SPAU 332
Hearing Aids I

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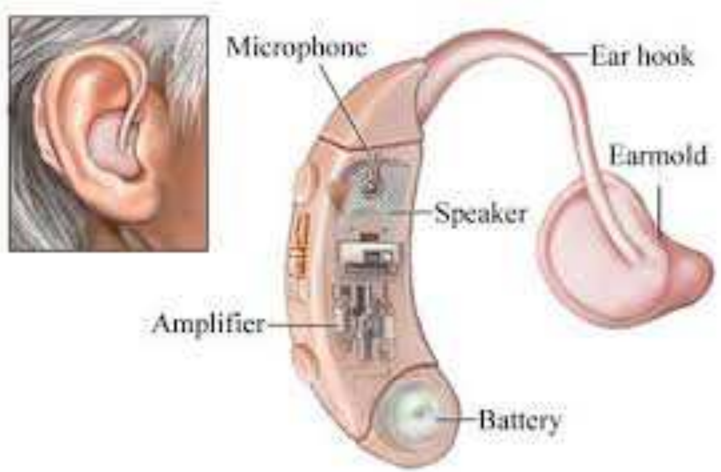
Hearing Aid Components and Principles

Aim of today's lecture:

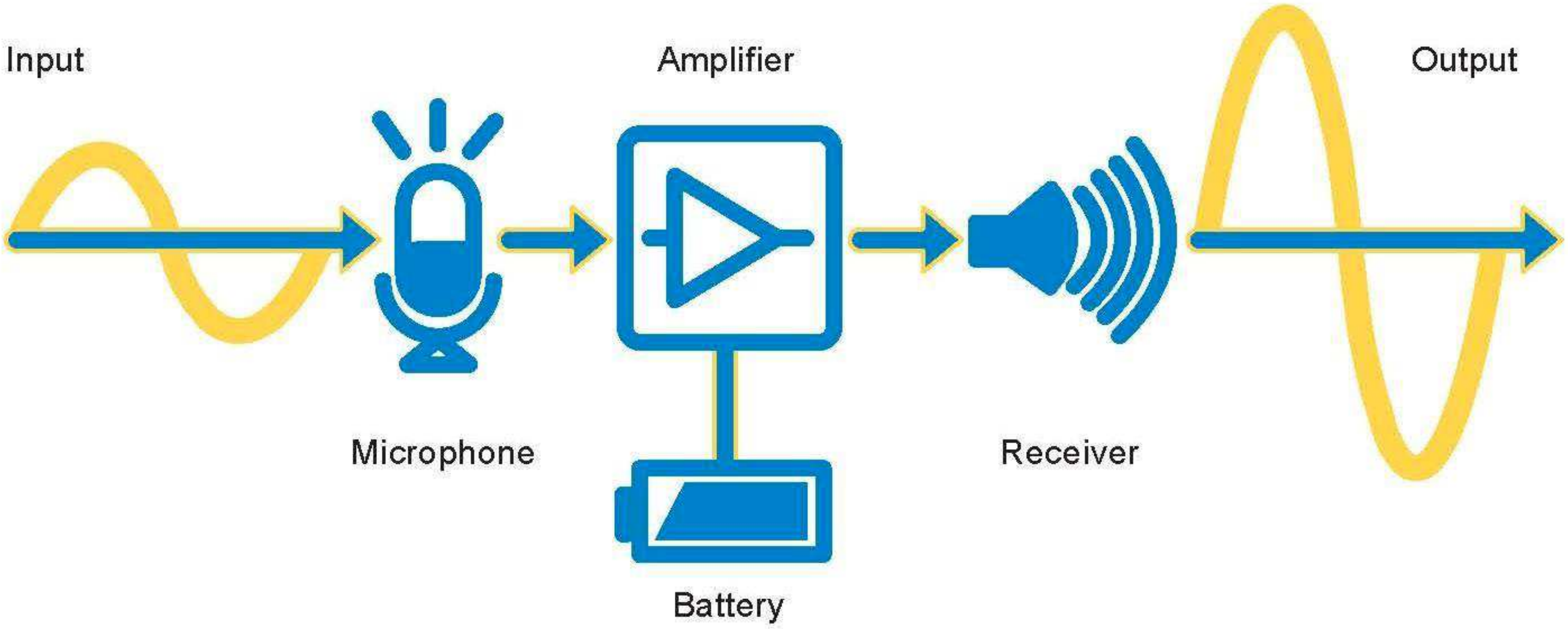
Demonstrate understanding of hearing instrument components

What hearing instruments do

- Miniature public address system
- Contains
  - Microphone
  - Amplifier
  - Speaker
  - Battery
- Amplifies the frequency range most important for understanding speech



What is amplification?



