DIGITAL HEARING AID
A REVIEW

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AN INDEPENDENT PROJECT SUBMITTED AS PART FULFILMINT FOR FIRST YEAR MASTER OF [SPEECH AND HEARtNG] TO THE UNIVERSITY OF MYSORE

ALL INDIA INSTITUTE OF SPEECH AND HEARING
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TO

MY GRAND PARENTS

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CERTIFICATE

This is to certify that the Independent Project entitled: Digital Hearing Aids:A Review is the bonafide work in part fulfilment for the degree of Master of Science (Speech and Hearing of the student with Register No.M8709.

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This is to certify that the Independent Project entitled: Digital Hearing Aid - A Review has been done under my supervision and guidance.

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DECLARATION

This Independent Project entitled Digital Hearing Aid - A Review is the result of my own study under the guidance of Dr.(Miss) S.Nikam Prof, and Head Department of Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier at any University for any other Diploma or Degree.

Mysore

May 1980 Register Mo.M8709

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INTRODUCTION

Changes are inevitable with time, as in the case of living being it is called evolution. This is true of evergrowing modern technology, where one sees Changes interms of designing, constructions and function - characteristics. These advances have resulted in greater economy. Hearing aids are no exception to this phenomenon.

A persual of the historical development of hearing aids reveals that we have came a long way, from the mechanical coupler like ear cup to sophisticated electronic hearing aids.

Pre electric Hearing Aids:

The actual discovery and development of the means of enhancing the reception of sounds are lost in prehistory, Perhaps a rolled up leaf, a broken sea shell or a hollowed - out animal horn were among the first aids used by hunders, warriors and sailors to called and direct sound energy. The slight gains provided by such devices probably added no more than a 10 to 15dB increase in power for low frequency sounds, but this amplification prabably helped some hearing impaired individuals. By 1800's a great variety of mechanical hearing aids were available. In 1880's scientific advancements created the possibility of harnessing electrical enexgy for use in amplifying elec-trical discoveries to the problems of sending and receiving speech.

The fixst patent oa the principie of electrical amplification frx the hearing impaired was issued 1880, a few yeax after the invention of the telephone. It was not until 1908, however, that a commercial hearing aid was built on the patented principle.

Carbon Hearing Aids:

Carbon hearing aids did not have amplification units until the 1920s. The amount of gain was appxoxi-mately same as for some pre electric aids. In the 1920s the amplifiers into the carbon hearing aid system ampli fication gains rose to 45 to 50dB. A volume control was also incorporated into carbon hearing aids. Carbon hearing aids were susceptable to damage from dampness. The quality or fidelity of voice reproduction was only fair because of the limited sensitivity of the carbon microphone. The carbon hearing aid era began to draw a close in the early 1980s. When the table vaccum-tube made its debut.

Vacuum tube Hearing aid:

Vacuum tubes were the outgrowth of experiments carried out by Thomas A. Edison in 1880s. In the 1920s a number of large vacuum tube table-model hearing aids were developed. In 1930s with the development of the pantode vaccum tube which utilized a more complex grid arrangements, it become possible to design a portable vacuum tabe hearing aid which had an almost unlimited capacity to amplify. The major drawbacks, however, was that a large power supply was needed. With increase in amplification afforded by the vacuum tube hearing aid came the need for a refinement of the potentio-meter register to control volume in the hearing aid. So pic&-clipping and automatic volume control was introduced. AVC was not included ia the circuitory of wearable hearing aids until 1940s.

The transistor Era:

After World War II, development at Bell telephone Labo-ratories brought a new Era ia electronics - the Era of transister. The first hearing aids to incorporate transistors appeared in later 1952. They were much smaller, lighter and more economical to operate because they needed only a single, small voltage battery as a power source. With in a few year the miniaturization of hearing aid components and circuitory was such that hearing aids could be fitted into the temples of eyeglass or small cases to be worn behind the ear.

Hearing aids are now quite sophisticated. The amplifier circuit is usually composed of three to four stages of power which include the transistors and their coupling circuits. A simplified diagram of stages of a simple amplifier is shown in Fig.(1).

Integrated circuit:

Integrated circuit was developed in 1965 which contains thousands of smaller components circuit. This has brought to revolution in hearing aid design by making the size of the hearing aid very compact and small. This has made possible high gain, behind-the-ear and all-in–the ear hearing aids practical.

Signal processing hearing aid:

Electronic signal processing in the hearing aids has improved the amplification of speech signals considerably. Signal processing is the manipulation of signal to enhance or extract the information that it contains.

BLOCK DIAGRAM OF AN ANALOG HEARING IN5TRUMENT

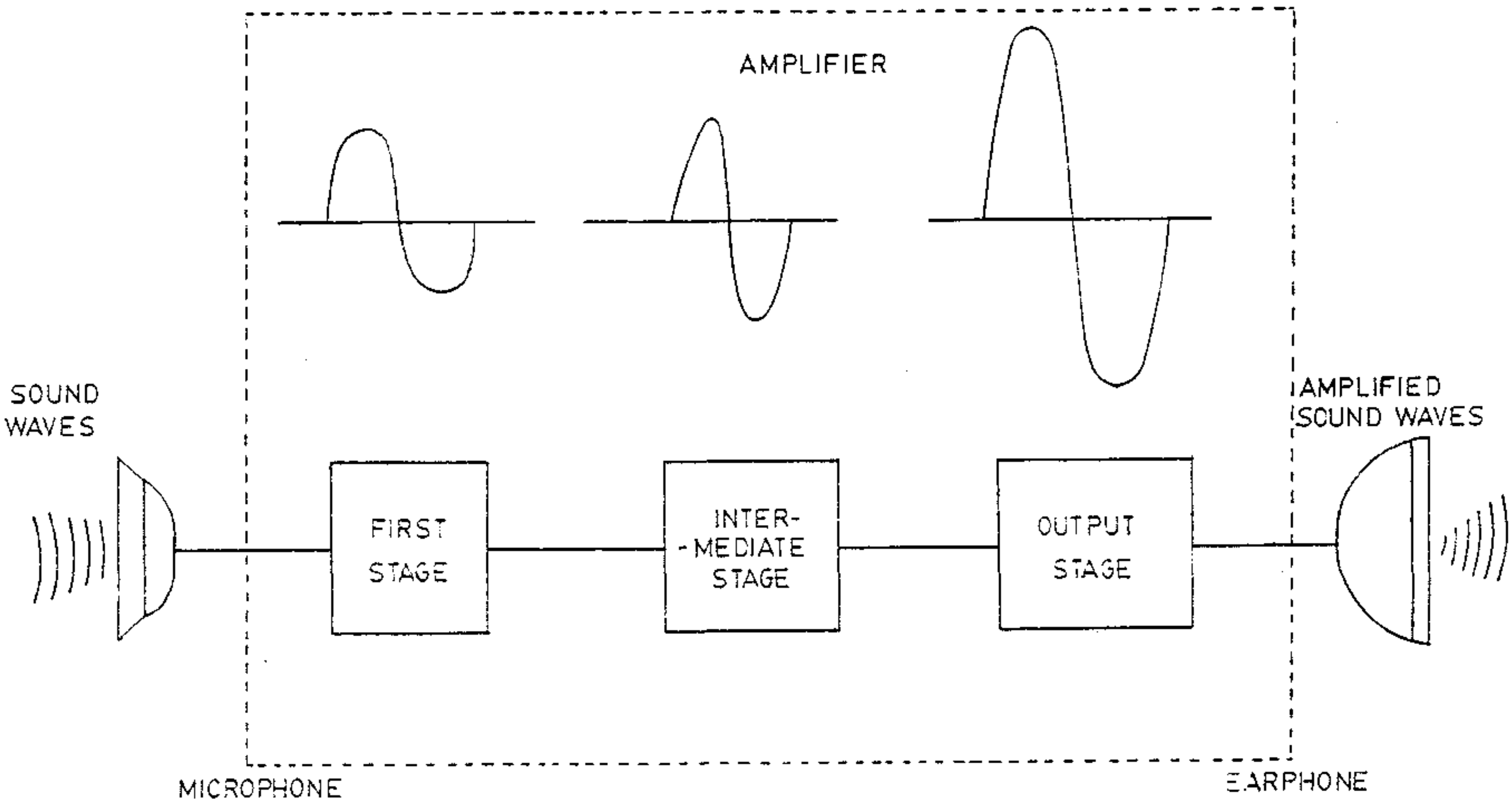


FIGURE (1)

This improves the intelligibility of speech in the noisy environment. Two types of signal processings are available.

