

Methods and Devices for Preprocessing signals for a loudspeakerField of the Invention

5 The present invention relates to methods and devices for preprocessing signals before they are transmitted to the transducer of a loudspeaker. In particular, the loudspeaker may be a directional ultrasonic loudspeaker for generating a directional beam of ultrasonic vibrations modulated by an audio-frequency signal.

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Background of the invention

Audio signals generated by conventional loudspeakers are usually almost omnidirectional (i.e. the vibrations are not emitted from the loudspeaker in a narrow angular range of directions). Directional emission of medium and low audio frequencies is only possible using an array of loudspeakers having a complex control arrangement.

However, it is well-known (see references [3]-[5]) that by transmitting a strong modulated ultrasonic wave in air, the nonlinear characteristics of air can automatically demodulate the envelope of the modulated ultrasonic signal. Since the ultrasonic wave can be easily focused to a narrow beam, the reproduction of the modulating signal can be confined by this narrow beam. If the modulating signal is audio signal, a narrow audio beam is formed. This is the basic mechanism of a directional loudspeaker.

In practice, the demodulated signal is not directly proportional to the envelope of amplitude modulation (AM). Based on the nonlinear theory of acoustics, it was shown by Berktaý that for an AM signal and under certain simplifying assumptions that if a primary wave of pressure $p_1(t)$ is transmitted, after the air demodulation, a secondary wave $p_2(t)$ is generated. The $p_1(t)$ and $p_2(t)$ are shown as follows:

$$p_1(t) = P_1 E(t) \sin(\omega_c t) \quad (1)$$



$$p_2(t) = \frac{\beta P_1^2 A}{16\pi\rho_0 c_0^4 z \alpha} \frac{\partial^2}{\partial t^2} E^2(\tau) \quad (2)$$

where P_1 is the amplitude of the pressure of primary wave, $E(t)$ is the modulation envelope, ω_c is the carrier frequency, β is the coefficient of non-linearity, ρ_0 is the ambient density of the medium, c_0 is the small-signal wave propagation speed, A is the beam's cross section area, z is the axial distance, α is the absorption coefficient of the medium (at frequency ω_c), and $\tau = t - z/c_0$ is the lag time. The relationship between modulation envelope $E(t)$ and the audio signal $f(t)$ is:

$$E(t) = 1 + mf(t) \quad (3)$$

where m is the AM depth. The main function of the pre-processing is to alleviate the nonlinear effect of equation (2) on $E(t)$, i.e. by pre-processing the $f(t)$ before the amplitude modulation, the secondary wave output should be made directly proportional to $f(t)$. This can be achieved by generating a modified version of $E(t)$ as:

$$E'(t) = [1 + \iint f(\tau) d\tau^2]^{1/2} \quad (4)$$

This seemingly simply pre-processing is very difficult to implement in practice. The main difficulty arises from the square-root operation. Because it is a nonlinear operation, it will increase the signal bandwidth vastly. This poses a very strict requirement for the bandwidth of the circuit and in particular of the ultrasonic transducer. It is very difficult to make a transducer simultaneously with wideband and high power-efficiency. The double integration is also difficult to implement due to the easy saturation of analog integrations.

To avoid directly using (4) and at the same time to reduce the nonlinear interference among different frequency components caused by square operation of (2), it is suggested in [3] to use a small modulating index m , but this

will reduce the power of the reproduced audio signal and therefore the power efficiency of the whole system. In [4], it was suggested to follow Eqn. (4) exactly, thus the problems mentioned in above would appear. As a compromise method, [7] suggested using single sideband (SSB) signal to replace the square-rooted double sideband (DSB) signal as the modulating signal. This method has the advantage of reducing the required frequency bandwidth greatly as compared with (4) and reducing the internal self-interference as compared with the direct DSB signal (i.e. Eqn. (3)), but this saving comes at the expense of a decrease in sound quality.

Preprocessing the SSB signal, to cause it to approach the ideal envelope of the square-rooted DSB signal can increase the audio fidelity, but the pre-processing incurs a significant processing cost.

Summary of the Invention

The present invention seeks to provide new and useful methods and devices for preprocessing audio signals before they are modulated onto an ultrasonic carrier wave and transmitted to the ultrasonic transducer of the loudspeaker.

