



Method and Apparatus for Generating A Directional Audio Signal

Field of the Invention



G00002

- 5 The present invention relates to generating a directional audio wave in air, and more specifically to a method and apparatus for pre-processing an audio signal and using the result to modulate ultrasonic signals for transmission by a directional ultrasonic loudspeaker. The invention further relates to the loudspeaker itself.

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Background of the Invention

- The acoustic field generated by a conventional loudspeaker is not directional, especially for low frequency signals. For this reason direct directional radiation
15 of audio signals is only possible using an array of loudspeakers controlled by sophisticated methods.

- However, it is well known [1-5] that if two different frequency signals are transmitted to air, the nonlinear effect of air will cause interaction between these
20 two signals and a new signal with a frequency corresponding to the difference of the two frequencies is generated. If the original two frequencies are high (in the ultrasonic frequency range), it is easy to focus them to a narrow beam. Therefore, the reproduced difference frequency signal is also confined to this narrow beam. If the difference frequency signal is in the audible range, a narrow
25 audio beam is formed. This is a very promising technology with a very wide range of possible applications, but to obtain a reproduced audio signal with high fidelity in a highly power-efficient way, appropriate pre-processing of the audio signal and appropriate selection of the ultrasonic transducer are necessary.

- 30 Figure 1 shows the overall arrangement of such a system. An input (which may comprise either or both of an analog component 1 and a digital component 3) is fed to a pre-processing section 5. The result is transmitted to a modulation and

power amplification section 7 and used to modulate an ultrasonic carrier signal generated by an oscillator 9. The result is transmitted to an ultrasonic transducer 11 which is arranged to generate a focused ultrasonic beam.

- 5 The theory underlying this system can be summarized as follows. Using the nonlinear theory of acoustics, it was shown in [5] that for two primary ultrasonic sound fields $q_1(r, z)$ and $q_2(r, z)$ with frequency f_1 and f_2 respectively, due to the non-linearity of the air, the reproduced difference frequency signal (secondary signal) is:

$$10 \quad q_-(r, z) = -\frac{\pi\beta k_-}{\rho_0 c_0^2} \int_0^z \int_0^\infty q_1(r', z') q_2^*(r', z') G_-(r, z/r', z') r' dr' dz' \quad (1)$$

where r is the off propagation axis distance, z is the coordinate along the axis of the beam, k_- is the wave number of difference frequency $f_1 - f_2$ (assuming $f_1 > f_2$), β is the coefficient of nonlinearity, ρ_0 is the ambient density of the medium, c_0 is the small-signal wave propagation speed, superscript * signifies
 15 the complex conjugate, $G_-(r, z)$ is the Green's function for difference frequency. Assuming that the initial sound pressure levels (SPL) of two primary waves are P_1 and P_2 , the strength of the reproduced difference frequency signal will be P_- and $P_- \propto P_1 P_2$.

- 20 It was further shown by Berktaý under certain simplifying assumptions [5] that if a double side band (DSB) amplitude modulation (AM) primary wave of pressure $p_1(t)$ is transmitted, after the air demodulation, a secondary wave $p_2(t)$ is generated:

$$p_1(t) = P_0 E(t) \sin(\omega_c t) \quad (2)$$

$$25 \quad p_2(t) = \frac{\beta P_0^2 A}{16\pi\rho_0 c_0^4 z \alpha} \frac{\partial^2}{\partial t^2} E^2(\tau) \quad (3)$$

where P_0 is the SPL of primary wave, $E(t)$ is the modulation envelope, ω_c is the angular frequency of the carrier wave, A is the beam's cross sectional area, α is the absorption coefficient of the medium (at ω_c), and $\tau = t - z/c_0$ is

the lag time. The relationship between modulation envelope $E(t)$ and the audio signal $a(t)$ is:

$$E(t) = 1 + ma(t)$$

(4)

5 where m is the AM depth. Based on Eqn.(3), it is found that the demodulated signal is not directly proportional to the envelope of the modulation. To reproduce the audio signal with high fidelity, a pre-distorting of the audio signal $a(t)$ is required. This means that by appropriately pre-processing the $a(t)$ before AM, the secondary wave output should be directly proportional to $a(t)$.

