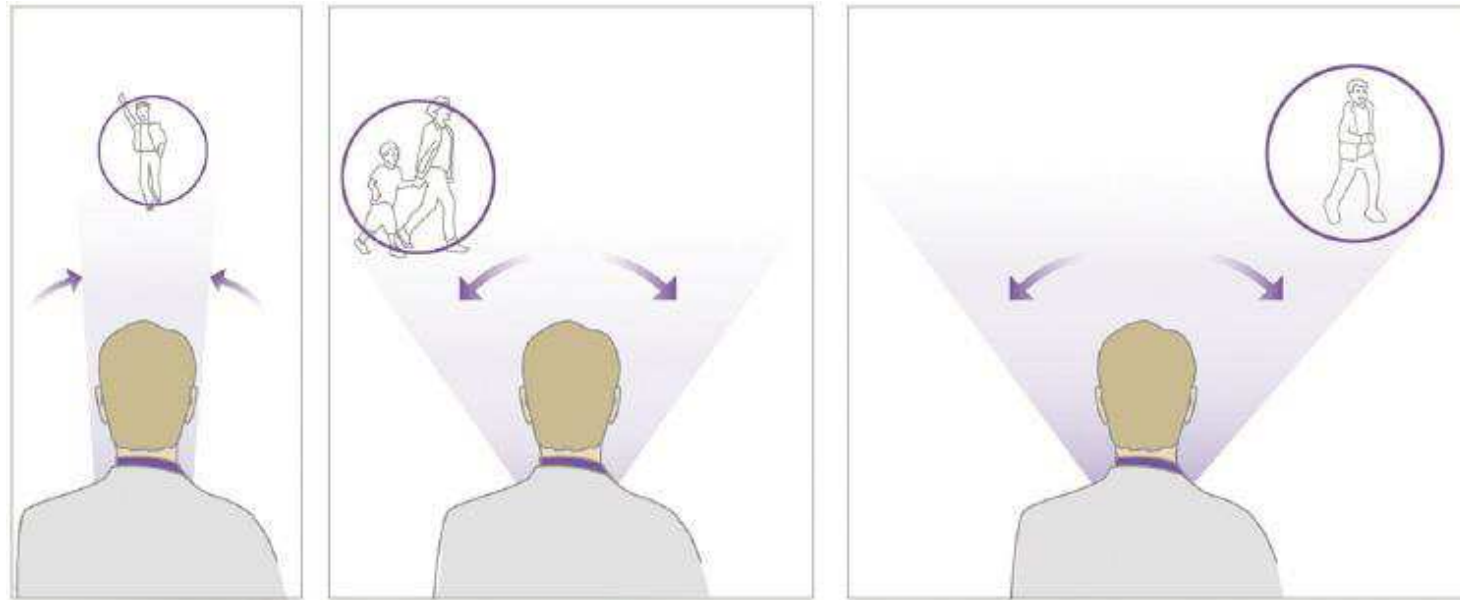
Benefit

- When most or all the noise is coming from a particularly location the hearing aid can put the point of maximum attenuation at this location to improve the signal-to-noise ratio

Limitation

- Systems are limited by the accuracy of the classification system and have no ability to determine the hearing aid user's intent in complex listening situations



