ISSN: 2690- 912X

Review Article

Journal of Genetic Engineering and Biotechnology Research

Explanation of Frequency Distinguishing in Human Ear (Using Computer Model)

Michael Ostrowski

***Corresponding author**
Michael Ostrowski.

Submitted: 25 Nov 2021; **Accepted:** 02 Dec 2021; **Published:** 22 Dec 2021

Citation: *Michael Ostrowski. (2021). Explanation of Frequency Distinguishing in Human Ear (Using Computer Model). J Gene Engg Bio Res, 3(2), 60-63.*

Abstract

The inspiration for this model ware possibilities of the human ear to distinguish the frequency of sounds and a diffraction grating. Detection takes place after max. 15 length of the wave (arbitrary choice). The range of frequencies to detect for tests – 800-3200 Hz: detection every 5 Hz in the range 800-1600 Hz and 10 Hz in the range 1600-3200 Hz (arbitrary choice). It can explain the ***residual hearing effect*** *(missing tone f is heard when harmonic tones 2f, 3f and 4f are played). The algorithm can be used as an* ***alternative for FFT****. Model uses only memory for delay line end for results, and adding operation, so it should be fast and cheep, and can work on-line in real-time. Testing program was written in Perl.*

Description of the Idea

Proposed explanation how a human ear works are very complicated and not explaining possibilities of human ear [1, 2]. Proposed method uses the idea of a diffraction grating. Figure 1 presents the idea. Sampling frequency – 50000 Hz.

On the first level, a cochlea is considered as:

- Delay-line (linear for simplicity) implemented as a vector
- Signal detectors along the delay-line – values in the DL
- Sophisticated low-pass filter along the delay-line (not shown on Figure 1)

Frequency detection takes place on the second and third level. It looks strange, but frequencies to detect should be arbitrary chosen. These values decide which delayed values are added ("legs" of each adder in Figure 1). It was arbitrarily chosen, that 15 values would be taken into consideration for each adder. Notice, that highest frequencies are detected at the beginning of the delay-line, the lowest are more scattered along DL – to the end for lowest detected frequency (fn and f1 in Figure 1). So after 15 periods of the frequency f_n this frequency in the signal in the rest of the delay-line is useless – look at the explanation of residual hearing effect.

In the used testing program, the file coef.sp2 is created. In it, it is possible to see the distribution of addresses along the DL, where the signals for adders are taken. These values are written as characters so it may be useful to look for similarities with connections of ortoneurons and spironeurons (or auditory nerve?) with hair cells in human ear. The Figure 2 presents the answer of the program for single frequency. It can detect frequencies little higher than highest expected (ultra sounds by bones).

Pictures, which show detected frequencies, are presented by the program as the PostScript file, and so inserted to the paper (look description of the program). Frequency discriminator in presentation simply changes the order of presented frequencies. To determine, which frequency was on input simply chose the local maximum, or construct the neuronal net, as it is sometimes suggested [4].

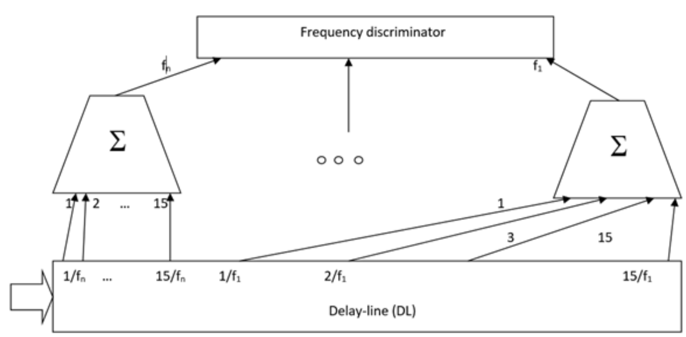

Figure 1: Idea of Frequency Detection and the Testing Program

/DL in the program is a vector, so is frequency discriminator – which reverts the order of frequencies (f1 - fn)/

