Basic and License Preparation Course Manual

Table of Contents

Abbreviations

AAA	American Academy of Audiology
AC	Air Conduction
A/B	Air/bone
A/D	Analog to Digital
AD	Right Ear
AIDS	Acquired Immune Deficiency Syndrome
ASHA	American Speech, Language and Hearing Association
AS	Left Ear
ASL	American Sign Language
AU	Both Ears
ADA	Americans with Disabilities Act or Academy of Doctors of Audiology
AGC	Automatic Gain Control
AI/DI	Articulation Index/Directivity Index
ALD	Assistive Listening Device
ANSI	American National Standards Institute
ASP	Automatic Signal Processing
AVC	Automatic Volume Control
ABR	Auditory Brainstem Response
BC	Bone Conduction
BBN	Broad-band Noise
BTE	Behind-the-Ear (hearing aid)
BILL	Bass Increases at Low Levels
CARF	Commission on Accreditation of Rehabilitation Facilities
CFA	Continuous Flow Adaptor
CIC	Completely-in-the-Canal (hearing aid)
CK	Compression Kneepoint
CL	Comfortable Level
cm3	Cubic Centimeter
CNS	Central Nervous System
CNT	Could Not Test
CPS	Cycles per Second
CR	Compression Ratio
CROS	Contralateral Routing of Signal
CSF	Cerebro-spinal Fluid
D/A	Digital to Analog
dB	deciBel
dB(A)	deciBel (A-weighted)
DFS	Digital Feedback Suppression
DI	Directivity Index
DNR	Did Not Respond or
Digital Noise Reduction	
DNT	Did Not Test
DR	Dynamic Range
DS	Discrimination Score
DSP	Digital Signal Processing
DTT	Difficult to Test
EAM	External Auditory Meatus (ear canal)
ENT	Ear, Nose and Throat (physician)

EM Effective Masking or

Earmold

EMI Earmold Impression

EPA Environmental Protection Agency

FDA Food and Drug Administration

FF Free Field

FOG Full-on Gain

HAE Hearing Aid Evaluation

HD Harmonic Distortion

HFA High Frequency Average

HIMSA Hearing Instrument Manufacturers Software Association

HIPAA Health Insurance Portability and Accountability Act

HIV Human Immune-deficiency Virus

HL Hearing Level

HTL Hearing Threshold Level

Hz Hertz

IA Inter-aural Attenuation

IC Integrated Circuit

ID Internal Diameter

IE In the Ear (hearing aid)

IHC Inner Hair Cell

IIC Invisible-in-the-Canal (hearing aid)

ITE In-the-Ear (hearing aid)

ITC In-the-Canal (hearing aid)

IL Intensity Level

IM Inter-modulation Distortion

IHS International Hearing Society

IIHIS International Institute for Hearing Instrument Studies

IROS Ipsilateral Routing of Signal

JCAHO Joint Commission for the Accreditation of Healthcare Organizations

JND Just Noticeable Difference

KEMAR Knowles Electronics Mannequin for Acoustic Research

KHz Kilo-Hertz

LL Loudness Level

LDFR Level-dependant Frequency Response

LDL Loudness Discomfort Level

mA milli-Ampere

MCL Most Comfortable Level

MD Masking Dilemma

MLV Monitored Live Voice

MRSA Methicillin Resistant Staphylococcus Aureus

MSP Multiple Signal Processor

MRI Magnetic Resonance Imaging

mAh milli-Amp hours

MPO Maximum Power Output

MTO Microphone/Telephone/Off (switch)

n Number of Subjects (in a study)

NAEL National Association of Earmold Laboratories

NAL National Acoustics Laboratory

NASED National Association of Special Equipment Distributors

NBN	Narrow-band Noise
NIHL	Noise-Induced Hearing Loss
NR	No Response
NTE	Non-test Ear
OD	Outside Diameter
OHC	Outer Hair Cells
OAE	Otoacoustic Emission
OSHA	Occupational Safety and Health Administration
OSPL	Output Sound Pressure Level
PB	Phonetically Balanced
PTA	Pure Tone Average
PTS	Permanent Threshold Shift
PILL	Programmable Increases at Low Levels
REAG	Real Ear Aided Gain
REAR	Real Ear Aided Response
REIG	Real Ear Insertion Gain
REIR	Real Ear Insertion Response
REM	Real Ear Measurement
REOG	Real Ear Occluded Gain
REOR	Real Ear Occluded Response
RESR	Real Ear Saturation Response
REUR	Real Ear Unaided Response
RIC	Receiver-in-the-Canal (hearing aid)
RITA	Receiver-in-the-Aid (hearing aid)
RTG	Reference Test Gain
RTP	Reference Test Position
SAT	Speech Awareness Threshold
SAV	Select-a-Vent
SD	Speech Discrimination
SDS	Speech Discrimination Score
SDT	Speech Detection Threshold
SF	Sound Field
SIN	Speech in Noise
SISI	Short Increment Sensitivity Index
SL	Sensation Level
SLM	Sound Level Meter
S/N	Signal to Noise
SNR	Signal to Noise Ratio
SNAFU	Situation Normal, All Fouled Up
SPL	Sound Pressure Level
SRT	Speech Reception Threshold
SSPL	Saturation Sound Pressure Level
T-coil	Telephone coil
TD	Threshold of Discomfort
TE	Test Ear
TK	Threshold Kneepoint
TILL	Treble Increases at Low Levels
THD	Total Harmonic Distortion
TL	Tolerance Level
TM	Tympanic Membrane
TMJ	Temporo-mandibular Joint
TTS	Temporary Threshold Shift
UCL	Uncomfortable Level

WNL	Within Normal Limits
WR	Word Recognition
WRS	Word Recognition Score
VC	Volume Control
VU	Volume Units
WDRC	Wide Dynamic Range Compression

Unit 1: Introduction

Objectives

Notes

Overview

If you are looking for a fast-paced career that involves working with people, not to mention the opportunity to earn a good living, you've come to the right place. You can begin by reading the first chapter of this manual.

Follow Up

- Attend a Basic Training Course
- Attend a License Preparation Course
- Complete clinical modules in the Learning Management System
- Work with your manager to identify areas in need of improvement

Lesson #1: Success is a Journey - YOU Lead the Way

Notes

Welcome to the exciting and challenging world of hearing care. I'm betting that most of you are very excited about getting started in your newly-chosen career. The word "career" cannot be over-emphasized. The most successful and well respected people in the hearing care industry are those that look at the big picture; taking care of their hearing impaired patients over the long haul. One of the first things you may notice are the words client, patient and customer are used interchangeably. Don't be confused by this. Your primary focus must always be to care for and serve the person sitting in front of you.

Hearing care is a unique profession, and one you should be proud to be in. Hearing aids are an incredible combination of art and science. The dispensing of them combines elements of physics, biology, psychology, electronics, medical and retail selling. Most importantly, it is about helping others to communicate more effectively. Throughout your career journey, your ultimate focus needs to be on helping others hear well. It is simple: If you want to excel in your new career as a hearing care professional, you must have a clear focus of consistently meeting the needs of every patient and work every day to learn something new about your profession. You have to make a commitment to learn something new every day.

The more you give, the more you will receive in return. As you move ahead in your new career, learning as much as you can each day, keep that profound thought in the back of your mind. The more time and energy you put into learning all facets of hearing care, the more productive you will be. The more time and energy you devote to meeting the needs of every patient, the more successful you will be. One thing that separates the best in our profession is this: The most successful hearing care professionals have a single focus of giving all they can to meet the needs of the patient.

It has been said that hearing loss is a silent disability. This is indeed true. There is a powerful relationship between hearing loss and many negative personality traits such as depression, anxiety and stress. The social impact of hearing loss is even more dramatic.

The good news is that hearing aids are effective. When professionally fit by a well-trained dispenser, hearing aids can profoundly improve all aspects of a persons' life, literally changing their quality of life. As you move forward in your career, you will see this turnaround firsthand. It is one of the many rewards of being a hearing care professional. We hope that this basic training course will be the beginning of a long career journey. Like any journey, you will have ups and downs. Right now, the most important thing you can do is to study and prepare. It is critical to your success, especially in the early stages of your career that you study every day.

Lesson #2: Manage Your Career like a Business

Notes

The first step of your career as a hearing aid dispenser will be the completion of this course. Because you have chosen hearing aid dispensing as your career, it is up to you to work hard and to improve each day. One of the things all good business people do is to keep records. We do the same thing and if you want to be successful, you should do the likewise. After reading each unit, you will be asked to take a test. This helps us to measure the effectiveness of our training program and it helps you to gauge your progress. Here's a secret: If you measure something, you will automatically improve it. You may not like to take tests but they really do help you to learn the material.

You must take the tests and quizzes.

Lesson #3: Your Personality is the Biggest Contributor to your Success

There is one fundamental characteristic all successful hearing care professionals possess: a people-¬oriented personality. This doesn't mean you have to be extroverted and it certainly doesn't mean you have to be the smartest person on your block. What it does mean is that if you put the interests and needs of others ahead of your own, you will be successful. As you embark on your journey, it might help to review this prerequisite personality checklist. How would you assess your own personality in relation to these traits?

Personality Checklist

Pleasant: Generally speaking are you the type of person who is friendly and accommodating to others.

Curiosity: In order to most effectively meet the needs of your patients, you need to have a genuine curiosity about their needs. Every patient has unique needs and if you are curious about learning their individual needs, you are likely to be successful.

Courage: Helping others requires you to sometimes ask questions that are difficult for the patient to answer. Hearing aid dispensing also requires you to have the courage to ask tough questions and ask for the sale. **Tenacity:** There will be patients who say no to your recommendations. You must be tenacious in order to succeed. Tenacity means you don't dwell on these negative experiences. You simply move ahead and focus on helping the next person.

Patience: The majority of your patients will be over the age of 65 and some will be considerably older. It takes an immense amount of patience to address the needs of your elderly patients. To be an effective professional, an endless supply of patience is required.

Common Sense: Working with the public, running a business, learning on-the-job skills and the technical aspects of your career is demanding work. Common sense goes a long ways when it comes to doing what's right for the patient and your business. Large doses of common sense go a long way when you're making decisions that have an impact on your patients, your business and your career.

"Coachability": The defining characteristic, separating the great salesperson from the good one, is your ability to be open-minded and to accept feedback from your teachers, managers and others. All of us have skills that need to be improved upon. If you are self-aware and open to suggestions, you can improve any area of your career.

Lesson # 4: Technical Competence is Essential

Skills Checklist

The following skills are essential to your long-term career success. In order to master them, you must study. This requires a commitment. You will need to put in at least 2 hours of reading and studying for each section of this course. With the proper amount of time and motivation, you will acquire the skills listed below. The Amplifon Professional Development Department is there to help you along the way; however, YOU are responsible for learning these skill sets.

- Basic Customer Service Skills
- Introduction to the Psychology of the Hearing Impaired
- Introduction to Acoustics (Physics of Sound)
- Basic Anatomy and Physiology of the Ear
- Introduction to the Audiogram/Basic Audiogram Interpretation
- Introduction to Hearing Disorders
- Basic Hearing Aid Electronics and Troubleshooting
- The Sales Process: Meeting Your Patients Needs

Notes

Lesson #5: You Must Put in the Necessary Study Time Learning the Core Skills

It helps to break things up into a step-by-step process when you first start on the journey of becoming a hearing aid dispenser. The first step is to learn the skills listed on page 6.. Along with this class you can expect to put in at least 60 hours of studying and reading.

There are certain aspects of hearing aid dispensing that make hands-on learning indispensable. However, before tackling these topics, it is critical that you work through the sections listed above as independently as possible. In each section you will have the opportunity to connect with field trainers, via the internet, and others who are taking these classes with you. Take advantage of this virtual classroom by asking questions and learning from others.

Once you have mastered this material through study and hands-on application, you can move on to the next step of preparing to get your state hearing aid dispensing license. After you have demonstrated your knowledge of the Basic Training material, you will be eligible to take the License Prep class.

Once you have successfully completed this workbook you will move to the next step which is the Basic Training Technical Laboratory. You will be given the opportunity to work in a small group with other people taking this class. The content of the Basic Training Lab is as follows:

1. Compression
2. Otoscopy
3. Audiometry
4. Ear Mold Impressions
5. Hearing Aid Programming and Troubleshooting
6. The Sales Process

With diligent study and preparation, you will be up to the challenges of your new career as a hearing aid dispenser. With the challenges come many great rewards. Helping others is a great way to make a living.

Notes

Your First Assignments

1. In 200 words or less, write a brief biography. Include your career goals and your reason for choosing hearing aid dispensing as a profession.

2. What do you think are your professional strengths and weaknesses? How do you plan to improve your weaknesses? Please be specific.

3. List the 5 lessons outlined in Unit 1 and your impressions of them.

Unit 2: Psychology of the Hearing Impaired

Objectives

Notes

Overview

Nearly every patient seeking your services exhibits some or all of the behaviors outlined in this section. In order to provide the best care and service to your patients, it is critical for you to understand—from the patient's perspective—why they are behaving in such a way. This section will help you do this.

Goals

Upon completion of this module, you will be able to:

- Be more familiar with some of the behaviors associated with hearing loss.
- Have a better understanding of why hearing impaired people have many of these behavioral traits.
- Develop insight as to how you should interact with patients in a positive and productive manner.

Follow Up

- *A Bridge to Healing: Hearing Loss and Hearing Aids*. Edited by Richard Carmen. Auricle Ink
- Publishers, Sedona, AZ, 1998. Taylor and Hansen. *To Change the Industry We Must Change, Part I*, The Hearing Journal, December 2002.
- Taylor and Hansen. *To Change the Industry We Must Change, Part II*. The Hearing Journal, January 2003.

Introduction

Notes

To paraphrase a classic Country & Western ballad, "What drives you insane about me is the very thing keeping me from losing my mind." This phrase, taken from the perspective of a hearing aid dispenser, means that our hearing aid patients sometimes have behaviors that are hard for us to understand. However, these challenging behaviors can be seen as a "normal" response to the stresses of hearing loss. The good news is you don't have to appreciate Country & Western music to understand the personality traits of the typical hearing impaired adult.

Over 35 million Americans, adults as well as children, suffer some degree of hearing loss. The most common type is sensorineural hearing loss. The encouraging fact is that people with this type of hearing loss can be helped with hearing aids.

For the person who experiences hearing difficulties, hearing loss is usually just the beginning of a series of social obstacles. Hearing loss is usually a communication disorder of gradual onset, progressing slowly over the course of many years. Typically, the hearing loss comes on so slowly that the individual may not even be aware of the change as it occurs. Even after they are finally aware of the change in their hearing, it takes the average person 7 to 10 years to come to an office for a hearing test. Unlike other health problems, hearing loss is not physically noticeable and is not painful. It is usually a spouse, friend or other loved one who notices the hearing loss first. All of us know someone who has trouble with hearing conversations. Many times we notice the problem before they even admit they have a problem. As you will read in this section, this is completely normal behavior.

Hearing loss has been linked to changes in behaviors. After you have completed this section you should have more insight into why patients with hearing loss oftentimes are withdrawn, suspicious of others, indecisive and even hostile. There can also be a loss of earning power on the job due to untreated hearing loss.

Notes

Developing a relationship with your hearing impaired patient will ultimately increase your chance to successfully help this person do something about his communication deficit. In addition, his ability to adapt to hearing aids may be enhanced as a result of your ability to understand his hearing loss and the associated behavioral traits. As a hearing health care professional, you have an opportunity to have a profound and lasting influence on his life that goes beyond simply fitting him with hearing aids.

Even with normal hearing, it is not easy to communicate and function comfortably in many of today's noisy listening environments. Take a moment and think about the last time you were in a popular, crowded restaurant on a Saturday night: It takes a lot of concentration to follow the conversation of the person sitting next to you. It is even more difficult—sometimes impossible—to hear in these important situations when you have a hearing loss. It's no wonder that people with hearing loss are withdrawn, embarrassed or agitated about this "hidden handicap".

Hearing Impairment and Behavior: The Emotions of Hearing Loss

Our ego is attached to our health. All of us would like to think of ourselves as leading healthy and productive lifestyles. When a person is first aware they are missing out on conversation it is normal behavior to deny the existence of a problem. The hearing loss goes against our perception of reality. It is not part of our self-image to have a deficit such as this. Think about how you felt the last time you were at a noisy social gathering and someone told a funny story, and you missed the punch line. Did you laugh like everyone else? Or, did you ask the person to repeat the part you missed? Most people just laugh and go along with the group—pretending to have heard. Most people would say this is an uncomfortable situation. Now, think about the hearing impaired person having to go through this many times a day.

Hands-on Exercise

Find out what it is like to have a hearing impairment. For an entire day wear earplugs. Go about your daily routine and make a record of your reactions and emotions in your surrounding with the temporary hearing loss. The next time you encounter a hearing impaired person acting in a negative way, think about what it is like to live with a hearing loss every minute of the day. In a hundred words or less write about your experiences with a temporary hearing loss here:

Notes

Notes

Let's examine six emotions and behavioral characteristics of the hearing impaired person and how you— the professional—may assist the individual in overcoming these negative self images. Not every hearing impaired person exhibits all of these traits, but you are likely to observe several of them in your office on a daily basis.

Denial

When something bad happens to us, it is normal behavior to deny the problem exists. Denial has an important function: It allows us to recover from the shock of a painful or negative experience. For the person experiencing a hearing loss for the first time it is easy to simply ignore the problem. When you are working with individuals with hearing loss, you will experience denial on a daily basis. Patients will say, for example, "I would hear fine if people would just stop mumbling". Fortunately, most patients do not strongly deny their hearing loss. They will acknowledge the existence of the problem, but the other behaviors stemming from the initial denial can cause a great deal of emotional pain and stress. Simply stated, ignoring the hearing loss often times leads to some of the behaviors we will discuss next.

Withdrawal and Avoidance

The easiest way to deal with the psychological pain of hearing loss is to avoid vulnerable situations. The hearing impaired person begins to withdraw from society, even from situations which previously may have been the focal point of his social interaction. For example, an individual who has been an active, participating lodge member may find it increasingly difficult to communicate at meetings. People with hearing loss gradually decrease how often they attend meetings and eventually stop attending at all. .

As you begin your Case History, note that there is generally a direct correlation between the length of time the individual has withdrawn from social situations and the length of time the person has noticed a hearing loss. Unfortunately, *the hearing impaired individual does not always relate this withdrawal to a hearing problem*, but to other external influences. This individual may even develop a false sense of wanting to be alone. It is always easier to create an excuse or blame someone else for the problem.

As the hearing impaired person becomes more and more withdrawn from the world around him he begins to avoid situations he once loved to attend. Unable to hear and being isolated is a lonely way to live. The longer people with hearing loss avoid seeking professional help, the more they become entrenched in a hearing loss that rules their life. Once people who have lived with hearing loss for many years finally make the choice to seek help from you, it is not uncommon for them to show feelings of anger and hostility.

Hostility and Anger

It is usual for family members and or friends to notice behavioral changes in the hearing impaired individual. Family members may comment that this person is "grouchy", or has become "difficult to live with." Hostility develops. Again, this is normal behavior. The hearing impaired person may become less tolerant of others as a result of hearing these kinds of comments over and over again.

Imagine you have a hearing problem. Every time you have to ask someone to repeat themselves, it's a reminder to you that you have a hearing loss. Eventually, you become resentful and angry at others over your own need to have things repeated. You already know you have a hearing problem and you don't want to be constantly reminded of it. This is an emotionally painful experience. To compound the problem, your family and friends feel you are being stubborn and are resistance to help. This initiates a vicious cycle of events in which your family and friends become angry at you because they think you are being stubborn, and you are angry with them because they keep reminding you about your "problem". This cycle of anger and hostility wears on relationships.

Selfishness and Suspicion

Living with someone who refuses to get the necessary help for his hearing impairment is a challenge. Because the development of hearing loss is typically a slow process, summoning the courage to make an appointment to get a hearing test is often a terrifying experience. The hearing impaired person's unwillingness to help themselves may be seen as a selfish act. Many persons with hearing loss come to expect that all their daily interactions with others will be arranged around their hearing loss.

Unable to trust his own ability to hear and understand what is being said, the hearing impaired person may become suspicious of others. The individual who is suspicious of others may believe that people are talking about them. Because of his inability to hear conversations clearly, this person finds it harder to depend upon information as accurate. The suspicious hearing-impaired person who has lived with slowly degenerating hearing is slow to develop trust.

Notes

Practical Point

A critical part of the hearing aid sales process is the development of trust. Patients are suspicious by nature. When the suspicious behaviors of a typical hearing-impaired person are added to this, it underscores the important role you have in creating a trusting environment for your patients. As a professional your first priority with patients is to build a relationship founded on trust.

Exercise

In a hundred words or less write about your first experiences with patients you met who were extremely suspicious of you. How did it make you feel? What did you do to put the patient at ease?

Notes

Indecision

Lack of confidence can result in the inability to accurately evaluate and make decisions about many things. The hearing impaired person often turns decision making over to someone else, often to a trusted family member or significant other. It is not uncommon to see a spouse, son, or daughter as the influential person when the hearing impaired person has to make a buying decision.

Practical Point

Because consumers generally involve a spouse or other family member in the buying decision. It is important to have a companion present during the hearing aid consultation appointment. This is especially important for the hearing impaired patient who is exhibiting indecision.

Sensitivity to Intent and “The Stall”

. All of us communicate and develop relationships through visual cues—this is even more of the case for those with hearing loss. You will notice that your patients will watch you with a great deal of concentration. Their powers of observation are more than lip reading. The hearing impaired person has learned to read facial expressions, body language and content of speech. All of us do this more than we realize, but the hearing impaired person uses a greater degree than the rest of us on modes of communication other than words. It is important to use all modes of communication

Probably the single most common response that hearing care professionals hear from patients, once the presence of hearing loss is explained, is this emphatic question: “Is my hearing loss bad enough that I need hearing aids?” This question shows that the patient recognizes the presence of hearing loss but is trying to find an excuse not to do anything about it. We call this “the stall.” One of your greatest professional challenges will be to recognize when someone is making an excuse for not getting help for the hearing loss, and then moving that person to acceptance of the problem. No amount of convincing, cajoling, or explaining will make the hearing impaired patient solve the problem. Allowing a patient to accept her hearing loss and take the necessary steps to fix the problem is a skill that takes years to develop. However, now is the time for you to start developing these critical counseling skills.

Notes

Taking Your Skills to the Next Level

Notes

Your Professional Role and Responsibility

Because hearing loss manifests itself in so many negative behaviors and emotions it is absolutely critical to your success as a professional to try to gain a better understanding of them. Understanding the emotional consequences of hearing loss requires a lifetime of study and experience. It is beyond the scope of this Basic Training course to delve into detail, nevertheless; let your journey begin now. Here are a few events that occur every day in hearing aid dispensing offices around the country. As you review these events reflect on some experiences you have had with difficult patients. After reviewing these points, ask yourself: “Is there anything I would have said or done differently?”

- It has required a great expenditure of will for the hearing impaired person to come to your office. Because so many people with hearing loss experience feelings of denial, it is a profound step for them to even be in your office for a hearing test. Realize that it truly is a big deal for this person to acknowledge that they might have a problem. Although these patients might react to you in a hostile or suspicious manner, it is your professional duty to first acknowledge the courage it took this person to arrive at your doorstep.

- A patient must first accept ownership of the visit before disclosing his communication difficulties. You have probably already encountered patients who are accompanied by a spouse at the initial consultation. Many times these patients will say that their spouse made them come for the hearing test. This patient does not own the visit. Ownership of the visit refers to the fact that the patient acknowledges he has a problem and is willing to talk about it. Until ownership of the visit occurs the helping professional is prevented from assessing the impact a potential hearing loss has on this person. As a professional you must ask open-ended questions that allow the patient to take ownership of the visit.

- A patient will not accept your recommendation to buy hearing aids until they feel they are truly understood. This is the reason that understanding the emotional consequences of hearing loss is so important. After a patient has taken ownership of the office visit, the next step is to connect with this patient on an emotional level. This requires two things on your part:

1. **Courage** - You must be courageous enough to ask the patient thought-provoking, open-ended questions.

2. **Curiosity** - You must be curious enough about the patient’s life experiences to listen intently to their answers. Once a patient feels emotionally connected to you, they are more likely to accept your recommendation for better hearing.

Notes

Hearing aids are effective in remediating hearing loss, but some of the emotional consequences of hearing loss will likely remain. Even after you have successfully fit someone with hearing aids, they still may struggle from time to time with the emotional consequences of the hearing loss. One of your duties as a professional is to establish long-lasting relationships with your patients. No hearing aid, no matter how sophisticated, will completely solve all the difficulties associated with communicating in various environments. You must be there when called upon by your patients to offer emotional support and guidance.

Hands-on Exercise

Take the first 30 minutes of your next work day and review the case history and counseling forms in your office. Ask your manager or colleague to help you find them. There are a series of open-ended questions on this form. One of the most important parts of every consultation you do is to ask these questions to your patient. Ask the question and patiently wait for a response. Always ask these questions with great sincerity. Practice by asking these questions to a colleague or friend.

Check Your Understanding

1. A patient upon hearing for the first time that he has a significant hearing loss says, “Everyone mumbles, I’ll come back in a year to have my hearing checked” is exhibiting signs of:

   a. Hostility

   b. Denial

   c. Indecision

   d. Pain

2. The typical hearing loss occurs suddenly. When this occurs most patients schedule an appointment immediately to get their hearing checked

   a. True

   b. False

3. One way to help the patient overcome feelings of isolation and loneliness would be to:

   a. Immediately do a hearing test

   b. Ask a series of open-ended questions and listen to their answers intently

   c. Take an ear mold impression

   d. Blame the patient for waiting so long to see you for help

4. You notice that a patient is staring at your lips as you take the initial case history. This is an example of:

   a. Indecision

   b. Denial

   c. Withdrawal

   d. Sensitivity to Intent

5. Which of the following is a reason a patient with a hearing loss will NOT purchase hearing aids? Select all that apply

    a. Denial of a hearing loss

    b. The significant other is not present for the appointment

    c. The patient does not trust you

    d. The patient knows he has a hearing loss

6. How long does the typical hearing impaired person wait before they visit your office for a hearing test?

    a. 7 to 10 months

    b. 12 years

    c. 7-10 years

    d. More than a decade

7. A common initial response from a patient who needs hearing aids might be?

    a. “Sign me up. I can’t wait to get started.”

    b. “Let’s get to work fixing my problem.”

    c. “Is my problem really that bad?”

    d. All of the above

8. Please list one open-ended question you have viewed on your office case history form.

Unit 3: A (Painless) Introduction to Acoustics (Physics of Sound)

Objectives

Notes

Overview

The way in which humans hear is a complex subject involving the fields of physiology, psychology and acoustics. In this section, we will focus on the acoustics (the branch of physics pertaining to sound) of hearing. In order to perform a hearing evaluation, to fit and adjust hearing instruments, and become a licensed and practicing professional in your state you will need to acquire a basic understanding of the acoustical properties of sound and become familiar with its terms and concepts.

Goals

Upon completion of this module, you will be able to:

- Explain the relationship between the behavior of sound waves and their relative importance to hearing testing and the fitting of hearing instruments and amplification

- Describe basic concepts of acoustics including reflection, diffraction and refraction

- Define and discuss the following terms: sine wave, compression & rarefaction, pitch & frequency, intensity & loudness, deciBel, pure & complex tones, fundamental frequency, harmonics and noise

Follow Up

- ***Understanding Physics.*** By Isaac Asimov. Chapters 11 & 12. pp. 148-180. Mentor Books: New York, 1961.

- ***Acoustics for Audiologists.*** By Edgar Villchur. Chapters 13. pp. 134. Singular Press: San Diego, 2000.

- ***Programmed Instruction on the Decibel in Clinical Audiology.*** By Charles Berlin. 1970.

- ***The Physics Classroom.*** Sound Waves and Music. By Tom Henderson. Lessons 1,2,3 & 4. ***http://www.glenbrook.k12.il.us/gbssci/phys/Class/sound/u11l1a.html***

Section 1 – The Sound Wave

Notes

Sound is part of our everyday sensory experience. The basis for understanding sound and hearing is the physics of waves. Sound is a wave that is created by vibrating objects and propagated through a medium from one location to another. After completing Section I, you will have a better understanding of some highly technical concepts.(Yes, physics is technical and relies heavily on mathematical calculation, but when you take most of the math out of the equation, physics can be fun as well!)

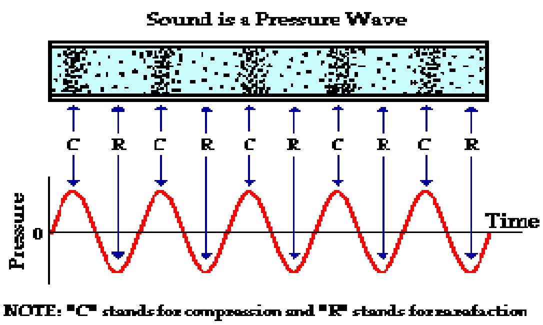
To begin, let us consider the question, “If a tree falls in the forest, and there is no one there to hear it, does it make a sound?” This question is not just a rhetorical one but also a query regarding of the nature of acoustics. Acoustics is the branch of physics pertaining to sound. As you will soon discover, the answer to the “falling tree” question can be found in the four elements required for sound to occur.

A wave may be described as a force or disturbance that travels through a medium transporting energy from one place to another. The medium is simply the material through which the disturbance moves and can be thought of as a series of interacting particles. For example, when our “tree in the forest” falls, a disturbance is typically created. Neighboring trees are shaken or moved. The ground quakes and the surrounding air vibrates as each displaced particle acts to displace an adjacent particle; subsequently this disturbance will travel through the entire forest. As the disturbance moves from tree to tree, along the ground and through the air, the energy that was originally introduced by the falling tree is transported along each medium from one location to another. The bigger the tree and the harder it falls, the larger the disturbance. The larger the disturbance, the greater the impact it will have. Let us assume, however, that our falling tree has no audience and revisit the question: Is it actually sound since no human ear is available to hear it? By definition, sound is only considered to be sound when each of these four very important elements is in place:

- An Energy Source (falling tree, electrical current, air from the lungs, a striking hammer, or a violin bow)
- A Vibrating Body (ground, diaphragm of a speaker, vocal cords, a tuning fork, or a violin string)
- A Medium (solid, liquid or gas) and
- A Receiver (the human ear)

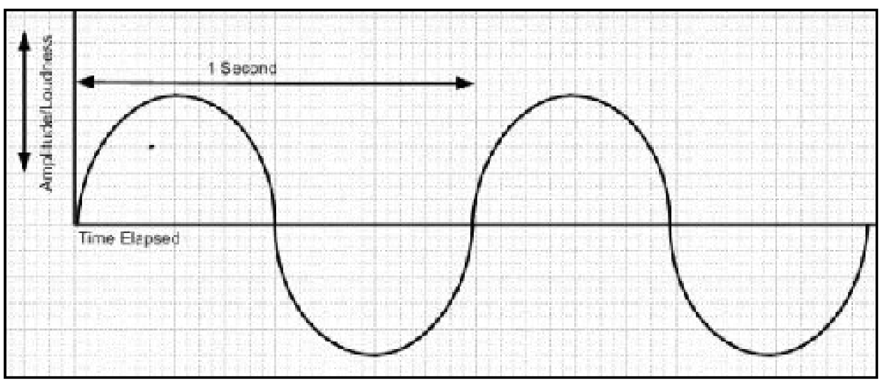
In our theoretical question there is no element #4, and if there is no element #4 or no human ear to receive the sound, then it was never heard, and logically if sound is not heard then it cannot be considered sound. Now you know!

Compression and Rarefaction in Those Traveling Waves



Sound pressure waves travel through a medium displacing particles, pushing and bumping and moving each other, coming together in tight groups and then dispersing in a series of what is known as **compressions** and **rarefactions**. When the molecules are close together you have compression and when they spread apart it is called rarefaction.

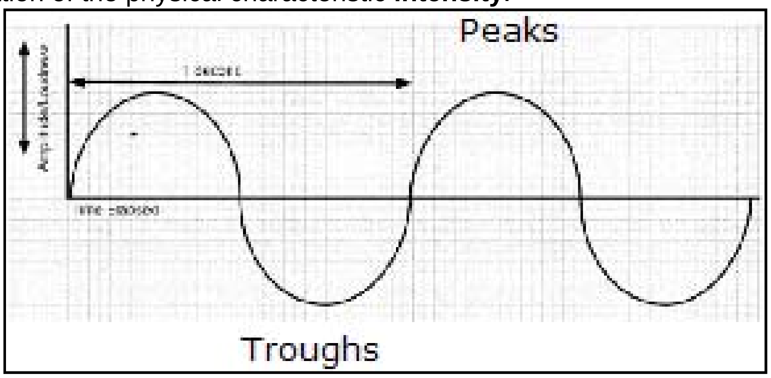
One completed cycle of a single compression and rarefaction occurring over one second in time can be drawn as a sine wave and expressed as one cycle per second (cps) or…1Hz.



Notes

Intensity & Loudness

The **intensity** of any sound is related to the largest pressure change via the displacement of particles, or the **amplitude** of the sound wave. The greater the amplitude or vertical distance between the peaks (maximum compressions) and troughs (maximum rarefactions) of the sound wave, the greater the **intensity**. The larger sine wave represents what would be the louder of two tones. It has larger **amplitude**, and therefore, greater **intensity**. Intensity is a physical characteristic of the sound and can be easily measured. **Loudness** is the psychological interpretation of the physical characteristic **intensity**.



Humans are equipped with very sensitive ears capable of detecting sound waves of extremely low intensity. The faintest sound that the typical human ear can detect has power intensity equal to 10-16 watts/cm2 and pressure intensity equal to .0002 dynes/cm2. A sound with an intensity of 10-16 watts/cm2 corresponds to a sound that will displace particles of air by a mere one billionth of a centimeter. The human ear is able to detect such a sound! This faintest sound that the human ear can detect is known as the **threshold of hearing**. The most intense sound that the ear can safely detect without suffering any physical damage is more than one billion times more intense than the threshold of hearing.

Since the range of intensities that the human ear can detect is so large, the scale that is frequently used by physicists to measure intensity is a scale based on multiples of 10. This type of scale is sometimes referred to as a **logarithmic scale**. The scale for measuring intensity is the deciBel scale. For the purpose of testing human hearing the threshold of hearing is assigned a sound level of 0 deciBels (*abbreviated **0 dB***); this sound corresponds to an intensity of 10-16 watts/cm2 and a pressure of .0002 dynes/cm2.

While the intensity of a sound is an objective quantity that can be measured with appropriate instrumentation, the **loudness** of a sound is a subjective response that varies given a number of factors. The same sound will not be perceived to have the same loudness to all individuals. One factor that affects the human ear's response to a sound is age. Obviously, hearing in seniors (think of your grandparents) is not what it used to be. The music at a rock concert would not be perceived to have the same quality of loudness to them as it would to you. Furthermore, two sounds with the same intensity but different frequencies will not be perceived to have the same loudness. Because of the human ear's tendency to amplify sounds having frequencies in the range from 1000 Hz to 5000 Hz, sounds with these intensities seem louder to the human ear.

Notes

Frequency & Pitch

The relationship between **frequency** and **pitch** is the same as the relationship between *intensity* and *loudness*. **Pitch** is the psychological interpretation or perception of **frequency**. Two people listening to the same mid-frequency (1000 Hz) sound may perceive and then describe its pitch differently. One person may perceive and describe it as a high pitched sound while the other may perceive and describe it as a midrange sound, being neither high pitched nor low pitched.

What Intensity & Loudness and Frequency & Pitch Will Mean to You

When you fit an individual with hearing loss with a hearing instrument you are not only providing amplification, you are creating a whole new listening environment for that person. In an ongoing effort to maintain the optimum listening environment for each patient you will have to make occasional adjustments to both the intensity and frequency of their hearing instruments. Being aware of subjective differences in each individual's ability to perceive and describe what and how they hear with their hearing instruments will ultimately contribute to your success as a hearing care professional. As you can see, there is a lot to learn!

Sound Measurement

As a hearing care professional you need to know about different measures of sound intensity using the deciBel *(abbreviated dB).* One measure, expressed as hearing level or hearing threshold level *(abbreviated dB HL or dB HTL, respectively)* is used primarily in reference to testing hearing levels using the audiometer. The other, expressed as sound pressure level *(abbreviated dB SPL),* is used in reference to the manufacturing and performance evaluation of hearing instruments, amplification, voice levels, and environmental sounds.

Hands-On Exercise

Most hearing aid manufacturer's software allows you to program the hearing aids in HL or SPL. In a product simulation mode, look at the graphs in both HL and SPL

Notes

Reflection and Absorption

Notes

Two key factors affect how sound eventually reaches the receiving ear: **Reflection** and **Absorption**. Reflection of sound waves off of surfaces can lead to one of two phenomena— echo or reverberation. **Reverberation** often occurs in a small room with height, width, and length dimensions of approximately seventeen meters or less. Perhaps you have observed reverberations when talking in an empty room or honking the horn while driving through a highway tunnel or underpass. These reverberations can mask other sounds, especially those having higher frequencies. In auditoriums and concert halls, reverberations occasionally occur and lead to the displeasing garbling of a sound. But reflection of sound waves in auditoriums and concert halls do not always lead to displeasing results, especially if the reflections are controlled by being purposefully built into the design. Smooth walls have a tendency to direct sound waves in a specific direction. Rough walls tend to diffuse sound, reflecting it in a variety of directions. For this reason, auditorium and concert hall designers prefer construction materials that are rough rather than smooth.

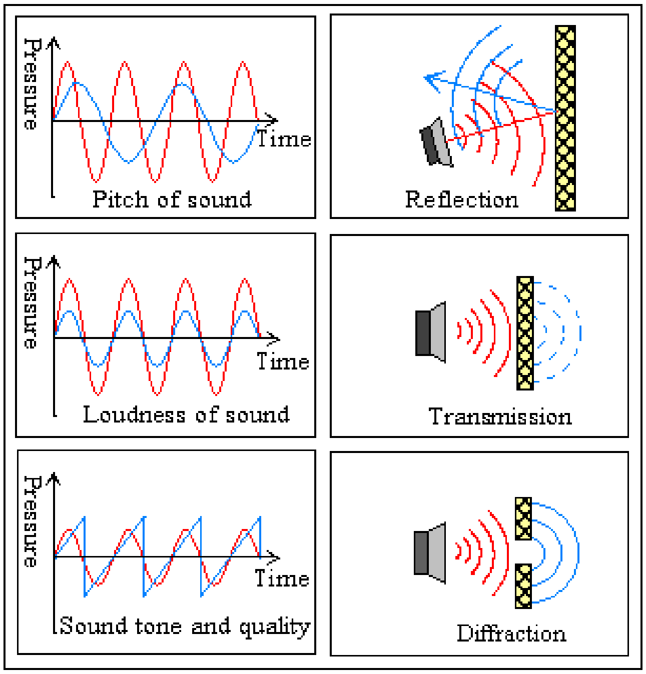
Reflection of sound waves also leads to **echoes**. Echoes are different from reverberations. Echoes are reflected sound, heard long enough after the original sound as to be perceived as a separate sound. Echoes typically are heard when the reflected sound reaches the ear more than a tenth of a second after the original sound was produced. An acoustically corrected room may, by design, eliminate these echoes and the loss of intelligibility they cause.

Absorption is the opposite of reflection. Certain materials, such as rubber, cork, and acoustic tiles, can absorb sound. Sound absorbing materials have high absorption coefficients. Soft, pliable items such as draperies, upholstered furniture, and carpeting help absorb sound and improve the listening environment for hearing impaired persons who experience more auditory distortion in the presence of reflection or reverberation than those of us with normal hearing. The absorption of sound is greater in warm rather than in cold and in moist conditions rather than dry. Knowing which materials absorb sound and which reflect it is very important in acoustical engineering projects such as the design of concert halls, which must be built to minimize unwanted effects.

Diffraction and Refraction

Notes

Like any wave, a sound wave doesn't just stop when it reaches the end of the medium or when it encounters an obstacle in its path. Rather, a sound wave will undergo certain behaviors when it encounters the end of the medium or an obstacle.

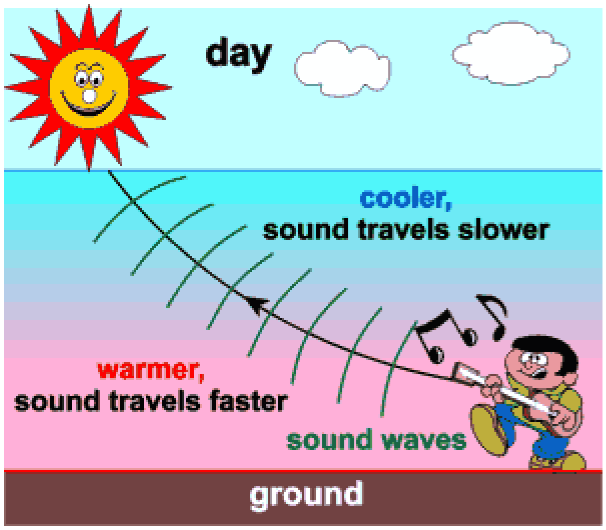


The **diffraction** of sound involves a directional about-face of waves as they pass through an opening or around a barrier in any medium. The **wavelength** of a wave is the distance that a disturbance travels along the medium in one complete wave cycle. The amount of diffraction (the sharpness of the change in direction) increases with increasing wavelength (low pitched sounds) and decreases with decreasing wavelength (high pitched sounds). In fact, when the wavelength of the waves is smaller than the obstacle or opening, no noticeable diffraction occurs.

Diffraction is a commonly-observed phenomenon. In our homes, sound literally bends around corners or slips through door openings, allowing us to hear other voices from other rooms. In nature, owls communicate across long distances because their low-pitched long-wavelength hoots are able to diffract around forest trees and carry farther than the high-pitched short-wavelength tweets of songbirds. *(Low-pitched sounds always carry further than high-pitched sounds.)*

Refraction involves a change in the direction of waves as they pass from one medium to another. Refraction is accompanied by a change in speed and wavelength (frequency) of the waves. Likewise, if there is a change in the medium (and its properties), the speed of the waves is changed.

Notes



Refraction of sound waves is most evident in situations in which the sound wave passes through a medium with gradually varying properties. For example, sound waves commonly refract when traveling over water. Since water has a moderating effect upon the temperature of air, the air directly above the water tends to be cooler than the air far above the water. Since sound waves travel more slowly in cooler air than they do in warmer air, that portion of the wave-front directly above the water is slowed down, while the portion of the wave-fronts far above the water speeds ahead. Subsequently, the direction of the wave changes, and refracts downwards towards the water**.**

A Matter of Frequency

As you have seen, nearly all objects, when hit, struck, strummed or otherwise disturbed, will vibrate. When you drop a pencil on the floor, it will begin to vibrate. When you pluck a guitar string, it will begin to vibrate. When you blow over the top of a pop bottle, the air inside will vibrate. When each of these objects vibrates, they tend to vibrate at a particular **frequency** or a **set of frequencies**. The frequency of a disturbance refers to how often the particles of the medium vibrate when a wave passes through the medium. The sensations of these frequencies are commonly referred to as the pitch of a sound. A high pitch sound corresponds to a high frequency and a low pitch sound corresponds to a low frequency.

The frequency, or frequencies, at which an object tends to vibrate when hit, struck, strummed or somehow disturbed is known as the **natural** or **resonant frequency** of the object. If the size or amplitude of the vibration is large enough and if natural frequency is within the range of human hearing (20 to 20,000 Hz), then the object will produce sound waves that can be picked up by the human ear. Any sound with a frequency below the audible range of human hearing (i.e., less than 20 Hz) is known as an **infrasound** and any sound with a frequency above the audible range of hearing (i.e., more than 20,000 Hz) is known as an **ultrasound.**

Notes

All objects have a natural frequency or set of frequencies at which they vibrate. Some objects tend to vibrate at a single frequency and they are often said to produce a **pure tone**. A flute tends to vibrate at as close to a single frequency as may be found in nature, producing a virtual pure tone.

Other vibrating objects produce more complex waves with a set of frequencies occurring at the same time in specific mathematical relationships to each other. The specific make-up and structure of these multiple frequencies is referred to as **timbre** or **spectrum**. Differing spectrum is the reason why different instruments that play in the same frequency range sound different from each other. The cello and bassoon both play in the bass range, but sound different due to their different spectra. The same is true for the difference in the vowels of speech. “Ah” sounds different from “ee” because of the different set of frequencies they comprise.

Some objects vibrate at a set of multiple frequencies that have no identifiable mathematical patterns between them. These objects are not musical and the sounds that they create are best described as *noise*. Noise is erratic, intermittent, or statistically random sound. When a pencil is dropped from a distance on a hard cement floor, it vibrates with a number of unrelated frequencies, producing a complex sound wave that is considered to be *noisy.*

Fundamental Frequency, First Harmonic & Timbre

Each natural frequency produced by an object or instrument has its own characteristic vibrational mode or standing wave pattern. These patterns only occur within the object or instrument at specific frequencies of vibration; these frequencies are known as harmonic frequencies, or harmonics. The lowest frequency produced by any vibrating body is known as the fundamental frequency. When a frequency is doubled or halved it is called an **octave**. The fundamental frequency is also called the **first harmonic (f1).**The frequency of the second harmonic **(f2)** is two times the frequency of the first harmonic. The frequency of the third harmonic **(f3)** is three times the frequency of the first harmonic.

For example, a vibrating body with a fundamental frequency of 400 Hz would have a 2nd harmonic at 800 Hz and a 3rd harmonic at 1200 Hz. A 4th harmonic would be found at 1600 Hz while a 5th would be at 2000 Hz etc. Of these, the harmonic at 800 Hz is one octave above the fundamental frequency, while the harmonic at 1600 Hz is 2 octaves above the fundamental frequency.

As the frequency of each harmonic increases the wavelength decreases. This is what is called an **inverse relationship**. For example, the wavelength of **(f2)** is one half (1/2) the wavelength of the first harmonic. The wavelength of **(f3)** is one third (1/3) the wavelength of the first harmonic.

Introducing the DeciBel

The basic unit for measuring sound pressure is the **microbar** or **dyne per square centimeter**. The microbar or dyne expresses what we refer to as effective sound pressure. One *dyne* is the force required to move one gram a distance of one centimeter in one second.

The softest sound that the best human ear can detect, in the best listening conditions, is an effective sound pressure of **.0002 dynes/cm2** . Conversely, the loudest sound the normal human ear can tolerate is at an effective sound pressure of about **1,000 dynes/cm^2**, just below the threshold of pain. At an effective sound pressure of **2000 dynes/cm^2** the human ear will feel pain and may suffer damage if the sound is sustained.

Using a dB SPL scale of measurement, the difference between the softest sound that the best human ear can hear and the loudest sound the normal ear can tolerate would be 5 million units. The difference between the softest effective sound pressure and pain would be 10 million units. Because we could not easily test human hearing with such large numerical differences in the range of sound, there needed to be an efficient way to express these values. Enter the **deciBel.**

The deciBel, developed by (and named after) Alexander Graham Bell literally means **1/10 of a Bel**. It is the term we use to describe intensity or loudness of sound. The deciBel is a logarithmic scale that reduces large numbers to the base of ten, giving them the number 10 plus an exponent. For the mathematically challenged, this logarithmic scale basically translates the unworkable range of **.0002 to 2000 dynes/cm^2** into a workable range of 0 to 140 dB HL. Table 1 on the next page shows the relationship between dynes/cm2 and the dB.

Notes

IL			SPL		
Power			Pressure		
Watt/cm²	Power Ratio	dB IL	dB SPL	Pressure Ratio	Dyne/cm²
10^{-2}	100,000,000,000,000:1	70	140	10,000,000:1	2000.
10^{-3}	10,000,000,000,000:1	65	130	3,160,000:1	632.
10^{-4}	1,000,000,000,000:1	60	120	1,000,000:1	200.
10^{-5}	100,000,000,000:1	55	110	316,000:1	63.2
10^{-6}	10,000,000,000:1	50	100	100,000:1	20.
10^{-7}	1,000,000,000:1	45	90	31,600:1	6.32
10^{-8}	100,000,000:1	40	80	10,000:1	2.0
10^{-9}	10,000,000:1	35	70	3,160:1	0.632
10^{-10}	1,000,000:1	30	60	1,000:1	0.2
10^{-11}	1000,000:1	25	50	316:1	0.0632
10^{-12}	10,000:1	20	40	100:1	0.02
10^{-13}	1,000:1	15	30	31.6:1	0.00632
10^{-14}	100:1	10	20	10:1	0.002
10^{-15}	10:1	5	10	3.16:1	0.000632
10^{-16}	Reference Level 1:1	0	0	Reference Level 1:1	0.0002

For example in the highlighted box above:

$I = P^2$ $\quad$ $I = 3.16 \text{ X } 3.16 = 10:1$

$\sqrt{I} = P$ $\quad$ $\sqrt{10} = 3.16:1$

dB IL x 2 = dB SPL $\quad$ 5 x 2 = 10 dB SPL

dB SPL ÷ 2 = dB IL $\quad$ 10 ÷ 2 = 5 dB IL

The decibel is not a whole number; rather it is a **ratio** between two pressures and has no fixed absolute value. A specific effective sound pressure is compared to .0002 dynes/cm2, the standard reference level for effective sound pressure, and expressed as **dB sound pressure level** *(abbreviated **dB SPL**).* Because the deciBel is a ratio and has no fixed absolute value, the term dB by itself, offers no information. It must be followed by a reference, i.e. dB HL or dB SPL, to identify what measurement scale you are using.

Notes

Confused? That's OK. Think temperature. If I tell you that it is 32˚ degrees outside and we are somewhere in the continental USA, then you will know that it is cold and put on your coat before you go outside. However, if we were on the European continent, it would be quite warm and you would be miserable (not to mention looking foolish) in that coat. This disparity is due to the fact that temperature measured in degrees (˚) also offers no information unless it is referenced to a specific scale of measurement, in this case, degrees ˚Fahrenheit or degrees ˚Celsius. For example, water freezes at 32˚F. It also freezes at 0˚C. Same temperature, different scale.

DeciBel Sound Pressure Level (dB SPL)

When talking about hearing instruments, voice levels, and environmental sounds, (which covers pretty much everything you encounter in everyday life) use **dB SPL**. With the exception of audiometric testing and audiograms you'll be working a lot with dB SPL, so there are a few things you need to know.

Because the deciBel is a calculated ratio, deciBels can only be added or subtracted exponentially. For example, if you could add one 50 dB SPL sound to another identical 50 dB SPL sound as you would 2+2=4 then the SPL level of the combined sounds would be 100 dB SPL. However, the actual combined level of 50 dB SPL + 50 dB SPL is **56 dB SPL.** ***When SPL values are doubled, there is an increase of 6 dB.*** Don't ask. Take my word for it.

This happens because when adding two sound pressures, you're actually combining two effective sound pressure (dyne/cm2) levels, not dB levels. When sound pressure levels increase from 134 to 140 dB SPL (or 6 dB) the dynes/cm2 double from 1000 to 2000 dynes/cm2.

In terms of loudness, you should know that **if the intensity of a sound increases by 10 dB, the subjective perception of loudness of that sound doubles**. For example, a 1000 Hz tone at 80 dB SPL is subjectively judged as 2 times louder than 70 dB SPL and 1/2 as loud as 90 dB SPL. Again, just take my word for it.

DeciBel Hearing Level (dB HL)

When performing a hearing test using an audiometer or referring to an audiogram, you will use **dB hearing level** (*abbreviated* ***dB HL***) to express the hearing threshold values.

We do this because the human ear is an incredibly interesting sensing organ. It is more sensitive to some sounds than others. Recall the human ear's tendency is to amplify sounds having frequencies in the range from 1000 Hz to 5000 Hz or those frequencies most important for detecting and understanding the speech sounds. Because of the ear's selectivity, many speech sounds do not have to be very intense (loud) for the ear to hear them.

Notes

On the other hand, those frequencies that are extremely low (below 1000 Hz) or high frequency sounds (above 5000 Hz) have to be much more intense for the human ear to detect them. If dB SPL were used as the reference level for audiometric testing where we are trying to find the lowest **threshold** (*softest sound that is heard and responded to 50% of the time*) at which our patients hear across all audiometric test frequencies, the extreme ends of the frequency range (250, 500, 6000 & 8000 Hz) would have to be presented at much higher intensity levels than those in the mid-frequency range. This would be problematic because different sound pressure levels at different frequencies would mean that, for the human ear, a dB SPL hearing level value obtained at a 1000 Hz would represent a different hearing level than that same dB SPL value obtained at 4000 Hz.

The American Standards Association (ASA) introduced the dB HL scale in 1951, creating the standard **Audiometric Zero** reference level. At each frequency, the different sound pressure intensity levels (dB SPL) required for the "best" human ear to hear the tone are "built" into the audiometer at 0 dB HL across all the test frequencies. The initial ASA standard was revised by the International Standards Association (ISO) in 1964, and revised again by the American National Standards Institute (ANSI) in 1969. Table 2 below illustrates dB SPL values for each of the test frequencies. While you may find these values important later when you begin to fit hearing instruments, it is not necessary to concern yourself with them while learning how to operate and perform hearing tests with the audiometer. For now you will simply record all hearing threshold values in dB HL.

Frequency	TDH 39/39D (dBSPL)
250	25.5
500	11.5
1000	7.0
2000	9.0
4000	9.5
8000	13.0

Sensation Level (dB SL)

Sensation level *(abbreviated **SL**)* may be used as a third scale of measurement or reference only after establishing a hearing threshold level in dB HL. By definition, a person's hearing threshold is the softest (lowest intensity) level at which they are able to hear a pure tone stimulus and respond 50% of the time. Sensation level refers to any audiometric procedure performed at a dB level above the patient threshold at any frequency also known as **supra-threshold** testing. **Sensation level** is the difference in deciBels between a threshold and a sound at the same frequency is *louder* than the threshold. Assume that patient A has a threshold at 1000 Hz of 10 dB HL. At that frequency, a sound that would be presented at a 20 dB sensation level, would be presented at 30 dB HL (10 dB HL threshold plus 20 dB sensation level).

Sensation level is often used during speech audiometry because starting intensity levels for some of the tests are determined relative to patient threshold levels.

The Pure-Tone Audiometer

Notes

The **audiometer** is, without a doubt, one of your most essential tools. It is the instrument you will use to measure a patient's ability to hear. The audiometer is a sound generator producing pure tones that you will present at various frequencies and intensity levels in order to establish **hearing threshold**.

In subsequent modules you will learn about the care and feeding of an audiometer, its operation, and how to perform accurate and complete hearing evaluations. There are specific protocols you will be expected to follow with each and every patient, (along with some hands-on lab work), and you should be able to learn how to administer a complete audiometric test battery in a relatively short period of time.

Audiometric Testing: Acoustics Applied

How Sound is Transmitted

Everyone with ability to hear perceives sound in two different ways. Sound can either be transmitted via sound waves in the air through the outer ear (ear canal to the eardrum), through the middle ear and to the inner ear (cochlea), or directly to the cochlea via the bones of the skull (for a more complete discussion of anatomy and how we hear, please review Unit 4: Anatomy and Physiology). When testing a patient's hearing using the air conductive pathway of sound from the outer to the inner ear we use headphones and perform **air conduction audiometry**. When we bypass the outer and middle ears and vibrate the bones of the head using an oscillator to stimulate the inner ear we are performing **bone conduction audiometry.**

Air Conduction Testing

When you place headphones over (or insert them in) a patient's ears, you are preparing to test their air conduction thresholds. The pure tone or speech stimulus that you introduce via the headphones or inserts travels through the outer ear and middle ears to the inner ear, then along the VIII nerve and finally to the auditory cortex of the brain where it is decoded. Sounds arriving at the auditory cortex traveling the entire auditory system are called **air conduction stimuli.** If our ears are open and functioning, all the sounds that we hear during normal day-to-day activities travel to us and into our brain via air conduction.

Bone Conduction Testing

When we place a bone conduction oscillator directly on the mastoid process of the temporal bone behind the ear, we are preparing to test the integrity of the inner ear or cochlea. A properly-placed oscillator vibrates the bones of the skull, stimulating sensory cells in the cochlea, then on to the auditory cortex via the VIII or auditory nerve. These vibrations directly move the structures of the inner ear and allow us to perceive sound in the exactly the same way we perceived the air conducted signal.

Notes

We test both of these auditory pathways so that we may compare the results. Comparing air and bone conduction thresholds helps us to determine the site of lesion of the hearing loss. Site of lesion testing tells us very important information about where the problem contributing to the hearing loss lies: in the outer, middle or inner ear or in combination. Physicians and the medical community are particularly interested in hearing loss resulting from problems in the outer or middle ear. Pathologies in these parts of the ear are usually treatable, either medically or surgically. As hearing care professionals, we are primarily interested in hearing loss resulting from damage to the inner ear or cochlea. Hearing loss originating in the cochlea is usually not treatable, either medically or surgically. Cochlear hearing loss does respond well to use of hearing aids, however.

It is important to understand that when any sound stimulus is presented to the head via bone conduction, both cochleas (given that both are functioning equally) will respond since the entire skull vibrates at essentially the same time. If the cochleas are not equally sensitive, **the better cochlea will respond to the lowest intensity sound, regardless of the placement of the bone oscillator**. So, when you place the bone oscillator behind the right ear, present a stimulus and obtain a threshold response, you may mark the results using the appropriate symbol for right bone conduction thresholds on the audiogram, but without further testing you cannot be sure exactly which cochlea heard the signal. You can only say that the best cochlea heard the signal and that a response was obtained with the bone conductor on the right side of the head.

Acoustic Resonators

Pick up an acoustic guitar and pluck one of the strings. You will notice that the sound fills the entire cavity behind the hole and resonates. Now, cover the hole behind the strings and strike the same note. Notice the difference. This time the sound just quickly dies off. It does not resonate. Any time sound fills an open cavity, like an ear canal, it will vibrate in a certain way. The way a sound resonates depends on factors like the composition of the medium it is traveling through and the barriers or walls it encounters along the way.

Within the human ear small differences like the thickness of the skin lining the ear canal or the sensitivity of the eardrum can greatly affect how sound resonates. It's also important to know that the human ear canal most fully resonates somewhere between 2000 and 3000 Hz for most persons, the average canal resonant frequency being 2700 Hz.

Hands-On Exercise

Set your audiometer for a solid Pure Tone Bone Conduction (BC) tone at 40 db at 1000hz. Place the oscillator to your forehead and close your eyes. See if you can tell where you hear the sound. Do you hear it at your forehead? Or is the sound going toward your right or left side of your head? You can also slide the bone oscillator up and down along your mastoid bone to see if there is an area that you hear the sound with more intensity.

Check Your Understanding

1. Which of the following is not one of the four essential elements of sound?
   a. Energy supply
   b. Medium
   c. Reflection
   d. Vibrating object
   e. Receiver

2. Of the following, which term is used to designate frequency?
   a. DeciBel
   b. Hz
   c. cm
   d. Bel
   e. Dynes

3. 200 Hz, 400 Hz 800 Hz are all examples of which of the following?
   a. 1^{st}, 2^{nd} and 3^{rd} harmonics
   b. Octaves
   c. Fundamental frequencies
   d. All of the above
   e. A and B

4. Pitch is to Frequency as:
   a. Intensity is to Pressure
   b. Amplitude is to Wavelength
   c. Loudness is to Intensity
   d. Frequency is to Intensity

5. Amplitude is to Loudness as:
   a. Wavelength is to Pitch
   b. Frequency is to Intensity
   c. Intensity is to Pressure
   d. Intensity is to Wavelength

6. We can hear voices from around corners in other rooms due to which of the following wave behaviors?
   a. Reverberation
   b. Diffraction
   c. Echo
   d. Absorption

7. Of the following mediums, sound would travel fastest through:
   a. Water
   b. Cool, dry air
   c. Warm, moist air
   d. Steel
   e. Wood

8. Rubber, cork, and acoustic tiles all affect sound in which of the following ways?
   a. They all reflect sound
   b. They all absorb sound
   c. They all diffract sound
   d. They all amplify sound

9. Match each of the decibel measurement scales to its reference.

a. dB HL	Patient's threshold
b. dB SPL	Audiometric Zero
c. dB SL	0002 dynes/cm^2

10. Match each of the decibel measurement scales to its primary application.

a. dB HL	Supra-threshold testing
b. dB SPL	Manufacturing Specs
c. dB SL	Audiometric Testing

11. The human ear by its design naturally amplifies which set of frequencies?
    a. 2000-8000 Hz
    b. 1000-5000 Hz
    c. 100-500 Hz
    d. 5000-6000 Hz

12. What do the different scales of temperature and loudness measurement have in common?
    a. Both scales are based on ratios rather than whole numbers
    b. Both scales require a reference in order to have meaning
    c. Both scales can be added and subtracted numerically
    d. All of the above
    e. A and B only

13. The **best** definition of a pure tone threshold is:
    a. The softest sound a customer can hear
    b. The softest sound a customer can respond to
    c. The softest sound a customer can hear and respond to 50% of the time
    d. None of the above

14. Air conduction sound is:
    a. Heard directly by the inner ear then to the brain
    b. Heard in all three parts of the ear then to the brain
    c. Presented through headphones
    d. Presented through an oscillator
    e. Both B & C
    f. Both A & D

15. A pure tone with 10 complete cycles of compression and rarefaction in one second would have a frequency of?
    a. 10 Hz
    b. 100 Hz
    c. 1 Hz
    d. 200 Hz

Bonus Question:

1. Would the frequency of the pure tone described in question #15 be audible to the human ear?
    a. Yes
    b. No

Unit 4: Anatomy and Physiology of the Ear: A General Overview

Objectives

Notes

Overview

The human ear is a marvel of elegant design, balanced structure, and extreme miniaturization allowing us to perceive and understand myriad sounds, the most important of which is speech. A basic understanding of the structures of the peripheral auditory system (anatomy) and how they work (physiology) is essential for the hearing care professional to understand the patient's hearing loss and to best recommend the most appropriate hearing instruments. This chapter will describe in some detail the structures of the peripheral auditory system and their functions.

Goals

Upon completion of this module, you will be able to:

- Recognize basic ear anatomy
- Describe how the outer, middle and inner ears work
- Explain how the three parts of the ear transmit sound to the brain
- Explain how the physiology of the ear affects hearing aid selection and fitting

Follow Up

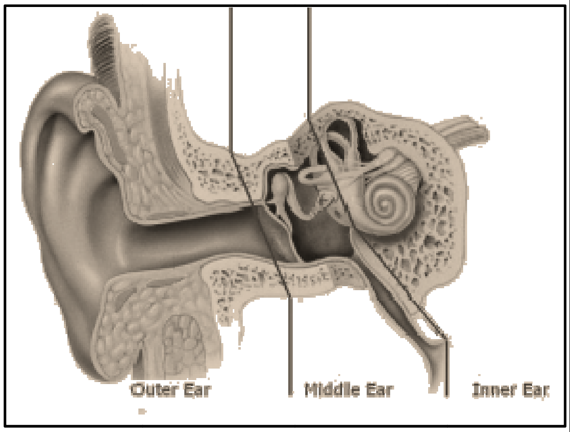
- Dallos, P. (1992). The Active Cochlea. *Journal of Neuroscience* , 12.
- Hallowell Davis. (1961). Anatomy and Physiology of the Ear. In *Hearing and Deafness* (p. Chapter 3). New York, NY: Holt.
- Venema, T. (1998). The Cochlea, Hair Cells and Compression. In *Compression for Clinicians* (p. Chapter 1). Sandiego, CA: Singular Press.

Section 1: The Flow of Communication

Notes

Some understanding of how the ear works is essential if one wants to understand the various types and degrees of hearing loss. The primary focus of this section is to introduce to you the most essential structures and functions of the three parts of the ear.

The outer, middle and inner ears comprise the peripheral auditory system. In any sensory system, the peripheral portion receives some signal from the outside world (in the case of hearing this would be the agitation of air molecules created by a vibrating body through a medium) and transform them into an electrical signal that will travel to the brain for interpretation and decoding. As hearing care professionals, we are mainly concerned with this part of the auditory system as this is where hearing aids will be fit and where they will have their principal effect.



Transduction is the process of changing energy from one form into another. In the peripheral auditory system, the main task is to change (transduce) acoustic energy (air molecules vibrating in the environment) into an electrical signal that can be transmitted through the nervous system to the brain. This cannot be accomplished in a single step. The components of the peripheral auditory system perform a series of transductions to complete this task.

Notes

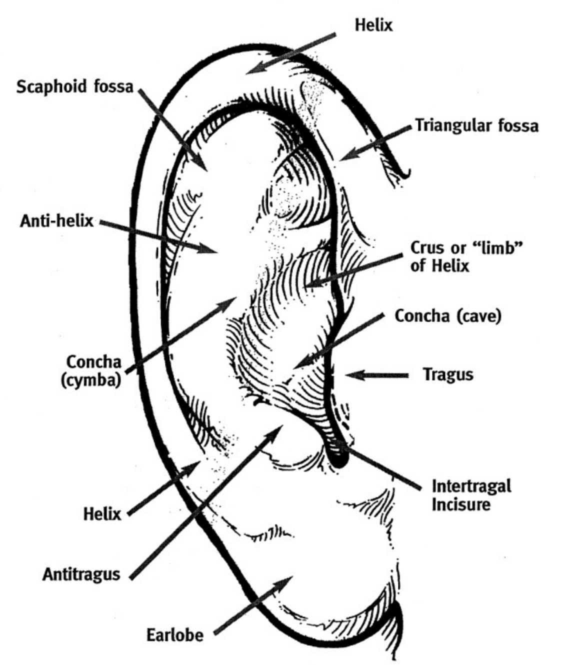
The Outer Ear

The outer ear changes acoustic energy into mechanical energy.

Pinna

The part of the outer ear that we see is called the **pinna**, or **auricle**. Besides being a place to hang earrings, the pinna is responsible for the gathering and localizing of sound. Due to the size and shape of the pinna, it acts as an acoustic resonator for sounds in the 2000 to 5000 Hz range. This means that the pinna provides a natural boost for sounds in this frequency range. Sounds in this range are some of the most important sounds for understanding human speech.

Diagram of the Ear



External Auditory Meatus (Ear Canal)

The external auditory meatus (*abbreviated **EAM***) extends from the bowl of the concha to the tympanic membrane *(abbreviated **TM**)* or eardrum.

The EAM is usually 1 - 1.5 inches long and approximately .25 inches in diameter. The outer one third of the EAM is **cartilaginous** (made of cartilage) and is comprised of relatively thick skin over cartilage. This portion contains hair cells, sebaceous (oil) glands and cerumen (ear wax) glands. The inner 2/3 of the EAM (past the second bend) is **osseous** (bony) and is comprised of relatively thin skin over bone. The osseous portion is more prone to irritation than the cartilaginous portion.

The ear canal is slightly "S"-shaped. There are two bends in the EAM, the first of which occurs at the **aperture** or entrance to the ear canal, and the second of which occurs at the **isthmus**, a slight narrowing of the ear canal occurring in the area where the cartilaginous and osseous portions meet.

Resonant Frequency

Resonant frequency is the frequency at which an object (or tube of air) naturally vibrates. It is also known as "natural frequency". The **resonant frequency** of the open ear canal occurs between 2500 and 3500 Hz, depending on its size and shape. This provides a natural boost of approximately 10 dB to sounds in that frequency range. When this is combined with the high frequency enhancement of the pinna, the natural amplification of high frequency sounds by the outer ear is 15-20 dB in the vicinity of 2700 Hz.

Notes

Vagus Nerve

A branch of the tenth cranial nerve – the vagus nerve – passes through the bottom of the EAM. When the vagus nerve is physically stimulated, as when an otoblock is placed for ear impressions, a cough reflex may result. This reflex is called Arnold's Reflex.

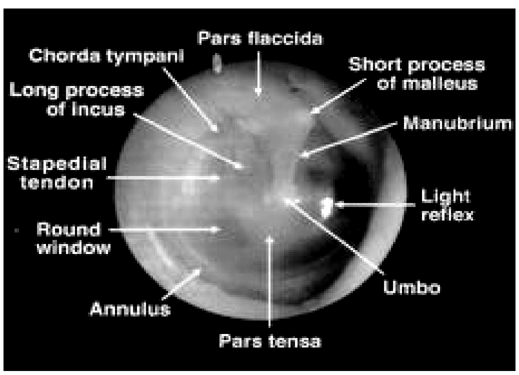
The Middle Ear

The middle ear boosts the mechanical energy that would otherwise be lost when vibrations in the air meet the fluid in the inner ear. This is known as the mass and stiffness effect.