10 This can be achieved by generating a modified version of $E(t)$ as [4], [7]:

$$E'(t) = [1 + m \iint a(\tau) d\tau^2]^{1/2} \quad (5)$$

This seemingly simple pre-processing is very difficult to implement in practice. The main difficult arises from the square-root operation. Because it is a
15 nonlinear operation, it will increase the signal bandwidth vastly. This poses a very strict requirement to the bandwidth of the circuit and ultrasonic transducer. In particular, it is very difficult to make a wideband and high power-efficiency ultrasonic transducer. The double integration is also difficult to implement due to the -12dB/octave amplitude weighting effect and the large frequency span
20 (20~20,000Hz, 10 octaves) of audio signal. Also, an analog integrator is easily saturated and difficult to debug in practice.

Some efforts have been made to design a wideband ultrasonic transducer [2], [3], [8], [9] for audio beam application. These designs mainly use polyvinylidene
25 fluoride (PVDF) piezoelectric material as the transducer. Although it is easy to design a PVDF ultrasonic transducer with a wideband, compared with the traditional piezoelectric ceramics (such as PZT), its power-efficiency is low due to the poor electromechanical coupling coefficient k and high insertion loss. Another disadvantage of PVDF is that it requires a much higher driving voltage
30 to generate the same SPL as PZT.

Even for a PVDF transducer, the bandwidth requirement of the square-rooted DSB AM signal (it may reach 200KHz) will push the center frequency of the transducer to a very high value (e.g. 200KHz). For example, to keep the harmonics generated by square-root operation up to its 5th order, the required

5 bandwidth for usual audio signal (20~20,000Hz) will be 200KHz (100KHz before modulation). It is impossible to build an ultrasonic transducer with such a bandwidth if their center frequency is below 100KHz. Only by using transducer with a higher center frequency (several hundred KHz to several MHz) is it possible to have the requirement bandwidth. However, high frequency

10 ultrasonic wave is attenuated in air much faster than low frequency one. This will reduce the reproduced audio power level. This can be illustrated by Eqn.(1) as it shows the reproduced difference frequency component is proportional to the integration of the primary sound field along z- coordinate. High frequency ultrasonic wave will greatly reduce the effective z-length due to the high

15 propagating loss. Actually, the analysis [11] shows that the best carrier frequency for audio beam application is around 40KHz. Another problem of wideband transducer is that it is usually not so power-efficient as compared with narrow band transducer because the latter usually works at its resonant frequency point with higher quality factor. Therefore, in practice, if we directly

20 use the square-rooted DSB AM, it is very difficult to simultaneously satisfy the requirements of transmitted audio signal bandwidth and the power-efficiency of the system.

One way to reduce the demand for the bandwidth of the transducer is to avoid

25 directly using Eqn. (5) as the pre-processing method. The conventional DSB AM can be used (i.e. without the square-root operation in Eqn. (5)). This, however, will generate a very high distortion if the modulating depth approaches 1. It is suggested in [3] to use a small modulating depth to reduce the distortion. However, this also reduces the power of the reproduced audio signal and

30 therefore the power-efficiency of the system.

Further analysis shows that part of the distortion for DSB AM comes from the cross action of upper and lower sidebands of the modulated signal spectrum. To reduce this effect, it is suggested in [7] to use a single sideband (SSB) signal to replace the DSB signal as the amplitude modulating signal. The frequency components of a single frequency (f_0) signal modulated onto an ultrasonic frequency f_c by a square-rooted DSB AM, DSB AM and SSB AM are illustrated in Figure 6a, Figure 6b and Figure 6c respectively.

SSB AM based pre-processing has the advantages of both relaxing the frequency bandwidth requirement and lessening the internal self-interference as compared with DSB AM. However, its interference is high compared with the ideal square-rooted DSB AM. This is because the interaction among different frequency components of single sideband still exists. The interference can be further suppressed by reducing the modulating depth. However, a small modulating depth reduces the reproduced audio SPL and also the power-efficiency of the system as well.

A problem common to both types of the modulating scheme is the vast dynamic range requirement caused by the double integration operation. It is well-known that double integration causes a -12dB/octave filtering for audio signal. Within the whole audio frequency range of 20 to 20,000Hz, there is a span of 10 octaves. This means that the highest frequency component gets to be attenuated by 120dB with reference to the lowest frequency component. Therefore, if a single signal path and transducer are used for the entire frequency range, they must have a vast dynamic range to keep all frequency components with sufficient signal-to-noise ratio. This greatly increases the system complexity.

