Advanced Audiometer: A
Novel Signal Generator Technique

Manish Kumar Nalamwar
(Roll No. 04307418)
Supervisor: Prof. P. C. Pandey

Abstract

Audiometry is the technique to identify and quantitatively determine the degree of hearing loss of a person by measuring his hearing sensitivity, so that suitable medical treatment or one of the appropriate hearing aids and assestive device can be prescribed. In audiological investigations, the hearing sensitivity is tested for pure tones, speech or other sound stimulus. The results when plotted graphically are termed as audiogram. The electronic instrument used for measuring hearing threshold level is called an audiometer. Test tones of different frequencies and levels are generated and presented to the subject. Hearing thresholds are determined on the basis of responses from the subject. Different audiometric tests, techniques, and various audiometers are discussed.

A Novel signal generator technique for advanced audiometer is suggested which gives finely tunable pure sine wave and noise signals with controlled spectrum. Because of revolution in electronic science all the methods and algorithms can be embedded in one chip with the help of hardware descriptive language and this chip can be a part of add on card, which may be used in any high performance medical instruments.

Key words: Audiogram, Direct digital synthesis (DDS), Tone decay test (TDT), Short increment sensitivity index (SISI), Programmable division factor (PDF).

1. Introduction

There could be various disorders in the various parts of the ear. Audiological investigations help to diagnose the nature of deafness and localise the site of disorder. The method by which subject's hearing sensitivity can be determined is termed as audiometry. It helps in assessing the nature, degree, and probable cause of the hearing impairment. In this technique, auditory stimuli with varying intensity levels are presented to the person who responds to these stimuli. The minimum intensity level of these stimuli to which consistent responses are obtained is taken as the threshold of hearing. Depending on this threshold, the subject's hearing sensitivity can be estimated by obtaining an audiogram. An audiogram is a plot of threshold intensity versus frequency.

The second section of paper discusses disorders of the auditory system. Third section describes the various investigations through audiometric techniques. Fourth section provides description of audiometers. In the fifth section different types of audiometer and their features including microcontroller based audiometer have been discussed [1]. A novel signal generator technique for advanced audiometer is discussed in the sixth section.

1.1 Auditory system

Our system of hearing comprises of two sections viz. a peripheral section which is our ear and a central section located in the brain, which carries the sensation from the ears to the auditory area of the cerebral cortex. The auditory area of the cerebral cortex (called auditory cortex) is the area of the brain, which is dedicated to and specialised in interpreting the sound. The ear receives the sound in the form of sound energy, which is a form of vibration. This vibrating energy enters the external part of the ear (called external auditory meatus) and vibrates the eardrum (technically known as tympanic membrane). This vibration of the tympanic membrane is picked up by a chain of small bones called malleus, incus and stapes, which conduct vibration to a specialised organ called cochlea [7].

The cochlea is the transducer of the hearing system. The function of the cochlea is to convert the vibratory energy into electrical energy. Once this has been achieved, this electrical energy enters the nerve of hearing (called auditory nerve) and carries the sensation through different parts of the brain to the auditory cortex, where the sensation of sound is analysed and interpreted. The phenomenon by which sound reaches the inner ear through the eardrum is called air conduction. Sound, particularly in the low frequency range, may reach the inner ear via the bones in the head rather than from the eardrum. This phenomenon is called bone conduction [6]. Wearing earplugs results in a greater percentage of the sound heard coming from bone conduction. Normally only a small fraction of sound is received in this way; however, deaf people whose inner ear still functions normally may be able to hear sound conducted to the ear in this way. For proper hearing each and every part of this system right from the external auditory meatus to the auditory cortex has to be normal.

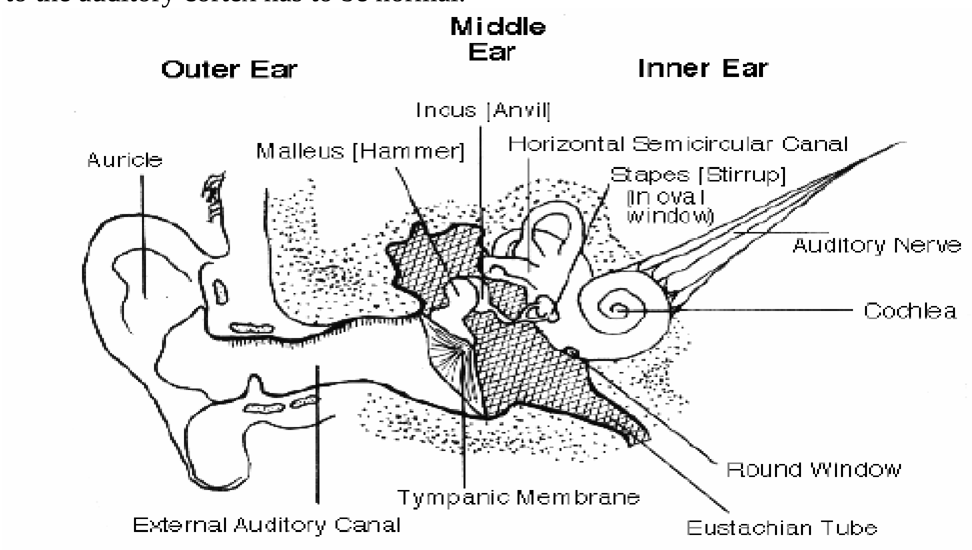

Fig. 1. The organ of hearing, consisting of the outer ear (auricle and pinna), the middle ear (ossicles) and the inner ear (cochlea). Adapted from [6].

1.2 Sound perception

Sound is generated in nature whenever an object vibrates in an elastic medium like air. Sounds in nature are complex and not pure tone or sine waves [2]. However, all complex sounds can be considered as a mixture of different pure tone sounds of different frequencies.

The ear is not equally sensitive to all frequencies, particularly in the low and high frequency ranges. The frequency response over the entire audio range has been charted, originally by Fletcher and Munson in 1933, with later revisions by other authors, as a set of curves showing the sound pressure levels of pure tones that are perceived as being equally loud. The curves are plotted for each 10 dB rise in level with the reference tone being at 1 kHz, also called loudness level contours and the Fletcher- Munson curves, as shown in Fig.2. The lowest curve represents the threshold of hearing, and the highest represents the threshold of pain.

