MICROPHONE WITH ELECTRO-MECHANIZED GAIN CONTROL

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Abstract—*Peak amplitudes of the input signals to the microphone generally affect the life and performance of the speakers and might also hurt our ears. Advertently, if the microphone receives weak sound waves it cannot convert into audio signals which means it fails to perform the function of a transducer. The solution is to control these input signals within the microphone itself. This can be done by designing few compact devices and integrating them into one single circuit which will be connected to the transducer and packed inside the microphone. Doing this will also help in attenuating whistles and howls caused by feedback loops.*

Keywords—Transducer; Compressor; Pre-amplifier

1. INTRODUCTION

A microphone is a transducer that changes vibrating air patterns caused by sound to audio signals. An audio signal is represented by electric signal patterns and these are asymmetric waves. A variety of mechanical techniques can be used in building microphones. Two different types of microphones namely: Dynamic and Condenser are used today in a large scale. Dynamic microphones have a magnetic design and condenser microphones have a variable capacitance design.

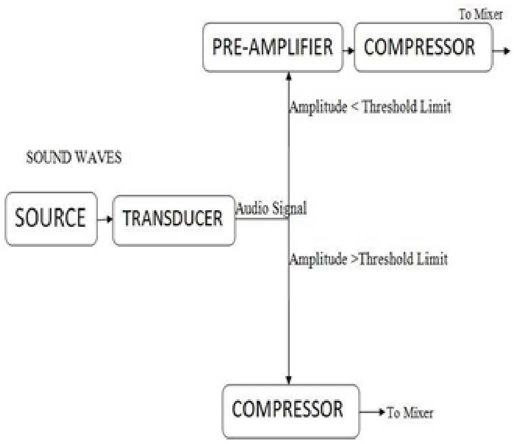
In the magneto-dynamic microphone, a thin metal diaphragm and a wire coil attached to it vibrates (to-fro movement) due to the incoming sound waves. A magnetic field round the coil is thus caused which moves the scroll within this area. This in- turn causes current to flow. The velocity of the apparent movement of diaphragm determines the sum of current produced. This is a velocity sensitive microphone.

Microphones produce a very small amount of current, hence to be useful for recording or other electronic processes, the signal must be amplified using a preamplifier contained in a mixer. Since the signal as a whole also contains interruptions such as electrical interference which is brought forth by the microphone, the signal along with the electrical noise will also be amplified, so even little amounts of noise is unbearable. Dynamic microphones are usually noise free, but the condenser type microphones have an electronic circuit built into it is a potential source of trouble, and hence careful designing and construction of premium partsare required.

In general, a dynamic microphone does not need a preamplifier on the input side, only the output signal from the microphone is to be expanded by the amplifier in the PA system. Since this project is using a dynamic microphone, it does not require a preamplifier on the input side of the microphone. The unwanted pickup of mechanical vibrations through the body of the microphone is also a source of noise.

The most usual origins of noise in microphones originates in the wire linking up the microphone to the console or tape deck and unwanted pickup of mechanical vibrations through the torso of the microphone. A microphone preamplifier is very similar to a radio receiver, so the cable must be prevented from becoming an antenna. The basic technique is to surround the wires that carry the current to and from the microphone with a flexible metallic shield, which deflects most of the radio energy. The second technique, which is to balance the line is a more effectual method for low frequency hum induced by the electric current and power in our surroundings.

2. BLOCK DIAGRAM



A. WAVEFORMS AND SOURCE CHARACTERISTICS

Speech is characterized into two types:-

i) Periodic sound,
ii) Aperiodic sound

Periodic sound produces voiced speech with defined intervals and aperiodic sounds producing voiceless speech. Thus the two different signals need a different audio source. The vocal chords of a human being work by separating out the tract shapes of the periodic or aperiodic sound with the aid of various sides of the tongue, lips etc.

Aperiodic sounds, when examined closely are observed to be a non-repeating or random pattern. No part of this waveform pattern is repeated at regular intervals, therefore the sound is said to be aperiodic.