1. Analog signal processing
2. Digital signal processing.

Analog signal processing or automatic signal Processing(ASP):

These began to appear in the mid-late 1970s. Quite a few behind-the-ear and all-in-the-ear hearing aids also use ASP circuitory and their performance have shown positive result.

Digital signal processing (DSP) hearing aids:

Before 1980 digital signal processing was mainly a theoretical term. In 1980 Graupe, Brek and Cousy reported on a prototype hybrid (partly analog and partly digital) electronic circuit for the purpose of reducing background noise.

Multichannel compression in a programmable hearing aid by a desk type computer was attempted by Mangold and Leijon in Sweden in 1981.

However, it was not until 1983, that the first digital hearing aid was designed and reported by Nunlay and stabb.

Lavitt and Morley in 1984 incorporated some of the basic technological advances in the digital signal processing used for hearing rehabilitation.

Digital hearing aids permit the dispensor to customize the performance of the hearing aid to meet the individual needs in the various hearing situations that he/she is exposed to. Ideally a digital hearing aid will have flexibility to adopt its performance in sort of the adverse listening environments.

this project was undertaken to give an overview of the digital hearing aids to the people who concerned with rehabilitation of the hard of hearing individuals.

The various areas touched upon by this review are:

1. Digital technology
2. Digital signal processing
3. Use of digital signal processing in hearing aid industry
4. Digital hearing aids and its advantages and disadvantages
5. Present status in hearing aid industry
6. Future perspective of hearing aid technology.

DIGITAL TECHNOLOGY

Digital electronics began in 1946 with the advent of electronic digital computeR. The concept can be traced to Charles Barbbage who developed a mechanical digital computer ia the 1830s but never completed a workable model. The first computer was built around 1944 by Harvard University Professor. It was essentially an electromechanical device, act electronic.

The term 'digital' refers to data in the form of an discrete units and to devices operating such data.

Digital techniques is a broad term. It refers to the process of converting analog data to digital numerical form by processing the numbers and reconverting them to analog form again. It also refers to the processing of coded data only. Eadie (1974) "digital techniques are methods of applying integers to the quantities that occur in a system and the process of instrumenting such systems with devices that base their operations on manipulation of integers".

Digital techniques, are applied only to that portion of the system Where the data in the form of integer. At the output end of the process after the reconversion to an analog form, the data are not in the form of integers, at these point they are analog. Only intermediate form the system is digital. Digital processing assumed that the analog signal can be adequately represented by a finite act of numbers (sample) and that ia possible to convert analog quantities into digital and vice versa. Processing of informations axe moat important here.

Digital technique are largely based oa digital signal processing (DSP) which will be discussed briefly.

Digital signal processing (DSP):

Digital signal processing ia the most general sense, means the creation, changing and/or detection of signals using digital implementation. This signal processing implies the 'real time' processing of a digitied analog signal to change that signal in some way.

DSP of analog signal is a sampling technique that eliminates the need for conventional analog components (transistors, precision resistors, capacitors and diodes) by implementing in software. What is normally implemented through the hardware devices are filters, limiters, oscillators, modulators and demodulators.

The basic DSP system is shown in Fig.(2). At the first stage analog signal is digitized in 'real time' by the analog to digital converter (ADC). In the second stage

DIGITAL SIGNAL PROCESSING (DSP)

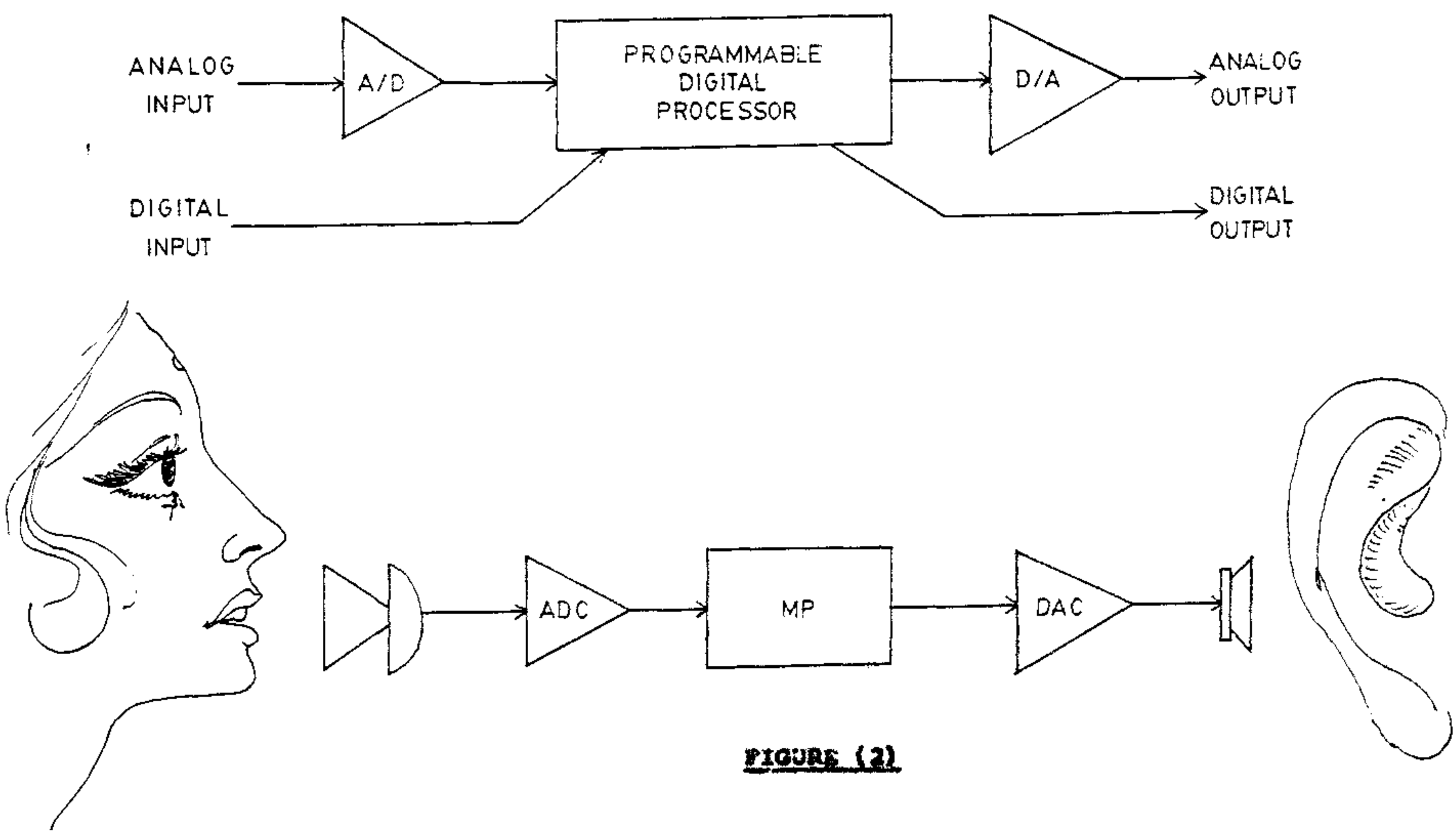


FIGURE (2)

the resulting digital stream is processed mathematically by a high speed processor as per the continually cycling instruments provided o the processor. In the third stage the resulting output digital or into analog from through the digital to analog converter. The main processing unit are baaed on following systems.

