Audio Meter System Bekesy

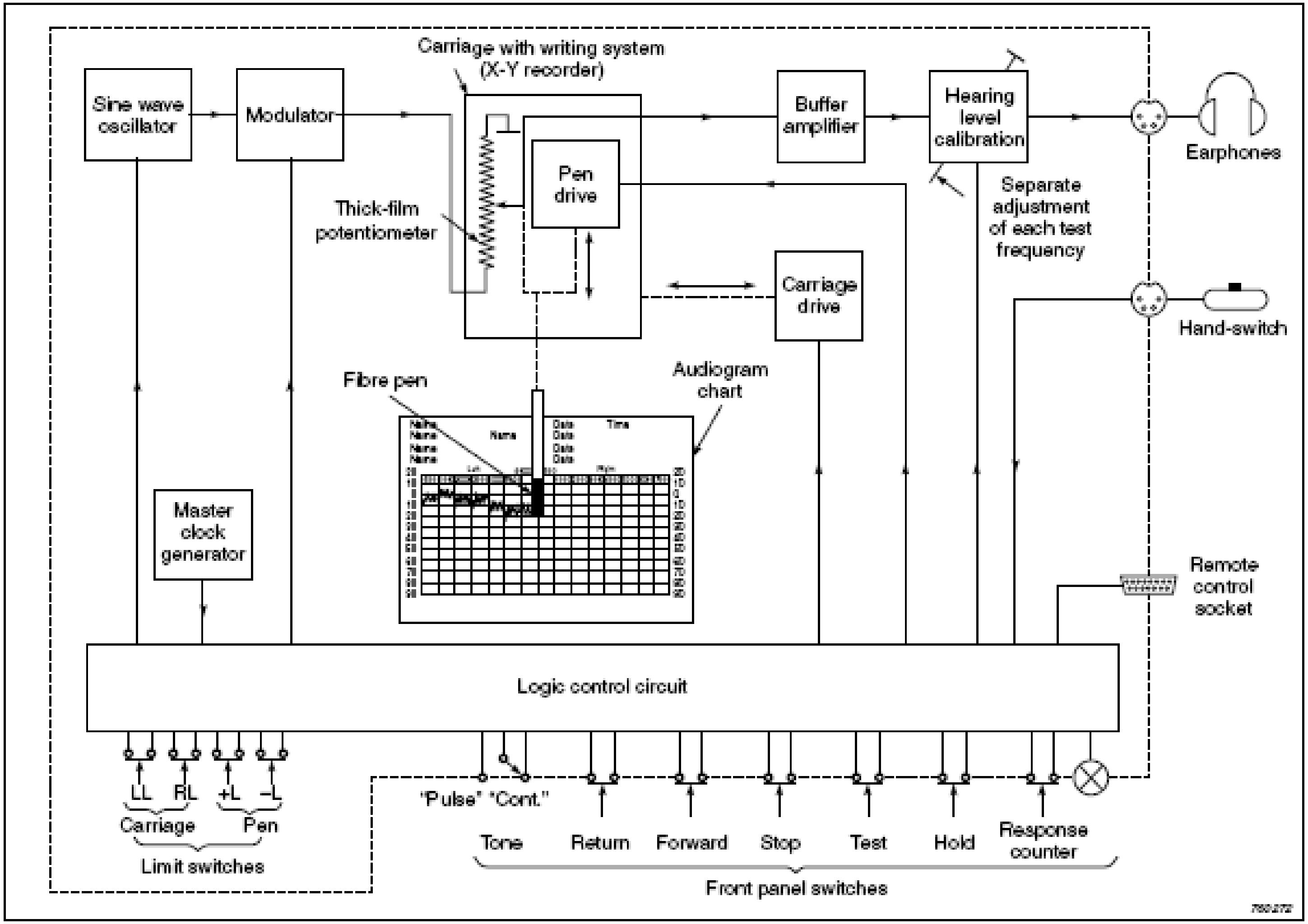
Bekesy System Contd..

George Van Bekesy, a Hungarian scientist, designed an automatic audiometric testing method for plotting the hearing threshold based on the patient's signal. A principal feature of the method, differentiating it from conventional pure-tone audiometric techniques, is the interdependence of the patient's response and stimulus intensity: responses govern intensity and are affected by changes they introduce in it. An audiogram traced by the Bekesy method represents the absolute threshold values at all frequencies in the range tested. In addition, it shows the difference, in decibels, between levels at which the patient just hears a signal of increasing intensity and those at which he just ceases to hear the signal when its intensity is decreasing. This latter characteristic often varies significantly with the type of hearing impairment, and can aid in establishing the site of lesion within the auditory system. On the basis of the audiograms, one can easily separate the conduction and perceptive hearing deficiencies from each other.

Bekesy System Contd..

Audiometers Bekesy are relatively simple for the patient to operate. The instrument generates a pure-tone signal, which is presented to him through an air-conduction earphone. The subject is told to press a switch when the tone is heard and to release the switch when it is not heard. This switch controls the motor-driven attenuator of the audiometer: when it is pressed, signal intensity decreases and when it is released, signal intensity increases. A pen connected to the attenuator traces a continuous record of the patient's intensity adjustments on an audiogram chart, producing a graphic representation of the subject's threshold. The test signal may be presented in a variety of ways, each suited to the investigation of a particular problem

Block Diagram



➤ Fig. 17.4 *Block diagram of the audiometer system Bekesy*

Electrical Switch

Sine Wave Oscillator: This oscillator generates test signals with frequencies of 125, 250, 500, 1000, 1500, 2000, 3000, 4000, 6000 and 8000 Hz. This sequence is first presented to the left ear automatically, each tone for 30 s, and then to the right ear, the shift between the frequencies being noiseless. After both ears have been tested, a 1 kHz tone is presented to the right ear to provide a useful indication of test reliability.

Modulator: From the oscillator the test signal is fed to the modulator, where the mode of operation is selected by the ‘Tone’ switch, via the logic control circuit. Two models, ‘Pulse’ or ‘Cont’, are available. In the ‘Pulse’ mode the test signal is modulated giving a signal, which is easily recognized by the patient. In the ‘Cont’ mode no modulation is applied, giving a signal suitable for use, when calibrating the audiometer.

Attenuator

Automatic Attenuator: The signal from the modulator feeds the automatic attenuator situated on the carriage together with the writing system. The attenuator consists of a logarithmic potentiometer which has its wiper attached to the pen drive so that the attenuation of the potentiometer corresponds to the position (y-axis) of the pen on the audiogram chart. The potentiometer has infinite resolution. The attenuation range is 100 dB, thereby covering the range of hearing levels from -10 to +90 dB. When the test is initiated, the attenuator starts at its top position of -10 dB and then increases the level with a rate of 5 dB/s. Also, when the test signal switches between the ears and when retesting at 1 kHz, the attenuator decreases the signal level to –10 dB to ensure that the right ear does not receive a tone at the elevated level possibly required at 8 kHz.

Buffer Amplifier

Hand Switch: The pen drive is controlled via the logic control circuit by means of the hand-switch operated by the patient. Pressing the switch decreases the output from the potentiometer and thereby the level in the earphones, while releasing the switch increases the output both ways with a speed of 5 dB/s.

Buffer Amplifier and Calibration Circuit: From the attenuator the signal is fed via a buffer amplifier to the hearing level calibration circuit. The buffer amplifier isolates the attenuator from the calibration circuit in order not to affect its output. The calibration circuit consists of seven potentiometers, one for each test frequency. During calibration, the potentiometers are adjusted one at a time until the correct level, measured in a coupler, is obtained in the earphones.

Earphones: The earphones are a matched pair with distortion, typically less than 1%.

Master Clock Generator: A stable clock generator supplies the necessary signals for the control of motor speed, attenuator speed, frequency shift, modulation and other timing functions. This makes the system independent of variations in line voltage and frequency.

Mechanical System

Carriage: The carriage with the writing system is driven by a stepping motor via a toothed belt. The speed and direction of rotation of the motor are automatically controlled via the logic control system. When the test is initiated and the patient indicates that he hears the signal by pressing the hand switch, the carriage moves along the X-axis (frequency axis) of the audiogram in agreement with the frequency of the test signal. When the frequency shifts, the carriage stops until the patient again, by pressing the hand switch, indicates that he hears the signal. This avoids wastage of recording space on the audiogram if a patient's hearing threshold varies from frequency to frequency or from left to right ear. When the complete test is finished the carriage and writing system return to the start position. To prevent carriage over-run, two limit switches are included in the carriage drive circuit.

