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Design of Low Complexity Adjustable Filter Bank for Personalized Hearing Aid Solutions

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Abstract: The auditory system is a very sensitive and complex network. Diseases, drugs, noise, trauma and aging may have resulted in varying degrees of hearing loss, which makes hearing impairments one of the most common sensory disturbances in the world. The most effective way to compensate hearing loss is to employ a hearing aid system which is an integration of voice amplification, noise reduction, feedback suppression, automatic program switching, environmental adaptation, and etc. The basic function of a hearing aid system is to amplify sounds selectively and then transfer the processed signal to the ear. The processed signal which is transferred to the ear enables the deaf people to hear the voice of the person who is near to them.

Keywords: Filter bank, Audiometer, ATMEL89S52 microcontroller, BAHA, Watch dog timers.

1. INTRODUCTION

Hearing is a complex process, so it should be no surprise that the causes of hearing loss are also complex. Hearing loss can occur because of damage to the ear, especially the inner ear. For example, infants may be born with hearing loss caused by a viral infection that was acquired during pregnancy. Other times the cause is genetic and therefore due to changes in the genes involved in the hearing process. Sometimes, hearing loss is due to a combination of genetic and environmental factors.

There is, for example, a genetic change that makes some people more likely to develop hearing loss after taking certain antibiotic medications. Understanding the genetic causes of deafness has important benefits. This knowledge not only allows doctors to inform families about their chances of having children with hearing loss, but it can also influence the way a person's deafness is treated. Whether a person's hearing loss is going to worsen can sometimes be predicted if the specific cause is known. Also, deafness may be only one of a group of medical problems that a person may have. For example, some people with hearing loss also have problems that affect other parts of the body, such as the heart, kidneys, or eyes. Knowing the genetic cause in these cases allows a doctor to predict the appearance of these other problems. It might seem reasonable to suspect a genetic cause of deafness only if the hearing loss runs in the family. But there are situations in which children have genetic deafness even though neither one of their parents are affected. This deafness can also be passed on to future generations. Genetic tests can therefore be helpful even if there is only one person in the family with hearing loss. In this proposed system we are going implement Signal-Processing Strategy for Restoration of Cross-Channel Suppression in Hearing-Impaired Listeners. It is very useful for hearing disability person. Voice input given to mike after that we use some filters to compress the voice. Cleared voice stored in storage device of controller, after processing of controller give another section depends upon the channel selection. In decompress section retrieve the given input voice but it has less amplitude so we have to amplify the signal. After amplification we use transducer for converting sound signal into vibrating signal.

2. EXISTING METHOD

- In the existing system, the bone anchored hearing aid (BAHA) is a surgically implanted device designed to provide a hearing aid to patients.
- The majority of the conventional hearing aid transmits sound through the medium of air conduction. BAHA stimulate the cochlea by transmitting the sound waves through the bones in our skull or bone conduction there by passing the outer and middle ear.

Figure1. BAHA Operated Image

3. PROPOSED SYSTEM

- In this project Signal-Processing Strategy for Restoration of Cross Channel Suppression is implemented in Hearing-Impaired Listeners.
- It is very useful for hearing disability person. The voice is given as the input.

- Gamma tone filter unit is used to separate the useful information and by the compressor unit the voice is compressed for the clear signal which is get transferred to the WT voice IC where the data recording is possible.
- During the recording of data some amount of error signal will occur which is clear out by the decompression unit. Its efficiency is improved by amplification process.
- Then by using the transducer the sound signal is converted into the vibration signal. The vibration signal induces the auditory nerves and reaches the inner ear drum.

3.1 Advantages

- Reduces the size of the aid.
- Suitable for different aged persons.
- It undergoes acceptable delay.
- It offers flexibility.