Summary of the Invention

The present invention aims to provide a new and useful method and apparatus for generating a modulated ultrasonic signal.

In general terms, the present invention proposes that an audio signal is divided into at least a low frequency range component and a high frequency range component, and the components used to drive different respective transducers.

5 Specifically, the low frequency range component is used to drive a first transducer (such as a PZT transducer) having a high power efficiency but a narrow bandwidth, while the high frequency range component is used to drive a second transducer (such as a PVDF transducer) having a wider bandwidth. The ultrasonic signals produced by both the transducers are formed into a highly
10 directed beam. In this way the system achieves a good compromise between the requirements of high audio fidelity and power efficiency.

Furthermore, since the first component contains only a relatively narrow range of frequencies, it is possible to process it by an advanced technique such as
15 one employing the square root operation of Eqn. (5) (e.g. square rooted DSB) without obtaining an excessive bandwidth.

The cutoff frequency value which marks the division between the high and low frequency ranges may be selected based on the bandwidth of the first
20 transducer. For example, it should preferably be much less than (e.g. no more than a third of) the bandwidth of the first transducer. This ensures that the bandwidth of the first transducer is sufficient to contain all the frequency components of the low frequency range component, even after performing a square rooted double sideband (DSB) amplitude modulation (AM) which
25 typically greatly increases the bandwidth. For example, for the 5th order harmonic of the frequency cutoff value to be broadcast the bandwidth of the first transducer must be 5 times as high (e.g. 1000Hz if the cutoff is 200Hz).

Specifically, in a first aspect the invention provides a method of generating an
30 ultrasonic signal, the method comprising:
receiving an audio signal;

obtaining from the audio signal a first lower-frequency-range component and a second higher-frequency-range component;

forming respective modulated signals from the first and second components; and

- 5 driving first and second transducers using the first and second modulated signals respectively, the second transducer having a wider bandwidth than the first transducer.

In a second aspect the invention provides an apparatus for processing an audio
10 signal, the apparatus comprising:

an input stage for receiving an audio signal;

a frequency separation section for separating the audio signal into a first, low frequency range component and a second, high frequency range component; and

- 15 a modulating section for modulating an ultrasonic signal using the first and second components respectively to generate respective first and second modulated ultrasonic signals.

In a third aspect the invention provides a loudspeaker for generating a
20 directional ultrasonic beam, the loudspeaker having:

first and second inputs for respectively receiving first and second ultrasonic input signals; and

first and second transducers arranged to generate ultrasonic signals based respectively on the first and second input signals;

- 25 the second transducer having a wider bandwidth than the first transducer.

In a fourth aspect the invention provides loudspeaker system which is a combination of an apparatus according to the second aspect of the invention and a loudspeaker according to the third aspect of the invention.

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Brief description of the Figures

An embodiment of the invention will now be described for the sake of example only with reference to the following figures, in which:

Fig. 1 is a block diagram of a known directional audio signal generating system;

Fig. 2, which is composed of Figs. 2(a) and 2(b) is a block diagram of an

5 embodiment of a loudspeaker system according to the present invention;

Fig. 3 is a comparison of the frequency response of PZT and PVDF transducers under same driving voltage;

Fig. 4(a) and Fig. 4(b) are two examples of the transducer arrangement of the embodiment of Fig. 2;

10 Fig. 5(a) represents the frequency response of the PVDF transducer of the embodiment, and 5(b) shows the frequency weights used to implement time domain double integration and transducer frequency response compensation in the embodiment of Fig. 2; and

Fig. 6(a) is an illustration of the frequency components of an ideal square-rooted DSB modulated signal where the frequency of the audio signal is assumed to be f_o and the carrier frequency is f_c , Fig. 6(b) illustrates the frequency components of a DSB modulated signal, Fig. 6(c) illustrates the frequency components of a SSB modulated signal.

20 Detailed description of the embodiments

Referring firstly to Fig. 2, an embodiment of a loudspeaker system according to the present invention is shown. Fig. 2 is composed of Figs. 2a and 2(b), the outputs at the right of Fig. 2(a) being the corresponding inputs to the left of Fig. 25 2(b).

The embodiment has the overall structure of Fig. 1, and for easy comparison the elements of Fig. 2 are labeled by reference numbers starting with the reference numerals of the components of Fig. 1 within which the elements of Fig. 2 are contained. For example, the components of the preprocessing section 5 in Fig. 1 are labeled 500 to 511 in Fig. 2.