Writing System

Writing System: The writing system is operated by the pen drive, which is driven by a stepping motor. The pen drive moves the pen, and with it the wiper of the automatic attenuator, along the Y-axis (hearing level axis) with a constant speed corresponding to the change in attenuation of 5 dB/s. The direction of movement of the pen is determined by the position of the hand switch operated by the patient. Limit switches are also included with the pen drive.

Audiogram Chart: The audiogram is printed in standard A5 format (148 X 210 mm). The recording space is large, 0.8 dB/mm, to enable easy reading. Space is provided on the audiogram side for registration of information on the patient, audiometer, operator, etc. while the other side has space for recording the patient's medical and occupational history. Four holes in the chart give precise and automatic location of the audiogram on the chart bed.

Hearing aids

Hearing loss has many forms. The most common is related to the body aging process and to long term cumulative exposure of the ear to sound energy. As one grows older, it becomes more difficult to hear. The ear becomes less sensitive to sound, less precise as a sound analyzer and less effective as a speech processor. Loss of hearing differs greatly in different individuals. Changes in the ear occur gradually over time. However, by the time the changes are manifested, it is estimated that approximately 30 to 50 percent or more of the sensory cells in the inner ear have suffered irreparable structural damage or are missing (Engebretson, 1994). Under these conditions, the only choice available for hearing-impaired individuals is to wear a hearing aid.

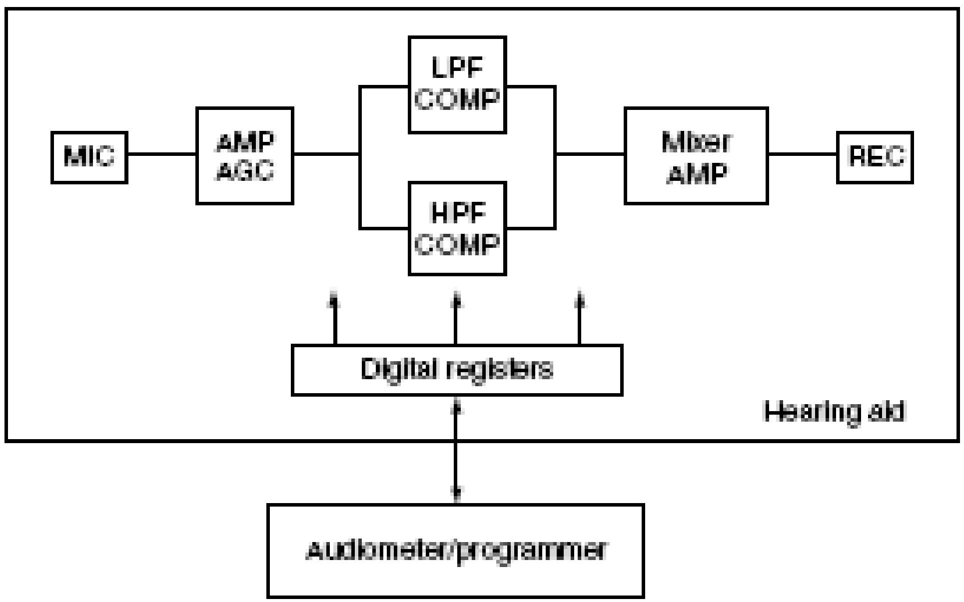
Hearing impairment is caused by either loss in sensitivity (loss in perceived loudness), or loss in the ability to discriminate different speech sounds or both. Loss of loudness may be due to either increased mechanical impedance between the outer ear and the inner ear or by the reduced sensitivity of the sensory organ of hearing. Loss of the discrimination ability is basically associated with damage to the sensory organ, although, other neural structures at higher levels may also be involved.

The modern hearing aid became possible with the invention of the transistor, which has enabled to develop small, power-efficient amplifier circuits that could be packed in a form that fits behind or in the ear. Even though the primary function of an hearing aid is to compensate for the loss of sensitivity of the impaired ear, in practice, it is not this simple. The ear behaves differently for soft sounds near the hearing threshold than it does for loud sounds. Therefore, a frequency response that restores normal hearing thresholds for soft sounds will not, in general be appropriate for louder sounds.

Furthermore, even when speech sounds are made audible for the hearing-impaired listener, it does not follow that he/she will be able to understand speech. Hearing-impaired listeners experience more difficulty in understanding speech in background noise than normal hearing listeners. In normal hearing the brain cells naturally filters background signal from speech and makes sense out of the signal. This would be a very difficult process for the hearing impaired persons and they also find loudness very uncomfortable.

Conventional Hearing Aid

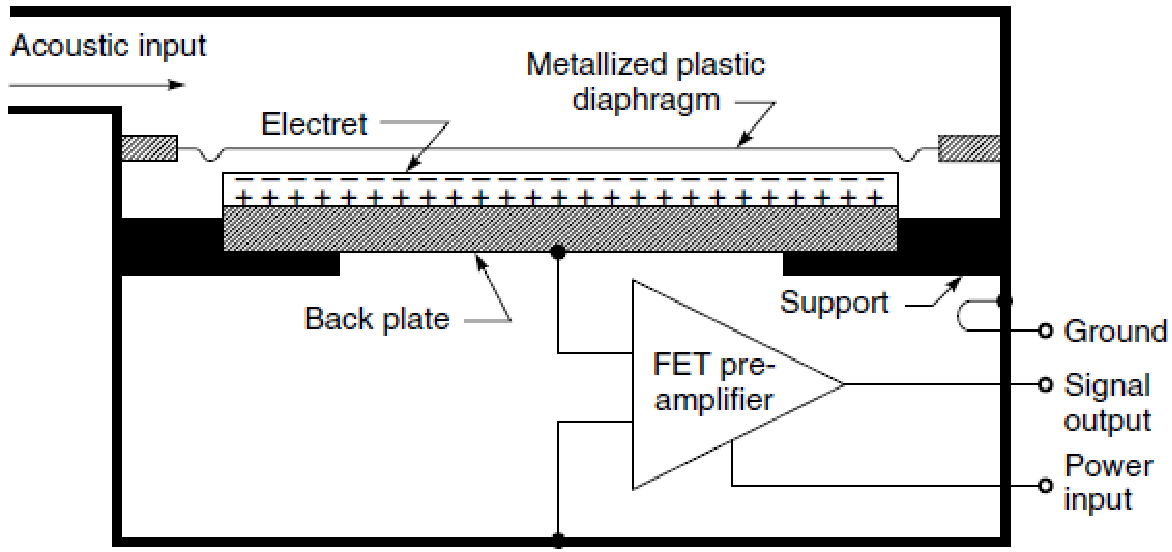
Modern hearing aids have evolved from single-transistor amplifiers to modern multi-channel designs containing hundreds and even thousands of transistors. A typical design is shown in Fig. 17.6. The basic functional parts include a microphone and associated preamplifier, an automatic gain control circuit (AGC), a set of active filters, a mixer and power amplifier, an output transducer or receiver. The total circuitry works on a battery. The use of multiple channels in this design provides different compression characteristics for different frequency ranges. Typically, the crossover frequencies of the channels and the compression characteristics can be adjusted with potentiometers. Most of the latest hearing aids are electronically programmable. The programmable parameters are downloaded from a computer-based system and stored in digital registers. The register outputs are used to switch resistor networks that control various analog circuitry. The active filters are adjusted to generally provide for low-frequency attenuation of up to 30–40 dB relative to the high-frequency response. This is because most hearing aid wearers require high frequency gain.



➤ **Fig. 17.6** *Conventional analog type hearing aid*

The transducer in a hearing aid, which is a microphone, can be realized in an integrated form with a field-effect transistor preamplifier (Fig. 17.7). The preamplifier is housed in the metallic, microphone case to shield its input from extraneous noise. On the other hand, the receiver is an electromagnetic device, which drives a miniature diaphragm to produce acoustic output. The acoustic output is routed to the ear-mould through a flexible tubing whose frequency response can be altered to boost the high-frequency response. This is done by tapering its inside diameter from the ear mould back to the receiver port end.