1. **Signal sampling:**

According to the signal sampling theorem (Shannan,1949) the information from ananalog system can be retained in a set of samples if taken regularly at a rate greater than twice the highest frequency contained in the signal. If there is any doubt about this frequency, then a generous sampling rate should be allowed or the signal is filtered to ensure that the upper frequency limit is known.

2. **Digital representating of analog quantitias:**

The sampled electrical analog signal amplitudes are represented in digital form. All the input is analyzed in terms of '0's and 'l's . These are called binary numbers Binary numbers are quite different from desimals although one can be written in the form of the other. This is in the form of positive (+) or negative (–) electrical voltages processed through semiconductor devices. The binary number used is most likely 8-bit, 12-bit of 16-bit, The number of bits determines the resolution of the digitizing process. Analog to digital converter helps to do this.

3. Logic circuit:

Digital signals are processed and manipulated by logic circuits. 'Combinational' logical circuit or 'gates' have several inputs and usually one output. The output is 'true' or '1' only when a specified pattern (i.e. combination) of 1's and o's are applied to the input.

4. Sequential circuits:

This refers to series of items which are placed in a definite order. It is a stable one and called as 'Bistable circuit'. Bistable circuit can store a set ('1' or 'o') until changed by set or reset signals. It is the basis for memorizing bit patterns. A variety of types can be fabricated by appropriate combination of logic circuits.

5. Analog-to-Digital (A/D) and Digital to Analog conversion:

a. **Analog-to-digital conversion:**

A/C conversion is the process by which analog quantity is converted into a digital form. This conversion is necessary when measured quantities must be in digital form for processing in a computer or for display or storage. ADC basically utilizes parallel differential comparators that compares the reference voltage with analog input voltage. The most popular A/D conversion method uses 10 bits or more which gives fast conversion in the successive approximation method.

b. **Digital to analog conversion:**

DAC is an important interface process for input operation in many applications. An example is where a voice signal has been digitized for processing or transmission and wust be changed back into an approximation of the original signal to ultimately drive a speaker.

It basically contains a resistor network with values that represent the binary weights of input bits of the digital code.

D/A performance characteristics.

a) Resolution: The resolution of a DAC is the reciprocal of the number of discrete steps in the D/A output. It is dependent en the number of input bits.

b) Accuracy: Accuracy involves comparison of the actual output of a DAC with expected output. It is expressed as a percentage of full scale or Maximum output voltage. The accuracy should be approximately $\pm$ .2 percent.

c) Linear error: A linear error is the deviation from the ideal straight line output of a DAC. A special ease is an affect error which is the amount of output voltage when the input bits are all zero.

d) Monotonicity: A DAC is monotonic if it does not miss a step or take any reverse steps when it is sequenced over it, through the entire range of input bit.

e) Settling time: This is normally defined aa the time taken by a DAC to settle within 1½ LSB of its final value when a change occurs in input code.
LSB : Least significant bit.

6. Sampling and hold techniques:

Whan sampling rapidly changing voltage, it is necessary to take samples in a very short time. It is used to avoid appropriate change of voltage during sampling period. It ia easier to use a fast sampling circuit which holds the sampled voltage adequately for conversion, by a alow converter. The circuit contains a switch and an amplifier. The hold time depends upon the size of the capacitor and the leakage of charge both through the switch and the amplifier.

7. Digital memory:

The team memory is generally used to describe the internal store of a computer, i.e, the immediate access memory. In its strictest sense, it refers to the storage locations that can be immediately addressed by programme controller of a central processor, rather then any backing store medium such as magnetic tape, magnetic disc etc. However, these backing stores are sometimes referred to as disc-file memory, in which the internal storage would be referred to as main memory.

a) Random access memory (RAM); The most versatile digital store for large numbers of bits ia the read/write random access memory, RAM in which each bit is held ia a separate memory 'cell'. The location of a given cell is called its address. The arrangement of cells is called a memory array and there are many ways in which the cells can be arranged or organized.

b) Read only memory(ROM): It contains permanently or semipermanently stored binary data which can be read from memory. It cannot be changed without a relatively complicated involved process.

A ROM differs from a RAM in that the ROM has no normal write cycle. ROM address caa be selected ia any gives sequence just as in a RAM. ROMs are used to store data that are used ever and over again in systems. Its applications are look up tables, code conversions, programmed instructions aad sequence for system and operation.

There are several categories of programmable read only memories.

1. **Programmable Read Only Memory:(PROM):** These are nonerasable memories that eaa be programmed by the user. But once programmed they can not be changed. These processes of programming involve storing a specified pattern of 1's and 0's in memory array, so that address corresponds to a given bit pattern.

2. **Erasable Programmable Read Only Memory (EPROM):** Unlike an ordinary PROM, an EPROM can be reprogrammed by first erasing an existing program ia the memory array. There are two basic types of SPROM (a) the ultraviolet light erasable PROM (b) electrically erasable PROM.

8. **Digital filters:**

Digital filters are used with sampled data. They may be incorporated in an analog filter system to replace the analog filters, where A/D and D/C converters are required. Alternatively they may operate in a purely digital environment as in a computer.

The cut-off frequency of digital filters are restricted to submultiples of the sampling rate. The filter can be divided into two groups depending upon their function.

1. The time domain
2. The frequency domain.

A Pearler filter is a frequency domain digital filter. Digital filtering eaa be accomplished directly in the frequency domain by using fast Founder transform {Cooley and Tokey 1965). This gives the spectrum of a signal in terms of phase and magnitude. The fast Fourier algorithm operates by eliminating all redundant multiplication in a direct digital calculation of the Pourier aeries of a set of discrete signal samples, it returns sine and cosine coefficients for all hocmonic components upto the Nyquest frequency (1 cycle per samples of the signal) for which the magnitude and phase values eaa be calculated.

Filtering can be achived by removing unwanted frequencies by setting their amplitude to zero. Inverse transformation to the time domain produces the filtered signal. However, abrupt filtering often yields undesirable transients in the waveform. Using digital filters there can be a vary good representation of speech signal. In the sense the output obtained by summing appropriately modulated band pass channels can be made indistinguishable from input (Schafer and Rabiner, 1971). Digital filters are easy to implement an a computer and have the advantages of precision and stability over analog filters (schafer and Rabiner, 1973).

9. Central processing unit:

The central processor is the nerve center of any digital computer system. Coordinates and controls the activities of all the other units and performs all the arithmetic and logical processes to be applied to data. All program instructions to be executed must be held within the central processor and all data to be processed wast first be loaded into this unit. It is (convenient to consider the CPU as three separate hardware sections:

1. Internal memory
2. Arithmetic Unit
3. Central section.

Internal memory: It consists of a series of magnetic storage devices (eg. magnetic memory). It is organized to hold data or program instructions in a series of locations as either words or characters, in computing systems the internal memory ia stored with in the cabinet housing.

Arithmetic unit: It consists of a series of special resistors and circuits which are able to perform arithmetic and logical operations upon one or mare operants selected from memory.

For example, the arithmetic operations of addition, substraction, multiplication and division may be performed, as well as various shift operations and comparision including logical operation can be done.

The Central Section: It has many complex function to perform.

The program central unit will select an instruction from memory and will store it with in one of a number of special central resistors. Here the instruction is decoded to ascertain the particular logical operation to perform and to ascertain the memory locations involved. Then the instruction is executed, causing the required operands to be selected from the specified memory locations and to be routed back again te the same or some other specified memory locations. The cycle is rapeated for each instruction in the program. Call aa instruction, decode and execute the instruction, call all instruction and so on.

Advantages of digital signal processing.

1. Flexibility
2. Repeatability
3. Freedom from parmanent drift.
4. Quick prototyping capability.
5. Lower implementation cost.
6. Better phase response filters.
7. Timesharing of digital filter is also possible.