Tympanic Membrane

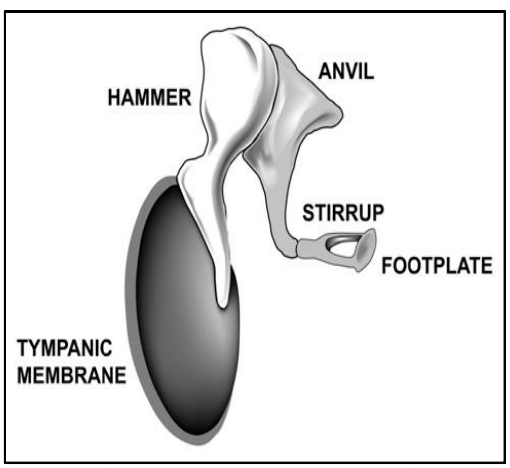
With enough light, when you gently pull back on the pinna and look into the ear canal with an otoscope, you can see the pearly white reflection of the tympanic membrane (TM)

The eardrum or **tympanic membrane** (*abbreviated **TM***) is the dividing line between the outer and middle ears. The greater part (the bottom 80%) of the eardrum is called the **pars tensa**. The small triangular shaped area at the top edge of the eardrum is called the **pars flaccida**. The **umbo** is the central most part of the TM. The **cone of light** or light reflex is a landmark of the normal tympanic membrane. It is produced by the reflection of the otoscope light from the concave eardrum. The TM is connected to the bony wall of the ear canal by a tough fibrous ring called the **annular ligament.**



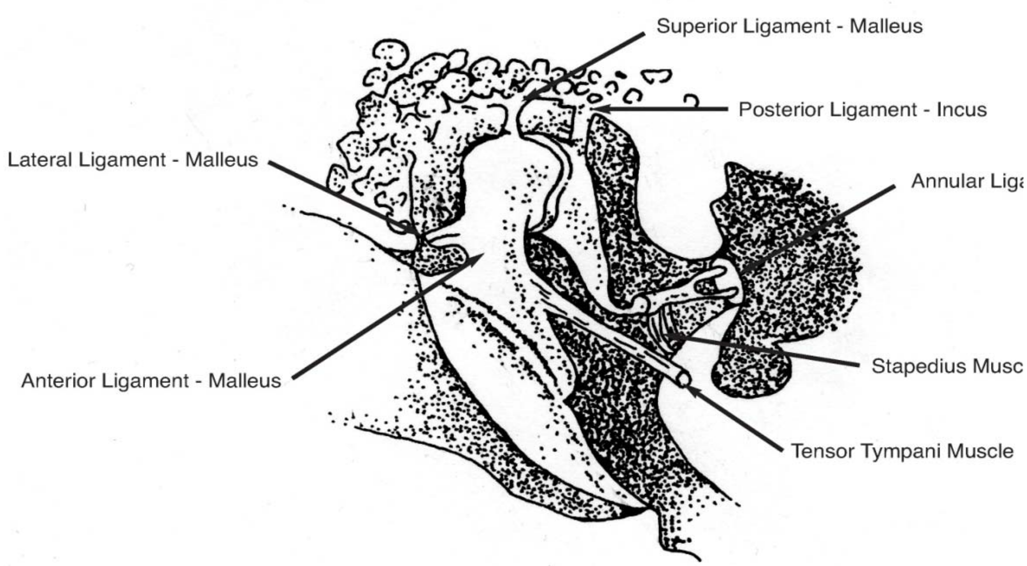
Ossicles

Within the middle ear and attached to the TM are the three smallest bones in the human body – the **ossicles**. These bones are sometimes called the hammer, anvil and stir-up, or more properly the **malleus**, **incus**, and **stapes**. The ossicles, or ossicular chain, have one essential function: They serve as an impedance matching device. Impedance is merely a technical term for resistance to flow. In the case of the ear, sound has to travel from the low impedance air pressure waves of sound to the high impedance hydraulic (fluid) system of the cochlea.



The ossicular chain with its lever and funnel action boosts sound as it travels between these two mediums, recovering energy that would have been lost. The stapes is seated in a flexible membrane called the **oval window,** which it vibrates in a piston-like manner, sending the vibrations of sound into the fluids of the cochlea. Without a fully functional ossicular chain a person would have about a 40 dB HL **conductive** hearing loss.

The Middle Ear



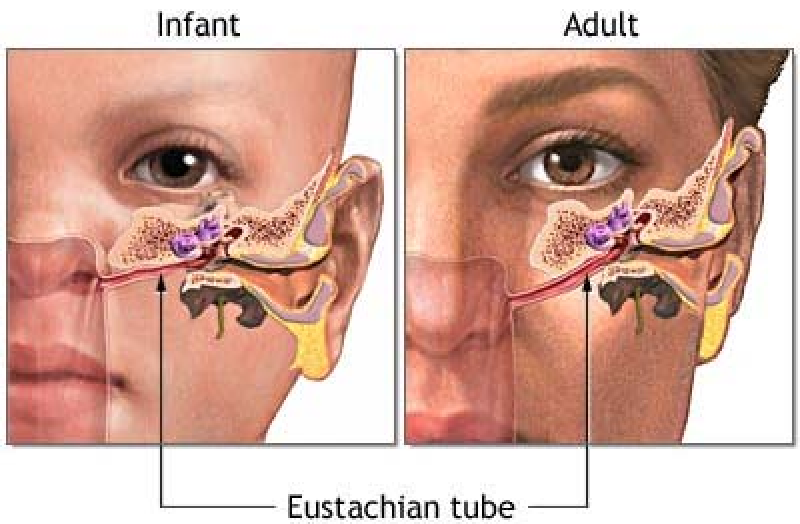
The **ossicular chain** is supported in the middle ear by ligaments and two muscles. The two muscles are called the **stapedius muscle** and the **tensor tympani muscle.** The stapedius muscle attaches to the stapes and draws the stapes in a posterior (rear) direction when it contracts. The tensor tympani attaches to the malleus. When the tensor tympani contracts it pulls forward, in opposition to the stapedius muscle, tightening the tympanic membrane.

The stiffening actions of these two muscles together create the **acoustic reflex**. The acoustic reflex is thought to provide some protection for the ear from loud sounds. However, because it takes 60 to 120 milliseconds to activate then tires or fades over time, the acoustic reflex does not protect the ear from either sudden impact sounds (e.g. gunfire, explosives etc) or sustained loud sounds (e.g. sirens, machinery noise, etc).

Notes

Eustachian Tube

The **Eustachian tube** is the middle ear's air-pressure-equalizing system. The middle ear is encased in bone and does not communicate with the outside atmosphere except through the Eustachian tube. The Eustachian tube is 35 to 40 mm long in adults, traveling first through 10 mm of bone and then through 25-30 mm of cartilage. In adults the Eustachian tube is tilted 30-40 degrees down from the horizontal. In children, it is closer to the horizontal plane, and is shorter and wider. The ability of the Eustachian tube to open periodically ventilates the middle ear space. Middle ear infections, the most common pathology among children, are the result of a Eustachian tube that is not working properly. Eustachian tube dysfunction, as it is commonly called, is a normal consequence of an immature Eustachian tube. Most children outgrow ear infections when their Eustachian tubes have completely developed to the correct size and angle. Though most common in childhood, Eustachian tube dysfunction is not uncommon in adult seniors (our primary clientele) with chronic allergy or sinus conditions.



Note: Notice the position of each.

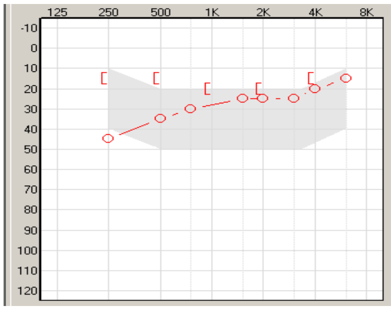
Notes

Stiffness/Mass Effect

Notes

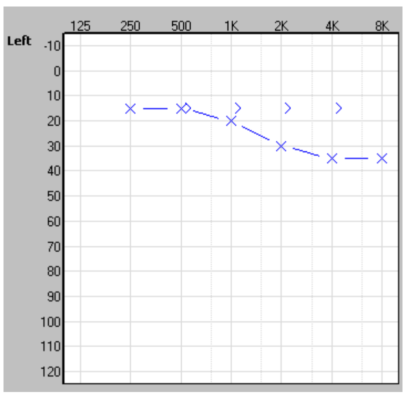
Depending upon the pathology responsible, a conductive hearing loss may have an audiometric configuration suggesting either the "stiffness effect" or the "mass effect".

Stiffness effect appears as an audiogram in which high frequencies are "passed" and the conductive component is mostly present in the low frequencies. The "stiffness effect" is usually present in Eustachian tube dysfunction, when the ear drum is likely to be more stiff than normal due to the negative pressure present in the middle ear.



Stiffness effect – right ear

In the **mass effect**, low frequencies are passed, and the conductive component is greater in the high frequencies. The most common "mass" responsible for a conductive component is likely to be fluid in the middle ear space. Therefore, otitis media will frequently produce an audiogram with the "mass effect".



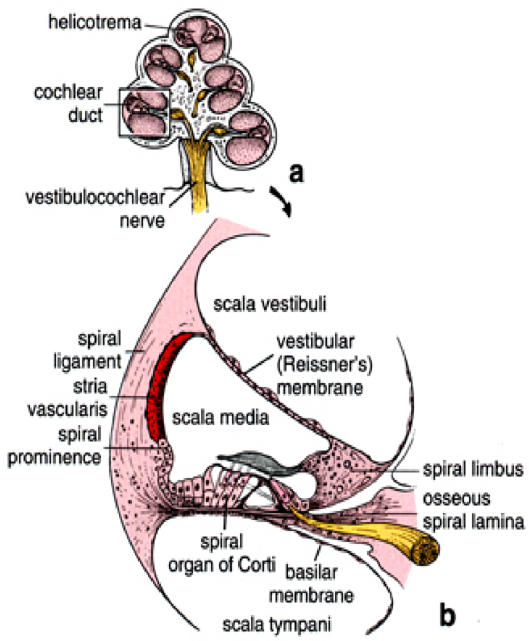
Mass effect – Left ear

The Inner Ear

The inner ear accomplishes the final transduction, changing hydraulic energy into an electrical signal that will travel down the Auditory (VIII) Cranial Nerve to the auditory processing centers of the brain.

Cochlea

The inner ear is a series of channels and chambers embedded deep within the temporal bone. It is also called the bony labyrinth. This term alone should tell you how convoluted the inner ear is. The inner ear, specifically the **cochlea**, is the part of the ear that you need to spend the most study-time on. Just about every patient who you will fit with hearing instruments will have a deficit that can be pinpointed to the inner ear. Having detailed knowledge regarding how the cochlea works will be critical to your professional success. The better you understand how normal cochlear function compares to that of a damaged cochlea, the more effective you will be in identify hearing loss and fitting hearing instruments. Now let's begin to tackle some of the details of cochlear physiology. You may find many of the details to be interesting.



The cochlea **transduces** (changes from one form to another) the **mechanical** vibrations of sound, via the tympanic membrane and the ossicular chain, into a sequence of electrical discharges that is the language of the auditory nervous system. This exquisitely engineered sense-organ completes the transduction in several stages. The mechanical vibrations of the stapes are delivered to the cochlear fluids, where they appear as a **hydro-mechanical** disturbance, or wave. This wave, traveling through the membranous structures of the inner ear, acts to displace the **cilia** projecting from two highly specialized types of sensory cells. These structures are called **inner and outer hair cells**.

These hair cells convert mechanical vibrations into **electro-mechanical** energy. **Outer hair cells** act as a sort of biological amplifier, boosting the electro-mechanical traveling wave. This process, in turn, produces synaptic transmission between the hair cells and the **neurons** (nerve cells) of the auditory portion of the **Eighth (VIII) Cranial Nerve** (the Auditory Nerve). Finally, the electrical potential created from this outer hair cell activity is directly transmitted from the inner hair cells to the eighth nerve dendrites. The electrical impulses then travel along the length of the eighth cranial nerve to the central nervous system. All of this is performed over and over again in a few milliseconds by an organ considerably less than one ml in volume. The intricacy of the cochlear mechanism is one of the most fascinating tales in sensory biology!

Notes

Notes

Structurally, the **cochlea** is an elongated, fluid-filled cavity housed in the **petrous portion of the temporal bone**. This cavity is coiled into a tight spiral that resembles the shell of a snail (or cockle shell –hence cochlea). The broad end of the spiral, which lies close to the middle ear, is called the **base**. The narrow end at the middle of the spiral is known as the **apex**. The cochlea is divided length-wise into three channels by the **basilar membrane** and **Reissner's membrane**. The channel formed by the upper bony wall of the cochlea is known as the **scala vestibuli**. The channel between the basilar membrane and the lower bony wall is called the **scala tympani**. The remaining channel, which lies between the two membranes, is called the **scala media** or the **cochlear duct**. One of the most important characteristics of the cochlear duct is its elasticity. Because it is bound on two sides by tissue membranes, it responds to pressure from either side by moving in either direction.

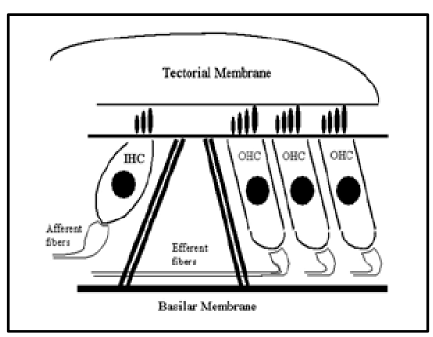
The **cochlear duct** throughout the length of the cochlea separates the **scala vestibuli** and **scala tympani**, except at the apical end, farthest from the middle ear. Here the cochlear duct ends blindly and the two canals communicate through an opening called the **helicotrema.**

The outer chambers, **scalae vestibuli** and **tympani**, are filled with a high sodium low potassium fluid called **perilymph** which is identical to the cerebro-spinal fluid that protects the brain. The cochlear duct contains a second fluid known as **endolymph,** which is perilymph's opposite, having low sodium and high potassium content.

The fluid-filled spaces of the cochlea are separated from the air spaces of the middle ear by the bony wall and two openings or **windows**. One opening, the **oval window**, leads from the middle ear directly into the scala vestibuli. The smallest bone of the middle ear, the **stapes**, fits loosely into the oval window via the **stapes footplate**. The footplate is held in place by the flexible **annular ligament** that seals in the perilymph while allowing the stapes to move in and out of the oval window creating the wave-like movements in the fluid-filled inner ear.

The second opening or window is known as the **round window**. This opening, which is covered by a thin, flexible membrane, leads directly into the scala tympani, on the side of the cochlear duct opposite the oval window. When pressure is applied to the cochlear fluids via the oval window, the round window acts as a pressure release valve by bulging outward, which compensates for the additional pressure.

The **basilar membrane**, which separates the cochlear duct from scala tympani, supports the structures that are directly responsible for sensory function in the cochlea. These include the **organ of Corti** and the **tectorial membrane**. Together along with the basilar membrane, these structures make up what is called **cochlear partition**. Because the cochlear partition plays a very important part in the hearing mechanism we will discuss its anatomic features in some detail.



The **cochlear partition** changes as it progresses from the base of the cochlea to its apex in three ways that are especially important to its function.

- **First,** the width of the partition increases, from base to apex, by approximately tenfold.

- **Second,** its mass increases with the width, primarily owing to an increase in the size and number of supporting cells in the organ of Corti.

- **Third**, the flexibility of the partition changes from being quite stiff at the base to becoming progressively more elastic toward the apex.

This change in elasticity is more than one hundred-fold from base to apex. The consequences of these physical characteristics will be discussed later in this unit.

Figure 4 (following page) provides a cross-sectional view of the cochlear partition. The **organ of Corti** rests directly on the **basilar membrane**. The organ of Corti consists of sensory cells embedded in a matrix of supporting cells. It is the supporting cells that make up the bulk of the organ, anchoring the embedded sensory cells firmly to the basilar membrane. Thus, as the basilar membrane moves, the sensory cells follow its motion closely. These sensory cells occur in rows that run along the organ from end to end of the cochlea. The top of each sensory cell forms part of the upper surface of the organ of Corti and each bears a cluster of stiff **cilia**, or hair-like cells. This is illustrated in Figure 5 on the following page. For this reason, these sensory cells are known as **hair cells**. Atop each sensory hair cell, tiny sensory “hairs” occur in several rows of increasing length, so that the bundles rise in staircase fashion above the surface of the organ of Corti. The tips of these “hairs” or **stereocilia** are connected to the sides of the next tallest **stereocilium** by longer filaments known as **tip links.**

There are two types of hair cells typically found in the mammalian cochlea. The **inner hair cells** *(abbreviated **IHCs**)* lie in a single row close to the inside of the cochlear spiral. They are flask-shaped, and are held firmly in place (non-motile) by supporting cells. Only the cilia of the IHCs move to the transmission of sound. The **outer hair cells** *(abbreviated **OHCs**)* form three rows that lie more toward the outer edge of the cochlea. Outer hair cells are able to move (motile) because they are in contact with the supporting cells only at their very top and bottom. The bulk of their elongated, cylindrical cell body is suspended in the fluid spaces that occur inside the organ of Corti. These spaces are filled with a fluid occasionally called **cortilymph**, which is similar to perilymph in its chemical composition.

Outer hair cell damage is the most common site of problems you will see. Most of your patients with sensorineural hearing loss have extensive damage to the outer hair cells due to noise exposure and/or age. Today’s digital hearing instrument technology is designed primarily to compensate for this outer hair cell damage.

Notes

Notes

Inner Ear

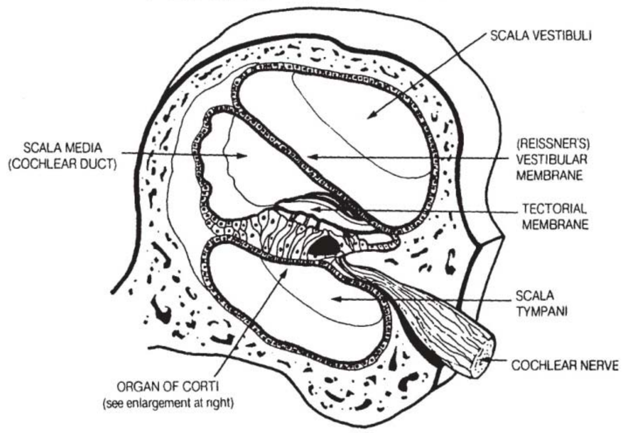


Figure 4

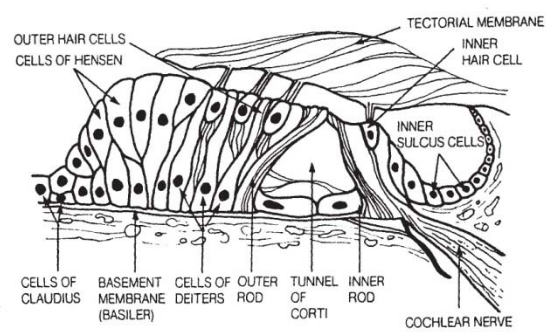
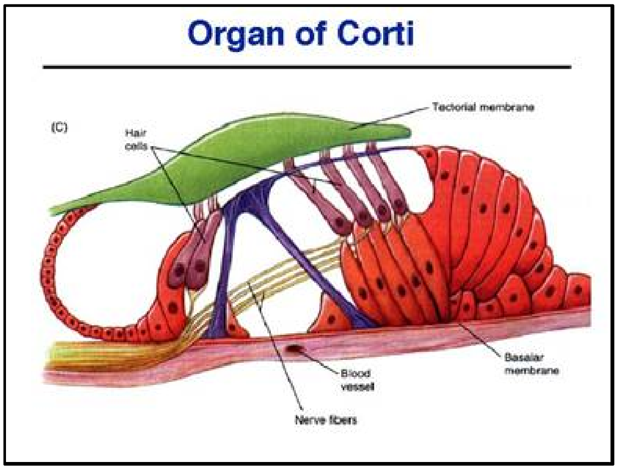


Figure 5

Lying directly above the organ of Corti, but separated from it by narrow space, is the **tectorial membrane**. This unicellular, gelatinous structure is attached at its inner edge to the lining of the bony cochlear wall. Otherwise it is suspended, loosely coupled to the organ of Corti by slender processes. The tallest hairs of the outer hair cells are in firm contact with the underside of the tectorial membrane, facilitating the transmission of movement between the membrane and the outer hair cells. The fluid underneath the tectorial membrane that bathes the cilia of the hair cells is **endolymph.**



The fibers of the eighth cranial nerve enter the cochlea through the center of the cochlear spiral. They emerge from tiny openings in the inner edge of the bony cochlear wall and enter the **organ of Corti**. 95% of these **afferent** fibers (sending signals from cochlea to brain) approach the closest inner hair cells to form a one-to-one connection, about twenty fibers per hair cell. The remaining 5% of these fibers travel across the organ of Corti, turning down the cochlea toward the base to connect with groups of outer hair cells. Each fiber may be connected to approximately twenty to fifty sensory cells, and each outer hair cell may receive processes from approximately twenty afferent fibers. Obviously, given the disparity in the pattern of innervation for these two types of hair cells, their behavior would be quite different. *It would appear that most of the sensory information going to the brain would originate from the inner hair cells and very little from the outer hair cells.*

The afferent nerve fibers contact both inner and outer hair cells at the bottom at the end farthest from the sensory hairs. Seen under high power magnification, the bottom of the hair cell appears to contain the pre-synaptic structures necessary for chemically transmitting the signals to eighth nerve cells. The **spiral ganglion**, just inside the bone of the modiolus from the Organ of Corti, contains the cell bodies of the eighth nerve cells.

A small number of the fibers in the eighth nerve are **efferent** (sending signals from brain to cochlea). These fibers arise from neurons whose cell bodies are located in the brain stem, mostly on the side opposite from the ear to which they travel.

Two populations of efferent nerve fibers have been identified in the cochlea. Each population originates from a distinct cell group in the brain stem, and they also differ in their biochemical makeup. Once these efferent fibers reach the cochlea they branch out to form a large number of nerve endings. One group contacts the afferent fibers beneath the inner hair cells while the other group terminates primarily at the outer hair cells. However, in some species this second group also supplies endings that contact the afferent fibers under the inner hair cells.

Notes

Notes

Being tiny does not stop the cochlea from consuming a great deal of energy; consequently, the inner ear possesses its own extensive network of blood vessels to supply oxygen and nutrients. The cochlear artery enters the inner ear alongside the eighth cranial nerve and then divides into two major pathways. One artery branches into an extensive network of capillaries that occupy the outer wall of the cochlear duct supplying the **stria vascularis, spiral prominence**, and **spiral ligament**. These structures consume large quantities of energy in order to maintain homeostasis, or a state of equilibrium, in the cochlea even when the inner ear is at rest. The outer wall structures and especially the stria vascularis are thought to be responsible for generation of the endocochlear potential (a resting potential critical to the function of the inner ear) and for the secretion of endolymph. The second source of arterial blood are the spiral vessels that run longitudinally through the **spiral ganglion** and **spiral limbus**, and just beneath the **basilar membrane**. It is this second arterial vessel that is thought to supply oxygen and nutrients to the organ of Corti.

The inner ear is not just an organ of hearing. It is also responsible for maintaining your balance via the fluid filled and cilia lined **semicircular canals** and **vestibule**. Collectively, this connected series of canals keeps you aware of your position (lateral, vertical or horizontal) in space as you move in different directions. The vestibule and semicircular canals share the same **perilymph** and **endolymph** found in the cochlea.

Because of this close relationship between the **cochlea** and the **semi-circular canals** it should be no surprise to you that disorders of hearing and disorders of balance frequently go hand-in-hand. As your career in the hearing healthcare industry develops you will undoubtedly see patients who have problems with their balance. Some, but not all, of these “dizzy” patients will have medical conditions affecting both the balance and hearing parts of the inner ear. These patients must be given special consideration (not to mention medical clearance) when considering the fitting of hearing instruments. You will find more references to this and other related concerns in Unit 5: Introduction to Hearing Disorders.

Section 2: How the Cochlea Analyzes Sound

Notes

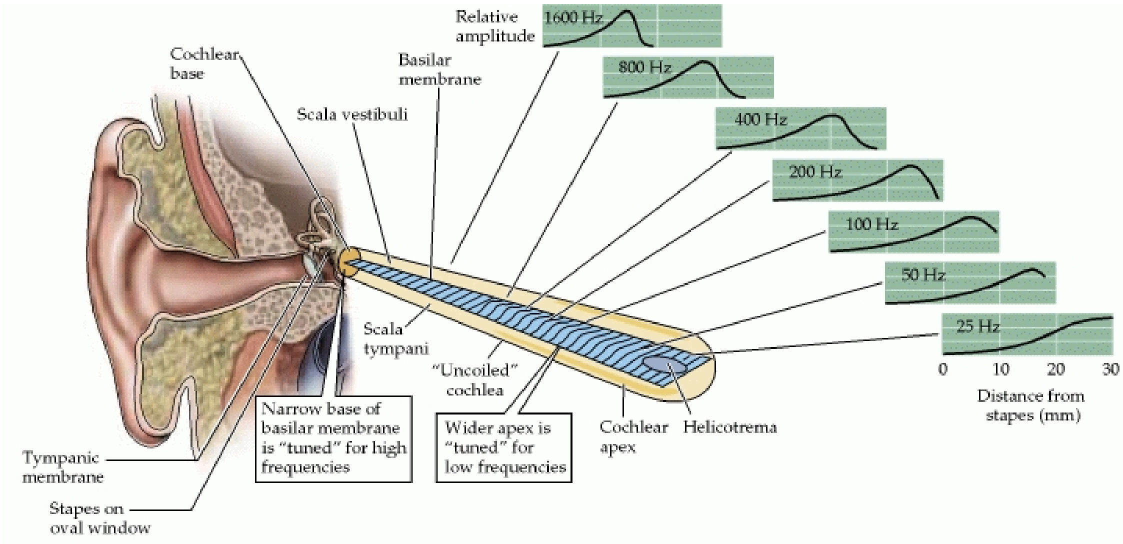
In this section we will discuss some of the acoustical properties of the traveling wave and how the cochlea processes sound. If you haven't yet tackled the module unit dedicated to acoustics you may be asking, "What wave and why does it travel?" Don't worry. These concepts are pretty straightforward. It's important to address this topic here for one important reason: It is the damaged cochlea's inability to precisely amplify sound that gives you a reason to fit hearing aids on about 90% of the patients you will see. You can assume over 90% of your patients will have sensorineural hearing loss.

Fitting the Ear with Hearing instruments is Different than Fitting the Eye with Glasses!

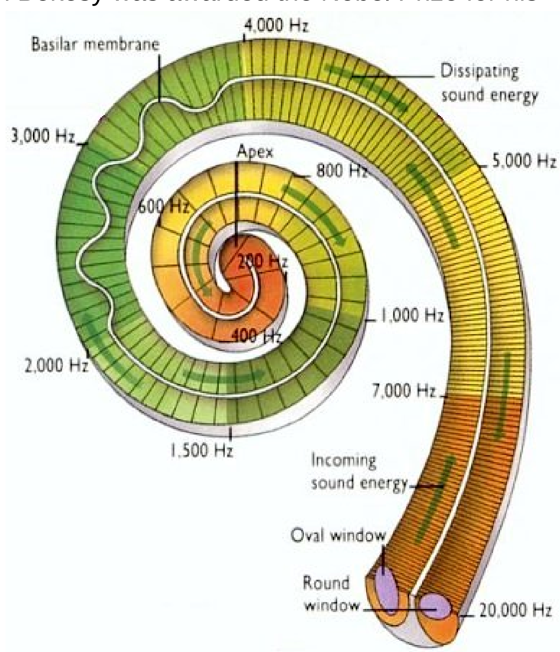
The human retina, like the cochlea, is a complex sensing organ; however, most people with vision problems have normal retinas. Most vision problems are **conductive**; e.g. they result when the eyeball shortens or lengthens and incoming light falls short of or overshoots the retina. Either way, the shortfall or an overshoot (near-sighted or far-sighted) may be corrected by simply refocusing the light on the retina. Once the optician is able to refocus the light through a prescriptive lens, normal vision is restored.

The job of correcting a sensorineural hearing loss is not as simple as fitting a pair of glasses. This is because, while some of our patients will have what we call a **"conductive"** loss of hearing (loss resulting from damage to the outer and/or middle ear), *most of the patients with hearing loss we will see will have damage to the cochlea*. For the latter, hearing instruments will not restore normal cochlear function. It is this damage to the cochlea that makes fitting hearing aids extremely challenging. Given time, you will learn that it is your knowledgeable application of what we call **compression** in a hearing instrument that will compensate for some of this irreversible damage to the cochlea.

You will remember that the basilar membrane of the cochlea is finely tuned to different frequencies or **tonotopically arranged**. Tonotopic means that specific parts of the cochlea are more sensitive to specific frequencies or pitches of sound. See picture on next page.



The **base** of the cochlea, which is narrow and stiff, is tuned to high frequency sounds. The **apex** of the cochlea, which is wider and heavier, is tuned to low frequency sounds. In other words, high frequency sound waves travel to and stimulate the base of the cochlea while low frequency sound waves travel to and stimulate the apex of the cochlea. George von Bekesy first theorized the concept of the **traveling wave** in the 1950's through his work with cadavers. (Gruesome yes, but keep in mind, the cochlea was, at that time, completely inaccessible in the living.) Besides, he was nobody's ghoul; von Bekesy was awarded the Nobel Prize for his research on the traveling wave.



From his observations, von Bekesy believed that the cochlea was passive, rather than sharply tuned (remember, he was working with cadavers). He believed that the acuity of human hearing ability did not occur in the cochlea; rather, he attributed all fine-tuning to central nervous pathways located higher up in the system. The cochlea was thought to function as a simple sound transmitter, sending auditory information along to wherever it would be "amplified" by the auditory nervous system. Though von Bekesy may have been brilliant, he was still wrong.

Notes

Twenty-five years ago we recognized the cochlea as being a sharply tuned **cochlear amplifier** and more recently that the OHCs are primarily responsible for this sharp tuning. As each point along the basilar membrane is precisely tuned to a specific sound frequency, the OHCs located at that point are equally sensitive to that same specific frequency. Through the active mechanics of the OHCs, the cochlea actually adds energy to the sound before it travels up to the brain via the auditory nerve. *OHCs are an active biological amplifier, not a passive mechanical filter.*

It is now well-accepted that both the traveling wave theory and the activity of the OHCs are primarily responsible for our keen ability to detect the smallest differences in intensity and frequency and it is this acuity that allows those of us with normal hearing ability to readily differentiate between similar speech sounds found in words like “buy” and “guy” and pick out one voice from a roomful of people talking.

So what happens when someone has a hearing loss?

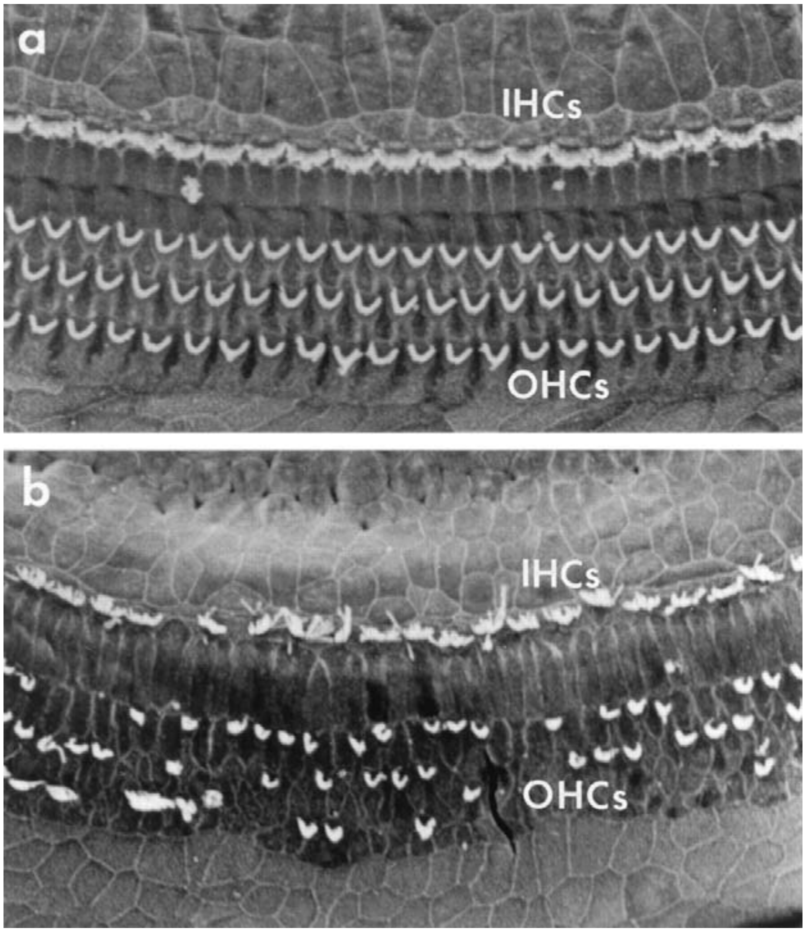
So far we have learned a lot about the cochlea’s complexity. We’ve seen that it not only transmits sound to the brain’s cortex via the auditory nerve, but that it also provides selective amplification to these sounds. We also know that in order to accomplish these incredibly intricate feats, the cochlea requires a steady blood supply while generating a stunningly awesome amount of biochemical activity. Most of us also know the old adage of “the more complex something is, the more that can go wrong with it”. This is also true of the human cochlea; noise exposure, infection, diet, medication, the aging process, disease, and other harmful agents can wreak havoc on an otherwise healthy inner ear.

Because outer hair cells are the primary site of all that biomechanical activity (they move faster than any muscle in the human body) they are the most susceptible to damage. When OHCs are damaged two important things happen:

1. There is a mild to moderate loss of hearing (Up to 50 dB)

2. The cochlea loses its ability for sharp tuning

Notes



When these two things occur at the same time, people will not only notice an inability to pick up softer sounds (e.g. birds, a baby crying from another room, the sound of a turn signal, etc) but will complain that they frequently miss or misunderstand words during normal conversation. Two of the most common types of damage to the cochlea are the result of aging (**presbycusis**) and exposure to high levels of sound (**noise-induced hearing loss**). Both of these generally affect OHC performance.

Outnumbered by OHCs (4 to 1), **inner hair cells** *(abbreviated **IHCs**)* are tougher and more resistant to damage. However, because IHCs are directly connected to the auditory nerve, the subsequent hearing loss is more severe when they are damaged.

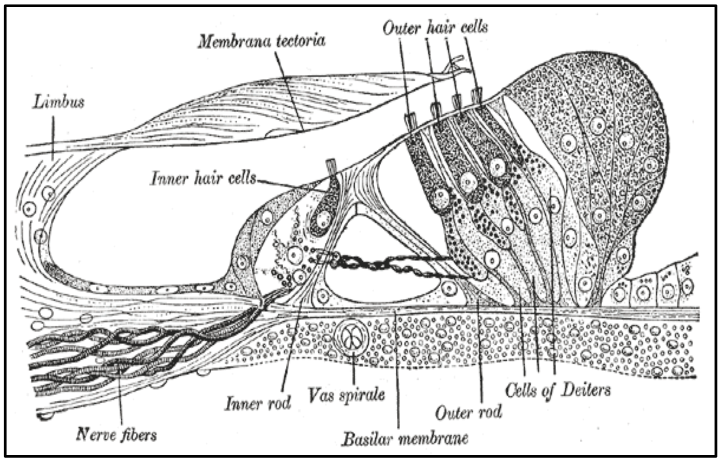
In the beginning, you won't have to be too concerned whether or not the impaired hearing of your patients is the result of IHC or OHC damage. For now when the hearing loss is both sensorineural and mild to moderate (<50dBHL), you can assume there is primarily OHC involvement, while more severe impairments (>50 dB HL) will typically involve both OHC and IHC damage.

A Quick Guide to Essential Cochlear Anatomy and Physiology

Notes

You may pat yourself on the back if you have read this entire unit on the anatomy and physiology of the cochlea. This is some heavy stuff! You should now appreciate the fact that when you are sitting before a patient, trying to solve their communication problems, you won't need to know or explain all the details. Just take a look at the following Quick Guide and familiarize yourself with the key differences between outer and inner hair cells. Any information you retain will help to emphasize these differences when helping patients in the selection of hearing instruments.

Outer Hair Cells (OHCs)



1. Shaped like a cylinder
2. End of OHC is embedded in the tectorial membrane
3. Efferent: they receive information from the auditory nerves of the lower brainstem
4. OHCs mechanically boost soft incoming sounds. OHCs are sometimes called cochlear amplifiers
5. OHCs sharpen the peaking of the traveling wave
6. OHC damage results in up to 50 dB hearing loss

Inner Hair Cells (IHCs)

1. Shaped like a flask
2. Do not touch the tectorial membrane
3. Afferent: they send information to the brain via the auditory nerves of the lower brainstem
4. IHC damage results in severe hearing loss (greater than 60 dB) and/or very poor word understanding ability. This is because sound is not getting to the brain.

Section 3: What are you actually doing when you fit hearing aids?

Notes

Now it is time to take a deep breath and relax. It is easy to get overwhelmed by all the terminology and jargon associated with fitting hearing aids. Just take a look at the hearing aid programming software on your computer if you need a reminder! It may help if we take steps here to simplify the hearing aid fitting process.

The cochlea is buried deep in the temporal bone of the skull right behind the pinna. Whenever someone pushes an amplifier encased in a hard plastic shell into their tight, humid, waxy ear canal they can literally apply more force to the eardrum and ossicles than they were ever designed to absorb.

When this amplified sound eventually reaches the damaged cochlea, its boosted sound energy excites any remaining undamaged outer hair cells. The primary purpose of hearing aids is to restore the missing sounds of quiet speech. Unfortunately, normal cochlear function cannot be restored. Hearing aids do a great job of restoring soft sounds, but at this point they cannot bring back cochlea's ability to sharpen the peaks of the traveling wave.

Why people who have hearing loss complain about hearing in noise:

For anyone who has spent more than a few days visiting with patients in a hearing aid office it is obvious: The most frequent complaint or comment by someone with a mild hearing loss is an inability to hear in noise. Why?

First, it is important to realize that it is hard for anyone, normal hearing or otherwise, to hear in noise. The din of a crowded restaurant has a lot more intensity than the typical volume of the average talker. Second, speech is a very dynamic signal. It has a characteristic distribution of frequency and intensity that is always shifting and changing. Third, people with mild hearing loss frequently miss the softest *consonant* sounds of speech in noisy environments. When this occurs, speech may sound like it is loud enough, but it is not really clear or distinct. These folks can hear the hubbub of the on-going noise, but because the damaged cochlea has lost some of its sharp tuning, it becomes difficult to accurately cue in on all the quick changes in intensity of speech, especially when it is **masked** or buried in a sea of noise. Confused? Think of the Where's Waldo puzzle pictures that were so popular a few years back. The task was to find the tiny Waldo figure (distinguishable mostly by his red and white striped cap) in a visual sea of bodies and activities. With normal vision it was difficult enough, but with eyes crossed it would have been impossible. Remember that damage to the OHCs in the cochlea have eliminated the fine tuning of hearing. Incoming sounds may be louder in noise, but they are no more distinguishable than red and green dots to a color blind person.

Hearing instruments, even advanced digital hearing aids with noise reduction, cannot correct for this problem completely enough for a hearing impaired patient to function as well as normal in all listening situations. Digital hearing aids with noise reduction features help to soften the hubbub of background noise, keeping the sound environment more comfortable, but they still do not make one speech sound more distinct than another (remember“buy”and“guy”?) They just can’t sharpen those peaks. Before you get too discouraged, it is good to know that hearing aids really do help people to hear better, and in many cases they keep them more comfortable while doing so. It’s just that there are no substitutes for a normally functioning, healthy cochlea. That said, you will learn in Unit 8: Introduction to Hearing Instruments, about highly specialized hearing aids that take advantage of directional microphones and do a great job of helping people to hear much better in noise. Stay tuned…the best is yet to come!

Notes

Check Your Understanding

1. Damage to OHCs typically result in: (select all that apply)
    a. Sensorineural hearing loss
    b. Mild to moderate hearing loss
    c. Severe to profound hearing loss
    d. None of the above

2. Damage to the IHCs typically results in: (select all that apply)
    a. Sensorineural hearing loss
    b. Mild to moderate hearing loss
    c. Severe hearing loss
    d. None of the above

3. Name four characteristics of OHCs that make them different than IHCs
    1) ______________________________
    2) ______________________________
    3) ______________________________
    4) ______________________________

4. The resonant frequency of the normal outer ear is:
    a. 500 to 750 Hz
    b. 1000 to 2000 Hz
    c. 2000 to 5000 Hz
    d. 10000 to 12000 Hz

5. Eustachian tube dysfunction may result in: (select all that apply)
    a. Conductive hearing loss
    b. Sensorineural hearing loss
    c. Otitis Media

6. The IHCs of the cochlea are responsible for your ability to hear the differences between very similar sounds.
   a. True
   b. False

7. The apex of the cochlea is tuned to high-pitched sounds.
   a. True
   b. False

8. The Eight (VIII) cranial nerve is also called the:
   a. Facial Nerve
   b. Auditory Nerve
   c. Organ of Corti
   d. Tensor Tympani

9. Which of the following fluids are not found in the cochlea? Select all that apply.
   a. Cortilymph
   b. Perilymph
   c. Endolymph
   d. Pseudolymph

10. Transduction refers to:
   a. The transportation of energy
   b. The changing of energy from one form to another
   c. The path of the traveling wave
   d. The boosting of sound via mechanical mean

Unit 5: Introduction to Hearing Disorders

Objectives

Notes

Overview

The primary purpose of this unit is to provide you with general information about the essential medical aspects of hearing impairment. The goal of this unit is *not* to turn you into an expert of hearing disorders; rather, the focus is on essential and practical issues that will help you to take better care of your patients.

Before reading this chapter it would be very beneficial to first read the Anatomy and Physiology, Audiogram Interpretation and Acoustics units. All of them contain valuable information that will help you to better understand this chapter. Knowing how the normal ear looks and works helps you to better understand the damaged or diseased ear.

Goals

Upon completion of this module, you will be able to:

- Recognize the landmarks of the normal eardrum
- Recognize some of the patterns on the audiogram that would require a referral to a physician.
- Determine when to refer a patient to a physician with a possible medical problem
- Determine how some common hearing disorders affect the hearing test results
- Differentiate between a conductive, sensorineural and mixed hearing loss

Follow Up

- Campbell, K. (1998). Essential Audiology for Physicians. San Diego, CA: Singular Press.
- Mueller, J. H. (1998). Auditory Desk Reference, Volume 1. San Diego, CA: Singular Press.

Section 1 – Introduction to Hearing Disorders

Notes

The first important lesson relating to hearing disorders is that you must know when to refer a patient to a physician for a medical evaluation. In fact, before you even begin to discuss hearing aids with a patient it is imperative that you have ruled out a treatable or dangerous medical problem involving the ears. This means you have to recognize what a hearing disorder looks like on an audiogram.

Probably your biggest professional obligation is to know the **8 Signs of a Medical Pathology.** As a hearing care professional, you are mandated by the US Food and Drug Administration (FDA) to refer a patient immediately to a physician if you see one of these problems. When you see one of these signs you should immediately refer your patient to a medical doctor for a complete evaluation. In many of these cases the medical doctor will evaluate and treat the patient and refer them back to you for a hearing aid evaluation. A good term to know is **RED FLAG**. A red flag is any of the 8 signs of a hearing disorder that show up on the audiogram or in your case history. Red flags need to be handled immediately by referring the patient to a medical doctor who specializes in diseases of the ear.

Here are the Eight Classic Red Flags of a Hearing Disorder as outlined by the FDA:

1. Visible deformity to the outer ear
2. Visible evidence of significant cerumen accumulation or a foreign body in the ear canal
3. Any history of or active drainage from the ear within the previous 90 days
4. Any history of sudden hearing loss within the previous 90 days
5. Any acute or chronic dizziness
6. A hearing loss in one ear of sudden or rapid onset within the previous 90 days
7. Ear pain or discomfort
8. An air bone gap on the audiogram of more than 15 dB at 500, 1000 and 2000Hz

Notes

When someone walks into your door to get a hearing test, they may be experiencing several symptoms of a hearing problem. It's important to know and understand what some of the most common symptoms are. Here is a list of the most common symptoms and their definitions:

Tinnitus: The perceived sensation of sounds in the ear or head with no external source. Tinnitus is often associated with sensory hearing loss

Vertigo: Dizziness. True vertigo is a severe spinning sensation usually of short duration .

Otalgia: This is a Latin term for *ear pain*.

Aural Fullness: The perceived sensation of a plugged ear

Hyperacusis: An abnormal sensitivity to sound

Recruitment: Abnormal growth of loudness

There are many signs and symptoms of a hearing disorder you need to know. In order to examine the most common hearing disorders, let's break the ear up into its three major parts: outer, middle and inner and look at each part separately in relation to the 8 Red Flags.

Check Your Understanding

In the FDA question regarding air/bone gaps, which of the following is a criterion for requirement of a medical referral?

a. A/B gaps of 15 db or greater at 1000, 2000 and 4000 Hz

b. A/B gaps of 15 db or greater at 500, 1000 or 2000 Hz

c. A/B gaps of 15 db or greater at 1000, 2000 or 4000 Hz

d. A/B gaps of 15 db or greater at 500, 1000 and 2000 Hz

Section 2: Outer Ear Disorders

Notes

During embryonic development, the growth of the structures of the outer ear may be interrupted or prevented altogether. This can result in a number of **congenital** (present at birth) conditions:

1. **Stenosis** – a narrowing of the ear canal
2. **Atresia** – the absence of an ear canal
3. **Microtia** – a smaller-than-normal pinna
4. **Anotia** – absence of one or both external ears

In cases of Atresia, Microtia, or Anotia, the inner ear structures may still be intact. If this is the case, the patient would benefit from either a bone conduction hearing aid or a bone-anchored hearing aid (BAHA).

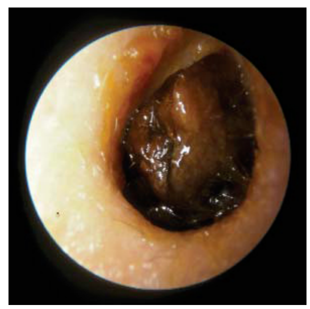
Below is an image of a **microtic pinna** with complete **Atresia** of the ear canal.

Other common disorders of the outer ear include:

Notes

- **Otitis externa** ("Swimmer's Ear") – a bacterial infection and inflammation of the walls of the EAM. It can cause conductive hearing loss and pain - generally in one ear.
- **Polyps** – growths of cartilage in the outer third of the ear canal
- **Otorrhea –** infected discharge (pus) from the ear canal or middle ear
- **Osteoma** – bony tumors in the inner two thirds of the ear canal

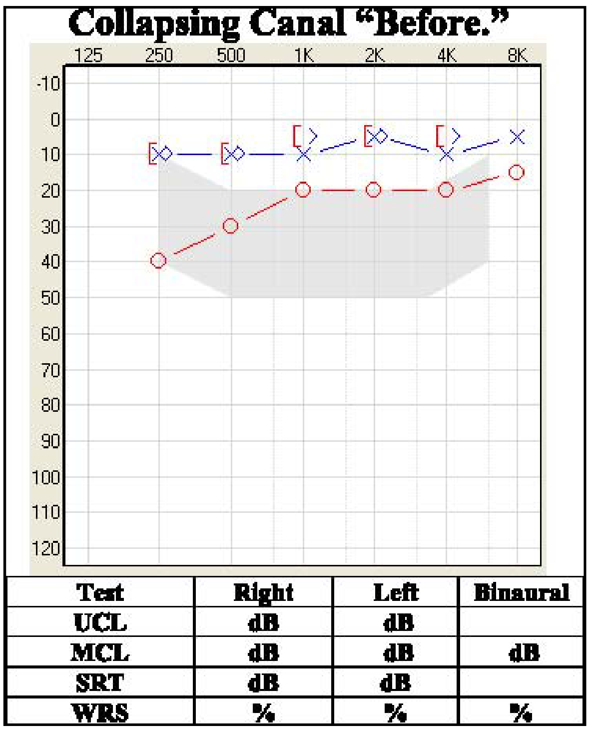
- **Cerumen impaction** – excessive earwax occluding the ear canal

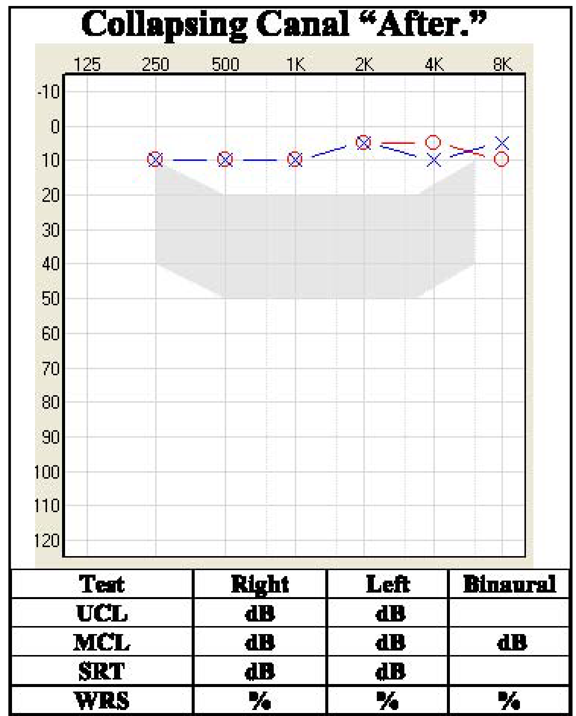


- Foreign object in the ear canal

Collapsing Ear Canal

Some people, particularly the elderly, have ear canals that collapse. This means that the tissues lining the ear canal have become very soft. This is a normal condition and does not cause hearing loss in the vast majority of cases: that is, until it is time to do a hearing test. When you place the headphones on someone with collapsing ear canals, you are actually causing a hearing loss because the soft tissues of the ear canal collapse, blocking the ear canal and obstructing sound. This condition results in an apparent conductive hearing loss and can be easily fixed by using insert earphones. Below is an example of an audiogram of a patient with collapsing ear canals. The audiogram on the right is after the use of insert phones. Note how the loss returns to near normal levels. If insert ear phones are not an option, a small piece of #13 tubing placed in the bottom of the ear canal under headphone can be a possible fix.





Check Your Understanding

Which of the following is a bacterial infection and inflammation of the walls of the EAM?

a. Otorrhea

b. External Otitis

c. Osteatoma

d. Collapsed canal

Notes

Section 3: Middle Ear Disorders

Notes

Otosclerosis

Otosclerosis is a condition that affects the bones of the middle ear. In simple terms, the bones of the middle ear lose their hardness and become spongy. When this occurs sound is not transmitted through the middle ear to the inner ear as efficiently and results in a conductive hearing loss. The condition is hereditary (runs in families).

Most cases of advanced otosclerosis result in a mixed hearing loss. Eventually, this condition may begin to affect the inner ear. In cases in which the loss only affects the middle ear, surgery can be performed to improve the hearing. **Figure 6** (shown on a later page) is an example of otosclerosis you might see in your office on an audiogram. Notice the characteristic notch in bone conduction thresholds at 2000 Hz. This is called "Carhart's Notch" and is a classic audiometric indication of otosclerosis.

Otitis Media

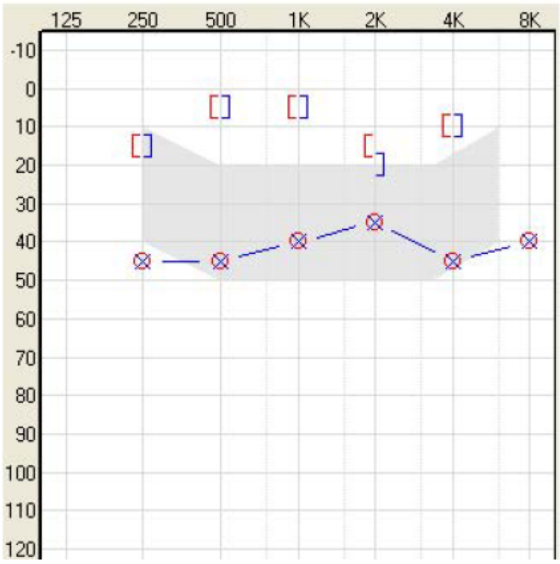
Otitis Media is the term for fluid in the middle ear. This is probably the most common type of hearing loss in people under the age of eighteen. However, adults may also present with otitis media. Otitis media causes a temporary conductive hearing loss. An example of Serous Otitis Media, with air bubbles, can be seen in the picture to the right.

Otitis media results from a fluid buildup in the air filled middle ear space. Often, while suffering through a common cold, the Eustachian tube, which allows ventilation (air flow) into the middle ear space, becomes swollen and closes. A normal consequence of this is otitis media. There are many stages of otitis media. If the fluid is infected (suppurative) it can be quite painful. If the fluid is not infected (non-suppurative or serous) it may not be painful at all. All cases have one thing in common; they result in a temporary conductive hearing loss. Figure 5 on the following page is a classic example of what a case of otitis media looks like on the audiogram.

Figure 5 is an audiogram from a case of acute bilateral otitis media. **Figure 6** is an example of bilateral otosclerosis. Notice the Carhart Notch at 2000 Hz.

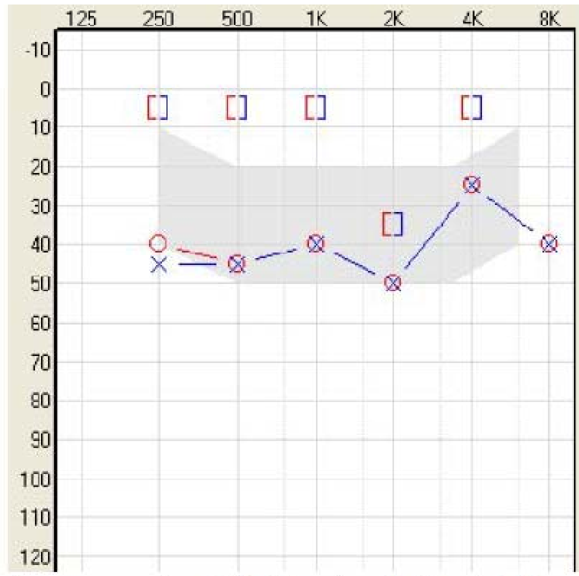
Notes

Figure 5



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	
SRT	40dB	40dB	
WRS	%	%	%

Figure 6



Test	Right	Left	Binaural
UCL	100 dB	100 dB	
MCL	75 dB	75 dB	
SRT	40 dB	40 dB	
WRS	96%	92%	96%

Other Disorders of the Middle Ear

- **Monomeric Spots** – a chalky scar on the TM as a result of a healed hole in eardrum (typically does not cause hearing loss)

- **Tympanosclerosis** – thickening and scarring of the TM as the result of recurrent perforations and ear infections. (more likely to cause a conductive hearing loss). This can be seen in the image below.

- **Perforated or Ruptured TM –** the picture below is an example of a perforated TM with a monomeric spot.

- **Cholesteatoma** – A “tumor” of the TM. Occasionally, in the process of healing a perforation, the layers of the TM do not come together evenly, creating a pocket into which debris, cerumen and dead skin can accumulate. This pocket is a cholesteatoma. It may invade into the middle ear space and eventually erode the ossicles.

Notes

Check Your Understanding

What is the name of uninfected fluid in the middle ear?

a. Perilymph fluid

b. Duct fluid

c. Serous Otitis Media

d. Non Serous Otitis Media

Section 4: Inner Ear Disorders

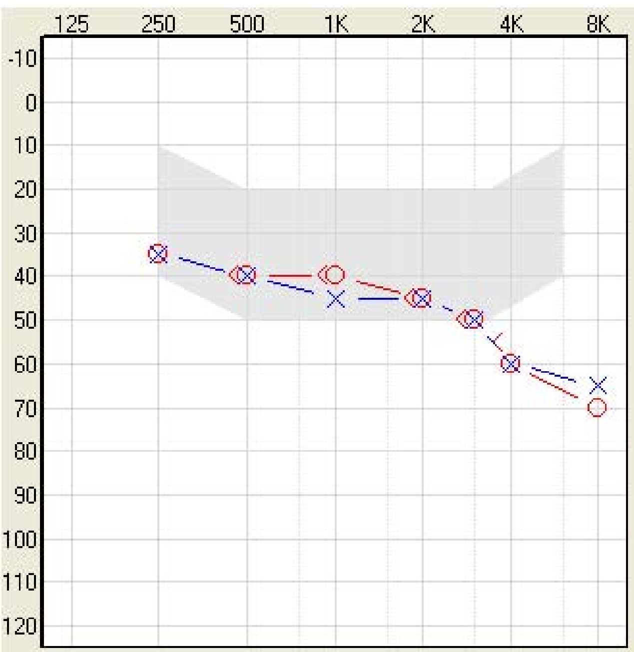
Notes

Presbycusis (Sensory)

Presbycusis is hearing loss in humans caused by the aging process or other factors that occur over a long life. Over time, the typical person is exposed to a wide variety of noise insults and other agents that can cause wear and tear on the ear. All of us are subject to the occasional firecracker, loud concert and the steady din of traffic. Over the course of your lifetime, this intermittent exposure can gradually cause a hearing loss. It is important to understand that hearing loss is not a *normal* consequence of the aging process. Not all older people have presbycusis.

Presbycusis affects all parts of the ear, but the primary site-of-lesion is the cochlea. The outer hair cells within the cochlea are particularly sensitive to the occasional exposure to loud sounds. Presbycusis is not a hearing disorder that requires medical treatment. Presently there is no cure for presbycusis; however, most patients with it do well with hearing aids. A classic case of cochlear presbycusis may be seen on the audiogram below. Note the symmetry and the gradually sloping shape of the audiogram.

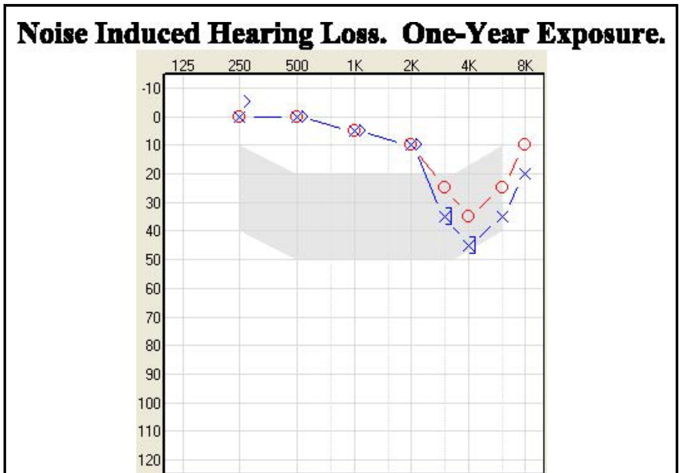
Cochlear Presbycusis



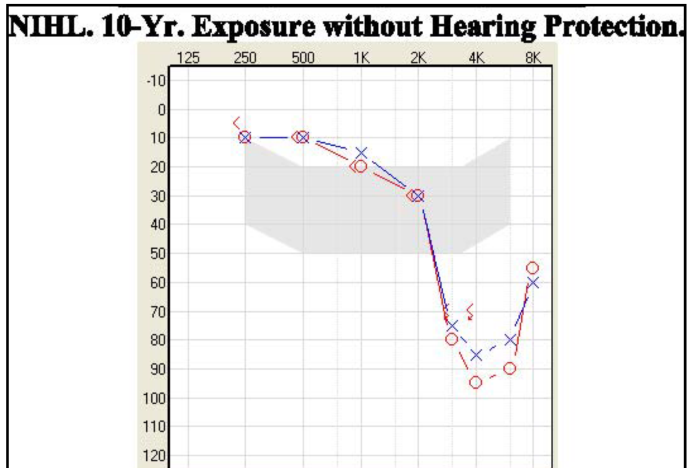
Notes

Noise Induced Hearing Loss (NIHL) (Mechanical)

NIHL affects the cochlea of the inner ear. Because of the shape of the cochlea and the resonant effects of the outer ear, classic cases of NIHL show a precipitous high frequency hearing loss with thresholds poorest around 4000 Hz and with some recovery in the highest frequencies. This audiometric pattern is called a "noise notch". NIHL can affect people of all ages. NIHL in its most common form is of very gradual onset. The two audiograms shown below are from the same factory worker taken ten years apart. Notice that the loss has gotten worse over the ten year period.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	5 dB	5 dB	
WRS	92 %	92 %	%



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	20 dB	15 dB	
WRS	84%	84%	%

People with NIHL can be fit with hearing aids. Many people with the hearing loss shown in this audiogram example say they can hear, but not understand speech completely. This is because the good low frequency hearing gives the false impression of normal hearing, while the poor high frequency thresholds impair perception of consonants such as /f/, /s/, or /t/. Understanding in the presence of background noise can be difficult as well.

Notes

Ototoxicity (Metabolic)

There are several drugs used for therapeutic treatment of diseases that have the potential side-effect of causing damage to the hair cells of the inner ear. Because the cochlea is such a delicate organ it is susceptible to damage from powerful medications and/or chemical agents. Such agents are considered to be **ototoxic**; that is, poisonous to the ears. Ototoxic drugs cause sensorineural hearing loss. The amount of ototoxic hearing loss depends on the exact dosage, duration of use and the individual's susceptibility. When you encounter a patient who has used or been exposed to an ototoxic medication or agent you should consult a physician or pharmacist. A list of medications being taken should always be obtained from your patient during the case history.

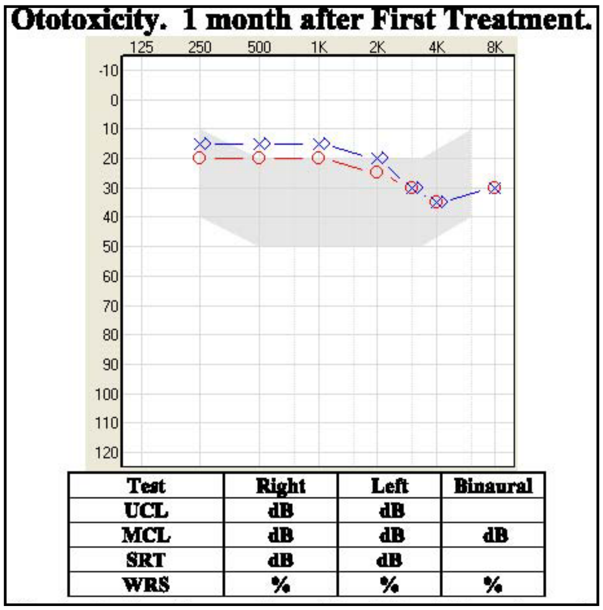
There are several ototoxic medications and agents. The most common ones are listed below along with their therapeutic uses. Whether the drugs cause a permanent or reversible hearing loss is also listed. The majority of drugs cause a permanent hearing loss, but in some cases the loss is reversible. This list is by no means exhaustive; rather, it is designed to represent a sample of the most common ototoxic agents you will encounter. New medications are always being introduced onto the market and it is best to consult with your local physician or pharmacist for the most current information. Medications are constantly being introduced and you will need to update the list regularly.

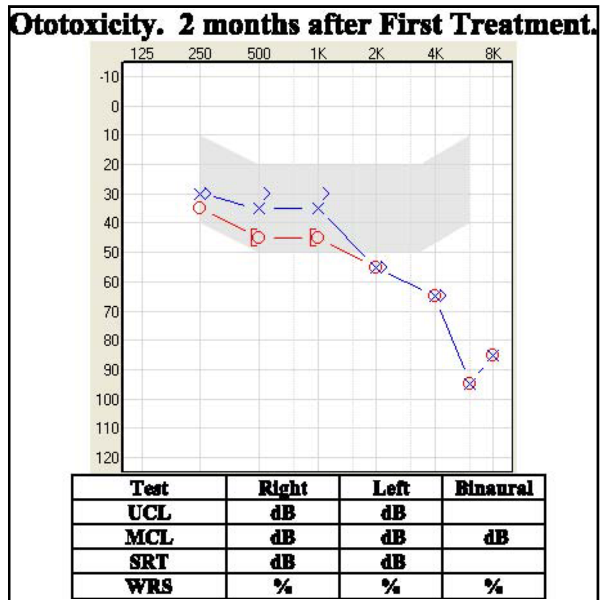
Type of Drug	Type of Hearing Loss	Reversible? (Y/N)
1. Aminoglycoside Antibiotics	Sensorineural	No
• Streptomycin • Gentamycin • Kanamycin • Vancocin		
2. Cancer Chemotherapeutics	Sensorineural	No
• Cisplatin • Carboplatin		
3. Loop Diuretics (Furosemide)	Sensorineural	Yes
• Lasix • Bumax		
4. Salilcylates	Sensorineural	Yes
• Aspirin		
5. Quinine	Sensorineural	Yes

Notes

Below are two audiograms from a patient who has been receiving large doses of cisplatin for lung cancer. The first audiogram is one month after the first treatment and the second audiogram is sixty days later. Note the difference in the thresholds due to the treatment duration.

As a dispensing professional you will probably not be directly involved in collecting these types of serial audiograms, however, it's important to note how and when various treatments may affect someone's hearing. Be aware if your patient complains that their hearing aid is not working properly; it is possible there has been a threshold shift due to medications.





Meniere's Disease

Meniere's disease occurs from an over secretion or under absorption of the fluids of the Inner ear. The resulting buildup in pressure within the inner ear may cause the following set of symptoms:

1. A usually unilateral, low frequency sensorineural hearing loss of sudden or rapid onset that sometimes fluctuates

2. A fullness or pressure sensation in the ear

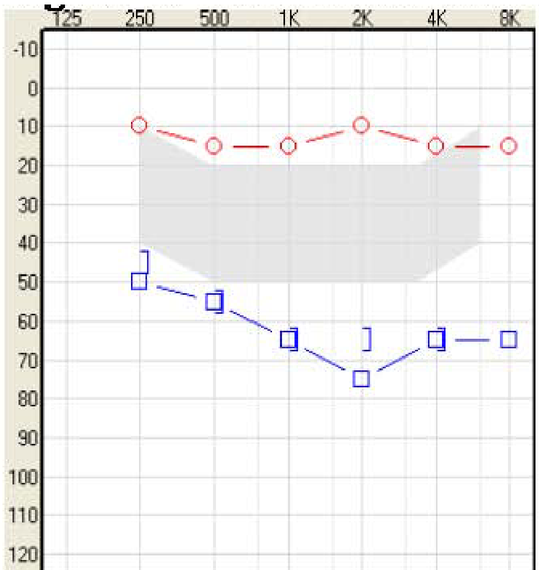
3. Brief and sudden episodes of severe dizziness (vertigo)

4. A roaring (tinnitus) in the affected ear

One or all of these symptoms require an immediate referral to a physician. There are many sub-categories of Meniere's disease beyond the scope of this tutorial. Many types of sudden hearing losses, although they are sensorineural, respond to medical treatment if referred immediately to a physician.

Notes

Below is a complete audiogram of a patient with Meniere's disease. Meniere's disease is characterized by a hearing loss of sudden onset in one ear accompanied by a pressure sensation in the ear, roaring tinnitus and sudden episodes of severe dizziness (vertigo). The patient will also exhibit very poor speech discrimination in the affected ear. Note the asymmetrical nature of the hearing loss. After this hearing loss has stabilized and the physician has given authorization, this person might be fit with a hearing aid in the affected ear.



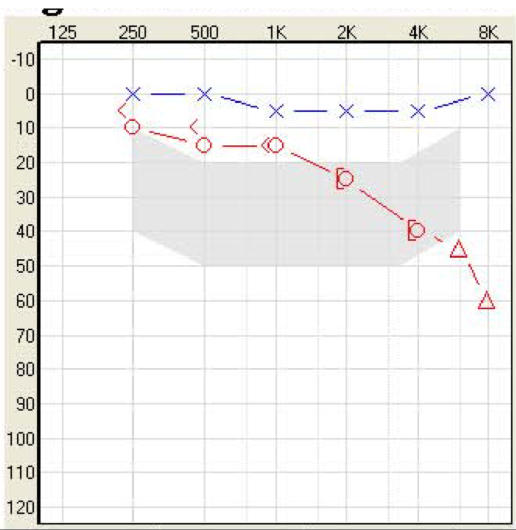
Test	Right	Left	Binaural
UCL	90 dB	95 dB	
MCL	55 dB	85 dB	dB
SRT	10 dB	65 dB	
WRS	100%	4%	%

Acoustic Neuroma

An acoustic neuroma is a benign tumor of the VIII (Auditory) Cranial Nerve.