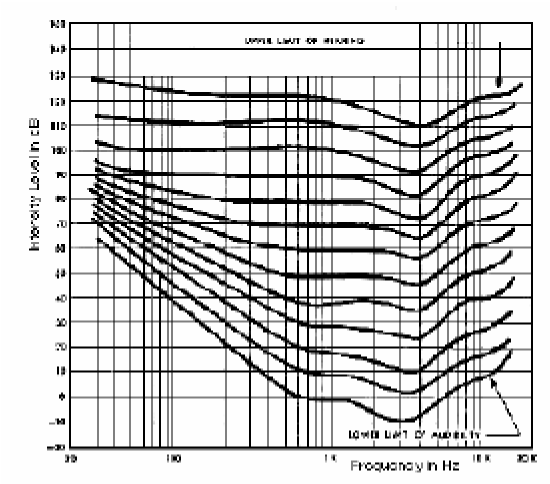

Fig.2 Curves based on the studies of Fletcher and Munson showing the response of the human hearing mechanism as a function of frequency and loudness levels. Adapted from [6].

The curves are lowest in the range from 1 to 5 kHz, with a dip at 4 kHz, indicating that the ear is most sensitive to frequencies in this range. The intensity level of higher or lower tones must be raised substantially in order to create the same impression of loudness. The phone scale was devised to express this subjective impression of loudness, since the decibel scale alone refers to actual sound pressure or sound intensity levels.

Human hearing ranges from 20 Hz to 20 kHz. There is little speech information above 8000 Hz. Perception of frequencies below 100 Hz is increasingly tactile in nature, making them difficult to assess. The loss of hearing sensitivity is observed first at high frequency (8 kHz) and later on as the loss progresses, its effect is observed in the mid-frequency region (1-2 kHz). By the time loss is observed in the low frequency region the subject nears to deafness. Hence, audiometric tests carried out in the low frequency region do not give any significant information about hearing loss. Therefore, audiologists routinely test only in the range of 250-8000 Hz, often in octave steps. Standardized frequencies tested include 250, 500, 1000, 1500, 2000, 3000, 4000, 6000, and 8000 Hz. This represents octave intervals, by convention, but intervening frequencies may also be tested.

In acoustic measurements, sound level is often given in dB, taking sound pressure of 20 μPa as the reference level, and is known as sound pressure level (SPL).

Sound level in dB SPL = 20 log (measured sound pressure / 20 μPa)

However, in audiometry the sound level of pure tones is given in dB by taking average hearing threshold of normal hearing young adults as the reference, and is known as hearing level (HL) [2].

Sound level in dB HL = 20 log (measured sound / average threshold of normal hearing)

The hearing threshold is frequency dependent. Hence, SPL corresponding to a given HL varies with frequency. Intensity levels in audiometers are indicated in HL. Table 1 gives the dB SPL (dB HL) threshold values of a normal person for standard frequencies. The "0 dB" hearing level in audiometry is a modal value derived from a large population of normal persons. Normal values for auditory thresholds were defined by the International Standards Organization (ISO) in 1984. These values are derived from large population studies of normal adults 18-30 years of age. The following table gives the dB SPL values corresponding to 0 dB HL for standard frequencies.

Table 1 Threshold values in dB SPL for 0 dB HL (ISO, 1984). Adapted from [5].

Frequency (Hz)	250	500	1k	1.5k	2k	3k	4k	6k	8k
dB SPL	25.5	11.5	7	6.5	9	10	9	10.5	13

Since both HL and SPL are logarithmic units, a certain increment in HL corresponds to the same value increment in SPL [2].

1.3 Audiogram

An audiogram is a plot of threshold intensity versus frequency. The intensity scale in HL increases downwards, and hence the audiogram resembles like an attenuation response, a lower point on the audiogram indicating higher loss. A typical audiogram (dB HL vs. frequency graph) comparing normal and impaired hearing is shown in Fig.3. The dip or notch at 4 kHz, or at 6 kHz is a symptom of noise-induced hearing loss.

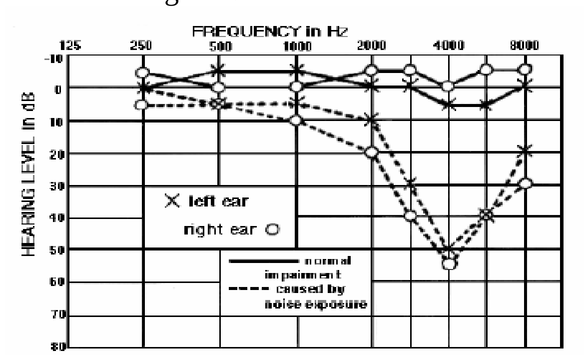

Fig. 3 Audiogram of normal ears and impaired ears. Adapted from [6].

Most thresholds are approximately 0 dB HL for a normal ear. Points below 0 dB HL denote louder threshold levels. Those above 0 dB HL are expressed in negative decibels and are less intense. An audiogram comprises of four separate curves. They are right ear air conduction (AC), right ear bone conduction (BC), left ear AC, and left ear BC. The symbols used on most audiograms are x - left air conduction, o - right air conduction.

2. Disorders of the auditory system

Each section of the ear has diseases specific to it and specific tests (investigations) are there to identify disorders in each portion. The common causes of disorder in the external auditory meatus is collection of either wax or fungal debris or foreign body in it. To diagnose this no investigation is required and doctor can see it directly and clean it with instruments. This deafness due to blockage of the external ear is usually very slight.

The middle ear comprises of the eardrum, the ossicles, and the air space within the cavity of the middle ear. The common diseases affecting this portion are perforation in the ear drum, a stiffness or damage to the chain of small bones in the ear, and collection of fluid in the middle ear space (called middle ear effusion). A perforation can usually be diagnosed just by visual inspection. The other middle ear disorders require special investigations for confirmation. Any deafness due to disorders in the external auditory meatus or in the middle ear is called conductive deafness because the primary function of these portions is the conduction of sound to the inner ear and disorders in these areas impede the conduction of sound to the cochlea. A perforation in the eardrum or a stiffness of the ossicular chain can be corrected surgically. Collection of fluid in the middle ear is usually treatable by medicines but sometimes surgery is required.