DIGITAL SIGNAL PROCESSING IN HEARING AID TECHNOLOGY

The application of DSP has revolutioned the hearing aid technology. For many years manufactures of hearing aids have tried to improve the performance of the hearing aid for each individual user. One method would be digital signal processing because it allowes individual adjustment of hearing aid characteristics by using a competent and appropriate software.

Digital signal processing hearing aids are different from other hearing aids because:

1. It provides true digitized input signal
2. These hearing aids are programmable
3. They are designed to "make decisions".

The use of digital computer in hearing aids are:

1. Use of digital circultary to control analog circuit.
2. use of digital signal processing to replace analog circuitary to accomplish standard hearing instrument functions.
3. Use of digital techniques to produce new kinds of signal processing, such as noise suppression.
4. Use of digital coumputer ia hearing ingtrument fitting.

DIGITAL HEARING AIDS

Digital hearing aids are the future direction in the hearing aid design.

A true digital hearing aids is a 'wearable computer' that will allow for software adjustment of a hearing aids parameters. 'Wearable' and 'Computer' are the key words (Stabb, 1985).

Basically a digital hearing aids is a controlled microproeessor. The use af a microprocessor alone, however is not an indication that the device is a digital hearing aid. From outside it does not look any different from other hearing aid. The essential difference lie in the interior circuit design of the aict and its inactions.

The electroacoustic characteristics of a digital hearing aids are well reproduciable and do not change with time. This is because its characteristic are dete—mined by & stored programme. This programme is independent of manufacturing tolerance. The program is also not subjected to gain but remains same throughout the entire life of the hearing aid.

Analog signal processing vs. Digital signal processing:

To understand the nature of a digital hearing aid, it is important to differentiate betwenn analog signal processing and digital signal processing.

Analog information refers to either data in the form of continuously changing physical quantities to to devices that operate oa them (sound waves), at every instant, the signal takes a value from aa infinite amount of possible values. An analog signal is aa AC or DC voltage that varies smoothly or continuously. It does net change abruptly or in steps.

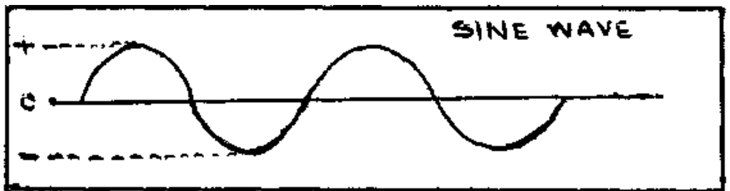


Fig. One of the most common analog signals, the sine wave.

Digital information refers to data in the form of descrete units and to devices operation on such data. Digital signals are essentially a series of pulses. It rapidly changes voltages levels that vary in discrete steps or increments between two fixed levels.

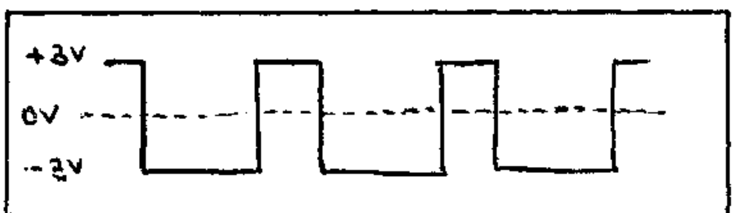


Fig. A digital signal in which the signal alternates between +3 and
This steplike characteristic is fundamental to digital signals.

A digital signal alters between +3 and -3 volts. This step like characteristics in fundamental to digital signals.

Digital signal processing in the hearing aids is becoming popular because of its several advantages over analog signal processing hearing aids.

There are 3 main advantages.

1. Extremely sophisticated signal processing function with better clarity of the decoding message.
2. Paster processing.
3. Reliable transmission of the signal even in a noisy channel using suitable coding in a digital form.

Speech processing function:

a. Typical analog hearing instrument corrupt the speech signal with hearing aid hiss and other distortion. This problem does net arise in digital hearing aids. To achieve this however, sufficient bits must be used in the digital representation.

b. Waveform coding: Here, the speech wave is assigned digital values and analyzed. This is possible by using the auto-correlation function in digital signal processing. Various methods can be used to incorporate this waveform coding, viz. the Deltamodulation.

This helps in identifying the voiced and voiceless sounds in the speech signal. Thus this sophisticated processing, waveform coding is mot a feature of Analog signal processor.

c. Compression: DSP performs the process of compression also for speech encoding. In an ASP, compression is done using amplifier, registers sad capasitera. If the time constraints are not maintained well, (eg. If released-time is long) a part of the signal may be lost and may not be amplified enough. This problem is not seen in a digital hearing aid. Even the high intensity vowels will not cause a higher gain reduction in the system because this is amplified by using two channel compression and quantization.

d. Speech in terms of time domain: Digital signal proessing represents the speech signals in terms of time domain, it is simple and useful for estimating the important feature of speech signal. Its fluctuations determines whether a particular waveform corresponding to speech or not or a selection of the signal is voiced, voiceless or noise. Thus digital hearing aid discards the irrelevant information and places the desired features clearly.

The digital hearing aid actually performs two functions: 1) analog
2) Digital

A block diagram of the digital hearing aid haa been ahown in Pig. 3.

The difference parta of a digital hearing aid are explained below:

1. sound pressure which impinge on the diaphragm of the microphone ia changed into an electric analog voltage.

2. The electrical analog signal is low pass filtered to limit the bandwidth of the signal to frequencies less than half the sampling rate. This is to avoid a condition called 'aliasing*. Filtering is required to pass only those frequencies which are equal to or less than half the sampling rate, such frequencies only can be reconstructed properly from a digital representation. Therefore, if a signal of a particular bandwidth is desired, the sampling rate and filtering requirements are essentially determined. The signal ia smoothed and ia limited to approximately 10KHz.