An asymmetrical sensorineural hearing loss, particularly with poorer-than-expected word recognition ability in the poorer ear is a classic sign of this on an audiogram.

Below is a complete audiogram of a patient diagnosed with an acoustic neuroma. Note that the difference between the two ears is not as great as in Meniere's Syndrome. Note also that the WRS for the RE is much poorer than would normally be expected given the relatively mild sensorineural hearing loss in that ear.



Test	Right	Left	Binaural
UCL	85dB	90dB	
MCL	55dB	45dB	55/50dB
SRT	20 dB	5 dB	
WRS	52%	100%	64%

Notes

Other Disorders of the Inner Ear

- **Recruitment** – abnormal growth of loudness. For many people who have sensorineural hearing loss, small increases in loudness are perceived as large increases in loudness.

- **Hyperacusis** – abnormally acute hearing due to heightened irritability of the sensory and/or neural mechanisms. Whereas the person with recruitment would be bothered by loud sounds, the person with hyperacusis may be bothered by moderate or even soft sounds.

- **Tinnitus** – ringing, buzzing, popping, clicking, roaring, hissing or other sounds that are perceived with no external stimulus. This is frequently, but not always, present with sensorineural hearing loss.

- **Congenital sensorineural hearing loss** – Hearing loss that is present at birth. Approximately 3-6 per 1,000 infants are born with significant sensorineural hearing loss. Congenital hearing loss frequently displays an atypical audiometric configuration, such as “cookie bite” or reverse slope.

Check Your Understanding

Noise induced hearing loss frequently shows an audiometric configuration in which the poorest threshold is present at what frequency?

a. 500 Hz

b. 1000 Hz

c. 2000 Hz

d. 4000 Hz

Notes

Check Your Understanding

1. Match the following hearing disorders with the appropriate type of hearing loss:

I.	Meniere's Disease	**<u>S</u>**ensorineural
II.	Ototoxicity	**<u>M</u>**ixed
III.	Otitis Media	**<u>C</u>**onductive
IV.	Collapsing Ear Canal	
V.	Presbycusis	
VI.	Cholesteatoma	
VII.	Otosclerosis	

2. An outer ear with a large, dark black spot might be a red flag for:
   a. Squamous cell cancer
   b. Ototoxicity
   c. Otitis Media
   d. NIHL

3. NIHL is characterized by:
   a. A flat conductive loss
   b. A reversible hearing loss
   c. A moderate sensorineural loss
   d. A precipitously sloping sensorineural loss

4. Presbycusis is a common disorder in: (select all that apply)
   a. Children
   b. Young adults
   c. Elderly men
   d. Elderly women

5. The normal eardrum or tympanic membrane:

   a. Is dark red in appearance

   b. Has a cone of light landmark

   c. Often emits a clear, odorless fluid

   d. Is a common cause of hearing loss

6. If someone walks into your office and reports vertigo, tinnitus in one ear and a very sudden hearing loss in one ear, you may have a case of: (select all that apply)

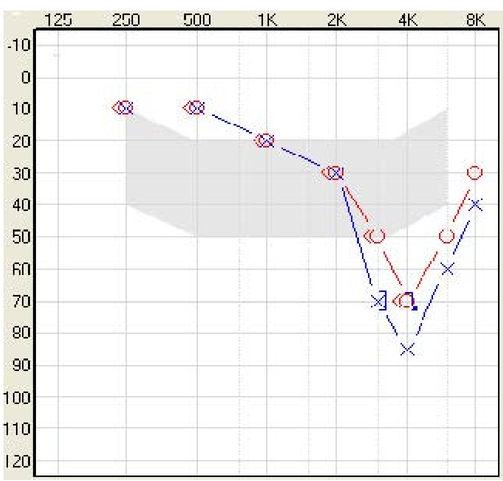
   a. Otitis Media

   b. Congenital Hearing Loss

   c. Meniere's Disease

   d. NIHL

7. The audiogram shown below is from a 45-year-old truck driver. He reports hearing loss of gradual onset. This audiogram is consistent with a diagnosis of:



   a. Otosclerosis

   b. NIHL

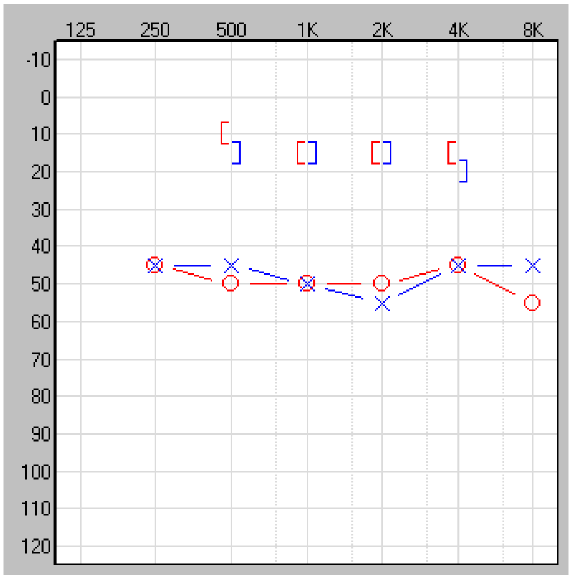
   c. Presbycusis

   d. Conductive

Speech Audiometry

Test	Right	Left	Binaural
UCL	95dB	95dB	
MCL	55dB	55dB	50/50dB
SRT	20dB	20dB	
Word Recognition	84%	76%	84%

8. A patient reports to you that she has had a cold for over two weeks, both ears are feeling stuffy and she cannot hear. Your audiogram shows the following results. What does the audiogram indicate and what should you do? Write your answer in the empty space to the right of the audiogram.

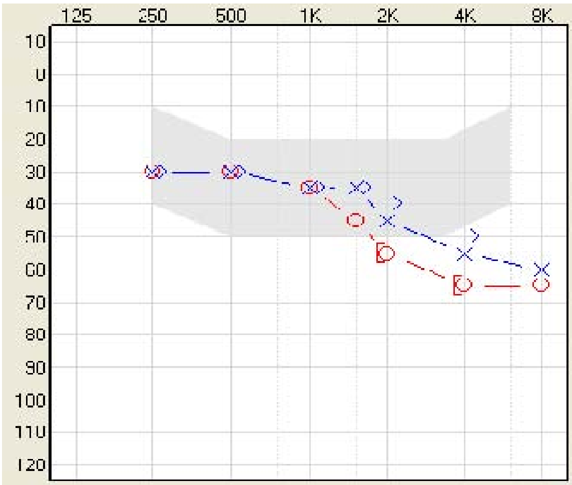


*insert earphones used

Speech Audiometry

Test	Right	Left	Binaural
UCL	>105dB	>105dB	
MCL	80dB	85dB	80/85dB
SRT	45 dB	55 dB	
Word Recognition	100%	100%	100%

9. Your patient is ninety-one years old, very healthy with no history of noise exposure. The audiogram looks like this. What is the likely cause of this hearing loss? Write your answer in the empty space to the right of the audiogram.



Speech Audiometry

Test	Right	Left	Binaural
UCL	85dB	80 dB	
MCL	65dB	60dB	65/65dB
SRT	45 dB	40 dB	
Word Recognition	80%	80%	80%

10. High doses of aspirin cause a temporary sensorineural hearing loss.

    a. True

    b. False

Unit 6: Basic Audiometry

Objectives

Notes

Overview

The time has come for you to start learning about the essential skills that you will need as a hearing care professional. This unit will focus on the audiometric tests that make up a complete hearing evaluation. Remember, you cannot help your patients to hear better until you have the full understanding of their hearing loss that you can only obtain by conducting a thorough evaluation. The information in this unit is designed to supplement the hands-on experience you will receive during the Basic Training Course.

Goals

Upon completion of this module, you will be able to:

- Describe all basic test procedures in the hearing evaluation battery
- Explain why each test needs to be completed
- Perform each test in an accurate and time-efficient manner

Follow Up

1. Do a complete battery of all the tests outlined in this manual on at least five people.
2. Audiology Clinical Protocols. By Robert Margolis. Protocol 1. pp. 5-16. Allyn and Bacon, 1997.
3. The Measurement of Hearing. By Ira Hirsch. Chapter 9. pp. 231-254.

Section 1: Introduction to Audiometric Testing

Notes

The following five core clinical skills must be mastered in order to become a successful hearing care professional. It will take time, practice and hard work to achieve this. Practice, Practice, Practice!

1. Otoscopy
2. Pure Tone Air Conduction Audiometry (including frequency-specific LDLs)
3. Pure Tone Bone Conduction Audiometry
4. Speech Audiometry (Including MCL, Speech UCL and Speech-in-Noise Testing and Quick SIN)
5. Masking

Technique Counts: Never Compromise It

The absolute importance of using a standardized technique for these five core skills cannot be emphasized enough. You will conduct four of these tests on virtually every patient you see in your office. (Masking is a technique you will have to apply with some of your patients and is discussed in a separate chapter.) In fact, you cannot complete the job of fitting hearing aids unless you do these tests professionally and accurately. The application of a consistent technique is important for a number of reasons, the most important being that using a consistent test technique allows you to be efficient and accurate. Accuracy and time sensitivity are crucial. The test techniques addressed below are all field-tested and accepted by licensed hearing care professionals and audiologists worldwide.

There is no substitute for direct hands-on learning of these core skills. At your Basic Training Course you will learn by actually doing these procedures over and over again with your colleagues and your instructor. Again, it is important to remember that practice will make you proficient.

Section 2: Otoscopy

Notes

In order to perform otoscopy you must first be familiar with the basic anatomy of the outer and middle ear. Take the time to review Unit 3: Anatomy and Physiology if you feel uncomfortable with this topic.

Purpose/Procedure

Otoscopy is the process of visually observing the ear canal and ear drum using the otoscope. The otoscopic examination is more than simply looking into the ear canal. Before even picking up the otoscope, it is important to carefully examine the outer ear (pinna) and the mastoid process behind the ear. Signs of previous ear surgery and other malformations should be noted. Remember to follow proper infection control protocols, outlined in the Unit 13 : Infection Control, each time you use your otoscope.

Equipment

The otoscope is like a magnifying flashlight. On the part of the otoscope that is placed into the ear there is a black cone-shaped object called a speculum. Specula come in different sizes. The size you use should correspond to the size of the patient's ear canal. If the patient is a child, a small-diameter speculum should be used; for large adult ear canals a larger size is necessary. Most patients require the standard size.

It is important that the otoscope have a strong light. Sanitation and patient safety are the other important concern at any time you are using equipment that will come in contact with your patient.

Instructions to the Patient

Prior to performing otoscopic inspection on your patient, prepare them by telling them a version of the following example in your own words:

"I'm going to use this light to look in to your ear. It might feel just a little uncomfortable, but it shouldn't hurt. Please hold real still while I take a look. I just need to see what the inside of your ear canal and the eardrum looks like."

After you complete this procedure briefly explain what you saw to the patient. For example, "Your ear canal is clear, it looks normal." Show them on the video otoscope what it looks like, or use a picture of a normal eardrum as a reference.

The Examination Process

1. Before you begin the examination process be sure to follow proper infection control procedures. These are outlined in Unit 13 of this manual for your reference.

2. After you have visually observed the outer ear and mastoid process, and noted the appearance, pick up the otoscope. Sanitize the specula with alcohol.

3. Turn on the otoscope and set it to the maximum brightness.

Notes

4. Place the largest size speculum appropriate for your patient's ear gently into the opening of the ear canal. Be sure to use the Bridge and Brace technique, which is shown to the right.

5. Hold the otoscope like a pencil with the lighted end about where the tip of the pencil would be. Face the light toward the patient's ear canal opening.

6. Place both hands (one still has the otoscope) up to the patient's ear along the patient's face. It is important to use both hands. This technique, called bracing and bridging, will prevent the speculum from scraping the ear canal if the patient moves suddenly*. It is important that you are on the same level as the patient.* If the patient is seated, you should be seated right next to him/her, straddling his/her chair, if possible. Get as close as possible. Carefully look into the patient's ear canal.

7. The posterior (back) portion of the pinna should be gently pulled up and back in order to straighten out the ear canal. While you are pulling back on the pinna with one hand, place the tip of the otoscope into the open ear canal. Look into the ear.

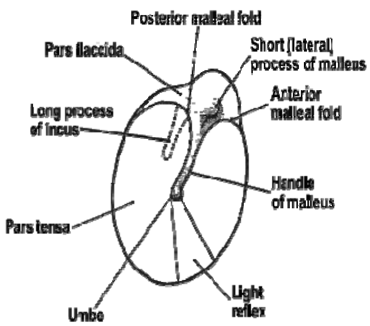
8. The first thing that should be noted is the condition of the external ear canal, including excessive amounts of cerumen. The normal ear canal should be smooth and pinkish in appearance. Scratches, blood, redness, wetness are all signs of an abnormality.

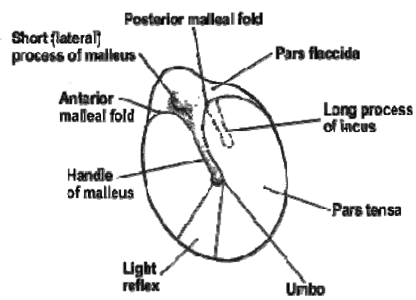
9. At the end of the canal lies the **tympanic membrane (TM)**, commonly called the **ear drum**. Both terms are acceptable. The ear drum should be a light gray color and should be shiny. The **cone of light**, your most important visual landmark, should be clearly visible in the lower/front (inferior/anterior) quadrant of each ear drum. The condition of the ear drum should always be noted on your history form.

Below are the views of normal right and left TMs.

Normal Tympanic Membrane

Right Ear **Left Ear**





Hands-On Exercise

Look into the ear canals of five different people. Use the same process listed above for each person. Don't forget to practice good infection/sanitation control techniques. Wash your hands and use a clean speculum with each new person you examine. A clean speculum should be used between looking into each ear to prevent cross contamination. When you are finished looking into the ear canals, write down what was observed. Note the color and appearance of the pinna, mastoid process, ear canal and TM. Use the pictures in this section as a guide.

Notes

Check Your Understanding

What landmark of the tympanic membrane should be visible in the inferior/anterior quadrant of the eardrum?

a. Annulus

b. Oval window

c. Cone of light

d. Umbo

Section 3: Introduction to the Hearing Test Battery

Notes

Well, it's not really a battery like you would find in your car, but when we do a group of tests we call it a battery. When you test someone's hearing in order to get a complete picture of the way the ear works, you must do a series of tests. The tests in this series complement each other. In a short while, when you have fully mastered audiometric interpretation, you will be able to look at a hearing test battery and explain the degree of the loss, the type of hearing loss and appropriate amplification. Right now let's learn how to do the hearing test battery.

Pure Tone Air Conduction Audiometry

You should already know that air conduction threshold testing is the standard hearing test performed during a hearing evaluation. Virtually every patient that you see will require a hearing evaluation. Review Unit 4: Introduction to Acoustics, if you are unfamiliar with the concepts of intensity (loudness), frequency (pitch), air conduction, bone conduction and threshold. A solid foundational knowledge of these concepts is vital to comprehension of this section.

Equipment Preparation

The equipment required to perform pure tone air conduction audiometry is an annually calibrated audiometer with calibrated headphones or insert earphones. When an audiometer is calibrated it can only be calibrated to one transducer, i.e. headphones or inserts. These are not interchangeable.

1. Identify the following parts of the audiometer

    a. Power switch

    b. Output: Right and left ear

    c. Input: Pure tone

    d. Frequency selector: 250 to 8000 Hz

    e. Hearing Level (HL) dial: 10 to 120 dB HL

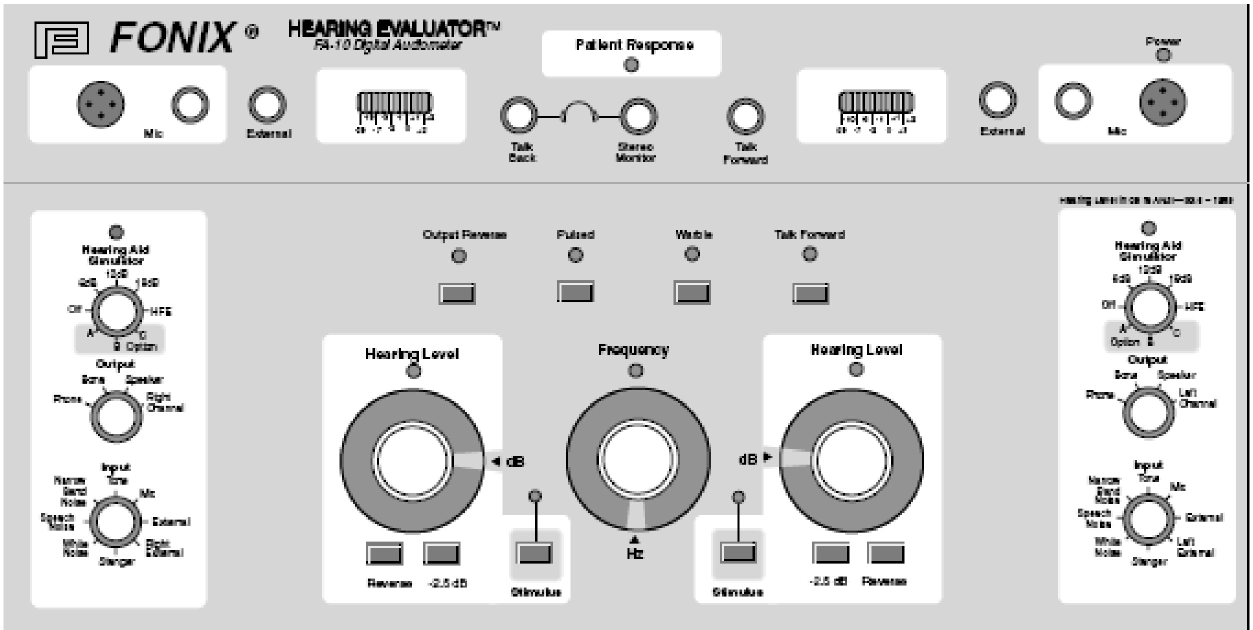
    f. Tone presentation bar

    g. Talk-over system: Allows the tester to talk to the patient through the earphones

    h. Talk-back system: Allows tester to hear the patient

Below is an image of the front panel of a Fonix FA-10 audiometer:

Notes



2. Turn on the audiometer

3. Perform a daily biological listening check on the audiometer at the start of each day. Do this by setting the audiometer to a 40 or 50dB HL tone and sweeping through all the frequencies. Then turn to a 1000 Hz tone and sweep through all the intensities (0 to 100dB HL), being careful not to exceed your tolerance level for loud sounds. Do this for both earphones and bone oscillator. Listen for intermittency and distortion. Wiggle the wires to see if a short occurs. This is a biological calibration and should be performed every time the equipment is turned on.

4. Set Channel 1 to tone and to the desired ear. Begin with either the right ear or the better ear.

5. Set the frequency selector to 1000 Hz and the tone presentation to 50 dB.

6. Ensure that both the talk-over and talk-back are set at the appropriate loudness.

Instructions to the Patient

Begin by addressing the patient by saying the following in your own words:

"After I place these phones over your ears, please listen for the beeping sounds and press the button (raise hand) each time you think you hear the tone. Listen carefully because it will sound as if the sounds are way off in the distance. I will begin the test in your right ear. Do you have any questions?"

Procedure

1. The patient should be seated so he cannot see you or the dials of the audiometer.

2. The response mode depends on the individual being tested. Either ask the patient to raise his hand when he hears the tone, or push a button when he hears the tone.

3. Place the headphones on the patient. Hair should be pushed away from the opening of the ear canals. Glasses, hearing aids, and earrings must be removed before placing the headphones on. The red earphone is placed on the right ear and the blue ear phone is placed on the left ear. The diaphragm of each earphone is placed directly over the ear canal of both ears. Finally, the headband is tightened to ensure that the earphones do not move. Below are images of TDH-39 headphones (left) and insert earphones.

4. Begin at 1000 Hz and 50 dB in the better ear, or the right ear if there is not a preference. This starting place is used because it is important that the first tone can be heard at a comfortable level.

5. Present the first tone. The duration of each tone presentation should be between 1 and 2 seconds.

6. It is extremely important to use random time intervals between tone presentations in order to prevent a pattern that might cue the patient to respond.

7. If there is no response at the starting level of 50dB raise the intensity +10dB until the patient responds to the presentation of the tone.

Notes

Notes

8. As soon as a response is elicited, the intensity of the tone is reduced (made softer) in 10 dB steps, until no response is given. At this point you may assume that the level of the tone is below the patient's threshold, and threshold bracketing may begin.

9. If no response is given the intensity of the tone is increased (made louder) in 5 dB steps, until a response is again observed.

10. As soon as a response is obtained, the intensity is reduced (made softer) by 10 dB.

11. If a response is not obtained, the intensity is increased in 5 dB steps until you see a response.

12. This procedure is repeated (reduce by 10, increase by 5). The lowest intensity level at which at least 2 out of 3 presentations produce responses is considered to be the threshold.

13. The threshold is recorded on the audiogram with a blue "X" for the left ear or a red "O" for the right ear. Record the patient's responses as neatly on the audiogram as possible at the appropriate frequency. Because 1000 Hz will be re-tested, for the first ear only place a dot on the audiogram, after reconfirming the threshold place the appropriate symbol.

14. After obtaining the threshold on one ear at 1000 Hz, continue testing the same ear and test in ascending order (2000 Hz, 3000 Hz, 4000 Hz, 6000 Hz, 8000 Hz). An optional technique would be, after the initial threshold at 1000 Hz, next test 500 and 250Hz, followed by a re-test of 1000 Hz, and progressing then to the higher frequencies. Please note that in some states, this is considered incorrect and the "upward first" technique is preferred.

15. It is required to test inter-octave frequencies (750, 1500, 3000 and 6000 Hz) when the difference in threshold between two adjacent octave frequencies is 20 dB or greater.

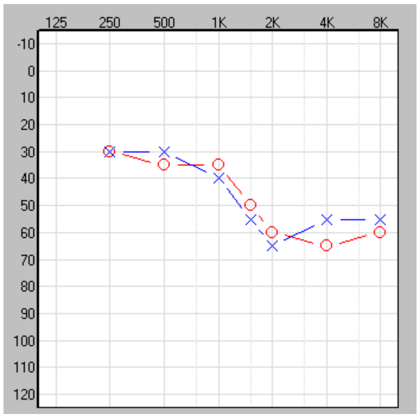
16. After testing at 8000 Hz, go back and recheck the threshold at 1000 Hz. This second threshold should be the same as the first one. If this is not the case, the reliability of the test is in question and the patient must be re-instructed and all previously measured thresholds must be re-tested.

17. After rechecking 1000 Hz, 500 Hz is tested followed by 250 Hz.

18. The other ear is now tested using the identical procedure.

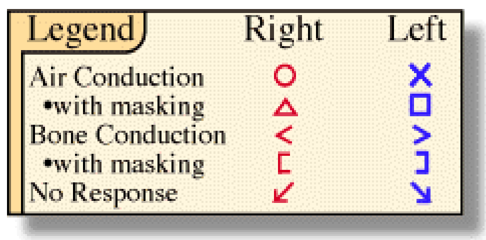
Notes

Below you will see a completed pure-tone air-conduction audiogram. Note the threshold obtained at 1500 Hz due to the difference in threshold between 1000 and 2000 Hz.



As stated previously, record the responses on the audiogram using a blue **X** for the left ear and a red **O** for the right ear.

Refer to the key at the bottom of the audiogram form. This gives you more information about what other symbols you might have to use. For example, if the patient does not respond to any tones, they are recorded as no response (NR). See the example at right for most of the commonly-used symbols.



Interpretation

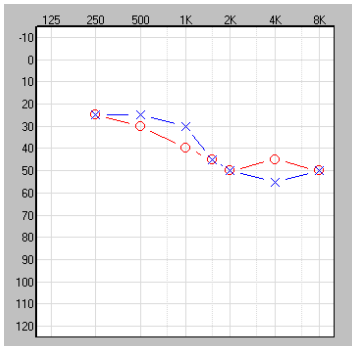
The air conduction pure tone thresholds tell you how MUCH hearing loss a person has. In simple terms, the amount or degree of hearing loss is the difference (in decibels) between 0 dB HL and the patient's threshold at each frequency.

The following are considered to be the degrees of hearing loss:

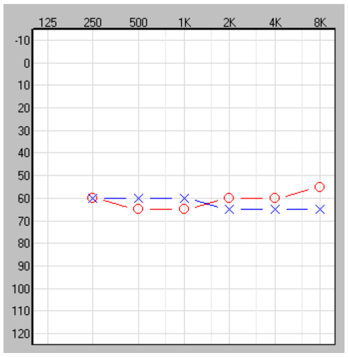
- 10 dB to 25 dB = Normal hearing
- 26 dB to 40 dB = Mild hearing loss
- 41 dB to 55 dB = Moderate hearing loss
- 56 dB to 70 dB = Moderately Severe hearing loss
- 71 dB to 90 dB = Severe hearing loss
- 91 dB or greater = Profound hearing loss

Notes

Because different frequency areas of the audiogram usually will have different degrees of hearing loss, hearing loss is typically described in terms of severity in the low frequencies, **to** severity in the high frequencies. Below you will see an audiogram that represents a bilateral mild to moderate hearing loss.



Here is a bilateral flat moderately severe hearing loss:



Pure Tone Average (PTA)

The average air conduction hearing loss is calculated using the **3-Frequency Pure Tone Average (***abbreviated* ***PTA*****).** To calculate the PTA, add the thresholds obtained at 500, 1000 and 2000 divide the sum by 3. This is the PTA for each ear. Use the PTA and the degree of hearing loss chart to summarize the amount of hearing loss for each ear. The audiogram interpretation section will elaborate on this.

Check Your Understanding

Notes

When performing a pure-tone air conduction threshold search, when the patient indicates he or she heard the tone, what do you do?

a) Increase the intensity of the tone by 5 dB

b) Decrease the intensity of the tone by 10 dB

c) Decrease the intensity of the tone by 5 dB

d) Increase the intensity of the tone by 10 dB

Frequency-Specific Loudness Discomfort Levels (LDL)

It is necessary to obtain and to document frequency-specific loudness discomfort levels (LDL - sometimes called threshold of discomfort or TD) in order that compression characteristics of hearing aids be set properly. Speech uncomfortable levels (UCL), discussed in the section on speech audiometry, are used for the same purpose. At a minimum, LDL/TDs should be obtained at 500 and 2000 Hz. When fitting hearing aids with multiple channels of output adjustment, it is best to obtain LDL/TDs at 500, 1000, 2000 and 4000 Hz.

Equipment Preparation

Obtain the LDL/TD at 500 Hz first. Output should be set to 'headphone'. The input will be 'pure tone'. Because the intensity of the signal will become somewhat loud, a straight pure tone may be uncomfortably piercing in quality. In order to reduce this quality, it may be a good idea to have the pure tone modified to "frequency modulation", sometimes called a "warble tone". Another option is to use narrow-band noise as the test signal.

When testing loudness levels, whether pure-tone or speech, it is important to orient the patient so that you can see his or her face, for an indication when the sound is becoming uncomfortably loud. In the first set of instructions below, the chart that is referenced is a "Categories of Loudness Chart" which may also be used in Most Comfortable Level and Speech UCL testing, described a bit later. A copy of this document in a format that may be used by the patient may be found at the end of this chapter. The categories of loudness are as follows:

7 – Uncomfortably loud

6 – Loud but OK

5 – Comfortable (slightly loud)

4 – Comfortable

3 – Comfortable (slightly soft)

2 – Soft

1 – Very soft

Notes

Instructions to Patient for Frequency-specific LDL/TD

"You will be listening to some tones that will gradually increase in loudness. Use the chart as a guide to tell me when the tone is too loud to listen to for more than a second. In this case we are determining how loud sounds have to be for you to judge them "Uncomfortably Loud". You see there is a level called "Loud but OK". If it is loud but you can tolerate it, we have not yet gotten to "Uncomfortably Loud". As soon as you tell me the sound is "Uncomfortably Loud", I will immediately turn it down. Do you have any questions?"

Alternative Instructions to Patient for Tone LDL/TD

Use the same instructions as above, or, as an alternative, the following:

"You will listen to some tones as they get louder. When they become loud raise your finger. When they become uncomfortably loud, but not painful, raise your hand, I will stop at that point"

Procedure

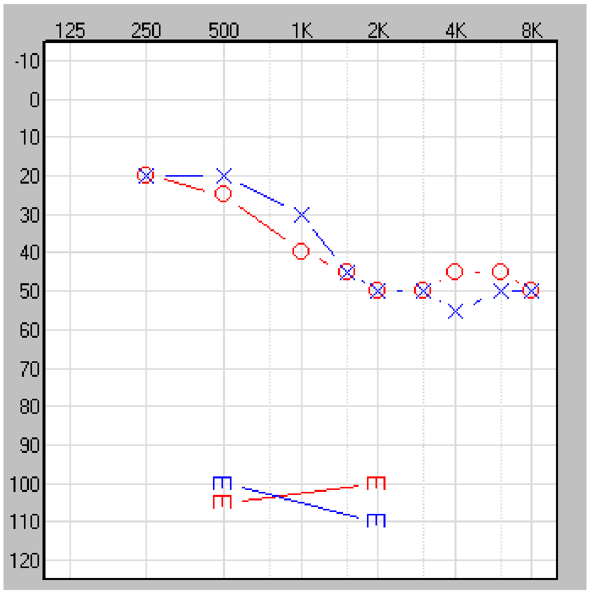
1. Begin presenting the pure tone or narrow-band noise 20 dB louder than the threshold at the frequency tested (20 dB SL) and increase the intensity of the audiometer every 2-3 seconds until the patient tells you the signal has reached a level that is too loud to listen to (#7 on the chart).

2. Immediately reduce the intensity to MCL and repeat.

3. Record the highest-intensity result for each ear on the audiogram. See below for an example of an audiogram with pure-tone thresholds and frequency-specific LDL/TDs at 500 and 2000 Hz.

Interpretation

The result of the pure-tone LDL determines the Residual Area of Processing for you patient: Frequency-Specific LDL/TD minus PureTone Threshold = Residual Area of Processing.



Pure-Tone Audiometry: Bone Conduction

Notes

Bone conduction (BC) audiometry is always done after you have completed air conduction (AC) audiometry. BC audiometry will determine if the patient has a sensorineural, conductive or mixed hearing loss.

Equipment Preparation

1. Turn on the audiometer
2. Set channel 1 to “tone” and “bone conduction”
3. Select 1000 Hz
4. Set the hearing level dial to the appropriate intensity (+20 dB HL above the AC threshold frequency specific)
5. BC testing is done for all octaves between 250 to 4000 Hz and for intensities between 10 to 70dB HL. (except 250 Hz where the limit is 45 dB)

Instructions to the Patient

Begin by addressing the patient by saying the following in your own words:

“You are going to hear more tones. This time you will hear them through this headset I have placed behind your ear. You may hear the sounds in either ear. Simply press the response button or raise your hand when you hear the tone. Listen for the softest sounds. Do you have any questions?”

Procedure

1. The patient is seated with their back to the audiometer—just as for AC testing.
2. The bone oscillator is placed, concave side down, on the patient’s head. In most cases the oscillator is placed on the mastoid bone behind the pinna. The oscillator’s concave surface should rest flat on the mastoid bone. Headphone or insert must be removed from the ear being tested.

3. Begin the test at 1000Hz, +20 dB above air conduction threshold (20 dB SL) at the frequency being tested.
4. Stimulus duration should be between one and two seconds
5. Keep in mind that if there is no response it may be a medical referral.

6. As soon as the patient responds, reduce the intensity in 10dB steps. The process of obtaining threshold has begun.

7. The intensity of the tone is raised in 5 dB steps until another response is seen. As soon as a response is obtained, the intensity is lowered by 10 dB.

8. If a response is not seen, the intensity is increased in 5dB steps until a response is seen. The procedure starts again “quieter 10 dB, louder 5 dB”

9. The lowest level where 2 responses are seen in 3 stimulations is considered to be the patient’s threshold. These must be ascending responses.

10. Record the threshold on the audiogram in the appropriate place. Use the key on the audiometer to determine the appropriate symbol to use. Right < and Left >

11. Thresholds are then obtained at other frequencies: 2K Hz, 4K Hz, 500 Hz and 250 Hz

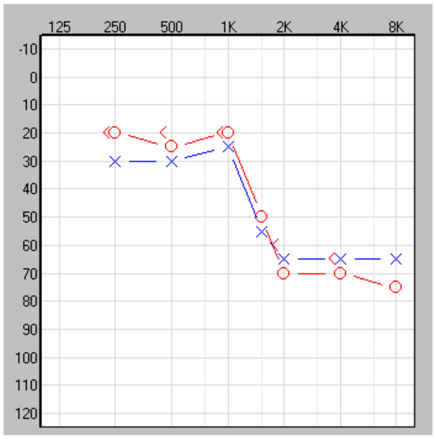
12. Move the oscillator to the opposite ear, and complete the same threshold procedure.

13. Determine the need to use effective masking.

Notes

Notes

Interpretation



- The difference between 0 dB HL and the BC threshold is the amount of hearing loss at each frequency that is sensorineural.
- The difference between the AC threshold and the BC threshold is called the Air/Bone Gap. This is the amount of conductive hearing loss at each frequency.
- Although bone conduction thresholds should theoretically never be poorer than air conduction thresholds, this is likely to occur occasionally. If the BC is poorer by 5 dB, document the threshold. If BC is poorer by more than 5 dB, move the oscillator and/or check your equipment to obtain a threshold closer to the AC.

Below you see a completed pure-tone audiogram with air and unmasked bone conduction thresholds, conducted with the bone oscillator behind the right ear.

Check Your Understanding

You should begin searching for a bone conduction threshold at what intensity?

a. At the same intensity as the air conduction threshold

b. 10 dB louder than the air conduction threshold

c. 0 dB HL

d. 20 dB louder than the air conduction threshold

Notes

Speech Audiometry

Since the most important signal human beings hear is speech, we must evaluate how our patients decode speech signals in several different ways. We evaluate the softest speech our patients can respond to (Speech Reception Threshold - SRT), what is a comfortable loudness for speech (Most Comfortable Level - MCL), how well they are able to understand individual words presented at a comfortable loudness (Word Recognition - WR), the loudest speech they can tolerate (Uncomfortable Level - UCL) and how well they understand speech in the presence of noise (QuickSIN). Below is an image of the area of the audiogram on which results of these tests are documented.

	Ear	**Left**	**Right**	**Binaural**
Speech	SRT	dB	dB	dB
	MCL	dB	dB	dB
	UCL	dB	dB	dB
	Word % Recognition	%	%	%
	QuickSIN	dB SNR	dB SNR	dB SNR

Speech Reception Threshold (SRT)

Purpose

The main purpose of the speechreception threshold (SRT) procedure is to check the reliability of the pure-tone audiogram thresholds. The SRT should be within +/¬- 10 dB of the 3-frequency pure-tone average for each ear. After this exercise, you should be able to determine an SRT and make judgments as to the reliability of pure-tone threshold results. If PTA and SRT do not agree, there is a chance that the patient is feigning hearing loss.

The stimuli to be used for SRT testing are **spondee words**, which are two syllable words that have equal stress on both syllables. Both syllables should peak at 0 on the VU meter of the audiometer. A list of spondees may be found at the end of this section.

The SRT is often the first test of the audiologic battery. When it is completed first, there is no bias in SRT testing, which can occur if pure-tone thresholds are already known, and it can be used as a reliability check, as described earlier. Speech testing can be delivered via CD, or by monitored live voice (MLV). MLV is the use of a microphone with careful visual attention paid to the volume units (VU) meter of the audiometer. CD testing for speech produces the most reliable results.

Notes

Equipment Preparation

1. Before testing the patient the level of the pre-amplifier for the VU meter must be calibrated.

2. If MLV (microphone live voice) is to be used the audiometer's input is set to "microphone". The microphone level knob is adjusted while presenting the spondees, until both syllables peak at 0 on the VU meter.

3. A calibration tone is provided on pre-recorded (CD or tape) materials that can be played while the "external" attenuator is adjusted until the VU meter settles at 0 dB.

4. Set the test channel to the "microphone" position for MLV, or to "disc", "tape" or "external" for prerecorded materials.

5. Set the output to the appropriate earphone.

6. The patient should not be allowed to see the tester's face.

Instructions to the Patient

The patient must be seated so that movements of the hearing care professional are not visible, especially if monitored live voice is used. Speech-reading the stimulus words can produce test results that suggest that the patient's hearing is better than it truly is. Tell the patient the following in your own words:

"You are going to hear some words. Repeat every word you hear. The words will get softer and softer, so soft, in fact that they will be very difficult for you to hear. It is very important that you try to repeat the words, even if you have to guess. Do you have any questions?"

Procedure

1. An important part of the SRT test is the initial patient familiarization with the test words. One way to familiarize the patient is to use live voice. The spondees are presented through the microphone/speech channel or through the talk-over channel on the audiometer at a comfortable loudness level (60 dB HL or louder). This method allows you to verify that the patient understands all the spondees. If the patient cannot repeat a given spondee at a comfortable loudness level, then that spondee should not be used to determine the SRT.

2. The starting level is + 20 dB HL above the PTA.

3. One spondee is presented.

Notes

4. If the patient repeats the word correctly, the hearing level is reduced by 10 dB steps while presenting one word at each level. This procedure is continued until a spondee is missed. .

5. When the patient misses the first spondee, the threshold determination procedure begins. At this point, the level is increased in 5 dB steps, presenting one spondee at each level, until the patient is able to repeat the spondee.

6. The procedure is the same “reduce by 10 dB, increase by 5 dB” one used for pure-tone threshold determination. Each time the patient gets a word correct, the hearing level is decreased 10 dB, and another spondee is presented. Every time the patient does not respond correctly at a given level, the level of the word is increased by 5 dB and another spondee is presented.

7. When a level is reached where at least 2 out of 3 of the spondees are repeated correctly ascending, the procedure is terminated, and the SRT is recorded.

8. There is a place on the audiometric worksheet for the SRT to be recorded in dB HL monaurally.

Interpretation

- The SRT should be within +/-10 dB of the pure-tone average, unless there is some unusual shape of the audiogram (e.g. sharply falling or sharply rising configurations).

- The difference (in decibels) between 0 dB HL and the patient’s SRT is the amount of hearing loss for speech.

- For the patient who has difficulty performing SRT testing, even following re-instruction, alternative test protocols are described in the section regarding Difficult-to-Test Patients.

Below is a list of spondee words.

airplane	eardrum	iceberg	railroad
armchair	farewell	inkwell	schoolboy
baseball	grandson	mousetrap	sidewalk
birthday	greyhound	mushroom	stairway
cowboy	hardware	northwest	sunset
daybreak	headlight	oatmeal	toothbrush
doormat	horseshoe	padlock	whitewash
drawbridge	hotdog	pancake	workshop
duckpond	hothouse	playground	woodwork

Check Your Understanding

Which two of the following are characteristics of spondee words?

a. Two-syllables

b. One syllable

c. Use a carrier phrase

d. Equal stress on both syllables

Most Comfortable Level (MCL)

Purpose

A Most Comfortable Level (MCL) is used as -the initial intensity at which Word Recognition testing is administered.

Equipment Preparation

1. The MCL is obtained in each ear separately under headphones

2. The microphone output is used for MLV presentation or tape output for CD presentation.

3. A Categories of Loudness Scale is given to the patient. (Provided at the end of this chapter).

Instructions to the Patient

Address the patient by paraphrasing the following in your own words and based on the materials/equipment chosen.

(If using a CD) "You will hear a male speaker talking about rainbows. As you listen to the speech, the loudness level will vary. As you listen, point to the level as you judge it on the chart I just gave you. When the speech is at a level that is the most comfortable, at a level at which you could listen all day without struggling or discomfort, point to Comfortable - #4 on the scale. Do you have any questions?"

Procedure

You may use a passage of cold running speech from a CD, commonly called the Rainbow Passage or some other continuous discourse. You may also use MLV.

1. Begin the test at SRT + 20 dB and every 5 seconds as the passage plays increase the intensity in 5 dB steps until the patient points to the number 4 on the chart. Increase the intensity 5 dB beyond their initial indication and ask if this is better or worse. Document their final selection.

Notes

Notes

2. Report this procedure if the MCL is the same stop. If it varies repeat the procedure a third time and take the lowest score.

3. Record the results for each ear on the audiogram.

4. This test needs to be done monaural (each ear) and binaural (both ears together).

Speech Audiometry: Word Recognition (WR)

Purpose

Word recognition (WR) testing is the first supra-threshold test in the audiometric evaluation. Supra-threshold means the test is done at a level louder than threshold. It is typically performed at a presentation level that is comfortable to the patient. The purpose of the test is to evaluate an individual's ability to recognize single syllable words from a phonetically balanced (PB) word list. It is critical that WR testing be conducted at a level in which the words are both comfortable and loud enough to hear and understand as well as possible. Start the test at MCL. WR testing is usually conducted at MCL or 5 to 10 dB louder.

Two of the most popular tests for this purpose are the NU-6 and CID W-22 word lists. They are 50 words in length, and either a full, 50-word list is presented (with a value of 2 percent per word), or a half, 25 word list is presented (with a value of 4 percent per word). Most clinicians use the 25 word list, but *no fewer than 25 words should be presented.*

1. The input is set to "tape" or "CD". CD is used whenever possible, because it provides control over dialectal differences and therefore gives us assurance of a consistent level and manner of presentation. If pre-recorded materials are not available, or if there is some other special need, MLV is used, and the audiometer is set to "microphone". The output is set to the test ear.

2. All pre-recorded tests of PB word lists contain a calibration tone. While the tone is playing, the level control for the tape or CD is adjusted to the point where the needle on the VU meter reads "0". This should be completed before the test begins. If pre-recorded materials are used, the CD or tape should be advanced so that the introduction to the test is not heard by the patient.

3. The speaker must have good microphone technique for MLV testing, and be experienced at using the audiometer's VU meter. The microphone should be approximately 6 inches from the speaker's mouth. The carrier phrase "Say the word______" is uttered so that the last word of the phrase strikes 0 VU, and the subsequent test word is uttered with normal stress.

4. The level of presentation for the words in the test ear is set at +20 dB above the SRT (20 dB SL) to begin. If the patient misses words (i.e. 2 or 3 out of 5) increase the presentation level 5 or 10 dB. This is so the patient has the best chance of doing as well as possible on the WR.

Notes

5. The WR should always be done monaurally and then binaurally (25 words minimum each ear). The reason for this is to determine a central issue. The rule is: The binaural score should be as good as or better than the better monaural score. When this is not the case it is called binaural degradation and there is a possibility that the problem is processing in the brain or a central issue.

Instructions to the Patient

The patient should not be allowed to watch the hearing care professional, especially if monitored live voice testing is used. A written response may be substituted for a verbal one, if desired.

"You are going to hear some sentences. Please repeat the last word in each sentence. For example, if you hear "Say the word BOY", just repeat the word "BOY". If you are unsure of a word, say whatever you think you heard. Do you have any questions?"

Procedure

1. Pre-recorded tests are preferred, but MLV may also be used. Word-recognition testing is always done with a carrier phrase. The last word of the carrier phrase should peak at "0" on the VU meter and the test word follows using natural, sentence-like stress. The traditional carrier phrase is: "Say the word______". This phrase must be used before every word in the list.

2. A count should be kept of the number of words the patient misses.

Interpretation

- The type of word list used (NU6 or CID W22) and the intensity at which the list was presented are always recorded on the audiometric worksheet.

- The Word Recognition Score (WRS) is calculated as follows: the number of words missed is counted, multiplied by 2 percent if all 50 words are given, or multiplied by 4 if only 25 words are used, and subtracted from 100 percent. This number is recorded in the appropriate box on the audiometric worksheet.

Check Your Understanding

How many words are in a full PB word list?

a. 10

b. 25

c. 100

d. 50

Uncomfortable Loudness Level (UCL) Speech

Notes

Purpose

The UCL speech is the intensity level at which speech becomes too loud to listen to. While the UCL has traditionally been used in the selection and fitting of hearing aids, with current digital technology, frequency-specific LDL/TDs are preferred.

Equipment Preparation

The steps to prepare your equipment for UCL testing are the same as those used when preparing for MCL testing with the exception of the direction in which the patient faces. Whether you are performing speech or tone testing the patient will need to be turned around so that he/she is facing you. This way you can watch their facial expressions for discomfort.

Instructions to Patient for UCL Speech

"You will be listening to some speech that gradually increases in loudness. Use the chart as a guide to tell me when the speech is too loud to listen to for more than a second. In this case we are determining how loud sounds have to be for you to judge them "Uncomfortably Loud". You see there is a level called "Loud but OK". If it is loud but you can tolerate it, we have not yet gotten to "Uncomfortably Loud". As soon as you tell me the sound is "Uncomfortably Loud", I will immediately turn it down. Do you have any questions?"

Alternative Instructions to Patient for speech UCL

Use the same instructions as above, or, as an alternative, the following:

"You will listen to speech as it gets louder. When they become loud raise your finger. When they become uncomfortably loud, but not painful raise your hand, I will stop at that point"

Procedure for Speech UCL

You may use either the recorded Rainbow Passage or cold running speech via MLV.

1. Start the presentation at MCL and increase the intensity of the audiometer every 2-3 seconds until the patient tells you the signal has reached a level that is too loud to listen to (#7 on the chart).

2. Immediately reduce the intensity to MCL *and repeat*.

3. Record the highest intensity result for each ear on the audiogram. This test may be administered binaurally.

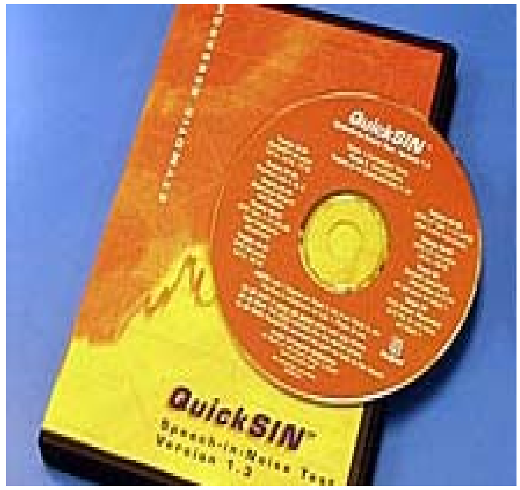
Notes

Interpretation

- The result of the speech UCL will determine dynamic range (DR) -
- UCL-SRT=DR.

QuickSIN

Purpose



QuickSIN is a test of your patients' ability to understand speech in noise. This is an important metric to obtain for two reasons. First, one of the most common complaints our patients have is that they have difficulty understanding speech in noise. And second, since the advent of fully digital hearing aids, directional microphones (meant to help the user to better understand speech in noise) have become a nearly universal feature.

Procedure

QuickSIN may be administered under headphones binaurally, or via sound field speakers. The test should be presented at an intensity somewhat louder than MCL – usually 70-80 dB HL. We present the sentences louder than MCL so that we may determine if audibility will be sufficient for the patient to understand speech in noise. If so, directional microphones may not be necessary.

Instructions for Patient

"You will hear a female speaker saying a series of sentences with people talking in the background. Just ignore the talking in the background. The sentences will be like 'The sun came up to light the eastern sky', or 'Take shelter in this tent but keep still'. With each sentence, the noise in the background will get a little louder. In the last sentence, the speech and the noise will be at the same level, which will be very difficult, no matter how good your hearing is. Just repeat each sentence out loud as accurately as possible. If you are not sure of a word, go ahead and guess. If you only get a word or two, just tell me the words you heard. I will be presenting three sets of six sentences. The first set will be practice and the second and third will be scored. Do you have any questions?"

Proceed by presenting Track 12 (List 10) for practice, and then Tracks 13 and 14 (Lists 11 and 12) for scoring. (Lists provided at end of Chapter)

Notes

Scoring QuickSIN

Keep track of the number of underlined words that are repeated *correctly* in each scored set and subtract that number from the number 25.5. Average the results of the two scored sets, and the result is the Signal-to-Noise-Ratio Loss (SNR loss). The SNR loss score represents the signal-to-noise-ratio required for the patient to understand 50% of the target words.

Here is an example of the practical implications of a mild/moderate SNR loss score. Assume the patient scores a 10 dB SNR loss. This means that, when the background noise in the environment is 75 dB, in order for the patient to understand 50% of the words said by the person s/he is talking to, that person would have to speak at an intensity of 85 dB. This would be a difficult task to maintain for any period of time.

Interpretation

- 0-4 dB SNR loss – Normal – Audibility is sufficient and directional microphones may not be necessary.
- 5-15 dB SNR loss – Mild/Moderate difficulty in noise– Directional microphones are indicated.
- >15 dB SNR loss – Severe difficulty in noise – Directional microphones may not be effective. Use of assistive devices and counseling regarding effective communication strategies are indicated.

Site-of-Lesion Tests

Notes

There are several audiometric tests that are not typically administered by hearing aid dispensers, but of which all hearing professionals should have some awareness. These are sometimes called "special audiometric tests" or "site-of-lesion" tests and are for the most part obsolete due to improved technology. However, they are occasionally referenced in licensure exams.

These tests are designed to identify hearing losses that originate beyond the outer, middle, and inner ear. Since the locations or site-of-lesion of disorders identified by these tests are beyond the cochlea, these disorders are called **retrocochlear**. Acoustic neuroma, discussed in Unit 5, is an example of a retrocochlear disorder.

The three most common audiometric site-of-lesion tests are **tone decay**, **acoustic reflex decay**, and **speech rollover** or **PIPB function**.

Tone decay is a test that determines whether very soft pure-tones remain audible. If a steady pure-tone becomes inaudible to the patient (**positive tone decay**), this may be a sign of a compromised auditory nerve.

Acoustic reflex decay is one of the tests that may be administered when doing tympanometry (a measurement of the movement of the structures of the middle ear). The acoustic reflex should be able to "hold" for 10 seconds when a loud sound is present. If it cannot maintain its hold for 10 seconds, this is considered a **positive acoustic reflex decay**.

Speech rollover or **positive PIPB function** is related to the Word Recognition test. In a normal auditory system, once a person establishes the intensity at which they obtain their maximum word recognition score, that score should not decrease with increased presentation intensity. If word recognition scores do decrease at increased intensities, this is considered "**speech rollover**".

If results of any of these tests are positive, then medical referral is required in order to rule out a retrocochlear lesion.

Check Your Understanding

QuickSIN can tell us the potential effectiveness of what hearing aid feature?

a. Directional microphone

b. Digital noise reduction

c. Digital feedback suppression

d. Compression kneepoint

Notes

Tympanometry

Tympanometry is a test that is routinely performed by audiologists and seldom administered by hearing aid dispensers. However, since a basic knowledge of tympanometry is required for many licensure exams, a brief description is included here.

Tympanometry is a test of middle ear function. It is designed to tell the hearing care provider whether the middle ear is functioning properly, and if not, what is likely to be causing the problem. In tympanometry, an airtight seal is made at the entrance to the ear canal. Positive and negative air pressure is presented to the ear canal so that measurements of the movement of the ear drum can be obtained. These movements are then represented graphically as a *tympanogram.*

There are three important parameters that are measured in tympanometry.

1. **Ear canal volume**: This is the volume of the space from the aperture of the ear canal to the ear drum. Normal adult ear canal volume ranges 0.6-1.5 cubic centimeters (cc).

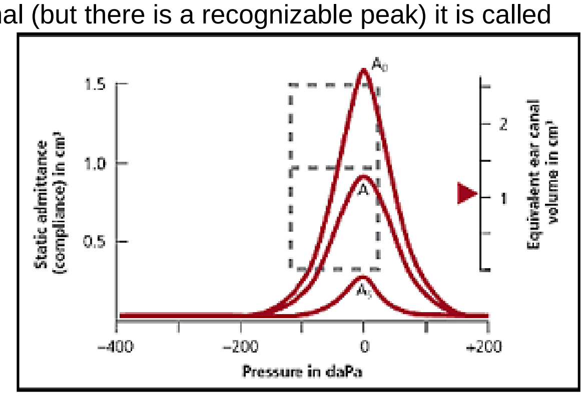
2. **Static Compliance**: This is a measure of how freely the eardrum moves. The normal range is 0.3-1.4 cc.

3. **Middle ear Pressure:** This measures the pressure of air in the middle ear space as compared to the pressure of air outside the middle ear. The range of pressures considered normal are from -150 to +100 millimeters of water (mmH2O) or decaPascals (dPa).

The vertical axis of a tympanogram represents static compliance and the horizontal axis represents middle ear pressure. Tympanogram shapes are labeled by letter - Types A, B and C.

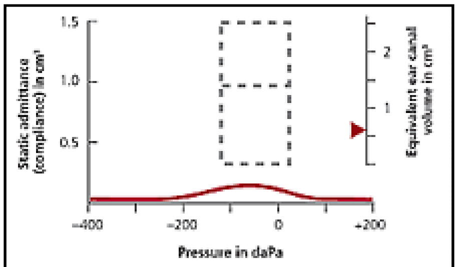
Type A Tympanogram

Type A tympanograms are characterized by a recognizable peak at or near “0”. If the peak occurs within the normal compliance range, it is considered a plain Type A. Type A tympanograms occur with normal middle ear function and would be expected when hearing is within normal limits and when there is a sensorineural hearing loss. If the compliance is lower than normal (but there is a recognizable peak) it is called Type As (‘s’ for shallow). This could mean the ossicles are fixed, perhaps by otosclerosis. If, on the other hand, the compliance is larger than normal, the tympanogram would be called Type Ad (‘d’ for deep), which could mean the ossicles are dislocated. To the right you will see a representation of all three Type A tympanograms. The dashed-line box represents the normal range.



Notes

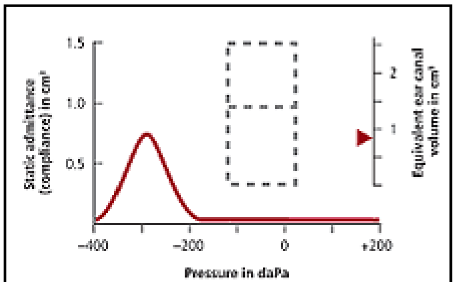
Type B Tympanogram



A. Type B tympanogram, normal ear canal volume.

The Type B tympanogram is characterized by a flat graph. This indicates that the eardrum is not moving well. If the tympanogram is Type B with normal ear canal volume (the most common configuration) it usually means there is fluid in the middle ear space (Otitis media) preventing the eardrum from moving properly. If it is Type B with a reduced ear canal volume, the cause is usually a plug of earwax fully occluding the ear canal. And if it is Type B with an abnormally large ear canal volume, the cause is likely to be a perforated ear drum. Above you see a Type B tympanogram from an ear with Otitis media.

Type C Tympanogram



E. Type C tympanogram with significantly negative peak pressure.

A Type C Tympanogram looks like a Type A in that it has a recognizable peak. The difference is that in a Type C, the peak is shifted off to the left. This usually indicates a Eustachian tube dysfunction. The Eustachian tube is not equalizing pressure properly and the resulting abnormally low middle ear pressure causes the peak to shift into the negative range. At right is a Type C tympanogram.

Other tests derived from tympanometry are *acoustic reflex* and *acoustic reflex decay*, which can provide insight regarding the status of the cranial nerves associated with the ear.

Hands-On Exercise

Notes

Perform the tests discussed in this unit on at least five people. Keep the audiograms that you collect for review at a later date by your instructor

Pure tone Air Conduction Testing

- Better ear or Right ear first. Then test opposite ear.
- 1000, 2000, 3000, 4000, 6000, 8000, Recheck 1000, 500, 250 Hz.
- Alternative: sequence 1000, 500, 250 Recheck 1000, 2000, 3000, 4000, 6000, 8000 Hz
- Begin at 50 dB HTL at each frequency.
- Use the “Quieter by 10 dB, Louder by 5 dB” method. Verify threshold twice ascending
- Do not fall into a rhythm and do not cue patient with movement.
- Record results. Right O (red), Left X (blue).
- Obtain frequency-specific LDL/TDs at 500 and 2000 Hz.

LDL/TD Loudness Discomfort Level/ Threshold of Discomfort Pure Tone

- Better ear first. Then opposite side.
- Frequencies to test 500, 1000, 2000, 3000 and 4000.
- Use 2 peep tone presentation 1 to 2 seconds long total
- Start presentation at 70 dB increase by 5 dB steps
- Be sure to watch patient’s facial expressions.
- Stop when patient raises their hand or indicates “7 - Uncomfortably Loud” on the Categories of Loudness Chart.

Pure Tone Bone Conduction Testing

- Better ear first. Then opposite ear.
- 1000, 2000, 4000, 500, 250 Hz.
- Begin at 20 dB louder than the air conduction threshold at the frequency being tested.
- Use the “Quieter by 10 dB, Louder by 5 dB” method. Verify threshold twice. ascending
- Record results <Right, Left>.

SRT: Speech Reception Threshold

- Better ear first. Then opposite side.
- Use Spondee words C.I.D. W-1 List A (ie. greyhound, schoolboy).
- Compute the Pure Tone Average (500+1000+2000 Hz)/3.
- Begin testing at 20 dB louder than the PTA.
- Use the “Quieter by 10 dB, Louder by 5 dB” method.. Verify SRT twice ascending.
- Record results. SRT = PTA within +/- 10 dB.

MCL: Most Comfortable Level

- Better ear first. Then opposite side. Then binaural.
- Use paired comparison sentences or continuous discourse.
- Begin at 20 dB louder than the SRT.
- Increase the repeat of each sentence by 5 dB or as continuous discourse continues
- Keep increasing until patient lets you know which was most comfortable. Verify MCL twice. Bracket by increasing 5dB over comfort level until slightly too loud then decrease by 5dB
- Record results.

Notes

UCL: Uncomfortable Level Speech

- Better ear first. Then opposite side.
- Use paired comparison sentences or continuous discourse.
- Begin at MCL.
- Be sure to watch patient's facial expressions.
- Increase each sentence 5 dB. Verify UCL twice.
- Bracket this as well going louder from where they tell you it is too loud
- Record results.

Word Recognition Scores (WRS)

- Better ear first. Then opposite side. Then binaurally.
- Use PB words, (phonetically balanced) C.I.D W22List 1A, or NU¬6 list.
- Conduct test at MCL (or 5 - 10 dB louder if needed for best results).
- Give minimum half list of 25 words per ear, then 25 words, binaurally. Record results. Record the incorrect words the patient says. Each word equals 4%.
- Record results as percentage correct.

Notes

Check Your Understanding

1. When you set the audiometer to a 40 or 50dB HL tone and sweep through all the frequencies. Then turn to a 1000 Hz tone and sweep through all the intensities (0 to 100dB HL) you are doing a

   a. Audiometric Test

   b. Pure Tone Test

   c. Weber Test

   d. Biological Test Calibration

2. QuickSIN is a test of your patients' ability to understand speech in noise. This is an important metric to obtain for two reasons.

   a. First, one of the most common complaints our patients have is that they have difficulty understanding speech in noise. And second, since the advent of fully digital hearing aids, feedback cancellation has become a nearly universal feature

   b. First, one of the most common complaints our patients have is that they have difficulty understanding speech in noise. And second, since the advent of fully digital hearing aids, directional microphones have become a nearly universal feature.

   c. First to find the supra-threshold, and second to evaluate an individual's ability to recognize single -syllable words from a phonetically balanced (PB) word list.

   d. First to find the supra-threshold, and second , since the advent of fully digital hearing aids, feedback cancellation has become a nearly universal feature

3. A patient has a flat 65dB hearing loss. The loss would be described as

   a. Moderately Severe hearing loss

   b. Moderate hearing loss

   c. Conductive Loss

   d. Sensorineural Loss

4. The ____________, your most important visual landmark, should be clearly visible in the lower front (inferior/anterior) quadrant of each ear drum.

    a. Tympanic Membrane

    b. Cone of light

    c. Stapes

    d. Basilar Membrane

5. (BC) Bone Conduction Threshold Testing begins at _______.

    a. Threshold of (AC) air conduction.

    b. Threshold of (AC) minus 5 dB

    c. Threshold of (AC) plus 10 dB

    d. Threshold of (AC) plus 20dB

6. The difference between the AC threshold and the BC threshold is called the____________. This is the amount of ____________hearing loss at each frequency.

    a. Air/Bone Gap; sensorineural

    b. Sensorineural hearing loss; conductive

    c. Air/Bone Gap; conductive

    d. Conductive hearing loss; Air/Bone Gap

7. Playground, Northwest, Baseball are examples of __________; and are used to find _____.

    a. Phonetically Balanced Words: MCL

    b. Spondee Words; SRT

    c. Phonetically Balanced Words; SRT

    d. Spondee Words; MCL

8. Dynamic Range is:

    a. UCL – SRT

    b. MCL – SRT’

    c. UCL – MCL

    d. MCL – SRT

9. A Binaural degradation may show up when doing the _______ testing.

    a. SRT

    b. MCL

    c. UCL

    d. WRS

10. Perform LDL testing in:

    a. Two High Frequencies

    b. Two Low Frequencies

    c. Running Speech

    d. One low and One High Frequency

11. During Pure Tone , and Bone Conduction testing:

    a. The lowest intensity level at which at least 2 out of 3 presentations produce responses is considered to be the threshold.

    b. The highest intensity level at which at least 2 out of 3 presentations produce responses is considered to be the threshold.

    c. The lowest frequency level at which at least 2 out of 3 presentations produce responses is considered to be the threshold.

    d. The highest frequency level at which at least 2 out of 3 presentations produce responses is considered to be the threshold

12. Always begin audiometric testing:

    a. On the Right Ear

    b. On the Left Ear

    c. On the patients perceived Better Ear

    d. On the patients perceived Worst Ear

Categories of Loudness

7 – Uncomfortably loud

6 – Loud but OK

5 – Comfortable (slightly loud)

4 – Comfortable

3 – Comfortable (slightly soft)

2 – Soft

1 – Very soft

QuickSIN Lists

Examples:

The sun came up to light the eastern sky.

Take shelter in this tent but keep still.

QUICKSIN 10-11-12

Tr. 12 – List 10

1. Dots of light betrayed the black cat.
2. Put the chart on the mantle and tack it down.
3. The steady drip is worse than a drenching rain.
4. A flat pack takes less luggage space.
5. The gloss on top made it unfit to read.
6. Seven seals were stamped on great sheets.

Tr. 13 – List 11

1. The marsh will freeze when cold enough.
2. A gray mare walked before the colt.
3. Bottles hold four kinds of rum.
4. He wheeled the bike past the winding road.
5. Throw out the used paper cup and plate.
6. The wall phone rang loud and often.

Tr. 13 – List 12

1. The hinge on the door creaked with old age.
2. The bright lanterns were gay on the dark lawn.
3. He offered proof in the form of a large chart.
4. Their eyelids droop for want of sleep.
5. There are many ways to do these things.
6. We like to see clear weather.

Unit 7: Masking

Objectives

Notes

Overview

Masking is a necessary component of audiometric testing. Every hearing care professional needs to be competent with the concepts and the procedures of masking. This module will discuss the criteria for determining the necessity of pure tone air conduction and bone conduction threshold testing and speech testing, as well as the proper techniques for determining masked thresholds

Goals

Upon completion of this module, you will be able to:

- Recognize the criteria for determination of the need to mask for pure tones and speech.
- Identify the rules and procedures for masking for pure tone air conduction testing.
- Identify the rules and procedures for masking for pure tone bone conduction testing.
- Identify the rules and procedures for masking for speech testing

Follow Up

- Donaldson, BCH-HIS, CCC-A, Linda L. *Practical Applications of Masking and Principles and Procedures (Third Edition)* National Institute for Hearing Instrument Studies

Section 1 – Why Mask?

Notes

The simplest explanation of the necessity for masking is:

We mask so that we can be certain which ear we are testing.

Under some circumstances, we cannot be sure which ear is actually responding. In those situations, we occupy the ear we do not want to test with masking noise, so that we know the responses we obtain are truly from the ear we want to test.

With apologies to Mr. Shakespeare, "To mask or not to mask? That is the question. Whether it is nobler to just ignore the problem or to really determine the true threshold?"

For the hearing care professional, there is only one answer: Determine the true threshold and do it correctly.

Definition of Terms

Let's begin with a list of terms that you will need to know to make sense of the process.

TE - Test Ear: The ear that is being tested. This is the ear receiving the test signal of interest, be it pure tone or speech. This will be the poorer ear.

NTE - Non-Test Ear: The other ear or better ear, This is the ear not being tested and that will be presented with the masking noise, either narrow band noise or speech noise.

NBN - **Narrow Band Noise**: Broad-band noise that is centered on a single frequency. This is the noise of choice to use when masking pure tones.

Speech noise: This is broad-band noise that has the envelope of long-term average speech. That is, most of its energy is in the mid frequencies (500-4000 Hz) with less energy in lower and higher frequencies. This is the preferred masker for speech testing.

IA - Inter-aural Attenuation: The reduction in intensity of a signal passing through the mass of the head. Inter-aural attenuation for air conduction is 40 dB for TDH-39 headphones and 70 dB for insert earphones. Inter-aural attenuation for bone conduction is 0 dB.

Crossover: Crossover occurs when inter-aural attenuation is exceeded and a test signal "crosses over" from the test ear to the non-test ear.

EM - Effective Masking: The intensity of masking noise necessary to occupy an ear and prevent crossover from occurring.

Overmasking: When the masking noise is intense enough to be at risk of exceeding inter-aural attenuation and becoming audible in the test ear.

Undermasking: Masking that is insufficient to prevent crossover of the test signal.

AC Masking: Air Conduction Masking – Masking that is administered to prevent crossover of the test signal in air conduction testing for either pure tones or speech.

BC Masking - Bone Conduction Masking: This is masking that is administered to prevent crossover of the test signal in bone conduction testing, usually for pure tones.

TDH-39 Circum-aural Headphones: Headphones used during audiometric testing. The inter-aural attenuation for an air conducted signal presented via TDH-39 headphones is 40 dB.

Insert Earphones: Earphones within foam inserts. The inter-aural attenuation for an air conducted signal presented via insert earphones is 70 dB.

Plateau: The four consecutive intensities of masking noise over which the signal in the test ear remains audible and is not increased.

Masking Dilemma: When the intensity of masking noise is, from the outset, intense enough to create an overmasking situation. This is most likely to occur when there are large conductive components in both ears.

True Threshold (masked threshold): The threshold that has been obtained after a masking plateau has been established.

SRT: Speech Reception Threshold

Spondee Words: Two syllable words with equal stress on each syllable. Used during SRT testing.

SDS: Speech Discrimination Score

WRS: Word Recognition Score

Phonetically Balanced Words (PB words): Used during word recognition testing

Notes

The following are the unmasked and masked pure-tone symbols:

The following are the unmasked and masked pure-tone symbols:

Unmasked air conduction – Right	O
Unmasked air conduction – Left	X
Masked air conduction – Right	Δ
Masked air conduction – Left	□
Unmasked bone conduction – Right	<
Unmasked bone conduction – Left	>
Masked bone conduction – Right	[
Masked bone conduction – Left	]

Check Your Understanding

Which of the following is the preferred masking noise for pure tones?

a. NBN

b. White noise

c. Speech noise

d. Loud noise

Notes

Section 2: Masking for Air Conduction

Notes

Masking is required for pure tone Air Conduction (*abbreviated **AC***) when crossover is possible. Crossover for AC occurs when the AC signal in the poorer ear vibrates the skull sufficiently to overcome inter-aural attenuation (*abbreviated **IA***) and stimulate the opposite-ear cochlea. When this happens, the better ear (non-test ear or NTE) must be kept occupied with masking noise while the true threshold of the poorer ear (test ear or TE) is determined.

Written as a formula, you will mask the better ear for AC when:

AC (TE) – IA ≥ AC (NTE) (at the same frequency)

In other words:

If the AC threshold of the Test Ear, minus Inter-Aural attenuation, is greater than or equal to the AC threshold of the non-test ear, at the same frequency – then masking is required for AC.

