



Method and Apparatus for Distortion Reduction in Ultrasonic beam audio systems

FIELD OF THE INVENTION

- 5 The present invention generally relates to systems for generating audio sound using a modulated ultrasonic beam. In particular, the invention relates to controlling such a system so as to increase the audio fidelity of the generated sound.

10 BACKGROUD OF THE INVENTION

- When two sound waves, so-called "primary waves", with finite amplitudes and different frequencies interact with each other in air, new sound waves, so-called "secondary waves", are generated. The secondary waves have frequencies which are the sum and difference of the primary waves. Many theoretical and
15 experimental works illustrate that this non-linear phenomenon can be used to produce audio sound (i.e. sound audible to humans) when at least one of the primary waves is ultrasonic (i.e. inaudible to humans). For example, the disclosure of the following references is incorporated here: Mark F. Hamilton, "Sound Beams" in *Nonlinear Acoustics*, Edited by Mark F. Hamilton and
20 Malcolm J. Crocker, Chapter 8, pp. 233–261, Academic Press, 1998, and James J. Croft and Joseph O. Norris, "Theory, History, And The Advancement



Of Parametric Loudspeakers: A Technology Overview," *White Paper*, American Technology Corporation 2001/2002. The ultrasonic primary wave is amplitude modulated by a desired audio signal, and the modulated signal is demodulated by air due to the non-linear fluid properties of air.

5 Berkday studied this phenomenon using the concept of a "modulation envelop" in "Possible exploitation of non-linear acoustics in underwater transmitting applications," J. Sound Vib. 2 (4): 435–461 (1965). Given that the primary wave has the form

$$p_1(t) = P_1 E(t) \sin 2\pi f_c t$$

10 (1)

where $E(t)$ is a function representing the envelop of the primary wave, and f_c is the ultrasonic carrier frequency, Berkday provided a far-field solution in which the primary signal yields a secondary demodulated signal at a point along a transducer's radiation axis having a pressure given by:

15 $p_2(t) \propto \frac{\partial^2}{\partial t^2} E(t)^2$

(2)

The term "far-field" refers here to an area beyond a critical distance from the sound emitter where the ultrasound has decayed dramatically. This solution is also valid for the near field locations (i.e. closer to the sound emitter than the
20 critical distance) but in these locations the ultrasonic primaries coexist with the demodulated signal. As shown by eqn. (2), the demodulated signal is

proportional to the second time derivative of the square of the envelope. The demodulation is nonlinear because of the nonlinear square operation.

One of the fundamental requirements for such an audio system is that, when an
5 audio signal is input to it and used to modulate the ultrasonic primary beam, the
audio signal is reproduced with high fidelity when the beam is demodulated. In
an ideal case, no signal distortion should occur. However, in practice all the
components in the system contribute to signal distortion due to their
imperfections. Furthermore, the nonlinear effect of air demodulation produces
10 other undesired sound components, which make the demodulated signal even
more distorted.

Figure 1 is a schematic block diagram of a known ultrasonic beam audio
system, which generates a beam of audio sound by utilizing non-linear air
15 demodulation. The system comprises a controller 110 and a plant 120. The
controller 110 comprises a signal processor 111 for processing an audio signal
input from the left of Fig. 1, and a modulator 112 for modulating the output of
the processor 111. The signal processor 111 is typically a digital signal
processor. The plant 120 comprises an amplifier 121 for amplifying the power
20 level of the modulated signal output by the controller 110, and a transducer 122
(i.e. a loudspeaker) for converting the amplified signal from electrical form to

acoustic form. The amplifier 121 enhances the signal to such a power level that the transducer 122 generates enough sound power. Ideally, the amplifier 121 should have a flat frequency characteristic in the required frequency range, since any other characteristic will result in signal distortion. Because at least one of the primary sound components has an ultrasonic frequency, the transducer 122 should be able to transmit an ultrasonic frequency. The gain characteristics of the transfer function of a typical ultrasonic transducer is shown in Figure 2. Another requirement for the transducer 122 is that it includes a high enough resonant peak to generate a sufficient sound pressure level and a wide bandwidth, for generating an undistorted audio signal. The transducer shown in Figure 2 has a resonant frequency around 48 kHz, at which the audio signal should be modulated to maximally increase the output sound pressure level. The bandwidth of the transducer is the frequency range where the gain falls 6 dB from the resonance peak, which is approximately 20.6 kHz for this transducer.

The modulator 112 is used to shift the processed signal output by the signal processor 111 to the resonant frequency of the transducer 122. According to Berkta, an envelope which contains the information of the modulation signal, should exist in the modulated signal. We now describe two conventional

modulation techniques, which are performed by the signal processor 111 and modulator 112 together.