A voice with a high-sounding pitch has a high frequency. Even the most purely voiced human speech sounds, such as vowels, are not perfectly periodic. Three vowel sounds, on the other hand, are periodic. Firstly, the frequency is constantly changing, this results from a combination of prosody (esp. Intonation) and segmental phenomena. In voicing fricatives, we can see repeating (periodic) large peaks with non-repeating (aperiodic) fine detail superimposed on them. To distinguish between a periodic sound like /iː/ and a weakly mixed voiced fricative, one must be able to distinguish between repeating and non-repeating fine detail. The relationship between frequency and pitch is not linear and depends on the characteristics of the ear and the auditory nervous system. The frequency of a sound wave refers to the rate of the vibrations of the sound travelling through air. This rate parameter decides whether a sound is perceived as high pitched or low pitched. In sound, frequency is also defined as Pitch. The frequency of the vibrating sound source is calculated in cycles per second. The SI Unit for Frequency is Hertz and can be defined be as '1/T' where T is the time period of the sound wave. Time period is defined as the time required for the wave to complete one cycle. The mathematical relationship between Wavelength and frequency of a sound wave are related as:

$$\text{Velocity of Sound} = \text{Frequency} * \text{Wavelength}$$

No pitch periods are perfectly steady in nature. Generally there is some noise generated at the glottis during normal phonation and noise can also be generated at other places in the vocal tract. The occasional perturbation of the vocal fold vibration can result in creaks and other imperfections in the voice. Without such variations in the voice source, the speech would sound inhuman and machinelike. The token of these vowel source soundswas very similar, but the wave shapes are very dissimilar because the different tongue and lip positions results in different vowel qualities. The vowel /iː/ consists of two repeating major peaks with a less intense finer detail pattern superimposed on it, with the fine details repeating quite accurately from cycle to cycle. Some spectral information can be deduced directly from these waveforms, whilst similar vowels cannot be distinguished. For this reason we must recourse to spectral displays for explicit frequency-domain information.

B. Compressor

Compressors are most commonly used in almost all audio work, but it is one which is most misused audio processors. Since there is some type of mandatory dynamic range modification with the sound of records, telephones, TV, radios and public address systems, compressed audio has become an everyday fact of modern life. Specialized amplifiers such as compressors and limiters are utilized to reduce the dynamic range i.e., the span between the softest and loudest sounds. The approximate dynamic range of human voice is estimated to be 10 Decibels, but a plucked or percussion instrument guesstimates to 15 Decibels or more difference. Human ears are subjected to compressing, responding to roughly the average loudness of a sound. Good compressor design includes a detector circuit that imitates the human ear by responding to average signal levels. Even better compressor designs employ a second detector that is destined to respond to peak signal levels, which has a provision to adjust to clamp peaks that occur at a specific level above the average signal level.

Threshold is the determined level of the incoming signal at which the compressor amplifier changes from a unity gain amplifier into a compressor, reducing gain. When the signal falls below the threshold set level, the compressor is designed to be inactive. Once the signal reaches or crosses the threshold level, the compressor is turned on. The compressor starts reducing the gain according to the amount through which the signal exceeds the threshold and the ratio control setting. The threshold level setting can be estimated as the sensitivity of the compressor expressed in Decibels (dB). The instant at which the compressor starts the reduction of gain is termed as "Knee".

Compression process can be categorized into two types, namely: intentional compressions and unintentional compressions. In this project, the intentional compression method is implemented. In intentional compression, which is also known as automatic gain control or audio level compression, which are used in devices called 'dynamic range compressors', the overall gain of the circuit changes actively in response to the level of the input over time. Due to this activity, the transfer function remains linear over a short period of time. There will be no changes in the wave appearance, but the wave gain characteristics will vary. For instance, a sine wave into such a system will still appear to be a sine wave at the output, but depending on the level of that sine wave, the overall gain is varied. Above a certain input level, the output sine wave will possess the same amplitude.

C. Pre-Amplifier

The prime function of a preamplifier is to extract the signal from the detector without remarkably degrading the intrinsic signal to Noise ratio. Therefore, for such a heedful extraction the preamplifier is located as close as possible to the detector, and the input circuits are designed to perfectly match the characteristics of the detector. In this project however, The preamplifier will be located just after the

compressor since our main focus is to control high peaking signals. Different pulse processing techniques are implemented, depending on whether the arrival time or the amplitude (energy) of the detected event must be measured. Pulse shaping for either application is normally employed in a subsequent module. This module can be located at some distance from the preamplifier, subsequently cautious that the signal fidelity is not degraded due to the length of the interconnecting coaxial cable. Several types of detectors produce moderately large signals at their outputs, and this mitigates the restrictions on the noise contribution from the preamplifier. Detectors that are subjected under such categories are: photodiodes operating with intense light pulses, photomultiplier tubes (PMT), scintillation detectors (scintillator mounted on a PMT), micro channel plate PMTs, micro channel plates, channeltrons, and electron multipliers. For such detectors, a wideband amplifier with low input impedance can be used directly at the detector output to generate short, fast-rising pulses for timing or counting purposes.