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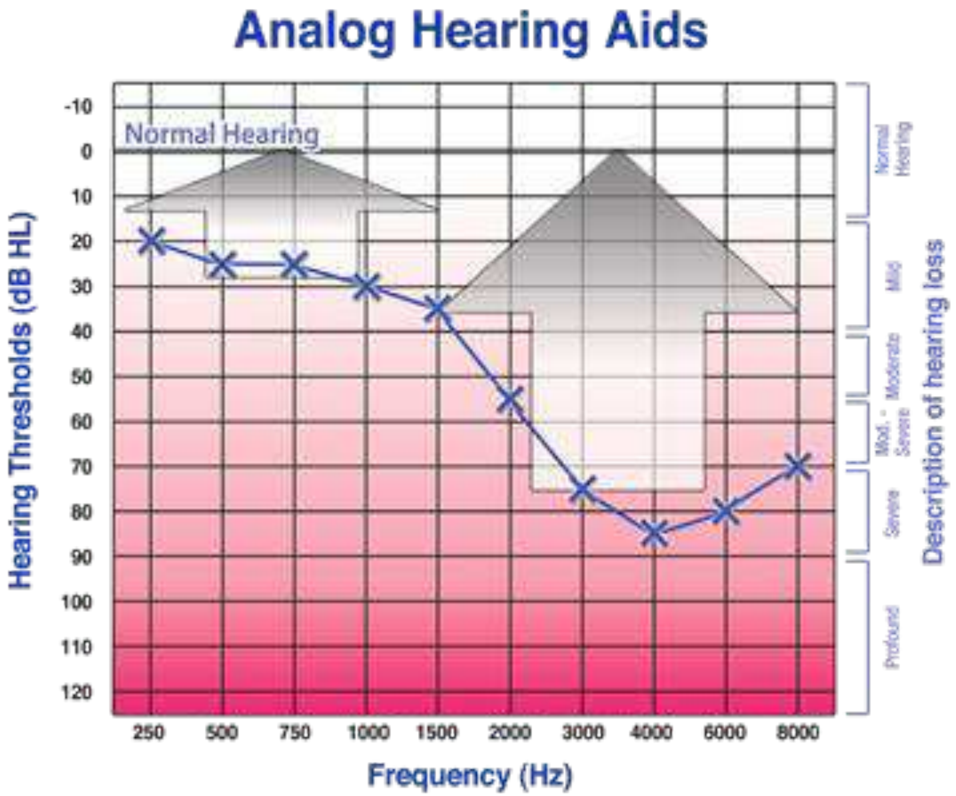
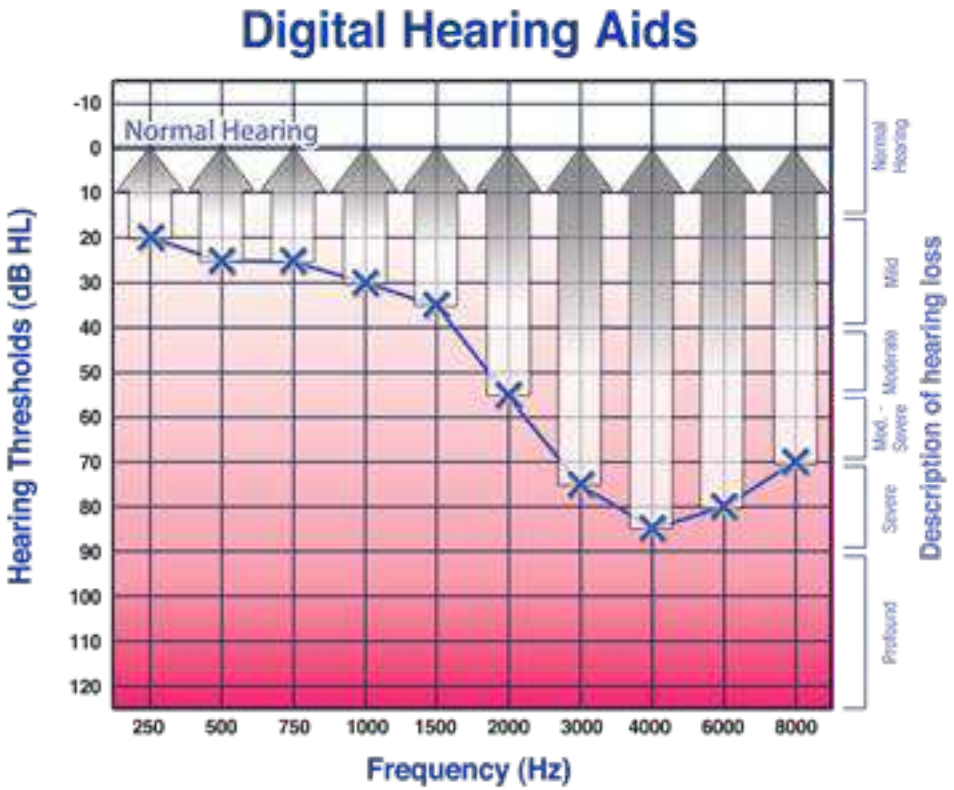




