Custom Hearing Aid Earshells and Earmolds

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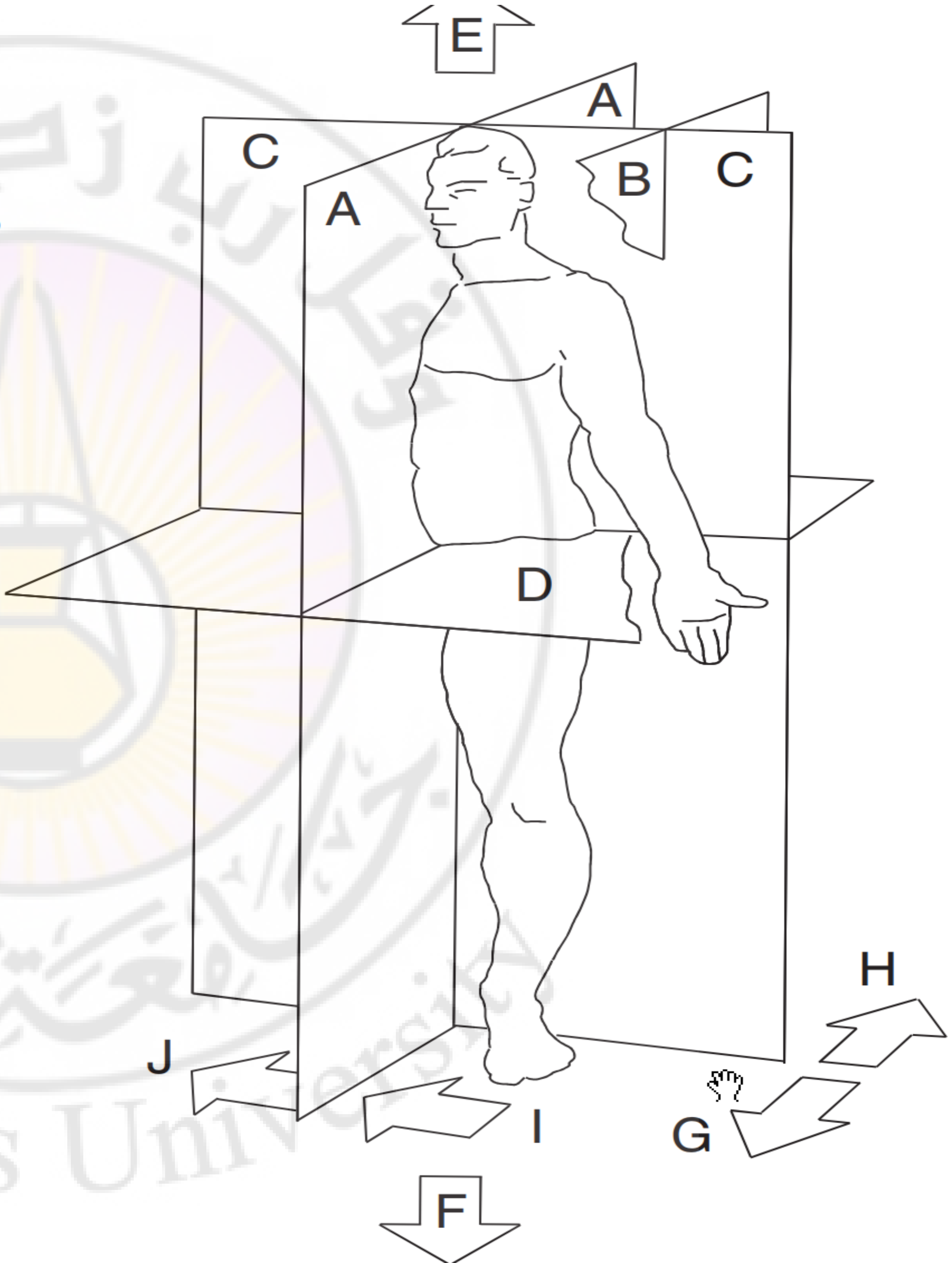
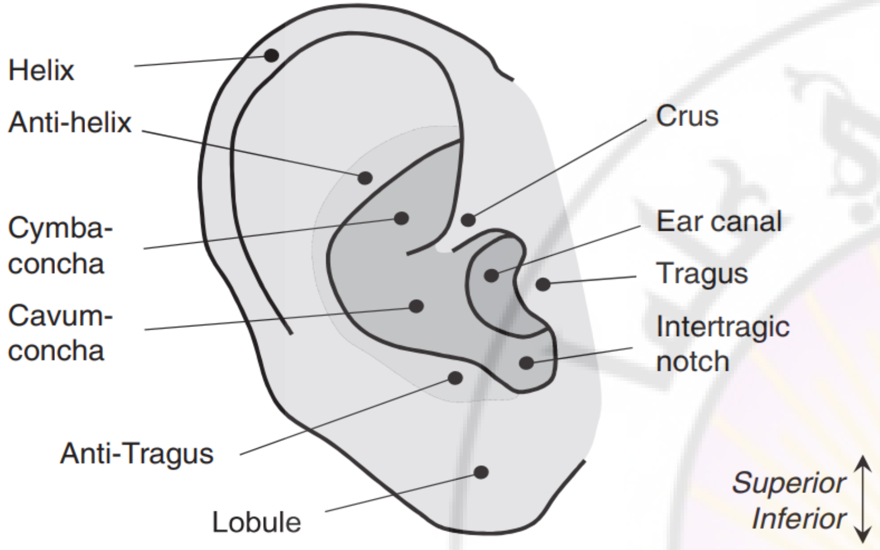
BODY PLANES

MEDIAN (A)
The median plane divides the body into left/right halves.
SAGITTAL (B)
The plane dividing the body into unequal left and right parts and parallel to the median plane. The terms median and lateral relate to this plane.
CORONAL, FRONTAL (C)
The plane dividing the body into equal/unequal front and back parts. The terms anterior/posterior relate to this plane.
TRANSVERSE, CROSS HORIZONTAL (D) The horizontal plane divides the body into upper (cranial) and lower (caudal) parts. Cross/transverse sections are perpendicular to the long axis of the body or other structures and may not be horizontal.

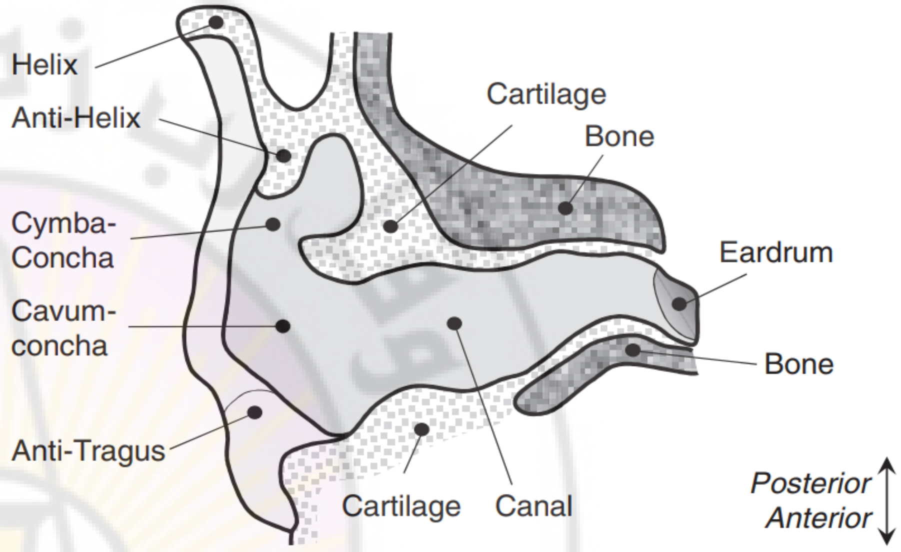
ANATOMICAL DIRECTIONS/POSITIONS

CRANIAL, SUPERIOR (E)
These terms refer to a structure being closer to the head or higher than another structure in the body.
CAUDIAL, INFERIOR (F)
These terms refer to a structure being closer to the feet or lower than another structure in the body.
ANTERIOR, VENTRAL (G)
These terms refer to a structure being more in back than another structure in the body.
POSTERIOR, DORSAL (H)
These terms refer to a structure being more in back than another structure in the body.
MEDIAL (I)
This term refers to a structure being closer to the median plane than another structure in the body.
LATERAL (J)
This term refers to a structure being further away from the median plane than another structure in the body.

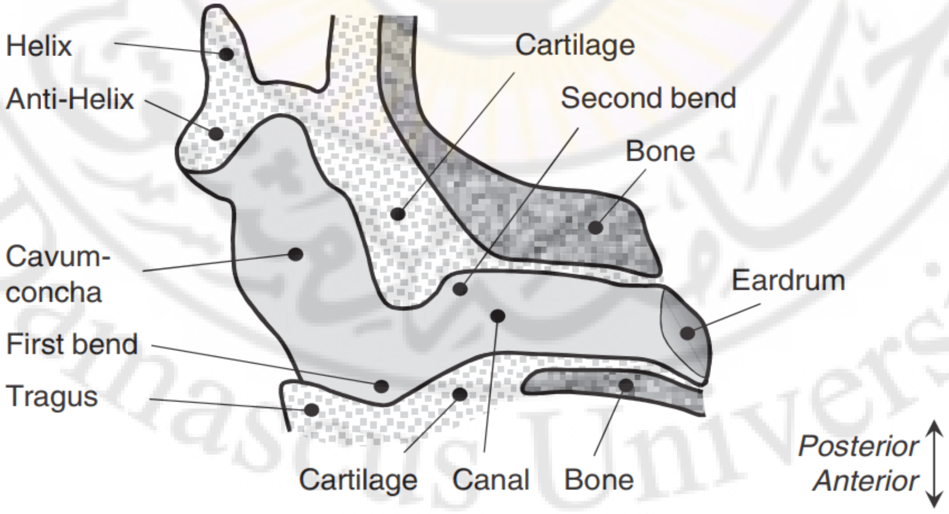




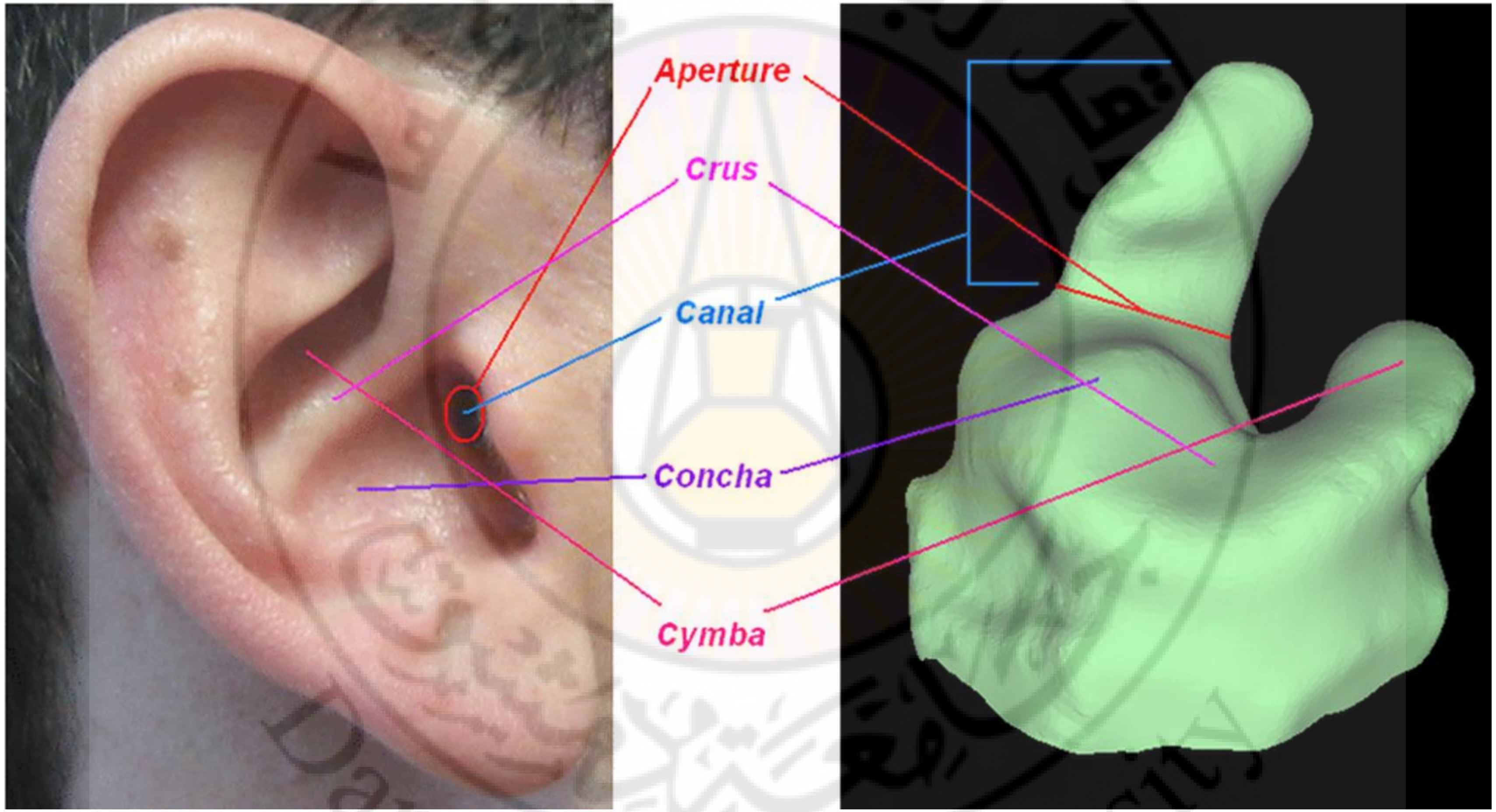
Sagittal View



Frontal View



Transverse View


Aperture
Crus
Canal
Concha
Cymba

What is the ear impression?

- A custom hearing aid requires an earpiece to couple the device with the ear canal, both acoustically and physically. For a BTE (behind-the-ear) hearing aid, the coupler is an earmold, for an in-the-ear hearing aid it is an earshell. To make a custom earmold or earshell, a model of the ear is required.
- The model, called an ear impression, is obtained by filling the external ear with soft putty plastic, commonly silicone.
- The silicone polymerizes and sets in the ear within minutes, becoming a true negative replica of the ear.

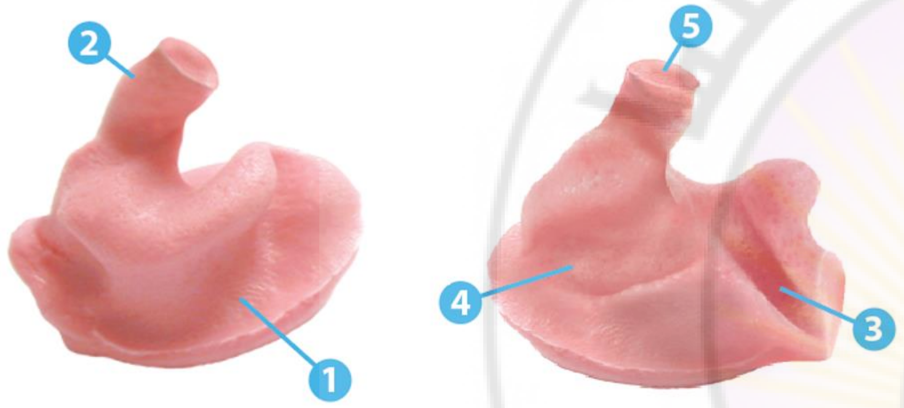
we can avoid to occurrence of feedback with correctly and precisely making ear impression and ear impression should have following characteristics for feedback free and being comfortable:

1. Have no areas such as gaps, weld marks, or air pockets

2. Illustrate the two bends of the auditory canal

3. Demonstrate the increase in the ear canal diameter resulting from the jaw's downward movement (via taking impression with open mouth).

Make a GREAT IMPRESSION



1. The CONCHA is round and full, no pockets or voids
2. The CANAL LENGTH is past the 2nd bend
3. The HELIX is complete through to the tip
4. The TRAGUS has been completely covered
5. The OTO-DAM has imprinted completely across the canal tip

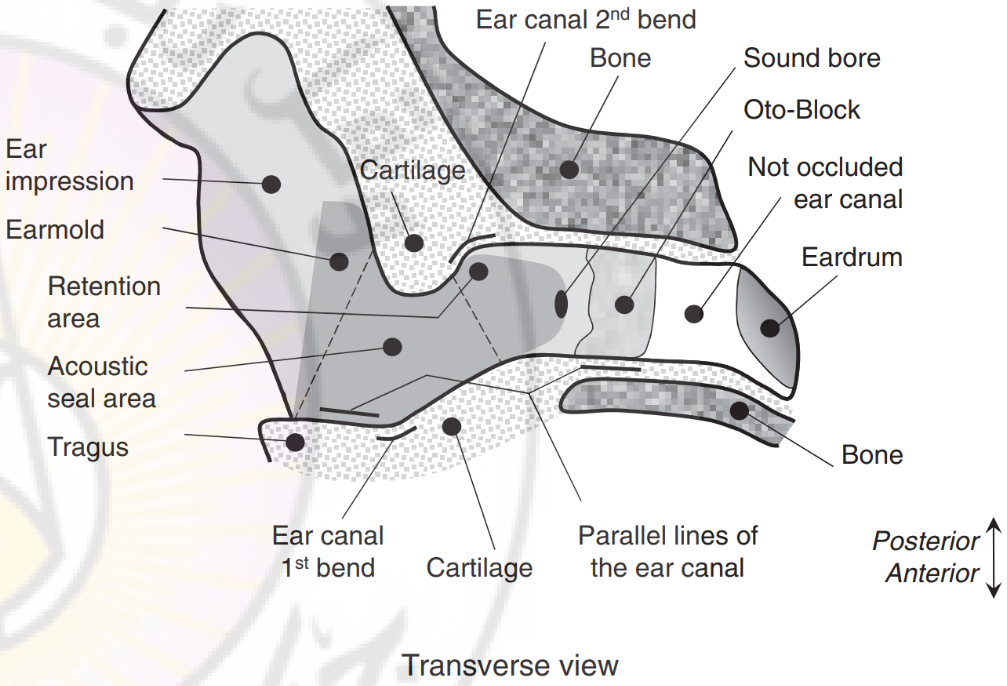
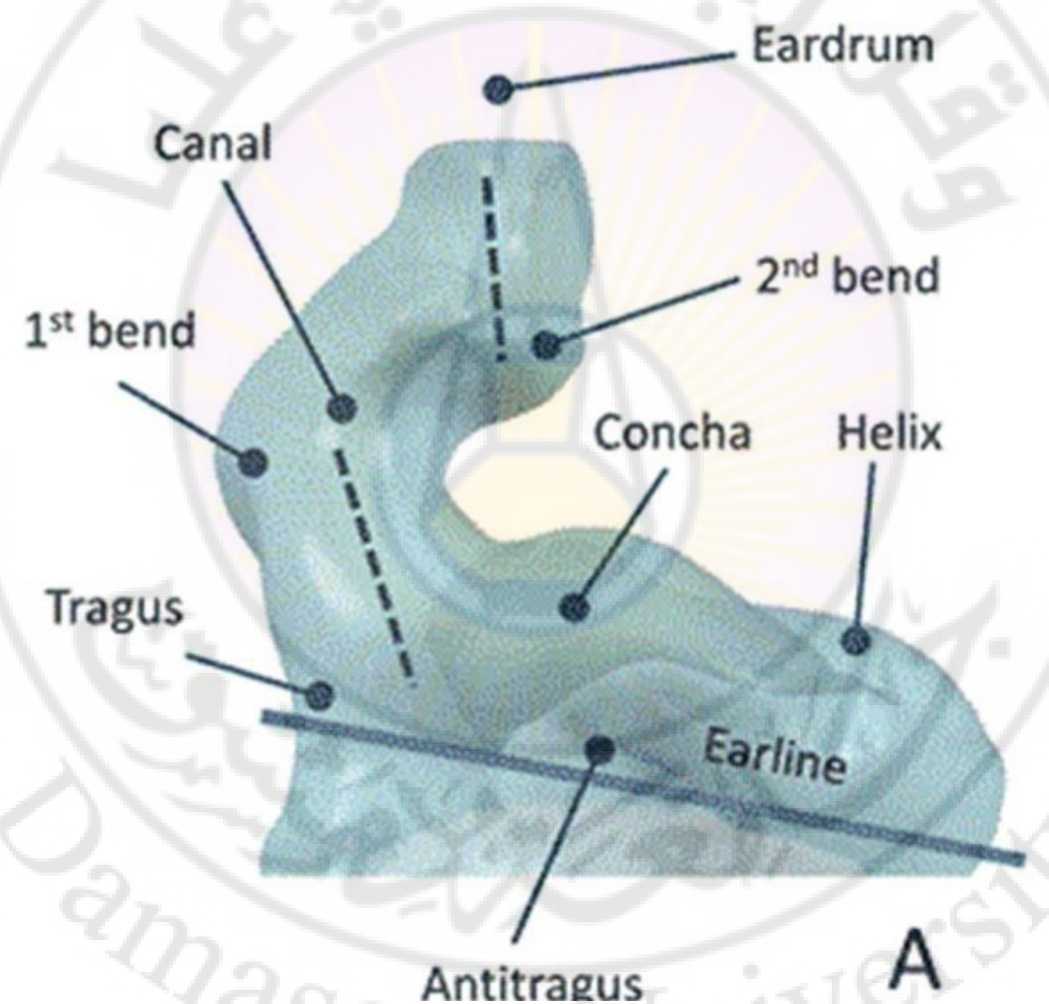
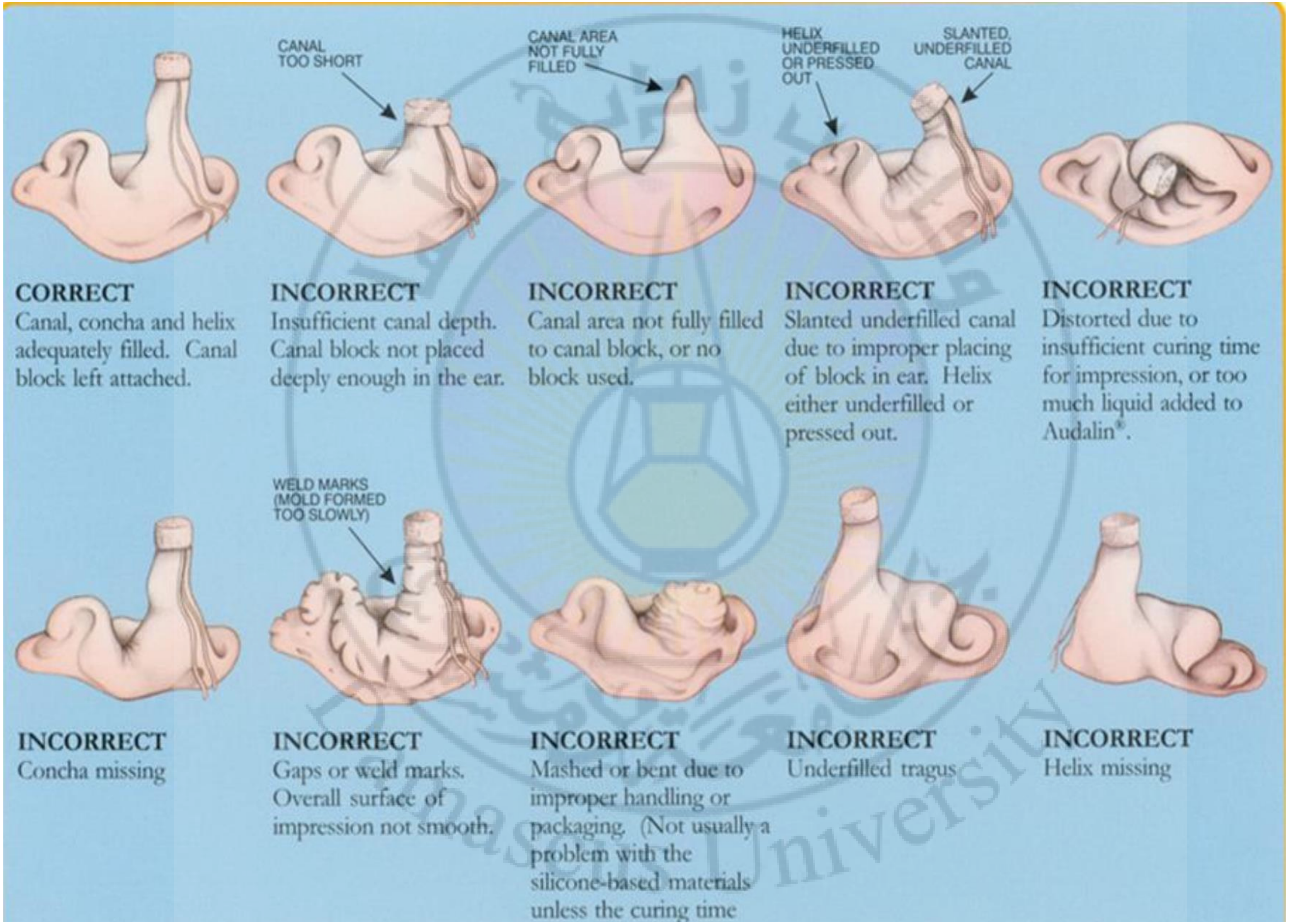


Figure 3–3. The transverse view of the human ear shown with an ear impression taken past the canal second bend and an earmold crafted from the impression.

Prefect impression




CORRECT
Canal, concha and helix adequately filled. Canal block left attached.
CANAL TOO SHORT
INCORRECT
Insufficient canal depth. Canal block not placed deeply enough in the ear.
CANAL AREA NOT FULLY FILLED
INCORRECT
Canal area not fully filled to canal block, or no block used.
HELIX UNDERFILLED OR PRESSED OUT
SLANTED, UNDERFILLED CANAL
INCORRECT
Slanted underfilled canal due to improper placing of block in ear. Helix either underfilled or pressed out.
INCORRECT
Distorted due to insufficient curing time for impression, or too much liquid added to Audalin®.
INCORRECT
Concha missing
WELD MARKS (MOLD FORMED TOO SLOWLY)
INCORRECT
Gaps or weld marks. Overall surface of impression not smooth.
INCORRECT
Mashed or bent due to improper handling or packaging. (Not usually a problem with the silicone-based materials unless the curing time
INCORRECT
Underfilled tragus
INCORRECT
Helix missing

preparation before tacking impression………

- wash your hand before and after taking impression or use gloves
- Clean the specula of the otoscope, ear-light tip, syringe, and so on, with an antiseptic solution
- Some equipment used in otoscopy is not meant for reuse, and therefore the cleaning of these pieces is inappropriate.
- the equipment for taking impression include:
  - Otoscope/ syringe/ pen-light/otoblock/ material for impression


3mm


LUXAMED


A
B


ENT
Diagnostic Products

Otoscopic examination…………………

- Check the ear canal for foreign objects, excessive wax, abnormal growths, or any condition that is out of the ordinary
- Determine whether or not wax removal is appropriate. If removal is contradicted or beyond your scope or comfort level, a referral to an otologist or otolaryngologist is recommended.
- Check the ear canal for size to determine how large an otoblock needs to be used.

Otoscopic examination…………………

- Check direction of the canal and bends. Look for the second anatomical bend. This will give you an idea for direction of push the block in the proper position.
- Encourage the patient to open and close their mouth to determine the effects of mandibular movements on the ear canal wall. Look at the anterior ear wall at the canal aperture. If any movement is apparent, taking an open-mouth impression is recommended.
- Check for landmarks in the canal. Look for any bumps and hollows and circle them later on the finished impression.
- After impression you should ensures that the earmold lab technician will not fill the voids assuming they were air pockets or other imperfections.

Tools and accessories for taking impression…………….

Oto-block

- the function of an oto-block is to prevent impression material from reaching the eardrum. Oto-blocks are made of cotton or foam and come in a variety of sizes.
- Cotton tends to be more comfortable for deeper canal impressions. However, in a draining ear or a postsurgical ear canal, cotton fibers may adhere to the canal wall and provide a host for bacterial infection.
- Otoblocks that are too small may get pushed down the canal by the impression material and allow the material to flow past the block. Blocks that are too large may not go far enough into the ear canal. If you are dealing with a postsurgical ear, use a larger otoblock than normal, or several otoblocks.

Mouth prop:

A mouth prop is a white Styrofoam block. It is used to take ear impressions with the client's mouth wide open. The prop must be inserted into the patient's mouth prior to syringing the impression material and must remain there until the impression hardens.

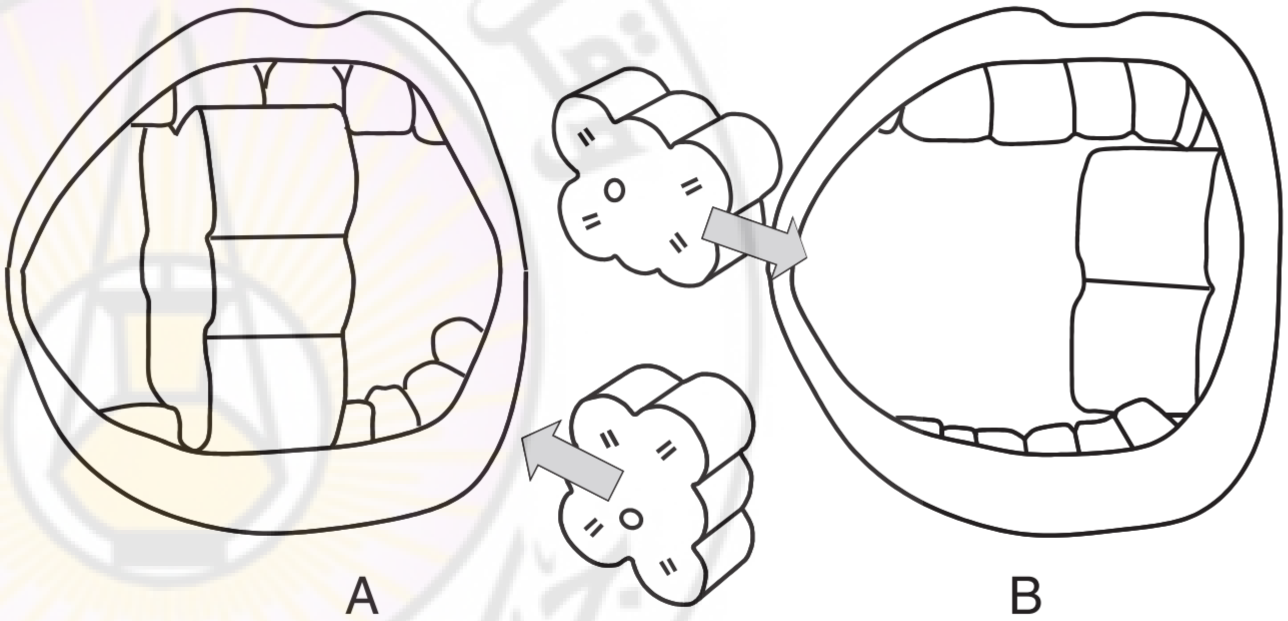


Figure 3–7. Options in placing a mouth prop for taking an open-mouth impression.

Tools and accessories for taking impression…………….

- Marking card:

When an impression for a hearing aid with a directional microphone is taken, use a marking card to imprint the horizontal plane of the ear in the hardening silicone. The mark in the impression will allow the earmold lab technician to properly align the microphones on the hearing aid faceplate.

- Impression syringes:

Syringes are most commonly used for impression taking. There are over 10 different models of syringes on the market. The most commonly recommended are syringes that are easy to load, self-cleaning, offer a good grip, and require minimum pressure to operate.

Tools and accessories for taking impression…………..

Penlight: for viewing the otoblocks in ear canal and making a pressure them in the appropriate direction.

Material for impression………

- There are different types of materials for taking impression.
- the right selection will increase the chance of successful fitting.
- The most commonly used impression materials are called silicones.
- The material is mixed by blending two putties (A and B) at 1:1 ratio.

Material for impression………

- Other impression materials include condensation-cure silicones (25:1 mixing ratio).

Material for impression………

Material for impression is very different in some characteristics such as: viscosity, contraction ratio, stress relaxation and shore value.

What is the viscosity?

- Viscous means having a thick or sticky consistency.
- Viscosity of an impression material is defined as a measure of the material consistency before polymerization.
- Hand-mixed silicones are medium or higher viscosity.
- A medium or higher viscosity silicone is better to use on patients having hair in their ears.

What is the shore value?

❖The shore value of an impression material refers to the finished impression hardness 1 hour after the impression was made.

❖ There is no apparent relationship between the viscosity of a silicone and its shore value.

❖ the value is very important for building an earmold.

What is the Contraction ratio?

- This parameter relates to the material shrinkage with time. Silicone impression materials typically shrink 0.1% to 0.7% in 7 days.
- Shrinkage of less than 1% is negligible in terms of the resulting dimensional accuracy and earmold fitting

What is the stress relaxation?

- Stress relaxation describes the viscoelastic nature of finished ear impression.
- An impression must not change shape as a result of its removal from the patient’s ear, during shipping or in-lab processing.
- Impressions made from silicone will recover from stress such as removal from the ear in 99%, which is quite satisfactory.

how to take an impression?

Procedure of Taking Impression

- Step 1. Gather necessary materials
- Step 2. Explanation of procedure
- Step 3. Infection control

Procedure of Taking Impression

- Step 4. Otoscopic examination
- Step 5. Placing the otoblock or ear dam
- Step 6. Mixing the impression material

Procedure of Taking Impression

Step 7. Injecting the material: You should follow this order:

1. Pull the pinna up and back gently to straighten out the ear canal.
2. Place the tip of the injection gun or syringe into the aperture of the ear canal.
3. Squeeze the syringe or injection gun smoothly. Never inject the material with force.
4. Once the entire concha bowl has been filled with EI material, let go of the pinna with your other hand.
5. Keep the tip of the injection gun or syringe in the material until it flows back around it.
6. Make sure you have filled the entire concha and helix area with material before removing the tip from the ear.
7. Allow the material to sit 5 to 10 minutes

Procedure of Taking Impression

- Step 8. Removing the EI
  - Gently pull the pinna up and out to loosen the seal.
  - Pull the helix portion (top part) of the EI out slightly away from the ear canal.
  - If the patient is not in too much discomfort, carefully and slowly continue to gently pull the EI out of the ear canal.
- Step 9. Inspect the ear canal and the EI
- Step 10 . Send the information to the manufacturer

Why should we take an impression with high quality?

- Ear mold impression procedure is an mildly invasive and it is better to has done one time.
- Poor quality ear impression resulted in poor quality of fitting hearing aid and reject of it.

Earmolds and Earshells

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2023 May

Ear-shell and ear-mold

- earmolds are designed to hold (or couple) the hearing aid to the ear.
- there are different types of earmolds for various types of hearing loss.
- generally, All earmold coupling systems have three essential components: the tubing, the vent, and the earmold itself.
- Most earmolds are custom-made in a laboratory.
- earmolds play a significant part in shaping the amplified sound before it reaches the tympanic membrane

Main landmarks in earmolds

Sometimes, there are some problems in the earmolds and you must communicate with lab. So, you must have a common terminology and identify the main landmarks of the earmolds.

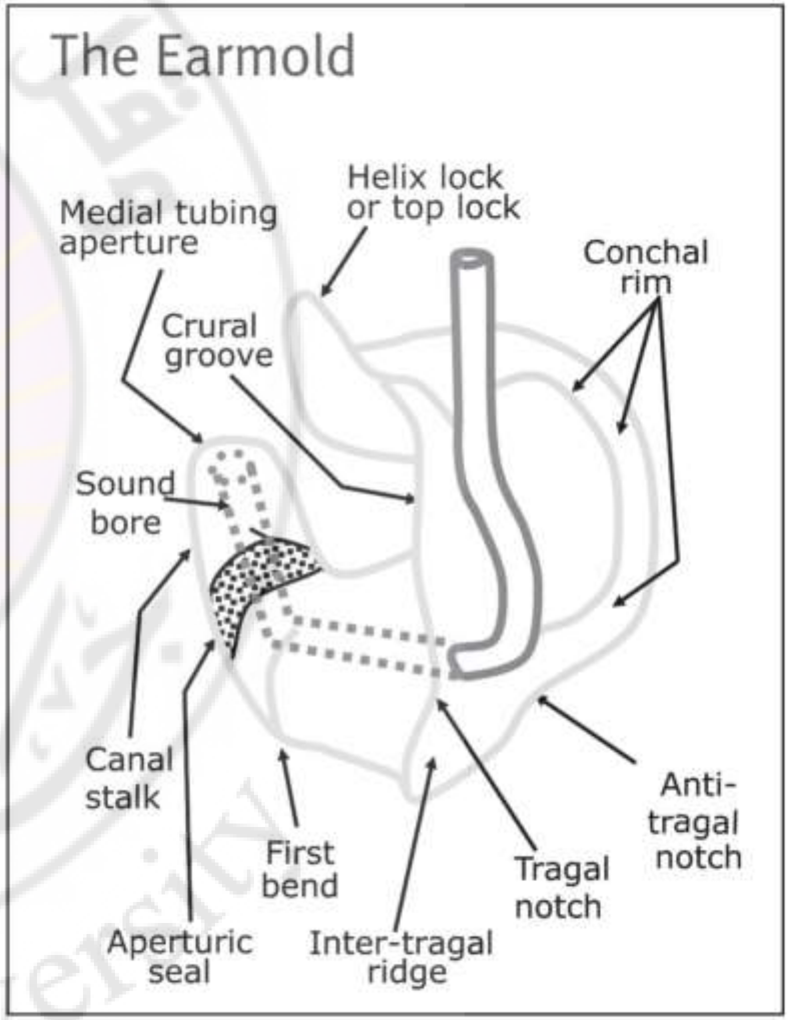


Figure 7–8. The key landmarks and common terminology of an earmold. Reprinted with permission from Dillon (2001) "Hearing Aids."

Earmold style:

- There are different types of earmold styles.
- As a general rule, the greater the hearing loss the more material is used to fill the ear canal.
- The names of different styles of earmolds in different companies are different.

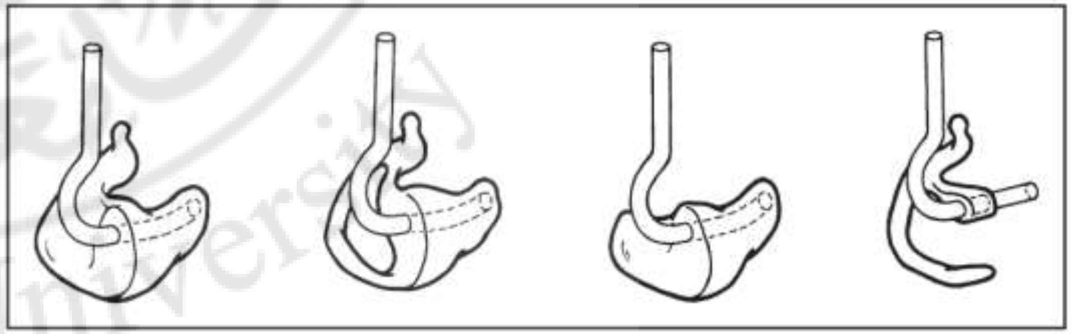


Figure 7–7. The four most common earmold styles from left to right; shell, skeleton, canal, and free-field/CROS. Reprinted with permission from Unitron. All rights reserved.

Earmold style:

- Before making a decision on the earmold style and material, several considerations must be taken.

- The evaluation should include: severity and type of the patient’s hearing loss, anatomical properties of the ear, patient’s manual skills, allergic conditions, personal preferences, and difficulty with any previous hearing instrument fitting

Earmold style:

- Earmolds are made in two basic styles: concha and canal.
- Each style has several sub-styles that differ mainly in cosmetics.

Concha earmolds:

- Concha earmolds occupy the ear concha and canal, and can be made in the standard, shell, skeleton, or nonoccluding configurations.
- The standard earmold fits in the entire concha.
- The shell earmold has the bowl shelled out for an enhanced cosmetic appeal.
- In the skeleton earmold the concha is reduced even further to a ring, or 1/2-ring

Canal style earmolds

- Canal style earmolds include the half-shell, canal, and canal-lock configurations.
- The half-shell earmold occupies the cavum-concha.
- The canal earmold fits in the canal aperture.
- The canal-lock earmold possesses a small plastic tail (lock) extended to the ear concha.

Different styles of earmolds

The receiver mold (confusingly called a standard or regular mold) is the only one that can be used for a body aid: a button receiver clips firmly into the ring on the surface of the mold.

The top seven earmolds of this figure are occluding earmolds and can be vented. The remaining six molds and the open dome can never be completely occluding, because the canal portion of the mold does not fill the entire cross-section of the ear canal at any point along its length

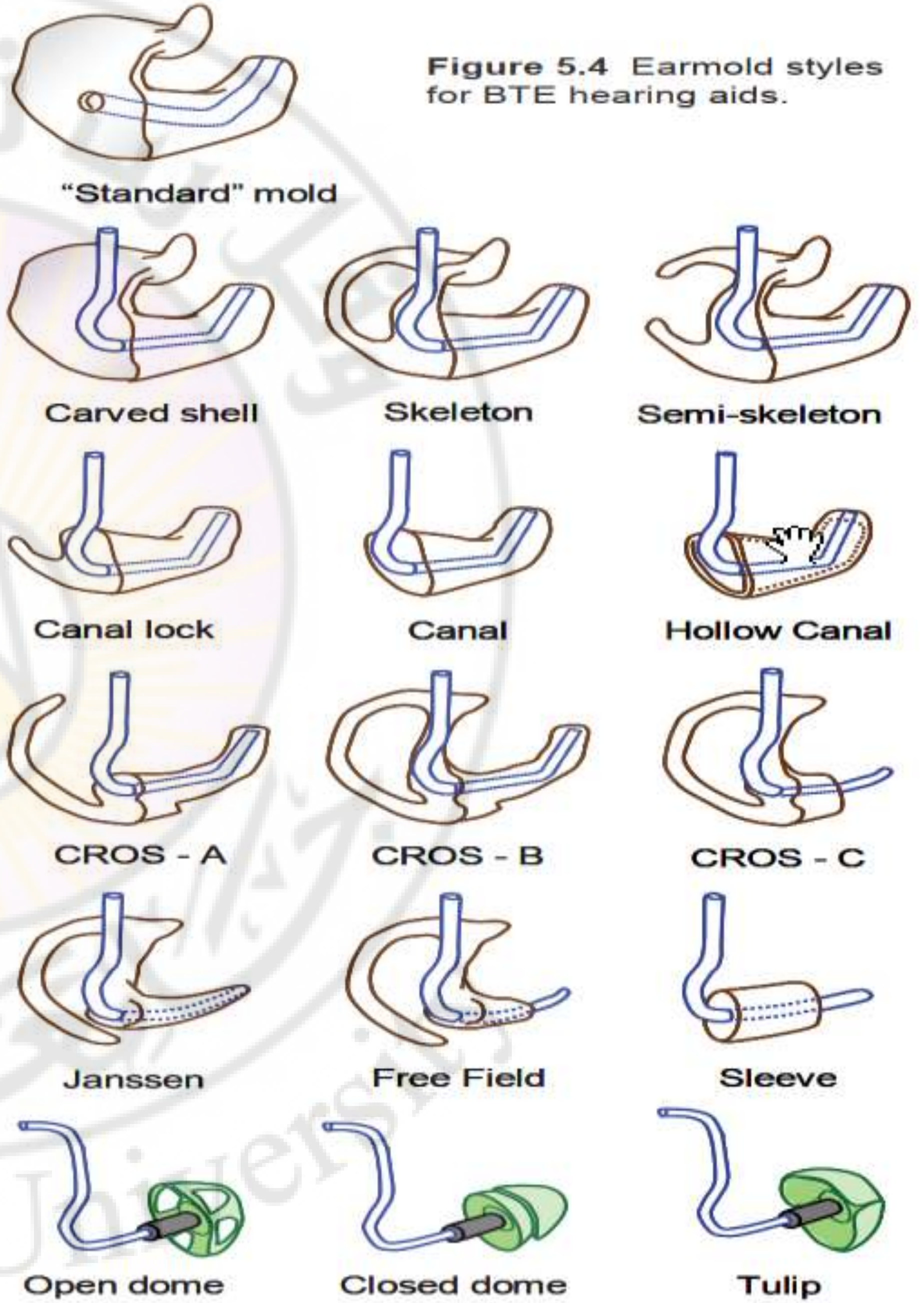


Figure 5.4 Earmold styles for BTE hearing aids.

Earmold material options

Another important consideration is the type of material used to make the earmold.

- In general, softer materials, such as silicon and vinyl, are used for hearing losses greater than 75 dB HL.
- The table outlines the three major earmold materials, along with their performance advantages.

Name	Characteristics	Advantages
Acrylic	Hard	Hard, durable Hypoallergenic Easy to modify in clinic More easily inserted Recommended for mild to severe hearing losses
Silicone	Semi-soft	Firm, semi-flexible Designed to provide added comfort over hard material May expand to reduce leaks Recommended for mild to profound hearing losses Recommended for pediatric patients Available in specialty colors and glitter options
Soft Silicone	Very-soft	Soft, flexible material with superior sealing properties Flexes to accommodate TMJ movement Recommended for severe to profound hearing losses Good choice for sports Recommended for pediatric patients

Earmold Characteristics

Regardless of the earmold style or material, there are three requirements of a properly fitted earmold:

1) Acoustic seal. The earmold needs to direct sound toward the eardrum without acoustic feedback.

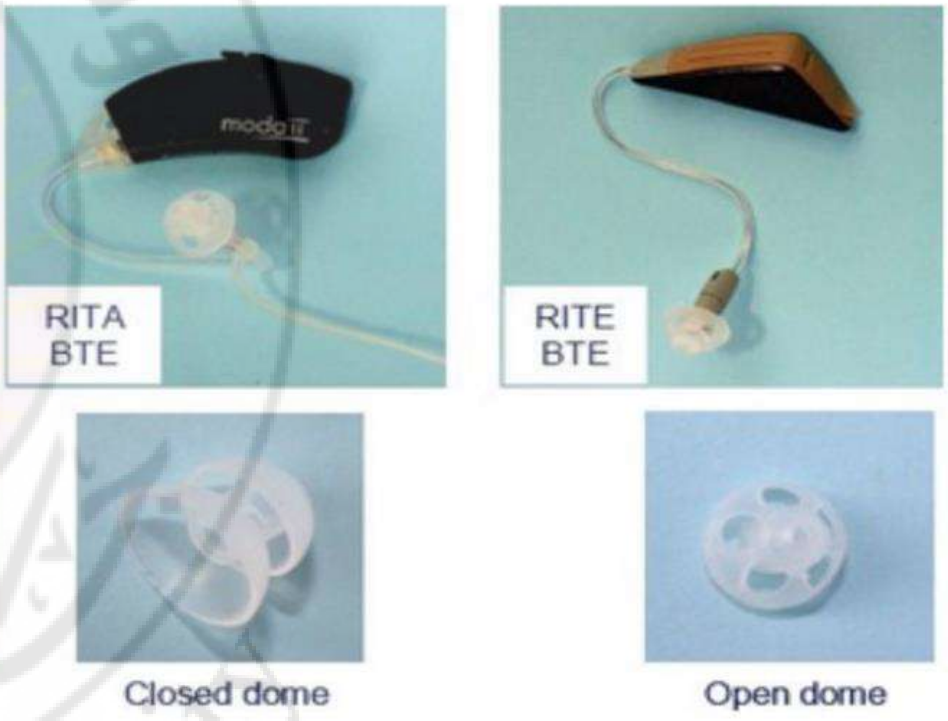
2) Comfort. The earmold should fit as comfortably as possible without causing irritation to the external ear.