In the time at which the pulse crosses the threshold of the timing discriminator, the noise added to the signal by the preamplifier causes an uncertainty or jitter. The result is a degradation of the time resolution. Therefore, it is important to choose a current-sensitive preamplifier whose rise time is similar to the rise time of the pulse at the detector output.

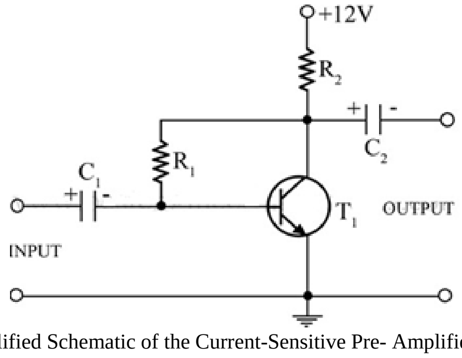
A preamplifier rise time that is much faster than the detector rise time does not improve the signal rise time, but it does contribute extra noise, because of the unnecessarily wide bandwidth. This excess noise will increase the timing jitter. Selecting a preamplifier rise time that is much slower than the detector rise time reduces the preamplifier noise contribution, but it is not enough to overcome the degradation in pulse rise time and amplitude.

Consequently, the timing jitter becomes worse. Although the optimum choice depends on the rise time and amplitude of the detector signal, as well as the characteristics of the preamplifier input stage, a good guideline is to choose a preamplifier rise time that is within a factor of 2 of the detector rise time (faster or slower). Most fast preamplifiers with gains in excess of 10 must employ a coupling between internal amplifying stages, to achieve fast rise times and to eliminate dc drift with respect to temperature. This is an excellent solution in the case where the average spacing between pulses is greater than 100 times the individual pulse width.

When the average spacing between pulses becomes comparable to the pulse width, the ac coupling causes the baseline between pulses to shift so that the preamplifier output signal circumscribes as much area above ground potential as it does below ground. This effect distorts the amplitude measurement in subsequent modules. Operating a photo multiplier tube with the cathode at high voltage with the anode being dc-coupled to the input of the Model 9305 Preamplifier, eliminates the baseline shift at high counting rates, and permits operation at much higher counting rates.

Most current-sensitive preamplifiers designed for timing applications have ac-coupled time constants in the range of a few hundred nanoseconds. Parasitic-Capacitance Preamplifiers Photomultiplier tubes, electron multipliers, micro channel plates, and micro channel plate. PMTs produce moderately large output signals with very fast rise times. Therefore, the most cost-effective preamplifier for pulse- amplitude measurements or energy spectroscopy with these detectors is the parasitic-capacitance preamplifiers. Parasitic- capacitance preamplifiers have high input impedance (~5 MΩ). This combined capacitance is typically 10 to 50 pF. The resulting signal is a voltage pulse having amplitudes proportional to the total charge in the detector pulse, and a rise time equal to the duration of the detector current pulse.

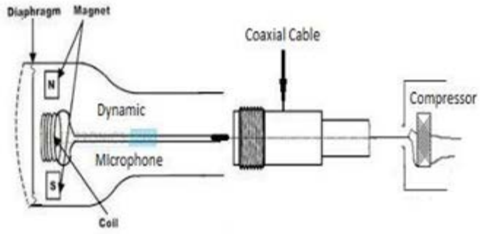
A resistor is connected in parallel with the input capacitance which causes an exponential decay of the pulse with a time constant ~50 µs .The output at the coaxial cable with a low impedance can be driven by an amplifier having a high input impedance and unity gain which is included as a buffer. By terminating the cable in its characteristic impedance, the 93-Ω resistor in series with the output absorbs reflected pulses in long cables. Parasitic-capacitance preamplifiers are not used with semi-conductor detectors because the gain of these preamplifiers is sensitive to small changes in the parasitic capacitance. In partially-depleted semiconductor detectors, the detector capacitance varies with the bias voltage applied to the detector diode.