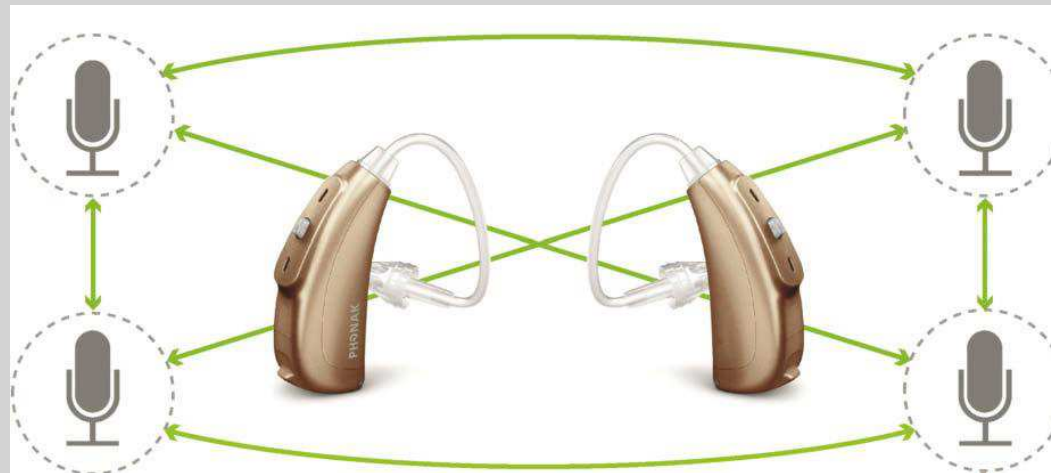


Beam forming

- Front beam can be narrowed or widened depending on signal level to the front

Audio data transfer between hearing aids

- Wireless technology
 - Left and right hearing aids coupled
 - Network of four microphones across two hearing aids
 - Improves beamforming capabilities



Directional microphones

- Key information
 - Improve SNR by 2-6 dB
 - Fixed and adaptive directional microphones perform equally in everyday listening situations
 - Automatic directional technology may be easier to use as at least one-third of hearing aid users either forget to switch or do not understand how to switch to a directional mode
 - Most adults will benefit from directional microphone technology to some degree. Will depend on:
 - Person's hearing loss
 - The signal-to-noise ratio of their common listening situation
 - For people who do not benefit sufficiently, personal assistive listening technology is available

Noise Reduction Algorithms

Noise reduction algorithms aim to reduce unwanted background noise.



Two approaches/algorithms:

Noise suppression

Spectral subtraction

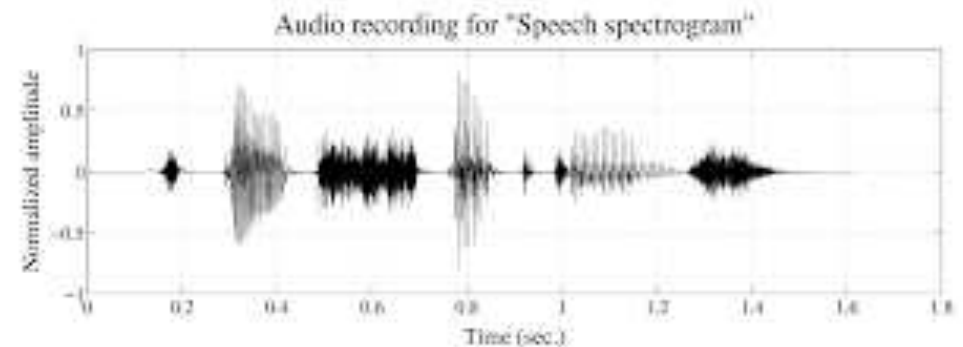
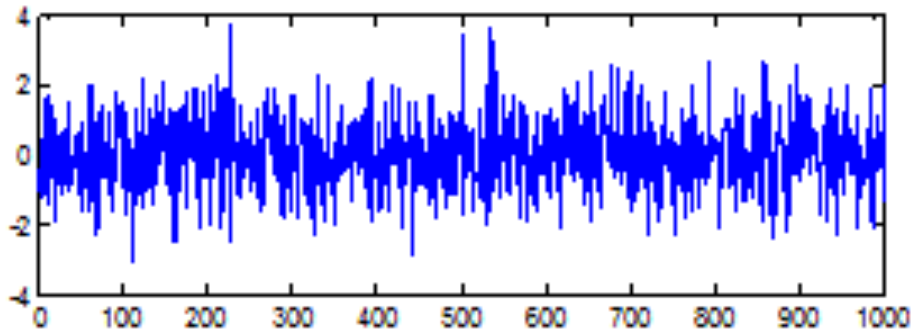