3) Aesthetic appearance.

Pre-molded fitting or domes:

- There are essentially two dome styles, the open dome with holes in the flange which is designed to leave the canal as open as possible, and the closed dome with no holes which is designed to seal the canal as completely as possible.

- the same designs are used for RITE hearing aids, except that the central portion of the dome contains the receiver, the thin tube from the hearing aid is replaced by an equally thin electrical connection to the receiver.




Closed Domes
Small
Medium
Large
Open Domes
Small
Medium
Large
Power Domes
9 mm
10 mm
11 mm
Tulip Dome

Advantages of pre-molded fittings

- Invisible and good appearance
- Facilitate same-day fitting as no ear-impression needed

Earmold, Earshell and Canal Fitting Acoustics

Acoustical characteristics of earmolds can affect on:

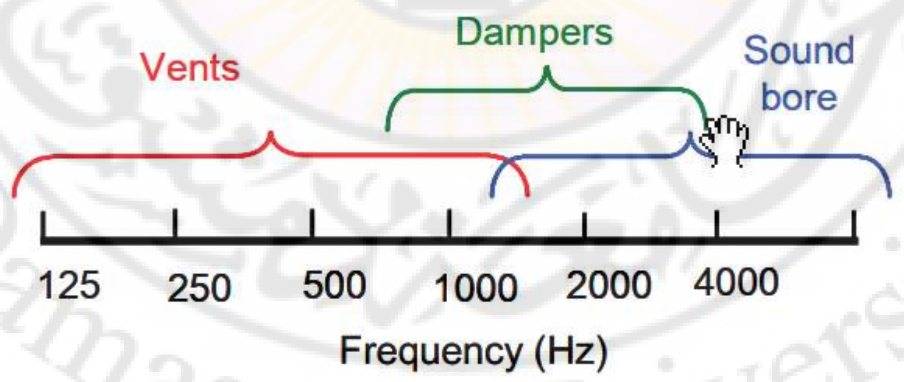
- the shape of the gain-frequency response of the aid when it is mounted in the ear, the
- The quality of the patient’s voice
- the likelihood of feedback oscillation.

Earmold, Earshell and Canal Fitting Acoustics

- There are three acoustic aspects of the coupling system:

  the sound bore, the damping, and the venting.

- These primarily affect the frequency response in different frequency regions



the sound bore, the damping, and the venting

- Sound bore dimensions affect only the mid and high frequency response (above 1 kHz for BTE aids and above 5 kHz for ITE/ITC/CIC aids).
- Damping mainly affects the response shape in the mid-frequency region (from 800 Hz to 2500 Hz for BTE aids, and from 1500 Hz to 3500 Hz for ITE/ITC/CIC aids).
- Venting mainly affects the low-frequency response, from 0 Hz up to approximately 1 kHz, although if the vent is large enough, such as with an open-canal fitting, it affects the entire frequency range.

Venting:

- A vent is a hole in the earmold or custom hearing aid that allows communication (air and sound) from the residual ear canal space to the outside world.
- All vents are designed to provide some reduction of amplified low frequency sound that is, allow low frequency sounds to leak out of the ear.
- That is, some of the low frequency sounds amplified by the hearing aid will not be transmitted through the middle ear into the inner ear.
- Venting also allows for some pressure relief. This problem is especially apparent when patients talk or chew.

Parallel vs. Diagonal Vents

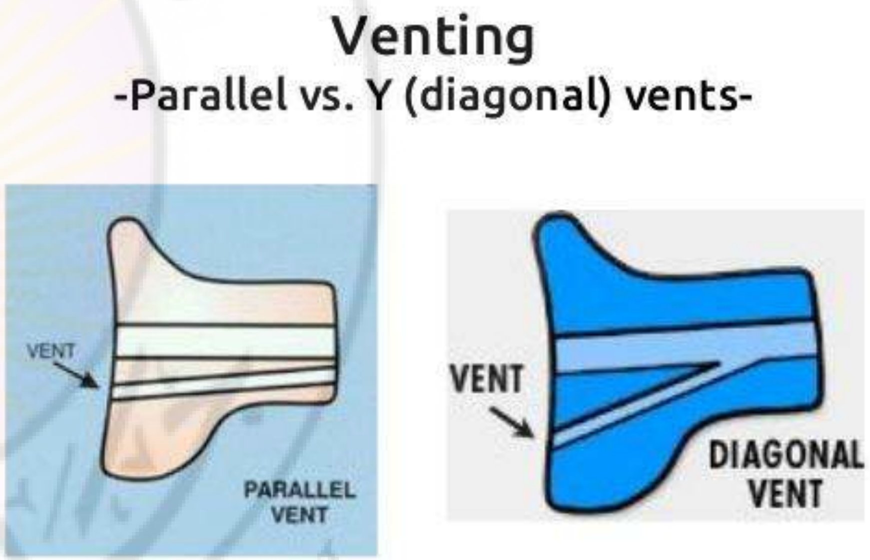
- There are two major types of vents found in a hearing aid shell or earmold: parallel, and diagonal.
- The most commonly used vent is the parallel type; however, there are instances when a professional may have to substitute a different style vent.
- diagonal vents reduce low frequency energy more than parallel vents with the same internal diameter.

Parallel vs. Diagonal Vent

Cross section of a Y-vent (or diagonal vent) in a BTE earmold

Venting:

Parallel vents are generally used, because not only do diagonal vents decrease low frequency gain like parallel vents, diagonal vents also decrease mid frequency (630-1600 Hz) gain by up to 10 dB.



Advantages to venting

There are at least 5 advantages to venting an earmold:

1) to reduce unnecessary low frequency gain and output from the ear canal/eardrum,

2) to allow unamplified sound to enter the ear canal,

3) to reduce the occlusion effect,

4) to relieve the feeling of pressure in the ear, and,

5) to reduce moisture build-up in the ear canal.

Disadvantages to venting

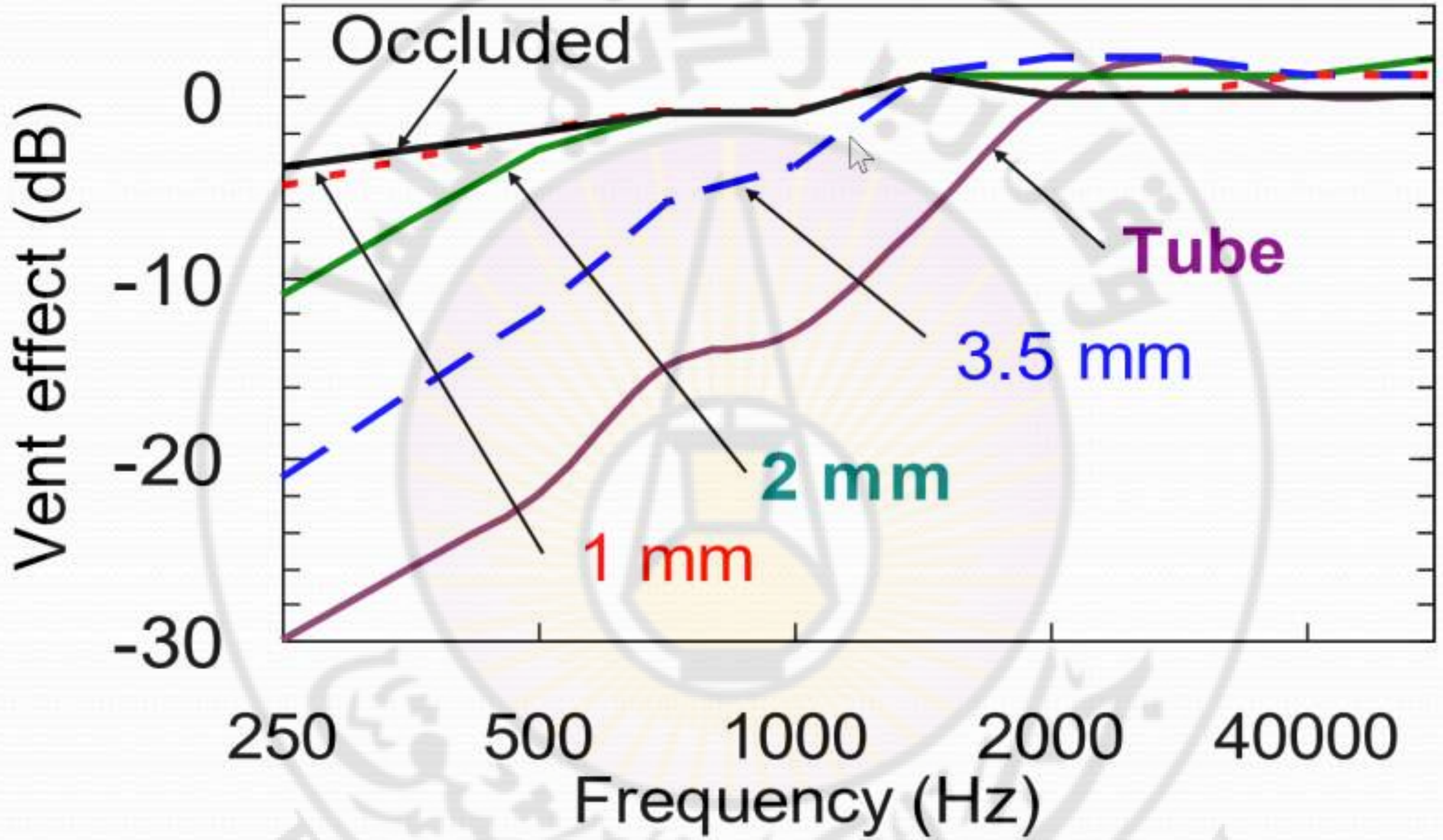
- One disadvantage to venting is that the volume of the vent combines with the ear canal cavity to produce a Helmholtz resonator (increased gain in the 250-1000 Hz region), sometimes resulting in an echo or barrel effect when the patient speaks.

- Another disadvantage of venting is, of course, feedback. As vent size increases, the amount of acoustic leakage increases, and therefore the probability of feedback increases.

The Effect of Parallel Vent Diameter/Canal Length

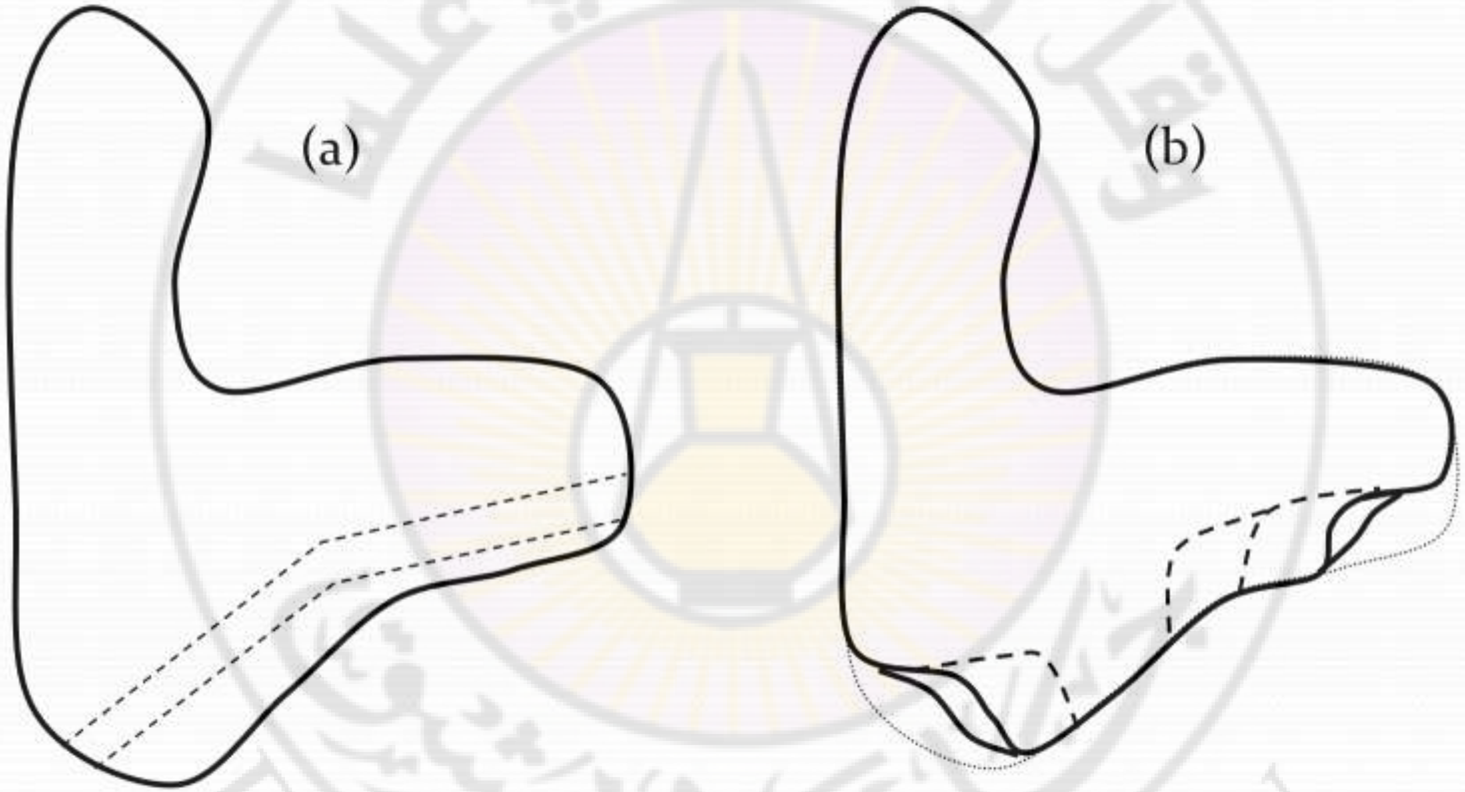
- The rule of thumb is that as the parallel vent size increases (1 to 3mm), low frequency gain decreases (up to 30dB) at 500 Hz and below.

- As canal length increases (6 to 22 mm), the effect of venting is reduced. Thus, low frequency gain decreases are less (15 dB versus 30 dB) with long canals



Effect of different sized vents on the frequency response of amplified sound, relative to the response with a tightly fitting earmold or earshell (Dillon, 1985).

Shortening the vent



An unmodified vent (a) and a shortened vent (b). The dashed lines in (a) indicate the position of the vent. The dashed lines in (b) indicate potential further stages of shortening, and the dotted line indicates the original profile.

So, what does this mean to you?

- If you want to decrease the low frequencies, you should order a short canal (6mm) and a large vent (3mm).

- Pressure equalization vents (.06-.8mm) on the other hand, no matter what the canal length is, generally only decrease gain up to 6 dB below 250 Hz.

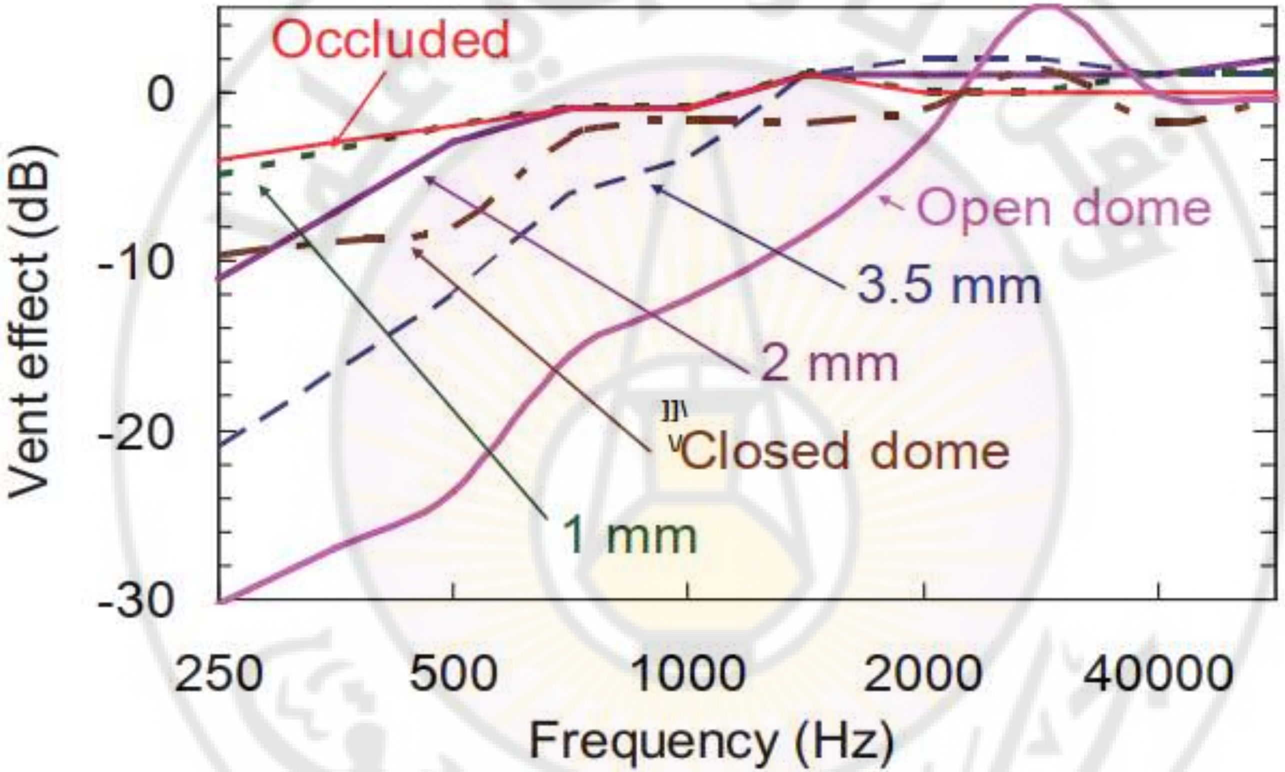


Figure 5.11 Effect of different sized vents on the frequency response of amplified sound, relative to the response with a tightly fitting earmold or earshell.[430, 431, 1355]

A Venting Selection Guide

- A guide for selecting the appropriate venting based on low frequency hearing loss.
- In general, if the hearing loss below 1000 Hz is 25 dB or less, an open mold (large vent 3.0 mm) is recommended.
- If the hearing loss is 30-45 dB in the low frequencies, an acoustic modifier or medium vent (2.0mm) is recommended.
- If the hearing loss is 50-60 dB in the low frequencies, a small vent (1.0 mm) or pressure vent (.06-.88 mm) is recommended.
- When the loss exceeds 60 dB, we usually do not put in a pressure vent, unless the patient is occluded or their voice bothers them.

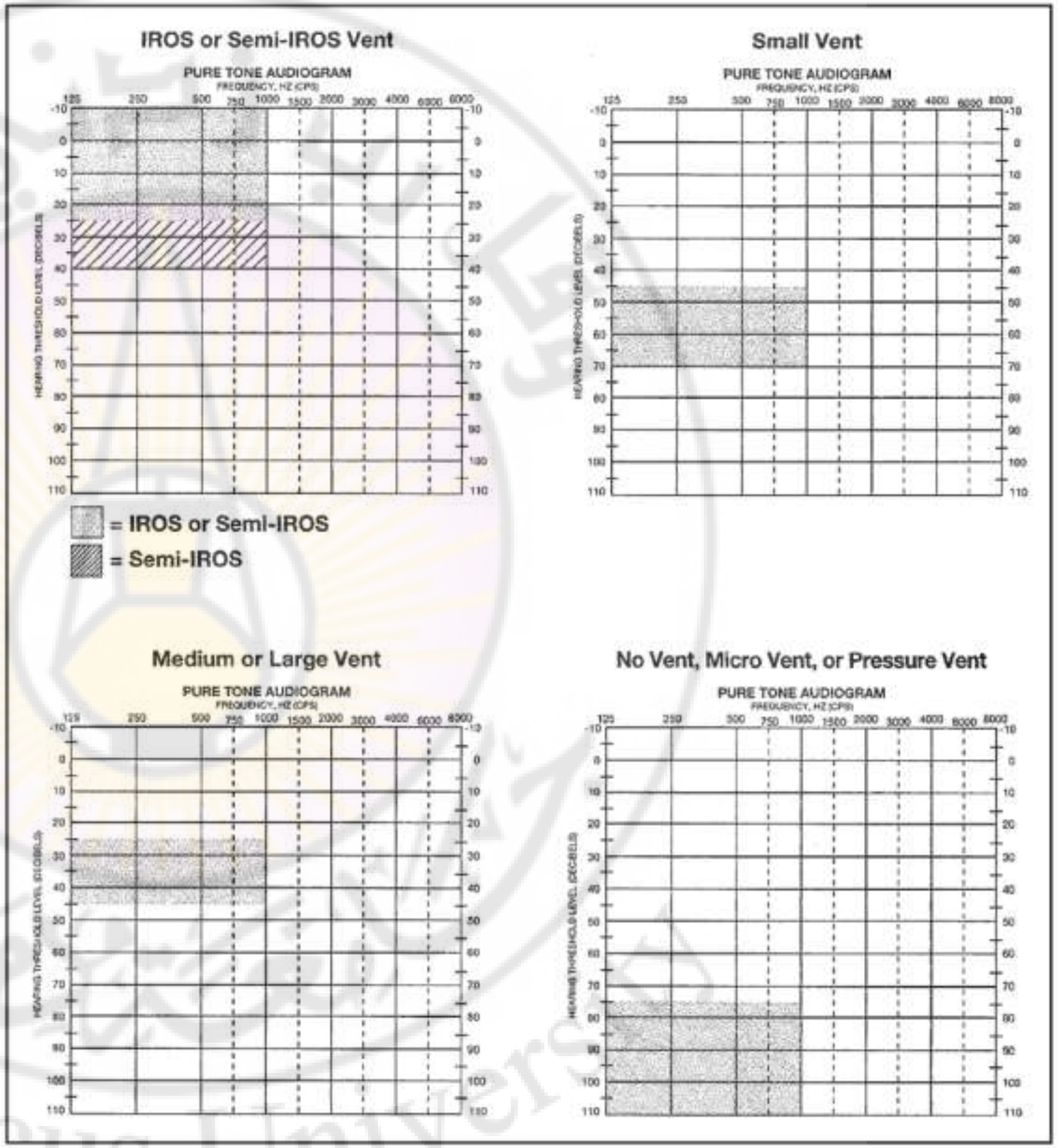
Venting:

Recommended vent diameters (in mm) based on the hearing loss at 500 Hz (in dB HL). For an open-ear fitting, the hearing loss at 1000 Hz should not exceed 60-70 dB HL so that sufficient gain may be provided

Degree of hearing loss at 500 Hz (in dBHL)	Recommended vent diameter (solid mold)
<20 dB	open
20-29 dB	3-4 mm
30-39 dB	2-3 mm
40-49 dB	1-2 mm
50-60 dB	0.5-1 mm

Another guideline for selecting an appropriate vent

Generally, large would constitute a vent larger than 2 mm, medium 1 to 2 mm, and small is less than 1 mm. Because it's not an exact science, and you will need to balance vent size based on patient comfort and adequate gain before feedback, it is better to order a select-a-vent (SAV) whenever possible. With SAV you can change the vent size in your office depending on the specific needs of your patient.



Tubing………………..

- Tubing can affect on the frequency response and gain of hearing aids.
- Tubing length and internal diameter can have a pronounced effect on the frequency response of the hearing aid.
- Tubing diameters are standardized according to the internal diameter.
- The most common tubing is size 13
- When the internal diameter of the tubing becomes smaller, there is a gradual reduction in the hearing aid's gain in the frequencies above 2000 Hz.

Different size of tubing

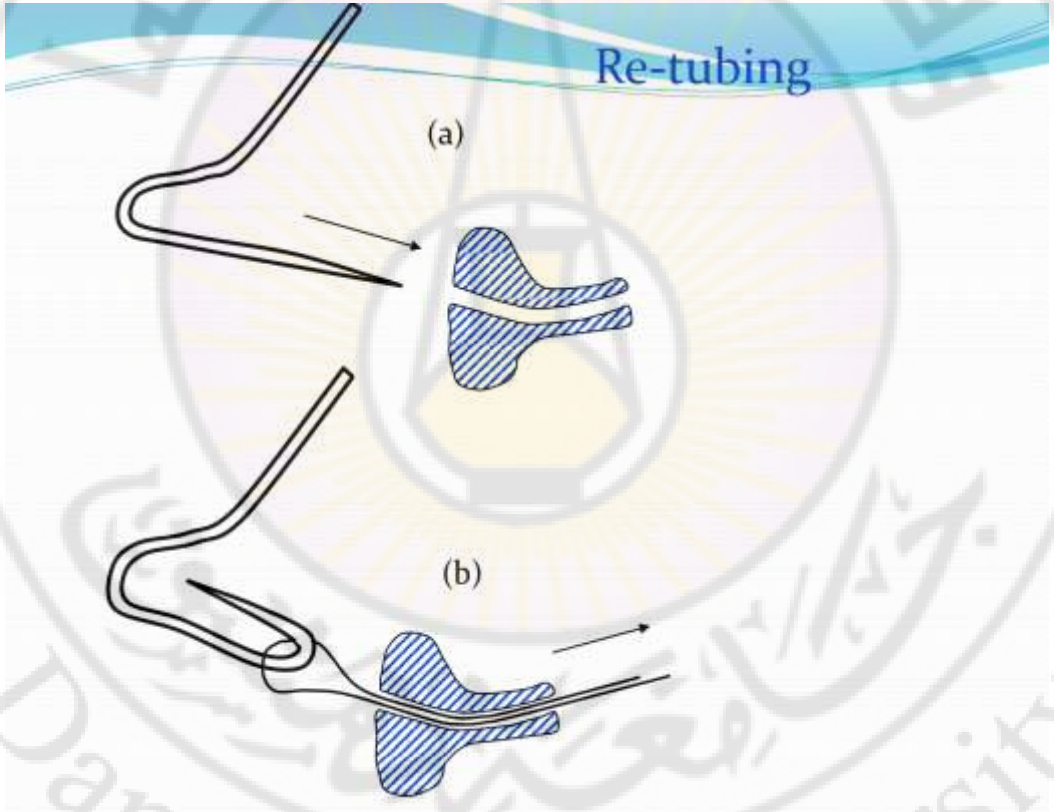
Tubing Size	Inside/Outside Diam
#12	2.16/3.18 mm
#13 standard	1.93/2.95 mm
#13 medium	1.93/3.10 mm
#13 thick	1.93/3.30 mm
#14	1.68/2.95 mm
#15	1.50/2.95 mm
Thin tube	0.90/1.40 mm

Tips……………………

- Tubing customarily is glued into the sound bore of the earmold, and needs to be replaced two to four times per year on average.

- Tell your patient not to remove the earmold by pulling on the tubing, but they'll probably do it anyway.
- After each re-tubing, re-adjusting or real ear measurement verifying is necessary.

Re-tubing



Insertion of tubing into an earmold by (a) pushing, or by (b) pulling with a loop of wire.

Sound bore and horn effect……………..

- Sound bore is a channel that sound pass through it to the ear canal.
- Changes in sound bore diameter affect the high frequency response of the hearing aid.
- Increases in bore diameter alone, however, yield little or no change in the high frequency response.

Sound bore and horn effect……………

- The gradual increase of internal diameter of sound bore is named horn effect. This create an increase on gain of high frequencies (2000-5000 Hz) and improve speech intelligibility.

- The standard sound bore is 2 to 3 mm diameter. When the sound bore is increased to 4 mm at the end of the sound channel, a horn effect may be achieved, and high-frequency gain is increased 2 to 4 dB. A smaller bore (1 to 2 diameters) can be used to slightly enhance low-frequency sound.

Sound bore and horn effect……………

- For an acoustic horn to work effectively, a gradual increase in internal diameter over a specific length is required. If the length is too long or too short, the increase in high frequency gain will not be attained.

- With a properly constructed horn, an increase in high frequency gain can be readily achieved without increasing the hearing aid's output.

- The 3 mm or 4 mm Libby horn is a proven method for obtaining more high frequency gain mechanically rather than electronically.

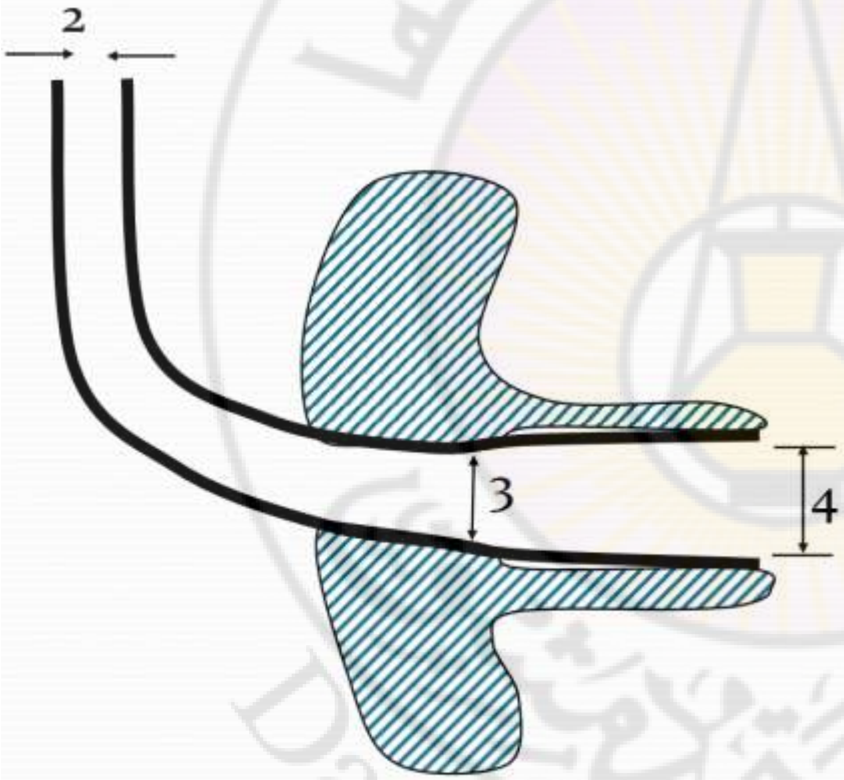
Sound bore and horn effect……………

- When using a Libby horn tube, it is important not to modify the length of the tubing, as shortening the length can render the horn ineffective.

- The mechanical action of a horn is governed by the length and taper of the horn.

- In simple terms, for any horn to be effective, the length of the horn must be at least 17 mm. With the proper length maintained, up to 12 dB of gain between 2 kHz and 5 kHz can be obtained.

Nowadays, horn effect almost are not used.



A Libby 4 mm horn fully inserted into the earmold. Diameters are in mm.

Dampers……………………….

- As sound travels through the tubing, standing-wave resonances occur.
- These create peaks in the frequency response of the hearing aid, resulting in acoustic feedback or poor sound quality.
- To some extent, these peaks and valleys can be smoothed out electronically using the fitting software.
- However, many manufacturers still rely on plastic or metal dampers placed in the earhook to smooth out the response.

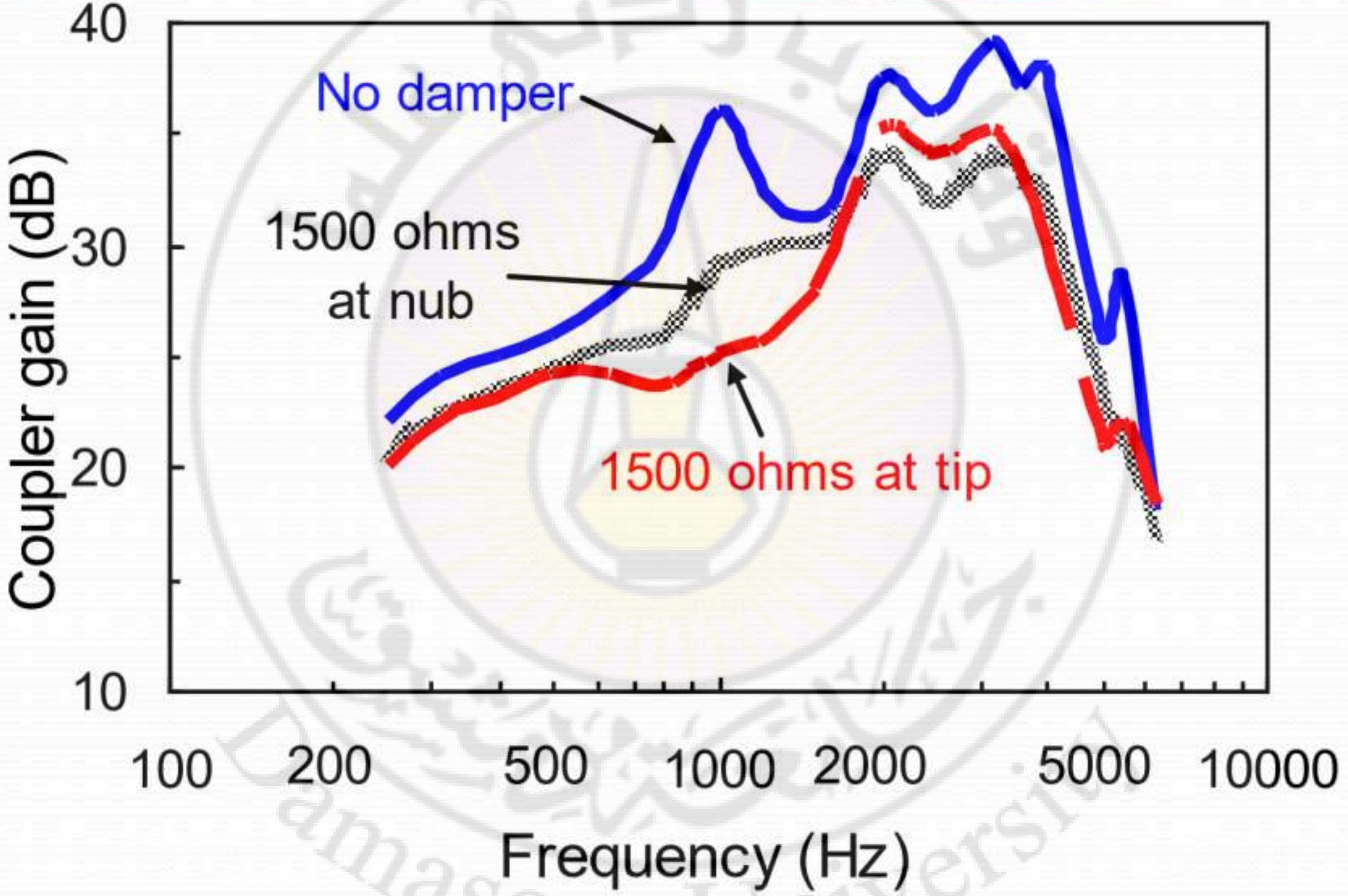
Dampers……………………………………

- The location of the damper also has a significant effect on the frequency response.

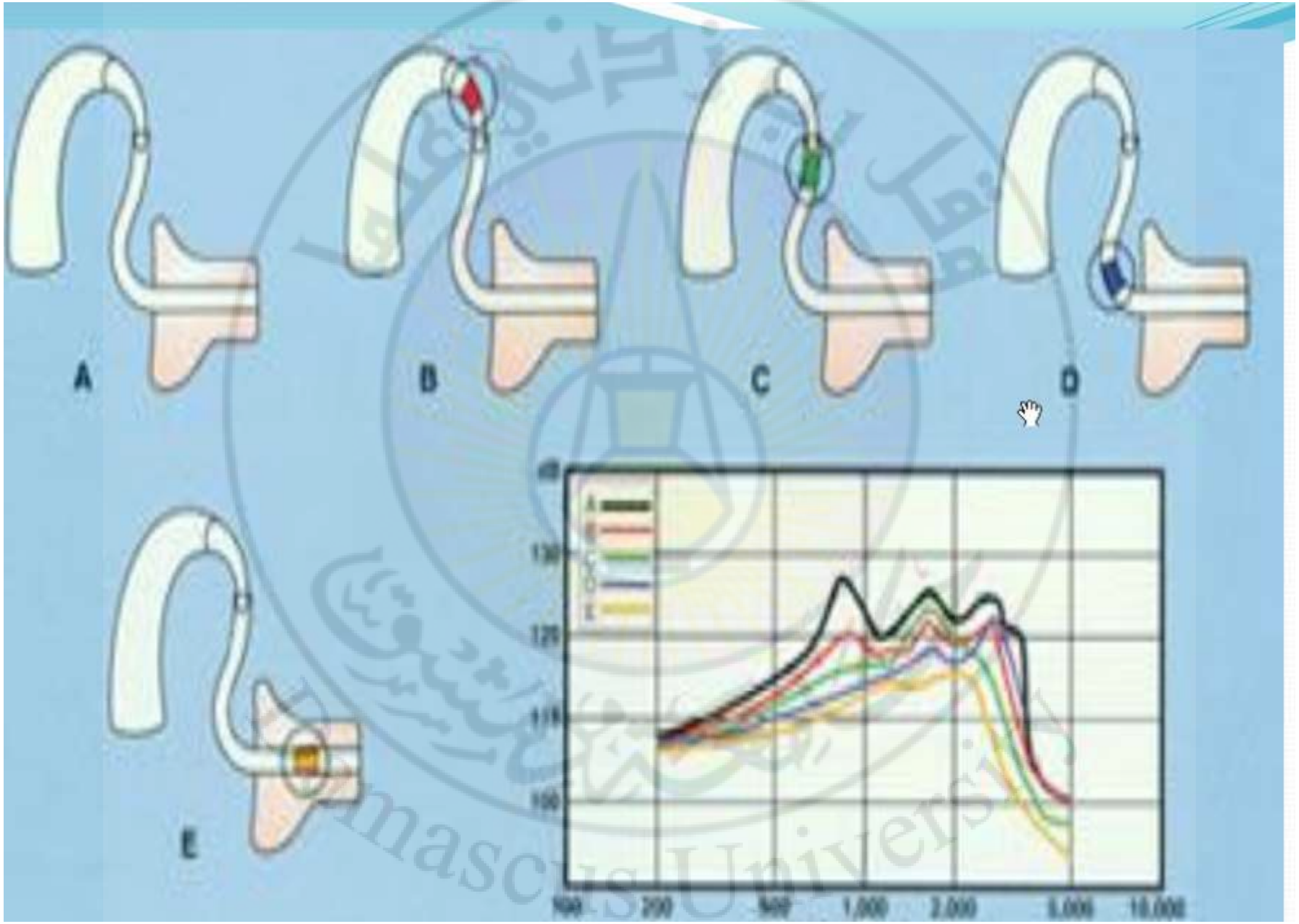
- For the clinician, knowing the type and location of a damper is essential when replacing it, as the damper impacts sound quality.

- Dampers, working in the mid-frequencies, primarily remove unwanted peaks in the response curve, and take several forms

Effects of dampers



Frequency response of a hearing aid with no damper, and with a 1500 ohm damper placed at each end of the earhook.


A
B
C
D
E
dB
130
120
110
100
100
200
500
1,000
2,000
5,000
10,000

Open hearing aids……………

- OC fittings (the ones that are truly open) maintain most of the resonance of the unoccluded ear canal.
- the advantages of OC include:
  - Maintenance of natural canal resonance, so, less gain
  - More natural sound specially in normal hearing at low frequency
  - Comfort own voice without occlusion effect.

What is the occlusion effect?

- The occlusion effect is a phenomenon that has been described by some hearing aid users, newer ones in particular, of their own voice sounding a bit "funny“ when they talk.
- Sometimes people feel like they're talking in a barrel, that what they're hearing is echoes of their own voice, or that their voice sounds "hollow" or "booming.
- People who have a large occlusion effect may also feel a sense of pressure or blockage in the ear when an earmold is inserted.

What is the occlusion effect?

It is disturbing enough so that if the occlusion effect is not solved or made acceptable in a reasonable time frame, some people will reject hearing aids.

How does the occlusion effect occur?

- An occlusion effect occurs when some object (like an unvented earmold) completely fills the outer portion of the ear canal.
- What this does is trap the bone-conducted sound vibrations of a person's own voice in the space between the tip of the earmold and the eardrum.

How does the occlusion effect occur?

- Ordinarily, when people talk (or chew) these vibrations escape through an open ear canal and the person is unaware of their existence.
- But when the ear canal is blocked by an earmold, the vibrations are reflected back toward the eardrum and increases the loudness perception of their own voice.
- Compared to a completely open ear canal, the occlusion effect may boost the low frequency (usually below 500 Hz) sound pressure in the ear canal by 20 dB or more.

How does the occlusion effect remove?

- The way to reduce or remove the occlusion effect is to not completely block the ear canal with an earmold.
- This permits the bone-conduction sound generated in the ear canal to escape the ear the way it is supposed to.
- When someone is wearing hearing aids, the only way to do this is to create a vent in the earmold.
- The amount of sound that escapes, and thus the magnitude of the occlusion effect, depends upon the size of the vent.
- The larger the vent, the more the occlusion effect can be reduced.

Hearing aid components

Block diagram

A block diagram shows what operations a device carries out on signals within the hearing aid, and in what order each of these operations is carried out. Block diagrams also usually show the location of fitter and user controls within the processing chain. This helps the clinician understand what effect varying a control will have, and, just as importantly, what effects it will not have.

Block Diagrams

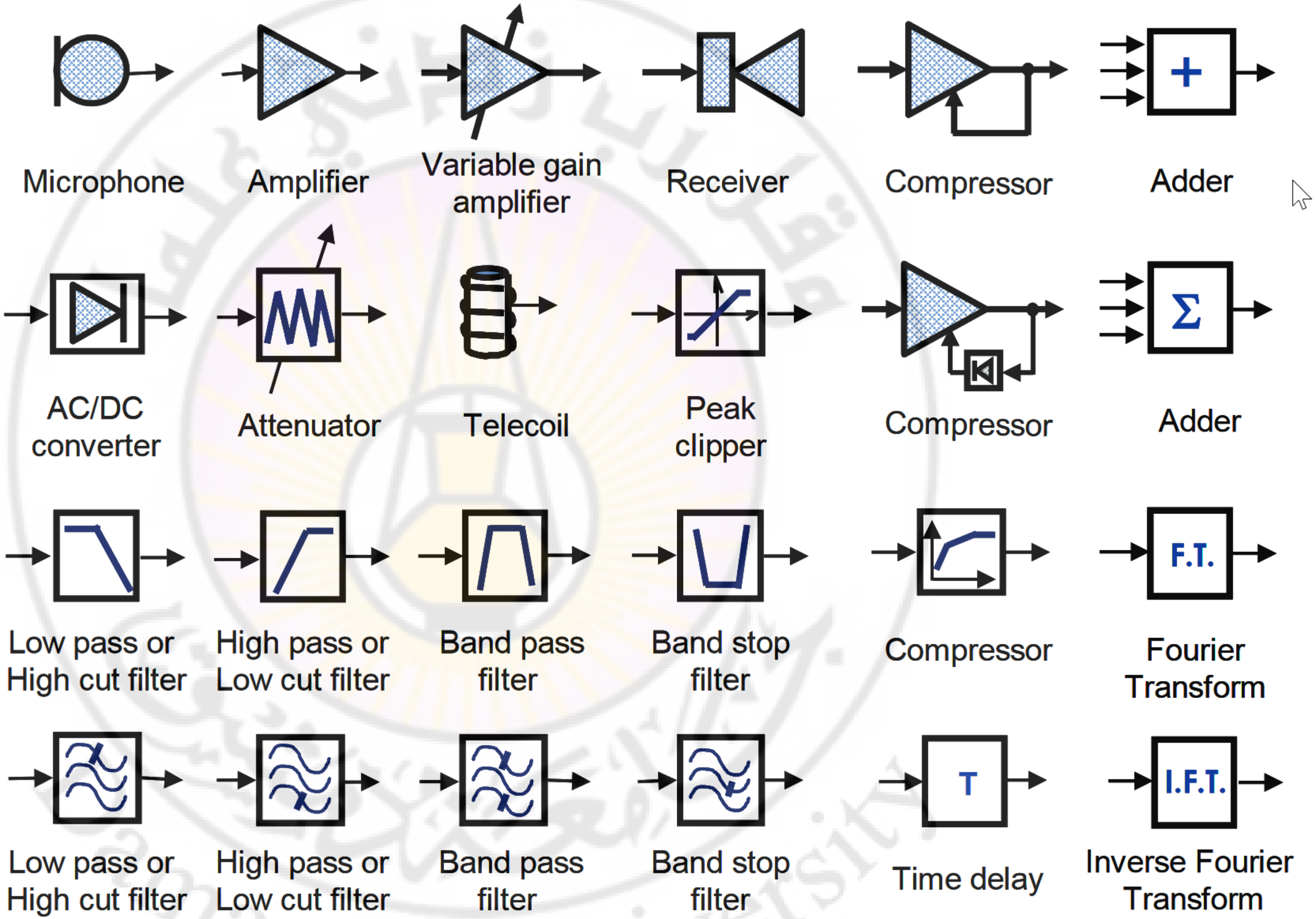


Figure 2.1 Symbols used in block diagrams.

Please explain this block diagram?

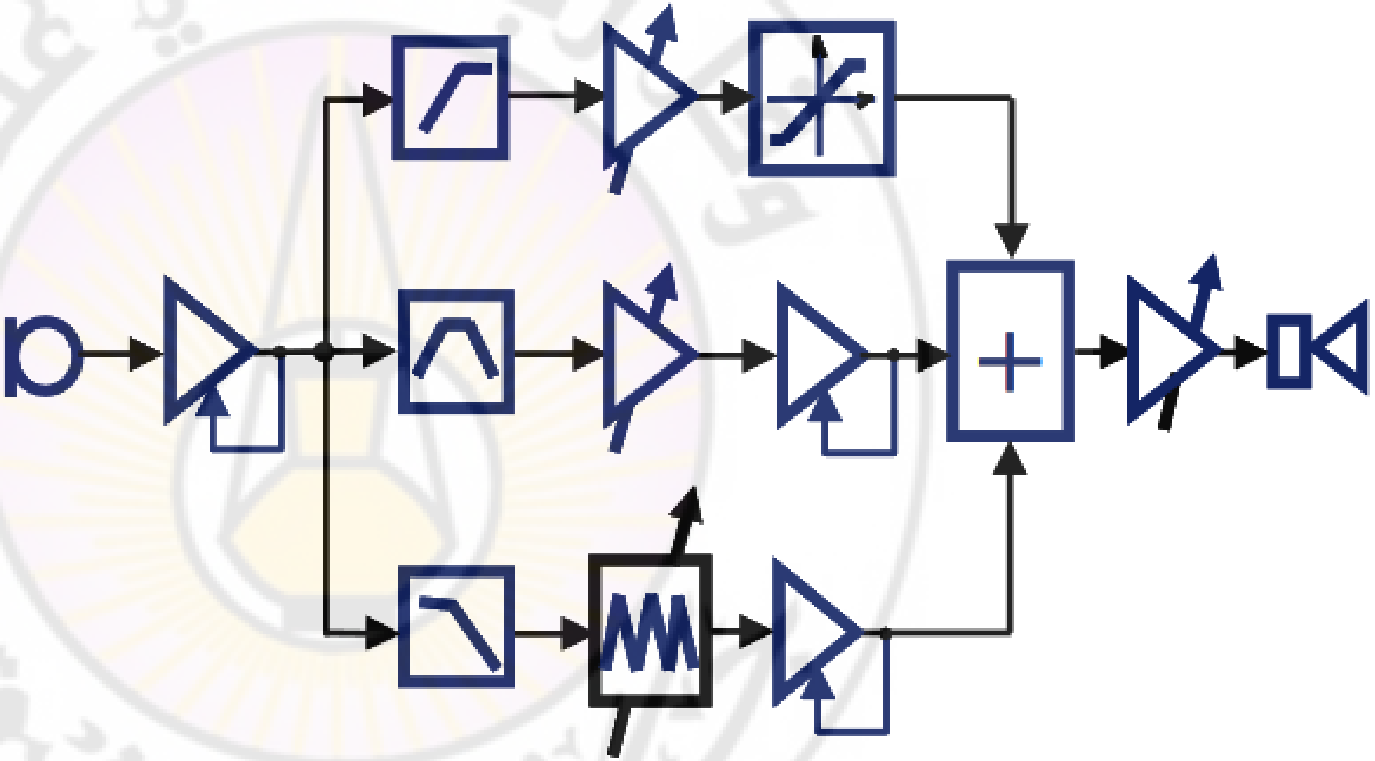


Figure 2.2 A three channel compression hearing aid.

