Design of Programmable Hearing Aid based on Frequency Filters Bands

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Abstract: ***Hearing disorders are the most prevalent disabling conditions reported all over the world that affect about 10% of the population. It adversely affects physical, cognitive, behavioral, and social functions, as well as the general quality of live. The first electrical hearing instruments was introduced 100 years ago, but only about 20% of those who could benefit from hearing aids wear them, moreover, surveys suggested that about only 50% of hearing aid users are satisfied with their hearing instruments. This indicates hearing technology even today is a prominent research area. This research work involves the design of programmable hearing aid with frequency filters bands to allow the user to tune the amplitudes selectively to a person's particular pattern of hearing loss to be amplified by PGA (Programmable Gain Amplifier) more than the rest. These filters are tested with the standard test input signal and also implemented. The result showed that output of filters bands are in the same phase and flattest group and lesser matching errors.***

Keywords: Sallen-key(VCVS) , PGA(Programmable Gain Amplifier), Programmable hearing aid

1. Introduction

Hearing aids can do wonders for faded hearing. With the technological advancement in the society, hearing aids can significantly enhance the quality of life for most people with hearing impairment. Therefore, the electronic hearing aid is designed to make sounds louder and therefore easier to hear. Also the design of the circuitry keeps the sound from becoming too loud and helps reduce the effects of background noise.

The digital hearing aids are designed and developed have more advantage than analog hearing aids because it has low power consumption, small size and low noise. Digital hearing instruments [1], [2] uses advanced digital signal processing like multichannel compression, multiple memories and intelligent signal processing, which improves the performances of the hearing instruments and the satisfaction of the user. With the development of VLSI microelectronics technology, it is now possible to incorporate greater function modules of electronic circuits in a very small area, and hearing instruments now can be positioned completely inside the ear canal.[3]

2. Literature Survey

Tomson Devis et al. proposed the application of multirate frameworks and filter banks for different procedures in personalized hearing assistants are investigated [4]. BALAJIS.SAWANT et al. designed digital FIR non-uniform reconfigurable filter bank. the complexity is greatly decreased by using only three prototype filters for subband generation and same for masking filters bank. This filter bank can achieve better matching than fixed filter bank due to its ability to distribute subbands flexibly[5]. NishaHaridas et al. proposes the use of a variable bandwidth filter, using Farrow subfilters, for this purpose. The design of the variable bandwidth filter is carried out for a set of selected bandwidthsResults show that lower order filters and better audiogram matching with lesser matching errors are obtained using Farrow structure. This, in turn reduces implementation complexity. The cost effectiveness of this technique also comes from the fact that, the user can reprogram the same device, once his hearing loss pattern is found to have changed in due course of time, without the need to replace it completely [6].

3. Method and Materials

Classifying hearing loss according to the type, degree and configuration of hearing loss is the primary information required to determine further test procedures and to direct medical and/or audiological interventions. As mentioned previously, the main goal of this work is to design and construct an efficient frequency bands filter of an adjustable hearing aid which helps the user to adjust the device according to the change in hearing loss pattern with time or age. First, Since this is an audio device, it is important to ensure that the outputs of the filters remain in phase. Since, We will be using second order Sallen-Key topologies (VCVS topology).

A VCVS filter uses a 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 VCVS filter allows high Q factor and passband gain without the use of inductors. A VCVS filter also has the advantage of independence: VCVS filters can be cascaded without the stages affecting each others tuning. A Sallen–Key filter is a variation on a VCVS filter that uses a unity-voltage-gain amplifier (i.e., a pure buffer amplifier). It was introduced by R. P. Sallen and E. L. Key of MIT Lincoln Laboratory in 1955.[7]

In our work four parallel filters are tuned separately to the optimum center frequencies and bandwidths to match each of the audiogram.

The first band consists of signals below 300Hz, so we use a standard low pass filter, This band corresponds to low frequencies that can be heard. The second band starts at 300Hz and continues to 3.4 kHz. This band is selected to correspond to the voice range. The third band begins at 3.4 kHz and extends to 8kHz. This corresponds to high frequencies that can still be heard. Finally, the fourth band extends beyond 8kHz.These filter are simulated by computer using NI Multism 13.0 version and frequency response ofeach are obtained , also are implanted on breadboard and tested with the standard test input signal.

