Digital Hearing Aids - A Poor Substitute for the Real Thing

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ABSTRACT

Modern digital hearing aids are programmable real-time signal processors many times more powerful than the computer that landed Neil Armstrong on the moon. However, they are a far cry from the amazing signal processing capabilities of the human hearing system. This talk discusses why the best that man can produce is so inferior to our natural God-given hearing system.

Introduction

When the first digital hearing aids became available one hearing aid manufacturer claimed that the circuit in their hearing aid was more powerful than the computer that landed the first man on the moon. That sounds pretty impressive and it prompted me to find out more about the computer that made it possible for Neil Armstrong to step onto the moon's surface.

The first landing of men on the moon on the 20th July, 1969 was surely one of the most exciting events in recent history but it would not have been possible without computers to make the rapid calculations needed to place the craft in orbit and guide the Lunar Module (LM) to the moon's surface.

The computer in the LM was an amazing instrument considering that integrated circuits were fairly new and large scale integration was still to be developed. It weighed 30 kg and had 5000 integrated circuits, 74 kBytes of ferrite core program memory and just 2 kBytes of ferrite core Random Access Memory (RAM) – that's 256000 times less than you'll find in the average PC.

Today we are putting computers into hearing aids that fit entirely in the ear canal and they are many times more powerful than the computer that put Neil Armstrong and Buzz Aldrin on the moon.

Digital Hearing aids

The computer in a modern in-the-canal hearing aid may be no more than 6 mm x 3 mm x 1.5 mm in size. It can process digitized audio signals in real time100 times faster than a digital music system with a supply voltage of only 1.2 Volts and a current consumption of 0.8 mA.

Digital signal processing means that the frequency response can be adjusted with virtually no limitations compared to analogue hearing aids where there is a limit to how many trimmers you can fit on the housing. In addition it is relatively easy to program the chip to automatically make adjustments to the frequency response and gain in real time. Some modern hearing aids use microphone arrays that steer the pickup towards signals identified as speech, while reducing the level of signals identified as noise.

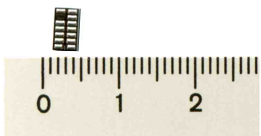

Figure 1. A digital hearing aid circuit.

With such amazing technology available we might wonder why many hearing aid users with instruments costing up to R30000.00 still resort to "Assistive Listening Devices" such as remote microphones and headphones for watching television and complain about abnormal interference from background noise in restaurants and shopping centres. One explanation seems to be related to the fact that many people use only one hearing aid.

Sound intensity differences at the ears

The human Hearing System (HS) has an amazing ability to separate meaningful sound from background noise by comparing the sound in two ears. This ability starts with a means of determining the position of a sound source by comparing the sound intensity at the ears. The ear closest to the sound source will receive a stronger signal. The intensity difference at the ears is $20.\log((D+d) / D)$ where d is the distance between the ears and D is the distance to the nearer ear. If the ears are 0.18 m apart a sound originating at a distance of say 4 meters would be 0.4 dB louder at the nearer ear which is the smallest difference in intensity that we can detect at 125 Hz. We are more sensitive to intensity differences at low frequencies than at high frequencies.

Figure 2. Kitty ears.

Our ears are positioned so that the difference in intensity is magnified at high frequencies where the dimensions of the head cast a sound shadow. This guarantees that the difference in sound intensity from left to right will always be at least 8 dB at about 2000 Hz regardless of the distance from the source

In most animals the head shadow seldom applies because their ears are mounted high on the head. However, animals can do something we can't – they can twist their ears to focus on a sound source. We would have to move the whole head from side to side to achieve the same result. I know a blind man who does exactly that to enhance his perception of the environment. When he enters a room he points his head slightly downwards and waves it from side to side. He talks a lot, makes swishing sounds with his mouth and clicks his fingers. His ability to locate walls and furniture is quite remarkable. The reason for lowering his head is probably because human ears are focused 30 degrees upwards from the horizontal plane at high frequencies.

Time differences at the ears

In addition to intensity differences there is a difference in the time that sound arrives at the ears and we also use that to determine the direction of sound. At high frequencies it is the onset of the signal that we detect, at low frequencies it is the peaks in the waveform, or phase, that we detect.