Microphones

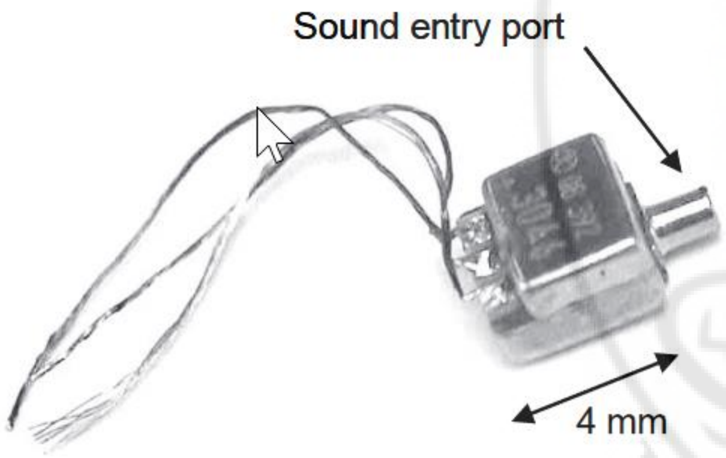


Figure 2.3 An electret microphone.

- the ratio of the size of the output voltage to the size of the input sound pressure is known as the sensitivity of the microphone.
- Typical hearing aid microphones have a sensitivity of about 16 mV per Pascal, which means that sounds of 70 dB SPL produce a voltage of around 1mV.

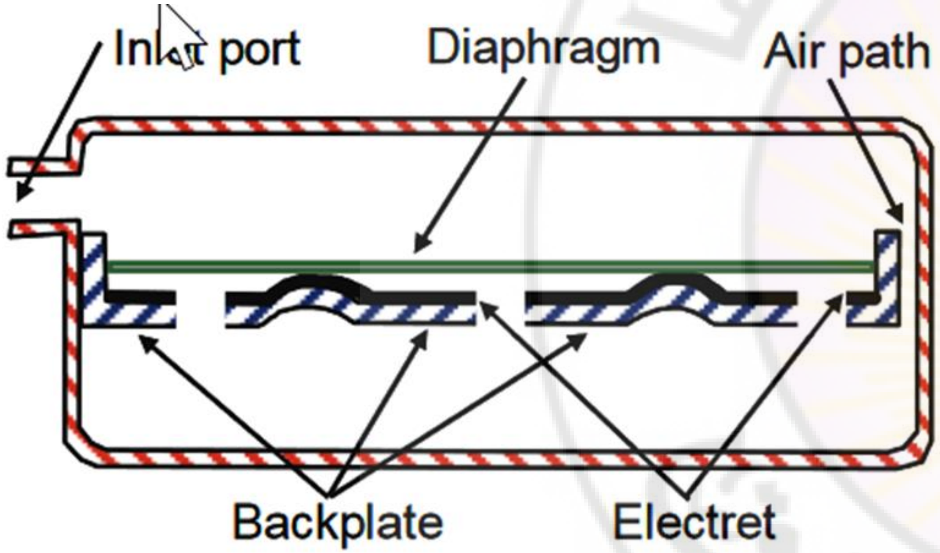


Figure 2.4 Cross section of an electret microphone.

Sound waves enter through the inlet port and reach one side of a very thin, very flexible plate with a metallized surface, called the diaphragm. Pressure fluctuations within the sound wave cause the diaphragm to move up and down (by an extremely small amount, invisible to the naked eye). A small air space separates the diaphragm from a rigid metal plate, called the back-plate. Coated onto the back-plate is some thin Teflon material called an electret. The diaphragm is held away from the back-plate by some bumps in the back-plate. The electret plate has a permanent electric charge comprising an excess of electrons on one side of it. When sound pressure forces the diaphragm towards or away from the electret, the changing distance between the diaphragm and the electret changes the electrical force between the opposing charges, which is another way of saying that the voltage between the back-plate and diaphragm varies.

Field effect transistor or buffer amplifier or follower

Frequency Response of Microphones

- The frequency response is the gain or output of Mic. as a function of frequency. It shows the amount of output in different frequencies.
- The frequency response of electret Mic. is flat. although variations from a flat response occur both by design and accident.
- the first variation is due to low cut filter. A low cut filter intentionally introduced into mic. this makes hearing aid less sensitivity to the intense low frequency sound that around us.
- how to reduce low frequencies in electret Mic. or how to makes low cut filter?
  - the response is easy. By a small passage-way between the front and back of the diaphragm allows low-frequency sounds to impact almost simultaneously on both sides of the diaphragm, thus reducing their effectiveness in moving the diaphragm.

Frequency Response of Microphones

- The second variation is due to acoustic resonance within the microphone case which is called Helmholtz resonance.

- This resonance causes a peak in the gain frequency response, typically about 5 dB somewhere in the range from 4 to 10 kHz.

- The shorter and wider the inlet port, the higher is the resonant frequency, and consequently the greater is the high frequency bandwidth.

Frequency Response of Microphones

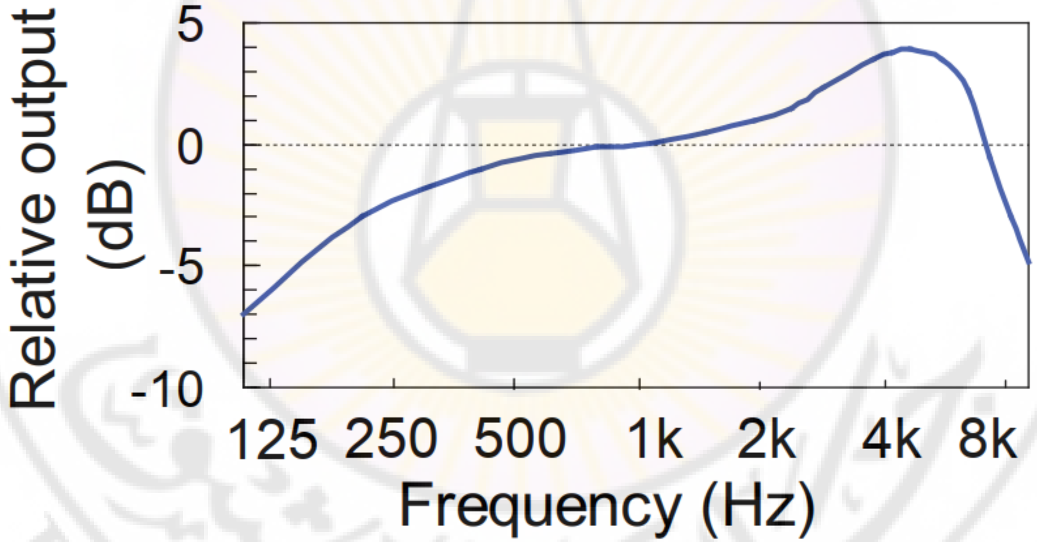


Figure 2.5 Frequency response of a typical electret microphone with tubing on its input port.

Microphone imperfections

- break down by adverse chemical agent like perspiration
- having random electrical noise
- very sensitive to vibration.
- will overload or distort sound when the internal sound pressure is very high.
- wind noise

Directional microphones

Directional microphones suppress noise coming from some directions, while retaining good sensitivity to sounds arriving from one direction

Directional microphones

- ❖ The directional sensitivity of microphones is usually indicated on a polar sensitivity pattern.
- ❖ This particular response shape is called a cardioid, because of its heart shape.
- ❖ The opposite of a directional microphone is an omni-directional microphone.
- ❖ Which has a single port and a polar pattern in the shape of a circle.

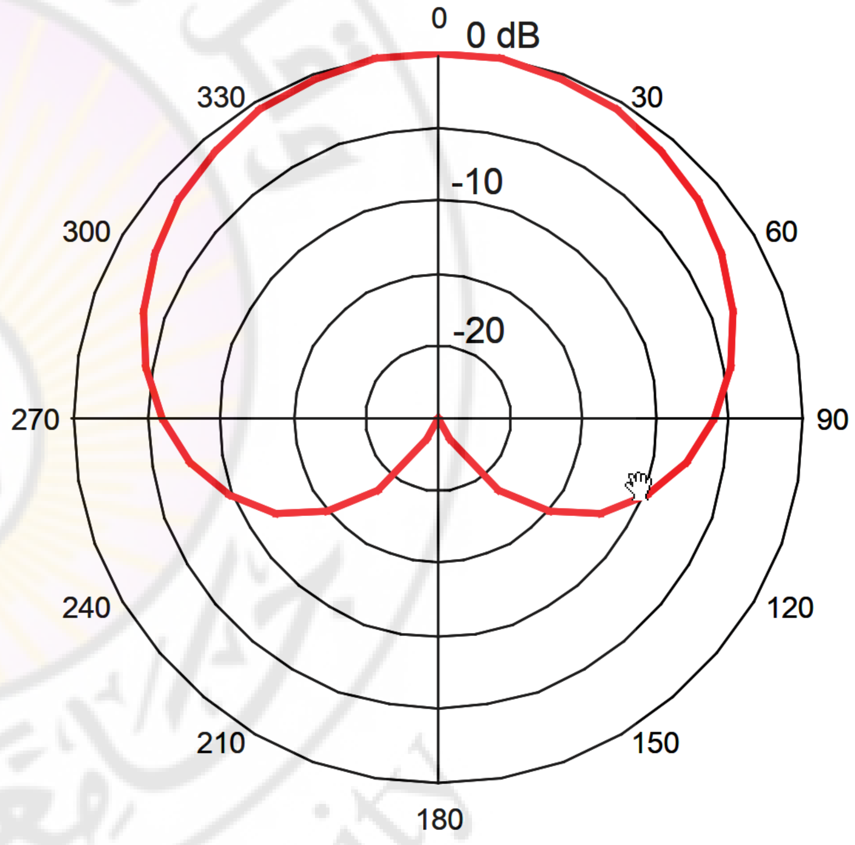


Figure 2.7 Directional sensitivity (in dB) of a microphone with a cardioid sensitivity pattern.

How does directional Mic. work?

By changing them ratio of the internal delay to the external delay, a whole family of patterns can be generated.

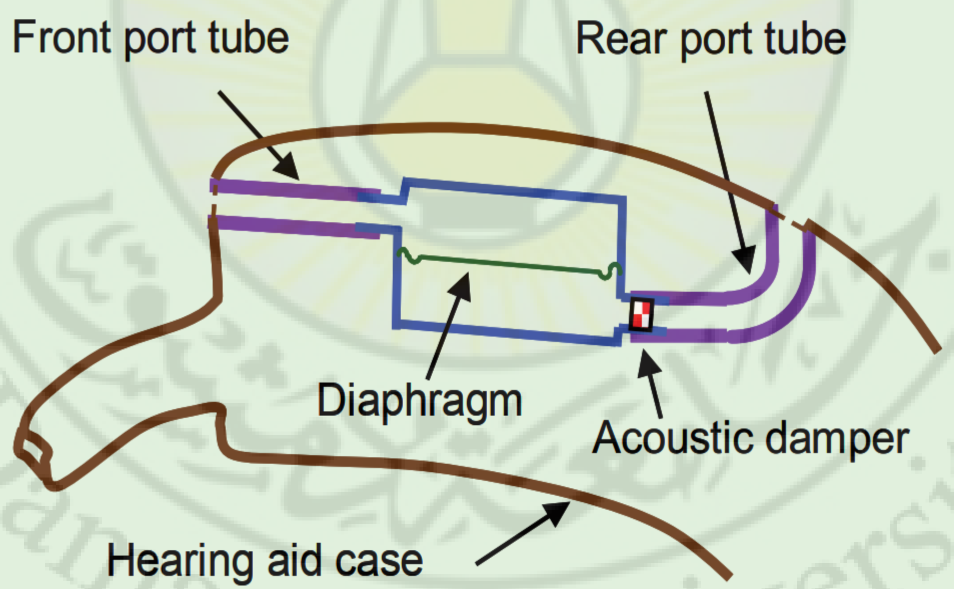


Figure 2.6 Diagram showing the sound paths in a directional microphone.

When directional pattern is the most desirable?

❖ In many real-life situations, unwanted noise arrives more or less equally from all directions…………………………………………….

❖ They are the only form of signal processing that can improve the signal-to-noise ratio (SNR) in a way that significantly improves intelligibility.

❖ The benefits of directional microphones are well established. The benefit decreases as the environment becomes more reverberant, unless the source of the wanted signal is very close to the listener.

What is the directivity index?

This ratio of sensitivity for frontal sounds relative to sensitivity averaged across all other directions is referred to as the directivity index (DI).b The “other directions” can be restricted to just the horizontal plane (giving a two-dimensional DI) or to all directions in space (giving a three-dimensional DI)

An omni-directional microphone has a DI of 0 dB when measured in free space, but when mounted on the head its DI varies from about -1 dB in the low frequencies to 0 to 2 dB in the high frequencies, depending on the microphone location.

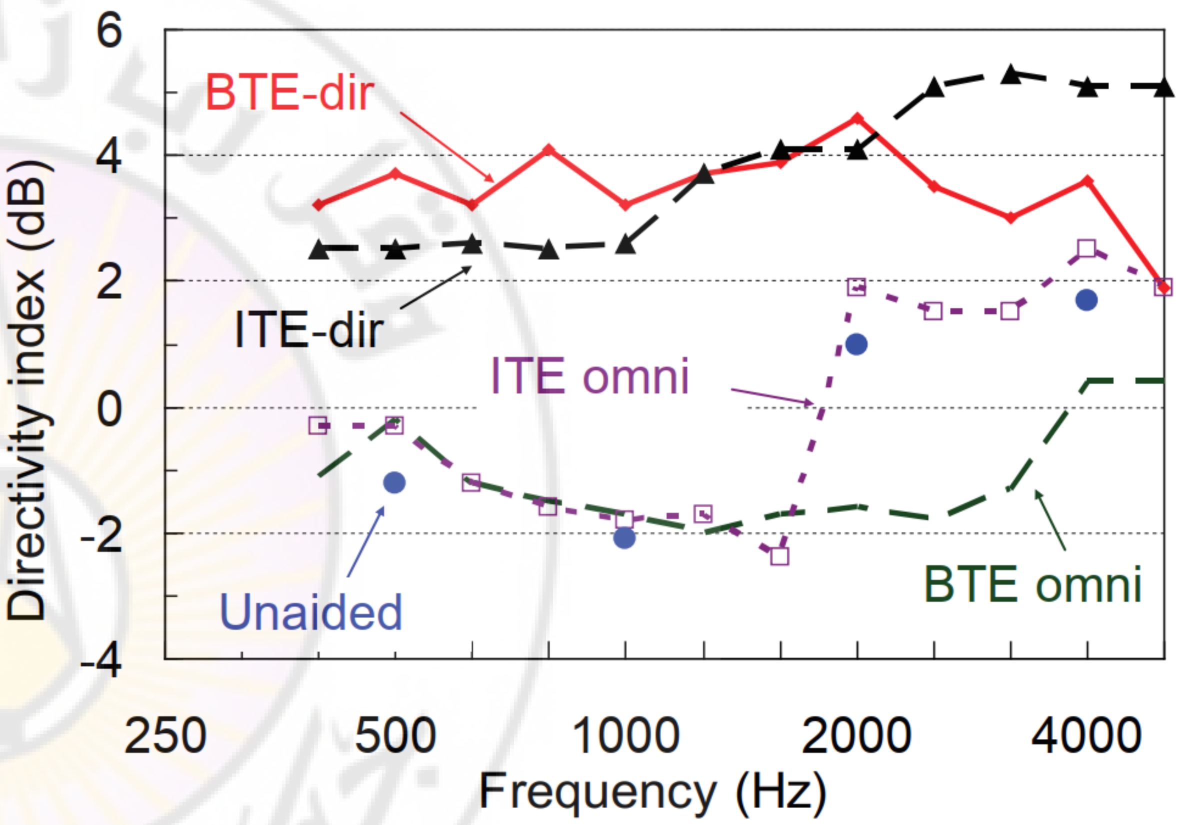


Figure 2.8 Directivity index, measured in the horizontal plane, for a directional and omnidirectional BTE and a directional and omnidirectional ITE.[1504] Also shown (as blue dots) is the directivity index for an unaided ear, measured on KEMAR.[454]

When the sound comes from behind the person……. what should we do?

1- The first is for a hearing aid to include both a directional microphone and an omni-directional microphone, and for the user (or the hearing aid itself) to select the microphone most appropriate to each situation.

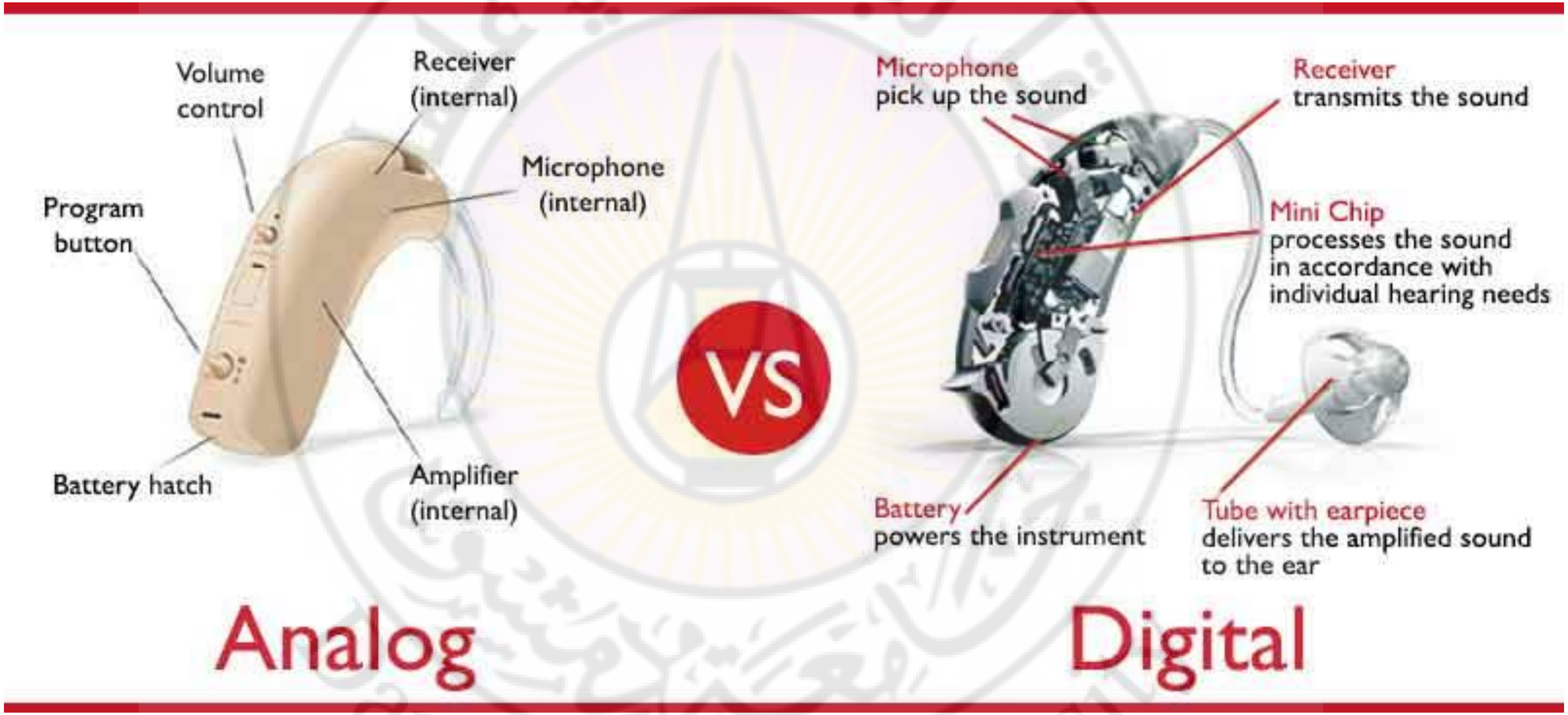
2- solution is for the hearing aid to incorporate two separate omni directional microphones, each with one inlet port. When an omni-directional sensitivity pattern is needed, the output of one microphone is selected or the two outputs are added together. When a directional sensitivity pattern is needed, the two microphones are used in a different combination. In this dual-microphone technique, the output from the second microphone is electronically delayed and subtracted from the first microphone.

Amplifiers

- The amplifier job is to make a small electrical signal into a larger electrical signal.
- Amplifiers can do three things:
  - they increase the voltage
  - they increase the current
  - they increase both of the current and voltage

Amplifier Technology

- Nearly all of today's hearing instruments include digital signal processing.
- Until the end of the 1980s, hearing health care professionals or audiologists dispensed analog hearing instruments that were adjusted according to the degree of hearing loss with screwdriver-controlled potentiometer trimmers.
- They make continuous sound waves louder by amplifying all sounds (speech and noise) in the same way.


Volume control
Receiver (internal)
Microphone (internal)
Program button
Battery hatch
Amplifier (internal)
Analog
VS
Microphone
pick up the sound
Receiver
transmits the sound
Mini Chip
processes the sound in accordance with individual hearing needs
Battery
powers the instrument
Tube with earpiece
delivers the amplified sound to the ear
Digital

VS

ANALOG HEARING AID

DIGITAL HEARING AID

- Then the advent of integrated circuit chip technology allowed for the development of analog hearing instruments that could be programmed digitally with dedicated programming devices or with personal computers.
- hearing aids were analog but they can fit with general programming device as called “Hypro” or dedicated programming device in each company.

Digital signal processing

- In 1996, the third type of hearing instrument was introduced; they were digitally programmable and offered digital signal processing.

What is the analog hearing instrument?

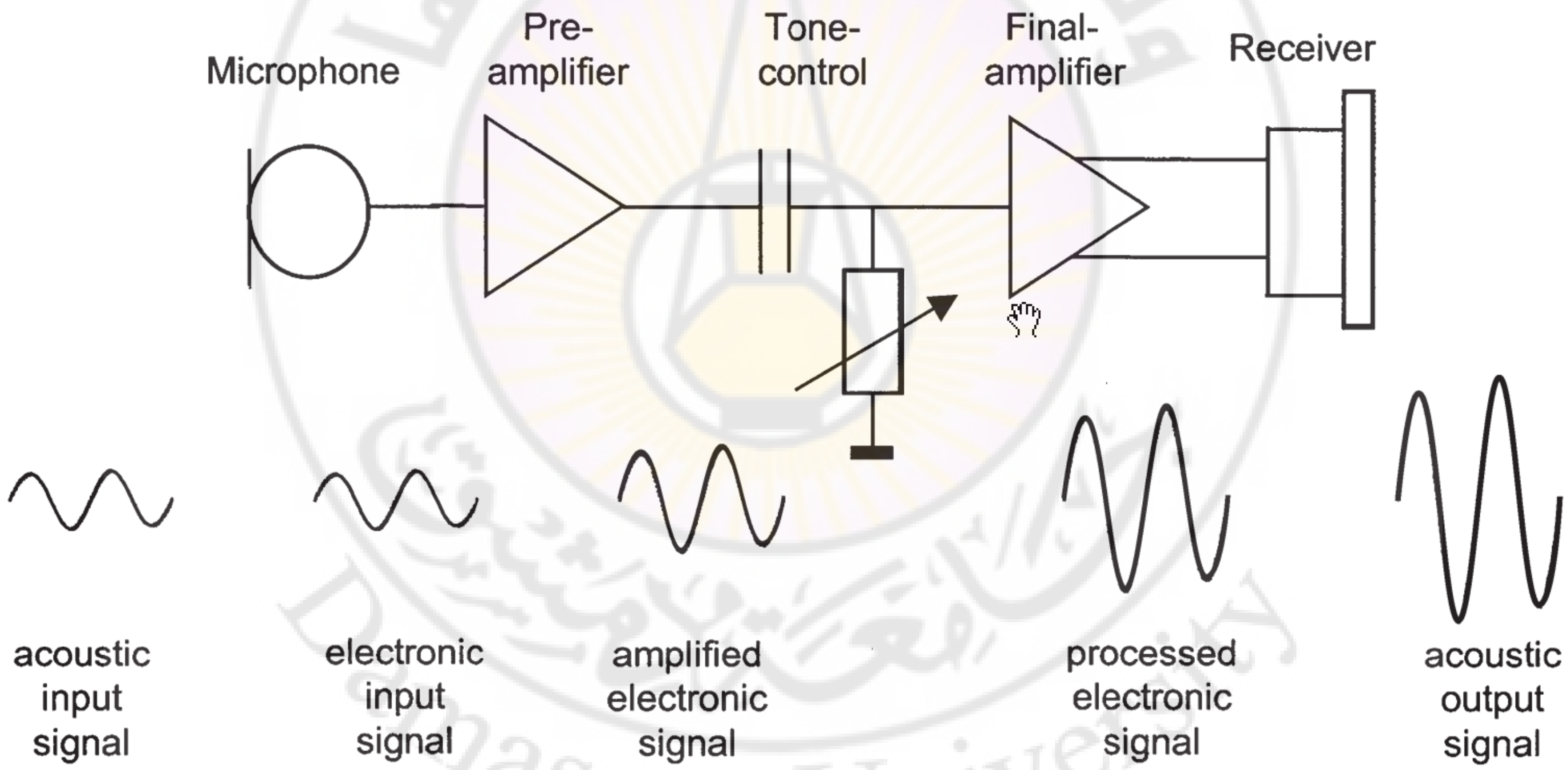


Figure 7–1. Block diagram of an analog hearing instrument.

Programable hearing aids

In analog hearing instruments that are digitally programmable, parameters of processing, such as the tone control or the compression ratio of the AGC, are stored in a memory and can be modified for different hearing instrument users

Block diagram digital hearing aids

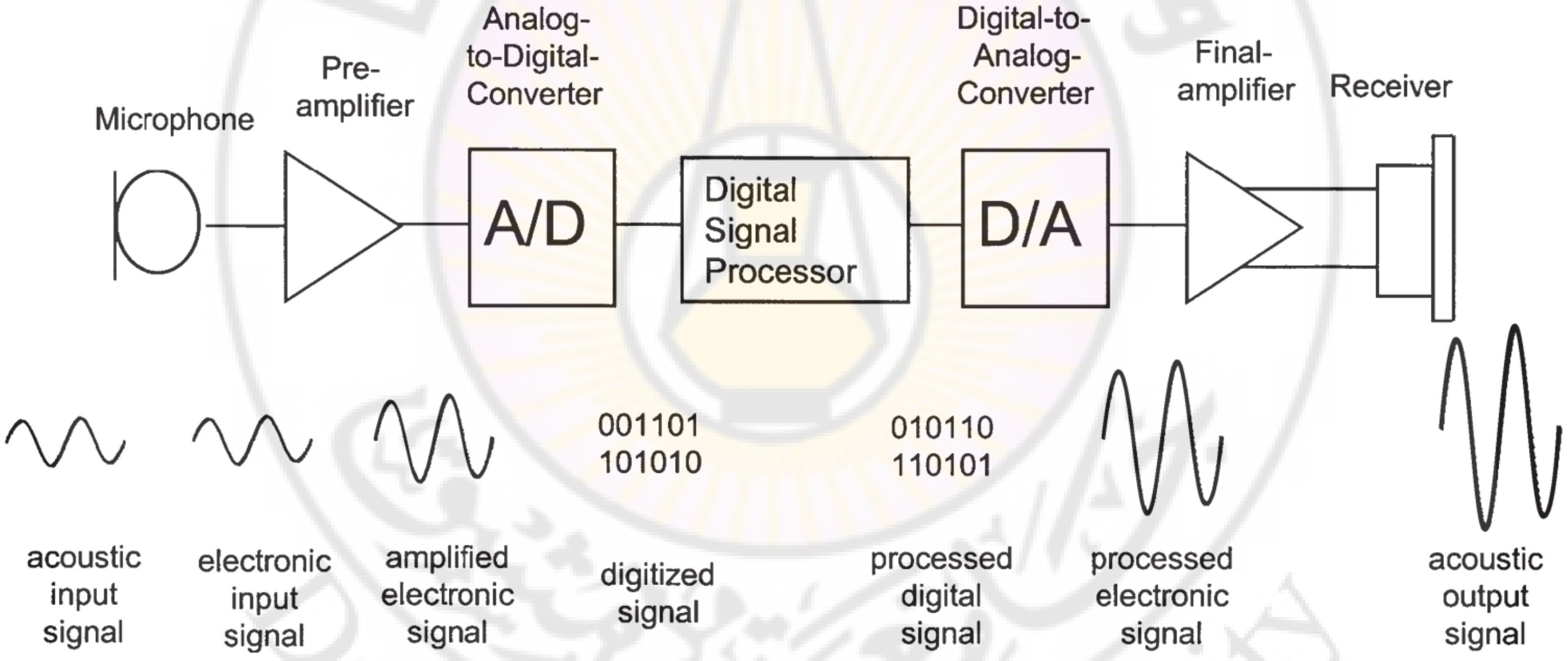


Figure 7–2. Block diagram of a digital hearing instrument.

Analog signal versus digital signal

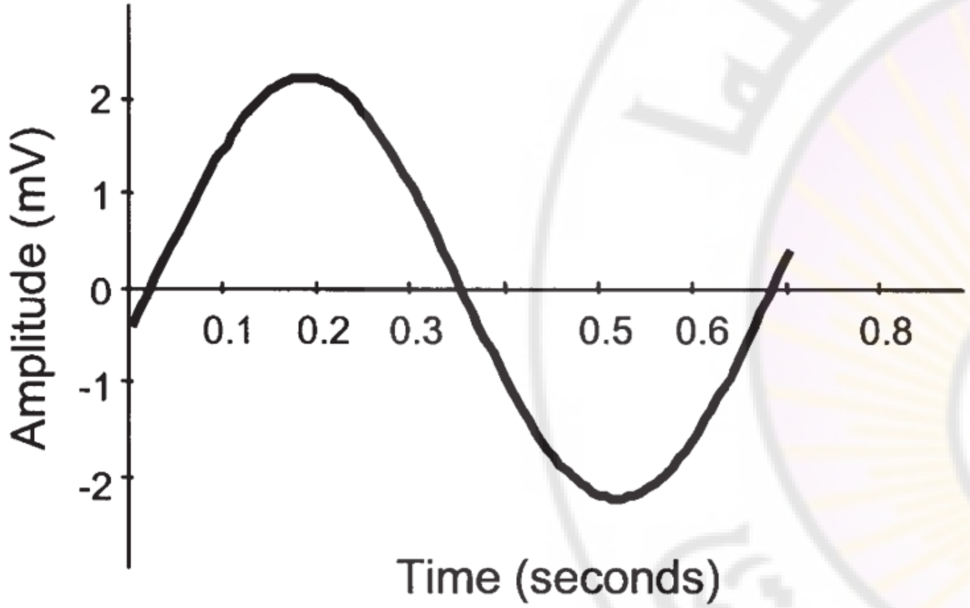


Figure 7–3. An analog signal with different levels at any given point of time.

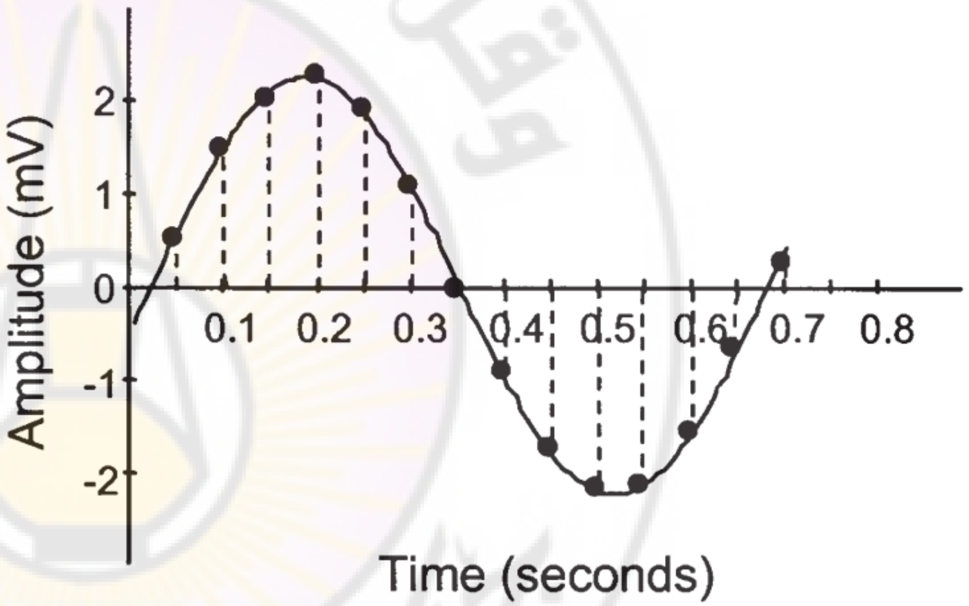


Figure 7–4. An analog signal sampled at discrete points in time.

Continues in time and amplitude, with this continuously varying signal, there are no “steps” in the signal curve.

The input signal is sampled at discrete points in time. corresponding to a sampling rate of 20 kHz.

Analog to digital convertor

- This sampling of the input signal is done in the analog-to-digital (A/D) converter.
- it is important to sample the input signal frequently enough to have a good representation.
- The analog input signal is first low-pass filtered with a cutoff frequency appropriate for a hearing instrument (e.g., 10 kHz). Then, the analog signal is sampled at a rate at least twice the cutoff frequency of the low-pass filter to avoid aliasing. In this last step, quantization of the samples is dependent on the number of bits available in the A/D converter

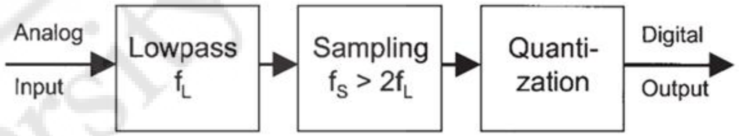


Figure 7–10. Components of an A/D-converter.

What is the aliasing?

This figure shows as an example a sine wave, labeled "input signal" that is sampled at four different discrete points in time. Note that only one or two points are sampled within each cycle. The same four sample points can also represent a different sine wave with a lower frequency, labeled "aliasing signal." This ambiguity results in the presence of a sine wave, which was not present in the original input, at the output of the digital system. This type of error is called aliasing. In an acoustic system, such as a hearing aid, an aliasing error introduces distortions.

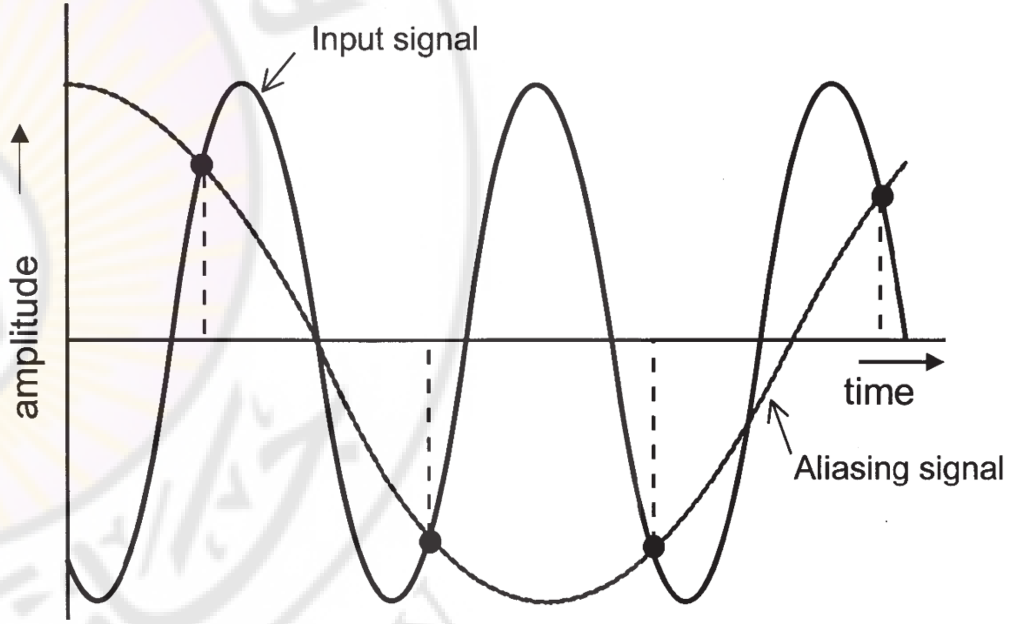


Figure 7–9. Two sine waves of different frequencies that can be represented by the same samples.

What is the quantization?

- The digital signal processor cannot use numbers with infinite precision for calculation. Therefore, each sample must be either truncated or rounded to a specific precision. This action called quantization.
- The quantized signal is characterized by bit values. A single bit can have two values (0 or 1).
- In this Figure , there is a digital signal in the upper panel. The signal is either on, with a value of 1, or off, with a value of 0. In the lower panel, the binary code or quantization value corresponding to the signal is given.

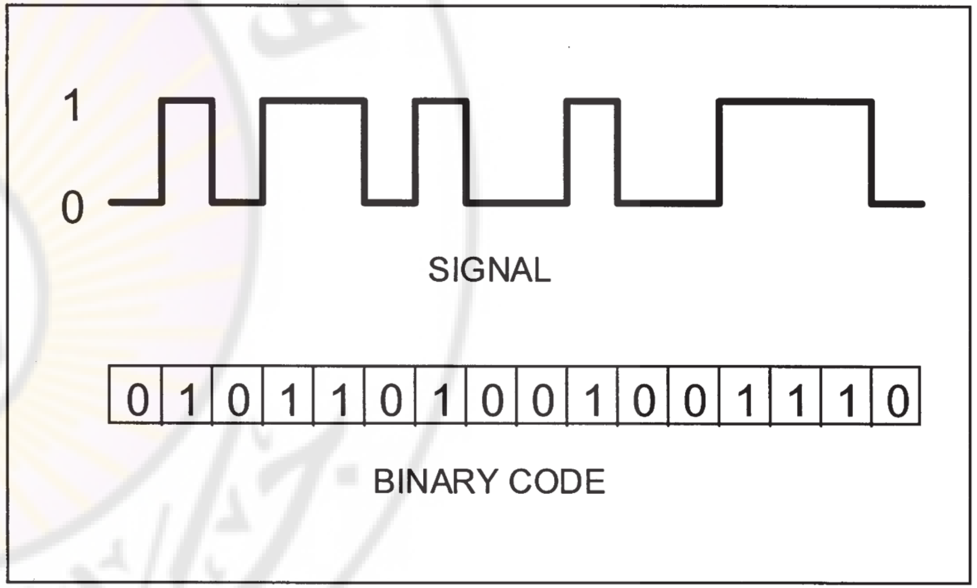


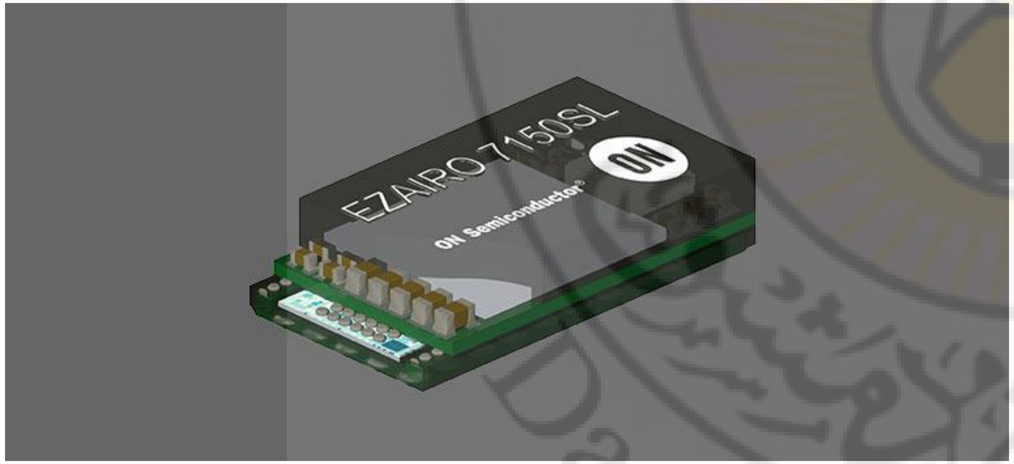
Figure 7–5. A digital signal with only two states, or 1 bit, and its corresponding binary code.

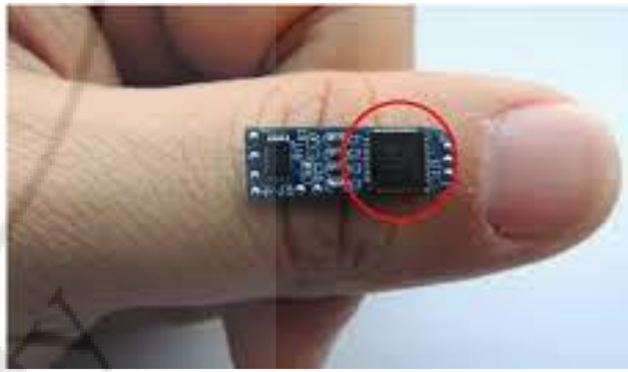
Processing algorithm

- After A/D conversion of the input signals, the digital signal processor can now use the digitized signals to process the algorithm with a predefined calculation procedure or algorithm.
- A simple example of a signal processing algorithm is multiplication of a digital number, resulting in amplification or attenuation of the input signal. For example, multiplication with a factor of 4 results in a gain of 12 dB.

Chip technology

Today’s hearing instruments generally contain one integrated circuit chip that contains its electronic parts. These electronic parts are simply small geometric patterns on the chip.



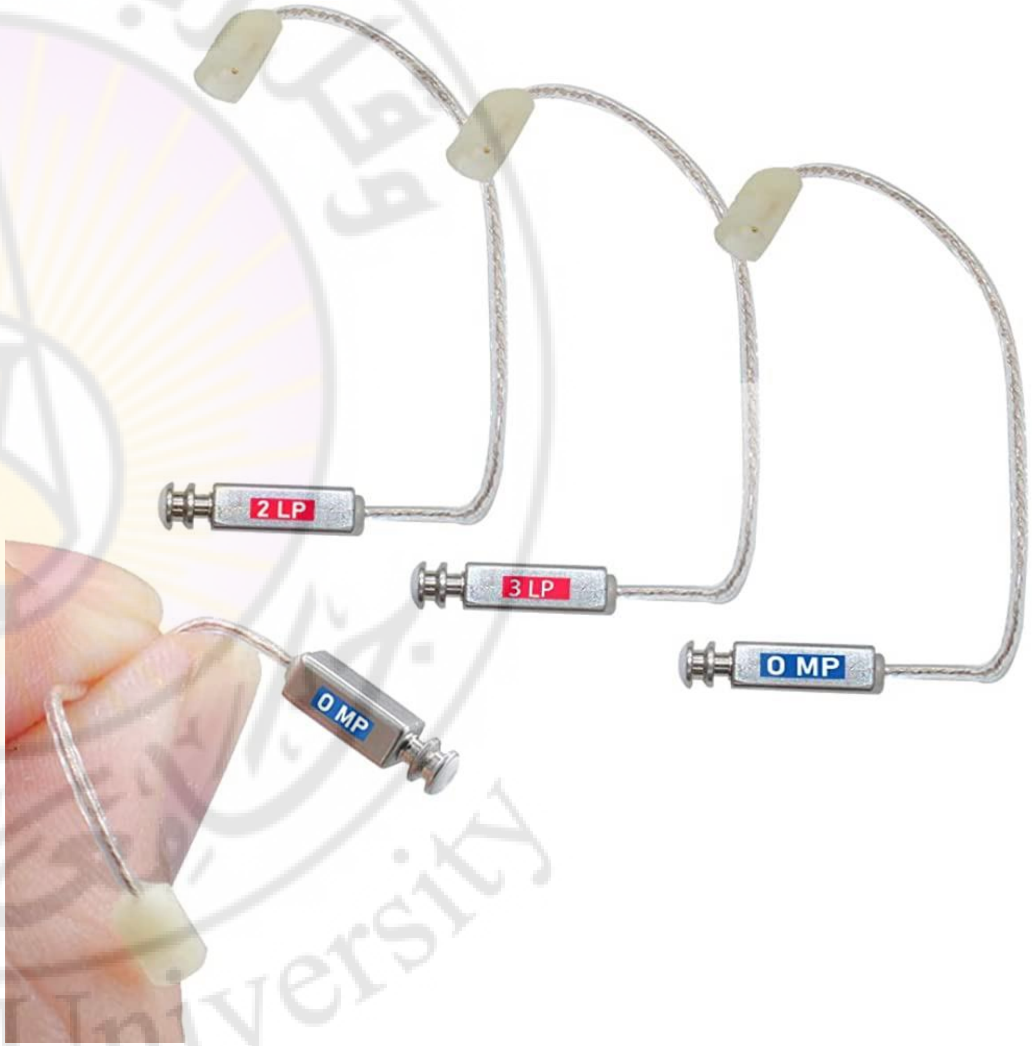


advantages of digital signal processing

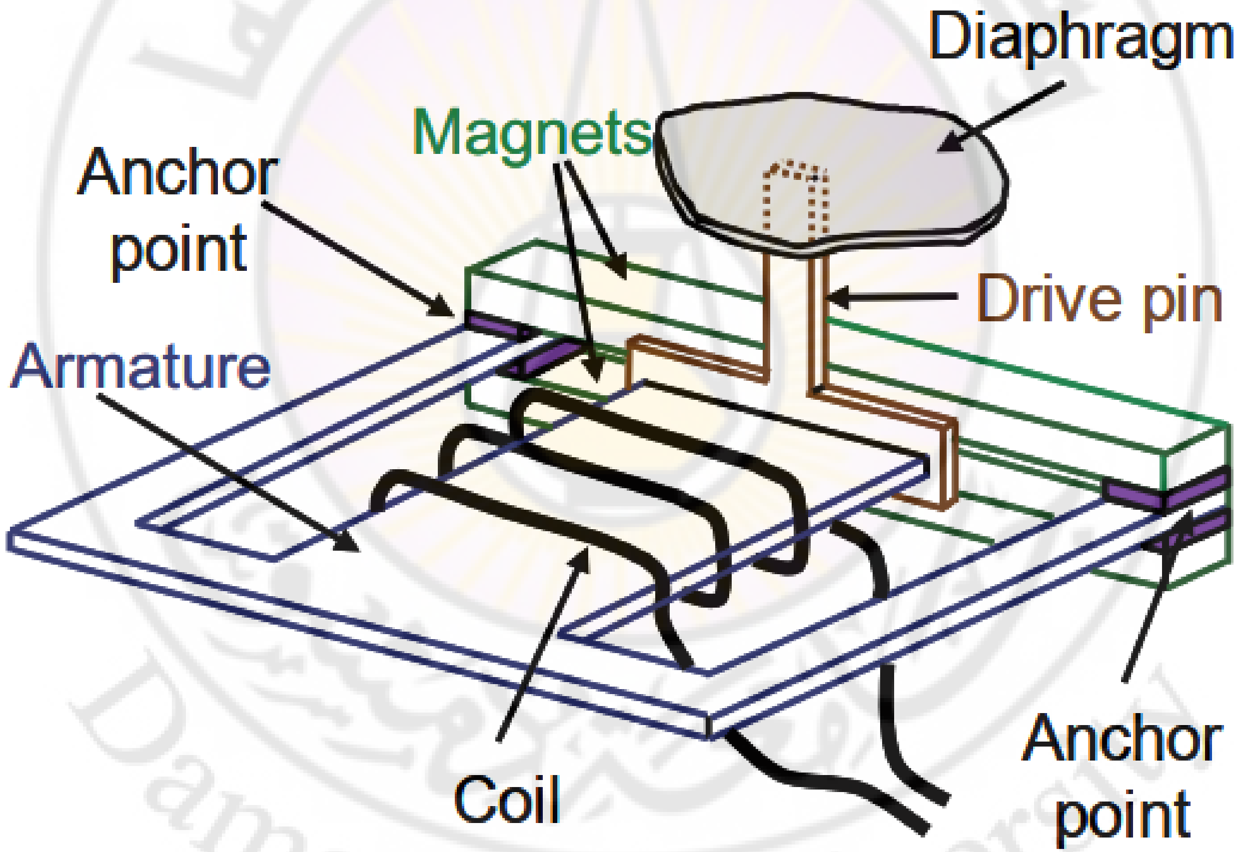
- miniaturization
- low power consumption
- low internal noise
- reproducibility
- stability, programmability
- signal processing complexity

Receivers

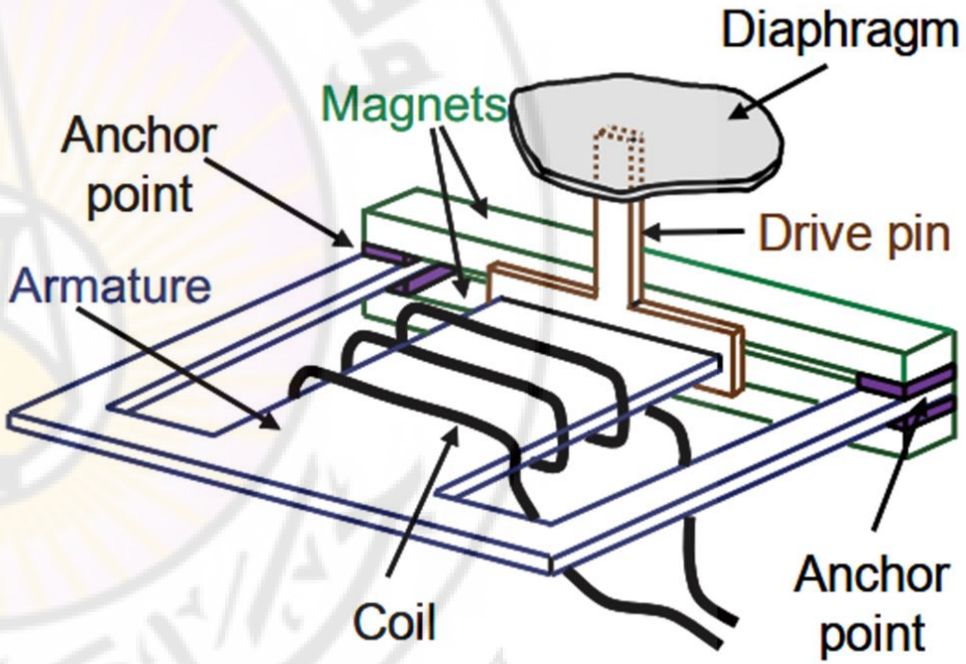
Receivers converts the amplified and modified electrical signal into an acoustic output signal



How do Receivers work?