The first such technique is amplitude modulation (AM), which preserves the information of the modulation signal. A typical conventional AM can be denoted as

$$x_c(t) = A_c(1 + \mu x(t))\cos 2\pi f_c t$$

(3)

where $x(t)$ is the audio signal input to the controller 110 from the left and which has a bandwidth W (which is typically up to 20kHz, but normally does not include frequencies below 20Hz, since these are inaudible to humans). The value μ , which is in the range $0 < \mu < 1$, is a factor called modulation index. $x_c(t)$ is the modulated signal output from the controller 110, A_c is the carrier amplitude and f_c is the carrier frequency. The spectrum of conventional AM is shown in Fig. 3, where $X_c(f)$ is the Fourier transform of $x_c(t)$. Fig. 3 shows that this approach requires a transmission bandwidth

$$B_T = 2W$$

(4)

The average transmitted power is

$$S_T = P_c + 2P_{sb}$$

(5)

where $P_c \geq S_T/2$ represents the unmodulated carrier power and $P_{sb} \leq S_T/4$ stands for power per sideband. Therefore, at least 50% of the total transmitted power resides in a carrier term that is independent of $x(t)$ and thus conveys no message information. AM is thus wasteful of both transmission power and

5 bandwidth.

The waste can be reduced if the modulation technique suppresses the carrier transmission power, and required transmission bandwidth can be reduced by suppressing one sideband, which is called Single Side Band (SSB) modulation.

10 The spectrum of the upper side band of Fig. 3 is shown in Figure 4. For a pure tone signal, that is

$$x(t) = \cos 2\pi f_m t$$

(6)

for some frequency f_m , the two SSBs can be written as:

$$\begin{aligned} x_c(t) &= A_c (x(t)\cos 2\pi f_c t \pm \hat{x}(t)\sin 2\pi f_c t) \\ &= A_c (\cos 2\pi f_m t \cos 2\pi f_c t \pm \sin 2\pi f_m t \sin 2\pi f_c t) \\ &= A_c \cos 2\pi (f_c \mp f_m) t \end{aligned}$$

15

(7)

where $\hat{x}(t)$ is the Hilbert transform of $x(t)$. The frequencies $f_c - f_m$ and $f_c + f_m$ correspond to upper SSB and lower SSB respectively. The envelope of the tone-modulated SSB wave is a constant and therefore envelope demodulation

20 does not work for SSB modulation. Since Berktay's solution requires the

envelope information, obviously the pure SSB modulation can not be used in ultrasonic audio beam applications. However, if the SSB is transmitted in combination with the unmodulated carrier, the combination allows for envelope demodulation as in conventional AM while retaining the bandwidth conservation of suppressed side band. This technique is commonly called Single Side Band Amplitude Modulation (SSB-AM), and is usually used for television video transmission, and in this case the output of the controller 110 is written as:

$$x_c(t) = A_c((1 + \mu x(t))\cos 2\pi f_c t \pm \mu \hat{x}(t)\sin 2\pi f_c t)$$

(8)

10 The envelope of the modulated signal is

$$\begin{aligned} A(t) &= A_c \sqrt{(1 + \mu x(t))^2 + \mu \hat{x}(t)^2} \\ &= A_c (1 + \mu x(t)) \sqrt{1 + \left(\frac{\mu \hat{x}(t)}{1 + \mu x(t)} \right)^2} \end{aligned}$$

(9)

If μ is chosen such that $|\mu \hat{x}(t)| \ll 1$, then

$$A(t) \approx A_c (1 + \mu x(t))$$

15 (10)

which is close to the case of conventional AM. In the SSB-AM approach, since one sideband has been suppressed, a quarter of the transmission power can be saved compared to conventional AM. The spectrum of SSA-AM is shown in Figure 5.

For a pure tone signal of (6) in the case of $\mu = 1$, the SSB-AM gives the modulated signal as

$$\begin{aligned} x_c(t) &= A_c((1 + \cos 2\pi f_m t)\cos 2\pi f_c t \pm \sin 2\pi f_m t \sin 2\pi f_c t) \\ &= A_c(\cos 2\pi f_c t + \cos 2\pi(f_c \mp f_m)t) \\ &= 2A_c \cos 2\pi \frac{f_m}{2} t \cos 2\pi \left(f_c \mp \frac{f_m}{2}\right) t \\ &= \sqrt{2}A_c \sqrt{1 + \cos 2\pi f_m t} \cos 2\pi \left(f_c \mp \frac{f_m}{2}\right) t \end{aligned}$$

(11)

- 5 This is a conventional AM signal with the envelope of $\sqrt{2}A_c \sqrt{1 + \cos 2\pi f_m t}$ and the carrier frequency of $(f_c - f_m/2)$ or $(f_c + f_m/2)$, depending on the choice of the plus/minus sign. Therefore, for a pure tone signal with single frequency, SSB-AM gives a desired envelope which results in no nonlinear distortion in the ultrasonic audio systems from equation (2). Moreover, compared to AM, SSB-
- 10 AM uses half of the transmission bandwidth and saves a quarter of transmission power. However, for an audio signal with multiple frequencies, the above relationship does not hold and the envelope of the modulated signal differs from the desired square root of the original audio signal. Thus, some signal distortion occurs.