In the case of intensity differences the head shadow amplifies the difference between the two ears. In the case of arrival time or phase there is a clever system that seems to give precedence to whichever ear hears the sound first. It has been suggested that the ear that hears the sound first sends an inhibiting signal to the opposite ear. For such a system to work we might assume that the nerve signal would have to get to the opposite side before the sound gets there, but therein lies a problem. The problem is that sound travels at 340 meters per second but neurons normally transmit signals at only 100 meters per second. Some quite clever systems have been proposed for how the hearing system gets around this obstacle.

Figure 3. Relationship of intensity and phase (from van Bergeijk et al[4]).

To determine how much the ear that hears the signal first attenuates the opposite ear audiological experiments have been done. They show that the intensity must be increased about 8 dB to offset a time difference of 2 mSec.

In fact it is not quite that simple. The effect of time delay only works for short duration or impulse signals. For continuous signals time of arrival is a meaningless concept. At low frequencies the HS treats the peaks in the wave as a series of sound pressure increments but at high frequencies the neurons can't respond fast enough to follow the instantaneous pressure variations. Experiments with pure tones prove that we cannot detect interaural phase differences at high frequencies. However, the HS can use high frequencies to detect interaural phase differences in complex signals such as speech.

Characteristics of speech

Figure 7 shows the wave form of the vowel */ae/ as in "back"* (American male voice). The raw signal is at the top. The centre trace shows the frequency components above 3000 Hz and the lower trace shows the components below 250 Hz.

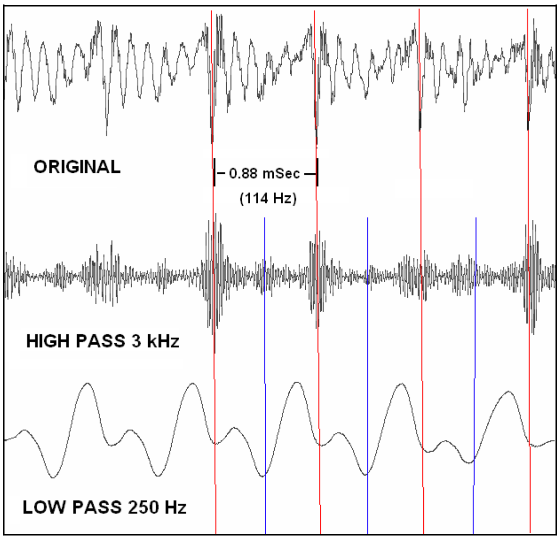

Figure 4. Characteristics of the vowel */ae/.*

When the vocal chords vibrate they do not produce a perfect sinusoidal wave. In this example the fundamental frequency of the vowel is 114 Hz but the uneven motion of the vocal chords produces a second harmonic at 228 Hz (lower trace). Superimposed on the distorted fundamental frequency are the high frequency formants (centre trace) which vary in intensity according to the instantaneous position of the vocal chords. The red and blue markers are inserted to show how closely the high frequency bursts are synchronized with the fundamental wave.

The synchronization of the high frequency envelope with the low frequency wave is an important characteristic for localizing speech because the HS is able to compare the phase of the high frequency envelope with the phase of the low frequency waveform. If either the low frequencies or the high frequencies are missing the ability to locate and identify the voice is impaired.

People with high frequency hearing loss invariably complain that they have abnormal difficulty in noisy or reverberant conditions. If only low frequencies are audible the sound image is relatively broad compared to normal hearing. When the HS can compare the high frequency envelope with the low frequency waveform it is easier to separate a given voice from a background of other voices.

It is fair to question how much the low frequencies can contribute to speech localization if the high frequency bursts alone supply both intensity and timing clues. By playing a recording of the */ae/* vowel with frequencies below 3000 Hz removed in a slightly reverberant room I noticed that the sound image was not only broader but the sound sometimes seemed to come from the wrong direction altogether. As soon as the low frequencies were replaced the sound image was sharp and it was easy to tell if the sound was coming from the front or the back.

In a similar fashion we can tell if a sound is coming from above or below the ears because the ears focus on high frequencies about 30 deg. vertically as well as 30 deg. from the front laterally. That's probably why in difficult conditions we tilt the head slightly downwards as well as to one side.

