A low cost 3-band gain equalizer using op-amps for hearing aid applications

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Abstract- Hearing loss is a common health issue that affects nearly 10% of the world population reported by international studies. The hearing impaired typically experience more mental disorientation than those with normal hearing levels. It is often difficult to analyze hearing loss that would closely match with prescribed real ear gain/frequency responses due to technological limitations. The conventional device for hearing impairment is an electronic circuit whose main components are transducers (microphone and receiver). This study is focused on the electronic components those are relatively cheap and available in local market. In this paper the conventional active equalizer circuit especially bass, treble, and mid control based equalizer is implemented by op-amp LM741. Overall gain is split between input and output buffers, for bass ±15dB in 100Hz, mid ±9dB for 1kHz and for treble ±15dB in 10kHz. A trimmer-adjusted, lighter, pocket –carry device can be developed in this case.

1. INTRODUCTION

Equalization (EQ) is one of the most important aspects of audio logical measurements. Equalizer is made of a combination of several filters such as low-pass, high-pass and band pass which can attenuate certain frequencies. There were many variations, but the general scheme that ended up being used by almost all manufacturers was the 'Baxandall' topology, named after its inventor Peter J Baxandall [7]. This technology is used to this day, but for audio production (as opposed to reproduction) the equalization available is much more complex and comprehensive. Over the past 40 years, numerous prescription formulas for hearing aid fitting have been proposed, each specifying a desired gain/frequency response based on the audio logical test results of an individual hearing-impaired patient [1, 2]. Equalizers can be classified into 4 categories – tone control, graphic, console and parametric [3], but for the purpose of this paper, we will only investigate active equalizers. A 3-band tone control is one of the simplest forms of equalizers which allows the user to boost/cut bass, mid and treble regions of the audio spectrum [3].

1.1 Bass Control:

Figure 4 shows a common passive arrangement that allows a nominal cut and boost of around within suitable dB range. Sometimes it is not symmetrical.

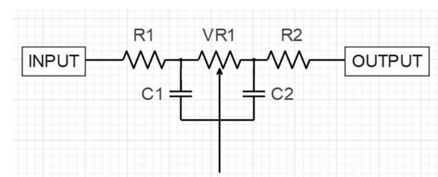

Fig.1. Passive Bass control circuit

Capacitor C1 and C2 and R2 affects the bass frequencies. If C2 is increased and R2 is decreased then the bass level will increase. The full bandwidth signal can pass through R1 but with the slider of VR1 at the bottom end of its resistance track, C1/R2 can form a high pass filter having a corner frequency of around 40 to 100Hz only frequencies appreciably higher than this are allowed to pass un-attenuated.

A tone control circuit should have a low output impedance, and this is difficult to achieve using a passive version. It is to be noted that the potentiometers are logarithmic - linear potentiometers do not work. The following stage must have a high input impedance, and was not uncommon. A gain of 10 is needed to restore the level with the controls set for a nominally flat response. [4]

1.2 Mid frequency control:

The Wien bridge stage is selected to work for the mid-frequency equalizer when it is fixed. Only significant limitation is the restricted range of boost and cut that is possible - about 9dB, due to the insertion loss of the Wien network. It's possible to increase the gain, but requires some gain in the Wien feedback path.

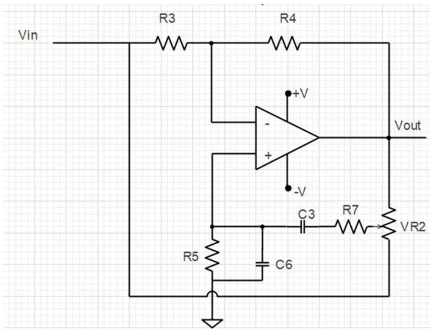

Fig.2. The Wien bridge circuit

As shown in the Figure 2 the output of the operational amplifier is fed back to both the inputs of the amplifier. One part of the feedback signal is connected to the inverting input terminal (negative or degenerative feedback) via the resistor divider network of R3 and R4 which allows the amplifiers voltage gain to be adjusted within narrow limits.