3.1 Low pass filter

Second order low pass filters are easy to design and are used extensively in many applications. An operational amplifier is used as the buffer here. The basic configuration for a Sallen-Key second order (two-pole) low pass filter is given as:

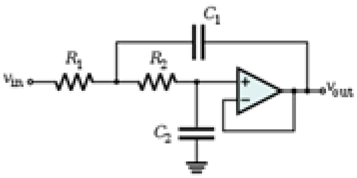

Figure 1: 2order sallen-key unity LPF

In active second order filters, the damping factor, ζ (zeta), which is the inverse of Q is normally used. Both Q and ζ are independently determined by the gain of the amplifier, A so as Q decreases the damping factor increases. Then Q, the quality factor, represents the “peakiness” of this resonance peak, that is its height and narrowness around the cut-off frequency point, fc. But a filters gain also determines the amount of its feedback and therefore has a significant effect on the frequency response of the filter.

$$A = 3 - (2 \times \zeta)$$

$$\text{Where}: \quad \zeta = \frac{3 - A}{2} = \frac{1}{2Q} \qquad \textbf{(1)}$$

$$\therefore\ A = 3 - \frac{1}{Q}$$

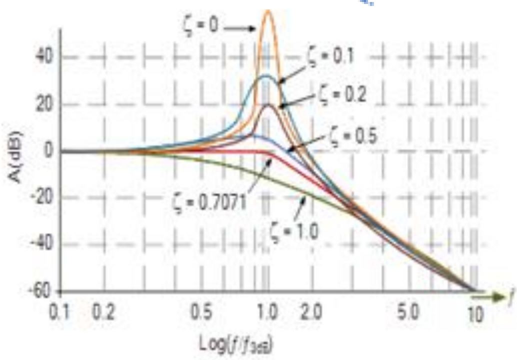

Figure 2: Second Order Filter Amplitude Response

From the circuit above we know that for equal resistances and capacitances, the cut-off frequency point, fc is given as:

$$f_C = \frac{1}{2\pi RC} \qquad (2)$$

And the gain is given as:

$$\text{Gain } (A_V) = 1 + \frac{R_A}{R_B} \qquad (3)$$

Since it is unity gain, Av will be=1, we will use the **MCP602** op-amp in an ll filters design.

Q will be =0.5 and ζ=1. Let the cutoff frequency (fc) =300Hz and C=100nF, Then
R = 1/ (2*π*300 * 100 x 10-9)=5.3KΩ

3.2 High pass filter

Since second order high pass and low pass filters are the same circuits except that the positions of the resistors and capacitors are interchanged, the design and frequency scaling procedures for the high pass filter are exactly the same as those for the previous low pass filter.

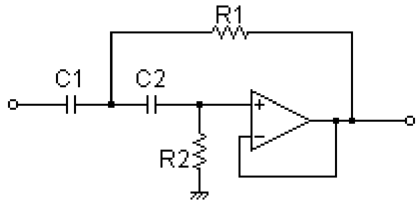

Figure 3: 2order sallen-key unity HPF

To design HPF the same equation (2) of LPF will be used, Let the(Fc)=8KHz and C=10nF,The R =1/ (2*π*8000 * 10 x 10-9)=2KΩ ,and Since it is unity gain it will be 1 and Q =0.5 , ζ=1.

3.3 Band pass filter

The architecture that has been used to implement the second order band-pass filter is the Sallen-Key Topology. This topology is chosen due to its simplicity compared to other known architectures such as multiple feedback and state variable. The circuit diagram below shows a second order Sallen-Key band pass filter

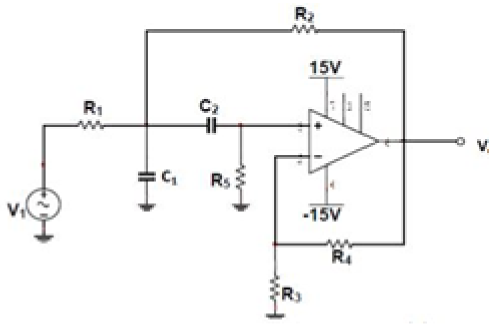

Figure 4: Sallen-key bandpass filter

The general transfer function of a second order band pass filter is given as:

$$H(s) = \frac{a_1 s}{s^2 + b_1 s + b_0} \qquad (1)$$

Where a_1,b_1 and b_0 are constant The transfer function of the second order Sallen-Key band pass filter in Fig(7).is:

$$H(s) = \frac{s c_2 R_2 R_5}{s^2 c_1 c_2 R_1 R_2 R_5 + s\,[c_2 R_2 R_5 (1-G) + c_2 R_1 R_5 + c_2 R_1 R_2 + c_1 R_1 R_2] + (R_1 + R_2)} \qquad (2)$$