3. The filtered electrical analog signal ia sampled every seconds (i.e., 40 millionth of a second)

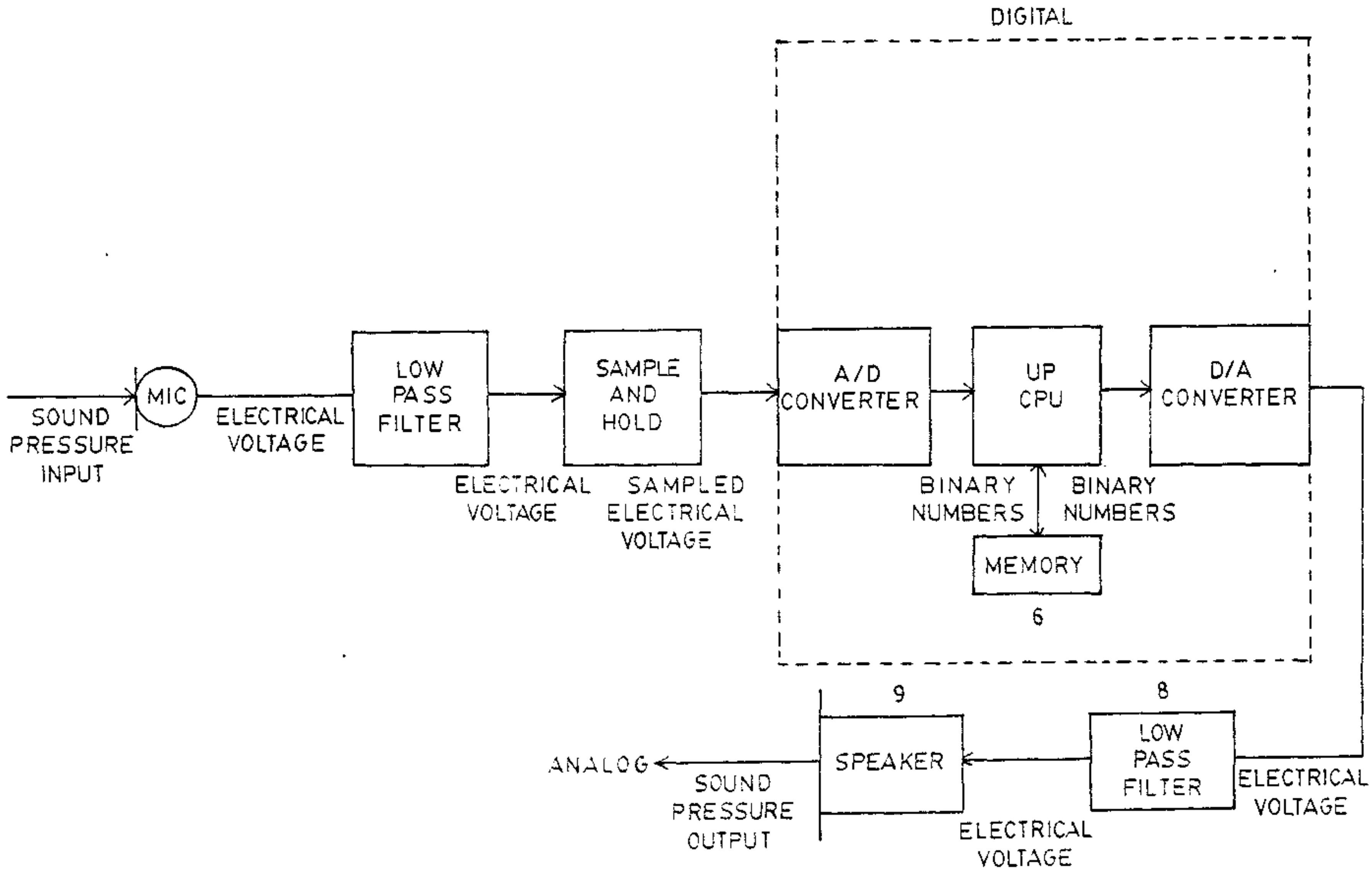
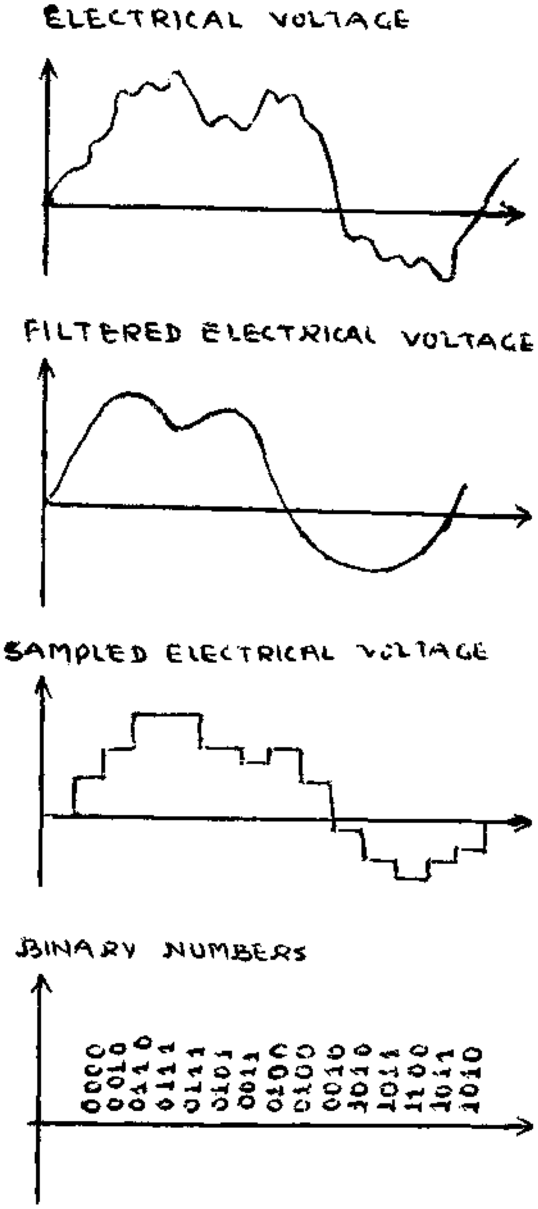


FIGURE (3) BLOCK DIAGRAM OF DIGITAL HEARING AID and 'held' so that the signal no longer contains an infinite amount of different values only 25,000 per second or one every 40 millionth of a second).

There are two important considerations of relevance in -sampling, both of which present deterioration of the signal when sampled. The first of those is called the 'Nyquist law(Shannan, 1949) which states that a sampled signal can be reconverted fully to the original signal by law pass filtering if the late is at least twice as high as the highest .frequency contained in the original analog signal for example, 25KHz for the filtering rate 10KHz as the low paas filtering.



Steps in signal change from analog to digital to analog.

The second consideration pertains to the quantizition noise which is due to step like input-output characteristics of the A/D converter. This noise can be kept to low level by choosing a sufficiently large number of bits.

4. The sampled electrical analog signal ia converted to digital information using the analog to digital converter. This is in the form of positive(+) or negative (-) electrical voltages processed through semiconductor. The binary number used ia most likely on 8-bit, 12-bit, 16-bit. The number of bits determined the resolution of the digitization process. It is critical to the issue of signal to noise (s/N) ratio of the digitized signal.

5. The central processing unit (CPU) or microprocessor is a single silicon chip computer. The entire electronics of a small computer are engraved onto this chip which can be programmed in many ways. The program consists of a series of instructions that is stored and executed in sequence to carry out some specific function. The instructions cause the computer to mathematically manipulate the data in some way. This is the 'heart'of the digital hearing aid. The digital representation caa be potentially operated in many ways. This infact is one of the difficulty task i.e, instructive the chip computer on how to respond.

Some of the ways that the digital information can be operated on includes

A. The adding or subtracting of information.

B. Weighted versions of past and present input

C. Emphasize or de-emphassize various aspects of the signal.

This is where the aid is 'personalized'. The CPU irresponsible for the automatic operation of the digital computer.

6. A memory retains the program, a sequence of operations and the intermediate results.

7. In the D/A converter, the computed digital output (binary number) is converted back into an anolog signal.

8. A low pass filter smoothes the output of the digital section for accurate reproduction of intelligible speech signals.

9. Electrical analog impulses are converted into output sound pressure by tne nearing aid receiver (speaker).

The advantages of digital hearing aids:

1. Extract speech from noise by identifying speech temporal characteristics.
2. Eliminates acoustic feedback.
3. Rejects unwanted harmonica (distortion control)

4. Reduce reverberation
5. Compensate for reduced frequency resolution
6. Throughly smooth the frequency response.
7. Effectively shape the response to any configuration needed.
8. Transpose one frequency band into a higher or lower frequency band.
9. Expands and/or compress signals without inducing the many distortion products inherent ia analog system.
10. A digital hearing aid has the potential to be programmed and reprogrammed an infinite number of times.
11. Earmold acoustics can be programmed into the hearing aid rather than relying on experiments.
12. The possibility exists for the elimination of switches and moving parts.
13. The digital aid would be much more stable than the present analog units. The reason is that the hearing aid characteristics are determined by the stored program. The characteristics would be stable and rapeatable, with no parameter drift.
14. The digital aid of future may even be able to 'improve' the original signal, since it may not only have to amplify and reproduce what is presented to the input.

Disadvantages of digital hearing aids:

The main disadvantages of digital hearing aids is seen in terms of its.

- performance
- coat
- size.

a) **Performance:** since digital hearing aids uses C-MOS technology, the power consumption is greater. This is because C-MOS device cannot function at low voltages and hence bipolar circuit is introduced at preamplifier and output stage. Hence, this extra in take of power supply. Naturally, overall it turns out to be less econnomcal and hence performance is hindered.

b) Cost: The coat of any digital hearing aid dependa upon its circuitary, components used and kind of energy consumption digital hearing aid exceeds in all these three when compared to other hearing aids which do not use digital technology.

c) Size Maintaining a small size, while increasing desirable features, has become a major draback in the development of electronic circuits for hearing aids.

Digital hearing aid faces similar problem. Due to the C-MOS technology, the size of digital hearing aid is more masking the device more cumhersome to handle, Especially when compared to in-the-ear devices which uses miniature circuits. Digital technology has to revolutionize itself to stay in competition with other hearing aids by reducing its size.