Headphones

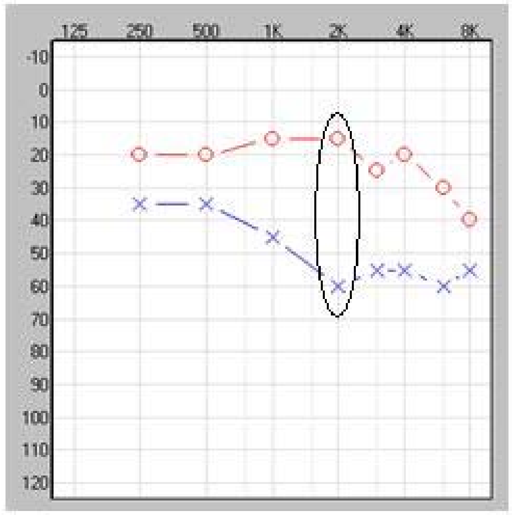
The criterion for when masking is required is different for TDH-39 headphones and for insert earphones. For TDH-39, inter-aural attenuation is 40 dB and for insert earphones inter-aural attenuation is 70 dB. This means that testing using TDH-39 headphones is more likely to require masking for air conduction than testing via insert earphones. For this reason, it is important that it is documented on the audiogram which transducer was used. Most state licensure exams will use the headphone rule, rather than the insert rule.

IIHIS Rule 1a:

Masking for air conduction is required if there is a greater than or equal to 40dB (70dB for inserts) difference between the better ear air conduction score and the poorer ear air conduction score at the same frequency.

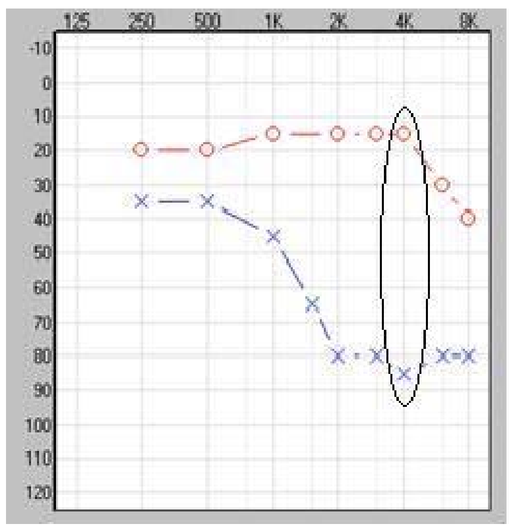
Application of Rule 1a Using TDH-39 Headphones

In the figure to the right, according to masking rule 1a, when TDH-39 headphones are used, the only frequency requiring masking in this audiogram would be 2000 Hz, where there is a 45 dB difference between the right and left ear thresholds. (If insert earphones were used, masking for air conduction would not be required.)



Application of Rule 1a Using Insert Earphones

In this figure, according to Rule 1a, using insert earphones this time, masking would be required only at 4000 Hz (75 dB difference between RE and LE thresholds). (If TDH -39 headphones were used, all frequencies above 1000 Hz would require masking for A/C.)



Notes

Masking for Air Conduction When AC and BC Scores Are Known

Because inter-aural attenuation is considered a bone conduction phenomenon, when BC thresholds are known, they must be taken into account. Which brings us to:

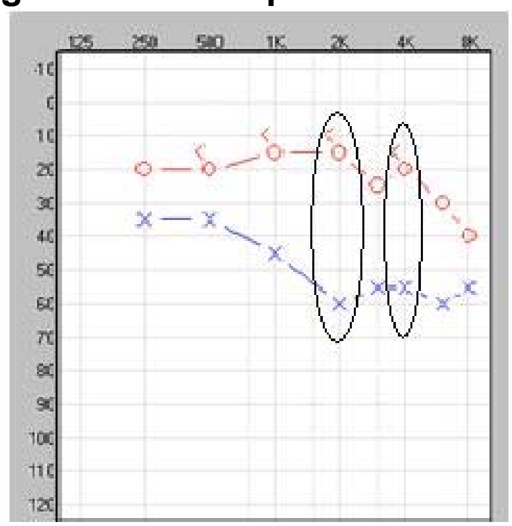
IIHIS Rule 1b:

Masking for air conduction is required if there is a greater than or equal to 40dB (70dB for inserts) difference between the better ear bone conduction score and the poorer ear air conduction score at the same frequency.

Note: Be sure to check your own state laws regarding requirements for air conduction masking. Each state sets its own rules and some exceptions apply.

Application of Rule 1b Using TDH-39 Headphones

This time (the figure to the right), with bone conduction scores known, we can see that, using TDH-39 headphones, masking would be required at both 2000 and 4000 Hz. (With inserts, no masking for AC would be required.)

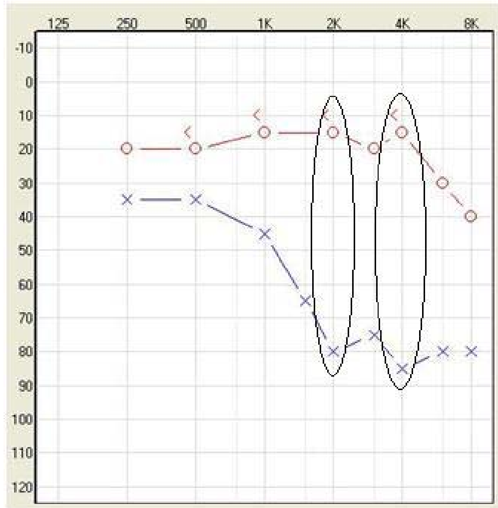


Notes

Application of Rule 1b Using Insert Earphones

With the addition of unmasked bone conduction thresholds, (using inserts) we can now see that masking for air conduction pure tones is required at 2000 Hz and 4000 Hz as well. This is shown in the audiogram to the right.

You see that, since there is a much greater allowable inter-aural attenuation for insert earphones, it is much more likely that masking for air conduction will be required for TDH-39 headphones than for insert earphones.



Masking for AC: Step-by-Step

If the air conduction masking rule was violated:

Give instructions. For example:

"Mrs. Jones, you will be hearing 'hissing' noise in your left ear. Simply ignore the hissing and continue to push the button when you hear the beeps, just as you have been."

Apply narrow-band noise to the non-test ear. The beginning level of the masking will be the non-test ear pure tone threshold plus a 10 dB pad. (The additional 10 dB assures that the masking noise is audible to the patient.)

Ask the patient if he or she hears the noise. If so, proceed. If the noise is inaudible, increase the intensity until it is audible.

Present pure tone to the test ear, beginning at the already obtained unmasked threshold intensity.

NOTE: The two levels should not be more than 40 dB apart for headphones, or 70 dB apart for insert earphones. If so, increase the NBN so that there is not a danger of undermasking.

If there *is* a response to the tone, increase the **masking** by 5 dB.

If there *is not* a response to the tone, increase the **tone** by 5 dB.

When masking you will never reduce the intensity of noise or tone. **Never increase both simultaneously!**

Notes

The masked threshold has been established when there has been a response to the same tone intensity in the test ear, with a corresponding 15 dB or greater increase in the NBN level in the non-test ear (four “yeses” in a row). You can’t be sure a threshold is true until you have challenged it four times and still get a response at the same level.

Check for overmasking. **Overmasking** is the level at which masking crosses over into the test ear and is louder than the maximum desired Plateau level.

Record masked AC threshold and final masking level at the bottom of the audiogram.

Three Important Things to Remember About the Masking Plateau

1. Intensity of signals only increase - they never decrease
2. Intensity always increases in 5 dB steps
3. Intensity increases for masking or for test signal - **never for both at the same time**

Check Your Understanding

Inter-aural attenuation for air conduction when using TDH-39 headphones is which of the following?

a. 70 dB

b. 60 dB

c. 50 dB

d. 40 dB

Notes

Section 3 - Masking for Bone Conduction

Inter-aural Attenuation for Bone Conduction is 0 dB. Therefore you are more likely to require masking for bone conduction than for air conduction.

IIHIS Rule 2: Masking for Bone Conduction

Masking for bone conduction is required if there is a greater than or equal to 15 dB difference between the air conduction score and the bone conduction score of the same ear at the same frequency.

Note: Be sure to check your own state laws regarding requirements for bone conduction masking. Each state sets its own rules and some exceptions apply.

Notes

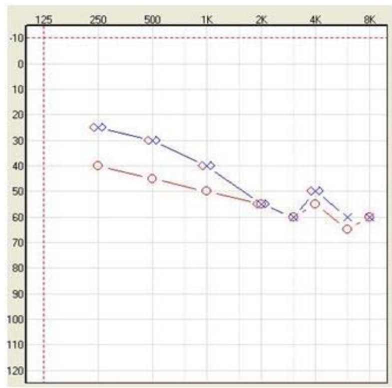
Masking for Bone Conduction

In this audiogram, masking for bone conduction, pure-tone thresholds is required at 250 Hz and 500 Hz.

Bone Conduction (BC) Masking Essentials

Place the bone oscillator on the mastoid process behind the ear with the potential air-bone gap (*abbreviated **ABG***). *This is the test ear*.

The air headphone/insert is placed *only on the ear to be masked* during bone conduction testing. This is the non-test ear. Leave the test ear unoccluded or uncovered (see below).



Masking for Bone Conduction: Step-by-Step

Give instructions. For example,

"You will hear a hissing sound in your [R/L] ear. Just ignore it. You will hear beeps through that thing behind your other ear. You may hear the beeps in either ear. Just continue to press the button whenever you hear the beeps."

Present pure tone bone conduction to the TE (test ear) at the obtained unmasked bone level for that frequency.

Present narrow-band noise (NBN) to the non-test ear. Additional NBN for low frequencies are required to compensate for the occlusion effect. Therefore, an additional 15dB at 250 and 500Hz, 10dB at 1000 Hz is required to start with.

This bears repeating!! To compensate for the occlusion effect, an **additional** 15 dB of initial masking is required at 250 and 500 Hz (so you will begin with NTE air conduction threshold, plus 10 plus 15 for a total pad of 25 dB) and an additional 10 dB at 1000 Hz (NTE AC threshold, plus 10, plus 10 for a total pad of 20 dB). No **additional** pad is required at 2000 and 4000 Hz. Remember to check your own state's specific rules.

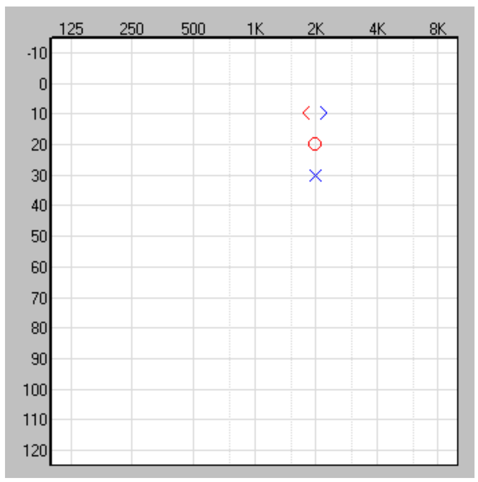
Notes

Notes

Once bone conduction masking is set up, the plateau method is the same as it was for air conduction. That is:

- For every response to the tone, increase the masking noise 5 dB.
- For every no-response to the tone, increase the tone 5 dB.
- When there has been a response to the same intensity of tone with four consecutive masking levels (15 dB plateau), the masked threshold has been established.
- Document the threshold on the audiogram and document the final masking level at the bottom of the audiogram.

Check for **overmasking**. Overmasking is much more common when masking for bone conduction testing. It occurs when the masking level is greater than the bone conduction threshold of the test ear plus inter-aural attenuation.



See the partial audiogram at right as an example. The right and left unmasked bone conduction thresholds were obtained at 10 dB HL and the right and left air conduction thresholds are at 20 and 30 dB HL respectively. When masking for bone conduction on the right side, a tone will be presented via the bone oscillator on the right mastoid at 10 dB HL, while 40 dB HL of masking noise is presented via headphones to the left ear. With a positive response to the tone, the masking noise is raised to 45 dB HL. Another positive response then raises the masking level to 50 dB HL. We are now in a potential overmasking situation. The difference between the test tone on the right (10 dB HL) and the masking noise on the left (50 dB HL) is 40 dB, which is the inter-aural attenuation of an air conduction signal. It is now possible that enough masking noise is present for that signal to cross to the test ear and mask the tone being presented to the right ear.

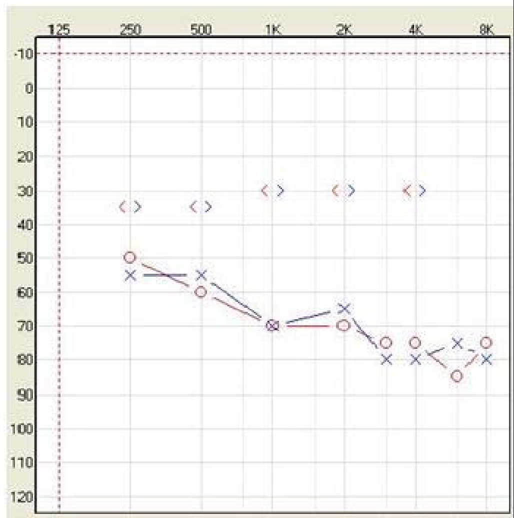
Sometimes overmasking is unavoidable. When this is the case, document the last threshold obtained before overmasking would be an issue, and note on the audiogram that the possibility of overmasking would have been a factor.

The Masking Dilemma

Notes

A masking dilemma occurs when both ears have significant A/B gaps - on the order of 40 dB or more. This may occur at one or multiple frequencies, with bilateral conductive or mixed hearing loss. If the minimum starting level for masking the NTE is at least equal to the AC score of that ear, and both ears have a 40 dB A/B gap, then you would be overmasking from the very beginning, even before you were to increase the levels of your plateau. It is not possible to mask in this situation. You are unable to establish a true threshold and a referral is required. Write “masking dilemma“ on the audiogram and proceed to the next test.

In the following audiogram, masking is required for bone conduction at 250, 500, 1000, 2000, and 4000 Hz. At 1000, 2000 and 4000 Hz, regardless which ear is masked, the beginning masking level would be so intense as to be able to cross over to the other ear (the BC thresholds). The masking level for 1000 Hz, for example, would have to be greater than 120 dB HL, which is at least 90 dB greater than the unmasked bone conduction threshold of 30 dB HL. This greatly exceeds AC inter-aural attenuation of 40 dB (the masking noise would inevitably be audible in the test ear), hence masking is impossible.



Sample AC Unmasked and Masked Audiograms

In Figure A, notice how after masking was applied, the air conduction thresholds at 250, 500, 1000 and 8000 Hz were shifted. This means that the unmasked thresholds at those frequencies actually represented the response of the right ear. That is, the tones presented to the left ear were loud enough to be heard in the right ear. When the right ear was “taken out of commission” by masking noise (Figure B), true air conduction thresholds could be obtained for the left ear.

Notes

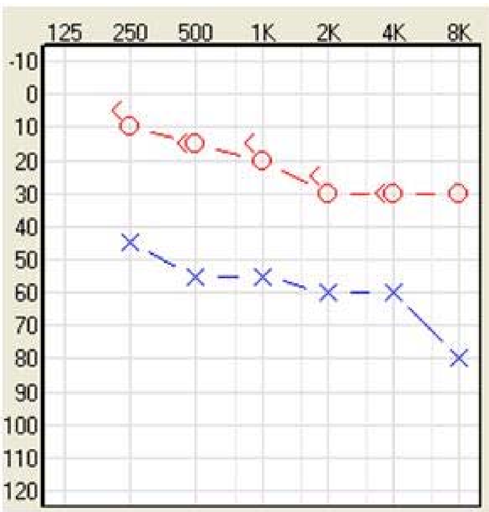
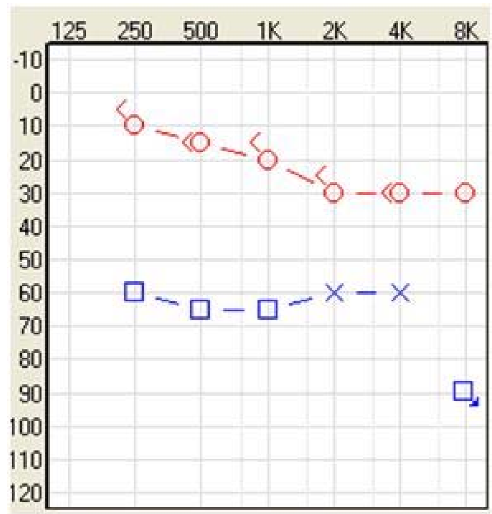


Figure A ←

→ Figure B



Sample BC Unmasked and Masked Audiograms

Notes

In Figure C, we have obtained only unmasked bone conduction thresholds. All we know is that one of the cochlea hears at about the same levels as the right ear air conduction thresholds. Since the better cochlea is responding, we need to occupy the right cochlea with masking noise to determine bone conduction thresholds for the left ear. When we do so at 250, 500 and 1000 Hz, we see in Figure D that, indeed, the unmasked thresholds did not represent the actual hearing levels for the left cochlea. We are now able to see that the hearing loss in the left ear is sensorineural (no significant air/bone gaps).

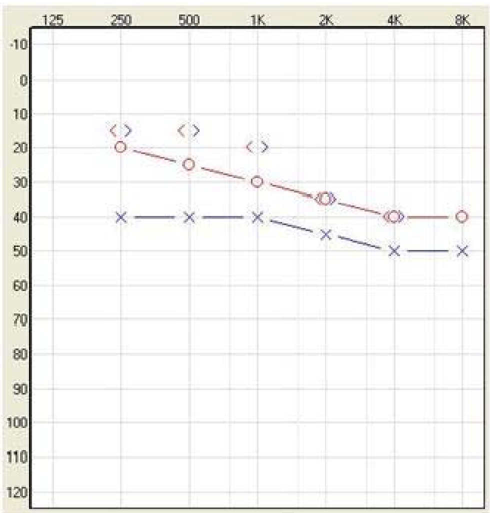
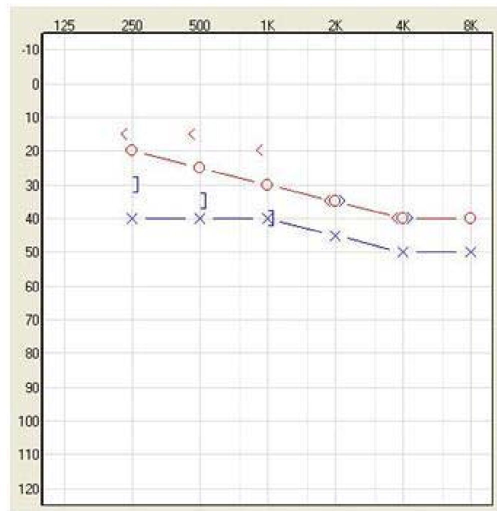


Figure C ←

→ Figure D



Check your Understanding

At 250 Hz the air conduction threshold of the non-test ear is 35 dB HL. At what intensity do you begin presenting masking noise to that ear?

a. 45 dB HL

b. 50 dB HL

c. 55 dB HL

d. 60 dB HL

Section 4: Masking for Speech SRT and WR

Notes

If it is necessary to mask for AC or BC pure-tone threshold testing then it may be necessary to mask for speech testing (*abbreviated* ***SRT)*** and Word Recognition. Below are the instructions for this.

Masking for SRT

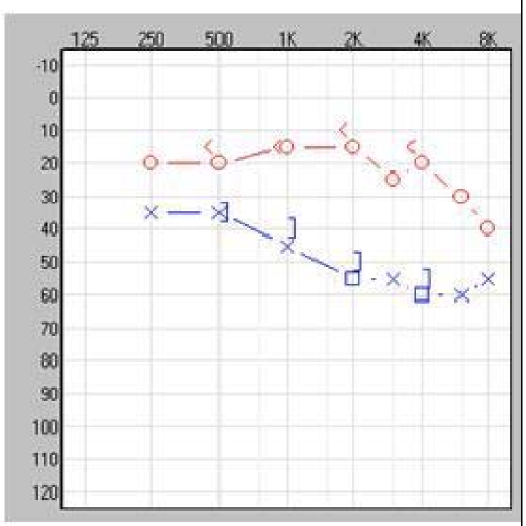
Speech will crossover in the same manner as pure tones. Speech is a complex signal comprising many frequencies simultaneously. The inter-aural attenuation for speech is 40dB for headsets, and 70dB for insert earphones, just as it was for pure tones. The rule for when masking for SRT, then, is very easy:

If you had to mask for Air Conduction, you will have to mask for SRT.

Since speech is a broad-band signal, if the criterion for AC masking has been met, then there is at least a small possibility that some part of the speech signal could cross over to the non-test ear, therefore masking is required.

This time we use *speech noise* to occupy the non-test ear. Begin speech-noise masking in the better (non-test) ear at its SRT + 10 dB. Begin spondees in the test ear at the unmasked SRT level. We then employ the same plateau method we used in pure-tone testing. For every correct response to a spondee word, increase the masking noise by 5 dB, and for every incorrect response, increase the intensity of the spondee word by 5 dB. Again, we will never increase both at the same time. Four correct responses at the same word intensity with three increases in masking noise level (four levels spanning 15 dB) establish the masked SRT threshold.

Consider the audiogram to the right. Testing was administered using TDH-39 headphones. For this patient the unmasked SRT for the right ear (RE) is 20 dB HL, and for the left ear (LE) it is 40 dB HL. Because masking was performed for AC thresholds at 2000 and 4000 Hz, masking is required for SRT in the LE. We will begin masking in the right ear (using speech noise) at 30 dB HL (SRT plus 10 dB) and begin presenting spondee words in the left ear at 40 dB HL (the level at which unmasked SRT was obtained). For each spondee word repeated accurately, the masking noise will be increased by 5 dB and for each spondee word not repeated accurately, the intensity will be increased for the spondee words and another word presented. When four consecutive spondee words have been repeated correctly at the same presentation level with three increases in masking (four total levels spanning 15 dB) the plateau has been established and a masked SRT determined.



Check Your Understanding

What is the inter-aural attenuation for air conduction speech using insert earphones?

a. 40 dB

b. 20 dB

c. 70 dB

d. 55 dB

Masking for Word Recognition

The presentation level for Word Recognition testing is usually the Most Comfortable Level (MCL) of the test ear or slightly louder. Because this is a supra-threshold (louder than threshold) intensity, it is reasonably likely that the presentation level for word recognition testing may be sufficient for crossover to the non-test ear to occur, even without large air conduction threshold asymmetries. Hence, we must mask for word recognition when the presentation level for word recognition in one ear minus inter-aural attenuation is greater than or equal to the SRT (or PTA) of the other ear.

In masking for Word Recognition, the Plateau Technique is not necessary since the presentation level is fixed. The masking level in the non-test ear is the presentation level of the test ear minus 20dB (or minus 30dB, if first masking level is "too loud". Masking noise should never exceed the patient's tolerance level in any masking procedure).

Once again, consider the audiogram on the previous page. Right ear (RE) SRT is 20 dB HL and masked left ear SRT is 40 dB HL. MCL for the right ear is 55 dB HL and for the left ear (LE) is 70 dB HL. Since a 70 dB AC speech signal presented to the LE is likely to cross over (70 minus inter-aural attenuation for TDH-39 headphones of 40 dB) the PB words could be perceived in the right ear at 30 dB HL, which is 10 dB louder than the 20 dB HL SRT for the RE. Masking is therefore required. Speech noise is presented to the RE at 50 dB HL (presentation level of 70 dB HL [the MCL for the left ear] minus 20 dB). The patient is asked if this is comfortable, and indicates that it is. (If it were not comfortable, the masking noise would be reduced to 40 dB HL.) 25 PB words are presented to the LE at 70 dB HL and the percentage is recorded on the audiogram.

Check Your Understanding

Which of the following is not necessary when masking for Word Recognition?

a. Masking with speech noise

b. Using the Plateau Technique

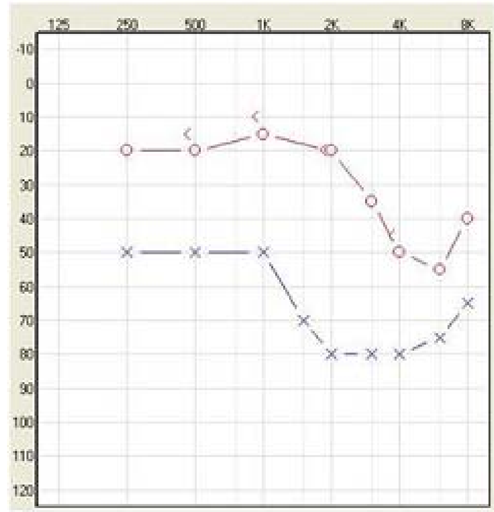
c. Presenting a complete list of PB words

d. Presenting words to the test ear at MCL

Notes

Check Your Understanding

For the following questions, refer to the audiogram to the right.



1. Assuming we are using TDH-39 headphones, for which frequencies is masking required for air conduction? Select all that apply.

    a. 1000

    b. 1500

    c. 2000

    d. 3000

2. At 1000 Hz, which is the test ear?

    a. Right

    b. Left

3. At 1000 Hz, in which ear will the masking noise be presented?

    a. Right

    b. Left

4. What kind of masking noise will you use?

    a. White noise

    b. Narrow-band noise – NBN

    c. Speech noise

    d. Bone noise

5. At 1000 Hz, at what intensity will you begin the masking noise in the non-test ear?

    a. 15 dB HL

    b. 20 dB HL

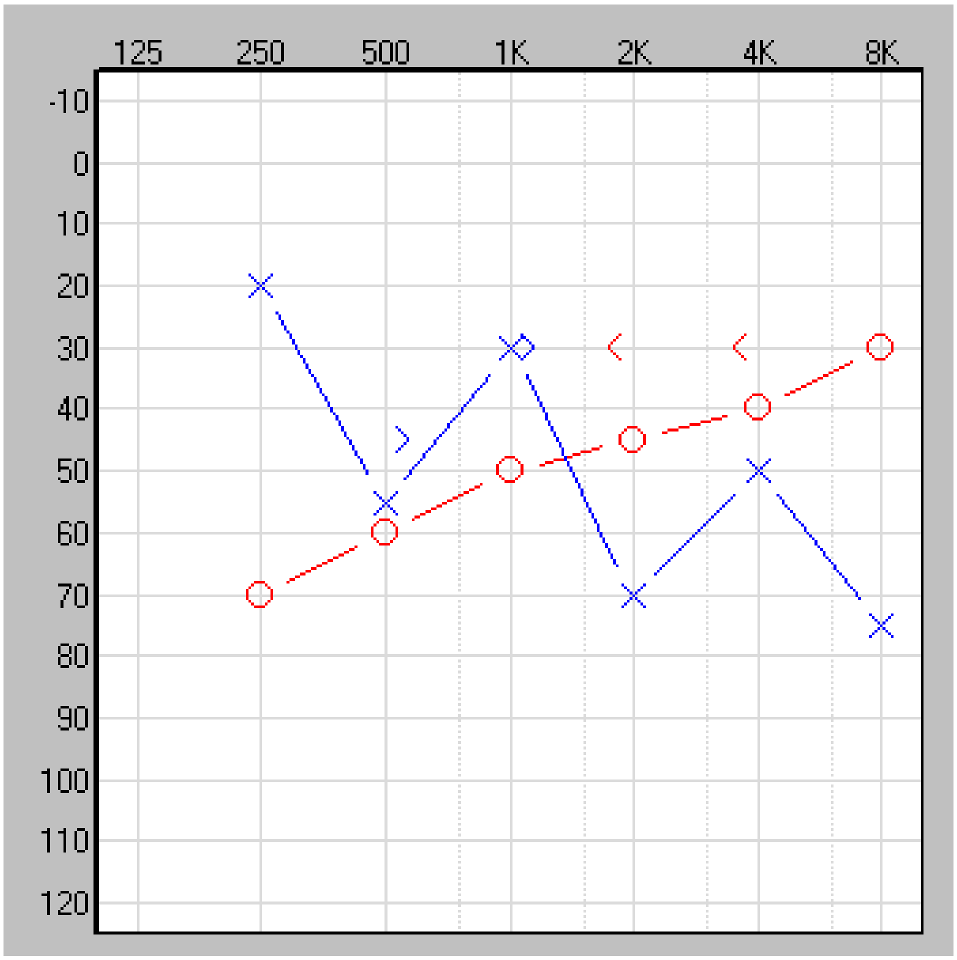
    c. 25 dB HL

    d. 35 dB HL

6. At 1000 Hz, at what intensity will you begin presenting pure tones in the test ear?
   a. 10 dB HL
   b. 15 dB HL
   c. 50 dB HL
   d. 60 dB HL
7. Which frequencies require masking for bone conduction?
   a. 500
   b. 1000
   c. 2000
   d. 4000
8. At 500 Hz, which ear is the test ear?
   a. Right
   b. Left
9. At what intensity will the masking noise begin in the non-test ear at 500 Hz for BC masking?
   a. 25 dB HL
   b. 45 dB HL
   c. 30 dB HL
   d. 50 dB HL
10. At what intensity will tones first be presented to the test ear at 500 Hz for BC masking?
   a. 15 dB HL
   b. 20 dB HL
   c. 30 dB HL
   d. 50 dB HL

Masking Exercise

Look carefully at the following audiogram and complete the two tables that follow.



Masking for Air Conduction

Frequency	Is masking required at this freq.? (Y/N)	Which ear will be masked (i.e. non-test ear)? (R/L)	Which ear will be the test ear? (R/L)	At what intensity will masking begin?	At what intensity will be the first presentation to the test ear?
250 Hz					
500 Hz					
1000 Hz					
2000 Hz					
4000 Hz					
8000 Hz					

Masking for Bone Conduction

Frequency	Is masking required at this freq.? (Y/N)	Which ear will be masked (i.e. non-test ear)? (R/L)	Which ear will be the test ear? (R/L)	At what intensity will masking begin?	At what intensity will be the first presentation to the test ear?
250 Hz					
500 Hz					
1000 Hz					
2000 Hz		1_______ 2	1_______ 2	1______ 2	1_________ 2
4000 Hz					
8000 Hz					

Masking Quick Guide

Masking for Air Conduction

The Rules

1A

Masking is required if there is a greater than or equal to 40dB difference between the better ear air conduction score and the poorer ear air conduction score at the same frequency.

1B

Masking is required if there is a greater than or equal to 40dB difference between the better ear bone conduction score and the poorer ear air conduction score at the same frequency.

Test ear – Poorer ear

Begin presenting tones at the previously obtained air conduction threshold.

Non-test (masked) ear – Better ear

Begin presenting narrow-band noise at the previously obtained air conduction threshold **plus 10 dB.**

Procedure – If there is a response to the tone, increase the masking by 5dB.

If there is not a response to the tone, increase the tone by 5dB.

Never increase both simultaneously!

Stop when there is a response to the same tone intensity with a corresponding 15 dB increase in the NBN level (4 yeses in a row).

Record masked AC threshold *and final masking level.*

Masking for Bone Conduction

The Rule -If there is a greater than or equal to 15 dB difference between the air conduction score and the bone conduction score of the same ear at the same frequency.

Test ear – the ear with the air/bone gap.

Begin presenting tones at the unmasked bone conduction threshold.

Non-test (masked) ear – the other ear.

Air conduction transducer is placed only over or in the masked ear. The test ear is left open. Beginning masking levels for BC vary, depending on the frequency – as follows.

(250) and 500 Hz - AC threshold plus 25 (AC+10+15)

1000 Hz - AC threshold plus 20 (AC+10+10)

2000 and 4000 Hz - AC threshold plus 10

Procedure - If there is a response to the tone, increase the masking by 5dB.

If there is not a response to the tone, increase the tone by 5dB.

Never increase both simultaneously!

Stop when there is a response to the same tone intensity with a corresponding 15 dB increase in the NBN level (4 yeses in a row).

Record masked AC threshold *and final masking level.*

Unit 8: Introduction to Hearing Instruments

Objectives

Notes

Overview

The very first hearing aid was probably created when early man cupped his hand behind his ear to hear distant sounds or his mate's voice over the roar of the campfire. It wasn't until the 20th century and the age of electronics that the hearing impaired person had much more than a cupped hand to improve his hearing ability. In today's age of digital electronics the number of calculations a technologically advanced hearing instrument will make in a hundredth of a second is truly staggering. In this unit you will be introduced to the inner workings of hearing instruments as well as many of the terms used to describe hearing instrument performance.

Goals

Upon completion of this module, you will be able to:

- Recite an overview of the history of hearing instruments
- Identify the basic components of hearing aids
- Recognize hearing aid styles/form factors
- Explain concepts of hearing aid compression
- Differentiate between output limiting and wide dynamic range compression

Follow Up

- In *Distance Learning for Professionals in Hearing Health Sciences* (pp. Lessons 23, 24 and 26-27, 29).
- Dillon, H. (2001). In *Hearing Aids* (pp. Chapters 2-6). New York: Allyn and Bacon.
- Hall, M. a. (1998). Hearing Instrument Science and Fitting Practices. In *Audiologist Desk Reference Volume II* (pp. 562-595). San Diego: Singular Press.
- Sandlin, R. In *Hearing Instrument Science and Fitting Practices* (pp. 380-414, 597-643, 699-841).
- Listen to and manipulate as many hearing aids and hearing aid software programs as possible.

Section 1: Introduction to Hearing Aids

Notes

A hearing aid may be described as an electronic sound amplifier specifically designed to make soft sounds audible for those persons with hearing loss. Today's programmable hearing instruments accomplish this through a series of electronically applied mathematical calculations called algorithms. As you will see, there are many brands, types, and models of hearing instruments to choose from. The good news is that almost all hearing instruments today share common components and that most of these components operate in the same general way. Let's start by looking at how a hearing instrument provides enhanced sound or what we call amplification.

What is amplification?

Amplification in today's technically advanced hearing instruments could well be considered an electronic sleight of hand as almost all hearing instruments manipulate or control sounds occurring naturally in the real world in the same manner. Typically, energy is first pulled from an outside source (the battery) as outside sounds enter the microphone and strike a diaphragm and are changed or **transduced** from an acoustic sound waveform into an analogous current of electricity. Energy is "transduced' when it is changed from one form to another. Yesterday's **analog hearing instruments** typically boosted this current or signal at the amplifier stage and sent it on to the receiver where it was changed or transduced back to an acoustic signal prior to delivery in the ear canal. Today's digital hearing instruments send this electrical current to an **analog-to-digital (A/D) converter** where it is changed from an electrical signal into a numeric code prior to the amplifier stage and then to a **digital-to-analog (D/A) converter** where it becomes an electrical current once more. In between the A/D converter and the Amplifier/D/A converter is the heart and brain of the circuit in a new modern hearing aid. Here is where mathematical changes are made. This may be called the **Advanced Features** department. The digital manipulation of sound allows for better control of environmental interference and is highly effective in digital hearing instruments.

Hearing Aid History and Overview

The hearing aid industry has advanced considerably in terms of both manufacturing techniques and products over the past few years. Currently, our feet are firmly planted in the digital era of hearing aid technology. Many hearing aid shells are now crafted using laser technology rather than the poured molds of 10 years ago or even the ultraviolet light process which was new only a few years ago. Computer-driven, multiple-channel-processing, multiple-memory, and multiple-microphone technology have replaced unsightly potentiometers and oversized microphones and are the logical solutions for maximizing wearer benefit. This was not always the case.

Pre-electronic hearing aids date back centuries and include such systems as hearing horns, or trumpets. If we were to describe amplification in physical measurement terms, the pre-electronic horns were capable of minimal acoustic gain primarily in the low frequency range of 500-1000 Hz, but little between 1000-4000 Hz.

Notes

The electronic era of hearing aids began with inventions initially appearing in other innovations of the day. An era sometimes called the **carbon era** began around 1900, and is characterized by the use of carbon material behind the diaphragm of the carbon style microphone. The carbon era style hearing aid was capable of providing about 20-30dB of amplification. This was appropriate for mild to moderate losses. This era lasted into the 1940's. These hearing aids were not wearable or portable. Some "shaping" of a frequency response - selectively amplifying certain frequencies to match hearing loss - was possible. Filtering, or limiting, of the hearing aid response was not feasible.

When the vacuum tube triode amplifier was invented in 1907, the **vacuum tube** era began. Vacuum tubes appeared in some hearing aids during the 1920's, and emerged more fully during the 1930's. They were very large and required multiple tubes and batteries. The power of hearing aids, however, did increase substantially to near 70dB gain and 130dB output. This would be appropriate for much more severe losses. During this era, filtering, shaping, and limiting were electronically possible. The first one-piece hearing aid was not introduced until around 1944. The wearable body aid was born! The vacuum tube era ended in the 1950's with the acceptance of the transistor style hearing aid.

Vacuum tube hearing aids

The **transistor era** began in 1947 and hearing aids began to use the technology in the early 1950's. In the transistor era, hearing aid components became smaller and more efficient. Hearing aids could now be worn on the head via eyeglasses or behind the ear. By 1964 transistors had become small enough to be put on a small integrated circuit chip, allowing for in-the-ear custom hearing aids. From a market share standpoint, in 1960, body aids accounted for approximately 25% of sales, eyeglass aids, 45%, and BTEs, 30%. In-the-ear sales were reported as 2% in 1961, did not reach 10% until 1967, and did not reach 30% until 1977.

The **digital era** began in the 1960's in the telephone industry, but didn't make much of an impact on the hearing aid industry until the mid 1990's. Programmable analog hearing aids were in existence beginning around 1986. They were a transitional technology in that the processing of the sound by the hearing aid was analog, but modifications to the frequency response were controlled by a computer.

In 1996 the first ear-level, fully digital hearing aid was introduced commercially. Digital hearing aid amplifiers allow for several advantages. **Digital signal processing** (DSP) could be used to control smaller areas or bands of information within the frequency response.

A professional using a computer interface could make adjustments. The result was more fitting flexibility and nearly infinite shaping of the desired response. Because they use mathematical bits of information, digital amplifiers also can act with speed that allows the possibility of advanced automatic gain adjustment, noise reduction, adaptive directional microphones, feedback algorithms and much more.

Section 2: Hearing Aid Components

Notes

Batteries

Because hearing instruments are electronic devices they need an energy source. Typically a dry cell zinc air battery provides this energy. Hearing aid batteries commonly have a reserve amount of storage of 1.4 volts and come in a variety of sizes. Commonly-used battery sizes for hearing aids include the 1.4volt AA battery for body aids, blue tabbed #675 for power BTEs, orange tabbed #13 for smaller BTEs and ITE contour aids, brown tabbed #312 for canal aids, receiver-in-the-canal (RIC) and receiver-in-the-aid (RITA) mini-BTEs, yellow tabbed #10 for small canal and CIC aids, and red tabbed #5 for very small CICs. Zinc-air batteries are activated by exposure to air. As long as the tab is in place, the battery is inert. Once the tab is removed, after several seconds the battery begins to function.

The formula for calculating battery life is as follows:

Battery life (hours) **Equals =** Battery rating capacity (mA hours – obtained from battery manufacturer) **Divided by /** Battery current drain (mA – obtained from hearing aid specifications or ANSI strip)

This number is then divided by the number of hours the patient wears the instrument.

For example, a battery with a 200 mA/hour battery rating capacity, if used in a hearing aid with a battery drain of 2.0 mA, would be expected to last for 100 hours (200 divided by 2). If the patient wears the hearing aid 10 hours a day, the battery would last for 10 days (100 divided by 10).

Check Your Understanding

12 volt batteries are commonly used in hearing aids.

True or False

Transducers

Notes

Transducer is a commonly-used engineering term for any device that changes energy from one form to another. The human ear is a transducer (transducing acoustic energy in the air into an electrical signal that can travel to the brain), so is a gerbil running on a wheel to power a light bulb (kinetic energy into electrical energy). In hearing instruments sound is changed from an acoustic wave signal to an electrical current and then back to an acoustic signal. The two transducers we most need to be concerned about are the **microphone (input)** and the **receiver (output)**.

Microphone

The first transduction in a hearing aid occurs at the input stage or microphone. The microphone picks up sounds in the environment, changing acoustical waveform properties into an analog electric wave with greater and lesser voltages matching the compressions and rarefactions of the original incoming sound signal. Since the microphone acts on or changes sound input coming into the hearing instrument it is sometimes referred to as an **input transducer**.

A microphone's main component is a diaphragm made of metalicized plastic that carries or holds a permanent electric charge. When sound strikes this diaphragm, it moves or vibrates to the input signal (not unlike how the eardrum moves or vibrates to sounds channeled through the ear canal). This causes changes in the material behind the diaphragm. See **Figures 1** and **2** below as examples of a single port (left) versus dual port (right) microphone example.

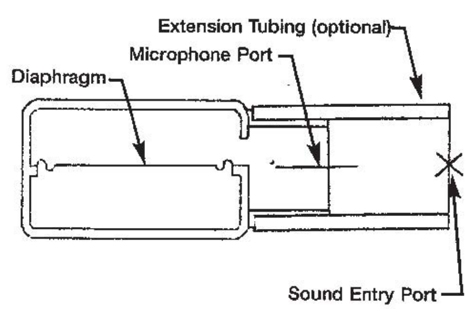


Illustration of omnidirectional microphone

Figure 1

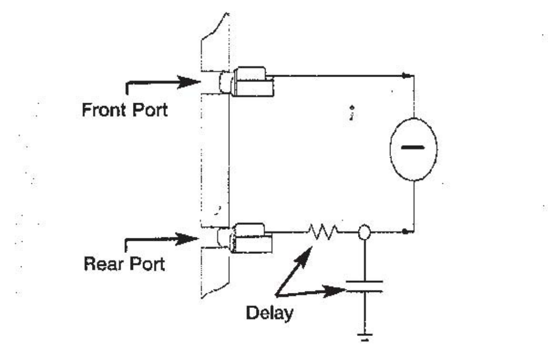


Illustration of dual microphone directional system consisting of two identically matched omnidirectional microphones.

Figure 2

Microphones may also be defined by the manner in which they pick up sound inputs. They can be either non-directional, otherwise known as **omnidirectional**, or they may be **directional**.

An **omnidirectional microphone** amplifies sound originating from all directions roughly equally.

A **directional microphone** may either comprise multiple sequenced microphones, or a single microphone with multiple ports or openings. Multiple microphones create a signature pattern where only incoming sounds from certain areas (usually in front of the wearer) are amplified. Patterns created by directional microphones are typically displayed as **polar diagrams or plots**. (See below for a description of polar plots.)

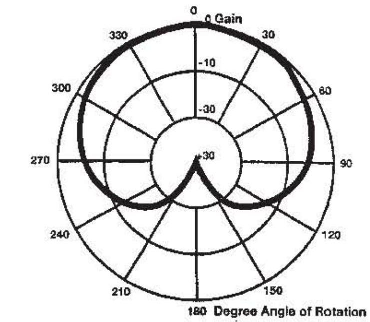
The desired effect of a directional microphone is to emphasize the foreground speech (the signal the wearer wants to hear) relative to the background noise (the sound that is causing interference), ultimately resulting in what is known as an improved **signal-to-noise ratio (SNR)**. Signal to noise ratio is not a ratio at all. It is a difference in dB between the intensity of a signal of interest and a competing signal. If, for example, the noise in a room is 45 dB SPL and the intensity of a speaker's voice is 65 dB SPL, the SNR would be plus 20 (signal is 20 dB louder than noise). If the situation were reversed, the SNR would be *minus* 20, i.e. noise is 20 dB louder than signal of interest.

The **polar plot** is the best way to show directional response. A polar plot is a graph showing the output of a hearing instrument, as a sound source of varying frequencies travels around the hearing aid's microphone in a full circle (from 0 to 360 degrees). It is represented as if looking directly from the top (the "north pole"). 0 is then directly to the front and 180 is directly to the rear. Various shapes or patterns (polar plots) can be achieved by a multi-microphone array: cardioid, hyper-cardioid, and super-cardioid, among others. See **figure 3** to the right for some examples of polar plots.

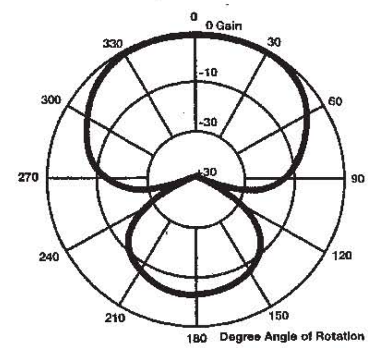
A **telecoil** is a type of input transducer worthy of mention. Many hearing impaired people find it difficult, if not impossible, to talk on the phone while wearing hearing instruments due to the feedback caused by the proximity of the microphone of the hearing instrument to the receiver of the telephone. A telecoil uses electromagnetic energy generated by all conventional land-line phones and turns it into an electrical signal the hearing aid can amplify.

Check Your Understanding

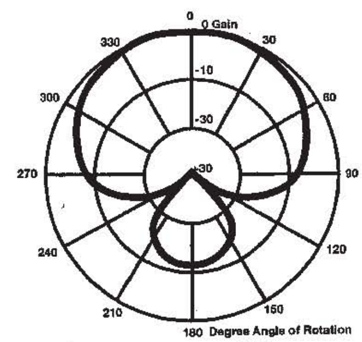
Signal of interest is 65 dB SPL and the noise in the room is 60 dB SPL; what is the SNR?



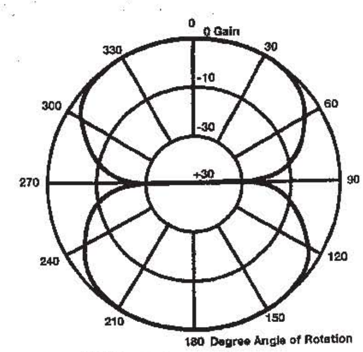
Polar plot showing cardioid response.



Polar plot showing hypercardioid response.



Polar plot showing supercardioid response.



Polar plot showing bidirectional response

Figure 3

Would you like to know more?

Notes

Omnidirectional

The omnidirectional microphone (**Figure 1** above) picks up sounds from all directions (omni) equally. It has one sound port and a diaphragm that moves back and forth in response to changes in air pressure across it. The diaphragm movement is converted to a proportional output voltage. *This microphone design provides the poorest performance in noise.*

Directional

The acoustic directional microphone (**Figure 2** above) has a front and back sound port leading to a separate small cavity, which is divided into two chambers by the diaphragm. The diaphragm senses differences in air pressure between the two chambers and converts these differences into an electrical output signal. In order to reduce the sensitivity to sounds coming from the rear, sounds from behind must be delayed for the time it takes them to travel from the rear port to the front port so they do not reach the diaphragm first. To produce delay, a fine mesh filter is added in the rear port. When sounds from both ports reach the diaphragm simultaneously, there is equal pressure on both sides of the diaphragm. This equal pressure results in sound cancellation for sounds from the rear because the diaphragm cannot move.

The effectiveness of a directional microphone and the pattern of directionality depend primarily upon:

- The distance between the microphone ports (external delay)
- The amount of time delay in the rear port (internal delay)
- The alignment of the ports on the hearing instrument faceplate/case.
- The reverberant ambient noise (room and environmental configuration)
- The size of the vent

Multiple Microphones

Directional performance from hearing instruments using multiple microphone systems will typically be better than from hearing aids using conventional single directional microphones.

Measuring Performance of Directional Microphones

There are several ways to measure and express the performance of directional microphones. Three of the most common measurements are polar sensitivity plots, directivity index (DI), and the articulation index/directivity index (AI-DI).

Directivity Index (DI) and Articulation Index Weighted Directivity Index (AIDI)

The DI represents the ratio of the microphone output for signals coming from 0 degrees azimuth (directly to the front) to sounds coming from all directions. It is expressed as a dB value and represents the calculated improvement in signal-to-noise ratio (SNR) provided by the microphone. The DI provides a good estimate of how helpful the directional microphone will be in difficult listening situations. The higher the DI, the easier it should be to hear and understand in difficult listening environments.

The AI-DI incorporates the Articulation Index as part of the DI. The Articulation Index is a calculation of the percentage of speech energy that is available for any given audiometric configuration that places increased emphasis on those frequencies that are critical to speech understanding in the form of "weightings". These frequency-weighting functions are combined with the DI to make up the AI-DI. Thus, if the DIs of two similar hearing instruments are close but one has a 2 dB higher DI at 1000 Hz and the second has a 2 dB higher DI at 2000 Hz, the latter hearing instrument will have the higher AI-DI because speech information at 2000 Hz is more important (has more "weight") than speech information at 1000 Hz. As with DIs, the higher the AIDI, the easier it should be to hear and understand speech in difficult listening environments. As one would suspect, the AI-DI seems to be a better overall indicator a hearing instrument's performance in noise than the DI alone.

Would you like to know more about SNR?

You may want to work through the following examples with your instructor.

Situation 1: Speech in Quiet

- Signal or message = 65 dB SPL, as in average conversational speech level.
- Noise or competition= 45 dB SPL, as in ambient room noise.
- Signal-to-Noise Ratio equals +20dB SNR.
- Maximum speech understanding (PB Max) would typically be easily obtained due to the high signal-to-noise ratio.

Situation 2: Speech in Noise

- Signal or message=65dB SPL
- Noise or competition=65dB SPL
- Signal-to-Noise Ratio equals 0dB SNR
- Speech understanding would typically be more difficult due to poorer signal-to-noise ratio.

Notes

Situation 3: Speech in high levels of noise

- Signal or message=70 dB SPL
- Noise or competition=85 dB SPL
- Signal-to-Noise Ratio equals –15 dB SNR

Speech understanding would typically be very difficult for people with normal hearing, and may be increasingly more difficult for hearing impaired due to extremely poor SNR. PB Max (i.e. the best performance for a word recognition test) is usually found for SNR >+5 dB for people with normal hearing.

Check Your Understanding

A SNR of 0dB is good for the best speech understanding.

True or False

Notes

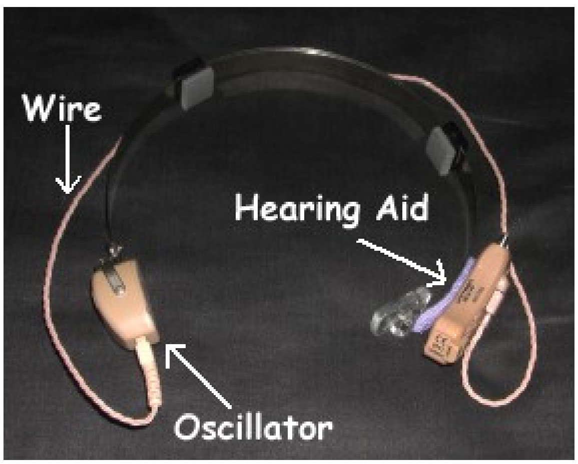
Receivers

Notes

Also known as **output transducers**, receivers are the loud speakers that provide the final or acoustic output of amplification. Receivers change amplified electrical signals coming from the amplifier back into an acoustic waveform ready for delivery into the ear canal.

Receiver Style.

Air conduction receivers also have a diaphragm but operate on a magnetic principle. Magnets in the speaker react to the amplified electrical current making the diaphragm move back and forth thus recreating acoustic sound much the same as the eardrum or the speakers of a stereo system. **Bone conduction receivers** vibrate a bone oscillator with the amplified sound sending it directly through the bones of the skull into the cochlea.



Bone Conduction Hearing Aid

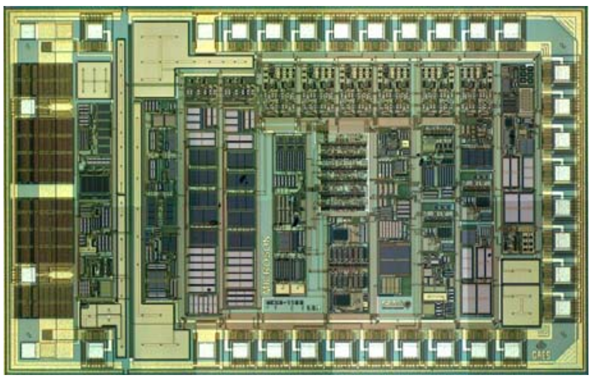
Notes

Amplifiers

Simply put, an amplifier boosts electrical signals that it receives from the microphone. Today's sophisticated amplifiers typically have three stages: a **preamp** that increases the weak electrical signal coming from the microphone, a **signal processing** stage, and an **output** stage. The amplifier is not a transducer since it is only boosting the electrical signal. An amplifier may be either analog or digital and supplies acoustic gain. **Acoustic gain** may be defined as "what is added to the input to achieve the output". The difference between the intensity of the sound entering the hearing aid (input) and the intensity of that sound as it leaves the hearing aid (output) is the gain. For example, if the input signal is 55 dB SPL and the output of the hearing aid is 85 dB SPL, the acoustic gain is 30 dB (85 minus 55 equals 30).

Integrated Circuit (IC) Design

A common design for integrated circuits (*abbreviated* ***IC***) is a circuit board on which an electrical pathway is etched. Transistors and resistors can then be wired into the electrical pathway. Integrated circuits consist of many transistors and resistors, and other electrical components on the circuit board. Both analog and digital amplifiers may use IC technology.



Hearing Aid Integrated Circuit

Analog amplifiers

Analog amplifiers reproduce output waveforms similar in shape to incoming electrical waveforms from the microphone. As sound pressures increase or decrease in the environment, the electrical waveform from the amplifier increases or decreases proportionately.

Digital amplifiers

Digital amplifiers require an analog-to-digital (A/D) converter to change the incoming electrical waveform into strings of mathematical bits.

Digital amplifiers employ a variety of algorithms to manipulate these bits of information at great speed, thus allowing for less internal noise and distortion, greater shaping flexibility, and the ability to perform noise suppression and feedback management. After processing, the digital waveform is changed back into an analog waveform via a digital-to-analog (D/A) converter.

Amplifier Signal Processing Stage

The second stage, the signal processing stage, is either linear or nonlinear. A linear signal processor provides constant (the same amount of) gain on an input/output graph until **saturation** (the intensity beyond which the system is incapable of providing further output, thus creating distortion). Non-linear signal processing typically applies reduced gain for increased inputs. Analog hearing aids can be either linear or non-linear.

Amplifier Output Stage

The output stage (3rd stage) of the amplifier is described as Class A, B, or D. Amplifiers are judged based on their distortion characteristics and effective use of battery current.

A **Class A** amplifier is simple and inefficient, since current flows through it even when there is no acoustic input. Output limitation is accomplished by "peak clipping" even as moderate inputs. For this reason distortion is nearly always present. These hearing aids, if used at all, are appropriate only for mild to moderate hearing losses.

A **Class B** amplifier combines two Class A amplifiers with switched polarities. One of the amplifiers "pushes" the compression phase of an acoustic signal and the other "pulls" the rarefaction phase. Hence, this circuit is known as a "push-pull" and is suitable for high-power applications. Because it comprises two Class A amplifiers, it is inefficient and very large.

Introduced in the late 1980's, the **Class D** amplifier (sometimes known as a "switching amplifier") is more efficient than the Class A and B because current only flows through the circuit when an input signal is present. The Class D is also superior since it does not peak-clip and produces a smooth output. Current digital hearing aids incorporate similar strategies in their signal processing.

Check Your Understanding

Which of the following is the input transducer of a hearing aid?

a. Amplifier

b. Receiver

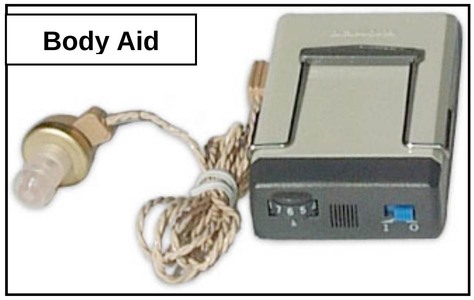
c. Integrated Circuit

d. Microphone

Notes

Hearing Aid Styles

Body aid


Body Aid

The body aid is a large non-custom style hearing aid that is typically worn in a pocket or special pouch that can be attached to a belt or other clothing. It is recommended for profound losses, especially for those cases in which there are severe physical limitations. The parts of the body style hearing aid are virtually the same as other hearing aids except for the receiver, which is external and worn at the ear. Until 1972 body aids comprised at least 10% of the hearing aid market, and prior to 1964 at least 20% of the market. At the writing of this chapter, few body aids persist in the market.

Eyeglass aids


Eyeglass Aids

In eyeglass aids, the components of the hearing aid lie in the bow of the eyeglass frame. Sound is led into the ear by a tube attached a custom earmold, or in some instances, by the tube alone. Eyeglass aids were at least 30% of the market until 1965, and at least 20% until 1975. Currently, eyeglass hearing aids are obsolete.

Behind the Ear (BTE)


BTE Aid

Also known as a post-auricular or over-the-ear (OTE) hearing system, the BTE is not a custom-fit hearing aid; however, the earmold coupling it to the ear is made-to-order. The BTE lies securely behind the auricle attached to the earmold by a plastic tip or ear hook and a length of tubing. The earmold rests in the concha and amplified sound is transferred into the ear canal via the sound bore. From 1966 to 1985 BTEs' were the hearing aids-of-choice by hearing healthcare professionals, achieving as much as 50% of market share. The versatile BTE, coupled with the appropriate custom earmold, can be used to fit nearly all hearing loss configurations, from mild to high-frequency ski-slope to profound.

Notes

Receiver-in-the-Canal (RIC) and Receiver-in-the-Aid (RITA)

Since 2004, a new style of hearing aid, commonly called **open-canal**, has gained great popularity. They are called "open canal" because they are frequently fit using non-custom ear couplings (no custom earmold) with essentially open ear canals (domes or tips). This style of hearing aid became possible with the introduction of digital signal processing and, specifically, digital feedback suppression. Because these hearing aid styles are not always fit with the ear canal open, the terms RIC and RITA are preferred descriptors.

The RITA style is essentially a mini-BTE with a very slender tube, called a slim-tube. All electronic components of the RITA reside in the case behind the ear. This hearing aid is designed for precipitous high-frequency hearing losses.

RITA

The RIC style looks similar to the RITA, however rather than an open tube, the RIC has a slender wire that connects to a receiver seated in the ear canal. Again, this aid was designed for high-frequency hearing losses, but if used with a custom earmold, may be appropriate for hearing loss configurations comprising poor low frequency thresholds as well.

In the Ear (ITE)

The in-the-ear model (full shell) fills the entire concha bowl and is a custom-fit instrument. The primary intended fitting range is for ski slope through moderately severe loss. There are other variations of the custom ITE. The half-shell or semi-canal covers the lower part of the concha. The canal ITE (or ITC) covers the **cavum** to the ear canal. The primary intended fitting range for ITC instruments is for mild through moderate losses. The CIC instrument fits entirely within the ear canal. The primary intended fitting range is for mild to moderate. The even smaller **invisible in-the-canal** *(abbreviated **IIC**)* is truly invisible, with the faceplate 2-4 mm beyond the aperture of the ear canal.

Behind and In-the-ear style hearing aids

Check Your Understanding

Which of the following styles of hearing aids is currently obsolete?

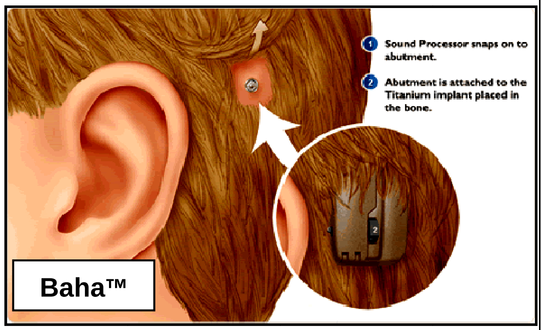
a. Receiver-in-the-ear

b. Body aid

c. Eyeglass hearing aid

d. Invisible-in-the-canal

Bone Conduction Hearing Aid



In some cases, use of a standard hearing aid may be precluded. For example, if a person has **atresia,** meaning there is no ear canal, or if the patient has a chronically draining ear and the physician will not clear the patient to wear standard hearing aids, and assuming the bone conduction thresholds are no poorer that 40 dB HL, a bone conduction hearing aid may be considered. Standard bone conduction hearing aids have a specially modified BTE hearing aid on a headband driving a bone oscillator (similar to that used in BC testing) that is placed behind the pinna. Another kind of bone conduction hearing aid is the Baha™, which is surgically implanted. The Baha™ may also be used for single-sided deafness, transmitting sounds from the dead-ear side, to the better ear via bone conduction.

Assistive Listening Device (ALD)

ALDs are electronic devices that work either in combination with or in lieu of a traditional hearing aid. There are several different types of ALDs on the market. Some are used for the telephone or watching TV. Want more information? Ask your facilitator.

There are four classes of hearing aids

- Class I – Single Channel and single memory
- Class II – Single channel and multi memory
- Class III – Multi channel and single memory
- Class IV – Multi channel and multi memory

Notes

Check Your Understanding

Most modern hearing aids process sound digitally.

True or False

Notes

Section 3: The Basics of Compression

Notes

Basic Amplification Terminologies

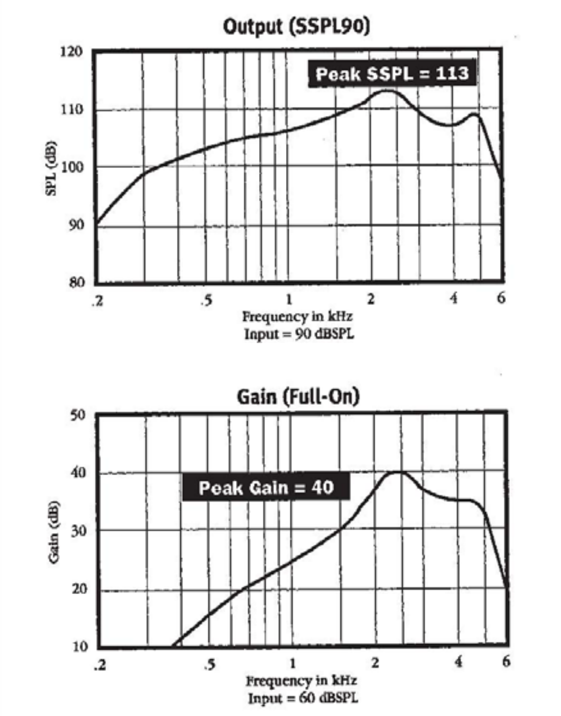
Hearing aid performance may be measured across several dimensions. You may even recall from the physics of sound section how the concepts of intensity and frequency relate to sound and to each other. Intensity is used to indicate how much energy a sound has. In the world of hearing instruments we typically talk of intensity in terms of **gain** and **output**. The amount of sound output the hearing aid produces in relation to the input is described by the term **compression**. Finally, the **frequency response** is defined by the amount of gain and output at each pitch or frequency.

Input: The intensity of the signal (in dB SPL) entering the hearing instrument microphone.

Output: The intensity of the signal (in dB SPL) that is delivered into the patient's ear after amplification.

Gain: Gain = Output – Input. Gain is the difference between the input signal level at the microphone and the output signal level at the receiver. It represents the amount of amplification provided by the hearing instrument.

Frequency Response refers both to the range of frequencies a hearing aid will amplify and how much amplification each frequency will receive. Most hearing aids have a measurable frequency response in the range between 200 Hz to 8000 Hz. A peaked response typically represents a poor quality sound hearing instrument. A smooth frequency response typically results in good sound quality.



Notes

Linear Amplification: Linear amplification occurs when there is a 1:1 relationship between the increase in the output relative to the increase in the input. For example, if the input signal increases by 10 dB, the output will also increase by 10 dB. A linear amplifier will work in this fashion until saturation is reached, at which point the output cannot be increased further. A linear amplifier provides constant gain until saturation occurs.

Peak Clipping: Used primarily in linear amplification, peak clipping effectively cuts off the peaks of high-level inputs either at the amplifier or receiver. While peak clipping provides a simple and effective way of protecting the wearer from sudden or ongoing loud sounds, it has the disadvantage of increased distortion above the limiting level that often results in compromised sound quality. This kind of processor is preferred for severe to profound losses as the intensity of the processor gives them speech and sound cues.

Compression: Occurs when amplification is reduced automatically in the presence of high-level input signals at some point prior to saturation. Imagine that there is a little person inside of the hearing instrument. When the input level at the microphone reaches or exceeds a predetermined level, the little person simply turns down the volume control, thereby effectively reducing gain and output of the hearing instrument.

Check Your Understanding

Output – Input =

a. Gain

a. Non-linear compression

b. Amplification

c. Saturation

Notes

The difference between linear and compression amplification can easily be seen in the comparative input/output curves shown in **Figure 5** and **11**. Figure 5 shows linear amplification for a 30 dB gain hearing instrument. In this figure, when the input increases from 60 dB to 70 dB, the output increases from 90 dB to 100 dB. A 10 dB increase in input = a 10 dB increase in output. Saturation occurs when the input reaches 90 dB SPL and the hearing aid can no longer maintain 30 dB of gain.

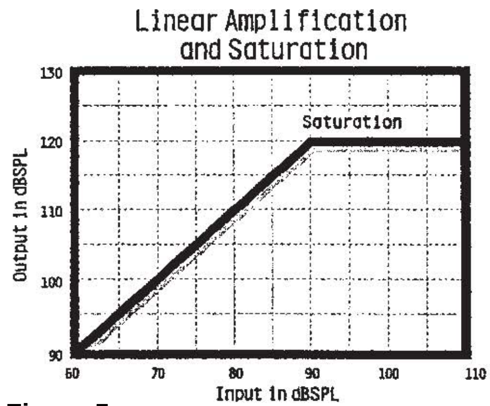


Figure 5

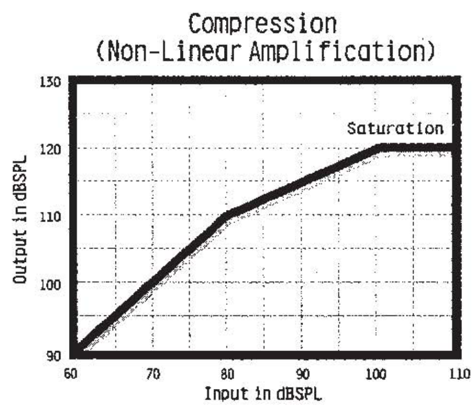


Figure 11

Figure 11 shows the input/output curve for compression in non-linear amplification. When the input level reaches 80 dB, the 1:1 relationship between input and output changes and the input/output curve no longer progresses at a 45° angle. While the input level increases from 80 dB to 90 dB, the output only increases from 110 dB to 115 dB SPL. A 10 dB change in input level results in a corresponding 5 dB change in output level (2:1), thereby reducing the gain from 30 to 25 dB. Whereas a peak clipping circuit provides a constant level of amplification until saturation occurs, a compression circuit will reduce the gain for loud sounds, avoiding saturation, minimizing distortion and preserving sound quality.

Compression Amplification Parameters

Notes

Compression systems are essentially defined by their parameters. These parameters are:

- Compression Threshold
- Compression Kneepoint or Threshold Kneepoint (CK, or TK)
- Compression Ratio (CR)
- Compression Range
- Attack and Release time.

Compression Threshold/Threshold Kneepoint (CK/TK)

Soft and many moderate-intensity sounds do not require compression. It is primarily loud sounds for which compression is necessary. The **compression kneepoint** (*abbreviated **CK***) is the intensity at which compression commences. Soft and moderate sounds may be processed in a linear fashion with little likelihood that the patient will find the resulting sounds uncomfortably loud. Hence, sounds that enter the hearing aid at and below the CK will receive full gain, and sounds entering the hearing aid at intensities greater than the CK will be compressed. CK is where the intensity at which amplification changes from linear (1:1) to non-linear (2:1, 3:1, 4:1 etc). See **Figure 13** for an example of Compression Kneepoint. Compression kneepoint affects gain for soft sounds in an inverse manner. ***When compression kneepoint is reduced, gain for soft sounds increases, and when compression kneepoint is increased, gain for soft sounds decreases.***

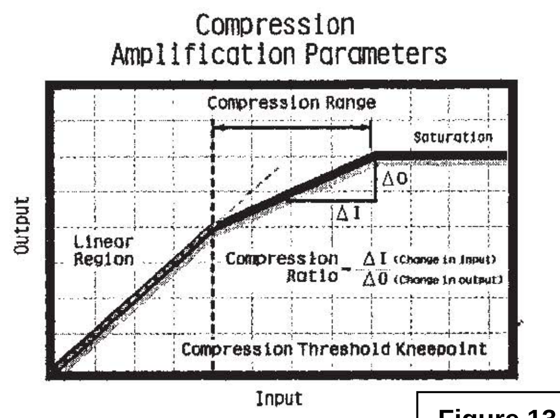


Figure 13

Compression Ratio (CR)

The **compression ratio** determines the amount of compression that will occur at the compression kneepoint (CK). It is the ratio of the change in input to the change in output. If the input at the microphone increases by 10 dB and the output at the receiver increases by 5 dB, then 10/5 = a compression ratio (CR) of 2:1. If the input increases by 15 dB and the output by 5 dB, then 15/5 = a CR of 3:1. If the input increases by 20 dB and the output by 5 dB, then 20/5 = a CR of 4:1. Simple, no?

Compression Range

The **compression range** is the range of input levels over which the compression amplifier operates. This would be above the linear range and below the saturation point. See **Figure 12** for an example.