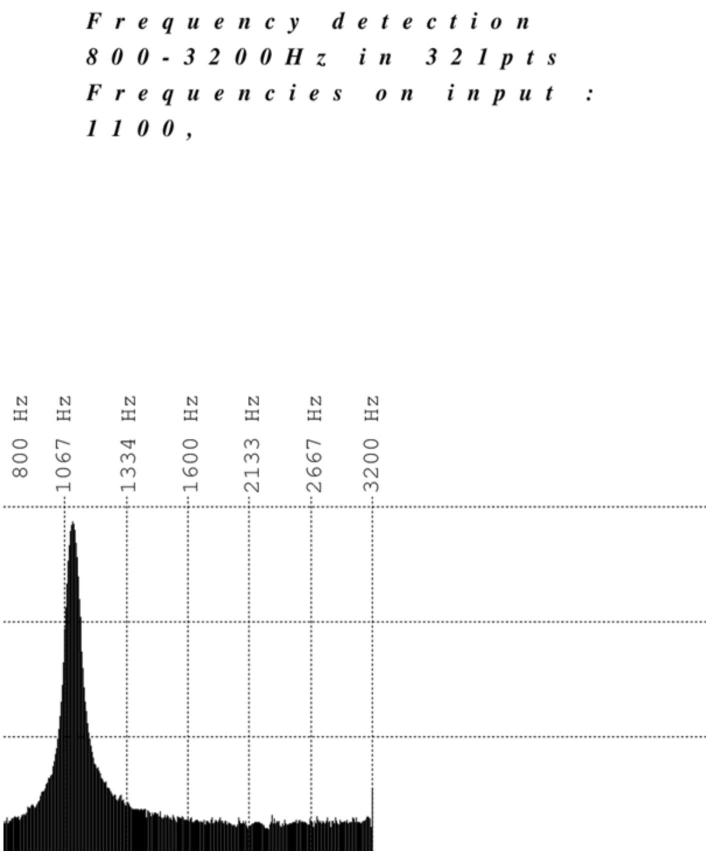

Figure 2: Single frequency 1.1 kHz detected. Each vertical bar represents frequency possible to detect.

Residual Hearing Effect

The side effect of the presented method is detection of the "ghost frequencies" (f/2 and f/3) – look Figure 3 (and Figure 4), but it can explain the residual hearing effect known in human hearing. After the time 15*1/f the signal has no meaning for f detection, but is still detected by f/2 and f/3 etc. adders – but only on a part of inputs of these adders. These detected signals are too strong. The solution of the problem can be the properties of a basilar membrane – Figure 5 (presented here because the figure is very illustrative). Basilar membrane is light, thin and stiff at the beginning and heavier, thicker and more elastic at the end [3, 4]. It can be considered as low-pass mechanical inertia filter distributed along the entire basilar membrane. It is not implemented in the program.

Notice, that for high frequencies input, "legs" for adders are all concentrated at the beginning of the DL, so the effect of low-pass filter is weak (Figure 1 & Figure 5).

The effect of the "ghost frequencies" can explain how ultra sounds coming by skull-bones can be detected (e.g. 30 kHz will be detected as 15kHz).

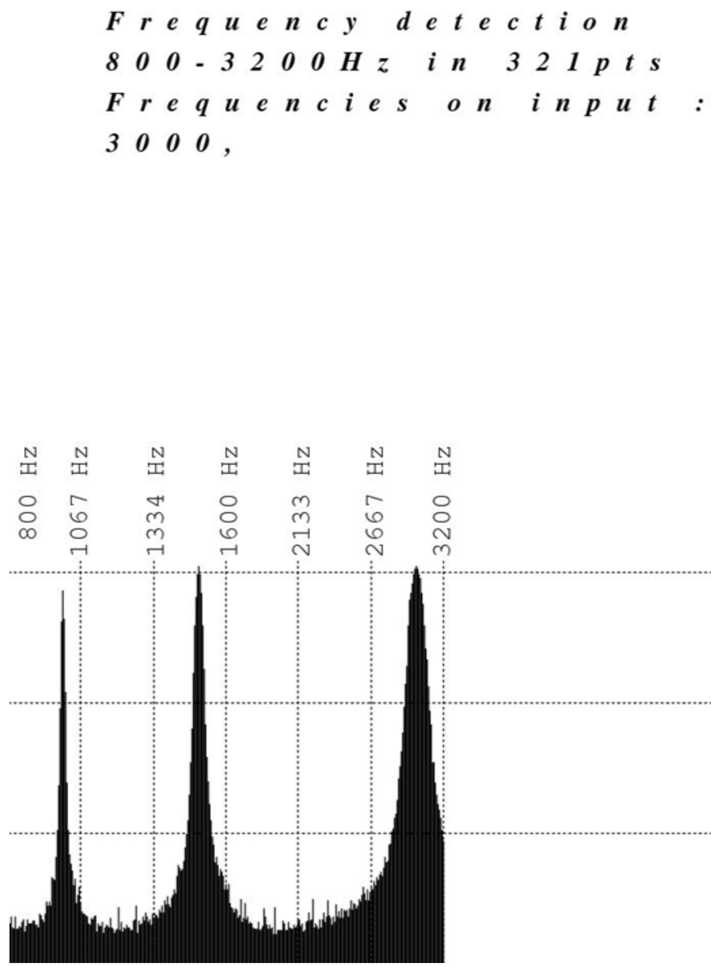

Figure 3: Detection of 3 kHz signal on Input /many periods of the input wave/.