The diseases of the inner ear, i.e. the cochlea is difficult to treat. Disorders of the inner ear not only cause a deafness called sensorineural (perceptive) deafness but also may case a peculiar sensation of buzzing sounds in the ear called tinnitus [6]. However tinnitus is not specific only of cochlear damage and sometimes disorders of the middle ear and / or the portion of the auditory nerve in the brain can also cause tinnitus. Deafness due to disorders of the inner ear is commonly refractory to medical and surgical methods and usually hearing aids are the only option. Deafness may also occur due to diseases of the nerve carrying the sensation from the cochlea to the brain.

3. Examination of sense of hearing

Different audiological investigations help us to diagnose the nature of deafness and localise the site of disorder. The simplest investigation for deafness is pure tone audiometry. It measures hearing acuity (i.e. how perfectly the subject can hear) and tells us whether is deafness is conductive (disorder in external auditory meatus and / or middle ear) or sensorineural (disorder in the inner ear or in the nerve of hearing in the brain) or whether the deafness is mixed, i.e. a disorder combining both the conductive apparatus as well as the inner ear / nerve of hearing.

Tympanometry is also a common audiological investigation. It assesses the structural integrity of the middle ear. It helps us to diagnose the nature of the disorder in the middle ear in cases of conductive or mixed deafness.

3.1 Audiometry

Audiometry is the technique to identify the nature of hearing loss and to determine the threshold of hearing by recording responses of the subject after presenting him with auditory stimuli with varying intensity levels. There are different audiometric techniques and procedures used for achieving this. For air conduction testing, stimuli are presented to each ear independently with specialized earphones. For bone conduction testing, a bone vibrator is placed onto the mastoid process of either right or left temporal bone; external auditory canals are not usually occluded. All equipment must be continually calibrated to conform with international standards. This ensures that a gradual loss of hearing noted on serial testing is truly valid and not due to machine error.

3.1.1 Masking in Audiometry

In audiometry, both ears are tested separately. In air and bone conduction audiometry where sound is applied to one ear, the contra lateral cochlea is also stimulated by transmission through the bone of the skull. In case the sound in one ear is sufficient to stimulate the second ear, it is called cross hearing. During the air conduction test, the stimuli while passing from test ear to cochlea of the non-test ear get attenuated. This loss of sound energy is called interaural attenuation and varies between 45 to 80 dB [2]. However, during bone conduction test, the cochleae of both sides are equally stimulated i.e. the inter-aural attenuation is of 0 dB. Hence, cross hearing is a serious concern in case of bone conduction test than it is for air conduction.

A simple procedure by which this can be done is to deliver a noise to the non-test ear in order to remove it from the test procedure by masking. Here masking noise which is loud enough to prevent the tone reaching and stimulating the non-test ear, but at the same time it should not mask the sensitivity of the test ear over masking [2]. Thus, an audiologist should provide appropriate level of masking. The masking noise is often selected to be a wide-band noise, or narrow band noise with the band centered about the test frequency. Wide-band noise has uniform power density spectrum over all the audible frequency range i.e. from 250 Hz to 8 kHz.

However the masking effect is actually contributed by frequency components centered on the test tone frequency, over a bandwidth of about 1/3 to 1/2 octave, known as critical band. Broadband noise band pass filtered with a band approximately corresponding to the critical band is known as narrow band noise, and compared to wide band noise it gives the same masking effect at a lower sound pressure level.

3.2 Audiometry Techniques

There are two types of audiometric techniques, subjective type and objective type.

3.2.1 Subjective Audiometry

In subjective test, the subject has to respond when he hears the presented sound. Subjective type audiometric test involves presentation of systematically varying acoustic stimuli to the subject and recording the responses.

3.2.2 Objective Audiometry

Objective test only requires co-operation from the subject towards attachment of the measuring electrodes or probes. There are different audiometeric procedures depending on the stimuli used.

3.3 Audiometric Procedures

There are different audiometric procedures depending on the stimulus used. In pure tone audiometry, the subject's threshold for hearing is measured. In speech audiometry, the subject threshold for the reception of speech is recorded. Different Audiometric techniques are described into the following sections.

3.3.1 Pure Tone Audiometry

Pure tone audiometry is a procedure for determination of the extent of hearing loss and the cause, i.e. conduction or sensorineural loss. The subject's hearing threshold for acoustic stimuli of different frequencies are measured. The initial level of the stimuli is selected by the audiologist.

Procedure

In this technique, at the outset, subject is instructed to signal the audiologist each time a tone is perceived. A variety of response signals may be employed - responding "yes" with each tone, tapping the rhythm of tones, or pointing to the ear where the tone is heard, or better by a response switch. For air conduction thresholds, earphones are comfortably positioned and the better ear tested first, if known. If not known, some audiologists will quickly screen each ear using the same initial frequency and the better ear tentatively determined.

Tones are often presented in an ascending series, that is, from low to high frequency. Initially a single frequency stimulus at some presumed level is presented to the subject. Initially a pure tone of 30 dB HL is presented to the subject. If the response is positive, the tone level is decreased in steps of 10 dB till the subject does not give response.

On the other hand, after applying 30 dB tone at first time, if the subject does not hear it, the level is raised in steps of 10 dB step until it is heard for first time. Once, the response is positive, the tone is decreased by 10 dB. If the subject hears this tone, the tone is again decreased by 5 dB. If the subject does not hear it, the tone is again raised by 5 dB. In this way by several presentations, the hearing threshold is obtained. Often, tone intensities slightly above and below this auditory threshold are tested to verify and help "hone in" on the precise threshold value. The minimum presentation level at which the subject responds at least 50% times (3 responses out of 6 tone presentations), is taken as the hearing threshold.