Hearing aid batteries

- **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

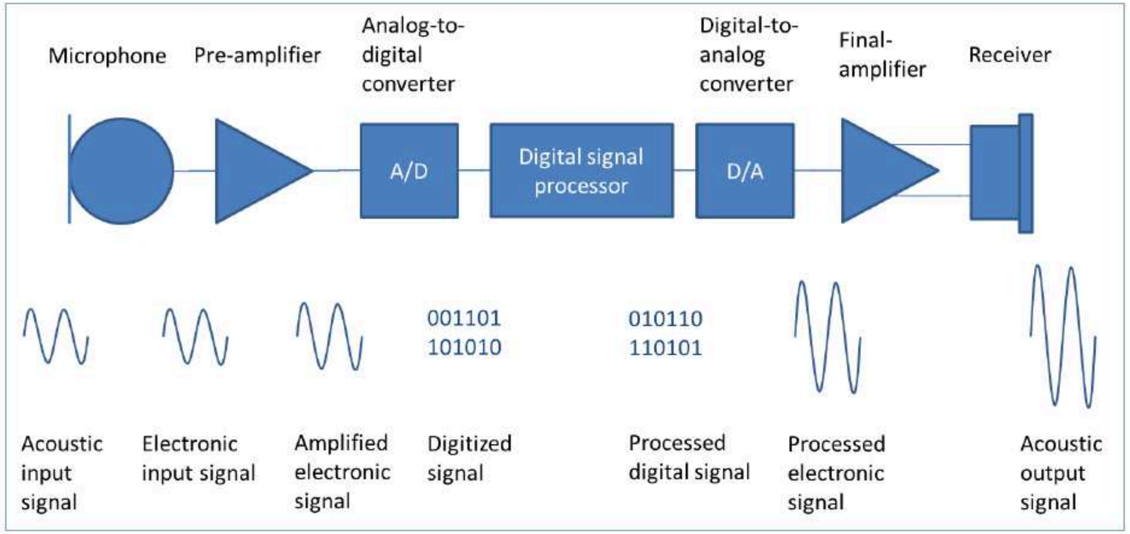
Hearing aid batteries

- **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



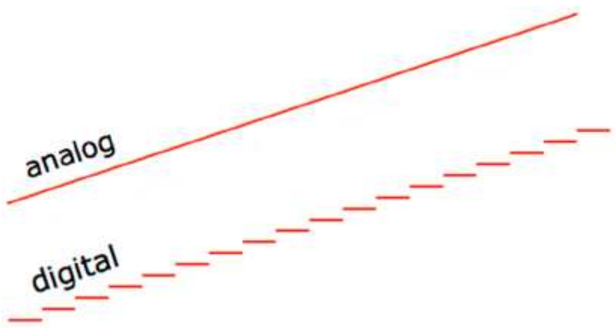
Microphones

- 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

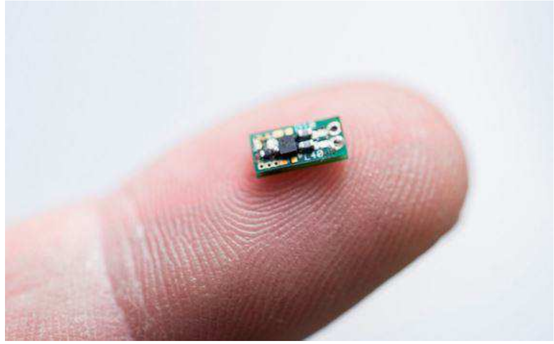


- Analog-to-digital converter
  - Digitizes electrical waveform
  - Samples at discrete points in time