In this proposed system, the voice is given as the input, then for the future process and to remove some amount of ripples, here we are using some filter unit to separate the useful information and by the compressor unit the voice is compressed for the clear signal which is get transferred to the voice IC here the data recording is possible during the recording of data some amount of error signal will occur which is clear out by the decompression unit. Then by using the transducer the sound signal is converted into the vibration signal. The voice input is given to the transformer which converts one form of physical quantity into another. Then it enters into the voltage regulator and microcontroller. Microcontroller provides the highly-flexible and cost-effective solution to many embedded control applications.

3.2 Block Diagram of Hearing Aid

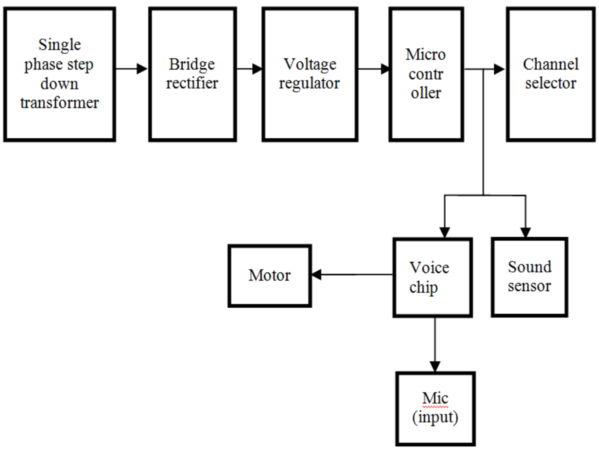

Figure2. Hearing Aid Block Diagram

• Transformer:
The potential transformer will step down the power supply voltage (0-230V) to (0-6V) level. Then the secondary of the potential transformer will be connected to the precision rectifier, which is constructed with the help of op–amp. The advantages of using precision rectifier are it will give peak voltage output as DC, rest of the circuits will give only RMS output.

• Voltage Regulator:
Voltage regulators comprise a class of widely used ICs. Regulator IC units contain the circuitry for reference source, comparator amplifier, control device, and overload protection all in a single IC. IC units provide regulation of either a fixed positive voltage, a fixed negative voltage, or an adjustable set voltage. The regulators can be selected for operation with load currents from hundreds of milli amperes to tens of amperes, corresponding to power ratings from milli watts to tens of watts.

• ATMEL89S52 Microcontroller:
The AT89S52 is a low-power, high-performance CMOS 8-bit microcontroller with bytes of in-system programmable Flash memory. The device is manufactured using Atmel's high-density nonvolatile memory technology and is compatible with the industry-standard 80C51 instruction set and pin out. The on-chip Flash allows the program memory to be reprogrammed in-system or by a conventional non volatile memory programmer. By combining a versatile 8-bit CPU with in-system programmable Flash on a monolithic chip, the Atmel AT89S52 is a powerful microcontroller which provides a highly-flexible and cost-effective solution to many embedded control applications.

Figure3. ATMEL89S52 Microcontroller

• Sensor/Transducer:
The word "Transducer" is the collective term used for both Sensors which can be used to sense a wide range of different energy forms such as movement, electrical signals, radiant energy, thermal or magnetic energy etc., and Actuators which can be used to switch voltages or currents. There are many different types of Sensors and Transducers, both analog and digital and input and output available to choose from. The type of input or output transducer being used, really depends upon the type of signal or process being "Sensed" or "Controlled" but we can define a sensor and transducers as devices that converts one physical quantity into another. Devices which perform an "Input" function are commonly called Sensors because they "sense" a physical change in some characteristic that changes in response to some excitation, for example heat or force and covert that into an electrical signal. Devices which perform an "Output" function are generally called Actuators and are used to control some external device, for example movement or sound.

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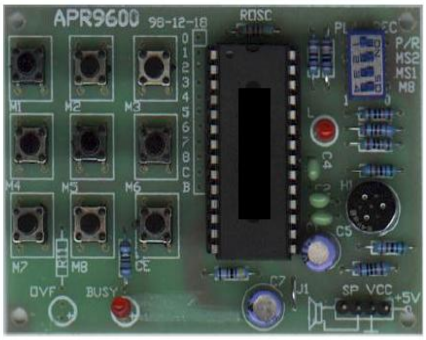
Electrical Transducers are used to convert energy of one kind into energy of another kind, so for example, a microphone (input device) converts sound waves into electrical signals for the amplifier to amplify (a process), and a loudspeaker (output device) converts these electrical signals back into sound waves.