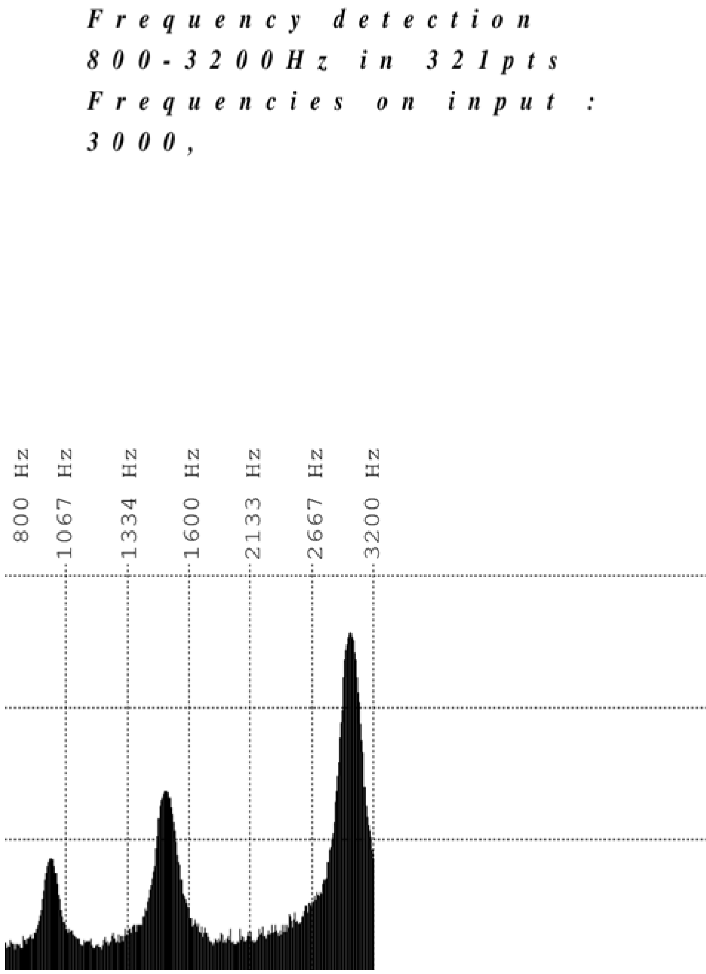

Figure 3a: Detection of 3 kHz signal after 15 periods of wave as the input signal.

Figure 3: and 3a show the reason of the residual hearing effect – see conclusions; Figure 3a shows the effect after short time – 15 periods of frequency 3000 Hz/

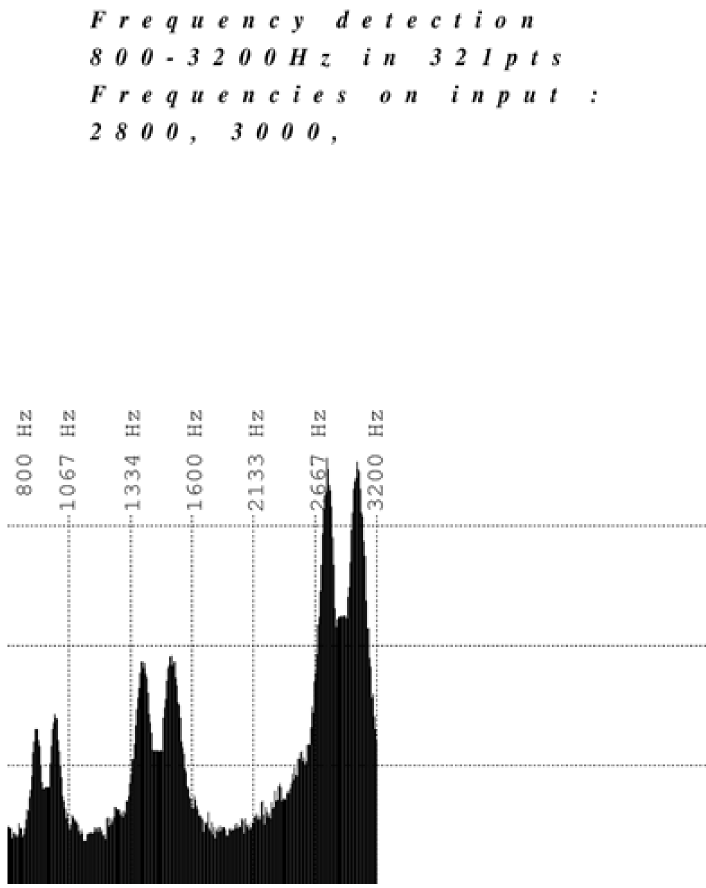

Figure 4: Detection of two frequencies (2.8 & 3 kHz) without correction of the "ghost frequencies"