15

A discussion of distortion in an information transmission system can be found in many textbooks, e.g. A. Bruce Carlson, *Communication Systems: an introduction to signals and noise in electrical communication*, third edition, McGraw-Hall Book Company, 1986. Distortionless transmission means that the

output signal has the same “shape” as the input. More particularly, given an input signal $x(t)$, the output $y(t)$ is deemed to be undistorted if it differs from the input only by a constant gain K and a finite time delay t_d ,

$$y(t) = Kx(t - t_d)$$

5 (12)

Therefore, the transfer function of a distortionless transmission system is

$$G(s) = \frac{Y(s)}{X(s)} = Ke^{-j\omega t_d}, \quad s = j\omega, \quad t_d > 0$$

(13)

In other words, a system giving distortionless transmission must have constant
10 amplitude response and negative linear phase shift over those frequencies
where the input signal has significant spectral content

$$|G(s)| = K \quad \angle G(s) = -2\pi t_d f \pm m \cdot 2\pi$$

(14)

where m is an integer and ω is written as $2\pi f$. There are three major types of
15 distortion:

- Amplitude distortion, which occurs when $|G(s)| \neq K$.
- Delay distortion, which occurs when $\angle G(s) \neq -2\pi t_d f \pm m \cdot 2\pi$.
- Nonlinear distortion, which occurs when the system includes nonlinear elements.

The first two types of distortions can be grouped under the general designation of linear distortion, which can be described in terms of a transfer function.

However, the human ear is curiously insensitive to delay distortion, so delay distortion should be no concern for the audio beam system.

5

For the third type of distortion, the non-linearity precludes the existence of a transfer function. The non-linearities generate output frequency components that are not present in the input. Furthermore, such components may fall in the bandwidth of the input and overlap with the input. The nonlinear distortion is the main factor degrading the system fidelity in ultrasonic audio beam systems since we utilize the nonlinear air-demodulation mechanism.

10

A system having nonlinear elements cannot be described by the transfer function as above. Instead, the instantaneous values of input and output are related by a function $y(t) = T[x(t)]$, which can be generally approximated to a polynomial:

15

$$y(t) = a_1x(t) + a_2x^2(t) + a_3x^3(t) + \dots \quad (15)$$

The higher order powers of $x(t)$ in this equation give rise to the nonlinear distortion.

20

The output spectrum of the system can be found by taking a Fourier transform of equation (15) and invoking the convolution theorem:

$$Y(f) = a_1 X(f) + a_2 X * X(f) + a_3 X * X * X(f) + \dots,$$

(16)

- 5 where $X * X(f)$ is the convolution of the Fourier transform of $x(t)$ with itself. If $x(t)$ only contains components with frequencies up to frequency W , then the non-linearity causes the output to have frequency components higher than W (in contrast with the linear case in which the output frequencies would all be within W). For example, $X * X(f)$ is band-limited to $2W$. Furthermore, since $X * X(f)$ may contain components having a frequency $|f| < W$, this portion of spectrum overlaps with that of $X(f)$ causing further nonlinear distortion.
- 10

A quantitative measure of nonlinear distortion is provided by taking a simple pure tone, $x(t) = \cos 2\pi f_0 t$, as the input. Then, (16) yields

$$15 \quad y(t) = \left(\frac{a_2}{2} + \frac{3a_4}{8} + \dots \right) + \left(a_1 + \frac{3a_3}{4} + \dots \right) \cos 2\pi f_0 t + \left(\frac{a_2}{2} + \frac{a_4}{4} + \dots \right) \cos 4\pi f_0 t + \dots$$

(17)

The second-harmonic distortion in percent is

$$d_2 = \left| \frac{\frac{a_2}{2} + \frac{a_4}{4} + \dots}{a_1 + \frac{3a_3}{4} + \dots} \right| \times 100\% \quad (18)$$

Since the effect of higher harmonics is usually much less, the second-harmonic distortion becomes our main concern. If the input is a sum of two waves, $x(t) = \cos 2\pi f_1 t + \cos 2\pi f_2 t$, the output will include all harmonics of f_1 and f_2 , plus cross-product terms which yield $f_2 - f_1$, $f_2 + f_1$, $f_2 - 2f_1$, etc. As the number of frequency components rises, the problem becomes more complex and this is the real situation of audio beam systems.

Conventional methods of reducing sound distortions in audio beam systems are briefly described below.

10

The linear distortion can be compensated by an equalizer. For a system with a transfer function $G(s)$, the transfer function of an ideal equalizer $Q(s)$ should be

$$Q(s) = \frac{K e^{-j\omega t_d}}{G(s)} \quad (19)$$

It is very rarely possible to design an ideal equalizer, but approximating such an equalizer reduces linear distortion to certain extent.