Figure4. Sensor

- **Voice Board:**

The APR9600 device offers true single-chip voice recording, non-volatile storage and playback capability for 40 to 60 seconds. The device supports both random and sequential access of multiple messages. Sample rates are user-selectable, allowing designers to customize their design for unique quality and storage time needs. Integrated output amplifier, microphone amplifier, and AGC circuits greatly simplify system design. The device is ideal for use in portable voice recorders, toys, and many other consumer and industrial applications. APLUS integrated achieves these high levels of storage capability by using its proprietary analog/multilevel storage technology implemented in an advanced Flash non-volatile memory process, where each memory cell can store 256 voltage levels. This technology enables the APR9600 device to reproduce voice signals in their natural form. It eliminates the need for encoding and compression, which often introduce distortion.



APR9600 Experimental board

Figure5. APR9600 Voice Board

In any electric motor, operation is based on simple electromagnetism. A current-carrying conductor generates a magnetic field when this is then placed in an external magnetic field, it will experience a force proportional to the current in the conductor, and to the strength of the external magnetic field. As you are well aware of from playing with magnets as a kid, opposite (North and South) polarities attract, while like polarities (North and North, South and South) repel. The internal configuration of a DC motor is designed to harness the magnetic interaction between a current-carrying conductor and an external magnetic field to generate rotational motion.

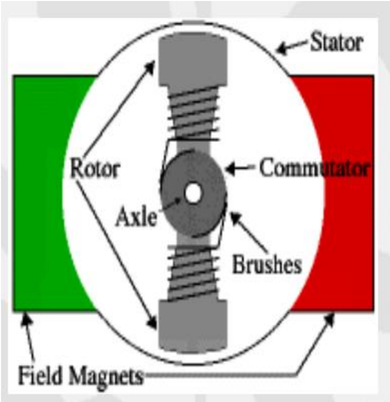


Figure6. Shunt motor

Every DC motor has six basic parts -- axle, rotor (a.k.a., armature), stator, commutator, field magnet(s), and brushes. In most common DC motors (and all that Beamers will see), the external magnetic field is produced by high-strength permanent magnets. The stator is the stationary part of the motor; this includes the motor casing, as well as two or more permanent magnet pole pieces. The rotor (together with the axle and attached commutator) rotates with respect to the stator. The rotor consists of windings (generally on a core), the windings being electrically connected to the commutator. The above diagram shows a common motor layout -- with the rotor inside the stator (field) magnets. The geometry of the brushes, commutator contacts, and rotor windings are such that when power is applied, the polarities of the energized winding and the stator magnet(s) are misaligned, and the rotor will rotate until it is almost aligned with the stator's field magnets. As the rotor reaches alignment, the brushes move to the next commutator contacts, and energize the next winding.

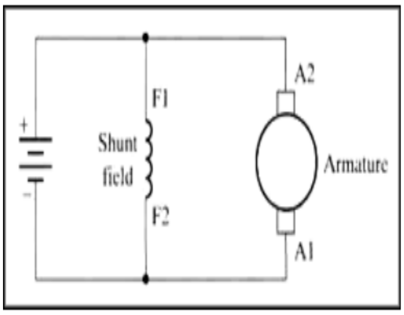


Figure7. Circuit Diagram of Shunt Motor

A shunt-wound motor is a direct-current motor in which the field windings and the armature may be connected in parallel across a constant-voltage supply. In adjustable speed applications, the field is connected across a constant-voltage supply and the armature is connected across an independent adjustable-voltage supply. Permanent magnet motors have similar control.