- 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

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

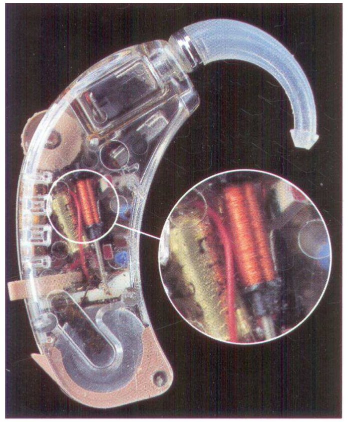
Easier to set up

What are Digital and Analog Signals? - Definition & Explanation

- https://www.youtube.com/watch?v=btgAUdbj85E

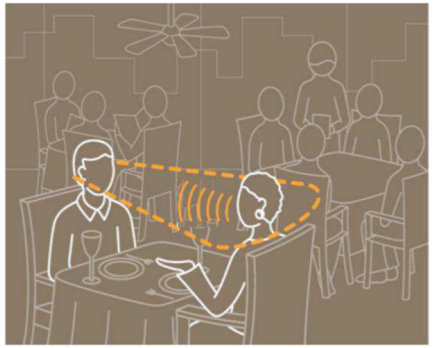
- https://study.com/academy/lesson/what-are-digital-and-analog-signals-definition-lesson-quiz.html

Telecoil



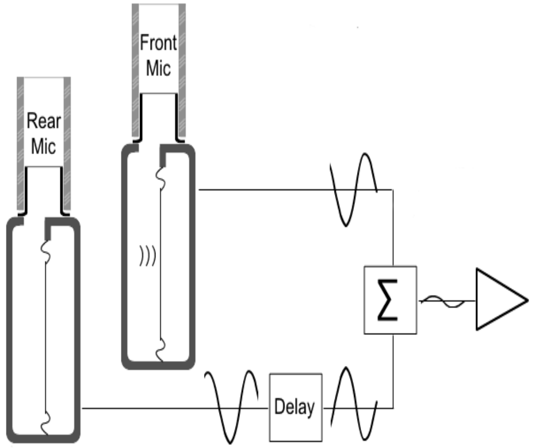
- Induction loop facilities found in facilities
  - E.g. public theatres, banks, places of worship
- Improves signal-to-noise ratio
- Prone to interference
- Another type of input transducer
- Uses electromagnetic energy present around telephone
- Bringing the phone close to the aid allows the magnetic 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



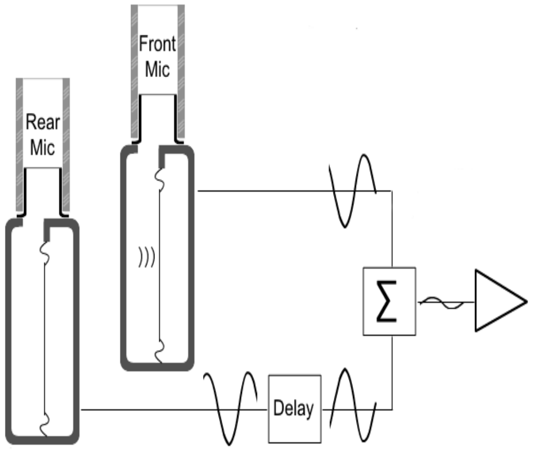
- Inability to hear speech in the presence of background noise is major reason hearing aids are rejected
- Directional microphones depend on noise being spatially separated from speech
- Reduce sound coming from behind rather than increasing sound from in front
- Improve signal-to-noise ratio (SNR)

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

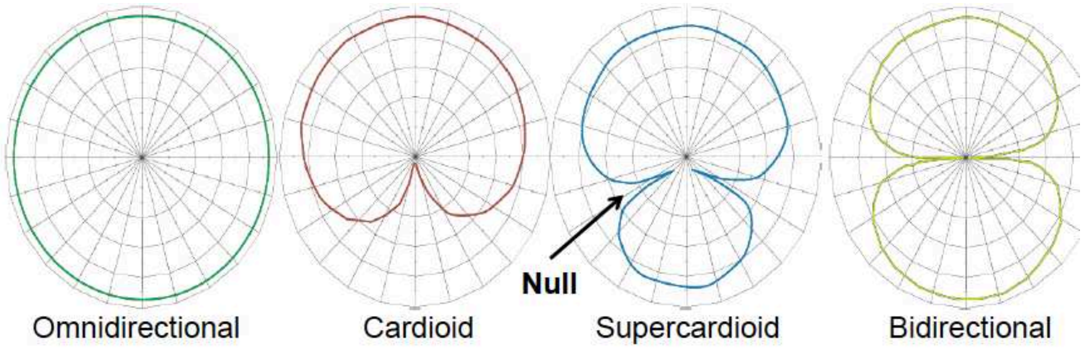
- A variety of spatial directional patterns can be achieved by changing the time delay applied to the signals
- Directivity index (DI)
  - Standard for measurement of effectiveness of directional microphones
- The greater the DI, the more effective the separation of signal and noise
- Current directional 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