Specific situations are as follows. If profound hearing loss is expected, frequencies from 125-500 Hz are tested first (some audiologists screen initially at 500 Hz then skip to 4000 Hz, if normal hearing expected). If a tone is not audible even at maximum audiometer output, "no response" is recorded [8]. If 100% correct response occurs at a minimal intensity, testing below 0 dB is possible. Thus, certain individuals may demonstrate greater hearing sensitivity and thresholds down to -20 dB are measurable.

The results of the audiometry are reported in an audiogram. Different shapes of audiograms are associated with different types of hearing loss [2]. When prescribing hearing aids the audiogram will guide the degree of amplification required at various frequencies. For site of lesion testing, "conductive" loss implies a lesion in the external auditory meatus, tympanic membrane, and / or middle ear. "Sensorineural" loss usually implies a lesion in the cochlea or acoustic nerve (cranial nerve VII), but not the cortex.

3.3.2 Tone Decay Test (TDT)

The purpose of this test is to quantify the deterioration in the auditory nerve. Here, a tone of particular frequency with threshold intensity is presented as a continuous tone and the time for which the subject able to hear is recorded. This test can be carried out with or without detecting the hearing threshold of the subject.

Procedure

This procedure is based on the Carhart's method of tone decay test [2]. In this method a pure tone stimulus is presented 10 dB below threshold and raised in 5 dB steps till the subject responds. As soon as the subject responds a stop-watch is started, the tone being constantly maintained. The subject is asked to indicate as soon as he fails to hear the tone, and the time on the stop-watch is noted. If the tone is heard for one full minute then the test is terminated. If the subject stops hearing the tone before completion of one minute then the time is recorded and next step is started immediately.

In this step tone is immediately raised by 5 dB without giving any time gap, and stop-watch is brought back to zero and started again. If the subject hears the tone for one full minute the test is terminated, but if the subject stop hearing before the end of one minute then time is recorded and this process repeated all over again. This raising of intensity of the tone by 5 dB steps is continued till the subject can hear the sound for one full minute. If it is found that the tone has been raised by 30 dB above threshold i.e. 30 dB (SL), and yet the subject is unable to hear the tone for one full minute the test is not continued further.

The result of this test is recorded as the difference in decibels between the hearing levels at which subject could hear the tone for one minute and the corresponding pure tone hearing threshold level for that particular frequency. If, in spite of increasing the sound level by 30 dB above hearing threshold the subject does not hear the sound for one minute, then the result of tone decay test is recorded as positive.

3.3.3 Short Increment Sensitivity Index (SISI) Test

The SISI test is used to detect the pathology in cochlear or retrocochlear lesions [2]. This test is normally carried out after finding the pure tone hearing threshold using normal pure tone audiometry this test determines the capacity of a subject to detect a brief 1 dB increment in intensity, provided at 5 seconds interval at a particular frequency.

Procedure

In SISI test, the operator selects the test frequency and sets the level to 20 dB suprathreshold level. The tone is presented with brief bursts of 1 dB modulation above the carrier tone at every 5 s. The 1 dB increment is presented for an interval of 300 ms, out of which the rise time and fall time are 50 ms each. The subject is asked to press the response button whenever he detects a change in the level [2]. Twenty such bursts are given and out of them, the number of bursts the subject is able to detect is recorded. The number of responses is converted to percentage and stored as the test results. The same procedure is repeated for each frequency, and the result is stored. A SISI audiogram is plotted on the basis of percent score for each of the test frequencies.

3.3.4 Bekesy Audiometry

This is another form of pure tone audiometry here a self-recording audiometer is used in which the changes in intensity as well as frequency are done automatically by means of a motor [2]. The change in frequency can occur in forward or in backward manner. Conventionally, a forward change is used. The motor drive attenuator is controlled by a switch, which is operated by the subject. The subject presses the switch as soon as he hears a sound and releases it as soon as he stops hearing the sound. The audiometer is so programmed that a tracing is recorded only when the subject presses the switch, the frequency being continually changed either in the forward or backward manner. A graphical representation of the subjects hearing threshold across the entire frequency range is thus obtained by the successive crossing and recrossing of the hearing threshold in the form of a jugged line. Two tracings are recorded for each ear, one by presenting a continuous tone and other by presenting a pulsed tone [2].

3.3.5 Speech Audiometry

Pure tone threshold testing attempts to assess sensitivity, speech audiometry testing attempts to address the integrity of the entire auditory system by assessing the ability to here clearly and to understand speech communication. The main use of speech audiometry is in the identification of neural types of hearing loss, in which both the reception as well as the discrimination of speech is impaired more markedly than in cochlear or conductive hearing loss. There are two types of speech audiometric tests, speech discrimination test and speech reception threshold test.

3.3.5.1 Speech discrimination test

In speech discrimination test, lists of monosyllable speech discrimination words are presented over earphones for each ear which subject is asked to repeat. The percentage of the total number of words presented which the subject is able to identify correctly gives the speech discrimination score (SDS). The SDS is determined when the subject repeats 50% of the words correctly. The result of this test is from 0 to 100 %. Generally, a high score is associated with normal hearing or conductive hearing loss and low score is associated with sensorineural loss.

3.3.5.2 Speech reception threshold test

This test is similar to the speech discrimination test except for the fact that this test uses two syllable words with equal stress (spondees) and the words are attenuated successively. The SRT (speech reception threshold) is the lowest hearing level in dB HL at which 50 % of a list spondee words are correctly identified by a subject. For estimating SRT, a group of 6 spondee words is presented at 25 dB above the average pure tone audiometry threshold for 500 Hz and 1000 Hz, and then at successively lower intensities. When the level is such that the subject is able to identify 3 words out of 6 correctly, the level is taken as SRT. The SRT of a normal subject is very closely related to his pure tone hearing threshold and the SRT is generally 2 dB lower than average of pure tone hearing level thresholds at 500 Hz and 1 kHz. A list of 36 such words in English language are prepared by the Central Institute for the Deaf. A way of differentiating between neural and other types of hearing loss is by graphically plotting the performance intensity function. This is done by ascertaining the speech discrimination score at different sensation levels and plotting the percentage of correctly identified words as a function of the intensity of presentation of the words.