Equation (2) can be simplified by putting R1 = R2 = R, R5 = 2R, C1 = C2 = C this is known as the equal component Sallen-Key band-pass filter

$$H(s) = \frac{SCRG}{S^2C^2R^2 + SCR(3-G) + 1} \qquad (3)$$

The poles of the transfer function are:

$$S_{1,2} = [(G-3)/RC \pm \sqrt{\{((3-G)/RC)^2 + 4(1/RC)^2\}}] / 2$$

Substituting, the above transfer function can be written in standard form as:

$$H(s) = \frac{SA(\omega_0/Q)}{s^2 + (\omega_0/Q)s + \omega_0^2} \qquad (4)$$

Resonant frequency (fo)	$1/2\pi CR$
Bandwidth (β)	f_0/Q
Op-amp gain (G)	$1+\frac{R_4}{R_3}$
Gain at ω_0 (A)	$\frac{G}{3-G}$

When designing band-pass filters, the parameters of interest are the gain at the mid frequency (A) and the quality factor (Q), which represents the selectivity of a band-pass filter. The quality factor (Q) is defined as the ratio of the mid frequency (f_0) to the bandwidth (B) of the second order.

Now to design two BPF of different bandwidth, the first one (300-3400) Hz.The second one (3400-8000) Hz.

BPF 1:
The bandwidth is (300-3400) so it will be =3100 Hz and f_0=1850 Hz
Let the design component be C=22nF, R3 & R4=10KΩ
R = 1/ (2*π*1850 * 22 x 10^{-9}) = 4KΩ and R5 =10KΩ
G = 1 + (10/10) = 2
A = 2/ (3 - 2) = 2
Q=1850/3100=0.59

BPF2:
The bandwidth is (3400-8000) so it will be =4600 Hz and f0=5700 Hz
Let the design component be C=22nF, R3 & R4=10KΩ
R = 1/ (2*π*5700 * 22 x 10^{-9}) = 1.3 KΩ and R5 =2.6KΩ
G = 1 + (10/10) = 2
A = 2/ (3 - 2) = 2
Q=1850/3100=0.59

Notice that when G approaches the value of 3, A and Q tend to infinitely increase and cause the circuit to oscillate, As G grows greater than 3, A becomes negative too so the R3 & R4 must be chosen accurately so that not to get a gain greater than 2.

4. Simulation and Results

The circuits of the second order filters using Sallen-Key topology is connected and simulated using MULTISIM 13.0 .The circuit of each filter composed of Op-amp. The amplifiers are based on the (MCP602) Op-amp circuit.The results of the each filter which are obtained in the Lab and simulation are shown below.

4.1 Lowpass filter (>300 Hz) result

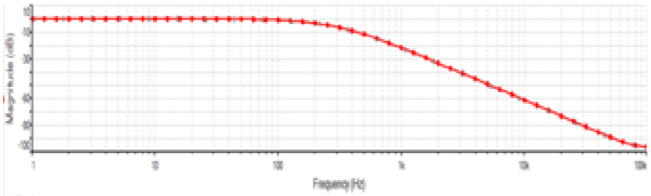

Figure 5: LPF Frequency response

Input voltage (mv)	Frequency (Hz)	Output voltage (mv)	Gain (dB)
20	20	200	20
20	30	200	20
20	40	200	20
20	50	200	20
20	100	180	19
20	200	150	17.5
20	300	100	14
20	500	70	10.8
20	800	20	0
20	1000	10	-6