15

Apart from the fact that the above equalizer approach can only yield an approximate solution, another drawback of this approach is that it works in an open loop. Therefore, any uncertainty with $Q(s)$ and $G(s)$ can not be compensated, and therefore causes errors.

20

There is no systematical solution to totally remove nonlinear distortion, but some methods are known to minimize its effects. To the best of the present inventors' knowledge, there are two kinds of compensation approaches for nonlinear distortion in audio beam systems: square root preprocessing and

5 SSB modulation with compensation.

Square root preprocessing of the input audio signal is quite natural since, according to Berkday's solution, the nonlinear air demodulation can be described by squaring the envelope of the modulated signal. Many early

10 researchers explored this approach, e.g. T. Kamakura, etc, "Studies for the realization of parametric loudspeaker", *J. Acoust. Soc. Japan*, vol. 41, No. 6, pp. 378–385, (June 1985) and F. Joseph Pompei, "The use of airborne ultrasonic for generation audio sound beams", Proceedings of 105th Convention of Audio Engineering Society, 1998. However, square root processing requires

15 a much wider bandwidth than can be realized, and the high order harmonics interact with each other to produce further distortion.

As for SSB modulation with compensation, James J. Croft and Joseph O. Norris, in "Theory, history, and the advancement of parametric loudspeakers: a

20 technology overview," *White Paper*, American Technology Corporation 2001/2002, American Technology Company (ATC), claimed that by using a

Single Side Band (SSB) with a recursive compensation algorithm, non-linear distortion can be much suppressed. The details of the technique were not disclosed, however, and anyway, as mentioned above, pure SSB modulation cannot be used in ultrasonic audio beam systems because there is no

5 information involved in the envelope of the modulated signal. However, as also mentioned above, if the carrier signal is added to the SSB, the resulting SSB-AM keeps the main advantages of SSB and has information involved in the envelope of the modulated signal. These factors make some people believe that an approach of combining the SSB-AM modulation with the square root

10 preprocessing would be a perfect solution. Nonetheless, after a further analysis, we can find two shortcomings in this approach. First, as mentioned above, for an audio signal with multi-frequency components, perfect matching with the square-rooting does not exist. Second, for such kinds of audio signal, it is hard, if not impossible, to implement the recursive algorithm for error compensation

15 since it requires finding the separate amplitude and phase for the various frequency components of the error.

In summary, there is no effective and efficient way to reduce signal distortion in ultrasonic audio sound beam systems.

SUMMARY OF THE INVENTION

The object of the present invention is to provide an apparatus which can suppress both linear and nonlinear distortion, and thereby improve the audio
5 fidelity of the ultrasonic audio beam system.

In general terms, the invention proposes that, in an ultrasonic sound beam generation system which includes a controller and a transducer, the controller
10 includes at least two feedback control loops which respectively compensate for linear and non-linear distortions.

Preferably, the two control loops are cascaded (i.e. nested), such that the output of one control loop (the inner control loop) is fed back as an input to the
15 other control loop (the outer control loop). Due to the cascaded structure the linear and non-linear distortions can be compensated for separately, so that their implementation can be clear, effective and easy to realize.

In particular, the feedback structure can be implemented straightforwardly in
20 software, so that the entire processing system may, if desired, be implemented on one or more integrated circuits.

The invention is not limited to a single form of modulation. In particular, as described in detail below, the processing system can be used for both AM and SSB-AM modulation techniques.

5

Specifically, the invention provides an ultrasonic sound beam generation system comprising:

a controller unit including a processing unit for processing an input signal, and a modulator for modulating a ultrasonic wave using an output of the processing unit, and

10

a transducer unit for transmitting the modulated ultrasonic wave as an acoustic beam;

15

the controller unit comprising at least two control feedback loops, at least one of the feedback loops being arranged to compensate for non-linear distortion and the other feedback loops arranged to compensate for linear distortion.

Furthermore, the invention provides a controller unit for use in an ultrasonic sound beam generation system as defined above.

20

BRIEF DESCRIPTION OF THE DRAWINGS

Further preferred features of the invention will be clear from the following detailed description of embodiments of the invention, with reference to the accompanying figures, in which like reference numerals refer to like elements throughout, and in which:

Figure 1 is a schematic block diagram of a prior art audio beam system.

Figure 2 shows the gain characteristics of a transfer function of a typical ultrasonic transducer.

10 Figure 3 illustrates the frequency spectrum of AM.

Figure 4 illustrates the frequency spectrum of single side band modulation.

Figure 5 illustrates the spectrum of SSB-AM.

Figure 6 is a schematic block diagram illustrating a first embodiment of an audio beam system which is an embodiment of the present invention.

15 Figure 7 schematically illustrates the detailed circuit of an inner feedback control loop with AM modulation of the first embodiment of Fig. 6.