Noise Suppression

- Involves two mode of action:
 - Modulation detection
 - Synchrony detection

Noise Suppression: Modulation Detection

- AKA amplitude modulation noise reduction
- There is an assumption: noise is stationary which means that there is almost no modulation/fluctuation in the amplitude)
- Reduces the gain in channels dominated by the stationary noise





Noise Suppression: Synchrony Detection

- Attempts to detect the presence of speech in different frequency bands based on the following assumptions
 - Voiced speech sounds have dominant low frequency fundamental frequency and harmonics
 - Identifies speech sounds by their fundamental frequencies (F_0) and harmonics (F_1 , F_2 , F_3)

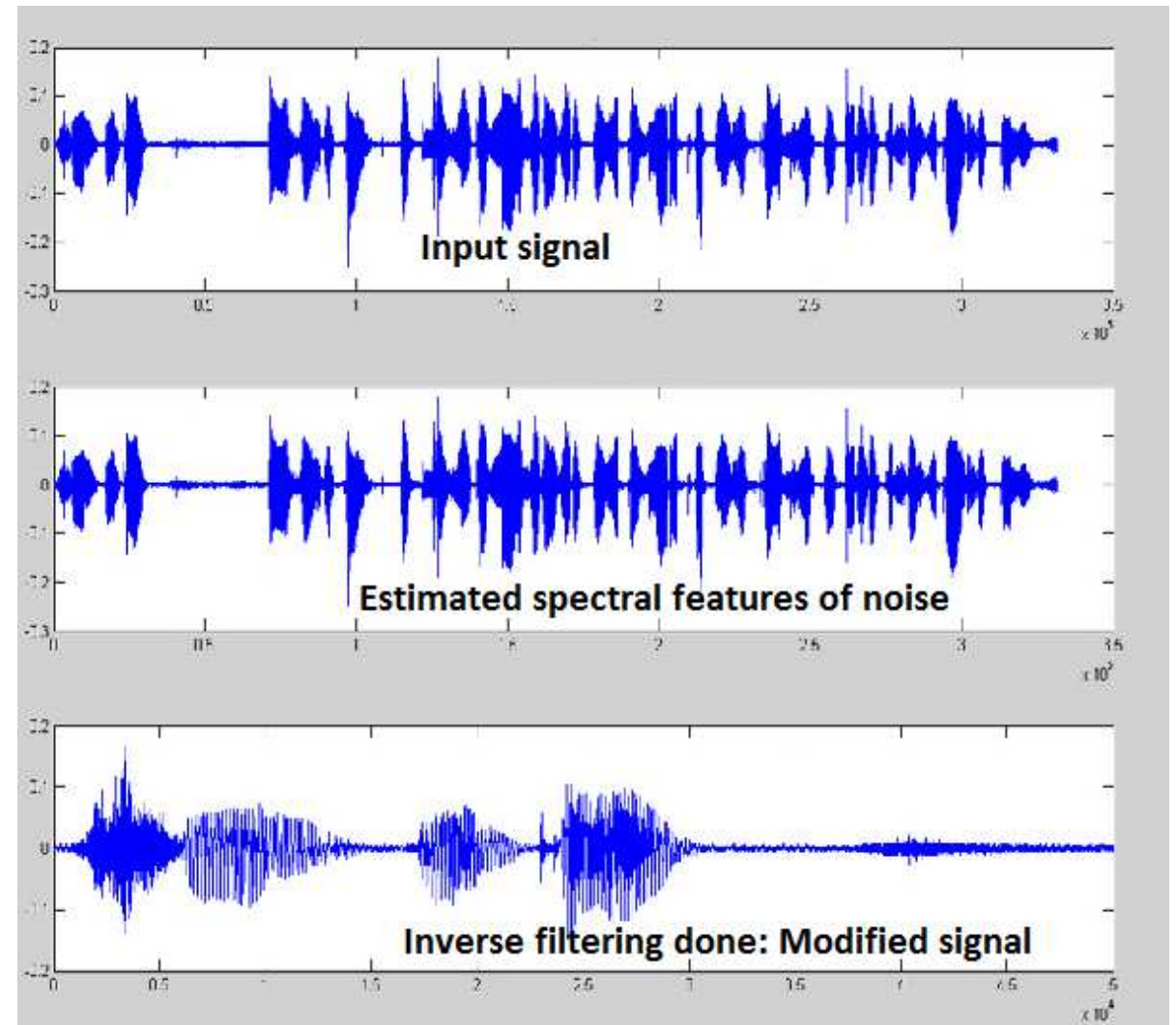


Noise Suppression: Combined Analysis

- In noise suppression, both modulation and synchrony detection work together.
- Synchrony detection accurately identifies the presence or absence of speech.
- Modulation detection provides info on the amount of noise in each channel.
- Therefore the gain is reduced at these channels accordingly.

Spectral Subtraction

- Aim: to estimate the spectral characteristics of the noise during pauses of speech
- Then subtract it from the speech using inverse filtering



Factors influencing the effectiveness of noise reduction

The accuracy of the identification of noise and speech

The type of noise

The number of channels

The amount of noise detected in each channel

Factors influencing the effectiveness of noise reduction