The other part, which forms the series and parallel combinations of R and C forms the feedback network and are fed back to the non-inverting input terminal (positive or regenerative feedback) via the RC Wien Bridge network and

The RC network is connected in the positive feedback path of the amplifier and has zero phase shift a just one frequency. Then at the selected resonant frequency, ($\boldsymbol{f}\mathbf{r}$) the voltages applied to the inverting and non-inverting inputs will be equal and "in-phase" so the positive feedback.

1.3 Treble control:

In Treble control circuit, the gain should be flat at low frequencies. At higher frequencies, the gain can be cut or boosted by turning the control knob Clockwise (CW) or Counter-Clockwise (CCW). A simple inverting operational amplifier and several cleverly placed components. [6] First, at low frequencies, the capacitors open up leaving this circuit.

For each excitation frequency to cover 10 points per octave in order to reach a steady state condition. Start frequency was set 20 to stop frequency 20000.

2.2 Circuit Model:

For the dual rail power supply in LTspice simulation we used +9V and -9V in op-amp thus the signal can be amplified in both side. In real hardware system, the microphone is biased with a pull-up 10k resistor. This electrical signal from environment is then fed to the R2C1 filter to receive signal from an initial frequency. In LTspice

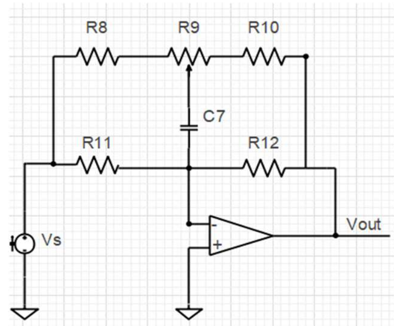

Fig.3. Treble control circuit

This is classic inverting amplifier with gain G = - R12 / R11. The R8-R10 resistor string just hangs between the input and output, not disturbing the summing point (negative input node) or the overall gain. Alternatively, at high frequencies the capacitor is shorted. As it is the inverting amplifier, but with its gain modified by the parallel resistors R8-R10.

With the pot turned fully CW, then is the gain (G = -R12/R11) will increased. Because R8 is parallel to R11, its reduced resistance in the equation's denominator causes the gain to increase. The maximum gain is calculated by,

$$G_{max} = \frac{R12||(R9+R10)}{R11||R8} \text{..........(1)}$$

R12 is also decreased to some degree. But, because R9 is very large, R12 is reduced by a much lesser amount.

In the high frequencies cut with the control knob fully counter-clockwise (CCW), we just slided the wiper of the pot forward, and see the effect on gain. The gain is reduced by R10 directly paralleling R12.

$$G_{max} = \frac{R12||R10}{R11||(R8+R9)} \text{..........(2)}$$

2. MATERIALS AND METHODS

2.1 LTspice model implementation

Next we will describe the creation of the LTspice netlist by means of a Newton iteration script. All calculations are done there beforehand, and the netlist file is set up line by line using strings of the appropriate SPICE code. The LTspice® script was executed for 3 different frequencies (100 Hz, 1kHz, 10 kHz) to a 4 stages of frequency selecting circuits. The AC transient simulation time was adjusted circuit model, an AC source is chosen instead of microphone for input signal with a magnitude about 1V. It is useful for small AC signal analysis (.AC). Initially the input filter is set to at 48Hz lower cut frequency. Each stage set separately to flat, half boost, half cut, full boost and full cut. There will be no interaction. The low frequency roll-off in first and last stages are kept same. As shown in the figure 4, stage gain is increased to restore overall circuit gain, U1 non-inverting subsection of LM741 is applied as buffer gain where input and output impedance is balanced. With R3, R4=33k is fed to

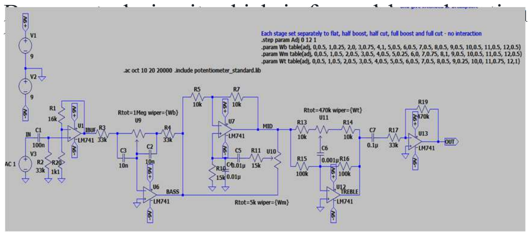

Fig.4. 3-band gain equalizer LTspice circuit model.