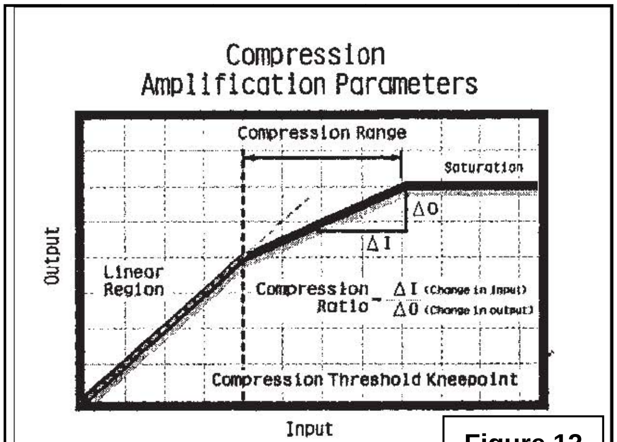


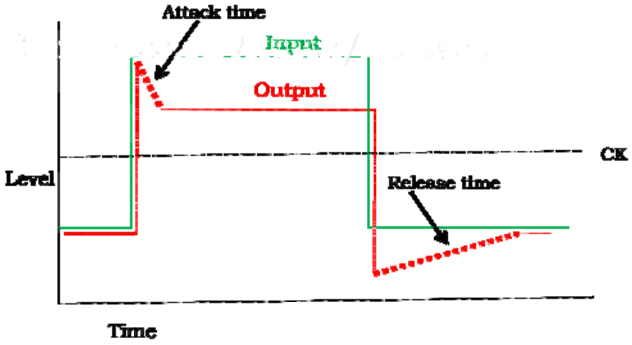
Figure 12

Attack and Release Time

Compression is a time-dependent form of gain regulation. There is a time delay from when sound triggers the compression until the hearing instrument gain stabilizes at its compressed level. This time period is called the **attack time.**

It also takes some time for the hearing instrument to restore the gain back to the linear or pre-compression level after the triggering sound is no longer present. This time period is called the **release** or **recovery time.**

While it is always preferable to have a fast attack time in response to a triggering sound input, this is not the case with the release time. Slower release times are needed for sound inputs that are ongoing. Faster release times are preferred for sounds that start and end quickly. Inappropriately fixed fast release times can cause gain fluctuations and a “pumping” or “fluttering” sound that is noticeable to the wearer. Conversely, inappropriately slow release times restore gain too gradually creating “dead spots” which are equally noticeable to the wearer. Neither situation is ideal. Today’s best hearing instruments have built in logic that allows them to react in accordance with the input sound environment. They are designed to release slowly when noise is constant and more quickly in environments with brief sporadic noise levels.



Notes

Input and Output Compression

Notes

A compression amplifier that has a feedback sampling loop, a level detection device, and a variable gain amplifier effectively samples incoming sound levels passing them through a level detection device. Below a certain level, the amplifier supplies linear gain. Above that same level, the variable gain amplifier goes into compression and reduces gain by a determined amount.

The position of the feedback sampling loop relative to the volume control will influence circuit behavior. If the feedback sampling is situated before the volume control, it is an input compression system. Conversely, if the feedback sampling is located after the volume control, it is an output compression system. **Figures 14** and **15** below show simple hearing instrument circuit designs for input and output compression systems.

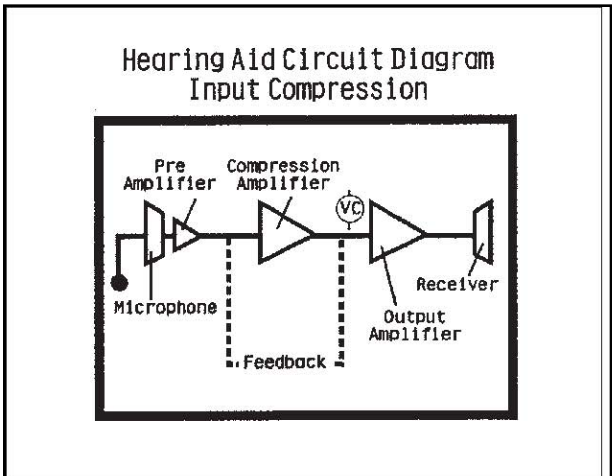


Figure 14

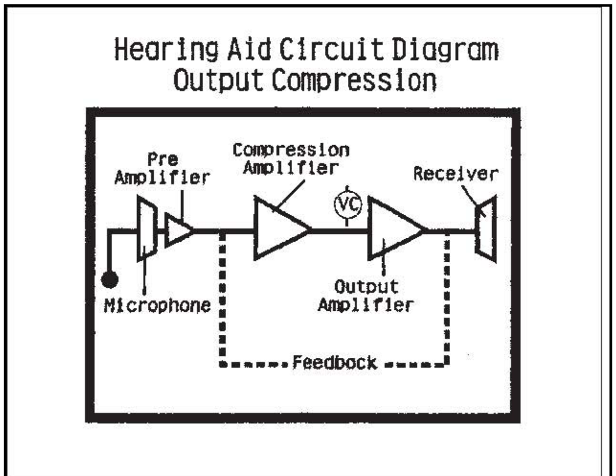


Figure 15

The position of the feedback sampling impacts the effect that the volume control will have on Compression Kneepoint (CK), gain and output.

With **input compression**, the CK will remain the same at all times. Changes in the volume control will affect both the gain and the maximum output. With **output compression**, when the volume control is adjusted both gain and the CK will change, but the maximum output will remain constant.

Check Your Understanding

When compression kneepoint is adjusted from 45 dB SPL to 65 dB SPL, which of the following occurs?

a. Gain for soft sounds increases

b. Gain for loud sounds increases

c. Gain for soft sounds decreases

d. Gain for loud sounds decreases

Now that we have reviewed the different characteristics of compression amplifiers, we are ready to tackle a more detailed discussion of how these characteristics can be configured to achieve different types of compression circuitry and how these compression types can be applied to hearing instrument fittings. **Figures 16** and **17** show input/output functions for input compression and output compression respectively.

Notes

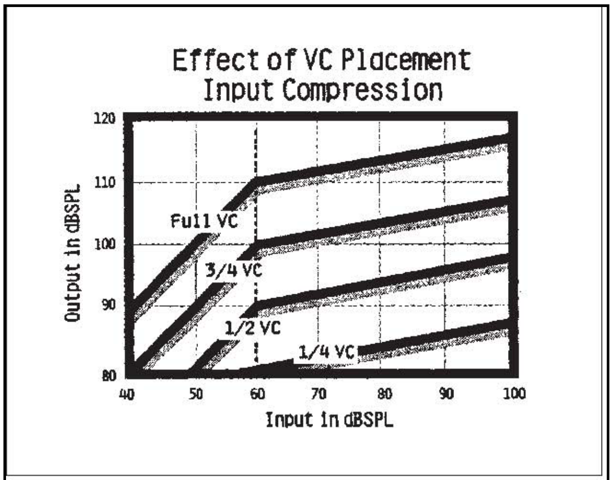


Figure 16

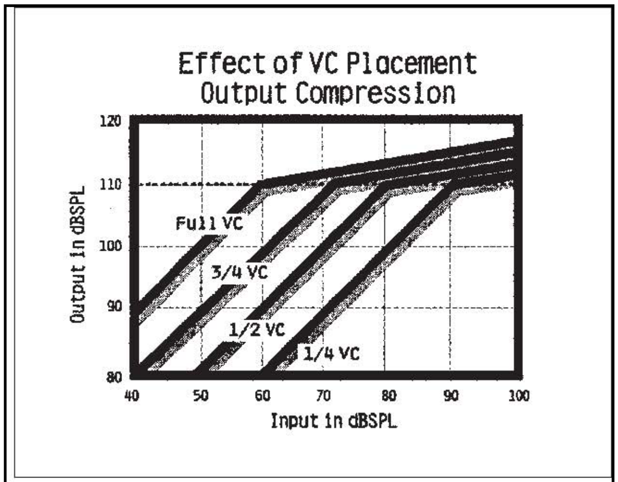


Figure 17

Section 4: Types of Compression Circuitry

Notes

Compression circuitry classifications depend on compression parameter configurations and the intended function or fitting objective. The two most common broad categories of compression are:

- Compression Limiting
- Wide Dynamic Range Compression

Compression Limiting

The objective of Compression Limiting (also known as high-level compression) is output limitation without amplifier saturation. Specifically, a Compression Limiting hearing instrument is designed to:

- Prevent the output from reaching a preset level.
- Minimize the distortion that occurs at moderate to high input levels.
- Operate linearly for a wide range of input levels.

In order to accomplish these things, compression parameters include:

- A high compression threshold/kneepoint (CK) (above 70 dB SPL)
- An aggressive compression ratio (as much as 10:1)
- A short attack time (2 to 10 ms)
- A short release time (100ms or less)

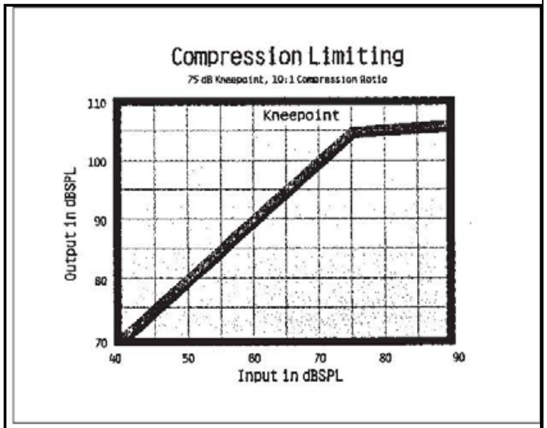


Figure 18

Figure 18 shows the input/output curve of an instrument set to Compression Limiting (TK 75 dB SPL, CR 10:1). When there are high kneepoint levels and very high compression ratios, the input/output curves resemble those from linear Peak Clipping instruments.

Notes

Compression limiting in hearing instruments is not new by any means. It has been available for many years. The primary advantage of a compression limiting hearing instrument over a peak clipping one is the way in which high intensity sound input is processed. By utilizing compression limiting the system avoids amplifier saturation and reduces distortion levels.

Compression limiting is the compression type of choice for patients with severe hearing impairment. Compression limiting type fittings can also be effective for those patients who find the inevitable distortion peak clipping type instruments provide in louder listening environments to be problematic. It can also be effective for those patients who are bothered by or who have a reduced tolerance for sudden loud sounds. Patients who can hear soft or average sound levels well enough, but complain that loud sounds are too loud and/or distorted, may be good candidates for compression limiting type settings as well. Compression limiting works in the background of most WDRC (see next section) hearing aids by limiting the MPO of these circuits.

Wide Dynamic Range Compression

The objective of Wide Dynamic Range Compression (*abbreviated* ***WDRC***) is to amplify the full range of speech within the hearing instrument wearers' reduced dynamic range or area of processing. The goal of WDRC is to

- Make inaudible sounds audible
- Make soft to moderate sounds (i.e. speech) more clear and distinct for better understanding
- Keep loud sounds loud but tolerable

Once these things are done, the patient can move from quiet listening environments to noisier ones with ease and greater confidence.

With WDRC type hearing instruments the need for volume control adjustment is minimized by:

- Using a low compression threshold kneepoint (less than 65 dB SPL)
- Using a low compression ratio (less than 4:1)
- Using a short attack and longer release times

Notes

Almost all digital hearing instruments utilize variable time constants and have WDRC capability. At WDRC settings (low CK and low CR) the compression operates over a wide range of input signals, providing maximum gain for soft sounds and minimal gain (comfort) for loud sounds. **Figure 19** shows the input output curve of a Wide Dynamic Range Compression Hearing Instrument with a 55 dB SPL compression kneepoint (CK) and a compression ratio (CR) of 2:1.

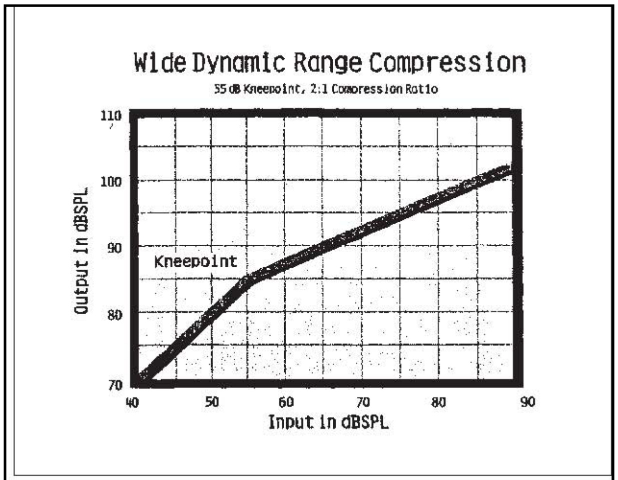


Figure 19

Hearing instruments with parameters set to Wide Dynamic Range Compression (WDRC) are typically best-suited for persons with mild to moderate hearing losses. Persons with moderately severe hearing losses (55-70 dB HL) may also benefit from this compression type. The following should be considered as good candidates for Wide Dynamic Range Compression settings.

- Persons who are frequently in changing listening environments, such as restaurants, places of worship, and small groups.
- Current wearers who complain that the volume of their hearing instruments are too low to hear soft or moderate sounds, or that conversational level sounds are OK, but soft sounds are too soft and loud sounds are too loud.
- “Nervous” new hearing instrument wearers who have been hesitant to try hearing aids because they are concerned that all sounds will be made too loud.
- Persons with reduced dynamic ranges (residual area of processing) (i.e., increased sensitivity to sudden changes in loudness)
- Persons who have very particular listening needs, such as musicians or singers.

Notes

As a recap, this table quickly shows the contrasts between Compression Limiting and Wide Dynamic Range Compression (WDRC) parameter settings:

Compression Limiting	**Wide Dynamic Range** Compression
High Kneepoint (CK > 70)	Low Kneepoint (TK < 65)
High Compression Ratio (CR>4:1)	Low Compression Ratio (CR < 4:1)
Variable time constraints	Variable time constraints

Multiple Channel Compression

Multiple channel compression hearing instruments are those that split the input signals into more than one frequency channel, processing each independently, and then recombining them before the amplification output stage. This allows the compression characteristics or parameters to be fine-tuned for different frequency regions. Frequency regions are simply bandwidths separated by a **crossover frequency**. Crossover frequencies are based on natural breaks in the audiometric configuration.

Multiple channel compression hearing instruments allow increased flexibility for tailoring the instrument's response and compression parameters to match the wearer's hearing requirements. For example, individuals with a steeply sloping sensorineural hearing loss will most likely have different **dynamic ranges** (i.e., the difference in dB from the softest sounds they can detect to the loudest sounds they can tolerate) for low and high frequency sound inputs. Multiple channel compression hearing instruments allow you to more effectively adjust the frequency response to provide audibility of high frequency speech sounds, while at the same time providing effective reduction of low frequency competing sound signals.

Level Dependent Frequency Response (BILL, TILL, PILL)

The term Level Dependent Frequency Response (LDFR) describes circuitry that uses filtering characteristics to provide a different frequency response for different sound input levels (soft, normal or average, or loud). There are three categories of LDFR circuits. They are BILL (Bass Increases at Low Levels), TILL (Treble Increases at Low Levels) and PILL (a Programmable combination of TILL and BILL…I know, I know! This profession loves its acronyms!) BILL and TILL are two different compression philosophies. See **Figures 20** and **21** below.

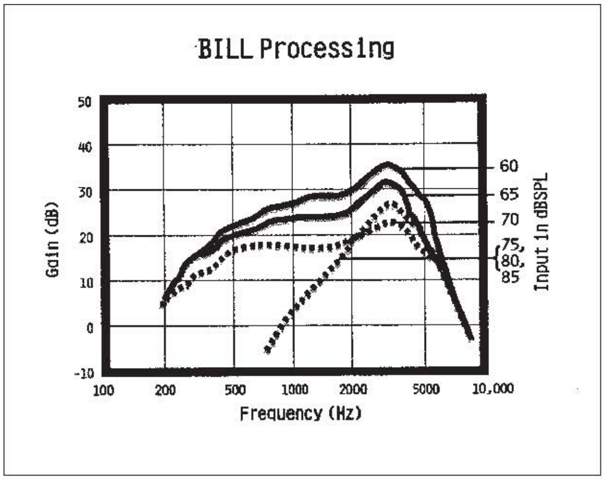


Figure 20

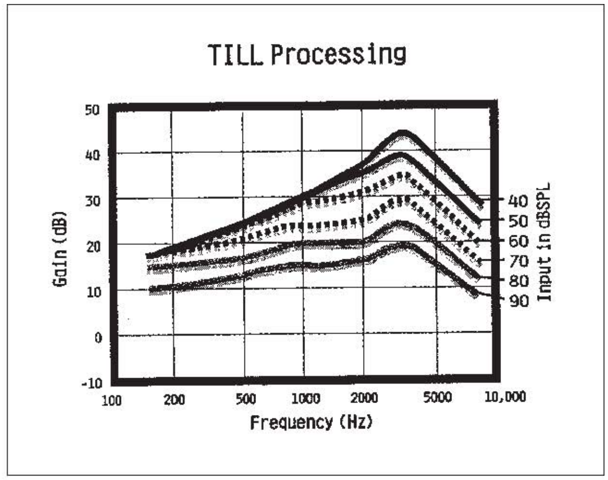


Figure 21

BILL (No, not that Bill…)

BILL circuits or circuit parameters are designed to provide a broad flat frequency response in quiet environments (low sound input levels) with automatic low frequency suppression of the amplified signal as the sound environment changes. Even though the name of the circuit refers to soft bass sounds being increased, the relevant effect is that ***loud** bass sounds are reduced.* The louder the sound environment becomes, the less low frequency gain is provided. This approach to sound management maintains sufficient high-frequency gain to provide improved audibility of high frequency speech information, while decreasing the amount of low frequency gain (where noise usually lives) as sounds levels increase. In other words, BILL settings are intended to provide good sound quality in quiet environments and improved listening in noisier environments by filtering out low frequency signals. BILL circuits or circuit parameters are typically recommended for flat or gradually sloping, mild-to-moderate sensorineural hearing losses.

TILL

TILL circuits operate very differently from BILL circuits. They are designed to provide a flat frequency response with a minimal amount of gain for loud sound environments and a high frequency emphasis with more gain for quieter environments. This design is intended to help ensure audibility of soft high frequency sounds while maintaining comfort for loud sounds (again, the take-home point for TILL circuits is that loud high frequency sounds are reduced). The rationale for TILL is based on the tenet that to not amplify loud sounds at all is better than to suppress them. TILL views suppression as kind of like throwing the baby out with the bath water…when you suppress noise, you suppress some important speech information as well.

Like BILL circuits, TILL circuits are typically recommended for mild to moderate gradually sloping hearing losses.

PILL

A PILL circuit (the "P" stands for programmable) is one that can be made into a BILL, TILL, neither BILL nor TILL, or a customized hybrid, by using multiple channel compression hearing instruments and employing unique compression settings for each instrument. The fitter has a great deal of flexibility with these types of hearing instrument.

Check Your Understanding

Input compression, the goal of which is to make soft sounds audible, loud sounds loud but OK and moderate sounds comfortable, is called which of the following?

a. Wide dynamic range compression

b. BILL and TILL response

c. Output limiting compression

Notes

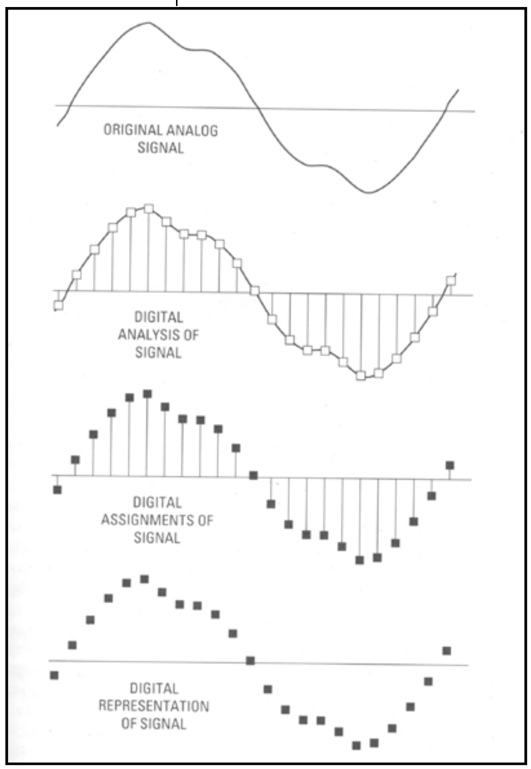
Advanced Features

Notes

It is digital (as opposed to analog) processing that allows us to dispense hearing aids with advanced features.

Analog refers to incoming sound being processed by a hearing aid as an analogous electrical current or signal.

Digital refers to incoming sound processed by a hearing aid through an analog-to-digital converter as binary code (1's and 0's i.e.-10101).The more quickly the incoming signal is processed, the faster the conversion, the more exact the binary code.



Advanced features are the heart and soul of the new modern hearing aid. Every manufacturer has unique names for the individual feature. The operation of the particular feature is in general the same, however. Here are few of these features :

Directional Microphones: Explained earlier in the unit

Expansion: The opposite of compression. This looks at soft input signals (refrigerator compressors, for example) and does not amplify them. This is usually a non-adjustable feature the dispenser has no control over (probably a good thing).

Environmental Classifier: This is the brains of the circuit. It determines the acoustic environment in which the hearing aid is operating (speech-in-quiet, noise, or music, for example). Based on this determination, the environmental classifier can then direct the hearing aid to employ a specific set of hearing aid responses best able to process that kind of sound.

Wide Dynamic Range Compression (WDRC): This is the input control of the circuit. It is designed to take all environmental sounds and fit them into the residual area of processing of the patient. This where the use of CK/CR (which were described earlier on) are employed.

Digital Noise Reduction (DNR): DNR is an algorithm that makes a determination whether the sound coming into a particular channel is likely to be speech (usually defined as a signal that changes frequently in frequency and amplitude) or noise (steady-state). In channels where noise is the predominant signal, gain can be reduced. This allows for greater comfort for the hearing aid wearer but in most cases has *not* been proven to improve speech understanding in noise.

The Hearing Aid Delivery

Notes

The delivery (or fitting) of hearing aids is a systematic process. If you carefully follow the steps outlined below, you will be better able to fit and counsel your patients during a 60 to 90 minute appointment.

Step 1: Before the patient arrives for appointment you should have run an ANSI test on the hearing instruments using the 2cc coupler to ensure the hearing instruments are performing to ANSI specifications as stated by the manufacturer.

Step 2: You should have also pre-programmed the hearing instruments via manufacturer's fitting software. To do this you will enter the patient's audiogram and follow the instructions outlined for you in Unit 10 of this manual.

Step 3: When the patient arrives make sure the hearing instruments are turned off before inserting them into the ear and checking the fit of the each instrument in the ear. By placing them in the patient's ear and carefully examining the way that they insert, sit, and how they look you will ensure that the hearing aids are neither too loose nor too tight.

Step 4: You must re-introduce realistic expectations at the fitting just as you did during the sales process. Take care to mention such universal side effects as speech in quiet vs. speech in noise and the occlusion (My voice sounds funny!) effect.

Step 5: Verify the performance of the hearing aids using probe microphone real-ear measurement (REM). If probe tube verification is not possible, verify via sound-field testing. For appropriate steps to sound field and probe microphone verification process please review the following section.

Step 6: Counsel or inform the patients regarding the performance they can reasonably expect for the first week (be sure to include a wearing schedule for first time wearers), hearing instrument insertion/removal, battery insertion/removal, care and use. This is a very important step requiring at least 30 minutes or more of your undivided attention.

Step 7: Schedule an appointment for the patient to return for a hearing aid check in one week to two weeks.

Step 8: Follow-up the delivery with a call to see how the patient is doing and whether or not they are wearing their hearing instruments. This should be done within 24 – 48 hours of delivery.

The Speech Index

The Audibility Index (AI), or Speech Index, is a result of studies that place speech sounds overlaid on an audiogram utilizing 100 dots that represent average intensity speech. A sample count-the-dots audiogram is included at the end of this unit .This can be helpful when verifying the hearing aid fit during the delivery or subsequent adjustment period.

The dynamic range for average intensity speech is about 30dB, falling between 20dB HL for the just audible parts of average, and 50 dB HL for the louder parts of average. The Speech Index can be used in verification to show the patient how many speech sounds the hearing aid is making audible, thus the term “audibility index.” The Speech Index is not the same as a word recognition score.

Follow-up and After-Care

A successful hearing care professional understands that hearing aid use begins with the follow-up. There is really no substitution for professional follow-up care. This begins the moment the patient walks out the door with hearing aids. In fact, it is considered best practice to call your patient the day after the fitting to see how they are doing (24 to 48 hour phone call). This reinforces to the patient the notion that you are interested in their success with hearing aids as well as providing an opportunity to address any potential problems at an early stage before they become serious issues.

After the hearing instruments have been delivered and verified, we begin post-fitting care. A synonymous term might be aural rehabilitation. Aural means *ear*, rehabilitation means, well, rehabilitation. Part of aural rehabilitation involves the hearing aid. The hearing aid may not be the only part of helping the hearing impaired. A hearing impairment has multi-faceted effects on the user. Rehabilitation may involve other aspects of life: medical, social, economic, psychological, biological.

While a successful hearing instrument fitting may be the most important factor in rehabilitation, total rehabilitation of the patient may be multifaceted.

- It may involve the family physician or ENT and the audiologist if the case history suggests progression of a medical condition of known or unknown symptoms.

- Family counseling and involvement of the family is important in that the family needs to support the efforts of use and care of the fitting, and to properly manage expectations of the patient and the entire family.

- Psychologically, the patient must learn to live with the limitations their hearing impairment. In extreme cases, a psychologist may help.

- Genetic counseling may be indicated for congenital issues.

Notes

During post-fitting aural rehabilitation, you will continue to counsel the patient on maintenance of the hearing aid as well as to continually monitor (verify) the fit of the hearing aids by using both subjective (subjective validation measures such as COSI) and objective measures (ANSI and real-ear measurement) to make sure aural rehabilitation is successful. You will also monitor the changes in hearing status at least annually.

Maintenance, Modification and Repair

Hearing aids are electro-acoustical devices. They have an electrical component consisting of the electrical parts of the hearing aid. They have an acoustical component consisting of the shell of the hearing aid, its depth in the ear, and the vent connecting the tip end of the aid to the faceplate end. Maintenance of the hearing aid is provided during delivery and subsequent follow-up visits. The patient and professional both have maintenance responsibilities.

In the area of modification, let’s understand some of the most common user complaints: poor fit, feedback, occlusion and poor speech understanding in quiet and in noise. For each of these, decide whether the most likely source of the complaint is electrical (coming from the electronics of the hearing aid/frequency response) or from the acoustical shell or vent. In some instances, the complaint may be both. Ask your facilitator for additional information.

There is no substitute for hands-on practice when it comes to successfully performing these procedures. Take the time to learn these skills from your manager, supervisor or colleague.

Check Your Understanding

1. Directional microphones:

   a. Greatly reduce background noise

   b. Increase signal-to-noise ratio

   c. Can make changes in signal processing at great speed

   d. All of the above

2. The most appropriate fitting use for a bone-conduction aid would be:

   a. A 70-year old with Meniere's disease

   b. Complete bilateral Atresia with measurable cochlear sensitivity

   c. Profound sensorineural loss, bilaterally

   d. A recovering cochlear implant patient

3. (ANSI) Total Harmonic Distortion of a hearing aid is measured using inputs of:

   a. 500, 1000 and 2000 Hz at 65 dB

   b. 70 dB at 500 and 800 Hz and 65 dB at 1600 Hz at full-on gain

   c. 70 dB at 500 and 800 Hz and 65 dB at 1600 Hz at RTP

   d. OSPL 90 minus 77 dB

4. A primary consideration for recommendation of a non-linear hearing aid fitting is:

   a. Dynamic range less than 40 dB

   b. Conductive Hearing Loss

   c. Elevated UCL

   d. Central hearing loss

5. Real-ear measurement terms ending in “G” (REIG, for example) mean: (select all that apply)

   a. Input was subtracted from output to obtain the gain

   b. The answer refers to gain produced by the hearing aid

   c. The response is in dB SPL measured at the eardrum.

6. Functional gain is obtained when: (select all that apply)

   a. Probe-tube measures are used

   b. Soundfield speakers are used

   c. 2cc coupler gain is added as a correction factor

7. A matrix of 115/45/25 would be most appropriate for:

   a. A mild hearing loss with a speech UCL of 85

   b. A moderate hearing loss with a speech UCL of 90

   c. A moderately severe loss with a speech UCL of 95

   d. A severe loss with a speech UCL of 100

8. If the speech UCL is 105 dB SPL, the most appropriate matrix is:

   a. 105/25/25

   b. 110/35/25

   c. 115/40/20

   d. 125/50/10

9. Which of the following is not true?

   a. Output minus gain equals input

   b. Input plus gain equals output

   c. Output minus input equals gain

   d. Output plus gain equals input

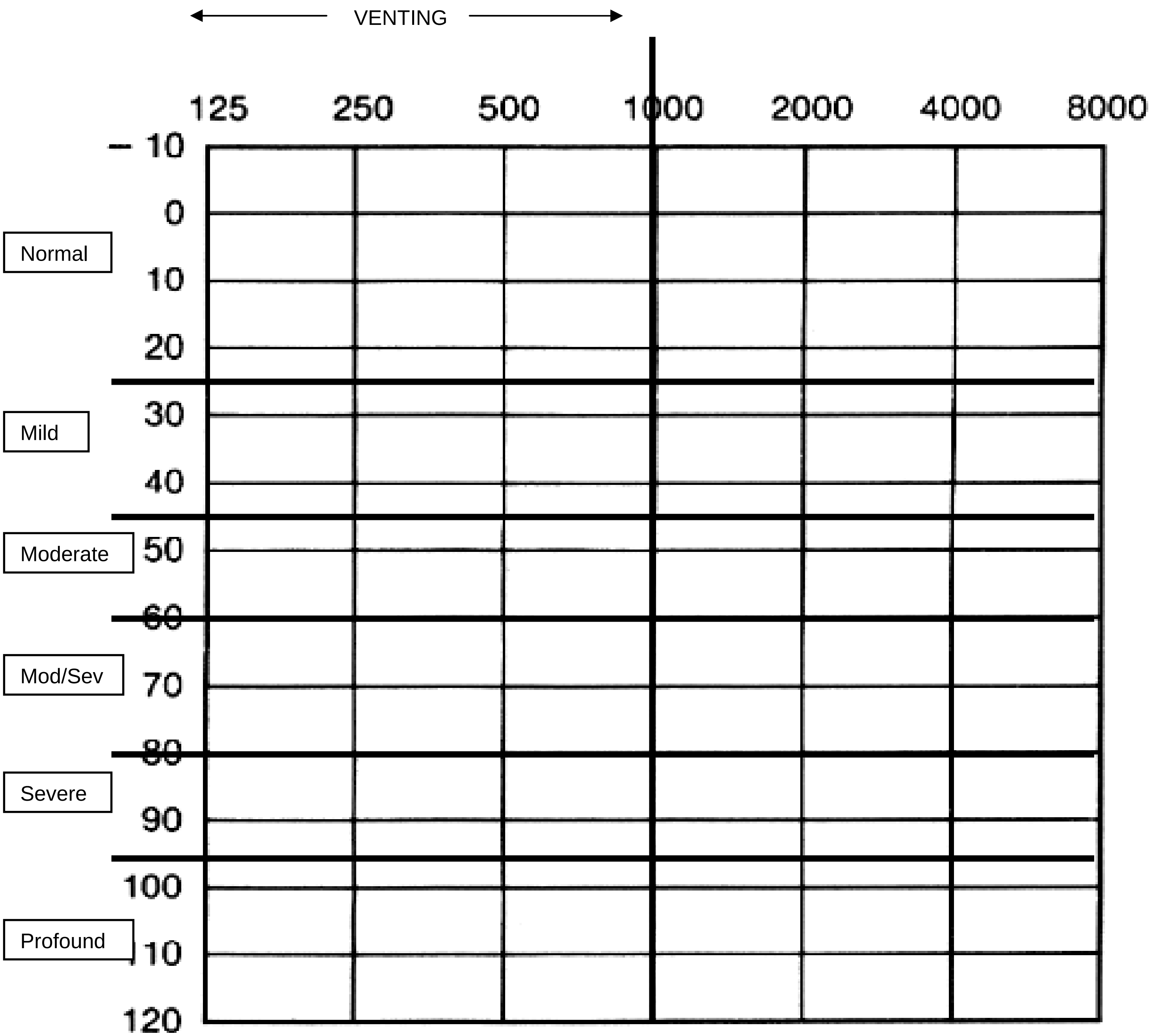
10. Patient B has a PTA of 80 dB in both ears. The fit most likely to be successful is: (select all that apply)

    a. Input compression in both ears

    b. Output compression in both ears

    c. Linear fit in the one ear with the best word recognition

Earmold and Venting Chart



Symbols Used in Hearing Aid Block Diagrams

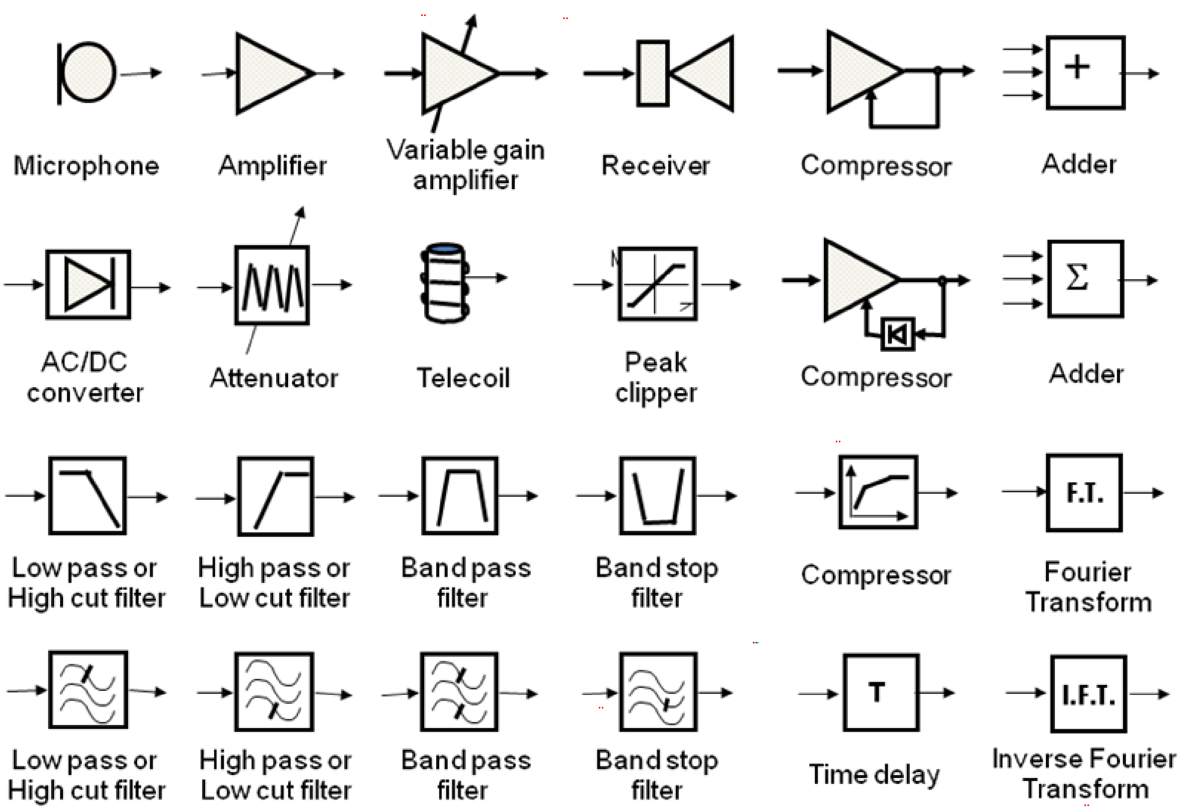


Figure 2.1 Symbols used in block diagrams.

Source: Dillon (2001): *Hearing Aids*

Unit 9: Basic Audiometric Interpretation

Objectives

Notes

Overview

The audiogram is the graph used to chart both the type and degree of hearing loss for every patient. In short, the audiogram represents the bread and butter of our profession because virtually every patient you will see requires a hearing test. When you do a hearing test, you will chart the outcome on an audiogram. This unit is designed to familiarize the reader with some basic concepts relating to the basic hearing test.

Goals

Upon completion of this module, you will be able to:

- Know what a complete audiogram (hearing test) looks like
- Read and interpret a basic audiogram
- Use this information to select hearing aids and counsel patients

Follow Up

- Northern, J. (1996). In *The Audiometric Evaluation* (p. Chapter 4). Boston, MA: Allyn and Bacon.

Section 1 – Introduction to the Audiogram

Notes

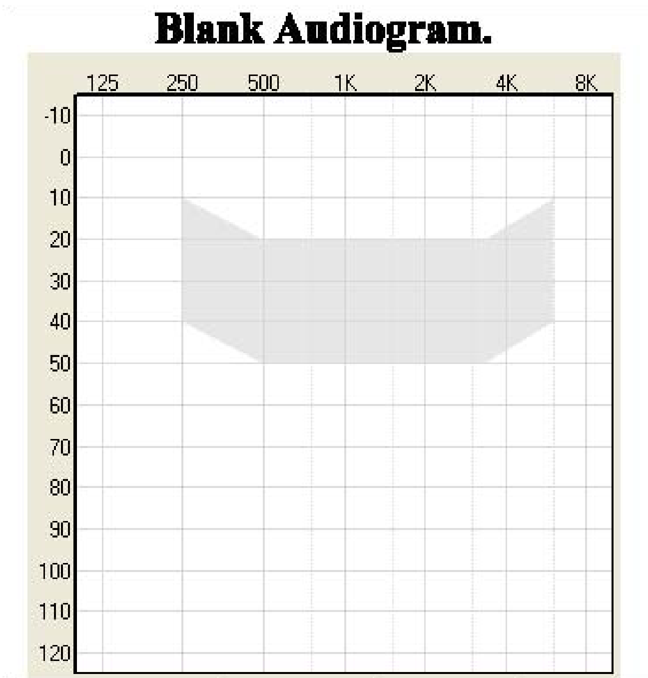
When we test someone's hearing there are two things we are interested in finding:

1. **How much hearing loss does the patient have?** This is commonly referred to as the degree of hearing loss. The degree of hearing loss is determined by finding a threshold. The definition of threshold is the softest sound to which a person responds 50% of the time when the sound is present. For those of you who are not technically minded that's really soft!

2. **What type of hearing loss does the patient have?** This is determined by looking at the relationship of the various symbols you place on the audiogram and the overall pattern of the symbols of the audiogram.

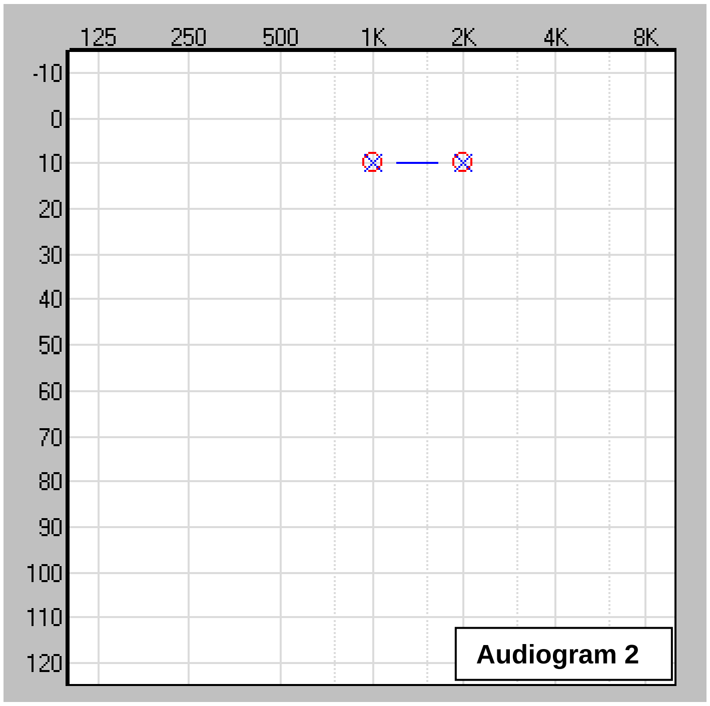
So there you have it, the audiogram tells us the type and degree of hearing loss.

There are a few key things you need to know about the blank audiogram, an example of which is shown to the right. Along the top horizontal line you will see the frequency of the sound presented to the patient. Frequency is described in Hertz *(abbreviated* ***Hz****)*. During the hearing test, the tester usually presents tones at six different frequencies (250, 500, 1000, 2000, 4000 and 8000 Hz. 125 Hz is usually not tested). Now look vertically along the left hand side. This is the intensity scale we use called **dB HL**. The further down the vertical line, the larger the dB HL number. The larger the dB HL number, the louder the sound being presented to the patient. For more information on frequency and loudness, please refer to the Unit 3: A (Painless) Introduction to Acoustics (Physics of Sound)



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

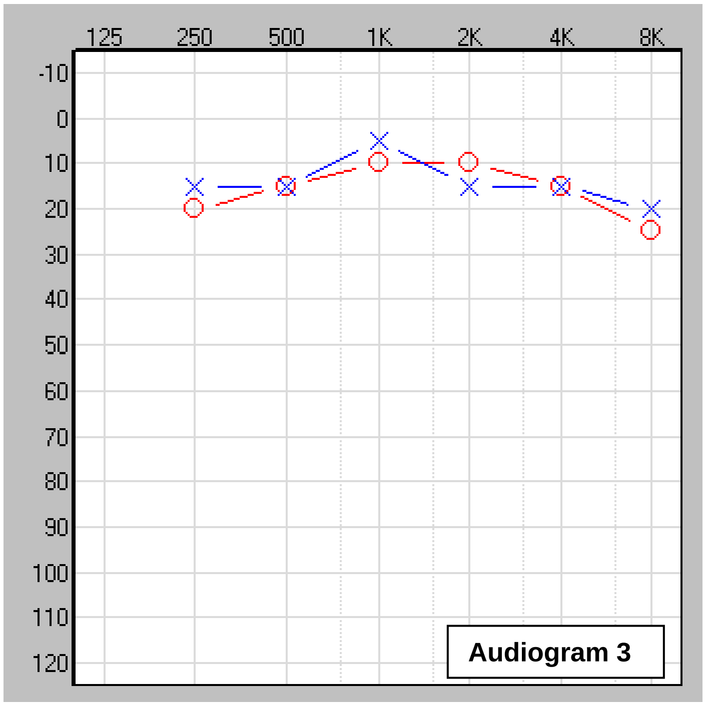
Notes



Audiogram 2 (left) shows a partially completed audiogram, for the left and right ears respectively, for someone with normal hearing. The left ear is represented by the **X** and the right ear is represented by the **O**. The symbols represented on Audiogram 2 are showing you where the patient's threshold is at two different frequencies. Threshold refers to the softest sounds a person can hear half the time. For example, a person with normal hearing has thresholds between –10 and 25 dB HL. In Audiogram 2, the patient can hear a tone 50% of the time when it is presented at 10dB HL at each frequency. In this example the threshold was determined at just 1000 and 2000 Hz for the left and right ears respectively.

The signals used for completing an audiogram are **pure tones**. For a complete discussion of pure tones, please refer to Unit 3: A (Painless) Introduction to Acoustics (Physics of Sound)

Let's review. **Threshold** is defined as the level at which the tone is so soft that it can only be perceived half of the time it is presented. Thresholds are classified on a deciBel scale.



Audiogram 3 (left) represents a completed **air conduction** (*abbreviated **AC***) audiogram for both the left and the right ears. In this example, the patient has thresholds in the normal range for all six of the frequencies tested. You might be wondering, what is air conduction? Remember from the earlier section on physics that when sound travels from the outer ear through the middle ear to the inner ear, it is referred to as an air conducted signal. When we place headphones or inserts on a patient we are performing an air conduction test. The results of the air conduction threshold test are represented by the X's and O's on the audiogram.

Degree of Hearing Loss

From the air conduction threshold we can determine the degree (or amount) of hearing loss. Using the dB scale on the left hand side of the graph, you can determine the degree of hearing loss. Table A outlines the various degrees of hearing loss classification.

Table A	
Normal: 0 to 25dB	**Moderate/Severe:** 56 to 70dB
Mild: 26 to 40 dB	**Severe:** 71 to 90dB
Moderate: 41 to 55dB	**Profound:** 91dB or greater

Check Your Understanding

On an audiogram, what do the following symbols represent?

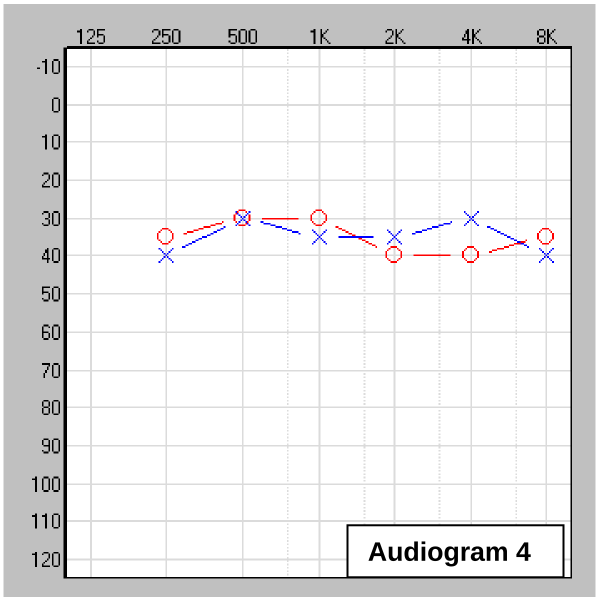
X = ____

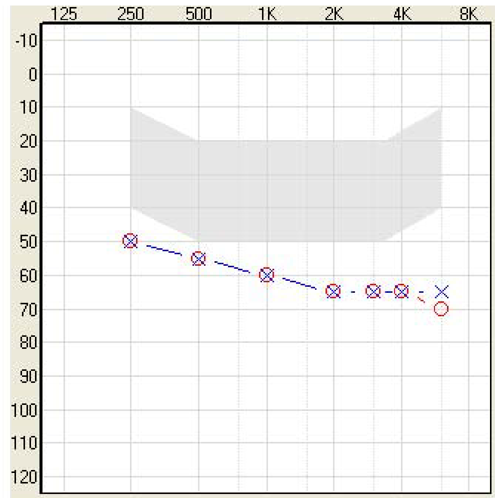
O=____

Notes

Notes

Audiogram 4 (below, left) and audiogram 5 (below, right) show the air conduction thresholds for two different amounts of hearing loss. Both of these hearing losses are called “flat” losses. This means that the thresholds are all at about the same dB level. More on that concept later. Look at the audiogram and determine how much loss is present for each.





Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Type of Hearing Loss

Notes

Before we discuss the audiogram further, let's examine the various types of hearing losses you will encounter in your daily practice. In simple terms, there are three distinct types of hearing loss:

1. Sensorineural (SN)

2. Conductive

3. Mixed

There are several kinds of hearing loss that are not represented by the conventional pure-tone audiogram but to which you may hear occasional reference:

A **central hearing loss** is one that occurs in the auditory processing centers of the brain. In this type of hearing loss, peripheral hearing (outer, middle and inner ear) may be normal, but when the signal gets to the brain, it cannot be interpreted.

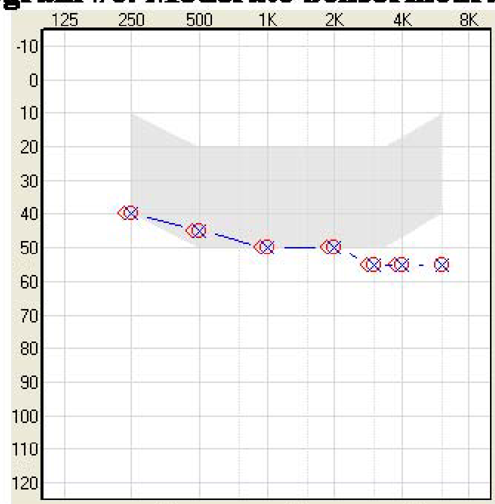
A **non-organic** or **functional hearing loss** (sometimes called **malingering**) is when hearing loss appears to be present with no auditory disorder. This may be hysterical in nature **-** that is, some mental disorder is responsible for the appearance of the hearing loss, or it may be a deliberate attempt by the patient to "fake" a hearing loss. Malingering must be ruled out whenever there is the likelihood of some gain or advantage to the patient for having a documented hearing loss.

A **retrocochlear hearing loss** is one that originates "beyond the cochlea". Acoustic neuroma, discussed in Unit 5 - Hearing Disorders, is one relatively common cause of retrocochlear hearing loss. Site-of-lesion tests, discussed in Unit 6, Basic Audiometry, are sometimes used to identify retrocochlear disorders.

Notes

In order to determine the second important variable - type of hearing loss - bone conduction (*abbreviated **BC***) thresholds are required. Bone conduction thresholds, as you will recall from an earlier chapter, refer to the hearing sensitivity of the inner ear. After air conduction thresholds are obtained with inserts or headphones, bone conduction thresholds are determined by placing a vibrating bone conduction oscillator over the mastoid bone located behind the pinna. Audiogram 6 (below) shows unmasked bone thresholds.

Audiogram #6. Moderate Sensorineural Loss.



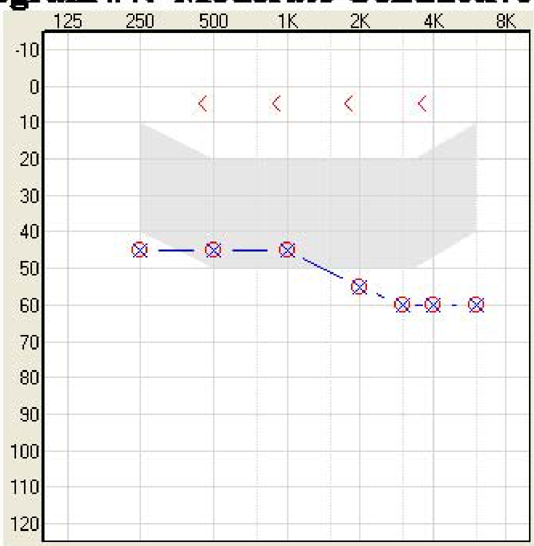
Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Notes

Bone conduction thresholds are represented by **<** and **>** on the audiogram. If you are concerned that you cannot remember that the < symbol is for the right ear and the > symbol is for the left ear, don't worry, all audiograms have a key at the bottom that you can refer to when needed.

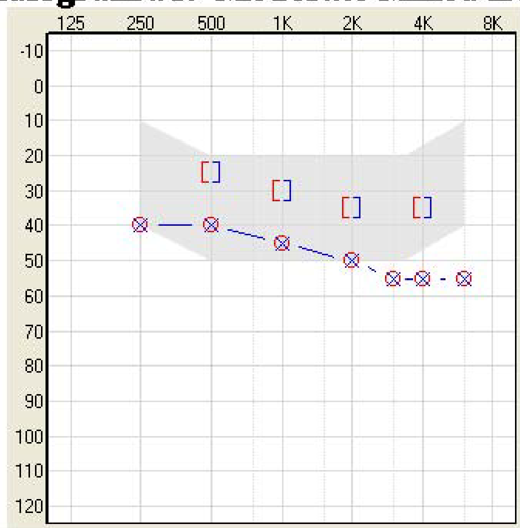
When determining the type of hearing loss on the pure tone audiogram, you need to look at the relationship between the air conduction thresholds represented by the X's and O's and the bone conduction thresholds represented by the < and >. When the symbols at a specific frequency are on the same line (the same intensity) or separated by less than 15 dB, the hearing loss is sensorineural. If there is more than a 10 dB separation between the air conduction thresholds (O's and X's) and the bone conduction thresholds (< and >) at the same frequency, the loss is either conductive or mixed. Let's look at a couple of examples. Audiograms 7 and 8 are good examples of air/bone gaps.

Audiogram #7. Moderate Conductive Loss.



Test	Right	Left	Binaural
UCL	105dB	100dB	
MCL	80 dB	80 dB	75/75dB
SRT	50 dB	50 dB	
WRS	96%	92%	96%

Audiogram #8. Moderate Mixed Loss.



Test	Right	Left	Binaural
UCL	100 dB	105 dB	
MCL	75 dB	75 dB	70/70dB
SRT	45 dB	45 dB	
WRS	92%	92%	96%

Check Your Understanding

Type of hearing loss can be determined by AC and BC.

True or False

Notes

Let's review. In order to determine the type of hearing loss you will need to look at both the air conduction and bone conduction thresholds on the audiogram. For a sensorineural hearing loss, the AC and BC thresholds symbols for a given frequency are written on the same line and within 10 dB of each other. For **conductive hearing loss** two things have to happen:

1. There will be a clear separation on the audiogram of more than 10 dB. This is called an air/bone gap.

2. The BC thresholds must be in the normal range (better than 25 dB HL) on the audiogram.

A mixed hearing loss will show both the AC and BC thresholds below normal (worse than 25dB HL on the audiogram) along with a greater than 10dB separation between the AC and BC thresholds. Please review the previous figures to see the distinct pattern of each type of hearing loss on the audiogram. Audiogram 8 from the previous page is an example of a mixed loss.

Symmetry

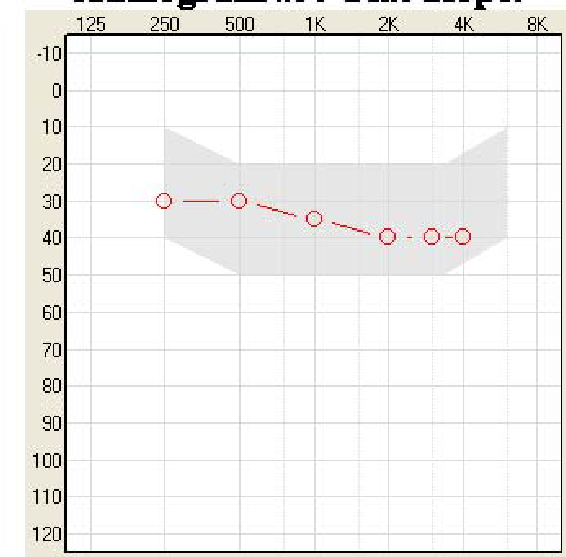
Another aspect of the audiogram is called symmetry. If both ears have roughly the same thresholds, they are considered **symmetrical**. That is, they are the same. When comparing the two ears, if all the frequencies tested are within 10 dB of each other, the hearing loss is considered symmetrical. When comparing the two ears tested, if the thresholds are different at two or more frequencies by >10 dB, the hearing loss is considered **asymmetrical**. *Asymmetrical hearing losses are a red flag for a possible medical problem and these patients should be referred to a physician. All the examples you have reviewed so far are symmetrical hearing loss.*

The shape of the audiogram: Patterns of hearing loss

Another variable in the interpretation of the audiogram is the shape. It's important to note that the shape of each ear's test results will be described individually by the audiogram. The terms you need to know describing the shape of the audiogram are the following:

Flat

Audiogram #9. Flat Slope.



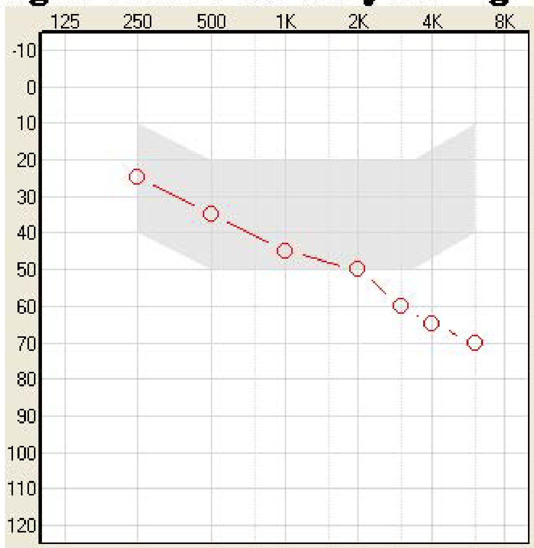
Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Notes

Notes

Sloping

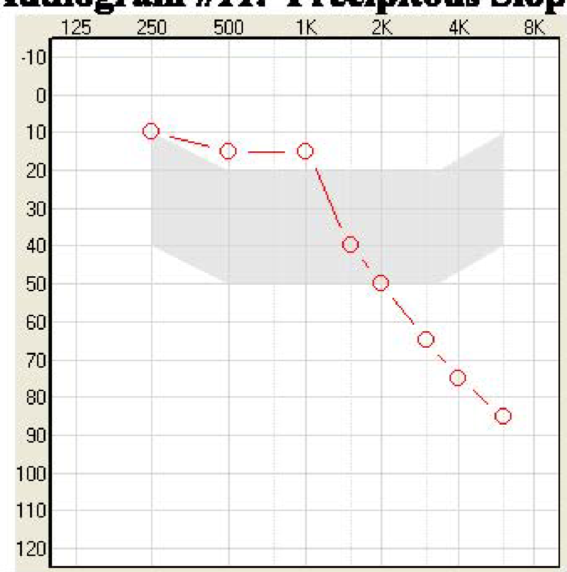
Audiogram #10. Gradually Falling Slope.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Steeply or Precipitously Sloping

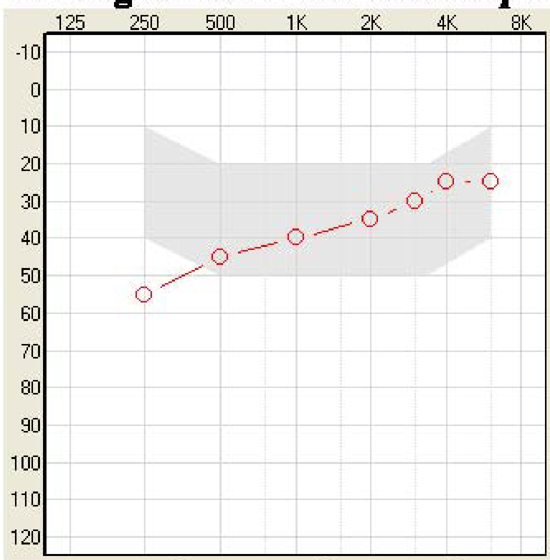
Audiogram #11. Precipitous Slope.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Reverse Slope or Rising

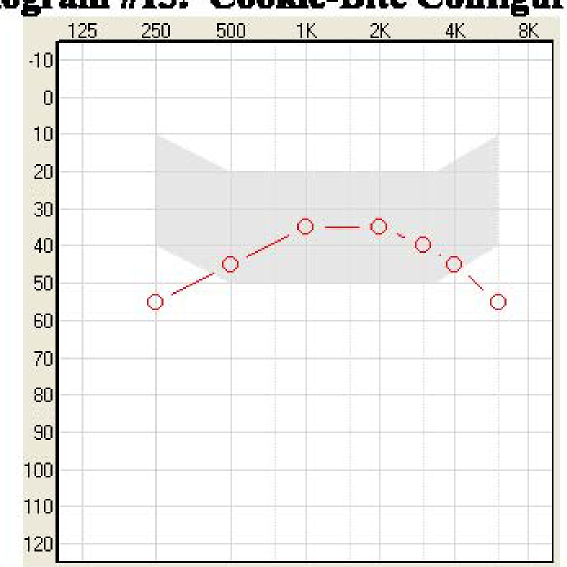
Audiogram #12. Reverse Slope.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Cookie Bite

Audiogram #13. Cookie-Bite Configuration.



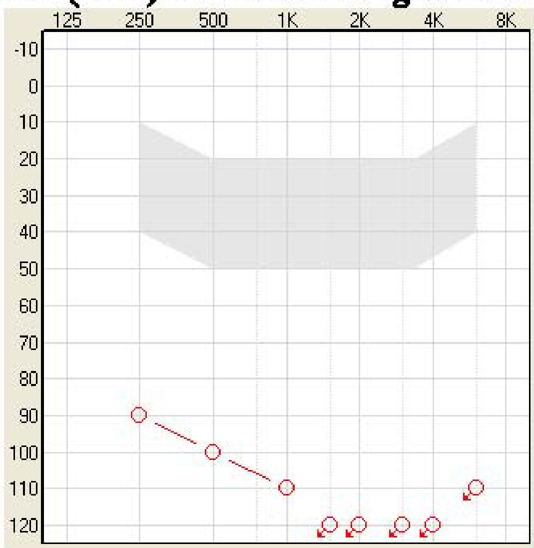
Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Notes

Left Corner Audiogram

Notes

Audiogram #14. (Left) Corner Configuration in Right Ear.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Check Your Understanding

Significant asymmetries require referral to a physician.

True or False

Section 2: Speech Audiometry – Beyond the Beeps

Notes

Now that you have been introduced to pure tone testing, it's time to turn our attention to speech audiometry. As the name implies, speech audiometry uses speech stimuli, either live voice or recorded, to assess the hearing mechanism. Pure tone and speech audiometry are designed to complement each other. Together they comprise your hearing test battery.

There are four speech audiometry tests you should know about:

1. Speech Reception Threshold (SRT)

2. Word Recognition Score (WRS), sometimes referred to as Speech Discrimination (SD)

3. Most Comfortable Loudness Level (MCL)

4. Most Uncomfortable Loudness Level (UCL)

Speech Reception Threshold

Let's start with Speech Reception Threshold (*abbreviated **SRT***). The SRT is the softest level at which speech can be heard 50% of the time. Like the pure tone threshold, it can be thought of as the floor of the audiogram. The SRT uses **spondee words** (two-syllable words with equal emphasis on both syllables) and the scoring is done when a patient can respond correctly to the words 50% of the time at the lowest volume level. The SRT is important because it serves as a way to double-check the results of the pure tone test. The SRT is compared to the pure tone average (PTA). PTA is obtained by adding the air conduction thresholds in each ear at 500, 1000 and 2000 Hz and dividing by 3. The result is the 3 frequency PTA. For an audiogram to be considered accurate, the PTA and SRT must agree. This means the PTA and SRT should be within +/-10 dB of each other for the entire test to be considered accurate.

Now that we know the floor of the audiogram it makes good sense to talk about the ceiling. The ceiling or upper limit of hearing and it is measured using a test called Uncomfortable Level (*abbreviated **UCL***) using speech. The UCL/LDL/TD may be measured with either pure tones or with speech. The pure tone loudness level is called Loudness Discomfort Level (LDL) or Threshold of Discomfort (TD). The UCL represents the intensity at which sound becomes uncomfortably loud. All of us have a UCL. Think about the last loud concert you went to. You likely experienced some sounds that exceeded your UCL. A person with hearing loss has a ceiling very close to that of a person with normal hearing. However, their thresholds, or ability to hear soft sounds has diminished. In other words their floor has been raised. Compared to someone who hears normally, a sound must be significantly louder before it is just audible, but not much louder than that before it becomes uncomfortably loud.

Notes

Now that we have talked about the floor and ceiling respectively of hearing we can address the area in between. This area is called the dynamic range (DR). The DR is the difference between the UCL and SRT. It is the area of what we call useable hearing. As you will see, when we fit hearing aids, the dynamic range becomes extremely critical because the hearing aid must repackage sound to fit into the individual dynamic range of the patient.

Word Recognition Score

Word Recognition Scores (*abbreviated **WRS***) measure how clearly someone hears speech in quiet at a comfortable volume level. This usually means we conduct WRS testing at Most Comfortable Level (*abbreviated **MCL***). Word recognitions scores will give the hearing care professional a good idea of how well speech is being processed. Word recognition testing is done monaurally and binaurally, which means that each ear is tested separately, then both ears are tested. In simple terms, the higher the score (greater percent correct), the more clear speech sounds to the patient. The WRS reflects the amount of speech information that is getting through to the brain. When someone has a WRS of, say, 40%, only 40% of the speech signal is getting through to the brain. This test does have some practical applications when it comes to hearing aid selection. Patients who score below 50% on the test often times do not do as well with hearing aids as those that score above 50%. Table B is a classification of word recognition scores. This classification system can be used to explain test results.

TABLE B
90-100% Excellent results
70-89% Good to Average Results
50-69% Fair Results
Less than 50% The results will be Poor

Just like pure tone thresholds, word recognition scores should be symmetrical. If the word recognition scores between the two ears are greater than 20%, then the WRS are considered asymmetrical and medical referral is required. The Binaural WRS should be as good as or better than the better monaural score. If the binaural score is *poorer than* the better monaural score, it is called **binaural degradation** and medical referral is required.

Speech in Noise Testing

Another important component of speech audiometry is speech in noise testing. These are tests in which words or sentences are presented to the patient along with background noise. We use these tests because we feel that they provide us with a better representation of how someone understands in everyday listening situations. The test we use is called the Quick SIN. You will be introduced to the Quick SIN during your basic training lab.

How to Read an Audiogram

1. Make sure your audiogram is legible.

2. Take a look at the thresholds and determine the degree of hearing loss

3. Look at the AC and BC scores to determine the type of loss

4. Determine the PTA and compare it to the SRT for each ear. If the PTA and SRT are more than +/-10 dB apart, then the accuracy of the audiogram is in question. You will need to redo at least part of the hearing test, especially AC.

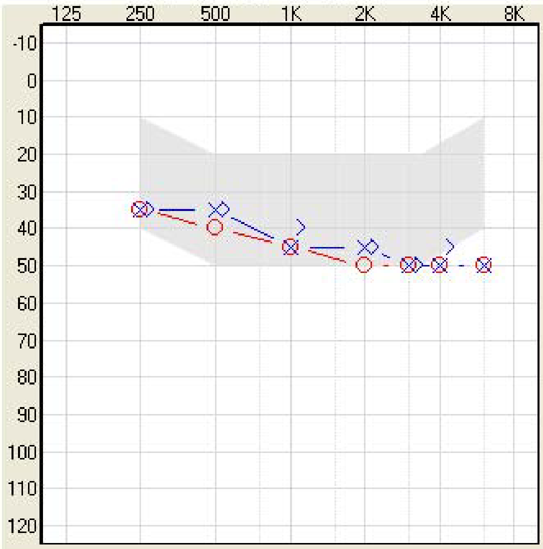
5. Review the word recognition scores. Use Table B on the previous page to classify the word recognition scores. If the scores are more than 20% apart they are asymmetrical. Remember, any significant asymmetry requires you refer the patient to a physician.

6. Finally, summarize the audiogram results in one sentence. Be sure to include the symmetry, slope, degree and type in your sentence. For example, “The patient has a symmetrical, mild to moderately-severe sloping sensorineural hearing loss.” Or “The patient has an asymmetrical hearing loss. For the right ear a mild to moderate sloping sensorineural hearing loss, and for the left ear a flat moderately severe mixed hearing loss.” For an asymmetrical loss always describe the better ear first when determining the above.

Notes

Exercise #1: Describe or summarize the following audiograms:

Exercise 1-1.



Test	Right	Left	Binaural
UCL	105 dB	105 dB	
MCL	60 dB	65 dB	dB
SRT	45 dB	50 dB	
WRS	92%	94%	%

Description

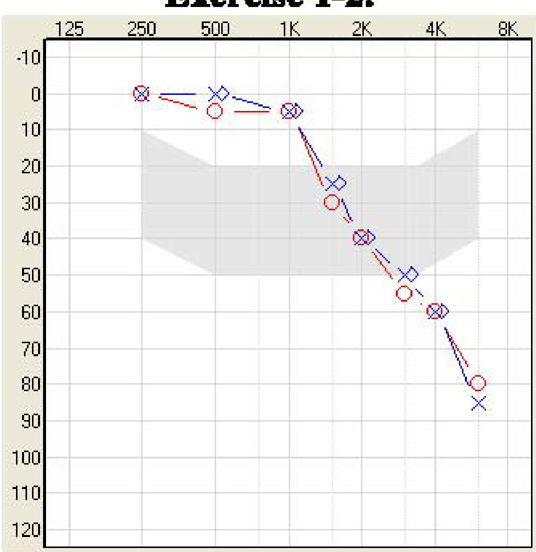
Symmetry:

Slope:

Degree:

Type:

Exercise 1-2.