The most common type of microphone used in hearing aids is the electret microphones, where the polyster diaphragm held a few microns away from the metallic background plate which hold he charge. The transducer works on the capacitive principle and the electrical equivalent representation changes as the diaphragm moves due to the sound waves.



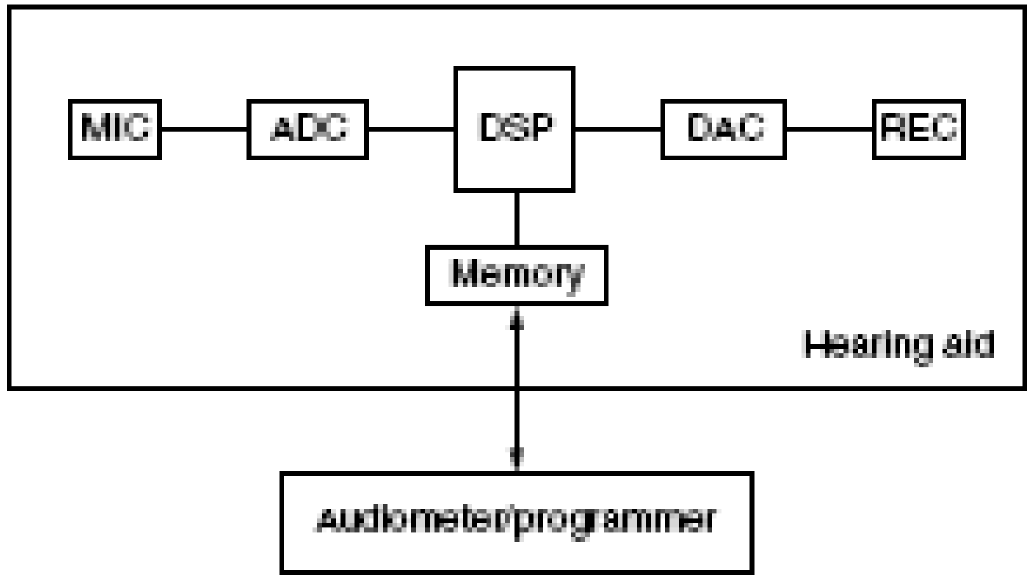
➤ **Fig. 17.7** *Integrated microphone and FET preamplifier (Electret microphone)*

All the electronics circuitry is packaged in a housing, which can be designed for fitting to the ear in any one of the following ways:

1. Placing all the components in a pocket-sized enclosure or box which is connected to the output transducer worn in the ear. The box can be carried in the shirt pocket or carried with a belt around the waist. With the availability of miniature-sized aids, this approach is no longer employed.
2. The components are packaged in a curved module, which is designed to fit comfortably behind the ear
3. The most popular design is in which the total package can be put inside the outer ear.
4. Using DSP processors we can go for improvement in performance in terms of the dynamic range (variable frequency and variable gain) reduction in power consumption, and decrease in size

Digital Hearing Aid

A typical digital hearing aid is illustrated in Fig.17.8. The major parts are the microphone, an analog-to-digital converter (ADC), the digital signal processor (DSP), the digital-to-analog converter (DAC), the receiver and a two port memory. Essentially, sound waves picked up by the microphone and transformed into electrical signals are converted into digital form by an A-D converter. A typical microphone will have an internal noise of 20 dB SPL (sound pressure level) when referred to the input and maximum undistorted output corresponding to a signal of about 90 dB SPL. Allowing some margin for peak performance, the total dynamic range required of the ADC is 80 dB. This requirement can be achieved with a 14 bit A–D converter.



DAC = Digital-to-analog converter
ADC = Analog-to-digital converter
MIC = Microphone
REC = Receiver
DSP = Digital signal processor

➤ **Fig.** 17.8 *Block dragram of a digital hearing aid (after Engebreston 1994)*

Since DAC has uniform spectrum, at high frequencies the low frequency signals will also be amplified and add to the noise, which could be above the threshold and can cause disturbance. The digital technology uses CMOS technology around 10nm. Around 10000 CMOS inverters are needed to implement the DSP functionality.

- The hearing aids both analog & digital can be adjusted by the audiologist for the user. The hearing aids will be custom made depending on the audiologist recommendations. Analog programmable hearing aids allow adjustment of the parameters. But the audiologist will have more flexibility in setting the parameters. Patients have to gradually get adjusted to the use of hearing aids.

You tube links

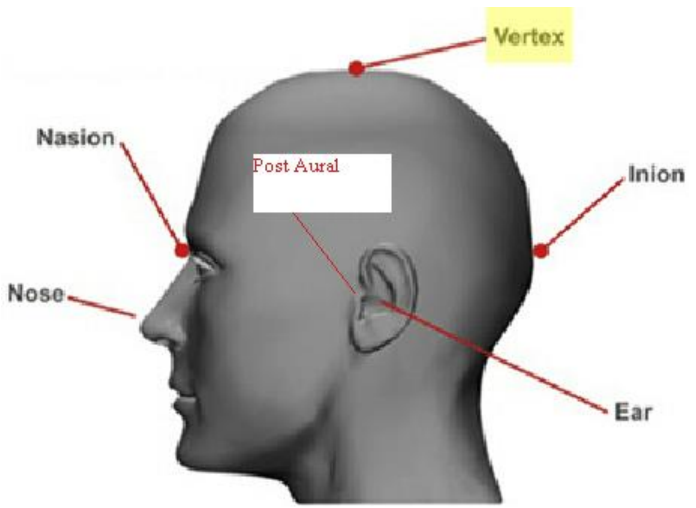
- https://www.youtube.com/watch?v=LhSpb36_1s4
- https://www.youtube.com/watch?v=AxzVyMcmRcs

Evoked Response Audiometry System

Evoked Response audiometry has been the subject of research for several years. This work has established evoked response electroencephalography resulting from an auditory stimulus above the hearing threshold. The acoustic stimulus of the human auditory receptors triggers a number of evoked potentials, the action potentials in the cochlear and the auditory pathway. Instruments based on this principle have been found particularly suitable for determining auditory threshold in the absence of voluntary response in subjects such as infants, uncooperative adults, or animals.

The system basically comprises a conventional wide range pure-tone audiometer, which operates under the control of an automatic programmer and provides a series of auditory stimuli to the subject via either a loudspeaker or standard earphones. The EEG signal is picked up by standard electrodes placed in contact with the subjects scalp. One electrode is usually placed on the vertex, one at the post auricular area, and a third (ground) on the earlobe or forehead. The instrument stores and evaluates that part of the EEG signal, which follows each individual stimulus presentation. At the end of the programmed series of stimuli, it writes out on a paper chart a waveform that is the average response to stimuli. The presence of characteristic amplitudes and latencies in this waveform give an indication that the test intensity exceeded the subject’s threshold at the test frequency. Similar trials at other intensity levels and other frequencies establish the threshold contour.

Points on the Scalp



ABR or BERA

Here the patient has to sit back and just listen to the clicks and sounds and does not have to do anything as the electrodes pick up the signal and finds out whether the sound reaches the brain (cerebellum or brain stem) and it is called Brain Stem Evoked Response Audiometry or BERA or ABR Audiometry Brainstem Response. It is mainly used to find out the abnormality in the brainstem and the auditory nerve.

The tones, duration and intensity are decided based on the gender and age group.