For reaching to the grater output, receivers must be made bigger and electrical power for them must be grater.

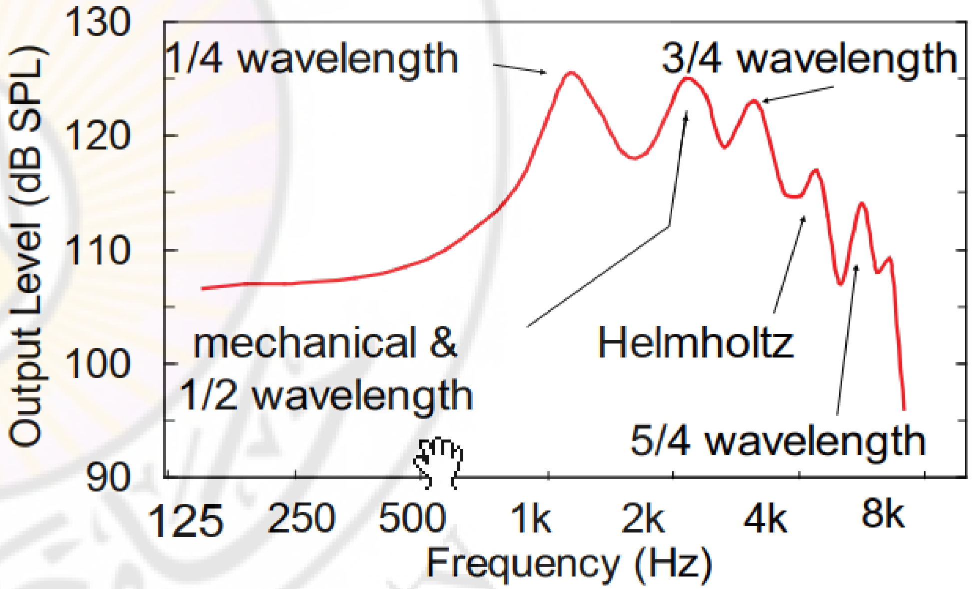


Frequency response of receivers

What causes all these bumps and dips?

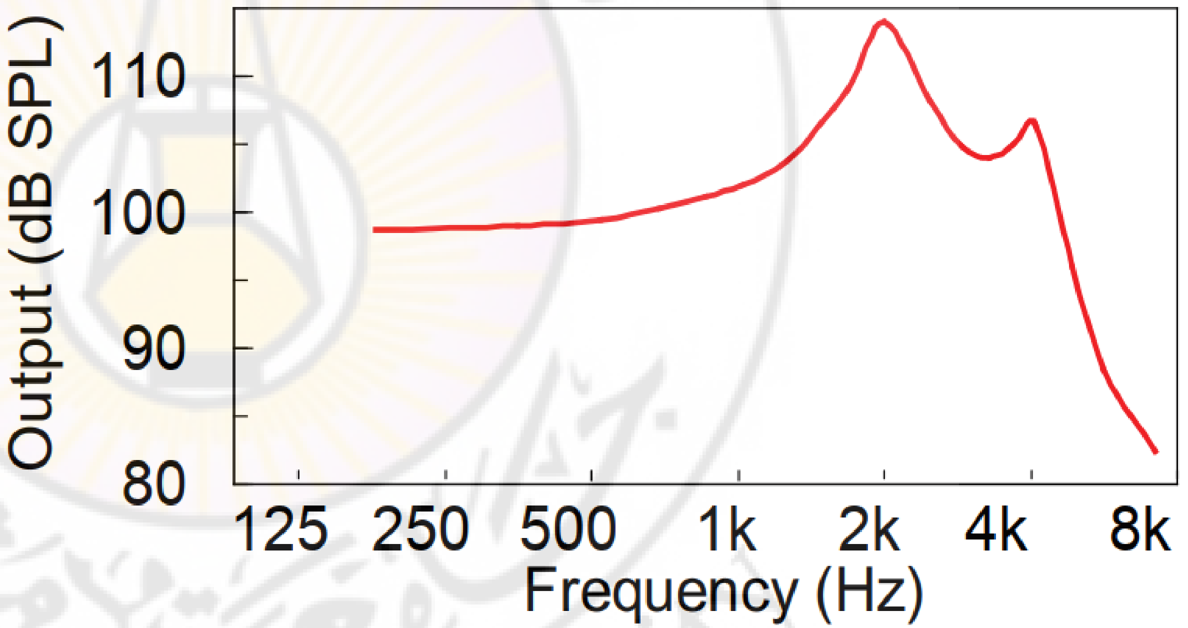
This receiver connected to the tubing used in the BTE.

These are related to tubing comprised a short length of tubing inside the hearing aid, and for a standard tube BTE, the ear hook, and finally the flexible tubing terminating at the tip of the earmold or the dome inside the ear canal.



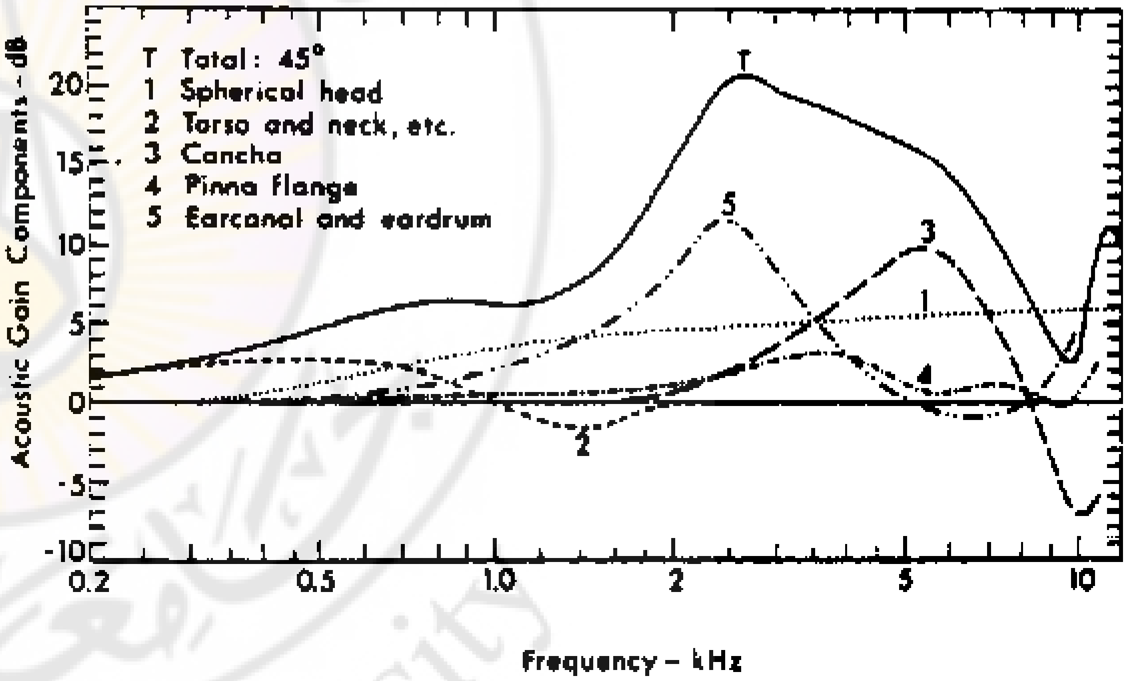
Frequency response of receivers

This is a frequency response of receivers in the ITE, ITC and RIC hearing aid.
In this device, there isn't tubing like that BTE. SO, the shape of FR is more smooth than the BTE



It is desirable for receivers to have a peak in the 2.5 to 3 kHz range because the unaided adult ear has a natural resonance in this frequency range .

Consequently, a receiver resonance at this frequency helps the hearing aid restore the natural resonance and gain that is lost when the hearing aid is inserted in the ear via any earmold that mostly closes the ear canal.



Frequency response of receivers

When coupled with suitable tubing and dampers, receivers can have a smooth, wide frequency response to 8 kHz or more that allows a very good sound quality to be achieved, however, to achieve a flat response out to 8 kHz requiring to a very high-power receiver.

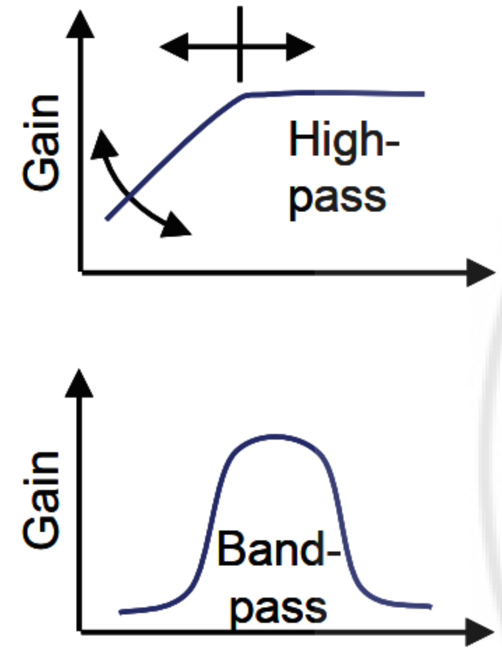
Filters and Tone Controls

They enable hearing aids to have different amplification characteristics in different frequency regions.

Filters

The basic electronic structure that causes gain to vary with frequency is the filter. Filters are known by their effect on signals:

- **High-pass filters** provide more gain to high frequency sounds than to low frequency sounds, which gives the sound a treble, or shrill quality.
- **Low-pass filters** provide more gain to low frequency sounds than to high frequency sounds, which gives the sound a muffled or boomy quality.
- **Band-pass filters** provide more gain to frequencies in a certain band than to either higher or lower frequencies.
- **Band-stop filters** provide less gain within restricted range of frequencies than for all other frequencies



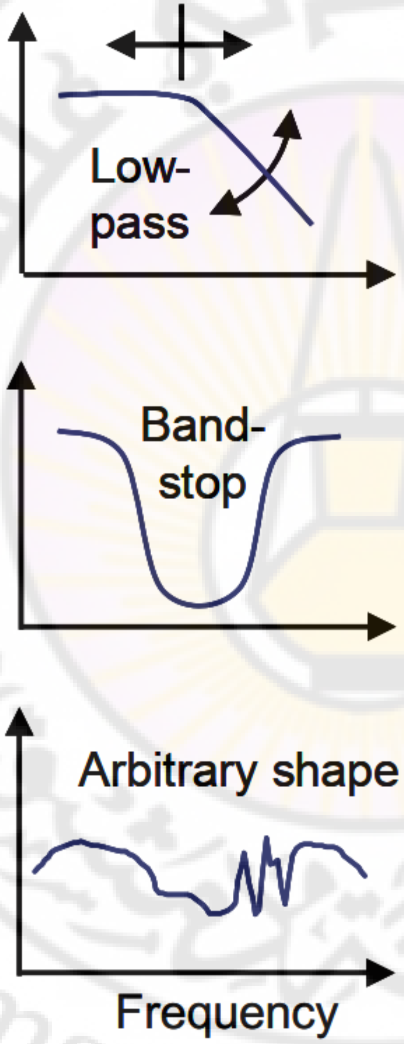
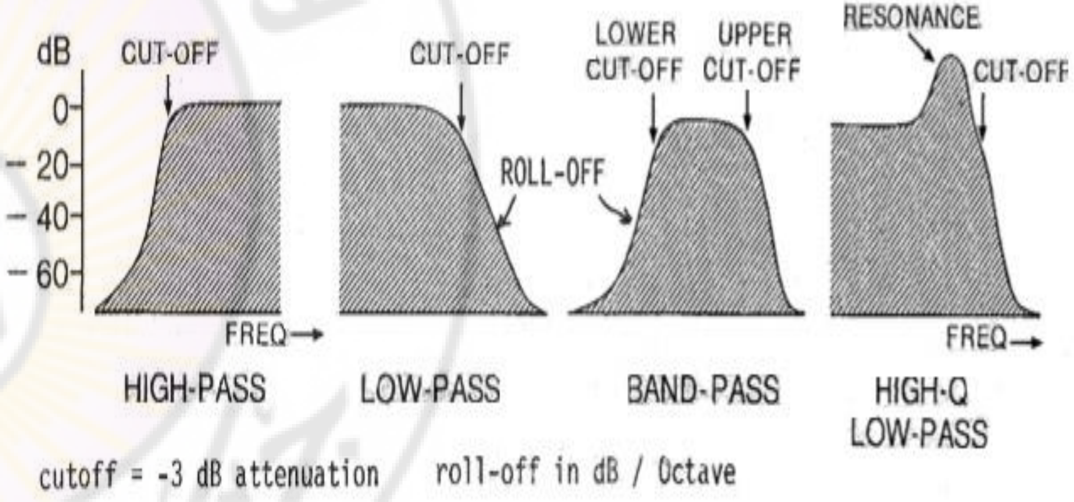


Figure 2.15 Gain frequency responses of high-pass, low-pass, band-pass, band-stop, and arbitrary shape filters or tone controls.



Different types of digital filters

- Nowadays, filtering in hearing aids is achieved by mathematical manipulations while the signal is in digital form, and therefore represented by numbers.
- One type of these filters is **finite impulse response (FIR).** In this filter, an output sample is calculated by adding a fraction of the current input sample to suitable fractions of each of the previous n samples, where n is the length of the filter.
- another type is **infinite impulse response(IIR)**. the output sample at a given time is also made to depend on the *output* samples at previous times. Every input signal will therefore have an effect on the output that theoretically lasts forever (though its effect continuously and rapidly gets smaller with time).
- IIR is better than FIR, because they have less time delay.
- Their disadvantage is that they can become unstable and oscillate, or alter the signal in unwanted ways, if

the precision of the computations is not sufficiently high

Tone controls

- They cause the gain of the amplifier to vary with frequency. Tone controls get their name because they affect the tonal quality, or timbre, of sounds passing through them.
- Tone controls are the same as adjustable filters.
- A high-pass filter, for example, can have its response varied by changing the corner frequency (also called the cut-off frequency) of the filter, or by changing the slope of the filter. Slopes of filters are commonly integer multiples of 6 dB per octave (e.g. 6, 12, 18, 24 dB per octave).

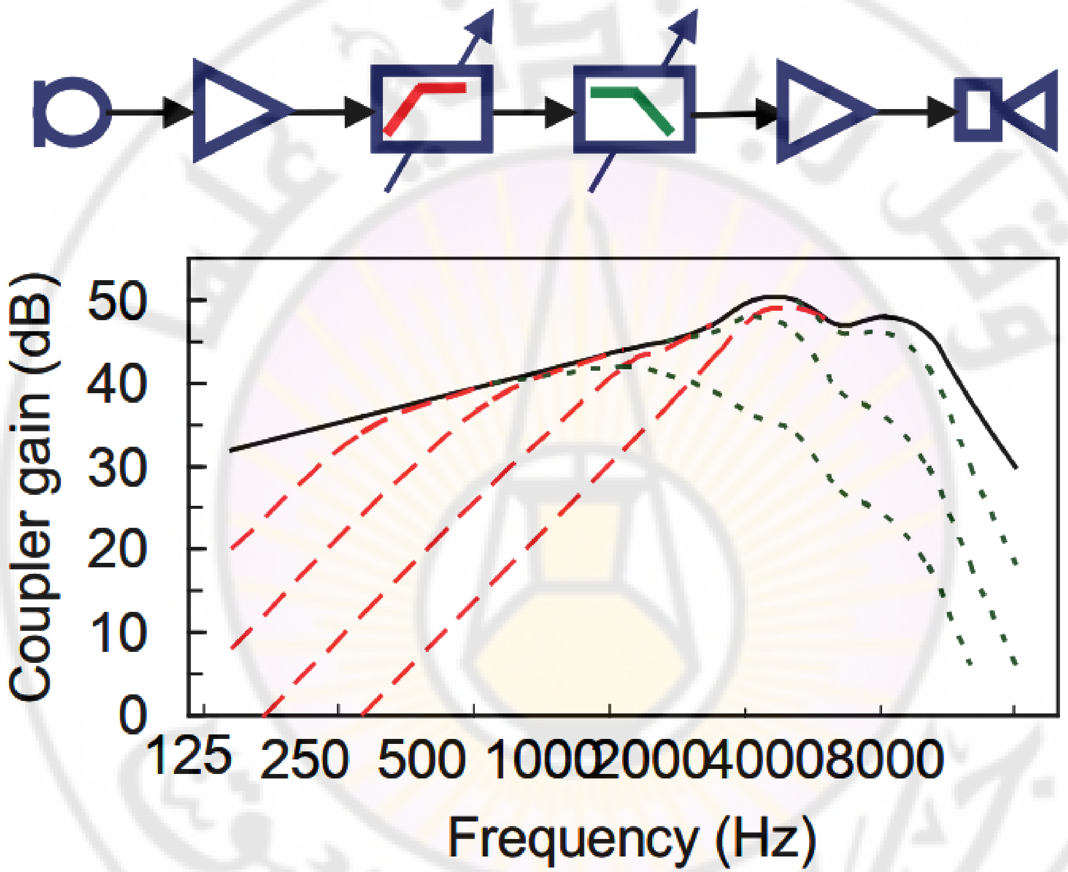
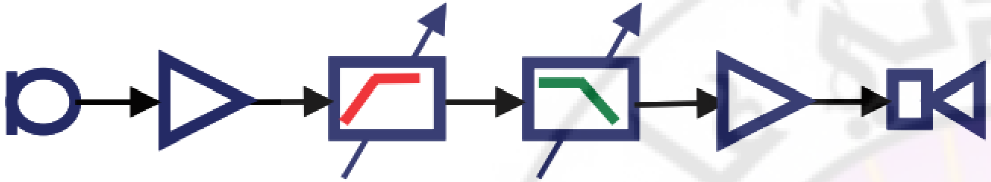
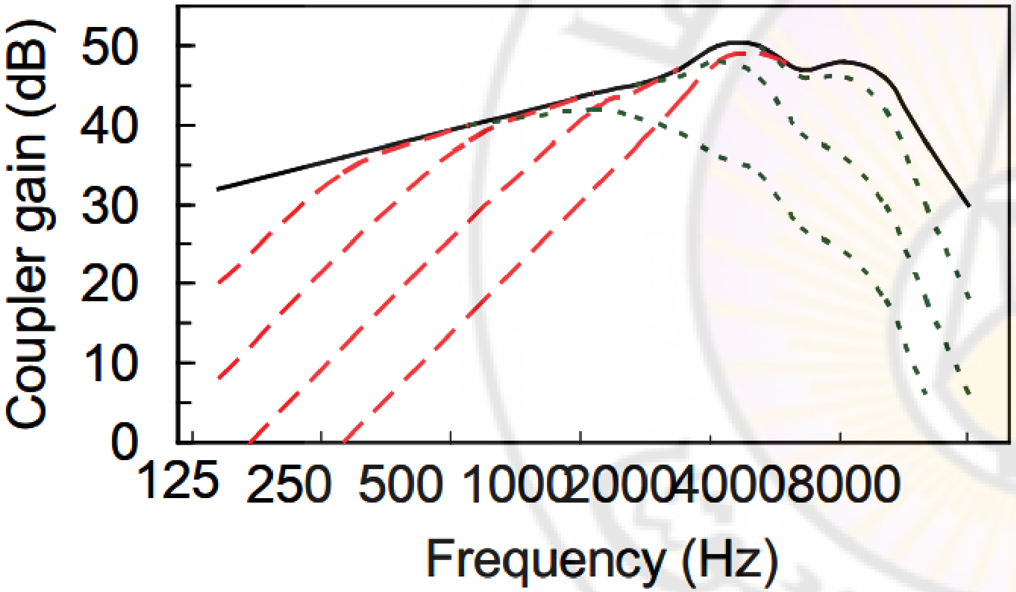


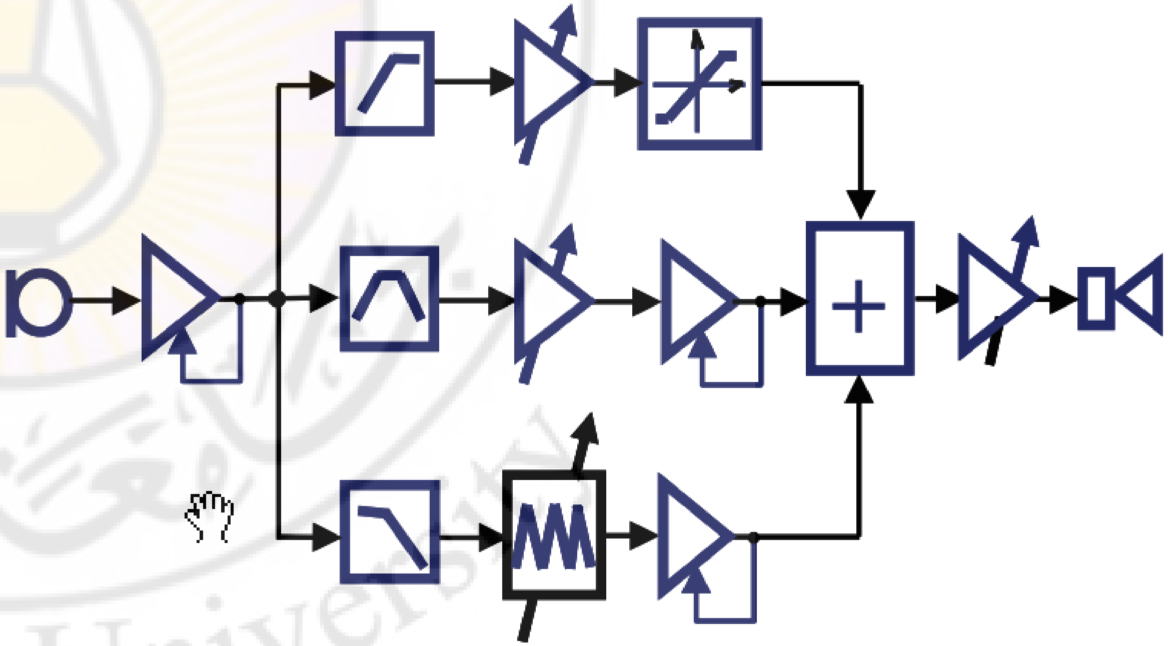
Figure 2.16 Block diagram of a serial structure, single-band hearing aid, and a range of low cut (dashed red curves) and high cut (dotted green curves) variations that might be made to the basic response (solid line).

Filter Structure

- ❑ serial structure: all sounds pass through all the blocks, one after the other. This arrangement was common in analog hearing aids, but is little used now.
- ❑ parallel structure: The filters divide the sound into adjacent frequency regions. These are variously called bands or channels. Parallel structures are simple conceptually, as sound in each frequency region can be amplified (or changed in various other ways) more or less independently of sound in other regions.
- ❑ After sounds are amplified in each band, parts are recombine in the adder.




Coupler gain (dB)
0
10
20
30
40
50
125
250
500
1000
2000
4000
8000
Frequency (Hz)



Filters Structures

- Serial-parallel structure: this method hasn't distortion and is the best method for filtering.
- The parallel bank of band-pass filters (or equivalently a Fourier transform) is used to determine the level and other characteristics of the signal present within each channel. These characteristics are then used to determine the a serial filter characteristics. If the serial filter is an FIR filter, it is relatively easy to ensure that the time and frequency distortions referred to do not occur

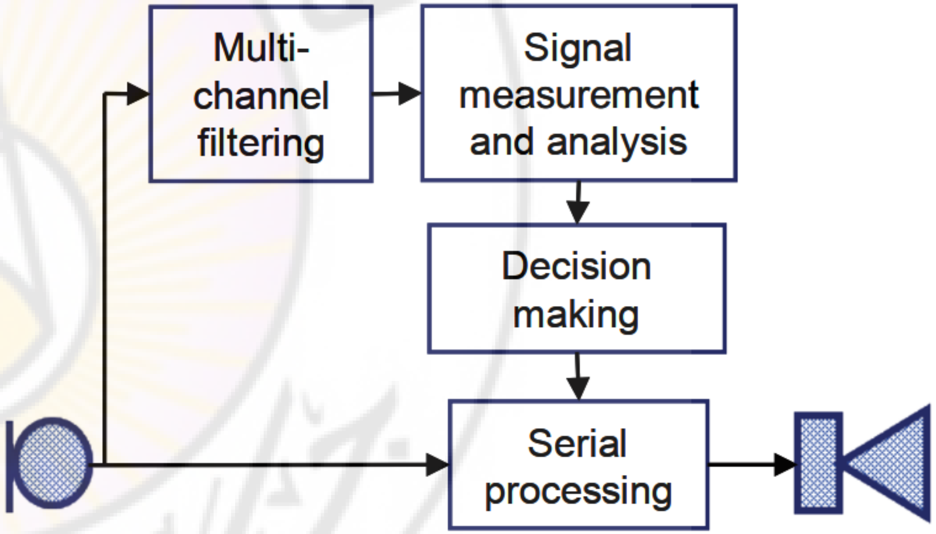


Figure 2.17 Series signal flow and parallel analysis structure used in some hearing aids. Modification of the signal occurs only in the serial path.

Multi-channels or Multi-bands in hearing aids

- The terms multi-band and multi-channel are usually used interchangeably, although some authors and hearing aid companies differentiate between them.
- Gain versus compression.

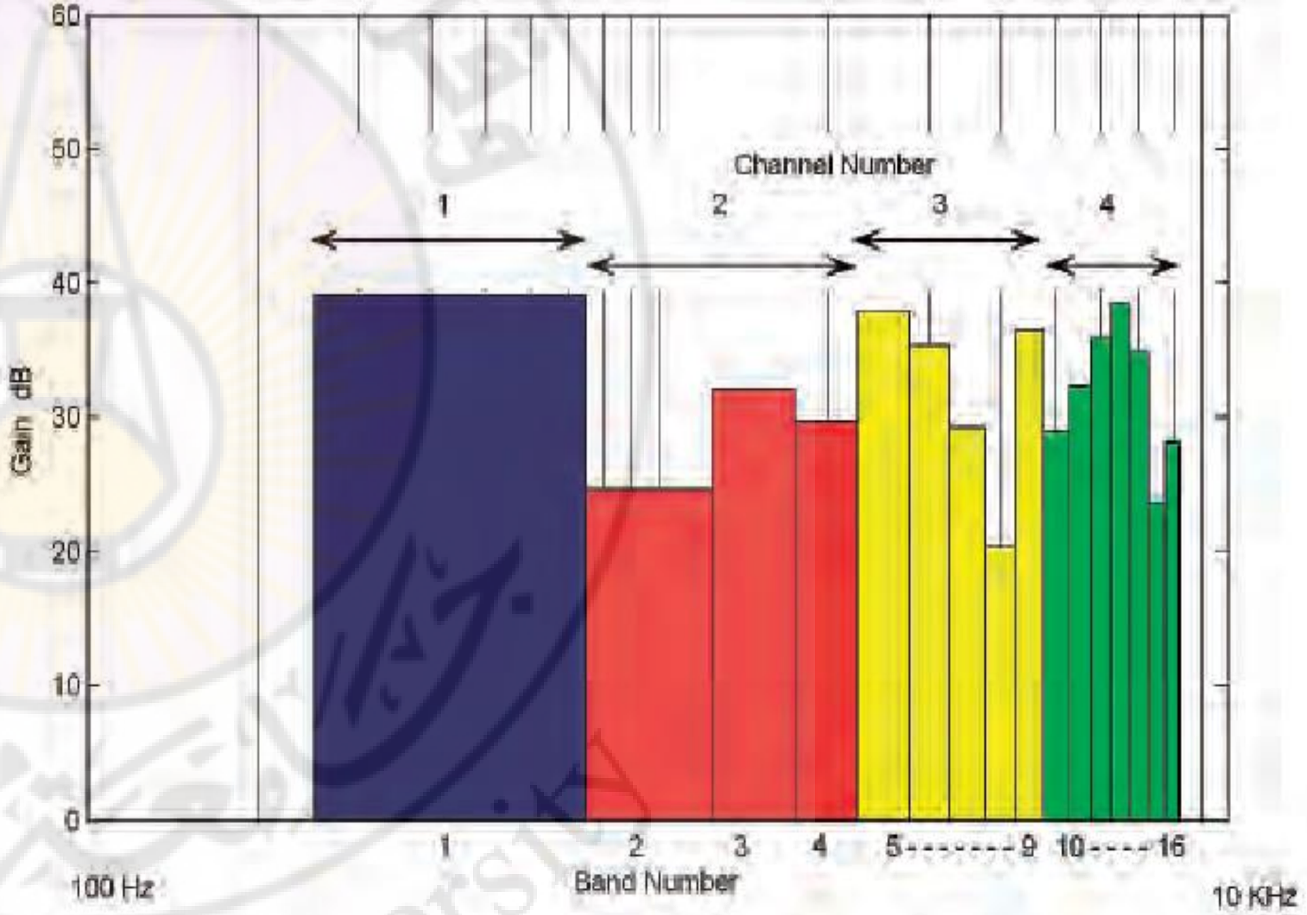


Fig. 1. A graphic representation of channels and bands within a digital hearing aid.

How many channels is good for hearing aid?

Studies suggest that as little as 8 channels are sufficient to restore audibility, even in environments where background noise is present.

Veterans Administration

Journal of Rehabilitation Research and Development Vol. 24 No. 4
Pages 161–180

Speech discrimination with an 8-channel compression hearing aid and conventional aids in background of speech-band noise*

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Abstract—The design for a multichannel compression hearing aid was developed from previous experimental and theoretical work in our laboratory concerning pitch perception in normal-hearing subjects. The new hearing aid, implemented with off-line digital signal processing, was tested on twenty subjects with sensorineural hearing loss using speech sounds in a background of speech-spectrum noise. Five signal-to-noise ratios (+15 to −5dB) were used at two noise levels (60 and 70 dB SPL). Hearing-loss subjects listened to these stimuli under three different conditions: a) processed by the new multichannel compression hearing aid; b) processed by a conventional hearing aid; and c) unprocessed. The performance of normal-hearing subjects with the unprocessed stimuli provided another condition against which the performance in the two hearing aid conditions could be evaluated. Both aided conditions provided improved performance over the unprocessed condition and the multichannel compression aid produced better performance than the conventional aid. In the case of 4 of the 20 subjects, with less severe gradually sloping hearing losses, the new multichannel compression aid produced near-normal performance even at low signal-to-noise ratios. Some aspects of the results also suggested that learning to use the aid was more important in the case of the multichannel compression aid than in the case of the conventional aid. These results indicate that a multichannel compression hearing aid can be very effective in some individuals with sensorineural hearing loss and is superior to a conventional hearing aid in most subjects.

*This work was supported by the VA Rehabilitation Research and Development Service. A preliminary report of this work was given at the 12th International Congress on Acoustics (26).

**Presently with the Smith-Kettlewell Eye Research Institute, San Francisco, CA.

INTRODUCTION

A multichannel compression hearing aid is one in which the amplification in each frequency band, or channel, decreases as the amplitude of the signal in that channel increases. The dependency of amplification upon amplitude also is usually different for the various channels. Using such an aid, an individual with sensorineural hearing loss (SNHL) could exhibit both normal thresholds and normal discomfort levels for narrow band stimuli and, in that very limited sense, may be said to exhibit "normal hearing." It is this obviously oversimplified view of multichannel compression that originally may have made it seem so promising and perhaps that still makes us continue to try to demonstrate its presumed superiority over simple linear amplification.

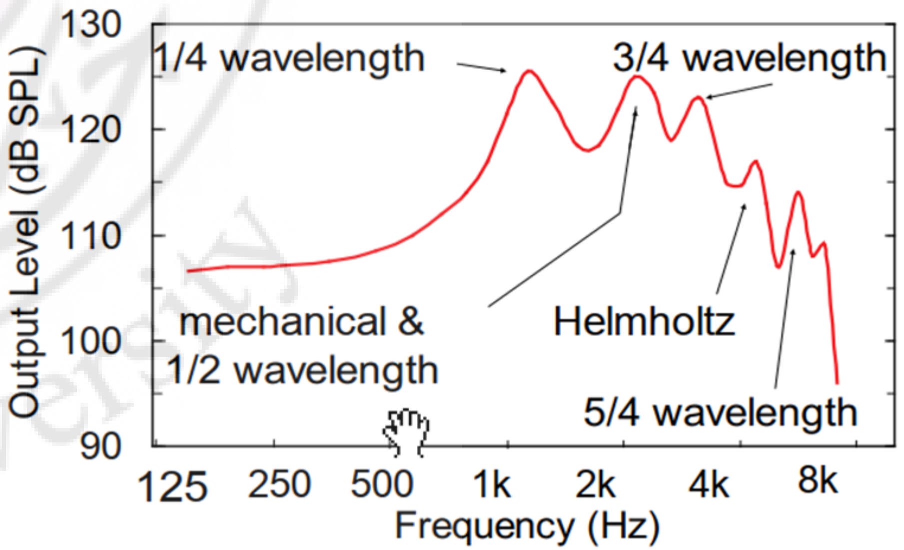
In one sense, however, such a view of multichannel compression cannot be dismissed as completely naive. Consider the pattern of neural signals, for some particular stimulus (either simple or complex to any degree), traveling up the auditory nerve in a normal-hearing subject. If it were possible to modify this stimulus in such a way that the modified stimulus produced the normal pattern of neural signals in the auditory nerve of a patient with cochlear hearing loss, then the patient should exhibit "normal hearing" for that stimulus, however complex it might be. A device that could "normalize" the auditory nerve signals for all possible stimuli for a particular SNHL patient, would be the "perfect" hearing aid for that patient. Although it may never be possible to make this "perfect" hearing aid, our current

Acoustic Dampers

Does it matter if the receiver response has bumps and dips, and does it matter what causes them?

the answer:

YES



NOW, what should we do to eliminate of bumps and dips in frequency response of receivers?

- Placing an acoustic resistor, also called a damper, in the tubing at an appropriate place decreases the peaks. One type of damper consists of a fine mesh.
- a damper will decrease the receiver output most at the resonant frequencies, but only if the damper is placed in an appropriate place.

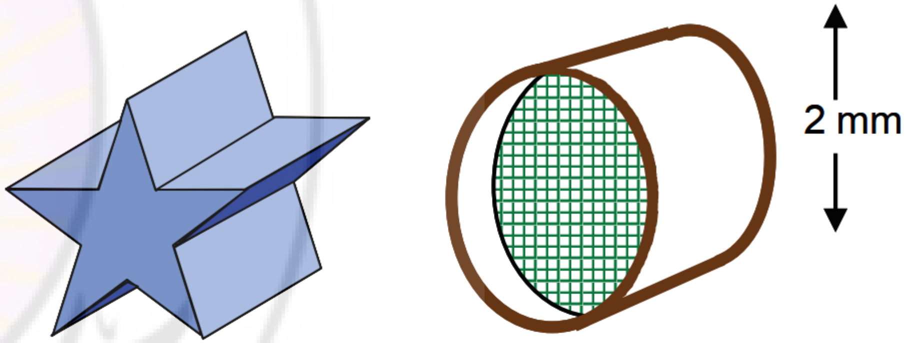


Figure 2.21 A star damper and a fused-mesh damper that can be inserted inside #13 tubing of internal diameter 1.93 mm.

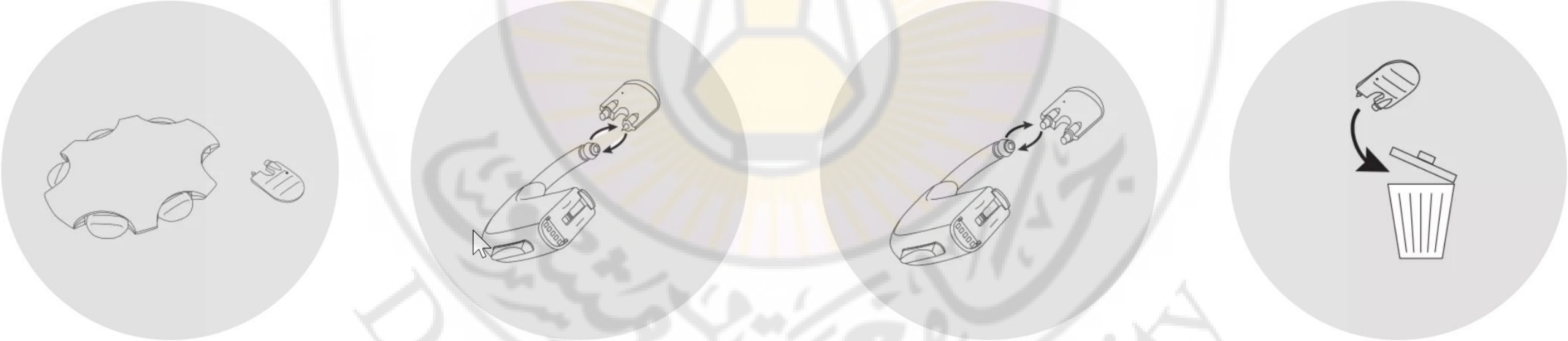
What is the dampers?

The damper works as an acoustic filter, shaping the frequency response, but also helps to stop moisture and earwax from getting into the hearing aid. If the sound output is reduced or there is no sound from the hearing aid, this may have been caused by a clogged damper and it should be replaced.

Dampers

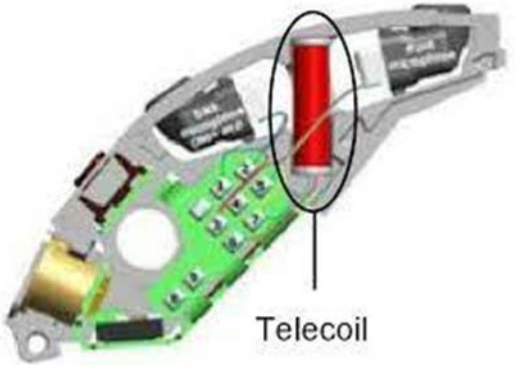
How to replace dampers?

Replace the damper as shown:



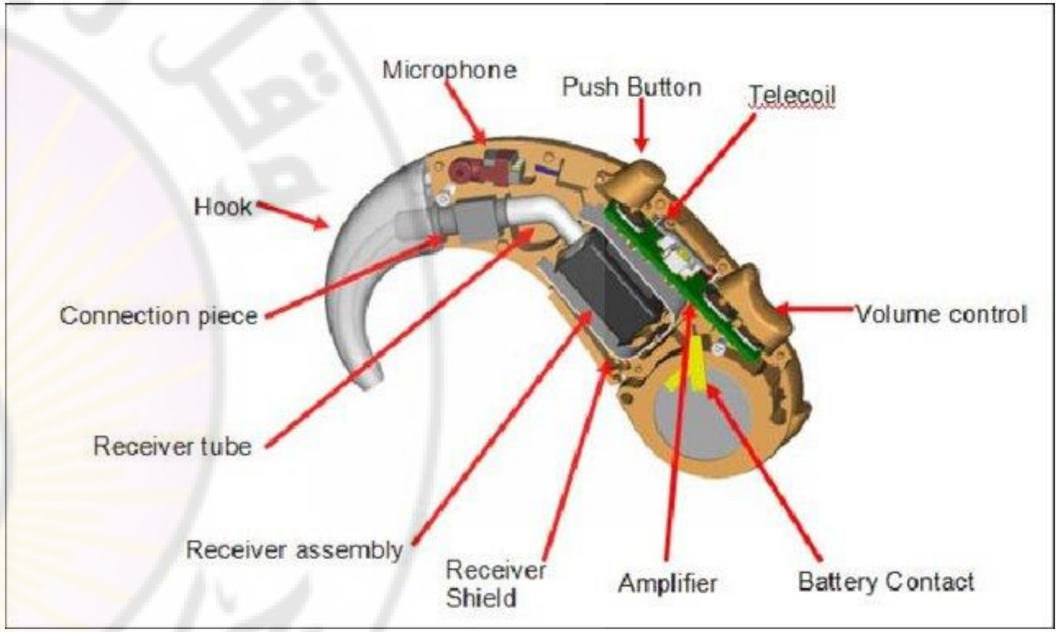
Telecoils

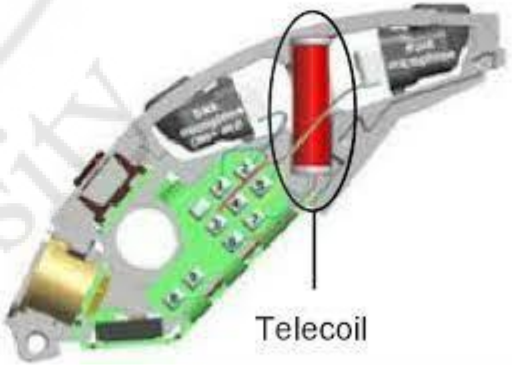
- ❖A telecoil is a small coil of wire that produces a voltage when an alternating magnetic field flows through it.
- ❖The magnetic field to be picked up by the telecoil is generated by an electrical current that has the same waveform as the original audio signal.
- ❖This magnetic field may occur as a by-product of some device, such as from a loudspeaker or a receiver in a telephone, or may be generated intentionally by a loop of wire around a room or other small area



Telecoils

- Not all hearing aids include a telecoil, although nearly all high-powered BTE hearing aids and many other BTE and ITE hearing aids do.
- When the hearing aid is switched to the T program and the input comes from a room loop, the only sounds amplified will be magnetic signals reaching the hearing aid





Audio Input

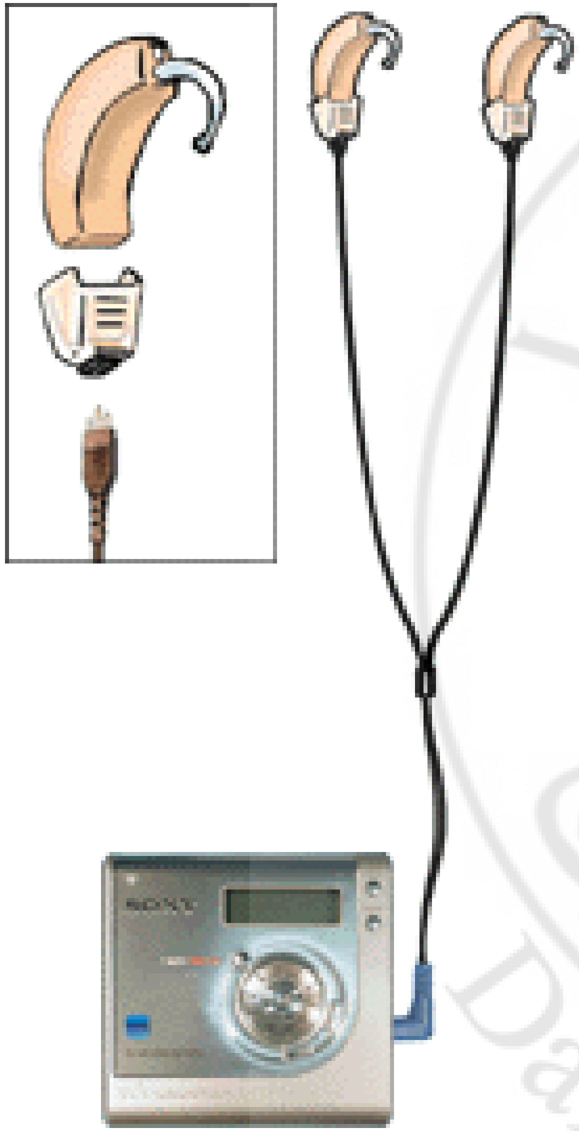
❖An alternative way to get an audio signal into a hearing aid is to connect it via an electrical cable. This is referred to as direct audio input (DAI).