Figure 8 schematically shows the detailed circuit of an inner feedback control loop with SSB-AM modulation of the first embodiment of Fig. 6.

Figure 9 shows the measured transfer function of a linearized plant with linear distortion in an outer feedback control loop of the first embodiment of Fig.6.

20

Figure 10 is a schematic block diagram illustrating the first implementation of the outer feedback control loop of the first embodiment of Fig. 6.

Figure 11 is a schematic block diagram illustrating the second implementation of the outer feedback control loop of the first embodiment of Fig. 6.

5 Figure 12 is a schematic block diagram illustrating the third implementation of the outer feedback control loop of the first embodiment of Fig. 6.

Figure 13 is a schematic block diagram illustrating a second embodiment of an audio beam system of the present invention.

10 DETAILED DESCRIPTION OF EMBODIMENTS OF THE INVENTION

The embodiments of the present invention described below have an overall structure which is the same as the block diagram Fig. 1 described above, and the explanation given above of the features shown in Fig. 1 applies also to the
15 present embodiment. In particular, in the context of the embodiments, the signal processor unit 111 is typically a digital signal processor, although other implementations, such as micro-controllers may be used. In the case that the signal processor 111 is a digital signal processor, the audio signal input to the left of the controller 110 is typically a digital signal, but if it is an analogue signal
20 the controller 110 includes an A/D converter for converting it to a digital signal before it is input to the processor 111. The modulator 112 may optionally be

realized by analog circuits, and in this case the signal output by the processor 111 before it is input to the modulator 112 through a D/A converter. However, it is also possible for the modulator 112 to be realized digitally, e.g. as part of the same integrated circuit which implements the signal processor 111, and in this case the output modulated signal may be put through a D/A converter before it is input to the plant 120.

Figure 6 is a schematic block diagram illustrating the overall structure of a first embodiment of the invention (i.e. it shows how the structure of Fig. 1 is implemented in an embodiment of the invention). The audio beam system of Fig. 6 consists of two parts: a controller 610 and a plant 620. These two units correspond respectively to the units 110 and 120 of Fig. 1. In particular, the plant 620 typically comprises an amplifier and a transducer; the function it performs is denoted as $P(s)$. Fig. 6 illustrates the controller 610 in terms of the control operation it performs, rather than the units it comprises (as in Fig. 1), but the controller 610 can be implemented (like the unit 110 of Fig. 1) as a structure including a signal processor and a modulator, although it includes feedback signal paths additional to those of the feed-forward paths shown in Fig. 1. In particular, the controller 610 includes two feedback loops: an inner feedback loop 615 and an outer feedback loop 616. As described in detail below, the main function of the inner loop 615 is to generate a signal which

includes an ultrasonic signal modulated square root of the biased audio signal input from the left of Fig. 6 (as discussed below, this modulation may be performed by AM or by SSB-AM within the scope of the invention, and indeed the invention is not limited to either). The inner loop 615 is designed with a

5 feedback loop structure to provide compensation of non-linear distortion, and the output loop 616 is provided with a feedback loop to provide compensation for linear distortion compensation. The loops 615, 616 are “cascaded”, i.e. they are nested, with the loop 615 entirely within the loop 616.

10 The outer feedback loop 616 includes a signal feedback from the plant 620. This signal may be the audio signal produced by air demodulation of the ultrasonic output of the transducer 122 of the plant 620 (the signal may for example be collected using a microphone).

15 The signals in the various locations of the system of Fig. 6 are denoted as follows. Firstly, the audio signal input to the system from the left of Fig. 6 is denoted y_2^* . This passes to an outer loop controller unit 617 which compensates for linear distortion by processing the audio signal y_2^* and the feedback signal from the plant 620. The output of the outer loop controller unit

20 617 is the input to the inner control loop 615, and is denoted as y_1^* . The output of the inner control loop 615 is denoted as y_1 , and is the input to the plant 620.

The output of the plant 620 (which is typically the audio signal generated by air demodulation of the output of the transducer 122) is denoted as y_2 . The inner feedback loop 615 includes an inner loop controller unit 618 and a unit 619 referred to here as the “inner plant”, which broadly speaking provides the modulation function. The signal output from the inner loop controller 618 to the unit 619 is denoted as u .

As mentioned above, the inner feedback loop 615 is used to produce the square root of the biased audio signal y_2^* . Ideally the square of the envelope of the modulated signal y_1 should be proportional to the output signal y_1^* from the outer loop controller 617.

$$y_1(t) = \sqrt{1 + y_1^*(t)} \cos 2\pi f_c t = A(t) \cos 2\pi f_c t$$

(20)

where $A(t) = \sqrt{1 + y_1^*(t)}$ is the envelope of $y_1(t)$. If the plant 620 is perfect, namely, the amplifier 121 and the transducer 122 in Figure 1 are both perfect, then according to Berkta's solution (2), the output audio beam y_2 yields

$$y_2(t) = k \frac{\partial^2}{\partial t^2} A(t)^2 = k \frac{\partial^2}{\partial t^2} (1 + y_1^*(t))$$

(21)

where k is a proportional factor.