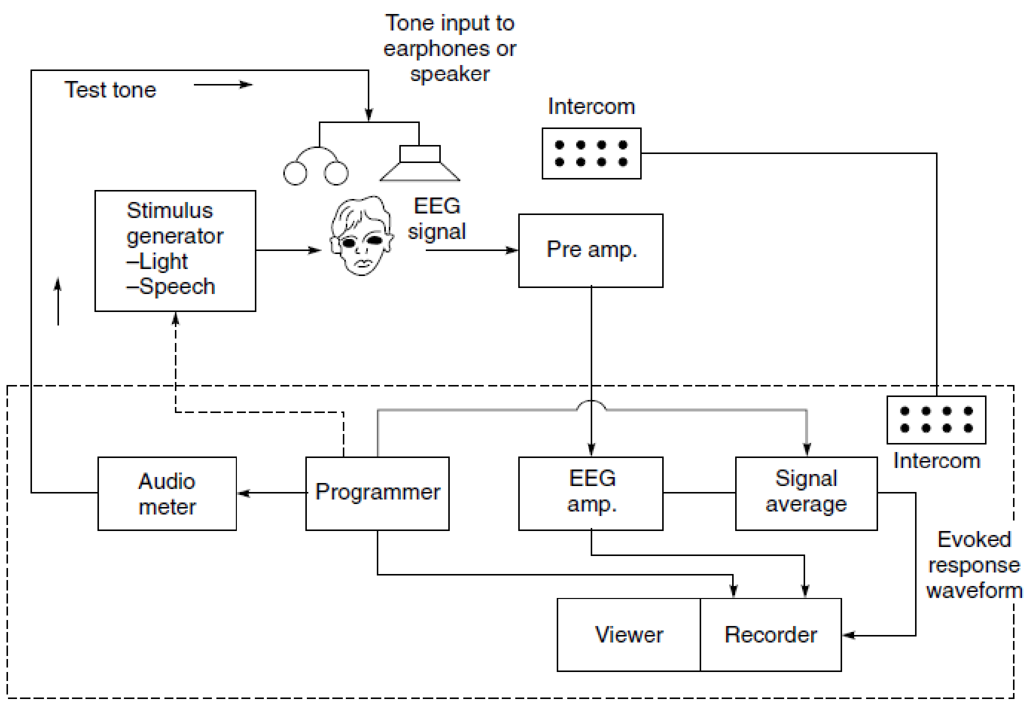


Fig. 17.5 *Block diagram of the evoked response audiometer*

The Tone Generator It is a wide range pure-tone audiometer whose frequency output can be selected at 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000 and 8000 Hz. The output power levels are adjustable from –5 to + 110 dB in 5 dB steps. In addition to the pure tones, internally generated broad-band noise may be used as the stimulus. Provision is also made for external input from other types of stimulus (speech). A special feature of the generator is the selectable rise/decay times for 1–100 ms. Outputs are provided for the left ear, right ear, or both. A variable intensity masking noise source is included in the generator. A power amplifier is incorporated to drive a speaker or tactile transducer. Total harmonic distortion should not exceed 2%. The brain responds 12ms, after the tone is presented.

EEG Amplifier It is a conventional high gain, high impedance, low noise amplifier. The first stage of the EEG amplifier is preferably kept in a separate "preamp head" located near the subject. Its design and location minimize power line frequency pick-up. An ohm meter provided in the preamp head enables the measurement of the electrode contact impedance and thus indicates when satisfactory contact resistance of the electrodes is obtained. The EEG preamplifier gain is fixed at 200. Generally a 5KΩ source resistors are used. The overall sensitivity is adjustable in steps of 10 to 1000 mV/div on the chart recorder. The amplifier also provides selectable roll-offs at the high and low ends of the spectrum of interest and a 60 Hz sharp notch filter. The low frequency roll-off points are at 0.15, 0.30.0.60, 1.5, 3, 6,10, 15,30, 60 Hz at 6dB (half amplitude), whereas high frequency points are at 1.5, 3, 6, 10, 15, 30, 60, 100 Hz at 6 dB (half amplitude) yielding a 12 dB/octave roll-off.

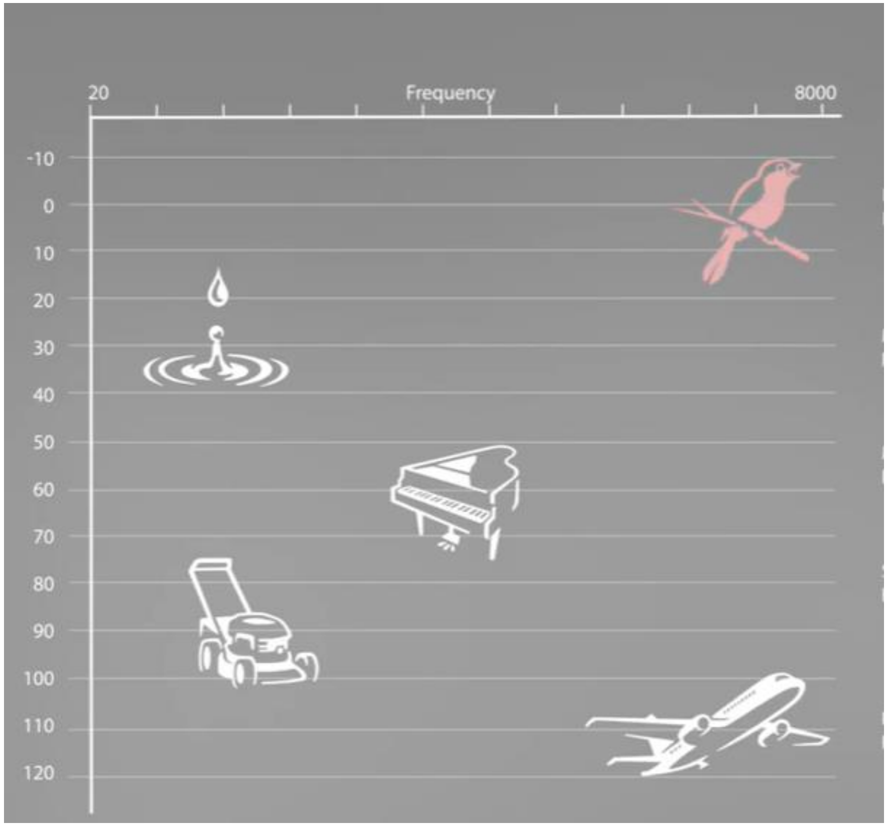
The Programmer: A logic device that controls the system operation in correct time sequence. It helps to have a selectable rate of stimulus presentation, stores the number of pulses that the operator chooses to constitute a run, starts the recorder at the beginning of the run, turns the audiometer tone generator on and off to provide the auditory stimuli at the proper time. It also speeds up the chart drive for the detailed signal samples, stops the recorder after providing for paper clearance, erases the signal averaging computer and clears and resets itself for the next run. Total count in the programmer is selectable from 1–109 stimuli with a selector switch. The pulse interval is normally kept as 0.1, 0.2, 0.5, 1, 2, 5, 10 and pulse duration as 1–10,000 ms.

Signal Averaging Computer: This separates evoked responses from the normal EEG activity by ignoring those components, which are not synchronized with stimuli. Because the waveform of the evoked potential will be essentially the same every time in response to the tone presentation, and the other electrical activity will vary randomly, the evoked response "grows" in the computer memory and the noise component tends to average to zero with repeated presentations. It may be provided with either 50 or 100 averaging points depending upon the degree of resolution required. The computer includes a provision for selection of integrating time constant (5, 10, 20, 50, 100, 200, 500 s) and sweep duration or analysis time (0.1, 0.2, 0.5, 1, 2, and 5 s). A delay circuit is incorporated to select a delay between the onset of the stimulus and the start of analysis by the signal averager. The delay time is selectable as 0, 0.2, 0.5 and 1 s. The amplitude of the evoked response may be normalized to that of the on-going EEG monitoring signal using a gain control (gain variable from 20 to 50). The computer provides outputs for display either on a conventional oscilloscope or on an X-Y plotter.