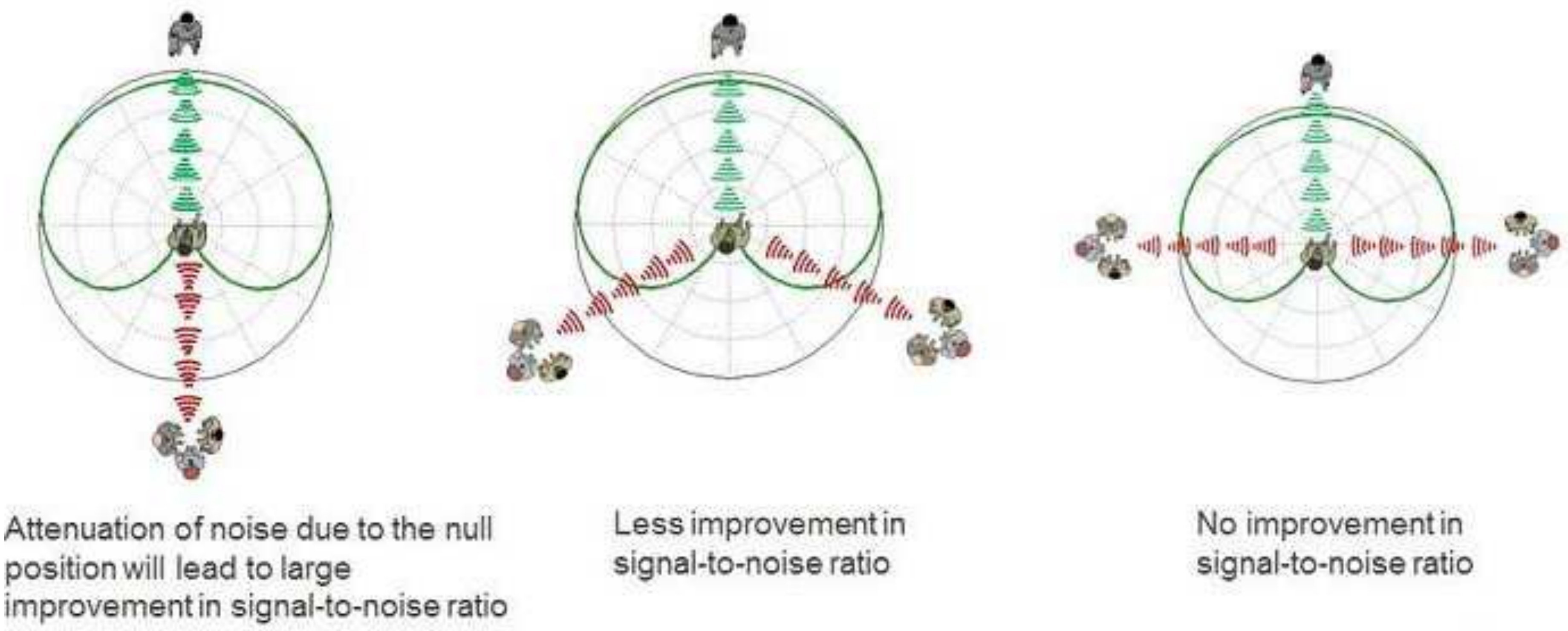
- Fixed directional
  - Nulls are always at the same angle of the pattern
  - Common directional polar plots:



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



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

Cord MT, SurrRK, Walden BE, DyrlundO. Relationship between laboratory measures of directional advantage and everyday success with directional microphone hearing aids. *J Am AcadAudiol.* 2004;15(5):353-64.