In general terms, the present invention proposes that the audio signals are processed in frequency space by weighting each of the frequency components by a predetermined weighting coefficient. Furthermore it is ensured that the positive frequencies, or alternatively the negative frequencies, are removed before the audio signal is converted back to the time domain, so that when the ultrasonic signal is modulated by the preprocessed audio signal, a single sideband (SSB) ultrasonic signal is obtained.

The weighting operation proposed here may be performed by a relatively simple processing operation, so that the processing requirement of an embodiment of the invention may be low. As discussed below, the weighting operation is sufficient to provide effective, precise and stable preprocessing at low

computational cost, and without producing a modulated ultrasonic signal which requires the transducer to have a wide bandwidth.

Specifically, a first expression of the invention is a method of operating an ultrasonic loudspeaker, the method comprising preprocessing an audio signal to generate a preprocessed signal, using the preprocessed signal to modulate an ultrasonic signal, and using the modulated ultrasonic signal to drive a transducer of the loudspeaker, the preprocessing of the audio signal including:

- transforming the audio signal into the frequency domain,
- weighting different frequency components of the transformed signal according to corresponding weights stored in a weights table;
- removing either the positive or the negative frequencies; and
- transforming the signal from the frequency domain into the time domain.

Note that this expression of the invention is not intended to imply an ordering of the steps of weighting the frequencies and removing the positive or negative frequencies. Rather, these two steps may be performed in any desired order, or even performed simultaneously.

A second expression of the invention is a preprocessing device for a loudspeaker, the preprocessing unit being operative to generate, from an audio signal, a preprocessed signal for modulating an ultrasonic wave, the preprocessing device comprising:

- transform means for transforming the audio signal into the frequency domain,
- weighting means for weighting different frequency components of the transformed signal according to corresponding weights stored in a weights table;
- means for removing either the positive or the negative frequencies; and
- means for transforming the signal from the frequency domain into the time domain.

As a first example of the value of the weighting operation, note that since a single sideband signal with proper AM depth is used, the square-root in Eqn. (4) can be avoided. In this case, in frequency space, the double integration in Eqn. (4) can be represented by a division of the frequency components by a factor
 5 proportional to the square of the corresponding frequency.

As a second example of the value of the weighting operation, the frequency components each be weighted by a corresponding numerical value which compensates for any frequency dependence of the ultrasonic transducer of the
 10 loudspeaker.

Although these two weighting operations are in principle logically independent, preferably they are combined, so that each frequency component is weighted by a weight proportional to $1/\omega^2 C(\omega + \omega_c)$, where ω is the frequency and
 15 $C(\omega + \omega_c)$ is a measure of the frequency response of the transducer at a frequency ω above a basic loudspeaker resonant frequency ω_c .

Preferably, the preprocessing is preformed digitally, but the preprocessing device is preferably capable of receiving signals which are any combination of
 20 analog and/or digital signals. An analog-to-digital converter is provided for converting any analog signals in the input into a digital format. Similarly, the output section of the preprocessing device preferably includes a digital-to-analog converter for generating a digital signal for use in the modulation.

25 Brief description of the Figures

Further preferred features of the invention will now be described in detail with reference to embodiments, which are given for the sake of example only:

30 Fig. 1 shows schematically the overall structure of a loudspeaker system which is an embodiment of the present invention;

Fig. 2 shows the preprocessing device of the embodiment of Fig. 1;

Fig. 3 illustrates a possible frequency response of the transducer of Fig. 1;

5 Fig. 4 illustrates a weighting scheme used in the embodiment of Fig. 1; and

Fig. 5, which is composed of Figs 5(a) and 5(b), compares the frequency components of the ideal square-rooted DSB modulated signal with the frequency components produced by the embodiment of Fig. 1 for an input signal
10 with angular frequency ω_0 .

Detailed description of the embodiments

Referring firstly to Fig. 1, a loudspeaker system which is an embodiment of the
15 invention includes a preprocessing device 10 which receives at least one audio signal and pre-processes it, a modulator 20 which receives an ultrasonic carrier wave generated by an oscillator 30 and amplitude modulates it (e.g. according to Eqn. (1)) using the preprocessed audio signal, and a directional ultrasonic transducer 40. The preprocessing device 10 is preferably arranged to be
20 capable of receiving either or both of analog or digital audio signals.

The structure of the preprocessing device 10 is shown in Fig. 2. It includes an input section 11 and a preprocessing section 12. The input section 11 comprises a filter and analog-to-digital converter 111 for receiving any input
25 analog signal, and a decoder 112 for receiving any input digital signal (e.g. from a compact disk player or MP3). In either case, the output of the input section 11 is a digital audio signal, which is passed to the preprocessing section 12.