If, more generally, the imperfectness of the plant 620 is taken into account by representing it as a transfer function $R(s)$, then the function performed by the whole plant 620 can be described as

$$Y_2(s) = kR(s)s^2 Y_1^*(s) = P(s)Y_1^*(s)$$

5 where $P(s) = kR(s)s^2$ is a linear transfer function. Therefore, if the inner loop could be arranged to operate perfectly, then the only distortion present in the system would be linear distortion, and the output control loop 616 is arranged to compensate for this.

10 Figures 7 and 8 show in more detail possible realizations of the inner feedback loop 615 in the system of Fig. 6.

Referring firstly to Fig. 7, a possible realization of the inner feedback loop 615 is illustrated. This realization performs an AM modulation.

15

The left block of Fig. 7 is the circuit of the inner loop controller 618 which receives the signal y_1^* , and the right block of Fig. 7 is the inner plant 619. The inner loop controller 618 includes a unit 711 for generating a biase signal, and an adder 712 for adding it to the signal y_1^* to produce a non-negative signal
20 denoted A^{*2} , which is the square of the desired envelope of the modulated signal.

$$A^{*2}(t) = 1 + y_1^*(t) = \left(\sqrt{1 + y_1^*(t)} \right)^2$$

(22)

The function of the signal A^{*2} is to be compared with a signal denoted A^2 which is feedback from the inner plant 619 via a unit 715 which performs a square operation. A subtractor unit 713 subtracts A^2 from A^{*2} , to produce a signal which is input to a unit 714. The unit 714 produces a correction term Δu , which is added by an adder unit 716 to the signal y_1^* to form the output u of the inner loop controller 618.

$$u(t) = y_1^*(t) + \Delta u(t)$$

10 (22)

The unit 619 includes a unit 717 which receives signal u and outputs a signal $(1+u/2)$, which is a truncated approximation of $\sqrt{1+u}$. The result is referred to as y_m , and signal y_m is input to a modulator 718 which uses it to modulate an ultrasonic carrier signal generated by the unit 719 to form the output y_1 .

A^2 is obtained by squaring the modulation signal y_m by the self-multiplier unit 715. Therefore, the square of the envelope y_m of the modulated signal envelope is A^2 , which is

$$A^2(t) = y_m^2(t) = \left(1 + \frac{u}{2} \right)^2 \quad (23)$$

where u is the control output from the adder 716. The truncation error represented by the difference between the square A^{*2} of the desired envelope (i.e. A^*) and the square A^2 of the real envelope is controlled by the inner feedback controller 714, whose function is to make the error $A^{*2} - A^2$ sufficiently small by providing a high enough gain in the interested frequency range, by setting the correction term Δu for the control output u . Thus, the structure of Fig. 7 provides compensation of the non-linear distortion due to the square-root operation.

Turning now to Fig. 8, the implementation of the inner feedback loop 615 in is shown for the case of SSB-AM. It can be seen that the detailed circuit of the inner loop controller 618 in Figure 8 is very similar to that of Figure 7, so the same reference numerals are used to label corresponding items. The only unit which is different is that the unit 715 of Fig. 7 is replaced by a unit 815, which, as explained below, receives a two-component input X and outputs the sum of the squares of the components, i.e. the square of the magnitude of the complex vector.

Certain elements of the plant 619 of Fig. 8 also correspond to those of Fig. 7, and are shown by corresponding reference numerals. The plant 619 in this case is an SSB-AM modulator, and the unit 717 of Fig. 7 is replaced by unit

821, unit 822 and unit 823. Unit 821 generates a bias signal, and an adder 822 adds the bias signal to the output u of the inner feedback controller 618. The inner plant 619 further comprises a Hilbert transformer 823 for obtaining the Hilbert transform $\hat{u}$ of the output u of the inner feedback controller 618. Instead
 5 of the multiplier 718 of Fig. 7, there are two multipliers 718, 818. The multiplier 718, like the multiplier 718 of Fig. 7, modulates the output $\cos 2\pi f_c t$ of an ultrasonic frequency generator 819 by the signal it receives from the adder 822. The multiplier 818 of Fig. 8 modulates a phase shifted ultrasonic carrier $\sin 2\pi f_c t$ generated by an ultrasonic frequency generator 891. An adder 824 adds the
 10 outputs of the multipliers 718, 818, to give the SSB-AM modulated signal, y_1 .

$$\begin{aligned}
 y_1(t) &= (1 + u(t))\cos 2\pi f_c t + \hat{u}(t)\sin 2\pi f_c t \\
 &= \cos 2\pi f_c t + [u(t)\cos 2\pi f_c t + \hat{u}(t)\sin 2\pi f_c t] \\
 &= y_{1am} + y_{1ssb}
 \end{aligned} \tag{23}$$