You may be interested to know that in spite of his large eyes the Barn Owl relies heavily on directional hearing to hunt in the dark. He can determine both the direction and elevation of his prey in spite of the fact that he does not have a pinna as we do. He has one ear positioned lower than the other and the lopsided arrangement allows him to tell the vertical angle of a sound.

Cross-correlation in the hearing system

If all we could do with stereophonic hearing was tell the direction from which a sound arrives it would not be much help in a noisy world. However the human brain is able to use the information from two ears to extract a desired signal from a noise background. The cat does it by moving its ears – we do it by steering our hearing in the brain. For example, sitting in a restaurant I can eavesdrop on a conversation at the table on my left, then without moving my head I can switch my attention to a table on the opposite side. While concentrating my attention on a given location I actually do not hear the other sounds, although it would be more correct to say that I am not listening to them.

It seems that the ability to extract a desired signal from a noise background is due to the brain's ability to compare complex signals from two ears. It does it so well, in fact, that signals that would be masked by noise in a single ear become audible when we hear the noise in both ears.

The effect known as binaural release from masking can be demonstrated with headphones. A pure tone is presented to both ears with equal loudness. Then a masking noise with equal loudness in each ear is introduced and increased until the tone is no longer audible in either ear. If the tone is then switched off in the left ear it becomes audible in the right ear, but only as long as the noise is still present in both ears. In other words, adding the noise to the opposite ear actually reduces the masking effect.

Binaural release from masking works only if the same noise is presented to both ears. If two random noise generators are used the experiment fails because the brain cannot find any correlation in the noise. In the real world we can use the binaural release from masking effect because the same noise is heard in both ears.

Time delay in digital hearing aids

If you have a digital "Home Theatre" system you can try an interesting experiment. Connect the output from the TV set to the Auxiliary input of the digital sound system then listen to the sound from the TV loudspeaker at the same time as the sound played through the digital music system. You will hear that there is a delay of about 200 mSec in the digital system due to the time it takes to do the signal processing.

The circuits in hearing aids must work in real time and are a lot faster but I have been told that some of the more sophisticated circuits take about 10 mSec to get the sound from the microphone to the receiver. That means the hearing aid user only hears the sound when it has travelled some 3 meters beyond him. The best I can do with my own hearing aids is a processing delay of 1.6 mSec. which means that I will only be slightly late when they call me for dinner.

However, a much more serious side effect is likely to occur if I were to wear two different hearing aid models with different time delays. Likewise, if one ear is normal some weird effects might occur due to phase shifts in one ear caused by the time delay.

A personal experience

As a result of an ear infection I developed a significant middle ear hearing loss in my right ear. Without amplification it is virtually useless. The left ear has a borderline cochlea hearing loss that requires only about 8 dB of amplification over most of the speech frequency range. It is one thing to learn from theory and laboratory experiments but after a lifetime of working with hearing aids I finally had the tools to test almost every aspect of hearing aid use in the real world. I fitted two hearing aids and was able to experience what I've been doing to other people for the last 49 years.

The first experiment

My first experiment was to compare two hearing aids with a single hearing aid on either the right or left ear. I noticed that the sound from the TV was not as clear with one ear as with two. Best results were obtained when neither ear sounded "dominant", even if the frequency response sounded different in the two ears. I had the distinct impression that when the loudness sensation in the two ears was balanced the room acoustics improved. Because of the severity of the hearing loss on the right ear I could not get a complete correction on that ear with my very small hearing aid and best results were obtained when the level in the better ear was reduced to match the loudness on the bad side. I judged that if I had to make a choice between correcting only the left or right ear I got better results with a less than ideal correction on the bad ear rather than a good correction on the better ear. In other words it seemed that the stereophonic effect was more important than perfect amplification in real world conditions with reverberation and background noise.

The second experiment

In order to test the importance of phase response I tried reversing the receiver wires on one hearing aid so that the two hearing aids would have exactly opposite phase responses. I listened to a familiar stereophonic recording of Strauss waltzes on a good sound system.

Initially it was obvious that something was different but it took me a little while to put my finger on it. With the phase reversed at the two ears the deep bass from the sub-woofer was missing but there was also something else that caused a slightly disturbing sensation - it was not possible to tell where the sound was coming from.