4. Audiometer

An audiometer is an instrument, which is used for carrying out audiometric tests and procedures. Audiometer can be of different types, depending upon the frequency range, range of acoustic output, mode of acoustic presentation, masking facility, procedures used, and types of acoustic stimuli. It is capable of generating pure tones at a specific frequency, specific intensity, and duration, either single or in series.

A conventional audiometer instrument has dials or knobs with calibrated scale for frequency selection and for tone masking noise level selection. The variation of the level of the stimulus is done manually by the audiologist after carefully observing the responses of the subject. The limitations and drawbacks of this conventional audiometer are that the interrupter switch is used for tone switching and needs to be mechanically silent. The presence of mechanical parts makes the instrument more susceptible to wear and tear. Calibration is necessary, at least, once in six months.

The advancement in technology has made the various switching tasks simple, flexible, and noise free. Application of microprocessor/PC in audiology offers many advantages in terms of flexibility and simplicity of use, over their conventional counterparts [2]. Increased accuracy and precision removes the need for frequent calibration of audiometer, which was required for earlier audiometers.

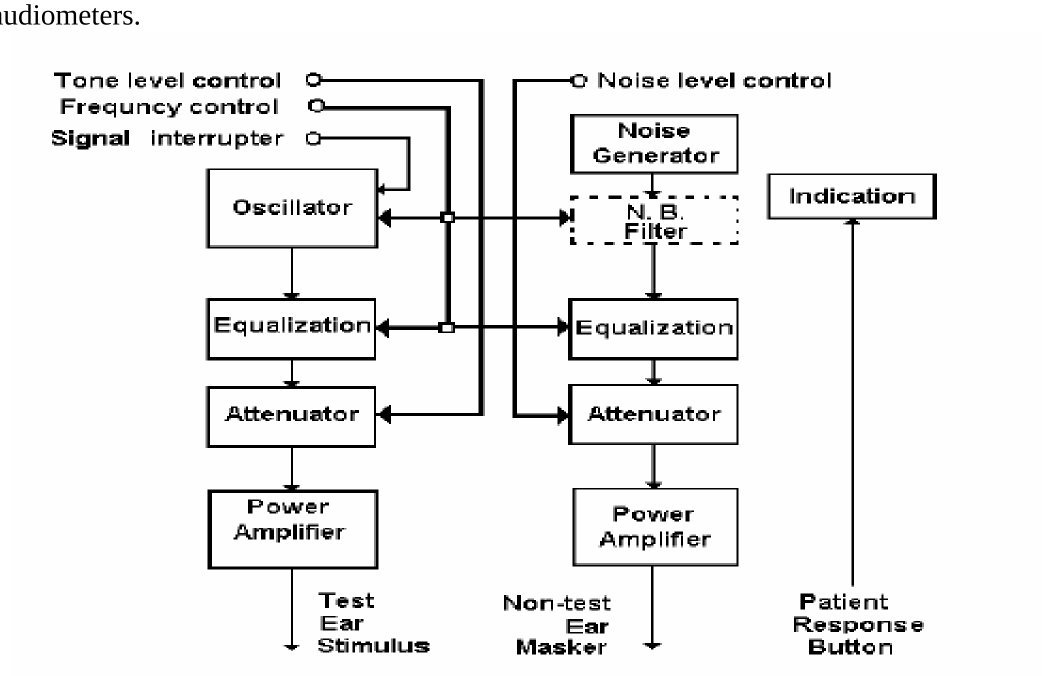

Fig. 4 General block diagram of an audiometer. Adapted from [1].

A general block diagram of an audiometer is shown in Fig.4. It consists of two channels, namely tone generator and noise generator, and each channel having an attenuator, equalization circuit, and power amplifier. The tone generator or oscillator should have a frequency range from 250 Hz to 8 kHz, controlled by frequency control. Each of the frequency should be within 3% of the indicated frequency. The generated tone should be stable. The equalization circuit is required firstly, to provide frequency dependent attenuation in order to calibrate the output sound levels in dB HL and secondly, to provide different amount of attenuation for different output devices used (headphone, loudspeaker, and vibrator). The attenuator, known as the as hearing or tone level control, should be capable of controlling the output sound level over a desired range in steps of 5 dB. Calibration should ensure the output sound level to be within ± 3 dB of the indicated value.

For the masking purpose, the noise generator should provide wide band noise, which has energy spectrum equally distributed over the test frequency range i.e. up to 8 kHz. There should also be a facility for narrow band noise, wherein the narrow band noise output should be distributed around the test frequency. The output power available from the power amplifier determines the maximum sound pressure level available from the headphones and the bone vibrator. The amplifier must have low distortion and a good S/N ratio to meet the standard requirements. A response switch is given to the subject, to indicate his response.

5. Detailed features and specifications of various audiometers

Tremetrics, Model: RA500

Audiometer type: microprocessor-based audiometers facility of air and bone conduction. Facility of Manual or automatic testing.
Test frequencies: 500, 1000, 2000, 3000, 4000, 6000, 8000 Hz.
Distortion: Total Harmonic Distortion below 40 dB
HL attenuator: Selectable from –10 dB to 100 dB in 5 dB steps.
HL accuracy: ± 1dB
Earphones: Telephonics TDH-39, 10 ohm earphones in Model 41 cushions
Mechanical: High impact GE Noryl U.L. approved plastic housing,”touch sensitive ” panel with super- twist LCD
Computer Interface: Two Built-in RS232C ports
Data Output: 300-19200 baud
Built-in Printer: High speed graphic printer, avg 6-8 seconds/audiogram
Power supply: 120 VAC, ± 10, 60 Hz.
240 VAC, ± 10, 50 Hz.
Weight: 4.313 kg.
Size: 17.5 x 12.25 x 4 inches [7]
Calibration: Non-volatile EEPROM memory provides for electronic calibration. Date displayed on screen and printed on screen and printed on each audiogram

Audiometrics, Inc (AZ26)

Audiometer type: microprocessor-based audiometers Type 4 with Wide Band, High Pass, Low Pass noise.
Facility of air and bone conduction. Facility of Manual or automatic testing. Painted metal cabinet
Frequencies: Frequency Accuracy ± 3%.
250 500 1000 2000 3000 4000 6000 8000 Hz.
Attenuator: -10 TO 120 dB HL in 5-dB steps.

