LOW COMPLEXITY VIRTUAL BASS ENHANCEMENT ALGORITHM FOR PORTABLE MULTIMEDIA DEVICE

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Earbuds have gained immense popularity with mobile multimedia systems recently owing to their convenience and easy low cost design. Unfortunately their design is subject to severe constraints especially affecting their low frequency signal performance. Bass signal performance contributes significantly to the user perceived sound quality and a good bass signal reproduction is essential. Increasing the sound energy in the bass signal range is an unviable solution since the gain required are exceedingly high and signal distortion occurs because of speaker overload. Recently methods are being proposed to invoke low frequency illusion using psychoacoustic phenomena of the missing fundamental. This paper proposes a simple and effective signal processing method to create bass signal illusion using the missing fundamental effect, at a complexity of 12 MIPS on Motorola 56371 audio DSP ideal for portable device implementation.

INTRODUCTION

Conventional Earbuds cannot reproduce high-quality bass sound because of the light mass of the speaker unit and size constraints. Figure 1 shows a typical Earbud speaker frequency response. Clearly the Bass frequency response is highly attenuated. The response drops below -20 dB for deep bass frequencies. With an increasing amount of low frequency content in popular music and film soundtracks, the consumer demands excellent Bass reproduction. While new developments are being made in the areas of physical design and materials recently the techniques of human perception are being applied to the goal of efficient earphone design. Such psychoacoustic techniques have been applied to the areas of audio coding as MPEG audio and Dolby AC-3 schemes, but use these techniques in speaker reproduction is very recent. Our work uses similar techniques to enhance the Bass Signal reproduction. Section 1 describes the missing fundamental effect and existing methods of Bass Enhancement. Section 2 describes our new developed harmonic generation method followed by overall DSP implementation results in section 3 and conclusions in section 4.

1 PSYCHOACOUSTIC BASS BACKGROUND

1.1 Missing Fundamental Effect

Pitch is a subjective, psychophysical quantity. For a pure tone, where the fundamental frequency corresponds to the frequency of the tone, the pitch is unambiguous and one can identify pitch with the frequency of the pure tone. For a complex tone, consisting of more than one frequency, pitch should then be measured by psychophysical experiments.

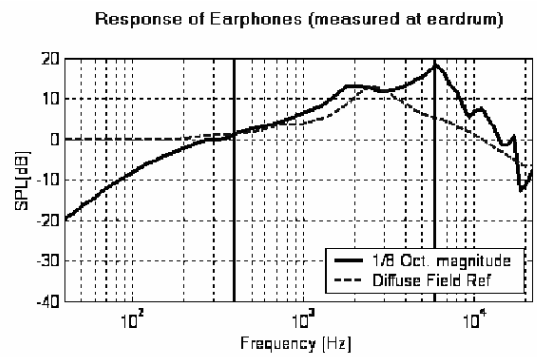

Figure 1: Earbud frequency response.

A pitch that is produced by a set of frequency components rather than by a single frequency is called a Residue. In Figure 2 the fundamental frequency is missing, yet will still be perceived as a residue pitch, which in this case is also called virtual pitch. The psychoacoustic phenomenon responsible for this effect is the "Missing Fundamental" effect, that the pitch perception of a set of harmonics is that of the fundamental frequency. Alternatively if we can produce the harmonics we can perceive the fundamental frequency. This phenomenon is used to improve the Bass frequency perception in our solution. If we can produce harmonics for each frequency in the Bass frequency we can perceive the Bass signal psychoacoustically.

1.2 Psychoacoustic Bass Enhancement Methods

Psychoacoustic Bass Enhancement methods involve extracting the low frequency signal below the speaker cutoff frequency, generating harmonics for each frequency component in the low frequency signal and

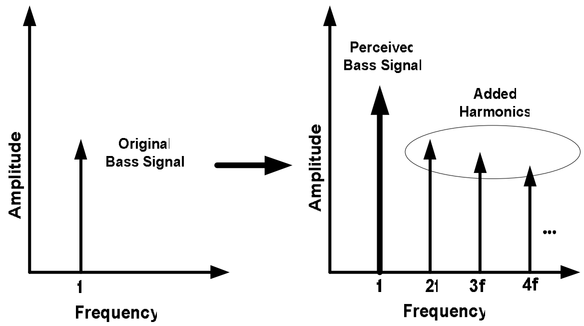

Figure 2: Pitch perception of a complex tone (Missing fundamental effect).

adding it back to the original signal. The original signal is usually high pass filtered above the speaker cutoff frequency to remove the low frequency component because it is anyway not reproduced acoustically by the transducer. The approach is shown in figure 3 (from [1]). The simplest method to produce harmonics of an input signal is to perform a non-linear operation on the signal. All non linear operations produce harmonic frequencies, the amplitude of harmonics depending on the type of non linearity used. The filters HFIL in figure 3 are high pass filters. The left and right channels are added together and the low frequency signal is extracted from the combined data using FIL1. The NLD portion is the nonlinear harmonic generation portion. The FIL2 is used to filter out DC and harmonics or distortion created in the bass frequency range. FIL2 can also be used to shape the harmonic structure produced by the NLD. After that a gain is applied and the harmonic signal is added to the high pass filtered signal.