4.2 Bandpass filter (300-3400) Hz result

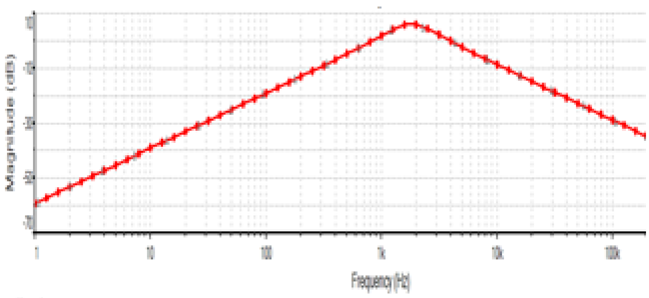

Figure 6: BPF 1 Frequency response

Input voltage (mv)	Frequency (Hz)	Output voltage (mv)	Gain (dB)
20	150	80	12
20	300	125	16
20	500	232	21.3
20	700	320	24
20	900	440	26.8
20	1000	500	28
20	2000	800	32
20	2500	720	31
20	3000	560	29
20	4300	480	27.6
20	6000	240	21.5
20	10000	80	12

4.3- Bandpass filter (3400-8000) Hz results

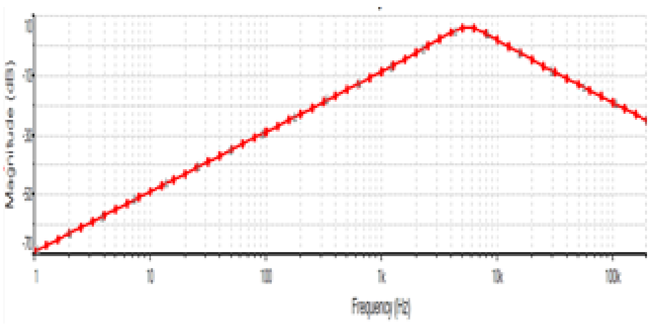

Figure 7: BPF 2 Frequency response

Input voltage (mv)	Frequency (Hz)	Output voltage (mv)	Gain (dB)
20	400	56	9
20	600	84	12.4
20	800	120	15.5
20	1000	140	17
20	3000	480	27.6
20	5000	800	32
20	6000	800	32
20	7000	720	31.1
20	8000	620	30
20	9000	480	27.6
20	12000	400	26
20	20000	100	14

4.4 Highpas filter (<8000 Hz) results

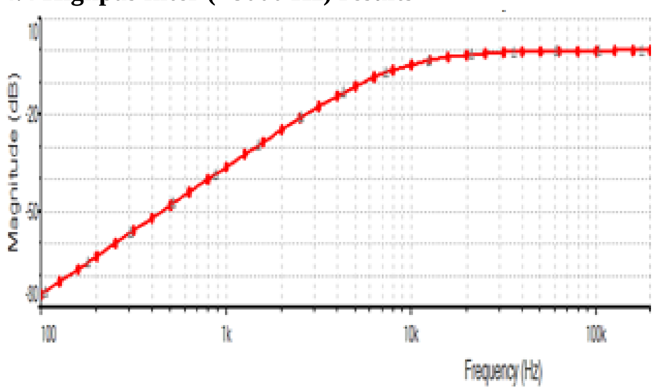

Figure 8: HPF Frequency response

Input voltage (mv)	Frequency (Hz)	Output voltage (mv)	Gain (dB)
20	1000	10	-6
20	2000	20	0
20	3000	40	6
20	5000	120	15.5
20	7000	180	19
20	8000	200	20
20	9000	220	20.8
20	12000	280	23
20	20000	340	24.6
20	60000	400	26
20	100000	400	26

5. Conclusion

An efficient method for the design of digital filters suitable for digital hearing aid is proposed in this paper. The method utilizes Sallen-Key topology based variable bandwidth filters. The Sallen-Key topology, also known as a voltage control voltage source (VCVS). It is one of the most widely used filter topologies due to its simplicity[8] and because it has the advantage of independence: VCVS filters can be cascaded without the stages affecting each others tuning. The circuit design for the four filters are implemented, simulated using MULTISM 13.0 and results are obtained in the Lab. The results showed that the outputs of the filters remain in phase and have the flattest group delay which is optimal result for every audio device especially the hearing aid.

6. Future Scope

Future trends and expected innovations in the hearing aid industry and new filter's topologies that are very accurate and noise reduction for patient comfort.

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Author Profile

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