Tone output range:

Frequency (Hz)	250	500	1000	2000	3000	4000	6000	8000
Air L_{max} (dBHL)	100	100	120	120	120	100	100	100

Minimum Amplitude: (air) 125 Hz -10 dB HL
Noise Generator: Wide Band, High Pass, Low Pass.
Channel Inputs: Tone, Speech Microphone, Noise.
Channel Outputs: Speaker, Earphones: TDH39P, Bone Vibrator: Radioear B-71
Auto Threshold Determination: Modified Hughson Westlake according to ISO 8253-1.
Interface: Built-in RS232C input/output computer interface.
Built-in Printer**:** Thermal printer. Paper width: 112mm
Power supply: 100, 110, 120, 220, 230 or 240 V, AC 50-60 Hz.
Weight: 9.5 kg.
Size: 19 x 16 x 6 in. [8].

Madsen, Orbiter Version 2 (OB 922-2)

Clinical Audiometer
Channels: 2 separate and identical channels
Outputs: Phones, bone, insert, free-field loudspeaker via external or internal amplifier
Tone Stimuli: Pulse, pure or warble tones
Frequency Range: Air FF: 125 to 20,000 Hz
Bone: 250 to 8,000 Hz
Frequency Resolution: Standard frequencies 6/12/24/48 points per octave, or 1Hz
Masking Signal: White Noise, Speech Noise and Narrow Band Noise
Attenuator: 1dB step resolution
Hearing Level Range:
Air: -10 to 120-125 dB HL at 500-6000 Hz
Bone: -10 to 70-80 dB HL at 500-4000 Hz
Speech Input: Microphone
Special Tests: S.I.S.I, ABLB, Tone Decay etc.
Data Communication: RS232C Serial Data Interface
Printer: Optional thermal printer
Display: 640 x 200 monochrome LCD display
Power Supply: AC 50-60 Hz or 200-240 V, ± 10 %
Subject Safety: Complies with EN 6060-1, Class I, Type B
Dimension: 20.5 x 16.5 x 6.1 in.
Net Weight: 9 kg. [9]

Audiometer developed at IIT Bombay

Audiometer type: dual channel microcontroller based audiometer, with pure/warble tone/AM tone stimulus and wide-band/narrow-band masking noise. Facility of air and bone conduction. Facility of auto testing, SISI test, tone decay test and speech audiometry.

Circuit size: two double-sided PCBs with PTH. PCB-1 of 14.5 cm x 13.5 cm and PCB-2 of 10 cm x 13.5 cm.
Stimulus: crystal controlled test tone frequencies, with intensity level controlled in 5 dB steps.

Tone output range: for air conduction and bone conduction are 0 to L_{max} (dBHL) for different frequencies as given below

Frequency (Hz)	250	500	1000	1500	2000	3000	4000	6000	8000
Air L_{max} (dBHL)	90	100	100	100	100	100	100	90	80
Bone L_{max} (dBHL)	40	50	50	50	50	50	50		

Warble tone: frequency deviation of ±10% with one sweep in two seconds.
Amplitude modulated tone: amplitude deviation of ± 5 dB with one sweep in one second.
Noise Generator: Masking noise: broadband/narrow-band noise over 0-60 dBHL range in 5 dB step.
Wide-band noise: flat spectrum up to 8 kHz, with approx. 12 dB/octave roll off on the higher side.
Narrow-band noise: centered at test tone frequency, 3-dB BW = 0.55 octave, 20-dB BW = 4 octave.
Channel Outputs: Headphone type TDH-39 (software calibration for other headphones, by changing a table).
Bone Vibrator type: Oticon 70127 (software calibration for others)
Control and indication: control through 4x4-matrix keypad of size 9x9 cm. 16 characters x 2 lines LCD display with font 5x7 or 5x10 dots.
Operation: software controlled menu driven manual / automated modes.
Result Storage: for one set of the test results with rewrite facility.
Interfacing: serial port (TxD, RxD, and GND), TTL level, baud rate of 2400 bits per second, 7 bit data, and even parity.
Self test: internal monitoring of output levels.
Power supply: +5V, 20 mA for digital and ±5V, 120 mA for analog [2].

Table 2 Comparison summary of audiometers

Model	Type of test	Freq. Range (Hz)	Masking noise		Output device	Control & Indication	Operation	Interface	Self test
			Wide band	Narrow band					
IITB-AUD2k1	Air conduction test Bone conduction test	250 – 8000	Up to 8000 12 dB / octave roll off	Centered at test tone Freq. 3 dB & 5 dB / octave	Head phone TDH-39 Bone vibrator Oticon 70127	Key Pad LCD	Manual / Auto	RS –232C port	Internal monitoring of output level
RA 500	Air conduction test	500 – 8000	-	-	Head phone	Graphic Printer	Manual / Auto	RS –232C port	Memory based
AZ 26	Air conduction test Bone conduction test	250 – 8000	*A	*A	Head phone	Built in printer	Manual / Auto	RS –232C port	-
OB 922-2	Air conduction test Bone conduction test	250 – 8000	*A	*A	Head phone	LCD Thermal printer	Manual / Auto	RS –232C port	Memory based

* A - Masking noise facility available but no details are provided in the data sheet.

6. Advanced audiometer

Medical instruments for ear, nose, and throat (ENT) such as audiometers, impedance meters, and evoked potential systems quality of audio signal is most important factor. In all of these instruments, audio signals, such as pure sine waves with variable frequency (from 100 to 20KHz), narrow band noise, speech noise and white noise are required. These signals can be generated using direct digital synthesis (DDS) for sinusoidal signal and pseudo-random sequence generators (PRSG) for noise generation [3].

One of the most critical requirements for sine wave generators is the stability and the resolution of the obtained frequency. Traditional analog sinusoidal sources allow a resolution of about 5 Hz to be achieved over the audio band. In the advanced audio instruments, resolutions lower than 1 Hz are required. To meet these specifications, an analog solution would require extremely precise and stable components, or online trimming and tuning, which increases the cost of the system. These requirements can be achieved with mixed analog/ digital IC's and several functions which were typically implemented with analog circuits can be implemented using digital techniques.

