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SPEAKER TESTING AND NOISE REMOVAL

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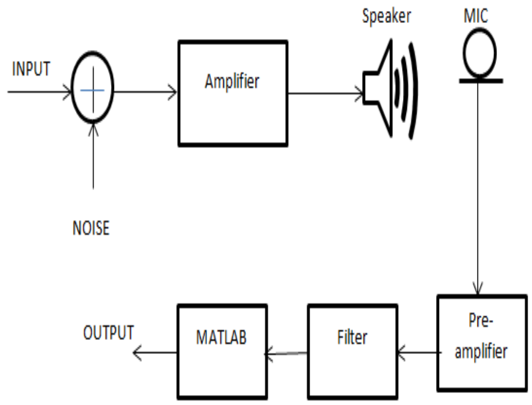
Abstract - *In this project we are going to test the speaker by detecting noise and reduced the noise that affects the speaker by using sallen and key filters. Noise samples are generated using MATLAB. Many times the speaker faces problem due to gaussian noise and some other aspects. Gaussian noise is nothing but unwanted sound signal made by signal received by the receiver or background sound signal that is atmospheric or air etc. in this project due to addition of this background sound in the original signal the speaker produces unwanted sound. In order to build an efficient reliable speaker testing equipment which is able to determine type of noise. For user purpose we are adding Graphical User Interface of the noise which is present in the speaker can be displayed. The reduction of noise is implemented using sallen and key filters and result will be displayed on matlab.*

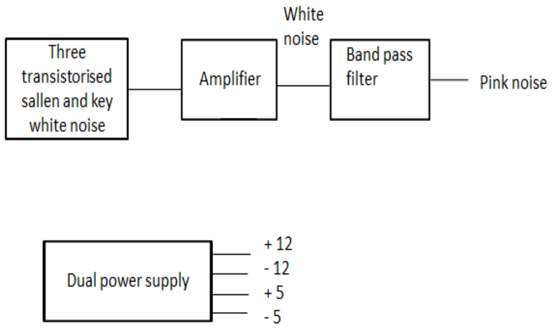
Key Words: **sallen n key filter, Gaussian noise, matlab, white noise, pink noise.**

1. INTRODUCTION

We propose a system that detects noise signal in the speakers, such as white noise , pink noise , gaussian noise, rub and buzz noise. under the laboratory environment, speaker recognition and testing has made great progress. but in real life, the performance of speaker testing system is vulnerable to various factors, especially environmental noise and background unwanted signal. due to this, noise performance of the speaker is reduced and it provide undesirable output. that can harm our ears. this paper studies the performance of speaker identification system when the tester suffers from the background unwanted sound. so we propose the method using different noises and pre-emphasis filter, process normal speech and test the speakers. so this system is based on reducing this unwanted noise that will provide efficient output . this system is also determined type of noise presently affecting the speaker.

II.BLOCK DIAGRAM





III.BLOCK DIAGRAM DESCRIPTION

1. POWER AMPLIFIER

It is used to increase power of the signal and increase the amplitude.It is system on chip amplifier. An audio power amplifier is an electronic amplifier that amplifies low-power electronic audio signals to a level that is strong enough for driving loudspeakers and making the signal whether it is recorded music or a live speech–audible to listeners. It is the final electronic stage in a typical audio playback chain before the signal is sent to the loudspeakers.

2. PRE AMPLIFIER

We are using TL081 pre amplifier. The signal coming from mic is amplified and given to filter. It is basically an op-amp having gain 10 ,which converts microvolt into volt. The TL081 is a low cost high speed JFET input operational amplifier with an internally trimmed input offset voltage. The device requires a low supply current and yet maintains a large gain bandwidth product and a fast slew rate. In addition, well matched high voltage JFET input devices provide very low input bias and offset currents. The TL081 is pin compatible with the standard LM741 and uses the same offset voltage adjustment circuitry.The TL081 may be used in applications such as high speed integrators, fast D/A converters, sample-and-hold circuits and many other circuits requiring low input offset voltage, low input bias current, high input impedance, high slew rate and wide bandwidth. The devices has low noise and offset voltage drift, but for applications where these requirements.