The operation of the embodiment is as follows:

- (1) The loudspeaker system of Fig. 2 receives an audio signal $a(t)$ which may include either or both of an analog component 1 and/or a digital component 3. Any analog component 1 of the audio signal is converted to digital form through a filter and analog-to-digital converter 501. Any digital component 2 (e.g. CD or MP3 audio etc.) is decoded by a decoder 502 to a sampled form of the audio signal. The results are selected to form a digital signal $a(t)$.
- (2) The digital signal $a(t)$ is transformed to the frequency domain by a transform unit 503 which performs a digital Fourier transform (DFT) or its fast implementation FFT. The transformed frequency is written $A(\omega)$.
- (3) In the frequency domain, the signal is divided into two components by a frequency separation unit 504. The first component is the low frequency range (e.g. $f \leq 200\text{Hz}$). The second component is the medium-to-high frequency range (e.g. $200\text{Hz} < f < 20,000\text{Hz}$). In practice, there may be some frequency overlap between these two components and the separating point between them may also be variable (e.g. 500Hz). The two components of the signal will be processed separately in the following blocks.
- (4) The double integration in time domain is equivalent to a frequency weighting in the frequency domain, i.e. weighting the $A(\omega)$ by $\frac{1}{\omega^2}$. This operation is performed on each of the two components of the signal optionally using weighting coefficients stored in the weighting coefficients memory 505. There is still a 12dB/octave attenuation for high frequency signals. But since the frequency range of each component is reduced, the dynamic range of each component is also reduced. For example, the dynamic range of the first component is 36dB (3 octaves) while that of the second component is 84dB (7 octaves). These values are much smaller than the entire range processing 120dB.

(5) After weighting, the first component is converted back to the time domain by an inverse FFT 503 with both positive and negative frequency components kept so that there is a double sideband signal. The resulting time domain signal is real.

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By contrast, the second component is further processed by the unit 505.

Specifically, this performs further a weighting by another weight $\frac{1}{C(\omega + \omega_{c1})}$ for

compensating the non-uniform frequency response of the PVDF transducer which will eventually be used to broadcast this signal (as discussed below).

10 Here $C(\omega + \omega_{c1})$ is the measured frequency response of the PVDF transducer and ω_{c1} is the center angular frequency of the PVDF transducer. This is possible because the following SSB modulation keeps the one-to-one frequency mapping relationship before and after modulation. In practice, the two-step weighting process for the second component signal performed by the unit 505

15 can be implemented in one step by storing the pre-computed combining weights in the weighting coefficients memory 506 which may be either a ROM or RAM. An example of the frequency weighting for the second component signal is shown in Fig. 5: in Fig. 5(a), the line 20 represents the value $C(\omega)$ and the arrow 21 indicates the passband of the PVDF transducer; and in Fig. 5(b) the

20 line 22 indicates the weights required to perform the $\frac{1}{\omega^2}$ function, with – 12dB/Octave to implement the double integration, and line 23 indicates the actual weights stored in the memory 506 applied to the second component signal: $\frac{1}{\omega^2 C(\omega + \omega_{c1})}$.

25 After this further compensation, the second component is fed to the unit 509, which retains only the positive (or negative) frequency part of the signal by setting the values of negative (or positive) frequency components to zero. The resultant signal derived from the second component of the audio signal is converted back to the time domain by IFFT. In practice, the weighting process

may be carried out by the unit 505 only for the finally kept frequency part (positive or negative) of the second component. The resulting time domain signal for the second component is a single sideband (SSB) complex signal, having a real part I and a complex part Q.

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(6) The first component is converted to an analog form by a unit 508 including a digital-to-analog (DAC) converter and a low pass filter. Similarly, the real part I and the complex part Q of the second component are converted to analog forms by a unit 510 having respective digital-to-analog converters (DAC) followed by respective low pass filters for each part.

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(7) The first component signal is processed by a square-root circuit 511, and is then modulated onto an ultrasonic carrier frequency 91 generated by a local oscillator (LO) 90 by a modulation unit 70. At the same time, the second component complex signal is used by a unit 71 to generate SSB AM signal. This requires the real part (I) and imaginary part (Q) of the complex signal to multiply the carrier signal 91 and its 90° shifted version 92 (also produced by the oscillator 90) separately. The two modulated signals are then added together by unit 73.