Test	Right	Left	Binaural
UCL	105 dB	105 dB	
MCL	65 dB	60 dB	dB
SRT	10 dB	5 dB	
WRS	96 %	96 %	%

Description

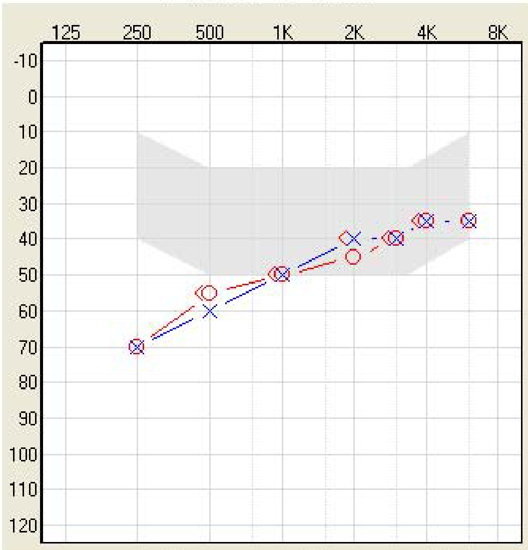
Symmetry:

Slope:

Degree:

Type:

Exercise 1-3.



Test	Right	Left	Binaural
UCL	105dB	100dB	
MCL	75 dB	70 dB	dB
SRT	65 dB	60 dB	
WRS	60%	82 %	%

Description

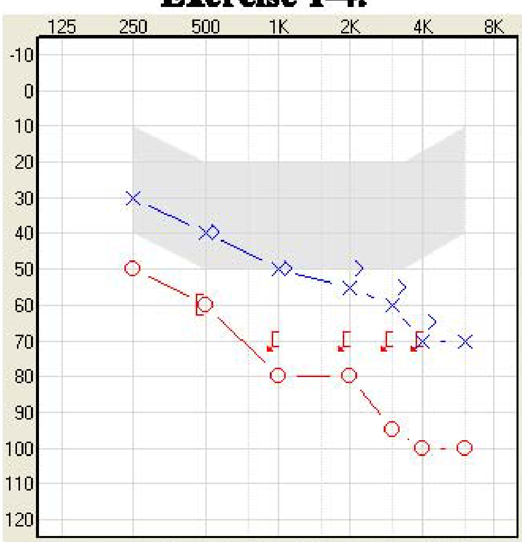
Symmetry:

Slope:

Degree:

Type:

Exercise 1-4.



Test	Right	Left	Binaural
UCL	110 dB	110 dB	
MCL	90 dB	70 dB	dB
SRT	80 dB	55 dB	
WRS	50%	68%	%

Description

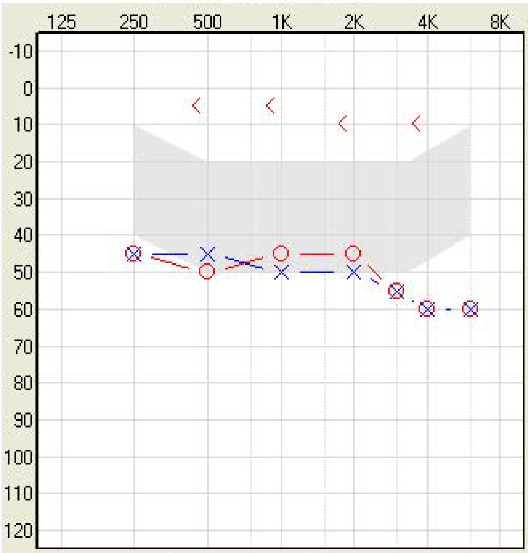
Symmetry:

Slope:

Degree:

Type:

Exercise 1-5.



Test	Right	Left	Binaural
UCL	110 dB	105 dB	
MCL	65 dB	70 dB	dB
SRT	40 dB	45 dB	
WRS	96%	98%	%

Description

Symmetry:

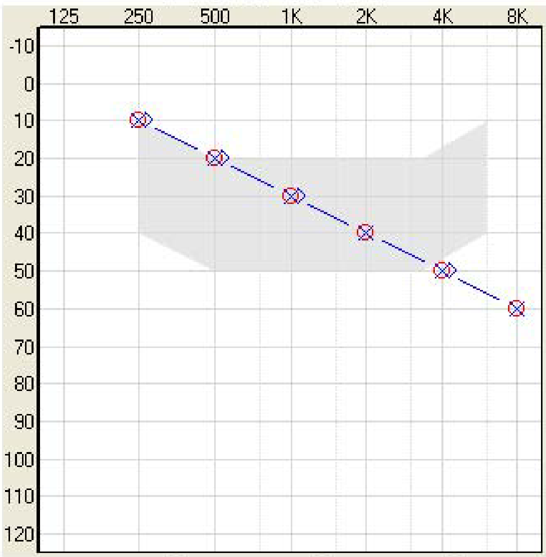
Slope:

Degree:

Type:

Exercise #2: List the missing components on each audiogram:

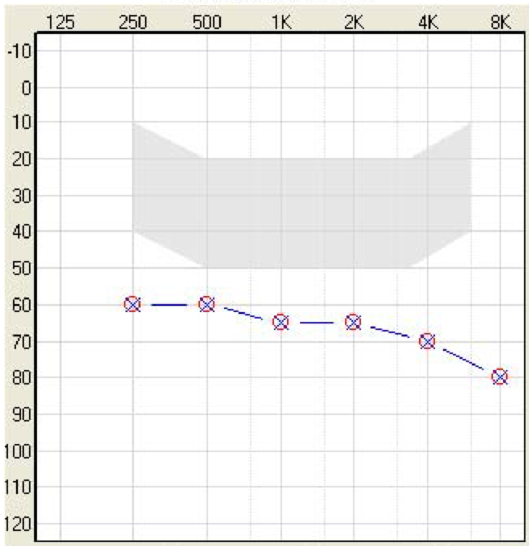
Exercise 2-1.



Test	Right	Left	Binaural
UCL	100 dB	dB	
MCL	65 dB	dB	dB
SRT	50 dB	dB	
WRS	84 %	%	%

Missing Components

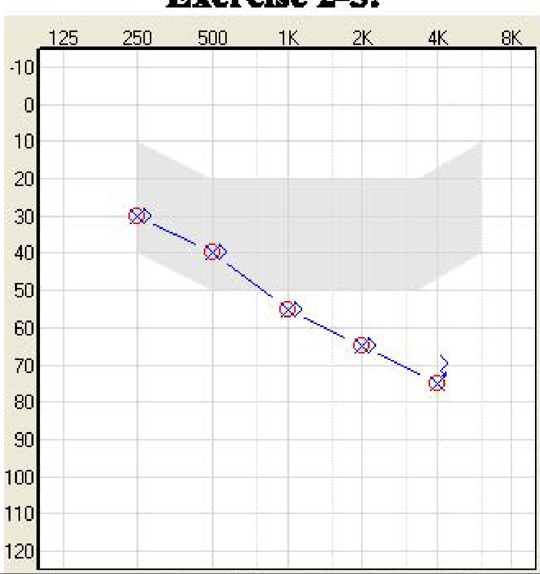
Exercise 2-2.



Test	Right	Left	Binaural
UCL	95 dB	100 dB	
MCL	65 dB	65 dB	dB
SRT	50 dB	50 dB	
WRS	76 %	76 %	%

Missing Components

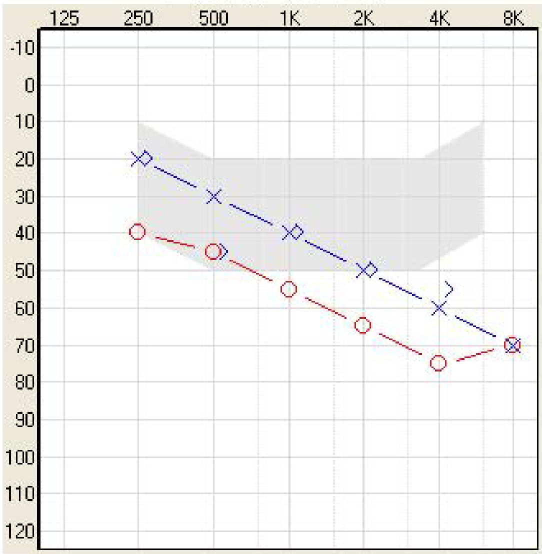
Exercise 2-3.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	65 dB	60 dB	
WRS	68 %	64 %	%

Missing Components

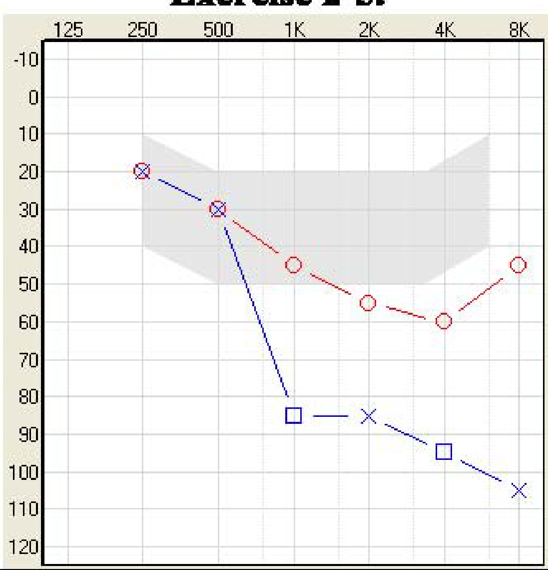
Exercise 2-4.



Test	Right	Left	Binaural
UCL	105 dB	105 dB	
MCL	60 dB	60 dB	dB
SRT	50 dB	40 dB	
WRS	80 %	84%	%

Missing Components

Exercise 2-5.



Test	Right	Left	Binaural
UCL	95 dB	110 dB	
MCL	60 dB	60 dB	dB
SRT	40 dB	40 dB	
WRS	84 %	80 %	%

Missing Components

Section 3: Putting It All Together

Audiogram Interpretation for Hearing Aid Selection & Counseling

This section is devoted to applying the results of the audiogram to the hearing aid selection and fitting process. One of the most important lessons you need to learn is that the audiogram is just one piece of information you need in order to select and fit hearing aids. Just as the personalities of two people are not exactly alike, the audiograms of two people are never exactly alike, but they can be similar. Still, we can make some general statements about how to use the audiogram to determine someone's ability to use hearing aids.

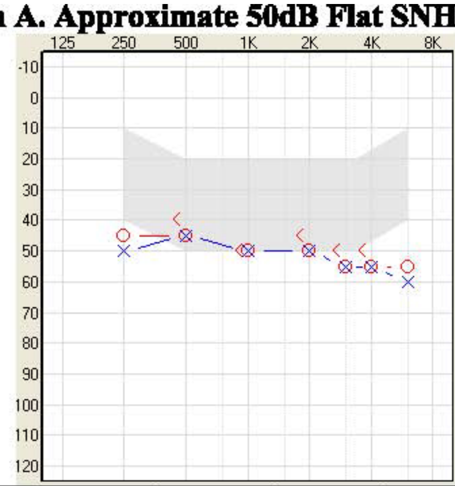
After the hearing test has been completed, and a possible medical problem has been ruled out, it is time to use the results of the audiogram to select hearing aids. The results of the audiogram are an important part of this selection process. The specific details of what style of hearing aid is best for various audiograms will be addressed in the hearing aid section. For now, let's begin with the basics by asking the following questions.

1. Is the patient a candidate for hearing aids?

Probably the best way to address this question is to go over some audiogram examples, which we will do in the following pages.

Here is a clear case of someone who is a candidate for hearing aids. As a rule-of-thumb, if the pure-tone average (also known as PTA and described in Unit 6 - Basic Audiometry) is greater than 25 dB HL, the patient should be considered a candidate for hearing aids.

Audiogram A. Approximate 50dB Flat SNHL. Binaural.



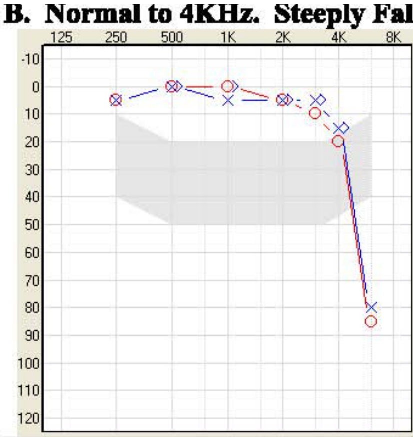
Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Notes

Notes

Here is an example of someone who is clearly not a candidate for hearing aids. Although this person has some hearing loss, a hearing aid will not provide any benefit for a loss exclusively in the highest frequencies.

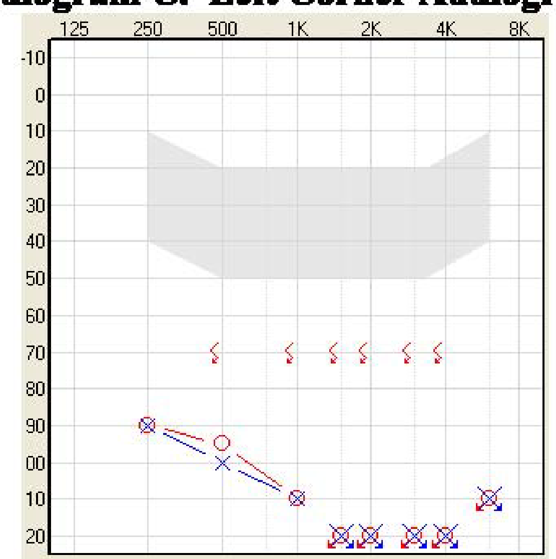
Audiogram B. Normal to 4KHz. Steeply Falling at 6KHz.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

Now let's look at some examples that are not so straightforward. You will see many patients who have normal hearing on the audiogram through 2 KHz, many of these patients are very good candidates. The rule of thumb for patients who have normal hearing through 2 KHz is that they are certainly possible candidates for hearing aids however, candidacy is largely determined by other factors that lie outside the audiogram. Motivation is a key factor for these people. Here is an example of a left corner audiogram.

Audiogram C. Left Corner Audiogram.



Test	Right	Left	Binaural
UCL	dB	dB	
MCL	dB	dB	dB
SRT	dB	dB	
WRS	%	%	%

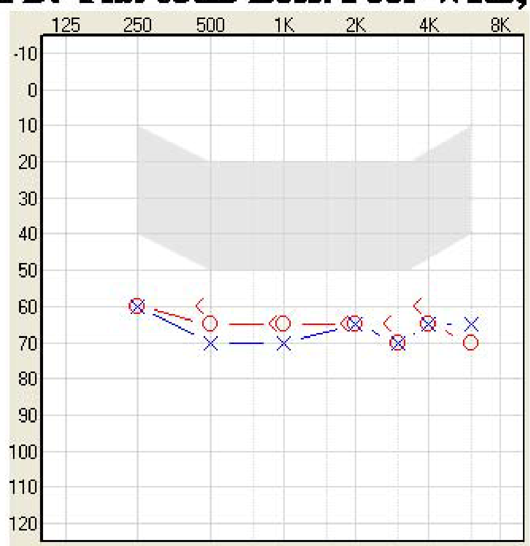
Individuals who have this degree of hearing loss are still good candidates for hearing aids, but the hearing loss is so severe that they will receive limited benefit. The goal of hearing aids for someone with this type of hearing loss would be to assist with lip-reading and environmental awareness. These types of hearing losses typically require powerful, behind-the-ear (BTE) hearing aids. Some patients with this amount of hearing loss also may opt to use a cochlear implant or not wear any devices and use American Sign Language (**ASL**) to communicate. Fortunately, most hearing losses like this are identified at a very early age, so if you see this amount of hearing loss, these folks have probably been tested and fitted with hearing aids several times.

2. How much improvement will the patient receive from hearing aids?

This question is extremely difficult to answer. Much of it depends on your ability to fit the hearing aids correctly, the motivation of the patient to accept help, and other factors. However, just looking at the audiogram can help us to counsel the patient on what to expect from hearing aids. Here are some examples to help you understand this concept.

Here is an example of someone with very poor word recognition ability.

Audiogram D. Flat 65dB Loss. Poor WRS, bilaterally.



Test	Right	Left	Binaural
UCL	105 dB	100 dB	
MCL	75 dB	75 dB	dB
SRT	65 dB	65 dB	
WRS	40 %	40 %	%

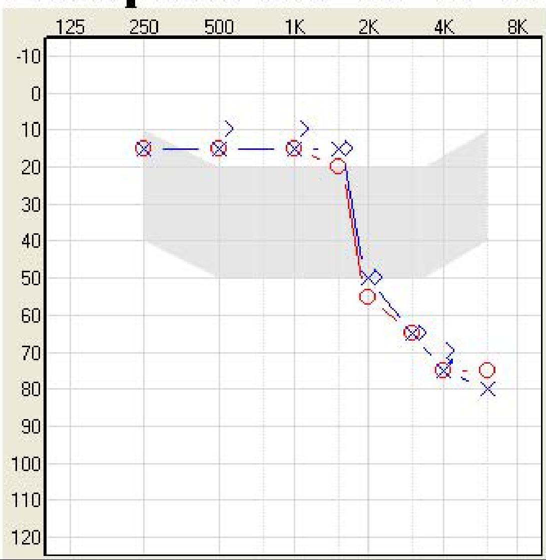
Notes

Notes

This type of situation occurs with the very elderly. The pure tone hearing loss is moderate, but the speech audiometry scores are poor, that is, below 50%. Certainly, these patients are hearing aid candidates, but due to the poorer than average word recognition ability, they need to be informed of some of the limitations of using them. In this example, the patient will experience an improvement in their hearing with hearing aids; however, speech will not always be clear and understandable.

Here is an example of someone with a precipitously sloping hearing loss.

Audiogram E. Precipitous Loss. <15dB ABG. Good WRS.



Test	Right	Left	Binaural
UCL	100 dB	105 dB	
MCL	60 dB	55 dB	dB
SRT	10 dB	10 dB	
WRS	80 %	80 %	%

This person is clearly an excellent hearing aid candidate. However, steeply sloping hearing losses that have areas in the low frequencies of normal hearing are often times challenging to fit. This patient needs to know that this loss can be fit quite successfully, but it might take a few extra visits to fine tune the devices. This is the kind of hearing loss for which open-canal instruments were designed. The two other factors to take in to account are:

1. How well does this person do with speech understanding in noise? Speech in noise testing is critical. How

2. How motivated is this patient to better their understanding in noise? This can be critical to a successful fitting.

Section 4: Hearing Aid Selection

Notes

Hearing Aid Selection

In order to select the proper hearing aid for a patient the five variables listed below are traditionally considered. During Basic Training your instructor will go over the details of properly selecting hearing aids using information from Five Key Areas of Audiometric Interpretation:

Five Key Areas of Audiometric Interpretation:

1. Degree of Loss (How Much?)

- Normal/Within Normal Limits: 0-25 dB PTA (ANSI)
- Mild: 26-40 dB HL
- Moderate: 41-55 dB HL
- Moderately/Severe: 56-70 dB HL
- Severe: 71-90 dB HL
- Profound: >90 dB HL

2. Type of Loss (What Kind?)

- Central
- Non-Organic
- Conductive
- Sensorineural
- Mixed Loss
- Neural Loss (retrocochlear)

The neural loss may show a normal or sensorineural pattern, but with positive findings for tone decay, speech rollover for PB max, and/or acoustic reflex decay. These tests are referred to as Site of Lesion testing (described in Unit 6 - Basic Audiometry). Large unexplained asymmetries for pure tones or word recognition are also “red flags” for neural involvement.

3. Slope or Shape of Loss

Typical Slopes	Atypical Slopes (Referral Based on Shape)
Rising	"U" or Cookie-Bite
Flat	Inverted "U" or Reverse Cookie Bite
Gradually-Falling	Corner Audiogram
Markedly-Falling	Islands of hearing/Fragment
Steeply-Falling	
Ski-slope/Precipitous	

4. Dynamic Range (DR)

- Speech – UCL minus SRT
- Pure Tone – Pure tone LDL/TD minus same Frequency Pure Tone Threshold = Residual Area of Processing

5. Speech Discrimination Score (SDS)/Word Recognition Score/(WRS)/SRS/PB%

- 90-100% = Excellent
- 70-89% = Good, Average
- 50-69% = Fair
- <50% = Poor

Matrix Selection: Some Practical Advice

A hearing aid matrix is a combination of numbers that define peak output, peak gain and the slope of the hearing aid response. For example if you see a combination of numbers on the specifications sheet such as, 115/40/15, you know that the 115dB is the maximum OSPL output, 40dB is the peak gain, and 15 is the slope. In new, modern matrices the slope number may not be given. These numbers are selected from what you know about the audiogram.

Maximum OSPL Output

The first number is peak output; the greater the number, the higher the output, or the greater the power of the hearing aid. Maximum OSPL output usually falls between 95 to 135 dB SPL. Maximum OSPL output can be estimated by adding 20 dB to the obtained speech UCL (or 10 dB to the pure tone LDL/TD between 2000 & 3000 Hz).

Notes

Peak Gain

The second number is peak gain; the greater the number the higher the gain. There are two ways peak gain can be calculated. One way is to use a prescriptive formula. This means you take the thresholds from the audiogram and enter them into a formula. NAL-R is the most popular linear prescriptive formula, but there are many others, including, POGO, Berger, Libby, 1/3 gain and 1/2 gain. The most popular non-linear (compression) prescriptive formulas are NAL-NL1, NAL-NL2 and DSL-i/o. They work the same as the other formulas, but instead of getting a single value for gain, you get three; one for soft sounds, average sounds and loud sounds.

Many of these formulas are essentially derived from the 1/2 gain rule. A good rule of thumb is to calculate the three frequency pure-tone (PTA) average and divide the number by 2. The number you get must be at least 25dB or more. You will then want to apply some reserve gain, so take the number you derive from the formula and add 5dB for binaural fit and 10dB for monaural fit. To recap, PTA/2 + reserve gain (5dB or 10dB) is the amount of gain you need to look for in your matrix.

Another traditional way to predict the gain is to take the binaural speech MCL, divide by 2 and add your reserve gain.

Note: losses with an air/bone gap require additional gain; 1/4 of the gap, up to 10 dB maximum.

Slope

That brings us to the final number on the matrix: slope. This refers to the shape of the frequency response. The slope is largely derived from the microphone of the hearing aid. In technical terms, slope equals peak gain of the hearing aid minus gain at 500 Hz. The smaller the number, the flatter will be the hearing aid’s frequency response. Some matrices leave the third value – slope – off the spec sheet. This is due to the fact that, with digital, multi-channel hearing aids, virtually any slope may be configured. See the image on the following page for a specifications sheet for a typical digital BTE hearing aid. Note that “output sound pressure level” and “gain” are documented, but not “slope”, for this hearing aid. Other specifications will be briefly discussed in the section on ANSI testing in Unit 10 - Real Ear Measurement (REM) and ANSI.

Notes

	Earhook		OpenTube	
	2 ccm coupler	Ear simulator	2 ccm coupler	Ear simulator
	Standard ANSI S3.22-2003; IEC 60118-7:2005	IEC 118-0/A1	Standard ANSI S3.22-2003; IEC 60118-7:2005	IEC 118-0/A1
Output Sound Pressure Level				
at 1.6 kHz	–	130 dB	–	117 dB
Peak	124 dB	132 dB	123 dB	127 dB
HFA[1]-OSPL 90	121 dB	–	111 dB	–
Gain				
Full-on Gain (FOG) at 1.6 kHz	–	52 dB	–	47 dB
Full-on Gain (Peak)	55 dB	65 dB	55 dB	60 dB
HFA-FOG	47 dB	–	40 dB	–
Reference Test Gain	44 dB	45 dB	34 dB	40 dB
Frequency Range (DIN 45605*)				
Low frequency limit	<100 Hz	<100 Hz*	<100 Hz	<100 Hz*
High frequency limit	7000 Hz	7500 Hz*	7100 Hz	7300 Hz*
Total Harmonic Distortion				
500 Hz	4 %	2 %	1 %	1 %
800 Hz	3 %	2 %	1 %	1 %
1600 Hz	1 %	1 %	1 %	1 %
Equivalent Input Noise	17 dB	17 dB	17 dB	19 dB
Inductive Coil Sensitivity				
MASL[2] (HFA-MASL*) (1mA/m)	78 dB*	84 dB	71 dB*	76 dB
HFA SPLITS[3] (left/right)	105 / 102 dB	–	95 / 92 dB	–
RSETS[4] (left/right)	1 / -2 dB	–	1 / -2 dB	–
AGC-O (fully activated)				
Attack time	5 ms	–	5 ms	–
Release time	600 ms	–	600 ms	–
Battery-Type 13				
Battery Voltage	1.3 V	1.3 V	1.3 V	1.3 V
Battery current drain	1.0 mA	0.8 mA	1.1 mA	0.8 mA
Battery life (Cell Zinc Air)	~220 h	~220 h	~200 h	~200 h
IRIL[5] IEC 118-13:2004 (bystander)				
800-960 MHz	-9 dB			
1400-2000 MHz	-12 dB			
AI-DI[6]	3.7			

[1]HFA– High Frequency Average [2]MASL– Magneto Acoustical Sensitivity Level [3]SPLITS–Coupler SPL for an Inductive Telephone Simulator
[4]RSETS–Relative Simulated Equivalent Telephone Sensitivity [5]IRIL–Input Related Interference Level [6]AI-DI–Articulation Index - Weighted Directivity Index

Ear Mold Selection

Selection of proper Behind-the-Ear and body aid style air conduction earmolds typically falls into three or four major areas of concern. First, the style of earmold is often selected based on four important criteria:

- Audiometric configuration
- Anatomical characteristics of the pinna and ear canal
- Physical dexterity of the patient
- Patient preferences

Acoustic characteristics such as venting and canal length are frequently based on the audiometric configuration and the desired frequency response. Changes in **tubing size** and **earhook dimensions** also provide acoustical modifications that must be taken into consideration

There are many earmold styles. For our purposes we will consider only four main groups:

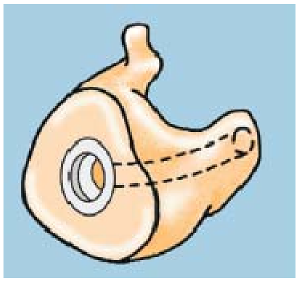
1. Receiver molds
2. Full shell molds
3. Skeleton molds
4. Open or non-occluding earmolds.

The earmolds we refer to here are those that serve as couplers for non-custom body and behind-the-ear (BTE) hearing instruments only. Custom in-the-ear (ITE) hearing instruments are circuitry and earmold coupler combined.

Notes

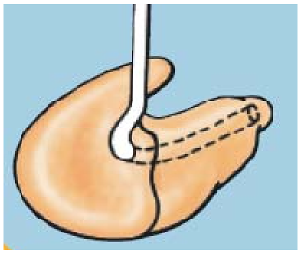
Receiver molds

Receiver-molds attach to body aids, coupling the external receiver to the ear. The microphone and amplifier are located in a transistor-sized housing that is worn, appropriately enough, on the body. Similar in appearance to a full concha mold, the receiver mold has a metal or plastic snap-ring that attaches to the external receiver. Due to the high power output of the typical body style aid, there is seldom any venting in the receiver style mold. Individuals with profound hearing losses typically wear body aids. Receiver molds may be made of either acrylic, silicon or a combination.



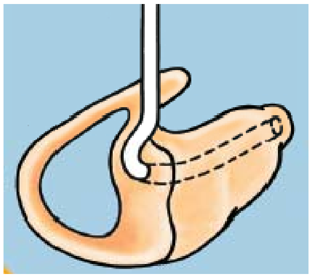
Full concha or Full shell molds

For individuals with severe to profound hearing losses a full concha mold coupled to a high power BTE typically works best. The full concha or full shell fills the entire concha of the ear. It typically has both a sound bore (through which the tubing is placed) and a vent. Venting options will be discussed later. Full concha style molds may be made of either acrylic, silicon or a combination.



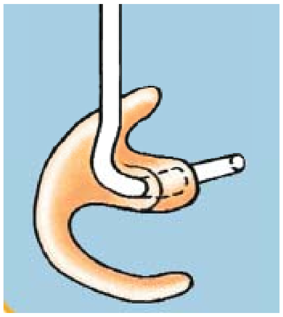
Skeleton molds

Skeleton molds are similar to full concha molds, except that portions of the concha are drilled out to allow for a lighter-weight fitting earmold. While recommended for individuals with mild to moderate or moderately severe hearing losses, the skeleton mold may also be fit to high-powered BTEs. As long as the drilled out portions of the mold do not compromise the sound bore, a tight feedback free coupling still might be achieved. Skeleton molds are typically made of acrylic.



Open or Non-occluding earmolds

Also sometimes referred to as the CROS mold, this earmold was designed to couple high frequency-emphasis BTEs to the ears of persons with normal low frequency and steeply sloping high frequency hearing loss. Utilizing hard acrylic composition and an open design (the earmold does not fill the canal), the open or non-occluding mold is the most effective way to achieve high frequency fitting emphasis.

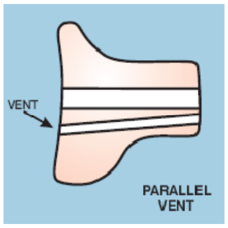


Notes

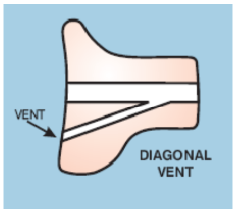
Venting considerations

Earmold venting may be classified three ways:

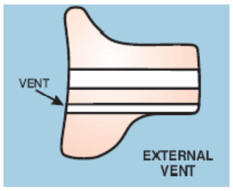
- Parallel venting
- Diagonal venting
- External venting



A **parallel vent** runs parallel to and is separated from the sound bore. Of the three venting choices, parallel venting is the most predictable and desired venting condition. It is also the easiest venting style to re-tube.



Diagonal venting is sometimes used when there is insufficient room for parallel venting, i.e., very small ears. Here the vent connects directly into the sound bore at some point in the earmold. Re-tubing this type of ear mold with a diagonal vent requires careful attention to detail so the vent is not occluded.



When there is not even enough room for a diagonal vent, **external venting**, sometimes called a **trench vent**, may be considered. An external vent is a channel drilled along the underneath exterior of the earmold along the bottom of the ear canal which, when open, may provide for air to enter the ear canal. As external vents have a tendency to fill up with skin debris and wax each time they are placed in the canal, they must be frequently inspected and cleaned by the wearer. This vent choice is effective for draining ears. External vents seldom produce the desired result.

Venting diameter or size is typically determined by both the degree of the loss and the size of the ear. The general rule is: the more severe the degree of loss, the more low frequencies that are needed to stay in the ear so the smaller the vent. No vents are recommended for those individuals with severe to profound hearing losses. Pressure or pinhole vents may be acceptable for those individuals with moderately severe loss. Those individuals with moderate loss typically get a small vent (.75mm) while wearers with a mild loss get a medium vent (>1.5mm). A large vent (3mm) or IROS vent is recommended for those individuals with normal lows and a steeply sloping loss.

Check Your Understanding

1. If the masked bone conduction scores are 15 dB or better than the air conduction scores and the bone conduction scores are within normal limits, this loss would be considered a:

    a. Sensorineural loss

    b. Conductive Loss

    c. Normal Hearing

    d. Mixed Loss

2. Difficulty with speech discrimination usually accompanies:

    a. Conductive loss

    b. Functional loss

    c. Sensorineural loss

    d. Otosclerotic loss

3. If air and bone scores are 25 dB HL or better with no air/bone gap, the loss is:

    a. Central

    b. Sensorineural

    c. Conductive

    d. Within Normal Limits

4. Which of the following loss types is not represented by the pure-tone audiogram?:

    a. Conductive

    b. Central

    c. Mixed

    d. Sensorineural

5. Otitis media is indicated by the following audiometric characteristics: (select all that apply)

   a. Flat or rising curve

   b. Slight air/bone gap

   c. Good discrimination

   d. Poor discrimination

6. A hearing loss with a pure-tone average of 65 dB HL would be considered which of the following:

   a. Within normal limits

   b. Profound hearing loss

   c. Severe hearing loss

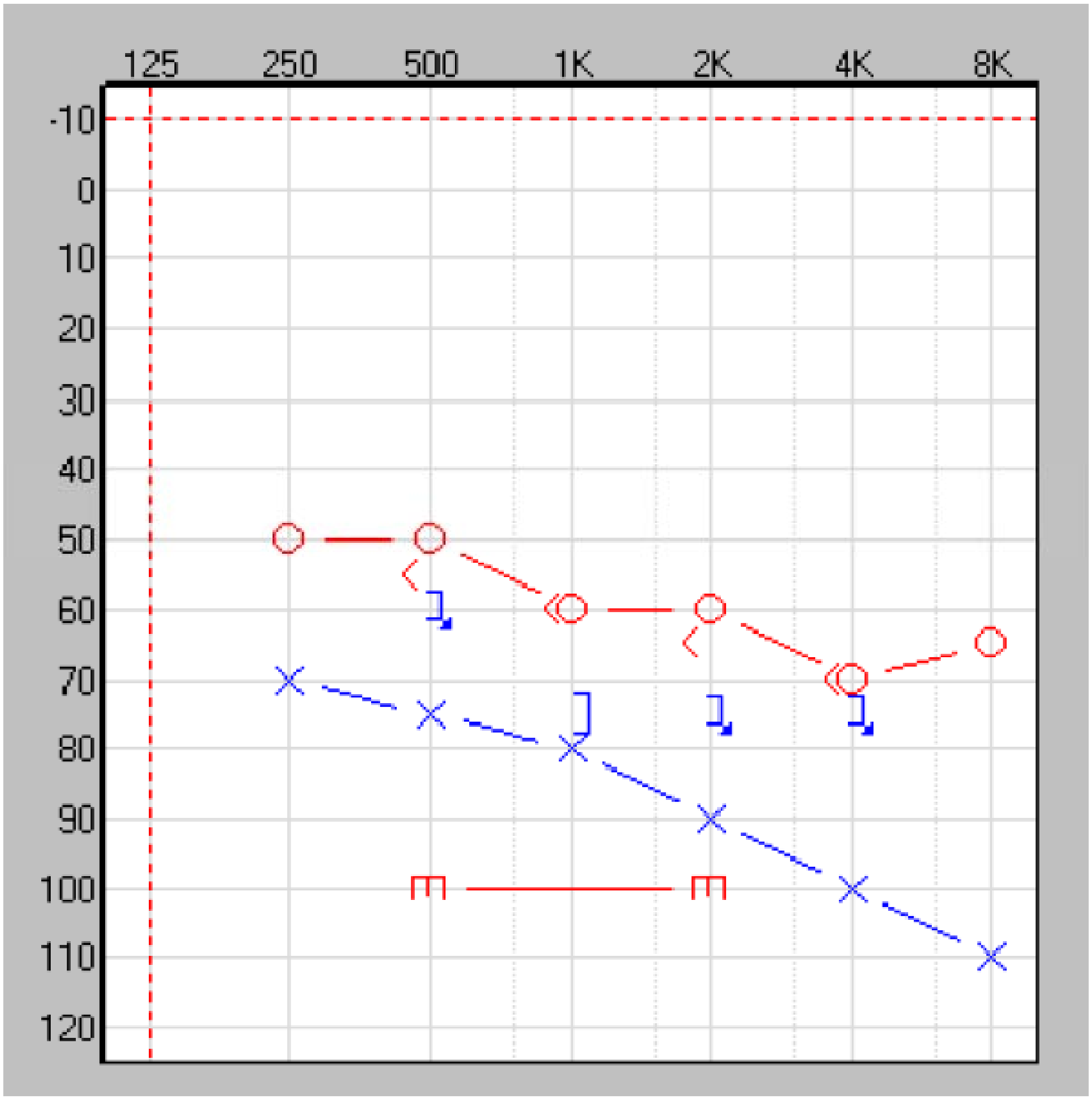
   d. Moderately severe hearing loss

On the following pages you will find a table followed by 10 audiograms for your review and interpretation. Examine the audiograms and describe the following conclusions derived from each, recommendations and descriptions you can obtain from the audiometric data.

1. What is the type, degree, slope of the hearing loss? If the audiogram is asymmetrical, describe both ears.
2. Is the audiogram valid?
3. Was masking required, and if so, was it done properly?
4. Can you determine the possible cause of the hearing loss from the data provided?
5. Is a medical referral required?
6. What style of hearing aid and or earmold would you recommend?
7. What matrix (MPO/peak gain) would you prescribe for this hearing loss?
8. What size venting would you recommend?
9. What are this patient's realistic expectations for success with hearing aids and why?
10. Are there any other significant considerations?

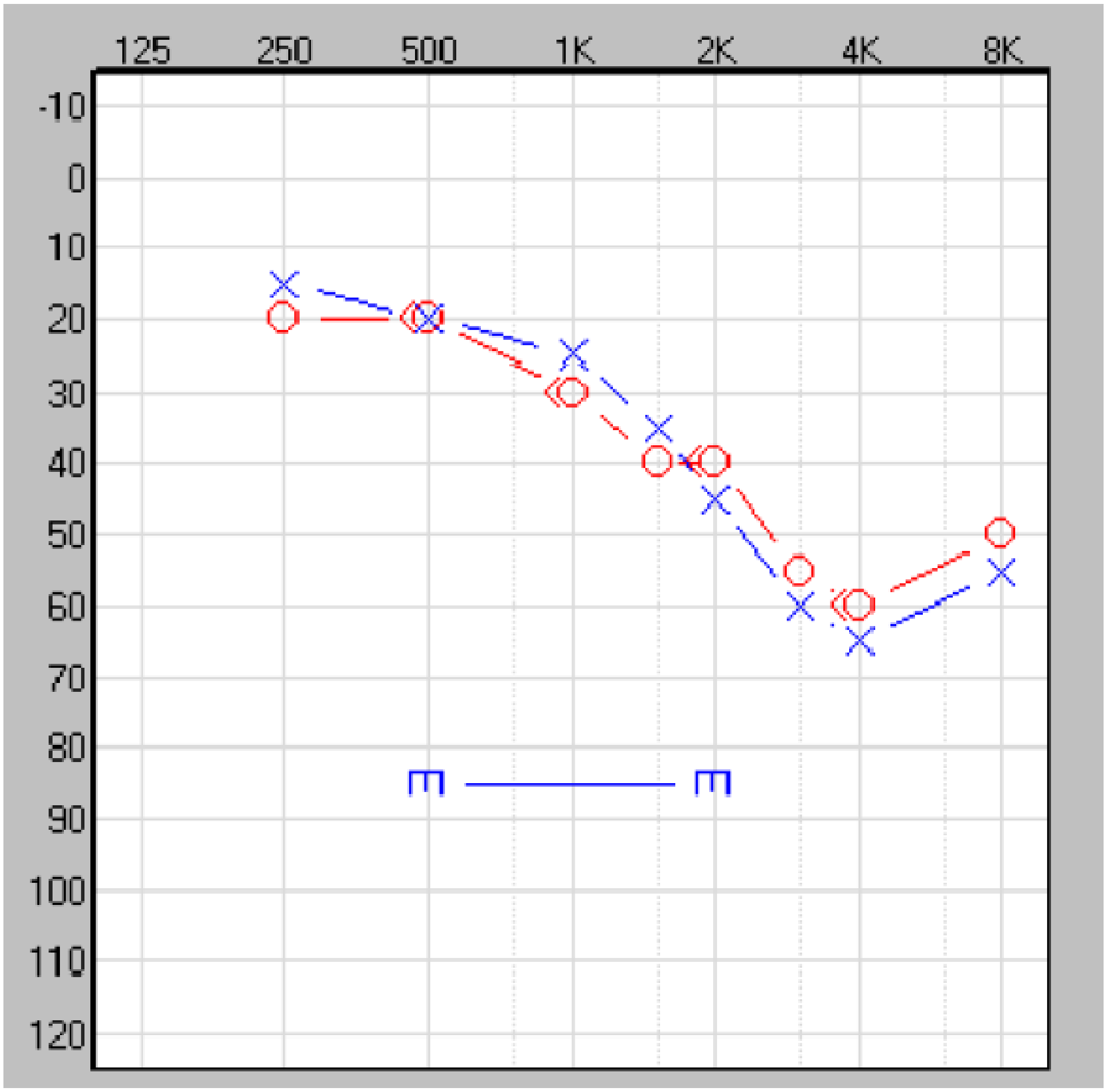
	Type, Degree, Slope	Valid?	Masking?	Cause (etiology)	Medical referral?	Style of hearing aid - EM	Matrix (MPO/gain)	Venting	Expectations?	Other
1										
2										
3										
4										
5										
6										
7										
8										
9										
10										

Audiogram Examples



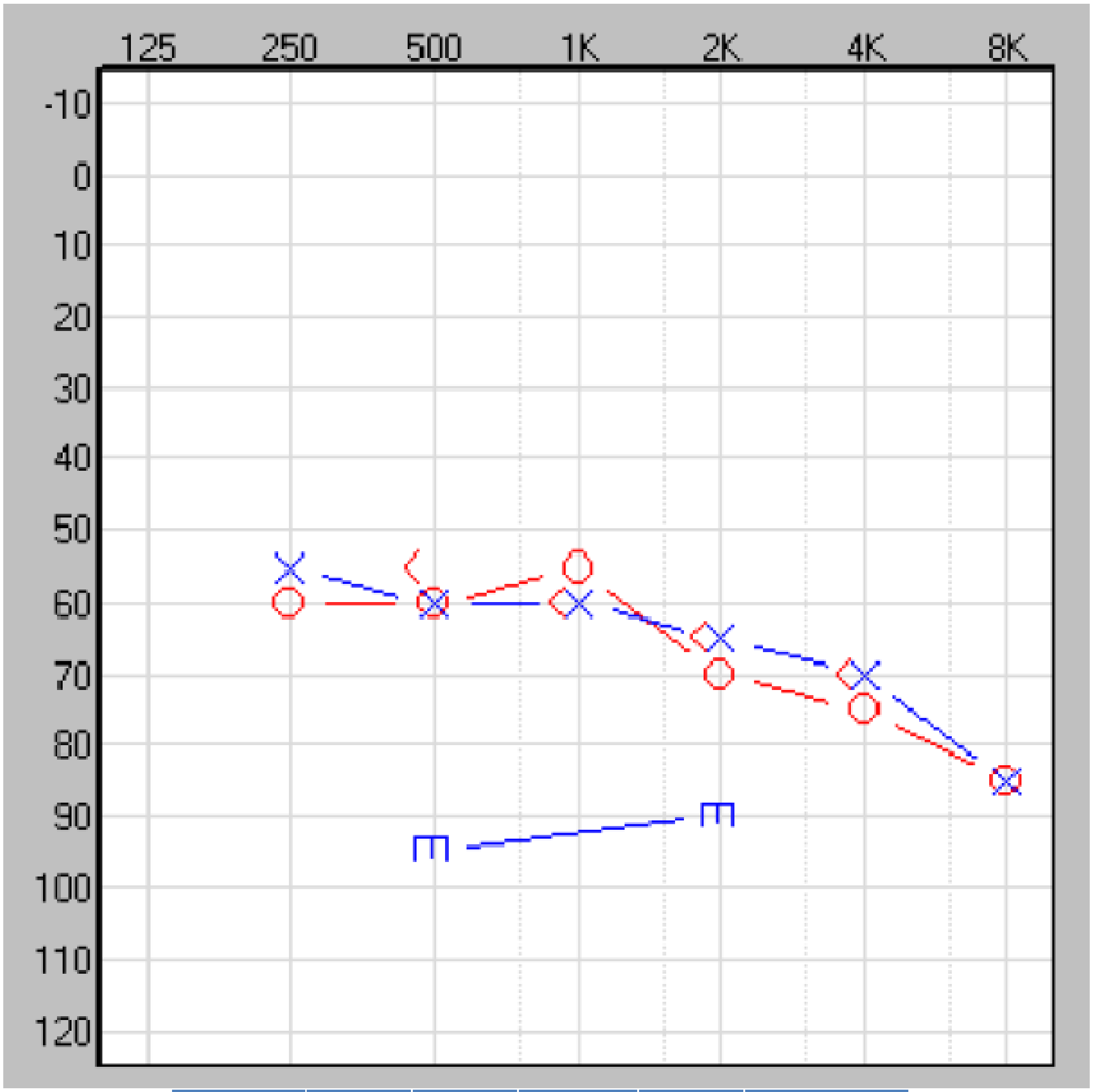
	SRT	MCL	WR%	UCL	SNR loss
Right	55	85	80	100	15
Left	80	90	36	100	

1



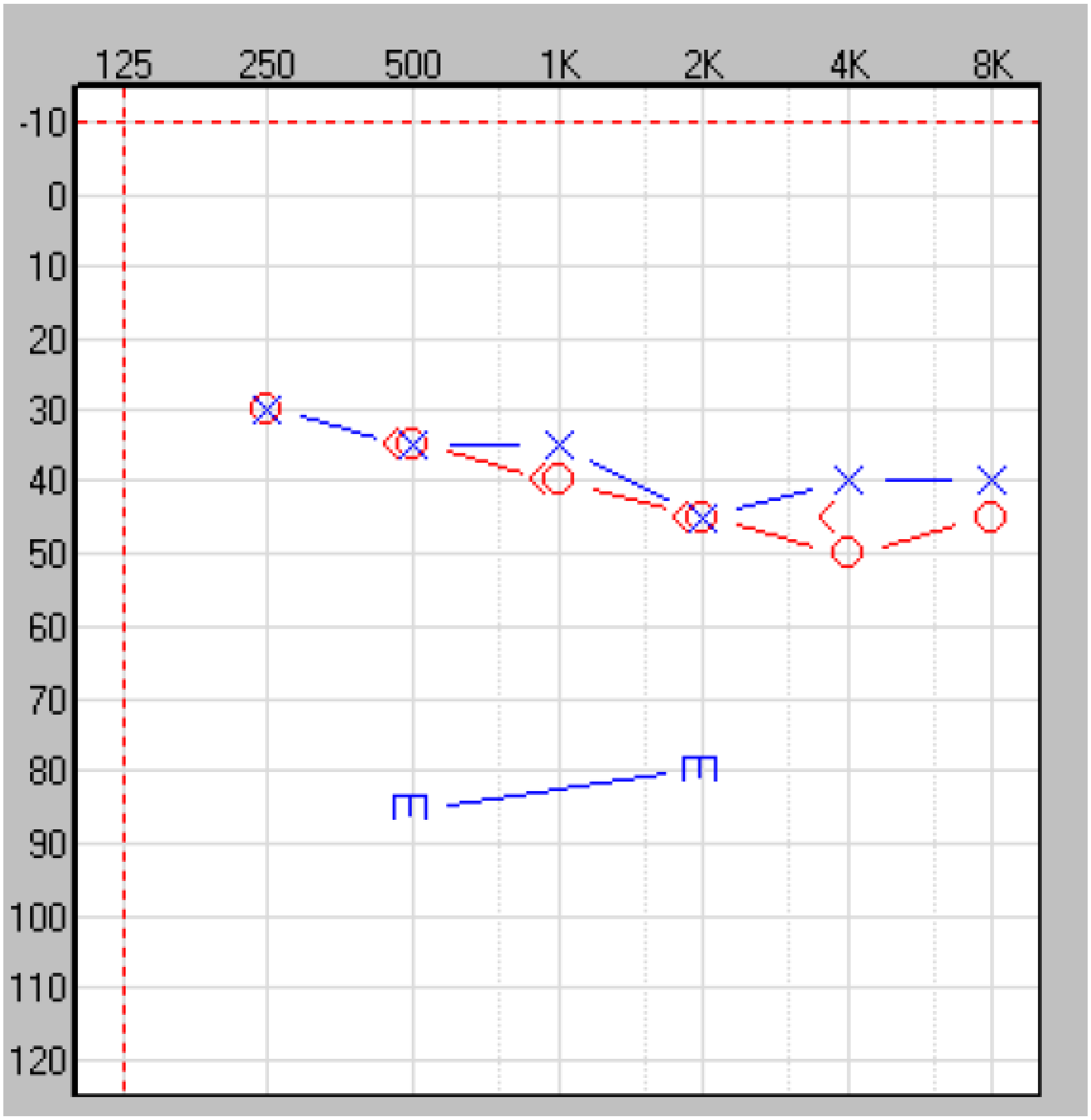
	SRT	MCL	WR%	UCL	SNR loss
Right	25	65	92	85	9
Left	30	65	92	85	

2



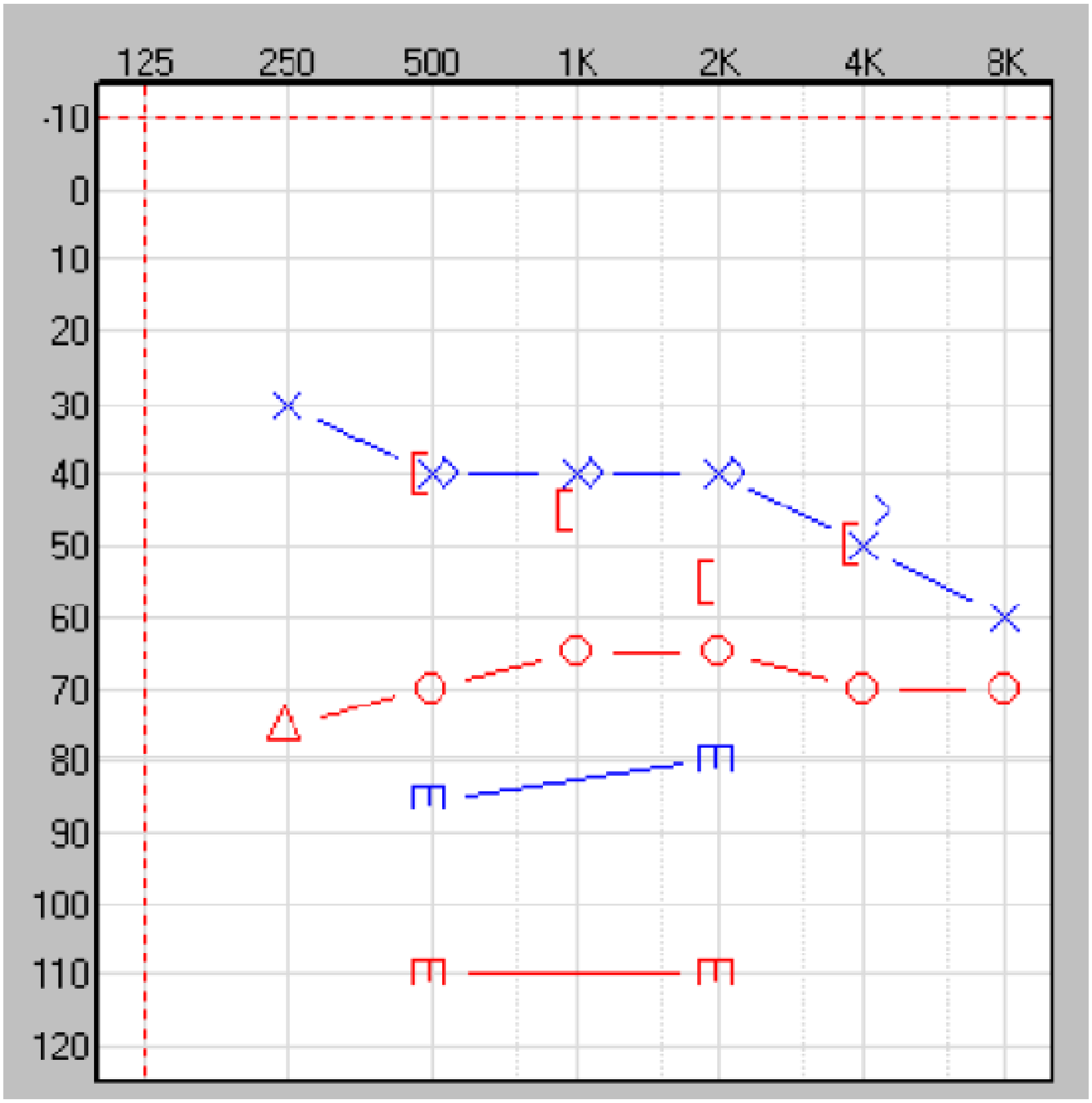
	SRT	MCL	WR%	UCL	SNR loss
Right	65	75	68	95	15
Left	65	75	72	95	

3



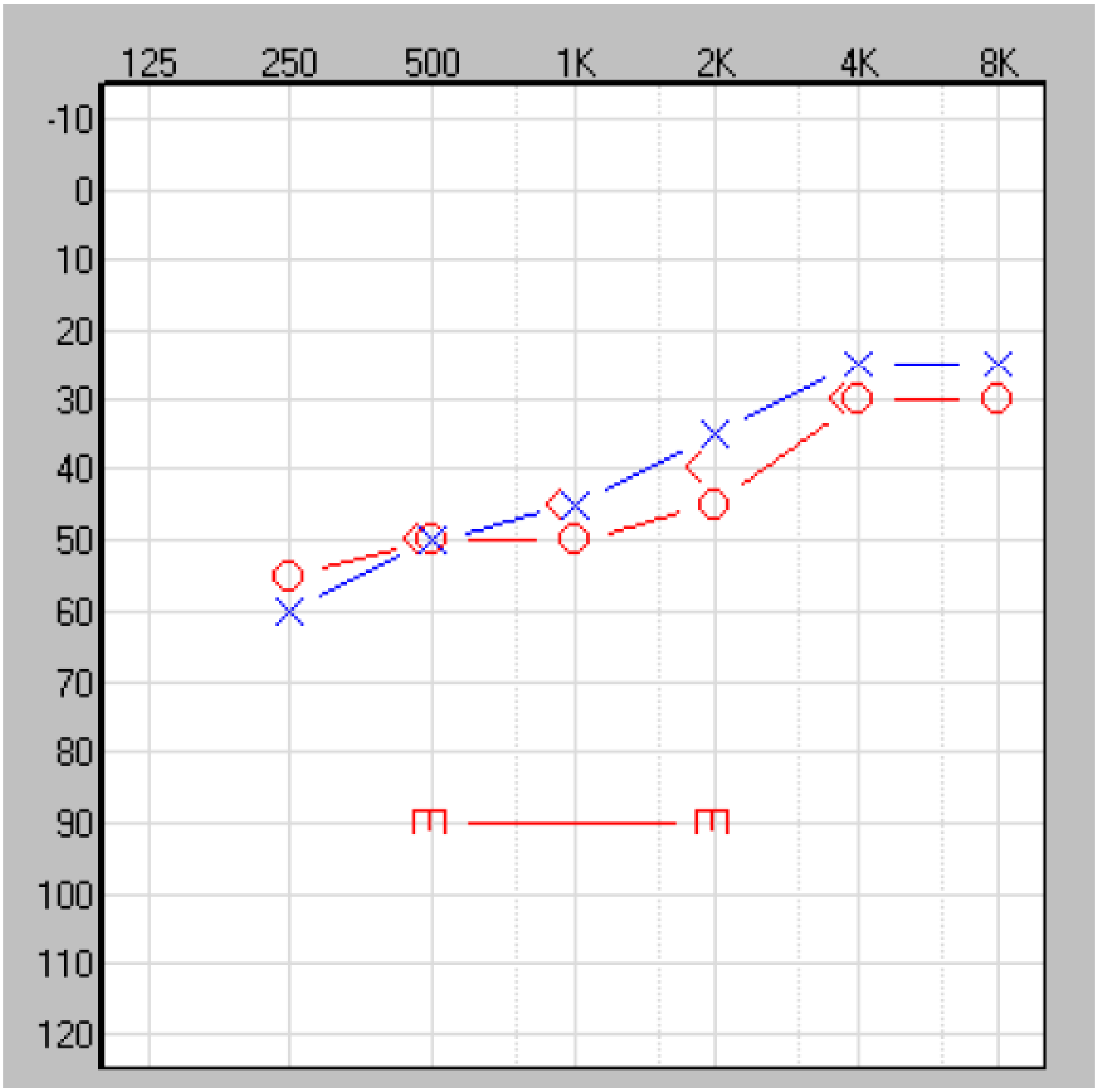
	SRT	MCL	WR%	UCL	SNR loss
Right	40	65	84	85	4
Left	40	65	88	85	

4



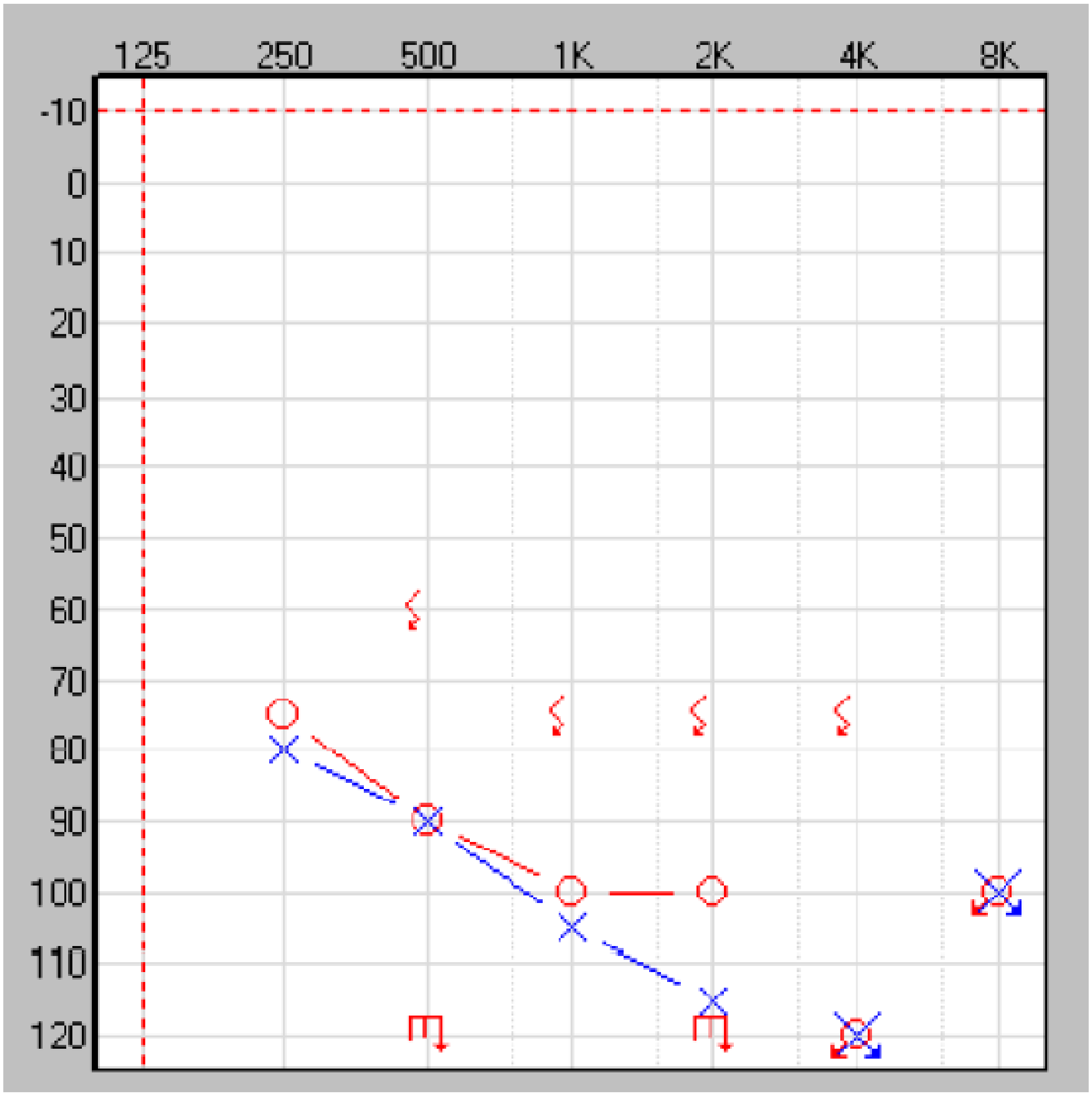
	SRT	MCL	WR%	UCL	SNR loss
Right	65	90	76	110	12
Left	40	65	80	90	

5



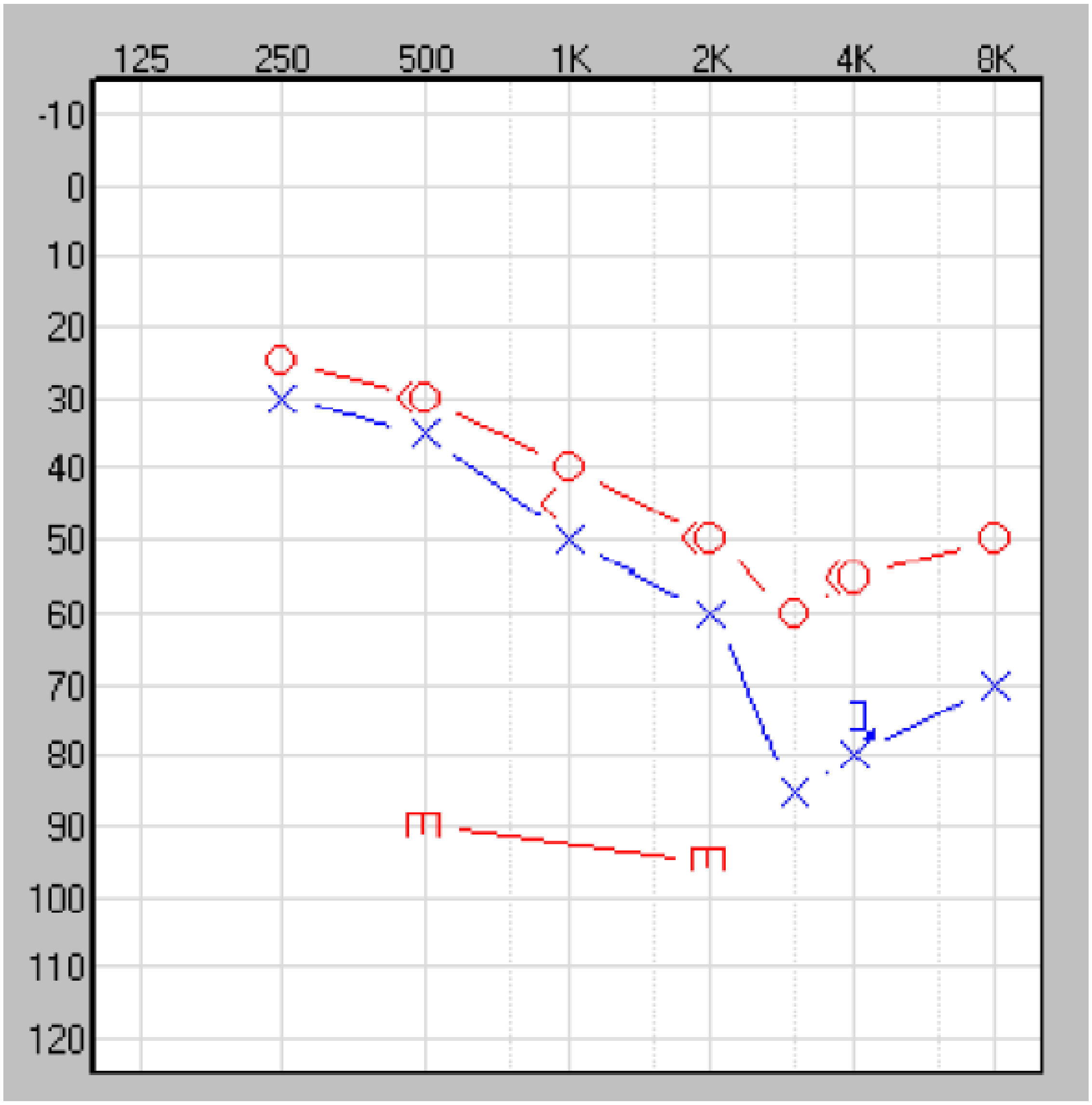
	SRT	MCL	WR%	UCL	SNR loss
Right	45	70	88	90	3
Left	40	70	92	90	

6



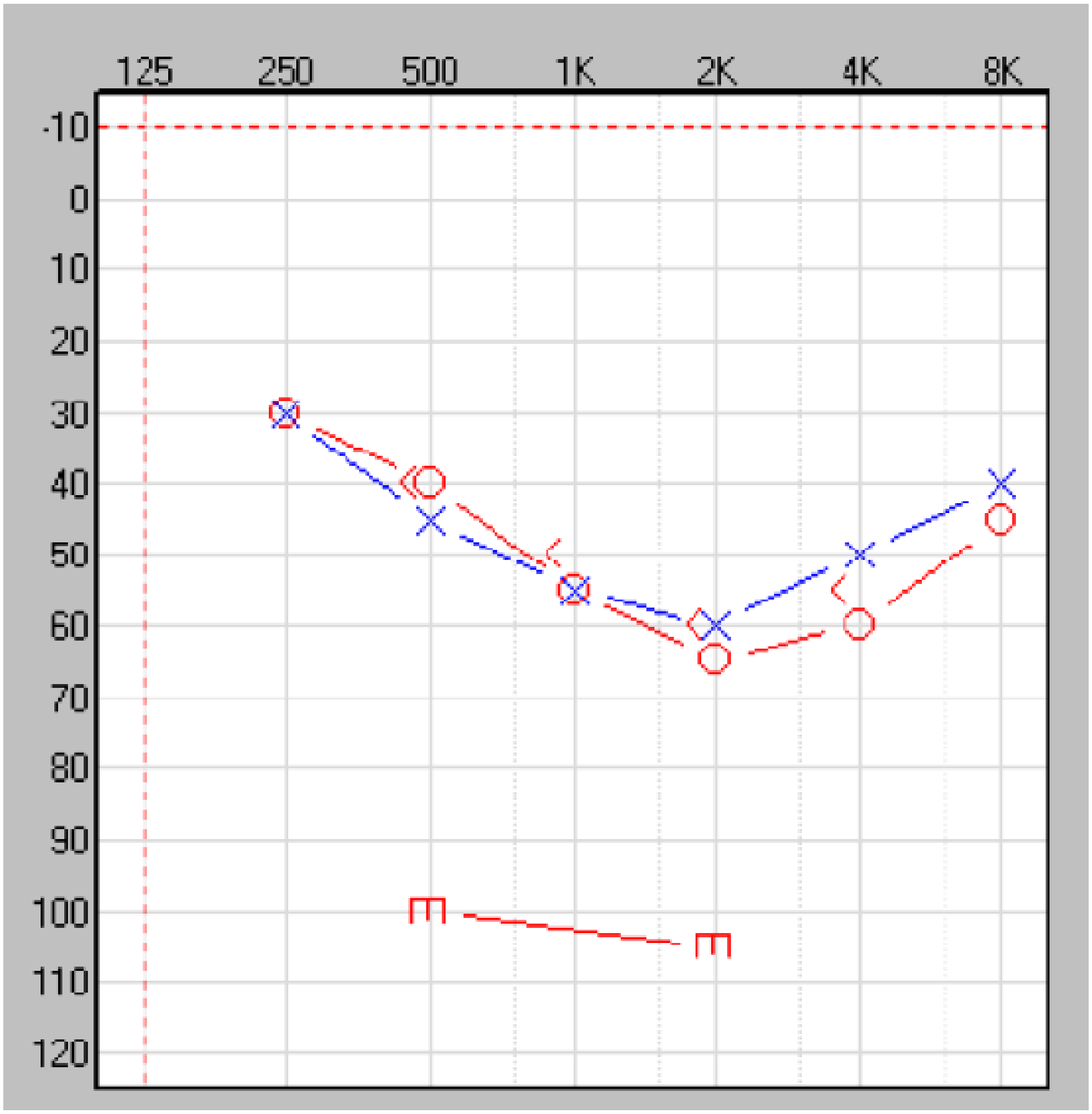
	SRT	MCL	WR%	UCL	SNR loss
Right	95	105	36	>120	22
Left	95	105	24	>120	

v 7



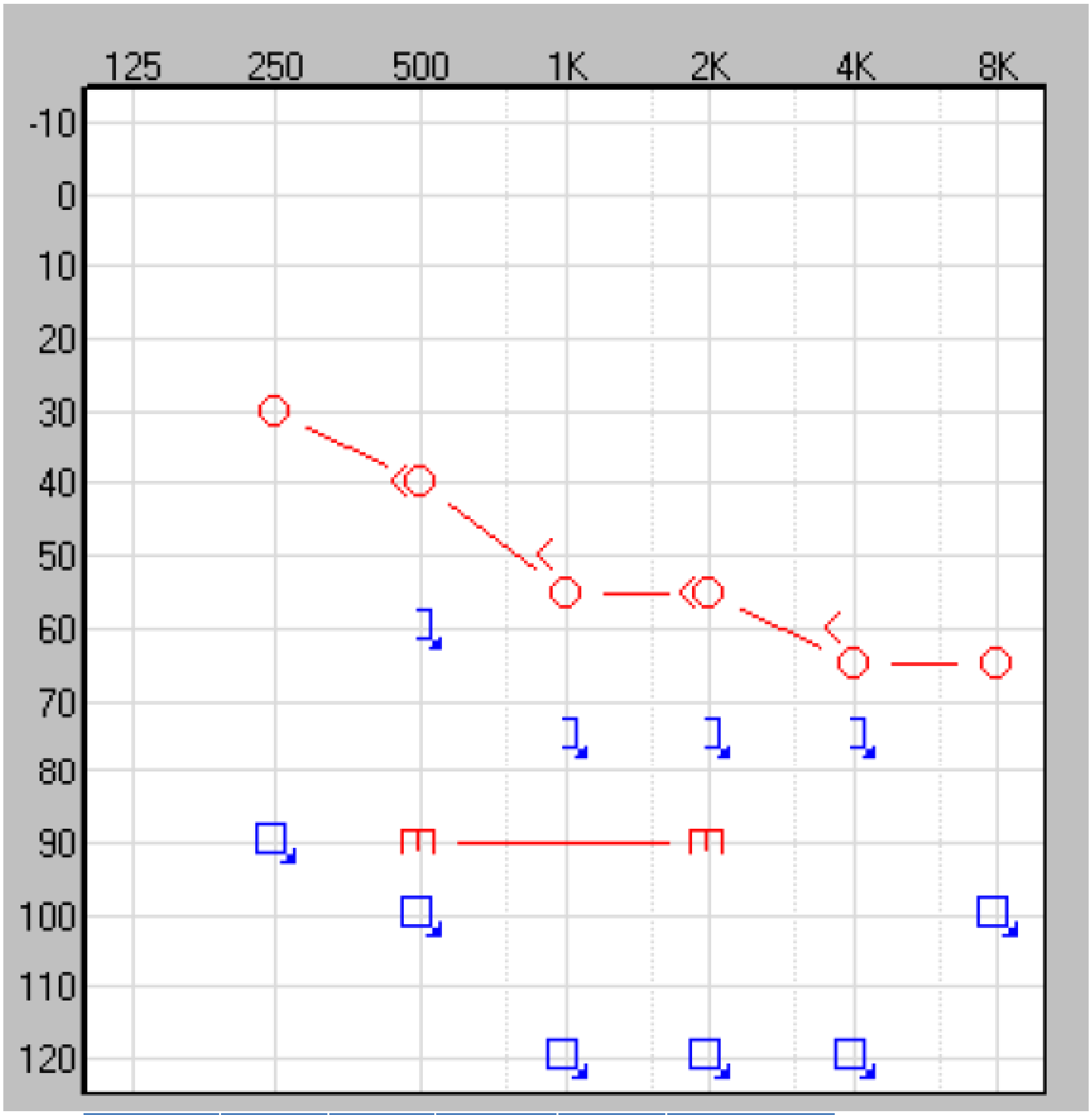
	SRT	MCL	WR%	UCL	SNR loss
Right	40	70	84	90	13
Left	45	75	76	90	

8



	SRT	MCL	WR%	UCL	SNR loss
Right	50	75	84	100	12
Left	50	75	84	100	

9



	SRT	MCL	WR%	UCL	SNR loss
Right	50	70	76	90	10
Left	NR		CNT		

10

Unit 10: Verification
Real Ear Measurement (REM) and ANSI

Objectives

Notes

Overview

How do we know that the hearing aids we fit on our patients are actually doing what we want and expect them to do? We evaluate, or confirm the hearing aids performance by **verification**. This is done by some kind of objective measurement on the performance of the hearing aid in use. The gold standard for hearing aid verification is Real Ear probe tube measurement. This will be the focus of Unit 10.