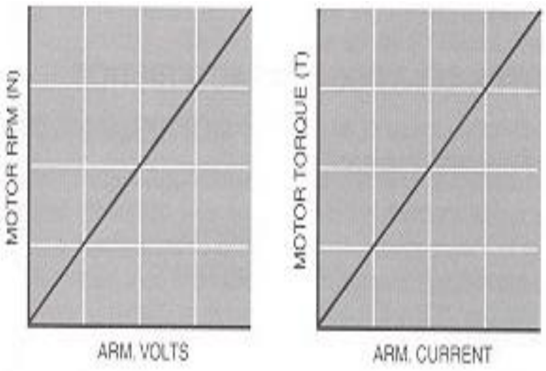


Figure8. Graphs for working of shunt motor

- **Speaker**

Speaker will control the electromagnetic wave into sound wave. It receives audio input from the device such as computer, audio receiver. This input may be either in analog or digital form. The sound produced by the speaker is defined by frequency and amplitude.

Figure9. Speaker

4. OUTPUT

The kit which involves the operation of the hearing aid is shown below. First the input voice is sensed by the sensor and it is then stored in the voice chip. This voice is processed and its efficiency is improved by the gamma tone filter which is in the voice IC.

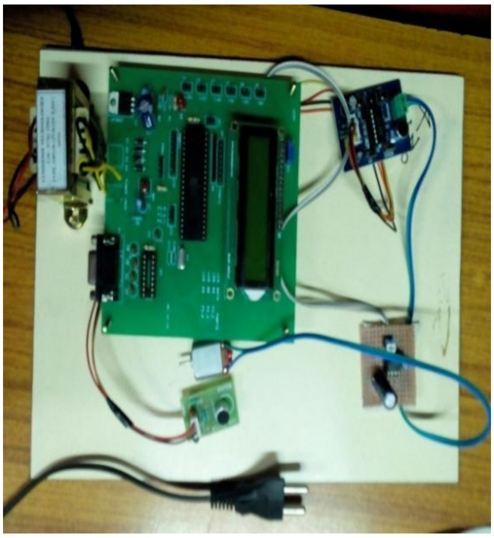
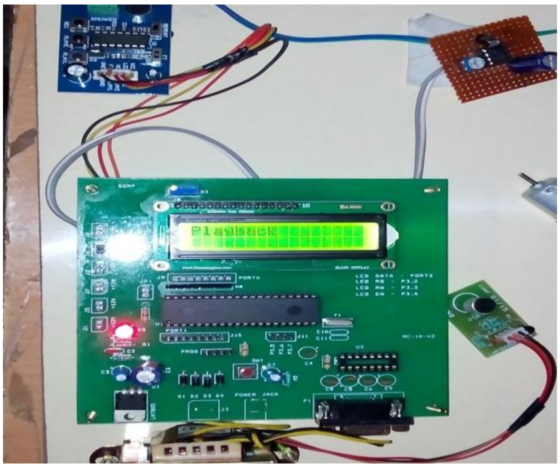


Figure5. Hearing Aid Kit

This processed voice then enters into the motor here the voice is converted into vibrations which can be audible by the person biting internal to their teeth. The supply required for this process is provided by the transformer which converts AC into DC (5V). This 5V supply is maintained constantly by the voltage regulator. The unwanted signals are removed by the filter which is placed next to the voltage regulator. This is overall working process of the hearing aid.

When the kit is recording the voice which is given as the input:

When the kit playbacks the voice which is recorded in the voice board and it is then heard by the deaf people:



5. CONCLUSION

A new strategy has been developed to reduce the complexity in filter bank for hearing aid solutions and to avoid the use of surgical operations. From these words the guilty of the deaf people while wearing this hearing aid is almost reduced. The complexity in the existing system is reduced to about 2.65 to 2.5 by using this method. It also reduces the irritation while hearing and to overcome the sound quality. This strategy is mainly used to reduce the complexity of the existing system by using filters. Hearing impairment has focused on very severe impairments, deafness and the hearing impairments of young people,

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rather than the moderate or partial hearing impairments that are common among elderly people. Most research on treatment for hearing impairment has focused on medical and surgical treatments rather than rehabilitative approaches.

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