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

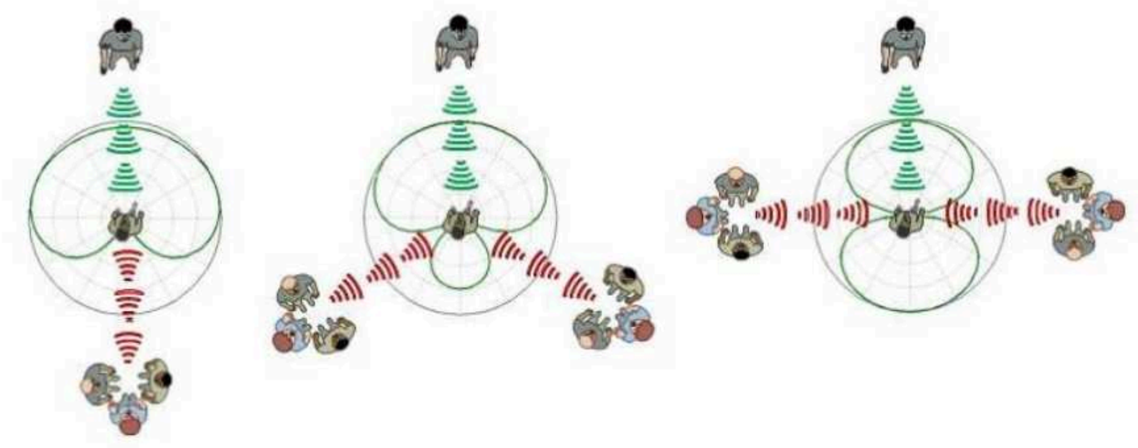
Adaptive directionality

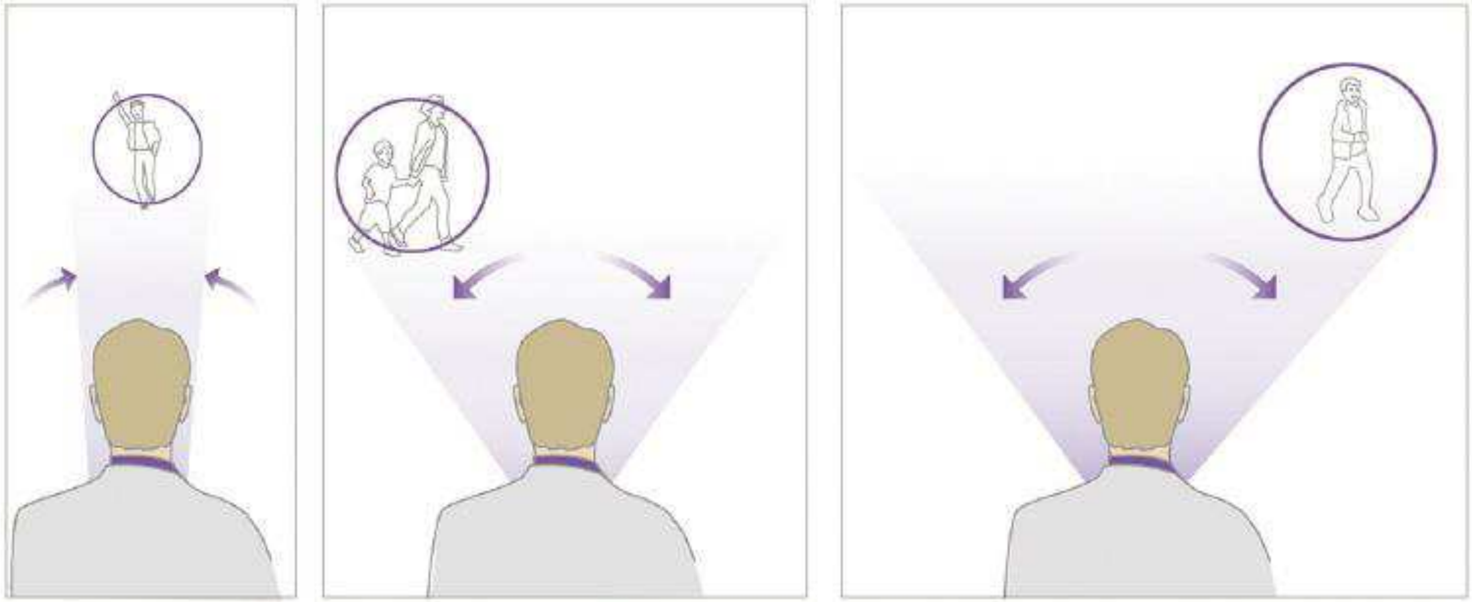
Benefit

- 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



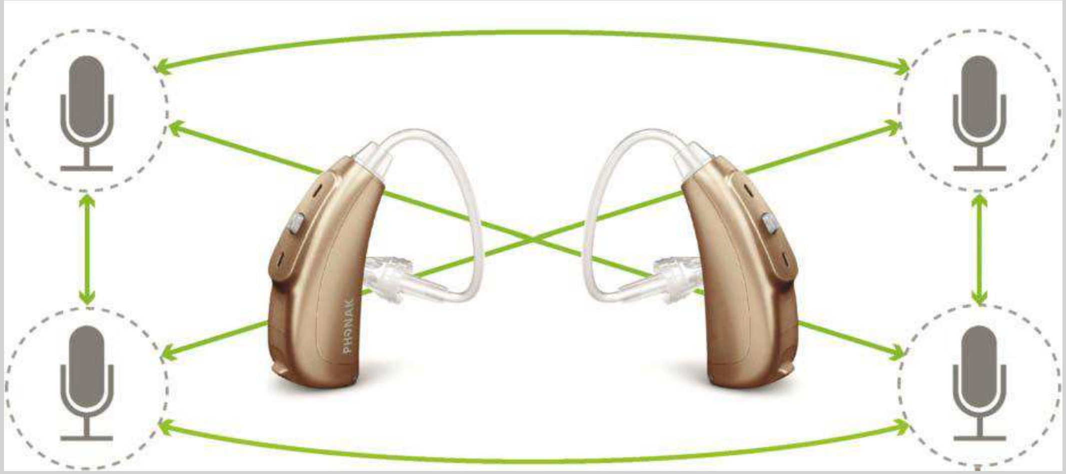


Beam forming

- 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:

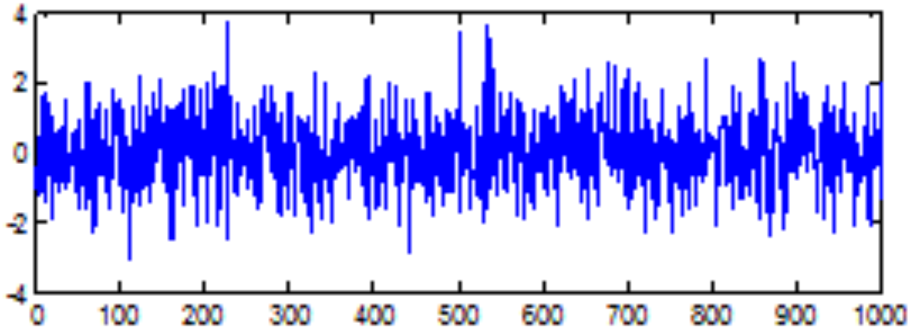
Noise suppression	Spectral subtraction

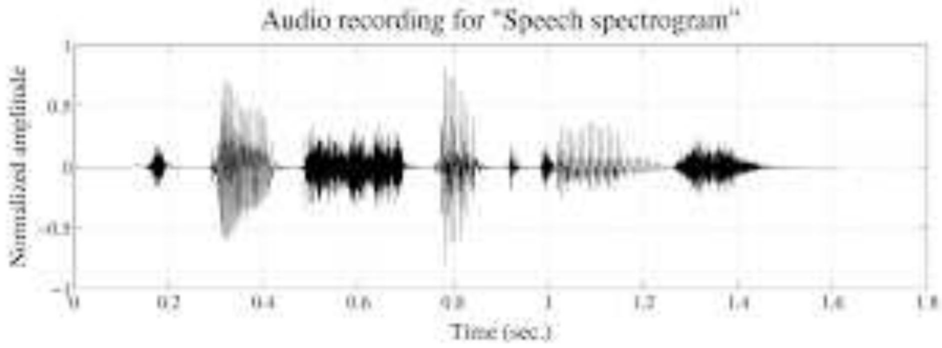
Noise Suppression

- 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

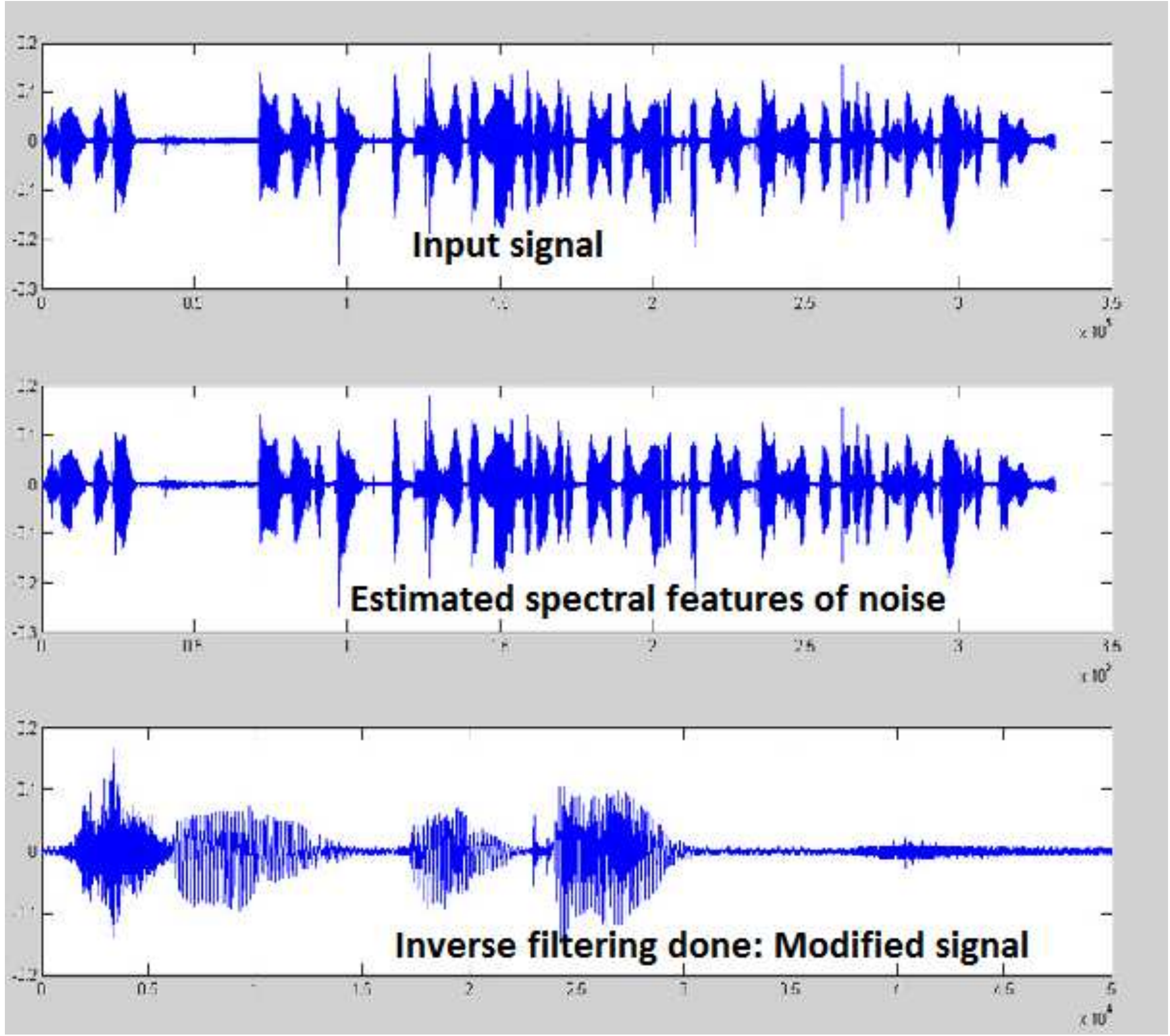
- Attempts to detect the presence of speech in different frequency bands based on the following assumptions
  - Voiced speech sounds have dominant low frequency fundamental frequency and harmonics
  - Identifies speech sounds by their fundamental frequencies (F0) and harmonics (F1, F2, F3)