Goals

Upon completion of this module, you will be able to:

- Explain why the verification of hearing aid fitting and technology is important
- Describe alternative methods of verification
- Provide an overview of different types of equipment terminology and operation

Follow Up

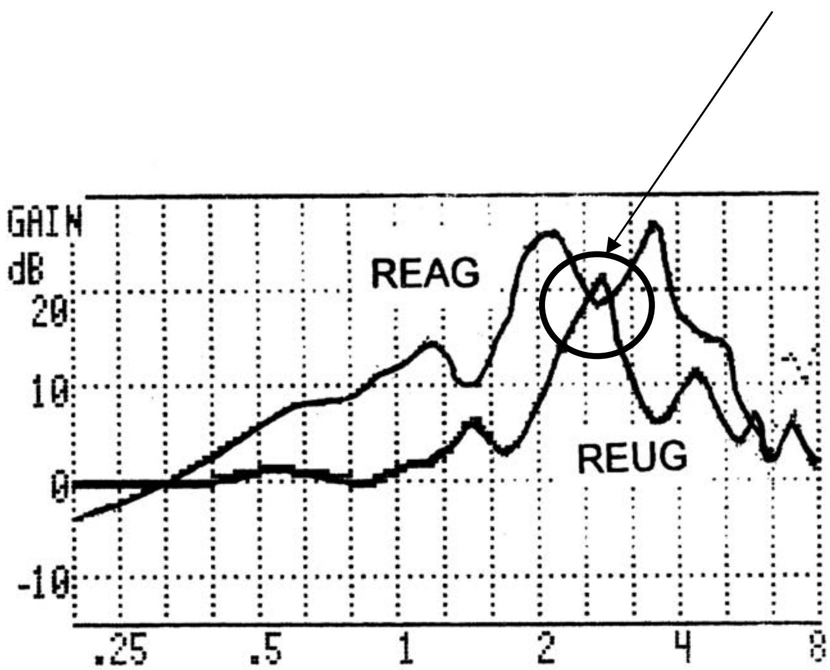
- Locate old working hearing aids, run ANSI strip on them and evaluate results
- Practice Real Ear measurements on co-workers to be prepared to work with patients.
- Review ANSI strips from current patient charts

Section 1 – Hearing Aid Verification

Notes

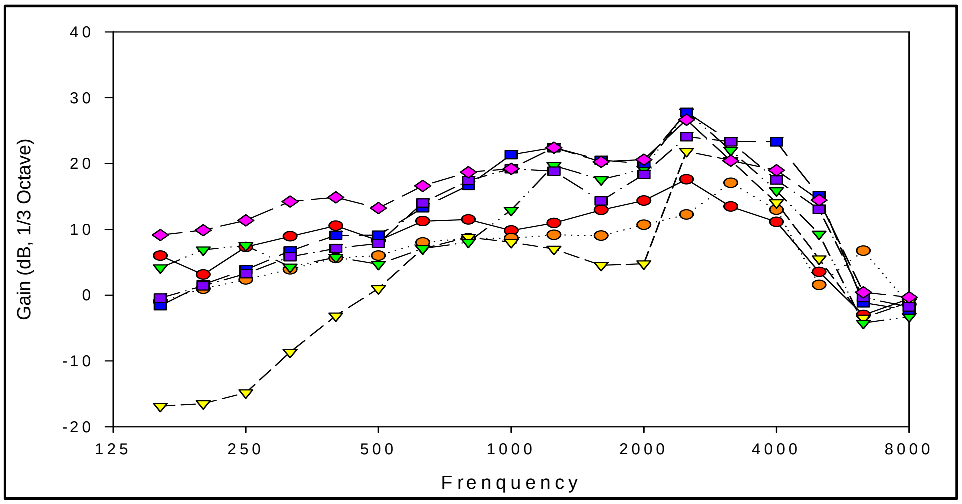
Verification procedures are more the exception than the rule. For example, less than 50% of dispensing professionals use probe microphone measures even though most professionals agree that there is no reasonable substitution for them . When it comes to validating the outcome of the fitting, things are not much better. 43% of dispensing professionals report that they “never or rarely” validate the outcome of their fittings. The bottom line: Both verification and validation are important. More on Validating at another time

Verification is about making sure hearing aid performance meets a certain fitting standard. Below is a probe measure of a hearing aid fitting. Note that at about 3000 Hz there is an insertion loss, meaning the unaided gain (REUG) is actually greater than the aided gain (REAG). .



Notes

First fit is not always the same for all manufacturers. The graph below is a first fit for hearing aids from six (6) different manufacturers. Note the differences. The provider would never know unless they were to measure using probe microphone measure that there was a difference. Not only can responses differ by manufacturer, but the acoustics of each individual ear will change the response as well.



What is the difference between verification and validation?

Verification is about making sure hearing aid performance meets a certain fitting standard/criteria.

Validation is about determining how much benefit the patient is receiving from the hearing aids. (More on Validation later)

Think of it this way: You are baking cookies. You are following an award-winning recipe. You read the recipe, go to the store, buy all the right ingredients, and then exactly follow the recipe as you make the cookies. You use your measuring cups and spoons to make sure that you get it right. Following the recipe and checking to make sure the amounts are exact is VERIFICATION.

Verification is not enough. What if your recipe has ingredients that some people just don't like? You may have followed the recipe, but the cookies still don't taste good. Tasting good is VALIDATION.

Notes

The quotes below are from the April 2010 article on Probe microphone measure and describe the reasons why this type of Verification is needed every time a hearing instrument is fit.

"In a July 2009 article, Consumer Reports concluded that about 2/3 of hearing aids are not fit correctly, that audiologists and HISs do not routinely conduct probe-microphone measures to assure that they are fit correctly, and that probe-microphone testing is a "must have" procedure for every consumer purchasing hearing aids

'In a recent issue of Audiology Today, Catherine Palmer, PhD, suggested that the failure to use probe microphone measures in the fitting of hearing aids is unethical practice."

'In a 2010 article, Sergei Kochkin and colleagues published an extensive article in Hearing Review on hearing aid outcomes, based on the data from MarkeTrak VIII.3 These data show that: (1) hearing aid satisfaction is related to the testing conducted at the time of the fitting, (2) more testing leads to more satisfaction, and (3) probe-microphone measures are one of the tests that affect these results."

Verification Using Sound Field and Probe Microphone Measures (Real-Ear Measurement)

There are two times to verify the performance of a hearing aid either during the delivery or at the first week follow-up. Both of them require the patient to be wearing the hearing aid(s). Aided sound field testing is much like doing a hearing test: however, the patient is wearing the hearing aids and the sound comes from speakers rather than the headphones

Real-ear measurement means measuring how the hearing aid(s) are performing in-situ, literally, in place, or as worn by the user. Another term commonly used for real-ear measurement is **probe microphone testing**. Either term is okay. **Speech**

Mapping is a specific way of performing real ear measurements that has become popular in recent years. The important thing is that you take the time to complete the test during the initial fitting or at the first week follow-up. The theory of Real-Ear measurement is to verify the performance of hearing instruments while the patient is actually wearing them.. As you have seen earlier ear canal resonance can be different from ear to ear and patient to patient. The most important **rule** regarding verification is **it must be done!**

Notes

Sound field audiometry is when hearing instruments are verified using an audiometer and loudspeakers attached to the audiometer. One speaker is needed to determine functional gain. Two speakers are needed if word recognition is being measured in competing noise. You must calibrate the soundfield (where the patient is seating) by a calibrated tone out of the audiometer and measured with a sound level meter (SLM) where the patient's head will be and set to 60 to 65dB.

During sound field audiometry, a procedure similar to the modified ascending/descending technique is used to determine a threshold at 500, 1000, 2000 and 4000 Hz. However, since the audiometer is calibrated for headphones and not sound field (some clinical audiometers may be calibrated for sound field), the professional uses the audiometer "dial reading" as a threshold value. The obtained value is not a true threshold, but can be used to determine increase or decrease of sensitivity, even though it is not a "threshold."

When we have obtained unaided and aided dial readings for sound field, the results we obtain are referred to as "functional gain." Functional gain is the difference between the unaided and aided thresholds at each frequency. Now we know whether or not the hearing aids are working, and how much gain they are providing at 4 frequencies. Speech testing may also be administered to show improvement with either SRT or WRS. Again this test is done unaided and then aided using the appropriate word list with recorded speech.

	500hz	1000hz	2000hz	3000hz	4000hz	6000hz
Unaided	30dB	40dB	45dB	55dB	60dB	70dB
Aided	15dB	25dB	25dB	35dB	50dB	55d

The Difference is Function Gain (used with a prescriptive formula

Check Your Understanding.

Real Ear Measure is not needed to fit hearing aids.

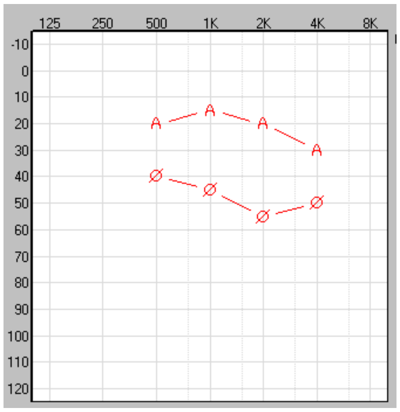
True or False

There are two different types of verification: **Soundfield** and **Probe Tube**. Both are similar in procedure, but different giving us different results. Soundfield uses an Audiometer and Speakers. This tutorial will explain each for procedure, results and why one might be better for today's modern hearing aid technology.

Aided Sound Field Testing

Sound field testing for verification has been around for a long time. This was used extensively when analog hearing aids were being fit. It measured frequency specific puretones and the result was Functional Gain at frequencies directly related to speech understanding.

In aided sound field testing, the patient is positioned one meter from a soundfield speaker and at 0 or 45 degrees azimuth relative to the speaker. The speaker is at ear level. Testing is administered first unaided, then aided, using a warble-tone or frequency modulated pure tone to prevent a standing wave. The distance and azimuth need to be the same for both measurements. For monaural testing, the non-test ear is plugged with a noise reduction plug, or air conduction masking is applied. The volume control of the hearing aid(s) is set at preferred user setting.



In the audiogram to the right, the functional gain (difference between aided and unaided thresholds) are 20 dB at 500 Hz, 30 dB at 1000 Hz, 35 dB at 2000 Hz and 20 dB at 4000 Hz.

Check Your Understanding

Fill in the blanks. The two different types of verification are ________________ and ________________________.

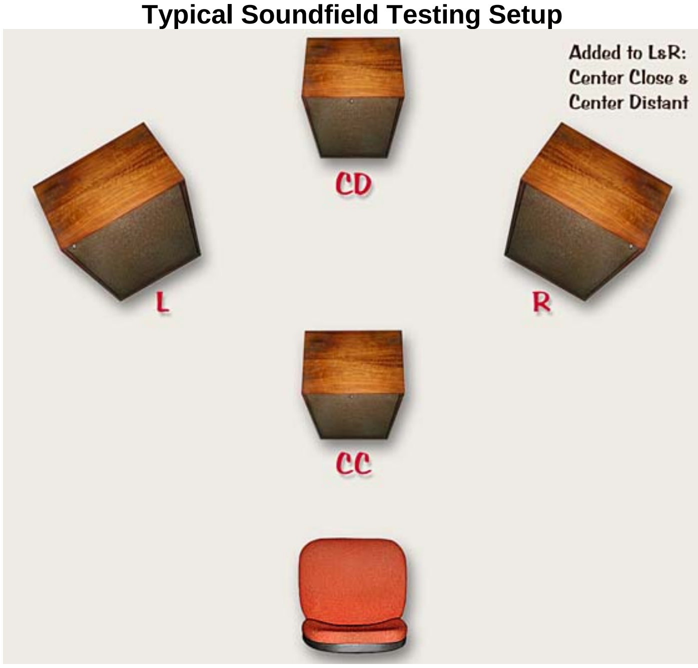
Notes

Notes

The other use for Soundfield testing was the ability to do speech testing, which is similar to puretones except this time using some kind of speech type signal (live voice or CD) with an Audiometer and speaker. This could be accomplished in a couple of different ways. The first using Spondee (SRT) word lists and comparing Unaided to Aided as explained previously. A positive result would show an improvement in scores. The other would be to use Word Recognition (PB) word lists. Again a positive result would be an improvement in percentage of words correct.

Soundfield could also be used to determine speech understanding in noise. This was done by placing another speaker, behind the patient and introducing competing noise. Special care needed to be used when setting up the Soundfield environment. See diagram below.

Typical Soundfield Testing Setup



Note: For licensing purposes in some states, professionals may be expected to apply a linear formula, such as NAL-R, POGO, Libby, Berger, 1⁄3, or 1⁄2 gain rule to an audiogram. The use of soundfield may also be required. These formulas use an Analog formula.

Notes

Examples of prescriptive formula are NAL-RP, Berger, POGO, 1/2 Gain etc. These where used to fit analog technology and the decided gain from the shape and degree of the loss. Two examples of calculations used for Berger and ½ gain rules.

Berger	1/2 Gain
Calculation method for the target gain:	
	250Hz AC/2 1500Hz AC/2
250Hz AC/2 1500Hz AC/1.55	500Hz AC/2 2000Hz AC/2
500Hz AC/2 2000Hz AC/1.5	750Hz AC/2 3000Hz AC/2
750Hz AC/2 3000Hz AC/1.7	1000Hz AC/2 4000Hz AC/2
1000Hz AC/1.6 4000Hz AC/1.9	6000Hz AC/2
6000Hz AC/2	
Correction:	
+ 0.25 * [AC - BC]	

Check Your Understanding

What components are needed for Soundfield testing? Select all that apply.

a. Speaker

b. Probe tube

c. Azimuth

d. Audiometer

Limitations of Soundfield Verification

Soundfield has limitations for verification of modern technology. Today we use Wide Dynamic Range Compression circuitry. This type of technology looks at multiple input levels (soft-average-loud) in multiple channels. These can be adjusted with a computer program which uses a different fitting formula developed for this technology. The formulas (NAL-NL1 and DSL I/O) are designed for use with the new WDRC technology. These are explained below. Most manufacturers use the NAL-NL1 to set “First Fit” parameters. From what you learned about use of probe microphone measure earlier and what the experts tell us is those first fit settings are not correct >90% of the time. If we take the time to measure the needs of our patient, we should use those measures most likely to manage their needs.

NAL-NL1

Notes

Prescribes amplification for non-linear hearing instruments. The new prescription procedure, called NAL-NL1, is similar to its predecessors (NAL-R and NAL-RP) in that it aims to maximize speech intelligibility. For average input levels, e.g. speech at conversational levels, NAL-NL1 prescribes similar gain as NAL-R and NAL-RP.

The NAL-NL1 formula has the additional aim that the overall loudness of sound for hearing-impaired people should be equal to or less than for individuals with normal hearing. Contrary to other fitting procedures for non-linear devices, which mainly aim at normalizing loudness in each frequency band, the NAL-NL1 prescription tends to equalize loudness of speech bands across frequencies at least for mild and moderate hearing loss.

Recently, a new version of this prescriptive formula has been released by the National Acoustics Laboratory – NAL-NL2. This new formula is to better fit the needs of the patient from the first fit. The formula was designed to obtain immediate wearer acceptance from amplification.

DSL I/O* (More common in Pediatric Fittings)

The calculation of the target formulae with DSL I/O is normally implemented based on narrowband test signals. For the fitting of some instruments, the DSL I/O targets were modified to also allow broadband signals. This modification is necessary because hearing instruments which incorporate multichannel dynamic compression, will process the channel specific power of these signals independent of the overall power of the broadband signal itself. For these, the correction of the target gain and output is achieved by calculating the level of the broadband test signals within octave bands and then using this signal level as the input for the DSL I/O target formulae.

The take-away message for the you in this chapter is even though there is a prescriptive formula it is based upon averaged data to get you to a starting point for your patient, you must verify the fitting *in the individual patients ear*. When you have taken the time to measure the needs of your patient (learned in another chapter) now is the time to meet those needs and verify with Real Ear Measures.

Probe Microphone (Real- Ear) Verification

A more familiar method of verification might be **probe tube measurement**. Probe tube measurement involves placing a flexible probe tube in the ear canal to within 3-5 mm of the eardrum. Advantages of probe tube REM include that minimal patient participation is necessary, reducing the total amount of time involved in verification.

Each manufacturer has their own guidelines for equipment usage. Make sure you read each manual so you know how to operate their equipment. Traditionally, pure tones or speech-weighted stimuli are used to measure the performance of the hearing aid in situ. One ear can be evaluated at a time, so no need to plug the non-test ear.

Notes

The following information is designed to acquaint you with probe microphone terminology.

Remember ANSI? ANSI S3.46 is the probe microphone standard established in 1998. Many of the terms used in the standard are different than those used today in clinics and those published in the book by Mueller, Hawkins, and Northern, “Probe Microphone Measurements; Hearing Aid Selection and Assessment.” Generally speaking:

- **RE** refers to *Real Ear* (the first two letters are always RE)
- The third letter **U** refers to *unaided*
- The third letter **A** refers to *aided*
- The third letter **O** refers to *occluded*
- The third letter **S** refers to *saturation*
- The third letter **I** refers to *insertion*
- If the measure refers to SPL measured in the ear canal, the last letter will be an **R** for *response* If the measure refers to a difference measure, generally used when input has been subtracted from output, the last letter will be a **G** for *gain*.

Probe Microphone Terms (ANSI S3.46)

REUR—Real-Ear Unaided Response—SPL as a function of frequency, at a specified measurement point in the ear canal with the ear canal unoccluded (i.e. no hearing aid in the ear).This is commonly referred to as “ear canal resonance.”

REOR—Real-Ear Occluded Response—SPL as a function of frequency, at a specified measurement point in the ear canal, for a specified sound field, with the hearing aid in place and turned off. This is used to examine the effect of a vent on the canal resonance.

REOG—Real-Ear Occluded Gain—Difference in decibels, as a function of frequency, between the SPL at a specified measurement point in the ear canal and the SPL at the field reference point, for a specified sound field with the hearing aid in place and turned off. **REOR (REOG)** measures are used to assess the effects of venting (planned or unplanned). To assure that unwanted vent resonance is not present. This is not a measure of the occlusion effect.

REAR—Real-Ear Aided Response—SPL as a function of frequency, at a specified measurement point in the ear canal, for a specified sound field, with the hearing aid in place and turned on.

Notes

REAG—Real-Ear Aided Gain—Difference in decibels, as a function of frequency, between the SPL at a specified measurement point in the ear canal and the SPL at the field reference point, for a specified sound field, with the hearing aid in place and turned on**. REAR & REAG** clinical use is to **(1)** validate a prescriptive approach, **(2)** calculate the REIG **(3)** to determine if special functions of the hearing aid, such as compression, are providing the desired effect. **REAR** of the hearing aid should have a fairly similar configuration to the curve that you see in the 2cc coupler.

REIG—Real-Ear Insertion Gain—Difference in decibels, as a function of frequency, between the REAG and the REUG, taken with the measurement point and the same sound field conditions.

REAG– REUG = REIG

REAR + REUR = REIR

These terms are not part of the ANSI specifications; however, they are good to know.

RESR—Real-Ear Saturation Response—SPL as a function of frequency, in the ear canal, with the hearing aid in place, turned on, with the VC adjusted to full-on (or just below feedback) with a 90 dB signal. This measurement is to determine if the maximum output of the hearing aid falls within the desired levels across frequencies based on the patient’s loudness discomfort levels (LDL’s). It is also used to determine if the maximum output of the hearing aid is at a “safe” level. You can predict the **RESR** if you measure the **RECD**.

Here is a simpler view of these probe microphone measures that may help in your understanding.

- **REUG**—the ear canal that they have had all their lives.
- **REOG**—What the hearing aid does in the ear as before you turn it on.
- **REAG**—What the hearing aid gives the patient when turned on (but not accounting for what might have been taken away).
- **REIG—**What the patient has when he walks out of the office with a hearing aid in his ear that he didn’t have when he walked in.

What we obtain with probe tube measurements is called insertion gain, not functional gain. Again, we calculate desired, or target, gain using the pure tone air conduction audiogram, which has already been obtained.

Target Matching

Now a word or two about target matching. How much is enough? How much is not enough? How much is too much? Probe tube measurement gives us a visual aspect of target matching. Generally speaking, it is important not to be more than 10dB below target at 2000 Hz, as *2000 Hz is the most important speech frequency.*

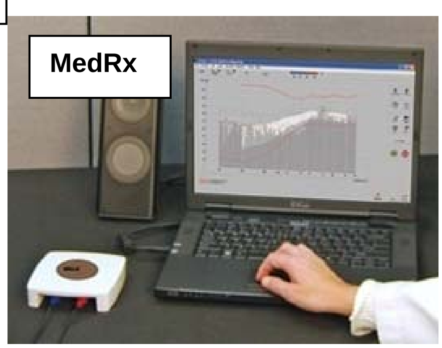
Note: It is important to remember, matching target is the goal, making sure the response follows the desired curve might be the best that can be done at the beginning of a fitting. Make sure the patient is comfortable with the sound when they leave your office. .

We get an increase of intensity due to binaural effects. Since we are verifying monaurally, being slightly below target is acceptable, especially if the most comfortable level (MCL) is reduced. If the patient's MCL is high, the patient might prefer to be slightly above target. We want to be as close to target as possible, **AND** achieve a positive subjective response to the fitting from the patient. It is only after a reasonable match to target has been achieved via real-ear that a subjective response (i.e. "So how does that sound to you?) should be requested.

These formulas are used with the Real Ear Measure (REM) by means of a probe tube placed down in the ear canal. This method gives us a more accurate measure of what is really happen in the individual ear canal. There are many types of equipment some are seen below.

AudioScan

Frye



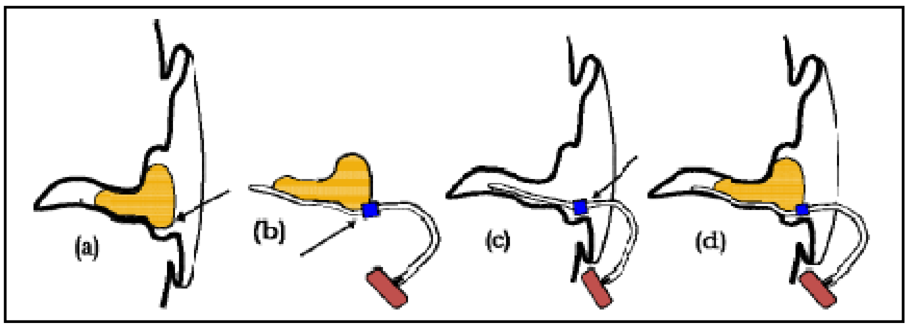
MedRx

Notes

Notes

Probe tube placement is individual to the equipment manufacturer (see individual probe assemblies). On the next page you will see four different types of probe tube assemblies.

To determine the proper placement of the probe tube, you may use a method such as that pictured below. The first drawing (a) shows in cross-section a canal style hearing aid in the ear canal. Taking the hearing aid out of the ear (b) place the movable marker (or mark the probe tube with a pen) at the point that would fall at the intertragal notch when the tip of the probe tube is situated sufficiently beyond the end of the canal portion of the hearing aid. Then (c) place the probe tube into the unaided ear with the marker at the intertragal notch. Finally (d) carefully insert the hearing aid over the probe tube.



Here are some general rules for probe tube placement in the ear.

30 MM Rule

- The length of the average ear canal is 25-mm.
- The distance from the opening of the ear canal to the intertragal notch is about 10-mm (a total of 35-mm)
- Therefore, if you place your tube marker at 30-mm, at the intertragal notch, the tip of the tube should be about 5-mm from the TM (the 5-mm rule)

5 MM Rule

- Try to place the tip of probe tube within 5-mm of TM
- Try to place the tip of probe tube 3-5-mm beyond tip of hearing aid (or earmold)

Check Your Understanding

What is the total length of the ear canal?

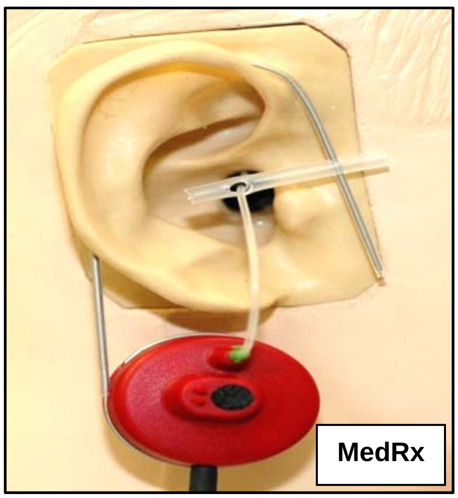
a. 25 mm

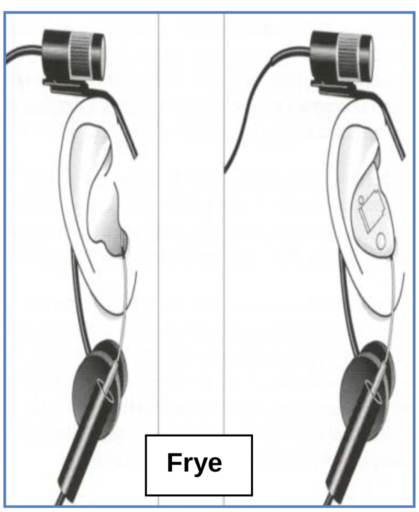
b. 35 mm

c. 30mm

d. 42mm

Probe assemblies





Notes

Section 2: Probe Microphone Results

Notes

The question of when to perform verification is always asked. The first and most important rule is "ALWAYS VERIFY". Hearing care professionals who are successful follow this simple rule. The "when" of the verifying part will differ from one hearing care professional to another. Some will verify at the time of the fitting, or first fit. Others say that if the patient is comfortable with their voice, third parties voice and the hearing care professional's voice then certainly verification needs to be done at the first week follow-up.

Definitions for REM terms used in the calculations of responses.

REUR (Real Ear Unaided Response) is the probe tube measured output dB SPL response curve without a hearing aid in the test ear. Related term: **"REUG"** (Real Ear Unaided Gain)

REAR (Real Ear Aided Response) is the probe tube measured output dB SPL response curve of a hearing aid, turned on, and measured in the ear canal for a given input signal. Related term: **"REAG"** (Real Ear Aided Gain)

REIG (Real Ear Insertion Gain) is the probe tube measured response gain curve with a functioning hearing aid in the test ear. **REAR-REUR=REIG**.

These measures are for a tone response.

The calculations and results from the Real Ear measure (REM) are displayed below.

The calculation is:

Aided Actual Performance Measured at the TM REAR	Minus	Unaided Ear Canal Resonance REUR

=	Actual Gain Actual REIG

Notes

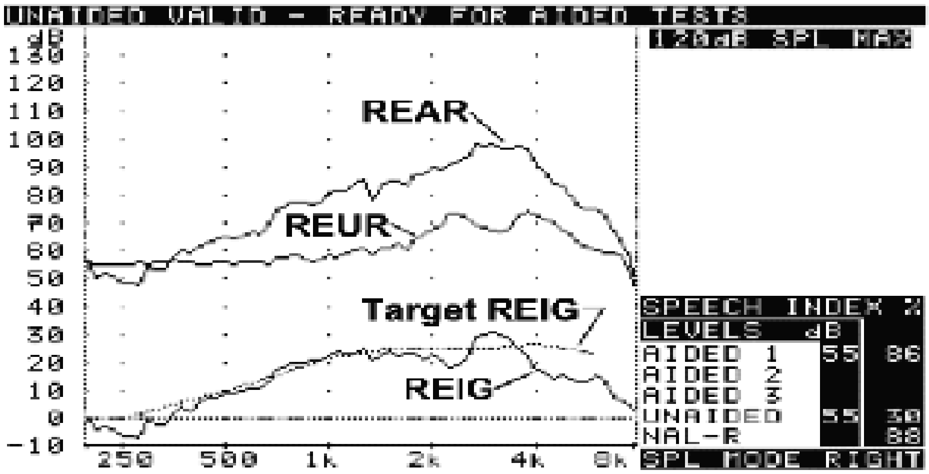
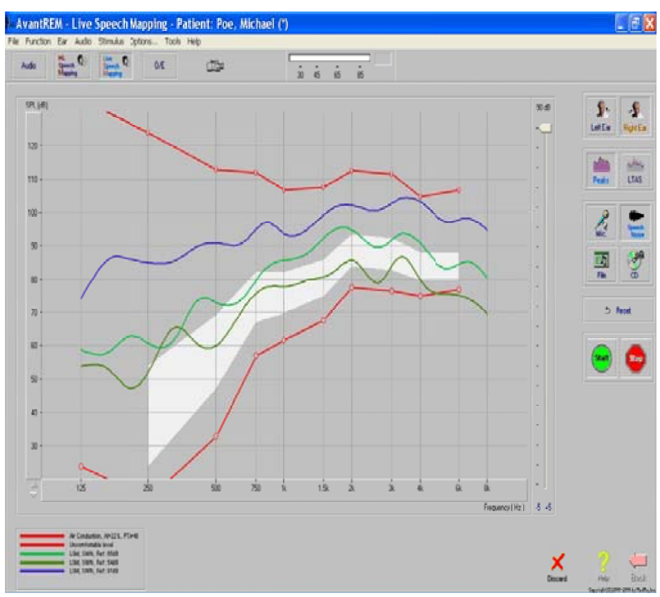


Figure 6. Example of the calculation of real-ear insertion gain (REIG) on the Audioscan RM500. Also shown is the target REIG and the previously measured real-ear unaided response (REUR) and real-ear aided response (REAR).

MedRx™ Results (Pure Tone)

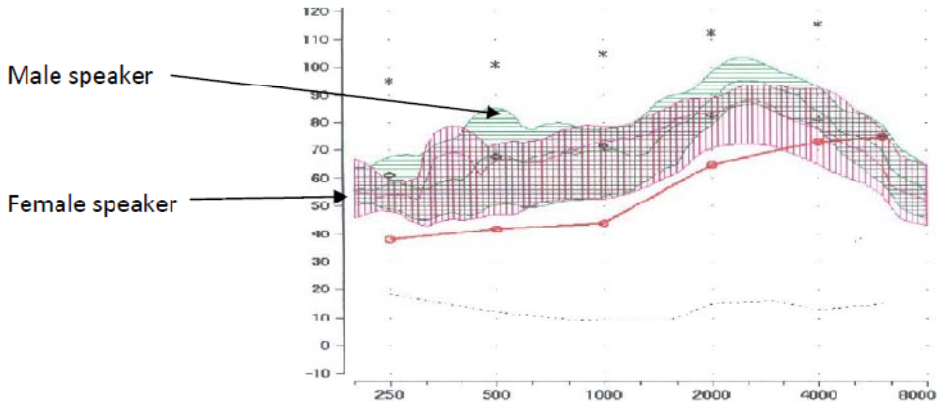
These last verification results are for puretone results. The experts have told us that speech type signals are a much better way to determine accurate response for WDRC processing. There are three different types of speech signals.



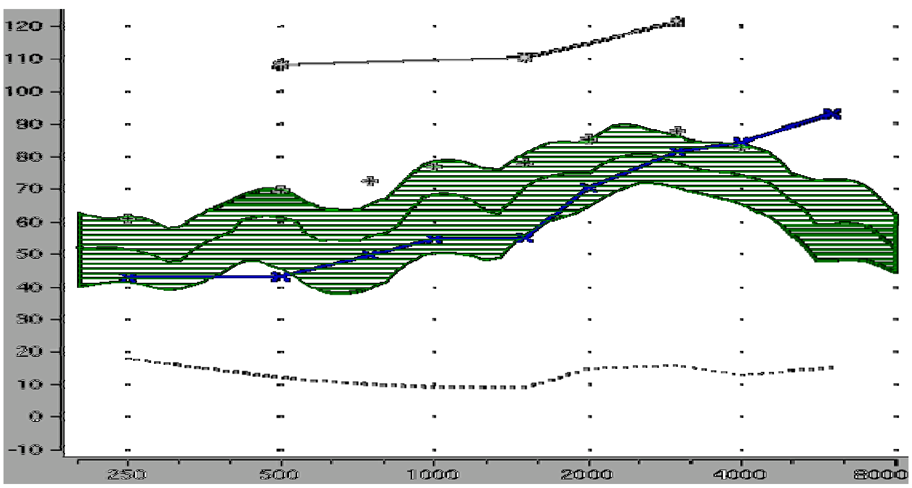
The first is **Live Speech**, meaning this is either the hearing care professional's voice or third parties' voice through the air picked up by the hearing aid measured by the probe tube. A positive to this type of REM is patient and third party can actual see their voice being used The best thing with live voice testing is you can actually program a hearing aid to the needs of the patient and the person they need to understand. . One problem with live voice is the ability to reproduce the results every time. Different voices produce different results. Below is an example of MedRx Live Speech

Notes

Below is an example of two different voices using live speech to measure.



The second type of speech signal is **recorded speech**. This allows the providers to test using a recorded speech signal. This has the advantage of being able to be verify and more importantly can be repeated even by different providers on the same patient using the same signal. Below is a AudioScan™ recorded speech signal (65 dB) average being used to match a target for average speech.



The third type of speech type testing is **recorded speech-like sounds, sometimes called ICRA signals**. These are not actual speech, but are complex sound signals that that have the "envelope" of a speech signal in that they change frequently in frequency and intensity. They are necessary for testing digital processors using digital noise reduction (DNR). A DNR algorithm may interpret a conventional real ear signal as noise and reduce hearing aid output. These speech-like signals may be called ICRA, Digi Speech or Composite speech.

Notes

Notes

What other types of verification can be done or should you consider when verification of the fit is accomplished? Below are just some of the types of tests you might consider. These will verify that the hearing instrument is working to the best possible response for specific circuitry options.

- Testing Directional Microphone Front/Back Ratio.
- Assessing DNR.
- Live Speech Mapping Unaided and Aided Demonstration.
- Visualizing feedback with Live Speech Mapping.
- Measuring the Occlusion Effect.

Bottom Line: **ALWAYS VERIFY** each hearing aid fitting **EVERY TIME.**

Check Your Understanding

Male live speech and Female live speech are the same.

True or False

The ANSI Test and the 2cc coupler

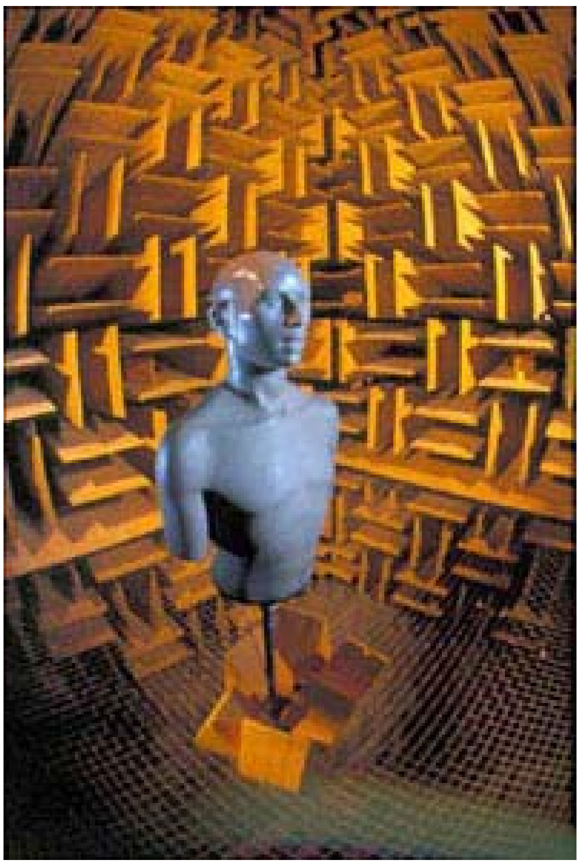
Upon receiving a hearing instrument order from the manufacturer you typically assume that the hearing aid will perform correctly, but how can you know *for sure* that it is performing correctly? Sometimes it will be necessary for you to perform your own quality control measurements to ensure each hearing instrument is performing to manufacturer specifications, therefore at some point you must learn how to measure hearing instrument performance using ANSI measurements.

ANSI stands for *American National Standards Institute*. This government agency provides specifications for almost all products manufactured in the US today, including hearing instruments. Done correctly, ANSI measurements can tell us whether or not a given hearing instrument is performing at manufacturing tolerance levels. Since a hearing instrument is a prescriptive device ordered to help correct a given hearing loss configuration as specified by the audiogram, the actual product needs to agree in its performance characteristics with data published in a product data manual. A manufactured hearing aid is measured at several points through the production line in a test box that may be given various names: Frye box, 2cc test box, HAT (Hearing Aid Test) box, for example.

A test box typically comprises an **anechoic chamber**, a **sound source**, and a **measurement device**. An anechoic chamber is a controlled environment designed to provide absolute sound absorption or no echo. The hearing instrument is placed inside the anechoic chamber, a proper ANSI testing protocol is followed (usually by locating and pressing the button marked “ANSI”), and a series of readings are produced.

NOTE: A contrast to the anechoic chamber that exercises strict controls on sound propagation is the “free field,” - an environment in which there are no boundaries to the propagation of sound. KEMAR (the **K**nowles **E**lectronics **M**annequin for **A**coustic **R**esearch) was designed to simulate “normal” human torso, head and ear acoustic effects, measuring hearing aid performance with a Zwislocki coupler in a free field.

The anechoic chamber uses a 2cc/2cm3 (cubic centimeter) coupler to measure hearing aid performance. This coupler was designed to approximate the size of the average male ear canal. Typically you will find two couplers available: The HA1 coupler is designed for testing custom in-the-ear hearing aids, the HA2 coupler for testing BTE/post-auricular/body/or eyeglass hearing instruments.



Notes

Setting Up the 2cc Anechoic Chamber

The protocol for setting up the ANSI test box is:

- When present the VC is set to the full-on position.
- Any trim pots or switches are set to the neutral position (i.e., having no effect on the hearing instrument response).
- Any vents are sealed off at the canal tip with putty or FunTak™.

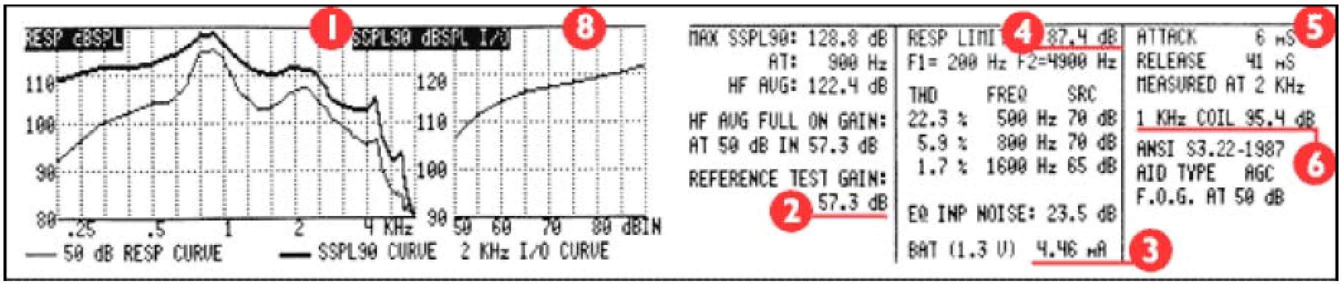
The hearing instrument is attached to the appropriate coupler and is placed inside the test box, the ANSI sequence is initiated, and a series of tests are run and displayed.

For more details on ANSI standards and using the 2cc coupler & test box see the table below.

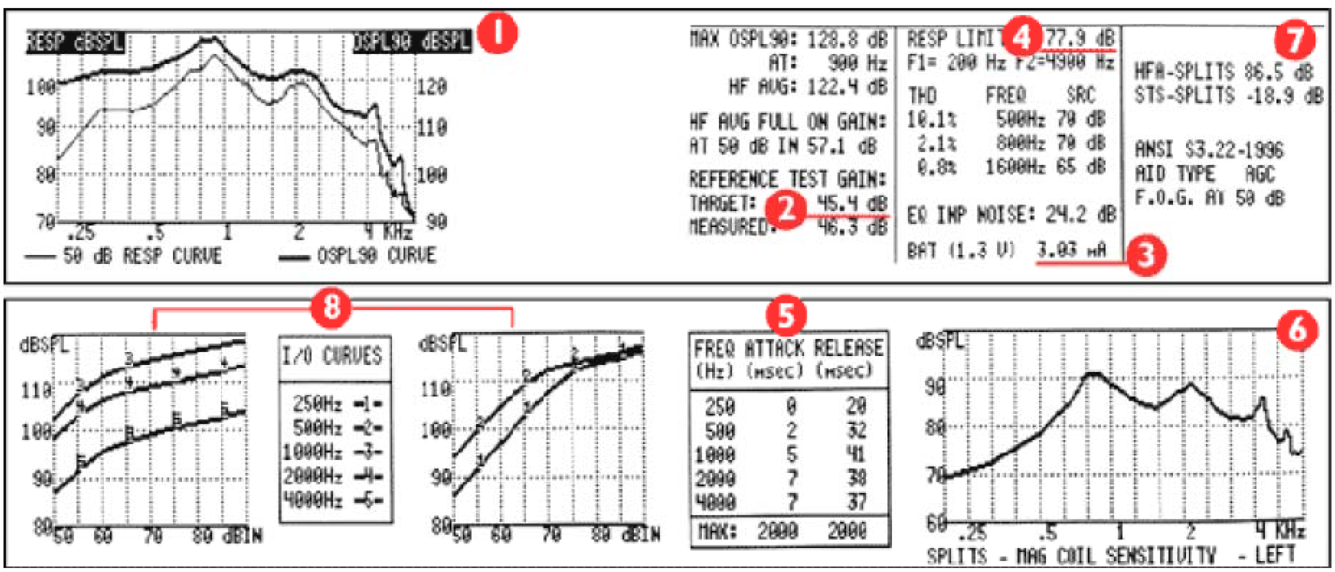
ANSI Test Standards

ANSI	Input SPL	Frequency (Hz)	VC/Gain Settings
1. **SSPL90** (ANSI 1989) **OSPL90** (ANSI 1996 & 2003)	90 dB	200 to 5000 Hz	Full on/Maximum
2. **Max SSPL90/OSPL90**	90 dB	Highest peak between 200 Hz to 5000 Hz	Full on/Maximum
3. **High Frequency Average (HFA) SSPL90/OSPL90**	90 dB	Average of points at 1000, 1600 & 2500 Hz	Full on/Maximum
4. **Full On Gain (FOG)**	Linear 60 dB Non Linear 50 dB	200 to 5000 Hz	Full on/Maximum
5. **Max FOG**	Linear 60 dB Non Linear 50 dB	Highest peak between 200 Hz to 5000 Hz	Full on/Maximum
6. **HFA FOG**	Linear 60 dB Non Linear 50 dB	Average of points at 1000, 1600 & 2500 Hz	Full on/Maximum
7. **Reference Test Gain Position (RTGP)**	60 dB	Average of points at 1000, 1600 & 2500 Hz	Varies SSPL90-17=RTGP
8. **Reference Test Gain (RTG)**			SSPL90-77=RTG
9. **Frequency Response**	60 dB	200 to 5000 Hz	RTGP or RTP
10. **Total Harmonic Distortion (THD)**	70 dB 70 dB 65 dB	500 Hz 800 Hz 1600 Hz	RTGP or RTP
11. **Equivalent Input Noise (EIN)**	60 dB	1000, 1600 & 2500 Hz	RTGP or RTP
12. **Telecoil**	Magnetic Field	1000 Hz	Full on/Maximum
13. **Battery Current**	70 dB	1000 Hz	Full on/Maximum
14. **Input/Output Curves (I/O Curves)**	50-90 dB	2000 Hz	Full on/Maximum
15. **Attack and Release Times**	55-80 dB	2000 Hz	Full on/Maximum
16. **Slope**	Difference between gain at 500 Hz and peak gain on the frequency response curve		

ANSI '87



ANSI '96



Key:

1: change in terminology
2: reference test gain
3: battery current
4: response limit
5: attack and release
6: telecoil measurement now full frequency test
7: new telecoil calculation
8: multiple input output curves

Check Your Understanding

1. In Aided sound field testing the sound source comes from:
    a. Hearing Aids
    b. Headphones
    c. Speakers
    d. Insert Plugs

2. Probe Tube Placement:
    a. The tip of the tube should be about 5-mm from the intertragal notch
    b. The tip of the tube should be about 30mm from the intertragal notch
    c. This tip of the tube should be about 25 mm from the TM
    d. The tip of the tube should be about 30 mm from the TM

3. Speech-in-noise testing may be done in:
    a. Verification Testing
    b. REM testing
    c. Probe Tube testing
    d. Sound field testing

4. For optimum speech understanding, REM measurement should not be more than 10dB below target:
    a. monaurally
    b. at 2000 Hz
    c. at 250 Hz
    d. of the fitting formula

5. Ear Canal Resonance is also known as:
    a. REIR
    b. REUR
    c. REAR
    d. REUG

6. REM gives a more accurate measure of what is really happening.
    a. in the individual ear canal
    b. in the hearing aid
    c. in the frequency response
    d. in the 2 cc coupler

7. ANSI stands for:
    a. American National Safety Institute
    b. American National Standards Institute
    c. American Noise Standards Institute
    d. American Noise Safety Institute

8. Aided Response – Unaided Response equals:
    a. RESR
    b. MCL
    c. Functional Gain
    d. Dynamic Range

9. Target Matching is:

    a. the “best fit” formula provided by the hearing aid manufacturers

    b. comparing prices at the big box retailers

    c. important to make sure that the hearing aids are meeting the patients prescriptive loss

    d. done in a hat box with a 2 cc coupler

Unit 11: Hearing Aid Programming and Troubleshooting

Objectives

Notes

Overview

Hearing professionals may deal with a multitude of manufacturers. Those manufacturers each offer a line of many different hearing aids. Each of these hearing aids may have a different set of digital and acoustic capabilities. In this section we will discuss many of the most common digital features in use today. Unfortunately, manufacturers frequently use proprietary names for these features. In this section some of the most common advanced features will be discussed in general terms. It is the responsibility of the hearing care professional to determine for themselves the nomenclature used by the vendors/manufacturers with whom they deal.

Goals

Upon completion of this module, you will be able to:

- Explain the functionality of modern Wide Dynamic Range Compression (WDRC) hearing aids
- Troubleshoot modern hearing aids using various advanced digital features
- Employ a troubleshooting decision-making matrix.
- Apply the principles of hearing aid acoustics to make changes in hearing aid response.
- Manipulate digital features/technology to address hearing aid complaints.
- Diagnose and remediate a variety of commonly-occurring hearing aid malfunctions.

Follow Up

- Read the Feature /Benefit Sheets for the Hearing Instruments that your office sells. Try to cross reference those features to the explanations in this chapter.
- If your office has demo hearing instruments, practice working with each of the programming features described in this chapter
- Ask if your office has any old hearing instruments that you can use, practice listening to them and practice repairs.

Section 1 – Programmable Hearing Aids

What are Digital Programmable Hearing Instruments?

Digital Programmable hearing instruments represent a more advanced technology than conventional hearing instruments. Programmed via a computer, these instruments provide a better way to accommodate patients' amplification needs thus allowing the hearing healthcare specialist/consultant greater flexibility and fine-tuning capability.

What are the advantages of digital signal processing (DSP) hearing instruments?

Digital hearing instruments are able to manipulate sounds in terms of Advanced Signal Processing and Digital Features. Features described below do not comprise an exhaustive discussion of these features, as new capabilities are continually being developed.

Multichannel capability

The amount of amplification (gain) a hearing instrument provides at each pitch or frequency is called a frequency response. Multichannel capability allows the hearing instrument to respond differently for low frequency sounds than for high frequency sounds. In current hearing aids the frequency response may be divided into as many as 16 or more channels. The other function of the Multi-Channel product is the ability to adjust compression kneepoint and compression ratios independently in a number of different channels.

Multi- memory capability

Some programmable hearing instruments have multiple memories, allowing them to store more than one frequency response or program. Multiple memories allow the wearer to choose from different frequency responses and signal processing options by pressing a button on the hearing instrument or remote control. The hearing healthcare specialist, and the hearing instrument wearer, working together, decides which hearing instrument responses to store in memory. The programs chosen are based on the wearer's most frequent and demanding listening situations (for example, business meetings, classrooms, amphitheaters, churches, etc.). In some hearing aids, the environmental classifier (discussed below) automatically determines the most appropriate program for the patient.

Notes

Notes

Advanced Signal Processing

This is signal-processing that automatically adjusts the amount of amplification (gain) the hearing instrument provides according to the loudness of the sound reaching the microphone.

A major component of the most advanced signal processing hearing aids is compression. In a compression hearing aid, the hearing instrument circuit constantly adjusts the amount of gain so that softer sounds can be more easily heard, while louder sounds are maintained at loud but comfortable levels. This signal processing scheme is called Wide Dynamic Range Compression (WDRC).

Wide Dynamic Range Compression (WDRC)

WDRC is input compression that employs low compression kneepoints (< 65 dB SPL) and low compression ratios (< 4:1) to assure that soft sounds are audible, moderate intensity sounds (speech) are comfortable/understandable, and loud sounds are not uncomfortably loud. You will find elsewhere in this manual an explanation of how compression works. For troubleshooting purposes, an important thing to remember is that WDRC is engaged only after the advance digital features, described below, are applied.

Maximum Power Output (MPO)

Most current hearing aids have an MPO control, the purpose of which is to set the maximum intensity OSPL that the hearing aid can produce. This is the final stage of the amplification and as such will keep all loud sounds from becoming uncomfortably loud.

Expansion

Expansion is a comfort control in most modern hearing aids, preventing annoying amplification of relatively unimportant low intensity ambient sounds (like the refrigerator compressor, for example) from being amplified. In the simplest terms expansion is the opposite of compression. Whereas compression is reduced amplification for loud sounds; expansion is reduced amplification for soft sounds. Another term used in relation to expansion is microphone noise reduction.

Check Your Understanding

Multi Channel and Multi Memory are the same.

True or False

Digital Features

Notes

Noise Reduction Capability

Some digital hearing instruments are designed to identify noise (as opposed to speech) in the listening environment and automatically reduce amplification (gain) in frequency regions where it is detected. The algorithm responsible for this defines “noise” as sounds that change little in frequency and intensity (for example engine noise or forced air) and speech as a signal that does change frequently in frequency and intensity. This may provide the wearer with increased comfort in background noise.

Environmental Classifier

The environmental classifier is an algorithm that makes a determination of what kind of acoustical environment in which the hearing aid is functioning at a given time. Based on the kind of sounds that are entering the processor, the algorithm is able to judge whether the acoustical environment is quiet or speech in quiet, noise or speech in noise, reverberant, or music. When the environment has been determined, the processor is then able to change the hearing aid program to the response that is most appropriate for that acoustical environment.

Automatic Feedback Cancellation

The automatic feedback cancellation mode of most digital hearing aids searches for feedback while the HA is being worn. If feedback is detected, the system either automatically positions a notch filter in the frequency area where the feedback is taking place, and/or generates a sound 180˚ out of phase with the feedback, creating a standing wave and eliminating the feedback. The optimized mode is able to measure the maximum amount of allowable gain prior to feedback and make certain that the gain of the hearing instrument is set below this level. Some manufacturers have advanced AFC circuits, so pay attention your particular manufacturer’s software.

Check Your Understanding

Compression and Expansion are opposites of each other.

True or False

Directional Microphones

Notes

Directional microphones are the only hearing aid technology that is proven to improve speech understanding in noise (SNR). They have been available since 1969, but only with the advent of digital processing have they become a routine component in most hearing aids. For that reason we will delve a bit more deeply into directional microphones at this time.

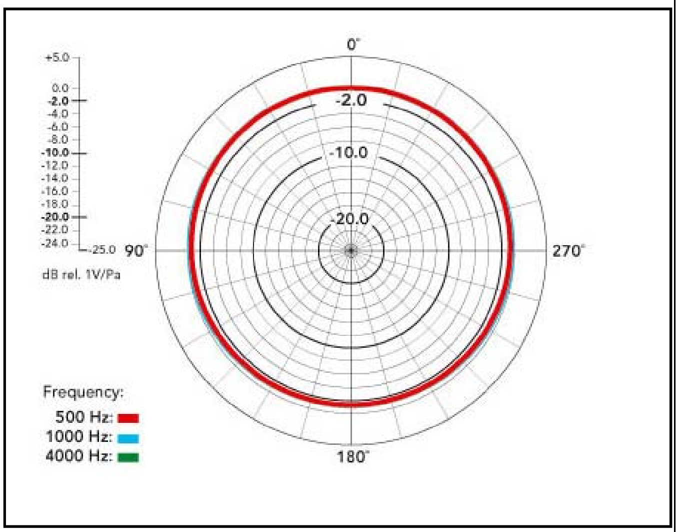
Why Directional Microphones?

Individuals with sensory hearing loss have particular difficulty understanding speech in the presence of noise. These individuals have a reduced ability to hear the high frequency components of speech so important for understanding. "Background noise" only serves to obliterate these sounds altogether.

How Do Directional Microphones Work?

Directional microphones focus amplification toward sounds originating from the front of the wearer (the most likely location of the speech signal of interest) and reduce amplification of sounds originating from the sides or rear of the wearer (most of which are likely to be unimportant to the listener , serving only to degrade the signal of interest).

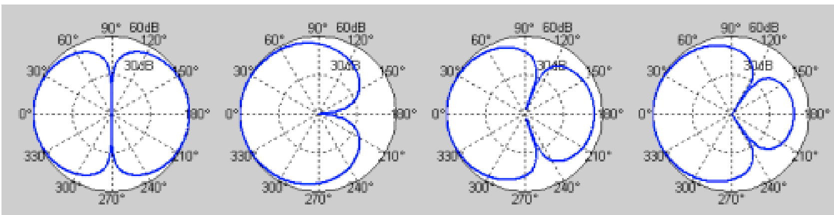
What is a Polar Plot?



A polar plot is a graphic representation of the output of a hearing aid for sounds that originate from all points 360° around a central microphone. It is a representation from above, with 0° representing the front of the head and 180° the rear, as if the observer is looking down at the "north pole" of the head – hence "polar" plot. To the right is an omni-directional polar plot. Note that the dark line is equidistant from the center of the circle (the location of the microphone) at all points of the compass.

Notes

A directional polar response, on the other hand, displays reduced output for sounds originating from certain directions. For example, one of the most common polar plots is the "cardioid" response – so called because of its resemblance to a heart (cardio). In the cardioid response, the least output, or null, is for sounds originating directly to the rear. Other common polar responses are the hyper-cardioid, super-cardioid, and bi-directional responses. In all of the plots below, the front of the head is to the left.



BI-DIRECTIONAL | CARDIOID | HYPERCARDIOID | SUPERCARDIOID

Automatic vs. Manual Directionality

The decision as to whether the hearing aid should be in directional or omnidirectional mode may be left to the user (manual) or to the hearing aid's environmental classifier (automatic). In manual mode, the user pushes a button, flips a toggle or uses a remote control device to switch back and forth from omni to directional mode. In automatic mode, it is the environmental classifier that makes the decision.

When in directional mode, the directionality may be either:

Fixed or Adaptive

When the directionality is fixed, a single polar plot is engaged (usually either cardioid or hyper-cardioid) and the plot does not change. When directionality is adaptive, the polar plot is changed automatically in response to the location of the primary source of noise.

Limitations of Directional Microphones

Notes

There are two groups of individuals for whom directional microphones may provide limited benefit.

1. Those individuals with elevated SNR loss (i.e. QuickSIN score of 15 or greater) have such difficulty in noisy situations that directional microphones would have limited benefit. For them, it is particularly important to stress the use of communication strategies (Aural Rehabilitation) and assistive listening devices to cope in such situations.

2. Hearing aids with large vents limit the benefit of directional microphones. Large (>1mm) vents, IROS vents or open domes are meant to allow natural audibility of sounds within the normal range of the hearing aid wearer. These sounds, coming into the ear naturally thru the vent, unprocessed by the hearing aid, will be omnidirectional regardless of the hearing aid microphone. Only higher frequency sounds, processed by the hearing aid, will be directional. In such instances the directional effect will be less dramatic than it would be in a more occluded fitting.

Section 2: Troubleshooting Hearing Instruments Concepts and Techniques

Notes

Particularly since the introduction of fully digital hearing instruments in 1996, hearing aids, and the approach to troubleshooting problems with hearing aids, have become more and more complex. In this module we will discuss the importance of clear communication with patients regarding difficulties they are experiencing, acoustic modifications and their effects on hearing aid performance, use of digital features in solving hearing aid problems, the general approach to troubleshooting a malfunctioning hearing aid, and how to deal with some specific hearing aid complaints.

The Troubleshooting Decision-making Matrix

Before you can begin to solve a hearing aid problem for a patient (or any other problem, for that matter), you must have a clear idea of exactly what is the problem with which you are dealing. The most efficient way to accomplish this is to follow a specific information-gathering protocol. If this is done, you will have a good chance of taking care of the problem the *first time*. This is important because, the longer a hearing aid problem takes to resolve (particularly in the early stages of hearing aid fitting and follow-up) the greater the likelihood the hearing aids will be returned. Therefore, always assume you have only one chance to solve a problem.

MarkeTrak VIII – Reasons for Non-Use

According to MarkeTrak VIII data, published by Sergei Kotchkin and the Better Hearing Institute in 2010, about 30% of new user fittings are probable failures (defined as fittings in which hearing aids are used less than 2 hours per day or not at all) and 8% of new hearing aid fittings end up in a drawer. MarkeTrak data has shown that the main reasons for non-use of hearing aids are:

- Poor benefit
- Poor fit and comfort
- Poor performance in noise

MarkeTrak VIII - Reasons for Dissatisfaction

MarkeTrak also shows that the top five reasons for patient dissatisfaction with signal processing and sound quality are:

- Chewing/swallowing sound
- Wind noise
- Sound of voice
- Richness of sound/fidelity
- Natural sounding

Appropriate troubleshooting can address nearly all of these issues.

Decision-making Matrix Steps

The steps of the Troubleshooting Decision-making Matrix are as follows:

1. Identify the problem
2. Identify possible solutions
3. Apply the most appropriate solution

Step 1: Identify the problem

This is perhaps the most important step. Without an accurate identification of the problem, none of the other steps will make any difference. Unfortunately, this is sometimes not as straight-forward as we would like. Our patients often do not have the vocabulary or understanding of hearing aid function and acoustics to accurately express the problem.

Notes

NOISE!!!

Notes

Take the simple complaint of "My hearing aids are noisy." This could mean any number of things:

- There is some "foreign noise" (hum, buzz, static etc.) in the hearing aid.
- I am annoyed by loud sounds.
- I am annoyed by low intensity soft sounds.
- I am annoyed by "circuit noise".
- The hearing aids are generally too loud.
- I have difficulty hearing in the presence of background noise.
- The presence of background sounds is annoying to me.
- Some sounds are distorted.

All of these fall into the large category of "noise", and all have different potential causes and solutions. Until the true problem is identified, beginning to search for a solution will be a waste of time and effort.

Talk to the Patient

We first need to ask probing questions to get to the bottom of the problem.

- What, exactly, does the problem sound like or exactly what is happening when the problem occurs?
- Does it happen all the time, or just occasionally?
- If just occasionally, is there anything about the acoustic environment (where they are and/or what kinds of sounds are present) that seems to be happening each time the problem occurs.
- Is there something that occurs physically to precipitate the problem – body position, furniture, walls, rooms, clothing, etc.
- Can the patient characterize the sound? What, exactly does it sound like?
- Can the patient tell you where the sound might be coming from?

Notes

- If the problem has to do with the manner in which the hearing aid processes sounds, identify the intensity of the sounds at issue. Are they soft sounds, medium intensity sounds or loud sounds.

- If the problem has to do with the manner in which the hearing aid processes sounds, identify as nearly as possible the frequency range of the sounds at issue. Are they high, medium or low frequency sounds. You may need to provide references for the patient since they may not describe sounds as you do. Are the sounds "squeaky, rumbling, sharp, piercing, booming etc?" Ask the patient if they can relate the sound to some other sound with which they are familiar.

Listen to the Hearing Aid

There is nothing like a good ear on the problem. Listen to the hearing aid through a hearing aid stethoscope. It is a good idea to use a consistent signal each time you listen to a hearing aid. In that way you will be able to compare one hearing aid to another. A useful set of speech sounds to use to evaluate a hearing aid is, "Oo, ee, ah, s, sh, t, k". This will reveal how the hearing aid responds to a range of vowel sounds as well as to high frequency consonants.

Early on in your experience with hearing aids, it is a good idea to listen to as many hearing aids as you can. Listen to hearing aids that are working well as well as to hearing aids that are malfunctioning. Since most consultants have normal hearing (or at least do not have the same hearing loss as our patients) hearing aids that are functioning perfectly well may sound odd to the normal ear. Familiarize yourself with the sound of as many hearing aids as you can.

ANSI and Real Ear

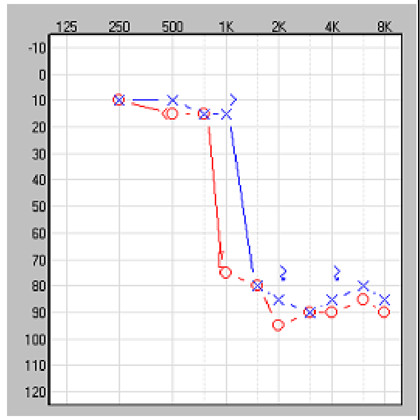
Electro-acoustic ANSI testing and real ear evaluation of hearing aid response gives a completely objective assessment of the function of a hearing aid. Real ear testing is the only way to evaluate the interaction of a hearing aid and a patient's ear. It should not be left as a last resort. Real ear should be a frequently-utilized tool in hearing aid problem-solving as it takes all the guesswork out of the troubleshooting process. ANSI testing is an under-used technology. Most real ear systems have hearing aid test boxes. It is a simple procedure to run a hearing aid and compare it to the test strip supplied from the manufacturer.

Audiometric/Physiological Problems

Unfortunately, there may not be a solution to the patient's problems that can be addressed by hearing aid manipulation. The most extreme example of this is the presence of an anacrustic (non-hearing or deaf) ear. As much as we and the patient want help, no hearing aid can restore function to an ear that has no remaining sensory cells. (I recall the patient with complete Atresia of the ear canals – he had no ear canals – who no longer wanted the bone conduction aid he had been using for years, but wanted to be fit with CICs. This was, of course, impossible.)

Extreme Precipitous Drop

A more common manifestation of audiometric/physiological realities that limit solutions is a precipitous severe to profound high frequency sensorineural hearing losses such as the one to the right. This can be an extremely problematic hearing loss for which few good solutions exist.



Step 2: Identify Possible Solutions

Much of the rest of this module will focus on a range of potential solutions to hearing aid problems and complaints. There are three categories of possible solutions:

- Compression
- Acoustic parameters
- Digital features

Step 3: Apply the Solution

This is the final step. Once the possible solutions have been identified, we select one at a time and attempt a correction. If a number of possible solutions present themselves, and we try to implement more than one, the effective solution may be masked by the application of a solution that only makes the situation worse.

We can apply "Occam's Razor" to decide which solution to attempt first. Occam's Razor says that the simplest or most obvious solution is usually the correct one. If the simplest solution is ineffective, we move on in a systematic way to the next most obvious solution, and so on until a solution is achieved. If changes are ineffective, go back to where you started and then make another change. Change upon change can frequently confound the solution.

Check Your Understanding

You should always listen to a hearing aid when troubleshooting.

True or False

Notes

Section 3: Acoustic Modifications

Notes

There are times when it is more appropriate to manipulate the acoustics of the hearing aid rather than the electronics of the hearing aid. The manner in which the hearing aid shell, earmold and tubing interact with the ear and ear canal can make a significant difference in how a hearing aid will sound. In this section we will discuss some of the factors involved in acoustic modification and how you may use them to address patient complaints. The two parameters we will concentrate on are:

- Venting
- Tubing

Venting

Selection and modification of venting is very important. An incorrect choice in venting can result in *excessive occlusion* if the vent is too small and *excessive feedback* if the vent is too large. In venting, therefore, a fine line must be walked. The vent must be large enough to allow an open sensation for the patient, but not so large as to create objectionable feedback.

The hearing aid vent is an open tube through the hearing aid shell or earmold. Its primary purpose is to bleed off or vent low frequencies. When an ear canal is occluded, there is an increase in low frequency sounds via bone conduction. This creates the so-called “occlusion effect”. Venting can relieve this sensation.

Venting also reduces the impact of low frequency amplification. The larger the vent, the less ability the consultant will have to increase amplification in the low frequencies. With a very open vent, low frequencies are essentially lost into the air.