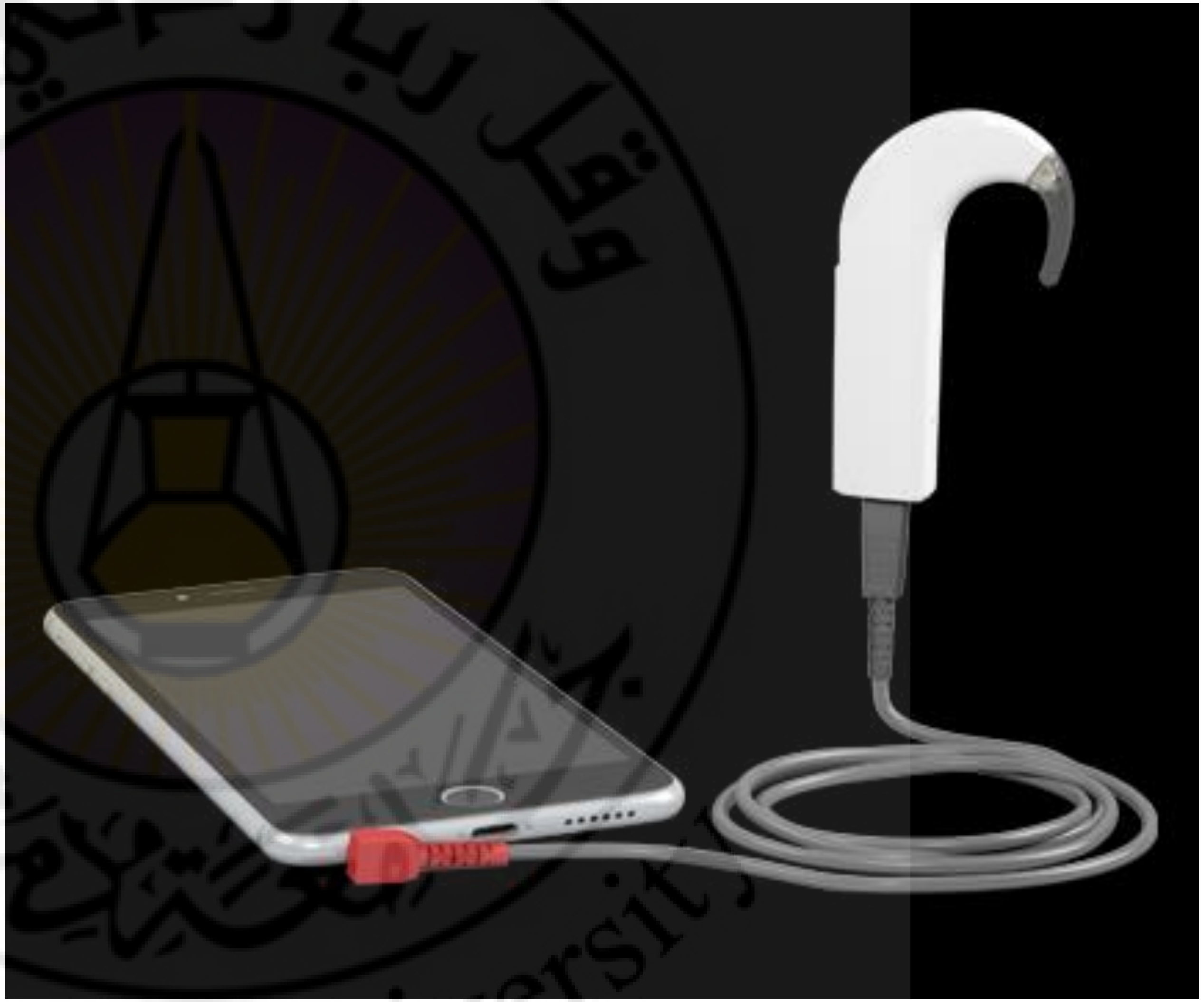
❖The electrical audio signal may have originated from equipment such as an MP3 player, a hand-held microphone, or an FM wireless receive.

❖ in this way, there isn't any reverberation or noise from the environment and SNR is the best.

❖The electrical current in this way is the same with the current connected to microphone, it means 1mA.

❖The patient can select ADI via: pushbutton, physical switch and remote control and automatically by DAI connector.





Remote controls

Remote Controls

Various methods of transmission are used in hearing aids currently available.

- ❖ Infrared
- ❖ Ultrasonic
- ❖ Radio wave
- ❖ Magnetic induction

A concern that is sometimes raised about the use of remote controls is their potential to interfere with Pacemakers.

Table 2.2 Advantages of different remote control technologies. *Interference* refers to interference of remote control operation by other devices.

	Ultrasonic	Infrared	Radio Waves	Magnetic Induction
Freedom from interference		✓		
Operated from any position			✓	✓
Simultaneous bilateral operation			✓	✓
Simple technology	✓	✓		✓

Air and bone conduction hearing aids

- Bone conductors are an alternative output transducer intended for people who, for various reasons, cannot wear a receiver coupled to the ear canal. Bone conductor transducers directly vibrate the skull, which via several transmission paths, transmits these vibrations to the cochlea.
- Disadvantages include:
  - high pressure into brain
  - small gain
  - lower quality than air conduction

Different shape of bone conductor hearing aids

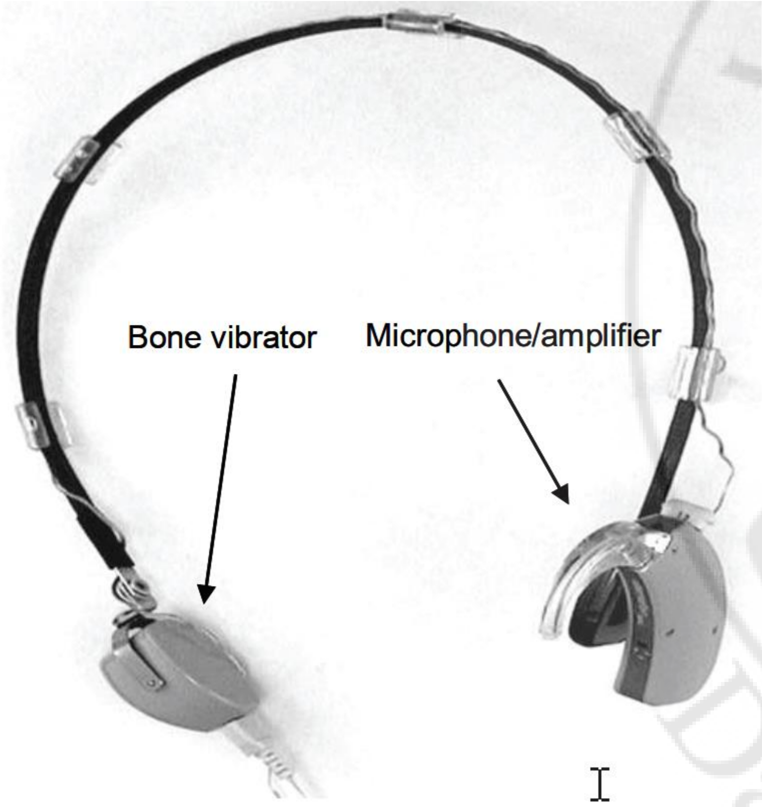
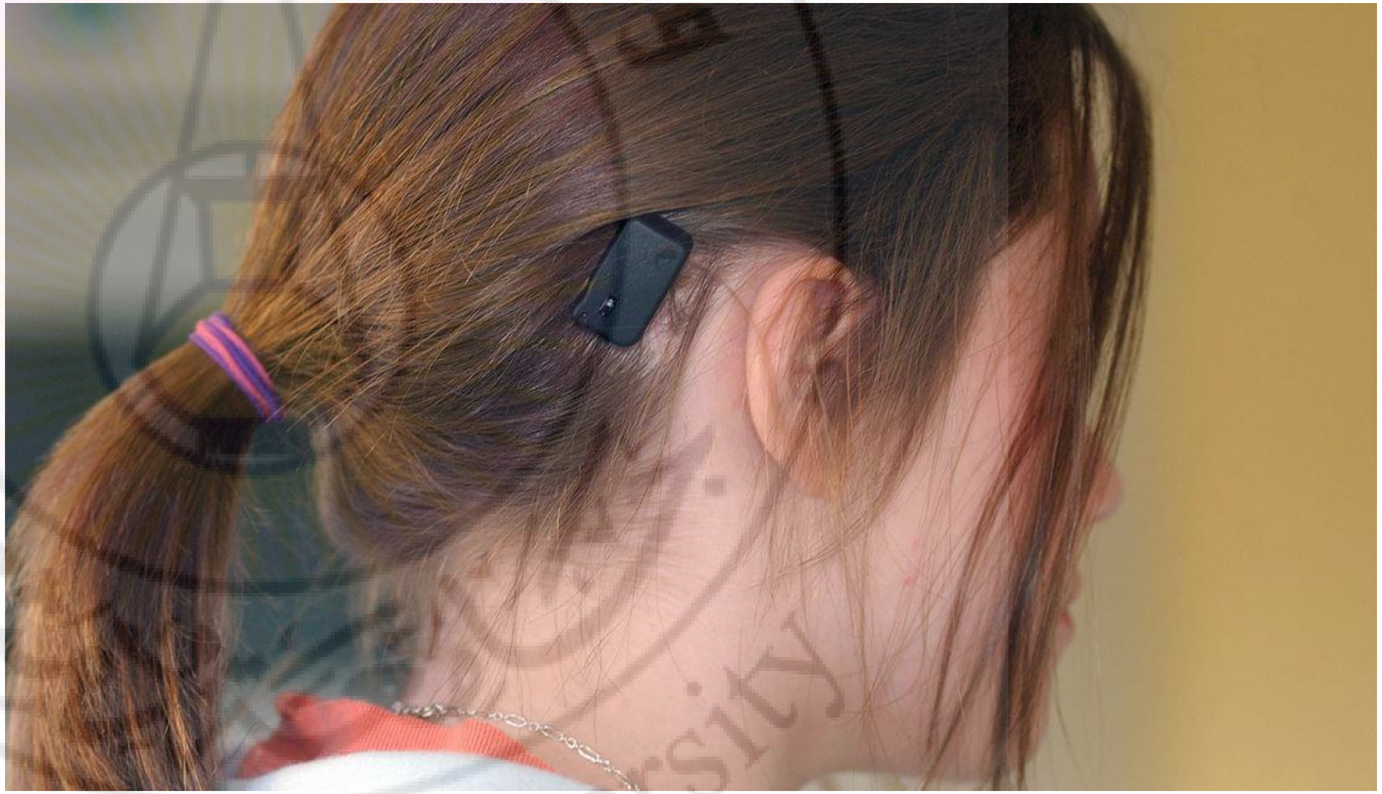


Figure 2.23 A bone conductor hearing aid



BAHA hearing aids with operation


MED°EL
ADHEAR
Bone Conduction Hearing System
- Adhesive Non-Surgical Solution
- Avoids Pressure on to Skin

Batteries

The important characteristics of the battery are its voltage, its capacity, the maximum current it can supply, its electrical impedance and its physical size.

- Voltage: The voltage generated by a battery depends solely on the type of materials used for the electrodes. The batteries most commonly used for hearing aids use Zinc and Oxygen as their negative and positive electrodes, respectively, so the batteries are known as Zinc-air batteries.

Batteries

❖ Capacity and physical size: Bigger batteries last longer than smaller batteries with the same chemistry. The electrical capacity of a battery is measured in milliamp hours (mAh). A battery with a capacity of 100 mAh, for example, can supply 0.5 mA for 200 hours, 1 mA for 100 hours, or 2 mA for 50 hours.

❖ If the current gets too high, even for a fraction of a second, the battery voltage will drop excessively because of the internal resistance of the battery

❖ Bigger batteries can generally supply bigger maximum currents, as well as having a larger mAh capacity. AA and AAA batteries have typical capacities of 2000 and 800 mAh, respectively.

❖ HP (High Performance or High Power) batteries, may have the prefix H in their name, are suitable for high power hearing aids.

Batteries

Peak-current demand can be assessed in a test box equipped with a battery pill, by applying a 500 Hz signal at 90 dB SPL with a high volume control setting. For open hearing aids, 2000 Hz signal is used.

Table 2.3 Names and typical capacities of zinc air batteries of various sizes and the hearing aid styles in which they are most commonly used.

Type	Standard Label	Capacity (mAh)	Hearing aid types
675	PR44	600	BTE
13	PR48	300	BTE, ITE
312	PR41	175	BTE, ITE, ITC
A10 (or 10A, or 230)	PR70	90	BTE, CIC
A5	PR63	35	CIC

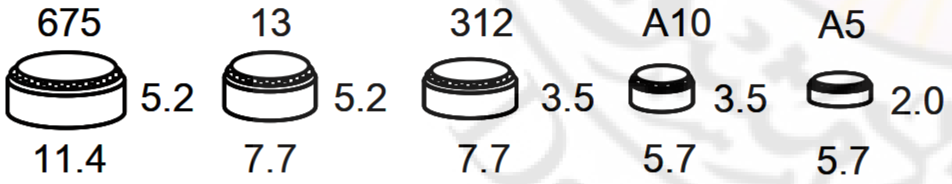


Figure 2.24 Hearing aid batteries of various types drawn full size, with typical dimensions shown in mm. Minimum and maximum allowable dimensions are 0.1 to 0.2 mm smaller and larger than these dimensions.

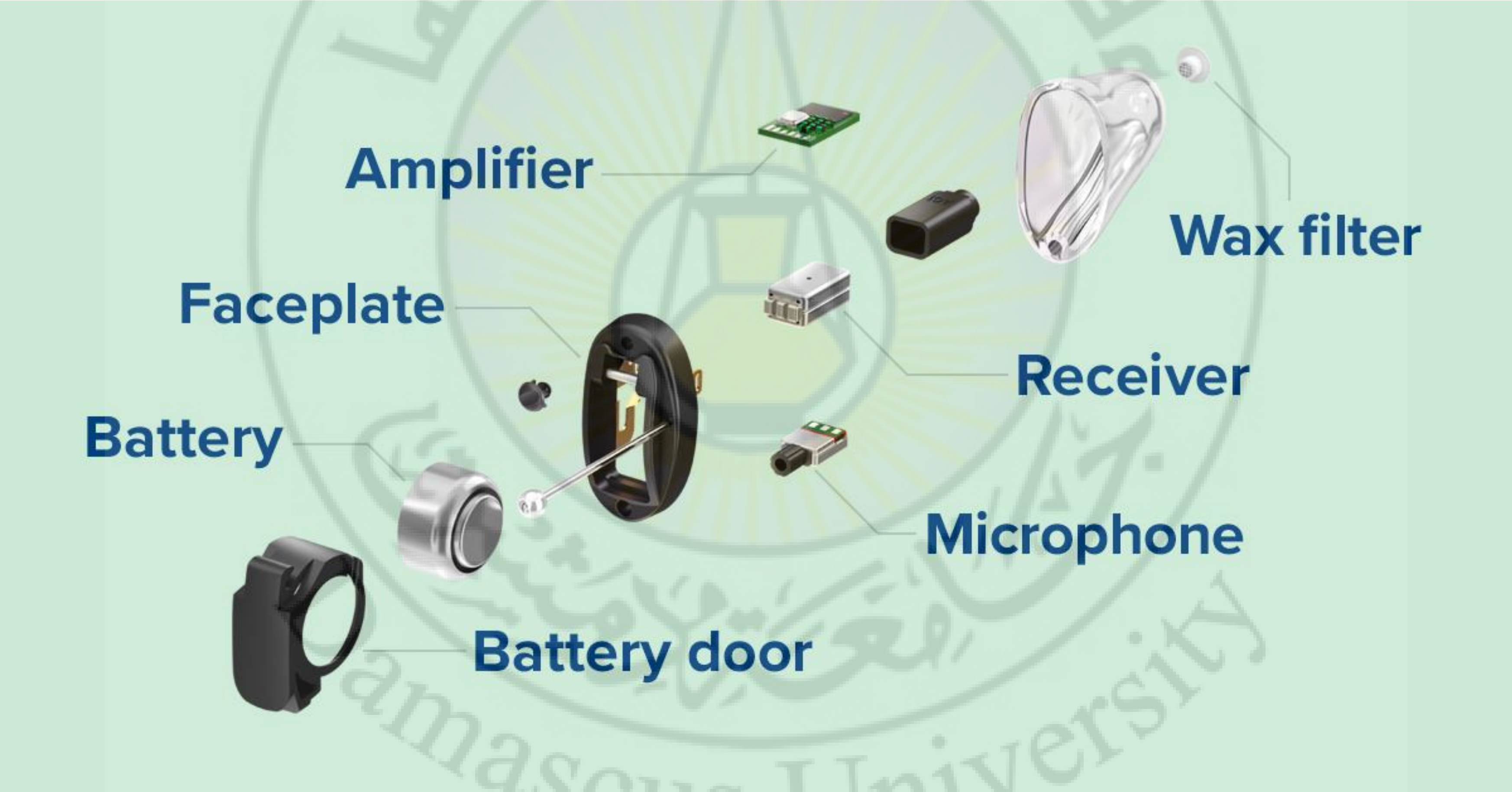
What is the green battery?

In the past, all zinc-air batteries contained small amounts of mercury, but mercury-free varieties are increasingly available and may colloquially be called green batteries. Because mercury is very hazardous for nature.

Rechargeable batteries……….

- Some manufacturers offer hearing aids with rechargeable batteries. Their major advantage is increased convenience from not having to change batteries.
- Rechargeable cells can be discharged and recharged several hundred times, so the battery only has to be replaced every one to three years.
- These batteries are very suitable for elderly people or for people without fine manipulation skills.
- The major disadvantages of rechargeable cells is that their capacity is only around 10% of that of a non-rechargeable cell of the same size, so that recharging must be performed regularly, as often as every night,

Different parts of CIC hearing aids



Different types of hearing aids

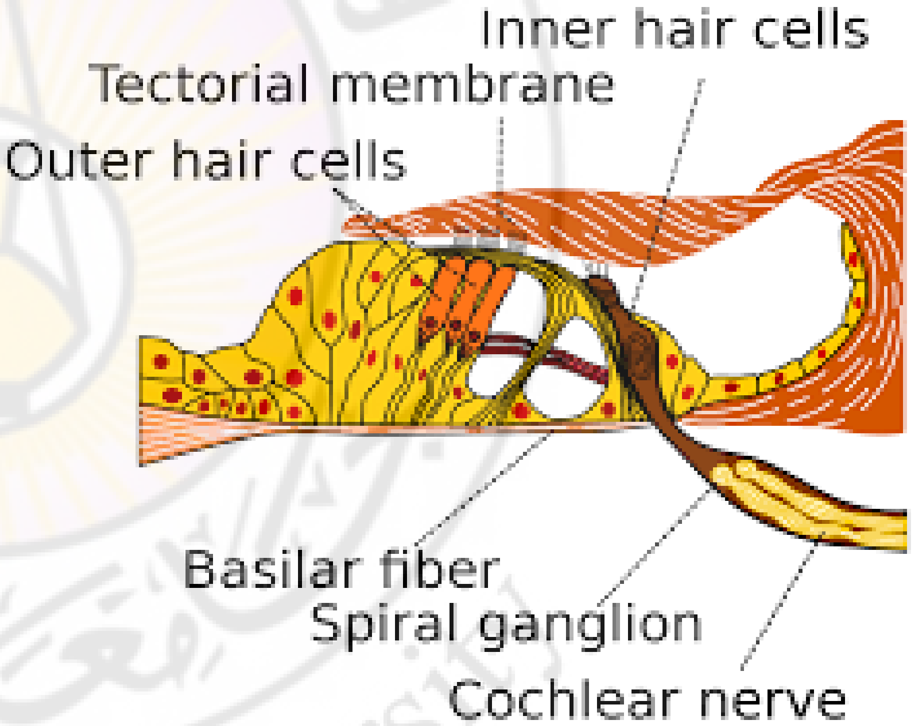
Advanced processing in hearing aids

Dr. Maryam Sadeghijam

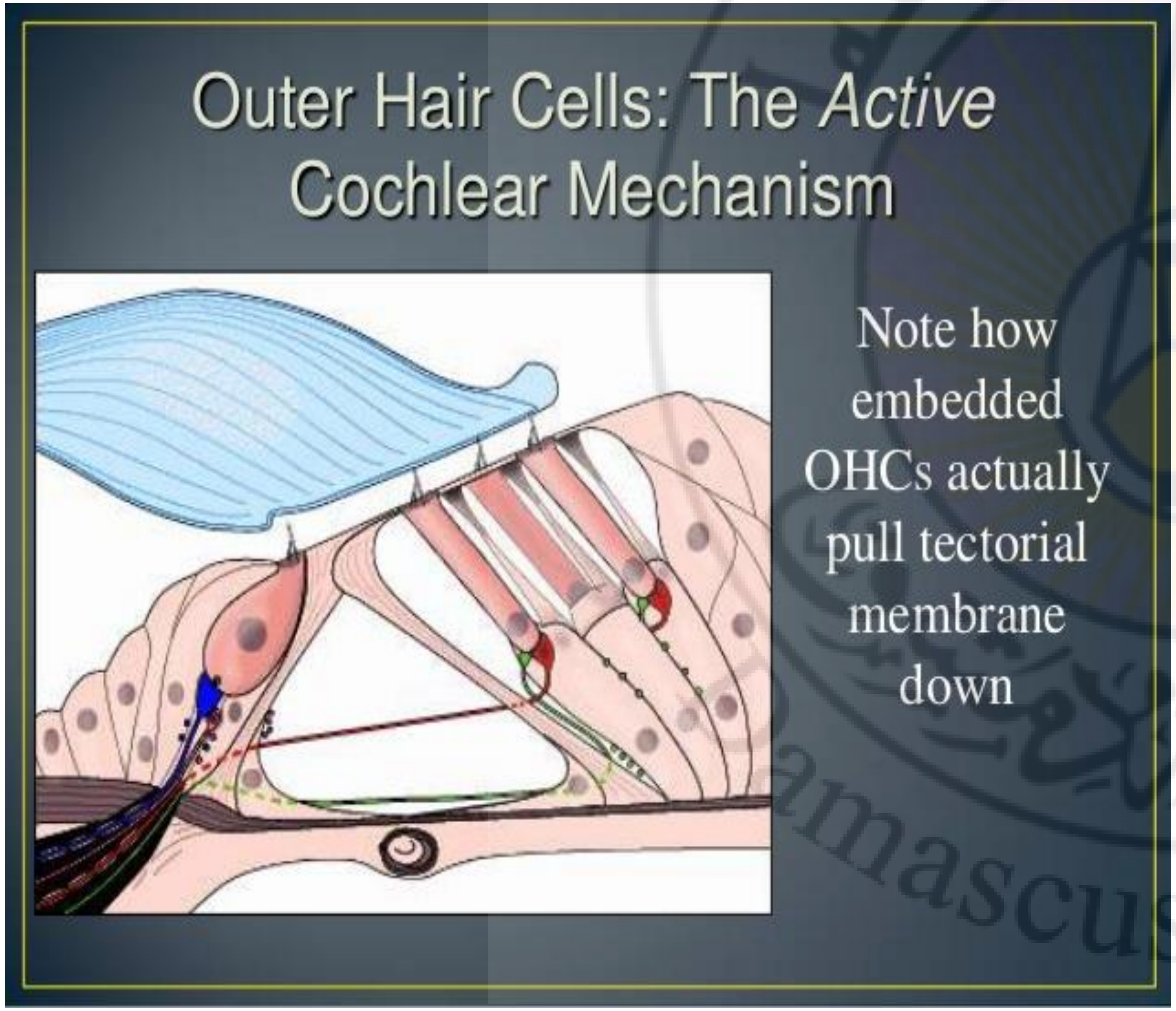
Assisstant professor of Iran University of Medical Sciences

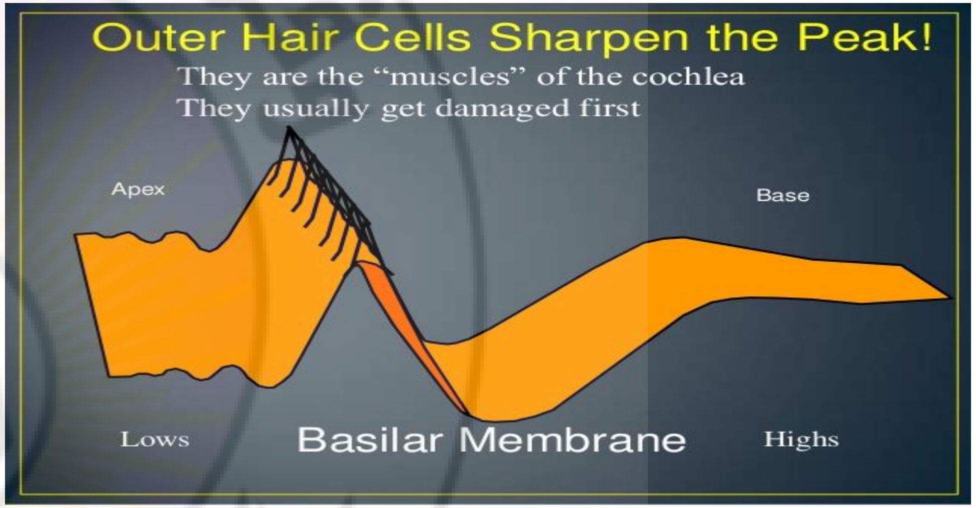
2323 May

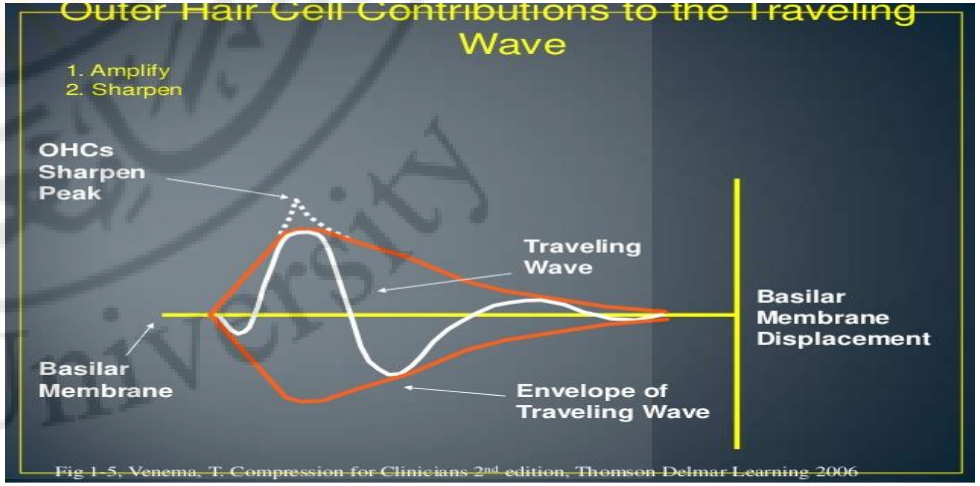
Amplitude Compression



OHCs and Hearing







HAIR CELL FUNCTION & PURPOSE

- In summary, outer hair cells have a twofold purpose:

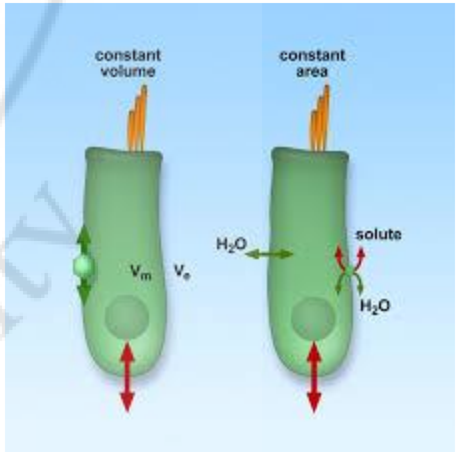
1. They amplify soft sounds (below forty to sixty decibels).
2. They “fine-tune” the frequency resolution of the basilar membrane.

Hearing Instruments ?

They don't grow new hair cells

HAIR CELL FUNCTION & PURPOSE

- **Hearing instruments cannot sharpen the peaks of the traveling wave. They will only increase the amplitude (size) of the wave.**
- **With more outer hair cells missing less resolution may be received by the brain—especially when noise is introduced.**

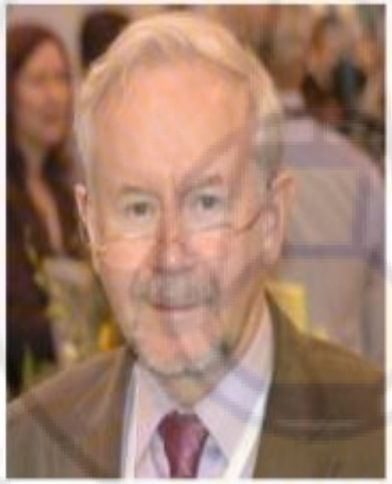


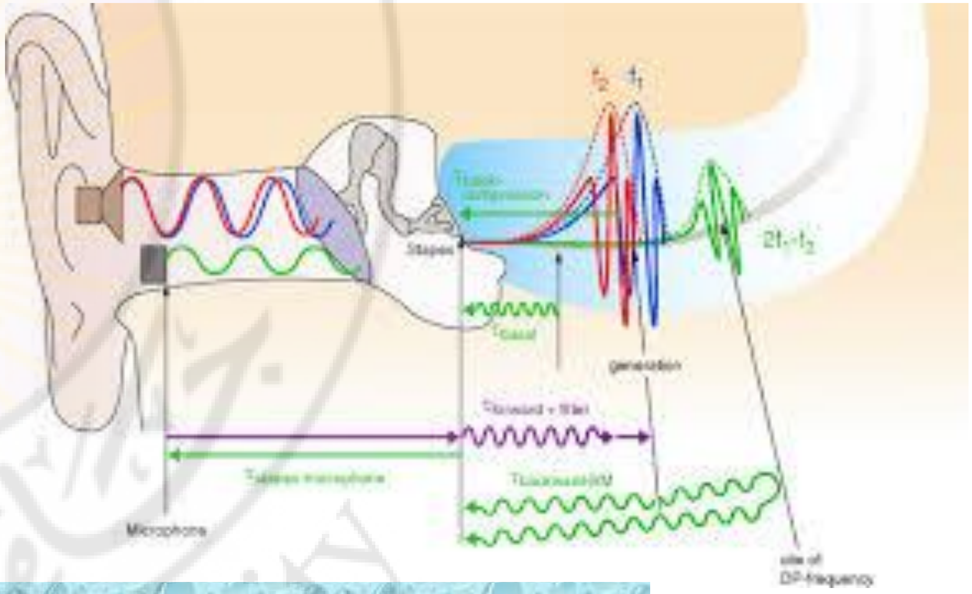
OAE and OHCs

History

David Kemp

- British physicist
- discovered OAEs in 1977
- In 1988, purchased patent rights from British govt. and founded Otodynamics which produced the ILO88 OAE system





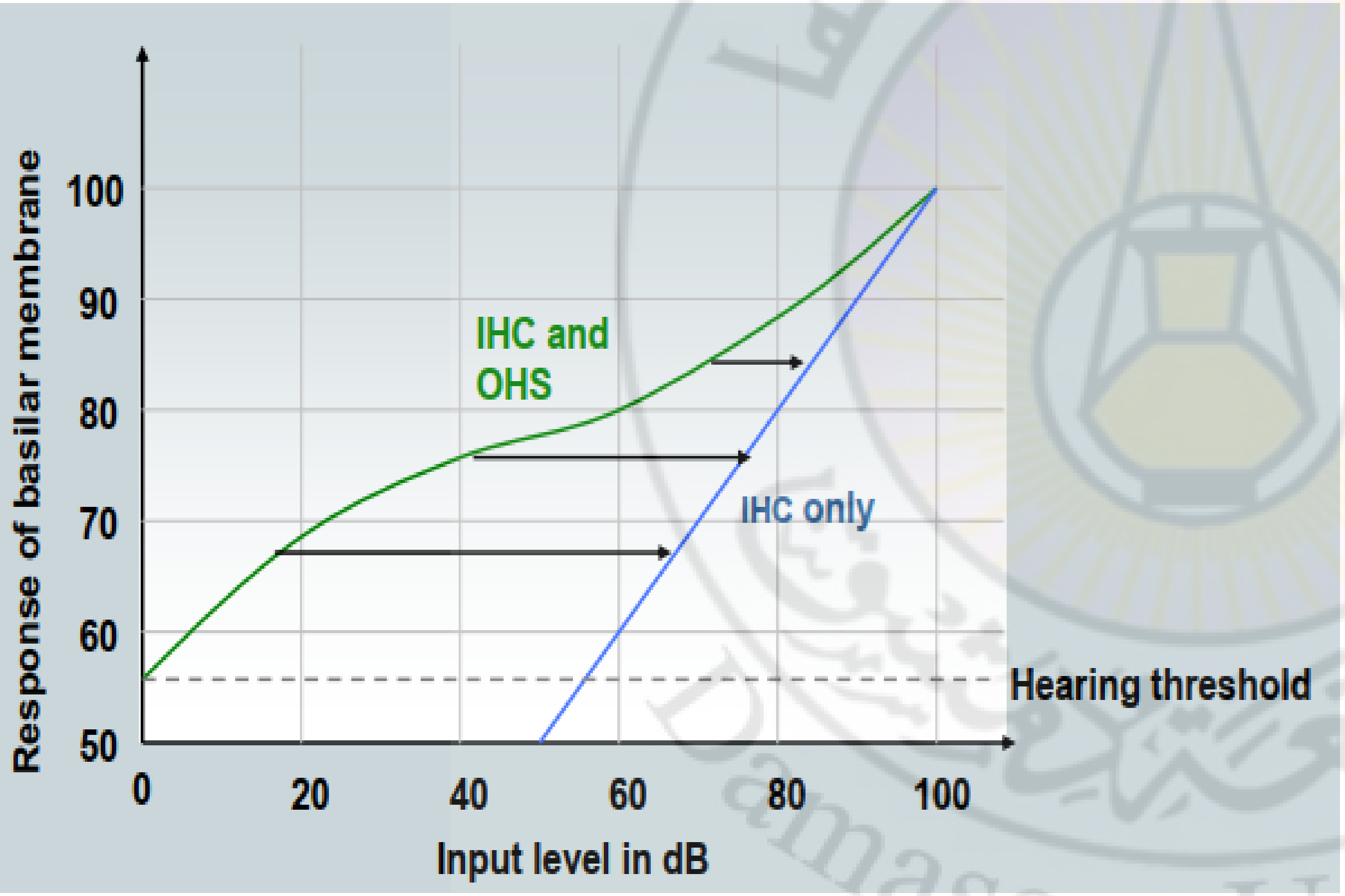
OAE: A revolutionary discover in the amplitude compression

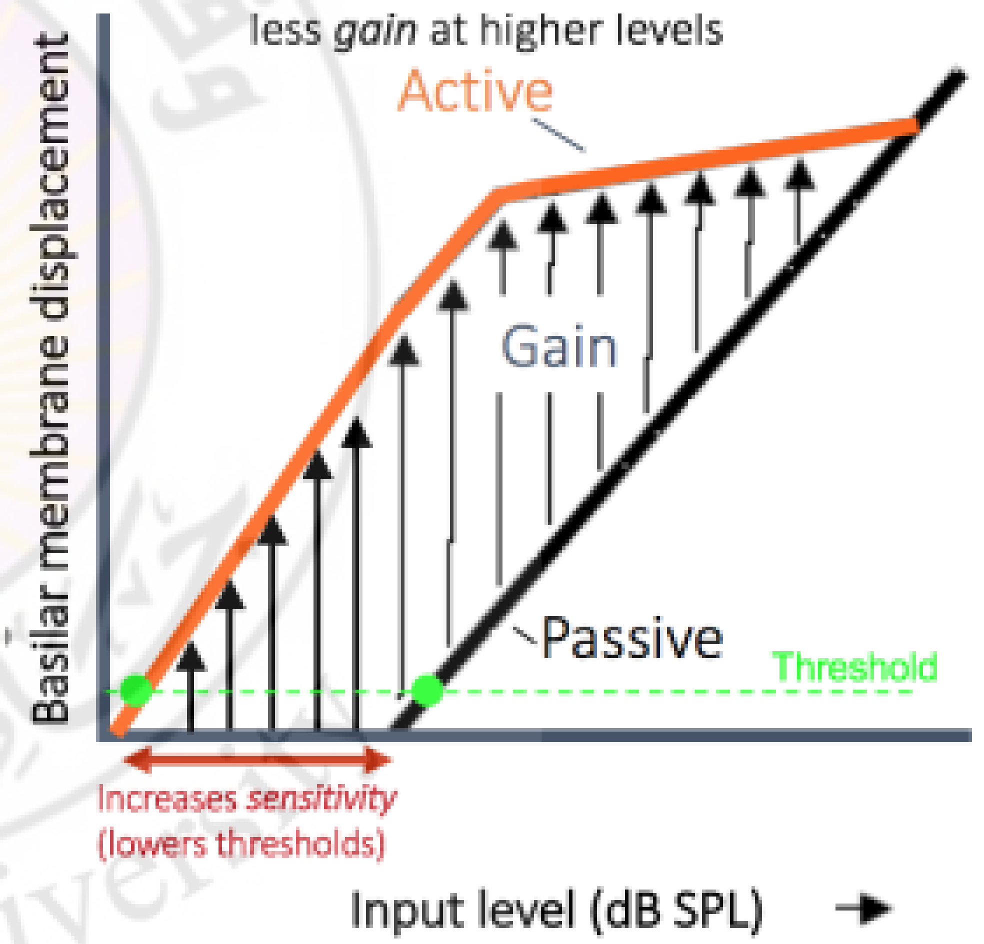
Cochlear function

- In low level sounds: OHCs (basilar membrane amplifier)
- In mid level sounds: IHCs
- In high level sound: compressive function of basilar membrane (protection of cochlea against loud and very loud sounds)

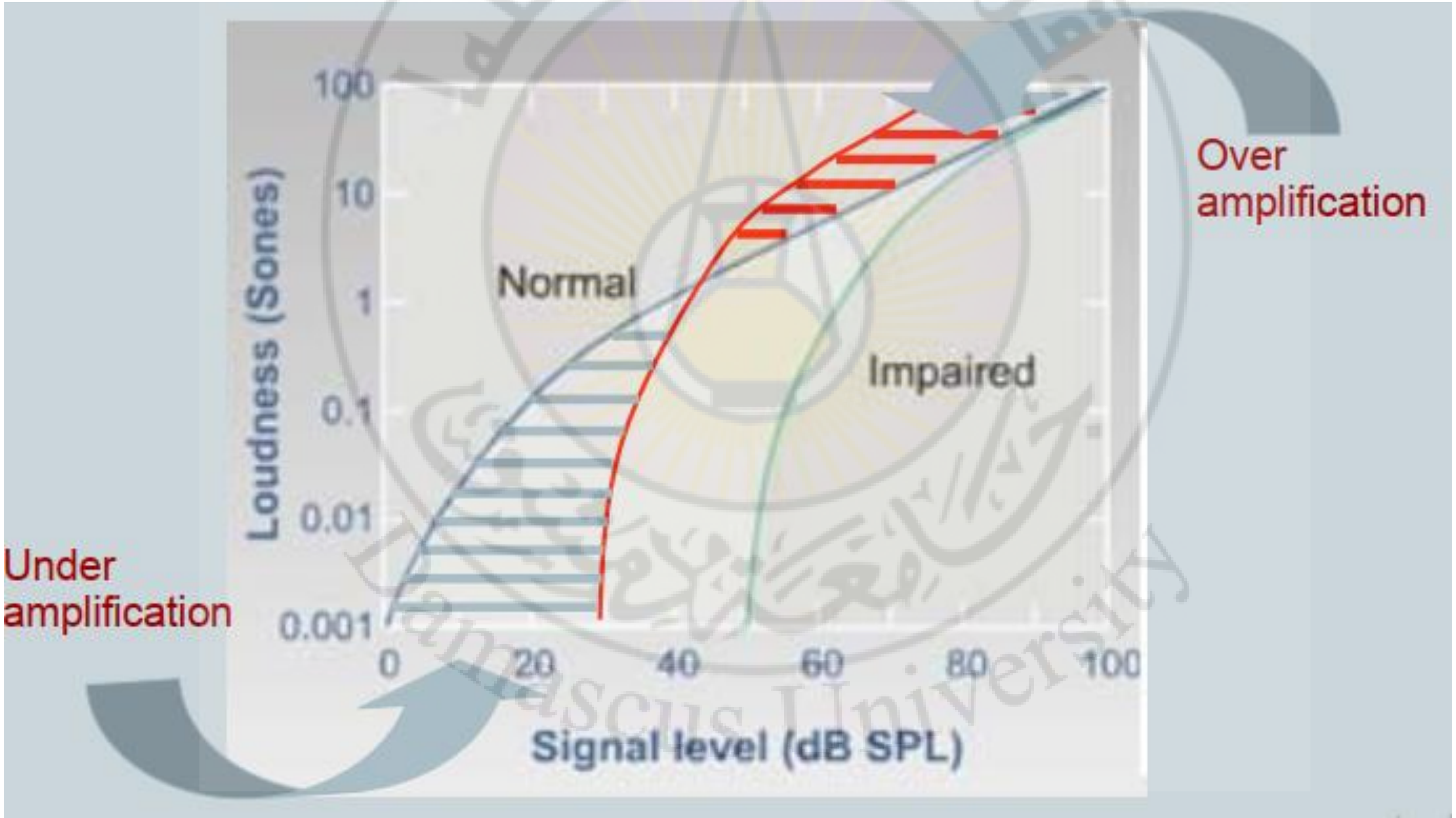
From a certain intensity, the basilar membrane will no longer an increase in travelling wave amplitude proportional to the increase in intensity.

Cochlear Function





Different Between Linear and Compression Amplification



Why Compression?
In Other Words……………….

In common single-channel linear hearing aid, it is possible to make virtually all soft speech sounds audible by turning up the volume control to full on (or the point just below where the hearing aid starts to feedback). All would be well for the patient for hearing soft sounds, that is, until a sound of average or loud intensity comes along, forcing the patient to turn the volume control wheel down, thus, making many soft speech sounds inaudible. Ensuring that the soft and medium intensity sounds of speech can be heard, while loud sounds are not too loud, is an essential requirement of all hearing aids.

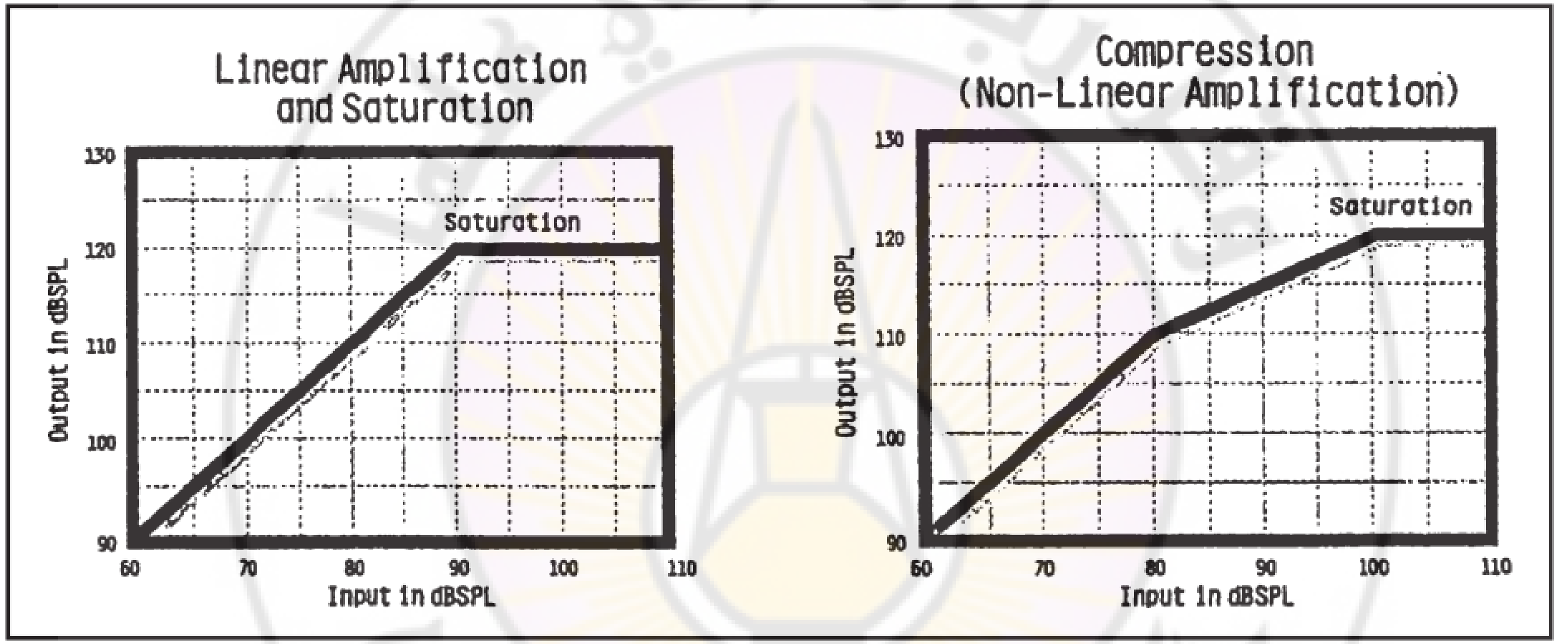


Figure 8–8. Input/output functions, along with saturation levels for a linear and compression hearing aid circuit. Notice on the graph on the right where the compression becomes activated 90 dB input). Most modern hearing aids begin compression for inputs around 40 dB SPL (80 dB input). Most modern hearing aids begin compression for inputs around 40 dB SPL.

For example………

Hearing threshold = 80 dB HL

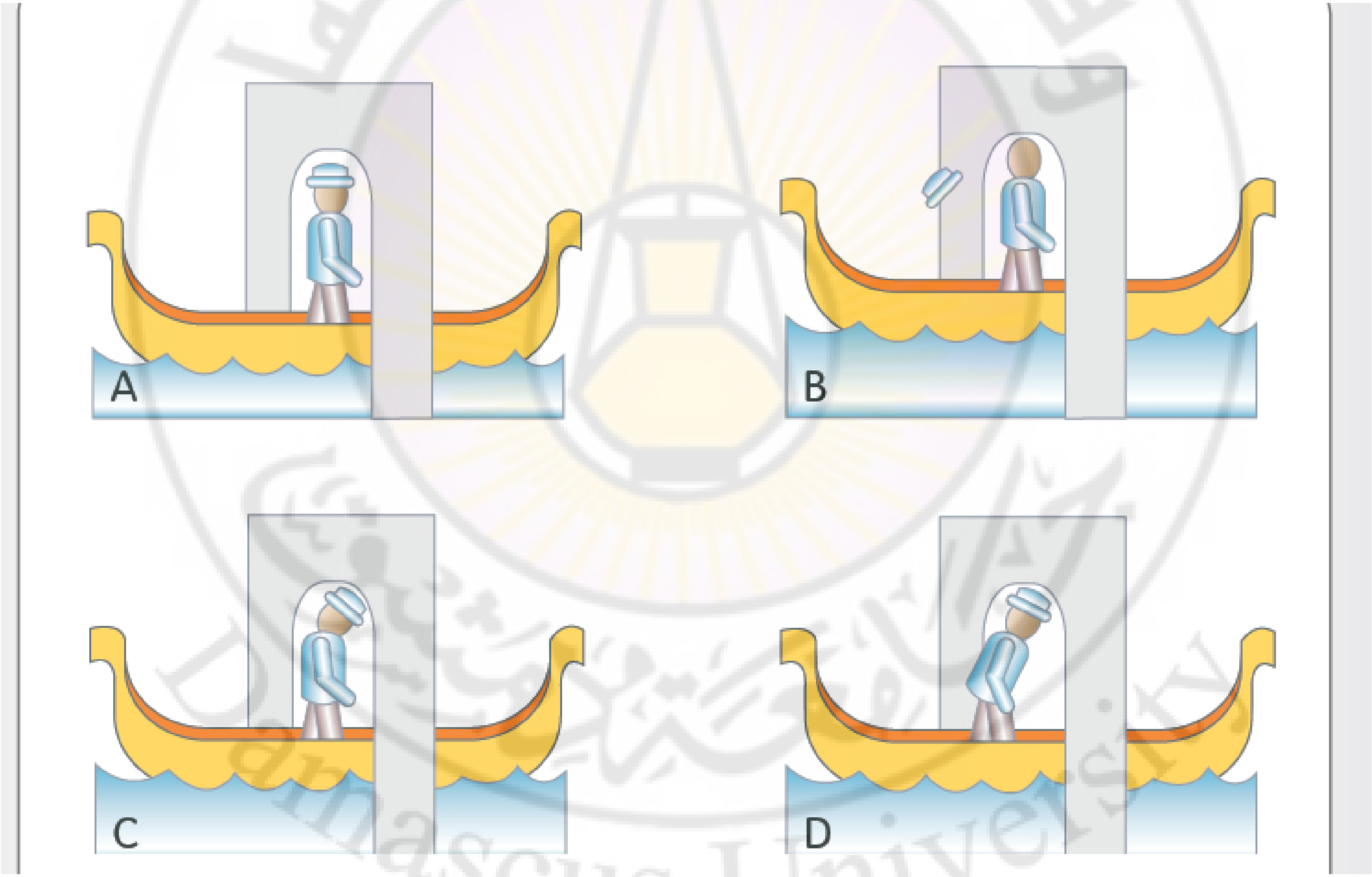
Loudness disconformable level = 120 dB HL

Dynamic range of patient = 40 dB

Real dynamic range in world = 120 dB (threshold = 0 dB HL and LDL = 120 dB HL)

So, we need to shrink the world

Amplitude Compression



Compression's Major Role: Reducing the Signal's Dynamic Range

Compression is like motherhood – everyone agrees it is a good thing, but there is much disagreement about the best way to do it. This chapter describes the different ways that compression can be applied in hearing aids. All of the compression methods have some advantages over linear/peak clipping amplification. All also have some disadvantages.