The digital audio signal from the input section 11 is processed first by a Fourier
30 transform unit 121, which is preferably a Fast Fourier transform (FFT) unit. The Fourier transform unit 121 converts the audio signal $f(t)$ in the time domain into $F(\omega)$ in the frequency domain.

Different frequency components of the output are multiplied by a weighting unit 122 by differing respective weights stored in a weights table 123. The setting of the weights in the weights table 123 will now be described with reference to
 5 Figs. 3 and 4. The weights table can be implemented in ROM or in RAM memory, and are preferably pre-computed.

Fig. 3 shows a typical profile of the transducer gain of the transducer 30 as a function of frequency. The profile has a frequency pass width marked as 25,
 10 which is typically approximately centered on a resonant frequency ω_c . This is preferably the ultrasonic frequency generated by the oscillator 30.

Fig. 4 shows how the weight values are selected. The dashed line 35 shows weight values which, if stored in the weights table 123 and used by the
 15 weighting unit 122, would perform the double integration of Eqn. (4) (without the square root). Typically, the value of the weights given by the dashed line 35 falls by about -12dB per Octave.

However, the weights actually stored in the weights table 123 and used by the
 20 weighting unit 122 are more preferably selected to be $1/\omega^2 C(\omega + \omega_c)$, where $C(\omega + \omega_c)$ is a measure of the frequency response of the transducer at frequency $\omega + \omega_c$ as illustrated in Fig. 3. The weights may be zero for frequencies outside the frequency pass band 25 of the loudspeaker, and are preferably zero for all frequencies which will be removed by the unit 124 as
 25 described below, but for other frequencies $C(\omega + \omega_c)$ may vary with (e.g. be equal to) the transducer gain shown in Fig. 3. Thus, the effect of the weighting unit 122 is not only to perform the double integration but also to compensate for variations in the frequency gain of the transducer.

30 Thus, the output of the weighting unit 122 is:

$$G(\omega) = \frac{1}{\omega^2 C(\omega + \omega_c)} F(\omega). \quad (5)$$

The output of the weighting unit 122 is then passed to a unit 124 which obtains the desired baseband signal by operating only on the negative (or positive) parts of $G(\omega)$, and converting that sideband into the time domain by an inverse Fourier transform (preferably inverse fast Fourier transform, IFFT). For example, if we write the negative frequency components of $G(\omega)$ as $G(\omega-)$, then the resulting complex valued signal is:

$$g(t) = g_I(t) + jg_Q(t) = IFFT(G(\omega-)). \quad (6)$$

$g_I(t)$ and $g_Q(t)$ are the real and imaginary parts of the baseband SSB signal respectively, and are shown in Fig. 2 as the outputs I and Q of the unit 124.

Within the unit marked as 125, $g_I(t)$ and $g_Q(t)$ are converted to analog signals by one or two DACs, and the analog forms of $g_I(t)$ and $g_Q(t)$ are filtered, to form the two respective outputs of the preprocessing unit 10.

The modulator unit 20 modulates $g_I(t)$ and $g_Q(t)$ onto the carrier signal and its 90° phase shifted version respectively. The output signal of the bandpass SSB signal is:

$$h(t) = (1 + mg_I(t)) \cos(\omega_c t) + (1 + mg_Q(t)) \sin(\omega_c t) \quad (7)$$

Signal $h(t)$ is further power amplified and used to drive the ultrasonic transducer 40.

Fig. 4(a) shows the ideal square-rooted DSB modulation derived from Eqn. (4) in the case that the audio signal includes only a single audio frequency ω_0 . The result has a high bandwidth with components spaced from the carrier frequency by multiples of ω_0 both above and below.

Fig. 4(b) shows the corresponding result in the case of the present invention. In addition to the carrier wave, there is only a single frequency component of $\omega_c - \omega_0$ (or $\omega_c + \omega_0$).

5 The resulting audio signal has better fidelity than the conventional DSB signal (Eqn. (3)), since self-interference among different frequency components of the upper sideband and lower sideband is removed. It is true that the embodiment provides poorer fidelity than the ideal square rooted DSB, since self-interference among different frequency components within the preserved
10 sideband still exists, but it should be noted that the ideal square-rooted DSB signal is impossible in practice to realize, since as illustrated in Fig. 5(a), it requires an unlimited bandwidth. Overall, the embodiment provides a better compromise between system complexity and audio fidelity than known methods mentioned above.