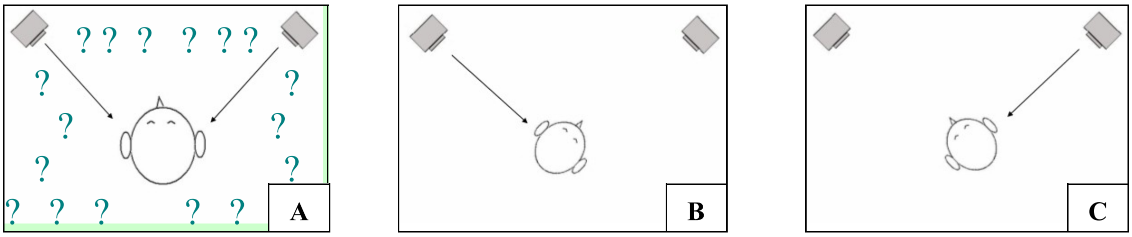

Figure 8. Phase reversed at one ear. A=Looking to front. B=Looking right. C=Looking left.

If I looked towards the right loudspeaker the sound appeared to be coming only from the left speaker. When I turned my head towards the left speaker the sound appeared to be coming only from the right speaker. If my head was pointed at a position midway between the loudspeakers the sound appeared to be coming from everywhere.

It seems that the reversed phase creates a situation so unnatural that the hearing system simply does not know how to deal with it. When the ears are positioned so that the intensity is greater at one ear it has only the sound intensity to work with so it assumes that the sound is coming only from that side.

After using the hearing aids with reversed phase over a weekend I corrected the wiring so that the phase was the same in both ears. When I listened to the music of Herr Strauss after the reversed phase experience the result was even more fascinating. Now it made no difference where I pointed my head – the instruments were always located in the same position - somewhere between the two loudspeakers. The perception of their location was totally independent of the angle of my head.

I subsequently experimented with reversed phase using a single loudspeaker producing a 200 Hz pure tone and a speech weighted composite noise which is a series of speech weighted clicks. I experienced the same effect as with the stereo music system. After restoring the phase response so that it was the same in both aids the ability to localize sound returned quite dramatically for both the 200 Hz tone and the speech weighted composite noise.

The sensation of occlusion

One of the more serious problems in hearing aid fitting is the disturbing “blocked ear” sensation that is caused by trapping sound from the user’s own voice in the ear canal. Low frequency sound vibrations from the larynx get to the ear canal via the cartilaginous tissues. Normally they would leak out of the open ear but if the ear is closed by a hearing aid the low frequency sound pressure is increased about 15 dB at the eardrum. Making a vent in the aid to leak away the low frequencies can help but if the vent is too large there will be acoustic feedback and the hearing aid will oscillate.

It occurred to me that with a single hearing aid part of the unpleasantness was due to the sound of my own voice being localized in my head on one side. I wondered if it would be possible to adjust the hearing aid in a way that would shift the position of my voice so that it would not sound as if it was in my head. Changing the gain at low frequencies had little effect but when I increased the high frequency gain to get the right balance between low and high frequencies my own voice moved out of the head towards my mouth. It was a lot less disturbing but it still sounded as if everything was coming from one side. It needed similar treatment on the opposite ear to make my own voice sound more natural, as if it were getting to the ears mainly from the mouth. Since I need about 8 dB of gain in the low frequencies anyway, I only needed to set the high frequencies about 6 dB too high to get the right balance.

Anyone who has listened to a stereophonic recording with headphones will know that the sound seems to be located left or right within the head. In their paper on externalization of sound Hartmann and Wittenberg[2] describe how it was quite possible to localize the source at the correct position with headphones provided the intensity and phase in the ear canal was the same as when the sound came from loudspeakers.

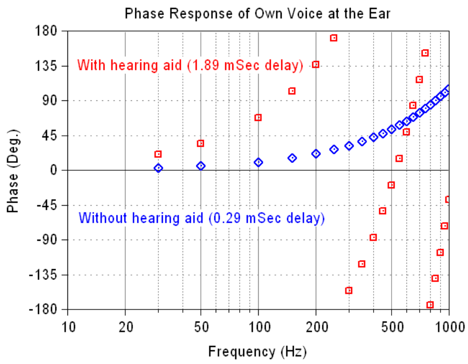

Figure 9. Phase of own voice at the ear.