The biased modulation signal, $1+u$, and the signal after Hilbert transformation,

$\hat{u}$, constitute a feedback vector X , which is fed back to the unit 618.

$$X(t) = [1 + u(t) \quad \hat{u}(t)]^T \tag{24}$$

Thus, feedback signal A^2 from the self-multiplier 815, which as mentioned above is the modulus square of the feedback vector X , can be written as:

$$A^2(t) = |X(t)|^2 = (1 + u(t))^2 + \hat{u}(t)^2 \tag{25}$$

Thus, the error represented by the difference between the square A^{*2} of the desired envelope (i.e. A^*) and the square A^2 of the real envelope is controlled by the inner feedback controller 714 by setting the correction term Δu for the control output u . Thus, the structure of Fig. 8, like that of Fig. 7, provides
5 compensation of the non-linear distortion due to the square-root operation.

Figure 9 shows experimental results of the transfer function of the linearized plant 620, which is obviously far from the desired flat characteristics in equation (5) due to the linear distortion, and we now turn to techniques used in
10 the embodiments for compensating for this linear distortion performed by the outer feedback loop 616. Moreover, the inner feedback loop 615 cannot be perfect and the resident error of the inner loop 615 can be considered as a disturbance, and also controlled to some extent by the outer feedback loop 616. The air demodulation produces a large time delay, which makes it hard to
15 design the outer loop controller 616 to completely remove linear distortion, but this delay is not significant to the audio signal's fidelity.

Before considering in detail possible realizations of the outer control loop 616 of Fig. 6, we consider the variant of Fig. 6 discussed above, which is the second
20 embodiment of the invention shown in Fig. 10. In Fig. 10 elements which correspond to those of Fig. 6 are given the same reference numerals. In

particular, the inner loop 615 has the same overall construction as the unit 615 of Fig. 6, including an inner feedback loop, and in particular may take either of the forms shown in Figs. 7 and 8. In comparison to the first embodiment of the invention shown in Fig. 6, in the second embodiment of Fig. 10 the feedback

5 from the plant 620 is omitted, and replaced by a signal (denoted as $\hat{y}_2$) derived from a model 1020 of the real plant 620 which is feedback in an outer control loop to the outer controller 617. The model 1020 represents all the dynamics of the real plant 620 except pure time delay due to the air demodulation, which means that the model 1020 can reflect all the imperfectness of the plant 620

10 except pure time delay. Therefore, if the model is perfect, the only difference between the output y_2 from the real plant 620 and the output $\hat{y}_2$ from the model 1020 is a pure time delay which is harmless to the audio system. The model 1020 feeds its output $\hat{y}_2$ reflecting all the imperfectness of the plant 620 back to the outer controller 617. The outer controller 617 includes a subtractor 1004

15 which subtracts the feedback signal $\hat{y}_2$ from the input signal y_2^* ; and a controller unit 1017 for compensating the linear distortion caused by the imperfectness of the plant 620, the double derivative operation of the air demodulation and the linear error of the inner feedback loop 615. In this implementation, the fact that the model 1020 does not model the pure time

20 delay means that designing the controller 1017 becomes easier.

Figures 11 to 12 show in more detail possible realizations of the first embodiment shown in Fig. 6, giving more detail of the structure of the outer feedback loop 616.

- 5 In the realization shown in Figure 11, the structure resembles that of Fig. 10 (so that corresponding units are shown by the same reference numerals), but there is additional feedback of the signal y_2 in the manner shown in Fig. 6.
- Specifically, the output $\hat{y}_2$ of the plant model 1020 is fed to a delay unit 1125 which models the delay between the output of the plant model 1020 and output
- 10 of the real plant 620. A subtractor unit 1124 for subtracts the delayed signal $\hat{y}_2$ from the feedback y_2 , which provides a difference value $(y_2 - \hat{y}_2)$ representing only the imperfectness of the plant 614. This value is input to a second outer loop controller 1121 for producing a disturbance signal d , which is used by a subtractor unit 1123 to correct the output $\hat{y}_2$ of the plant model 1020 to a value
- 15 $\bar{y}$, so that it better represents the real plant 620. Thus, as compared to Fig. 10, the system of Fig. 11 includes an additional feedback control unit (via the units 1121 and 1124) which provides an improved control value $\bar{y}$ to be transmitted to the unit 1017.
- 20 Another realization of the outer loop controller unit 617 of Fig. 6 is shown in Fig. 12 (where corresponding units are labeled by the same reference numerals).

Whereas in Figure 11 the additional feedback loop via the units 1121 and 1124 operates more slowly than the outer feedback loop from the plant model 1020 to the unit 1017, in Fig. 12 the output d of the unit 1121 is fed directly into the input of the plant model 1020. This means that, from the point of view of control theory, the additional loop works like an observer.