15 Although the invention has been described with reference to only a single embodiment, many variations are possible within the scope of the invention as will be clear to a skilled reader.

20 For example, the embodiment can be modified to incorporate further processing in the frequency/time domains to increase further the fidelity of the final audio output. For example, this may be performed conveniently in the frequency domain.

25 Furthermore, although invention has been described with reference to processing "units" it is to be understood that these units need not be physically separate, but may for example be realized as sections of software to be operated on any number of physically separate processing units.

30 References

The following references are incorporated in their entirety in this disclosure by reference:

- [1]. Tsuneo Tanaka, Mikio Iwasa, Youichi Kimura and Akira Nakamura.
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- [6]. Elwood G. Norris and James J. Croft III, "Method and Device for Developing
15 A Virtual Speaker Distant from the Sound Source" U.S. Patent No. 6,229,899,
2001.
- [7]. Tomoo Kamakura, Tasahide Yoneyama and Kazuo Ikegaya "Studies for the
Realization of Parametric Loudspeaker" J. Acoust. Soc. Japan, No. 6 Vol. 41,
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Claims

1. A method of operating an ultrasonic loudspeaker, the method comprising preprocessing an audio signal to generate a preprocessed signal, using the
5 preprocessed signal to modulate an ultrasonic signal, and using the modulated ultrasonic signal to drive a transducer of the loudspeaker, the preprocessing of the audio signal including:
 - transforming the audio signal into the frequency domain,
 - weighting different frequency components of the transformed signal
10 according to corresponding weights stored in a weights table;
 - removing either the positive or the negative frequencies; and
 - transforming the signal from the frequency domain into the time domain.
2. A method according to claim 1 in which the weights vary inversely with
15 the square of the corresponding frequency.
3. A method according to claim 1 in which the weights vary proportionally to $1/\omega^2 C(\omega + \omega_c)$, where ω is the frequency corresponding to the weight, ω_c is a constant frequency, and $C(\omega + \omega_c)$ is a measure of the frequency response of
20 the transducer at frequency ω_c .
4. A method according to claim 1, claim 2 or claim 3 in which the audio signal is digital and the preprocessing is performed digitally, the method further including generating the audio signal from an input signal including an analog
25 signal.
5. A method according to claim 4 in which the input signal further includes a digital signal.
- 30 6. A method according to claim 4 or claim 5 in which the preprocessing includes converting the preprocessed digital signal into an analog signal, which is filtered before being used in the modulation.

7. A method according to any preceding claim in which the ultrasonic beam is a directional beam.
- 5 8. A preprocessing device for a loudspeaker, the preprocessing unit being operative to generate, from an audio signal, a preprocessed signal for modulating an ultrasonic wave, the pre-processing device comprising:
- transform means for transforming the audio signal into the frequency domain,
- 10 weighting means for weighting different frequency components of the transformed signal, according to corresponding weights stored in a weights table;
- means for removing either the positive or the negative frequencies; and
- means for transforming the signal from the frequency domain into the
- 15 time domain.
9. A preprocessing device according to claim 8 in which the weights of the weights table vary inversely with the square of the corresponding frequency.
- 20 10. A preprocessing device according to claim 8 in which the weights vary proportionally to $1/\omega^2 C(\omega + \omega_c)$, where ω is the frequency corresponding to the weight, ω_c is a constant frequency, and $C(\omega + \omega_c)$ is a measure of the frequency response of the transducer at frequency $\omega + \omega_c$.
- 25 11. A preprocessing device according to claim 8, claim 9 or claim 10 comprising an input section which receives an input signal, the input section including an analog-to-digital converter for converting any analog component in the input signal into a digital audio signal.
- 30 12. A preprocessing device according to claim 11 in which the input section is further adapted to receive a digital input signal.

13. A preprocessing device according to claim 11 or claim 12 including a digital-analog converter for converting the preprocessed digital signal into an analog signal, which is filtered before being used in the modulation.

5 14. A loudspeaker system comprising a preprocessing device according to any of claims 8 to 13, and further comprising:

a modulator arranged to receive the preprocessed signal, and to modulate an ultrasonic signal using the preprocessed signal to generate a modulated ultrasonic signal; and

10 an ultrasonic transducer for generating a ultrasonic beam from the modulated ultrasonic signal.

15. A loudspeaker system according to claim 14 in which the ultrasonic transducer generates a directional beam.