Correcting both the level and the phase of my own voice at all frequencies was complicated by the time delay in the hearing aid. Figure 9 shows the normal phase response of one's own voice (blue) that is due to the time it takes for the voice to get from the mouth to the ear. It is delayed about 0.3 mSec relative to the sound trapped in the canal by the hearing aid. With the hearing aid the sound from the mouth is delayed by an additional 1.6 mSec. (red) which causes a 180 deg. phase reversal near 300 Hz.

Figure 10 shows the calculated low frequency phase errors caused by the delay in the hearing aid. With normal receiver wiring the phase is correct within 45 deg. only below the speech frequencies and near 600 Hz. When the receiver wires are reversed the phase is correct within 45 deg. from 235 to 375 Hz and from 840 to 1000 Hz.

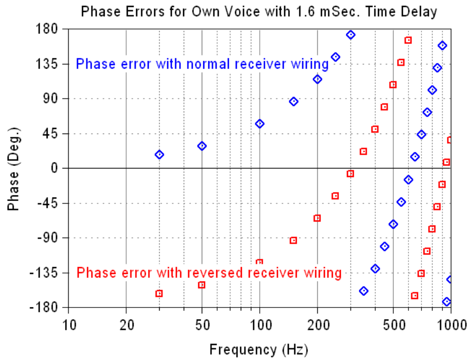

Figure 10. Phase errors with 1.6 mSec delay.

With reversed receiver wiring I noticed an improvement in externalization of my own voice - before I did the calculations. Hartmann and Wittenberg[2] noted that "getting the IPD (Interaural Phase Difference) correct in the 400- to 600-Hz range is effective in keeping the sound image out of the head."

In our efforts to match target gain curves derived from pure tone threshold audiograms we use very versatile digital filters. There are several ways to modify the frequency response with a digital filter. Some filters are fast, some are slow, some preserve the phase response and some don't. In their efforts to provide versatile frequency response adjustments hearing aid manufacturers will more often than not ignore the phase response. My own hearing aids have a digital filtering system that is very versatile but it causes phase errors. Matching the frequency response of the hearing aids up to 1000 Hz was the only way I could get the low frequency phase response matched. The phase improvements were only about 45 degrees but I became more comfortable with my own voice and there was a noticeable sharpening of the sound image. The phase differences above 1000 Hz are difficult to control but that is probably not important because the HS senses the phase of intensity variations in the wave envelope at high frequencies.

Optimizing the correlation coefficient

To be able to converse in a background of speech noise we must not only be able to focus on the desired voice, we must also be able to identify it by its "colour" (timbre). To both localize and identify sounds the HS correlates (compares) frequency content, loudness and phase.

All three parameters are necessary for accurate localization and identification of the source. For any given parameter the correlation coefficient is 1 when there is a perfect match and 0 when the signals have nothing in common. With a correlation coefficient of 1 (perfect coherence) for all three parameters a hearing aid user should hear just as well as a normal hearing person in noisy conditions.

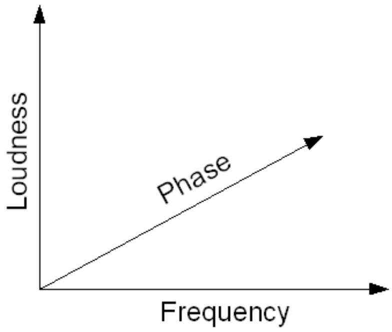

Figure 11. Parameters for speech identification and localization.

To identify a voice the HS looks for frequency content and phase coherence across frequencies. To separate the identified voice from noise and reverberation the HS compares the sound at both ears so that it can focus on the direction and distance of the voice. At the target position the bilateral correlation coefficient will be highest for frequency, loudness and phase. At all other positions it is possible to have a high correlation coefficient for one parameter, say loudness, but not all three.