Figure 6.1 shows three gain could change as left panel, gain starts level rises above *weak* level has been reached decreased, and linear for all higher input ing of the dynamic

There is always a balance between making sounds audible and making them comfortable—extreme audibility can work against sound quality and listening comfort.

Compression is like motherhood – everyone agrees it is a good thing, but there is much disagreement about the best way to do it. This chapter describes the different ways that compression can be applied in hearing aids. All of the compression methods have some advantages over linear/peak clipping amplification. All also have some disadvantages.

6.1 Compression's Major Role: Reducing the Signal's Dynamic Range

The major role of compression is to decrease the dynamic range of signals in the environment so that all signals of interest can fit within the restricted dynamic range of a hearing-impaired person (see Section 1.1.2 and Figure 1.2 in particular). This means that intense sounds have to be amplified less than weak sounds. A compressor is an amplifier that automatically reduces its gain as the signal level somewhere within the hearing aid rises (Section 2.3.3). There are, however, many ways in which the gain can be varied to decrease the dynamic range of a signal.

Figure 6.1 shows three ways in which the amount of gain could change as the input level changes. In the left panel, gain starts reducing as soon as the input level rises above *weak*. By the time a moderate input level has been reached, the gain has been sufficiently decreased, and linear amplification can then be used for all higher input levels. The necessary squashing of the dynamic range of the signal has all been accomplished for low signal levels, so we could refer to this as low-level compression. This can be seen in the upper picture as the lower levels coming closer together after amplification than before, while the spacing of the upper levels is not affected by amplification. In the lower figure, the same squashing (i.e. compression) of levels appears as the decreased slope of the input-output (I-O) function for low-level signals, whereas the linear amplification of higher level signals appears as a 45° slope (see also Section 4.1.5).

The opposite approach can be seen in the right panel of Figure 6.1: low-level sounds are amplified linearly, but the inputs from moderate to intense sounds are squashed into a narrower range of outputs. In gen-

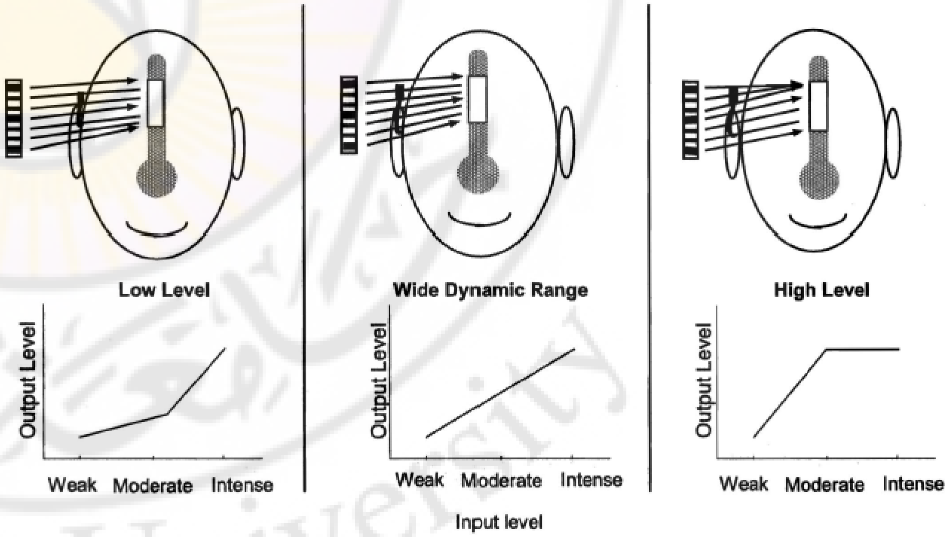


Figure 6.1 Three ways in which the dynamic range of signals can be reduced. In each case, the upper figure shows the spacing of different signal levels before amplification (the left end of the lines) and after amplification (the right end of the lines). The lower figure shows the same relationships, but as an input-output function. In each case compression is occurring in the red region.

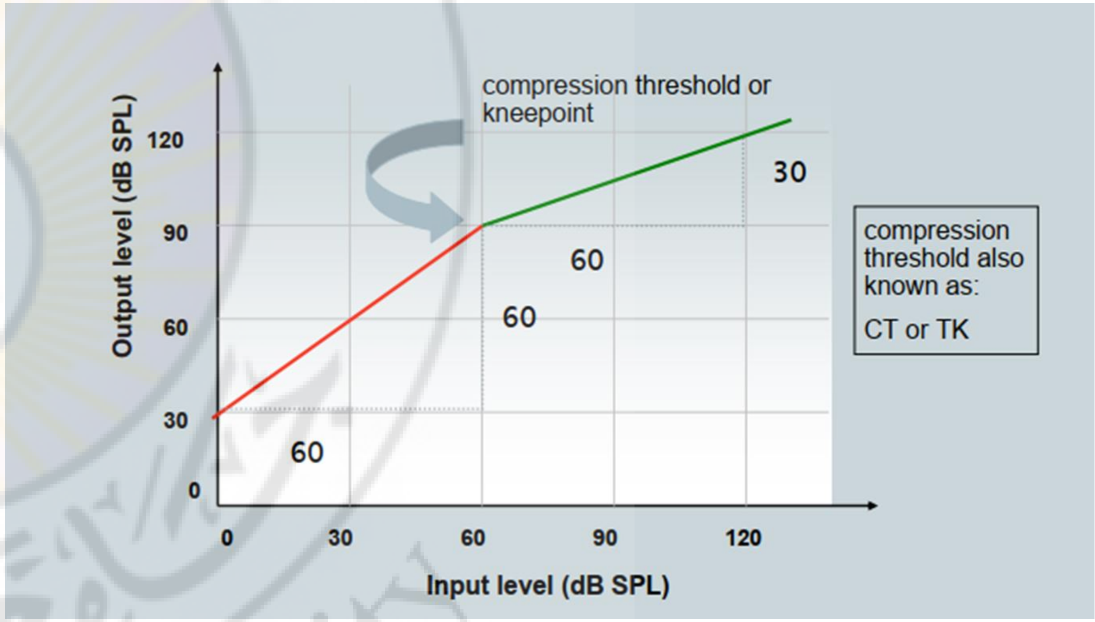
The characteristics of Amplitude Compression:

- **Static characteristics:**
  - Compression Threshold (CT)
  - Compression Ratio (CR)

- **Dynamic characteristics:**
  - Compression Time Consonant (attack time and release time)

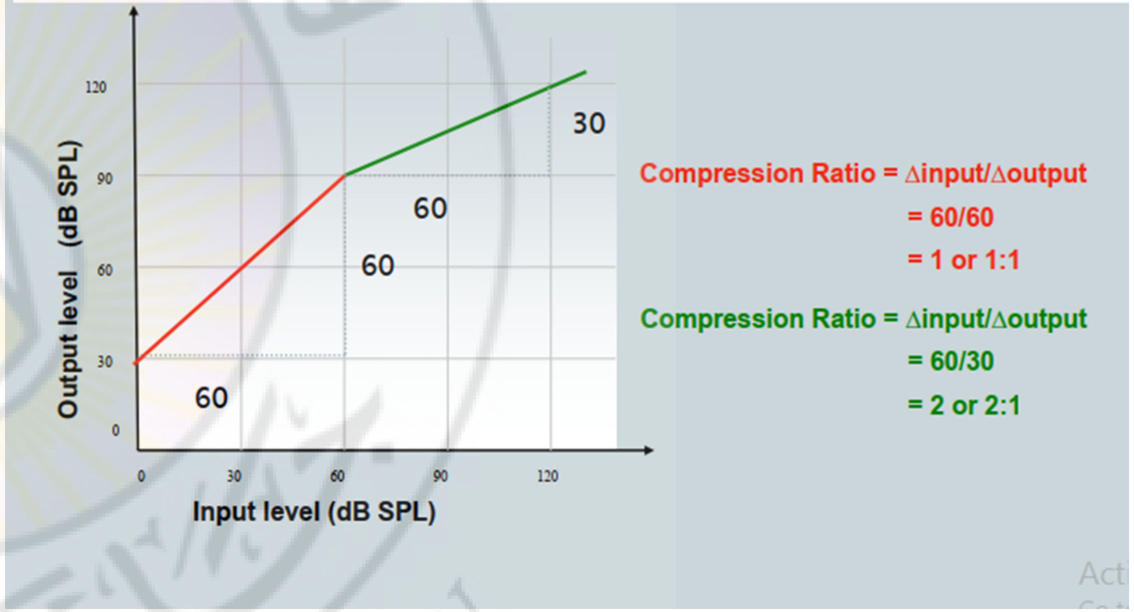
Compression threshold…………..

- In the I/O curve, The point where any gain line suddenly takes a bend is called the compression "threshold," or "knee-point," and it is at this point that "compression" begins. The "kneepoint" of compression above the input axis shows the input SPL where compression begins.
- In the certain channel, the loudest sounds will always activate compression system.



Compression ratio……………..

The compression ratio is the ratio of increase in input level to increase in output level. For example, a compression ratio of 2:1 means that for every 10 dB increase in the input signal, the output signal increases by 5 dB.



Attack and release time……….

- The dynamic characteristics of a compressor refer to the length of time required for the compression circuit to respond to a sudden change in the input.

- Attack time (AT) is the time delay that occurs between the onset of an input signal loud enough to activate compression (i.e., input signal exceeds the TK) and the resulting reduction of gain to its target value.

- Release (or recovery) time (RT) is the time delay that occurs between the offset of an input signal sufficiently loud to activate compression (i.e., input signal falls below the TK) and the resulting increase of gain to its target value.

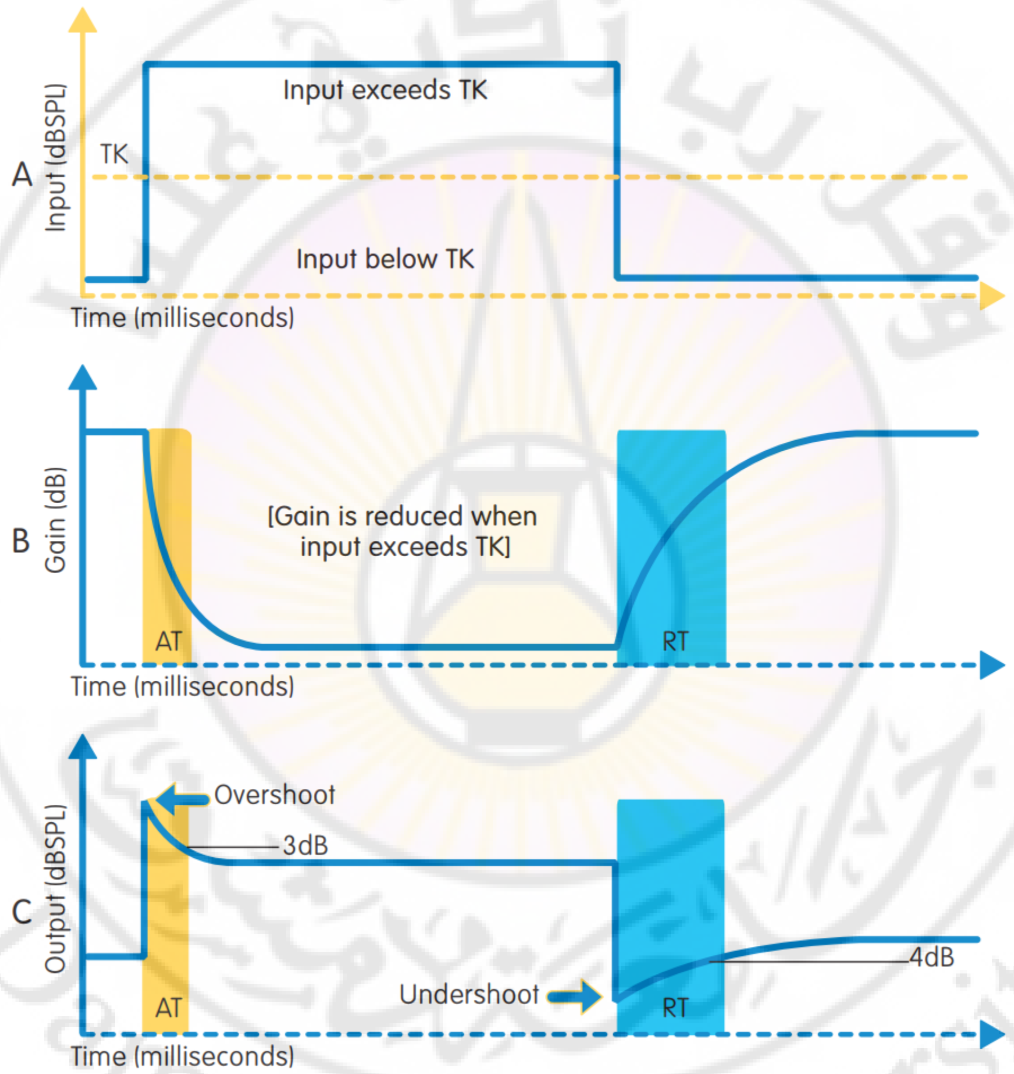


Figure 2-8
Schematic representation of the behavior of a hearing aid associated with changes in input. The input level, gain and output of the hearing aid are shown in panels A, B and C, respectively. The attack time (AT) and release time (RT) are indicated by the shaded areas. TK = Threshold kneepoint.

Attack and release time………..

- Depending on the purpose, a compression system may have fast or slow attack and release times.

- ATs as fast as 5 ms are especially desirable when compression is used to limit the maximum output of a hearing aid.

- The RT is generally longer than the AT. An RT of 20 ms is considered fast.

- The disadvantage of ATs and RTs between 100 ms and 2 s is that it causes the compressor to respond to brief sounds, or lack thereof, in the environment and create a phenomenon called pumping.

Attack and release time……….

- Attack and release times of 2 s or slower respond to changes in the overall level of sound in the environment rather than to individual events.

- Some compression circuits incorporate adaptive or variable release times – that is, the RT is adjusted based upon the duration of the triggering signal.

Different Types of Amplitude Compression Circuits

- Input compression
- Output compression
- Wide Dynamic Range Compression (WDRC)
- Compression Limiting (CL)
- Treble-Increase-at-low-levels (TILL)
- Bass-Increase-at-low-levels (BILL)

Input compression (AGC-I)

With input-controlled compression (AGC-I), the level detector is located before the volume control and compression acts on the input to the hearing aid. That is, once the input exceeds the TK, the compressor is activated and gain is reduced at the preamplifier.

Therefore, the volume control setting has no impact on the compression parameters.

Input controlled compression

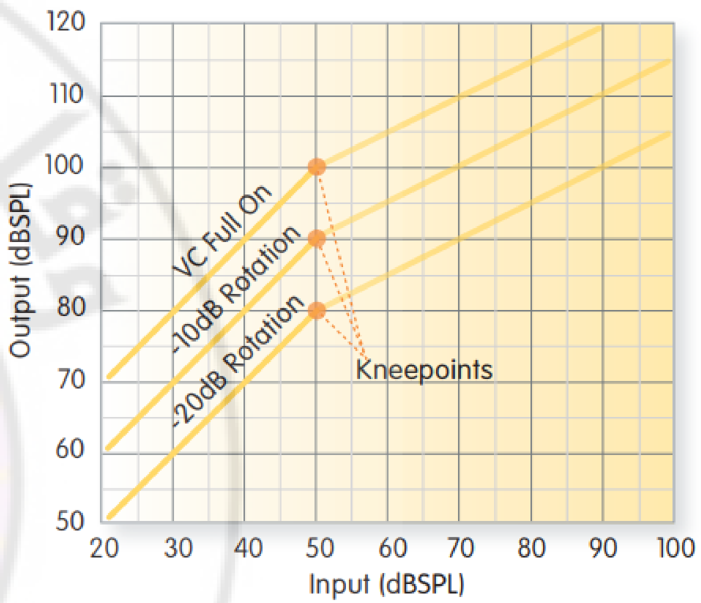


Figure 2-12
Sample input/output functions for an input-controlled compression (AGC-1) circuit showing the effects of volume control rotation.

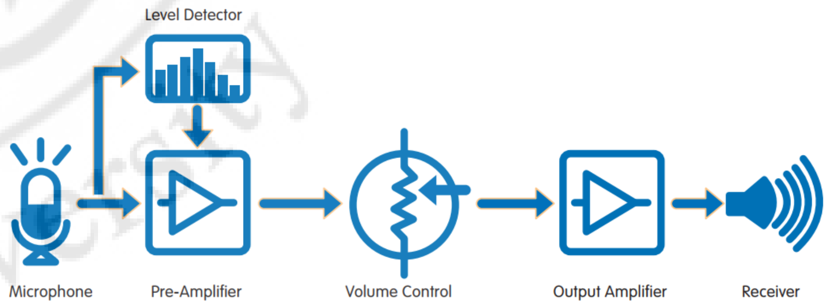


Figure 2-11
Schematic representation of an input-controlled compression (AGC-1) circuit.

Output compression (AGC-O)

- In output-controlled compression (AGC-o), the level detector is located after the volume control (Figure 2-13) and compression acts on the output of the hearing aid. That is, the compressor is activated once the output exceeds the TK.
- Therefore, the volume control setting has impact on the compression parameters.
- As the user increases the volume control, gain increases, but there are no changes to maximum output.

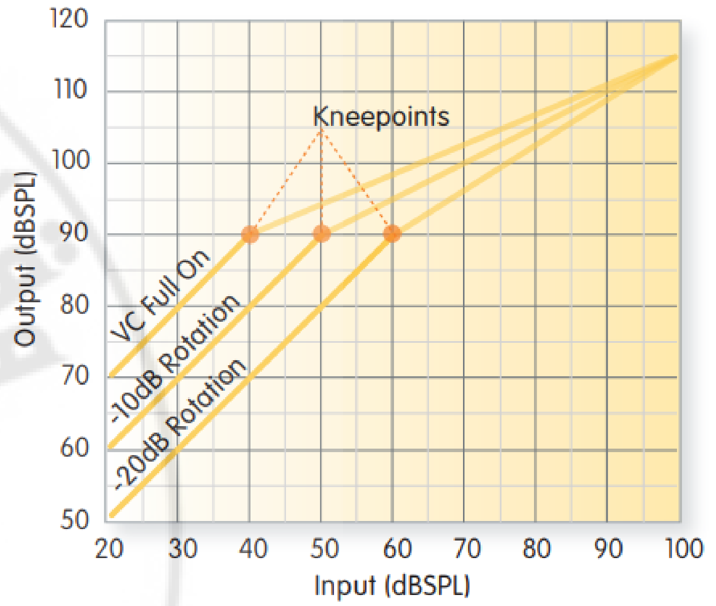


Figure 2-14
Sample input/output functions for an output-controlled compression (AGC-O) circuit showing the effect of volume control rotation.

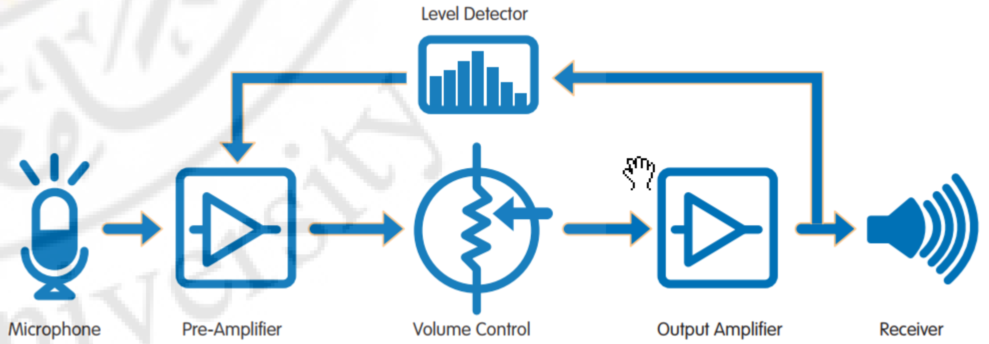


Figure 2-13
Schematic representation of an output-controlled compression (AGC-O) circuit.

some Important tips…………………

- In the old hearing aids, AGC-I and AGC-O had different effects on the hearing aids, but in the modern hearing aid, their effects on the hearing aids are similar.

- Input Compression : Mild to Moderate HL

- Output Compression : Severe to Profound HL

Multi channel compression……

- The ability to change gain individually in bands and/or channels is useful for two reasons.
- First, there is a wide variety of audiometric configurations for which given hearing aid may be useful. The ability to adjust gain and compression in discrete frequency regions permits customization of amplification to the individual's needs.
- Another advantage of multichannel compression is that acoustic events in a discrete frequency region do not affect the response of the hearing aid at all frequencies.

Advantages of multi channel compression………

Often, Noise and speech occur in discrete frequencies

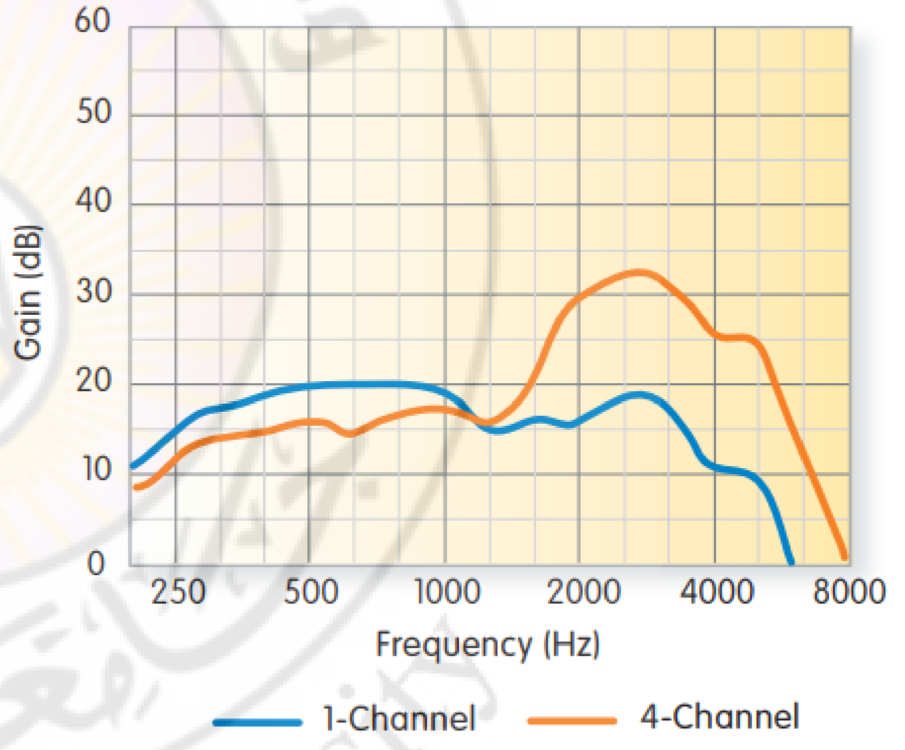
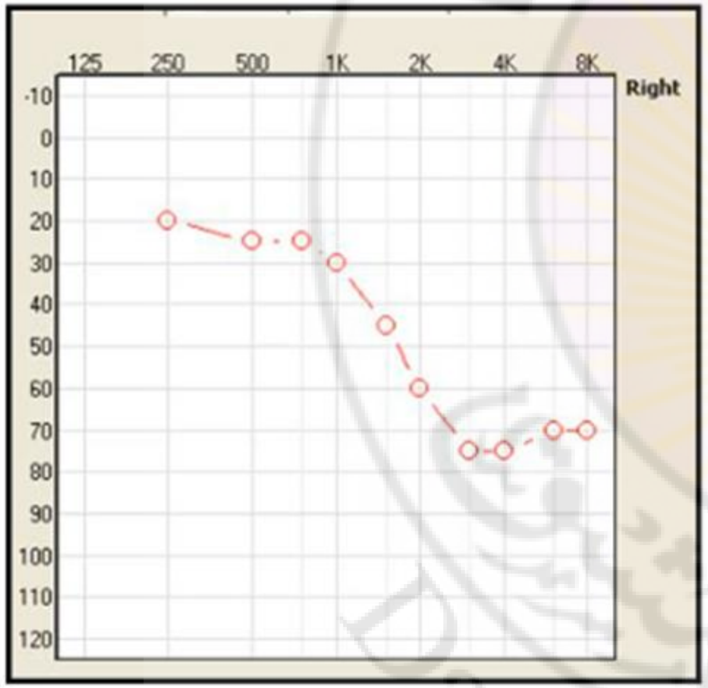
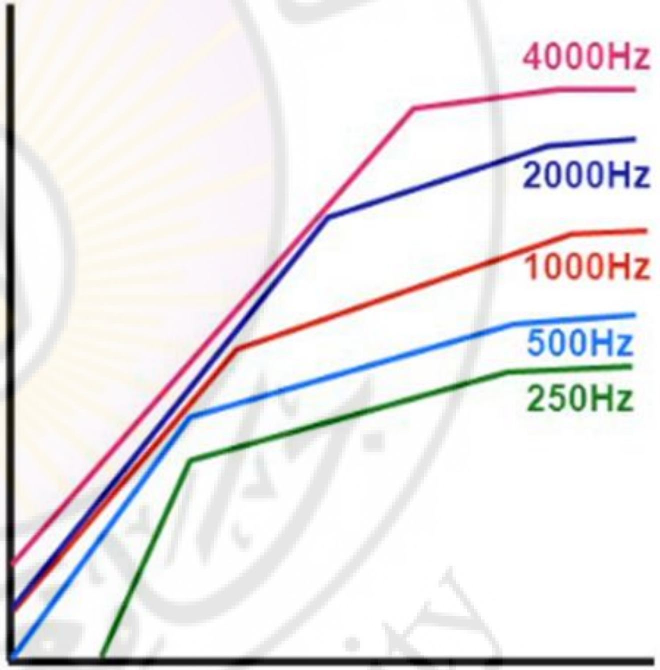


Figure 2-15
Sample frequency-gain curves for a 1-channel and a 4-channel hearing aid in the presence of a 90dBSPL pure tone at 500 Hz.

Multi channel compression….

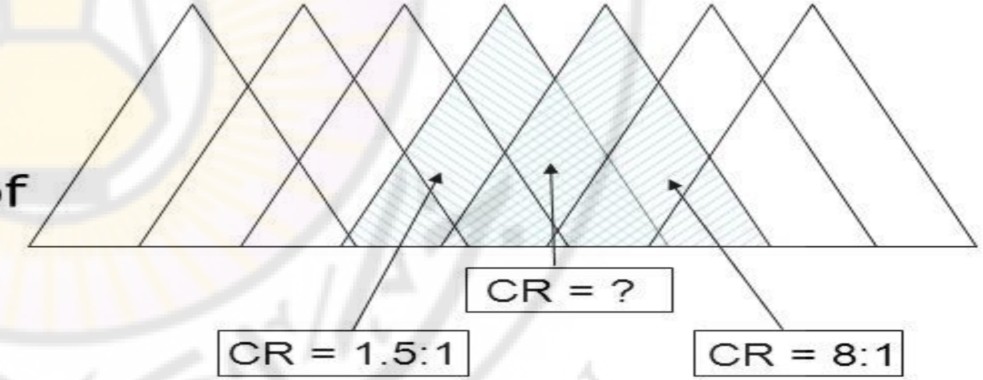




Multi Channel Compression

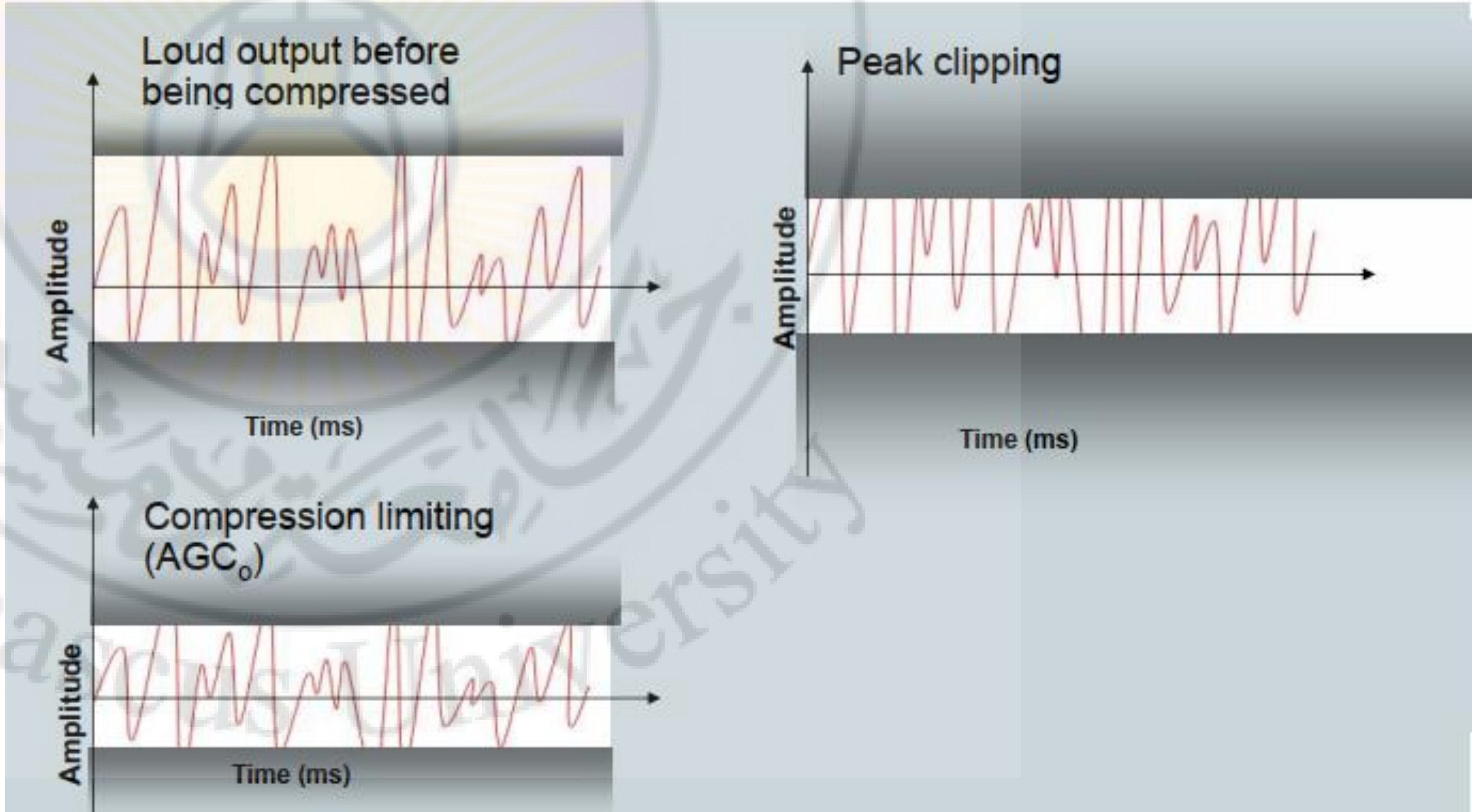
Channels and Compression

- Compression can vary in multi-channel hearing aids.
- In order to maintain a smooth response (no distortion) averaging of compression characteristics may occur.
- Averaged levels are less precise than actual levels.



Limiting Maximum Output

- **Peak clipping**
- **compression limiting (AGCo)**



Wide dynamic range compression: a kind of AGC-I

- The compression type utilized in nearly all hearing aids is AGCi, and when the AGCi kneepoint is relatively low (~55 dB SPL or below), this is referred to as wide dynamic range compression (WDRC). It is called this because a "wide" range of average speech is in compression.

- It has low compression ratios (less than 4:1, most commonly around 2:1).

Wide dynamic range compression: a kind of AGC-I

- In WDRC compression, as input goes up, gain goes down.

- this compression type is suitable foe mild to moderate SNHL. In this compression, there is a little or no gain in loud sounds and high gain in soft sounds.

- The fitting result of WDRC is that soft sounds become louder, but compared to linear processing, WDRC itself doesn't make soft sounds louder— it makes average sounds softer. If average sounds are softer, you (via programming) or the patient (via the VC) will turn up gain. When you turn up gain—soft sounds become louder.

Compression limiting : a kind of AGC-O

- Output limiting compression is associated with high compression kneepoints and high compression ratios.

- A high kneepoint means that the hearing aid begins to compress at relatively high input levels.

- Below the high kneepoint, the hearing aid would have linear gain if there were no companion AGCi circuit

Compression limiting : a kind of AGC-O

- The kneepoints used for output limiting are usually around 100 to 115 dB (re: 2-cc coupler). Why?

- Because this corresponds to the LDL of the average hearing impaired patient (when converted to 2-cc coupler values) .

- The compression ratio of an output limiting compressor is usually around 10 to 1, which means that there is only a 1 dB corresponding increase in output for a 10 dB change in input, once the signal is above the kneepoint.

Using WDRC and CL, together…….

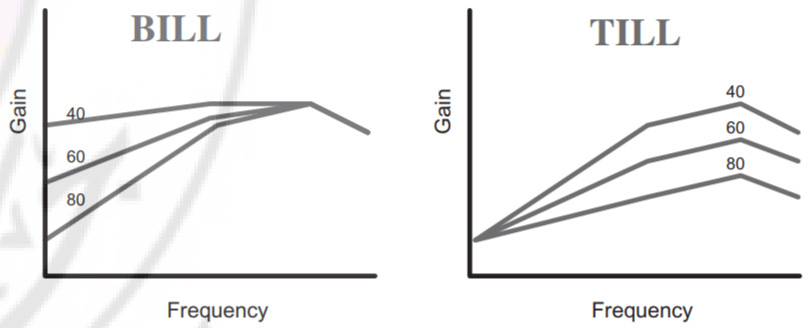
- Consider that output-limiting compression is used as a partner with WDRC. WDRC takes care of the soft to loud speech sounds; output limiting takes care of the very loud sounds.

- On post fitting visits, it’s important to know which is which when you start making “mouse clicks.” Start changing the WDRC when the problem really is with the AGCo, and the patient who walked in with one problem, might just leave with two!

Treble-Increase-at-low-levels (TILL) and Bass-Increase-at-low-levels (BILL)

- When hearing aids were only two channels, these two terms were coined to describe variations of compression applications.
- BILL was used to describe hearing aids that had more compression in the low then in the high frequencies.
- TILL was used to describe hearing aids that had more compression in the high then in the low frequencies.

BILL vs. TILL
Two Types of WDRC



Syllabic compression……

A term used to describe a relatively short release time (e.g., <150 milliseconds). The origin of this term is the notion that if a patient is using a hearing aid with a release time this quick, he would not miss more than one syllable before the hearing aid restored gain to the new input level.

Adaptive or Dual Compression….

The release time is related to the duration of the input signal. For most inputs, a long release is in effect, but if a short duration signal occurs (e.g., door slam), the short release time will be activated and temporarily will replace the long release.

Compression in digital hearing aids

Digital hearing aids simply combine all sorts of compression types.

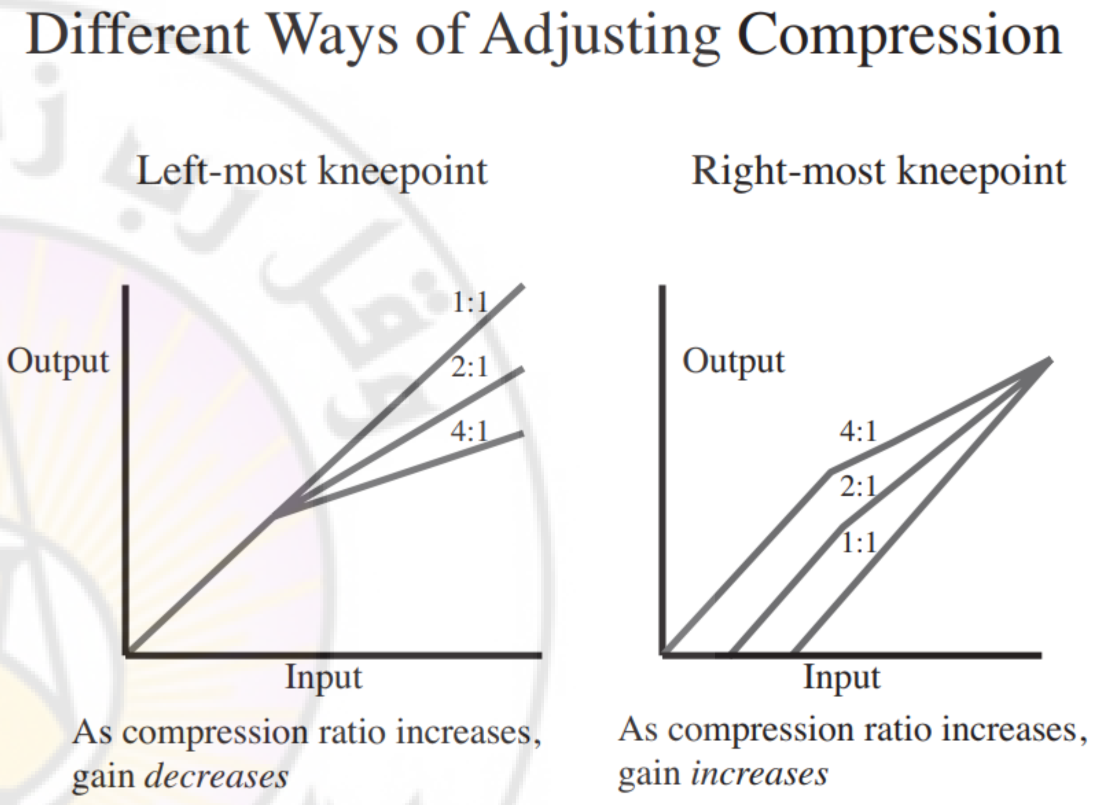


Figure 5–14. Different ways of adjusting compression. The left panel shows compression adjustments in the way that most clinicians conceive of them. When hinging from a left-most knee-point, linear (1:1) gain implies greatest gain. Increased compression ratios imply progressively less gain. The right panel shows compression as it originally became available with high-end multichannel analog hearing aids of the mid to late 1990s. This type of compression adjustment was based on the model of normal loudness growth. With compression hinging from the right-most knee-point, linear gain is actually the least amount of gain offered, while increased compression ratios imply increased gain. The objective here was to approximate normal loudness growth with increased compression ratios. Today's digital hearing aids commonly utilize both of these ways of adjusting compression.

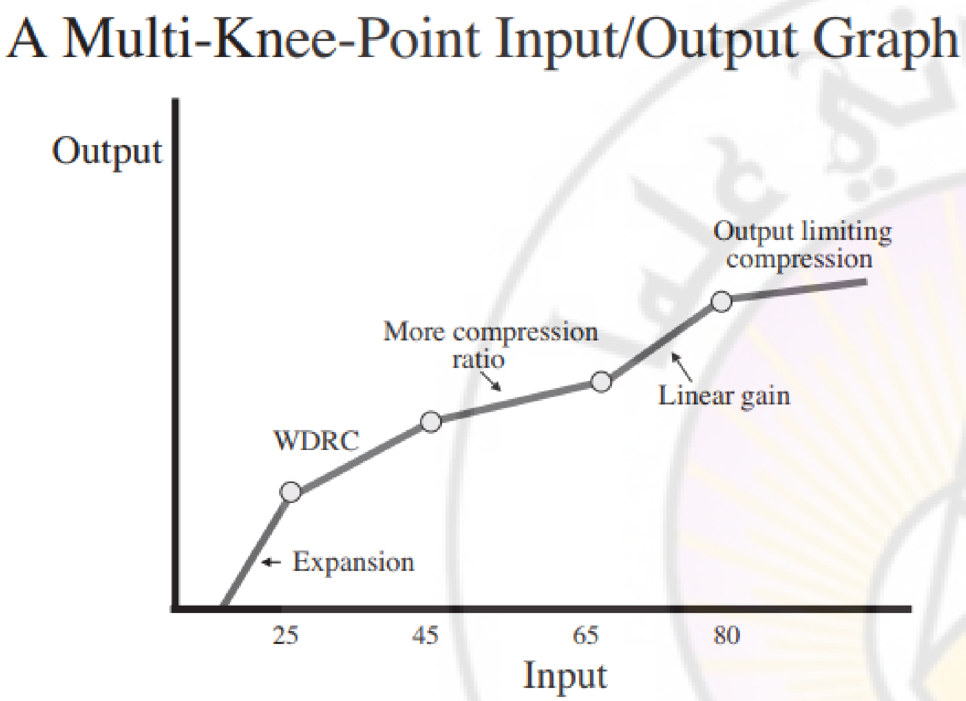


Figure 5–18. A multi knee-point input/output graph. The software for fitting many digital hearing aids often shows input/output graphs that have more than two knee-points. Each can be adjusted by the software, or they can be automatically set to a "best-fit" as determined by the fitting software. Below the left-most knee-point, either linear gain or expansion (commonly called "soft noise squelch") can be selected. Only expansion is shown on this figure. Linear gain, of course, would be a 45° angle.

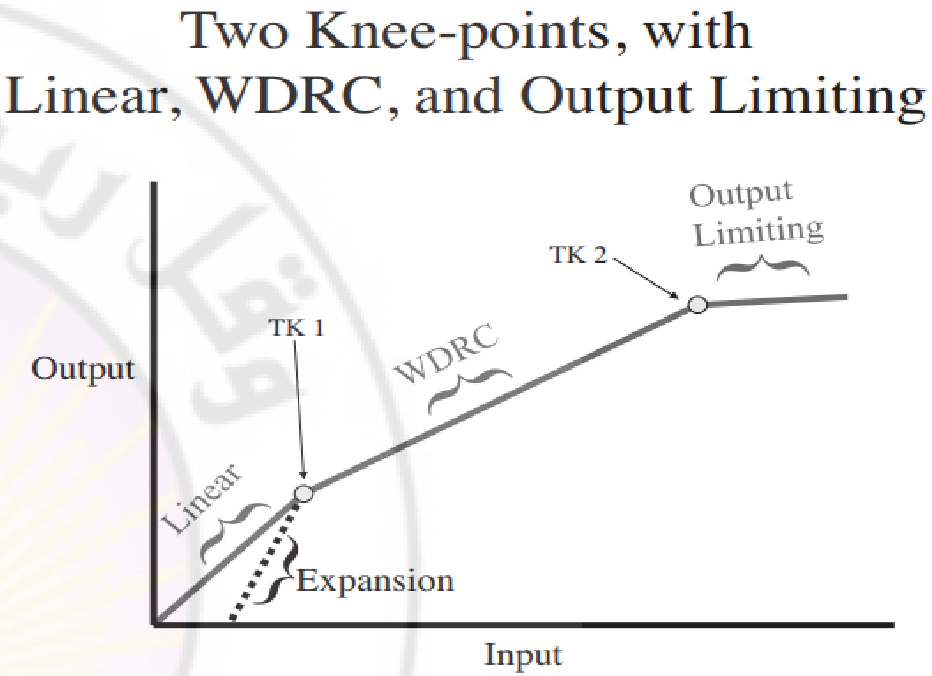


Figure 5–15. Two knee-points, with linear, WDRC, and output limiting. Digital hearing aid fitting software commonly enables clinicians to visualize input/output graphs showing two (or more) knee-points. Each knee-point can be adjusted both vertically and horizontally, to best meet the needs of the client. Linear gain takes place below the left-most knee-point; WDRC occurs between the two knee points, while output limiting compression occurs to the right of the right-most knee-point. In many digital hearing aids, expansion is also offered. Expansion provides greater than linear gain, and it occurs below the left-most knee-point.

Frequency lowering

Dr. Maryam Sadeghijam
Assistant Professor of Iran University of Medical Sciences

2023, May

What is the cochlear dead region?

- Cochlear damage may result in complete loss of function of inner hair cells over a certain area of cochlea called a **dead region**.
- The nerve cells that are dead can not transmit any sound of specific frequencies to the brain.
- People with extensive dead region often do not benefit much from a hearing aid, specially for listening of /s/ and /z/.

Dead Regions In Cochlea……………………

- When a pure-tone signal “falls” into a dead region, it can be heard by neighboring hair cells, if the intensity of the signal is loud enough.
- This is because the pure tone produces sufficient basilar-membrane vibrations in neighboring areas of the cochlea, where there are surviving IHCs and neurons. This phenomenon is defined as “Off Frequency Listening”.
- Clinically, this will be presented as a threshold on the traditional pure tone audiogram, but it may not be the real threshold.

How do we identify the dead region?

There is a test for identifying dead region called “TEN test”.

The threshold equalizing noise (TEN) test measures pure tone thresholds with a special masking noise called TEN. The TEN test is a quick and easy way to identify cochlear dead regions.

Characteristics that could indicate the presence of a cochlear dead region (Moore, 2009):

- ❖ Severe to profound hearing loss
- ❖ Absolute threshold at a specific (suspected) frequency is 70dB HL or greater
- ❖ Steeply sloping hearing loss
- ❖ Complaints of distortion
- ❖ Extremely poor speech discrimination

TEN test procedure……………….

- The TEN test is performed ipsilaterally, meaning that the tone and the noise are presented in the same ear.