PRESENT STATUS

Considering the present state of affairs, the signal processing techNology has added a new dimension to the hearing aid industry. The application of signal processing is an engineering approach for improving the signal-to-noise ratio which inturn will benefit the hearing aid wearers to hear and understand in the presence of background noise.

Patients with sensorineural hearing loss with high-frequency involvement, recruitment, raised threshold and consequent loss of dynamic range may receive greater benefits from using a signal processing hearing aid than from a conventional hearing aid.

Signal processing:

Signal processing (SP) is the manipulation of the signal to enhance or extract its information content (Ketes, 1985). Burnved (1985) defines signal processing as "any device that acoustically, electrically or electronically will alter a signal in frequency, intensity or time."

Signal processing can be broken down into two areas

1. Analog signal processing.
2. Digital signal processing.

Analog signal processing:

Analog signal procesaiag has primarily been, employed in hearing instruments to improve the S/N ratio. It may be of two types.

1. Non—adaptive signalprocessing
2. adaptive signal processing.

1. <u>The former includes:</u>

1. Directional microphone hearing instruments,
2. Binaural hearing instruments
3. Earmold coupling modification
4. Limiting systems
5. Trimmers (linear processing)
6. Damping.
7. Horn effects.
8. Passive filtering and some active filtering.
9. Automatic gain control systems.

The performance of aon-adaptive signal processing do not change once the control is set.

2. <u>The later includes:</u>

1. Adaptive signal processor
2. Adaptive compressors.
3. Adaptive noise compressor
4. Adaptive filtering
5. Some active filter system and non-linear processing.

Adaptive signal processing is a processing function which changes the performance of the hearing instrument when it is used ia changing input signal environments.

The hearing instruments circuit adapts and changes its own parameters automatically. The parameters are mostly frequency or intensity.

One of the classic example of adaptive signal processing circuit is multiple signal processing (MSP) which has been shwon in fig.(4). It is reported by Stabb and Nunley (1987).

They used an elaborate integrated citcuits which create a movable filter band which reduces low frequency background noise and simultaneously enhances the high frequency perfoxmance of the hearing aid through added gain.

The MSP improved the s/N ratio in competing noise, regardless of whether the noise was steady or intermittent.

The MSP aid has fitting capability that allows its use for individuals with hearing loss ranging from mild to severe degree, for individual with high frequency hearing loss and for gradually sloping loss and for those with flat loss.

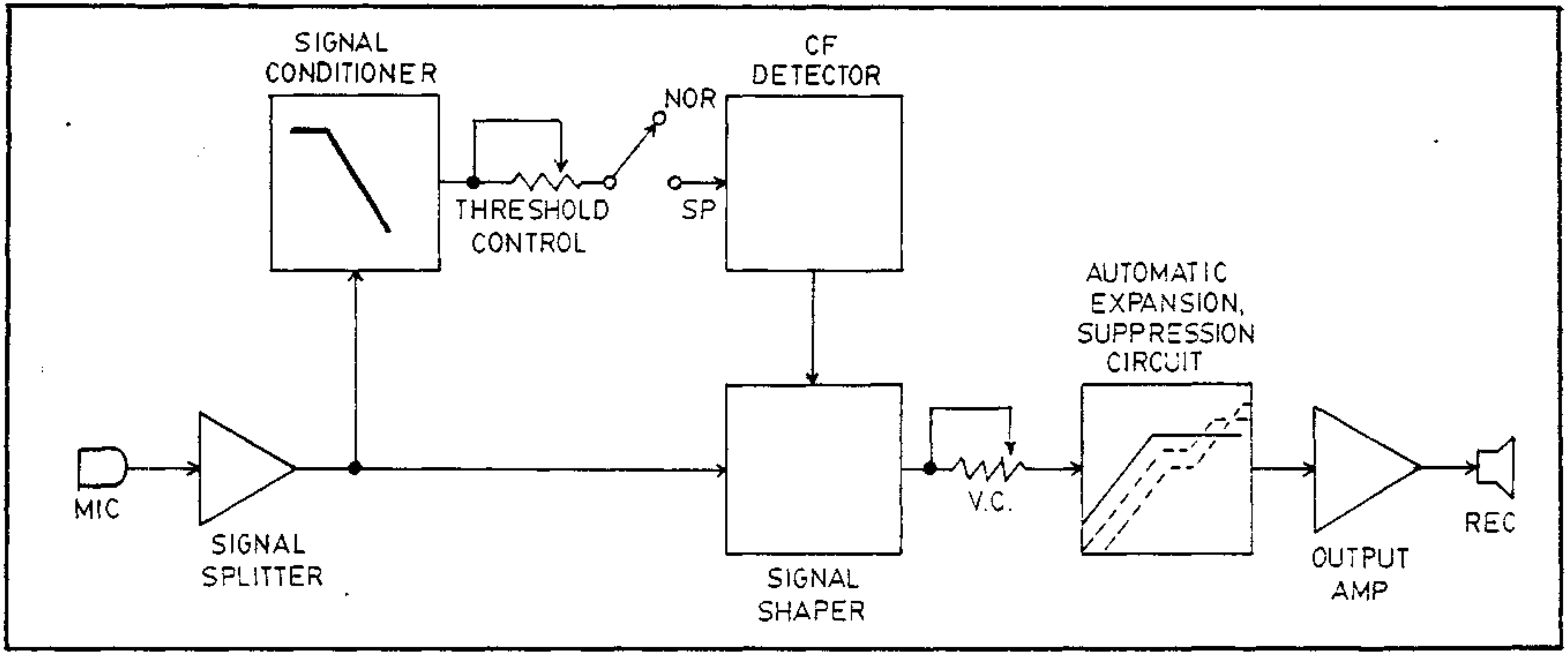


FIGURE-(4) BLOCK DIAGRAM OF MULTIPLE SIGNAL PROCESSOR.

Digital signal procession hearing aids:

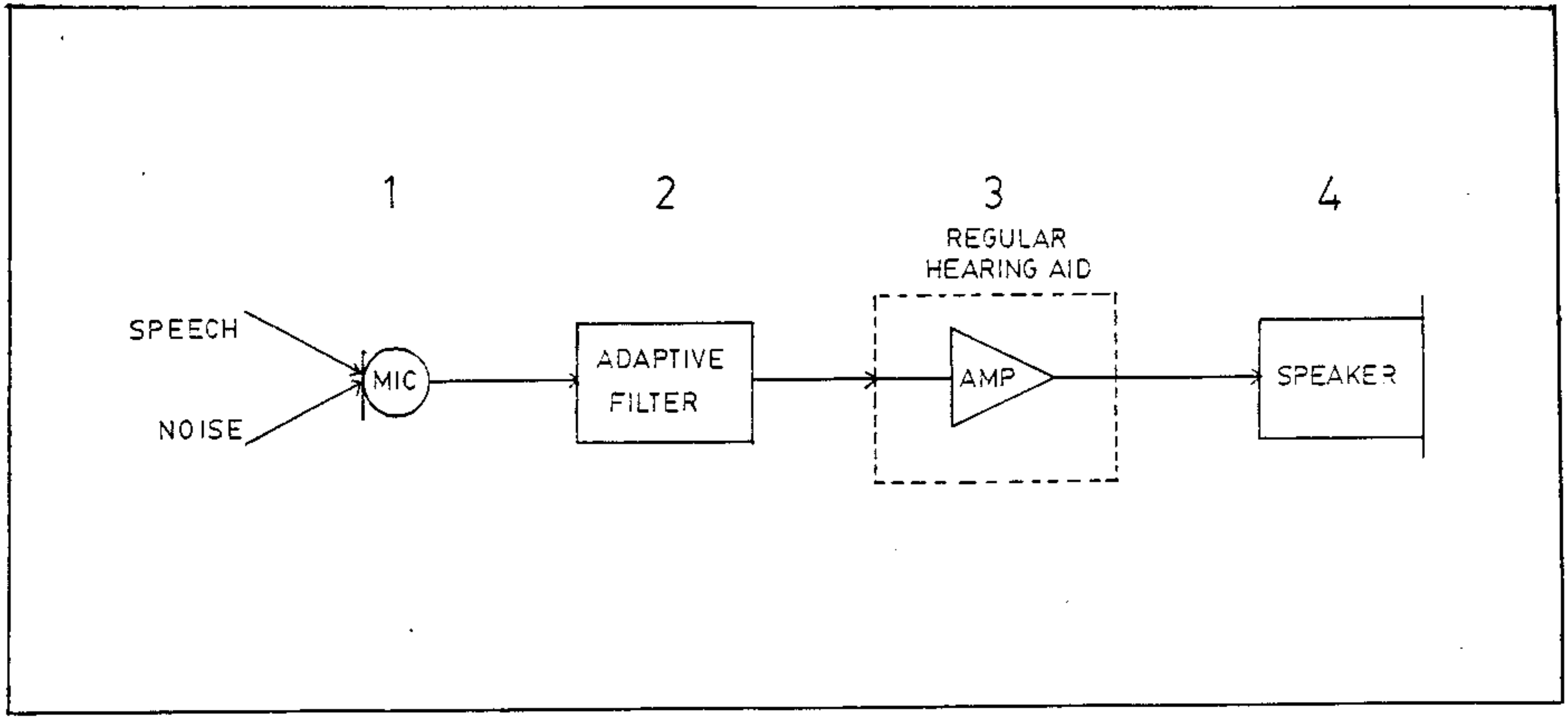
It is the other hand of interest because it holds the promise of extending the signal processing capability to assignificant extent and performance features of hearing instruments. Most specific and voiced signal control is possible. DSP approaches can be described best by referring to the method used for signal processing.