From the LTspice slandered potentiometer library and the user defined directive variables .PARAM statement allows to create adjustable table where the potentiometer slider can alter in a particular range. As stated above the mid and treble control can be designed in similar ways using U7 and U12 sections. At the last stage inverting gain – R19/R17 can reduced stage gain to restore high frequency bandwidth and provide intended low frequency breakpoint.

3. RESULTS AND OBSERVATIONS

In the Bass control part in U6, when U9=1M is rotated fully clock wise, C3 gets shorted and maximum bass boost = 20log(33+33/33)=6.02dB. When U9 rotates fully anticlockwise then, C2 gets shorted and minimum Bass cut=20log (33/33+33) = -6.02dB.

4. DISCUSSIONS / CONCLUSIONS

The data shown indicate substantial improvement in the 'Baxandall' topology capability of hearing aids to achieve good approximations to desired gain/frequency responses. With two- or more pole filters for bass, mid and treble frequency tone controls, and wide-band OP-AMP output stages, adequate high frequency gain can be achieved without over amplification of the mid frequencies. The flexibility of the three-band hearing aid with active equalization is sufficient to supply an

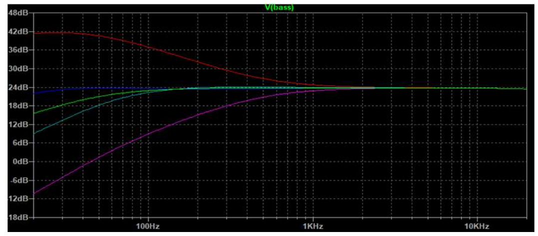

Fig.5. Bass control frequency response curve. In this case the gain value can be find both in positive and negative side of origin.

But since gain is taken from output of U13 section with certain gain multiplication then 23 dB to -10 dB gain can be obtainable.

Applying Wein bridge circuit using U7 subsection, the feedback network R11C5 is constructed. As shown in the ±9dB to ±15dB gain can be achievable in this design.

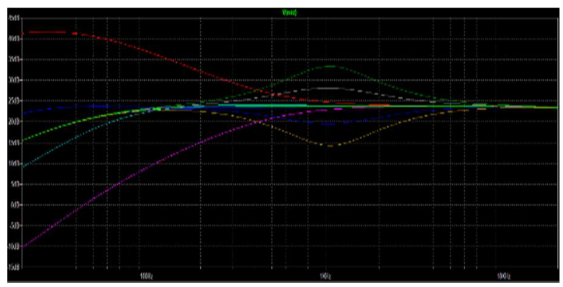

Fig.6 Mid control frequency response curve. At 1 to 2 kHz the gain curve gives 14 to 28dB output.

For Treble control circuit, with op-amp U12, the maximum boost treble is calculated when U10 rotates clockwise then from equation (1) ,

G_{max}=100k||(470k+10k)/100k||10k

=20log(9.15)=19.22dB,

and from equation (2) When U11 rotates anticlockwise then,

G_{min}=-100k||10k/100k||(470k+100k)

=20log(0.109)

=-19.25dB.

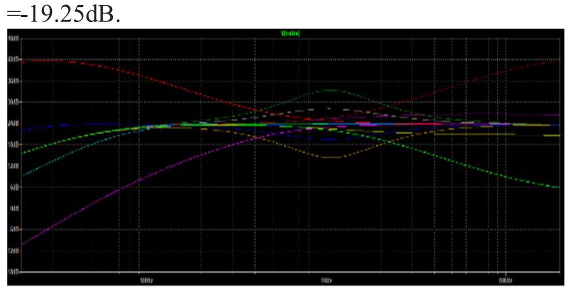

Fig.7 Treble control frequency response curve.

As shown in figure 7 at the high frequency like 10 kHz the gain is more than ±15dB.

accurate match even with more unusual audiograms that have previously been quite difficult to fit. This improved flexibility also provides greater opportunity for successful revision of a patient's gain/frequency response, if this is considered desirable after the initial fitting to prescribed values. Because prescriptive formulas are based on mean data for several parameters, the optimal frequency response for an individual patient may in fact differ from the prescribed target. Modifications to the desired frequency response for a patient whose speech recognition performance is poorer than expected would typically be in the direction of increased high frequency gain relative to low frequency gain.

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