A Simplified Schematic of the Current-Sensitive Pre- Amplifier Circuit

3. IMPLEMENTATION

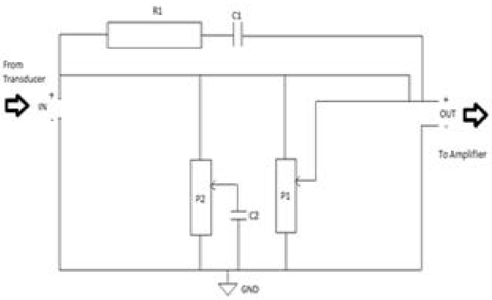
A. Design



A dynamic microphone as shown in the figure 1.2 contains a diaphragm below which a coil is present. This coil induces current by the current whenever diaphragm vibrates. The current produced by the microphone will flow down the coaxial cable, carrying the signal along with it.It is then plugged in to the compressor circuit by means of a micro-dual polarity receiver which is created as a component of the compressor itself.

B. Working

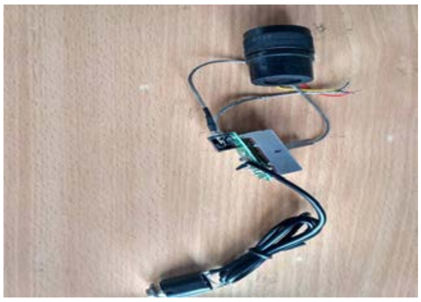
The compressor takes the input from the transducer (in the microphone) and transmits the output to the amplifier. The working means of the compressor are determined by theresistance and capacitance values, R1 and C1 in the circuit below.



The potentiometer P1 is used to set the threshold level. Potentiometer P2 with the help of capacitor C2, changes the characteristics of the wave which is to be compressed. It must be mentioned that the compressor circuit does not require

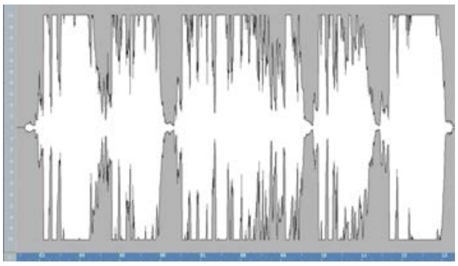
any external voltage. The voltage needed for the circuit to work is given by the input signal itself. Therefore, this is a passive circuit.

The input from the microphone is passed to this circuit by means of a coaxial transmission line. A coaxial cable has a much greater Signal to Noise Ratio(SNR) when compared to twisted pair cable. This circuit also gates ineffective signals which are most probably noise. The portion of the circuit that does this job are the pot and capacitor pair P2 and C2. According to the set P2 level, the weak signals are eliminated to the ground through C2.

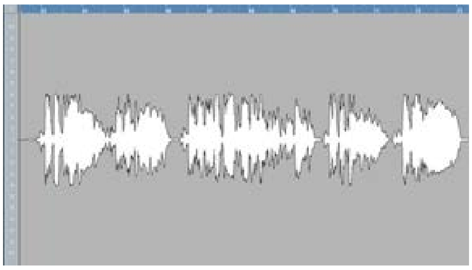


4. RESULTS

When the microphone is kept too close to the source it generally takes in signals of high voltage which causes peaking. When this signal is preamplified by the mixer or amplifier in the PA system, it distorts the output signal as well as the speaker which outputs it. The signal from the microphone without compression is presented beneath:



In the above simulation figure, it is the recording of a normal audio. We can observe that the peaks are ranging beyond the maximum limit. Hence the peaks of the waves are clipped. Clipping results in a loss of information such high scale audio. The below simulation is the same audio recorded through the microphone designed in this project.



Here, the clipped flat ends will be finished as curves after compression. In other words, the clipped wave ends are compressed to a rounded wave peak. The mean or the average amplitude remains the same in both the recordings, but the peaks will differ. The amplitude range (difference between high and low amplitudes) will be very high before compression, but after compression the range will be minimized, making the signal sweet sounding

5. CONCLUSION

To minimize the distorting effects in speakers due to very high amplitudes, of input signals from a sound source, there was a need to control these signals at the source itself, which was the aim of this project. The project was successfully implemented and the output wave characteristics were determined. Upon testing with a PA system, it yielded the desired output.Putting this microphone in practice would enhance the tone of the sound and durability of speakers when compared to the microphones presently used.

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