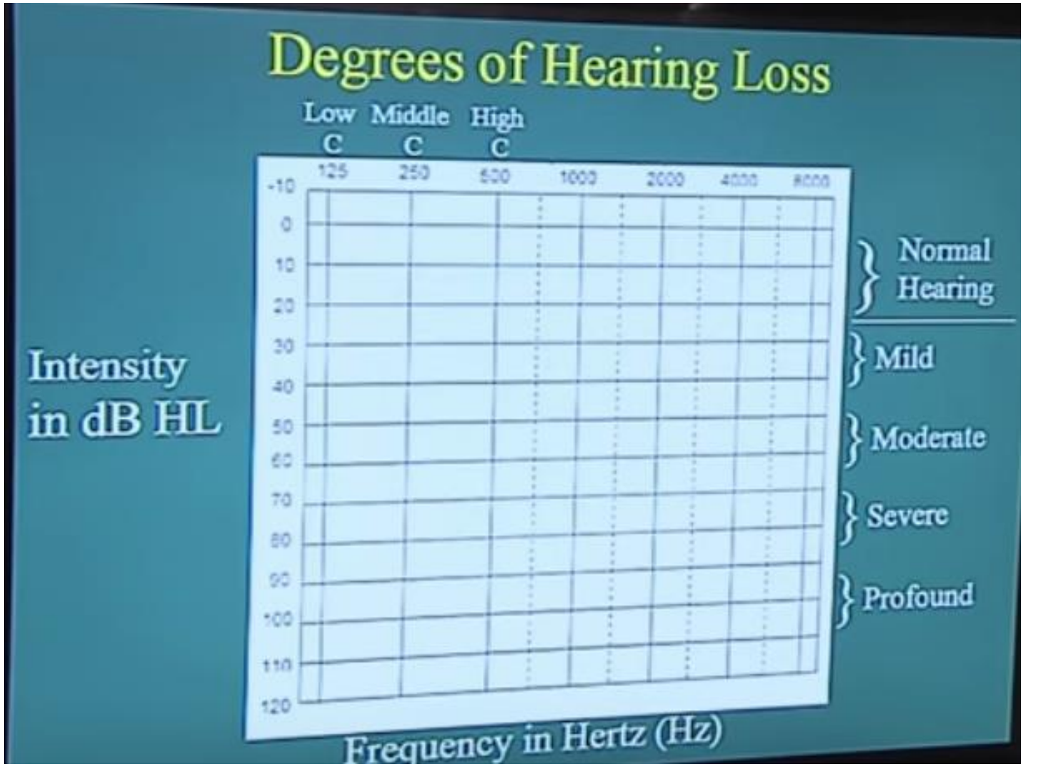
Chart Recorder: It is a two channel recorder. One of the two channels is used to display the averaged response after it has been processed by the computer and the other displays unprocessed EEG. There are two event markers, one of which is activated by each gating pulse from the programmer to show the beginning and duration of each stimulus and the other is available for registering any mark at the desired instant. The chart can be driven at four different speeds (1, 5, 25, 125 mm/s), which are automatically switched by the programmer. Translucent chart paper is usually employed so that records may be compared by overlaying one on another on the illuminated opal-glass viewer. Evoked response audiometer systems also contain a provision for 'External' mode of operation where any other type of stimulus generator can be connected into the system and controlled from the programmer. This could include narrow band noise, speech, a high frequency auditory signal for animal research or even a photic or tactile stimulator.

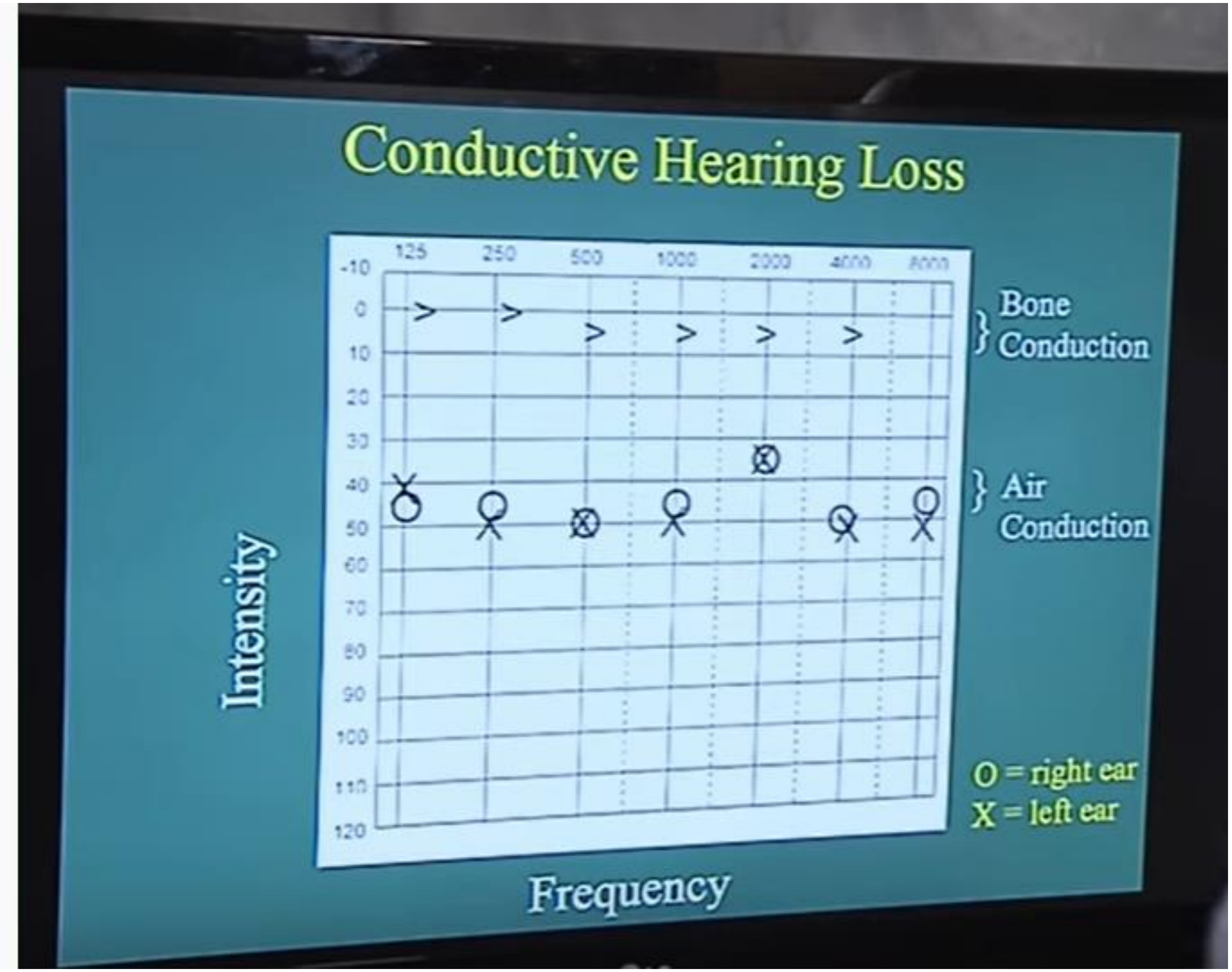
Modern evoked response audiometers are built around microcomputers. In these instruments, the stimulators, preamplifiers and amplifiers are all digitally controlled via the central processing unit, which automatically avoids undesirable parameters. They have no push button or dials as the parameters are varied by means of the keyboard. The parameters can be controlled to a very wide range, which would not be possible with conventional knobs and switches. Stimuli from 12 to 16 kHz are included to facilitate investigation into high frequency hearing loss and ototoxic drug effects. Texts and waveforms are displayed on a large size TV screen. A built-in chart recorder helps to make recordings under the control of the keyboard.