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

Boymanset 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

Gain Reduction

Reducing the overall gain (old approach)

- Disadvantage: compromises audibility

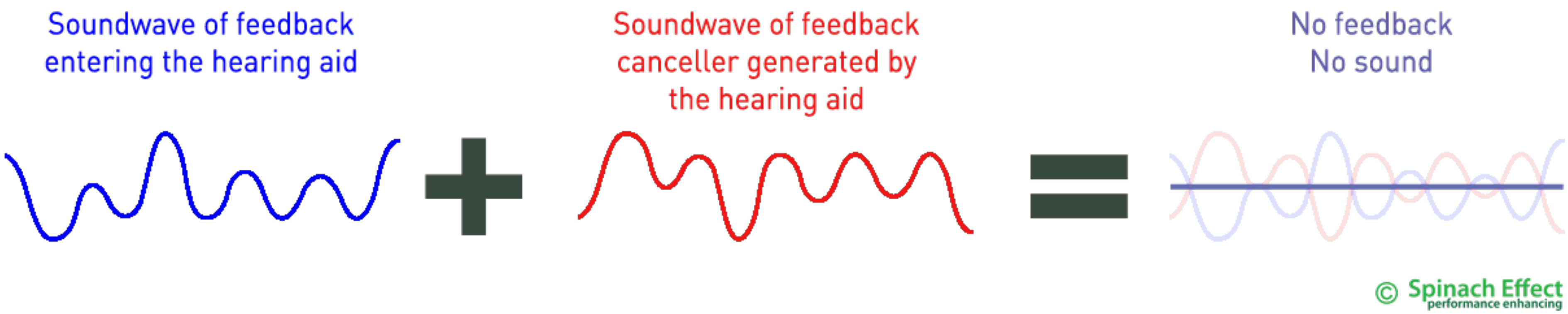
Identifies the frequencies at which the feedback is occurring and reduces the gain in this frequency region using a narrow band filter.

- 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



https://www.hearingpro.com.au/hearing-aids/features-of-modern-hearing-aids