The novel signal generator based on the direct digital synthesis (DDS) technique, which fulfills the requirements of advanced audiometric systems is discussed. Two most common problems of the audiometer are discussed and solutions are provided. The generation of a finely tunable pure sine wave and the generation of noise signals with controlled spectrum. In order to achieve tuning capabilities at 20 kHz with 1 Hz or at 100 Hz with 15 pHz, a fractional division of a 40 MHz master clock with noise shaping of the resulting error is performed.

The performance of a digital solution is determined by very stable clock signals available and the method of white noise generators [4]. Analog methods of white noise generation are typically based on high value resistors, which are very sensitive to interferences and require on-line calibration of temperature effects. The pseudo-random sequence generators (PRSG) gives perfectly white and repeatable spectrum with very simple hardware. Following table shows the comparison between switch capacitor based method and DDS based method of signal generation.

Table 3 Comparison of different signal generation methods

Parameter	Switched capacitor filter method	DDS method
Frequency range	250 Hz – 8000 Hz	100 Hz – 20000 Hz
Warble tone	±10% with one sweep in two seconds	- 10%
Masking noise band	10000 Hz	16000 Hz
Amplitude modulation	±5 dB with one step in one second	Implimentable

*Switched capacitor filter method [1].
** DDS method [3].

6.1 Pure Tone Generator

The principle of the proposed DDS system is illustrated in Fig.6. It consists of a programmable divider, a digital counter, a look-up table containing the digitized codes of a sine wave period (Sine ROM) and a digital to analog converter (DAC). The programmable division factor (PDF) is provided to the system by an external input device, such as a microprocessor.

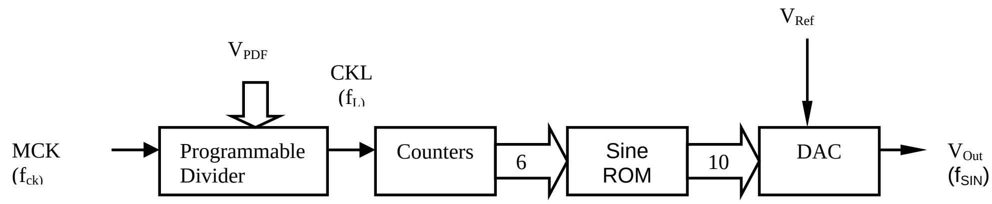

Fig.6 Basic Principle of direct digital synthesis of sinusoidal signal. Adapted from [3].

The detailed block diagram of the programmable divider is shown in Fig.7. The master clock signal MCK (f_{ck} = 40 MHz) is accumulated by an n_{PDI} bit synchronous counter. The resulting signal is compared during each master clock period with an n_{PDI} bit reference word. When the two words become equal, the output signal of the divider (CKL) is set to one and the synchronous counter is reset. The clock signal obtained (CKL) drives the input of a 6-bit counter, whose output word represents the address for the 6 x 10 bit look-up table containing a sampled sine wave period (n_{per} = 64 samples/period). The digital output code of the look-up table is finally converted into the analog domain by a 10 bit digital to analog converter (DAC)**.** The frequency of signal CKL (f_L) obtained at the output of the programmable divider is given by

$$f_L = f_{ck} / PDF$$

The output sine wave frequency (f_{SIN}) expression is as follows

$$f_{SIN} = f_L / (n_{per} \times M)$$

where n_{per} denotes the number of sampled points of the sine wave period and M is an additional division factor used to generate the clock signal for the switched-capacitor reconstruction filters required after the DAC [3]. With the assumption PDF to be an integer, the frequency resolution of the sinusoidal signal is limited by the given equation

$$\Delta f = \frac{f_{ck}}{M \times n_{per} \times (PDF\,(PDF+1))}$$

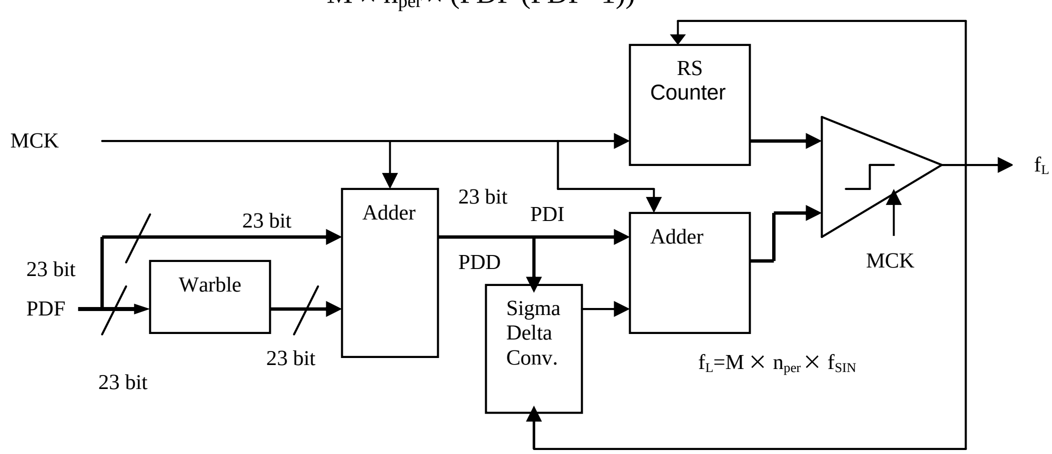

Fig.7 Block Diagram of the programmable divider. Adapted from [3].