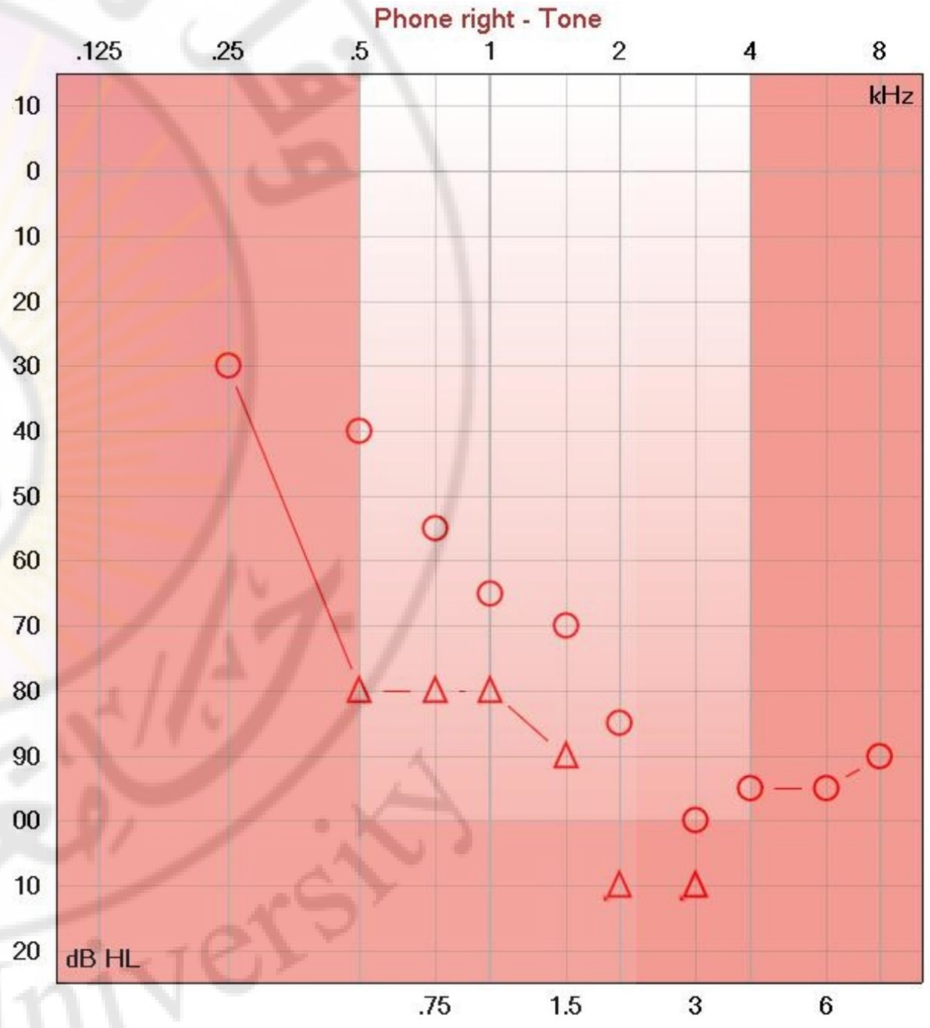
- Select two channel audiometer. The tone is presented in channel 1 and TEN signal or noise is presented in channel 2, with the stimulus in channel 2 reversed (Rev) to have a continuous masking signal.

TEN test procedure……………….

- Conduct a threshold search using traditional method for air conduction
- Conduct a threshold search while presenting noise. Set the intensity levels of noise:
  - For frequencies where hearing loss up to 60dB HL: set the TEN level to 70dB
  - For frequencies where hearing loss is 70dB or more: set the TEN level 10dB above the audiometric threshold at that frequency. E.g., if the audiometric threshold is 75dB HL, set the TEN level to 85dB HL
  - If the TEN is too loud, or if the maximum TEN level of 90dB HL is reached, then set the TEN level equal to the audiometric threshold. This should still give a definitive result.

TEN test procedure………………..

A dead region at a particular frequency is indicated when a masked threshold is at least 10dB or more above the level of the TEN and the masked threshold is at least 10dB above the non-masked threshold.



Frequency lowering…….

- Although audibility usually is a good thing, there are three cases when we might not want high-frequency gain in the instruments when we are fitting hearing aids:
  - The hearing aid is not able to deliver the desired high-frequency gain without continued feedback problems.
  - The patient's hearing loss is so severe that it is not practical to attempt to provide audibility.
  - The patient has cochlear dead regions (which often is consistent with Reason #2, but dead regions also can be present in milder cochlear hearing losses (e.g., 60 to 70 dB HL). Because of this, added gain does not result in added speech recognition.

How can we audible the frequencies in the dead region?

- For audible sound that are in the dead region, we can use frequency lowering. For frequency lowering, there are some strategies:
  - **digital frequency compression**
  - **frequency compression**
  - **frequency transposition**
  - **Frequency translation**
- Current algorithms are designed to move or compress high frequencies (e.g., around 3000 to 6000 Hz and higher) to lower frequencies.

Digital Frequency Compression (DFC)

- This strategy involves the compression of the entire frequency range, which means that none of the original signal will be preserved.
- **Nowadays, this technique isn't used.**

Frequency Compression………

- With frequency compression (e.g. SoundRecover by Phonak) high frequencies are brought to lower frequencies by squeezing frequency content together in a smaller space.
- on the other words, with this strategy, frequencies above the so-called cutoff frequency (1600 Hz or higher) will be compressed (i.e., squeezed together), while frequencies below the cutoff frequency will be amplified in the conventional manner.

Frequency compression……………….

The frequency compression knee point, identified here as the cutoff frequency, defines where the compression effect begins. Input frequencies below this cutoff are not compressed and frequencies above this cutoff are compressed. The magnitude of this compression is defined by the compression ratio deployed.

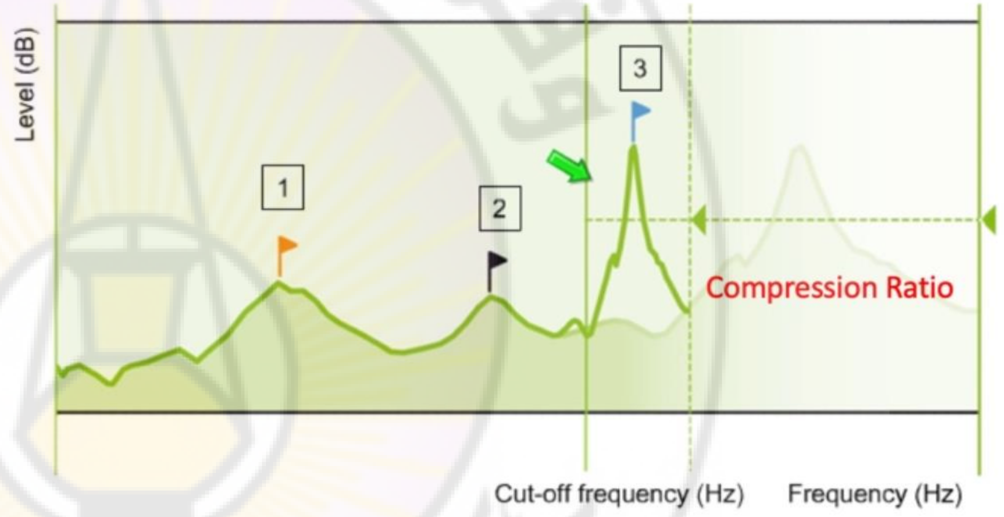


Figure 11. Representation of output amplitude across frequency (from Nyffeler (2009), with permission).

Figure 16. Illustration of frequency compression (from Starkey, with permission Crukley (2019)).

Frequency compression (advanced)

- In Sound Recover 2, Phonak's latest version of their frequency compression technology, they have added an adaptive element to their frequency compression procedure.
- CT 1 (the lower cutoff), represents the first compression knee point. In Sound Recover 2, there is a second cutoff frequency, labeled as CT 2, placed within the range of frequencies targeted for compression.
- Now all frequencies above CT 2 (above 8 kHz) are always compressed. However, for input frequencies between CT 1 and CT 2, frequency compression is adaptive.
- In other words, frequency compression won't be activated in this region unless there are relevant speech cues identified as present within this input bandwidth. Otherwise, this bandwidth will continue to be processed linearly.

Sound recover 2……………………

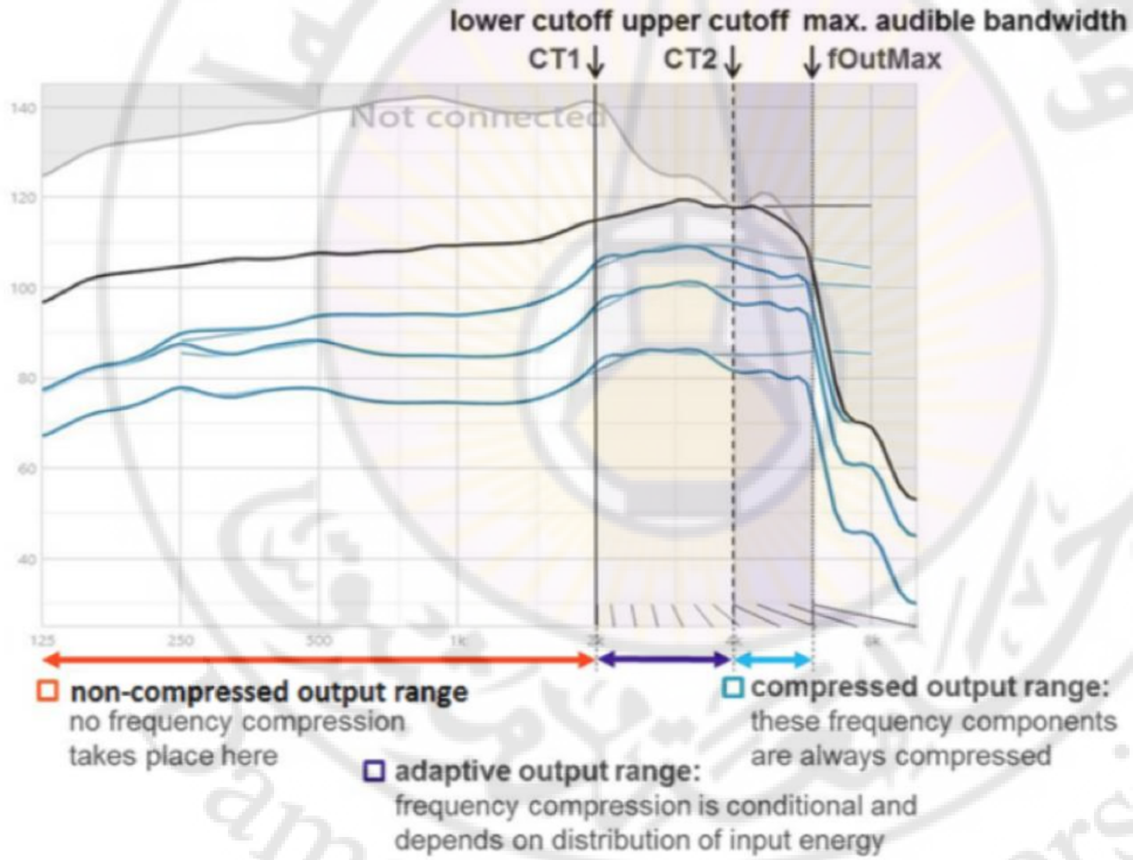


Figure 12. Adaptive frequency compression (from Phonak, with permission Jones (2019)).

Frequency transposition or audibility extender………………….

- This feature performs what is referred to as linear frequency shifting. The aim is to preserve the harmonic structure and modulation ratio of the original signal.
- In the basic mode, the algorithm is designed to cut or copy a portion of this unaidable region (source region) and paste it within a lower frequency region that is aidable. We've labeled this region as the ‘Target Region’.

Frequency transposition…………..

As you can see in Figure 3, once frequency transposition has been employed, there are now two components of input energy occupying this output frequency space. One component is the transposed input energy, and the other component is the input energy that was in the target region, to begin with.

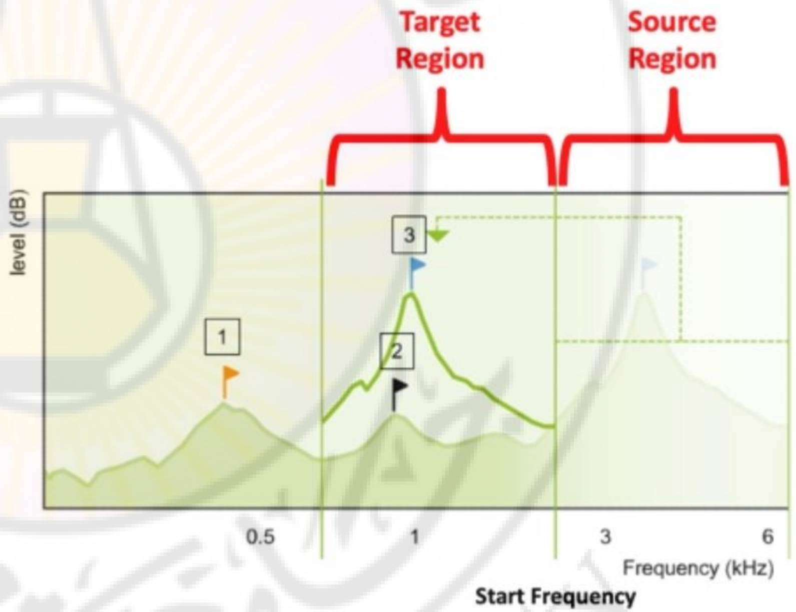


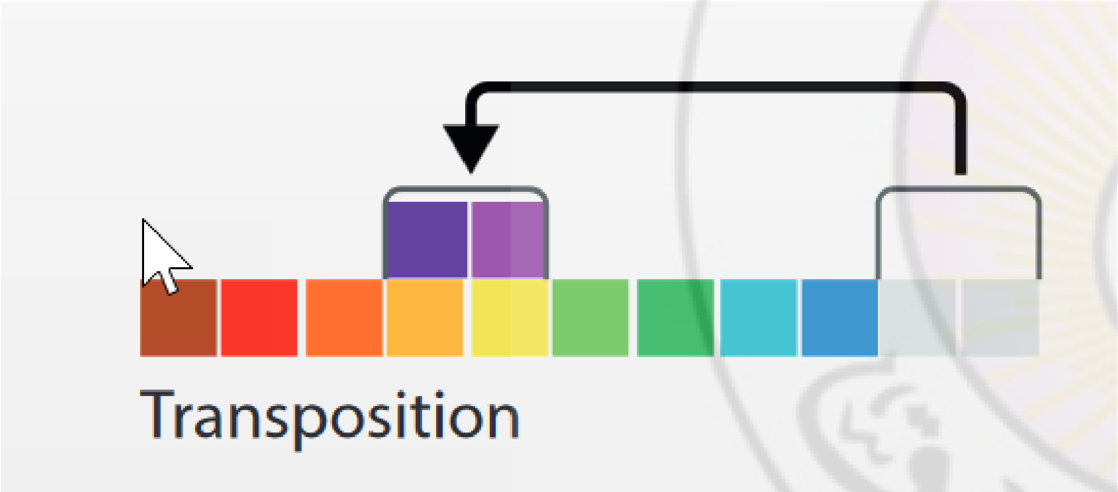
Figure 3. Frequency transposition approach (from Nyffeler (2009), with permission).

Frequency transposition………..

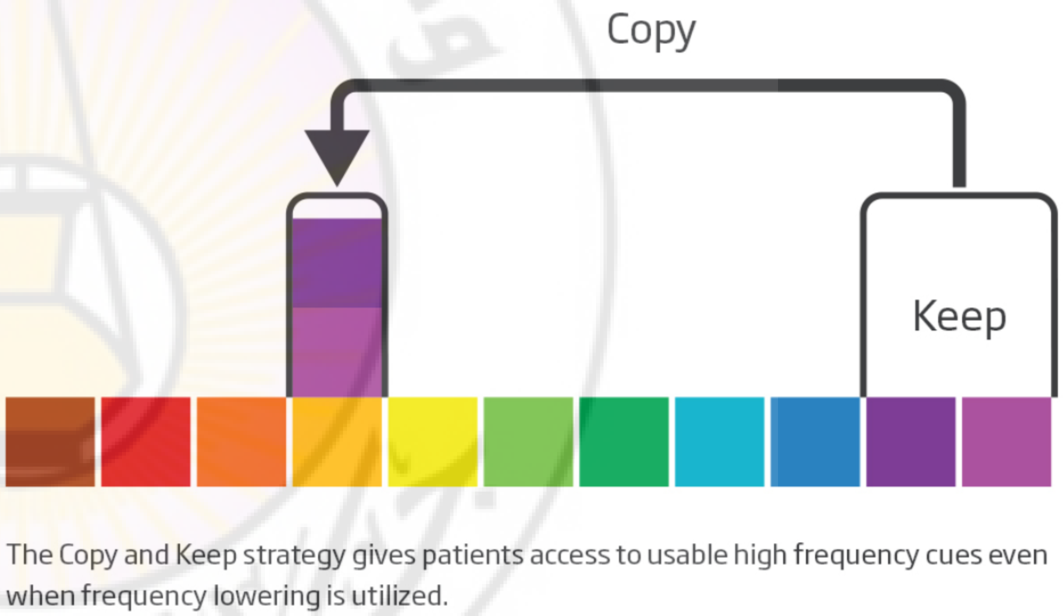
- Frequency transposition technology can be found in Widex products, identified as "Audibility Extender" (which utilizes a cut and paste approach), and in Oticon products, identified as "Speech Rescue" (which utilizes a copy and paste approach).
- In the expanded mode, the feature analyzes two source regions and places the most prominent parts of these two at the same octave in the target region. The Audibility Extender selects the source region by a so-called start frequency that can be as low as 630 Hz and as high as 6000 Hz.
-

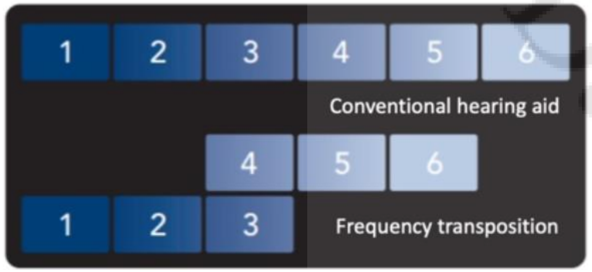
Frequency transposition……….

Audibility extender



Speech Rescue

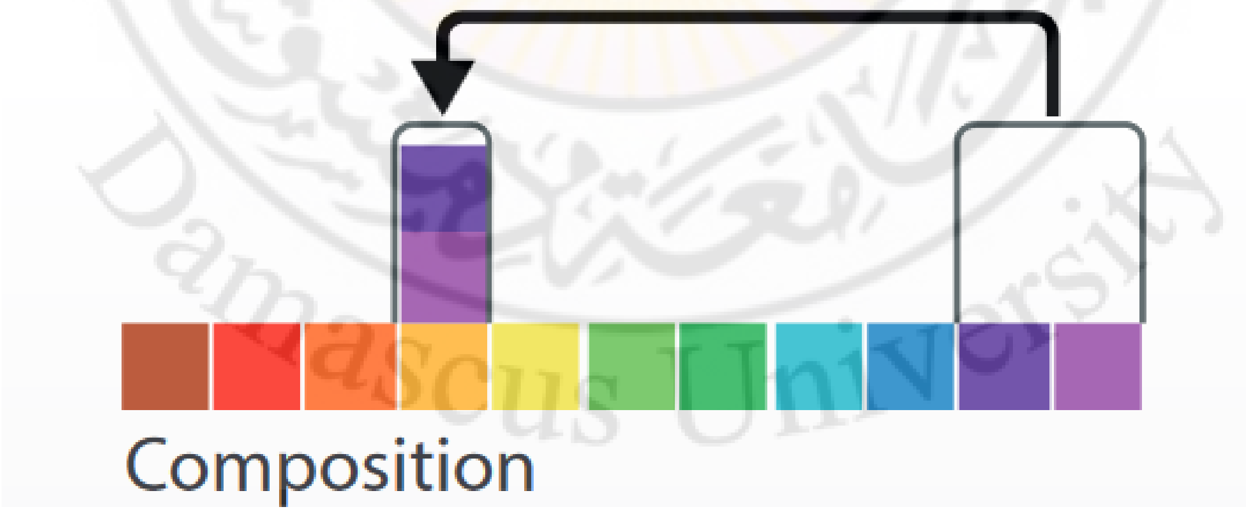




Frequency transposition in old Starkey hearing aids

Frequency Composition…………………

superimposes a high-frequency source band on a low-frequency destination band, but it first divides the source band into 2 or 3 segments and then overlaps them in the destination band in order to present information from a wider input region in a narrower output region



Spectral IQ or Frequency Translation……………………….

- This feature analyzes the content of the input signal in search for appropriate spectral cues such as /s/ and /t/ sounds, and transposes them to an audible area.
- When high frequency spectral cues are absent, Spectral IQ is inactive
- The bandwidth of the total signal is not altered, and thus retains the full bandwidth accessible to the user

Frequency translation……………………

In this method, an "S" phoneme has been identified as part of the input content residing in band 6. Having identified the presence of this phoneme, if frequency translation is activated, it will create a supplemental representation of this phoneme in a lower band. In this case, the supplemental phoneme was placed in band 3. The original "S" input content is not removed from the output. Rather, it is supplemented with a lower frequency representation of that phoneme. If no speech phoneme is detected in an upper band, no frequency translation is activated.

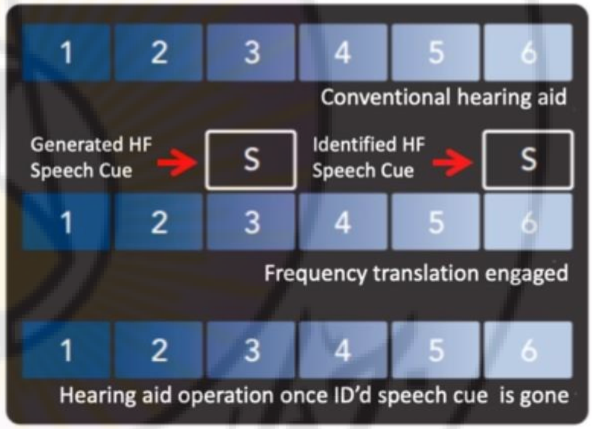


Figure 17. Illustration of frequency translation (from Starkey, with permission Crukley (2019)).

Important Tips……………………..

- Each of these algorithms has some disadvantages:
  - In the transposition strategy, the harmonic structure and modulation ratio of the original signals is preserved, this technique is equally capable of shifting both speech and environmental sound, but sometimes speech intelligibility decrease due to overlapping of input signal with output signal after transposition.
  - In the compression strategy, the harmonic structure isn’t preserved.
- Research on speech intelligibility has shown that both frequency compression and frequency transposition can improve speech perception and production for hearing aid users.

Important Tips……………………..

- However, in spite of the efforts of the manufacturers, the sound produced by any of the frequency lowering algorithms will sound unnatural at first and require acclimatization and training.

- The brain needs time to adapt to a new sound scheme, and this period can be as long as 3 to 6 months.

- So, it is important to test the effectiveness of this technology at least 3 months after its use.

Important Tips……………………

Keep in mind that efficiency of this strategy must be evaluated after activation **by Real Ear Measurement** verification.

It is important to know if indeed the desired signals have been made audible when frequency lowering has been activated.

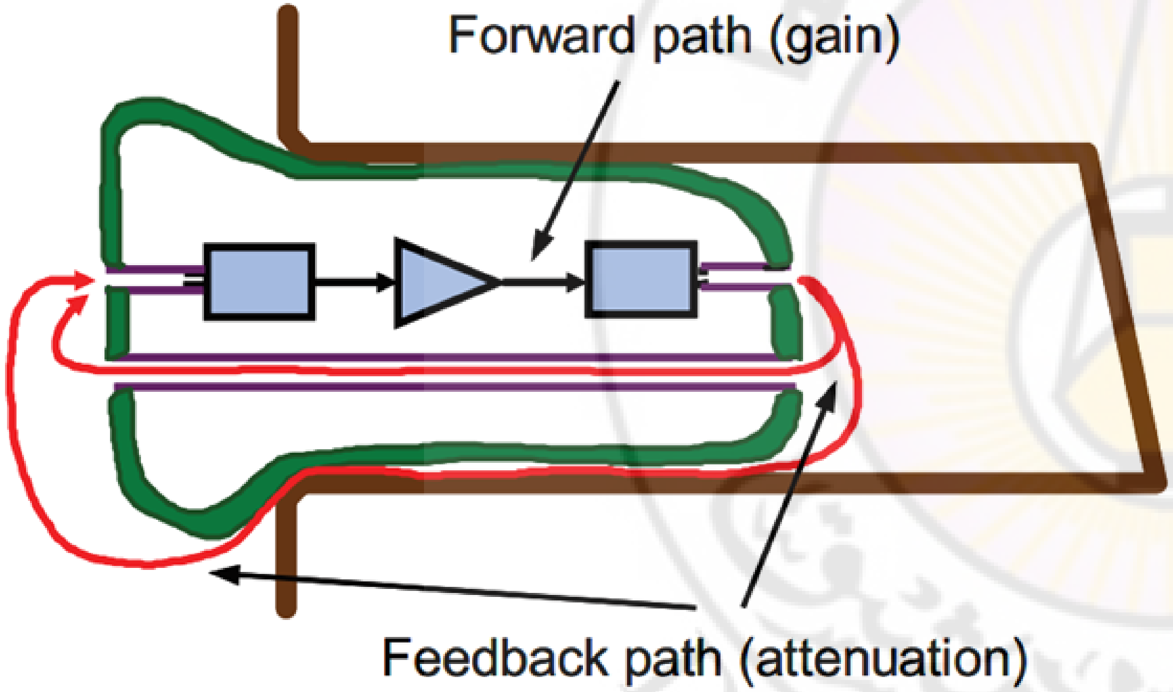
Feedback Cancelation

Dr. Maryam Sadeghijam
Assistant professor of Iran university of medical sciences
2023, May

What is the feedback in the hearing aids?

- Feedback oscillation (when a hearing aid whistles) is a major problem with hearing aids.
- The term feedback means that some of the output of the hearing aid manages to get back to the input of the aid (i.e., it is fed back to the input). Of course, when it does get back to the input, it is amplified along with every other signal arriving at the input. If it has grown stronger while traversing around that loop once, then it will grow stronger still the next time, and the next and the next, and so on.

Feedback mechanism……………….



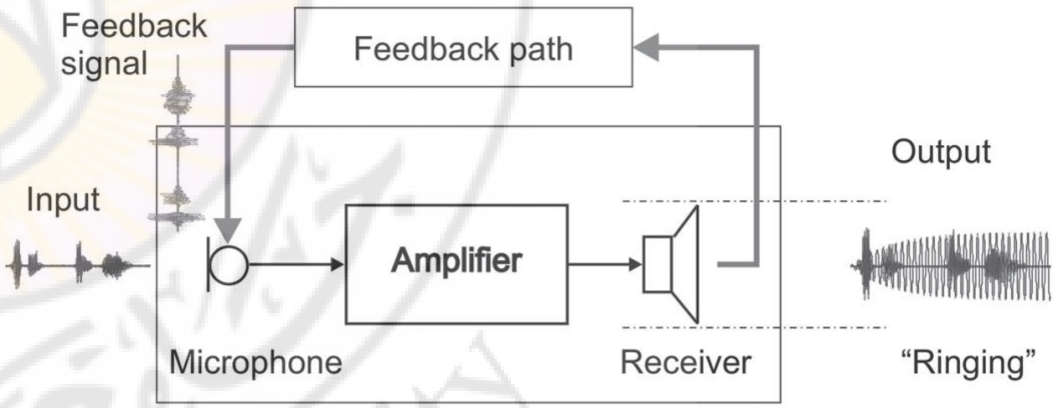


Figure 7–61. Schematic of a condition with acoustic feedback. (From "Hearing Aid Technology," by I. Holube and V. Hamacher, 2005, in J. Blauert (Ed.), *Communication Acoustics*, Springer, p. 270. Copyright 2005 by Springer. Adapted with permission.)

How much signal has to be feedback to create this unwanted oscillation?

- oscillations can happen only if the amount of amplification through the hearing aid is greater than the amount of attenuation from the ear canal back to the microphone. Thus, only if the real-ear aided gain of the hearing aid, (from input to the residual ear canal volume) is greater than the attenuation (from the residual ear canal volume back to the microphone) can continuous feedback oscillations occur. And if these sounds that come back to the microphones would be in phase, they will be added. So, if phase shifting would be very large (less than 360 degree), it is possible that feedback is cancelled.
- In the other word, it happen when the open loop gain of the hearing aid (the total gain travelling forward through the hearing aid amplifier and transducers, and then backward through the leakage path) has to be greater than 0 dB.

For example:

what happen in this situation?

Suppose, for example, that a test signal emerging from a hearing aid had an SPL of 90 dB SPL in the residual ear canal, but had an SPL of 60 dB SPL by the time it leaked back to the microphone inlet via a vent.

Feedback mechanism…….

- A hearing aid will oscillate at any frequency at which the forward gain is greater than the leakage attenuation, and at which the phase shift around the entire loop is an integer multiple of 360°.
- we have two types of feedback**: positive and negative feedback.**
- When the sound combines with the sound already in the ear canal in this constructive way, it is called **positive feedback**
- Positive feedback acts to increase the gain of the hearing aid. Indeed, a whistling hearing aid can be considered to have infinite gain at the frequency of oscillation.

Feedback mechanism…………..

- When the complete loop has a phase shift of 180°, 540°, or 900°, and so on in 360° steps, the sound fed back partially cancels any incoming sound.
- The effective gain of the hearing aid is decreased and we refer to this process as **negative feedback**.
- Negative feedback cannot cause oscillations.
- As with everyday use of these technical terms, positive feedback causes something to increase, whereas negative feedback causes something to decrease. For hearing aids, the "something" is their gain

What things increase the occurrence of feedback oscillation?

- positioning a sound reflector near the hearing aid, such as a telephone, or the brim of a hat; even movements of a telephone away from the ear by as little as 10 mm might avoid the problem.
- talking or chewing, such that the ear canal changes shapes and creates a sound path past the mold.
- growth of the ear canal, particularly in children
- shrinkage of the mold when it becomes old
- venting / open hearing aids
- increases in the high power hearing aids

What algorithms can prevent feedback from occurring?

- There are some methods for cancelling feedback, but none of them can cancel feedback oscillations entirely.
- The need for carefully made impressions and earmolds or shells is very important.
- Feedback path cancellation has closed enough to avoid feedback oscillation and achieve sufficient gain to make the hearing aid.

The benefits of feedback cancellation systems……

- The benefit of advanced feedback cancellation algorithms, however, is significant, especially with open canal (OC) fittings.

- A good feedback suppression system can provide 15 dB or more of added stable gain (ASG)—the difference in gain with the feature turned “on” versus “off.”

- This can make a huge difference in the audibility of speech (especially soft speech), and subsequently improve speech understanding.

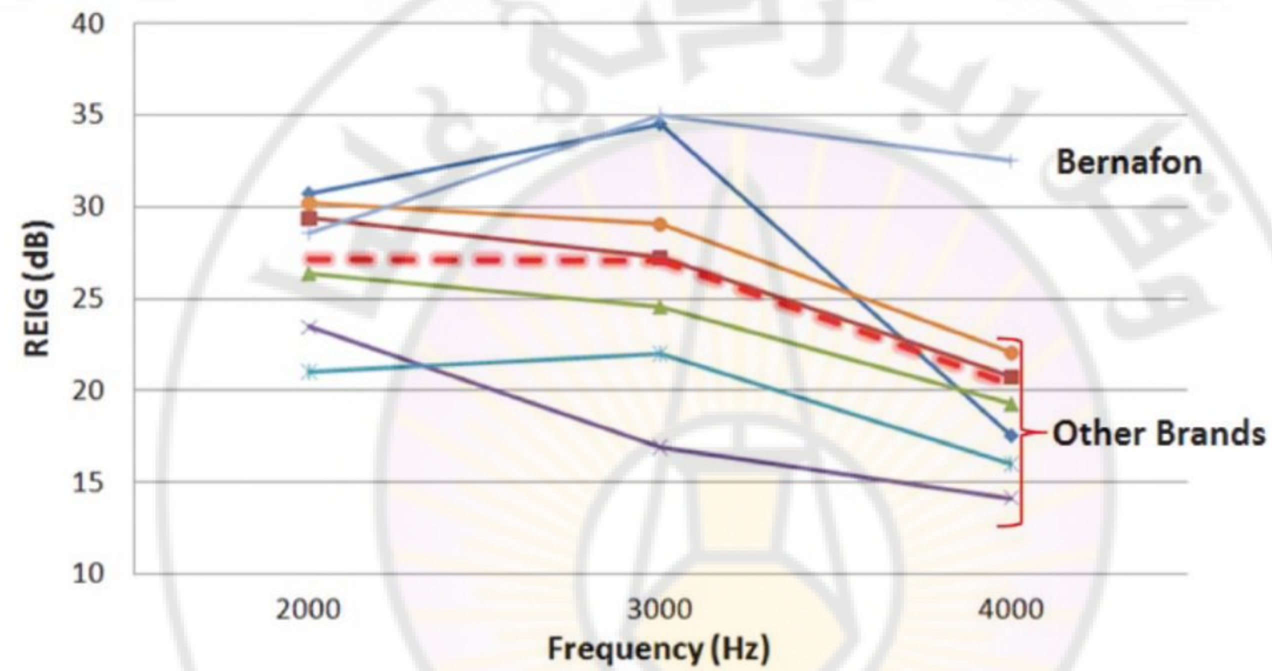


Figure 1: Maximum real ear insertion gain before feedback

Fig. 1 shows seven solid lines of different color, each representing the performance of a particular hearing aid brand. In addition, there is a dashed red line that shows the average performance of all brands. First of all, the diagram shows a clear distinction between brands. Whereas the performance of all other brands drops to below 25 dB at 4 kHz, the AFC Plus exceeds 30 dB at both 3 and 4 kHz – with an advantage of 10 to 15 dB compared to the competitors' average.

What is the critical gain measurement or feedback curve or static gain reduction at the beginning of the software fitting?

- Most fitting software will allow you to run a “feedback curve” prior to the fitting.
- By comparing the gain delivered to the ear canal to the gain leaking out of the ear, the software will predict if and where feedback will occur, and you can choose to implement feedback reduction to prevent this.
- the fitting system at the time of fitting – this is achieved by performing an in-situ feedback test, in which the fitting system automatically raises the gain in each channel until it detects oscillation occurring.
- Static gain limitation measures typically strongly limit the gain and thus the fitting possibilities of open hearing aids to compensate severe hearing losses.
- We prefer to save this approach as a “last-ditch” measure when other gain adjustments were not effective.

Feedback control by gain reduction: stable or adaptive

- one way to avoid feedback oscillations is to decrease the gain at all those frequencies where these conditions are met. This can be done in several ways.
- turn down the volume or gain control below the point require. This is obviously unsatisfactory, as it will give the patient inadequate loudness, audibility, and intelligibility.
- A better alternative is to decrease the gain at only those frequencies where feedback oscillation is a possibility. These frequencies are usually the peaks in the frequency gain response. In the other words, Once the feedback frequency is detected, a narrowband notch filter is applied at this frequency to eliminate the feedback.
- So, multichannel hearing aids provide a more reliable way to decrease gain in only one frequency region.
- digital filtering that control feedback precisely. Because the bandwidth of their effectiveness is very narrow.

Feedback reduction by gain-frequency response control

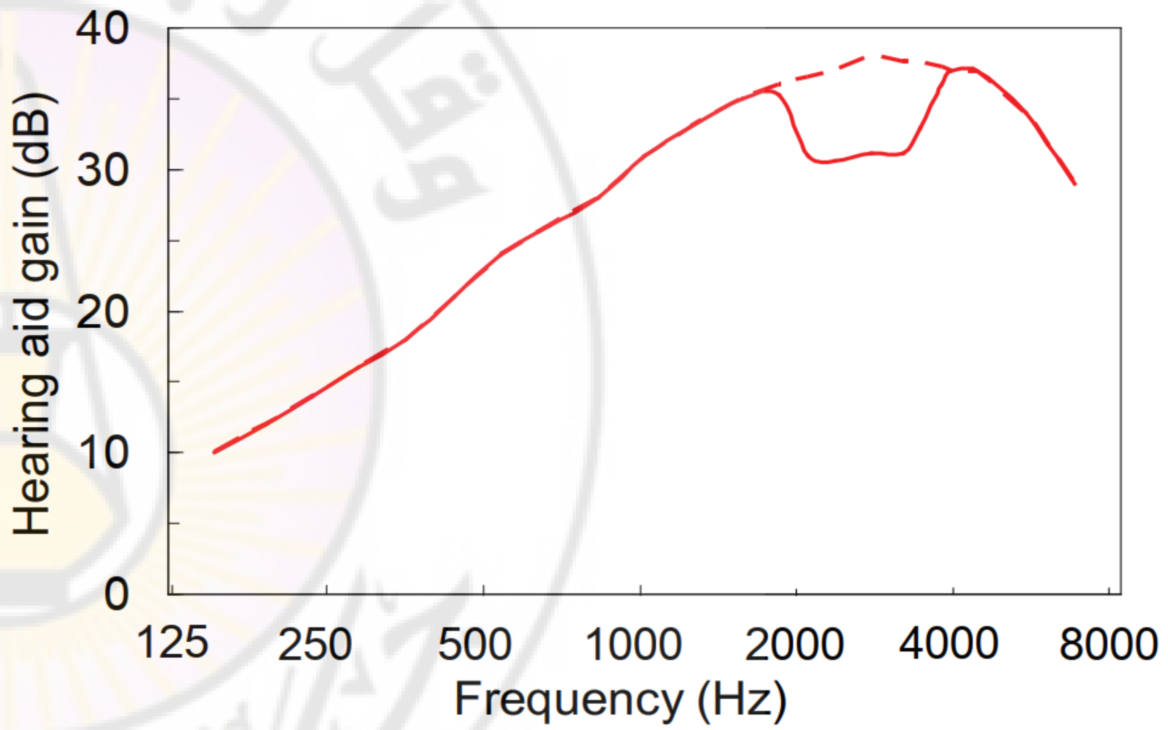


Figure 8.6 The gain-frequency response of a (hypothetical) four-channel hearing aid, where feedback oscillation has been avoided by decreasing the gain of the band from 2 kHz to 4 kHz (solid line) from the original response (dashed line).

Phase cancellation/frequency cancellation

- Once the feedback frequency is detected, the hearing aid introduces a signal that is 180 degrees out of phase from the feedback signal, which then serves to eliminate the feedback(phase cancellation).
- In some cases, the frequency also is slightly shifted, which makes the process more effective(frequency cancellation).
- In theory, once the phase cancellation has been activated, the resulting real ear output is relatively unchanged from what it was before the feedback occurred.

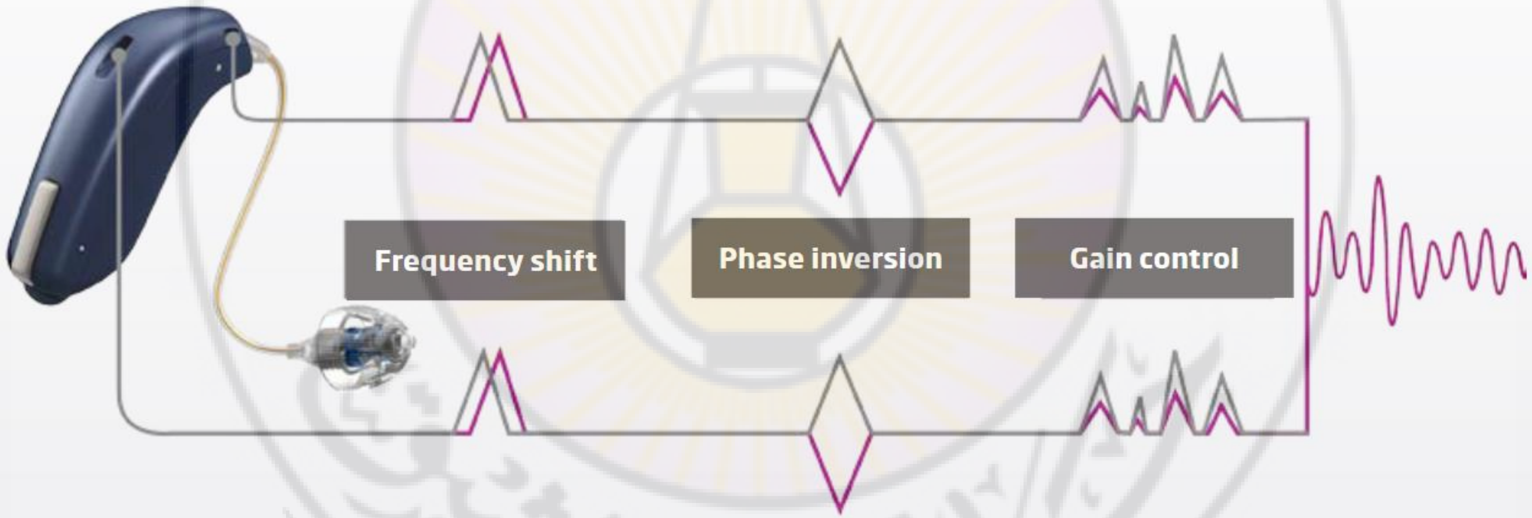


Figure 2. All three feedback strategies are continuously engaged. Here shown for the two separate microphones.

A most important tip…………………………

- It’s important to note that adaptive feedback control does not make up for a bad fitting (e.g., a bad ear impression resulting in an unusually “leaky” fitting).

- Its primary purpose is for transient feedback that might occur when the patient’s hand is placed close to the ear (when adjusting the VC or changing programs), or when an object like a telephone receiver is placed near the ear.

Open fitting and feedback cancellation………

- OC fittings are very popular, reportedly accounting for 75% of the fittings in some offices.
- We dare say that if they had not the adaptive feedback algorithms like what are now available, they would be far less popular.
- With OC fittings, there is nearly always a substantial amount of sound leaking out of the ear canal. With OC instruments, a good feedback system usually allows us to achieve as much as 10 to 15 dB more stable gain than we could obtained if this feature were not available.

Feedback cancellation and its downsides…………

- There are not many downsides to a good feedback reduction algorithm, except increased battery drain if it is running continuously or most of the time.

Feedback cancellation and its downsides…………

- One downside, but minor, is when that the adaptive feedback system “thinks” tonal sounds, such as a musical passage is acoustic feedback from the hearing aid, it tries to reduce the gain of the incoming tonal sound. This particular side effect causes the hearing aids to produce a warbling like sound called entrainment.
- sometimes we can solve this problem with going into the fitting software and making some adjustments to how the adaptive feedback settings.
- In recent years, manufacturers have mostly solved this problem by having the hearing aid “remember” the feedback frequency.

Directional Microphones

Dr. Maryam Sadeghijam

Assistant professor of Iran university of medical sciences

Directional microphones……………………..

- Directional microphone systems depend on noise to be spatially separated from speech, or other sounds the listener wants to hear.
- the effectiveness of directional HA in the reverberant situations is the worst.
- there have been two ways directionality has been achieved in a hearing aid:
  - Single microphone with two inlet ports and acoustical delay, that is seldom used in modern hearing aids.
  - Two omnidirectional microphone and electronic delay, with more flexibility and as tunning can be automatic and adaptive during use.

How does directional Mic. work?

By changing them ratio of the internal delay to the external delay, a whole family of patterns can be generated.

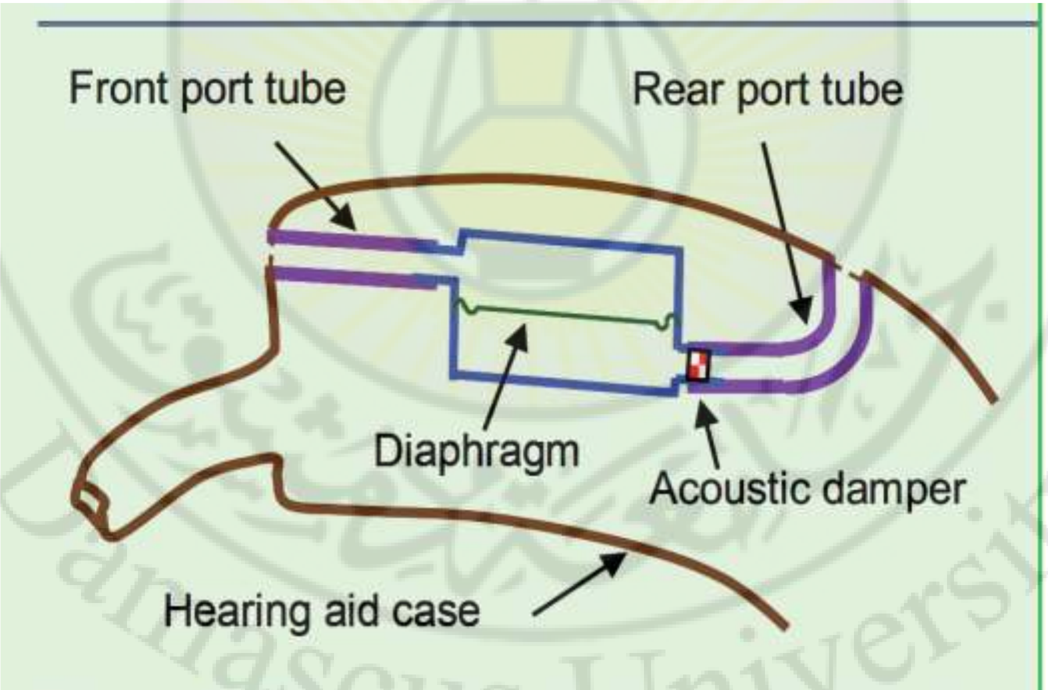
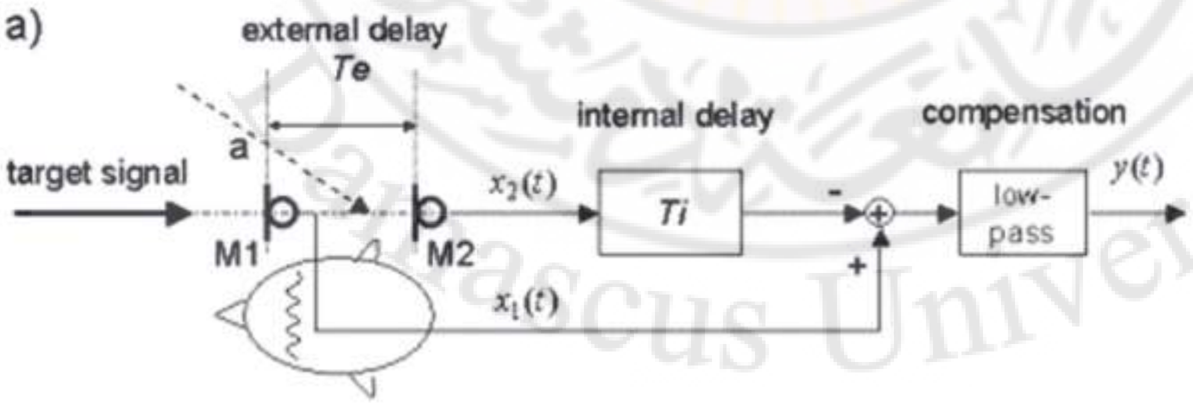


Figure 2.6 Diagram showing the sound paths in a directional microphone.

Two omnidirectional method……………

- Subtracting the output of the rear microphone from the output of the front microphone and adding an electronic time delay to the output of the rear microphone provides an improved signal-to-noise ratio.
- The distance between the two microphone inlets determines the amount of delay to the signal from the rear microphone.



What is the directivity index?

This ratio of sensitivity for frontal sounds relative to sensitivity averaged across all other directions is referred to as the directivity index (DI). The “other directions” can be restricted to just the horizontal plane (giving a two-dimensional DI) or to all directions in space (giving a three-dimensional DI)

An omni-directional microphone has a DI of 0 dB when measured in free space, but when mounted on the head its DI varies from about -1 dB in the low frequencies to 0 to 2 dB in the high frequencies, depending on the microphone location.