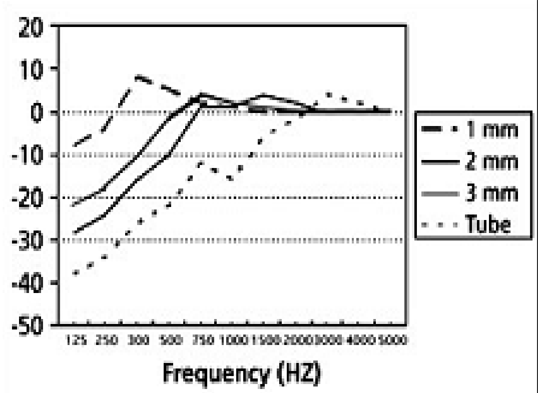
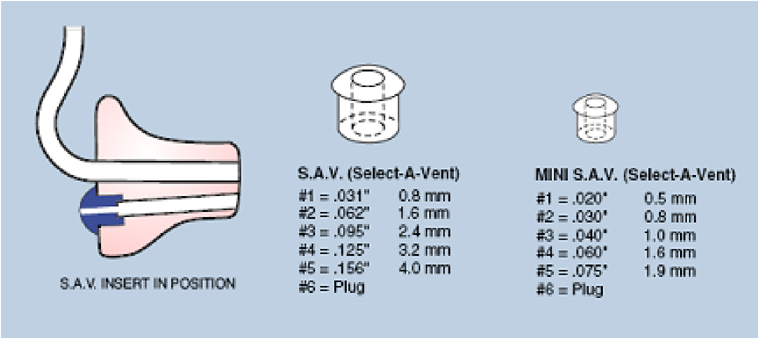


FIGURE 3. Vent parameters for different parallel vent diameters and for an open “tube” fitting with no canal occlusion. Based on Lybarger.4

In the graph at right you can see that the larger the vent, the less energy there is in the low frequencies. The “tube fitting” showing the greatest attenuation of low frequency sounds is the equivalent of “open tip/dome” fittings used in receiver-in-the-aid (RITA) and receiver-in-the-canal (RIC) mini-BTEs.

Select-a-Vent (SAV)

The most convenient way to adjust the size of the vent is by using a Select-a-vent (SAV). By changing the SAV insert, you can quickly and easily change the size of the vent.



The On-Off Test

When a patient complains of occlusion-like problems ("I don't like the sound of my own voice." Or "It sounds like my head is in a barrel", for example) the problem may be true occlusion (i.e. that the physical presence of the earmold or shell is causing the problem) or the problem may be what we will call ampclusion (meaning it is something about the manner in which the electronics of the hearing aid process sound that is causing the problem). Each of these problems has a different set of possible solutions. In order to determine which set of solutions to apply, do an on-off test.

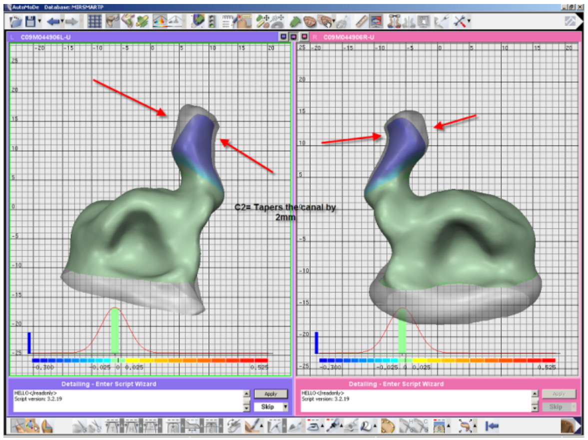
- With the hearing aid on and in the ear, have the patient produce a sustained /ee/ sound. Verify that the offensive sound is present.
- Keep the hearing aid in the ear, but turn it off or open the battery door and have the patient produce the /ee/ again. Is the sound still present?

Yes, the offensive sound is still there.

Then the problem is **oc**clusion.

No, the offensive sound is gone.

Then the problem is **amp**clusion



Notes

Possible solutions for OCclusion

Solutions for occlusion must be acoustic modifications because occlusion is caused by the fit of the shell.

- Open vent diameter
- Reduce vent length
- Taper canal portion (at left)
- Shorten canal portion
- Lengthen canal portion into osseous part of ear canal
- Semi-IROS or "step vent" the end of canal portion

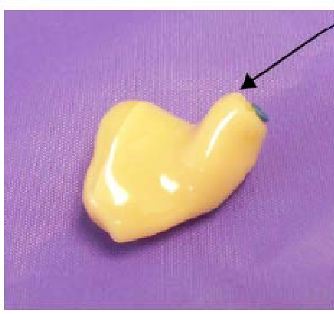
What is a "Semi-IROS" or "step vent" modification?

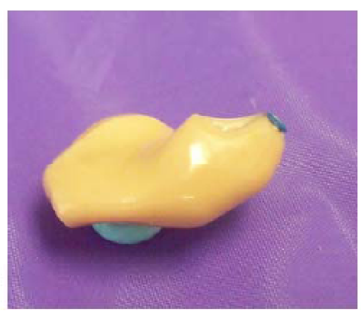
In a semi-IROS modification, the vent at the canal end of the hearing aid is "stepped back" to allow a more open feel. This is a modification you can do on either BTE earmolds or on custom hearing instruments. Please practice on discarded or non-functioning hearing aids before attempting on a patient's hearing aid.

Note: When modifying an earmold (grinding or buffing), ALWAYS have a battery (good or bad) in the hearing aid. This prevents static charge build up in the circuit.

Material in this area removed by Dremel tool.

Step vent/semi-IROS before modification





Step vent/semi-IROS after modification

Notes

Options for correcting AMPclusion

Solutions for ampclusion must be programming modifications because the problem is in the electronics of the hearing aid.

- Reduce gain in lows/increase gain in lows
- Reduce overall gain
- Increase compression in highs
- Reduce gain in highs
- Reduce MPO

Be sure to make small, incremental changes each direction and verify the effect.

Tubing

For BTE hearing aids, the internal diameter of the tubing and the size of the sound bore of the earmold can have a significant effect on the frequency response of a hearing aid. The narrower the tubing or diameter of the sound bore, the more reduction will occur in the high frequency response of the hearing aid. See below for a graphic representation of the difference between response for #13 tubing and a slim tube.

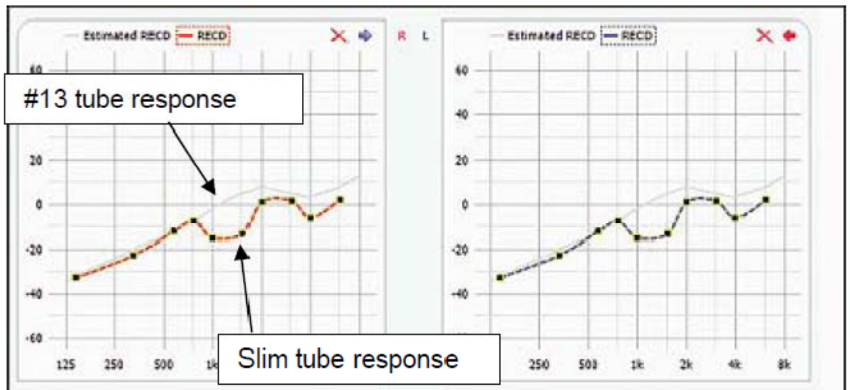


FIGURE 7. Comparison between RECD for standard earhook and #13 tubing (solid curve) and 0.8-mm diameter "slim tube" (circles) for a BTE hearing instrument.

If a patient complains of "clarity" or "speech understanding" problems in hearing aids using slim tubes, switching to standard tubing can improve high frequency response.

Tubing thickness can have an impact on the susceptibility of a BTE hearing aid to feedback. The more powerful the hearing aid, the thicker the tubing should be. By allowing less sound to escape through the tubing wall, double wall tubing provides 2 dB more attenuation than is provided by standard tubing.

Notes

Section 4: Digital Features

Notes

The plethora of digital features that can be manipulated to address patient concerns presents the biggest advantage of digital hearing instruments. In this section we will discuss some of the more common digital features:

- Directional microphones
- Digital Feedback Suppression
- Digital Noise Reduction
- Intelligent Peak Smoothing

Directional Microphone

Directional microphones are the only proven way to improve signal-to-noise ratio with hearing aids. They work by focusing amplification on sounds to the front, and reducing amplification to the sides and/or the rear. They may accomplish this automatically (i.e. the environmental classifier algorithm in the hearing aid determines when the hearing aid should be in directional mode and automatically switches) or under manual control. Directionality can also be adaptive, meaning the directionality changes depending on the location of the greatest noise or the primary speech signal. For hearing aids having the Voice Target option, this may mean the hearing aid focuses on signals to the rear. If a patient is having problems understanding speech in noise, directional microphones may be the solution.

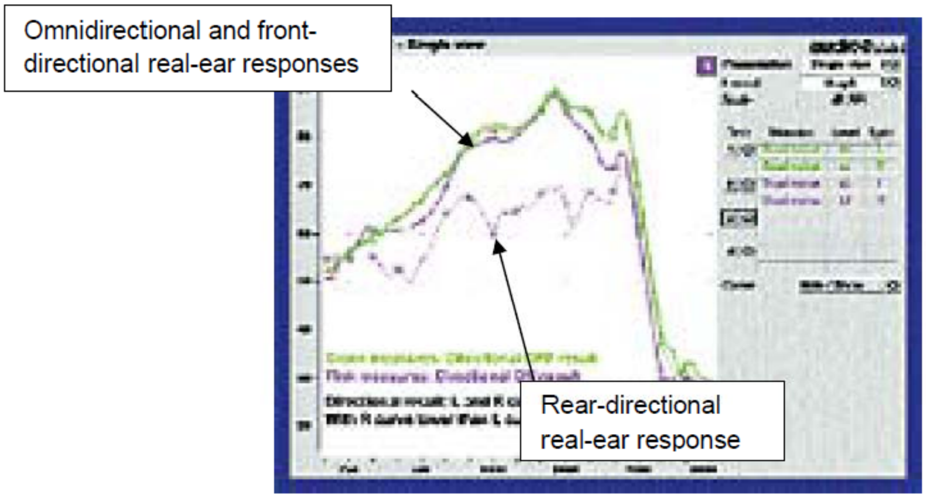
But what do you do if the patient has directional microphones and they are not improving speech understanding in noise?

Here are some things to check:

Are the Directional Microphones Functioning Properly?

Obviously, if the directional microphones are not working, they will be ineffective. You can listen to the hearing aid on a hearing aid stethoscope. With the hearing aid in directional mode (not adaptive), listen as you speak into the hearing aid from the front and then from the rear. The response of the hearing aid should be audibly less intense from the rear. You can also run real ear from the front then the rear using the same signal. Similarly, the response from the rear should be reduced. On the next page, you can see the difference that should be observed between omnidirectional, directional front, and directional rear responses when measured via real ear.

Notes



Directional microphone clogged

A simple reason that directional microphones may be malfunctioning is that one or the other of the directional microphones is clogged with debris. Visual inspection will usually reveal this. It may be a good idea to brush off the microphones each time you inspect a hearing aid.

Directional Microphones Reversed or Out of Phase

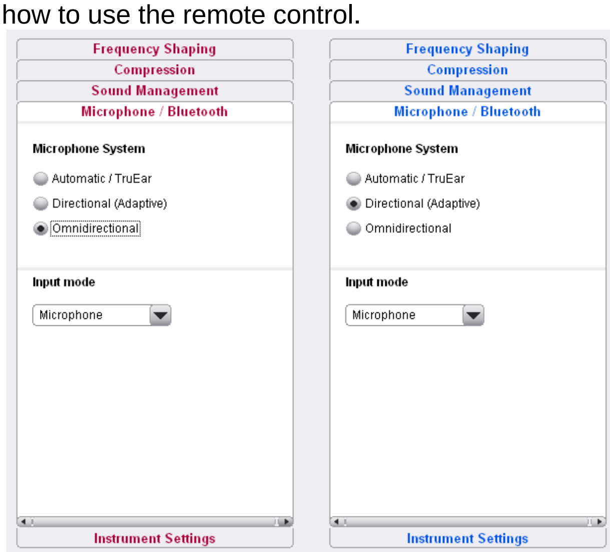
Again, either listen on a listening tube front and back, or run real ear front and back to check. If the d mics are reversed, the signal will be louder or the response greater *from the rear*, rather than from the front.

Is the patient using the directional microphone control properly?

Perhaps the patient is confused about when to use the manual directional control or they are forgetting to do so. It may also be that they cannot locate the control on the hearing aid or are uncertain about how to use the remote control.

Is the Hearing Aid Programmed Correctly?

Is the hearing aid programmed to automatically go into directional mode? In the example to the right, the left aid is programmed to automatically switch to directional in noisy environments, but the right is set for omnidirectional.



Notes

Does the Patient have a High QuickSIN score?

If QuickSIN was administered, the score will tell you how much benefit the patient is likely to get from directional microphones. If the SNR loss is greater than >15, the benefit from directional microphone is likely to be limited.

Venting and Directional Microphones

It is important to consider the size of the vent with directional microphones. The larger the vent, the less dramatic will be the directional effect. Keep in mind that sounds that get to the ear through the vent will not be processed by the hearing aid and are omni-directional in nature.

Directional Equalization and Directional Microphones

Since noise is primarily low frequency energy, and directional microphones are used primarily in the presence of noise, many manufacturers build in a reduction in low frequencies in the directional microphone program. If the patient has a significant low frequency hearing loss, however, this reduction in gain and output for the low frequencies may interfere with speech understanding (audibility). It may then be necessary to manually add gain back to the low frequencies when the hearing aid is in directional mode.

Digital Feedback Suppression (DFS)

It is the introduction of digital feedback suppression (DFS) via phase cancellation that has allowed many of the most widely-adopted hearing aid innovations. Open canal instruments were impractical before the advent of DFS, and are only possible today because of it. DFS allows as much as 25 dB or more of additional stable gain before feedback commences. This has allowed hearing aid manufacturers to introduce hearing aids using much larger vents. Most hearing aid software has a feedback manager of some sort – sometimes called the critical gain measurement. This is a function that presents a swept pure tone to the hearing aid and measures at what frequencies feedback is likely to occur. Where feedback is identified, the feedback manager will usually reduce gain in the frequency areas that feedback is found. This may be a substantial reduction and could interfere with clarity of speech. Therefore, unless the feedback manager must be run in order to activate the DFS, *run the feedback manager or critical gain measurement only if you are encountering problems with feedback.*

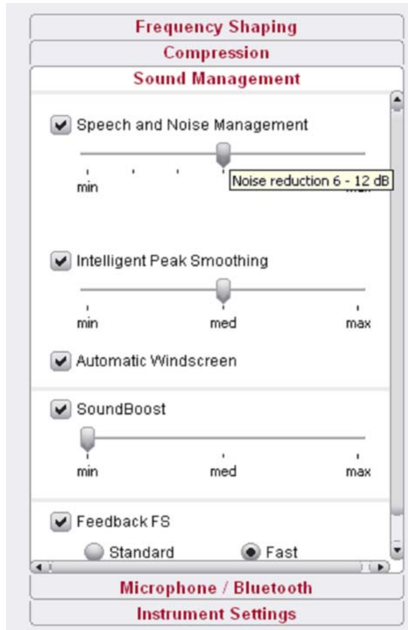
Check Your Understanding

WDRC circuitry is output compression.

True or false

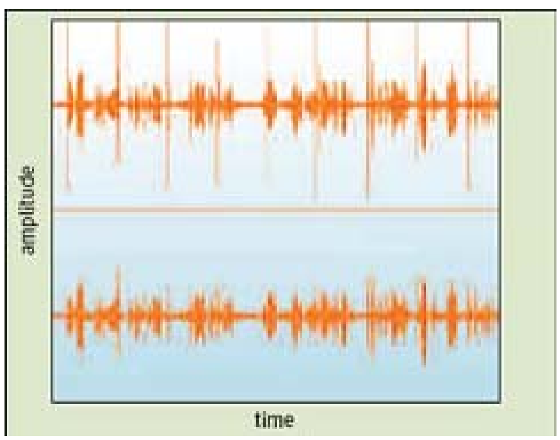
Digital Noise Reduction (DNR)

Digital noise reduction runs in the background of virtually all digital hearing aids. The processor has an algorithm that analyzes incoming signals in each of its channels and determines whether the sound entering the hearing aid is likely to be noise (i.e. steady-state such as forced air or engine noise) or likely to be speech (that is, variable in frequency and intensity). In channels where noise is identified, the hearing aid will automatically reduce gain, and where speech is identified, gain will not be reduced. With the speech and noise management control, you can alter the degree to which noise is reduced. See figure at right.



Speech and Noise Management is a DNR control. In this example it is set to a mid level reduction – 6-12 dB. The maximum is 24 dB, which may sound unnatural.

Intelligent Peak Smoothing (IPS)



Intelligent Peak Smoothing is a very fast-acting compression that affects loud transient sounds without interfering with the speech occurring at the same time. The kinds of sound reduced by IPS are such things as doors slamming, clanking dishes, or crinkling paper. In a traditional compression, if these sounds were acted upon by the compression algorithm of the hearing aid, all sounds occurring in that frequency range would be reduced at the same time, and a relatively long release time would keep the compression engaged, reducing the amplification of any simultaneous speech. At right is an image of speech with occasional transient peaks (top) and the same speech during application of IPS (bottom).

Notes

Section 5: General Approach to a Malfunctioning Hearing Aid

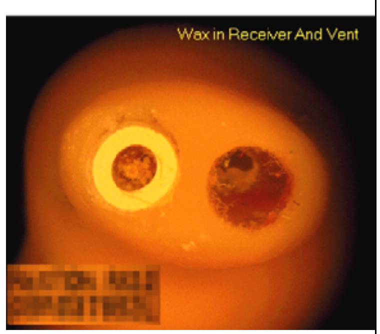
In this section we will outline the steps to take in evaluating a malfunctioning hearing aid:

- Determine the problem
- Visually inspect the hearing aid
- Check the battery
- Listening check
- Check switches/potentiometers - programming parameters
- Check earmold, tubing, earhook
- Check microphone and receiver ports
- Problems likely to require the hearing aid to be sent out for repair.
- Can the patient relate the sound to something that is similar to the problematic sound

Determine the Problem

- Ask the patient to describe the problem as accurately as possible.
- Under what conditions does the problem occur?
- How often does the problem occur?
- If applicable, for what frequency/intensity inputs does the problem occur?
- Can the patient duplicate the problem?

Visually Inspect the Hearing Aid



- Check for cracked or broken case.
- See if any potentiometers, volume control, switches or toggles are broken.
- Look for anything out of the ordinary.
- Is a vent blocked with wax, skin, or debris?

Notes

Check the Battery

- Is it the correct size and type?
- Was it inserted correctly?
- Is it dead: check with a voltage vmeter?
- Are the battery contacts clean or damaged?
- Is the battery leaking?
- Does the battery make proper contact with the battery contacts?
- Does the battery door close properly?

Tab still on battery

Battery inside aid

Battery in door backwards

Notes

Listening Check

As described at the beginning of this module, listen to the hearing aid on a stethoscope or listening tube.

Listen for distortion, static, weakness, humming, buzzing, clicks, or any other unusual sounds. Use a new battery and turn VC and tone pots, if any, while listening.

Check Potentiometers and Switches

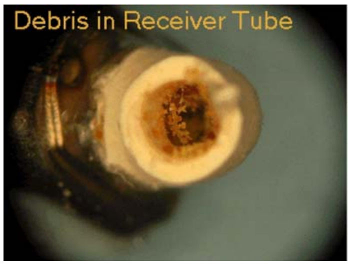
- Is the hearing aid turned on?
- Is the MTO switch in microphone position? (MTO is a control found on most BTE hearing aids that determines whether the hearing aid is in **M**icrophone, **T**-coil, or **O**ff mode)
- Is the volume control (VC) turned up?
- Turn the VC and trim pots, if any, to determine if there are any dead spots.

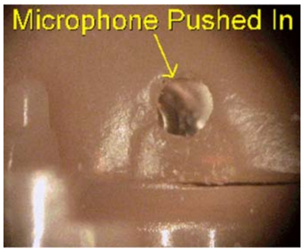
Check Earmold, Tubing and Earhook

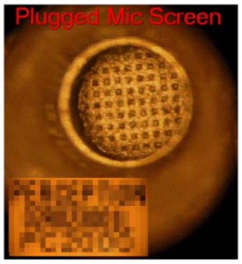
- Is the tubing damaged, bent, blocked, cracked, split or brittle?
- Is the earmold blocked with earwax or other material?
- Is the earhook connected to the tubing correctly?
- Is the earhook cracked?
- Is there moisture in the tube?

Check the Microphone and Receiver Ports

The most common reason for a dead hearing aid is earwax occluding the receiver. This may be a simple fix. A bit less common is the microphone being blocked. This usually occurs when the microphone has a wind screen. If the aid is dead, listen to the hearing aid while rubbing on the microphone. If rubbing is heard, the microphone is blocked – usually by hair spray or other debris clogging the wind screen. Correct this by brushing the debris out of the wind screen.







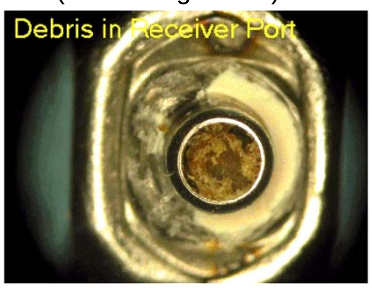
Notes

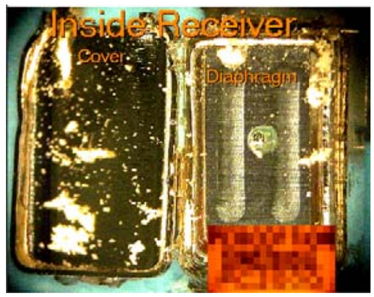
Problems Likely to Require the Hearing Aid to be Sent Out for Factory Repair

Look with an otoscope or video otoscope into the receiver tube, looking for cerumen blocking the receiver. The presence of cerumen, if deep in the receiver, may require the receiver to be replaced.

If the hearing aid is dead and a click is heard when engaging the battery (closing the battery door) the microphone or receiver is dead and the hearing aid requires factory repair.

If the hearing aid is dead and when rubbing on the microphone only line noise is heard (no rubbing sound) then the microphone is dead and factory repair is required.





Notes

Section 6: Addressing Specific Complaints

Notes

In this final section, we will discuss strategies for addressing a number of specific complaints. As noted below, some have been discussed in previous sections.

- Hearing aid is dead
- Patient does not like the sound of his/her own voice
- Patient is having difficulty hearing in noise
- Hearing aid feedback
- Hearing aid is weak or distorted
- Moisture problems
- Hearing aid lacks clarity

Feedback

Hearing aid feedback is perhaps the most common problem encountered by the hearing professional. Feedback occurs when the amplified sound of the hearing aid leaks out of the ear and gets back into the microphone, creating a feedback loop. As discussed above, DFS has been an effective tool in reducing the incidence of feedback. However, DFS does not eliminate feedback altogether. Because DFS has allowed the fitting of more severe precipitous high frequency hearing losses than in the past, we continue to push the envelope and continue to have to deal with feedback.

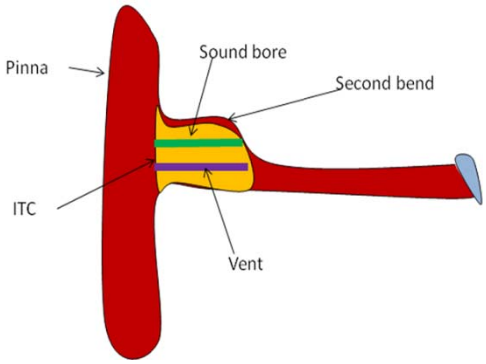
Feedback – Diagnosing the Problem

In general, there are four primary causes of feedback:

1. Cerumen or other debris in the ear canal
2. Fit-related issues
3. High frequency gain
4. Damage to the shell or earmold

To determine the precise frequency of feedback, real ear measures can be useful. An improperly seated hearing aid (left) is likely to produce feedback.

Cerumen or Other Debris in the Ear Canal



The first step in diagnosing the cause of feedback is to look in the ear canal. An accumulation of earwax or other debris can cause the amplified sound from the hearing aid to “bounce off” the debris and cause feedback. (A similar situation can be present if there is a sharp bend in the ear canal and the end of the canal portion of the hearing aid is in close proximity to the canal wall.)

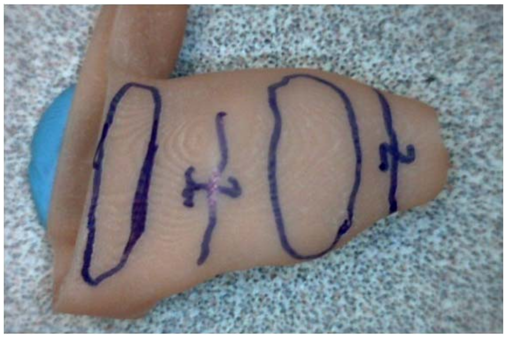
In this schematic, a narrowing of the ear canal at the second bend places the end of the sound bore in close proximity to the canal wall. This may either create a feedback situation, or block off the sound bore.

What to do about Cerumen?

If removal of earwax is within your professional scope of practice, and you have the training and equipment to accomplish it safely, you may remove the cerumen or debris yourself. Check with your own state laws regarding scope of practice.

If, as is likely, this is not an option, you must refer the patient to a physician or healthcare (urgent care) facility for removal. Consult with your supervisor regarding where patients should be sent for cerumen removal.

Fit-Related Issues



Amplified sound leaking out of the ear canal is the next most common cause for feedback. This may occur if the earmold or shell is too loose, if the vent is too large, or if the hearing aid is not seated in the ear correctly.

In the picture to the right, the circled areas frequently require build-up to prevent feedback. First and second bends are indicated by lines.

Notes

Notes

What to do about Fit-Related Issues

If the hearing aid is loose, the most common remedy is to have the shell or the earmold remade. You may want to take the impression for the remake with mouth open, which sometimes creates a better seal. You may be able to build up the canal portion of the earmold or shell with monomer or other material, if you have been trained how to do so.

If the vent is too large, the easiest solution is to narrow it with a select-a-vent.

If the hearing aid is not seated correctly, re-instructing the patient regarding proper insertion is usually required.

The top two impressions were made with mouth closed, the bottom 2 with mouth open. Notice the difference in circumference of the canal portions.

High-Frequency Gain

Since it is usually in the high frequencies that our patients have most of their hearing loss, and that this loss is what creates most of the communicative difficulty they encounter, *reducing high-frequency gain to solve feedback issues should be a last resort. That said, there are certainly instances where this is necessary and appropriate.* (Note – Raising CK in the high frequencies can also reduce feedback by reducing gain for soft sounds.)

What to do about High-Frequency Gain

As has been previously described, you can run the feedback manager. Sometimes the high frequency reduction recommended by this procedure can be more aggressive than may be necessary. You can run the manager, observe the recommended high-frequency cut, not apply the change, and manually make a more conservative reduction. This will frequently resolve the issue.

Cochlear Dead Spots

Sometimes it is justified to roll off high frequencies altogether because the hearing loss in the high frequencies is so severe that very few, if any, sensory cells are present to respond to the high frequencies. This is called a “cochlear dead spot”. One should not assume that a dead spot exists however.

A cochlear dead spot may be identified by a perception by the patient of pure tones that lack "tonal quality". This means that a pure tone, rather than sounding like a whistle or beep, sounds like a scratch, hiss, crackle, or thud or has some other non-tonal quality. During pure-tone testing, presence of tonal quality can be screened. When testing hearing thresholds above 2000 Hz with thresholds greater than 65 dB HL, enquire from the patient whether the sound they perceive is a "tone" or does it have some other quality such as those described above. If the response is that the sound is non-tonal, it is justified to roll off high frequencies in that region. Attempting to apply gain where there is not a perception of tonal quality may actually interfere with speech understanding.

Check Your Understanding

When troubleshooting a hearing aid, you should always make the most aggressive change first.

True or False

Notes

Troubleshooting Feedback in a BTE

Notes

Because of the complexity of the BTE hearing aid, with earmold, tubing, earhook and hearing aid itself, diagnosing cause of feedback can be a challenge. Any of the components can be the source of the feedback, therefore a step-by-step approach is the most efficient way to proceed.

- With the hearing aid out of the ear, turn the VC full-on
- With fingertips, firmly close off the sound bore at the end of the canal portion as well as the vent (see illustration at right). Listen for feedback.
- *If there is no feedback*, earmold fit, venting or conditions in the ear canal (cerumen, bends so that canal portion is adjacent to the canal wall or too short canal portion) are causing the problem.
- *If feedback persists*, remove the earmold and tube from the earhook and seal the earhook opening with finger. (see illustration at right)

- *If there is no feedback*, the problem is with the tubing (split or cracked) or with the earmold (cracked). Retube or remake earmold as indicated.
- *If feedback persists*, remove the earhook and place finger over the end of the receiver.

- *If there is no feedback,* the problem is the earhook. It may be split or cracked. Replace the earhook.
- *If feedback persists*, the hearing aid is demonstrating *internal feedback*. This may occur due to a split case or defective internal components and the hearing aid will require factory repair.

Troubleshooting Feedback in a Custom Instrument

Notes

Troubleshooting a custom instrument, because there is less "plumbing" involved, is a simpler affair.

- With VC full-on, seal the end of the canal portion and vent firmly with your finger.
- *If feedback persists*, the problem may be internal feedback and the aid will require factory repair.
- *If there is no feedback*, the problem may be that the aid is too loose, the vent is too large, the canal portion may be adjacent to the ear canal wall, or there may be cerumen in the ear canal or the hearing aid receiver tube. Correct the problem as indicated.

Weak Aid

The following is a summary of many of the possible causes of a weak hearing aid, how it may be diagnosed, and suggested steps to solve the problem.

Possible Cause	Diagnosis	Remedy
Weak battery	Test battery or try a new one	Replace battery
Dirty battery contacts	Visual inspection	Clean with eraser
Corroded battery contacts	Visual inspection	Clean with abrasive paper or send out for repair
Clogged sound bore or receiver	Visual inspection	Clean with wax loop
Clogged wax guard (custom instrument)	Visual inspection and output restored when wax guard is removed	Replace wax guard
Clogged damper/filter	Output restored when earhook or damper is removed	Replace damper or earhook.
Clogged mic inlet port/wind screen	Visual inspection or hear scratch when mic port is rubbed	Clean inlet port or wind screen with a fine pick or brush
Inadvertent deprogramming or reprogramming	Check program settings	Reprogram and send out for factory repair if the fault recurs
Faulty microphone	Aid works on T or DAI setting - internal noise present at high VC setting	Send out for factory repair

Moisture Problems

As with any electronic device, moisture can cause problems. Moisture from high humidity, perspiration, condensation or immersion in water can cause intermittencies, reduced gain/output, or corrosion. In BTE hearing aids, moisture can sometimes be observed in the tubing. Among the possible solutions are the use of silica gel crystals, other dessicant or a Dri-Aid Kit, use of a Dry-and-Store (seen at right) or other hearing aid conditioning system, replace tubing, replace filters, or have the patient use sweat bands or sweat socks for BTE

Notes

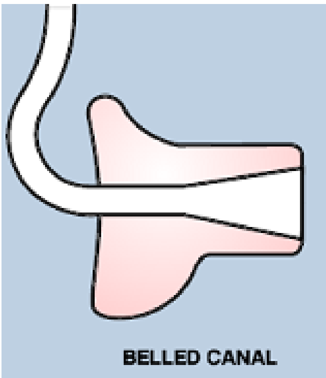
Clarity – Speech Understanding

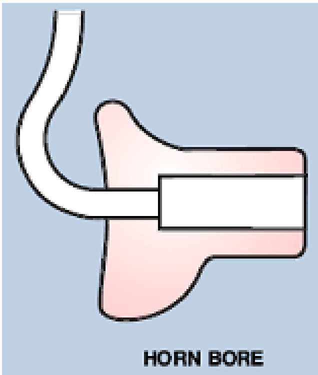
When the issue is clarity or speech understanding, the problem is usually insufficient amplification in the high frequencies. High frequencies equal consonants, particularly unvoiced consonants such as s, f, sh, t, etc. In order to allow high frequencies to “pop” there are three hearing aid processing adjustments that can be tried:

- Increase high frequency gain
- Lower CK in the high frequencies (remember that reducing CK may increase gain for soft sounds)
- Reduce Compression Ratio. If the compression ratio made more linear (changing from 2.5:1 to 2:1, for example) high frequencies may be “crisper” and clearer.

Clarity - Acoustic Solution

There is also an acoustic approach to clarity that may be used with standard BTE hearing aids using custom earmolds. “Belling” or creating a “horn” the end of the canal portion will provide additional high frequency gain and extend high frequency response. See below for images of bells/horns. If there is sufficient space in the canal portion, a bell can be fashioned by use of a Dremel tool or Red Wing drill.





Real Ear and Electroacoustic Evaluation

As always, objectively assessing hearing aid function via real ear testing and/or electroacoustic evaluation is a valuable tool in the determination of the source of hearing aid problems. Real ear can be your friend!!

Notes

Check Your Understanding

1. Multi-Channel is the same as Multi-Memory
   a. True
   b. False

2. ________________ allows the hearing instrument to respond differently for low frequency sounds than for high frequency sounds.
   a. Multi-Channel
   b. Directional Mics
   c. Multi-Memory
   d. Expansion

3. When a wearer chooses from different frequency responses and signal processing options by pressing a button, they are utilizing the ____________ feature.
   a. Multi-Channel
   b. Directional Mics
   c. Multi-Memory
   d. Expansion

4. The only hearing aid technology that is proven to improve speech understanding in noise (SNR) is:
   a. Multi-Channel
   b. Directional Mics
   c. Multi-Memory
   d. Expansion

5. In a Bi-Directional Polar Plot the null/s are:
    a. To the back of the head
    b. To the front of the head
    c. To the shoulders
    d. There is no null

6. QuickSIN scores (above 15) and large venting can adversely affect the benefit of:
    a. Frequency Response gain adjustments
    b. MPO
    c. Directional Mics
    d. Multi-Memories

7. A cause for Mechanical Feedback is:
    a. If the receiver tube is pushed in
    b. If the mic is blocked
    c. If the battery contact is broken
    d. If the vent is too large

8. When the polar plot changes automatically to the location of the primary source of noise, this is the:
    a. Manual mode
    b. Adaptive mode
    c. Bi-polar mode
    d. Fixed mode

9. When an aid has an algorithm that recognizes “noise” as sounds that change little in frequency and intensity and speech as a signal that does change frequently in frequency and intensity it is an example of an aid with ________ _________ ________.

   a. Adaptive Directional Microphones

   b. Automatic Feedback cancellation

   c. Noise reduction capability

   d. Multi Memory capability

10. A good rule of thumb for troubleshooting is:’

    a. “always do the least invasive procedure first.”

    b. “always send the aid in for repair”

    c. “always check the aid with the battery the patient had in the aid.”

    d. “always re-tube the hearing instrument”

11. Wide Dynamic Range Compression (WDRC) ensures that:

    a. Soft sounds are audible, moderate intensity sounds are comfortable, and loud sounds are not uncomfortably loud.

    b. Moderate sounds are audible, soft intensity sounds are comfortable, and loud sounds are not uncomfortably loud.

    c. Soft sounds are comfortable, moderate intensity sounds (speech) are audible, and loud sounds are not uncomfortably loud.

    d. Soft sounds are uncomfortably loud, moderate intensity sounds (speech) are audible, and loud sounds are comfortable

12. When the processor is able to change the hearing aid program to the response that is most appropriate for a specific acoustical environment such as music, that is a hearing aid with:

    a. Multi Channels

    b. Multi-Memory

    c. Expansion

    d. Environmental Classifier

Unit 12: Earmold Impressions – Concepts and Techniques

Objectives

Notes

Overview

The ability to take an ear impression confidently and proficiently is essential to your success for a variety of reasons:

- Satisfied patients are comfortable with the fit of their instruments
- Reduces the number of remakes due to fitting issues
- Reduces the number of returns due to fitting issues

Goals

Upon completion of this module, you will be able to:

- Explain the impression taking procedures to your patient
- Complete appropriate infection control procedures during the impression taking process
- Use proper bridging and bracing techniques
- Properly conduct an examination of the ear with your otoscope
- Place the cotton/foam otoblock properly
- Mix and inject the impression material to get the best results
- Remove the impression in a manner that is comfortable to your patient
- Explain what to look for with your otoscope before and after the impression is taken.
- Appraise the quality of the impression being made

Follow Up

- Dillon, Harvey. (2001) *Hearing Aids*. Boston, MA: Allyn and Bacon

Section 1 – Concepts and Techniques

Once you have successfully completed the "sales process" and gained agreement with your customer, hearing instruments or ear molds will need to be ordered. If the selected hearing aids are custom in-the-ear or behind-the-ear models requiring an earmold, impressions of the ears must be taken.

Taking a high quality ear mold impression and using proper technique is critical for two reasons.

1. The act of taking an ear mold impression *(abbreviated **EMI**)* is the most invasive procedure we perform. If you do not take the time to learn the correct technique you can cause physical harm to the customer.

2. A hearing aid is not likely to fit well or work properly if it is made using a poor quality EMI.

As you will see, taking an EMI is a fairly straightforward procedure, as long as you follow a step-by-step procedure. Just like a hearing test, taking an EMI should be done the same way every time. In short, you must always use the techniques you learn here and in the Basic Training Lab. Follow the steps listed below.

Notes

Procedures

You will learn the steps to follow to ensure a comfortable and exact fit.

1. Required materials
2. Explanation of the impression process to the patient
3. Infection control procedures
4. Bridging and Bracing technique
5. Identify texture of the ear
6. Otoscopic examination before the impression is made
7. Re-tying and placing the otoblock
8. Mixing the impression material
9. Injecting the material into the ear
10. Removing the finished impression
11. Rechecking the ear canal with the otoscope
12. Appraising the quality of the impression

Notes

Required Materials

- White Towel
- Hand sanitizer
- Alcohol wipes
- Otoscope with specula
- Impression gun or syringe
- Impression material
- Ear light
- Otoblocks

It is important when performing impression-making procedures, especially in the beginning, that you follow a protocol so you develop good habits. You may need to take impressions in front of a proctor at your license test. Developing a routine in the beginning is one way to alleviate the nervousness that is often felt while taking your licensure exam.

Explanation of Impression Process

Use a diagram or replica of the ear to help the patient to visualize what you are about to do. Tell your patient that you will be placing the otoblock approximately half way down their ear canal and that it will be nowhere near the eardrum. Explain that it may feel a little strange because that area is never touched, but it will not cause undo discomfort. Let them know that you will be injecting impression material into their ear canal and bowl of the ear. Let them know that it will feel cool and the ear will feel full and that the material will set up quickly, usually in about five minutes. Finally, instruct the patient on how they should hold their jaw (this will depend on whether the impression is being obtained with an open or closed mouth).

Infection Control Procedures

Set out all materials on a clean white towel. Use anti-bacterial hand cleaner to wash your hands. With an alcohol or anti-bacterial wipe, wipe the otoscope tip, ear light and the end of the syringe/impression gun. Repeat before taking an impression of the opposite ear. Every time you pick up a piece of equipment or use a piece of equipment, you must follow appropriate infection-control procedures.

Notes

Bridging & Bracing

Always use the bridging and bracing technique when using the otoscope, earlight and syringe. For otoscope and ear light, use your little finger and ring finger of the hand holding the implement braced against the patient's skull to prevent an accident should the patient move or jerk suddenly using the bridging and bracing technique.

When using the syringe or gun, hold your opposite hand with its back away from you. While placing the backs of the middle, ring and pinky fingers on the customer's head, hold the tip of the syringe or gun with the thumb and forefinger. This should be practiced with a partner or using the rubber ear until you are comfortable with the technique. In your Basic Training Lab the instructor will demonstrate proper technique which will help you become proficient with this procedure.

Identify Texture of the Ears

The first part of examining the patient is to check the texture of the ear. Generally, when you touch the ear to examine the canal area, you will be able to classify the ear texture as soft, medium or hard. This can be important when selecting the viscosity of the impression material. *Viscosity* refers to the texture of the impression material as it goes into the ear canal. Low viscosity mans the material is more "runny" and high viscosity means the material is more dense or "stiff". A soft-textured ear requires a lower viscosity (less "stiff") so that ear structures are not stretched or distended during the procedure. Impressions for firmer ears can be made with a higher viscosity material.

Notes

Otoscopic Examination

Notes

Before you start your examination, ensure you have followed appropriate infection control procedures. Always tell your patient what you are doing; don't sneak up on them with any equipment. To properly examine the ear canal, you will need an otoscope with a good light source and magnification. Use the Otoscopic techniques you learned in a previous unit. As you examine the ear and canal, you should note the following:

- Length and course of the canal
- Overall volume of the concha and the ear canal
- Any foreign objects in the ear canal
- Excessive cerumen/wax in the ear canal
- Identify anything that is out of the ordinary
- Check for a prolapsed (collapsing) ear canal
- Check for enlarged ear canal, usually due to surgery, particularly in the osseous portion of the ear canal (just past the second bend)
- Identify any unusual bumps, moles, pimples, ridges, etc. that should be circled on the impression
- Observe tempero-mandibular (jaw) joint motion

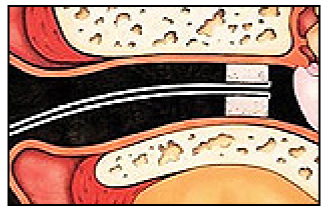
Re-tying a Knot and Placing the Otoblock

Placing the Otoblock is the most critical step in the EMI process. An EMI should never be made without an Otoblock, and the Otoblock must be placed in position so that impression material does not "blow by" the block and touch the TM, or in the case of a surgical ear go down into the open ear cavity. Both of these situations could result in pain or discomfort to the patient and most likely an immediate physician or ER referral.

Before placing the otoblock, tie another knot around the otoblock. To ease insertion, you may lightly wet the otoblock with a water-based product like KY Gel or oto-ease. This will ease insertion of the otoblock and facilitate removal of the impression. Tell your patient, "I am now going to place the otoblock into your ear canal just as I showed you on the diagram. Again, the reason for this is to protect your ear drum from the impression material." Bridge & Brace when inserting the otoblock.

Notes

- Otoblocks are generally cotton or foam and come in different sizes
- Use an otoblock that is the same size or slightly larger than the diameter of the second bend
- On some surgically repaired ears, several blocks may be necessary
- The otoblock must be placed at 2 to 4 mm beyond the second bend
- Before injecting the impression material, recheck with the otoscope to ensure that there are no gaps around the otoblock. Remember to bridge & brace and sanitize.



With your fingers or round-tip tweezers, place the otoblock at the aperture of the ear canal. Then using the ear light, gently push the otoblock down the canal, moving around the top, bottom and sides. If you believe the otoblock is beyond the second bend, use the otoscope and assure the canal is completely blocked with no gaps. If this is the case, you are ready to mix the impression material.

Mixing the Impression Material

There are two common methods for inserting the impression material, Syringe method and the injection gun method.

(Check the licensure laws in your State to find out which method is allowed at your Practicum.)

- **Injection gun method**. The silicon and hardener mix in the tip as it is being injected
- **Syringe method.** This is used for both powder/liquid and silicon ear impression materials. Read and follow manufacturer's instructions for the designated proportions.

Injecting the Impression Material

Important: Your full concentration needs to be on the impression-taking process. You will want to minimize any distractions during this time.

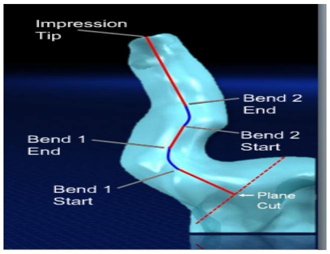
- Place mixed material into the syringe, insert the plunger and force the material to the tip of the syringe to eliminate air pockets

- Pull the pinna up and back to straighten the canal
- Be sure to bridge and brace the syringe or gun
- Place the tip into the aperture of the canal

- Using moderate pressure, squeeze smoothly, forcing the material up to the block
- Fill the canal and release the pinna
- Keep the tip in the material until it flows back

- Work upwards and fill the helix, then the concha bowl, cover the anti-helix rim, crus area, inter-tragal notch, and finally the tragus, pointing the tip of the implement into the recesses

- If there is a history of feedback or a hearing aid that works out of the ear, you may want to take an open-mouth impression. This is done the same way but the patient's mouth is held open with a dental bite block or by separating their front teeth slightly by biting softly on their thumb. Be sure if you use both impression procedures to mark each as open or closed before sending to the manufacturer.



- Allow the material to set 5 to 10 minutes

Notes

Removing the Impression

- Touch the impression with the tip of the ear light or finger nail and lightly push. If no mark is made, the impression is ready.
- Pull the ear up and back to loosen the seal
- Pull the helix out and, holding firmly, rock the impression front to back
- Ask the patient how it feels and if he says it is fine, carefully and slowly pull the impression out rolling the top of the impression forward toward the nose while gently lifting up on the impression. For difficult removal, have the patient open their mouth and tilt their head back.

Inspect the Ear Canal

As soon as you remove the impression, set it aside and inspect the ear canal **IMMEDIATELY** upon removal. DO NOT LOOK AT THE IMPRESSION UNTIL AFTER YOU HAVE INSPECTED THE EAR CANAL

- No impression material should be left in the ear canal
- Note any trauma that may have occurred
- Commonly seen or reported: redness, hematoma, “blood blister”, throbbing sensation or soreness

Let the patient know if everything is alright, but also tell them if there is a little bleeding, that it is no cause for concern.

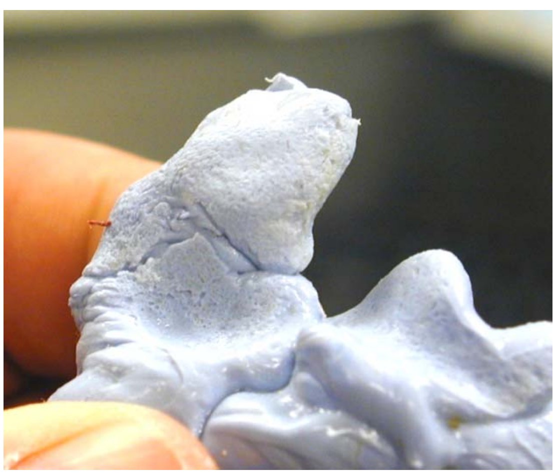
Notes

Appraising the Quality of the Impression

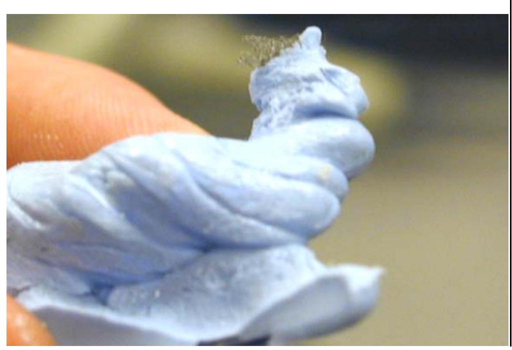
Take a careful look at the EMI to make sure it is correct. If your ear impression is not correct, take another impression.

The images above are examples of good impressions.

In the following picture we don't see both bends and the rough edges and cracking can be due air pockets in the syringe.



In the example to the right, the rolling effect was caused by the tip of the syringe or gun being pulled back too quickly, which also caused the canal length to be too short.



Notes

This picture gives another example of a canal length that is too short.

Notes

In this example the canal is not well formed, and the impression stops at the beginning of the second bend.

Summary and Assignment

Notes

Your ability to take a high quality EMI is critical to your ability to provide a consistently high quality of care to your patient. You must take the time to learn the proper technique. You have to practice.

You should take a set of EMIs of someone you know. After you take the EMIs send them to your Basic Training instructor. Your instructor will evaluate your work and call you with the results.

Key points to Remember:

- **Follow proper infection control procedures**
- **Bridge & Brace**
- **Patient education**

Check Your Understanding

1. EMI is the Acronym for:
   a. Ear Mold Insertion
   b. Ear Mold Impression
   c. Ear Model Impression
   d. Ear Model Insertion

2. A good quality ear impression reduces the chances of:
   a. Patient Discomfort
   b. Feedback
   c. Remakes and Returns
   d. All of the Above

3. The reason for "Bridging and Bracing" is:
   a. To be able to put the gun or syringe tip past the second bend
   b. To prevent an accident, should the patient jerk or move.
   c. To be to keep the gun or syringe immobile.
   d. To be able to pull the pinna up and out.

4. The Otoblock needs to be placed at the:
   a. 2 to 4 mm beyond the second bend
   b. 1 to 2 mm beyond the second bend
   c. 2 to 4 mm beyond the first bend
   d. 1 to 2 mm beyond the second bend

5. Proper Otoblock Insertion:
   a. is to protect the patient's TM
   b. is to make sure cerumen is not dislodged
   c. is to make sure cerumen is dislodged
   d. is to gage the length of the ear impression

6. After making an EMI during inspection it is common to see:
   a. a torn TM
   b. a disarticulated ossicular chain
   c. a pearly grey TM
   d. a hematoma

7. The viscosity of the impression material refers to:
   a. the color
   b. the texture
   c. how quickly it sets up
   d. the taste

8. A common procedure for a patient who has had ongoing feedback issues is to:
   a. request that the factory make a new aid or ear mold
   b. change hearing aid models
   c. make a new impression with the "open mouth" procedure
   d. tell the patient that feedback is expected in all hearing aids

9. To remove the EMI pull up on the pinna to break the seal, then:
    a. rotate the impression toward the back of the head
    b. rock, pull and push the impression inward and outward
    c. ask the patient to remove the impression
    d. rotate the impression forward toward the nose

10. EMI's are required for:
    a. All hearing instrument fittings
    b. All hearing assessments
    c. Custom Hearing Aids
    d. Open fit hearing aids

Unit 13: Infection Control

Objectives

Notes

Overview

Since the beginning of awareness of human immunodeficiency virus (HIV-1), the cause of acquired immune deficiency syndrome (AIDS), a greater concern over potential cross infection of healthcare professionals and patients has resulted in an awareness and institution of infection control protocols throughout the healthcare field. More recently, the rise of antibiotic resistant bacteria such as Methicillin Resistant Staphylococcus Aureus (MRSA) has further raised awareness of the importance of effective infection control measures. Each day, the hearing aid office conducts procedures that potentially risk cross contamination, and create attendant risks to personal health, risks of malpractice suits or citation by the Occupational Safety and Health Administration (OSHA). In this module we will discuss the who, what, why and how " of infection control in the hearing instrument dispensing clinic.

Goals

Upon completion of this module, you will be able to:

- Recognize the parts of the human immune system, varieties of infectious agents, and organizations responsible for policing infection control in the US.
- Describe OSHA infection control requirements and procedures applicable for the hearing aid dispensing clinic.

Follow Up

- Familiarize yourself with the infection control policies of the clinic where you work. Make sure any required materials are stocked and available.

Section 1 – Human Immune System, Infectious Agents, and Responsible Organizations

Notes

Why is Effective Infection Control Important?

Healthcare Associated Infections (HAIs) cause significant morbidity, mortality and cost every year. HAIs are responsible for illness in nearly 2 million individuals annually, accounting for 100,000 deaths. HAIs incur $4.5 to 5.7 billion in patient care costs annually. Transmission of pathogens occurs most often via the contaminated hands of the healthcare worker. An estimated 1/3 of these infections can be prevented through proper hand hygiene. Generally, compliance with proper hand hygiene guidelines is less than 50-60%.

What Causes Infection?

Microbes are microscopic single-cell organisms. Microbes come in millions of forms, but occur in several large categories.

- **Non-Pathogenic Microbes** – These microbes do not cause disease, and in some instances, may actually be beneficial.

- **Pathogenic Microbes** – These microbes are capable of producing infection – sometimes merely causing annoying symptoms, and in other cases causing deadly disease such as tuberculosis or hepatitis-B. Pathogenic microbes are categorized as either:

  - **Inherently pathogenic** – that is they are always capable of causing disease

    – or

  - **Opportunistic** – that is, they are harmless until a compromise in the host's defense mechanism initiates an infection.

There are three main categories of microbe.

1. **Virus** – A virus is a submicroscopic, potentially infectious particle, made up of DNA or RNA. It is the smallest known living organism. Viruses are unable to multiply on their own. In order to procreate, viruses appropriate the reproductive mechanism of a living cell to make copies of themselves.

2. **Bacterium** – A bacterium is much larger than a virus. It is a one-celled microorganism that multiplies by simple division. Unlike the parasitic virus, a bacterium can live independently.

3. **Fungus** – A fungus is a microscopic plant that is capable of growing either on dead organic matter or in living hosts as parasites.

How Does the Body Resist Infection?

Notes

The human body has multiple levels of defense against microscopic outside invaders.

Non-Specific System

The first line of defense is the non-specific system. This comprises the skin and mucous membranes. These structures represent a physical barrier to prevent foreign microbes or materials from entering the body. Unfortunately, the eyes, ears, nose, mouth and other openings in the body represent natural portals of entry to unwanted microorganisms. Scrapes, cuts and abrasions can also allow germs direct access to our bodies.

Specific System

Once the first line of defense has been breached, the specific system takes over. Once inside the body, the next line of defense is ***Peripheral Lymphoid Tissue***. The lymphatic system transports various microbe destroying cells around the body. Some of the elements of the lymphatic system are lymph nodes, lymphatic vessels, the spleen and mucosa-associated lymphoid tissue, including the tonsils and adenoids.

The next level of defense is accomplished by ***Central Lymphoid Tissue***, including bone marrow (which produces infection-fighting cells such as erythrocytes, platelets, and B-cell lymphocytes) and the thymus (which generates T-cell lymphocytes). These cells are distributed throughout the body by the lymphatic system and work to destroy the tiny invaders.

How is Infection Spread?

There are four main modes of transmission.

1. Contact
2. Vehicle
3. Airborne
4. Vector borne

Notes

Contact Transmission

Contact transmission is the most important and frequent means of transmission in the healthcare setting. It can be divided into three subcategories: direct contact, indirect contact, and droplet contact.

- Direct contact – the direct physical transfer of microorganisms between a susceptible host and an infected person. For example, if a sick person shakes hands with an uninfected person, the direct hand-to-hand contact can pass the infectious microbe from one to the other.

- Indirect contact – personal contact of the susceptible host with an intermediate object that has been touched by the infected person. In the hearing aid office, such intermediate objects may include otoscope specula, earlight, ear impression syringe, hearing aids, earmolds, headphones, and all working surfaces, just to name a few.

- Droplet contact – Infectious organisms come in contact with the mucous membranes that line the eyelids, nose or mouth of a susceptible person as a result of coughing, sneezing, or talking to a person who is infected. (This is considered contact transmission rather than airborne because droplets usually travel no more than about three feet.)

Vehicle Route of Transmission

This applies to diseases transmitted by contaminated items, such as water (such as Legionnaire’s Disease), food (such as salmonella) and blood or body substances (such as hepatitis-B).

Airborne Transmission

This occurs by dissemination of either droplet nuclei (residue of evaporated droplets that may remain suspended in the air for long periods of time), or dust particles in the air containing the infectious agents. These organisms can be widely dispersed by air currents before being inhaled by or deposited on a susceptible host.

Vector borne Transmission

This occurs when an animal or insect carries the pathogen and infects a susceptible host. As an example, in the Middle Ages, fleas on rats brought to Europe from Asia on ships initiated the Black Plague that killed millions of people.

Who is Responsible for Policing Infection Control?

Notes

There are a number of government and non-governmental entities that have assumed responsibility for setting guidelines and enforcing compliance with infection control in a variety of healthcare settings.

Occupational Safety and Health Administration (OSHA)

OSHA is a federal agency that regulates the work place to assure that all workers have a safe working environment. Rules concerning workplace infection control were published in the Federal Register in 1991 (29 CFR Part 1910.1030 – Occupational Exposure to Bloodborne Pathogens; Final Rule). Through the power of federal law, OSHA mandates infection control programs in all healthcare settings and oversees and enforces these regulations.

Joint Commission for the Accreditation of Healthcare Organizations (JCAHO)

JCAHO is a non-governmental organization representing the American Dental Association, American College of Physicians, American College of Surgeons, American Hospital Association, American Medical Association and the general public. The JCAHO establishes standards and conducts voluntary accreditation programs for a large variety of healthcare facilities. JCOHA sets guidelines for infection control programs in these facilities and conducts inspections every three years for accreditation, which is highly valued since it usually determines Medicare and Medicaid eligibility.

Commission on Accreditation of Rehabilitation Facilities (CARF)

This organization has a function similar to JCAHO for facilities providing rehabilitation services to persons with disabilities such a spinal cord injuries, chronic pain and emotional disorders.

State and Local Health Agencies

A large variety of state and local agencies enforces and oftentimes builds on OSHA regulations.

Environmental Protection Agency (EPA)

The EPA administers the Federal Insecticide, Fungicide and Rodenticide Act (FIFRA), that registers and regulates the use of all chemical disinfectants and sterilants intended for use on inanimate objects and/or environmental surfaces.

Food and Drug Administration (FDA)

The FDA registers certain products and devices used on or in the human body. Such products and devices include sterilants, disinfectants and sterilizing devices.

Section 2 – OSHA Requirements and Procedures for the Hearing Aid Dispensing Clinic

Notes

OSHA Requirements

All infection control programs are designed to reduce the number of pathogens in the working environment and eliminate cross contamination and cross infection. See below how many persons are at risk of cross infection by the hearing healthcare provider, and how each of those persons could spread infection to their own circle of contacts.

Universal Precautions

Notes

Universal precautions recommendations and guidelines were developed by the Centers for Disease Control and Prevention in the 1980s for minimizing cross infection of healthcare workers by blood borne diseases. The success of an effective infection control depends upon the assumption by the healthcare worker that *every patient* is a potential carrier of an infectious disease. Hence these precautions should be taken ***universally.***

Guideline 1 (personal barriers)

All personnel must wear appropriate personal barriers (i.e. gloves, masks, eye protection, gowns) when performing procedures that may expose personnel to infectious substances.

Gloves are not reusable. It is advisable to double-glove when the risk of blood contact is high. Carefully remove gloves, turning them inside out and avoiding contact of skin with the exterior of the glove. Wash hands after gloves are removed.

Guideline 2 (hand washing)

Hands must be washed before and after every patient contact and after glove removal.

When hands are visibly dirty, they should be washed. A medical grade antibacterial soap should be used. Remove all jewelry and wash all skin surfaces, nails and wrists. Blot hands dry with paper towels and use the towels to turn off the sink and open the washroom door.

When hands are not visibly soiled, an alcohol-based hand sanitizer rub may be used for routine decontamination.

Guideline 3 (“touch” and “splash” surfaces)

All “touch” or “splash” surfaces must be precleaned and disinfected with an EPA registered, hospital-grade disinfectant, or covered with an environmental barrier. Precleaning is always necessary prior to disinfection.

Touch surfaces are those that the patient, clinician, or potentially contaminated instruments, regularly touch. Splash surfaces are those that may be hit by the splash of irrigation, sneezes or other bodily substances.

Cleaning refers to the process of removing gross contaminants from a surface without killing germs. **Disinfection** refers to the use of a chemical solution that will kill particular microbes in the vegetative state. Germicidal disposable cloths may be used for this purpose in most clinical settings. Note that baby wipes are not germicidal and are not intended for this purpose.

Guideline 4 (sterilization of instruments)

Notes

All critical instruments (those that contact blood or other potentially infectious substances) must be sterilized. Precleaning is necessary prior to sterilization.

Sterilization is the most complete form of decontamination, killing 100% of the vegetative microbes and their resistant spores. Cold sterilization using a 2% glutareldehyde (such as Sporox™ or Wavicide™) or 7% or higher hydrogen peroxide solution is preferred in hearing aid offices.

The suggested procedure for sterilization is as follows:

1. Place used instruments in designated covered containers
2. While wearing gloves, clean the surfaces of the instruments with a towelette
3. Place instruments in the tray containing cold sterilants (completely submerged)
4. Cover and soak according to the manufacturer's specifications
5. Remove gloves and wash hands

Guideline 5 (disposal of infectious waste)

Potentially infectious waste must be disposed of appropriately.

Waste that is contaminated with blood, ear drainage or cerumen containing blood or ear drainage can be placed in regular trash receptacles unless the amount of blood or mucous is significant. Gross amounts of hazardous material should be disposed of in impermeable bags labeled with the symbol for bio-hazardous waste and should be picked up by a waste hauler licensed for medical waste disposal.

What Procedures are Specifically Appropriate for the Hearing Aid Clinic?

Otoscopy

If there is evidence of drainage from the ear or open sores on the pinna, don gloves. If not using gloves, wash hands or use an alcohol-based hand sanitizer rub prior to examining ears. Examine the first ear. If drainage or evidence of inflammation or fungal infection is present, use a new speculum for the other ear. If none of these conditions exist, clean the speculum with an alcohol prep pad or disinfectant wipe before using on the other ear. If using gloves, remove them and wash hands. If not using gloves, wash hands or use an alcohol-based hand sanitizer rub.

Notes

Audiometry

If using insert earphones, the foam tips are meant for single-use. Standard TDH-39 headphones may be cleaned before and after use with disposable germicidal cloths. Alternatively, single-use disposable earphone covers may be used. The bone oscillator should also be cleaned with germicidal cloths before and after use.

Making Ear Impressions

This procedure is one of those most likely to cause exposure to blood or other bodily fluids in the hearing dispensing clinic. Prior to taking the ear impression, wash hands or use an alcohol-based hand sanitizer rub. If there is evidence of drainage from the ear or open sores on the pinna, don gloves. Carefully insert the otoblock and inject the impression material. Properly disinfect the syringe tip using the syringe or impressions gun manufactures recommendations. Afterwards remove gloves if being used, and wash hands or use alcohol sanitizer rub. Before removing impression, if using gloves, put on new ones. Place impression in box without touching the surface of the earmold. If using gloves, carefully remove. Wash hands or use alcohol rub and disinfect surfaces using a disinfectant wipe.

Handling Hearing Instruments

A hearing professional may handle dozens of hearing instruments each day. The risk for cross contamination is thus quite high. When taking hearing aids from the patient the use of small single use paper cups is recommended. Also consider the listening tube or listening stethoscope, used many times a day to diagnose hearing aid problems. Unless the hearing aid and the tip of the listening tube/stethoscope is cleaned before and after every use, it potentially contains contaminants from every hearing aid that has been attached to it, and has contaminated every hearing aid with which it has come into contact. Therefore, the hearing aid and the tip of the listening tube should be cleaned with a disinfectant wipe before and after each use.

When handling hearing aids, if there is evidence of drainage from the ear, otitis externa, or open sores on the pinna, don gloves. Clean and disinfect the hearing aid with a disinfectant towelette. Use protective eye wear when buffing or grinding hearing aids. All hearing aid cleaning tools should be cleaned, disinfected or sterilized, as appropriate. All surfaces that have come in contact with the hearing aid or hands while handling hearing aids should be disinfected with a disinfectant towelette. After handling hearing aid, remove gloves, if used, and wash hands or use alcohol-based sanitizer rub.

Check Your Understanding

1. Match the following terms to their definitions

Virus	A one-celled microorganism that multiply by simple division
Bacterium	A submicroscopic, potentially infectious particle made up of DNA or RNA
Fungus	A microscopic plant that is capable of growing either on dead organic matter or in living hosts as parasites

2. Skin and mucous membranes are part of the non-specific system of defense from infection.

   a. True

   b. False

3. Vectorborne transmission refers to diseases transmitted by contaminated items, such as water, food, blood or body substances.

   a. True

   b. False

4. Which of the following is the federal agency that regulates the work place to assure that all workers have a safe working environment?

   a. FDA

   b. OSHA

   c. EPA

   d. JCAHO

5. Match the following terms to their definitions

Cleaning	The use of a chemical solution that will kill particular microbes in the vegetative state
Disinfecting	The most complete form of decontamination, killing 100% of the vegetative microbes and their resistant spores
Sterilization	The process of removing gross contaminants from a surface without killing germs

Answer Key

Unit 2: Psychology of the Hearing Impaired

Answers to Check Your Understanding

1. b
2. b
3. b
4. d
5. a, b, c
6. c
7. c
8. Your answer could have included anything similar to the following: How does that make you feel? Tell me more about that situation. Tell me about the locations.

Unit 3: A (Painless) Introduction to Acoustics (Physics of Sound)

Answers to Check Your Understanding

1. c
2. b
3. e
4. c
5. a
6. b
7. d
8. b
9. HL – Audiometric Zero, SPL - .0002 dynes/cm^2, SL – Patient's Threshold
10. HL – Audiometric Testing – SPL – Manufacturing Specs, SL – Supra-Threshold Testing
11. b
12. e
13. c
14. e
15. a

Bonus Question: No

Unit 4: Anatomy and Physiology of the Ear

Answers to Check Your Understanding

1. a, b, c
2. a, c, d
3. b, c, d
4. a, b
5. a, c
6. b
7. b
8. b
9. d
10. b

Unit 5: Introduction to Hearing Disorders

Answers to Check Your Understanding (within unit)

Page 67: d

Page 70: b

Page 73: c

Page 80: d

Answers to Check Your Understanding (end of unit)

1. I: S

   II: S

   III: C

   IV: C

   V: S

   VI: C

   VII: M

2. a
3. d
4. c, d
5. b
6. c
7. b
8. Otitis Media/refer
9. Presbycusis/sensory
10. A

Unit 6: Basic Audiometry

Answers to Check Your Understanding (within unit)

Page 90: b

Page 97: b

Page 101: d

Page 105: a, d

Page 107: c

Page 111: a

Answers to Post-Text Check Your Understanding (end of unit)

1. d
2. b
3. a
4. b
5. d
6. c
7. b
8. a
9. d
10. d
11. a
12. c

Unit 7: Masking

Answers to Check Your Understanding (within unit)

Page 128: a

Page 133: d

Page 139: d

Page 141: (top) c

Page 141: (bottom) b

Answers to Post-Text Check your Understanding (end of unit)

1. a, c, d
2. b
3. a
4. b
5. c
6. c
7. a, b, c, d
8. b
9. b
10. d

Unit 8: Introduction to Hearing Instruments

Answers to Check Your Understanding (within unit)

Page 152: False

Page 154: +5 dB

Page 157: False

Page 160: d

Page 163: c

Page 164: True

Page 166: a

Page 171: c

Page 177: a

Answers to Check Your Understanding (end of unit)

1. b
2. b
3. c
4. a
5. a, b
6. b
7. c
8. d
9. d
10. b

Unit 9: Basic Audiometric Interpretation

Answers to Section 2 Exercises

Exercise 1 - 1 Symmetrical, flat, mild to moderate, SN

Exercise 1 - 2 Symmetrical, steeply falling, normal to severe, SN

Exercise 1 - 3 Symmetrical, reverse, mod/sev to moderate, SN

Exercise 1 4 Asymmetrical, gradually falling, L-mild to moderate R-moderate to severe, SN bin

Exercise 1 - 5 Symmetrical, flat, moderate, conductive

Exercise 2 -1 BC-RT all, BC-LT @2Khz, All SP tests LT, Bin MCL & WRS

Exercise 2 -2 BC RT & LT, Bin MCL & WRS

Exercise 2 -3 BC RT, UCL & MCL Bin, Bin MCL & WRS

Exercise 2 - 4 BC RT, Bin MCL & WRS

Exercise 2 - 5 BC RT & LT, Bin MCL & WRS

Answers to Check Your Understanding (within unit)

Page 192: Blue X represents the **Left Ear**. Red O represents the **Right Ear**

Page 196: True

Page 201: True

Answers to Check Your Understanding (end of unit)

1. c
2. c
3. d
4. a
5. b, c
6. d

Unit 10: Verification

Answers to Check Your Understanding (within unit)

Page 241: False

Page 242: Soundfield/Probe Tube

Page 244: a, c, d

Page 250: b

Page 255: False

Answers to Check Your Understanding (end of unit)

1. c
2. b
3. d
4. b
5. b
6. a
7. b
8. c
9. c

Unit 11: Hearing Aid Programming and Troubleshooting

Answers to Check Your Understanding (within unit)

Page 265: False

Page 266: True

Page 274: True

Page 281: False

Page 290: False

Answers to Post-Text Check Your Understanding (end of unit)

1. false
2. a
3. c
4. b
5. c
6. c
7. a
8. b
9. c
10. a
11. a
12. d

Unit 12: Earmold Impressions – Concepts and Techniques

Answers to Check Your Understanding

1. b
2. a, b, c
3. b
4. a
5. a
6. d
7. b
8. c
9. d
10. c

Unit 13: Infection Control

Answers to Check Your Understanding

1. Match the following terms to their definitions:

Virus	A submicroscopic, potentially infectious particle, made up of DNA or RNA.
Bacterium	A one-celled microorganism that multiplies by simple division.
Fungus	A microscopic plant that is capable of growing either on dead organic matter or in living hosts as parasites.

2. True
3. False
4. OSHA
5. Match the following terms to their definitions:

Cleaning	The process of removing gross contaminants from a surface without killing germs.
Disinfecting	The use of a chemical solution that will kill particular microbes in the vegetative state.
Sterilization	The most complete form of decontamination, killing 100% of the vegetative microbes and their resistant spores.