Frequency Range of Different Sounds

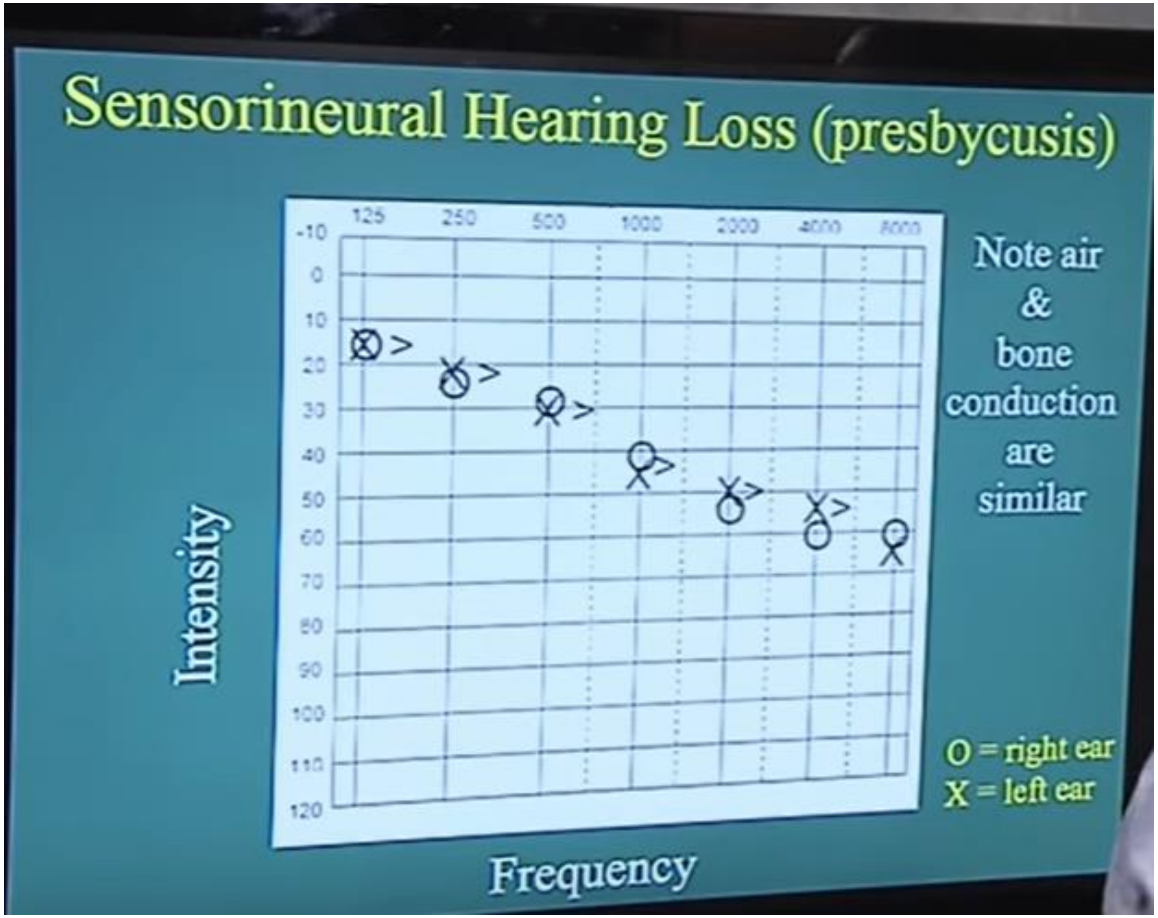


0-25 dB Normal Hearing
25-45 dB Mild Hearing Loss
45-65 dB Moderate Hearing Loss
65-85 dB Severe Hearing Loss
90-110 dB profound Hearing Loss

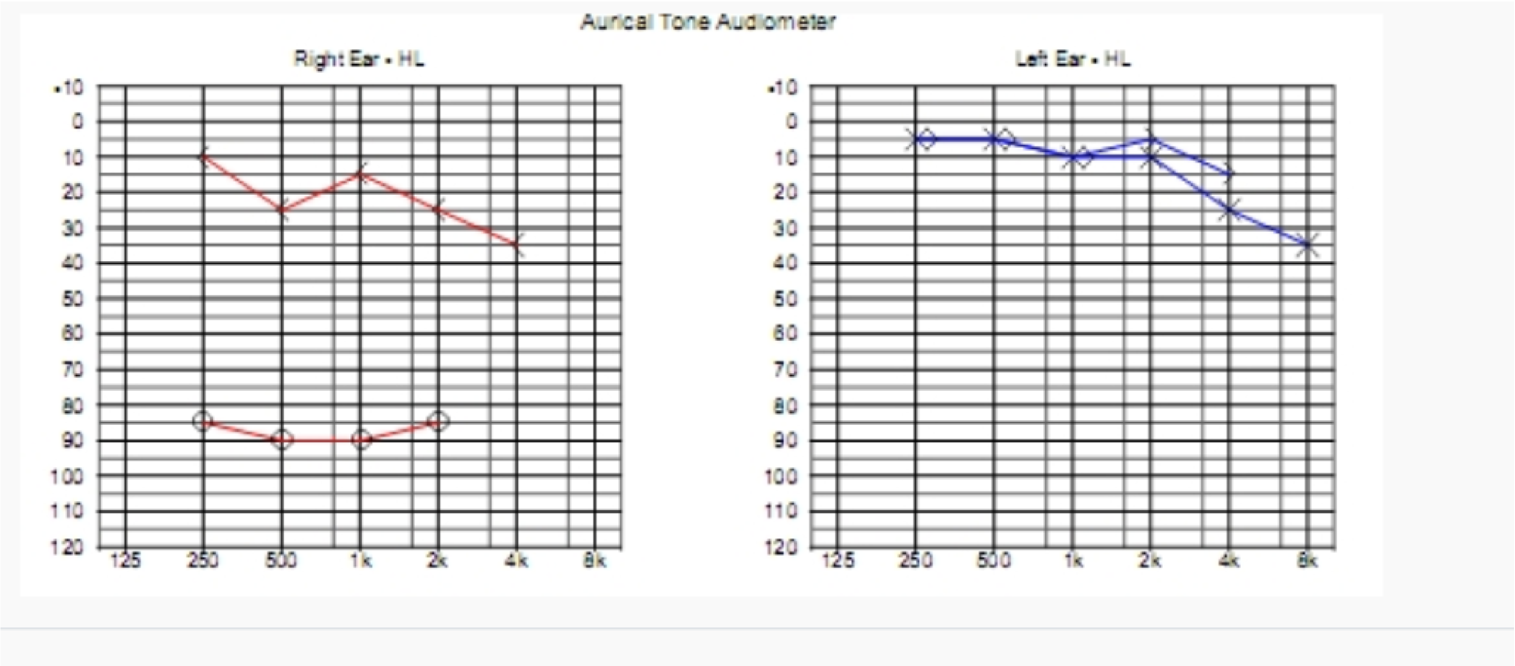




Bone conduction is flat and normal and the graph shows hearing loss in outer ear, because of air conduction loss or plugged airway



The graph shows a decline in hearing loss for both air & bone conduction. This is a problem of sensory nerves and the hearing loss is normal to moderate and this commonly found as ageing sets in.


Aurical Tone Audiometer
Right Ear - HL
Left Ear - HL
-10
0
10
20
30
40
50
60
70
80
90
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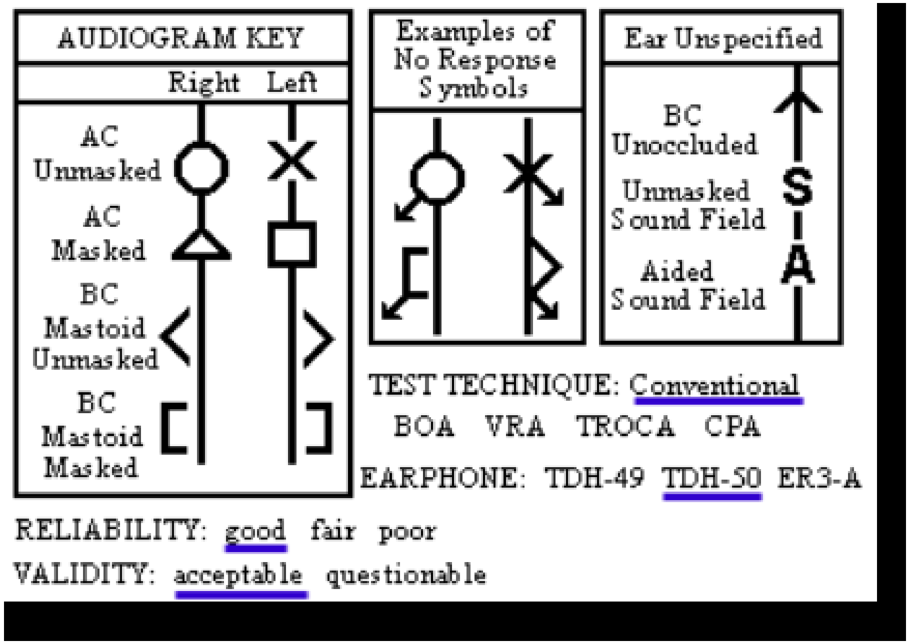
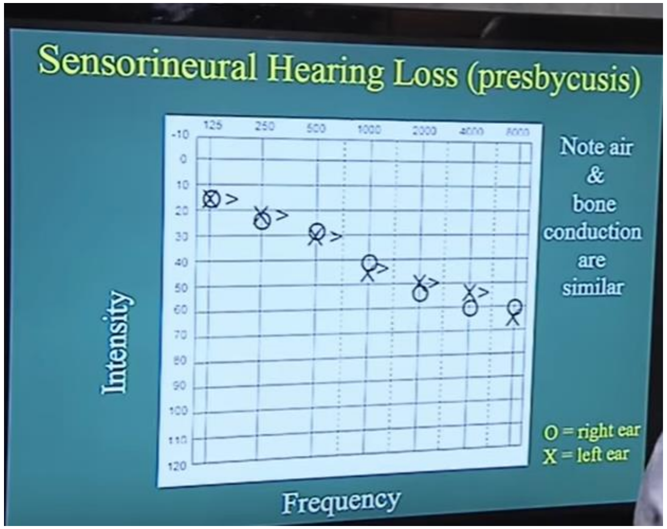
125
250
500
1k
2k
4k
8k