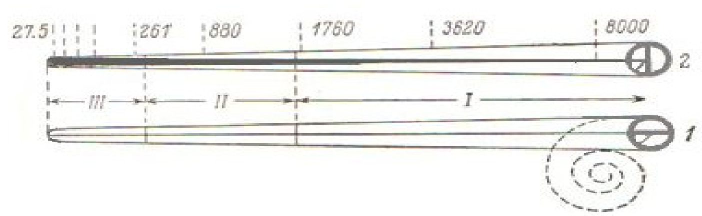

Figure 5: The shape of the basilar membrane/ [4],[5]/ can explain how detection of the input frequency as the "ghost frequencies" visible on figures 3, 3a and 4 can be reduced (decreased) if membrane is low-pass filter along all its length.

The Alternative for Fft

As shown above, the algorithm can be used for fast detecting of frequencies in the input signal. It uses only memory – as DL and frequency discriminator, and adders – which can work as separate tasks (Figure 1). For one octave signal (e.g. 1 MHz-2 MHz) detection is very simple (no need of "ghost frequencies" correction - Figure 2).The main difference is that at the beginning we must decide what frequencies we want to detect (or range of frequencies and its density).

The detected frequencies can be taken as the greatest local values (see Figure 2, 3 and 4).The problem is with detected "ghost frequencies", but they can be removed in mathematical way: when frequency f is detected, "ghost frequencies" f/2, f/3 etc. have to appear, and it is known, how the detected signals look. So is possible to subtract part of the values in f-adder and around it, from f/2 and around, and so for other "ghost frequencies". More, frequency f is detected 2 times faster than "ghost" f/2, and is 2 times smaller (when the basic signal is short <=15 wave periods in this example) so it can be used in real-time reaction (Figure 3a and 4).

For the light, diffraction gratings have 120,000 lines to the inch (approx. 4,724 lines per mm). so for very high frequencies detection, more than 15 adders should be probably used.

The Testing Program

The program is simple (idea is presented on Figure 1) - for tests. It shows how the method works. It was written in Perl. Perl (as interpreter) is convenient to make changes and to see the result fast. Perl 5.6.0 (version for Windows 7 /but was tested on Windows 10 too/) was used, with the library Tk – for implementation of input and output. Perl is open source development, so it can be implemented for free. As the output, PostScript file is created. Each detected frequency is presented as the vertical line. On output maximum signal amplitude values from the summers are presented. On input a pure sinusoid is given (may be given few sinusoids). Amplitudes should be less than 1.5 - the reason is the results presentation. PostScript files can be viewed by IrfanView (or other PS viewer). The input is amplified. The main algorithm has about 30 lines (plus preparing the table of DL addresses of input values for each adder detecting the frequency).

The program ends, when signal from input comes to the end of DL. If it should work longer or in real-time, memory of a DL should be organized as a ring or such a ring can be simulated using modulo arithmetic (modulo DL-length) – for calculation of addresses in such DL- to avoid of shifting values in the DL.

Conclusions

The paper presents only a part of mechanism of frequency detection – to show, that it works. Algorithm is simple, fast, and can be used in general signal processing – alternatively to FFT. It can explain some possibilities of a human ear. I have tested detection of every frequency using 15 delayed signals. More signals will provide better distinguishing of frequencies. E.g. near frequency 1000Hz selectiveness should be 1Hz (in this model – 5Hz). Does the same number of delayed signals should be added for all frequencies? Another question: in this model high frequencies are detected very fast, comparing with low frequencies (waiting for 15 wavelengths). Is it important, or can be used in other areas of frequencies detection?

The simulation of distributed low-pass filter along the DL is too complicated for me.

References

1. David, S. V. (2018). Incorporating behavioral and sensory context into spectro-temporal models of auditory encoding. Hearing research, 360, 107-123.
2. Verhulst, S., Altoe, A., & Vasilkov, V. (2018). Computational modeling of the human auditory periphery: Auditory-nerve responses, evoked potentials and hearing loss. Hearing research, 360, 55-75.
3. Vartanian I.A. Sound- Audition-Brain(1981).(in Russian) Leningrad "Science".
4. Tadeusiewicz R. Speech Signal(1988). (in Polish) WKL.
5. Lindsay, P. H., & Norman, D. A. (2013). Human information processing: An introduction to psychology. Academic press.