3. FILTER

A filter is device or process that removes some unwanted component or feature from signal. Removing some frequencies and not others in order to suppress interfering signals and reduce background noise. However, filters do not exclusively act in the frequency domain. Correlations can be removed for certain frequency components and not for others without having to act in the frequency domain. There are many different bases of classifying filters and these overlap in many different ways; there is no simple hierarchical classification. It is adaptive filter. It filters out noise from the input signal. Noise is having high frequency that is above 20 kHz range, which is neglected.

4. THREE TRANSISTORISED SALLEN KEY

The Sallen–Key topology is an electronic filter topology used to implement second-order active filters that is particularly valued for its simplicity. It is a degenerate form of a voltage-controlled voltage-source (VCVS) filter topology. A VCVS filter uses a unity-gain voltage amplifier with practically infinite input impedance and zero output impedance to implement a 2-pole low-pass, high-pass, bandpass, bandstop, or allpass response. The unity-gain amplifier allows very high Q factor and passband gain without the use of inductors. A Sallen–Key filter is a variation on a VCVS filter that uses a unity-gain amplifier Because of its high input impedance and easily selectable gain, an operational amplifier in a conventional non-inverting configuration is often used in VCVS implementations Implementations of Sallen–Key filters often use an operational amplifier configured as a voltage follower; however, emitter or source followers are other common choices for the buffer amplifier.

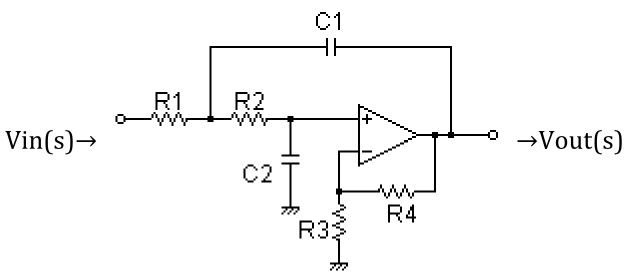
VCVS filters are relatively resilient to component tolerance, but obtaining high Q factor may require extreme component value spread or high amplifier gain. Higher-order filters can be obtained by cascading two or more stages

5. DUAL POWER SUPPLY

Dual power supply units are common equipment in electrical engineering and electronics. They supply positive polarity as well as negative polarity and ground potential. In particular cases both the positive and negative rails are required for the proper operation of your circuit. For example, some Op-Amps need dual power sources.

7812 and 7912 AN7812 is the Positive Voltage Regulator. It regulates the voltage from (almost) 24vDC to 12vDC. AN7912 is the Negative Voltage Regulator. It regulates the voltage from -24vDC to -12vDC. A transformer output must be between 12vAC to 24vAC @ 500mA. Input of transformer.

IV.MATHEMATICAL MODEL



Cut-off frequency:

$$fc = \frac{1}{2\pi\sqrt{R1C1R2C2}}$$

Transfer function:

$$\frac{Vout(s)}{Vin(s)} = \frac{G(2\pi f_c)^2}{s^2 + 2\zeta(2\pi f_c)s + (2\pi f_c)^2}$$

$$Q = \frac{1}{2\zeta}$$

$$G = \frac{R3 + R4}{R3}$$

V. ALGORITHM

1. Start.
2. Make white noise from three stage transistor oscillator.
3. Convert the white signal into pink noise.
4. Output of noise goes to switch.
5. Adding audio signal with respective noise signal.
6. Apply added signal to pre amplifier and then to power amplifier.
7. Signal is applied to mic via speaker.
8. Then apply to MicStarlizer and Pre Filter.
9. Give to PC Using Stereo Socker.
10. Check Noise type and remove using simulinc.
11. Display Spectrum on PC.
12. End.

3. CONCLUSIONS

Thus we can find all type of distortion in speaker i.e. Cone break down, Human interference, Surface crack, Harmonic distortion, Noise interference. It can check all type of speaker up to 1000 Watt.

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