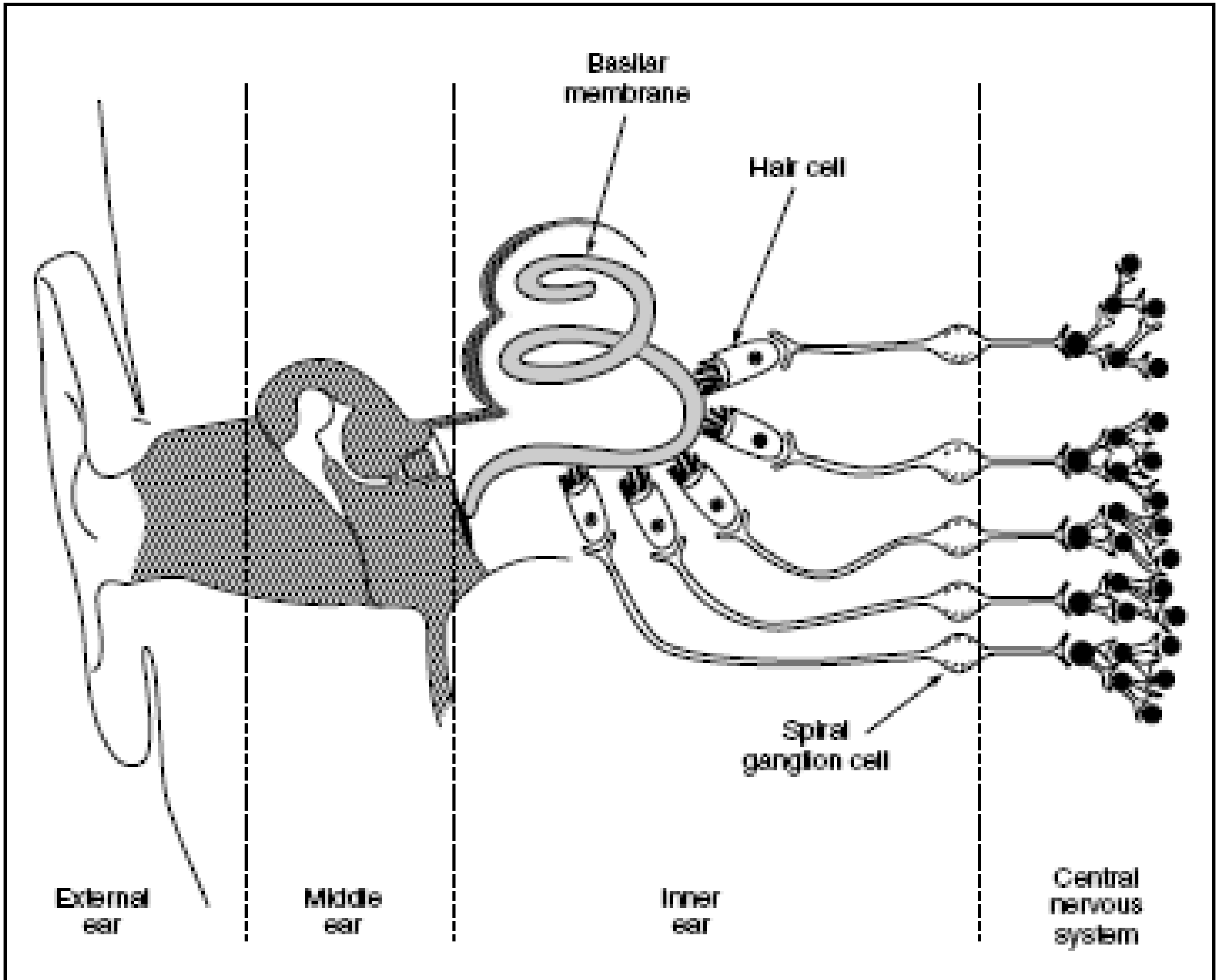
Audiogram Legends



Cochlear implants

The cure for sensorineural deafness (Presbycusis) is the cochlear impalnts. Sensori-neural deafness can be caused either by cochlear damage or by damage within the auditory nerve or to the neurons of the central auditory system. The hair cells are the sensory cells that transduce mechanical motion into signals that can be recognized by auditory neurons. The auditory neurons carry information from the hair cells to the cochlear nucleus in the brainstem and, via the cochlear nucleus, to higher nuclei in the brain (Fig. 17.9).

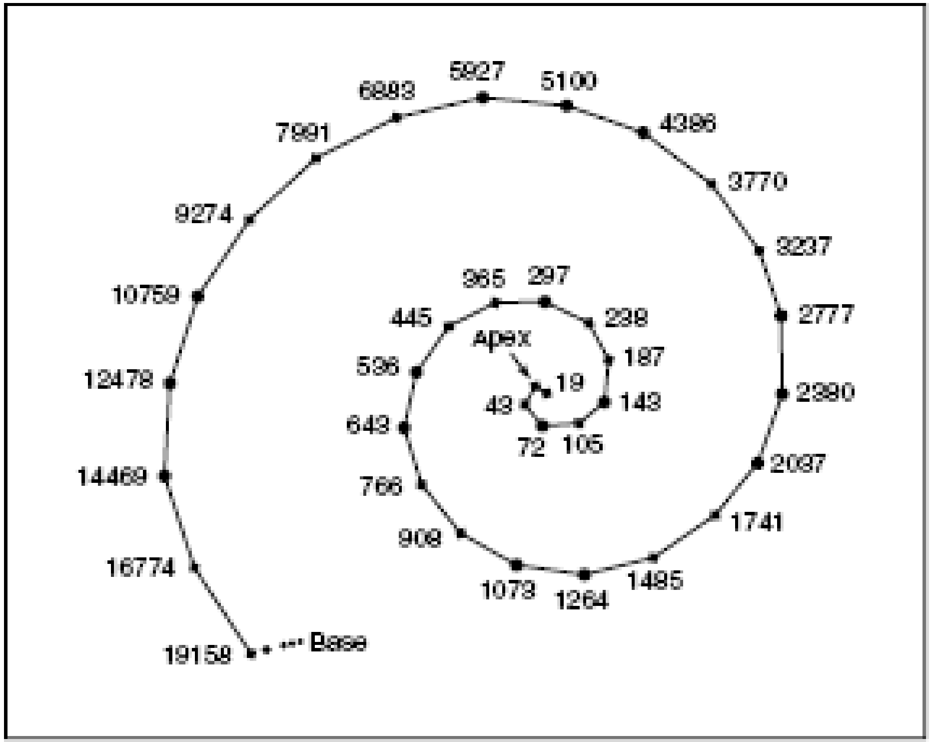




➤ Fig. 17.9 *Details of cochlear part of the human ear (after Loizou, 1999)*

The normal cochlea and the associated neurons of the central auditory system provide information about both the frequency content and intensity of the auditory signal. Information is conveyed to the acoustic nerve about frequency content by the mechanically tuned properties of the basilar membrane. The inner hair cells, which connect to the vast majority of afferent neurons, are thought to be the sensory cells of the cochlea whereas the role of the outer hair cells is still under investigation. The location of hair cells along the cochlea determines their optimal response to frequency: hair cells at the apex are responsive to low frequencies, while hair cells at the base are responsive to high frequencies. The distribution of frequencies along the spiral is logarithmic (Fig. 17.10).