The master clock frequency is already high; it is not possible to fulfill the system specifications of frequency resolution by further increasing the time discretization. This problem is solved by using non-integer division of the master clock period. The digital word programmable division factor (PDF) actually consists of two coefficients: integer part of the division factor (PDI) and fractional part of the division factor (PDD). The word length of PDI (n_{PDI}) is determined by the minimum sine wave frequency (100 Hz). The equation of PDI is given as

$$n_{PDI} = \ln_2 (f_{ck} / n_{per} \times M \times f_{SIN})$$

with the value of f_{SIN} = 100 Hz we have got n_{PDI} = 10. The word length of the fractional part PDD (n_{PDD}) is determined by the required resolution at the maximum sine wave frequency (20 KHz) is computed as follows

$$n_{PDD} = \ln_2 (n_{per} \times M \times (f_{SIN}-1) / f_{ck})$$

with the value of f_{SIN} = 20 KHz we have got n_{PDD} = 13. Therefore the word length of the PDF is 23 bits.

It is tough task to achieve fractional division of the master clock period while keeping spurious signals produced by distortion and phase noise below -60 dB. To meet the requirement over sampling and noise shaping techniques are used. The fractional component of PDF (PDD) is applied at the input of a first order digital sigma-delta (ΣΔ) modulator, as shown in Fig.7.

The obtained bit stream is added to PDI. The actual division factor is, therefore, alternatively PDI or PDI +1, according to the output of the sigma-delta modulator (0 or 1). The effect of the resulting frequency modulation of the clock signal CKL is illustrated in Fig.8 .

The average frequency of the obtained sine wave, considering a suitable number of master clock periods (at least $2^{13)}$, is very close to the desired value.

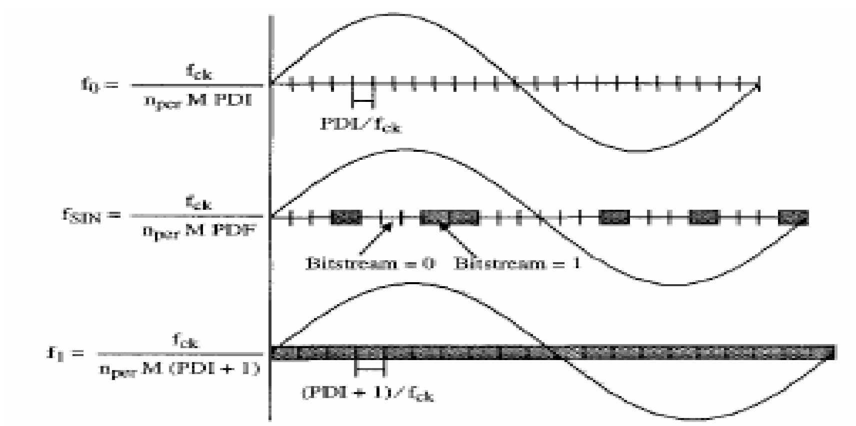

Fig.8 Sigma-delta modulation effect. Adapted from [3].

In audiometric applications, when hearing tests are performed in open spaces (without earphones), it is necessary to periodically modulate the frequency of the generated sine wave (from f_{SIN} to 0.9 f_{SIN} and back in 200 ms with 100 steps), in order to avoid stationary waves. This function, called as warble can be easily implemented with the proposed DDS technique. The detailed schematic diagram of the warble circuit is shown in Fig.9. It consists of a shift register and accumulator. Shift register determines the step amplitude according to the following equation and digital accumulator, which increment each time step (1 msec) the value added to PDF giving rise to the frequency shift.

$$\Delta PDF_{WB}=(f_{ck}/(n_{per}\times M))\times 10^{-3}/(f_{SIN}\times(1-10^{-3}))$$

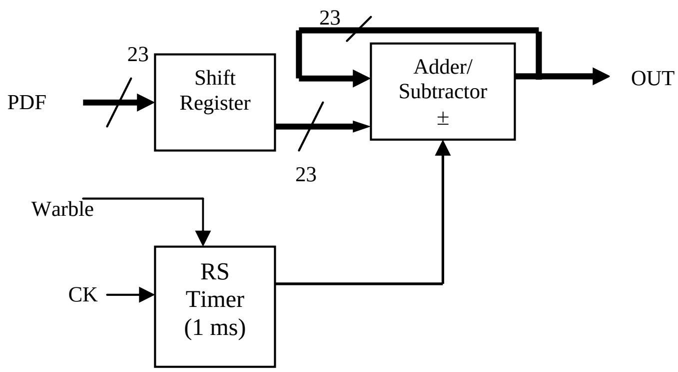

Fig.9 Schematic Diagram of the warble circuit. Adapted from [3].

6.2 White Noise Generator

In the audiometric tests noise sources as audio signals required. Several noise signals, such as white noise, pink noise, narrow-band noise or speech noise are used as masking sounds, it can be derived by properly amplifying and filtering a good quality white noise source. Therefore it is required to design a primary white noise source.

Analog instruments are typically based on high value resistors operating as white noise sources. To avoid the drawbacks associated with high-value resistors, digital synthesis technique is proposed. This has been accomplished by means of pseudo-random sequence generator (PRSG).

For effective audiometric tests a minimum sequence repetition period of 5 s is required. If a clock frequency of 32 kHz, the random sequence has to be at least 1.6×10^5 samples long, which corresponds to 18 bits. Such a maximal-length sequence can be generated implementing the primitive polynomial ($\varnothing(x)=1\oplus x^7\oplus x^{18}$).

The 18-bit output word (N) is then digitally processed in order to reduce the total length to 10 bits, as required by the DAC. A simple truncation of the output word produces a low-pass filtering effect, which degrades the quality of the white spectrum. To overcome this problem a compensating high-pass filtering is implemented. The output word (OUT) is

$$OUT = N\,[9:0]+N\,[18:10](1-z^{-1})$$

A prototype implementation of the proposed techniques was integrated in a 0.8 µm double-poly double-metal CMOS technology. The signal generators (pure tone and white noise) as well as a microprocessor interface were designed [3] using the Verilog HDL description language and synthesized using Synergy.

7. Conclusion

In this paper various human ear problems and their diagnosis techniques are discussed. Because of revolution in electronic science all the methods and algorithms can be embedded in one chip with the help of hardware descriptive language and this chip can be a part of add on card, which may be used in any high performance medical instruments [3]. The retrofitting of all available biomedical instruments to make them compatible to such new universal add on cards needs to be defined by specifc standards of industry in terms of size, protocols, and power supply requirements need to be debated and defined for universal application of such systems.

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