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The carrier signals 90, 91 generated from the local oscillator 90 have a desired frequency f_0 (e.g. 40KHz). This is preferably the center frequency of the PZT transducer 111 described below.

(8) The modulated first component signal is preferably passed through an equalizer 72 which is designed to equalize the non-uniform frequency response of the PZT transducer 111.

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(9) The first component (i.e. the output of the unit 73) and the second component (i.e. the output of the unit 73) pass through respective power amplifiers 74, 75 separately to generate the necessary power and voltage required to drive PZT transducers and PVDF transducers respectively.

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The resultant signals are then sent to a loudspeaker 110 having one or more PZT transducers 111 and one or more PVDF transducers 112. The first component signals are input to the PZT transducer(s) 111, and the second
 5 component signals are input to PVDF transducer(s) 112.

The PZT transducers 111 and PVDF transducers 112 should be arranged such that the generated ultrasonic waves from these two kinds transducers are collimated (i.e. the sound field is overlapped well) to facilitate the nonlinear
 10 interaction. Two possible forms of the loudspeaker 110 having different arrangements of the PZT and PVDF transducers are shown in Figure 4(a) and 4(b). In Fig. 4(a), a big PZT transducer 111 is in the center of the whole loudspeaker 110 surrounded by many PVDF transducers 112. In Fig. 4(b), several relatively big PZT transducers 111 are arranged in a ring and also
 15 surrounded by PVDF transducers 112. There are many other kinds of possible arrangements for the ultrasonic loudspeaker 110.

An example of the frequency responses of PZT and PVDF transducers is shown in Figure 3. A PZT transducer 111 has a frequency response 19 which is
 20 narrow band but highly power-efficient, while PVDF transducer 112 has a frequency response 18 (which corresponds to that shown as line 20 in Fig. 5(a), but is drawn with a lower level of detail) which is wideband but less power-efficient.

25 In practice, as shown in Fig. 3, the center frequencies of the PVDF transducer 112 and PZT transducer 111 may be chosen to be slightly different. This is because the signal frequency components of PVDF and PZT transducers are different. For example, the center frequency of the PVDF transducer(s) 112 can be selected as $f_c + 200$ Hz or higher as they only need to output signal
 30 frequencies from $f_c + 200$ Hz to $f_c + 20,000$ Hz. This arrangement will further improve the power-efficiency of the system.

The audio fidelity and system power-efficiency are further analyzed here.

Firstly, considering the first, low frequency component of the audio signal, since almost ideal square-rooted DSB AM signal is transmitted within the efficient
 5 bandwidth of the PZT transducer 111, there is no limitation for the modulating depth and the fidelity of reproduced audio is very high. Also, because of the high power-efficient PZT transducer 111, a high power of carrier frequency signal is also efficiently generated. Since the piezoelectric constant of PZT
 10 material is much higher than that of the PVDF material, its driving voltage can be much lower than that of the PVDF transducer 112 if the same ultrasonic SPL is to be generated. In other words, if the same driving voltage is used for both the PZT transducer(s) 111 and the PVDF transducer(s) 112, a much stronger ultrasonic SPL is obtained from PZT transducer(s) as shown in Figure (3). This is especially suited to generate a strong carrier frequency signal.

15 Secondly, considering the second, medium-to-high frequency component of the audio signal, as mentioned above it is SSB amplitude modulated and transmitted through PVDF transducer(s) 112. Since the bandwidth of the PVDF transducer(s) 112 is wide, it is sufficient for the required signal bandwidth
 20 (20KHz). According to Eqn. (1), it is known that when two different primary signals are transmitted, they will interact with each other and generate a new signal with the frequency which is the difference of the two primary signal frequencies. For the SSB signal, the interaction between the carrier signal frequency and each individual modulated audio frequency is desired, but for an
 25 audio signal with multiple frequency components, there also exist many unavoidable interactions among these different frequencies components and generation of undesirable interference signals. However, since the reproduced difference frequency signal strength P_- is proportional to the product of the two primary wave strengths $P_1 P_2$, if a strong carrier frequency P_c is transmitted then
 30 the desirable reproduced audio signal strength $P_i \propto P_c P_1$, whereas the strength of undesirable interference signal generated by the interaction between two modulated audio frequency components is $P_i \propto P_1 P_2$, where P_2 is used to

designate the strength of another modulated audio frequency component. By increasing P_c and decreasing P_2 , the ratio of the interference to desired signal will be decreased as: $\frac{P_i}{P_r} \propto \frac{P_1 P_2}{P_1 P_c} = \frac{P_2}{P_c}$. Since the PVDF transducer(s) 112 only transmit the relatively weak P_2 , the difference between the driving voltages of the PZT transducer(s) 111 and the PVDF transducer(s) 112 can be made smaller.