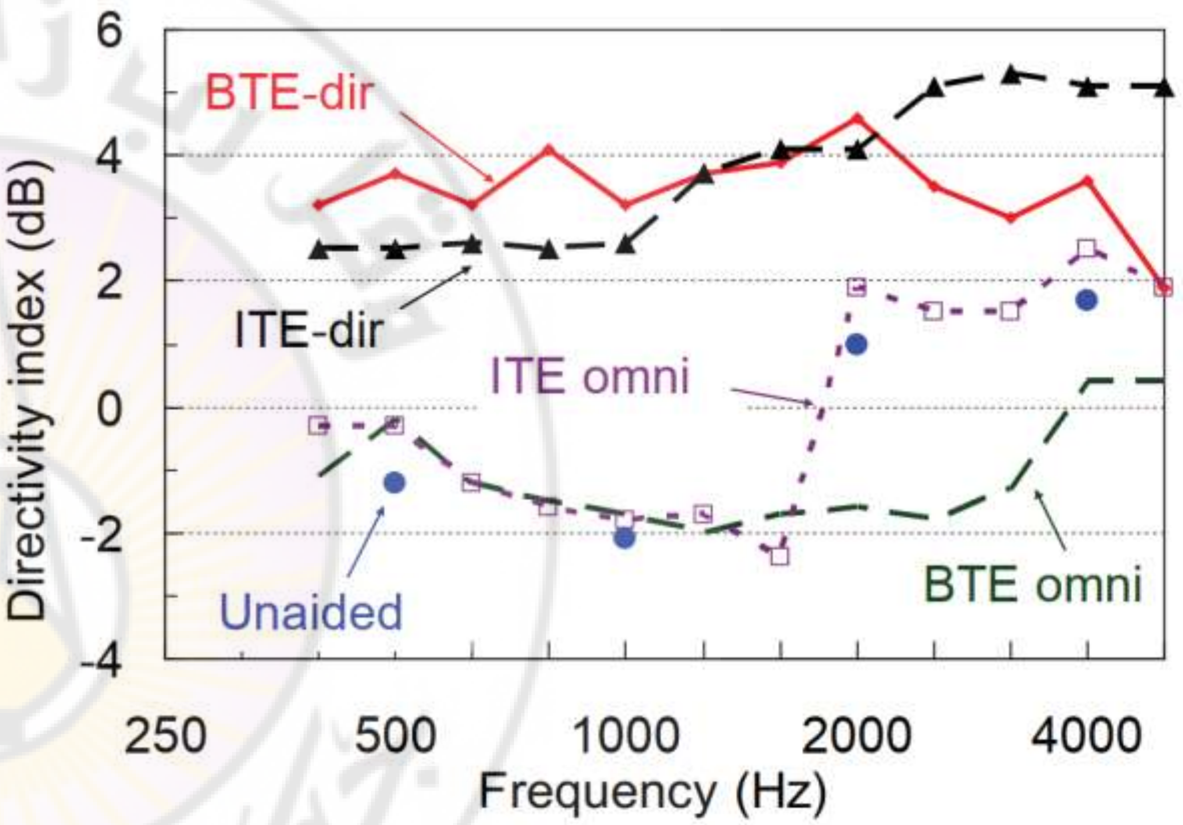


Figure 2.8 Directivity index, measured in the horizontal plane, for a directional and omnidirectional BTE and a directional and omnidirectional ITE.[1504] Also shown (as blue dots) is the directivity index for an unaided ear, measured on KEMAR.[454]

Some important tips……………………

- Relative sensitivity of two microphones (equal sensitivity at all frequencies) is very important and it should be quiet similar.
- If relative sensitivity of them isn't same, electronic drift creates when the two microphone system is out of electronic alignment.
- The dirt and debris that has collected in the ports can create drift and decrease the directivity index.
- So, Clean Ports Are Critical.

Frequency Response Equalization……………

- Directional microphones inherently reduce low-frequency output. This is referred to as an unequalized frequency response.
- The exact amount of low-frequency gain reduction varies, but typically there is a 6-dB per octave roll-off of low frequency energy beginning around 1000 Hz.
- When the low-frequency response is increased to be the same the omnidirectional response, this is referred to as an "equalized frequency response.
- To equalize, it is necessary to increase low-frequency amplifier gain, which can make the hearing aid sound "noisy".

Polar patterns……………………….

- the most common laboratory method for evaluating directionality is polar patterns.
- A polar plot is constructed by measuring the output of the hearing aid at several points within an imaginary sphere around the hearing aid microphone.
- These results are plotted relative to the output at a 0 degrees azimuth in both the horizontal and vertical planes.

Polar patterns……………………

❖Polar plots can be expressed in two ways.

❖One way is to show polar plots for several key frequencies (0.5, 1, 2, and 4 kHz), because the directivity and gain is not equal for each of them.

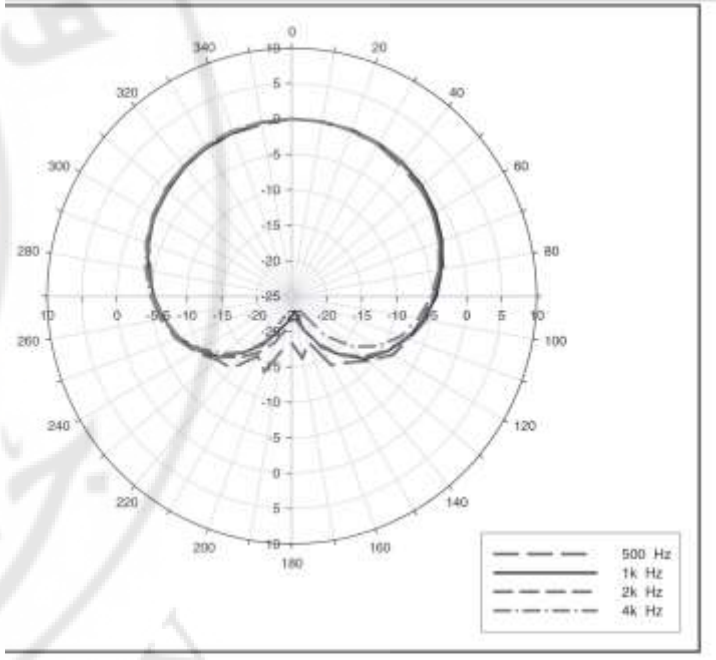
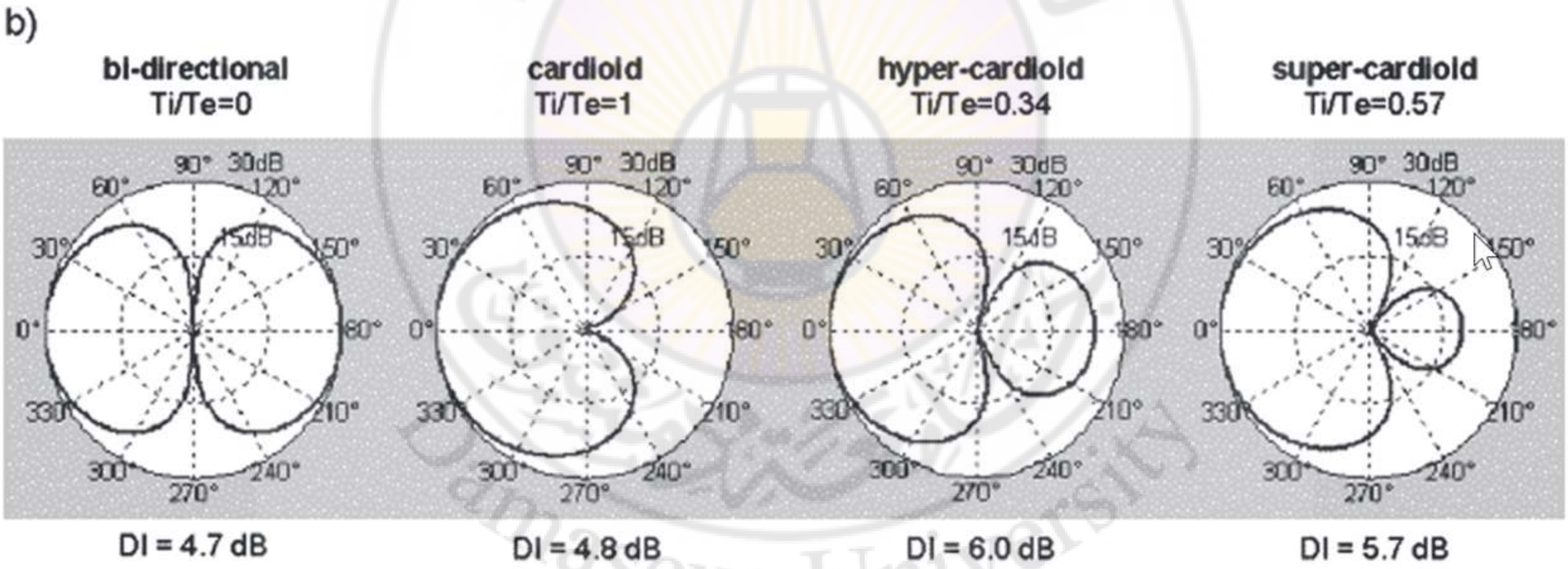


Figure 9–6. Frequency-specific polar plot for four frequencies. Data provided by Y. Wu, University of Iowa.

Polar patterns…………………

- Another way is to average the four key frequencies together and express the polar plot as a single line.
- Polar patterns can be measured with the hearing aid positioned in the sound field (e.g., attached to a microphone stand), or with the hearing aid placed in or on the ear of the KEMAR (an electronic manikin used for acoustic research).

Polar patterns……………………………



Polar patterns……………………

- Evaluating of polar patterns at the KEMAR curves, is somewhat more “real world” as they show how the directionality actually works on the head, which also includes the directional effects of head reflections and head shadow.
- It’s common for a custom instrument to be “more directional” with the KEMAR measure, as concha effects can enhance the directivity in the higher frequencies

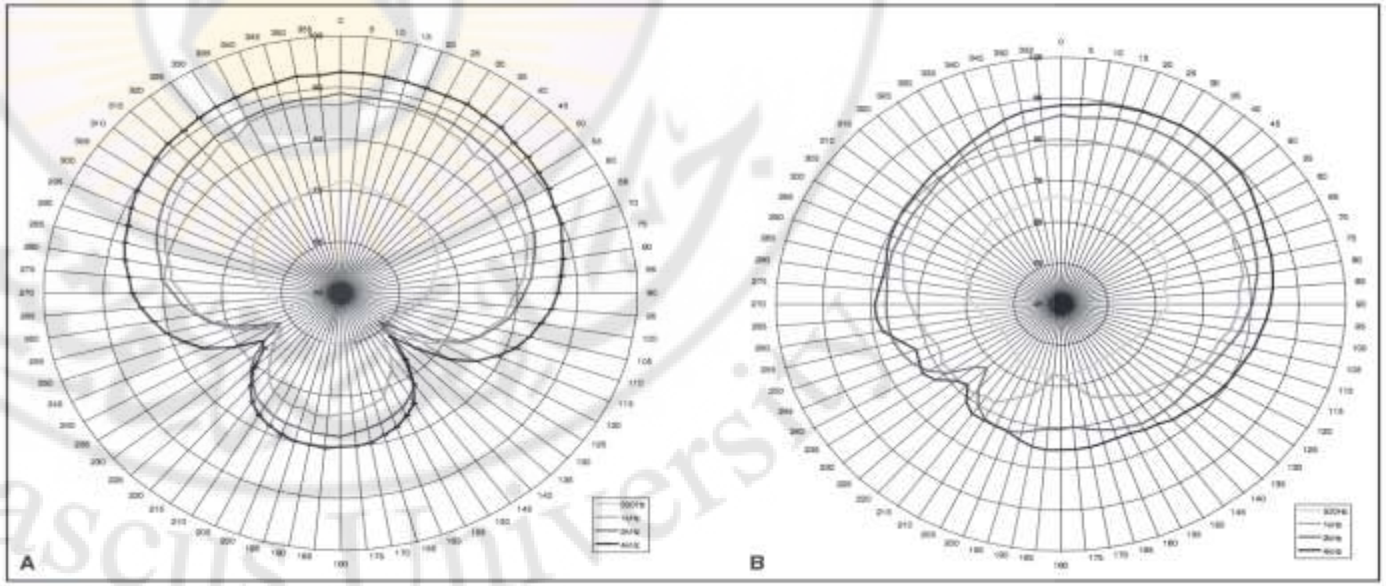
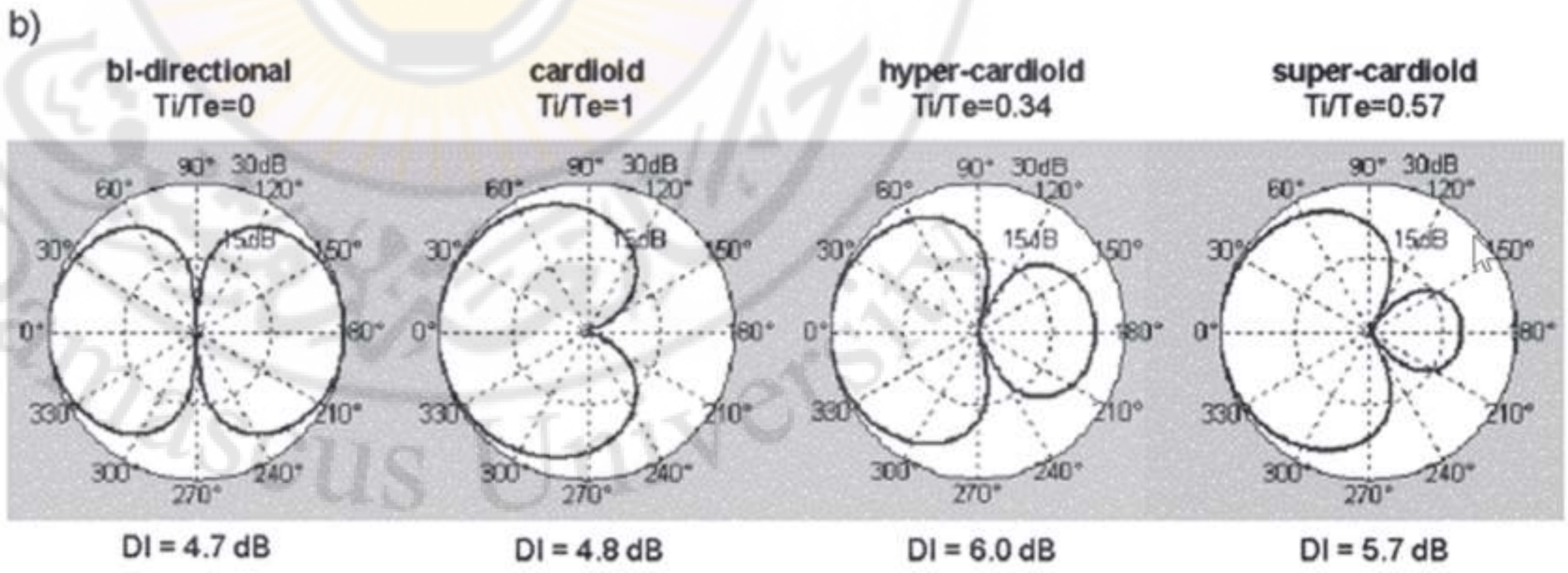


Figure 9–7. **A.** The polar plots for a hearing aid measured in the free field. **B.** The polar plots for the same hearing aid measure the KEMAR.

Different types of polar patterns………..

- Cardioid (upside-down heart appearance)
- Hyper cardioid (more reduction from back, but not at 180 degrees)
- Super cardioid (more reduction from back, but not at 180 degrees)
- Bidirectional (figure-8 pattern).



The Directivity Index…………..

- The DI is a ratio that compares the output of the signal at 0-degree azimuth to the output of the average of all other azimuths.
- For example, a given product at 2000 Hz, with a cardioid polar pattern, might have an output of 90 dB SPL at 0-degree azimuth and 70 dB SPL at 180-degree azimuth: a front-to-back difference of 20 dB.
- However, if we average the output from all the measured azimuths between 0 and 360 degrees, we might find that the average is 85 dB SPL — the DI therefore would be 5.0 dB (90 dB minus 85 dB).

Common Directional Hearing Aid features…………….

- **Automatic Switching:** With most directional products, the hearing aid will automatically switch to directional when the user is in a noisy situation, and will automatically switch to omnidirectional when the user is in a quiet listening environment.

- **Adaptive Polar Pattern:** Directional hearing aids with dual microphones easily can be adjusted to different polar patterns, depending on the electronic delay that is introduced. Different patterns have polar nulls at different azimuths.

One important point…………..

If there is not one specific source of noise, or the room is reverberant and the noise is bouncing around the room, the hearing aid will classify the listening situation “diffuse field” and will default to the best algorithm for that condition (usually hyper cardiod).

Adaptive Directional Microphones (Beamforming) and Audio Data Transfer Between Hearing Aids……

- In reality, beamforming describes the polar pattern of the directional microphones and their ability to change based on the environment.
- In recent years, wireless technology has provided the opportunity to couple the right and left hearing aids, thereby improving the beamforming properties of the devices, essentially turning a pair of hearing aids into a single system that works together.
- So, directional microphones, mounted on the left and right ears, respectively to work together as a pair.

Binaural directionality……….

- The automatic adjustment of polar plots by both hearing aids via wireless audio data transfer is commonly referred to as null steering.
- Manufacturers may call this feature “binaural directionality,” although it is really “bilateral directionality.”

Relation Between DI and Speech Understanding……

- It seems likely that the DI measures should provide a reasonable prediction of speech understanding in noisy situations.
- For mild to moderate-severe losses, and when the patient is more or less surrounded by noise, an approximate 7% to 10% improvement in speech understanding can be expected for every 1 dB improvement in the DI.
- For severe to profound losses a 3.5% improvement can be expected for every 1 dB of improvement.

Factors Affecting Directional Microphone Performance……………

- Venting effects
- Microphone port alignment effects
- Distance and Reverberation effects
- Low frequency gain
- Compression
- Real world benefits of directional microphones

Venting and directionality…………..

- Studies have shown that directivity is significantly **reduced** with increasing vent size.
- Consequently, maximum directivity is achieved with no vent.

- Venting disappears the directionality effects at low frequencies. But, hearing aids with large vents and open-canal fittings still maintain good directionality in the higher frequencies.

- Behavioral research has shown that indeed this high-frequency directionality does result in improved speech recognition in noise, although not as great as if there also was a significant directional effect in the lows.

Microphone Port Alignment Effects……….

- Clinical studies have demonstrated that directivity is negatively impacted with as little as a 10- to 15-degree deviation between the microphone port alignment and the horizontal plane.

- Microphone port alignment can be a particular problem with the mini-BTE instruments that have become popular in recent years.

- Of course, comfortable is more important than the a few directionality.

Distance and Reverberation…….

- Reverberation time is the parameter commonly used to describe room acoustics, although background noise and distance from talker also are important factors in determining if communication is successful.

- As a speaker moves farther from the hearing aid user, speech intelligibility in noise decreases. Also, as room reverberation increases, directional benefit decreases, thus decreasing speech intelligibility in noise.

Low frequency gain……..

The rule of thumb for all losses greater than 40 dB HL at 500 Hz is to always equalize (increase the gain) the frequency response in the directional mode (Figure 9–8)

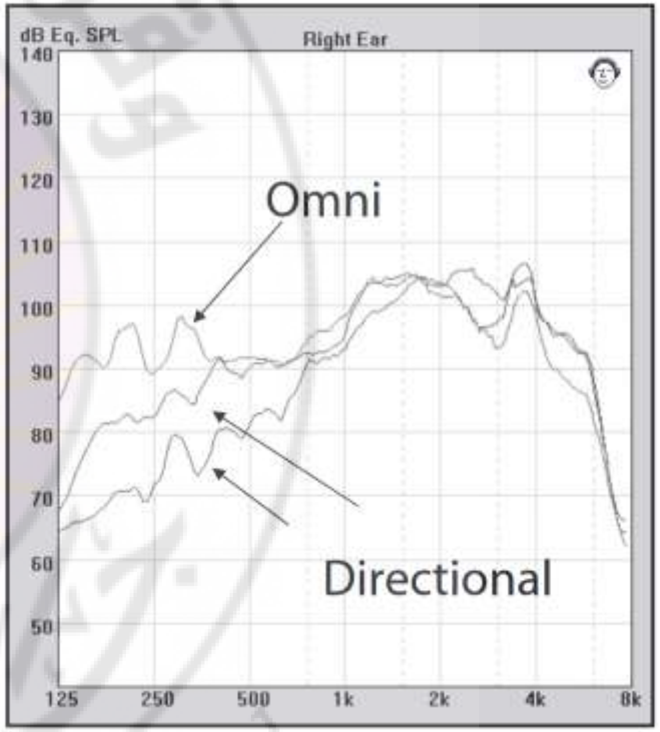


Figure 9–8. The low-frequency reduction of a typical hearing aid in the directional microphone mode (*lower curve*). The middle curve is the result of applying additional low-frequency gain.

Compression…………………

- An interaction between compression with low kneepoints (40 to 45 dB SPL) and directivity is possible due to the way in which directional microphones work.
- WDRC hearing aids provide more gain for sounds arriving from azimuths for which amplitude is reduced by the directional microphone than for those sounds arriving from azimuths for which there is little or no gain reduction. That is, the directional microphone reduces the input, but WDRC boosts it back up to some degree because it provides more gain for soft sounds.
- This interaction results in a reduction in the magnitude of directivity on hearing aids with low kneepoints compared to hearing aids with higher kneepoints.

The good news is that studies have shown that compression does not affect directivity (better or worse) when it is measured in real-world listening environments in which noise is arriving at the microphone from multiple sources at the same time as the speech signal.

Real world……………………..

- The studies suggest that the performance of directional microphones in everyday listening is highly dependent on the characteristics of the listening environment.
- Location of the talker, the talker's distance from the user, amount of noise, and reverberation can affect directional microphone performance.
- These studies suggest that the omnidirectional mode is preferred in quiet listening environments and in the presence of background noise when the talker was not located directly in front of listener and/or when the talker was more than 10 to 12 feet from the user.
- The directional mode appears to be preferred when background noise was present and the talker was located in front of and/or 10 feet or closer to the hearing aid user.

Noise Reduction in hearing aids

Dr. Maryam Sadeghijam

Assistant professor of Iran university of medical sciences

History…………………

- Starting with body aids, in the 1940s there was a low-frequency reduction available that was advertised as “noise reduction.
- In the 1980s, single-channel analog hearing aids were equipped with active low-cut tone controls as a way to manually reduce low-frequency types of ambient sound.
- Also in the 1980s, automatic signal processing was introduced as a method for automatically reducing low-frequency sounds, a technology made famous when President Ronald Reagan was fitted.

History……………..

- ASP and BILL/TILL/ Manhattan II, Adaptive Compression and the Zeta Noise Blocker, were also introduced around this time with mixed results.
- But, all of above noise reduction, reduce speech as well as noise and degraded the speech intelligibility.
- todays, we have digital noise reductions.

How does DNR work?

- Often, the interworking of noise reduction algorithms is shrouded in mystery.

There are some commonalities that can help us demystify the black box

Modulation-Based Noise Reduction

- signal classification system on board the hearing aid analyses the signal, looking at the number and depth of modulations as well as many other characteristics (e.g., speech usually has 4 to 6 modulations/second).
- Modulation-based noise reduction systems work under the premise that speech has fewer modulations (Hz) with more depth (dB) than noise stimuli. Typically, modulation frequency and depth are analyzed independently in each channel of the instrument.

Modulation-Based Noise Reduction

- If the input signal is classified as noise the intensity of the signal is reduced, and if it's classified as speech the intensity level may be increased, or more commonly, the signal in that channel remains at programmed gain.
- The important point to remember regarding modulation based noise reduction is that when noise is found to be the dominant signal in a given channel, gain for everything is reduced: noise and speech.
- There is not an improvement in the signal to noise ratio within that channel and, hence, we would not expect an improvement in speech understanding.

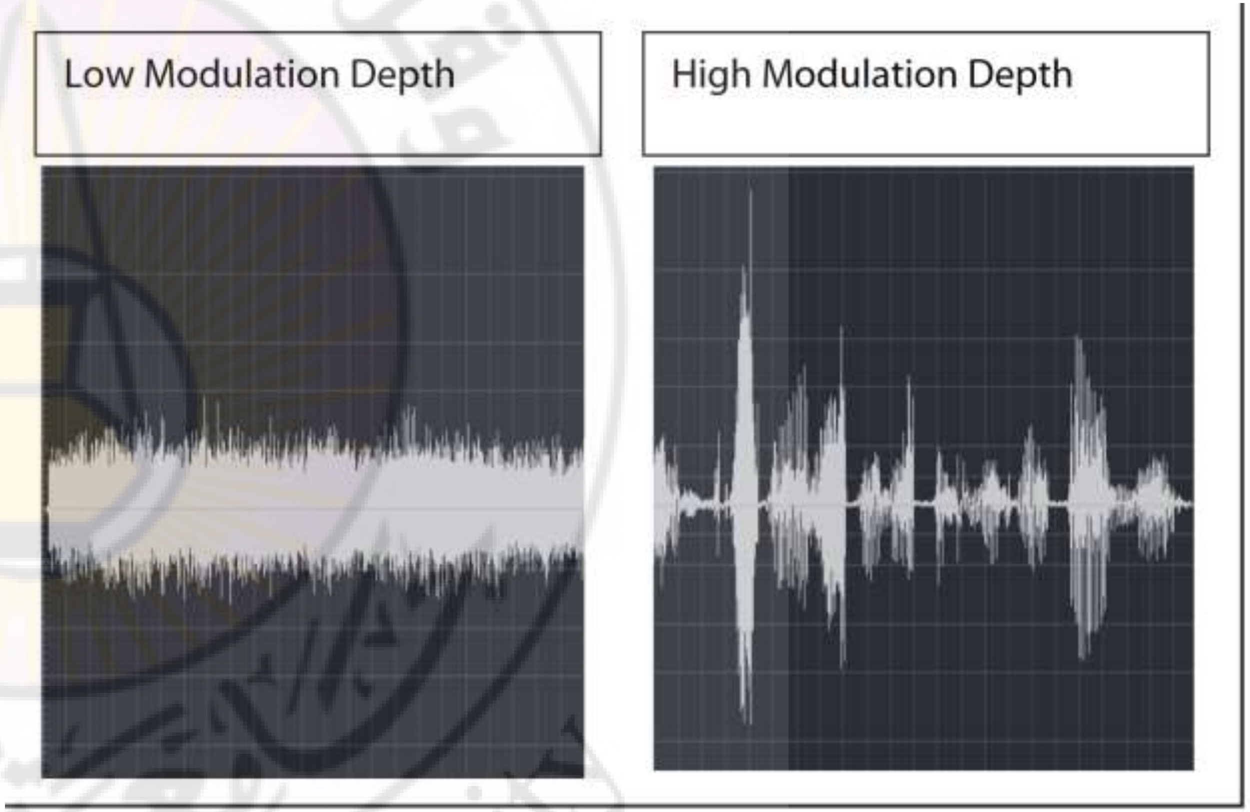


Figure 9–4. A comparison of modulation depth. The left signal has a low modulation depth. The signal on the right has a high modulation depth. Modulation depth is one of many characteristics used by the hearing aid's on-board signal classification to identify non-speech and speech-like signals.

Filtering and Subtraction

- The most hearing aids employ some combination of filtering, spectral subtraction, or co-modulation detection.
- These types of DNR systems are geared toward cleaning up the speech signal when speech is at least somewhat the dominant signal.
- One method is to look for gaps between speech signals and reduce the “noise” for this short duration. The trick of course, is to pull out the noise without also pulling out some of the speech.
- These systems work simultaneously with the previously mentioned modulation-based algorithms.

Impulse noise…………………

- Impulse noise reduction (termed things like “sound smoothing,”
- “sound relax,” etc.) is not geared toward speech understanding, but simply designed to reduce annoyance.
- That is, the classification system looks for very sharp peaks in the onset of a signal. If found, it’s assumed that this is noise (probably an irritating one) and is not speech.
- The initial peak of this signal is then reduced.

Selecting DNR parameters……….

- Gain reduction
- Gain enhancement
- Active time (onset and offset time)
- SNR and level effects
- Noise reduction and directional technology

Gain reduction……………….

- All products reduce gain in varying amounts across the frequency range, and might reduce it differently for different noises.
- With some products the "max" setting may result in a 4 to 6 dB reduction, while with other products, "max" could mean a 12 to 15 dB reduction for some noise inputs.
- And this will also vary depending on the spectrum of the noise and probably for the intensity of the noise.

Gain enhancement……………………

- The amount of gain enhancement varies across frequency as well.
- Some products have been known to even boost high-frequency gain when the SNR is adverse.
- We recommend using your probe-mic system with a real-speech input, and then measure the output for DNR-on versus DNR-off.
- This will help you understand what is happening when DNR is activated.

Active time (onset and offset time)

- once the input signal has been classified as noise, how long does it take to reduce the noise signal? To reduce it to its maximum? This could be as fast as a second or two, or as long as 5 to 10 seconds.
- Or, when the input signal that was classified as noise is no longer present, or speech is present that is more intense than the noise, how long does it take for programmed gain to recover?
- Again, this is something very easy to observe in the real ear with your probe-mic testing.

SNR and Level Effects

- Two other factors that can impact the DNR effects are the SNR of the signal, and the overall level of the noise.
- At what SNR is the noise reduction scheme activated, and how much is gain reduced at various signal-to-noise ratios?
- Is the noise reduction the same when speech is present?

Noise Reduction and Directional Technology

- Linking the DNR and directional microphone in the hearing aids is very helpful for understanding of speech.
- Consider a situation when the classification system has determined that our patient is listening to speech from the front, and is surrounded by noise.
- The system will then automatically implement directional technology with a front-facing beam. But if the classifier knows that the signal of interest is from the front, then it also can “amp up” noise reduction for sounds from the back, and improve the overall SNR by 1 to 2 dB compared to when they are not linked.

Some tips……………….

- Studies have shown that DNR couldn’t increase speech intelligibility, they just decrease listening fatigue and improve dual tasks.
- DNR makes listening in noise comfort and easy.
- Noise can affect on cognitive function. So, DNR can increase attention and cognitive functions.

Wireless Audio Data Transfer Between Hearing Aids

- This feature makes two hearing aids to “talk” to one another when they are worn by the patient. Today’s hearing aids employ this feature to accomplish many things.
- In the simplest form, adjusting the volume control or changing the program at the both of hearing aids by pushing a bottom on one hearing aid.
- This can be a big convenience, especially for patients with hand or finger dexterity problems. This technology has been around since 2004.

Wireless Audio Data Transfer Between Hearing Aids

- But today, Wireless audio data transfer between hearing aids is different.
- It is used to optimize the performance of features like directional microphones and digital noise reduction.
- For examples,
  - the patient can hold the phone up to one ear, and through wireless audio data transfer, hear the speech signal from the phone in both ears.
  - The patient can optimize speech understanding while riding as a passenger in a car, or for trying to hear his favorite nephew who always sits to his right at a crowded restaurant.

Multiple Memories

- Todays, this is a standard feature in the hearing aids (switching from memory to another memory automatically).
- If the signal classification is working properly, a lot of good things can happen within a single memory, automatically. However, in many cases, it’s still useful to have a dedicated memory for certain types of listening so that the user can override the automatic functioning.
- Some of today’s products automatically switch to the telecoil memory when the phone is brought to the ear.
- In each program, it is important to activate some advanced features. For example, in the music program, WDRC is not activated and AGCO has a higher knee point.

Multiple Memories

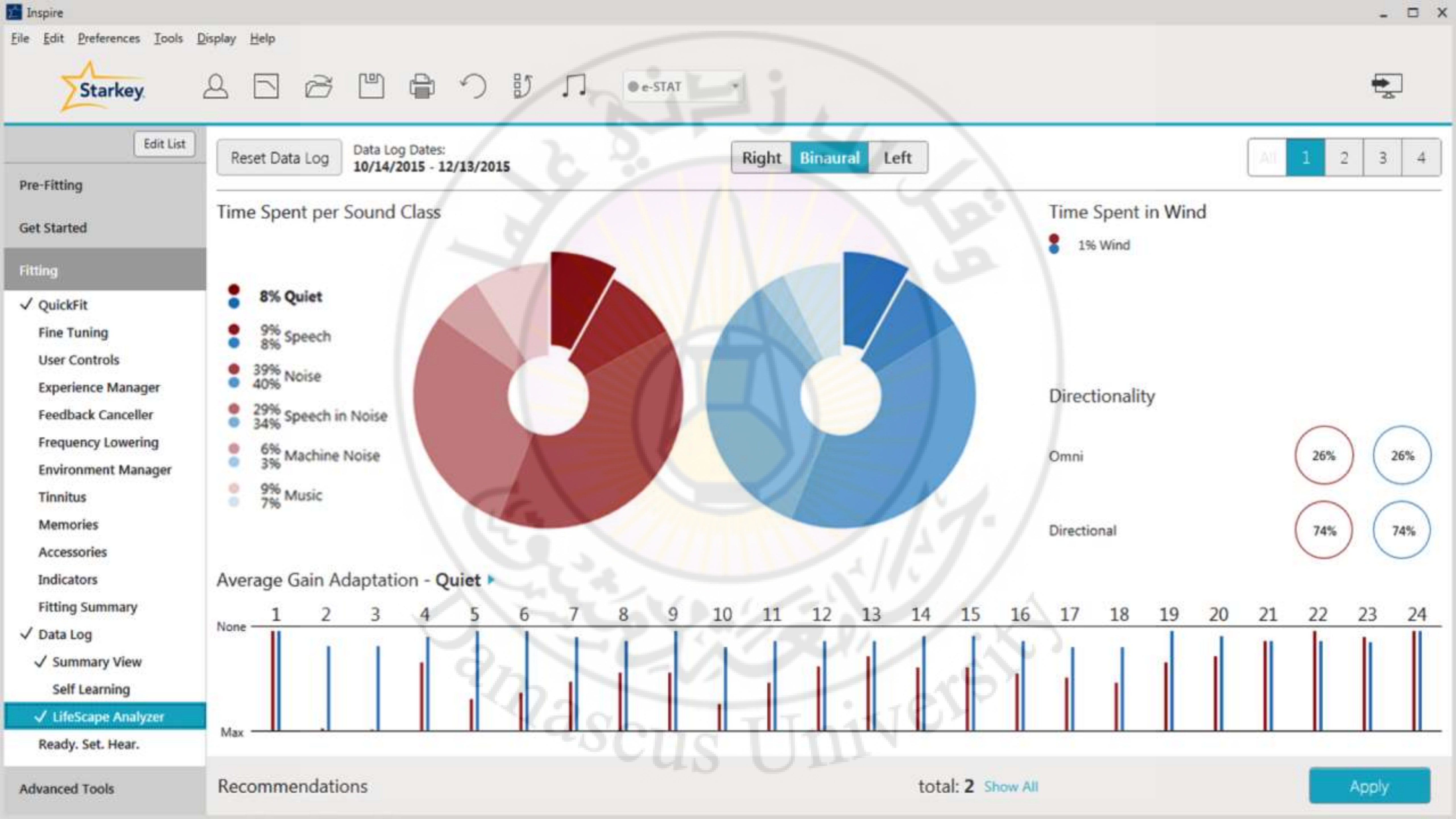
- The memories sometimes can be used to assist in the fitting.
- For example, it is difficult to determine when a patient might benefit from frequency lowering. You could easily program traditional amplification in one memory, and frequency lowering in a second memory, allowing the patient to switch back and forth in many of his or her real-world listening situations.
- With more sophisticated classification systems, the need for different memories is decreased. For example, today’s hearing aids can automatically detect that music is present, and then automatically turn off directional technology, DNR, adjust gain, input and output compression, and so forth, all within the same memory.

Data logging

- Hearing aid can store all information that achieved by classification systems from different situations and store all actions that take place: VC changes, on-off changes, use of DNR, use of directional, changes of programs, and so forth. When all this information is stored, it’s called “data logging.”
- Data logging can be used at different times and for various purposes throughout the fitting process.
- The four most common general uses appear to be: counseling at the time of the fitting, routine counseling during the post fitting visits, troubleshooting patient complaints, and using data logging results to change the programming of the hearing aids (often related to hearing aid “training”)

Data logging

- Some information that achieved by data logging include:
  - Minimal hearing aid use: What’s the problem? Poor performance? Unrealistic expectations? A change in lifestyle? Illness?
  - Minimal use for only one hearing aid: Poorer performance with two versus one? Has the patient given two hearing aids a fair shot? A cosmetics issue? Uncomfortable fit?
  - Much less use than verbally reported: Why the discrepancy? Trying to please dispenser or family members? Using a dead battery?
  - Much more use than verbally reported: Neglecting to turn off hearing aids at night, or when not using them?
  - Minimal use of additional programs: How does this compare with environment logging? Understand the purpose of the different programs? How to switch? Are all the additional programs really necessary?


Inspire
File Edit Preferences Tools Display Help
Starkey
e-STAT
Edit List
Pre-Fitting
Get Started
Fitting
✓ QuickFit
Fine Tuning
User Controls
Experience Manager
Feedback Canceller
Frequency Lowering
Environment Manager
Tinnitus
Memories
Accessories
Indicators
Fitting Summary
✓ Data Log
✓ Summary View
Self Learning
✓ LifeScape Analyzer
Ready. Set. Hear.
Advanced Tools
Reset Data Log
Data Log Dates:
10/14/2015 - 12/13/2015
Right Binaural Left
All 1 2 3 4
Time Spent per Sound Class
8% Quiet
9% 8% Speech
39% 40% Noise
29% 34% Speech in Noise
6% 3% Machine Noise
9% 7% Music
Time Spent in Wind
1% Wind
Directionality
Omni 26% 26%
Directional 74% 74%
Average Gain Adaptation - Quiet
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24
None
Max
Recommendations
total: 2 Show All
Apply

Data logging for data training

- Today, data logging can record the users preferences for different listening environments (based on signal classification), for all input levels, including gain preferences for loudness and frequency response for each environment being logged. The hearing aid can then be trained to automatically reprogram to these desired setting, when the situation is detected.
- In other words, ongoing gain, frequency, and compression training, automatically switching to different settings for speech-in-quiet, noise, music, and so forth, all in the same hearing aid program. This certainly will be much closer to a “tailored fitting,”

Trainable hearing aids

- Data logging is the basis of the trainable hearing aids.
- The first generation of products only had learning for overall gain, but now we have products that also can learn frequency response, microphone strategy, and compression.
- Today there are trainable hearing aids that automatically “train” for as many as six different listening situations identified by the signal classification system.

Automatic Acclimatization

- Some dispensers have the patient come back at periodic intervals and gradually increase gain. But, as the name indicates, some hearing aid can be programmed so that the gain of the instrument can increase at a prescribed amount over a prescribed time frame.

- For example, you could fit a patient 5 dB below the NAL-NL2 target and then have the acclimatization feature increase 1 to 2 dB per week until the desired gain setting was obtained.

Wireless connectivity

- General, today we usually think of “wireless” as a radio frequency or electromagnetic signals that carry some type of communication signal over a desired pathway.
- A wireless technology, introduced in 1998, that is rapidly growing in use is the Bluetooth protocol. Bluetooth is radio technology for exchanging data over short distances and developing personal area networks.
- FM is a wireless application often used with hearing-impaired children. This is an effective method of overcoming the negative effects of background noise and talker distance, resulting in an improvement in the signal-to-noise ratio at the listener’s ear.

Wireless connectivity

- There are two types wireless communication in hearing aids:

  1- Hearing Aid to Hearing Aid Communication:

  - This feature is the audio data transfer of information between two hearing aids.
  - In 2004, wireless transmission was introduced that used magnetic transmission to communicate between the two hearing instruments, and also between hearing instruments and an optional remote control accessory.
  - This makes symmetric steering of important functions such as digital noise reduction and directional technology.

Wireless connectivity

2-Using Bluetooth as a streaming device the hearing aid user can connect to his cell phone, television, stereo system, MP3 player, or other audio products:

- The end result of using Bluetooth transmission is that the signal-to-noise ratio of the listening situation has been improved. In other words, the microphone is placed closer to the sound source and Bluetooth is used to transmit the signal directly to the amplifier and receiver in the hearing aid.
- In this system, a streamer or gateway device is used.

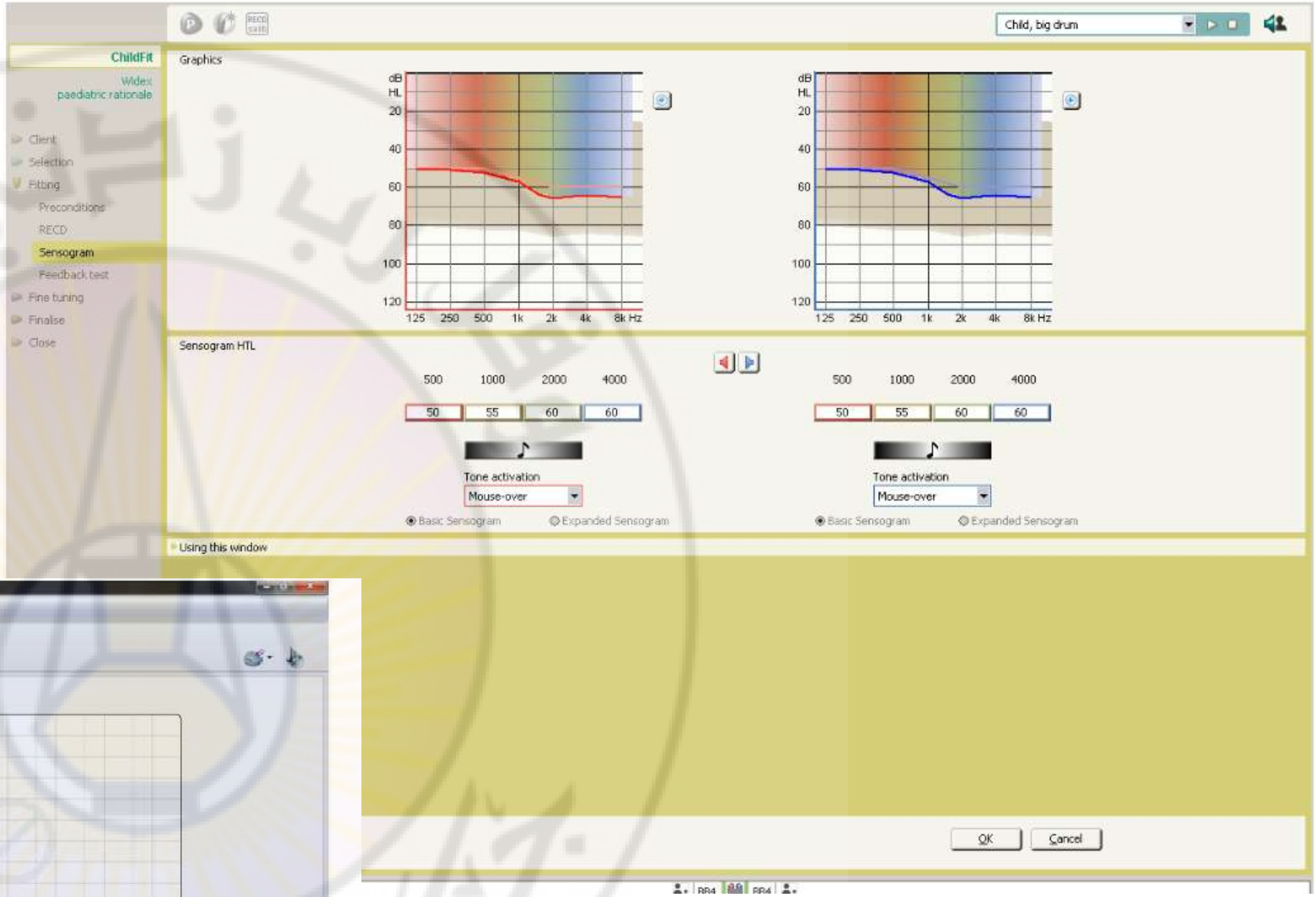
Bluetooth and 2.4 GHz Transmission

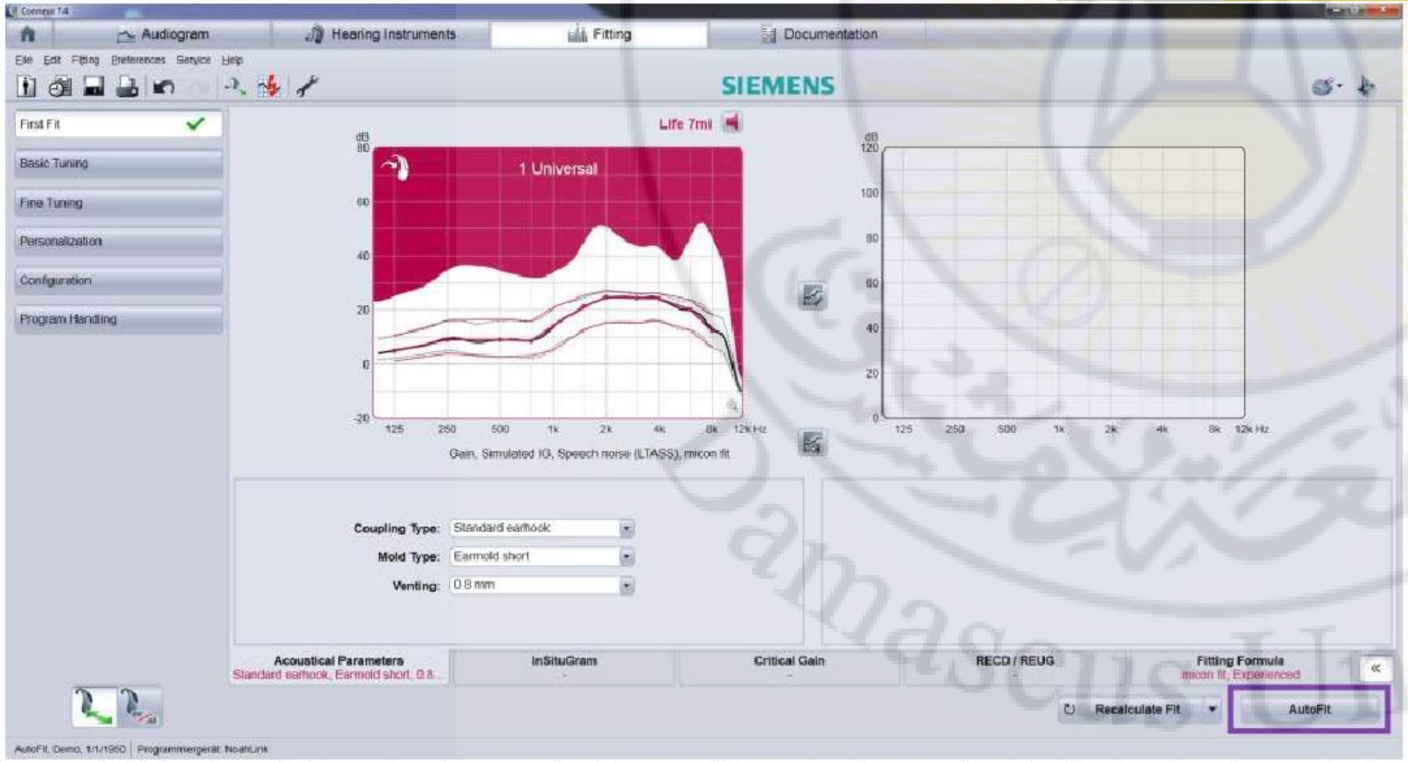
- Recently, just about all hearing aid manufacturers have introduced accessories like remote microphones that can be paired to hearing aids without a gateway device.

- In order to remove the gateway device that is often worn around the neck, a 2.4 GHz transmission signal is needed.

In situ testing

- The hearing aid is being used in place of an audiometer.
- The hearing aid send audio signals to the ear and audiologist can evaluate the auditory thresholds in all frequencies by the hearing aid.
- This technique is currently used to estimate thresholds, loudness levels, or LDLs, but could also be used to present speech material, or conduct special tests such as gap detection or the TEN test.





5. The AutoFit dialog is available under the AutoFit tab, accessible on the bottom right beside the First Fit / Recalculate Fi