At present time there are three approaches to signal processing, they are discussed below:

1. Computer-based adaptive noise reduction circulatory:

This approach uses digital technology in the chip development to develop the filter bat remains basically an analog system. The signalis always in analog form and the circuitary is non-software controlled. This approach allows the use of low voltage and low current digital logic. It is primarily designed for adaptive noise suppression. The distinguishing feature is the addition of a special circuit added to a standard hearing instrument circuit has been shown in Fig.(5).

The most frequently identified example of this application is the Zeta Noise blockers which attempts to



<u>FIGURE (5)</u>: BLOCK DIAGRAM OF COMPUTER-BASED ADAPTIVE NOISE REDUCTION CIRCUITRY ADDED TO A STANDARD HEARING INSTRUMENT CIRCUIT.

separate speech from steady-state noise in several frequency bonds. This is accomplished by determining how much the energy in each frequency bond changes ever time, If many changes are detected, the circuitary considers the signal to be speech and does not suppress the signal in that frequency bond. On the otherhand, if little change in energy occurs, the chip assumes that the predominant signal in that frequency band is noise. The chip automatically reduces the gain in that frequency band without affecting the gain in the other band(s). The design employes basic algorithms which direct the adaptive filter in its function. The future of these systems must be directed in learning how to improve the execution and to expand the digital portion of the analog-digital system.

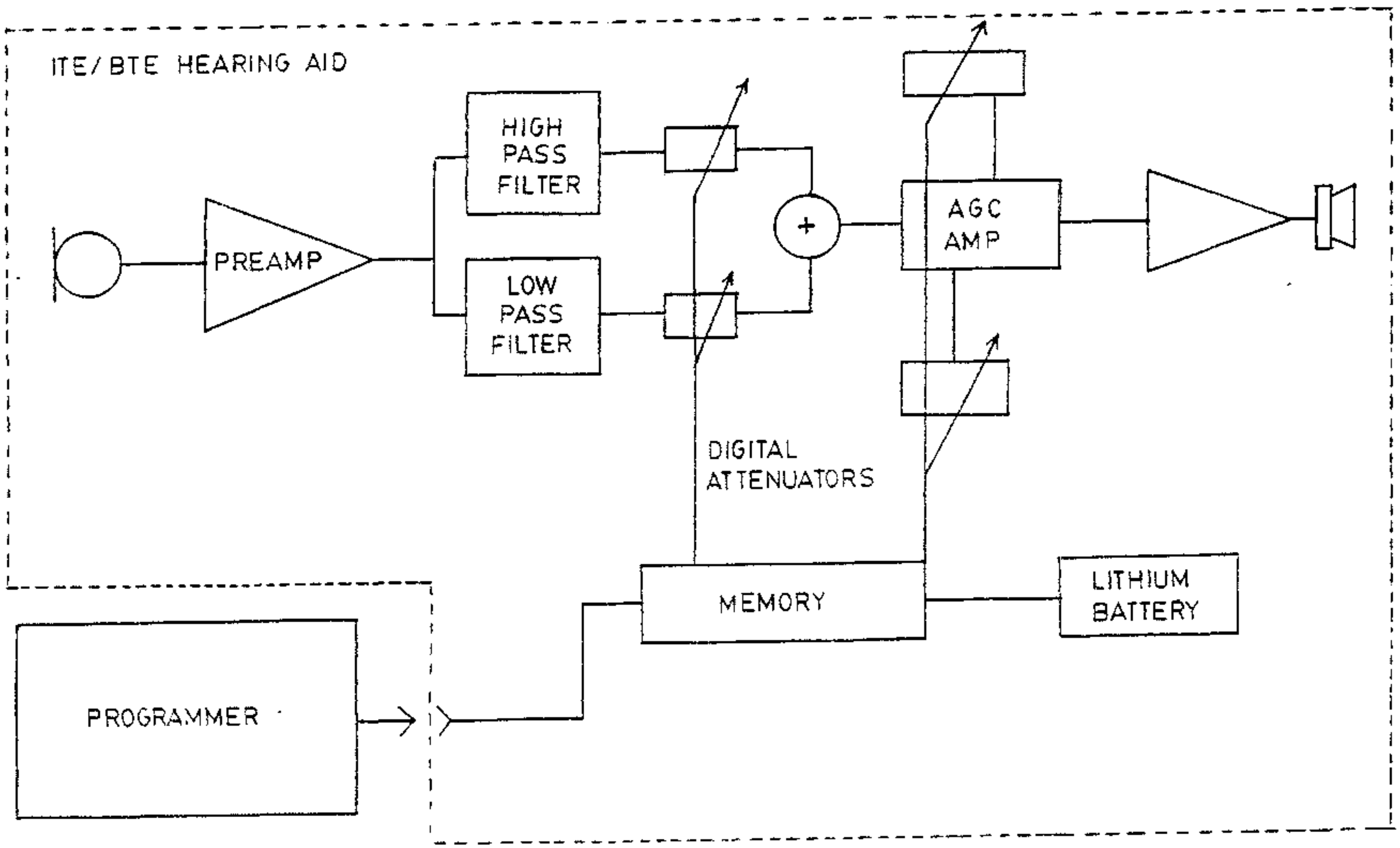
2. The programmable hearing instrument:

It is a practical approach to solving the size and power consumption problems. The programmable instrument utilizes a digital memory which takes the place of conventional trimmers in the hearing instrument. The system consists of two primary components.

a. the programmable hearing instrument with a memory chip.

b. the programmer which is used by the filter for circuit selection.

This has been shown in Fig.No.(6).



<u>FIGURE (6)</u>: BLOCK DIAGRAM OF A PROGRAMMABLE HEARING INSTRUMENT. THE SOLID SQUARE SHOWS THE PROGRAMMING COMPUTER, AND THE AREA WITHIN THE DASHED LINE IS ALL PART OF THE HEARING INSTRUMENT.

This system is the first to allow for the 'in situ' circuit selection. In this particular case, the saturation output, gain, low frequency response and high frequency response can be selected for the wearer while the instrument is being worn. Additionally, variables such as canal size canal resonance, insertion loss and subjective toae preference can be adjusted for while the person wearing his/her owa hearing instrument and through his/her own earmold. Such an experimental instrument was shown by Dahlberg in 1985.