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

For example, although the embodiment described above uses a digital implementation of the pre-processing circuit, including an ADC at the beginning and a DAC at the end, this is not necessary to the invention, which can alternatively be implemented in an analog way using a filter to separate the low and high frequency components, by directly using analog double integration and by using a Hilbert filter to obtain the SSB signal.

References

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Claims

1. A method of generating an ultrasonic signal, the method comprising:
receiving an audio signal;
5 obtaining from the audio signal a first lower-frequency-range component
and a second higher-frequency-range component;
forming respective modulated signals from the first and second
components; and
driving first and second transducers using the first and second modulated
10 signals respectively, the second transducer having a wider bandwidth than the
first transducer.
2. A method according to claim 1 in which at least one of the first and
second components is converted to the frequency domain and there subject to
15 a multiplication of its frequency components by respective weighting factors
selected to effect a square root operation in the time domain.
3. A method according to claim 2 in which the second component is subject
in the frequency domain to a multiplication of its frequency components by
20 respective weighting factors based on the frequency response of the second
transducer.
4. A method according to any preceding claim in which the first frequency
component is subject in the time domain to a square root operation.
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5. An method according to any preceding claim in which the positive, or
alternatively negative, frequency frequencies of the second component are set
to zero, and the real and complex components of the result are used to
modulate an ultrasonic carrier frequency and a phase shifted ultrasonic carrier
30 frequency, the results being summed to form the second modulated signal.

6. A method according to any preceding claim in which the first modulated signal is equalized in the time domain according to the frequency response of the first transducer.

- 5 7. An apparatus for processing an audio signal, the apparatus comprising:
an input stage for receiving an audio signal;
a frequency separation section for separating the audio signal into a first,
low frequency range component and a second, high frequency range
component; and
10 a modulating section for modulating an ultrasonic signal using the first
and second components respectively to generate respective first and second
modulated ultrasonic signals.

8. An apparatus according to claim 7 in which the input stage is arranged to
15 transmit the audio signal to a Fourier transform section for converting the audio
signal into frequency space, the Fourier transform section transmitting the audio
signal in frequency space to the frequency separation section.

9. An apparatus according to claim 8 in which at least one of the first and
20 second components is subject to a multiplication of its frequency components
by weighting factors selected to effect a double integration in the time domain.

10. An apparatus according to claim 8 or claim 9 including a square root unit
for subjecting the first component to a square root operation, before it is used to
25 modulate a carrier frequency.

11. An apparatus according to any of claims 8 to 10 including a unit for
setting the positive, or alternatively negative, frequency frequencies of the
second component to zero.

12. An apparatus according to any of claims 8 to 11 comprising a unit for compensating the first modulated ultrasonic signal according to the frequency response profile of a transducer.
- 5 13. An apparatus according to claim 7 in which the filter separation section is implemented as an analog filter.
14. An apparatus according to claim 13 further including an analog double integration unit and a Hilbert filter unit to generate a single sideband.
- 10 15. A loudspeaker for generating a directional ultrasonic beam, the loudspeaker having:
first and second inputs for respectively receiving first and second ultrasonic input signals; and
15 first and second transducers arranged to generate ultrasonic signals based respectively on the first and second input signals;
the second transducer having a wider bandwidth than the first transducer.
- 20 16. A loudspeaker according to claim 15 in which there are a plurality of second transducers, all generating ultrasonic signals based on the second input signal.
- 25 17. A loudspeaker according to claim 15 or claim 16 in which the first transducer is a PZT transducer and the second transducer is a PVDF transducer.
- 30 18. A loudspeaker system comprising in combination a preprocessing circuit according to any of claims 7 to 15 and a loudspeaker according to any of claim 15 to 17, the loudspeaker being arranged to receive the first and second modulated signals via its first and second inputs respectively.