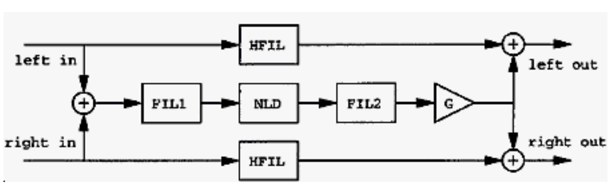

Figure 3: Psychoacoustic Bass Enhancement Method.

The method used to generate the harmonics i.e. the non linearity used is a topic of research and various methods have been proposed. Amongst the simplest methods used is the full wave rectification of the input signal [1]. Full wave rectification produces harmonic components at 2f, 4f, 6f and so on for the input frequency f. Although this method is very simple but the pitch perceived is 2f rather than f because only even harmonics are produced. [1] Also proposes the full wave integration approach to harmonic generation.

2 PROPOSED HARMONIC GENERATOR

The properties essential for a harmonic generator is that it should produce all (even and odd) harmonics, harmonics should be independent of input signal level, relative amplitudes of the harmonics should be controllable and the method should be very low in complexity. The new proposed harmonic generator satisfying the above described properties using a modified envelope detector has been suggested. The detector does not rectify the signal, has a fast rise time and a slow fall time. When a signal is passed through the signal zero crossings with positive slope are preserved but the points at which the signal goes from positive to negative (negative slope) can be shifted by varying the fall time of the circuit. This processing generates signal harmonics satisfying the desired properties. Figure 4 shows the harmonic signal waveform generated by using various settings of the fall time. Figure 5 shows the effect of varying the fall time on the harmonic signal spectrum envelope. Figure 6 shows the block diagram of the harmonic generator**.** For every sample the input signal is compared with the last output signal. If the input signal is greater than the last output a fast rise constant is selected or else a slow fall constant is selected. Changes in the fall time change the shape of the harmonic spectrum envelope as shown in figure 5.

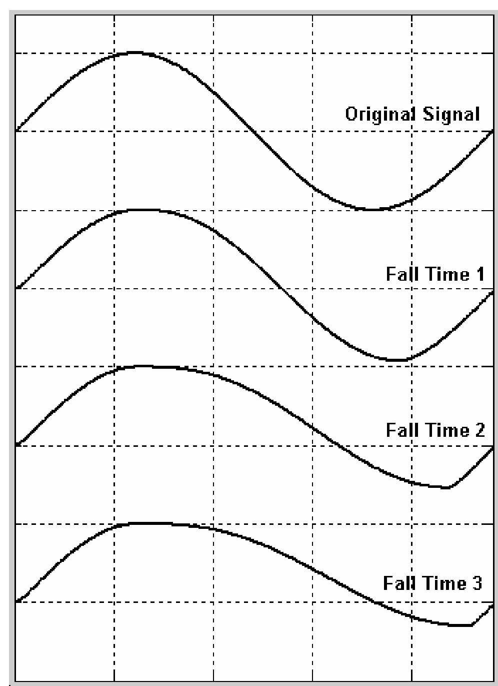

Figure 4: Method 1 harmonic signal waveform using 3 fall times.

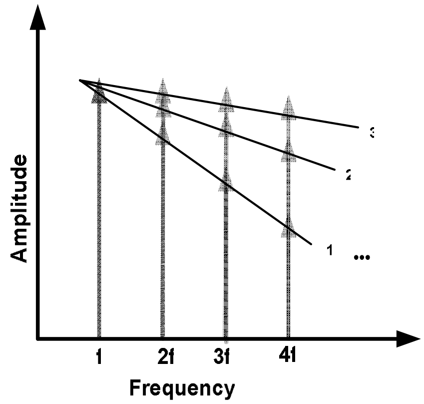

Figure 5: Harmonic spectrum envelope changes using 3 different fall time settings.

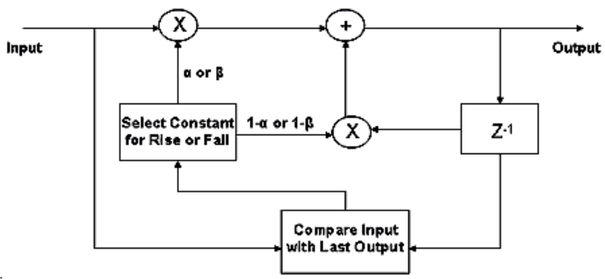

Figure 6: Harmonic generation method 1 block diagram.

An alternate method with clipping for negative waveform portions can also be used for harmonic generation as shown with the output waveforms in figure 7. The method is a modified scheme of the first method in which the output is set to zero whenever the input signal is negative. The output of method 2 is fed to a harmonic shaping filter which changes the harmonic spectrum for the desired Timbre. Figure 8 shows the block diagram of the proposed Method 2.