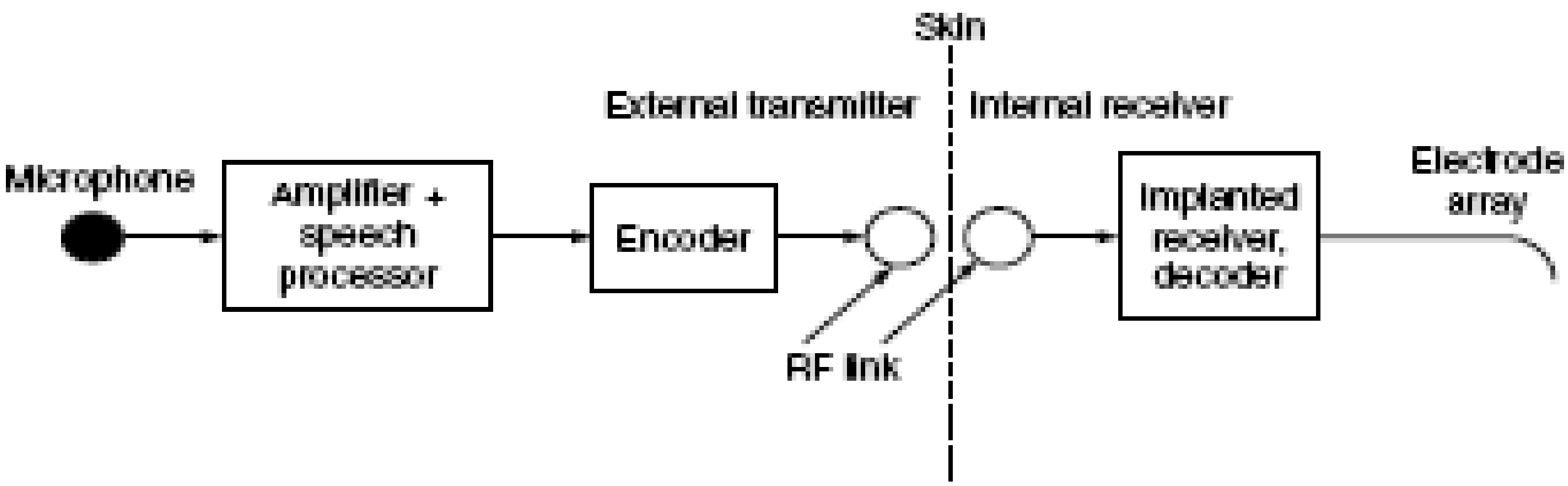
If the damage to the auditory system is peripheral in the inner ear, then a cochlear implant can be used. In the general design of a cochlear implant, the sound is decomposed into frequency bands of use for the transmission and reception of speech, and critical features of the signals within those frequency bands are delivered to auditory neurons via an array of electrodes.



➤ Fig. 17.10 *Diagram of the basilar membrane showing the base and the apex. The position of maximum displacement in response to sinusoid of different frequency (in Hz) is indicated*

A block diagram of a generic cochlear implant is shown in Fig. 17.11. The microphone converts acoustic signals into electrical signals. The electrical signals are amplified and encoded in various ways in the block called stimulus encoder. In the vast majority of implants, the stimulus encoder is worn outside the head, producing a serially coded signal that is transmitted with a transcutaneous link, most often inductive. The link sends both data and power to an internal circuit that decodes the serial data stream and decomposes it into signals that are delivered to the current sources that drive the electrodes of the cochlear electrode array. Each electrode of the array is driven with either a pulse or an analog electrical signal. The signals traverse the tissues of

Block Diagram of Cochlear implant

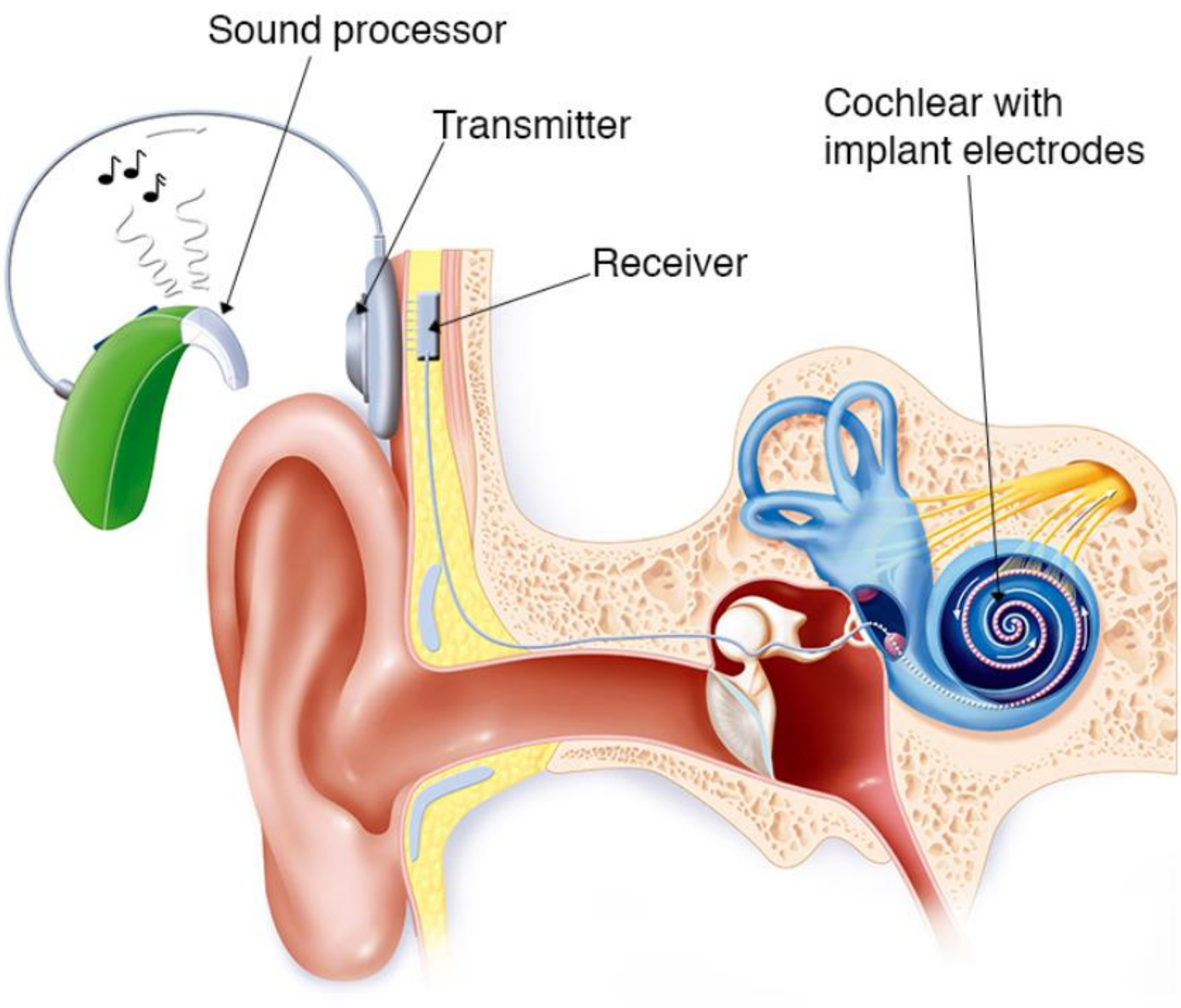


➤ Fig. 17.11 *Schematic of a cochlear prosthesis (after Spelman, 1999)* the inner ear, usually the fluids of the scala tympani, and excite the auditory neurons. The excitation depends upon the number of intact neurons that remain, the proximity of the electrode array to the neurons, and the spatial and temporal characteristics of the current-density fields that affect the neurons.

In single channel implants, only one electrode is used. In multi-channel cochlear implants, an electrode array is inserted in the cochlear so that different auditory nerve fibres can be stimulated at different places, thereby exploiting the place mechanism for coding frequencies.

- Different electrodes are stimulated, depending on the frequency of the signal. Electrodes near the base of the cochlea are stimulated with high frequency signals while electrodes near the apex are stimulated with low frequency signals. The signal processor is responsible for breaking the input signal into different frequency bands or channels and delivering the filtered signals to the appropriate electrodes.

Cochlear Implants



Cochlear implants have been a spectacular success story for biomedical engineers.

More yet to come

You tube links

- https://www.youtube.com/watch?v=LhSpb36_1s4
- https://www.youtube.com/watch?v=AxzVyMcmRcs