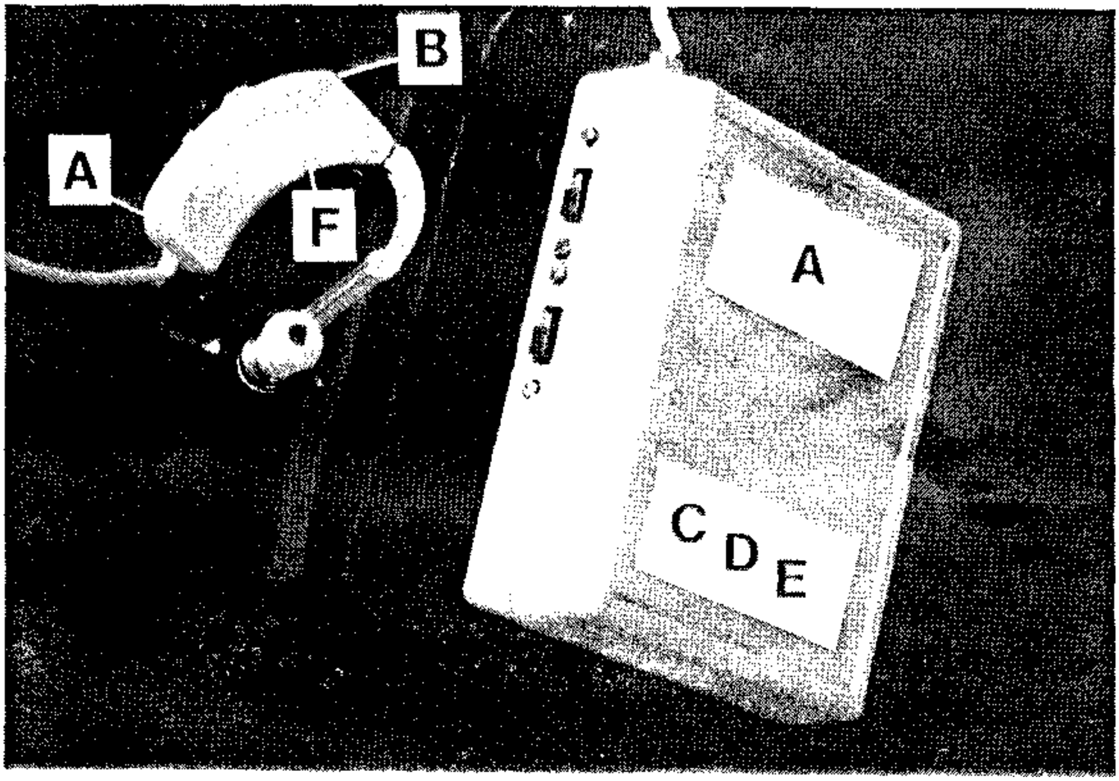
The experimental hearing instrument contained a CMOS ((Complimentary metal oxide semiconductor) memory module which kept the program in memory even when no battery is present or the instrument is dead. This is accomplished by means of a built in witnin cell backup. The programmer is a microprocessor containing as EFRCM and can access each of the memory modals locations in the hearing instrument to set its performance parameters. The software ia versatile enough to program a variety of hearing instrument models. An advantage of this type of system is that the hearing instrument caa be programmed as often as desired. This type of system would be desirable in the following instances.

1. For hearing instruments where trimers are hard to aid (I-T-E and canal hearing aid)
2. Where adjustment by a new wearer is difficult.
3. For children from whom specific hearing information is not available.
4. For patients with fluctuating losses.
5. Being reprogrammable, electroacoustic performance can be easily changed.
6. Patient with multiple handicaps.

At present four types of storage mechanisms such as RAM, PROM, EPROM, EEPHOM are available in the programmable hearing instruments.

3**. Digital processing system:**

These system uses digital circuitary. They are the true digital hearing aid. The first wearable system was reported by stabb (1983). It consisted of a body packed connected to a behind the ear instrument housing the microphone and receiver. The microprocessor and power supply was contained in the body pack. The primary function of this system is the elimination of noise that is being amplifier. A prototype digital hearing aid has been shown in Fig. No.(7).



PROTOTYPE OF FIRST DIGITAL HEARING AID EVER DEVELOPED
AS A WEARABLE UNIT: (A) POWER SUPPLY (B) MICROPHONE
(c) A/b CONVERTER (c) CPU (E) D/A CONVERTER (F) RECIVER.

FUTURE PRESPECTIVE

Hearing aid technology has come a long way through. It has with stood and undergone the basic research, product development and laboratory verification which in turn has helped it to reach the current state of affairs. Digital hearing aid is one of the recent typhons of changes noticed in our dynamic world. This will positively pave new pathways for better diagnostic instruments which will help in assessing the loss, determining the type of aid to be prescribed and programming a user based flexible hearing aid. In fact, experimentation with these aids may give new insights about our hearing mechanism and its functioning better than any other method being currently followed.

It is impossible to predict how soon these digital hearing aids will be available in the market.

However, infiltration of digital systems into the field of hearing instruments have practical advantages in terms of functioning, dispensing and availing (utilizing). This review has aspired at exemplifying these feature in a nutshell.

GLOSSARY

1. A/b - Analog to Digital.
2. Algorithm: A step by step process for the solution of a problem, usually developed in outline or as a flow Chart before coding (i.e. setting it in computer language).
3. Aliasing: The introduction of error into the Fourier analysis of a discrete sampling of continuous data when components with frequencies too great to be analyzed with the sampling interval being used contribute to the amplitudes of lower frequency components.
4. Arithmetic logic unit (ALU); The element in a computer that can perform the basic data manipulation in the central processor.
5. Analog: Refers to data in the form of continuously changing physical quantities – waves or devices that operate on it.
6. C-MOS: Complementary metal oxide semi-conducter.
7. Computer - A programmable electronic device that can store, retrieve and process data.
8. Central processing unit (CPU): The computer module in change of fetching decoding and executing instructions.
9. D/A:Digital to Analog.
10. Digital: Refers to data in the form of discrete unit on/off or high/low states-and to devices operating on such data.

11. Digital signal processing: In the most general sense, means the creation, changing and/or detection of signals using digital implementation.
12. Digitize: To put data into digital from for use in a digital computer.
13. Input/output(I/b): The communication of information to and form a computer or peripheral device.
14. Large scale integration: The simultaneous achievement of large area circuit chips and optimum density of components packaging for the express purpose of cost reduction by maximization of the number of system interconnections made at the chiplord.
15. Memory: The collecting and holding of pertinent information until it is needed by She computer.
16. Microprocessors LSI implementation of a complete pro cessor (ALU- central section) on a single chip.
17. Program: A sequence of instructions that results in the execution of an algorithm. Programs are essentially written at three levels (1) Binary (3) Assemble language (3) High level language requiring a compiler or interpreter.
18. Real time - Processing that operates on all previous data and results in aa output before the next input sample isrequired, so that there is little time delay.

19. Sequential access: The method in which data is accessed by scanning blocks or records sequentially.

20. simulator : A device or computer that performs simulation.

21. Software: The programs or routines and supporting documentation, which instruct the operations of a computer.

22. subroutine: A series of computer instructions to perform a specific task for many other routines. It is distinguishable from a 'main routine' in that it requires, as one of its parameters, a location specifying where to return to the main program after its function has been accomplished.

23. Time sharing: Occurs whea a single computer has multiple users who are each getting a 'slice' of each second of computer processing time.

24. VLSI: Very large scale integration. In practice, the compressing of more than 10,000 transistors on a single chip.

25. Zeta-Noise Plocker: This is also called adaptive filter. A filter which, on real time basis, identifies the spectrum and repetitiveness of back grand noise and adjusts its own frequency response several times per second accordingly to maximize speech intelligibility. Commercially available such adoptive filter is known as z-N-B.

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