With my own hearing aids I used the 2 cc coupler to match the phase at all frequencies below 1000 Hz and it produced quite sharp images at 0 deg. incidence and from 60 deg. to 90 deg. on both sides but the image was somewhat blurred (diffuse) at 30 deg. from the front on both sides. I like to describe this as “Auditory Astigmatism”. To sharpen the image at 30 deg. requires bilateral intensity matching at frequencies above 1000 Hz but in the case of hearing loss we cannot do this with objective measurements because it is the perceived intensity (loudness) that is important.

I used a “white noise” with constant phase as the test signal (see Appendix) and was required to match the loudness and <u>timbre</u> at an angle of 60 deg. where the angle of the head is not too critical. I surmised that if I could get the two ears to sound the same the relative loudness levels of the frequency components should be reasonably well matched.

This result was quite encouraging. The sound image became sharp and correctly localized at all angles of incidence. I found that I could afford to reduce the gain slightly which made the aids more comfortable for loud signals.

Summary

Some of the effects that have been described were based on the experience of just one person who could have been biased, which means that they do not meet the criteria for scientific investigation. Nevertheless, effects such as auditory astigmatism and occlusion reduction appear to be quite acute and methods to correct them with minimal training of subjects appear to be feasible.

The effect of time delay and phase distortion in hearing aids appears to be grossly underrated. Considering that bilateral phase correlation is known to be a major factor in sound localization[2] and speech discrimination in competing speech noise[1] it seems incredible that it has received virtually no attention in the design of modern hearing aids.

We might question how much improving signal-to-noise ratios with directional microphones is spoiling the natural ability of the HS to exclude the residual noise. Compression systems are used to improve the dynamic range of the ear but their finite attack and release times cause phase errors in the high frequency wave envelope. Correcting the frequency response below 1000 Hz may also cause phase imbalance and poor correlation with the high frequency wave envelope if the hearing loss in the two ears is different.

Finally, we must question whether we should be allowing hearing aids to decide what or who we should hear. Imagine the possible frustration when you’re trying to listen to the soothing music of Mr. Strauss and the hearing aid gives precedence to the human voice (Figure 12)..

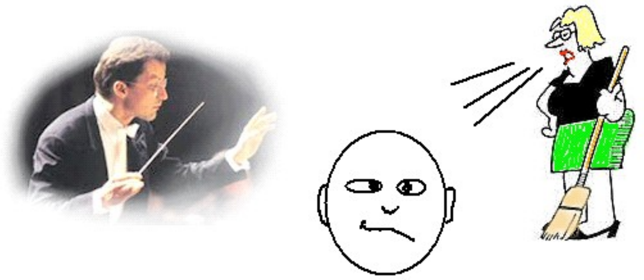

Figure 12 Directional Dilemma.

Appendix

To test the sharpness of acoustic images a single small loudspeaker with a diameter of 45 mm was used. That size was chosen because it is close to the size of the human mouth. The “Constant Phase White Noise” was synthesized by a series of Dirac pulses as illustrated in Figure 13. The high frequency limit is determined by the duration of the flat-topped pulse and the low frequency limit is determined by the time interval between pulses. The phase spectra of the random and constant phase signals are compared in Figure 14. For the Dirac series the phase is 90 deg. at all frequencies.

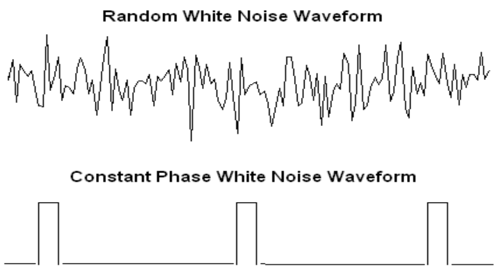

Figure 13. Comparison of random and constant phase (Dirac series) waveforms.