Fig. 13 shows a third embodiment of the invention in which, unlike the first two embodiments, the linear distortion is compensated by two main feedback loops, which results in a structure with three main loops. This implementation comprises an outer feedback loop 1350, a first inner feedback loop 1310 and a second inner feedback loop 1340. As in other embodiments these are for providing a modulated signal to a plant 620. The structure further includes a section 1330 which models the operation of the real plant 620, and comprises a power amplifier 1342, a transducer 1341, a demodulator 1333 (for modeling air demodulation), a non-linear element 1332 which produces a square operation, and a double derivative operator 1331.

The first inner feedback loop 1310 consists of a first inner controller 1311, a first inner plant 1312 which includes a modulator 1313, and has the function of compensation of the non-linear distortion. For this it uses the element 1332 in a

way very similar to how the arrangements of Figs. 7 and 8 use the square operators 715, 815, and therefore this will not be described further.

The second inner feedback loop 1340 comprises a second inner loop controller
5 1343 and a part of the model 1330 including amplifier 1342 and transducer 1341. The second inner loop controller 1343 compensates the linear distortion due to imperfections in the power amplifier 1342 and the transducer 1341.

The outer feedback loop 1350 includes a first outer loop controller 1300, for
10 compensation of the remaining linear distortion, which is due to the double derivative operation performed in the air demodulation. For this purpose, the outer loop uses the unit 1331 which models the double derivative operation.

The above description of the invention is intended to be illustrative and not
15 limiting. Various changes or modifications in the embodiments described above are straightforward to those skilled in the art and these can be made without departing from the scope of the invention.

In particular, a skilled reader will appreciate that, although the elements of the
20 embodiment have been described above as separate units, this is only one possibility. More preferably, some or all the functions of each of the

embodiments can be implemented in software. For example, the functions of all the elements included in all the feedback control loops of the embodiments can be implemented as software run by one or more integrated circuits. Indeed, the fact that the embodiments illustrated above can be straightforwardly

5 implemented in this way provides one of the advantages of certain embodiments of the invention.

CLAIMS

1. An ultrasonic sound beam generation system comprising:
a processing unit for modulating a ultrasonic wave using an input
signal,
5 a transducer unit for transmitting the modulated ultrasonic wave as
an acoustic beam;
the processing unit comprising at least two control feedback loops, a
first of the feedback loops being arranged to compensate for non-linear
distortion and a second of the feedback loops for arranged to compensate for
10 linear distortion.
2. A system according to claim 1, wherein the feedback loops include
an inner feedback loop and an outer feedback loop,
the output of the inner feedback loop being fed to the transducer,
15 the outer feedback loop being arranged to receive the input signal
and additionally an outer feedback signal which is generated from the output of
the inner feedback loop, and to generate using them an output which is fed to
the inner feedback loop.

3. A system according to claim 2, wherein the inner feedback loop is for compensation of non-linear distortion, and the outer feedback loop is for compensation of linear distortion.
- 5 4. A system according to claim 3 in which the outer feedback loop includes a model of the transducer, the model receiving the output of the inner feedback loop.
5. A system according to claim 3 or claim 4 in which the outer feedback
10 loop includes the transducer, the outer feedback signal being generated using the output of the transducer.
6. A system according to claim 5 when dependent on claim 4 further including a delay unit for delaying the output of the model, and a comparator
15 unit for obtaining a difference signal indicative of a difference between the delayed output of the model and a signal obtained from the output of the transducer.
7. A system according to claim 6 in which the difference signal is used
20 to modify the input to the model.

8. A system according to claim 6 in which the difference signal is used to modify the output of the model.

9. A system according to any of claims 3 to 8 in which the output of the
5 inner feedback loop is passed to a modulator for modulating the ultrasonic wave based on the output of the inner feedback loop.

10. A system according to claim 9 in which the output of the inner feedback loop is fed back via a squaring unit which produces a signal which is a
10 square of the output of the inner feedback loop, the inner feedback loop comprising a controller for modifying the output of the inner feedback loop based on the output of the squaring unit, whereby the inner feedback loop compensates for non-linear distortion caused by air demodulation of the ultrasonic beam.

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11. A system according to claim 10 in which the ultrasonic wave is modulated according to AM modulation.

12. A system according to claim 10 in which the output of the inner
20 feedback loop comprises two components, which are respectively used to modulate two ultrasonic waves differing in phase, the two waves being

combined to form a combined modulated ultrasonic wave which is passed to the transducer, the squaring unit generating a signal indicative of the sum of the squares of the components.

5 13. A system according to claim 12, wherein the combined modulated ultrasonic wave is modulated according to SSB-AM modulation.

14. A system according to any preceding claim in which at least the two feedback loops are implemented as software running inside a single integrated
10 circuit device.

15. A processing unit for use in a system according to any preceding claim.