Ricketts & Hornsby (2005): reduced loudness and annoyance of noise without reducing speech perception: spectral characteristics of speech and noise overlap which doesn't result in improved SNR

Less cognitive load/effortful listening

Boymans et al. (1999): no improvements in speech intelligibility but it doesn't decrease speech intelligibility either

Evidence is inconclusive

Acoustic Feedback

- Caused by the leakage of the sound from the HA speaker (rec) back to the mic
- This sound wave leakage from the OP back to the IP produces a form of instability, resulting in an audible feedback sound
- Feedback typically occurs b/w 2-5 KHz and is often initiated by high-frequency gain of the HA
- Reduced the max amplification that can be used in the HA without making it unstable



Feedback Reduction Algorithms

- Two approaches:
 - Gain reduction method
 - Feedback phase cancellation

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Gain Reduction

Reducing the overall gain
(old approach)

- Disadvantage: compromises audibility

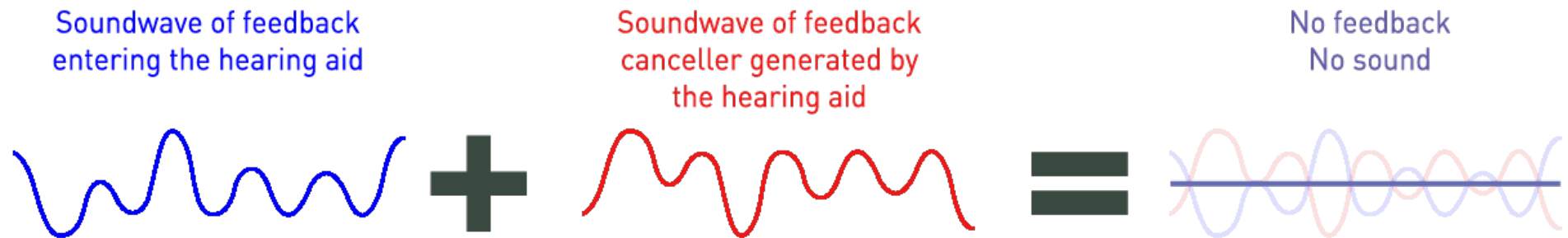
Identifies the frequencies at
which the feedback is
occurring and reduces the
gain in this frequency region
using a narrow band filter.

- Disadvantage: may compromise speech components in the targeted

Feedback Phase Cancellation

Identifies the feedback signal then inverts the signal making it out of phase → cancels the signal

- Advantage: tries to cancel just the feedback noise



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