3 SYSTEM IMPLEMENTATION

The psychoacoustic methods described above were applied to improving the bass frequency range audio signal performance of an earphone system. The speaker cutoff frequency was about 100 Hz with response dropping considerably in the deep bass frequency range. The standard psychoacoustic Bass enhancement method configuration of figure 3 was implemented.

As the first step both the left and right channel signals were combined and low pass filtered to extract the Bass frequencies. The design of the low pass filter is critical. It is best to use a FIR filter because of the sensitivity of low frequency signal to phase distortions. We instead used an IIR filter because of complexity constraints in the overall system. Extreme care was taken in the design of the IIR filter to minimize the group delay. We used the Bessel method to design our IIR filters with almost constant group delay in the pass band.

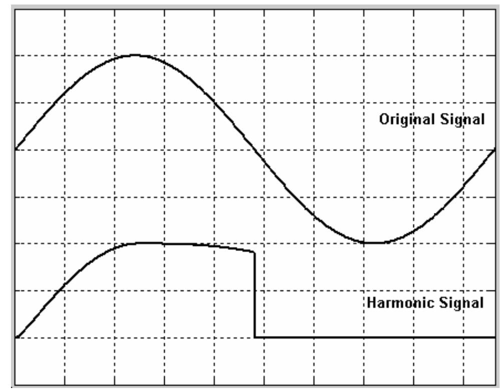

Figure 7: Method 2 harmonic signal waveform.

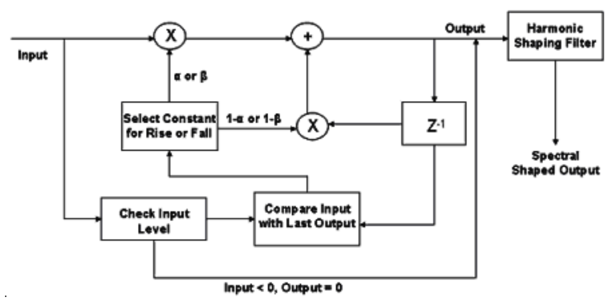

Figure 8: Harmonic generation method 2 block diagram.

The filter order was 4th order Bessel filter. In our filter design experiments we found that cascading lower order filters gives better group delay performance than using higher order filters. After low pass filtering the signal is passed through the harmonic generator as described before. Experiments were done with both Method 1 and Method 2 harmonic generator. The rise time was very fast at 1 msec. The fall time was set at 5 around msecs.

After the harmonic generation the signal was passes through the shaping filter. The harmonic shaping filter (or FIL2 from figure 3) is a band pass filter. The high pass bank frequency was same as the cutoff of the earphones at about 100 Hz. The high pass bank is essential to remove the low frequency intermodulation distortion that might have been generated because of the nonlinear process. The low pass bank frequency was about 300 Hz. At this value at least first 2 harmonics would be present for all the bass frequencies which are sufficient for psychoacoustic bass perception. Increasing the low pass bank cutoff frequency to beyond 300 Hz would lead to a much sharper timbre which can also sound distorted. 4th order Bessel IIR filters were also used for both the high pass and low pass bank of the harmonic shaping filter.

The generated harmonics are added back to the left and right channels after giving a suitable gain. The bass frequency components are removed from the left and right channels using high pass filters HFIL. A 2nd order Butterworth filter was used for the high pass filtering. The high pass filtered left and right channels are also delayed to compensate for the approximately constant group delay introduced by the filtering of the bass signal and its harmonics.

The system was implemented on the Motorola 56371 24 bit audio DSP. The complexity of implementation at 48 KHz sampling was 12 MIPS including low pass and high pass filtering to separate bass frequencies, harmonic generation algorithm, harmonic shaping filtering and high pass signal, harmonic signal addition operations. The data memory used was 1.7K words and the program memory was 0.5K words.

Extensive listening tests were subsequently performed on the developed Bass Enhancement method. A good effect could be observed in most sequences. Slight distortion was observed in music samples having multiple bass instruments which was attributed to the intermodulation distortion produced during the nonlinear operation. Also in a few cases where the bass signal has a sudden increase in amplitude (such as that with a bass drum) phase distortion issues were observed. The generated psychoacoustic bass seemed time misaligned with the rest of the high pass signal. Perhaps a better design of the IIR filter to reduces nonlinear phase effects would be good although as previously stated its best to use a FIR filter if the complexity is acceptable.

4 CONCLUSIONS

This paper describes the development of a psycho acoustical bass enhancement method applied to the enhancement of sound quality in an earphone speaker system. New harmonic generation methods were proposed in our work and the system design was described in details. The method was implemented on Motorola DSP 56371 at a complexity of 12 MIPS.

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[4] U.S. Patent No. 5930373 Shashoua et al.

[5] U.S. Patent No. 6111960 Aarts et al.

[6] U.S. Patent No. 6285767 Klayman.