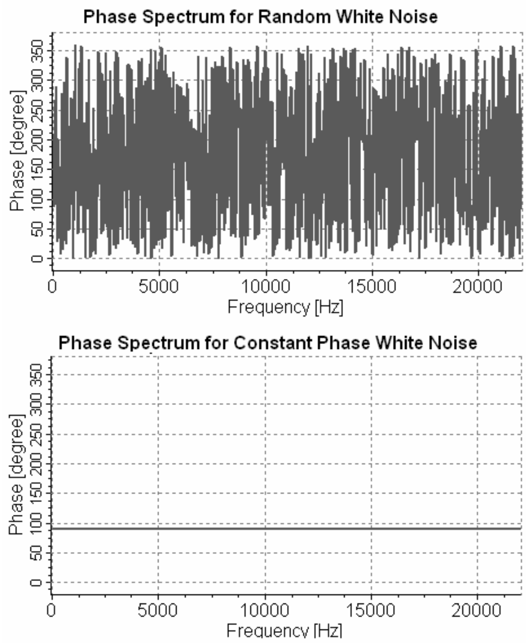

Figure 14. Phase spectrum of Dirac series compared with random white noise.

According to Ohm's acoustical law the relative phases of the component frequencies in a complex wave should not affect the quality of the sound. This is indeed true for signals consisting of only a small number of frequencies.

For signals with a large number of frequencies the phase relationships can have a profound effect on the Crest Factor (CF), which is the ratio of the Peak Level to the Root Mean Square (RMS) level. For a signal such as the Dirac series the CF is typically about 16 dB but it is dependent on the bandwidth of the signal. For a signal consisting of only two frequencies phase shifting cannot change the CF by more than 1.8 dB and we would have difficulty hearing the difference. The Dirac series has a 16 dB CF and the random white noise has a CF of 9 dB. They sound quite different. In addition the relatively high CF actually enables the hearing system to identify the 20 Hz component that cannot be reproduced by the small loudspeaker.

In spite of the fact that the random and constant phase signals sound quite different their frequency spectra are the same - as shown in Figure 15. It seems that phase may be playing a much more important role in hearing than is generally assumed.

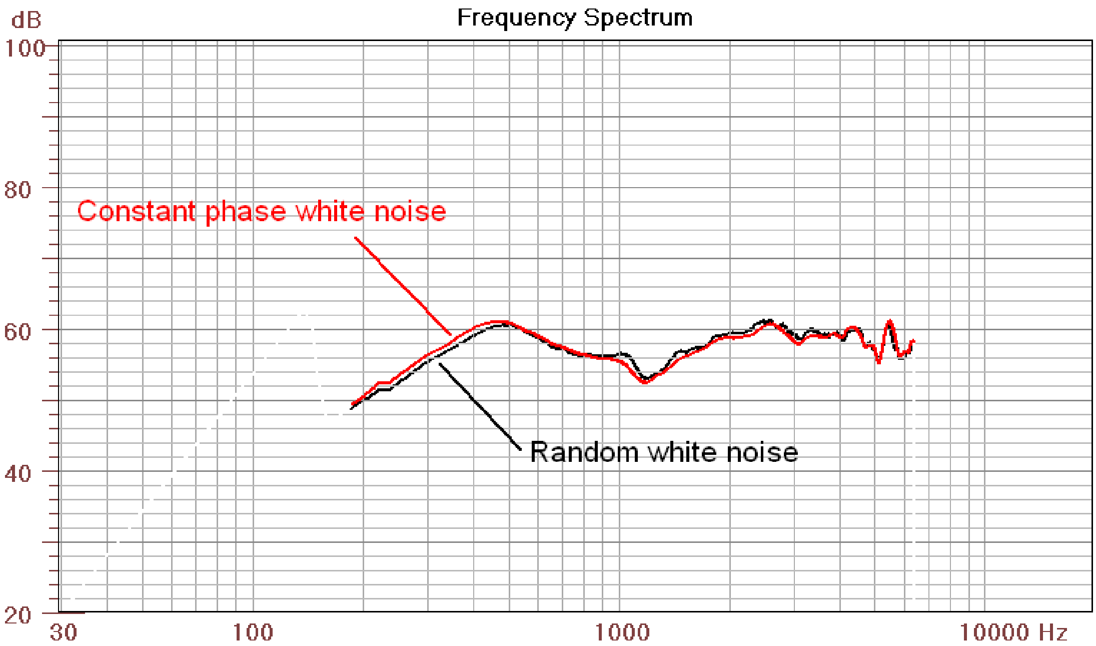

Figure 15. Measured frequency spectra of random white noise and 45 µSec Dirac pulses at 50 mSec intervals as reproduced by a 45 mm diameter loudspeaker (average of 16 samples).

Acknowledgements

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